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**Information technology — Coding of  
audio-visual objects —**

**Part 3:  
Audio**

*Technologies de l'information — Codage des objets audiovisuels —  
Partie 3: Codage audio*

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## Foreword

ISO (the International Organization for Standardization) and IEC (the International Electrotechnical Commission) form the specialized system for worldwide standardization. National bodies that are members of ISO or IEC participate in the development of International Standards through technical committees established by the respective organization to deal with particular fields of technical activity. ISO and IEC technical committees collaborate in fields of mutual interest. Other international organizations, governmental and non-governmental, in liaison with ISO and IEC, also take part in the work. In the field of information technology, ISO and IEC have established a joint technical committee, ISO/IEC JTC 1.

International Standards are drafted in accordance with the rules given in the ISO/IEC Directives, Part 3.

The main task of the joint technical committee is to prepare International Standards. Draft International Standards adopted by the joint technical committee are circulated to national bodies for voting. Publication as an International Standard requires approval by at least 75 % of the national bodies casting a vote.

ISO/IEC 14496-3 was prepared by Joint Technical Committee ISO/IEC JTC 1, *Information technology*, Subcommittee SC 29, *Coding of audio, picture, multimedia and hypermedia information*.

This second edition cancels and replaces the first edition (ISO/IEC 14496-3:1999), which has been technically revised. It incorporates Amd.1:2000 and Cor.1:2001.

ISO/IEC 14496 consists of the following parts, under the general title *Information technology — Coding of audio-visual objects*:

- *Part 1: Systems*
- *Part 2: Visual*
- **Part 3: Audio**
- *Part 4: Conformance testing*
- *Part 5: Reference software*
- *Part 6: Delivery Multimedia Integration Framework (DMIF)*
- *Part 7: Optimized software for MPEG-4 visual tools*
- *Part 8: Carriage of MPEG-4 contents over IP networks*

Annexes 2.E, 3.C, 4.A and 5.A form a normative part of this part of ISO/IEC 14496. Annexes 1.A to 1.C, 2.A to 2.D, 3.A, 3.B, 3.D to 3.F, 4.B, 5.B to 5.F, 6.A and 7.A are for information only.

Due to its technical nature, this part of ISO/IEC 14496 requires a special format as several standalone electronic files and, consequently, does not conform to some of the requirements of the ISO/IEC Directives, Part 2.

## Introduction

### Overview

ISO/IEC 14496-3 (MPEG-4 Audio) is a new kind of audio standard that integrates many different types of audio coding: natural sound with synthetic sound, low bitrate delivery with high-quality delivery, speech with music, complex soundtracks with simple ones, and traditional content with interactive and virtual-reality content. By standardizing individually sophisticated coding tools as well as a novel, flexible framework for audio synchronization, mixing, and downloaded post-production, the developers of the MPEG-4 Audio standard have created new technology for a new, interactive world of digital audio.

MPEG-4, unlike previous audio standards created by ISO/IEC and other groups, does not target a single application such as real-time telephony or high-quality audio compression. Rather, MPEG-4 Audio is a standard that applies to every application requiring the use of advanced sound compression, synthesis, manipulation, or playback. The subparts that follow specify the state-of-the-art coding tools in several domains; however, MPEG-4 Audio is more than just the sum of its parts. As the tools described here are integrated with the rest of the MPEG-4 standard, exciting new possibilities for object-based audio coding, interactive presentation, dynamic soundtracks, and other sorts of new media, are enabled.

Since a single set of tools is used to cover the needs of a broad range of applications, *interoperability* is a natural feature of systems that depend on the MPEG-4 Audio standard. A system that uses a particular coder—for example a real-time voice communication system making use of the MPEG-4 speech coding toolset—can easily share data and development tools with other systems, even in different domains, that use the same tool—for example a voicemail indexing and retrieval system making use of MPEG-4 speech coding.

The remainder of this clause gives a more detailed overview of the capabilities and functioning of MPEG-4 Audio. First a discussion of concepts, that have changed since the MPEG-2 audio standards, is presented. Then the MPEG-4 Audio toolset is outlined.

### New concepts in MPEG-4 Audio

Many concepts in MPEG-4 Audio are different than those in previous MPEG Audio standards. For the benefit of readers who are familiar with MPEG-1 and MPEG-2 we provide a brief overview here.

- **MPEG-4 has no standard for transport.** In all of the MPEG-4 tools for audio and visual coding, the coding standard ends at the point of constructing a sequence of access units that contain the compressed data. The MPEG-4 Systems (ISO/IEC 14496-1:2001) specification describes how to convert the individually coded objects into a bitstream that contains a number of multiplexed sub-streams.

There is no standard mechanism for transport of this stream over a channel; this is because the broad range of applications that can make use of MPEG-4 technology have delivery requirements that are too wide to easily characterize with a single solution. Rather, what is standardized is an interface (the Delivery Multimedia Interface Format, or DMIF, specified in ISO/IEC 14496-6:1999) that describes the capabilities of a transport layer and the communication between transport, multiplex, and demultiplex functions in encoders and decoders. The use of DMIF and the MPEG-4 Systems bitstream specification allows transmission functions that are much more sophisticated than are possible with previous MPEG standards.

However, LATM and LOAS were defined to provide a low overhead audio multiplex and transport mechanism for natural audio applications, which do not require sophisticated object-based coding or other functions provided by MPEG-4 Systems.

The following table gives an overview about the multiplex, storage and transmission formats for MPEG-4 Audio currently available within the MPEG-4 framework:

	Format	Functionality defined in:	Functionality redefined in:	Description
Multiplex	FlexMux	ISO/IEC 14496-1:2001 (MPEG-4 Systems) (Normative)	-	Flexible multiplex scheme
	LATM	ISO/IEC 14496-3:2001 (MPEG-4 Audio) (Normative)	-	Low Overhead Audio Transport Multiplex
Storage	ADIF	ISO/IEC 13818-7:1997 (MPEG-2 Audio) (Normative)	ISO/IEC 14496-3:2001 (MPEG-4 Audio) (Informative)	(MPEG-2 AAC) Audio Data Interchange Format, AAC only
	MP4FF	ISO/IEC 14496-1:2001 (MPEG-4 Systems) (Normative)	-	MPEG-4 File format
Transmission	ADTS	ISO/IEC 13818-7:1997 (MPEG-2 Audio) (Normative, Exemplarily)	ISO/IEC 14496-3:2001 (MPEG-4 Audio) (Informative)	Audio Data Transport Stream, AAC only
	LOAS	ISO/IEC 14496-3:2001 (MPEG-4 Audio) (Normative, Exemplarily)	-	Low Overhead Audio Stream, based on LATM, three versions are available: AudioSyncStream() EPAudioSyncStream() AudioPointerStream()

To allow for a user on the remote side of a channel to dynamically control a server streaming MPEG-4 content, MPEG-4 defines backchannel streams that can carry user interaction information.

- **MPEG-4 Audio supports low-bitrate coding.** Previous MPEG Audio standards have focused primarily on transparent (undetectable) or nearly transparent coding of high-quality audio at whatever bitrate was required to provide it. MPEG-4 provides new and improved tools for this purpose, but also standardizes (and has tested) tools that can be used for transmitting audio at the low bitrates suitable for Internet, digital radio, or other bandwidth-limited delivery. The new tools specified in MPEG-4 are the state-of-the-art tools that support low-bitrate coding of speech and other audio.
- **MPEG-4 is an object-based coding standard with multiple tools.** Previous MPEG Audio standards provided a single toolset, with different configurations of that toolset specified for use in various applications. MPEG-4 provides several toolsets that have no particular relationship to each other, each with a different target function. The Profiles of MPEG-4 Audio (subclause 1.5.2) specify which of these tools are used together for various applications.

Further, in previous MPEG standards, a single (perhaps multi-channel or multi-language) piece of content was transmitted. In contrast, MPEG-4 supports a much more flexible concept of a *soundtrack*. Multiple tools may be used to transmit several *audio objects*, and when using multiple tools together an *audio composition* system is used to create a single soundtrack from the several audio substreams. User interaction, terminal capability, and speaker configuration may be used when determining how to produce a single soundtrack from the component objects. This capability gives MPEG-4 significant advantages in quality and flexibility when compared to previous audio standards.

- **MPEG-4 provides capabilities for synthetic sound.** In natural sound coding, an existing sound is compressed by a server, transmitted and decompressed at the receiver. This type of coding is the subject of many existing standards for sound compression. In contrast, MPEG-4 standardizes a novel paradigm in which synthetic sound descriptions, including synthetic speech and synthetic music, are transmitted and then *synthesized* into sound at the receiver. Such capabilities open up new areas of very-low-bitrate but still very-high-quality coding.
- **MPEG-4 provides capabilities for Error Robustness.** Improved error robustness for AAC is provided by a set of error resilience tools. These tools reduce the perceived degradation of the decoded audio signal that is

caused by corrupted bits in the bitstream. Improved error robustness capabilities for all coding tools are provided through the error resilient bitstream payload syntax. This tool supports advanced channel coding techniques, which can be adapted to the special needs of given coding tools and a given communications channel. This error resilient bitstream payload syntax is mandatory for all error resilient object types.

The error protection tool (EP tool) provides unequal error protection (UEP) for MPEG-4 Audio in conjunction with the error resilient bitstream payload. UEP is an efficient method to improve the error robustness of source coding schemes. It is used by various speech and audio coding systems operating over error-prone channels such as mobile telephone networks or Digital Audio Broadcasting (DAB). The bits of the coded signal representation are first grouped into different classes according to their error sensitivity. Then error protection is individually applied to the different classes, giving better protection to more sensitive bits.

- **MPEG-4 provides capabilities for Scalability.** Previous MPEG Audio standards provided a single bitrate, single bandwidth toolset, with different configurations of that toolset specified for use in various applications. MPEG-4 provides several bitrate and bandwidth options within a single bitstream, providing a scalability functionality that permits a given bitstream to scale to the requirement of different channels and applications or to be responsive to a given channel that has dynamic throughput characteristics. The tools specified in MPEG-4 are the state-of-the-art tools providing scalable compression of speech and audio signals.

As with previous MPEG standards, MPEG-4 does not standardize methods for encoding sound. Thus, content authors are left to their own decisions as to the best method of creating bitstreams. At the present time, methods to automatically convert natural sound into synthetic or multi-object descriptions are not mature; therefore, most immediate solutions will involve interactively-authoring the content stream in some way. This process is similar to current schemes for MIDI-based and multi-channel mixdown authoring of soundtracks.

## Capabilities

### Overview of capabilities

The MPEG-4 Audio tools can be broadly organized into several categories:

*Speech* tools for the transmission and decoding of synthetic and natural speech.

*Audio* tools for the transmission and decoding of recorded music and other audio soundtracks.

*Synthesis* tools for very low bitrate description and transmission, and terminal-side synthesis, of synthetic music and other sounds.

*Composition* tools for object-based coding, interactive functionality, and audiovisual synchronization.

*Scalability* tools for the creation of bitstreams that can be transmitted, without recoding, at several different bitrates.

*Upstream* tools for the dynamic control the streaming of the server for bitrate control and quality feedback control.

*Error robustness* (including error resilience as well as error protection).

Each of these types of tools will be described in more detail in the following subclauses.

### MPEG-4 speech coding tools

Two types of speech coding tools are provided in MPEG-4. The *natural* speech tools allow the compression, transmission, and decoding of human speech, for use in telephony, personal communication, and surveillance applications. The *synthetic* speech tool provides an interface to text-to-speech synthesis systems; using synthetic speech provides very-low-bitrate operation and built-in connection with facial animation for use in low-bitrate video conferencing applications. Each of these tools will be discussed.

### Natural speech coding

The MPEG-4 speech coding toolset covers the compression and decoding of natural speech sound at bitrates ranging between 2 and 24 kbit/s. When variable bitrate coding is allowed, coding at even less than 2 kbit/s, for example an average bitrate of 1.2 kbit/s, is also supported. Two basic speech coding techniques are used: One is a parametric speech coding algorithm, HVXC (Harmonic Vector eXcitation Coding), for very low bit rates; and the other is a CELP (Code Excited Linear Prediction) coding technique. The MPEG-4 speech coders target

applications from mobile and satellite communications, to Internet telephony, to packaged media and speech databases. It meets a wide range of requirements encompassing bitrate, functionality and sound quality, and is specified in subparts 2 and 3.

**MPEG-4 HVXC** operates at fixed bitrates between 2.0 kbit/s and 4.0 kbit/s using a bitrate scalability technique. It also operates at lower bitrates, typically 1.2-1.7 kbit/s, using a variable bitrate technique. HVXC provides communications-quality to near-toll-quality speech in the 100-3800 Hz band at 8kHz sampling rate. HVXC also allows independent change of speed and pitch during decoding, which is a powerful functionality for fast access to speech databases. HVXC functionalities including 2.0-4.0 kbit/s fixed bitrate mode and 2.0 kbit/s maximum variable bitrate mode.

Error Resilient (ER) HVXC extends operation of the variable bitrate mode to 4.0 kbit/s to allow higher quality variable rate coding. The ER HVXC therefore provides fixed bitrate modes of 2.0 - 4.0kbit/s and a variable bitrate of either less than 2.0kbit/s or less than 4.0kbit/s, both in scalable and non-scalable modes. In the variable bitrate modes, non-speech parts are detected in unvoiced signals, and a smaller number of bits are used for these non-speech parts to reduce the average bitrate. ER HVXC provides communications-quality to near-toll-quality speech in the 100-3800 Hz band at 8kHz sampling rate. When the variable bitrate mode is allowed, operation at lower average bitrate is possible. Coded speech using variable bitrate mode at typical bitrates of 1.5kbit/s average, and at typical bitrate of 3.0kbit/s average has essentially the same quality as 2.0 kbit/s fixed rate and 4.0 kbit/s fixed rate respectively. The functionality of pitch and speed change during decoding is supported for all modes. ER HVXC has a bitstream syntax with the error sensitivity classes to be used with the EP-Tool, and the error concealment functionality is supported for use in error-prone channels such as mobile communication channels. The ER HVXC speech coder targets applications from mobile and satellite communications, to Internet telephony, to packaged media and speech databases.

**MPEG-4 CELP** is a well-known coding algorithm with new functionality. Conventional CELP coders offer compression at a single bit rate and are optimized for specific applications. Compression is one of the functionalities provided by MPEG-4 CELP, but MPEG-4 also enables the use of one basic coder in multiple applications. It provides scalability in bitrate and bandwidth, as well as the ability to generate bitstreams at arbitrary bitrates. The MPEG-4 CELP coder supports two sampling rates, namely, 8 and 16 kHz. The associated bandwidths are 100 – 3800 Hz for 8 kHz sampling and 50 – 7000 Hz for 16 kHz sampling. The silence compression tool comprises a Voice Activity Detector (VAD), a Discontinuous Transmission (DTX) unit and a Comfort Noise Generator (CNG) module. The tool encodes/decodes the input signal at a lower bitrate during the non-active-voice (silent) frames. During the active-voice (speech) frames, MPEG-4 CELP encoding and decoding are used.

The silence compression tool reduces the average bitrate thanks to compression at a lower-bitrate for silence. In the encoder, a voice activity detector is used to distinguish between regions with normal speech activity and those with silence or background noise. During normal speech activity, the CELP coding is used. Otherwise a Silence Insertion Descriptor (SID) is transmitted at a lower bitrate. This SID enables a Comfort Noise Generator (CNG) in the decoder. The amplitude and the spectral shape of this comfort noise are specified by energy and LPC parameters in methods similar to those used in a normal CELP frame. These parameters are optionally re-transmitted in the SID and thus can be updated as required.

MPEG has conducted extensive verification testing in realistic listening conditions in order to prove the efficacy of the speech coding toolset.

### Text-to-speech interface

Text-to-speech (TTS) capability is becoming a rather common media type and plays an important role in various multi-media application areas. For instance, by using TTS functionality, multimedia content with narration can be easily created without recording natural speech. Before MPEG-4, however, there was no way for a multimedia content provider to easily give instructions to an unknown TTS system. With **MPEG-4 TTS Interface**, a single common interface for TTS systems is standardized. This interface allows speech information to be transmitted in the International Phonetic Alphabet (IPA), or in a textual (written) form of any language. It is specified in subpart 6.

The **MPEG-4 Hybrid/Multi-Level Scalable TTS Interface** is a superset of the conventional TTS framework. This extended TTS Interface can utilize prosodic information taken from natural speech in addition to input text and can thus generate much higher-quality synthetic speech. The interface and its bitstream format is scalable in terms of this added information; for example, if some parameters of prosodic information are not available, a decoder can

generate the missing parameters by rule. Normative algorithms for speech synthesis and text-to-phoneme translation are not specified in MPEG-4, but to meet the goal that underlies the MPEG-4 TTS Interface, a decoder should fully utilize all the provided information according to the user's requirements level.

As well as an interface to Text-to-speech synthesis systems, MPEG-4 specifies a joint coding method for phonemic information and facial animation (FA) parameters and other animation parameters (AP). Using this technique, a single bitstream may be used to control both the Text-to-Speech Interface and the Facial Animation visual object decoder (see ISO/IEC 14496-2 Annex C). The functionality of this extended TTS thus ranges from conventional TTS to natural speech coding and its application areas, from simple TTS to audio presentation with TTS and motion picture dubbing with TTS.

### MPEG-4 general audio coding tools

MPEG-4 standardizes the coding of natural audio at bitrates ranging from 6 kbit/s up to several hundred kbit/s per audio channel for mono, two-channel-, and multi-channel-stereo signals. General high-quality compression is provided by incorporating the MPEG-2 AAC standard (ISO/IEC 13818-7), with certain improvements, as **MPEG-4 AAC**. At 64 kbit/s/channel and higher ranges, this coder has been found in verification testing under rigorous conditions to meet the criterion of "indistinguishable quality" as defined by the European Broadcasting Union.

Subpart 4 of MPEG-4 specifies the AAC tool set, **MPEG-4 Twin-VQ** and BSAC, in the General Audio (GA) coder. This coding technique uses a perceptual filterbank, a sophisticated masking model, noise-shaping techniques, channel coupling, and noiseless coding and bit-allocation to provide the maximum compression within the constraints of providing the highest possible quality. Psychoacoustic coding standards developed by MPEG have represented the state-of-the-art in this technology since MPEG-1 Audio; MPEG-4 General Audio coding continues this tradition.

For bitrates ranging from 6 kbit/s to 64 kbit/s per channel, the MPEG-4 standard provides extensions to the GA coding tools, AAC, TwinVQ, and BSAC, that allow the content author to achieve the highest quality coding at the desired bitrate. Furthermore, various bit rate scalability options are available within the GA coder. The low-bitrate techniques and scalability modes provided within this tool set have also been verified in formal tests by MPEG.

The **MPEG-4 low delay** coding functionality provides the ability to extend the usage of generic low bitrate audio coding to applications requiring a very low delay in the encoding / decoding chain (e.g. full-duplex real-time communications). In contrast to traditional low delay coders based on speech coding technology, the concept of this low delay coder is based on general perceptual audio coding and is thus suitable for a wide range of audio signals. Specifically, it is derived from the proven architecture of MPEG-2/4 Advanced Audio Coding (AAC) and all capabilities for coding of 2 (stereo) or more sound channels (multi-channel) are available within the low delay coder. It operates at up to 48 kHz sampling rate and uses a frame length of 512 or 480 samples, compared to the 1024 or 960 samples used in standard MPEG-2/4 AAC to enable coding of general audio signals with an algorithmic delay not exceeding 20 ms. Also the size of the window used in the analysis and synthesis filterbank is reduced by a factor of 2. No block switching is used to avoid the "look-ahead" delay due to the block switching decision. To reduce pre-echo artefacts in the case of transient signals, window shape switching is provided instead. For non-transient portions of the signal a sine window is used, while a so-called low overlap window is used for transient portions. Use of the bit reservoir is minimized in the encoder in order to reach the desired target delay. As one extreme case, no bit reservoir is used at all.

The **MPEG-4 BSAC** is used in combination with the AAC coding tools and replaces the noiseless coding of the quantized spectral data and the scalefactors. The MPEG-4 BSAC provides fine grain scalability in steps of 1 kbit/s per audio channel, i.e. 2 kbit/s steps for a stereo signal. One base layer bitstream and many small enhancement layer bitstreams are used. To obtain fine step scalability, a bit-slicing scheme is applied to the quantized spectral data. First the quantized spectral values are grouped into frequency bands. Each of these groups contains the quantized spectral values in their binary representation. Then the bits of a group are processed in slices according to their significance. Thus all most significant bits (MSB) of the quantized values in a group are processed first. These bit-slices are then encoded using an arithmetic coding scheme to obtain entropy coding with minimal redundancy. In order to implement fine grain scalability efficiently using MPEG-4 Systems tools, the fine grain audio data can be grouped into large-step layers and these large-step layers further grouped by concatenating large-step layers from several sub-frames. Furthermore, the configuration of the payload transmitted over an Elementary Stream (ES) can be changed dynamically (by means of the MPEG-4 backchannel capability) depending on the environment, such as network traffic or user interaction. This means that BSAC can allow for real-time adjustments to the quality of service. In addition to fine grain scalability, it can improve the quality of an



audio signal that is decoded from a bitstream transmitted over an error-prone channel, such as a mobile communication networks or Digital Audio Broadcasting (DAB) channel.

Subpart 7 of MPEG-4 specifies the parametric audio coding tool **MPEG-4 HILN** (Harmonic and Individual Lines plus Noise) to code non-speech signals like music at bitrates of 4 kbit/s and higher using a parametric representation of the audio signal. The basic idea of this technique is to decompose the input signal into audio objects which are described by appropriate source models and represented by model parameters. Object models for sinusoids, harmonic tones, and noise are utilized in the HILN coder. HILN allows independent change of speed and pitch during decoding. Furthermore HILN can be combined with MPEG-4 parametric speech coding (HVXC) to form an integrated parametric coder covering a wider range of signals and bitrates.

The Parametric Audio Coding tools combine very low bitrate coding of general audio signals with the possibility of modifying the playback speed or pitch during decoding without the need for an effects processing unit. In combination with the speech and audio coding tools in MPEG-4, improved overall coding efficiency is expected for applications of object based coding allowing selection and/or switching between different coding techniques.

This approach allows one to introduce a more advanced source model than just assuming a stationary signal for the duration of a frame, which motivates the spectral decomposition used in e.g. the MPEG-4 General Audio Coder. As known from speech coding, where specialized source models based on the speech generation process in the human vocal tract are applied, advanced source models can be advantageous, especially for very low bitrate coding schemes.

Due to the very low target bitrates, only the parameters for a small number of objects can be transmitted. Therefore a perception model is employed to select those objects that are most important for the perceptual quality of the signal.

In HILN, the frequency and amplitude parameters are quantized according to the “just noticeable differences” known from psychoacoustics. The spectral envelope of the noise and the harmonic tones are described using LPC modeling as known from speech coding. Correlation between the parameters of one frame and those of consecutive frames is exploited by parameter prediction. Finally, the quantized parameters are entropy coded and multiplexed to form a bitstream.

A very interesting property of this parametric coding scheme arises from the fact that the signal is described in terms of frequency and amplitude parameters. This signal representation permits speed and pitch change functionality by simple parameter modification in the decoder. The HILN Parametric Audio Coder can be combined with MPEG-4 Parametric Speech Coder (HVXC) to form an integrated parametric coder covering a wider range of signals and bitrates. This integrated coder supports speed and pitch change. Using a speech/music classification tool in the encoder, it is possible to automatically select the HVXC for speech signals and the HILN for music signals. Such automatic HVXC/HILN switching was successfully demonstrated and the classification tool is described in the informative Annex of the MPEG-4 standard.

### **MPEG-4 Audio synthesis tools**

The MPEG-4 toolset providing general audio synthesis capability is called **MPEG-4 Structured Audio**, and it is described in subpart 5 of ISO/IEC 14496-3. MPEG-4 Structured Audio (the SA coder) provides very general capabilities for the description of synthetic sound, and the normative creation of synthetic sound in the decoding terminal. High-quality stereo sound can be transmitted at bitrates from 0 kbit/s (no continuous cost) to 2-3 kbit/s for extremely expressive sound using these tools.

Rather than specify a particular method of synthesis, SA specifies a flexible language for describing methods of synthesis. This technique allows content authors two advantages. First, the set of synthesis techniques available is not limited to those that were envisioned as useful by the creators of the standard; any current or future method of synthesis may be used in MPEG-4 Structured Audio. Second, the creation of synthetic sound from structured descriptions is normative in MPEG-4, so sound created with the SA coder will sound the same on any terminal.

Synthetic audio is transmitted via a set of *instrument* modules that can create audio signals under the control of a *score*. An instrument is a small network of signal-processing primitives that control the parametric generation of sound according to some algorithm. Several different instruments may be transmitted and used in a single Structured Audio bitstream. A score is a time-sequenced set of commands that invokes various instruments at specific times to contribute their output to an overall music performance. The format for the description of

instruments—SAOL, the Structured Audio Orchestra Language—and that for the description of scores—SASL, the Structured Audio Score Language—are specified in subpart 6.

Efficient transmission of sound samples, also called *wavetables*, for use in sampling synthesis is accomplished by providing interoperability with the MIDI Manufacturers Association Downloaded Sounds Level 2 (DLS-2) standard, which is normatively referenced by the Structured Audio standard. By using the DLS-2 format, the simple and popular technique of wavetable synthesis can be used in MPEG-4 Structured Audio soundtracks, either by itself or in conjunction with other kinds of synthesis using the more general-purpose tools. To further enable interoperability with existing content and authoring tools, the popular MIDI (Musical Instrument Digital Interface) control format can be used instead of, or in addition to, scores in SASL for controlling synthesis.

Through the inclusion of compatibility with MIDI standards, MPEG-4 Structured Audio thus represents a unification of the current technique for synthetic sound description (MIDI-based wavetable synthesis) with that of the future (general-purpose algorithmic synthesis). The resulting standard solves problems not only in very-low-bitrate coding, but also in virtual environments, video games, interactive music, karaoke systems, and many other applications.

### **MPEG-4 Audio composition tools**

The tools for audio composition, like those for visual composition, are specified in the MPEG-4 Systems standard (ISO/IEC 14496-1). However, since readers interested in audio functionality are likely to look here first, a brief overview is provided.

*Audio composition* is the use of multiple individual “audio objects” and mixing techniques to create a single soundtrack. It is analogous to the process of recording a soundtrack in a multichannel mix, with each musical instrument, voice actor, and sound effect on its own channel, and then “mixing down” the multiple channels to a single channel or single stereo pair. In MPEG-4, the multichannel mix itself may be transmitted, with each audio source using a different coding tool, and a set of instructions for mixdown also transmitted in the bitstream. As the multiple audio objects are received, they are decoded separately, but not played back to the listener; rather, the instructions for mixdown are used to prepare a single soundtrack from the “raw material” given in the objects. This final soundtrack is then played for the listener.

An example serves to illustrate the efficacy of this approach. Suppose, for a certain application, we wish to transmit the sound of a person speaking in a reverberant environment over stereo background music, at very high quality. A traditional approach to coding would demand the use of a general audio coding at 32 kbit/s/channel or above; the sound source is too complex to be well-modeled by a simple model-based coder. However, in MPEG-4 we can represent the soundtrack as the conjunction of several objects: a speaking person passed through a reverberator added to a synthetic music track. We transmit the speaker’s voice using the CELP tool at 16 kbit/s, the synthetic music using the SA tool at 2 kbit/s, and allow a small amount of overhead (only a few hundreds of bytes as a fixed cost) to describe the stereo mixdown and the reverberation. Using MPEG-4 and an object-based approach thus allows us to describe in less than 20 kbit/s total a bitstream that might require 64 kbit/s to transmit with traditional coding, at equivalent quality.

Additionally, having such structured soundtrack information present in the decoding terminal allows more sophisticated client-side interaction to be included. For example, the listener can be allowed (if the content author desires) to request that the background music be muted. This functionality would not be possible if the music and speech were coded into the same audio track.

With the **MPEG-4 Binary Format for Scenes (BIFS)**, specified in MPEG-4 Systems, a subset tool called AudioBIFS allows content authors to describe sound scenes using this object-based framework. Multiple sources may be mixed and combined, and interactive control provided for their combination. Sample-resolution control over mixing is provided in this method. Dynamic download of custom signal-processing routines allows the content author to exactly request a particular, normative, digital filter, reverberator, or other effects-processing routine. Finally, an interface to terminal-dependent methods of 3-D audio spatialisation is provided for the description of virtual-reality and other 3-D sound material.

As AudioBIFS is part of the general BIFS specification, the same framework is used to synchronize audio and video, audio and computer graphics, or audio with other material. Please refer to ISO/IEC 14496-1 (MPEG-4 Systems) for more information on AudioBIFS and other topics in audiovisual synchronization.

### **MPEG-4 Audio scalability tools**

Many of the bitstream types in MPEG-4 are *scalable* in one manner or another. Several types of scalability in the standard are discussed below.

Bitrate scalability allows a bitstream to be parsed into a bitstream of lower bitrate such that the combination can still be decoded into a meaningful signal. The bitstream parsing can occur either during transmission or in the decoder. Scalability is available within each of the natural audio coding schemes, or by a combination of different natural audio coding schemes.

Bandwidth scalability is a particular case of bitrate scalability, whereby part of a bitstream representing a part of the frequency spectrum can be discarded during transmission or decoding. This is available for the CELP speech coder, where an extension layer converts the narrow band base layer encoder into a wide band speech coder. Also the general audio coding tools which all operate in the frequency domain offer a very flexible bandwidth control for the different coding layers.

Encoder complexity scalability allows encoders of different complexity to generate valid and meaningful bitstreams. An example for this is the availability of a high quality and a low complexity excitation module for the wideband CELP coder allowing to choose between significant lower encoder complexity or optimized coding quality.

Decoder complexity scalability allows a given bitstream to be decoded by decoders of different levels of complexity. A subtype of decoder complexity scalability is *graceful degradation*, in which a decoder dynamically monitors the resources available, and scales down the decoding complexity (and thus the audio quality) when resources are limited. The Structured Audio decoder allows this type of scalability; a content author may provide (for example) several different algorithms for the synthesis of piano sounds, and the content itself decides, depending on available resources, which one to use.

### MPEG-4 Audio Upstream

The **MPEG-4 upstream** or backchannel allows a user on a remote side to dynamically control the streaming MPEG-4 content from a server. Backchannel streams carrying the user interaction information.

### MPEG-4 Audio Error robustness

The error robustness tools provide improved performance on error-prone transmission channels. They are comprised of codec specific error resilience tools and an common error protection tool.

#### Error resilience tools for AAC

Several tools are provided to increase the error resilience for AAC. These tools improve the perceived audio quality of the decoded audio signal in case of corrupted bitstreams, which may occur e. g. in the presence of noisy transmission channels.

- The *Virtual CodeBooks tool (VCB11)* extends the sectioning information of an AAC bitstream. This permits the detection of serious errors within the spectral data of an MPEG-4 AAC bitstream. Virtual codebooks are used to limit the largest absolute value possible within a any scalefactor band that uses escape values. While using to the same codebook used by codebook 11, the sixteen virtual codebooks introduced by VCB11 provide sixteen different limitations of the spectral values belonging to the corresponding subclause. Therefore, errors in the transmission of spectral data that result in spectral values exceeding the indicated limit can be located and appropriately concealed.
- The *Reversible Variable Length Coding tool (RVLC)* replaces the Huffman and DPCM coding of the scalefactors in an AAC bitstream. The RVLC uses symmetric codewords to enable both forward and backward decoding of the scalefactor data. In order to have a starting point for backward decoding, the total number of bits of the RVLC part of the bitstream is transmitted. Because of the DPCM coding of the scalefactors, also the value of the last scalefactor is transmitted to enable backward DPCM decoding. Since not all nodes of the RVLC code tree are used as codewords, some error detection is also possible.
- The *Huffman codeword reordering (HCR)* algorithm for AAC spectral data is based on the fact that some of the codewords can be placed at known positions so that these codewords can be decoded independent of any error within other codewords. Therefore, this algorithm avoids error propagation to those codewords,

the so-called priority codewords (PCW). To achieve this, segments of known length are defined and those codewords are placed at the beginning of these segments. The remaining codewords (non-priority codewords, non-PCW) are filled into the gaps left by the PCWs using a special algorithm that minimizes error propagation to the non-PCWs codewords. This reordering algorithm does not increase the size of spectral data. Before applying the reordering algorithm, the PCWs are determined by sorting the codewords according to their importance.

### Error protection

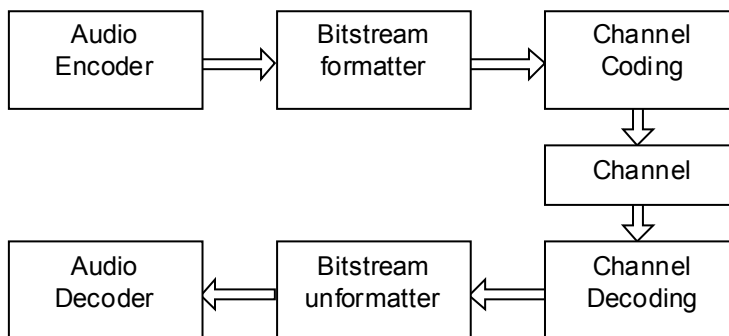
The EP tool provides unequal error protection. It receives several classes of bits from the audio coding tools, and then applies forward error correction codes (FEC) and/or cyclic redundancy codes (CRC) for each class, according to its error sensitivity.

The error protection tool (EP tool) provides the unequal error protection (UEP) capability to the set of ISO/IEC 14496-3 codecs. Main features of this tool are:

- *providing a set of error correcting/detecting codes with wide and small-step scalability, both in performance and in redundancy*
- *providing a generic and bandwidth-efficient error protection framework, which covers both fixed-length frame bitstreams and variable-length frame bitstreams*
- *providing a UEP configuration control with low overhead.*

### Error resilient bitstream reordering

Error resilient bitstream reordering allows the effective use of advanced channel coding techniques like unequal error protection (UEP), which can be perfectly adapted to the needs of the different coding tools. The basic idea is to rearrange the audio frame content depending on its error sensitivity in one or more instances belonging to different error sensitivity categories (ESC). This rearrangement can be either data element-wise or even bit-wise. An error resilient bitstream frame is build by concatenating these instances.



The basic principle is depicted in the figure above. A bitstream is reordered according to the error sensitivity of single bitstream elements or even single bits. This new arranged bitstream is channel coded, transmitted and channel decoded. Prior to audio decoding, the bitstream is rearranged to its original order. The reordered syntax, that is the syntax of the bitstream transmitted over the channel, is defined in the corresponding subparts.

# **Information technology — Coding of audio-visual objects —**

## **Part 3: Audio**

### **Structure of this part of ISO/IEC 14496:**

This part of ISO/IEC 14496 contains seven subparts:

Subpart 1: Main

Subpart 2: Speech coding — HVXC

Subpart 3: Speech coding — CELP

Subpart 4: General Audio coding (GA) — AAC, TwinVQ, BSAC

Subpart 5: Structured Audio (SA)

Subpart 6: Text To Speech Interface (TTSI)

Subpart 7: Parametric Audio Coding — HILN

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## Subpart 1: Main

### 1.1 Scope

ISO/IEC 14496-3 (MPEG-4 Audio) is a new kind of audio standard that integrates many different types of audio coding: natural sound with synthetic sound, low bitrate delivery with high-quality delivery, speech with music, complex soundtracks with simple ones, and traditional content with interactive and virtual-reality content. By standardizing individually sophisticated coding tools as well as a novel, flexible framework for audio synchronization, mixing, and downloaded post-production, the developers of the MPEG-4 Audio standard have created new technology for a new, interactive world of digital audio.

MPEG-4, unlike previous audio standards created by ISO/IEC and other groups, does not target a single application such as real-time telephony or high-quality audio compression. Rather, MPEG-4 Audio is a standard that applies to every application requiring the use of advanced sound compression, synthesis, manipulation, or playback. The subparts that follow specify the state-of-the-art coding tools in several domains; however, MPEG-4 Audio is more than just the sum of its parts. As the tools described here are integrated with the rest of the MPEG-4 standard, exciting new possibilities for object-based audio coding, interactive presentation, dynamic soundtracks, and other sorts of new media, are enabled.

Since a single set of tools is used to cover the needs of a broad range of applications, *interoperability* is a natural feature of systems that depend on the MPEG-4 Audio standard. A system that uses a particular coder—for example, a real-time voice communication system making use of the MPEG-4 speech coding toolset—can easily share data and development tools with other systems, even in different domains, that use the same tool—for example, a voicemail indexing and retrieval system making use of MPEG-4 speech coding. A multimedia terminal that can decode the Natural Audio Profile of MPEG-4 Audio has audio capabilities that cover the entire spectrum of audio functionality available today and into the future.

### 1.2 Normative references

The following normative documents contain provisions which, through reference in this text, constitute provisions of this part of ISO/IEC 14496. For dated references, subsequent amendments to, or revisions of, any of these publications do not apply. However, parties to agreements based on this part of ISO/IEC 14496 are encouraged to investigate the possibility of applying the most recent editions of the normative documents indicated below. For undated references, the latest edition of the normative document referred to applies. Members of ISO and IEC maintain registers of currently valid International Standards.

ISO/IEC 11172-3:1993, *Information technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s – Part 3: Audio*

ITU-T Rec. H.222.0 (1995) | ISO/IEC 13818-1:2000, *Information technology – Generic coding of moving pictures and associated audio information: Systems*

ISO/IEC 13818-3:1998, *Information technology – Generic coding of moving pictures and associated audio information – Part 3: Audio*

ISO/IEC 13818-7:1997, *Information technology – Generic coding of moving pictures and associated audio information – Part 7: Advanced Audio Coding (AAC)*

ITU-T Rec. H.223 (2001), *Multiplexing protocol for low bit rate multimedia communication*

MIDI Manufacturers Association, 1996, *The Complete MIDI 1.0 Detailed Specification* v. 96.2

MIDI Manufacturers Association, 1998, *The MIDI Downloadable Sounds Specification* v. 98.2

## 1.3 Terms and definitions

- 1.3.1. **AAC**: Advanced Audio Coding.
- 1.3.2. **API**: Application Programming Interface.
- 1.3.3. **AudioBIFS**: The set of tools specified in ISO/IEC 14496-1 (MPEG-4 Systems) for the composition of audio data in interactive scenes.
- 1.3.4. **audio buffer**: A buffer in the system target decoder (see ISO/IEC 13818-1:2000) for storage of compressed audio data.
- 1.3.5. **backward compatibility**: A newer coding standard is backward compatible with an older coding standard if decoders designed to operate with the older coding standard are able to continue to operate by decoding all or part of a bitstream produced according to the newer coding standard.
- 1.3.6. **bitrate**: The rate at which the compressed bitstream is delivered to the input of a decoder.
- 1.3.7. **bitstream; stream**: An ordered series of bits that forms the coded representation of the data.
- 1.3.8. **bitstream verifier**: A process by which it is possible to test and verify that all the requirements specified in ISO/IEC 14496-3 are met by the bitstream.
- 1.3.9. **BSAC**: Bit Sliced Arithmetic Coding
- 1.3.10. **byte**: Sequence of 8-bits.
- 1.3.11. **byte aligned**: A bit in a coded bitstream is byte-aligned if its position is a multiple of 8-bits from either the first bit in the stream for the Audio\_Data\_Interchange\_Format (see subclause 1.A.2.1) or the first bit in the syncword for the Audio\_Data\_Transport\_Stream Format (see subclause 1.A.2.2).
- 1.3.12. **CELP**: Code Excited Linear Prediction.
- 1.3.13. **channel**: A sequence of data representing an audio signal intended to be reproduced at one listening position.
- 1.3.14. **coded audio bitstream**: A coded representation of an audio signal.
- 1.3.15. **coded representation**: A data element as represented in its encoded form.
- 1.3.16. **composition (compositing)**: Using a scene description to mix and combine several separate audio tracks into a single presentation.
- 1.3.17. **compression**: Reduction in the number of bits used to represent an item of data.
- 1.3.18. **constant bitrate**: Operation where the bitrate is constant from start to finish of the coded bitstream.
- 1.3.19. **CRC**: The Cyclic Redundancy Check to verify the correctness of data.
- 1.3.20. **data element**: An item of data as represented before encoding and after decoding.
- 1.3.21. **decoded stream**: The decoded reconstruction of a compressed bitstream.
- 1.3.22. **decoder**: An embodiment of a decoding process.
- 1.3.23. **decoding (process)**: The process that reads an input coded bitstream and outputs decoded audio samples.



- 1.3.24. **de-emphasis**: Filtering applied to an audio signal after storage or transmission to undo a linear distortion due to emphasis.
- 1.3.25. **digital storage media; DSM**: A digital storage or transmission device or system.
- 1.3.26. **downmix**: A matrixing of  $n$  channels to obtain less than  $n$  channels.
- 1.3.27. **editing**: The process by which one or more coded bitstreams are manipulated to produce a new coded bitstream. Conforming edited bitstreams must meet the requirements defined in part 3 of ISO/IEC 14496.
- 1.3.28. **elementary stream (ES)**: A sequence of data that originates from a single producer in the transmitting MPEG-4 Terminal and terminates at a single recipient, e.g. an AVObject or a Control Entity in the receiving MPEG-4 Terminal. It flows through one FlexMux Channel.
- 1.3.29. **emphasis**: Filtering applied to an audio signal before storage or transmission to improve the signal-to-noise ratio at high frequencies.
- 1.3.30. **encoder**: An embodiment of an encoding process.
- 1.3.31. **encoding (process)**: A process, not specified in ISO/IEC 14496, that reads a stream of input audio samples and produces a valid coded bitstream as defined in part 3 of ISO/IEC 14496.
- 1.3.32. **entropy coding**: Variable length lossless coding of the digital representation of a signal to reduce statistical redundancy.
- 1.3.33. **EP**: Error Protection
- 1.3.34. **ER**: Error resilience or Error Resilient (as appropriate)
- 1.3.35. **FFT**: Fast Fourier Transformation. A fast algorithm for performing a discrete Fourier transform (an orthogonal transform).
- 1.3.36. **flag**: A variable which can take one of only the two values defined in this specification.
- 1.3.37. **forward compatibility**: A newer coding standard is forward compatible with an older coding standard if decoders designed to operate with the newer coding standard are able to decode bitstreams of the older coding standard.
- 1.3.38. **F<sub>s</sub>**: Sampling frequency.
- 1.3.39. **Hann window**: A time function applied sample-by-sample to a block of audio samples before Fourier transformation.
- 1.3.40. **HCR**: Huffman codebook reordering
- 1.3.41. **HILN**: Harmonic and Individual Lines plus Noise
- 1.3.42. **Huffman coding**: A specific method for entropy coding.
- 1.3.43. **HVXC**: Harmonic Vector Excitation Coding (very low bit rate speech)
- 1.3.44. **IPQF**: inverse polyphase quadrature filter
- 1.3.45. **low frequency enhancement (LFE) channel**: A limited bandwidth channel for low frequency audio effects in a multichannel system.
- 1.3.46. **LPC**: Linear Predictive Coding.

- 1.3.47. **LSP**: Line Spectrum Pair.
- 1.3.48. **LTP**: Long Term Prediction.
- 1.3.49. **MIDI**: The Musical Instrument Digital Interface standards. Certain aspects of the MPEG-4 Structured Audio tools provide interoperability with MIDI standards.
- 1.3.50. **multichannel**: A combination of audio channels used to create a spatial sound field.
- 1.3.51. **multilingual**: A presentation of dialogue in more than one language.
- 1.3.52. **Nyquist sampling**: Sampling at or above twice the maximum bandwidth of a signal.
- 1.3.53. **OD**: Object Descriptor.
- 1.3.54. **padding**: A method to adjust the average length of an audio frame in time to the duration of the corresponding PCM samples, by conditionally adding a slot to the audio frame.
- 1.3.55. **parameter**: A variable within the syntax of this specification which may take one of a range of values. A variable which can take one of only two values is a flag or indicator and not a parameter.
- 1.3.56. **parser**: Functional stage of a decoder which extracts from a coded bitstream a series of bits representing coded elements.
- 1.3.57. **PNS**: Perceptual Noise Substitution.
- 1.3.58. **PQF**: polyphase quadrature filter
- 1.3.59. **prediction**: The use of a predictor to provide an estimate of the sample value or data element currently being decoded.
- 1.3.60. **prediction error**: The difference between the actual value of a sample or data element and its predictor.
- 1.3.61. **predictor**: A linear combination of previously decoded sample values or data elements.
- 1.3.62. **presentation channel**: An audio channel at the output of the decoder.
- 1.3.63. **PSNR**: Peak Signal to Noise Ratio.
- 1.3.64. **random access**: The process of beginning to read and decode the coded bitstream at an arbitrary point.
- 1.3.65. **reserved**: The term "reserved" when used in the clauses defining the coded bitstream indicates that the value may be used in the future for ISO/IEC defined extensions.
- 1.3.66. **RVLC**: Reversible Variable Length Coding
- 1.3.67. **Sampling Frequency (Fs)**: Defines the rate in Hertz which is used to digitize an audio signal during the sampling process.
- 1.3.68. **SAOL**: The MPEG-4 Structured Audio Orchestra language, used to transmit the description of synthesis algorithms in MPEG-4.
- 1.3.69. **SASBF**: The MPEG-4 Structured Audio Sample Bank Format, an efficient format for the transmission of blocks of wavetable (sample data) compatible with the MIDI method for the same.
- 1.3.70. **SASL**: The MPEG-4 Structured Audio Score Language, used to transmit synthesis control parameters in MPEG-4.

- 1.3.71. **side information**: Information in the bitstream necessary for controlling the decoder.
- 1.3.72. **Structured audio**: Audio coding by means of transmitting descriptions that are synthesized into sound as they are received. See subpart 5.
- 1.3.73. **stuffing (bits); stuffing (bytes)**: Code-words that may be inserted at particular locations in the coded bitstream that are discarded in the decoding process. Their purpose is to increase the bitrate of the stream which would otherwise be lower than the desired bitrate.
- 1.3.74. **surround channel**: An audio presentation channel added to the front channels (L and R or L, R, and C) to enhance the spatial perception.
- 1.3.75. **syncword**: A code embedded in audio transport bitstreams that identifies the start of a transport frame.
- 1.3.76. **TLSS**: Tools for Large Step Scalability. The TLSS tools comprise the frequency selective switch (FSS) tool and the upsampling filter tool, described in subpart 4. In addition, all other methods, which are required to implement all the scalability modes of the generic audio coder, as defined in subpart 4, are also included in the TLSS tools.
- 1.3.77. **TTSI**: Text to Speech Interface.
- 1.3.78. **TwinVQ**: Transform domain Weighted Interleave Vector Quantization.
- 1.3.79. **variable bitrate**: Operation where the bitrate varies with time during the decoding of a coded bitstream.
- 1.3.80. **variable length code (VLC)**: A code word assigned by variable length encoder (See variable length coding).
- 1.3.81. **variable length coding**: A reversible procedure for coding that assigns shorter code-words to frequent symbols and longer code-words to less frequent symbols.
- 1.3.82. **variable length decoder**: A procedure to obtain the symbols encoded with a variable length coding technique.
- 1.3.83. **variable length encoder**: A procedure to assign variable length codewords to symbols.
- 1.3.84. **VCB11**: Virtual Codebooks for codebook 11
- 1.3.85. **virtual codebook**: If several codebook values refer to one and the same physical codebook, these values are called virtual codebooks.

## 1.4 Symbols and abbreviations

The mathematical operators used in this part of ISO/IEC 14496 are similar to those used in the C programming language. However, integer division with truncation and rounding are specifically defined. The bitwise operators are defined assuming two's-complement representation of integers. Numbering and counting loops generally begin from zero.

### 1.4.1 Arithmetic operators

- |    |   |
|----|---|
| +  | Addition.   |
| –  | Subtraction (as a binary operator) or negation (as a unary operator). |
| ++ | Increment.  |
| -- | Decrement.  |

## ISO/IEC 14496-3:2001(E)

*	Multiplication.
^	Power.
/	Integer division with truncation of the result toward zero. For example, $7/4$ and $-7/-4$ are truncated to 1 and $-7/4$ and $7/-4$ are truncated to $-1$ .
//	Integer division with rounding to the nearest integer. Half-integer values are rounded away from zero unless otherwise specified. For example $3//2$ is rounded to 2, and $-3//2$ is rounded to $-2$ .
DIV	Integer division with truncation of the result towards $-\infty$ .
	Absolute value. $ x  = x$ when $x > 0$ $ x  = 0$ when $x == 0$ $ x  = -x$ when $x < 0$
%	Modulus operator. Defined only for positive numbers.
Sign( )	Sign. $\text{Sign}(x) = 1$ when $x > 0$ $\text{Sign}(x) = 0$ when $x == 0$ $\text{Sign}(x) = -1$ when $x < 0$
INT ( )	Truncation to integer operator. Returns the integer part of the real-valued argument.
NINT ( )	Nearest integer operator. Returns the nearest integer value to the real-valued argument. Half-integer values are rounded away from zero.
sin	Sine.
cos	Cosine.
exp	Exponential.
$\sqrt{\quad}$	Square root.
log <sub>10</sub>	Logarithm to base ten.
log <sub>e</sub>	Logarithm to base e.
log <sub>2</sub>	Logarithm to base 2.

### 1.4.2 Logical operators

	Logical OR.
&&	Logical AND.
!	Logical NOT

### 1.4.3 Relational operators

>	Greater than.
>=	Greater than or equal to.
<	Less than.

<= Less than or equal to.

== Equal to.

!= Not equal to.

max [...,] the maximum value in the argument list.

min [...,] the minimum value in the argument list.

#### **1.4.4 Bitwise operators**

A two's complement number representation is assumed where the bitwise operators are used.

& AND

| OR

>> Shift right with sign extension.

<< Shift left with zero fill.

#### **1.4.5 Assignment**

= Assignment operator.

#### **1.4.6 Mnemonics**

The following mnemonics are defined to describe the different data types used in the coded bitstream.

bslbf	Bit string, left bit first, where "left" is the order in which bit strings are written in ISO/IEC 11172. Bit strings are written as a string of 1s and 0s within single quote marks, e.g. '1000 0001'. Blanks within a bit string are for ease of reading and have no significance.
L, C, R, LS, RS	Left, center, right, left surround and right surround audio signals
rpchof	Remainder polynomial coefficients, highest order first. (Audio)
uimsbf	Unsigned integer, most significant bit first.
vlclbf	Variable length code, left bit first, where "left" refers to the order in which the VLC codes are written.
window	Number of the actual time slot in case of block_type==2, 0 <= window <= 2. (Audio)

The byte order of multi-byte words is most significant byte first.

#### **1.4.7 Constants**

$\pi$  3,14159265358...

e 2,71828182845...

### 1.4.8 Method of describing bitstream syntax

The bitstream retrieved by the decoder is described in the syntax section of each subpart. Each data item in the bitstream is in bold type. It is described by its name, its length in bits, and a mnemonic for its type and order of transmission.

The action caused by a decoded data element in a bitstream depends on the value of that data element and on data elements previously decoded. The decoding of the data elements and the definition of the state variables used in their decoding are described in the subclauses following the syntax section of each subpart. The following constructs are used to express the conditions when data elements are present, and are in normal type:

Note this syntax uses the 'C'-code convention that a variable or expression evaluating to a non-zero value is equivalent to a condition that is true.

<pre>while ( condition ) {   <b>data_element</b>   ... }</pre>	<p>If the condition is true, then the group of data elements occurs next in the data stream. This repeats until the condition is not true.</p>
--	--

<pre>do {   <b>data_element</b>   ... } while ( condition )</pre>	<p>The data element always occurs at least once. The data element is repeated until the condition is not true.</p>
---	--

<pre>if ( condition ) {   <b>data_element</b>   ... } else {   <b>data_element</b>   ... }</pre>	<p>If the condition is true, then the first group of data elements occurs next in the data stream</p> <p>If the condition is not true, then the second group of data elements occurs next in the data stream.</p>
--	---

<pre>for (expr1; expr2; expr3) {   <b>data_element</b>   ... }</pre>	<p>Expr1 is an expression specifying the initialisation of the loop. Normally it specifies the initial state of the counter. Expr2 is a condition specifying a test made before each iteration of the loop. The loop terminates when the condition is not true. Expr3 is an expression that is performed at the end of each iteration of the loop, normally it increments a counter.</p>
--	--

Note that the most common usage of this construct is as follows:

<pre>for ( i = 0; i &lt; n; i++) {   <b>data_element</b>   ... }</pre>	<p>The group of data elements occurs n times. Conditional constructs within the group of data elements may depend on the value of the loop control variable i, which is set to zero for the first occurrence, incremented to one for the second occurrence, and so forth.</p>
--	---

As noted, the group of data elements may contain nested conditional constructs. For compactness, the {} may be omitted when only one data element follows.

<p><b>Data_element [ ]</b></p>	<p>data_element [ ] is an array of data. The number of data elements is indicated by the context.</p>
<p><b>Data_element [n]</b></p>	<p>data_element [n] is the n+1th element of an array of data.</p>
<p><b>Data_element [m][n]</b></p>	<p>data_element [m][n] is the m+1,n+1 th element of a two-dimensional array of data.</p>

**Data\_element [l][m][n]** data\_element [l][m][n] is the l+1,m+1,n+1 th element of a three-dimensional array of data.

**data\_element [m..n]** data\_element [m..n] is the inclusive range of bits between bit m and bit n in the data\_element.

While the syntax is expressed in procedural terms, it should not be assumed that this implements a satisfactory decoding procedure. In particular, it defines a correct and error-free input bitstream. Actual decoders must include a means to deal with incorrect bitstreams and to find the start of the described elements.

**Definition of bytealigned function**

The function bytealigned ( ) returns 1 if the current position is on a byte boundary, that is the next bit in the bitstream is the first bit in a byte. Otherwise it returns 0.

**Definition of nextbits function**

The function nextbits ( ) permits comparison of a bit string with the next bits to be decoded in the bitstream.

The number of bits for each data element is written in the second column. "X..Y" indicates that the number of bits is one of the values between X and Y including X and Y. "{X;Y}" means the number of bits is X or Y, depending on the value of other data elements in the bitstream.

**1.5 Technical overview**

**1.5.1 MPEG-4 audio object types**

**1.5.1.1 Audio object type definition**

**Table 1.1 – Audio Object Type definition**

Tools	13818-7 main	13818-7 LC	13818-7 SSR	PNS	LTP	TLSS	Twin VQ	CELP	HVXC	TTSI	SA tools	SASBF	MIDI	BSAC	HILN	Low Delay AAC	HVXC 4kbs VR	Silence Compression	Error Robust	Hierarchy	Object Type ID	
Null																					0	
AAC main	X			X																2)	1	
AAC LC		X		X																	2)	2
AAC SSR			X	X																		3
AAC LTP		X		X	X																2)	4
(Reserved)																						5
AAC Scalable		X		X	X	X																6
TwinVQ					X		X															7
CELP								X														8
HVXC									X													9
(Reserved)																						10
(Reserved)																						11
TTSI										X												12
Main synthetic											X	X	X								3)	13
Wavetable synthesis												X	X								4)	14
General MIDI													X									15

Algorithmic Synthesis and Audio FX												X								16
ER AAC LC		X		X														X		17
(Reserved)																				18
ER AAC LTP		X		X	X													X	5)	19
ER AAC scalable		X		X		X												X		20
ER TwinVQ							X											X		21
ER BSAC				X								X						X		22
ER AAC LD				X	X									X				X		23
ER CELP								X										X	X	24
ER HVXC									X								X	X		25
ER HILN														X				X		26
ER Parametric									X					X		X		X		27
(Reserved)																				28
(Reserved)																				29
(Reserved)																				30
(Reserved)																				31

Notes -

- 1) The error-robust tool is composed of error resilience and error protection. With respect of the decoder it is mandatory to equip the bit parsing function for error protection. However, it is optional to have the error detection and correction function.
- 2) Contains AAC-LC.
- 3) Contains Wavetable synthesis and Algorithmic Synthesis and Audio FX.
- 4) Contains General MIDI.
- 5) Contains ER AAC LC.

**1.5.1.2 Description**

**1.5.1.2.1 NULL Object**

The NULL object provides the possibility to feed raw PCM data directly to the audio compositor. No decoding is involved, However, an audio object descriptor is used to specify the sampling rate and the audio channel configuration.

**1.5.1.2.2 AAC - Main Object**

The AAC Main object is very similar to the AAC Main Profile that is defined in ISO/IEC 13818-7. However, additionally the PNS tool is available. The AAC Main object type bitstream syntax is compatible with the syntax defined in ISO/IEC 13818-7. All the MPEG-2 AAC multi-channel capabilities are available. A decoder capable to decode a MPEG-4 main object stream can also parse and decode a MPEG-2 AAC raw data stream. On the other hand, although a MPEG-2 AAC coder can parse an MPEG-4 AAC Main bitstream, decoding may fail, since PNS might have been used.

**1.5.1.2.3 AAC - Low Complexity (LC) Object**

The MPEG-4 AAC Low Complexity object type is the counterpart to the MPEG-2 AAC Low Complexity Profile, with exactly the same restrictions as mentioned above for the AAC Main object type.

**1.5.1.2.4 AAC - Scalable Sampling Rate (SSR) Object**

The MPEG-4 AAC Scalable Sampling Rate object type is the counterpart to the MPEG-2 AAC Scalable Sampling Rate Profile, with exactly the same restrictions as mentioned above for the AAC Main object type.



#### 1.5.1.2.5 AAC - Long Term Predictor (LTP) Object

The MPEG-4 AAC LTP object type is similar to the AAC Main object type. However, a Long Term Predictor replaces the MPEG-2 AAC predictor. The LTP achieves a similar coding gain, but requires significantly lower implementation complexity. The bitstream syntax for this object type is very similar to the syntax defined in ISO/IEC 13818-7. An MPEG-2 AAC LC profile bitstream can be decoded without restrictions using an MPEG-4 AAC LTP object decoder.

#### 1.5.1.2.6 AAC Scalable Object

The scalable AAC object uses a different bitstream syntax to support bitrate- and bandwidth- scalability. A large number of scalable combinations are available, including combinations with TwinVQ and CELP coder tools. However, only mono, or 2-channel stereo objects are supported.

#### 1.5.1.2.7 TwinVQ Object

The TwinVQ object belongs to the GA coding scheme that quantizes the MDCT coefficients. This coding scheme is based on fixed rate vector quantization instead of Huffman coding in AAC.

Low bit rate mono and stereo audio coding is available. Scalable audio coding schemes are also available in the Scalable Audio Profile combined with AAC scalable object type.

#### 1.5.1.2.8 CELP Object

The CELP object is supported by the CELP speech coding tools, which provide coding at 8 kHz and 16 kHz sampling rate at bit rates in the range 4-24 kbit/s. Additionally, bit rate scalability and bandwidth scalability are available in order to provide scalable decoding of CELP bitstreams. CELP Object always contains exactly one mono audio signal.

#### 1.5.1.2.9 HVXC Object

The HVXC object is supported by the parametric speech coding (HVXC) tools, which provide fixed bitrate modes (2.0-4.0kbit/s) in a scalable and a non-scalable scheme, a variable bitrate mode (< 2.0kbit/s) and the functionality of pitch and speed change. Only 8 kHz sampling rate and mono audio channel are supported.

#### 1.5.1.2.10 TTSI Object

The TTSI object is supported by the TTSI tools described in subpart 6. It allows very-low-bitrate phonemic descriptions of speech to be transmitted in the bitstream and then synthesized into sound. No particular speech synthesis method is specified in MPEG-4; rather, the TTSI tools specify an *interface* to non-normative synthesis methods. This method has a bit rate ranging 200 ~ 1200 bit/s. The TTSI object also supports synchronization of the synthesized speech with the facial animation defined in ISO/IEC 14496-2.

#### 1.5.1.2.11 Main Synthetic Object

The main synthetic object allows the use of all MPEG-4 Structured Audio tools (described in subpart 5 of this standard). It supports flexible, high-quality algorithmic synthesis using the SAOL music-synthesis language; efficient wavetable synthesis with the SASBF sample-bank format; and enables the use of high-quality mixing and postproduction in the Systems AudioBIFS toolset. Sound can be described at 0 kbit/s (no continuous cost) to 3-4 kbit/s for extremely expressive sounds in the MPEG-4 Structured Audio format.

#### 1.5.1.2.12 Wavetable Synthesis Object

The wavetable synthesis object is supported only by the SASBF format and MIDI tools. It allows the use of simple „sampling synthesis“ in presentations where the quality and flexibility of the full synthesis toolset is not required.

#### 1.5.1.2.13 General Midi Object

The General MIDI object is included only to provide interoperability with existing content. Normative sound quality and decoder behavior are not provided with the General MIDI object.

#### 1.5.1.2.14 Algorithmic Synthesis and Audio FX Object

The Algorithmic Synthesis object provides SAOL-based synthesis capabilities for very low-bitrate terminals. It is also used to support the AudioBIFS **AudioFX** node in content where sound synthesis capability is not needed.

#### 1.5.1.2.15 Error Resilient (ER) AAC Low Complexity (LC) object type

The Error Resilient (ER) MPEG-4 AAC Low Complexity object type is the counterpart to the MPEG-4 AAC Low Complexity object, with additional error resilient functionality.

#### 1.5.1.2.16 Error Resilient (ER) AAC Long Term Predictor (LTP) object type

The Error Resilient (ER) MPEG-4 AAC LTP object type is the counterpart to the MPEG-4 AAC LTP object, with additional error resilient functionality.

#### 1.5.1.2.17 Error Resilient (ER) AAC scalable object type

The Error Resilient (ER) MPEG-4 AAC scalable object type is the counterpart to the MPEG-4 AAC scalable object, with additional error resilient functionality.

#### 1.5.1.2.18 Error Resilient (ER) TwinVQ object type

The Error Resilient (ER) TwinVQ object type is the counterpart to the MPEG-4 TwinVQ object, with additional error resilient functionality.

#### 1.5.1.2.19 Error Resilient (ER) BSAC object type

The ER BSAC object is supported by the fine grain scalability tool (BSAC: Bit-Sliced Arithmetic Coding). It provides error resilience as well as fine step scalability in the MPEG-4 General Audio (GA) coder. It is used in combination with the AAC coding tools and replaces the noiseless coding and the bitstream formatting of MPEG-4 version 1 GA coder. A large number of scalable layers are available, providing 1 kbit/s/ch enhancement layer, i.e. 2 kbit/s steps for a stereo signal.

#### 1.5.1.2.20 Error Resilient (ER) AAC LD object type

The AAC LD object is supported by the low delay AAC coding tool. It also permits combinations with the PNS tool and the LTP tool. AAC LD object provides the ability to extend the usage of generic low bitrate audio coding to applications requiring a very low delay of the encoding / decoding chain (e.g. full-duplex real-time communications).

#### 1.5.1.2.21 Error Resilient (ER) CELP object type

The ER CELP object is supported by silence compression and ER tools. It provides the ability to reduce the average bitrate thanks to a lower-bitrate compression for silence, with additional error resilient functionality.

#### 1.5.1.2.22 Error Resilient (ER) HVXC object type

The ER HVXC object is supported by the parametric speech coding (HVXC) tools, which provide fixed bitrate modes (2.0-4.0 kbit/s) and variable bitrate modes (< 2.0 kbit/s and < 4.0 kbit/s) both in a scalable and a non-scalable scheme, and the functionality of pitch and speed change. The syntax to be used with the EP-Tool, and the error concealment functionality are supported for the use for error-prone channels. Only 8 kHz sampling rate and mono audio channel are supported.

### 1.5.1.2.23 Error Resilient (ER) HILN object type

The ER HILN object is supported by the parametric audio coding tools (HILN: Harmonic and Individual Lines plus Noise) which provide coding of general audio signals at very low bitrates ranging from below 4 kbit/s to above 16 kbit/s. Bitrate scalability and the functionality of speed and pitch change are available. The ER HILN object supports mono audio objects at a wide range of sampling rates.

### 1.5.1.2.24 Error Resilient (ER) Parametric object type

The ER Parametric object is supported by the parametric audio coding and speech coding tools HILN and HVXC. This integrated parametric coder combines the functionalities of the ER HILN and the ER HVXC objects. Only 8 kHz sampling rate and mono audio channel are supported.

## 1.5.2 Audio profiles and levels

### 1.5.2.1 Profiles

Eight Audio Profiles have been defined:

1. The **Speech Audio Profile** provides a parametric speech coder, a CELP speech coder and a Text-To-Speech interface.
2. The **Synthetic Audio Profile** provides the capability to generate sound and speech at very low bitrates.
3. The **Scalable Audio Profile**, a superset of the Speech Audio Profile, is suitable for scalable coding of speech and music, for transmission methods such as Internet and Digital Broadcasting.
4. The **Main Audio Profile** is a rich superset of all the other Profiles, containing tools for natural and synthetic audio. The **Main Audio Profile** is a superset of the other three profiles (scalable, speech, synthesis).
5. The **High Quality Audio Profile** contains the CELP speech coder and the Low Complexity AAC coder including Long Term Prediction. Scalable coding coding can be performed by the AAC Scalable object type. Optionally, the new error resilient (ER) bitstream syntax may be used.
6. The **Low Delay Audio Profile** contains the HVXC and CELP speech coders (optionally using the ER bitstream syntax), the low-delay AAC coder and the Text-to-Speech interface TTSI.
7. The **Natural Audio Profile** contains all natural audio coding tools available in MPEG-4.
8. The **Mobile Audio Internetworking Profile** contains the low-delay and scalable AAC object types including TwinVQ and BSAC. This profile is intended to extend communication applications using non-MPEG speech coding algorithms with high quality audio coding capabilities.

Table 1.2 – Audio Profiles definition

Audio Object Type	Speech Audio Profile	Synthetic Audio Profile	Scalable Audio Profile	Main Audio Profile	High Quality Audio Profile	Low Delay Audio Profile	Natural Audio Profile	Mobile Audio Internet working Profile	Object Type ID
Null									0
AAC main				X			X		1
AAC LC			X	X	X		X		2
AAC SSR				X			X		3
AAC LTP			X	X	X		X		4
(reserved)									5
AAC Scalable			X	X	X		X		6
TwinVQ			X	X			X		7
CELP	X		X	X	X	X	X		8
HVXC	X		X	X		X	X		9
(reserved)									10
(reserved)									11
TTSI	X	X	X	X		X	X		12
Main synthetic		X		X					13

Wavetable synthesis		(Subset of Main Synthetic)		(Subset of Main Synthetic)					14
General MIDI		(Subset of Main Synthetic)		(Subset of Main Synthetic)					15
Algorithmic Synthesis and Audio FX		(Subset of Main Synthetic)		(Subset of Main Synth)					16
ER AAC LC					X		X	X	17
(reserved)									18
ER AAC LTP					X		X		19
ER AAC Scalable					X		X	X	20
ER TwinVQ							X	X	21
ER BSAC							X	X	22
ER AAC LD						X	X	X	23
ER CELP					X	X	X		24
ER HVXC						X	X		25
ER HILN							X		26
ER Parametric							X		27
(reserved)									28
(reserved)									29
(reserved)									30
(reserved)									31

**1.5.2.2 Complexity units**

Complexity units are defined to give an approximation of the decoder complexity in terms of processing power and RAM usage required for processing MPEG-4 Audio bitstreams in dependence of specific parameters.

The approximated processing power is given in “Processor Complexity Units” (PCU), specified in integer numbers of MOPS. The approximated RAM usage is given in “RAM Complexity Units” (RCU), specified in mostly integer numbers of kWords (1000 words). The RCU numbers do not include working buffers that can be shared between different objects and/or channels.

If a profile level is specified by the maximum number of complexity units, then a flexible configuration of the decoder handling different types of objects is allowed under the constraint that both values for the total complexity for decoding and sampling rate conversion (if needed) do not exceed this limit.

The following table gives complexity estimates for the different object types. PCU values are given in MOPS per channel, RCU values in kWords per channel.

**Table 1.3 – Complexity of Audio Object Types and SR conversion**

Object Type	Parameters	PCU (MOPS)	RCU	Remark
AAC Main	fs = 48 kHz	5	5	1)
AAC LC	fs = 48 kHz	3	3	1)
AAC SSR	fs = 48 kHz	4	3	1)
AAC LTP	fs = 48 kHz	4	4	1)
AAC Scalable	fs = 48 kHz	5	4	1), 2)
TwinVQ	fs = 24 kHz	2	3	1)
CELP	fs = 8 kHz	1	1	
CELP	fs = 16 kHz	2	1	
CELP	fs = 8/16 kHz (bandwidth scalable)	3	1	
HVXC	fs = 8 kHz	2	1	

TTSI		-	-	4)
General MIDI		4	1	
Wavetable Synthesis	fs = 22.05 kHz	depends on bitstreams (3)	depends on bitstreams (3)	
Main Synthetic		depends on bitstreams (3)	depends on bitstreams (3)	
Algorithmic Synthesis and AudioFX		depends on bitstreams (3)	depends on bitstreams (3)	
Sampling Rate	rf = 2, 3, 4, 6	2	0.5	
ER AAC LC	fs = 48 kHz	3	3	1)
ER AAC LTP	fs = 48 kHz	4	4	1)
ER AAC Scalable	fs = 48 kHz	5	4	1), 2)
ER TwinVQ	fs = 24 kHz	2	3	1)
ER BSAC	fs = 48 kHz (input buffer size=26000bits)	4	4	1)
	fs = 48 kHz (input buffer size=106000bits)	4	8	
ER AAC LD	fs = 48 kHz	3	2	1)
ER CELP	fs = 8 kHz	2	1	
	fs = 16 kHz	3	1	
ER HVXC	fs = 8 kHz	2	1	
ER HILN	fs = 16 kHz, ns=93	15	2	6)
	fs = 16 kHz, ns=47	8	2	
ER Parametric	fs = 8 kHz, ns=47	4	2	5),6)

## Definitions:

- fs = sampling frequency
- rf = ratio of sampling rates

## Notes -

- 1) PCU proportional to sampling frequency.
- 2) Includes core decoder.
- 3) See ISO/IEC 14496-4.
- 4) The complexity for speech synthesis is not taken into account.
- 5) Parametric coder in HILN mode, for HVXC mode see ER HVXC.
- 6) PCS depends on fs and ns, see below.

## PCU for HILN:

The computational complexity of HILN depends on the sampling frequency fs and the maximum number of sinusoids ns to be synthesized simultaneously. The value of ns for a frame is the total number of harmonic and individual lines synthesized in that frame, i.e. the number of starting plus continued plus ending lines. For fs in kHz, the PCU in MOPS is calculated as follows:

$$\text{PCU} = (1 + 0.15 \cdot \text{ns}) \cdot \text{fs} / 16$$

The typical maximum values of ns are 47 for 6 kbit/s HILN and 93 for 16 kbit/s HILN bitstreams.

**1.5.2.3 Levels within the profiles**

A number of 0 stages of interleaving for the EP-tool indicates that the EP is not used in that particular level. The notation used to specify the number of audio channels indicates the number of full bandwidth channels and the number of low-frequency enhancement channels. For example, "5.1" indicates 5 full bandwidth channels and one low-frequency enhancement channel.

Note: For the case of scalable coding schemes, only the first instantiation of each object type will be counted to determine the number of objects relevant to the level definition and complexity metric. For example, in a scalable

coder consisting of a CELP core coder and two enhancement layers implemented by means of AAC scalable objects, one CELP object and one AAC scalable object and their associated complexity metrics are counted since there is almost no overhead associated with the second (and any further) GA enhancement layer.

- **Levels for Speech Audio Profile**

Two levels are defined by number of objects:

1. One speech object.
2. Up to 20 speech objects.

- **Levels for Synthetic Audio Profile**

Three levels are defined :

1. Synthetic Audio 1: All bitstream elements may be used with:
  - "Low processing" (exact numbers in ISO/IEC 14496-4:2000)
  - Only core sample rates may be used
  - No more than one TTSI object
2. Synthetic Audio 2: All bitstream elements may be used with:
  - "Medium processing" (exact numbers in ISO/IEC 14496-4:2000).
  - Only core sample rates may be used.
  - no more than four TTSI objects.
3. Synthetic Audio 3: All bitstream elements may be used with:
  - "High processing" (exact numbers in ISO/IEC 14496-4:2000).
  - no more than twelve TTSI objects.

- **Levels for Scalable Audio Profile**

Four levels are defined by configuration; complexity units define the fourth level:

1. Maximum 24 kHz of sampling rate, one mono object (all object Types).
2. Maximum 24 kHz of sampling rate, one stereo object or two mono objects (all object Types).
3. Maximum 48 kHz of sampling rate, one stereo object or two mono objects (all object Types).
4. Maximum 48 kHz of sampling rate, one 5.1 channels object or multiple objects with at maximum one integer factor sampling rate conversion for a maximum of two channels.  
Flexible configuration is allowed with PCU < 30 and RCU < 19.

- **Levels for Main Audio Profile**

Main Audio Profile contains all natural and synthetic object types. Levels are then defined as a combination of the two different types of levels from the two different metrics defined for natural tools (computation-based metrics) and synthetic tools (macro-oriented metrics).

For Object Types not belonging to the Synthetic Profile four levels are defined:

1. Natural Audio 1: PCU < 40, RCU < 20
2. Natural Audio 2: PCU < 80, RCU < 64
3. Natural Audio 3: PCU < 160, RCU < 128
4. Natural Audio 4: PCU < 320, RCU < 256

For Object Types belonging to the Synthetic Profile the same three Levels are defined as above, i.e. Synthetic Audio 1, Synthetic Audio 2 and Synthetic Audio 3.

Four Levels are then defined for Main Profile:

1. Natural Audio 1 + Synthetic Audio 1
2. Natural Audio 2 + Synthetic Audio 1
3. Natural Audio 3 + Synthetic Audio 2
4. Natural Audio 4 + Synthetic Audio 3

- Levels for the High Quality Audio Profile

Table 1.4 – Levels for the High Quality Audio Profile

Level	Max. channels/object	Max. sampling rate [kHz]	Max PCU <sup>*2</sup>	Max RCU <sup>*2</sup>	EP-Tool: Max. redundancy by class FEC <sup>*1</sup>	EP-Tool: Max. # stages of interleaving per object
1	2	22.05	5	8	0 %	0
2	2	48	10	8	0 %	0
3	5.1	48	25	12 <sup>*3</sup>	0 %	0
4	5.1	48	100	42 <sup>*3</sup>	0 %	0
5	2	22.05	5	8	20%	9
6	2	48	10	8	20%	9
7	5.1	48	25	12 <sup>*3</sup>	20%	22
8	5.1	48	100	42 <sup>*3</sup>	20%	22

\*1: This number does not cover FEC for the EP header, i. e. FEC for the EP header is always permitted. In case of several audio objects the limit is valid independently for each audio object. This value is the maximum redundancy for the Audio object, which has the longest maximum frame length, in each profile & level. For audio object types supporting variable frame lengths and arbitrary bitrates (i. e. any AAC audio object type) this does just extend the required decoder input buffer but does not affect the amount of redundancy actually applied. Nevertheless, the application of any class FEC is not permitted if 0% is specified.

\*2: Level 5 to 8 do not include RAM and computational complexity for the EP tool.

\*3: Sharing of work buffers between multiple objects or channel pair elements is assumed.

- Levels for the Low Delay Audio Profile

Table 1.5 – Levels for the Low Delay Audio Profile

Level	Max. channels/object	Max. sampling rate [kHz]	Max PCU <sup>*2</sup>	Max RCU <sup>*2</sup>	EP-Tool: Max. redundancy by class FEC <sup>*1</sup>	EP-Tool: Max. # stages of interleaving per object
1	1	8	2	1	0 %	0
2	1	16	3	1	0 %	0
3	1	48	3	2	0 %	0
4	2	48	24	12 <sup>*3</sup>	0 %	0
5	1	8	2	1	100%	5
6	1	16	3	1	100%	5
7	1	48	3	2	20%	5
8	2	48	24	12 <sup>*3</sup>	20%	9

\*1: This number does not cover FEC for the EP header, i. e. FEC for the EP header is always permitted. In case of several audio objects the limit is valid independently for each audio object. This value is the maximum redundancy for the Audio object, which has the longest frame length, in each profile & level. For audio object types supporting variable frame lengths and arbitrary bitrates (i. e. any AAC audio object type) this does just extend the required decoder input buffer but does not affect the amount of redundancy actually applied. Nevertheless, the application of any class FEC is not permitted if 0% is specified.

\*2: Level 5 to 8 do not include RAM and computational complexity for the EP tool.

\*3: Sharing of work buffers between multiple objects or channel pair elements is assumed.

- Levels for the Natural Audio Profile

**Table 1.6 – Levels for the Natural Audio Profile**

Level	Max. sampling rate [kHz]	Max PCU <sup>*2</sup>	EP-Tool: Max. redundancy by class FEC <sup>*1</sup>	EP-Tool: Max. # stages of interleaving per object
1	48	20	0 %	0
2	96	100	0 %	0
3	48	20	20%	9
4	96	100	20%	22

\*1: This number does not cover FEC for the EP header, i. e. FEC for the EP header is always permitted. In case of several audio objects the limit is valid independently for each audio object. This value is the maximum redundancy for the Audio object, which has the longest frame length, in each profile & level. For audio object types supporting variable frame lengths and arbitrary bitrates (i. e. any AAC audio object type) this does just extend the required decoder input buffer but does not affect the amount of redundancy actually applied. Nevertheless, the application of any class FEC is not permitted if 0% is specified.

\*2: Level 3 and 4 do not include computational complexity for the EP tool.

No RCU limitations are specified for this profile.

- Levels for the Mobile Audio Internetworking Profile

**Table 1.7 – Levels for the Mobile Audio Internetworking Profile**

Level	Max. channels/object	Max. sampling rate [kHz]	Max PCU <sup>*3</sup>	Max RCU <sup>*2 *3</sup>	Max. # audio objects	EP-Tool: Max. redundancy by class FEC <sup>*1</sup>	EP-Tool: Max. # stages of interleaving per object
1	1	24	2.5	4	1	0 %	0
2	2	48	10	8	2	0 %	0
3	5.1	48	25	12 <sup>*4</sup>	-	0 %	0
4	1	24	2.5	4	1	20%	5
5	2	48	10	8	2	20%	9
6	5.1	48	25	12 <sup>*4</sup>	-	20%	22

\*1: This number does not cover FEC for the EP header, i. e. FEC for the EP header is always permitted. In case of several audio objects the limit is valid independently for each audio object. This value is the maximum redundancy for the Audio object, which has the longest frame length, in each profile & level. For audio object types supporting variable frame lengths and arbitrary bitrates (i. e. any AAC audio object type) this does just extend the required decoder input buffer but does not affect the amount of redundancy actually applied. Nevertheless, the application of any class FEC is not permitted if 0% is specified.

\*2: The maximum RCU for one channel in any object in this profile is 4. For the ER BSAC, this limits the input buffer size. The maximum possible input buffer size in bits for this case is given in PCU/RCU (Table 1.3).

\*3: Level 4 to 6 do not include RAM and computational complexity for the EP tool.

\*4: Sharing of work buffers between multiple objects or channel pair elements are assumed.



## 1.6 Interface to ISO/IEC 14496-1 (MPEG-4 Systems)

### 1.6.1 Introduction

The header streams are transported via MPEG-4 systems. These streams contain configuration information, which is necessary for the decoding process and parsing of the raw data streams. However, an update is only necessary if there are changes in the configuration.

The payloads contain all information varying on a frame to frame basis and therefore carry the actual audio information.

### 1.6.2 Syntax

#### 1.6.2.1 AudioSpecificConfig

Table 1.8 – Syntax of AudioSpecificConfig()

Syntax	No. of bits	Mnemonic
AudioSpecificConfig ()		
{		
<b>audioObjectType;</b>	<b>5</b>	<b>bslbf</b>
<b>samplingFrequencyIndex;</b>	<b>4</b>	<b>bslbf</b>
if ( samplingFrequencyIndex==0xf )		
<b>samplingFrequency;</b>	<b>24</b>	<b>uimbsf</b>
<b>channelConfiguration;</b>	<b>4</b>	<b>bslbf</b>
if ( audioObjectType == 1    audioObjectType == 2    audioObjectType == 3    audioObjectType == 4    audioObjectType == 6    audioObjectType == 7 )		
GASpecificConfig();		
if ( audioObjectType == 8 )		
CelpSpecificConfig();		
if ( audioObjectType == 9 )		
HvxcSpecificConfig();		
if ( audioObjectType == 12 )		
TTSSpecificConfig();		
if ( audioObjectType == 13    audioObjectType == 14    audioObjectType == 15    audioObjectType==16)		
StructuredAudioSpecificConfig();		
if ( audioObjectType == 17    audioObjectType == 19    audioObjectType == 20    audioObjectType == 21    audioObjectType == 22    audioObjectType == 23 )		
GASpecificConfig();		
if ( audioObjectType == 24 )		
ErrorResilientCelpSpecificConfig();		
if ( audioObjectType == 25 )		
ErrorResilientHvxcSpecificConfig();		
if ( audioObjectType == 26    audioObjectType == 27 )		
ParametricSpecificConfig();		
if ( audioObjectType == 17    audioObjectType == 19    audioObjectType == 20    audioObjectType == 21    audioObjectType == 22    audioObjectType == 23    audioObjectType == 24    audioObjectType == 25    audioObjectType == 26    audioObjectType == 27 ) {		
<b>epConfig;</b>	<b>2</b>	<b>bslbf</b>
if ( epConfig == 2    epConfig == 3 ) {		
ErrorProtectionSpecificConfig();		
}		
}		

<pre> if ( epConfig == 3 ) {     <b>directMapping;</b>     if ( ! directMapping ) {         /* tbd */     } } </pre>	<p><b>1</b></p>	<p><b>bslbf</b></p>
--	-----------------	---------------------

**1.6.2.1.1 HvxcSpecificConfig**

Defined in ISO/IEC 14496-3 subpart 2.

**1.6.2.1.2 CelpSpecificConfig**

Defined in ISO/IEC 14496-3 subpart 3.

**1.6.2.1.3 GASpecificConfig**

Defined in subclause in ISO/IEC 14496-3 subpart 4.

**1.6.2.1.4 StructuredAudioSpecificConfig**

Defined in ISO/IEC 14496-3 subpart 5.

**1.6.2.1.5 TTSSpecificConfig**

Defined in ISO/IEC 14496-3 subpart 6.

**1.6.2.1.6 ParametricSpecificConfig**

Defined in subclause in ISO/IEC 14496-3 subpart 7.

**1.6.2.1.7 ErrorProtectionSpecificConfig**

Defined in subclause 1.8.2.1.

**1.6.2.1.8 ErrorResilientCelpSpecificConfig**

Defined in subclause ISO/IEC 14496-3 subpart 3.

**1.6.2.1.9 ErrorResilientHvxcSpecificConfig**

Defined in subclause ISO/IEC 14496-3 subpart 2.

**1.6.2.2 Payloads**

For the NULL object the payload shall be 16 bit signed integer in the range from -32768 to +32767. The payloads for all other audio object types are defined in the corresponding parts. These are the basic entities to be carried by the systems transport layer. Note that for all natural audio coding schemes the output is scaled for a maximum of 32767/-32768. However, the MPEG-4 System compositor expects a scaling. Payloads that are not byte aligned should be zero-padded at the end for transport schemes which require byte alignment.

The following table shows an overview about where the Elementary Stream payloads for the Audio Object Types can be found and where the detailed syntax is defined.

Table 1.9 – Audio Object Types

Audio Object Type	definition of elementary stream payloads and detailed syntax
AAC MAIN	ISO/IEC 14496-3 subpart 4
AAC LC	ISO/IEC 14496-3 subpart 4
AAC SSR	ISO/IEC 14496-3 subpart 4
AAC LTP	ISO/IEC 14496-3 subpart 4
AAC scalable	ISO/IEC 14496-3 subpart 4
TwinVQ	ISO/IEC 14496-3 subpart 4
CELP	ISO/IEC 14496-3 subpart 3
HVXC	ISO/IEC 14496-3 subpart 2
TTSI	ISO/IEC 14496-3 subpart 6
Main synthetic	ISO/IEC 14496-3 subpart 5
Wavetable synthesis	ISO/IEC 14496-3 subpart 5
General MIDI	ISO/IEC 14496-3 subpart 5
Algorithmic Synthesis and Audio FX	ISO/IEC 14496-3 subpart 5
ER AAC LC	ISO/IEC 14496-3 subpart 4
ER AAC LTP	ISO/IEC 14496-3 subpart 4
ER AAC scalable	ISO/IEC 14496-3 subpart 4
ER Twin VQ	ISO/IEC 14496-3 subpart 4
ER BSAC	ISO/IEC 14496-3 subpart 4
ER AAC LD	ISO/IEC 14496-3 subpart 4
ER CELP	ISO/IEC 14496-3 subpart 3
ER HVXC	ISO/IEC 14496-3 subpart 2
ER HILN	ISO/IEC 14496-3 subpart 7
ER Parametric	ISO/IEC 14496-3 subpart 2 and 7

## 1.6.3 Semantics

### 1.6.3.1 AudioObjectType

A five bit field indicating the audio object type. This is the master switch which selects the actual bitstream syntax of the audio data. In general, different object type use a different bitstream syntax. The interpretation of this field is given in the Audio Object Type table in subclause Table 1.1.

### 1.6.3.2 samplingFrequency

The sampling frequency used for this audio object. Either transmitted directly, or coded in the form of **samplingFrequencyIndex**.

### 1.6.3.3 samplingFrequencyIndex

A four bit field indicating the sampling rate used. If **samplingFrequencyIndex** equals 15 then the actual sampling rate is signaled directly by the value of **samplingFrequency**. In all other cases **samplingFrequency** is set to the value of the corresponding entry in table Table 1.10.

Table 1.10 – Sampling Frequency Index

samplingFrequencyIndex	Value
0x0	96000
0x1	88200

0x2	64000
0x3	48000
0x4	44100
0x5	32000
0x6	24000
0x7	22050
0x8	16000
0x9	12000
0xa	11025
0xb	8000
0xc	7350
0xd	reserved
0xe	reserved
0xf	escape value

**1.6.3.4 channelConfiguration**

A four bit field indicating the audio output channel configuration:

**Table 1.11 – Channel Configuration**

value	number of channels	audio syntactic elements, listed in order received	channel to speaker mapping
0	-	-	defined in GASpecificConfig
1	1	single_channel_element	center front speaker
2	2	channel_pair_element	left, right front speakers
3	3	single_channel_element, channel_pair_element	center front speaker, left, right front speakers
4	4	single_channel_element, channel_pair_element, single_channel_element	center front speaker, left, right center front speakers, rear surround speakers
5	5	single_channel_element, channel_pair_element, channel_pair_element	center front speaker, left, right front speakers, left surround, right surround rear speakers
6	5+1	single_channel_element, channel_pair_element, channel_pair_element, lfe_element	center front speaker, left, right front speakers, left surround, right surround rear speakers, front low frequency effects speaker
7	7+1	single_channel_element, channel_pair_element, channel_pair_element, channel_pair_element, lfe_element	center front speaker, left, right center front speakers, left, right outside front speakers, left surround, right surround rear speakers, front low frequency effects speaker
8-15	-	-	reserved

**1.6.3.5 epConfig**

This data element signals what kind of error robust configuration is used.

Table 1.12 – epConfig

epConfig	Description
0	All instances of all error sensitivity categories belonging to one frame are stored within one access unit. There exists one elementary stream per scalability layer, or just one elementary stream in case of non-scalable configurations.
1	Each instance of each sensitivity category belonging to one frame is stored separately within a single access unit, i.e. there exist as many elementary streams as instances defined within a frame.
2	The error protection decoder has to be applied. The definition of EP classes is not normatively defined, but defined at application level. However, the restrictions imposed on EP classes as defined in subclause 1.8 must be satisfied.
3	The error protection decoder has to be applied. The mapping between EP classes and ESC instances is signaled by the data element directMapping.

### 1.6.3.6 direct mapping

This data element identifies the mapping between error protection classes and error sensitivity category instances.

Table 1.13 – directMapping

directMapping	Description
0	Reserved
1	Each error protection class is treated as an instance of an error sensitivity category (one to one mapping), so that the error protection decoder output is equivalent to epConfig=1 data.

## 1.6.4 Upstream

### 1.6.4.1 Introduction

Upstreams are defined to allow for user on a remote side to dynamically control the streaming of the server.

The need for an up-stream channel is signaled to the client terminal by supplying an appropriate elementary stream descriptor declaring the parameters for that stream. The client terminal opens this up-stream channel in a similar manner as it opens the downstream channels. The entities (e.g. media encoders & decoders) that are connected through an up-stream channel are known from the parameters in its elementary stream descriptor and from the association of the elementary stream descriptor to a specific object descriptor.

An up-stream can be associated to a single downstream or a group of down streams. The stream type of the downstream to which the up-stream is associated defines the scope of the up-stream. When the up-stream is associated to a single downstream it carries messages about the downstream it is associated to. The syntax and semantics of messages for MPEG-4 Audio are defined in the next subclause.

### 1.6.4.2 Syntax

Table 1.14 – Syntax of AudioUpstreamPayload()

Syntax	No. of bits	Mnemonic
AudioUpstreamPayload() {		
<b>upStreamType</b> ;	<b>4</b>	<b>uimsbf</b>
switch ( upStreamType ) {		
case 0: /* scalability control */		
<b>numOfLayer</b> ;	<b>6</b>	<b>uimsbf</b>
for ( layer = 0; layer < numOfLayer; layer++ ) {		
<b>avgBitrate[layer]</b> ;	<b>24</b>	<b>uimsbf</b>
}		
}		

```

        break;
    case 1: /* BSAC frame interleaving */
        numOfSubFrame;           5           uimsbf
        break;
    case 2: /* quality feedback */
        multiLayOrSynEle;       1           uimsbf
        if ( multiLayOrSynEle ) {
            layOrSynEle;        6           uimsbf
        }
        else {
            layOrSynEle = 1;
        }
        numFrameExp[layOrSynEle]; 4           uimsbf
        lostFrames[layOrSynEle];  numFrameExp
                                [layOrSynEle]
        break;
    case 3: /* bitrate control */
        avgBitrate;             24          uimsbf
        break;
    default: /* reserved for future use */
        break;
}

```

1.6.4.3 Definitions

**upStreamType** A 4-bit unsigned integer value representing the type of the up-stream as defined in the following Table 1.15

Table 1.15 – Definition of upStreamType

UpStreamType	Type of Audio up-stream
0	scalability control
1	BSAC frame interleaving
2	quality feedback
3	bitrate control
4 – 15	reserved for future use

**avgBitrate[layer]** The average bitrate in bits per second of a large step layer, which the client requests to be transmitted from the server.

**numOfSubFrame** A 5-bit unsigned integer value representing the number of the frames which are grouped and transmitted in order to reduce the transmission overhead. The transmission overhead is decreased but the delay is increased as numOfSubFrame is increased.

**multiLayOrSynEle** This bit signals, whether or not a multi-channel or multi-layer configuration is used. Only in that case a layer number or a syntactic element number needs is transmitted.

**layOrSynEle** A 6-bit unsigned integer value representing the number of the syntactic elements (in case of multi-channel setup) or the number of the layers (in case of multi-layer setup), to which the following quality feedback information belongs. This number refers to one of the layers or one of the syntactic elements contained within the associated Audio object. If the Audio object does neither support scalability nor multi-channel capabilities, this value is implicitly set to 1.

**numFrameExp[layOrSynEle]** This value indicates the number of last recently passed frames  $(2^{numFrameExp} - 1)$  considered in the following lostFrames value.

**lostFrames[layOrSynEle]** This field contains the number of lost frames with respect to the indicated layer or syntactic element within the last recently passed frames signalled by numFrameExp.

**avgBitrate** The average bitrate in bits per second of the whole Audio object, which the client requests to be transmitted from the server.

#### 1.6.4.4 Decoding process

First, **upStreamType** is parsed which represents the type of the up-stream. The remaining decoding process depends upon the type of the up-stream.

##### 1.6.4.4.1 Decoding of scalability control

Next is the value **numOfLayer**. It represents the number of the data elements **avgBitrate** to be read. **avgBitrate** follows.

##### 1.6.4.4.2 Decoding of BSAC frame interleaving

The data element to be read is **numOfSubFrame**. It represents the number of the sub-frames to be interleaved in BSAC tool. BSAC can allow for runtime adjustments to the quality of service. When the content of upstream is transmitted from the client to the server to implement a stream dynamically and interactively. BSAC data are split and interleaved in the server. The detailed process for implementing an AU payload in the server is described in Annex 4.B.17.

##### 1.6.4.4.3 Decoding of quality feedback

The real frame loss rate in percent can be derived using the following formula:

$$frameLossRate[layOrSynEle] = \frac{lostFrames[layOrSynEle]}{2^{numFrameExp[layOrSynEle]} - 1} * 100\%$$

##### 1.6.4.4.4 Decoding of bitrate control

**avgBitrate** is parsed.

## 1.7 MPEG-4 Audio transport stream

### 1.7.1 Overview

This subclause defines a mechanism to transport ISO/IEC 14496-3 (MPEG-4 Audio) streams without using ISO/IEC 14496-1 (MPEG-4 Systems) for audio-only applications. Figure 1.1 shows the concept of MPEG-4 Audio transport. The transport mechanism uses a two-layer approach, namely a multiplex layer and a synchronization layer. The multiplex layer (Low-overhead MPEG-4 Audio Transport Multiplex: LATM) manages multiplexing of several MPEG-4 Audio payloads and AudioSpecificConfig() elements. The synchronization layer specifies a self-synchronized syntax of the MPEG-4 Audio transport stream which is called Low Overhead Audio Stream (LOAS). The Interface format to a transmission layer depends on the conditions of the underlying transmission layer as follows:

- **LOAS shall be used for the transmission over channels where no frame synchronization is available.**
- **LOAS may be used for the transmission over channels with fixed frame synchronization.**
- **A multiplexed element (AudioMuxElement()/EPMuxElement()) without synchronization shall be used only for transmission channels where an underlying transport layer already provides frame synchronization that can handle arbitrary frame size.**

The details in the LOAS format and the AudioMuxElement() format are described in subclauses 1.7.2 and 1.7.3, respectively.

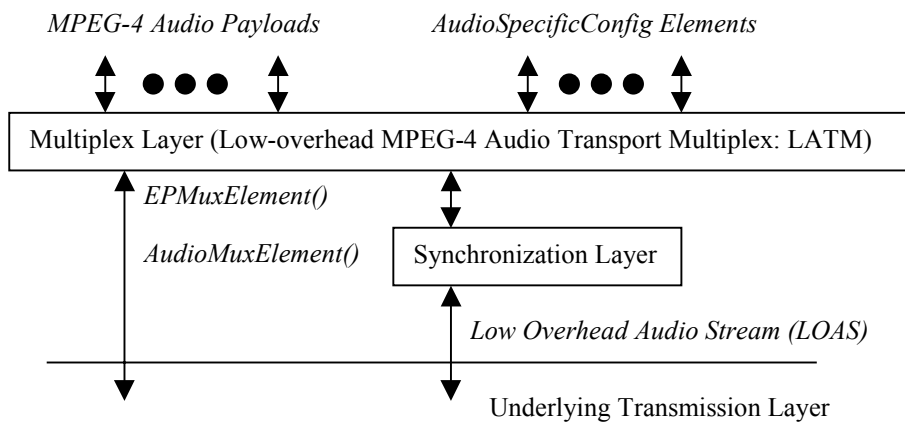


Figure 1.1 – Concept of MPEG-4 Audio Transport

The mechanism defined in this subclause should not be used for transmission of TTSI object (12), Main Synthetic object (13), Wavetable Synthesis object (14), General MIDI object (15) and Algorithmic Synthesis and Audio FX object (16). For these object types, other multiplex and transport mechanisms might be used, e.g. those defined in MPEG-4 Systems.

### 1.7.2 Synchronization Layer

The synchronization layer provides the multiplexed element with a self-synchronized mechanism to generate LOAS. The LOAS has three different types of format, namely AudioSyncStream(), EPAudioSyncStream() and AudioPointerStream(). The choice for one of the three formats is dependent on the underlying transmission layer.

- AudioSyncStream()

AudioSyncStream() consists of a syncword, the multiplexed element with byte alignment, and its length information. The maximum byte-distance between two syncwords is 8192 bytes. This self-synchronized stream shall be used for the case that the underlying transmission layer comes without any frame synchronization.

- EPAudioSyncStream()

For error prone channels, an alternative version to AudioSyncStream() is provided. This format has the same basic functionality as the previously described AudioSyncStream(). However, it additionally provides a longer syncword and a frame counter to detect lost frames. The length information and the frame counter are additionally protected by a FEC code.

- AudioPointerStream()

AudioPointerStream() shall be used for applications using a underlying transmission layer with fixed frame synchronization, where transmission framing can not be synchronized with the variable length multiplexed element. Figure 1.2 shows synchronization in AudioPointerStream(). This format utilizes a pointer indicating the start of the next multiplex element in order to synchronize the variable length payload with the constant transmission frame.



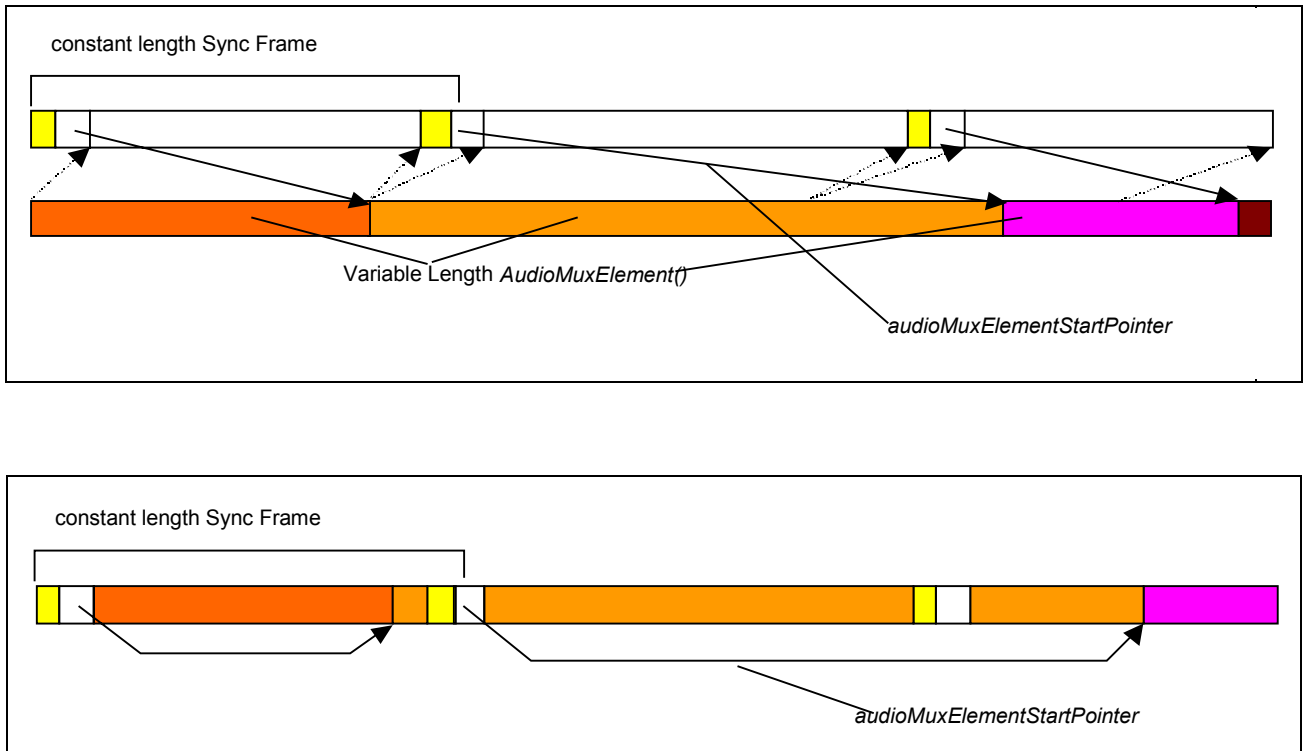


Figure 1.2 – Synchronization in AudioPointerStream()

1.7.2.1 Syntax

Table 1.16 – Syntax of AudioSyncStream()

Syntax	No. of bits	Mnemonic
AudioSyncStream() {		
while (nextbits() == 0x2B7) { /* syncword */	11	bslbf
audioMuxLengthBytesLast;	13	uimbsf
AudioMuxElement( 1 );		
ByteAlign();		
}		
}		

Table 1.17 – Syntax of EPAudioSyncStream()

Syntax	No. of bits	Mnemonic
EPAudioSyncStream() {		
while (nextbits() == 0x4de1) { /* syncword */	16	bslbf
futureUse;	4	uimbsf
audioMuxLengthBytes;	13	uimbsf
frameCounter;	5	uimbsf
headerParity;	18	bslbf
EPMuxElement( 1, 1 );		
}		
}		

```
}
}
```

**Table 1.18 – Syntax of AudioPointerStream()**

Syntax	No. of bits	Mnemonic
AudioPointerStream( length )		
{		
ByteAlign();		
<b>audioMuxElementStartPointer</b> ;	ceil(ld(length))	<b>uimsbf</b>
AudioMuxElement( 1 );		
}		

**1.7.2.2 Semantics**

- audioMuxLengthBytesLast** A 13-bit field indicating the byte length of the multiplexed element with byte alignment.
- futureUse** A 4-bit field for future use.
- audioMuxLengthBytes** A 13-bit field indicating the byte length of the multiplexed element.
- frameCounter** A 5-bit field indicating a sequential number which is used to detect lost frames. The number is continuously incremented for each multiplexed element as a modulo counter.
- headerParity** A 18-bit field which contains a BCH (36,18) code shortened from BCH (63,45) code for the elements **audioMuxLengthBytes** and **frameCounter**. The generator polynomial is  $x^{18}+x^{17}+x^{16}+x^{15}+x^9+x^7+x^6+x^3+x^2+x+1$ . The value is calculated with this generation polynomial as described in subclause 1.8.4.3.
- audioMuxElementStartPointer** A field indicating the start point of the multiplexed element. The number of bits required for this field is calculated as  $nbits = \text{ceil}(\log_2(\text{the transmission frame length}))$ . The transmission frame length should be provided from the transmission layer. The maximum possible value of this field is reserved to signal that there is no start of an access unit in this sync frame.
- AudioMuxElement()** A multiplexed element as specified in subclause 1.7.3.
- EPMuxElement()** An error resilient multiplexed element as specified in subclause 1.7.3.

**1.7.3 Multiplex Layer**

The LATM layer multiplexes several MPEG-4 Audio payloads and AudioSpecificConfig() syntax elements into one multiplexed element. The multiplexed element format is selected between AudioMuxElement() and EPMuxElement() depending on whether error resilience is required in the multiplexed element itself, or not. EPMuxElement() is an error resilient version of AudioMuxElement() and may be used for error prone channels.

The multiplexed elements can be directly conveyed on transmission layers with frame synchronization. In this case, the first bit of the multiplexed element shall be located at the first bit of a transmission payload in the underlying transmission layer. If the transmission payload allows only byte-aligned payload, zero-padding bits for byte alignment shall follow the multiplexed element. The number of the padding bits should be less than 8. These padding bits should be removed when the multiplexed element is de-multiplexed into the MPEG-4 Audio payloads. Then, the MPEG-4 Audio payloads are forwarded to the corresponding MPEG-4 Audio decoder tool.

Usage of LATM in case of scalable configurations with CELP core and AAC enhancement layer(s):

- *Instances of the AudioMuxElement() are transmitted in equidistant manner.*
- *The represented timeframe of one AudioMuxElement() is similar to a multiple of a super-frame timeframe.*
- *The relative number of bits for a certain layer within any AudioMuxElement() compared to the total number of bits within this AudioMuxElement() is equal to the relative bitrate of that layer compared to the bitrate of all layers.*

- In case of  $coreFrameOffset=0$  and  $latmBufferFullness=0$ , all core coder frames and all AAC frames of a certain super-frame are stored within the same instance of `AudioMuxElement()`.
- In case of  $coreFrameOffset>0$ , several or all core coder frames are stored within previous instances of `AudioMuxElement()`.
- Any core layer related configuration information refers to the core frames transmitted within the current instance of the `AudioMuxElement()`, independent of the value of  $coreFrameOffset$ .
- A specified  $latmBufferFullness$  is related to the first AAC frame of the first super-frame stored within the current `AudioMuxElement()`.
- The value of  $latmBufferFullness$  can be used to determine the location of the first bit of the first AAC frame of the current layer of the first super-frame stored within the current `AudioMuxElement()` by means of a backpointer:

$$backPointer = -meanFrameLength + latmBufferFullness + currentFrameLength$$

The backpointer value specifies the location as a negative offset from the current `AudioMuxElement()`, i. e. it points backwards to the beginning of an AAC frame located in already received data. Any data not belonging to the payload of the current AAC layer is not taken into account. If  $latmBufferFullness==0$ , then the AAC frame starts after the current `AudioMuxElement()`.

### 1.7.3.1 Syntax

Table 1.19 – Syntax of `EPMuxElement()`

Syntax	No. of bits	Mnemonic
<code>EPMuxElement( epDataPresent , muxConfigPresent )</code>		
{		
if( epDataPresent ) {		
<b>epUsePreviousMuxConfig;</b>	1	<b>bslbf</b>
<b>epUsePreviousMuxConfigParity;</b>	2	<b>bslbf</b>
if( !epUsePreviousMuxConfig ) {		
<b>epSpecificConfigLength;</b>	10	<b>bslbf</b>
<b>epSpecificConfigLengthParity;</b>	11	<b>bslbf</b>
ErrorProtectionSpecificConfig();		
ErrorProtectionSpecificConfigParity();		
}		
ByteAlign();		
EPAudioMuxElement( muxConfigPresent );		
} else {		
AudioMuxElement( muxConfigPresent );		
ByteAlign();		
}		
}		

Table 1.20 – Syntax of `AudioMuxElement()`

Syntax	No. of bits	Mnemonic
<code>AudioMuxElement( muxConfigPresent )</code>		
{		
if( muxConfigPresent ) {		
<b>useSameStreamMux;</b>	1	<b>bslbf</b>
if( !useSameStreamMux )		
StreamMuxConfig();		
}		
if( audioMuxVersion == 0 ) {		
for( i=0; i<=numSubFrames; i++ ) {		
PayloadLengthInfo();		
PayloadMux();		
}		
}		

```

    }
    if( otherDataPresent ) {
        for( i=0; i<otherDataLenBits; i++ ) {
            otherDataBit;
        }
    }
}
else {
    /* tbd */
}
}

```

**Table 1.21 – Syntax of StreamMuxConfig()**

Syntax	No. of bits	Mnemonic
StreamMuxConfig() {		
<b>audioMuxVersion;</b>	1	bslbf
if (audioMuxVersion == 0) {		
streamCnt = 0;		
<b>allStreamsSameTimeFraming;</b>	1	uimsbf
<b>numSubFrames;</b>	6	uimsbf
<b>numProgram;</b>	4	uimsbf
for ( prog = 0; prog <= numProgram; prog++ ) {		
<b>numLayer;</b>	3	uimsbf
for ( lay = 0; lay <= numLayer; lay++ ) {		
progSIdx[streamCnt]=prog; laySIdx[streamCnt]=lay;		
streamID [ prog][ lay] = streamCnt++;		
if ( prog == 0 & lay == 0 ) {		
AudioSpecificConfig();		
} else {		
<b>useSameConfig;</b>	1	uimsbf
if ( !useSameConfig )		
AudioSpecificConfig();		
}		
<b>frameLengthType</b> [streamID[prog][ lay]];	3	uimsbf
if ( frameLengthType[streamID[prog][lay] == 0 ) {		
<b>latmBufferFullness</b> [streamID[prog][ lay]];	8	uimsbf
if ( ! allStreamsSameTimeFraming ) {		
if ((AudioObjectType[lay]==6		
AudioObjectType[lay]== 20) &&		
(AudioObjectType[lay-1]==8		
AudioObjectType[lay-1]==24)) {		
<b>coreFrameOffset;</b>	6	uimsbf
}		
}		
} else if( frameLengthType[streamID[prog][ lay]] == 1 ) {		
<b>frameLength</b> [streamID[prog][lay]];	9	uimsbf
} else if ( frameLengthType[streamID[prog][ lay]] == 4		
frameLengthType[streamID[prog][ lay]] == 5		
frameLengthType[streamID[prog][ lay]] == 3 ) {		
<b>CELPframeLengthTableIndex</b> [streamID[prog][lay]];	6	uimsbf
} else if ( frameLengthType[streamID[prog][ lay]] == 6		
frameLengthType[streamID[prog][ lay]] == 7 ) {		
<b>HVXCframeLengthTableIndex</b> [streamID[prog][ lay]];	1	uimsbf
}		
}		
}		
}		
}		

<b>otherDataPresent;</b>	<b>1</b>	<b>uimsbf</b>
if (otherDataPresent) {		
otherDataLenBits = 0; /* helper variable 32bit */		
do {		
otherDataLenBits = otherDataLenBits * 2^8;		
<b>otherDataLenEsc;</b>	<b>1</b>	<b>uimsbf</b>
<b>otherDataLenTmp;</b>	<b>8</b>	<b>uimsbf</b>
otherDataLenBits = otherDataLenBits + otherDataLenTmp;		
} while (otherDataLenEsc);		
}		
<b>crcCheckPresent;</b>	<b>1</b>	<b>uimsbf</b>
if( crcCheckPresent ) <b>crcChecksum;</b>	<b>8</b>	<b>uimsbf</b>
}		
else {		
/* tbd */		
}		

Table 1.22 – Syntax of PayloadLengthInfo()

Syntax	No. of bits	Mnemonic
<pre> PayloadLengthInfo() {     if( allStreamsSameTimeFraming ) {         for ( prog = 0; prog &lt;= numProgram; prog++ ) {             for ( lay = 0; lay &lt;= numLayer; lay++ ) {                 if( frameLengthType[streamID[prog][lay]] == 0 ) {                     do { /* always one complete access unit */                         <b>tmp;</b>                         MuxSlotLengthBytes[streamID[prog][lay]] += tmp;                     } while( tmp==255 );                 } else {                     if ( frameLengthType[streamID[prog][lay]] == 5                            frameLengthType[streamID[prog][lay]] == 7                            frameLengthType[streamID[prog][lay]] == 3 ) {                         <b>MuxSlotLengthCoded</b>[streamID[prog][lay]];                     }                 }             }         }     } else {         <b>numChunk;</b>         for ( chunkCnt=0; chunkCnt &lt;= numChunk; chunkCnt++ ) {             <b>streamIdx;</b>             prog = progCIdx[chunkCnt] = progSIdx[streamIdx];             lay = layCIdx[chunkCnt] = laySIdx [streamIdx];             if( frameLengthType[streamID[prog][lay]] == 0 ) {                 do { /* not necessarily a complete access unit */                     <b>tmp;</b>                     MuxSlotLengthBytes[streamID[prog][lay]] += tmp;                 } while ( tmp == 255);                 <b>AuEndFlag</b>[streamID[prog][lay]];             } else {                 if ( frameLengthType[streamID[prog][lay]] == 5                        frameLengthType[streamID[prog][lay]] == 7                        frameLengthType[streamID[prog][lay]] == 3 ) {                     <b>MuxSlotLengthCoded</b>[streamID[prog][lay]];                 }             }         }     } } </pre>	<p><b>8</b></p> <p><b>2</b></p> <p><b>4</b></p> <p><b>4</b></p> <p><b>8</b></p> <p><b>1</b></p> <p><b>2</b></p>	<p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>bslbf</b></p> <p><b>uimsbf</b></p>

```

    }
  }
}

```

**Table 1.23 – Syntax of PayloadMux()**

Syntax	No. of bits	Mnemonic
<pre> PayloadMux() {   if( allStreamsSameTimeFraming ) {     for ( prog = 0; prog &lt;= numProgram; prog++ ) {       for ( lay = 0; lay &lt;= numLayer; lay++ ) {         payload [streamID[prog][ lay]];       }     }   } else {     for ( chunkCnt=0; chunkCnt &lt;= numChunk; chunkCnt++ ) {       prog = progCIdx[chunkCnt];       lay = layCIdx [chunkCnt];        payload [streamID[prog][ lay]];     }   } } </pre>		

**1.7.3.2 Semantics**

In order to parse an AudioMuxElement(), a muxConfigPresent flag shall be set at the underlying layer. If muxConfigPresent is set to 1, this indicates multiplexing configuration (StreamMuxConfig()) is multiplexed into AudioMuxElement(), i.e. in-band transmission. If not, StreamMuxConfig() should be conveyed through out-band means, such as session announcement/description/control protocols. For parsing of EPMuxElement(), an epDataPresent flag shall be additionally set at the underlying layer. If epDataPresent() is set to 1, this indicates EPMuxElement() has error resiliency. If not, the format of EPMuxElement() is identical to AudioMuxElement(). The default for both flags is 1.

muxConfigPresent	Description
0	out-band transmission of StreamMuxConfig()
1	in-band transmission of StreamMuxConfig()

epDataPresent	Description
0	EPMuxElement() is identical to AudioMuxElement()
1	EPMuxElement() has error resiliency

**epUsePreviousMuxConfig** A flag indicating whether the configuration for the MPEG-4 Audio EP tool in the previous frame is applied in the current frame.

epUsePreviousMuxConfig	Description
0	The configuration for the MPEG-4 Audio EP tool is present
1	The configuration for the MPEG-4 Audio EP tool is not present. The previous configuration should be applied

**epUsePreviousMuxConfigParity** A 2-bits element which contains the parity for **epUsePreviousMuxConfig**. Each bit is a repetition of **epUsePreviousMuxConfig**. Majority decides.

**epHeaderLength** A 10-bit field to indicate the size of **ErrorProtectionSpecificConfig()**

**epHeaderLengthParity:** A 11-bit field for **epHeaderLength**, calculated as described in subclause 1.8.4.3 with “1)Basic set of FEC codes”.  
Note: This means shortened Golay(23,12) is used

**ErrorProtectionSpecificConfig()** Configuration information for the EP tool which is applied to **AudioMuxElement()** as defined in subclause 1.8.2.1.

**ErrorProtectionSpecificConfigParity()** The parity bits for **ErrorProtectionSpecificConfig()**, calculated as described in subclause 1.8.4.3 with “1) Basic Set of FEC codes”.

**EPAudioMuxElement()** Error resilient multiplexed element that is generated by applying the EP tool to **AudioMuxElement()** as specified by **ErrorProtectionSpecificConfig()**. Therefore data elements in **AudioMuxElement()** are subdivided into different categories depending on their error sensitivity and collected in instances of these categories. Following sensitivity categories are defined:

elements	error sensitivity category
useSameStreamMux + <b>StreamMuxConfig()</b>	0
<b>PayloadLengthInfo()</b>	1
<b>PayloadMux()</b>	2
otherDataBits	3

Note 1: There might be more than one instance of error sensitivity category 1 and 2 depending on the value of the variable **numSubFrames** defined in **StreamMuxConfig()**. Figure 1.3 shows an example for the order of the instances assuming **numSubFrames** is one (1).



**Figure 1.3 – Instance order in EPAudioMuxElement()**

Note 2: **EPAudioMuxElement()** has to be byte aligned, therefore **bit\_stuffing** in **ErrorProtectionSpecificConfig()** should be always on.

**useSameStreamMux** A flag indicating whether the multiplexing configuration in the previous frame is applied in the current frame.

useSameStreamMux	Description
0	The multiplexing configuration is present.
1	The multiplexing configuration is not present. The previous configuration should be applied.

**audioMuxVersion** A data element to signal the bitstream syntax version. possible values: 0 (default), 1 (reserved for future extensions).

**otherDataBit** A 1-bit field indicating the other data information.

**allStreamsSameTimeFraming** A flag indicating whether all payloads, which are multiplexed in **PayloadMux()**, share a common time base.

**numSubFrames** A field indicating how many **PayloadMux()** frames are multiplexed (**numSubFrames+1**). If more than one **PayloadMux()** frame are multiplexed, all **PayloadMux()** share a common **StreamMuxConfig()**.The minimum value is 0 indicating 1 subframe.

**numProgram** A field indicating how many programs are multiplexed (**numProgram+1**). The minimum value is 0 indicating 1 program.

**numLayer** A field indicating how many scalable layers are multiplexed (**numLayer+1**). The minimum value is 0 indicating 1 layer.

**useSameConfig** A flag indicating whether **AudioSpecificConfig()** for the payload in the previous layer or program is applied for the payload in the current layer or program.

useSameConfig	Description
0	AudioSpecificConfig() is present.
1	AudioSpecificConfig() is not present. AudioSpecificConfig() in the previous layer or program should be applied.

**frameLengthType**

A field indicating the frame length type of the payload. For CELP and HVXC objects, the frame length (bits/frame) is stored in tables and only the indexes to point out the frame length of the current payload is transmitted instead of sending the frame length value directly.

frameLengthType	Description
0	Payload with variable frame length. The payload length in bytes is directly specified with 8-bit codes in PayloadLengthInfo().
1	Payload with fixed frame length. The payload length in bits is specified with frameLength in StreamMuxConfig().
2	Reserved
3	Payload for a CELP object with one of 2 kinds of frame length. The payload length is specified by two table-indexes, namely CELPframeLengthTableIndex and MuxSlotLengthCoded.
4	Payload for a CELP or ER_CELP object with fixed frame length. CELPframeLengthTableIndex specifies the payload length.
5	Payload for an ER_CELP object with one of 4 kinds of frame length. The payload length is specified by two table-indexes, namely CELPframeLengthTableIndex and MuxSlotLengthCoded.
6	Payload for a HVXC or ER_HVXC object with fixed frame length. HVXCframeLengthTableIndex specifies the payload length.
7	Payload for an HVXC or ER_HVXC object with one of 4 kinds of frame length. The payload length is specified by two table-indexes, namely HVXCframeLengthTableIndex and MuxSlotLengthCoded.

**latmBufferFullness**

state of the bit reservoir. It is transmitted as the number of available bits in the bit reservoir divided by the number of audio channels divided by 32 and truncated to an integer value (mean number of 32 bit words per channel remaining in the encoder buffer after encoding the first audio frame in the AudioMuxElement()). A value of hexadecimal FF signals that the bitstream is a variable rate bitstream. In this case, buffer fullness is not applicable.

**coreFrameOffset**

identifies the first CELP frame of the current super-frame. It is defined only in case of scalable configurations with CELP core and AAC enhancement layer(s) and transmitted with the first AAC enhancement layer. The value 0 identifies the first CELP frame following StreamMuxConfig() as the first CELP frame of the current super-frame. A value > 0 signals the number of CELP frames that the first CELP frame of the current super-frame is transmitted earlier in the bitstream.

**frameLength**

A field indicating the frame length of the payload with frameLengthType of 1. The payload length in bits is specified as  $8 * (\text{frameLength} + 20)$ .

**CELPframeLengthTableIndex** A field indicating one of two indexes for pointing out the frame length for a CELP or ER\_CELP object. (Table 1.25 and Table 1.26)

**HVXCframeLengthTableIndex** A field indicating one of two indexes for pointing out the frame length for a HVXC or ER\_HVXC object. (Table 1.24)

**otherDataPresent**

A flag indicating the presence of the other data than audio payloads.



otherDataPresent	Description
0	The other data than audio payload otherData is not multiplexed.
1	The other data than audio payload otherData is multiplexed.

otherDataLenBits  
**otherDataLenEsc**  
**otherDataLenTmp**  
**crcCheckPresent**

A helper variable indicating the length in bits of the other data.  
 A field indicating whether there is more than one otherDataLenTmp data element following.  
 A field used to calculate otherDataLenBits.  
 A flag indicating the presence of CRC check bits for StreamMuxConfig() elements.

crcCheckPresent	Description
0	CRC check bits are not present.
1	CRC check bits are present.

**crcChecksum**  
**tmp**

A field indicating the CRC check bits.  
 A field indicating the payload length of the payload with frameLengthType of 0. The value 255 is used as an escape value and indicates that at least one more **tmp** value is following. The overall length of the transmitted payload is calculated by summing up the partial values.

**MuxSlotLengthCoded**  
**numChunk**

A field indicating one of two indexes for pointing out the payload length for CELP, HVXC, ER\_CELP, and ER\_HVXC objects.  
 A field indicating the number of payload chunks (numChunk+1). Each chunk may belong to an access unit with a different time base; only used if allStreamsSameTimeFraming is set to zero. The minimum value is 0 indicating 1 chunk.

**streamIndx**  
**chunkCnt**  
**progSindx,laySindx**  
**progCindx,layCindx**  
**AuEndFlag**

A field indicating the stream. Used if payloads are splitted into chunks.  
 Helper variable to count number of chunks.  
 Helper variables to identify program and layer number from **streamIndx**.  
 Helper variables to identify program and layer number from **chunkCnt**.  
 A flag indicating whether the payload is the last fragment, in the case that an access unit is transmitted in pieces.

AuEndFlag	Description
0	The fragmented piece is not the last one.
1	The fragmented piece is the last one.

**Table 1.24 – Frame length of HVXC [bits]**

frameLengthType[]	HVXCframeLengthTableIndex[]	MuxSlotLengthCoded			
		00	01	10	11
6	0	40			
6	1	80			
7	0	40	28	2	0
7	1	80	40	25	3

**Table 1.25 – Frame Length of CELP Layer 0 [bits]**

CELPframeLengthTable Index	Fixed-Rate frameLengthType[] =4	1-of-4 Rates (Silence Compression) frameLengthType[]=5				1-of-2 Rates (FRC) frameLengthType[]=3	
		MuxSlotLengthCoded				MuxSlotLengthCoded	
		00	01	10	11	00	01
0	154	156	23	8	2	156	134

1	170	172	23	8	2	172	150
2	186	188	23	8	2	188	166
3	147	149	23	8	2	149	127
4	156	158	23	8	2	158	136
5	165	167	23	8	2	167	145
6	114	116	23	8	2	116	94
7	120	122	23	8	2	122	100
8	126	128	23	8	2	128	106
9	132	134	23	8	2	134	112
10	138	140	23	8	2	140	118
11	142	144	23	8	2	144	122
12	146	148	23	8	2	148	126
13	154	156	23	8	2	156	134
14	166	168	23	8	2	168	146
15	174	176	23	8	2	176	154
16	182	184	23	8	2	184	162
17	190	192	23	8	2	192	170
18	198	200	23	8	2	200	178
19	206	208	23	8	2	208	186
20	210	212	23	8	2	212	190
21	214	216	23	8	2	216	194
22	110	112	23	8	2	112	90
23	114	116	23	8	2	116	94
24	118	120	23	8	2	120	98
25	120	122	23	8	2	122	100
26	122	124	23	8	2	124	102
27	186	188	23	8	2	188	166
28	218	220	40	8	2	220	174
29	230	232	40	8	2	232	186
30	242	244	40	8	2	244	198
31	254	256	40	8	2	256	210
32	266	268	40	8	2	268	222
33	278	280	40	8	2	280	234
34	286	288	40	8	2	288	242
35	294	296	40	8	2	296	250
36	318	320	40	8	2	320	276
37	342	344	40	8	2	344	298
38	358	360	40	8	2	360	314
39	374	376	40	8	2	376	330
40	390	392	40	8	2	392	346
41	406	408	40	8	2	408	362
42	422	424	40	8	2	424	378
43	136	138	40	8	2	138	92
44	142	144	40	8	2	144	98
45	148	150	40	8	2	150	104
46	154	156	40	8	2	156	110
47	160	162	40	8	2	162	116
48	166	168	40	8	2	168	122
49	170	172	40	8	2	172	126
50	174	176	40	8	2	176	130
51	186	188	40	8	2	188	142
52	198	200	40	8	2	200	154
53	206	208	40	8	2	208	162
54	214	216	40	8	2	216	170
55	222	224	40	8	2	224	178
56	230	232	40	8	2	232	186
57	238	240	40	8	2	240	194
58	216	218	40	8	2	218	172
59	160	162	40	8	2	162	116
60	280	282	40	8	2	282	238

61	338	340	40	8	2	340	296
62-63	reserved						

Table 1.26 – Frame Length of CELP Layer 1-5 [bits]

CELPframeLengthTableIndex	Fixed-Rate frameLengthType []=4	1-of-4 Rates (Silence Compression) frameLengthType[]=5			
		MuxSlotLengthCoded			
		00	01	10	11
0	80	80	0	0	0
1	60	60	0	0	0
2	40	40	0	0	0
3	20	20	0	0	0
4	368	368	21	0	0
5	416	416	21	0	0
6	464	464	21	0	0
7	496	496	21	0	0
8	284	284	21	0	0
9	320	320	21	0	0
10	356	356	21	0	0
11	380	380	21	0	0
12	200	200	21	0	0
13	224	224	21	0	0
14	248	248	21	0	0
15	264	264	21	0	0
16	116	116	21	0	0
17	128	128	21	0	0
18	140	140	21	0	0
19	148	148	21	0	0
20-63	reserved				

## 1.8 Error protection

### 1.8.1 Overview of the tools

For error resilient audio object types, the error protection (EP) tool may be applied. The usage of this tool is signalled by the **epConfig** field. The input of the EP tool decoder consists of error protected access units. In case usage of the EP tool decoder is signalled by epConfig, the following restrictions apply:

- There exists one elementary stream per scalability layer, or just one elementary stream in case of non-scalable configurations.
- The output of the EP decoder is a set of several EP classes. The concatenation of EP classes at the output of the EP decoder is identical to epConfig = 0 data.

The definition of an EP class depends on **epConfig** and **directMapping**. For epConfig = 2, EP classes are not strictly defined. Their exact content is to be defined at application level, although the above mentioned restrictions have to be fulfilled. For epConfig = 3, the mapping between EP classes and instances of error sensitivity categories (ESCs) is normatively defined. In this case, the mapping is signalled by **directMapping**. In case directMapping = 1, each EP class maps exactly to one instance of an error sensitivity class. The EP decoder output then is identical to the case in which epConfig = 1. Figure 1.4 summarises the usage of EP classes, depending on the value of epConfig.

The error protection tool (EP tool) provides the unequal error protection (UEP) capability to the ISO/IEC 14496-3 codecs. The main features of the EP tool are as follows:

- providing a set of error correcting/detecting codes with wide and small-step scalability, in performance and in redundancy
- providing a generic and bandwidth-efficient error protection framework, which covers both fixed-length frame bitstreams and variable-length frame bitstreams
- providing a UEP configuration control with low overhead

The basic idea of UEP is to divide the frame into sub-frames according to the bit error sensitivities (these sub-frames are referred to be as classes in the following subclauses), and to protect these sub-frames with appropriate strength of FEC and/or CRC. If this would not be done, the decoded audio quality is determined by how the most error sensitive part is corrupted, and thus the strongest FEC/CRC has to be applied to the whole frame, requiring much more redundancy.

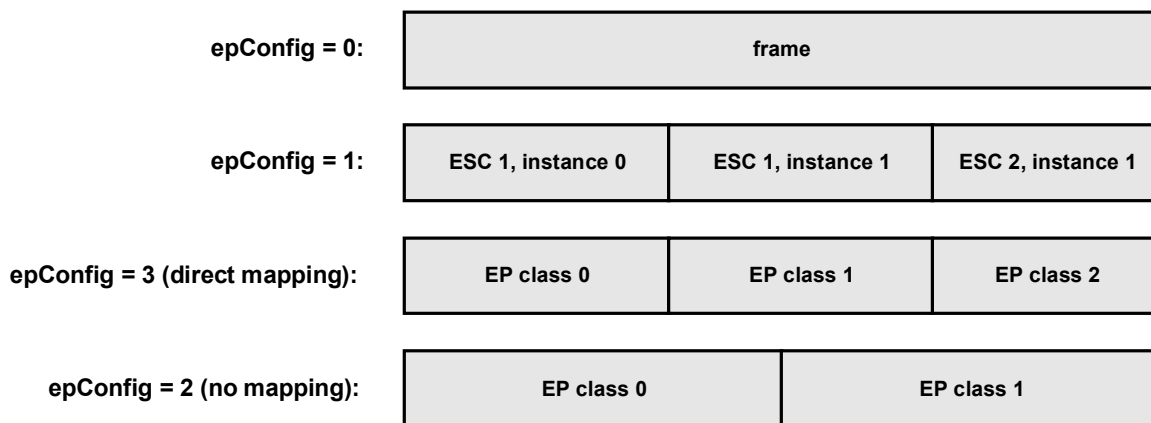


Figure 1.4 – EP classes for different epConfig values

In order to apply UEP to audio frames, the following information is required:

1. Number of classes
2. Number of bits each class contains
3. The CRC code to be applied for each class, which can be presented as a number of CRC bits
4. The FEC code to be applied for each class

This information is called as “frame configuration parameters” in the following sections. The same information is used to decode the UEP encoded frames; thus they have to be transmitted. To transmit them effectively, the frame structures of MPEG-4 audio algorithms have been taken into account for this EP tool.

The MPEG-4 audio frame structure can be categorized into three different approaches from the viewpoint of UEP application:

1. All the frame configurations are constant while the transmission (as CELP).
2. The frame configurations are restricted to be one of the several patterns (as Twin-VQ).
3. Most of the parameters are constant during the transmission, but some parameters can be different frame by frame (as AAC).

To utilize these characteristics, the EP tool uses two paths to transmit the frame configuration parameters. One is the out-of-band signaling, which is the same way as the transmission of codec configuration parameters. The parameters that are shared by the frames are transmitted through this path. In case there are several patterns of configuration, all these patterns are transmitted with indices. The other is the in-band transmission, which is made by defining the EP-frame structure with a header. Only the parameters that are not transmitted out-of-band are

transmitted through this path. With this parameter transmission technique, the amount of in-band information, which is a part of the redundancy caused by the EP tool, is minimized.

With these parameters, each class is FEC/CRC encoded and decoded. To enhance the performance of this error protection, an interleaving technique is adopted. The objective of interleaving is to randomize burst errors within the frames, and this is not desirable for the class that is not protected. This is because there are other error resilience tools whose objective is to localize the effect of the errors, and randomization of errors with interleaving would have a harmful influence on such part of bitstream.

The outline of the EP encoder and EP decoder is figured out in Figure 1.5 and Figure 1.6.

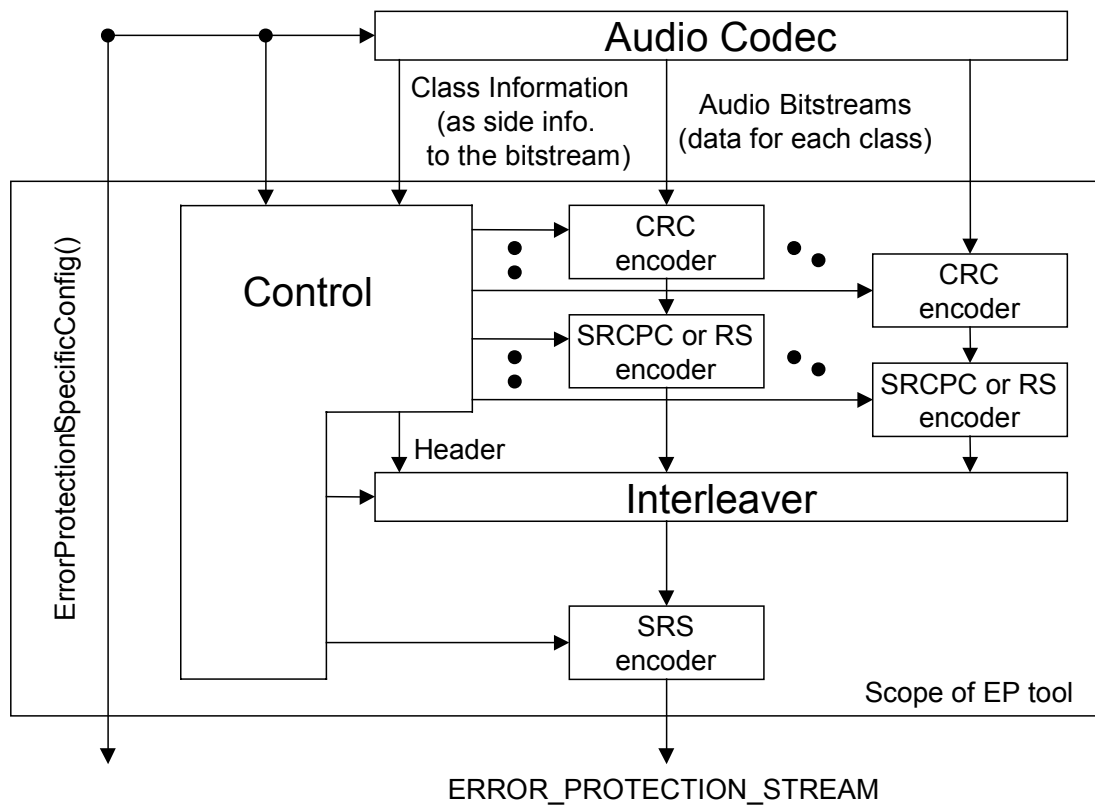


Figure 1.5 – Outline of EP encoder

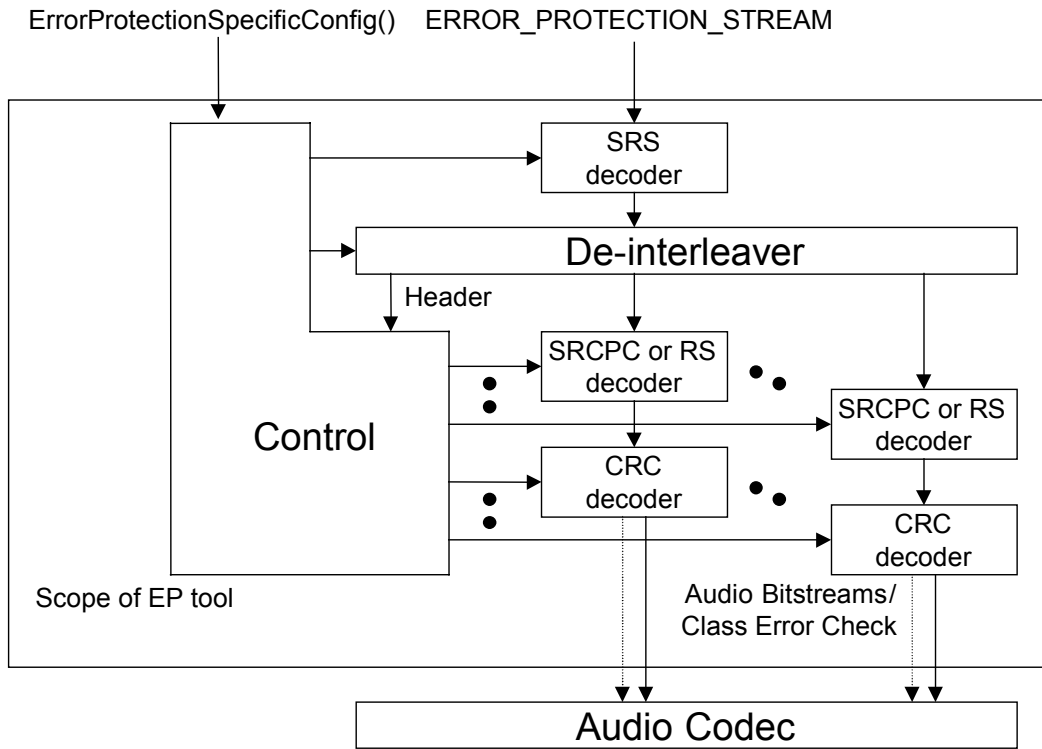


Figure 1.6 – Outline of EP decoder

## 1.8.2 Syntax

### 1.8.2.1 Error protection specific configuration

This part defines the syntax of the specific configuration for error protection.

Table 1.27 – Syntax of ErrorProtectionSpecificConfig()

Syntax	No. of bits	Mnemonic
ErrorProtectionSpecificConfig()		
{		
<b>number_of_predefined_set;</b>	8	<b>uimsbf</b>
<b>interleave_type;</b>	2	<b>uimsbf</b>
<b>bit_stuffing;</b>	3	<b>uimsbf</b>
<b>number_of_concatenated_frame;</b>	3	<b>uimsbf</b>
for ( i = 0; i < number_of_predefined_set; i++ ) {		
<b>number_of_class[i];</b>	6	<b>uimsbf</b>
for ( j = 0; j < number_of_class[i]; j++ ) {		
<b>length_escape[i][j];</b>	1	<b>uimsbf</b>
<b>rate_escape[i][j];</b>	1	<b>uimsbf</b>
<b>crclen_escape[i][j];</b>	1	<b>uimsbf</b>
if ( number_of_concatenated_frame != 1 ) {		
<b>concatenate_flag[i][j];</b>	1	<b>uimsbf</b>
}		
<b>fec_type[i][j];</b>	2	<b>uimsbf</b>
if( fec_type[i][j] == 0 ) {		
<b>termination_switch[i][j];</b>	1	<b>uimsbf</b>
}		
}		
}		
if (interleave_type == 2) {		

interleave_switch[i][j];	2	uimsbf
}		
class_optional;	1	uimsbf
if ( length_escape[i][j] == 1 ) { /* ESC */		
number_of_bits_for_length[i][j];	4	uimsbf
}		
else {		
class_length[i][j];	16	uimsbf
}		
if ( rate_escape[i][j] != 1 ) { /* not ESC */		
if(fec_type[i][j]){		
class_rate[i][j]	7	uimsbf
}else{		
class_rate[i][j]	5	uimsbf
}		
}		
if ( crclen_escape[i][j] != 1 ) { /* not ESC */		
class_crclen[i][j];	5	uimsbf
}		
}		
class_reordered_output;	1	uimsbf
if ( class_reordered_output == 1 ) {		
for ( j = 0; j < number_of_class[i]; j++ ) {		
class_output_order[i][j];	6	uimsbf
}		
}		
}		
header_protection;	1	uimsbf
if ( header_protection == 1 ) {		
header_rate;	5	uimsbf
header_crclen;	5	uimsbf
}		
rs_fec_capability;	7	uimsbf
}		

### 1.8.2.2 Error protection bitstream payloads

This part defines the syntax of the error protected audio bitstream payload. This kind of syntax can be selected by setting epConfig=2. It is common for all audio object types. If MPEG-4 Systems is used, one rs\_ep\_frame() is directly mapped to one access unit.

Table 1.28 – Syntax of rs\_ep\_frame ()

Syntax	No. of bits	Mnemonic
rs_ep frame()		
{		
ep_frame();		
rs_parity_bits;	Nrsparity	bslbf
}		

Nrsparity: see subclause 1.8.4.7

**Table 1.29 – Syntax of ep\_frame ()**

Syntax	No. of bits	Mnemonic
ep_frame() { if (interleave_type == 0){ ep_header(); ep_encoded_classes(); } if (interleave_type == 1){ <b>interleaved_frame_mode1;</b> } if (interleave_type == 2){ <b>interleaved_frame_mode2;</b> } <b>stuffing_bits;</b> }	1 -  1 -  Nstuff	<b>bslbf</b>  <b>bslbf</b>  <b>bslbf</b>

**Table 1.30 – Syntax of ep\_header ()**

Syntax	No. of bits	Mnemonic
ep_header() { <b>choice_of_pred;</b> <b>choice_of_pred_parity;</b> class_attrib(); <b>class_attrib_parity;</b> }	<b>N<sub>pred</sub></b> <b>N<sub>pred_parity</sub></b>  <b>N<sub>attrib_parity</sub></b>	<b>uimsbf</b> <b>bslbf</b>  <b>bslbf</b>

N<sub>pred</sub>: the smallest integer value greater than log<sub>2</sub> (# of pre-defined set).  
 N<sub>pred\_parity</sub>: See subclause 1.8.4.3  
 N<sub>attrib\_parity</sub>: See subclause 1.8.4.3

**Table 1.31 – Syntax of class\_attrib ()**

Syntax	No. of bits	Mnemonic
class_attrib(class_count, length_escape, rate_escape, crclen_escape, frame_pred) { for(j=0; j<class_count; j++){ if (length_escape[frame_pred][j] == 1){ <b>class_bit_count[j];</b> } if (rate_escape[frame_pred][j] == 1){ <b>class_code_rate[j];</b> } if (crclen_escape[frame_pred][j] == 1){ <b>class_crc_count[j];</b> } } if (bit_stuffing == 1){ <b>num_stuffing_bits;</b> } }	<b>Nbitcount</b>  <b>3</b>  <b>3</b>  <b>3</b>	<b>uimsbf</b>  <b>uimsbf</b>  <b>uimsbf</b>  <b>uimsbf</b>

class\_count: number of class for this frame  
 frame\_pred: selected predefined set for this frame



Table 1.32 – Syntax of ep\_encoded\_classes ()

Syntax	No. of bits	Mnemonic
<pre>ep_encoded_classes(class_count) {     for(j=0; j&lt;class_count; j++){         ep_encoded_class[j];     } }</pre>		<b>bslbf</b>

### 1.8.3 General information

#### 1.8.3.1 Definitions

<b>ErrorProtectionSpecificConfig ():</b>	Error protection specific configuration that is out-of-band information.
<b>number_of_predefined_set</b>	The number of pre-defined set.
<b>interleave_type</b>	This variable defines the interleave type. (interleave_type == 0) means no interleaving, (interleave_type == 1) means intra-frame interleaving and (interleave_type == 2) enables interleaving fine tuning for each class. For details see subclause 1.8.4.8. (interleave_type==3) is reserved.
<b>bit_stuffing</b>	Signals whether the bit stuffing to ensure the byte alignment is used with the in-band information or not: 1 indicates the bit stuffing is used. 0 indicates the bit stuffing is not used. This implies that the configuration provided with the out-of-band information ensure the EP-frame is byte-aligned.
<b>number_of_concatenated_frame</b>	The number of concatenated source coder frames for the constitution of one error protected frame.

Table 1.33 – concatenated frames depending on number\_of\_concatenated\_frame

Codeword	000	001	010	011	100	101	110	111
number of concatenated frame	reserved	1	2	3	4	5	6	7

<b>number_of_class[i]</b>	The number of classes for i-th pre-defined set.
<b>length_escape[i][j]</b>	If 0, the length of j-th class in i-th pre-defined set is fixed value. If 1, the length is variable. Note that in case “until the end”, this value should be 1, and the <b>number_of_bits_for_length[i][j]</b> value should be 0.
<b>rate_escape[i][j]</b>	If 0, the SRCPC code rate of j-th class in i-th pre-defined set is fixed value. If 1, the code rate is signaled in-band.
<b>crclen_escape[i][j]</b>	If 0, the CRC length of j-th class in i-th pre-defined set is fixed value. If 1, the CRC length is signaled in-band.
<b>concatenate_flag[i][j]</b>	This parameter defines whether j-th class of i-th pre-defined set is concatenated or not. 0 indicates “not concatenated” and 1 indicates “concatenated”. (See subclause 1.8.4.4)
<b>fec_type[i][j]</b>	This parameter defines whether SRCPC code (“0”) or RS code (“1” or “2”) are used to protect the j-th class of i-th pre-defined set. Note that the class length which is signaled to be protected by RS code shall be byte aligned, in either case that the length is signaled in the out-of-band information or that the length is signaled as in-band information. If this field is set to “2”, it indicates that this class is RS encoded in conjunction with next class as one RS code. Note that more than two succeeding classes have the value “2” for this field, it means these classes are concatenated and RS encoded as one RS code. If this field is “1”, it indicates that this class is not concatenated with next class. This means this class is the last class to be concatenated before RS encoding, or this class is RS encoded independently.
<b>termination_switch[i][j]</b>	This parameter defines whether j-th class of i-th pre-defined set is terminated or not when it is SRCPC encoded. See subclause 1.8.4.6.2.

<b>interleave_switch[i][j]</b>	This parameter defines how to interleave j-th class of i-th pre-defined set. 0 – not interleaved 1 – interleaved without intraclass-interleaving: the interleaving width is same as the number of bits within the current class if ( fec_type == 0 ), but same as the number of bytes within the current class if ( fec_type==1    fec_type == 2 ) 2 – interleaved with intraclass-interleaving: interleaving width is 28 if ( fec_type == 0 ); but this value is reserved if ( fec_type == 1    fec_type == 2 ) 3 – concatenated (see subclause 1.8.4.8.2.2)
<b>class_optional</b>	This flag signals, whether the class is mandatory (class_optional == 0) or optional (class_optional == 1). This flag can be used to reduce the redundancy within ErrorProtectionSpecificConfig. Usually it would be necessary to define 2 <sup>N</sup> predefinition sets, where N equals the number of optional classes.(See subclause 1.8.4.2)
<b>number_of_bits_for_length[i][j]</b>	This field exists only when the <b>length_escape[i][j]</b> is 1. This value shows the number of bits for the class length in-band signaling. This value should be set considering possible maximum length of the class. The value 0 indicates the “until the end” functionality (see subclause 1.8.4.1).
<b>class_length[i][j]</b>	This field exists only when the <b>length_escape[i][j]</b> is 0. This value shows the length of the j-th class in i-th pre-defined set, which is the fixed value while the transmission.
<b>class_rate[i][j]</b>	This field exists only when the <b>rate_escape[i][j]</b> is 0. In case <b>fec_type[i][j]</b> is 0, this value shows the SRCPC code rate of the j-th class in i-th pre-defined set, which is the fixed value while the transmission. The value from 0 to 24 corresponds to the code rate from 8/8 to 8/32, respectively. In case <b>fec_type[i][j]</b> is 1 or 2, this value shows the number of erroneous bytes which can be corrected by RS code (see subclause 1.8.4.7). All the classes which is signaled to be concatenated with <b>fec_type[i][j]</b> shall have the same value of <b>class_rate[i][j]</b> .
<b>class_crclen[i][j]</b>	This field exists only when the <b>crclen_escape[i][j]</b> is 0. This value shows the CRC length of the j-th class in i-th pre-defined set, which is the fixed value while the transmission. The value should be 0 – 18, which represents CRC length 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 16, 24 or 32. (See subclause 1.8.4.5)
<b>class_reordered_output</b>	If this value is “1”, the classes output from ep decoder is re-ordered. If “0”, no such processing is made. See subclause 1.8.4.9.
<b>class_output_order[i][j]</b>	This field only exists when class_reordered_output is set to “1”, to signal the order of the class after re-ordering. The j-th class of i-th pre-defined set is output as ( <b>class_output_order[i][j]</b> )-th class from ep decoder. See subclause 1.8.4.9.
<b>header_protection</b>	This value indicates the header error protection mode. 0 indicates the use of basic set of FEC, and 1 indicates the use of extended header error protection, as defined in subclause 1.8.4.3. The extended header error protection is applied only if the length of the header exceeds 16 bits.
<b>header_rate, header_crclen</b>	These values have the same semantics with <b>class_rate[i][j]</b> and <b>class_crclen[i][j]</b> respectively, while these error protection is utilized for the protection of header part.
<b>rs_fec_capability</b>	This field indicates the correction capability of SRS code to protect the whole frame (see subclause 1.8.4.7). This capability is given as number of erroneous bytes that can be corrected. The value “0” indicates SRS is not used.
<b>rs_ep_frame()</b>	Reed-Solomon error protected frame that is applied Reed-Solomon code.
<b>rs_parity_bits</b>	The Reed-Solomon parity bits for ep_frame(). See subclause 1.8.4.7.
<b>ep_frame()</b>	error protected frame.
<b>ep_header()</b>	EP frame header information.
<b>ep_encoded_classes()</b>	The EP encoded audio information.
<b>interleaved_frame_mode1</b>	The information bits after interleaving with interleaving mode 1. See subclause 1.8.4.1 and subclause 1.8.4.8.
<b>interleaved_frame_mode2</b>	The information bits after interleaving with interleaving mode 2. See subclause 1.8.4.1 and subclause 1.8.4.8.
<b>stuffing_bits</b>	The stuffing bits for the EP frame octet alignment. The number of bits Nstuff is signaled in class_attrib(), and should be in the range of 0...7.
<b>choice_of_pred</b>	The choice of pre-defined set. See subclause 1.8.4.2.
<b>choice_of_pred_parity</b>	The parity bits for <b>choice_of_pred</b> . See subclause 1.8.4.2.
<b>class_attrib_parity</b>	The parity bits for class_attrib(). See subclause 1.8.4.2.

<code>class_attrib()</code>	Attribution information for each class
<code>class_bit_count[j]</code>	The number of information bits included in the class. This field only exists in case the <code>length_escape</code> in out-of-band information is 1 (escape). The number of bits of this parameter <code>Nbitcount</code> is also signaled in the out-of-band information.
<code>class_code_rate[j]</code>	The coding rate for the audio data belonging to the class, as defined in the table below. This field only exists in case the <code>rate_escape</code> in out-of-band information is 1 (escape).

Table 1.34 – The coding rate for the audio data belonging to the class

Codeword	000	001	010	011	100	101	110	111
Puncture Rate	8/8	8/11	8/12	8/14	8/16	8/20	8/24	8/32
Puncture Pattern	FF, 00 00, 00	FF, A8 00, 00	FF, AA 00, 00	FF, EE 00, 00	FF, FF 00, 00	FF, FF AA, 00	FF, FF FF, 00	FF, FF FF, FF

<code>class_crc_count[j]</code>	The number of CRC bits for the audio data belonging to the class, as defined in the table below. This field only exists in case the <code>crclen_escape</code> in out-of-band information is 1 (escape).
---------------------------------	--

Table 1.35 – The number of CRC bits for the audio data belonging to the class

Codeword	000	001	010	011	100	101	110	111
CRC bits	0	6	8	10	12	14	16	32

<code>num_stuffing_bits</code>	the number of stuffing bits for the EP frame octet alignment. This field only exists in case the <b>bit_stuffing</b> in out-of-band information is 1.
<code>ep_encoded_class[j]</code>	CRC/SRCPC encoded audio data of j-th class. Note that if <code>class_bit_count[j] == 0</code> , audio data of j-th class is not encoded by CRC/SRCPC/SRS.

## 1.8.4 Tool description

### 1.8.4.1 Out-of-band information

The content of out-of band information is represented by means of `ErrorProtectionSpecificConfig()`. Some configuration examples are provided in Annex 1.B.

The length of the last class can be specified to last “until the end”. In MPEG-4 Systems, the systems layer guarantees the audio frame boundary by mapping one audio frame to one access unit. Therefore, the length of the “until the end” class can be calculated from the length of other classes and the total EP-encoded audio frame length.

The flag `class_optional` might be used to reduce the redundancy within `ErrorProtectionSpecificConfig()`. However, the EP tool still works with the same number of pre-defined sets. If there are N classes with (`class_optional == 1`), this pre-defined set is extended to  $2^N$  pre-defined sets. Unwrapping of the predefinition sets is described within the following subclause.

### 1.8.4.2 Derivation of pre-defined sets

This subclause describes the post processing, whose input is `ErrorProtectionSpecificConfig()` with “`class_optional`” switch and whose output are pre-defined sets used for the `ep_frame()` parameters.

General procedure:

- Each pre-defined set expands  $2^{NCO[i]}$  pre-defined sets, where  $NCO[i]$  is the number of classes with ( $class\_optional == 1$ ) in  $i$ -th original pre-defined set. Hereafter, any class with ( $class\_optional == 1$ ) is referred to as **optClass**.
- These expanded pre-defined sets start from “all the optClasses don’t exist” to “all the optClasses exists”.

Algorithm:

```

transPred = 0;
for ( i = 0; i < nPred; i++ ) {
    for ( j = 0; j < pow ( 2, NCO[i] ); j++ ) { /* for all predefinition sets */
        for ( k = 0; k < NCO[i]; k++ ) { /* unwrapping */
            if ( j & ( 0x01 << k ) ) { /* for all optional classes */
                optClassExists[k] = 1;
            }
            else {
                optClassExists[k] = 0;
            }
        }
        DefineTransPred(transPred, i, optClassExists);
        transPred ++;
    }
}

```

where,

**optClassExists[k]** signals whether  $k$ -th **optClass** of the pre-defined set exists (1) or not (0) in the defining new pre-defined set.

**DefineTransPred ( transPred, i, optClassExists)** defines  $transPred$ -th new pre-defined set used for the transmission. This new pre-defined set is a copy of  $i$ -th original pre-defined set, except it don't have **optClasses** whose **optClassExists** equals to 0.

**Example**

ErrorProtectionSpecificConfig() defines pre-defined sets as follows:

**Table 1.36 – The example of pre-defined set**

Pred #0		Pred #1	
Class A	class_optional = 1	Class E	class_optional = 1
Class B	class_optional = 0	Class F	class_optional = 0
Class C	class_optional = 1		
Class D	class_optional = 0		

After the pre-processing described above, the pre-defined sets used for ep\_frame() becomes as follows:

**Table 1.37 – The example of pre-defined sets after the pre-processing**

Pred #0	Pred #1	Pred #2	Pred #3	Pred #4	Pred #5
Class B	Class A	Class B	Class A	Class F	Class E
Class D	Class B	Class C	Class B		Class F
	Class D	Class D	Class C		
			Class D		

### 1.8.4.3 In-band information

The EP frame information, which is not included in the out-of-band information, is the in-band information. The parameters belonging to this information are transmitted as an EP frame header. The parameters are:

- The choice of pre-defined set
- The number of stuffing bits for byte alignment
- The class information which is not included in the out-of-band information

The EP decoder cannot decode the audio frame information without these parameters, and thus they have to be error protected stronger than or equal to the other parts. On this error protection, the choice of pre-defined set has to be treated differently from the other parts. This is because the length of the class information can be changed according to which pre-defined set is chosen. For this reason, this parameter is FEC encoded independently from the other parts. At decoder side, the choice of pre-defined set is decoded first, and then the length of the remaining header part is calculated with this information, and decodes that.

The FEC applied for these parts are as follows:

#### Basic set of FEC codes:

**Table 1.38 – Basic set of FEC codes for in-band information**

Number of bit to be protected	FEC code	total number of bits	Length of codeword
1-2	majority (repeat 3 times)	3-6	3
3-4	BCH(7,4)	6-7	6-7
5-7	BCH(15,7)	13-15	13-15
8-12	Golay(23,12)	19-23	19-23
13-16	BCH(31,16)	28-31	28-31
17-	RCPC 8/16 + 4-bit CRC	50 -	-

$N_{pred\_parity}$  (or  $N_{attrib\_parity}$ ) = total number of bits – Number of bit to be protected

Note that Number of bit to be protected is  $N_{pred}$  (or the total number of bits for  $class\_attrib()$ ).

#### Extended FEC:

If a header length exceeds 16 bits, this header is protected in the same way as the class information. The SRCPC code rate and the number of CRC bits are signaled. The encoding and decoding method for this is the same as described below within the CRC/SRCPC description.

The generation polynomials for each FEC is as follows:

$$\text{BCH}(7,4): x^3+x+1$$

$$\text{BCH}(15,7): x^8+x^7+x^6+x^4+1$$

$$\text{Golay}(23,12): x^{11}+x^9+x^7+x^6+x^5+x^3+x+1$$

$$\text{BCH}(31,16): x^{15}+x^{11}+x^{10}+x^9+x^8+x^7+x^5+x^3+x^2+x+1$$

With these polynomials, the FEC ( $n, k$ ) for  $l$ -bit information encoding is made as follows:

Calculate the polynomial  $R(x)$  that satisfies

$$M(x) x^{n-l} = Q(x)G(x) + R(x)$$

M(x): Information bits. Highest order corresponds to the first bit to be transmitted

G(x): The generation polynomial from the above definition

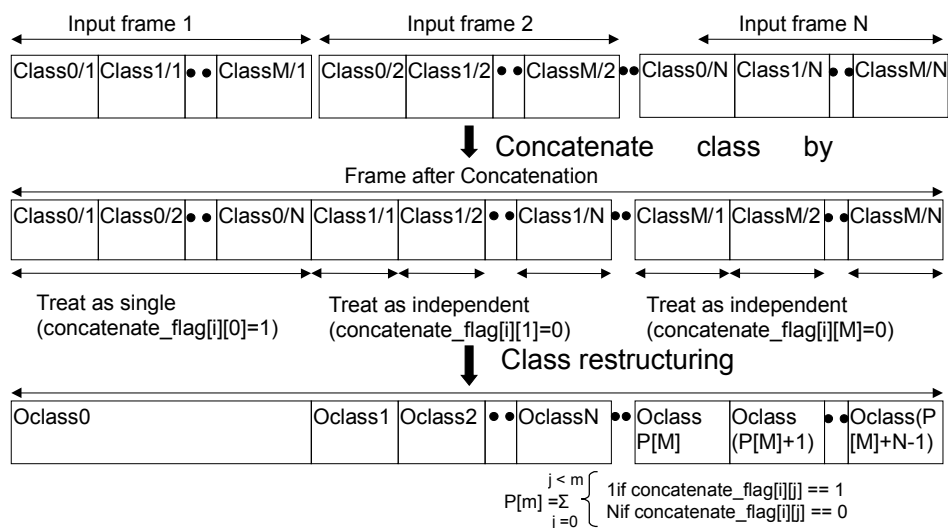
This polynomial R(x) represents parity to **choice\_of\_pred** or **class\_attrib()**, and set to **choice\_of\_pred\_parity** or **class\_attrib\_parity** respectively. The highest order corresponds to the first bit. The decoder can perform error correction using these parity bits, while it is optional operation.

**1.8.4.4 Concatenation functionality**

EP tool has a functionality to concatenate several source coder frames to build up a new frame for the EP tool. In this concatenation, the groups of bits belonging to the same class in the different source coder frames are concatenated in adjacent, class by class basis. The concatenated groups belonging to the same class is either treated as a single new one class or independent class in the same manner as before the concatenation.

The number of frames to be concatenated is signaled as **number\_of\_concatenated\_frame** in **ErrorProtectionSpecificConfig()**, and the choice whether the concatenated groups belonging to the same class is treated as single new one class or independent class is signaled by **concatenate\_flag[i][j]** (1 indicate “single new one class”, and 0 indicates “independent class”). This process is illustrated in Figure 1.7.

The same pre-defined set shall be used for all concatenated frames. No escape mechanism shall be used for any class parameter.



\*Oclass: Output class to be proceed for CRC/FEC

**Figure 1.7 – Concatenation procedure**

**1.8.4.5 CRC**

The CRC provides error detection capability. The information bits of each class is CRC encoded as a first process. In this tool, the following set of the CRC is defined:

- 1-bit CRC **CRC1**:  $x+1$
- 2-bit CRC **CRC2**:  $x^2+x+1$
- 3-bit CRC **CRC3**:  $x^3+x+1$
- 4-bit CRC **CRC4**:  $x^4+x^3+x^2+1$
- 5-bit CRC **CRC5**:  $x^5+x^4+x^2+x+1$

6-bit CRC **CRC6** :  $x^6+x^5+x^3+x^2+x+1$

7-bit CRC **CRC7** :  $x^7+x^6+x^2+1$

8-bit CRC **CRC8** :  $x^8+x^2+x+1$

9-bit CRC **CRC9** :  $x^9+x^8+x^5+x^2+x+1$

10-bit CRC **CRC10** :  $x^{10}+x^9+x^5+x^4+x+1$

11-bit CRC **CRC11** :  $x^{11}+x^{10}+x^4+x^3+x+1$

12-bit CRC **CRC12** :  $x^{12}+x^{11}+x^3+x^2+x+1$

13-bit CRC **CRC13** :  $x^{13}+x^{12}+x^7+x^6+x^5+x^4+x^2+1$

14-bit CRC **CRC14** :  $x^{14}+x^{13}+x^5+x^3+x^2+1$

15-bit CRC **CRC15** :  $x^{15}+x^{14}+x^{11}+x^{10}+x^7+x^6+x^2+1$

16-bit CRC **CRC16** :  $x^{16}+x^{12}+x^5+1$

24-bit CRC **CRC24** :  $x^{24}+x^{23}+x^6+x^5+x+1$

32-bit CRC **CRC32** :  $x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^8+x^7+x^5+x^4+x^2+x+1$

With these polynomials, the CRC encoding is made as follows:

Calculate the polynomial  $R(x)$  that satisfies

$$M(x)x^k = Q(x)G(x) + R(x)$$

$M(x)$ : Information bits. Highest order corresponds to the first bit to be transmitted

$G(x)$ : The generation polynomial from the above definition

$k$ : The number of CRC bits.

With this polynomial  $R(x)$ , the CRC encoded bits  $W(x)$  is represented as:

$$W(x) = M(x)x^k + R(x)$$

Note that the value  $k$  should be chosen so that the number of CRC encoded bits does not exceed  $2^{k-1}$ .

Using these CRC bits, the decoder should perform error detection. When an error is detected through CRC, error concealment may be applied to reduce the quality degradation caused by the error. The error concealment method depends on MPEG-4 audio algorithms. See the informal annex (example of error concealment).

#### 1.8.4.6 Systematic rate-compatible punctured convolutional (SRCPC) codes

Following to the CRC encoding, FEC encoding is made with the SRCPC codes. This subclause describes the SRCPC encoding process.

The channel encoder is based on a systematic recursive convolutional (SRC) encoder with rate  $R=1/4$ . The CRC encoded classes are concatenated and input into this encoder. Then, with the puncturing procedure described in the subclause later, we obtain a Rate Compatible Punctured Convolutional (RCPC) code whose code rate varies for each class according to the error sensitivity.

1.8.4.6.1 SRC code generation

The SRC code is generated from a rational generator matrix by using a feedback loop. A shift register realization of the encoder is shown in Figure 1.8.

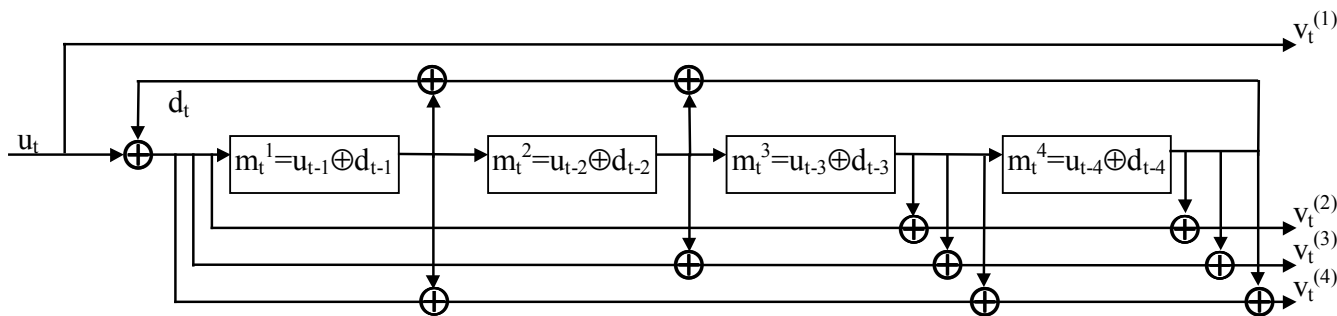


Figure 1.8 – Shift register realization for systematic recursive convolutional encoder

To obtain the output vectors  $v_t$  at each time instant  $t$ , one has to know the content of the shift registers  $m_t^1, m_t^2, m_t^3, m_t^4$  (corresponds to the state) and the input bit  $u_t$  at time  $t$ .

We obtain the output  $v_t^{(2)}, v_t^{(3)}$  and  $v_t^{(4)}$

$$v_t^{(2)} = m_t^4 \oplus m_t^3 \oplus (u_t \oplus d_t)$$

$$v_t^{(3)} = m_t^4 \oplus m_t^3 \oplus m_t^2 \oplus (u_t \oplus d_t)$$

$$v_t^{(4)} = m_t^4 \oplus m_t^3 \oplus m_t^1 \oplus (u_t \oplus d_t)$$

with

$$d_t = m_t^4 \oplus m_t^2 \oplus m_t^1, m_t^4 = u_{t-4} \oplus d_{t-4}, m_t^3 = u_{t-3} \oplus d_{t-3}, m_t^2 = u_{t-2} \oplus d_{t-2}, m_t^1 = u_{t-1} \oplus d_{t-1}$$

Finally we obtain for the output vector  $\underline{v}_t = (v_t^{(1)}, v_t^{(2)}, v_t^{(3)}, v_t^{(4)})$  at time  $t$  depending on the input bit  $u_t$  and the current state  $\underline{m}_t = (m_t^1, m_t^2, m_t^3, m_t^4)$ :

$\begin{aligned} V_t^{(1)} &= u_t \\ V_t^{(2)} &= m_t^4 \oplus m_t^3 \oplus (u_t \oplus d_t) = m_t^3 \oplus m_t^2 \oplus m_t^1 \oplus u_t \\ V_t^{(3)} &= m_t^4 \oplus m_t^3 \oplus m_t^2 \oplus (u_t \oplus d_t) = m_t^3 \oplus m_t^1 \oplus u_t \\ V_t^{(4)} &= m_t^4 \oplus m_t^3 \oplus m_t^1 \oplus (u_t \oplus d_t) = m_t^3 \oplus m_t^2 \oplus u_t \end{aligned}$
--

with  $\underline{m}_1 = (m_1^1, m_1^2, m_1^3, m_1^4) = (0, 0, 0, 0) = \underline{0}$

The initial state is always  $\underline{0}$ , i.e. each memory cell contains a 0 before the input of the first information bit  $u_t$ .

1.8.4.6.2 Termination of SRC code

In case the SRC coded class is indicated as terminated with `termination_switch[i]` in `ErrorProtectionSpecificConfig()`, or SRC code is used for the protection of in-band information, the SRC encoder shall add the tail bits at the end of this class, and start the succeeding SRC encoding with initial state (all the encoder shift register shall reset to be 0).

The tail bits following the information sequence  $\underline{u}$  for returning to state  $\underline{m}_n = \underline{0}$  (termination) depends on the last state  $\underline{m}_{n-3}$  (state after the input of the last information bit  $u_{n-4}$ ). The termination sequence for each state described by  $\underline{m}_{n-3}$  is given in Table 1.39. The receiver may use these tail bits (TB) for additional error detection.



The appendix  $(u_{n-3}, u_{n-2}, u_{n-1}, u_n)$  to the information sequence can be calculated with the following condition:

$$\text{for all } t \text{ with } n-3 \leq t \leq n: u_t \oplus d_t = 0$$

Hence we obtain for the tail bit vector  $\underline{u}' = (u_{n-3}, u_{n-2}, u_{n-1}, u_n)$  depending on the state  $\underline{m}_{n-3} = (m_{n-3}^1, m_{n-3}^2, m_{n-3}^3, m_{n-3}^4)$

$u_{n-3} = d_{n-3} = m_{n-3}^4 \oplus m_{n-3}^2 \oplus m_{n-3}^1$
$u_{n-2} = d_{n-2} = m_{n-2}^4 \oplus m_{n-2}^2 \oplus m_{n-2}^1 = m_{n-3}^3 \oplus m_{n-3}^1 \oplus 0 = m_{n-3}^3 \oplus m_{n-3}^1$
$u_{n-1} = d_{n-1} = m_{n-1}^4 \oplus m_{n-1}^3 \oplus m_{n-1}^2 = m_{n-3}^2 \oplus 0 \oplus 0 = m_{n-3}^2$
$u_n = d_n = m_{n-3}^1 \oplus 0 \oplus 0 = m_{n-3}^1$

**Table 1.39 – Tail bits for systematic recursive convolutional code**

state $\underline{m}_{n-3}$	$m_{n-3}^4$	$m_{n-3}^3$	$m_{n-3}^2$	$m_{n-3}^1$	$u_{n-3}$	$u_{n-2}$	$u_{n-1}$	$u_n$
0	0	0	0	0	0	0	0	0
1	0	0	0	1	1	1	0	1
2	0	0	1	0	1	0	1	0
3	0	0	1	1	0	1	1	1
4	0	1	0	0	0	1	0	0
5	0	1	0	1	1	0	0	1
6	0	1	1	0	1	1	1	0
7	0	1	1	1	0	0	1	1
8	1	0	0	0	1	0	0	0
9	1	0	0	1	0	1	0	1
10	1	0	1	0	0	0	1	0
11	1	0	1	1	1	1	1	1
12	1	1	0	0	1	1	0	0
13	1	1	0	1	0	0	0	1
14	1	1	1	0	0	1	1	0
15	1	1	1	1	1	0	1	1

**1.8.4.6.3 Puncturing of SRC for SRCPC code**

Puncturing of the output of the SRC encoder allows different rates for transmission. The puncturing tables are listed in Table 1.40.

**Table 1.40 – Puncturing tables (all values in hexadecimal representation)**

Rate r	8/8	8/9	8/10	8/11	8/12	8/13	8/14	8/15	8/16	8/17	8/18	8/19	8/20
$P_r(0)$	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
$P_r(1)$	00	80	88	A8	AA	EA	EE	FE	FF	FF	FF	FF	FF
$P_r(2)$	00	00	00	00	00	00	00	00	00	80	88	A8	AA
$P_r(3)$	00	00	00	00	00	00	00	00	00	00	00	00	00

Rate r	8/21	8/22	8/23	8/24	8/25	8/26	8/27	8/28	8/29	8/30	8/31	8/32
$P_r(0)$	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
$P_r(1)$	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
$P_r(2)$	EA	EE	FE	FF	FF	FF	FF	FF	FF	FF	FF	FF
$P_r(3)$	00	00	00	00	80	88	A8	AA	EA	EE	FE	FF

The puncturing is made with the period of 8, and each bit of Pr(i) indicates the corresponding vt(i) from the SRC encoder is punctured (not transmitted) or not (transmitted). Each bit of Pr(i) is used from MSB to LSB, and 0/1 indicates not-punctured/punctured respectively. The code rate is a property of the class, thus the choice of the table is made according which class the current bit belongs to. After this decision which bits from vt(i) is transmitted, they are output in the order from vt(0) to vt(3).

**1.8.4.6.4 Decoding process of SRCPC code**

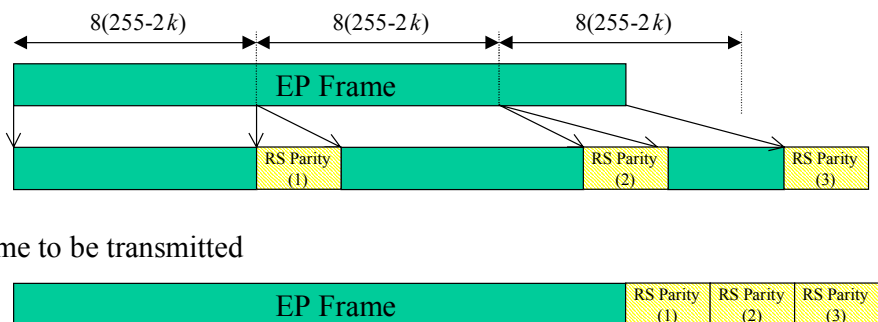
At the decoder, the error correction should be performed using this SRCPC code, while it is the optional operation and the decoder may extract the original information by just ignoring parity bits.

Decoding of SRCPC can be achieved using Viterbi algorithm for the punctured convolutional coding.

**1.8.4.7 Shortened Reed-Solomon codes**

Shortened RS codes RS(255-l, 255-2k-l) defined over GF(2<sup>8</sup>) is used to protect EP encoded frame or to protect each class. Here, k is the number of correctable errors in one RS codeword. l is for the shortening.

First, the EP encoded frame is divided into N parts, so that its length is less than or equal to (255-2k) octets. This division is made from the beginning of the frame so that the length of the sub-frame becomes (255-2k) octets, except the last part. Then for each of N sub-frames, the parity digits are calculated. For the transmission, these N parity digits are appended at the end of the EP frame. This process is illustrated in Figure 1.9.



**Figure 1.9 – RS encoding of EP frame**

In case RS code is used for EP for whole frame, the correction capability of SRS code t is transmitted within the out-of-band information as **rs\_fec\_capability**. This value can be selected as an arbitrary integer value satisfying 0 ≤ 2k ≤ 254. The SRS code defined in the Galois Field GS(2<sup>8</sup>) is generated from a generator polynomial g(x) = (x - α)(x - α<sup>2</sup>)... (x - α<sup>2k</sup>), where α denotes a root of the primitive polynomial m(x) = x<sup>8</sup> + x<sup>4</sup> + x<sup>3</sup> + x<sup>2</sup> + 1. The binary representative of α<sup>i</sup> is shown in the Table 1.41 below, where the MSB of the octet is transmitted first.

**Table 1.41 – Binary representation for α<sup>i</sup> (0 ≤ i ≤ 254) over GF(2<sup>8</sup>)**

a <sup>i</sup>	binary rep.	a <sup>i</sup>	binary rep.	a <sup>i</sup>	binary rep.	a <sup>i</sup>	binary rep.
0	00000000	a <sup>63</sup>	10100001	a <sup>127</sup>	11001100	a <sup>191</sup>	01000001
a <sup>0</sup>	00000001	a <sup>64</sup>	01011111	a <sup>128</sup>	10000101	a <sup>192</sup>	10000010
a <sup>1</sup>	00000010	a <sup>65</sup>	10111110	a <sup>129</sup>	00010111	a <sup>193</sup>	00011001
a <sup>2</sup>	00000100	a <sup>66</sup>	01100001	a <sup>130</sup>	00101110	a <sup>194</sup>	00110010
a <sup>3</sup>	00001000	a <sup>67</sup>	11000010	a <sup>131</sup>	01011100	a <sup>195</sup>	01100100
a <sup>4</sup>	00010000	a <sup>68</sup>	10011001	a <sup>132</sup>	10111000	a <sup>196</sup>	11001000
a <sup>5</sup>	00100000	a <sup>69</sup>	00101111	a <sup>133</sup>	01101101	a <sup>197</sup>	10001101
a <sup>6</sup>	01000000	a <sup>70</sup>	01011110	a <sup>134</sup>	11011010	a <sup>198</sup>	00000111
a <sup>7</sup>	10000000	a <sup>71</sup>	10111100	a <sup>135</sup>	10101001	a <sup>199</sup>	00001110
a <sup>8</sup>	00011101	a <sup>72</sup>	01100101	a <sup>136</sup>	01001111	a <sup>200</sup>	00011100
a <sup>9</sup>	00111010	a <sup>73</sup>	11001010	a <sup>137</sup>	10011110	a <sup>201</sup>	00111000

a <sup>10</sup>	01110100	a <sup>74</sup>	10001001	a <sup>138</sup>	00100001	a <sup>202</sup>	01110000
a <sup>11</sup>	11101000	a <sup>75</sup>	00001111	a <sup>139</sup>	01000010	a <sup>203</sup>	11100000
a <sup>12</sup>	11001101	a <sup>76</sup>	00011110	a <sup>140</sup>	10000100	a <sup>204</sup>	11011101
a <sup>13</sup>	10000111	a <sup>77</sup>	00111100	a <sup>141</sup>	00010101	a <sup>205</sup>	10100111
a <sup>14</sup>	00010011	a <sup>78</sup>	01111000	a <sup>142</sup>	00101010	a <sup>206</sup>	01010011
a <sup>15</sup>	00100110	a <sup>79</sup>	11110000	a <sup>143</sup>	01010100	a <sup>207</sup>	10100110
a <sup>16</sup>	01001100	a <sup>80</sup>	11111101	a <sup>144</sup>	10101000	a <sup>208</sup>	01010001
a <sup>17</sup>	10011000	a <sup>81</sup>	11100111	a <sup>145</sup>	01001101	a <sup>209</sup>	10100010
a <sup>18</sup>	00101101	a <sup>82</sup>	11010011	a <sup>146</sup>	10011010	a <sup>210</sup>	01011001
a <sup>19</sup>	01011010	a <sup>83</sup>	10111011	a <sup>147</sup>	00101001	a <sup>211</sup>	10110010
a <sup>20</sup>	10110100	a <sup>84</sup>	01101011	a <sup>148</sup>	01010010	a <sup>212</sup>	01111001
a <sup>21</sup>	01110101	a <sup>85</sup>	11010110	a <sup>149</sup>	10100100	a <sup>213</sup>	11110010
a <sup>22</sup>	11101010	a <sup>86</sup>	10110001	a <sup>150</sup>	01010101	a <sup>214</sup>	11111001
a <sup>23</sup>	11001001	a <sup>87</sup>	01111111	a <sup>151</sup>	10101010	a <sup>215</sup>	11101111
a <sup>24</sup>	10001111	a <sup>88</sup>	11111110	a <sup>152</sup>	01001001	a <sup>216</sup>	11000011
a <sup>25</sup>	00000011	a <sup>89</sup>	11100001	a <sup>153</sup>	10010010	a <sup>217</sup>	10011011
a <sup>26</sup>	00000110	a <sup>90</sup>	11011111	a <sup>154</sup>	00111001	a <sup>218</sup>	00101011
a <sup>27</sup>	00001100	a <sup>91</sup>	10100011	a <sup>155</sup>	01110010	a <sup>219</sup>	01010110
a <sup>28</sup>	00011000	a <sup>92</sup>	01011011	a <sup>156</sup>	11100100	a <sup>220</sup>	10101100
a <sup>29</sup>	00110000	a <sup>93</sup>	10110110	a <sup>157</sup>	11010101	a <sup>221</sup>	01000101
a <sup>30</sup>	01100000	a <sup>94</sup>	01110001	a <sup>158</sup>	10110111	a <sup>222</sup>	10001010
a <sup>31</sup>	11000000	a <sup>95</sup>	11100010	a <sup>159</sup>	01110011	a <sup>223</sup>	00001001
a <sup>32</sup>	10011101	a <sup>96</sup>	11011001	a <sup>160</sup>	11100110	a <sup>224</sup>	00010010
a <sup>33</sup>	00100111	a <sup>97</sup>	10101111	a <sup>161</sup>	11010001	a <sup>225</sup>	00100100
a <sup>34</sup>	01001110	a <sup>98</sup>	01000011	a <sup>162</sup>	10111111	a <sup>226</sup>	01001000
a <sup>35</sup>	10011100	a <sup>99</sup>	10000110	a <sup>163</sup>	01100011	a <sup>227</sup>	10010000
a <sup>36</sup>	00100101	a <sup>100</sup>	00010001	a <sup>164</sup>	11000110	a <sup>228</sup>	00111101
a <sup>37</sup>	01001010	a <sup>101</sup>	00100010	a <sup>165</sup>	10010001	a <sup>229</sup>	01111010
a <sup>38</sup>	10010100	a <sup>102</sup>	01000100	a <sup>166</sup>	00111111	a <sup>230</sup>	11110100
a <sup>39</sup>	00110101	a <sup>103</sup>	10001000	a <sup>167</sup>	01111110	a <sup>231</sup>	11110101
a <sup>40</sup>	01101010	a <sup>104</sup>	00001101	a <sup>168</sup>	11111100	a <sup>232</sup>	11110111
a <sup>41</sup>	11010100	a <sup>105</sup>	00011010	a <sup>169</sup>	11100101	a <sup>233</sup>	11110011
a <sup>42</sup>	10110101	a <sup>106</sup>	00110100	a <sup>170</sup>	11010111	a <sup>234</sup>	11111011
a <sup>43</sup>	01110111	a <sup>107</sup>	01101000	a <sup>171</sup>	10110011	a <sup>235</sup>	11101011
a <sup>44</sup>	11101110	a <sup>108</sup>	11010000	a <sup>172</sup>	01111011	a <sup>236</sup>	11001011
a <sup>45</sup>	11000001	a <sup>109</sup>	10111101	a <sup>173</sup>	11110110	a <sup>237</sup>	10001011
a <sup>46</sup>	10011111	a <sup>110</sup>	01100111	a <sup>174</sup>	11110001	a <sup>238</sup>	00001011
a <sup>47</sup>	00100011	a <sup>111</sup>	11001110	a <sup>175</sup>	11111111	a <sup>239</sup>	00010110
a <sup>48</sup>	01000110	a <sup>112</sup>	10000001	a <sup>176</sup>	11100011	a <sup>240</sup>	00101100
a <sup>49</sup>	10001100	a <sup>113</sup>	00011111	a <sup>177</sup>	11011011	a <sup>241</sup>	01011000
a <sup>50</sup>	00000101	a <sup>114</sup>	00111110	a <sup>178</sup>	10101011	a <sup>242</sup>	10110000
a <sup>51</sup>	00001010	a <sup>115</sup>	01111100	a <sup>179</sup>	01001011	a <sup>243</sup>	01111101
a <sup>52</sup>	00010100	a <sup>116</sup>	11111000	a <sup>180</sup>	10010110	a <sup>244</sup>	11111010
a <sup>53</sup>	00101000	a <sup>117</sup>	11101101	a <sup>181</sup>	00110001	a <sup>245</sup>	11101001
a <sup>54</sup>	01010000	a <sup>118</sup>	11000111	a <sup>182</sup>	01100010	a <sup>246</sup>	11001111
a <sup>55</sup>	10100000	a <sup>119</sup>	10010011	a <sup>183</sup>	11000100	a <sup>247</sup>	10000011
a <sup>56</sup>	01011101	a <sup>120</sup>	00111011	a <sup>184</sup>	10010101	a <sup>248</sup>	00011011
a <sup>57</sup>	10111010	a <sup>121</sup>	01110110	a <sup>185</sup>	00110111	a <sup>249</sup>	00110110
a <sup>58</sup>	01101001	a <sup>122</sup>	11101100	a <sup>186</sup>	01101110	a <sup>250</sup>	01101100
a <sup>59</sup>	11010010	a <sup>123</sup>	11000101	a <sup>187</sup>	11011100	a <sup>251</sup>	11011000
a <sup>60</sup>	10111001	a <sup>124</sup>	10010111	a <sup>188</sup>	10100101	a <sup>252</sup>	10101101
a <sup>61</sup>	01101111	a <sup>125</sup>	00110011	a <sup>189</sup>	01010111	a <sup>253</sup>	01000111
a <sup>62</sup>	11011110	a <sup>126</sup>	01100110	a <sup>190</sup>	10101110	a <sup>254</sup>	10001110

In case RS is used for UEP for each class, this is indicated by **fec\_type** in ErrorProtectionSpecificConfig(). The SRS code shall be applied for each class, as the same manner with SRCPC encoding. One limitation for SRS code is that it can only be applied for the class whose length is known with ErrorProtectionSpecificConfig() or ep\_header(), i. e. SRS code cannot be applied to the class whose length is defined as "Until the End".

If the target information bits for SRS code are not byte-aligned, bits with value '0' shall be added before SRS encoding, and deleted before transmission. At the decoder side, if SRS decoding is performed, same number of '0's should be added before SRS decoding procedure, and deleted again after SRS decoding.

The decoder should perform error correction using these parity bytes, while this is an optional operation and the decoder may ignore these parity bytes added.

Before the SRS encoding, the EP frame is divided into sub-frames so that the length is less than or equal to  $255-2k$ . The length of sub-frames are calculated with as follows:

$L$ : The length of EP frame in octet

$N$ : The number of sub-frames

$l_i$ : The length of  $i$ -th sub-frame

$N = \text{minimum integer small than } (L / (255-2k))$

$l_i = 255-2k$ , for  $i < N$

$L \bmod (255-2k)$ , for  $i = N$

For each of these sub-frames, the SRS parity digits with length of  $2k$  octets are calculated using  $g(x)$  as follows:

$u(x)$ : polynomial representative of a sub-frame. Lowest order corresponds to the first octet.

$p(x)$ : polynomial representative of the parity digits. Lowest order corresponds to the first octet.

$$p(x) = x^{2k} \cdot u(x) \bmod g(x)$$

The number of bits for rs\_parity\_bit is as follows:

$$N_{rsparity} = 8N$$

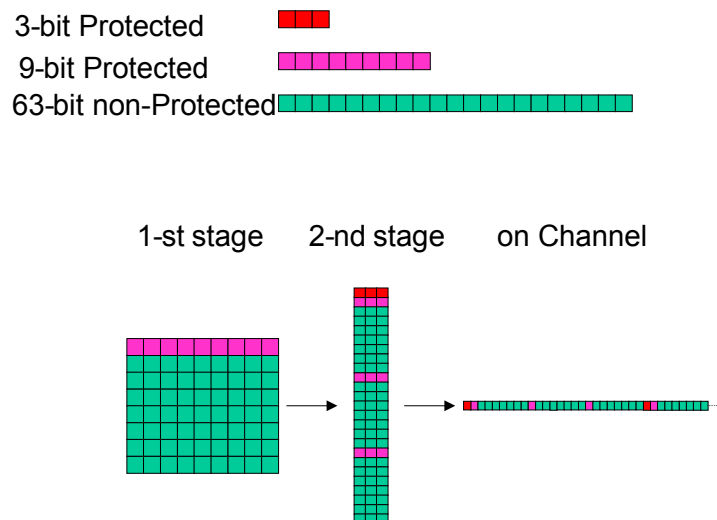
#### 1.8.4.8 Recursive interleaving

The interleaving is applied in multi-stage manner. Figure 1.10 shows the interleaving method.



**Figure 1.10 – One stage of interleaving**

In the multistage interleaving, the output of this one stage of interleaving is treated as a non-protected part in the next stage. Figure 1.11 shows the example of 2 stage interleaving.



**Figure 1.11 – Example of multi-stage interleaving**

By choosing the width  $W$  of the interleave-matrix to be the same as the FEC code length (or the value 28 in case of SRCPC codes), the interleaving size can be optimized for all the FEC codes.

In actual case, the total number of bits for the interleaving may not allow to use such rectangular. In such case, the matrix as shown in Figure 1.12 is used.

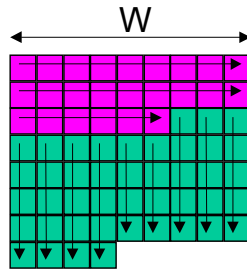


Figure 1.12 – Interleave matrix in non-rectangular case

1.8.4.8.1 Definition of recursive interleaver

Two information streams are input to this interleaver,  $X_i$  and  $Y_j$ .

$$X_i, 0 \leq i < l_x$$

$$Y_j, 0 \leq j < l_y$$

where  $l_x$  and  $l_y$  is the number of bits for each input streams  $X_i$  and  $Y_j$ , respectively.  $X_i$  is set to the interleaving matrix from the top left to the bottom right, into the horizontal direction. Then  $Y_j$  is set into the rest place in vertical direction.

With the width of interleaver  $W$ , the size of interleaving matrix is shown as Figure 1.13. Where,

$$D = (l_x + l_y) / W$$

$$d = l_x + l_y - D * W$$

Where ‘/’ indicates division by truncation.

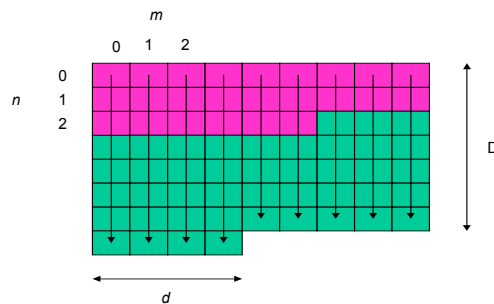


Figure 1.13 – The size of interleaving matrix

The output bitstream  $Z_k$  ( $0 < k \leq l_x+l_y$ ) is read from this matrix from top left to bottom right, column by column in horizontal direction. Thus the bit placed  $m$ -th column,  $n$ -th row ( $m$  and  $n$  starts from 0) corresponds to  $Z_k$  where:

$$k = m * D + \min(m, d) + n$$

In the matrix,  $X_i$  is set to

$$m = i \text{ mod } W, \quad n = i / W,$$

Thus  $Z_k$  which is set by the  $X_i$  becomes:

$$Z_k = X_i, \text{ where } k = (i \bmod W) * D + \min(i \bmod W, d) + i / W$$

The bits which are set with  $X_i$  in the interleaving matrix are shown as Figure 1.14 where:

$$D' = I_x / W$$

$$d' = I_x - D' * W$$

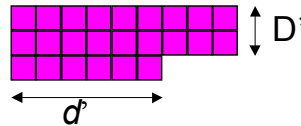


Figure 1.14 – The bits which are set with  $X_i$  in the interleaving matrix

Thus, in the  $m$ -th row,  $Y_j$  is set from the  $n$ -th row where  $n = D' + (m < d' ? 1 : 0)$  to the bottom. Thus  $Z_k$  set by  $Y_j$  is represented as follows:

```

Set j to 0;
for m = 0 to D-1 {
  for k = m * D + min(m, d) + D' + (m < d' ? 1 : 0) to (m+1) * D + min(m+1, d) - 1 {
     $Z_k = Y_j$ ;
    j ++;
  }
}

```

#### 1.8.4.8.2 Modes of interleaving

Two modes of interleaving, mode 1 and mode 2 are defined in the following subclauses.

##### 1.8.4.8.2.1 Interleaving operation in mode 1

Multi-stage interleaving is processed for **ep\_encoded\_class** from the last class to first class, and then class attribution part of `ep_header()` (which is `class_attrib()` + **class\_attrib\_parity**), and the pre-defined part of `ep_header()` (which is **choice\_of\_pred** + **choice\_of\_pred\_parity**), as illustrated in Figure 1.15.

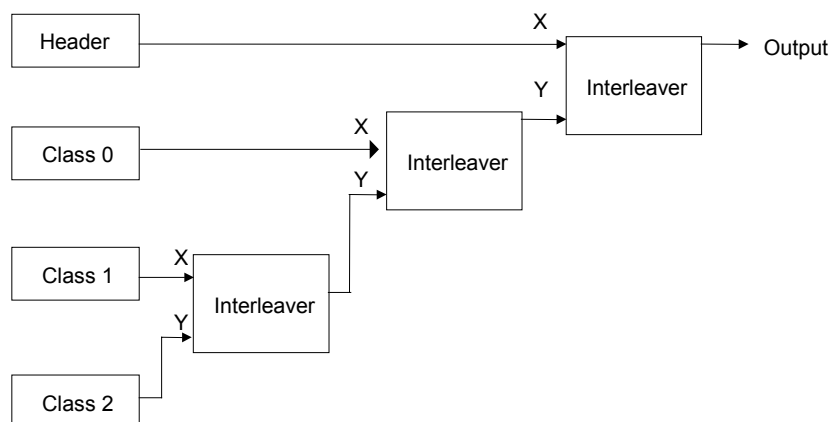


Figure 1.15 – Interleaving process of mode1 specification

1.8.4.8.2.2 Interleaving operation in mode 2

In mode 2, a flag indicates whether the class is processed with interleaver, and how it is interleaved. This flag `interleave_switch` is signaled within the out-of-band information. The value 0 indicates the class is not processed by the interleaver. The value 1 indicates the class is interleaved by the recursive interleaver, and the length of the class is used as the width of the interleaver (either the length in bits in case of SRCPC or the length in bytes in case of SRS). The value 2 indicates the class is interleaved by the recursive interleaver, and the width is set to be equal to 28 (permitted only in case of SRCPC). The value 3 indicates the class is concatenated but not interleaved by the recursive interleaver. The interleaving operation for the `ep_header` is same as mode 1.

Figure 1.16 shows the interleaving scheme principle for `fec_type == 1` or `2` (SRS) and `interleave_switch == 1`. The width is set to be the number of bytes in the class. The bits in the class are written into the interleaving matrix byte by byte for each column.

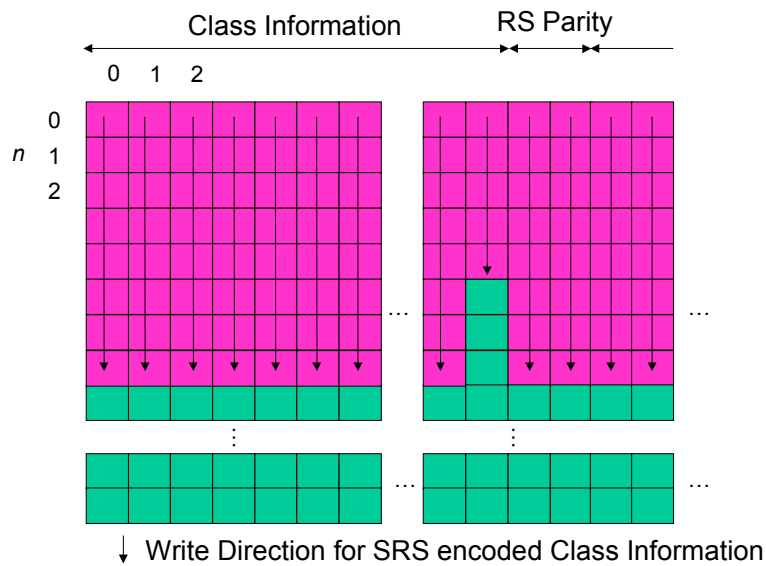


Figure 1.16 – Interleaving matrix in RS encoded class case

The interleaving process to obtain `interleaved_frame_mode2` is as follows (N: number of classes):

```

clear buffer BUF_NO /* Buffer for non-interleaved part. */
clear buffer BUF_Y /* Buffer for Y input in the next stage */
for j = 0 to N-1 {
    if ( interleave_switch[i][j] == 3 ) {
        concatenate ep_encoded_class[j] at hte end of BUF_NO;
    }
}
set BUF_NO into BUF_Y;
clear buffer BUF_NO
for j = N-1 to 0 {
    if ( interleave_switch[i][j] == 0 ) {
        concatenate ep_encoded_class[j] at the end of BUF_NO;
    } else if ( interleave_switch[i][j] != 3 ) {
        if ( interleave_switch[i][j] == 1 ) {
            set the size of the interleave window to be the length of ep_encoded_class[j];
        } else if ( interleave_switch[i][j] == 2 ) {
            set the size of the interleave window to be 28;
        }
        input ep_encoded_class[j] into the recursive interleaver as X input;
        input BUF_Y into the recursive interleaver as Y input;
        set the output of the interleaver into BUF_Y;
    }
}
}
    
```

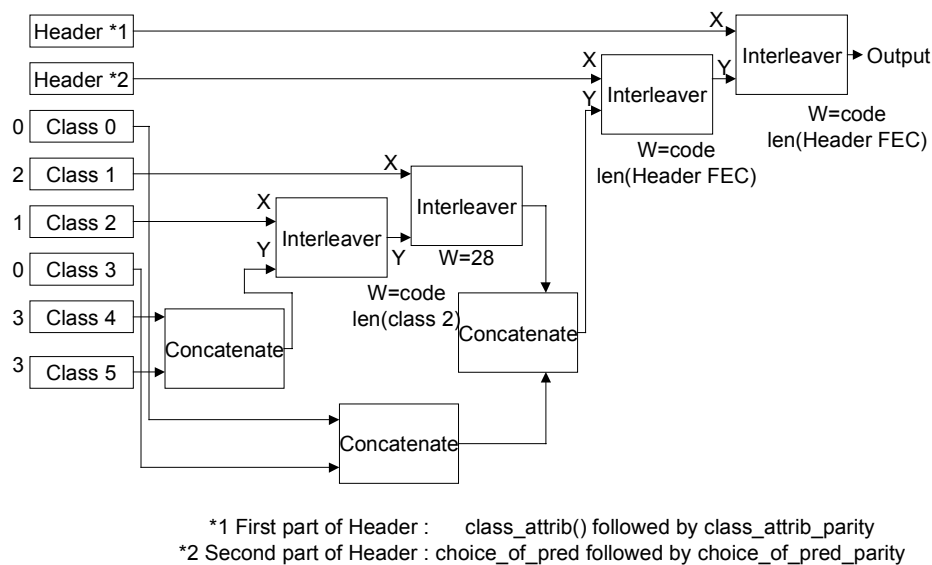


```

concatenate BUF_NO at the end of BUF_Y;
input class_attrib() followed by class_attrib_parity into the recursive interleaver as X
input;
input BUF_Y into the recursive interleaver as Y input;
set the output of the interleaver into BUF_Y;
input choice_of_pred followed by choice_of_pred_parity into the recursive interleaver as X
input;
input BUF_Y into the recursive interleaver as Y input;
set the output of the interleaver into BUF_Y;
set BUF_Y into interleaved_frame_mode2;

```

Interleave Switch ? (0: No interleaving, 1: Inter without Intra, 2: Inter with Intra)



**Figure 1.17 – Interleave process with class-wise control of interleaving**

The width of interleave matrix is chosen according to the FEC used. In case block codes are used, i.e. In-band information is protected with “Basic set of FEC codes” as shown in Table 1.38, the length of the codeword is used as this width (see subclause 1.8.4.3). In case the RCPC code is used, 28-bit is used as this width.

#### 1.8.4.9 Class reordered output

EP tool has a function to re-order the class output to the audio decoder, in order to align the bitstream order as defined for that codec, independently from the EP tool configuration and audio encoder to EP tool interface. Note that this interface is out of the scope of this specification, and is up to the implementation, while the interface to the audio decoder shall be aligned to the standard bitstream to avoid additional signaling from the encoder side.

The order of the class after this re-ordering is signaled as **class\_output\_order[i][j]** in the out-of-band information. The ep decoder re-orders the classes in the EP frame transmitted using *i*-th pre-defined set, so that the *j*-th class of EP frame is output as (**class\_output\_order[i][j]**)-th class when output to the audio decoder.

## Annex 1.A (informative)

### Audio Interchange Formats

#### 1.A.1 Introduction

The full capabilities and flexibility of MPEG-4 Audio, like the composition of audio scenes out of multiple audio objects, synthetic audio, and text to speech conversion, is available only if MPEG-4 Audio is used together with MPEG-4 Systems (ISO/IEC-14496-1). The interchange formats, defined in here in Annex A, only support a small subset of the capabilities of MPEG-4 Audio, by defining formats for the storage and transmission of a single mono or stereo or multi-channel audio object, very similar to the formats defined in MPEG-1 and MPEG-2.

As already stated in the introduction to subpart 1, the normative elements in MPEG-4 Audio end with the definition of the payloads (roughly equivalent to a bitstream frame in MPEG-1 and MPEG-2), and the coder configuration structures (resembling the MPEG-1/2 header information). However, there is no normative definition in MPEG-4 Audio how these elements are multiplexed, as this is required only for a limited number of applications. Nevertheless, this informative annex describes such a multiplex. However, MPEG-4 decoders are not obliged to comply to these interface formats.

#### 1.A.2 Interchange format streams

##### 1.A.2.1 MPEG-2 AAC Audio\_Data\_Interchange\_Format, ADIF

Table 1. A.1 – Syntax of adif\_sequence

Syntax	No. of bits	Mnemonic
<pre>adif_sequence() {     adif_header()     raw_data_stream() }</pre>		

Table 1. A.2 – Syntax of adif\_header()

Syntax	No. of bits	Mnemonic
<pre>adif_header() {     <b>adif_id</b>     <b>copyright_id_present</b>     if( <b>copyright_id_present</b> )         <b>copyright_id</b>     <b>original_copy</b>     <b>home</b>     <b>bitstream_type</b>     <b>bitrate</b>     <b>num_program_config_elements</b>     for ( i = 0; i &lt; num_program_config_elements + 1; i++ ) {         if( <b>bitstream_type</b> == '0' )             <b>adif_buffer_fullness</b> </pre>	<pre> 32 1 72 1 1 1 23 4 20 </pre>	<pre> <b>bslbf</b> <b>bslbf</b> <b>bslbf</b> <b>bslbf</b> <b>bslbf</b> <b>uimsbf</b> <b>bslbf</b> <b>uimsbf</b> </pre>

```

    program_config_element()
  }
}

```

Table 1. A.3 – Syntax of raw\_data\_stream()

Syntax	No. of bits	Mnemonic
<pre> raw_data_stream() {   while (data_available()) {     raw_data_block()     byte_alignment()   } } </pre>		

### 1.A.2.2 Audio\_Data\_Transport\_Stream frame, ADTS

Table 1. A.4 – Syntax of adts\_sequence()

Syntax	No. of bits	Mnemonic
<pre> adts_sequence() {   while (nextbits()==syncword) {     adts_frame()   } } </pre>		

Table 1. A.5 – Syntax of adts\_frame()

Syntax	No. of bits	Mnemonic
<pre> adts_frame() {   byte_alignment()   adts_fixed_header()   adts_variable_header()   adts_error_check()   for( i=0; i&lt;no_raw_data_blocks_in_frame+1; i++) {     raw_data_block()   } } </pre>		

#### 1.A.2.2.1 Fixed Header of ADTS

Table 1. A.6 – Syntax of adts\_fixed\_header()

Syntax	No. of bits	Mnemonic
<pre> adts_fixed_header() {   <b>Syncword</b>   <b>ID</b>   <b>Layer</b>   <b>protection_absent</b>   <b>Profile_ObjectType</b> </pre>	<p>12</p> <p>1</p> <p>2</p> <p>1</p> <p>2</p>	<p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>uimsbf</b></p> <p><b>bslbf</b></p> <p><b>uimsbf</b></p>

<b>sampling_frequency_index</b>	<b>4</b>	<b>uimsbf</b>
<b>private_bit</b>	<b>1</b>	<b>bslbf</b>
<b>channel_configuration</b>	<b>3</b>	<b>uimsbf</b>
<b>original/copy</b>	<b>1</b>	<b>bslbf</b>
<b>home</b>	<b>1</b>	<b>bslbf</b>
<b>Emphasis</b>	<b>2</b>	<b>bslbf</b>
}		

1.A.2.2.2 Variable Header of ADTS

Table 1. A.7 – Syntax of adts\_variable\_header()

Syntax	No. of bits	Mnemonic
adts_variable_header()		
{		
<b>copyright_identification_bit</b>	<b>1</b>	<b>bslbf</b>
<b>copyright_identification_start</b>	<b>1</b>	<b>bslbf</b>
<b>aac_frame_length</b>	<b>13</b>	<b>bslbf</b>
<b>adts_buffer_fullness</b>	<b>11</b>	<b>bslbf</b>
<b>no_raw_data_blocks_in_frame</b>	<b>2</b>	<b>uimsbf</b>
}		

1.A.2.2.3 Error detection

Table 1. A.8 – Syntax of adts\_error\_check

Syntax	No. of bits	Mnemonic
adts_error_check()		
{		
if (protection_absent == '0')		
<b>crc_check</b>	<b>16</b>	<b>Rpchof</b>
}		

1.A.3 Decoding of interface formats

1.A.3.1 Audio\_Data\_Interchange\_Format (ADIF)

1.A.3.1.1 Definitions: Bitstream elements for ADIF:

- **adif\_sequence():** A sequence according to the Audio\_Data\_Interchange\_Format (Table 6.1).
- **adif\_header():** Header of the Audio\_Data\_Interchange\_Format located at the beginning of an adif\_sequence (Table 6.2).
- **adif\_id:** ID that indicates the Audio\_Data\_Interchange\_Format. Its value is 0x41444946 (most significant bit first), the ASCII representation of the string „ADIF“ (Table 6.2).
- **copyright\_id\_present:** Indicates whether **copyright\_id** is present or not (Table 6.2).
- **copyright\_id:** The field consists of an 8-bit copyright\_identifier, followed by a 64-bit copyright\_number (Table 6.2). The copyright identifier is given by a Registration Authority as designated by SC 29. The copyright\_number is a value which identifies uniquely the copyrighted material. See ISO/IEC 13818-3, subclause 2.5.2.13.
- **original\_copy:** See ISO/IEC 11172-3, subclause 2.4.2.3 definition for **copyright**.
- **home:** See ISO/IEC 11172-3, subclause 2.4.2.3 definition for **original/copy**.

- **bitstream\_type**: A flag indicating the type of a bitstream (Table 6.2): '0' constant rate bitstream. This bitstream may be transmitted via a channel with constant rate '1' variable rate bitstream. This bitstream is not designed for transmission via constant rate
- **bitrate**: A 23 bit unsigned integer indicating either the bitrate of the bitstream in bits/sec in case of constant rate bitstream or the maximum peak bitrate (measured per frame) in case of variable rate bitstreams. A value of 0 indicates that the bitrate is not known (Table 6.2).
- **num\_program\_config\_element**: Number of program config elements specified for this adif\_sequence().
- **adif\_buffer\_fullness**: State of the bit reservoir after encoding the first raw\_data\_block() in the adif\_sequence. It is transmitted as the number of available bits in the bit reservoir.
- **program\_config\_element()**: Contains information about the configuration for one program. See subpart 4 for the definition.
- **raw\_data\_block()**: Defined in subpart 4.

### 1.A.3.1.2 Description

The raw\_data\_block() contains all data which belongs to the audio (including ancillary data). Beyond that, additional information like sampling\_frequency is needed to fully describe an audio sequence. The Audio\_Data\_Interchange\_Format (ADIF) contains all elements that are necessary to describe a bitstream according to this standard.

The Audio\_Data\_Interchange\_Format (ADIF) contains one header at the start of the sequence followed by a raw\_data\_stream (). The raw\_data\_stream() may not contain any further channel\_configuration\_elements.

As such, the ADIF is useful only for systems with a defined start and no need to start decoding from within the audio data stream, such as decoding from disk file. It can be used as an interchange format in that it contains all information necessary to decode and play the audio data.

### 1.A.3.2 Audio\_Data\_Transport\_Stream (ADTS)

#### 1.A.3.2.1 Definitions: Bitstream elements for ADTS:

- **original\_copy**: See ISO/IEC 11172-3, subclause 2.4.2.3 definition for **copyright**.
- **home**: See ISO/IEC 11172-3, subclause 2.4.2.3 definition for **original/copy**.
- **adts\_sequence()**: A sequence according to Audio\_Data\_Transport\_Stream ADTS (Table A.4).
- **adts\_frame()**: An ADTS frame, consisting of a fixed header, a variable header, an optional error check and a specified number of raw\_data\_blocks() (Table A.5).
- **adts\_fixed\_header()**: Fixed header of ADTS. The information in this header does not change from frame to frame. It is repeated every frame to allow random access into a bitstream bitstream (Table A.6).
- **adts\_variable\_header()**: Variable header of ADTS. This header is transmitted every frame as well as the fixed header, but contains data that changes from frame to frame (Table A.7).
- **adts\_error\_check()**: CRC error detection data generated as described in ISO/IEC 11172-3, subclause 2.4.3.1.

The following bits are protected and fed into the CRC algorithm in order of their appearance:

- all bits of the headers
- first 192 bits of any
  - single\_channel\_element (SCE)
  - channel\_pair\_element (CPE)
  - coupling\_channel\_element (CCE)
  - low frequency enhancement channel (LFE).

In addition, the first 128 bits of the second individual\_channel\_stream in the channel\_pair\_element must be protected. All information in any program configuration element or data element must be protected. For any element where the specified protection length of 128 or 192 bits exceeds its actual length, the element is zero padded to the specified protection length for CRC calculation.

Note that "all bits of the header" refers to the bits in the adts\_fixed\_header and adts\_variable\_header; that the id\_syn\_ele bits shall be excluded from CRC protection, and that if the length of a CPE is shorter than 192 bits, zero data are appended to achieve the length of 192 bits. Furthermore, if the first ICS of the CPE ends at the Nth bit ( $N < 192$ ), the first  $(192 - N)$  bits of the second ICS are protected twice. For example, if the second ICS starts at the

190th bit of CPE, the first 3 bits of the second ICS are protected twice. Finally, if the length of the second ICS is shorter than 128 bits, zero data are appended to achieve the length of 128 bits.

- **byte\_alignment():** If called from within a raw\_data\_block then align with respect to the first bit of the raw\_data\_block, else align with respect to the first bit of the header.
- **syncword:** The bit string '1111 1111 1111'. See ISO/IEC 11172-3, subclause 2.4.2.3.
- **ID:** MPEG identifier, set to '1' if the audio data in the ADTS stream are MPEG-2 AAC (See ISO/IEC 13818-7) and set to '0' if the audio data are MPEG-4. See also ISO/IEC 11172-3, subclause 2.4.2.3.
- **layer:** Indicates which layer is used. Set to '00'. See ISO/IEC 11172-3, subclause 2.4.2.3.
- **protection\_absent:** Indicates whether error\_check() data is present or not. Same as syntax element 'protection\_bit' in ISO/IEC 11172-3, subclause 2.4.1 and 2.4.2.
- **profile\_ObjectType:** The interpretation of this bitstream element depends on the value of the ID bit. If ID is equal to '1' this field holds the same information as the profile field in the ADTS stream defined in ISO/IEC 13818-7. If ID is equal to '0' this element denotes the MPEG-4 Audio Object Type according to the table defined in subclause 5.1.1.

**Table 1. A.9 – MPEG-2 Audio profiles and MPEG-4 Audio object types**

• MPEG-2 profile (ID == 1)	• MPEG-4 object type (ID == 0)
• Main profile	• AAC Main
• Low Complexity profile (LC)	• AAC LC
• Scalable Sampling Rate profile (SSR)	• AAC SSR
• (reserved)	• AAC LTP

- **sampling\_frequency\_index:** Indicates the sampling frequency used according to the table defined in subclause 6.2.2.
- **sampling\_frequency:** Integer value of the sampling frequency used.
- **private\_bit:** See ISO/IEC 11172-3, subclause 2.4.2.3.
- **channel\_configuration:** Indicates the channel configuration used. If channel\_configuration is greater than 0, the channel configuration is given by Table 6.3, see subclause 6.3.4. If channel\_configuration equals 0, the channel configuration is not specified in the header and must be given by a program\_config\_element following as first bitstream element in the first raw\_data\_block after the header, or by the implicit configuration (see subpart 4) or must be known in the application.
- **copyright\_identification\_bit:** One the bits of the 72-bit copyright identification field (see copyright\_id above). The bits of this field are transmitted frame by frame; the first bit is indicated by the copyright\_identification\_start bit set to '1'. The field consists of an 8-bit copyright\_identifier, followed by a 64-bit copyright\_number. The copyright identifier is given by a Registration Authority as designated by SC29. The copyright\_number is a value which identifies uniquely the copyrighted material. See ISO/IEC 13818-3, subclause 2.5.2.13.
- **copyright\_identification\_start:** One bit to indicate that the copyright\_identification\_bit in this audio frame is the first bit of the 72-bit copyright identification. If no copyright identification is transmitted, this bit should be kept '0'.
  - '0' no start of copyright identification in this audio frame
  - '1' start of copyright identification in this audio frame
 See ISO/IEC 13818-3, subclause 2.5.2.13.
- **aac\_frame\_length:** Length of the frame including headers and error\_check (Table A.8) in bytes.
- **adts\_buffer\_fullness:** State of the bit reservoir after encoding the first raw\_data\_block() in the ADTS frame. It is transmitted as the number of available bits in the bit reservoir divided by the number of audio channels divided by 32 and truncated to an integer value. A value of hexadecimal 7FF signals that the bitstream is a variable rate bitstream. In this case, buffer fullness is not applicable.
- **no\_raw\_data\_blocks\_in\_frame:** A field indicating how many raw data blocks are multiplexed (number\_of\_raw\_data\_blocks\_in\_frame+1). The minimum value is 0 indicating 1 raw\_data\_block() (Table 1. A.7).
- **data\_available():** Function that returns '1' as long as data is available, otherwise '0' (Help elements).

### 1.A.3.2.2 Description

The `raw_data_block()` contains all data which belongs to the audio (including ancillary data). Beyond that, additional information like `sampling_frequency` is needed to fully describe an audio sequence. Furthermore, additional information that varies from block to block (e.g. to enhance the parsability or error resilience) may be required. Therefore transport streams may be designed for a specific application and are not specified in this standard. However, one non-normative transport stream, called `Audio_Data_Transport_Stream (ADTS)`, is described. It may be used for applications in which the decoder can parse this stream.

The `Audio_Data_Transport_Stream (ADTS)` is similar to syntax used in ISO/IEC 11172-3 and 13818-3. This will be recognized by ISO/IEC 11172-3 decoders as a "Layer 4" bit-stream.

The fixed header of the ADTS contains the syncword plus all parts of the header which are necessary for decoding and which do not change from frame to frame. The variable header of the ADTS contains header data which changes from frame to frame.

The ADTS only supports a `raw_data_stream()` with only one program. The program may have up to 7 channels plus an independently switched coupling channel.

## Annex 1.B (informative)

### Error protection tool

Text file format of out-of-band information, and its example for AAC, Twin-VQ, CELP, and HVXC are presented. In addition, the example of error concealment is presented.

#### 1.B.1 Text format of out-of-band information

The ASCII text representation of out-of-band information is as follows.

```

/* BEGIN */
  printf("%d\n", number_of_predefined_set);
  printf("%d\n", interleave_type);
  printf("%d\n", bit_stuffing);
printf("%d\n", number_of_concatenated_frame);
  for(i=0; i<number_of_predefined_set; i++){
    printf("%d\n", number_of_class[i]);
    for(j=0; j<number_of_class[i]; j++){

      printf("%d %d %d %d %d\n", length_escape[i][j], rate_escape[i][j], crclen_escape[i][j]
, concatenate_flag[i][j], fec_type[i][j]);
      if( fec_type[i][j] == 0)
        printf("%d\n", termination_switch[i][j] );
      if(interleave_type == 2)
        printf("%d\n", interleave_switch[i][j]);
printf("%d\n", class_optional) ;
      if(length_escape[i][j] == 1) /* ESC */
        printf("%d\n", number_of_bits_for_lenghtgh[i][j]);
      else /* not ESC */
        printf("%d\n", class_length[i][j]);
      if(rate_escape[i][j] != 1) /* not ESC */
        printf("%d\n", class_rate[i][j]);
      if(crclen_escape[i][j] != 1) /* not ESC */
        printf("%d\n", class_crclen[i][j]);
    }
  }
printf("%d\n", class_reordered_output);
if (class_reordered_output ==1){
  for(j=0 ; j<number_of_class[i] ;j++){
    printf("%d\n", class_output_prder[i][j]) ;
  }
}
printf("%d\n", header_protection);
  if( header_protection == 1){
printf("%d\n", header_rate);
printf("%d\n", header_crclen);
  }
printf("%d\n", rs_fec_capability);

/* END */

```



## 1.B.2 Example of out-of-band information

### 1.B.2.1 Example for AAC

based on the error sensitivity category assignment described within the normative part, the following error protection setup could be used, while sensitivity categories are directly mapped to classes. This example shows just a simple setup using one channel and no extension\_payload().

class	length	interleaving	SRCPD puncture rate	CRC length
0	6 bit in-band field	intra-frame	8/24	6
1	12 bit in-band field	intra-frame	8/24	6
2	9 bit in-band field	inter-frame	8/8	6
3	9 bit in-band field	none	8/8	4
4	until the end	inter-frame	8/8	none

### 1.B.2.2 Example for Twin-VQ

This subclause describes examples of the bit assignment of UEP to the Scalable Audio Profile (TwinVQ object).

Here two encoding modes, PPC (Periodic Peak Component) - enable mode and disable mode, are described. Normally, encoder can adaptively select the PPC switch, but we force the switch always ON or always OFF in this experiment. If PPC is ON, 43 bits are assigned to quantize periodic peak components and these bits should be protected as side information.

For each mode we show bit assignment of four different bitrates, 16 kbit/s mono, 32 kbit/s stereo, 8 kbit/s + 8 kbit/s scalable mono and 16 kbit/s + 16 kbit/s stereo for each mode.

In all cases, error correction and detection tools are applied to only 10 % of bits for the side information. Remaining bits for the index of MDCT coefficients have no protection at all. As a result of these bit allocations, increase of bitrate comparing with the original source rate is around 10 % in case of PPC switch is ON, and less than 10% in case of the switch is OFF.

(A) PPC(Periodic Peak Component) enable version

- 16 kbit/s mono  
Class 1: 121 bit(fixed),SRCPD code rate 8/12, 8 bit CRC  
Class 2: 839 bit(fixed),SRCPD code rate 8/8,no CRC

```

1      /* number of predefined sets */
1      /* interleaving */
0      /* bitsstuffing */
1      /* number_of_concatenated_frame */
2      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
121    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
8      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
839    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0   /* class_reordered_output, header_protection, rs_fec_capability */

```

## ISO/IEC 14496-3:2001(E)

- 32 kbit/s stereo  
Class 1: 238 bit(fixed),SRCPC code rate 8/12, 10 bit CRC  
Class 2: 1682 bit(fixed),SRCPC code rate 8/8,no CRC

```
1      /* number of predefined sets */
1      /* interleaving */
0      /* bitstuffing */
1      /* number_of_concatenated_frame */
2      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
238    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
10     /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
1682   /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0   /* class_reordered_output, header_protection, rs_fec_capability */
```

- 8 kbit/s + 8 kbit/s scalable mono  
Class 1: 121 bit(fixed),SRCPC code rate 8/12, 8 bit CRC  
Class 2: 359 bit(fixed),SRCPC code rate 8/8,no CRC  
Class 3: 72 bit(fixed),SRCPC code rate 8/12, 8 bit CRC  
Class 4: 408 bit(fixed),SRCPC code rate 8/8,no CRC

```
1      /* number of predefined sets */
1      /* interleaving */
0      /* bitstuffing */
1      /* number_of_concatenated_frame */
4      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
121    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
8      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
359    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
72     /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
8      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
408    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0   /* class_reordered_output, header_protection, rs_fec_capability */
```

- 16 kbit/s + 16 kbit/s scalable stereo  
Class 1: 238 bit(fixed),SRCPC code rate 8/12, 10 bit CRC  
Class 2: 722 bit(fixed),SRCPC code rate 8/8,no CRC

Class 3: 146 bit(fixed),SRCPC code rate 8/12, 10 bit CRC  
 Class 4: 814 bit(fixed),SRCPC code rate 8/8,no CRC

```

1      /* number of predefined sets */
1      /* interleaving */
0      /* bitstuffing */
1      /* number_of_concatenated_frame */
4      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
238    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
10     /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
722    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
146    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
10     /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
814    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0   /* class_reordered_output, header_protection, rs_fec_capability */
  
```

#### (B)PPC disable version

- 16 kbit/s mono  
 Class 1: 78 bit(fixed),SRCPC code rate 8/12, 8 bit CRC  
 Class 2: 882 bit(fixed),SRCPC code rate 8/8,no CRC

```

1      /* number of predefined sets */
1      /* interleaving */
0      /* bitstuffing */
1      /* number_of_concatenated_frame */
2      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
78     /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
8      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
882    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0   /* class_reordered_output, header_protection, rs_fec_capability */
  
```

- 32 kbit/s stereo  
 Class 1: 152 bit(fixed),SRCPC code rate 8/12, 10 bit CRC

Class 2: 1768 bit(fixed),SRCPC code rate 8/8,no CRC

```

1      /* number of predefined sets */
1      /* interleaving */
0      /* bitstuffing */
1      /* number_of_concatenated_frame */
2      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
152    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
10     /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
1768   /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0   /* class_reordered_output, header_protection, rs_fec_capability */

```

- 8 kbit/s + 8 kbit/s scalable mono
  - Class 1: 78 bit(fixed),SRCPC code rate 8/12, 8 bit CRC
  - Class 2: 402 bit(fixed),SRCPC code rate 8/8,no CRC
  - Class 3: 72 bit(fixed),SRCPC code rate 8/12, 8 bit CRC
  - Class 4: 408 bit(fixed),SRCPC code rate 8/8,no CRC

```

1      /* number of predefined sets */
1      /* interleaving */
0      /* bitstuffing */
1      /* number_of_concatenated_frame */
4      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
78     /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
8      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
402    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
72     /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
8      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
408    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0   /* class_reordered_output, header_protection, rs_fec_capability */

```

- 16 kbit/s + 16 kbit/s scalable stereo
  - Class 1: 152 bit(fixed),SRCPC code rate 8/12, 10 bit CRC
  - Class 2: 808 bit(fixed),SRCPC code rate 8/8,no CRC
  - Class 3: 146 bit(fixed),SRCPC code rate 8/12, 10 bit CRC

Class 4: 814 bit(fixed),SRCPC code rate 8/8,no CRC

```

1      /* number of predefined sets */
1      /* interleaving */
0      /* bitstuffing */
1      /* number_of_concatenated_frame */
4      /* number of classes */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
152    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
10     /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
808    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0      /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
146    /* bits used for class length (0 = until the end) */
4      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
10     /* crc length */
0 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
0 0 0   /* fec_type, termination_switch, class_optional */
814    /* bits used for class length (0 = until the end) */
0      /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */

0      /* crc length */

0 0 0 /* class_reordered_output, header_protection, rs_fec_capability */

```

### 1.B.2.3 Example for CELP

The following tables provide an overview of the number of bit that are assigned to each error sensitivity category, dependent on the configuration.

#### 1.B.2.3.1 MPE-Narrowband Mode

Table 1. B.1 – Overview of bit assignment for MPE-narrowband

MPE-Mode	subframes	Bit/Frame	bitrate	ECR0	ECR1	ECR2	ECR3	ECR4
0	4	154	3850	6	13	20	37	78
1	4	170	4250	6	13	20	41	90
2	4	186	4650	6	13	20	45	102
3	3	147	4900	5	11	16	36	79
4	3	156	5200	5	11	16	39	85
5	3	165	5500	5	11	16	42	91
6	2	114	5700	4	9	12	29	60
7	2	120	6000	4	9	12	31	64
8	2	126	6300	4	9	12	33	68
9	2	132	6600	4	9	12	35	72
10	2	138	6900	4	9	12	37	76
11	2	142	7100	4	9	12	39	78
12	2	146	7300	4	9	12	41	80
13	4	154	7700	6	13	20	41	74
14	4	166	8300	6	13	20	45	82
15	4	174	8700	6	13	20	49	86
16	4	182	9100	6	13	20	53	90
17	4	190	9500	6	13	20	57	94

18	4	198	9900	6	13	20	61	98
19	4	206	10300	6	13	20	65	102
20	4	210	10500	6	13	20	69	102
21	4	214	10700	6	13	20	73	102
22	2	110	11000	4	9	12	33	52
23	2	114	11400	4	9	12	35	54
24	2	118	11800	4	9	12	37	56
25	2	120	12000	4	9	12	39	56
26	2	122	12200	4	9	12	41	56
27	4	186	6200	6	13	20	49	98
28	reserved							
29	reserved							
30	reserved							
31	reserved							

1.B.2.3.2 MPE-Wideband Mode

Table 1. B.2 – Overview of bit assignment for MPE-wideband

MPE-Mode	subframes	Bit/Frame	bitrate	ECR0	ECR1	ECR2	ECR3	ECR4	
0	4	218	10900	17	20	27	45	109	
1	4	230	11500	17	20	27	49	117	
2	4	242	12100	17	20	27	53	125	
3	4	254	12700	17	20	27	57	133	
4	4	266	13300	17	20	27	61	141	
5	4	278	13900	17	20	27	65	149	
6	4	286	14300	17	20	27	69	153	
			7	reserved					
8	8	294	14700	17	32	43	61	141	
9	8	318	15900	17	32	43	69	157	
10	8	342	17100	17	32	43	77	173	
11	8	358	17900	17	32	43	85	181	
12	8	374	18700	17	32	43	93	189	
13	8	390	19500	17	32	43	101	197	
14	8	406	20300	17	32	43	109	205	
15	8	422	21100	17	32	43	117	213	
16	2	136	13600	17	14	19	29	57	
17	2	142	14200	17	14	19	31	61	
18	2	148	14800	17	14	19	33	65	
19	2	154	15400	17	14	19	35	69	
20	2	160	16000	17	14	19	37	73	
21	2	166	16600	17	14	19	39	77	
22	2	170	17000	17	14	19	41	79	
			23	reserved					
24	4	174	17400	17	20	27	37	73	
25	4	186	18600	17	20	27	41	81	
26	4	198	19800	17	20	27	45	89	
27	4	206	20600	17	20	27	49	93	
28	4	214	21400	17	20	27	53	97	
29	4	222	22200	17	20	27	57	101	
30	4	230	23000	17	20	27	61	105	
31	4	238	23800	17	20	27	65	109	

1.B.2.3.3 RPE-Wideband Mode

Table 1. B.3 – Characteristic parameters for wideband CELP with RPE

RPE Mode	subframes	Bit/frame	bitrate	ERC0	ERC1	ERC2	ERC3	ERC4
0	6	216	14400	40	24	34	25	93
1	4	160	16000	32	18	26	21	63
2	8	280	18667	48	30	42	29	131
3	10	338	22533	56	36	50	33	163

1.B.2.4 Example for HVXC

- 2 kbit/s source coder  
 Class 1: 22bit(fixed), SRCPC code rate 8/16, 6 bit CRC  
 Class 2: 4 bit(fixed), SRCPC code rate 8/8, 1 bit CRC  
 Class 3: 4 bit(fixed), SRCPC code rate 8/8, 1 bit CRC  
 Class4: 10bit(fixed), SRCPC code rate 8/8, no CRC

```

1          /* number of predefined sets */
2          /* bit interleaving */
0          /* bitstuffing */
2          /* 2 frame concatenate */
4          /* number of classes */
0 0 0 1 0 0 2 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
22         /* bits used for class length (0 = until the end) */
8          /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
6          /* crc length */
0 0 0 0 0 0 3 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4          /* bits used for class length (0 = until the end) */
0          /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
1          /* crc length */
0 0 0 0 0 0 3 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4          /* bits used for class length (0 = until the end) */
0          /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
1          /* crc length */
0 0 0 1 0 0 3 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
10         /* bits used for class length (0 = until the end) */
0          /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0          /* crc length */
    
```

- 4kbit/s source coder  
 Class 1: 33bit(fixed), SRCPC code rate 8/16, 6bit CRC  
 Class 2: 22bit(fixed), SRCPC code rate 8/8, 6bit CRC  
 Class 3: 4bit(fixed), SRCPC code rate 8/8, 1bit CRC  
 Class 4: 4bit(fixed), SRCPC code rate 8/8, 1bit CRC  
 Class 5: 17bit(fixed), SRCPC code rate 8/8, no CRC

```

1          /* number of predefined sets */
2          /* 1 bit interleaving */
0          /* bitstuffing */
2          /* 2 frame concatenate */
5          /* number of classes */
0 0 0 1 0 0 2 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
33         /* bits used for class length (0 = until the end) */
8          /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
6          /* crc length */
    
```

```

0 0 0 1 0 0 3 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
22 /* bits used for class length (0 = until the end) */
0 /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
6 /* crc length */
0 0 0 0 0 0 3 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4 /* bits used for class length (0 = until the end) */
0 /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
1 /* crc length */
0 0 0 0 0 0 3 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4 /* bits used for class length (0 = until the end) */
0 /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0 /* crc length */
0 0 0 1 0 0 3 0 /* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
17 /* bits used for class length (0 = until the end) */
0 /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0 /* crc length */

```

**1.B.2.5 Example for ER BSAC**

This subclause describes examples of the bit assignment of Unequal Error Protection (UEP) to the ER BSAC Object Type.

The lower error sensitivity category (ESC) described in subclause 4.5.2.6.3 of subpart 4 indicates the class with the higher error sensitivity, whereas the higher ESC indicates the class with the lower sensitivity. Based on the error sensitivity category of the BSAC syntax, the following error protection setup example could be used, This example shows just a simple setup where sensitivity categories are directly mapped to classes.

Class	category	length	interleaving	SRCPC puncture rate	CRC length
0	0	9 bit in-band field	intra-frame	8/24	6
1	others	11 bit in-band field	no	8/8	none

In this example, error correction and detection tools are applied to only the general side information. Remaining bits for the index of MDCT coefficients have no protection at all. As a result of these bit allocations, increase of bitrate comparing with the original source rate is around 10 %.

And, the predefined set for UEP can be set up in addition to the UPE class setup as follows :

```

1 /* number of predefined sets */
1 /* interleaving */
0 /* bitstuffing */
1 /* number_of_concatenated_frame */
2 /* number of classes */
1 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
9 /* number of bits for length */
16 /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
6 /* crc length */
1 0 0 0 /* length_esc, srcpc_esc, crc_esc, concatenate_flag */
11 /* number of bits for length */
0 /* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0 /* crc length */

```

**1.B.3 Example of error concealment**

The error concealment tool is an optional decoder tool to reduce the quality degradation of decoded signals when the decoder input bitstream is affected by errors, such as bitstream transmission error. This is especially valid in



applying the MPEG-4/Audio tools to radio applications. The bitstream of a frame is compensated. The error detection and decision method to replace a frame bitstream are not defined in this subclause and decided based on each application.

### 1.B.3.1 Example for CELP

#### 1.B.3.1.1 Overview of the error concealment tool

The error concealment tool is used with the MPEG-4 CELP decoder described in ISO/IEC 14496-3. This tool reduces unpleasant noise when the MPEG-4 CELP decodes speech from erroneous input frame data. It also enables the MPEG-4 CELP to decode speech even if the input frame data is lost.

The tool has two operating modes: a Bit Error (BE) mode and a Frame Erasure (FE) mode. The mode is switched based on the availability of frame data at the decoder. When frame data is available (the BE mode), decoding is done using the frame data received in the past and the usable subpart of the current frame data. When frame data is not available (the FE mode), the decoder generates the speech using only the past frame data.

This tool operates according to a flag (the *BF\_flag*) that indicates whether the frame data is complete (*BF\_flag*=0), or is damaged by corruption and/or lost (*BF\_flag*=1). The flag is usually given by the channel coder or the transmission system.

This tool operates in Coding Mode II (subclause 3.1.2.1 of ISO/IEC 14496-3:2001), which has narrow band, wideband and band width scalable modes that use Multi-Pulse Excitation at sampling rates of 8 and 16 kHz.

#### 1.B.3.1.2 Definitions

BE:	Bit Error
BWS:	BandWidth Scalable
FE:	Frame Erasure
LP:	Linear Prediction
LSP:	Line Spectral Pair
MPE:	Multi-Pulse Excitation
NB:	Narrow Band
RMS:	Root Mean Square (the frame energy)
WB:	WideBand

#### 1.B.3.1.3 Helping variables

<i>frame_size</i> :	the number of samples in a frame
<i>g_ac</i> :	the adaptive codebook gain
<i>g_ec</i> :	the MPE gain
<i>lpc_order</i> :	the order of LP
<i>signal_mode</i> :	the speech mode
<i>signal_mode_pre</i> :	the speech mode of the previous frame

#### 1.B.3.1.4 Specifications of the error concealment tool

The error concealment tool operates based on the transition model with six states depicted in Figure 1.B.1. The state indicates the quality of the transmission channel. The bigger the state number is, the worse the channel quality is. Each state has a different concealment operation. The initial state in decoding is State 0 and the state is transited based on the *BF\_flag*. Each concealment operation is described in the following subclauses.

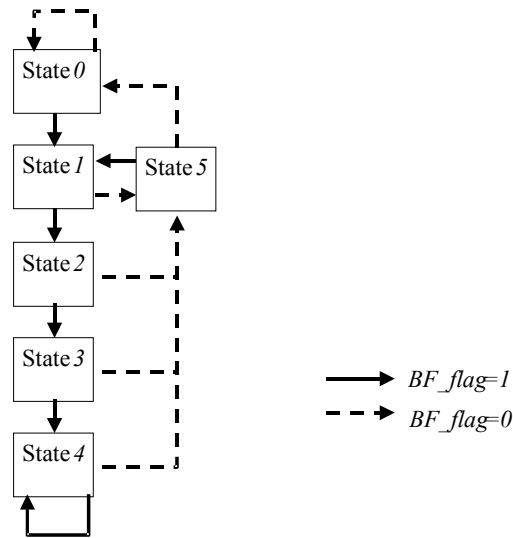


Figure 1.B.1 – State transition model for controlling the error concealment

#### 1.B.3.1.4.1 Operations in States 0 and 5

The decoding process is identical to that of the MPEG-4 CELP decoder with the following exceptions regarding the adaptive codebook and the excitation codebook gains:

In State 0 following State 5, and in State 5:

(1) for the first 80 samples in and after the frame where the *BF\_flag* is changed from 1 to 0, the gains  $g_{ac}$  and  $g_{ec}$  are calculated from the gains  $g_{ac}'$  and  $g_{ec}'$  decoded from the current frame data as follows:

```
if ( $g_{ac} > 1.0$ ) {
     $g_{ec} = g_{ec}' / g_{ac}'$ ;
     $g_{ac} = 1.0$ ;
}
```

(2) for the 160 samples that follow the first 80 samples, the gains are calculated as:

```
if( $g_{ac} > 1.0$ ){
     $g_{ec} = g_{ec}' * (g_{ac}' + 1.0) / g_{ac}' / 2.0$ ;
     $g_{ac} = (g_{ac}' + 1.0) / 2.0$ ;
},
```

where these operations continue in, at most, four subframes.

#### 1.B.3.1.4.2 Operations in States 1, 2, 3 and 4

The decoding process is identical to that of the MPEG-4 CELP decoder with the exceptions described in the following subclauses.

##### 1.B.3.1.4.2.1 Speech mode

**FE mode:**

The speech mode (*signal\_mode*) is decoded from the previous frame data.

**BE mode:**

The speech mode (*signal\_mode*) is decoded from the current frame data for *signal\_mode\_pre*=0 or 1. Otherwise, the mode is decoded from the previous frame data.

**1.B.3.1.4.2.2 Multi-Pulse Excitation (MPE)****FE mode:**

The MPE is decoded from the randomly generated frame data.

**BE mode:**

The MPE is decoded from the frame data received in the current frame.

**1.B.3.1.4.2.3 RMS**

For State 1 through  $K$ , the RMS in the last subframe of the previous frame is used after being attenuated in the first subframe of the current frame. In the following subframes, the RMS in the previous subframe is used after being attenuated. The attenuation level  $P_{att}$  depends on the state as follows:

$$P_{att} = \begin{cases} 0.4 \text{ dB} & \text{for State } 1, \dots, K \\ 1.2 \text{ dB} & \text{for State } K + 1, \dots \end{cases}$$

where  $K$  is the smaller number of 4 and  $K_0$ , and  $K_0$  is the maximum integer satisfying  $frame\_size \times K_0 \leq 320$ .

**1.B.3.1.4.2.4 LSP**

The LSPs decoded in the previous frame are used. In the BWS mode, the LSP codevectors should be buffered for the interframe prediction described in subclause 3.5.6.3.3 of ISO/IEC 14496-3:2001. However, when the frame data is corrupted or lost, the correct codevector can not be obtained. Therefore, the buffered codevector ( $b/sp[0][i]$ ) is estimated from the LSP ( $q/sp\_pre[i]$ ) in the previous frame, the predicted LSP ( $vec\_hat[i]$ ) and the prediction coefficient ( $cb[0][i]$ ) in the current frame as follows:

```
for ( i = 0; i < lpc_order; i++ ) {
    b/sp[0][i] = (q/sp_pre[i] - vec_hat[i])/cb[0][i];
}
```

**1.B.3.1.4.2.5 Delay of the adaptive codebook****FE mode:**

All the delays of the adaptive codebook are decoded from the delay index received in the last subframe of the previous frame.

**BE mode:**

(1) When *signal\_mode\_pre*=0, the delays are decoded from the current frame data.

(2) When *signal\_mode\_pre*=1, and if the maximum difference between the delay indices of the adjacent subframes in the frame are less than 10, the delays are decoded from the delay indices in the current frame. In each subframe where the difference in the delay indices between the current and the previous subframes is equal to or greater than 10, the delay is decoded from the index of the previous subframe.

(3) When *signal\_mode\_pre*= 2 or 3, the delay is decoded from the index in the last subframe of the previous frame.

**1.B.3.1.4.2.6 Gains**

**1.B.3.1.4.2.6.1 Index operation**

**FE mode:**

All the gains have the same value, which is decoded from the gain index received in the last subframe of the previous frame.

**BE mode:**

The gains are decoded from current frame data.

**1.B.3.1.4.2.6.2 Adjustment operation**

**FE mode:**

(1) When  $signal\_mode\_pre=0$ , the gains  $g_{ac}'$  and  $g_{ec}'$  are decoded from the current frame data. The gains  $g_{ac}$  and  $g_{ec}$  are then obtained by multiplying  $g_{ac}'$  by 0.5 and  $g_{ec}'$  by  $X$ , respectively.  $X$  satisfies the following equation and is calculated in each subframe:

$$(g_{ac}' * g_{ac}')A + (g_{ec}' * g_{ec}')B = ((0.5 * g_{ac}') * (0.5 * g_{ac}'))A + ((X * g_{ec}') * (X * g_{ec}'))B$$

where

$$A = norm_{ac} \times norm_{ac}$$

$$B = norm_{ec} \times norm_{ec} .$$

$norm_{ac}$  and  $norm_{ec}$  are the respective RMS values of the adaptive and the excitation codevectors.

(2) When  $signal\_mode\_pre=1$ , the gains are decoded from the current frame data.

(3) When  $signal\_mode\_pre=2$  or 3, gains for the first 320 samples in or after the frame where the  $BF\_flag$  is changed from 0 to 1, are calculated as:

$$g_{ac} = 0.95 \times 10^{(-0.4/20)}$$

$$g_{ec} = X \times 10^{(-0.4/20)} .$$

After the first 320 samples,

$$g_{ac} = 0.95 \times 10^{(-1.2/20)}$$

$$g_{ec} = X \times 10^{(-1.2/20)} ,$$

where  $X = 0.05 \times \frac{norm_{ac}}{norm_{ec}} .$

**BE mode:**

(1) When  $signal\_mode\_pre=0$  or  $1$ , the gains are calculated using the gains  $g\_ac'$  and  $g\_ec'$  decoded from the current frame data and the gains  $g\_ac\_pre$  and  $g\_ec\_pre$  of the previous subframe so that the calculated gains fall in a normal range and generate no unpleasant noise as follows:

```

if (g_ac > 1.2589){
    g_ec = g_ec' * 1.2589/g_ac';
    g_ac = 1.2589;
}
if (g_ec > 1.2589*g_ec_pre){
    g_ac = g_ac' * 1.2589*g_ec_pre/g_ec';
    g_ec = g_ec_pre * 1.2589;
}
if (signal_mode=1 & g_ac_pre < 1.2589 & g_ac < g_ac_pre*0.7943){
    g_ac = g_ac_pre*0.7943;
    g_ec = g_ec_pre*0.7943;
}

```

(2) When  $signal\_mode\_pre=2$  or  $3$ , the operation is identical to that for  $signal\_mode\_pre=2$  or  $3$  in the FE mode.

**1.B.3.2 Error concealment for the silence compression tool**

In frames where the bitstream received at the decoder is corrupted or lost due to transmission bit errors, error concealment is performed. When the received  $TX\_flag$  is  $1$ , the decoding process is identical to that of the error concealment for the MPEG-4 CELP. For  $TX\_flag=0, 2$  or  $3$ , the decoding process for the  $TX\_flag=0$  is used.

**1.B.4 Example of EP tool setting and error concealment for HVXC**

This subclause describes one example of the implementation of EP (Error Protection) tool and error concealment method for HVXC. Some of perceptually important bits are protected by FEC (forward error correction) scheme and some are checked by CRC to judge whether or not erroneous bits are included. When CRC error occurs, error concealment is executed to reduce perceptible degradation.

It should be noted that error correction method and EP tool setting, error concealment algorithm described below are one example, and they should be modified depending on the actual channel conditions.

**1.B.4.1 Definitions**

--- 2/4kbit/s common parameters ---

LSP1	LSP index 1	(5 bit)
LSP2	LSP index 2	(7 bit)
LSP3	LSP index 3	(5 bit)
LSP4	LSP index 4	(1 bit)
VUV	voiced/unvoiced flag	(2 bit)
Pitch	pitch parameter	(7 bit)
SE_shape1	spectrum index 0	(4 bit)
SE_shape2	spectrum index 1	(4 bit)
SE_gain	spectrum gain index	(5 bit)
VX_shape1[0]	stochastic codebook index 0	(6 bit)
VX_shape1[1]	stochastic codebook index 1	(6 bit)
VX_gain1[0]	gain codebook index 0	(4 bit)
VX_gain1[1]	gain codebook index 1	(4 bit)

--- only 4kbit/s parameters ---

LSP5	LSP index 5	(8 bit)
SE_shape3	4k spectrum index 0	(7 bit)
SE_shape4	4k spectrum index 1	(10 bit)
SE_shape5	4k spectrum index 2	(9 bit)
SE_shape6	4k spectrum index 3	(6 bit)
VX_shape2[0]	4k stochastic codebook index 0	(5 bit)
VX_shape2[1]	4k stochastic codebook index 1	(5 bit)
VX_shape2[2]	4k stochastic codebook index 2	(5 bit)
VX_shape2[3]	4k stochastic codebook index 3	(5 bit)
VX_gain2[0]	4k gain codebook index 0	(3 bit)
VX_gain2[1]	4k gain codebook index 1	(3 bit)
VX_gain2[2]	4k gain codebook index 2	(3 bit)
VX_gain2[3]	4k gain codebook index 3	(3 bit)

### 1.B.4.2 Channel coding

#### 1.B.4.2.1 Protected bit selection

According to the sensitivity of bits, encoded bits are classified to several classes. The number of bits for each class is shown in the Table 1. B.4, Table 1. B.5 (2kbit/s), Table 1. B.6 and Table 1. B.7 (4kbit/s). As an example, bitrate setting of 3.5kbit/s(for 2kbit/s) and 6.2kbit/s(for 4kbit/s) are shown. In these cases, two source coder frames are processed as one set. Suffix “p” means parameters of the “previous” frame, and “c” means those of the “current” frame.

For 3.5kbit/s mode, 6 classes are used. CRC check is applied for class I, II, III, IV, and V bits. Class VI bits are not checked by CRC.

The Table 1. B.4 below shows protected/unprotected(class I...VI) bit assignment in the case where both previous and current frames are voiced.

**Table 1. B.4 – Number of protected/unprotected bits at 3.5kbit/s(voiced frame)**

para-meters	voiced frame						total
	class I bits	class II bits	class III bits	class IV bits	class V bits	class VI bits	
LSP1p/c	5/5	-	-	-	-	-	10
LSP2p/c	2/2	-	-	-	-	5/5	14
LSP3p/c	1/1	-	-	-	-	4/4	10
LSP4p/c	1/1	-	-	-	-	-	2
VUVp/c	2/2	-	-	-	-	-	4
Pitchp/c	6/6	-	-	-	-	1/1	14
SE_gainp/c	5/5	-	-	-	-	-	10
SE_shape1p	-	4	-	-	-	-	4
SE_shape1c	-	-	-	4	-	-	4
SE_shape2p	-	-	4	-	-	-	4
SE_shape2c	-	-	-	-	4	-	4
<b>total</b>	<b>44</b>	<b>4</b>	<b>4</b>	<b>4</b>	<b>4</b>	<b>20</b>	<b>80</b>

The Table 1. B.5 shows protected/unprotected(class I...VI) bit assignment in the case where both previous and current frames are unvoiced.

**Table 1. B.5 – Number of protected/unprotected bits at 3.5kbit/s(unvoiced frame)**

para-meters	unvoiced frame							total
	class I bits	class II bits	class III bits	class IV bits	Class V bits	class VI bits		
LSP1p/c	5/5	-	-	-	-	-	-	10
LSP2p/c	4/4	-	-	-	-	-	3/3	14
LSP3p/c	2/2	-	-	-	-	-	3/3	10
LSP4p/c	1/1	-	-	-	-	-	-	2
VUVp/c	2/2	-	-	-	-	-	-	4
VX_gain1[0]p/c	4/4	-	-	-	-	-	-	8
VX_gain1[1]p/c	4/4	-	-	-	-	-	-	8
VX_shape1[0]p/	-	-	-	-	-	-	6/6	12
VX_shape1[1]p/	-	-	-	-	-	-	6/6	12
<b>total</b>	<b>44</b>	<b>0</b>	<b>0</b>	<b>0</b>	<b>0</b>	<b>0</b>	<b>36</b>	<b>80</b>

When the previous frame is unvoiced and the current frame is voiced, or when the previous frame is voiced and the current frame is unvoiced, the same protected/unprotected bit assignment rules as shown above are used.

For 6.2kbit/s mode, 7 classes are used. CRC check is applied for class I, II, III, IV, V, and VI bits. Class VII bits are not checked by CRC.

The Table 1. B.6 shows protected/unprotected(class I...VII) bit assignment in the case where both previous and current frames are voiced.

**Table 1. B.6 – Number of protected/unprotected bits at 6.2kbit/s(voiced sound)**

Para-meters	voiced sound							total
	class I bits	class II bits	class III bits	class IV bits	class V bits	class VI bits	class VII bits	
LSP1p/c	5/5	-	-	-	-	-	-	10
LSP2p/c	4/4	-	-	-	-	-	3/3	14
LSP3p/c	1/1	-	-	-	-	-	4/4	10
LSP4p/c	1/1	-	-	-	-	-	-	2
LSP5p/c	1/1	-	-	-	-	-	7/7	16
VUVp/c	2/2	-	-	-	-	-	-	4
Pitchp/c	6/6	-	-	-	-	-	1/1	14
SE_gainp/c	5/5	-	-	-	-	-	-	10
SE_shape1p	-	-	4	-	-	-	-	4
SE_shape1c	-	-	-	-	4	-	-	4
SE_shape2p	-	-	-	4	-	-	-	4
SE_shape2c	-	-	-	-	-	4	-	4
SE_shape3p/c	5/5	-	-	-	-	-	2/2	14

SE_shape4p/c	1/1	9/9	-	-	-	-	-	<b>20</b>
SE_shape5p/c	1/1	8/8	-	-	-	-	-	<b>18</b>
SE_shape6p/c	1/1	5/5	-	-	-	-	-	<b>12</b>
<b>Total</b>	<b>66</b>	<b>44</b>	<b>4</b>	<b>4</b>	<b>4</b>	<b>4</b>	<b>34</b>	<b>160</b>

The Table 1. B.7 below shows protected/unprotected(class I...VII) bit assignment in the case where both previous and current frames are unvoiced.

**Table 1. B.7 – Number of protected/unprotected bits at 6.2kbit/s(unvoiced sound)**

Para-meters	unvoiced sound							total
	class I bits	class II bits	class III bits	class IV bits	class V bits	class VI bits	class VII bits	
LSP1p/c	5/5	-	-	-	-	-	-	<b>10</b>
LSP2p/c	4/4	-	-	-	-	-	3/3	<b>14</b>
LSP3p/c	1/1	-	-	-	-	-	4/4	<b>10</b>
LSP4p/c	1/1	-	-	-	-	-	-	<b>2</b>
LSP5p/c	1/1	-	-	-	-	-	7/7	<b>16</b>
VUVp/c	2/2	-	-	-	-	-	-	<b>4</b>
VX_gain1[0]p/c	4/4	-	-	-	-	-	-	<b>8</b>
VX_gain1[1]p/c	4/4	-	-	-	-	-	-	<b>8</b>
VX_shape1[0]p/	-	-	-	-	-	-	6/6	<b>12</b>
VX_shape1[1]p/	-	-	-	-	-	-	6/6	<b>12</b>
VX_gain2[0]p/c	3/3	-	-	-	-	-	-	<b>6</b>
VX_gain2[1]p/c	3/3	-	-	-	-	-	-	<b>6</b>
VX_gain2[2]p/c	3/3	-	-	-	-	-	-	<b>6</b>
VX_gain2[3]p/c	2/2	-	-	-	-	-	1/1	<b>6</b>
VX_shape2[0]p/	-	-	-	-	-	-	5/5	<b>10</b>
VX_shape2[1]p/	-	-	-	-	-	-	5/5	<b>10</b>
VX_shape2[2]p/	-	-	-	-	-	-	5/5	<b>10</b>
VX_shape2[3]p/	-	-	-	-	-	-	5/5	<b>10</b>
<b>total</b>	<b>66</b>	<b>0</b>	<b>0</b>	<b>0</b>	<b>0</b>	<b>0</b>	<b>94</b>	<b>160</b>

When the previous frame is unvoiced and the current frame is voiced, or when the previous frame is voiced and the current frame is unvoiced, the same protected/unprotected bit assignment rules as shown above are used.

The bit order for UEP input is shown from Table 1. B.8 through Table 1. B.11 (for 2kbit/s), and from Table 1. B.12 through Table 1. B.15 (for 4kbit/s). These tables show the bit order for each of the combinations of V/UV condition of 2 frames. For example, if previous frame is voiced and current frame is voiced at 2kbit/s mode, Table 1. B.8 is used. The bit order is arranged according to the error sensitivity. The column "Bit" denotes the bit index of the parameter. "0" means LSB.

**Table 1. B.8 – Bit Order for 2kbit/s(voiced frame -- voiced frame)**

voiced frame – voiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class I Bit</b>			<b>28</b>	SE_gainc	1	<b>54</b>	SE_shape1c	1
<b>0</b>	VUVp	1	<b>29</b>	SE_gainc	0	<b>55</b>	SE_shape1c	0
<b>1</b>	VUVp	0	<b>30</b>	LSP1c	4	<b>Class V Bit</b>		



2	LSP4p	0	31	LSP1c	3	56	SE_shape2c	3
3	SE_gainp	4	32	LSP1c	2	57	SE_shape2c	2
4	SE_gainp	3	33	LSP1c	1	58	SE_shape2c	1
5	SE_gainp	2	34	LSP1c	0	59	SE_shape2c	0
6	SE_gainp	1	35	Pitchc	6	<b>Class VI Bit</b>		
7	SE_gainp	0	36	Pitchc	5	60	LSP2p	4
8	LSP1p	4	37	Pitchc	4	61	LSP2p	3
9	LSP1p	3	38	Pitchc	3	62	LSP2p	2
10	LSP1p	2	39	Pitchc	2	63	LSP2p	1
11	LSP1p	1	40	Pitchc	1	64	LSP2p	0
12	LSP1p	0	41	LSP2c	6	65	LSP3p	3
13	Pitchp	6	42	LSP3c	4	66	LSP3p	2
14	Pitchp	5	43	LSP2c	5	67	LSP3p	1
15	Pitchp	4	<b>Class II Bit</b>			68	LSP3p	0
16	Pitchp	3	44	SE_shape1p	3	69	Pitchp	0
17	Pitchp	2	45	SE_shape1p	2	70	LSP2c	4
18	Pitchp	1	46	SE_shape1p	1	71	LSP2c	3
19	LSP2p	6	47	SE_shape1p	0	72	LSP2c	2
20	LSP3p	4	<b>Class III Bit</b>			73	LSP2c	1
21	LSP2p	5	48	SE_shape2p	3	74	LSP2c	0
22	VUVc	1	49	SE_shape2p	2	75	LSP3c	3
23	VUVc	0	50	SE_shape2p	1	76	LSP3c	2
24	LSP4c	0	51	SE_shape2p	0	77	LSP3c	1
25	SE_gainc	4	<b>Class IV Bit</b>			78	LSP3c	0
26	SE_gainc	3	52	SE_shape1c	3	79	Pitchc	0
27	SE_gainc	2	53	SE_shape1c	2			

Table 1. B.9 – Bit Order for 2kbit/s(voiced frame – unvoiced frame)

voiced frame – unvoiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class I Bit</b>			28	VX_gain1[0]c	0	54	LSP2c	0
0	VUVp	1	29	VX_gain1[1]c	3	55	LSP3c	2
1	VUVp	0	30	VX_gain1[1]c	2	<b>Class V Bit</b>		
2	LSP4p	0	31	VX_gain1[1]c	1	56	LSP3c	1
3	SE_gainp	4	32	VX_gain1[1]c	0	57	LSP3c	0
4	SE_gainp	3	33	LSP1c	4	58	VX_shape1[0]c	5
5	SE_gainp	2	34	LSP1c	3	59	VX_shape1[0]c	4
6	SE_gainp	1	35	LSP1c	2	<b>Class VI Bit</b>		
7	SE_gainp	0	36	LSP1c	1	60	LSP2p	4
8	LSP1p	4	37	LSP1c	0	61	LSP2p	3
9	LSP1p	3	38	LSP2c	6	62	LSP2p	2
10	LSP1p	2	39	LSP2c	5	63	LSP2p	1
11	LSP1p	1	40	LSP2c	4	64	LSP2p	0
12	LSP1p	0	41	LSP2c	3	65	LSP3p	3
13	Pitchp	6	42	LSP3c	4	66	LSP3p	2
14	Pitchp	5	43	LSP3c	3	67	LSP3p	1
15	Pitchp	4	<b>Class II Bit</b>			68	LSP3p	0
16	Pitchp	3	44	SE_shape1p	3	69	Pitchp	0

17	Pitchp	2	45	SE_shape1p	2	70	VX_shape1[0]c	3
18	Pitchp	1	46	SE_shape1p	1	71	VX_shape1[0]c	2
19	LSP2p	6	47	SE_shape1p	0	72	VX_shape1[0]c	1
20	LSP3p	4	<b>Class III Bit</b>			73	VX_shape1[0]c	0
21	LSP2p	5	48	SE_shape2p	3	74	VX_shape1[1]c	5
22	VUVc	1	49	SE_shape2p	2	75	VX_shape1[1]c	4
23	VUVc	0	50	SE_shape2p	1	76	VX_shape1[1]c	3
24	LSP4c	0	51	SE_shape2p	0	77	VX_shape1[1]c	2
25	VX_gain1[0]c	3	<b>Class IV Bit</b>			78	VX_shape1[1]c	1
26	VX_gain1[0]c	2	52	LSP2c	2	79	VX_shape1[1]c	0
27	VX_gain1[0]c	1	53	LSP2c	1			

Table 1. B.10 – Bit Order for 2kbit/s(unvoiced frame – voiced frame)

unvoiced frame – unvoiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class I Bit</b>			28	SE_gainc	1	54	SE_shape1c	1
0	VUVp	1	29	SE_gainc	0	55	SE_shape1c	0
1	VUVp	0	30	LSP1c	4	<b>Class V Bit</b>		
2	LSP4p	0	31	LSP1c	3	56	SE_shape2c	3
3	VX_gain1[0]p	3	32	LSP1c	2	57	SE_shape2c	2
4	VX_gain1[0]p	2	33	LSP1c	1	58	SE_shape2c	1
5	VX_gain1[0]p	1	34	LSP1c	0	59	SE_shape2c	0
6	VX_gain1[0]p	0	35	Pitchc	6	<b>Class VI Bit</b>		
7	VX_gain1[1]p	3	36	Pitchc	5	60	VX_shape1[0]p	3
8	VX_gain1[1]p	2	37	Pitchc	4	61	VX_shape1[0]p	2
9	VX_gain1[1]p	1	38	Pitchc	3	62	VX_shape1[0]p	1
10	VX_gain1[1]p	0	39	Pitchc	2	63	VX_shape1[0]p	0
11	LSP1p	4	40	Pitchc	1	64	VX_shape1[1]p	5
12	LSP1p	3	41	LSP2c	6	65	VX_shape1[1]p	4
13	LSP1p	2	42	LSP3c	4	66	VX_shape1[1]p	3
14	LSP1p	1	43	LSP2c	5	67	VX_shape1[1]p	2
15	LSP1p	0	<b>Class II Bit</b>			68	VX_shape1[1]p	1
16	LSP2p	6	44	LSP2p	2	69	VX_shape1[1]p	0
17	LSP2p	5	45	LSP2p	1	70	LSP2c	4
18	LSP2p	4	46	LSP2p	0	71	LSP2c	3
19	LSP2p	3	47	LSP3p	2	72	LSP2c	2
20	LSP3p	4	<b>Class III Bit</b>			73	LSP2c	1
21	LSP3p	3	48	LSP3p	1	74	LSP2c	0
22	VUVc	1	49	LSP3p	0	75	LSP3c	3
23	VUVc	0	50	VX_shape1[0]p	5	76	LSP3c	2
24	LSP4c	0	51	VX_shape1[0]p	4	77	LSP3c	1
25	SE_gainc	4	<b>Class IV Bit</b>			78	LSP3c	0
26	SE_gainc	3	52	SE_shape1c	3	79	Pitchc	0
27	SE_gainc	2	53	SE_shape1c	2			

Table 1. B.11 – Bit Order for 2kbit/s(unvoiced frame – unvoiced frame)

unvoiced frame – unvoiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class I Bit</b>			<b>28</b>	VX_gain1[0]c	0	<b>54</b>	LSP2c	0
<b>0</b>	VUVp	1	<b>29</b>	VX_gain1[1]c	3	<b>55</b>	LSP3c	2
<b>1</b>	VUVp	0	<b>30</b>	VX_gain1[1]c	2	<b>Class V Bit</b>		
<b>2</b>	LSP4p	0	<b>31</b>	VX_gain1[1]c	1	<b>56</b>	LSP3c	1
<b>3</b>	VX_gain1[0]p	3	<b>32</b>	VX_gain1[1]c	0	<b>57</b>	LSP3c	0
<b>4</b>	VX_gain1[0]p	2	<b>33</b>	LSP1c	4	<b>58</b>	VX_shape1[0]c	5
<b>5</b>	VX_gain1[0]p	1	<b>34</b>	LSP1c	3	<b>59</b>	VX_shape1[0]c	4
<b>6</b>	VX_gain1[0]p	0	<b>35</b>	LSP1c	2	<b>Class VI Bit</b>		
<b>7</b>	VX_gain1[1]p	3	<b>36</b>	LSP1c	1	<b>60</b>	VX_shape1[0]p	3
<b>8</b>	VX_gain1[1]p	2	<b>37</b>	LSP1c	0	<b>61</b>	VX_shape1[0]p	2
<b>9</b>	VX_gain1[1]p	1	<b>38</b>	LSP2c	6	<b>62</b>	VX_shape1[0]p	1
<b>10</b>	VX_gain1[1]p	0	<b>39</b>	LSP2c	5	<b>63</b>	VX_shape1[0]p	0
<b>11</b>	LSP1p	4	<b>40</b>	LSP2c	4	<b>64</b>	VX_shape1[1]p	5
<b>12</b>	LSP1p	3	<b>41</b>	LSP2c	3	<b>65</b>	VX_shape1[1]p	4
<b>13</b>	LSP1p	2	<b>42</b>	LSP3c	4	<b>66</b>	VX_shape1[1]p	3
<b>14</b>	LSP1p	1	<b>43</b>	LSP3c	3	<b>67</b>	VX_shape1[1]p	2
<b>15</b>	LSP1p	0	<b>Class II Bit</b>			<b>68</b>	VX_shape1[1]p	1
<b>16</b>	LSP2p	6	<b>44</b>	LSP2p	2	<b>69</b>	VX_shape1[1]p	0
<b>17</b>	LSP2p	5	<b>45</b>	LSP2p	1	<b>70</b>	VX_shape1[0]c	3
<b>18</b>	LSP2p	4	<b>46</b>	LSP2p	0	<b>71</b>	VX_shape1[0]c	2
<b>19</b>	LSP2p	3	<b>47</b>	LSP3p	2	<b>72</b>	VX_shape1[0]c	1
<b>20</b>	LSP3p	4	<b>Class III Bit</b>			<b>73</b>	VX_shape1[0]c	0
<b>21</b>	LSP3p	3	<b>48</b>	LSP3p	1	<b>74</b>	VX_shape1[1]c	5
<b>22</b>	VUVc	1	<b>49</b>	LSP3p	0	<b>75</b>	VX_shape1[1]c	4
<b>23</b>	VUVc	0	<b>50</b>	VX_shape1[0]p	5	<b>76</b>	VX_shape1[1]c	3
<b>24</b>	LSP4c	0	<b>51</b>	VX_shape1[0]p	4	<b>77</b>	VX_shape1[1]c	2
<b>25</b>	VX_gain1[0]c	3	<b>Class IV Bit</b>			<b>78</b>	VX_shape1[1]c	1
<b>26</b>	VX_gain1[0]c	2	<b>52</b>	LSP2c	2	<b>79</b>	VX_shape1[1]c	0
<b>27</b>	VX_gain1[0]c	1	<b>53</b>	LSP2c	1			

Table 1. B.12 – Bit Order for 4kbit/s(voiced frame – voiced frame)

voiced frame – voiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class 1bit</b>			<b>55</b>	LSP2c	3	<b>Class III Bit</b>		
<b>0</b>	VUVp	1	<b>56</b>	SE_shape3c	6	<b>110</b>	SE_shape1p	3
<b>1</b>	VUVp	0	<b>57</b>	SE_shape3c	5	<b>111</b>	SE_shape1p	2
<b>2</b>	LSP4p	0	<b>58</b>	SE_shape3c	4	<b>112</b>	SE_shape1p	1
<b>3</b>	SE_gainp	4	<b>59</b>	SE_shape3c	3	<b>113</b>	SE_shape1p	0
<b>4</b>	SE_gainp	3	<b>60</b>	SE_shape3c	2	<b>Class IV Bit</b>		
<b>5</b>	SE_gainp	2	<b>61</b>	LSP3c	4	<b>114</b>	SE_shape2p	3
<b>6</b>	SE_gainp	1	<b>62</b>	LSP5c	7	<b>115</b>	SE_shape2p	2
<b>7</b>	SE_gainp	0	<b>63</b>	SE_shape4c	9	<b>116</b>	SE_shape2p	1
<b>8</b>	LSP1p	4	<b>64</b>	SE_shape5c	8	<b>117</b>	SE_shape2p	0
<b>9</b>	LSP1p	3	<b>65</b>	SE_shape6c	5	<b>Class V Bit</b>		

10	LSP1p	2	<b>Class II Bit</b>			118	SE_shape1c	3
11	LSP1p	1	66	SE_shape4p	8	119	SE_shape1c	2
12	LSP1p	0	67	SE_shape4p	7	120	SE_shape1c	1
13	Pitchp	6	68	SE_shape4p	6	121	SE_shape1c	0
14	Pitchp	5	69	SE_shape4p	5	<b>Class VI Bit</b>		
15	Pitchp	4	70	SE_shape4p	4	122	SE_shape2c	3
16	Pitchp	3	71	SE_shape4p	3	123	SE_shape2c	2
17	Pitchp	2	72	SE_shape4p	2	124	SE_shape2c	1
18	Pitchp	1	73	SE_shape4p	1	125	SE_shape2c	0
19	LSP2p	6	74	SE_shape4p	0	<b>Class VII Bit</b>		
20	LSP2p	5	75	SE_shape5p	7	126	LSP2p	2
21	LSP2p	4	76	SE_shape5p	6	127	LSP2p	1
22	LSP2p	3	77	SE_shape5p	5	128	LSP2p	0
23	SE_shape3p	6	78	SE_shape5p	4	129	LSP3p	3
24	SE_shape3p	5	79	SE_shape5p	3	130	LSP3p	2
25	SE_shape3p	4	80	SE_shape5p	2	131	LSP3p	1
26	SE_shape3p	3	81	SE_shape5p	1	132	LSP3p	0
27	SE_shape3p	2	82	SE_shape5p	0	133	LSP5p	6
28	LSP3p	4	83	SE_shape6p	4	134	LSP5p	5
29	LSP5p	7	84	SE_shape6p	3	135	LSP5p	4
30	SE_shape4p	9	85	SE_shape6p	2	136	LSP5p	3
31	SE_shape5p	8	86	SE_shape6p	1	137	LSP5p	2
32	SE_shape6p	5	87	SE_shape6p	0	138	LSP5p	1
33	VUVc	1	88	SE_shape4c	8	139	LSP5p	0
34	VUVc	0	89	SE_shape4c	7	140	Pitchp	0
35	LSP4c	0	90	SE_shape4c	6	141	SE_shape3p	1
36	SE_gainc	4	91	SE_shape4c	5	142	SE_shape3p	0
37	SE_gainc	3	92	SE_shape4c	4	143	LSP2c	2
38	SE_gainc	2	93	SE_shape4c	3	144	LSP2c	1
39	SE_gainc	1	94	SE_shape4c	2	145	LSP2c	0
40	SE_gainc	0	95	SE_shape4c	1	146	LSP3c	3
41	LSP1c	4	96	SE_shape4c	0	147	LSP3c	2
42	LSP1c	3	97	SE_shape5c	7	148	LSP3c	1
43	LSP1c	2	98	SE_shape5c	6	149	LSP3c	0
44	LSP1c	1	99	SE_shape5c	5	150	LSP5c	6
45	LSP1c	0	100	SE_shape5c	4	151	LSP5c	5
46	Pitchc	6	101	SE_shape5c	3	152	LSP5c	4
47	Pitchc	5	102	SE_shape5c	2	153	LSP5c	3
48	Pitchc	4	103	SE_shape5c	1	154	LSP5c	2
49	Pitchc	3	104	SE_shape5c	0	155	LSP5c	1
50	Pitchc	2	105	SE_shape6c	4	156	LSP5c	0
51	Pitchc	1	106	SE_shape6c	3	157	Pitchc	0
52	LSP2c	6	107	SE_shape6c	2	158	SE_shape3c	1
53	LSP2c	5	108	SE_shape6c	1	159	SE_shape3c	0
54	LSP2c	4	109	SE_shape6c	0			

Table 1. B.13 – Bit Order for 4kbit/s(voiced frame – unvoiced frame)

voiced frame – unvoiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class 1bit</b>			<b>55</b>	VX_gain2[0]c	2	<b>Class III Bit</b>		
<b>0</b>	VUVp	1	<b>56</b>	VX_gain2[0]c	1	<b>110</b>	SE_shape1p	3
<b>1</b>	VUVp	0	<b>57</b>	VX_gain2[0]c	0	<b>111</b>	SE_shape1p	2
<b>2</b>	LSP4p	0	<b>58</b>	VX_gain2[1]c	2	<b>112</b>	SE_shape1p	1
<b>3</b>	SE_gainp	4	<b>59</b>	VX_gain2[1]c	1	<b>113</b>	SE_shape1p	0
<b>4</b>	SE_gainp	3	<b>60</b>	VX_gain2[1]c	0	<b>Class IV Bit</b>		
<b>5</b>	SE_gainp	2	<b>61</b>	VX_gain2[2]c	2	<b>114</b>	SE_shape2p	3
<b>6</b>	SE_gainp	1	<b>62</b>	VX_gain2[2]c	1	<b>115</b>	SE_shape2p	2
<b>7</b>	SE_gainp	0	<b>63</b>	VX_gain2[2]c	0	<b>116</b>	SE_shape2p	1
<b>8</b>	LSP1p	4	<b>64</b>	VX_gain2[3]c	2	<b>117</b>	SE_shape2p	0
<b>9</b>	LSP1p	3	<b>65</b>	VX_gain2[3]c	1	<b>Class V Bit</b>		
<b>10</b>	LSP1p	2	<b>Class II Bit</b>			<b>118</b>	VX_shape1[1]c	4
<b>11</b>	LSP1p	1	<b>66</b>	SE_shape4p	8	<b>119</b>	VX_shape1[1]c	3
<b>12</b>	LSP1p	0	<b>67</b>	SE_shape4p	7	<b>120</b>	VX_shape1[1]c	2
<b>13</b>	Pitchp	6	<b>68</b>	SE_shape4p	6	<b>121</b>	VX_shape1[1]c	1
<b>14</b>	Pitchp	5	<b>69</b>	SE_shape4p	5	<b>Class VI Bit</b>		
<b>15</b>	Pitchp	4	<b>70</b>	SE_shape4p	4	<b>122</b>	VX_shape1[1]c	0
<b>16</b>	Pitchp	3	<b>71</b>	SE_shape4p	3	<b>123</b>	VX_shape2[0]c	4
<b>17</b>	Pitchp	2	<b>72</b>	SE_shape4p	2	<b>124</b>	VX_shape2[0]c	3
<b>18</b>	Pitchp	1	<b>73</b>	SE_shape4p	1	<b>125</b>	VX_shape2[0]c	2
<b>19</b>	LSP2p	6	<b>74</b>	SE_shape4p	0	<b>Class VII Bit</b>		
<b>20</b>	LSP2p	5	<b>75</b>	SE_shape5p	7	<b>126</b>	LSP2p	2
<b>21</b>	LSP2p	4	<b>76</b>	SE_shape5p	6	<b>127</b>	LSP2p	1
<b>22</b>	LSP2p	3	<b>77</b>	SE_shape5p	5	<b>128</b>	LSP2p	0
<b>23</b>	SE_shape3p	6	<b>78</b>	SE_shape5p	4	<b>129</b>	LSP3p	3
<b>24</b>	SE_shape3p	5	<b>79</b>	SE_shape5p	3	<b>130</b>	LSP3p	2
<b>25</b>	SE_shape3p	4	<b>80</b>	SE_shape5p	2	<b>131</b>	LSP3p	1
<b>26</b>	SE_shape3p	3	<b>81</b>	SE_shape5p	1	<b>132</b>	LSP3p	0
<b>27</b>	SE_shape3p	2	<b>82</b>	SE_shape5p	0	<b>133</b>	LSP5p	6
<b>28</b>	LSP3p	4	<b>83</b>	SE_shape6p	4	<b>134</b>	LSP5p	5
<b>29</b>	LSP5p	7	<b>84</b>	SE_shape6p	3	<b>135</b>	LSP5p	4
<b>30</b>	SE_shape4p	9	<b>85</b>	SE_shape6p	2	<b>136</b>	LSP5p	3
<b>31</b>	SE_shape5p	8	<b>86</b>	SE_shape6p	1	<b>137</b>	LSP5p	2
<b>32</b>	SE_shape6p	5	<b>87</b>	SE_shape6p	0	<b>138</b>	LSP5p	1
<b>33</b>	VUVc	1	<b>88</b>	VX_gain2[3]c	0	<b>139</b>	LSP5p	0
<b>34</b>	VUVc	0	<b>89</b>	LSP2c	2	<b>140</b>	Pitchp	0
<b>35</b>	LSP4c	0	<b>90</b>	LSP2c	1	<b>141</b>	SE_shape3p	1
<b>36</b>	VX_gain1[0]c	3	<b>91</b>	LSP2c	0	<b>142</b>	SE_shape3p	0
<b>37</b>	VX_gain1[0]c	2	<b>92</b>	LSP3c	3	<b>143</b>	VX_shape2[0]c	1
<b>38</b>	VX_gain1[0]c	1	<b>93</b>	LSP3c	2	<b>144</b>	VX_shape2[0]c	0
<b>39</b>	VX_gain1[0]c	0	<b>94</b>	LSP3c	1	<b>145</b>	VX_shape2[1]c	4
<b>40</b>	VX_gain1[1]c	3	<b>95</b>	LSP3c	0	<b>146</b>	VX_shape2[1]c	3
<b>41</b>	VX_gain1[1]c	2	<b>96</b>	LSP5c	6	<b>147</b>	VX_shape2[1]c	2
<b>42</b>	VX_gain1[1]c	1	<b>97</b>	LSP5c	5	<b>148</b>	VX_shape2[1]c	1
<b>43</b>	VX_gain1[1]c	0	<b>98</b>	LSP5c	4	<b>149</b>	VX_shape2[1]c	0

44	LSP1c	4	99	LSP5c	3	150	VX_shape2[2]c	4
45	LSP1c	3	100	LSP5c	2	151	VX_shape2[2]c	3
46	LSP1c	2	101	LSP5c	1	152	VX_shape2[2]c	2
47	LSP1c	1	102	LSP5c	0	153	VX_shape2[2]c	1
48	LSP1c	0	103	VX_shape1[0]c	5	154	VX_shape2[2]c	0
49	LSP2c	6	104	VX_shape1[0]c	4	155	VX_shape2[3]c	4
50	LSP2c	5	105	VX_shape1[0]c	3	156	VX_shape2[3]c	3
51	LSP2c	4	106	VX_shape1[0]c	2	157	VX_shape2[3]c	2
52	LSP2c	3	107	VX_shape1[0]c	1	158	VX_shape2[3]c	1
53	LSP3c	4	108	VX_shape1[0]c	0	159	VX_shape2[3]c	0
54	LSP5c	7	109	VX_shape1[1]c	5			

Table 1. B.14 – Bit Order for 4kbit/s(unvoiced frame – voiced frame)

unvoiced frame – voiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class 1bit</b>			<b>55</b>	LSP2c	3	<b>Class III Bit</b>		
0	VUVp	1	56	SE_shape3c	6	110	VX_shape1[1]p	4
1	VUVp	0	57	SE_shape3c	5	111	VX_shape1[1]p	3
2	LSP4p	0	58	SE_shape3c	4	112	VX_shape1[1]p	2
3	VX_gain1[0]p	3	59	SE_shape3c	3	113	VX_shape1[1]p	1
4	VX_gain1[0]p	2	60	SE_shape3c	2	<b>Class IV Bit</b>		
5	VX_gain1[0]p	1	61	LSP3c	4	114	VX_shape1[1]p	0
6	VX_gain1[0]p	0	62	LSP5c	7	115	VX_shape2[0]p	4
7	VX_gain1[1]p	3	63	SE_shape4c	9	116	VX_shape2[0]p	3
8	VX_gain1[1]p	2	64	SE_shape5c	8	117	VX_shape2[0]p	2
9	VX_gain1[1]p	1	65	SE_shape6c	5	<b>Class V Bit</b>		
10	VX_gain1[1]p	0	<b>Class II Bit</b>			118	SE_shape1c	3
11	LSP1p	4	66	VX_gain2[3]p	0	119	SE_shape1c	2
12	LSP1p	3	67	LSP2p	2	120	SE_shape1c	1
13	LSP1p	2	68	LSP2p	1	121	SE_shape1c	0
14	LSP1p	1	69	LSP2p	0	<b>Class VI Bit</b>		
15	LSP1p	0	70	LSP3p	3	122	SE_shape2c	3
16	LSP2p	6	71	LSP3p	2	123	SE_shape2c	2
17	LSP2p	5	72	LSP3p	1	124	SE_shape2c	1
18	LSP2p	4	73	LSP3p	0	125	SE_shape2c	0
19	LSP2p	3	74	LSP5p	6	<b>Class VII Bit</b>		
20	LSP3p	4	75	LSP5p	5	126	VX_shape2[0]p	1
21	LSP5p	7	76	LSP5p	4	127	VX_shape2[0]p	0
22	VX_gain2[0]p	2	77	LSP5p	3	128	VX_shape2[1]p	4
23	VX_gain2[0]p	1	78	LSP5p	2	129	VX_shape2[1]p	3
24	VX_gain2[0]p	0	79	LSP5p	1	130	VX_shape2[1]p	2
25	VX_gain2[1]p	2	80	LSP5p	0	131	VX_shape2[1]p	1
26	VX_gain2[1]p	1	81	VX_shape1[0]p	5	132	VX_shape2[1]p	0
27	VX_gain2[1]p	0	82	VX_shape1[0]p	4	133	VX_shape2[2]p	4
28	VX_gain2[2]p	2	83	VX_shape1[0]p	3	134	VX_shape2[2]p	3
29	VX_gain2[2]p	1	84	VX_shape1[0]p	2	135	VX_shape2[2]p	2
30	VX_gain2[2]p	0	85	VX_shape1[0]p	1	136	VX_shape2[2]p	1

31	VX_gain2[3]p	2	86	VX_shape1[0]p	0	137	VX_shape2[2]p	0
32	VX_gain2[3]p	1	87	VX_shape1[1]p	5	138	VX_shape2[3]p	4
33	VUVc	1	88	SE_shape4c	8	139	VX_shape2[3]p	3
34	VUVc	0	89	SE_shape4c	7	140	VX_shape2[3]p	2
35	LSP4c	0	90	SE_shape4c	6	141	VX_shape2[3]p	1
36	SE_gainc	4	91	SE_shape4c	5	142	VX_shape2[3]p	0
37	SE_gainc	3	92	SE_shape4c	4	143	LSP2c	2
38	SE_gainc	2	93	SE_shape4c	3	144	LSP2c	1
39	SE_gainc	1	94	SE_shape4c	2	145	LSP2c	0
40	SE_gainc	0	95	SE_shape4c	1	146	LSP3c	3
41	LSP1c	4	96	SE_shape4c	0	147	LSP3c	2
42	LSP1c	3	97	SE_shape5c	7	148	LSP3c	1
43	LSP1c	2	98	SE_shape5c	6	149	LSP3c	0
44	LSP1c	1	99	SE_shape5c	5	150	LSP5c	6
45	LSP1c	0	100	SE_shape5c	4	151	LSP5c	5
46	Pitchc	6	101	SE_shape5c	3	152	LSP5c	4
47	Pitchc	5	102	SE_shape5c	2	153	LSP5c	3
48	Pitchc	4	103	SE_shape5c	1	154	LSP5c	2
49	Pitchc	3	104	SE_shape5c	0	155	LSP5c	1
50	Pitchc	2	105	SE_shape6c	4	156	LSP5c	0
51	Pitchc	1	106	SE_shape6c	3	157	Pitchc	0
52	LSP2c	6	107	SE_shape6c	2	158	SE_shape3c	1
53	LSP2c	5	108	SE_shape6c	1	159	SE_shape3c	0
54	LSP2c	4	109	SE_shape6c	0			

Table 1. B.15 – Bit Order for 4kbit/s(unvoiced frame – unvoiced frame)

unvoiced frame – unvoiced frame								
No.	Item	Bit	No.	Item	Bit	No.	Item	Bit
<b>Class 1bit</b>			55	VX_gain2[0]c	2	<b>Class III Bit</b>		
0	VUVp	1	56	VX_gain2[0]c	1	110	VX_shape1[1]p	4
1	VUVp	0	57	VX_gain2[0]c	0	111	VX_shape1[1]p	3
2	LSP4p	0	58	VX_gain2[1]c	2	112	VX_shape1[1]p	2
3	VX_gain1[0]p	3	59	VX_gain2[1]c	1	113	VX_shape1[1]p	1
4	VX_gain1[0]p	2	60	VX_gain2[1]c	0	<b>Class IV Bit</b>		
5	VX_gain1[0]p	1	61	VX_gain2[2]c	2	114	VX_shape1[1]p	0
6	VX_gain1[0]p	0	62	VX_gain2[2]c	1	115	VX_shape2[0]p	4
7	VX_gain1[1]p	3	63	VX_gain2[2]c	0	116	VX_shape2[0]p	3
8	VX_gain1[1]p	2	64	VX_gain2[3]c	2	117	VX_shape2[0]p	2
9	VX_gain1[1]p	1	65	VX_gain2[3]c	1	<b>Class V Bit</b>		
10	VX_gain1[1]p	0	<b>Class II Bit</b>			118	VX_shape1[1]c	4
11	LSP1p	4	66	VX_gain2[3]p	0	119	VX_shape1[1]c	3
12	LSP1p	3	67	LSP2p	2	120	VX_shape1[1]c	2
13	LSP1p	2	68	LSP2p	1	121	VX_shape1[1]c	1
14	LSP1p	1	69	LSP2p	0	<b>Class VI Bit</b>		
15	LSP1p	0	70	LSP3p	3	122	VX_shape1[1]c	0
16	LSP2p	6	71	LSP3p	2	123	VX_shape2[0]c	4
17	LSP2p	5	72	LSP3p	1	124	VX_shape2[0]c	3

18	LSP2p	4	73	LSP3p	0	125	VX_shape2[0]c	2
19	LSP2p	3	74	LSP5p	6	<b>Class VII Bit</b>		
20	LSP3p	4	75	LSP5p	5	126	VX_shape2[0]p	1
21	LSP5p	7	76	LSP5p	4	127	VX_shape2[0]p	0
22	VX_gain2[0]p	2	77	LSP5p	3	128	VX_shape2[1]p	4
23	VX_gain2[0]p	1	78	LSP5p	2	129	VX_shape2[1]p	3
24	VX_gain2[0]p	0	79	LSP5p	1	130	VX_shape2[1]p	2
25	VX_gain2[1]p	2	80	LSP5p	0	131	VX_shape2[1]p	1
26	VX_gain2[1]p	1	81	VX_shape1[0]p	5	132	VX_shape2[1]p	0
27	VX_gain2[1]p	0	82	VX_shape1[0]p	4	133	VX_shape2[2]p	4
28	VX_gain2[2]p	2	83	VX_shape1[0]p	3	134	VX_shape2[2]p	3
29	VX_gain2[2]p	1	84	VX_shape1[0]p	2	135	VX_shape2[2]p	2
30	VX_gain2[2]p	0	85	VX_shape1[0]p	1	136	VX_shape2[2]p	1
31	VX_gain2[3]p	2	86	VX_shape1[0]p	0	137	VX_shape2[2]p	0
32	VX_gain2[3]p	1	87	VX_shape1[1]p	5	138	VX_shape2[3]p	4
33	VUVc	1	88	VX_gain2[3]c	0	139	VX_shape2[3]p	3
34	VUVc	0	89	LSP2c	2	140	VX_shape2[3]p	2
35	LSP4c	0	90	LSP2c	1	141	VX_shape2[3]p	1
36	VX_gain1[0]c	3	91	LSP2c	0	142	VX_shape2[3]p	0
37	VX_gain1[0]c	2	92	LSP3c	3	143	VX_shape2[0]c	1
38	VX_gain1[0]c	1	93	LSP3c	2	144	VX_shape2[0]c	0
39	VX_gain1[0]c	0	94	LSP3c	1	145	VX_shape2[1]c	4
40	VX_gain1[1]c	3	95	LSP3c	0	146	VX_shape2[1]c	3
41	VX_gain1[1]c	2	96	LSP5c	6	147	VX_shape2[1]c	2
42	VX_gain1[1]c	1	97	LSP5c	5	148	VX_shape2[1]c	1
43	VX_gain1[1]c	0	98	LSP5c	4	149	VX_shape2[1]c	0
44	LSP1c	4	99	LSP5c	3	150	VX_shape2[2]c	4
45	LSP1c	3	100	LSP5c	2	151	VX_shape2[2]c	3
46	LSP1c	2	101	LSP5c	1	152	VX_shape2[2]c	2
47	LSP1c	1	102	LSP5c	0	153	VX_shape2[2]c	1
48	LSP1c	0	103	VX_shape1[0]c	5	154	VX_shape2[2]c	0
49	LSP2c	6	104	VX_shape1[0]c	4	155	VX_shape2[3]c	4
50	LSP2c	5	105	VX_shape1[0]c	3	156	VX_shape2[3]c	3
51	LSP2c	4	106	VX_shape1[0]c	2	157	VX_shape2[3]c	2
52	LSP2c	3	107	VX_shape1[0]c	1	158	VX_shape2[3]c	1
53	LSP3c	4	108	VX_shape1[0]c	0	159	VX_shape2[3]c	0
54	LSP5c	7	109	VX_shape1[1]c	5			

### 1.B.4.3 EP tool setting

#### 1.B.4.3.1 Bit assignment

The Table 1. B.16 below shows an example bit assignment for the use of the EP tool. In this table, bit assignments for both of the 2kbit/s and 4kbit/s source coder are described.



Table 1. B.16 – The bit assignment for the use of the EP tool

	2kbit/s source coder	4kbit/s Source Coder
<b>Class I</b>		
Source coder bits	44	66
CRC parity	6	6
Code Rate	8/16	8/16
Class I total	100	144
<b>Class II</b>		
Source coder bits	4	44
CRC parity	1	6
Code Rate	8/8	8/8
Class II total	5	50
<b>Class III</b>		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class III total	5	5
<b>Class IV</b>		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class IV total	5	5
<b>Class V</b>		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class V total	5	5
<b>Class VI</b>		
Source coder bits	20	4
CRC parity	0	1
Code Rate	8/8	8/8
Class VI total	20	5
<b>Class VII</b>		
Source coder bits		34
CRC parity		0
Code Rate		8/8
Class VII total		34
Total Bit of All Classes	140	248
Bitrate	3.5 kbit/s	6.2 kbit/s

Class I:

CRC covers all the Class I bits, and Class I bits including CRC are protected by convolutional coding.

Class II-V(2kbit/s), II-VI(4kbit/s):

At least one CRC bits cover the source coder bits of these classes.

Class VI(2kbit/s), VII(4kbit/s) :

The source coder bits are not checked by CRC nor protected by any error correction scheme.

#### 1.B.4.4 Error concealment

When CRC error is detected, error concealment processing (bad frame masking) is carried out. An example of concealment method is described below.

A frame masking state of the current frame is updated based on the decoded CRC result of Class I. The state transition diagram is shown in Figure 1.B.2. The initial state is state=0. The arrow with a letter "1" denotes the transition for a bad frame, and that with a letter "0" a good frame.

**1.B.4.4.1 Parameter replacemet**

According to the state value, the following parameter replacement is done. In error free condition, state value becomes 0, and received source coder bits are used without any concealment processing.

**1.B.4.4.1.1 LSP parameters**

At state=1..6, LSP parameters are replaced with those of previous ones.

When state=7, If LSP4=0 (LSP quantization mode without inter-frame prediction), then LSP parameters are calculated from all LSP indexes received in the current frame. If LSP4=1 (LSP quantization mode with inter-frame coding), then LSP parameters are calculated with the following method.

In this mode, LSP parameters from LSP1 index are interpolated with the previous LSPs.

$$LSP_{base}(n) = p \cdot LSP_{prev}(n) + (1 - p)LSP_{1st}(n) \quad \text{for } n=1..10 \quad (1)$$

$LSP_{base}(n)$  is LSP parameters of the base layer,  $LSP_{prev}(n)$  is the previous LSPs,  $LSP_{1st}(n)$  is the decoded LSPs from the current LSP1 index, and  $p$  is the factor of interpolation.  $p$  is changed according to the number of previous CRC error frames of Class I bits as shown in Table 1. B.17. LSP indexes LSP2, LSP3 and LSP5 are not used and  $LSP_{base}(n)$  is used as current LSP parameters.

**Table 1. B.17 – p factor**

frame	p
0	0.7
1	0.6
2	0.5
3	0.4
4	0.3
5	0.2
6	0.1
>7	0.0

**1.B.4.4.1.2 Mute variable**

According to the “state” value, a variable “mute” is set to control output level of speech.

The “mute” value below is used.

In state=7, the average of 1.0 and “mute” value of the previous frame( = 0.5 ( 1.0 + previous “mute value” ) ) is used, but when this value is more than 0.8, “mute” value is replaced with 0.8.

**Table 1. B.18 – mute value**

State	mute
0	1.000
1	0.800
2	0.700
3	0.500
4	0.250
5	0.125
6	0.000
7	Average/0.800

#### 1.B.4.4.1.3 Replacement and gain control of “voiced” parameters

In state=1..6, spectrum parameter SE\_shape1, SE\_shape2, spectrum gain parameter SE\_gain, spectrum parameter for 4kbit/s codec SE\_shape3 .. SE\_shape6 are replaced with corresponding parameters of the previous frame. Also, to control volume of output speech, harmonic magnitude parameters of LPC residual signal “ $Am[0..127]$ ” is gain controlled as shown in Eq.(1). In the equation,  $Am_{(org)}[i]$  is computed from the received spectrum parameters from the latest error free frame.

$$Am[i] = mute * Am_{(org)}[i] \quad \text{for } i=0..127 \quad (1)$$

If previous frame is unvoiced and current state is state=7, Eq.(1) is replaced with Eq.(2).

$$Am[i] = 0.6 * mute * Am_{(org)}[i] \quad \text{for } i=0..127 \quad (2)$$

As described before, SE\_shape1 and SE\_shape2 are individually protected by 1 bit CRC. In state=0 or 7, when CRC errors of these classes are detected at the same time, the quantized harmonic magnitudes with fixed dimension  $Am_{qnt}[1..44]$  are gain suppressed as shown in Eq.(3).

$$Am_{qnt}[i] = s[i] * Am_{qnt(org)}[i] \quad \text{for } i=1..44 \quad (3)$$

$s[i]$  is the factor for the gain suppression.

**Table 1. B.19 – factor for gain suppression ‘s[0..44]’**

$i$	1	2	3	4	5	6	7..44
$s[i]$	0.10	0.25	0.40	0.55	0.70	0.85	1.00

At 4kbit/s, SE\_shape4, SE\_shape5, and SE\_shape6 are checked by CRC as Class II bits. When CRC error is detected, the spectrum parameter of the enhancement layer is not used.

#### 1.B.4.4.1.4 Replacement and gain control of “unvoiced” parameters.

In state=1..6, stochastic codebook gain parameter VX\_gain1[0], VX\_gain1[1] are replaced with the VX\_gain1[1] from the latest error free frame. Also stochastic codebook gain parameter for 4kbit/s codec VX\_gain2[0]..VX\_gain2[3] are replaced with the VX\_gain2[3] from the latest error free frame.

Stochastic codebook shape parameter VX\_shape1[0], VX\_shape1[1], and stochastic codebook shape parameter for 4kbit/s codec are generated from randomly generated index values.

Also, to control volume of output speech, LPC residual signal  $res[0..159]$  is gain controlled as shown in Eq.(4). In the equation,  $res_{(org)}[i]$  is computed from stochastic codebook parameters.

$$res[i] = mute * res_{(org)}[i] \quad (0 \leq i \leq 159) \quad (4)$$

1.B.4.4.1.5 Frame Masking State Transitions

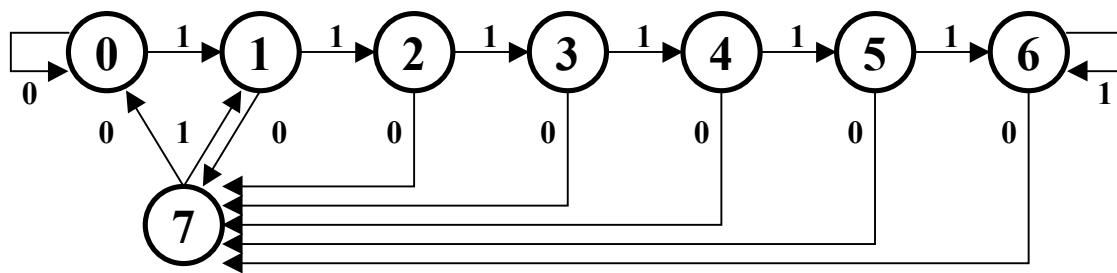


Figure 1.B.2 – Frame Masking State Transitions

## Annex 1.C (informative)

### Patent statements

The International Organization for Standardization (ISO) and the International Electrotechnical Commission (IEC) draw attention to the fact that it is claimed that compliance with this part of ISO/IEC 14496 may involve the use of patents, as indicated in the following table.

ISO and IEC take no position concerning the evidence, validity and scope of these patent rights.

- Notes:
1. The presence of a name of a company in the list below indicates that a patent statement has been received from that company
  2. The presence of a cross indicates that the statement identifies the part of the MPEG-4 standard to which the statement applies
  3. No cross in a line indicates that the statement does not identify which part of the standard the statement applies

	<b>Company</b>	<b>Part 1</b>	<b>Part 2</b>	<b>Part 3</b>	<b>Part 5</b>	<b>Part 6</b>
1.	Alcatel	X	X	X	X	X
2.	Apple	X			X	
3.	AT&T	X	X	X		
4.	BBC	X	X	X	X	X
5.	Bosch	X	X	X	X	X
6.	British Telecommunications	X	X	X	X	X
7.	Canon	X	X	X	X	X
8.	CCETT	X	X	X	X	X
9.	Columbia Innovation Enterprise	X	X	X	X	X
10.	Columbia University	X	X	X	X	X
11.	Creative	X		X	X	
12.	CSELT			X		
13.	DemoGraFX	X	X	X	X	X
14.	DirectTV	X	X	X		
15.	Dolby	X	X	X	X	X
16.	EPFL	X	X	X	X	
17.	ETRI	X	X	X	X	X
18.	France Telecom	X	X	X	X	X
19.	Fraunhofer	X	X	X	X	X
20.	Fujitsu	X	X	X	X	X
21.	GC Technology Corporation	X	X	X		
22.	General Instrument		X		X	
23.	Hitachi	X	X	X	X	X
24.	Hyundai	X	X	X	X	X
25.	IBM	X	X	X	X	X
26.	Institut fuer Rundfunktechnik	X	X	X		X
27.	Intertrust					
28.	JVC	X	X	X	X	X
29.	KDD Corporation	X	X			
30.	KPN	X	X	X	X	X
31.	LG Semicon					
32.	Lucent					
33.	Matsushita Electric Industrial Co., Ltd.	X	X	X	X	X
34.	Microsoft	X	X	X	X	X
35.	MIT					
36.	Mitsubishi	X	X	X	X	X
37.	Motorola		X		X	
38.	NEC	X	X	X	X	X
39.	NHK	X	X	X	X	X
40.	Nokia	X	X	X	X	

**ISO/IEC 14496-3:2001(E)**

41.	NTT	x	x	x	x	x
42.	NTT Mobile Communication Networks			x		
43.	OKI	x	x	x	x	x
44.	Optibase	x			x	
45.	Philips	x	x	x	x	x
46.	PictureTel Corporation		x		x	
47.	Rockwell	x	x	x	x	x
48.	Samsung	x	x	x	x	
49.	Sarnoff	x	x	x	x	x
50.	Scientific Atlanta	x	x	x	x	x
51.	Sharp	x	x	x	x	x
52.	Siemens	x	x	x	x	x
53.	Sony	x	x	x	x	x
54.	Sun	x				
55.	Telenor	x	x	x	x	x
56.	Teltec DCU		x		x	
57.	Texas Instruments					
58.	Thomson	x	x	x	x	x
59.	Toshiba		x			
60.	Unisearch Ltd.		x		x	
61.	Vector Vision		x			

The holders of these patent rights have assured ISO and IEC that they are willing to negotiate licences under reasonable and non-discriminatory terms and conditions with applicants throughout the world. In this respect, the statements of the holders of these patent rights are registered with ISO and IEC.

Attention is drawn to the possibility that some of the elements of this part of ISO/IEC 14496 may be the subject of patent rights other than those identified above. ISO and IEC shall not be held responsible for identifying any or all such patent rights.

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## Subpart 2: Speech coding - HVXC

### 2.1 Scope

MPEG-4 parametric speech coding uses Harmonic Vector eXcitation Coding (HVXC) algorithm, where harmonic coding of LPC residual signals for voiced segments and Vector eXcitation Coding (VXC) for unvoiced segments are employed. HVXC allows coding of speech signals at 2.0 kbps and 4.0 kbps with a scalable scheme, where 2.0 kbps decoding is possible not only using the 2.0 kbps bit-stream but also using a 4.0 kbps bit-stream. HVXC also provides variable bit rate coding where a typical average bit-rate is around 1.2-1.7 kbit/s. Independent change of speed and pitch during decoding is possible, which is a powerful functionality for fast data base search. The frame length is 20 ms, and one of four different algorithmic delays, 33.5 ms, 36ms, 53.5 ms, 56 ms can be selected.

Furthermore as an extension of HVXC, ER\_HVXC object type offers error resilient syntax and the 4.0kbit/s variable bitrate mode.

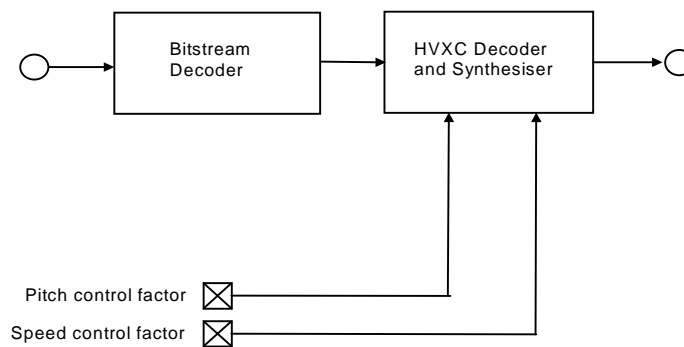


Figure 2.1 - Block diagram of the parametric speech decoder

### 2.2 Definitions

2.2.1. **DFT:** Discrete Fourier Transform.

2.2.2. **dimension conversion:** A method to convert a dimension of a vector by a combination of low pass filtering and linear interpolation.

2.2.3. **excitation:** A time domain signal which excites the LPC synthesis filter.

2.2.4. **FFT:** Fast Fourier Transform.

2.2.5. **fundamental frequency:** A parameter which represents signal periodicity in frequency domain.

2.2.6. **harmonics:** Samples of frequency spectrum at multiples of the fundamental frequency.

2.2.7. **harmonic magnitude:** Magnitude of each harmonic.

2.2.8. **harmonic synthesis:** A method to obtain a periodic excitation from harmonic magnitudes.

2.2.9. **HVXC:** Harmonic Vector eXcitation Coding (parametric speech coding)



**2.2.10. IFFT:** Inverse Fast Fourier Transform.

**2.2.11. initial phase:** A phase value at the onset of voiced signal in harmonic synthesis.

**2.2.12. interframe prediction:** A method to predict a value in the current frame from values in the previous frames. Interframe prediction is used in VQ of LSP.

**2.2.13. LPC:** Linear Predictive Coding.

**2.2.14. LPC synthesis filter:** An IIR filter whose coefficients are LPC coefficients. This filter models the time varying vocal tract.

**2.2.15. LPC residual signal:** A signal filtered by the LPC inverse filter, which has a flattened frequency spectrum.

**2.2.16. LSP:** Line Spectral Pairs.

**2.2.17. mixed voiced frame:** A speech segment which has both voiced and unvoiced components.

**2.2.18. pitch:** A parameter which represents signal periodicity in the time domain. It is expressed in terms of the number of samples.

**2.2.19. pitch control:** A functionality to control the pitch of the synthesized speech signal without changing its speed.

**2.2.20. postfilter:** A filter to enhance the perceptual quality of the synthesized speech signal.

**2.2.21. sinusoidal synthesis:** A method to obtain a time domain waveform by a sum of amplitude modulated sinusoidal waveforms.

**2.2.22. spectral envelope:** A set of harmonic magnitudes.

**2.2.23. speed control:** A functionality to control the speed of the synthesized speech signal without changing its pitch or phonemes.

**2.2.24. SQ:** Scalar Quantization.

**2.2.25. unvoiced frame:** A speech segment which looks like random noise with no periodicity.

**2.2.26. variable bit rate:** The number of bits representing a coded frame varies frame by frame over time.

**2.2.27. voiced frame:** A speech segment which has periodicity and a relatively high energy.

**2.2.28. VQ:** Vector Quantization.

**2.2.29. V/UV decision:** Decision whether the current frame is voiced or unvoiced or mixed voiced.

**2.2.30. VXC:** Vector eXcitation Coding. It is also called CELP (Coded Excitation Linear Prediction). In HVXC, no adaptive codebook is used.

**2.2.31. white Gaussian noise:** A noise sequence which has a Gaussian distribution.

A general glossary and list of symbols and abbreviations is located in Section 1.

## 2.3 Bitstream syntax

An MPEG-4 Natural Audio Object HVXC/ER\_HVXC object type is transmitted in one or two Elementary Streams: The base layer stream and an optional enhancement layer stream.

When the HVXC tool is used with an error protection tool, such as an MPEG-4 EP tool, the bit order arranged in accordance with the error sensitivity should be used. The HVXC with the error resilient syntax and 4.0 kbit/s variable bitrate mode described in subclauses 2.3.3 and 2.3.4 are called ER\_HVXC.

The bitstream syntax is described in pseudo-C code.

### 2.3.1 Decoder configuration (HvxcSpecificConfig)

The decoder configuration information for HVXC object type is transmitted in the DecoderConfigDescriptor() of the base layer and the optional enhancement layer Elementary Stream (see Section 1, subclause 1.6).

The following HvxcSpecificConfig() is required:

```
HvxcSpecificConfig ( ) {
    isBaseLayer          1  uimsbf
    if (isBaseLayer) {
        HVXCconfig()
    }
}
```

HVXC object type provides unscalable modes and a 2kbit/s base layer plus a 2kbit/s enhancement layer scalable mode. In this scalable mode the basic layer configuration must be as follows:

- HVXCvarMode = 0      HVXC fixed bit rate
- HVXCrateMode = 0    HVXC 2kbps
- isBaseLayer=1      base layer

**Table 2.1 - Syntax of HVXCconfig()**

Syntax	No. of bits	Mnemonic
HVXCconfig() {		
<b>HVXCvarMode</b>	<b>1</b>	<b>uimsbf</b>
<b>HVXCrateMode</b>	<b>2</b>	<b>uimsbf</b>
<b>extensionFlag</b>	<b>1</b>	<b>uimsbf</b>
if (extensionFlag) {		
< to be defined in MPEG-4 Version 2 >		
}		
}		

**Table 2.2 - HVXCvarMode**

HVXCvarMode	Description
0	HVXC fixed bit rate
1	HVXC variable bit rate

**Table 2.3 - HVXCrateMode**

HVXCrateMode	HVXCrate	Description
0	2000	HVXC 2 kbit/s
1	4000	HVXC 4 kbit/s
2	3700	HVXC 3.7 kbit/s
3	(reserved)	

**Table 2.4 - HVXC constants**

NUM_SUBF1	NUM_SUBF2
2	4

### 2.3.2 Bitstream frame (*alPduPayload*)

The dynamic data for HVXC object type is transmitted as AL-PDU payload in the base layer and the optional enhancement layer Elementary Stream (see Section 1, subclause 1.6).

#### HVXC Base Layer -- Access Unit payload

```
alPduPayload {
    HVXCframe();
}
```

#### HVXC Enhancement Layer -- Access Unit payload

To parse and decode the HVXC enhancement layer, information decoded from the HVXC base layer is required.

```
alPduPayload {
    HVXCenhaFrame();
}
```

**Table 2.5 - Syntax of HVXCframe()**

Syntax	No. of bits	Mnemonic
<pre>HVXCframe() {     if (HVXCvarMode ==0) {         HVXCfixframe(HVXCrate)     }     else {         HVXCvarframe()     } }</pre>		

#### 2.3.2.1 HVXC bitstream frame

Table 2.6 - Syntax of HVXCfixframe(rate)

Syntax	No. of bits	Mnemonic
<pre> HVXCfixframe(rate) {   if (rate &gt;= 2000){     idLsp1()     idVUV()     idExc1()   }   if (rate &gt;= 3700){     idLsp2()     idExc2(rate)   } } </pre>		

Table 2.7 - Syntax of HVXCenhaFrame()

Syntax	No. of bits	Mnemonic
<pre> HVXCenhaFrame() {   idLsp2()   idExc2(4000) } </pre>		

Table 2.8 - Syntax of idLsp1()

Syntax	No. of bits	Mnemonic
<pre> idLsp1() {   LSP1   LSP2   LSP3   LSP4 } </pre>	<p>5</p> <p>7</p> <p>5</p> <p>1</p>	<p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p>

Table 2.9 - Syntax of idLsp2()

Syntax	No. of bits	Mnemonic
<pre> idLsp2() {   LSP5 } </pre>	8	uimsbf

Table 2.10 - Syntax of idVUV()

Syntax	No. of bits	Mnemonic
<pre> idVUV() {   VUV } </pre>	2	uimsbf

Table 2.11 - VUV (for fixed bit rate mode)

VUV	Description
0	Unvoiced Speech
1	Mixed Voiced Speech-1
2	Mixed Voiced Speech-2
3	Voiced Speech

Table 2.12 - Syntax of idExc1()

Syntax	No. of bits	Mnemonic
idExc1() { if (VUV!=0){ <b>Pitch</b> <b>SE_shape1</b> <b>SE_shape2</b> <b>SE_gain</b> } else{ for (sf_num=0;sf_num<NUM_SUBF1;sf_num++){ <b>VX_shape1 [sf_num]</b> <b>VX_gain1 [sf_num]</b> } } }	7 4 4 5  6 4	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b>

Table 2.13 - Syntax of idExc2(rate)

Syntax	No. of bits	Mnemonic
idExc2(rate) { if (VUV!=0){ <b>SE_shape3</b> <b>SE_shape4</b> <b>SE_shape5</b> if (rate>=4000){ <b>SE_shape6</b> } } else{ for (sf_num=0;sf_num<NUM_SUBF2-1;sf_num++){ <b>VX_shape2[sf_num]</b> <b>VX_gain2[sf_num]</b> } if (rate>=4000){ <b>VX_shape2[3]</b> <b>VX_gain2[3]</b> } } }	7 10 9 6  5 3  5 3	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b>

idLsp1(), idExc1(), idVUV() are treated as base layer in case of scalable mode.

idLsp2(), idExc2() are treated as enhancement layer in case of scalable mode.

**Table 2.14 - Syntax of HVXCvarframe()**

Syntax	No. of bits	Mnemonic
HVXCvarframe() { idvarVUV() idvarLsp1() idvarExc1() }		

**Table 2.15 - Syntax of idvarVUV()**

Syntax	No. of bits	Mnemonic
idvarVUV() { <b>VUV</b> }	<b>2</b>	<b>uimsbf</b>

**Table 2.16 - VUV (for variable bit rate mode)**

VUV	Description
0	Unvoiced Speech
1	Background Noise
2	Mixed Voiced Speech
3	Voiced Speech

**Table 2.17 - Syntax of idvarLsp1()**

Syntax	No. of bits	Mnemonic
idvarLsp1() { if (VUV != 1){ <b>LSP1</b> <b>LSP2</b> <b>LSP3</b> <b>LSP4</b> } }	<b>5</b> <b>7</b> <b>5</b> <b>1</b>	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

**Table 2.18 - Syntax of idvarExc1()**

Syntax	No. of bits	Mnemonic
idvarExc1() { if (VUV != 1){ if (VUV != 0){ <b>Pitch</b> <b>SE_Shape1</b> } } }	<b>7</b> <b>4</b>	<b>uimsbf</b> <b>uimsbf</b>

<b>SE_Shape2</b>	<b>4</b>	<b>uimsbf</b>
<b>SE_Gain</b>	<b>5</b>	<b>uimsbf</b>
<code>}</code>		
<code>else{</code>		
<code>  for (sf_num=0;sf_num&lt;NUM_SUBF1;sf_num++){</code>		
<code>    <b>VX_gain1[sf_num]</b></code>	<b>4</b>	<b>uimsbf</b>
<code>  }</code>		
<code>}</code>		
<code>}</code>		
<code>}</code>		

### 2.3.3 Decoder configuration (ErrorResilientHvxcSpecificConfig)

The decoder configuration information for ER\_HVXC object type is transmitted in the DecoderConfigDescriptor() of the base layer and the optional enhancement layer Elementary Stream.

The following ErrorResilientHvxcSpecificConfig() is required:

```
ErrorResilientHvxcSpecificConfig() {
  isBaseLayer          1 uimsbf
  if ( isBaseLayer ) {
    ErHVXCconfig();
  }
}
```

ER\_HVXC object type provides unscalable modes and scalable modes. In the scalable modes the base layer configuration must be as follows:

```
HVXCrateMode = 0   ER_HVXC 2kbit/s
isBaseLayer=1     base layer
```

**Table 2.19 – Syntax of ErHvxcConfig()**

Syntax	No. of bits	Mnemonic
ErHVXCconfig()		
{		
<b>HVXCvarMode;</b>	<b>1</b>	<b>uimsbf</b>
<b>HVXCrateMode;</b>	<b>2</b>	<b>uimsbf</b>
<b>extensionFlag;</b>	<b>1</b>	<b>uimsbf</b>
if(extensionFlag) {		
<b>var_ScalableFlag;</b>	<b>1</b>	<b>uimsbf</b>
}		
}		

**Table 2.20- HVXCvarMode**

HVXCvarMode	Description
0	ER_HVXC fixed bitrate
1	ER_HVXC variable bitrate

**Table 2.21 - HVXCrateMode**

HVXCrateMode	HVXCrate	Description
0	2000	ER_HVXC 2 kbit/s
1	4000	ER_HVXC 4 kbit/s
2	3700	ER_HVXC 3.7 kbit/s
3 (reserved)		

**Table 2.22 – var\_ScalableFlag**

var_ScalableFlag	Description
0	ER_HVXC variable rate non-scalable mode
1	ER_HVXC variable rate scalable mode

**2.3.4 Bitstream frame (alPduPayload)**

The dynamic data for ER\_HVXC object type is transmitted as AL-PDU payload in the base layer and the optional enhancement layer Elementary Stream.

**ER\_HVXC Base Layer -- Access Unit payload**

```
alPduPayload {
    ErHVXCframe();
}
```

**ER\_HVXC Enhancement Layer -- Access Unit payload**

To parse and decode the ER\_HVXC enhancement layer, information decoded from the ER\_HVXC base layer is required.

```
alPduPayload {
    ErHVXCenhaFrame();
}
```

**Table 2.23 – Syntax of ErHVXCframe()**

Syntax	No. of bits	Mnemonic
ErHVXCframe() { if (HVXCvarMode ==0) { ErHVXCfixframe(HVXCrate); } else { ErHVXCvarframe(HVXCrate); } }		

**Table 2.24 – Syntax of ErHVXCenhaframe()**

Syntax	No. of bits	Mnemonic
ErHVXCenhaframe() { if (HVXCvarMode ==0) { ErHVXCenh_fixframe(); }		



```

    }
    else {
        ErHVXCenh_varframe();
    }
}

```

### 2.3.4.1 Bitstream syntax of the fixed rate mode

Table 2.25 – Syntax of ErHVXCfixframe()

Syntax	No. of bits	Mnemonic
ErHVXCfixframe(rate) { if (rate == 2000) { 2k_ESC0(); 2k_ESC1() 2k_ESC2(); 2k_ESC3(); } else if (rate >= 3700) { 4k_ESC0(rate); 4k_ESC1(rate) 4k_ESC2(); 4k_ESC3(); 4k_ESC4(rate); } }		

Table 2.26 – Syntax of 2k\_ESC0()

Syntax	No. of bits	Mnemonic
2k_ESC0() { <b>VUV, 1-0;</b> if (VUV!=0) { <b>LSP4, 0;</b> <b>SE_gain, 4-0;</b> <b>LSP1, 4-0;</b> <b>Pitch, 6-1;</b> <b>LSP2, 6;</b> <b>LSP3, 4;</b> <b>LSP2, 5;</b> } else { <b>LSP4, 0;</b> <b>VX_gain1[0], 3-0;</b> <b>VX_gain1[1], 3-0;</b> <b>LSP1, 4-0;</b> <b>LSP2, 6-3;</b> <b>LSP3, 4-3;</b> } }	  <b>2</b>  <b>1</b> <b>5</b> <b>5</b> <b>6</b> <b>1</b> <b>1</b> <b>1</b>   <b>1</b> <b>4</b> <b>4</b> <b>5</b> <b>4</b> <b>2</b>	  <b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>   <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 2.27 – Syntax of 2k\_ESC1()

Syntax	No. of bits	Mnemonic
2k_ESC1() { if (VUV!=0) { <b>SE_shape1, 3-0;</b> } else { <b>LSP2, 2-0;</b> <b>LSP3, 2;</b> } }	<b>4</b>  <b>3</b> <b>1</b>	<b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b>

Table 2.28 – Syntax of 2k\_ESC2()

Syntax	No. of bits	Mnemonic
2k_ESC2() { if (VUV!=0) { <b>SE_shape2, 3-0;</b> } else { <b>LSP3, 1-0;</b> <b>VX_shape1[0], 5-4;</b> } }	<b>4</b>  <b>2</b> <b>2</b>	<b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b>

Table 2.29 – Syntax of 2k\_ESC3()

Syntax	No. of bits	Mnemonic
2k_ESC3() { if (VUV!=0) { <b>LSP2, 4-0;</b> <b>LSP3, 3-0;</b> <b>Pitch, 0;</b> } else { <b>VX_shape1[0], 3-0;</b> <b>VX_shape1[1], 5-0;</b> } }	<b>5</b> <b>4</b> <b>1</b>  <b>4</b> <b>6</b>	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b>

Table 2.30 – Syntax of 4k\_ESC0()

Syntax	No. of bits	Mnemonic
4k_ESC0() { <b>VUV, 1-0;</b> if (VUV!=0) { <b>LSP4, 0;</b> <b>SE_gain, 4-0;</b> <b>LSP1, 4-0;</b> <b>Pitch, 6-1;</b> }	<b>2</b>  <b>1</b> <b>5</b> <b>5</b> <b>6</b>	<b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

<b>LSP2, 6-3;</b>	<b>4</b>	<b>uimsbf</b>
<b>SE_shape3, 6-2;</b>	<b>5</b>	<b>uimsbf</b>
<b>LSP3, 4;</b>	<b>1</b>	<b>uimsbf</b>
<b>LSP5, 7;</b>	<b>1</b>	<b>uimsbf</b>
<b>SE_shape4, 9;</b>	<b>1</b>	<b>uimsbf</b>
<b>SE_shape5, 8;</b>	<b>1</b>	<b>uimsbf</b>
if (rate>=4000) {		
<b>SE_shape6, 5;</b>	<b>1</b>	<b>uimsbf</b>
}		
}		
else {		
<b>LSP4, 0;</b>	<b>1</b>	<b>uimsbf</b>
<b>VX_gain1[0], 3-0;</b>	<b>4</b>	<b>uimsbf</b>
<b>VX_gain1[1], 3-0;</b>	<b>4</b>	<b>uimsbf</b>
<b>LSP1, 4-0;</b>	<b>5</b>	<b>uimsbf</b>
<b>LSP2, 6-3;</b>	<b>4</b>	<b>uimsbf</b>
<b>LSP3, 4;</b>	<b>1</b>	<b>uimsbf</b>
<b>LSP5, 7;</b>	<b>1</b>	<b>uimsbf</b>
<b>VX_gain2[0], 2-0;</b>	<b>3</b>	<b>uimsbf</b>
<b>VX_gain2[1], 2-0;</b>	<b>3</b>	<b>uimsbf</b>
<b>VX_gain2[2], 2-0;</b>	<b>3</b>	<b>uimsbf</b>
if (rate>=4000) {		
<b>VX_gain2[3], 2-1;</b>	<b>2</b>	<b>uimsbf</b>
}		
}		
}		
}		

Table 2.31 – Syntax of 4k\_ESC1()

Syntax	No. of bits	Mnemonic
4k_ESC1(rate)		
{		
if (VUV!=0) {		
<b>SE_shape4, 8-0;</b>	<b>9</b>	<b>uimsbf</b>
<b>SE_shape5, 7-0;</b>	<b>8</b>	<b>uimsbf</b>
if (rate>=4000) {		
<b>SE_shape6, 4-0;</b>	<b>5</b>	<b>uimsbf</b>
}		
}		
else {		
if (rate>=4000) {		
<b>VX_gain2[3], 0;</b>	<b>1</b>	<b>uimsbf</b>
}		
<b>LSP2, 2-0;</b>	<b>3</b>	<b>uimsbf</b>
<b>LSP3, 3-0;</b>	<b>4</b>	<b>uimsbf</b>
<b>LSP5, 6-0;</b>	<b>7</b>	<b>uimsbf</b>
<b>VX_shape1[0], 5-0;</b>	<b>6</b>	<b>uimsbf</b>
<b>VX_shape1[1], 5;</b>	<b>1</b>	<b>uimsbf</b>
}		
}		
}		

Table 2.32 – Syntax of 4k\_ESC2()

Syntax	No. of bits	Mnemonic
4k_ESC2()		
{		
if (VUV!=0) {		

SE_shape1, 3-0;	4	uimsbf
} else { VX_shape1[1], 4-1;	4	uimsbf
} }		

Table 2.33 – Syntax of 4k\_ESC3()

Syntax	No. of bits	Mnemonic
4k_ESC3() { if (VUV!=0) { SE_shape2, 3-0;	4	uimsbf
} else { VX_shape1[1], 0;	1	uimsbf
VX_shape2[0], 4-2;	3	uimsbf
} }		

Table 2.34 – Syntax of 4k\_ESC4()

Syntax	No. of bits	Mnemonic
4k_ESC4(rate) { if (VUV!=0) { LSP2, 2-0;	3	uimsbf
LSP3, 3-0;	4	uimsbf
LSP5, 6-0;	7	uimsbf
Pitch, 0;	1	uimsbf
SE_shape3, 1-0;	2	uimsbf
} else { VX_shape2[0], 1-0;	2	uimsbf
VX_shape2[1], 4-0;	5	uimsbf
VX_shape2[2], 4-0;	5	uimsbf
if (rate>=4000) { VX_shape2[3], 4-0;	5	uimsbf
} } }		

2.3.4.2 Bitstream syntax for the scalable mode

Bitstream syntax of the base layer for scalable mode is the same as that of ErHVXCfixframe(2000). Bitstream syntax of enhancement layer, ErHVXCenhFrame(), for scalable mode is shown below.

Table 2.35 – Syntax of ErHVXCenh\_fixframe()

Syntax	No. of bits	Mnemonic
ErHVXCenh_fixframe() { Enh_ESC0(); Enh_ESC1(); Enh_ESC2();		

```
}

```

Table 2.36 – Syntax of Enh\_ESC0()

Syntax	No. of bits	Mnemonic
Enh_ESC0()		
{		
if (VUV!=0) {		
SE_shape3, 6-2;	5	uimsbf
LSP5, 7;	1	uimsbf
SE_shape4, 9;	1	uimsbf
SE_shape5, 8;	1	uimsbf
SE_shape6, 5;	1	uimsbf
SE_shape4, 8-6;	3	uimsbf
}		
else {		
LSP5, 7;	1	uimsbf
VX_gain2[0], 2-0;	3	uimsbf
VX_gain2[1], 2-0;	3	uimsbf
VX_gain2[2], 2-0;	3	uimsbf
VX_gain2[3], 2-1;	2	uimsbf
}		
}		

Table 2.37 – Syntax of Enh\_ESC1()

Syntax	No. of bits	Mnemonic
Enh_ESC1()		
{		
if (VUV!=0) {		
SE_shape4, 5-0;	6	uimsbf
SE_shape5, 7-0;	8	uimsbf
SE_shape6, 4-0;	5	uimsbf
}		
else {		
VX_gain2[3], 0;	1	uimsbf
LSP5, 6-0;	7	uimsbf
VX_shape2[0], 4-0;	5	uimsbf
VX_shape2[1], 4-0;	5	uimsbf
VX_shape2[2], 4;	1	uimsbf
}		
}		

Table 2.38 – Syntax of Enh\_ESC2()

Syntax	No. of bits	Mnemonic
Enh_ESC2()		
{		
if (VUV!=0) {		
LSP5, 6-0;	7	uimsbf
SE_shape3, 1-0;	2	uimsbf
}		
else {		
VX_shape2[2], 3-0;	4	uimsbf
VX_shape2[3], 4-0;	5	uimsbf
}		

}

2.3.4.3 Bitstream syntax of variable bitrate mode

Table 2.39 – Syntax of ErHVXCvarframe()

Syntax	No. of bits	Mnemonic
ErHVXCvarframe(rate) { if (rate == 2000) { if (var_ScalableFlag == 1) { BaseVar_ESC0() BaseVar_ESC1(); BaseVar_ESC2(); BaseVar_ESC3(); } else { Var2k_ESC0(); Var2k_ESC1(); Var2k_ESC2(); Var2k_ESC3(); } } } else { Var4k_ESC0(); Var4k_ESC1(); Var4k_ESC2(); Var4k_ESC3(); Var4k_ESC4(); } }		

Table 2.40 – Syntax of Var2k\_ESC0()

Syntax	No. of bits	Mnemonic
Var2k_ESC0() { <b>VUV, 1-0;</b> if (VUV==2    VUV==3) { <b>LSP4, 0;</b> <b>SE_gain, 4-0;</b> <b>LSP1, 4-0;</b> <b>Pitch, 6-1;</b> <b>LSP2, 6;</b> <b>LSP3, 4;</b> <b>LSP2, 5;</b> } else if (VUV==0) { <b>LSP4, 0;</b> <b>VX_gain1[0], 3-0;</b> <b>VX_gain1[1], 3-0;</b> <b>LSP1, 4-0;</b> <b>LSP2, 6-3;</b> <b>LSP3, 4-3;</b> } }	2  1 5 5 6 1 1 1  1 4 4 5 4 2	<b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 2.41 – Syntax of Var2k\_ESC1()

Syntax	No. of bits	Mnemonic
Var2k_ESC1() { if (VUV==2    VUV==3) { <b>SE_shape1, 3-0;</b> } else if (VUV==0) { <b>LSP2, 2-0;</b> <b>LSP3, 2;</b> } }	4  3 1	<b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b>

Table 2.42 – Syntax of Var2k\_ESC2()

Syntax	No. of bits	Mnemonic
Var2k_ESC2() { if (VUV==2    VUV==3) { <b>SE_shape2, 3-0;</b> } else if (VUV==0) { <b>LSP3, 1-0;</b> } }	4  2	<b>uimsbf</b>  <b>uimsbf</b>

Table 2.43 – Syntax of Var2k\_ESC3()

Syntax	No. of bits	Mnemonic
Var2k_ESC3() { if (VUV==2    VUV==3) { <b>LSP2, 4-0;</b> <b>LSP3, 3-0;</b> <b>Pitch, 0;</b> } }	5 4 1	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 2.44 – Syntax of Var4k\_ESC0()

Syntax	No. of bits	Mnemonic
Var4k_ESC0() { <b>VUV, 1-0;</b> if (VUV==2    VUV==3) { <b>LSP4, 0;</b> <b>SE_gain, 4-0;</b> <b>LSP1, 4-0;</b> <b>Pitch, 6-1;</b> <b>LSP2, 6-3;</b> <b>SE_shape3, 6-2;</b> <b>LSP3, 4;</b> <b>LSP5, 7;</b> <b>SE_shape4, 9;</b> <b>SE_shape5, 8;</b> }	2  1 5 5 6 4 5 1 1 1 1	<b>uimsbf</b>  <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

SE_shape6, 5;	1	uimsbf
}		
else if (VUV==0) {		
LSP4, 0;	1	uimsbf
VX_gain1[0], 3-0;	4	uimsbf
VX_gain1[1], 3-0;	4	uimsbf
LSP1, 4-0;	5	uimsbf
LSP2, 6-3;	4	uimsbf
LSP3, 4-3;	2	uimsbf
}		
else {		
UpdateFlag, 0;	1	uimsbf
if (UpdateFlag==1) {		
LSP4, 0;	1	uimsbf
VX_gain1[0], 3-0;	4	uimsbf
LSP1, 4-0;	5	uimsbf
LSP2, 6-3;	4	uimsbf
LSP3, 4-3;	2	uimsbf
}		
}		
}		

Table 2.45 – Syntax of Var4k\_ESC1()

Syntax	No. of bits	Mnemonic
Var4k_ESC1()		
{		
if (VUV==2    VUV==3) {		
SE_shape4, 8-0;	9	uimsbf
SE_shape5, 7-0;	8	uimsbf
SE_shape6, 4-0;	5	uimsbf
}		
else if (VUV==0) {		
LSP2, 2-0;	3	uimsbf
LSP3, 2-0;	3	uimsbf
VX_shape1[0], 5-0;	6	uimsbf
VX_shape1[1], 5-0;	6	uimsbf
}		
else {		
if (UpdateFlag==1) {		
LSP2, 2-0;	3	uimsbf
LSP3, 2-0;	3	uimsbf
}		
}		

Table 2.46 – Syntax of Var4k\_ESC2()

Syntax	No. of bits	Mnemonic
Var4k_ESC2()		
{		
if (VUV==2    VUV==3) {		
SE_shape1, 3-0;	4	uimsbf
}		
}		



Table 2.47 – Syntax of Var4k\_ESC3()

Syntax	No. of bits	Mnemonic
Var4k_ESC3() { if (VUV==2    VUV==3) { <b>SE_shape2, 3-0;</b> } }	4	uimsbf

Table 2.48 – Syntax of Var4k\_ESC4()

Syntax	No. of bits	Mnemonic
Var4k_ESC4() { if (VUV==2    VUV==3) { <b>LSP2, 2-0;</b> <b>LSP3, 3-0;</b> <b>LSP5, 6-0;</b> <b>Pitch, 0;</b> <b>SE_shape3, 1-0;</b> } }	3 4 7 1 2	uimsbf uimsbf uimsbf uimsbf uimsbf

Table 2.49 – Syntax of BaseVar\_ESC0()

Syntax	No. of bits	Mnemonic
BaseVar_ESC0() { <b>VUV, 1-0;</b> if (VUV==2    VUV==3) { <b>LSP4, 0;</b> <b>SE_gain, 4-0;</b> <b>LSP1, 4-0;</b> <b>Pitch, 6-1;</b> <b>LSP2, 6;</b> <b>LSP3, 4;</b> <b>LSP2, 5;</b> } else if (VUV==0) { <b>LSP4, 0;</b> <b>VX_gain1[0], 3-0;</b> <b>VX_gain1[1], 3-0;</b> <b>LSP1, 4-0;</b> <b>LSP2, 6-3;</b> <b>LSP3, 4-3;</b> } else { <b>UpdateFlag, 0;</b> if (UpdateFlag==1) { <b>LSP4, 0;</b> <b>VX_gain1[0], 3-0;</b> <b>LSP1, 4-0;</b> <b>LSP2, 6-3;</b> <b>LSP3, 4-3;</b> } } }	2 1 5 5 6 1 1 1 1 1 1 1 4 4 5 4 2 1 1 4 5 4 2	uimsbf uimsbf

Table 2.50 – Syntax of BaseVar\_ESC1()

Syntax	No. of bits	Mnemonic
BaseVar_ESC1() { if (VUV==2    VUV==3) { <b>SE_shape1, 3-0;</b>	<b>4</b>	<b>uimsbf</b>
} else if (VUV==0) { <b>LSP2, 2-0;</b>	<b>3</b>	<b>uimsbf</b>
<b>LSP3, 2;</b>	<b>1</b>	<b>uimsbf</b>
} else { if (UpdateFlag==1) { <b>LSP2, 2-0;</b>	<b>3</b>	<b>uimsbf</b>
<b>LSP3, 2-0;</b>	<b>3</b>	<b>uimsbf</b>
} } }		

Table 2.51 – Syntax of BaseVar\_ESC2()

Syntax	No. of bits	Mnemonic
BaseVar_ESC2() { if (VUV==2    VUV==3) { <b>SE_shape2, 3-0;</b>	<b>4</b>	<b>uimsbf</b>
} else if (VUV==0) { <b>LSP3, 1-0;</b>	<b>2</b>	<b>uimsbf</b>
<b>VX_shape1[0], 5-4;</b>	<b>2</b>	<b>uimsbf</b>
} }		

Table 2.52 – Syntax of BaseVar\_ESC3()

Syntax	No. of bits	Mnemonic
BaseVar_ESC3() { if (VUV==2    VUV==3) { <b>LSP2, 4-0;</b>	<b>5</b>	<b>uimsbf</b>
<b>LSP3, 3-0;</b>	<b>4</b>	<b>uimsbf</b>
<b>Pitch, 0;</b>	<b>1</b>	<b>uimsbf</b>
} else if (VUV==0) { <b>VX_shape1[0], 3-0;</b>	<b>4</b>	<b>uimsbf</b>
<b>VX_shape1[1], 5-0;</b>	<b>6</b>	<b>uimsbf</b>
} }		

## 2.3.4.4 Enhancement Layer of the scalable mode of the variable bitrate mode

Table 2.53 – Syntax of ErHVXCenh\_varframe()

Syntax	No. of bits	Mnemonic
ErHVXCenh_varframe() { EnhVar_ESC0(); EnhVar_ESC1(); EnhVar_ESC2(); }		

Table 2.54 – Syntax of EnhVar\_ESC0()

Syntax	No. of bits	Mnemonic
EnhVar_ESC0() { if (VUV==2    VUV==3) { <b>SE_shape3, 6-2;</b> <b>LSP5, 7;</b> <b>SE_shape4, 9;</b> <b>SE_shape5, 8;</b> <b>SE_shape6, 5;</b> <b>SE_shape4, 8-6;</b> } }	   <b>5</b> <b>1</b> <b>1</b> <b>1</b> <b>1</b> <b>3</b>	   <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 2.55 – Syntax of EnhVar\_ESC1()

Syntax	No. of bits	Mnemonic
EnhVar_ESC1() { if (VUV==2    VUV==3) { <b>SE_shape4, 5-0;</b> <b>SE_shape5, 7-0;</b> <b>SE_shape6, 4-0;</b> } }	   <b>6</b> <b>8</b> <b>5</b>	   <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 2.56 – Syntax of EnhVar\_ESC2()

Syntax	No. of bits	Mnemonic
EnhVar_ESC2() { if (VUV==2    VUV==3) { <b>LSP5, 6-0;</b> <b>SE_shape3, 1-0;</b> } }	   <b>7</b> <b>2</b>	   <b>uimsbf</b> <b>uimsbf</b>

## 2.4 Bitstream semantics

### 2.4.1 Decoder configuration (*HvxcSpecificConfig, ErrorResilientHvxcSpecificConfig*)

**HVXCvarMode:** A flag indicating HVXC variable rate mode (Table 2.1).

**HVXCrateMode:** A 2 bit field indicating HVXC bit rate mode (Table 2.1).

**extensionFlag:** A flag indicating the presence of MPEG-4 Version 2 data (Table 2.1).

**var\_ScalableFlag:** A flag indicating ER\_HVXC variable scalable mode (Table 2.22).

**isBaseLayer:** A one-bit identifier representing whether the corresponding layer is the base layer (1) or the enhancement layer (0).

### 2.4.2 Bitstream frame (*alPduPayload*)

**LSP1:** This 5 bits field represents the index of the first stage LSP quantization (base layer, Table 2.8 and Table 2.17).

**LSP2:** This 7 bits field represents the index of the second stage LSP quantization (base layer, Table 2.8 and Table 2.17).

**LSP3:** This 5 bits field represents the index of the second stage LSP quantization (base layer, Table 2.8 and Table 2.17).

**LSP4:** This 1 bit field represents the flag whether a interframe prediction is used or not in the second stage LSP quantization (base layer, Table 2.8 and Table 2.17).

**LSP5:** This 8 bits field represents the index of the third stage LSP quantization (enhancement layer, Table 2.9).

**VUV:** This 2 bits field represents V/UV decision mode. It should be noted that this field has a different meaning according to HVXC variable rate mode (Table 2.10 and Table 2.15)

**Pitch:** This 7 bits field represents the index of the linearly quantized pitch lag ranging from 20 to 147 samples (Table 2.12 and Table 2.18).

**SE\_shape1:** This 4 bits field represents the index of the spectral envelope shape (base layer, Table 2.12 and Table 2.18).

**SE\_shape2:** This 4 bits field represents the index of the spectral envelope shape (base layer, Table 2.12 and Table 2.18).

**SE\_gain:** This 5 bits field represents the index of the spectral envelope gain (base layer, Table 2.12 and Table 2.18).

**VX\_shape1[sf\_num]:** This 6 bits field represents the index of the sf\_num-th subframe's VXC shape (base layer, Table 2.12 and Table 2.18).

**VX\_gain1[sf\_num]:** This 4 bits field represents the index of the sf\_num-th subframe's VXC gain (base layer, Table 2.12 and Table 2.18).

**SE\_shape3:** This 7 bits field represents the index of the spectral envelope shape (enhancement layer, Table 2.13).

**SE\_shape4:** This 10 bits field represents the index of the spectral envelope shape (enhancement layer, Table 2.13).

**SE\_shape5:** This 9 bits field represents the index of the spectral envelope shape (enhancement layer, Table 2.13).

**SE\_shape6:** This 6 bits field represents the index of the spectral envelope shape (enhancement layer, Table 2.13).

**VX\_shape2[sf\_num]:** This 5 bits field represents the index of the sf\_num-th subframe's VXC shape (enhancement layer, Table 2.13).

**VX\_gain2[sf\_num]:** This 3 bits field represents the index of the sf\_num-th subframe's VXC gain (enhancement layer, Table 2.13).

**UpdateFlag:** This 1 bit field represents a flag to indicate update noise frame(only for ER\_HVXC 4kbit/s variable rate mode).

## 2.5 HVXC decoder tools

### 2.5.1 Overview

HVXC provides an efficient coding scheme for Linear Predictive Coding (LPC) residuals based on harmonic and stochastic vector representation. Vector quantization (VQ) of the spectral envelope of LPC residuals with a weighted distortion measure is used when the signal is voiced. Vector excitation coding (VXC) is used when the signal is unvoiced. The major algorithmic features are:

- Weighted VQ of variable dimension spectral vector.
- A fast harmonic synthesis algorithm by IFFT.
- Interpolative coder parameters for speed/pitch control

Also, functional features include:

- As low as 33.5 ms of total algorithmic delay
- 2.0-4.0 kbps scalable mode
- Variable bit rate coding for rates less than 2.0 kbps

#### 2.5.1.1 Framing structure and block diagram of the decoder

Figure 2.2 shows the overall structure of the HVXC decoder. The HVXC decoder tools allow decoding of speech signals at 2 kbit/s and higher, up to 4 kbit/s. HVXC decoder tools also allow decoding with variable bit rate mode at an average bit rate of around 1.2-1.7 kbps. The basic decoding process is composed of four steps; de-quantization of parameters, generation of excitation signals for voiced frames by sinusoidal synthesis (harmonic synthesis) and noise component addition, generation of excitation signals for unvoiced frames by codebook look-up, and LPC synthesis. To enhance the synthesized speech quality spectral post-filtering is used.

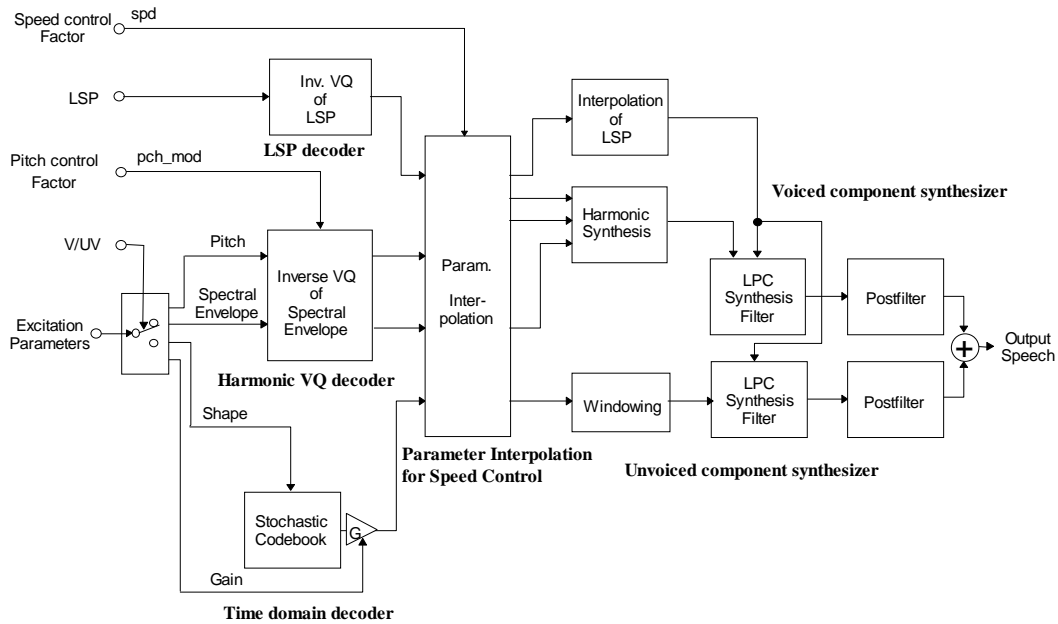


Figure 2.2 - Block diagram of the HVXC decoder

### 2.5.1.2 Delay mode

HVXC coder/decoder supports low/normal delay encode/decode mode, allowing any combinations of delay mode at 2.0-4.0 kbps with a scalable scheme. The figure below shows the framing structure of each delay mode. The frame length is 20 ms for all the delay modes. For example, use of low delay encode and low delay decode mode results in a total coder/decoder delay of 33.5 ms.

In the encoder, the algorithmic delay could be selected to be either 26 ms or 46 ms. When 46 ms delay is selected, one frame look ahead is used for pitch detection. When 26 ms delay is selected, only the current frame is used for pitch detection. For both cases, syntax is common, all the quantizers are common, and bitstreams are compatible. In the decoder the algorithmic delay can be selected to be either 10 ms (normal delay mode) or 7.5 ms (low delay mode). When 7.5 ms delay is selected, the decoder frame interval is shifted by 2.5 ms (20 samples) compared to the 10 ms delay mode. In this case, the excitation generation and LPC synthesis phase is shifted by 2.5 ms. Again, for both cases, syntax is common, all the quantizers are common, and bitstreams are compatible.

In summary, any independent choice of encoder/decoder delay from the following combination is possible:

Encoder delay: 26 ms or 46 ms

Decoder delay: 10 ms or 7.5 ms

One or some combinations of the delay mode shall be supported depending on the application.

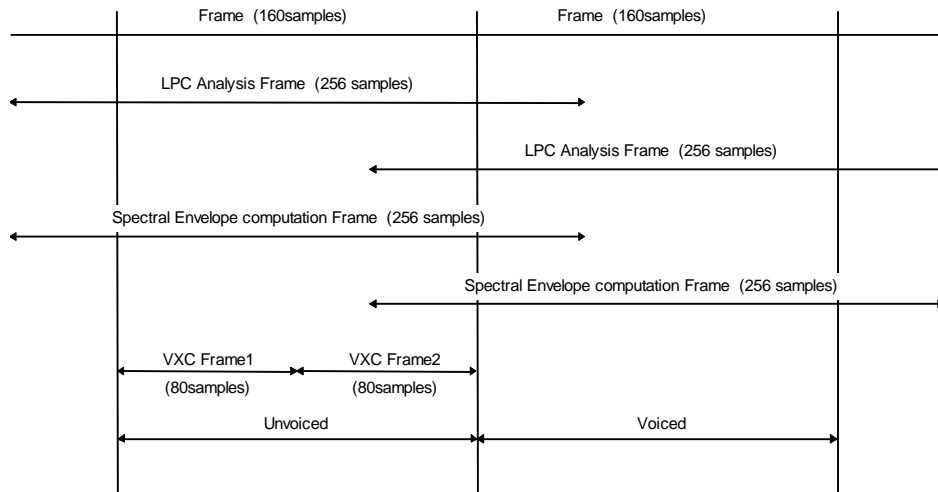


Figure 2.3 - Framing structure of HVXC

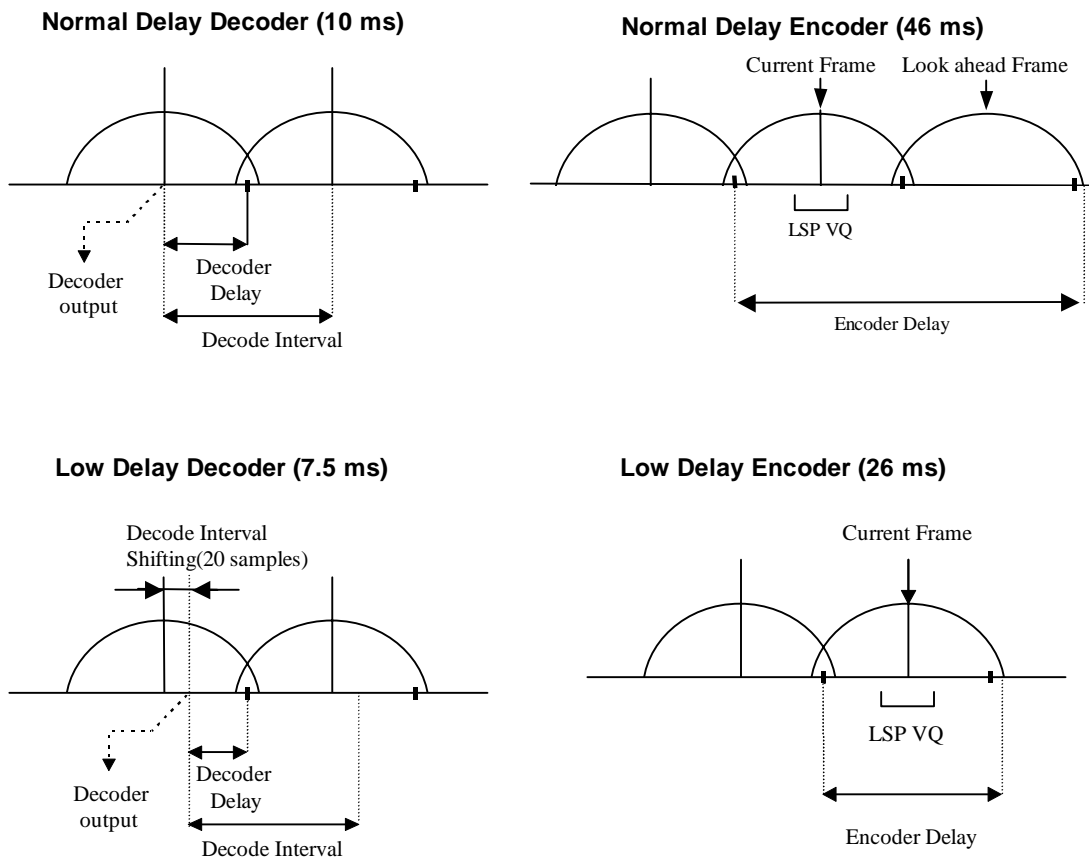


Figure 2.4 - Delay mode of the HVXC encoder and decoder

2.5.2 LSP decoder

2.5.2.1 Tool description

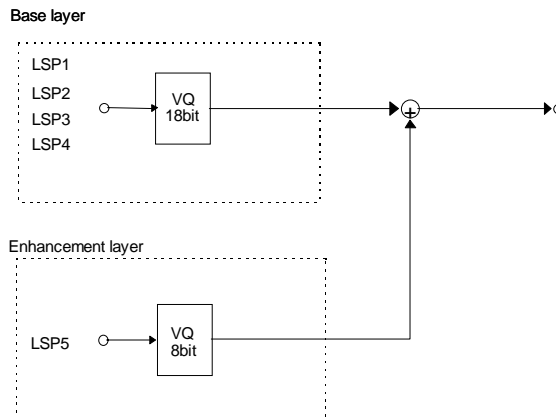


Figure 2.5 - LSP decoder

For the quantization of the LSP parameters, a multistage quantizer structure is used and the output vectors from each stage have to be summed up to obtain the LSP parameters.

When the bitrate is 2 kbps, the LSPs of the current frame, which are coded by split and two-stage vector quantization, are decoded using a two-stage decoding process. At 4 kbps, a 10-dimensional vector quantizer, which has an 8 bit codebook, is added to the bottom of the LSP quantizer scheme of 2.0 kbps coder. The bits needed for the LSPs are increased from 18bits/20msec to 26bits/20msec.

Table 2.57 - Configuration of the multistage LSP VQ

1st stage	10 LSP VQ	5 bits
2nd stage	(5+5) LSP VQ	(7+5+1) bits
3rd stage	10 LSP VQ	8 bits

2.5.2.2 Definitions

Definitions of constants

*LPCORDER* : LPC analysis order (=10)

*dim*[][] : dimensions for the split vector quantization

*min\_gap* : minimum distance between adjacent LSP coefficients (base layer, =4.0/256.0)

*ratio\_predict* : LSP interframe prediction ratio (=0.7)

*THRSLD\_L* : minimum distance between adjacent LSP coefficients

(low frequency part of enhancement layer, =0.020)

*THRSLD\_M* : minimum distance between adjacent LSP coefficients

(middle frequency part of enhancement layer, =0.020)

*THRSLD\_H* : minimum distance between adjacent LSP coefficients



(high frequency part of enhancement layer, =0.020)

### Definitions of variables

$qLsp[]$  : quantized LSP parameters

$LSP1$  : the index of the first stage LSP quantization (base layer)

$LSP2, LSP3$  : the indices of the second LSP quantization (base layer)

$LSP4$  : the flag showing whether a interframe prediction is used or not (base layer)

$LSP5$  : the index of the third LSP quantization (enhancement layer)

$lsp\_tbl[][][]$  : look-up tables for the first stage decoding process

$d\_tbl[][][]$  : look-up tables for the second stage decoding process of the VQ without interframe prediction

$pd\_tbl[][][]$  : look-up tables for the second stage decoding process of the VQ with interframe prediction.

$vqLsp[][]$  : look-up table for the enhancement layer

$sign$  : sign of code vector for the second stage decoding process

$idx$  : unpacked index for the second stage decoding process

$lsp\_predict[]$  : the LSPs predicted from  $lsp\_previous[]$  and  $lsp\_first[]$

$lsp\_previous[]$  : the LSPs decoded at the previous frame

$lsp\_current[]$  : the LSPs decoded at the current frame

$lsp\_first[]$  : the LSPs decoded at the first stage decoding process

### 2.5.2.3 Decoding process

The decoding process of the LSP parameters for the base layer (2.0 kbps) is the same as that of the narrowband CELP. The decoding process is as described below.

### Converting indices to LSPs

The LSPs of the current frame ( $lsp\_current[]$ ), which are coded by split and two-stage vector quantization, are decoded with a two-stage decoding process. The dimension of each vector is described in the tables below. The **LSP1** and **LSP2, LSP3** represent indices for the first and the second stage respectively.

**Table 2.58 - Dimension of the first stage LSP vector**

Split Vector Index: $i$	Vector Dimension: $dim[0][i]$
0	10

**Table 2.59 - Dimension of the second stage LSP vector**

Split Vector Index: $i$	Vector Dimension: $dim[1][i]$
0	5
1	5

In the first stage, the LSP vector of the first stage *lsp\_first[]* is decoded simply by looking up the table *lsp\_tbl[][][]*. (The table *lsp\_tbl[][][]* is shown in Annex 2.E)

```
for (i = 0; i < dim[0][0]; i++) {
    lsp_first[i] = lsp_tbl[0][LSP1][i];
}
```

In the second stage, there are two types of decoding processes, namely, decoding process of VQ without interframe prediction and VQ with interframe prediction. The flag **LSP4** indicates which process should be selected.

**Table 2.60 - Decoding process for the second stage**

LSP Index: <b>LSP4</b>	Decoding process
0	VQ without interframe prediction
1	VQ with interframe prediction

**Decoding process of VQ without interframe prediction**

In order to obtain LSPs of the current frame *lsp\_current[]*, the decoded vectors in the second stage are added to the decoded first stage LSP vector *lsp\_first[]*. The MSB of the **LSP2** and **LSP3** represents the sign of the decoded vector, and the remaining bits represent the index for the table *d\_tbl[][][]*. (The table *d\_tbl[][][]* is shown in Annex 2.E)

```
sign = LSP2>>6;
idx = LSP2&0x3f;
if (sign == 0) {
    for (i = 0; i < dim[1][0]; i++) {
        lsp_current[i] = lsp_first[i] + d_tbl[0][idx][i];
    }
}
else {
    for (i = 0; i < dim[1][0]; i++) {
        lsp_current[i] = lsp_first[i] - d_tbl[0][idx][i];
    }
}
sign = LSP3>>4;
idx = LSP3&0x0f;
if (sign == 0) {
    for (i = 0; i < dim[1][1]; i++) {
        lsp_current[dim[1][0]+i] = lsp_first[dim[1][0]+i] + d_tbl[1][idx][i];
    }
}
else {
    for (i = 0; i < dim[1][1]; i++) {
        lsp_current[dim[1][0]+i] = lsp_first[dim[1][0]+i] - d_tbl[1][idx][i];
    }
}
```

**Decoding process of VQ with interframe prediction**

In order to obtain LSPs of the current frame *lsp\_current[]*, the decoded vectors of the second stage are added to the LSP vector *lsp\_predict[]* which are predicted from the decoded LSPs of the previous frame *lsp\_previous[]* and the decoded first stage LSP vector *lsp\_first[]*. As with the decoding process of VQ without interframe prediction, the MSB of the **LSP2** and **LSP3** represents the sign of the decoded vector, and the remaining bits represent the index for the table *pd\_tbl[][][]*. (The table *pd\_tbl[][][]* is shown in Annex 2.E)

```

for (i = 0; i < LPCORDER; i++) {
    lsp_predict[i] = (1-ratio_predict)*lsp_first[i]
                    + ratio_predict*lsp_previous[i]
}

sign = LSP2>>6;
idx = LSP2&0x3f;
if (sign == 0) {
    for (i = 0; i < dim[1][0]; i++) {
        lsp_current[i] = lsp_predict[i] + pd_tbl[0][idx][i];
    }
}
else {
    for (i = 0; i < dim[1][0]; i++) {
        lsp_current[i] = lsp_predict[i] - pd_tbl[0][idx][i];
    }
}
sign = LSP3>>4;
idx = LSP3&0x0f;
if (sign == 0) {
    for (i = 0; i < dim[1][1]; i++) {
        lsp_current[dim[1][0]+i] = lsp_predict[dim[1][0]+i] + pd_tbl[1][idx][i];
    }
}
else {
    for (i = 0; i < dim[1][1]; i++) {
        lsp_current[dim[1][0]+i] = lsp_predict[dim[1][0]+i] - pd_tbl[1][idx][i];
    }
}
}

```

### Stabilization of LSPs

The decoded LSPs *lsp\_current*[] are stabilized in order to ensure stability of the LPC synthesis filter which is derived from the decoded LSPs. The decoded LSPs are arranged in ascending order, having a minimum distance between adjacent coefficients.

```

for (i = 0; i < LPCORDER; i++) {
    if (lsp_current[i] < min_gap) lsp_current[i] = min_gap;
}
for (i = 0; i < LPCORDER-1; i++) {
    if (lsp_current[i+1]-lsp_current[i] < min_gap) {
        lsp_current[i+1] = lsp_current[i]+min_gap;
    }
}
for (i = 0; i < LPCORDER; i++) {
    if (lsp_current[i] > 1-min_gap) lsp_current[i] = 1-min_gap;
}
for (i = LPCORDER-1; i > 0; i--) {
    if (lsp_current[i]-lsp_current[i-1] < min_gap) {
        lsp_current[i-1] = lsp_current[i]-min_gap;
    }
}

for (i = 0; i < LPCORDER; i++) {
    qLsp[i] = lsp_current[i];
}

```

### Storing the coefficients

After the LSP decoding process, the decoded LSPs have to be stored in memory, since they are used for prediction at the next frame.

```
for (i = 0; i < LPCORDER; i++) {
    lsp_previous[i] = lsp_current[i];
}
```

It must be noted that the stored LSPs *lsp\_previous*[] must be initialized as described below when the whole decoder is initialized.

```
for (i = 0; i < LPCORDER; i++) {
    lsp_previous[i] = (i+1)/(LPCORDER+1);
}
```

### Decoding process for the enhancement layer

For the enhancement layer (4.0 and 3.7 kbps), additional code vectors and the base layer's LSPs are summed up as follows.

```
for (i = 0; i < LPCORDER; i++) {
    qLsp[i] += vqLsp[LSP5][i];
}
```

After the calculation, the LSPs are stabilized again.

```
for (i = 0; i < 2; i++) {
    if (qLsp[i+1] - qLsp[i] < 0) {
        tmp = qLsp[i+1];
        qLsp[i+1] = qLsp[i];
        qLsp[i] = tmp;
    }
    if (qLsp[i+1] - qLsp[i] < THRSLD_L) {
        qLsp[i + 1] = qLsp[i] + THRSLD_L;
    }
}

for (i = 2; i < 6; i++) {
    if (qLsp[i+1] - qLsp[i] < THRSLD_M) {
        tmp = (qLsp[i+1] + qLsp[i])/2.0;
        qLsp[i+1] = tmp + THRSLD_M/2.0;
        qLsp[i] = tmp - THRSLD_M/2.0;
    }
}

for (i = 6; i < LPCORDER-1; i++) {
    if (qLsp[i+1] - qLsp[i] < 0) {
        tmp = qLsp[i+1];
        qLsp[i+1] = qLsp[i];
        qLsp[i] = tmp;
    }
    if (qLsp[i+1] - qLsp[i] < THRSLD_H) {
        qLsp[i] = qLsp[i+1] - THRSLD_H;
    }
}
```

#### 2.5.2.4 Tables

See LSP quantizer table of Annex 2.E.

## 2.5.3 Harmonic VQ decoder

### 2.5.3.1 Tool description

The decoding process consists of two major steps for the base layer, that is, inverse vector quantization of spectral envelope vectors and dimension conversion. For the enhancement layer, additional inverse-quantizers are used. The operations of each step are given below.

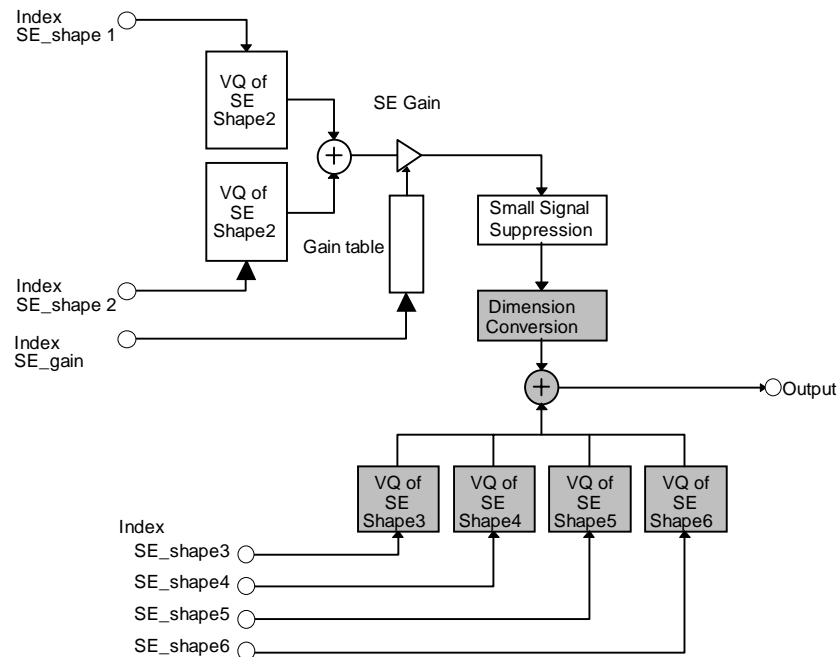


Figure 2.6 - Structure of Harmonic VQ decoder

### 2.5.3.2 Definitions

#### Definitions of constants

*SAMPLE* : the number of samples in frequency spectrum ranging from 0 to  $2\pi$  (=256)

*R* : over sampling rate in the dimension conversion (=8)

*vqdim0* : vector dimension of spectral envelope quantization (=44)

*JISU* : order of the over sampling filter in up sampled domain (=9)

*f\_coef*[] : over sampling filter coefficients

#### Definition of variables

*Pitch* : the index of linearly quantized pitch lag value

*pch* : pitch lag value in the current frame

*pch\_mod* : pitch modification factor

*w0f* : the original fundamental frequency where *SAMPLE* represents  $2\pi$

*send* : the number of harmonics in the current frame (between 0 and 3800Hz)

*w0* : target fundamental frequency after the dimension conversion where  $SAMPLE \times R$  represents  $2\pi$

HVXCrate : operating bit rate of the decoder

*qedvec*[] : quantized spectral envelope vector in the fixed dimension

*SE\_gain* : the index of spectral envelope gain (base layer)

*SE\_shape1*, *SE\_shape2* : the indices of spectral envelope shape (base layer)

*SE\_shape3*, *SE\_shape4*, *SE\_shape5*, *SE\_shape6* : the indices of spectral envelope shape (enhancement layer)

*g0*[] : *SE\_gain* codeword

*cb0*[][] : *SE\_shape1* codeword

*cb1*[][] : *SE\_shape2* codeword

*cb4k[0]*[][] : *SE\_shape3* codeword

*cb4k[1]*[][] : *SE\_shape4* codeword

*cb4k[2]*[][] : *SE\_shape5* codeword

*cb4k[3]*[][] : *SE\_shape6* codeword

*re*[] : input of the dimension conversion

*rel0*, *rel1* : 8 times over sampled values in the dimension conversion

*ip\_ratio* : linear interpolation ratio of the dimension conversion

*target*[] : reconstructed vector due to the vector quantizers for the enhancement layer

*am*[] : reconstructed harmonic magnitudes vector

*feneq* : rms of quantized spectral envelope vector in the current frame

*feneqold* : rms of quantized spectral envelope vector in the previous frame

### 2.5.3.3 Decoding process

#### Pitch index decoding

The pitch lag value in the current frame,  $pch$ , is decoded from the pitch index **Pitch** as follows:

$$pch = Pitch + 20.0$$

Pitch modification can be done by dividing  $pch$  by pitch modification factor  $pch\_mod$ :

$$pch = pch / pch\_mod$$

If the pitch modification is controlled by the **pitch** field in the AudioSource BIFS node, the modification factor is:

$$pch\_mod = pitch$$

It should be noted that the modulated pitch lag value must be within a range from 8.0 to 147.0.

Then the number of harmonics in a frequency range between 0 and 3800Hz,  $send$ , and the fundamental frequency,  $w0$  (where  $SAMPLE \times R$  represents  $2\pi$ ), are calculated as follows:

$$send = (\text{int})(0.95 \times pch \times 0.5)$$

$$w0 = \frac{SAMPLE \times R}{pch}$$

#### Harmonic magnitudes decoding

Decoding of harmonic magnitudes consists of the following steps;

- (S1) Inverse vector quantization of the base layer**
- (S2) Suppression of small signals**
- (S3) Dimension conversion of base layer output**
- (S4) Inverse vector quantization of the enhancement layer**

For the 2.0kbps mode, **S1**, **S2** and **S3** above are executed to obtain the harmonic magnitudes. For the 4.0 and 3.7kbps mode, **S4** is executed in addition to **S1**, **S2** and **S3**.

In the 2.0 kbps mode, a combination of two-stage shape vector quantization and a scalar gain quantization is used, whose indices are **SE\_shape1**, **SE\_shape2** and **SE\_gain** respectively. The dimension of the two shape codebooks is fixed (=44). In **S1**, two shape vectors represented by **SE\_shape1** and **SE\_shape2** are added and then multiplied by the gain represented by **SE\_gain**. The spectral envelope vector obtained in **S1** covers frequency range from 0 to 3800 Hz. The spectral envelope vector of very small energy is then suppressed in **S2**. In order to obtain the harmonic magnitudes vector of the original dimension,  $send$ , dimension conversion is then applied to the spectral envelope vector in **S3**. In the 4.0 kbps mode, additional stage with a split VQ scheme composed of four vector quantizers is used for the enhancement layer. **SE\_shape3**, **SE\_shape4**, **SE\_shape5** and **SE\_shape6** represents the indices of the quantizers for the enhancement layer. In **S4**, the output of these quantizers is added to the output of the **S3** for harmonic magnitudes at the lowest 14 harmonic frequencies. When 3.7kbps mode is selected, **SE\_shape6** is not available and **S4** is carried out for the harmonic magnitudes of only the lowest 10 harmonic frequencies.

**Table 2.61 - Configuration of the multistage harmonic VQ**

2.0 kbps two-stage VQ dimension	4bits shape + 4bits shape + 5bits gain			
	44			
4.0 kbps split VQ dimension	7bits	10bits	9bits	6bits
	2	4	4	4

**Dimension conversion algorithm**

The theoretical background of the dimension conversion algorithm used in this tool is explained below.

The number of points which composes the spectral envelope varies depending on the pitch value, since the spectral envelope is a set of the estimates of the magnitudes at each harmonic. The number of harmonics ranges from 9 to 70. In order to obtain the variable number of harmonic magnitudes, the decoder has to convert a fixed-dimension (=44) codevector to a variable dimension vector. The number of points, which represent the shape of the spectral envelope, should be modified without changing the shape. For this purpose, a dimension converter consisting of a combination of a low pass filter and a 1st order linear interpolator is used. An FIR low pass filter with 7 sets of coefficients, each set consisting of 8 coefficients, is used for the first stage 8-times over-sampling. The 7 sets of the filter coefficients are obtained by grouping 8 every coefficients from a windowed sinc,  $f\_coef[i]$ , with the offsets of 1 through 7, where

$$f\_coef[i] = \frac{\sin \pi(i-32)/8}{\pi(i-32)/8} (0.5 - 0.5 \cos 2\pi i / 64) \quad 0 \leq i \leq 64$$

This FIR filtering allows decimated computation, in which only the points used at the next stage are computed. They are the left and right adjacent points of the final output of the dimension converter.

At the second over-sampling stage, 1st order linear interpolation is applied to obtain the necessary output points. In this way, harmonic magnitudes vectors of variable-dimension are obtained from spectral envelope vectors of fixed dimension (=44).

**(S1) Inverse vector quantization of the base layer:**

```
qedvec[0] = 0.0f;
for (i = 0; i < vqdim0; i++)
    qedvec[i+1] = g0[SE_gain]*(cb0[SE_shape1][i]+cb1[SE_shape2][i]);
```

**(S2) Small signal suppression:**

```
feneq = 0.0f;
for (i = 0; i < vqdim0; i++)
    feneq += qedvec[i+1]*qedvec[i+1];
feneq = sqrt(feneq/(float)vqdim0);
if (feneq < 1.0f || 0.5f*(feneqold+feneq) < 1.4f) {
    for (i = 0; i < vqdim0; i++) qedvec[i+1] = 0.0f;
}
feneqold = feneq;
```

**(S3) Dimension conversion of the base layer output:**

```
for (i = 0; i < (JISU-1)/2; i++)
```



```

    re[i] = 0.0f;
    for (i = 0; i <= vqdim0; i++)
        re[i+(JISU-1)/2] = qedvec[i];
    for (i = 0; i < (JISU-1)/2; i++)
        re[i+vqdim0+1+(JISU-1)/2] = qedvec[vqdim0];

    w0f = (float)(SAMPLE*0.5*0.95)/(float)vqdim0;

    ii = 0;
    for (i = 0; i <= vqdim0 && ii <= send; i++) {
        for (p = 0; p < R && ii <= send; p++) {
            ip_ratio = (i*R+p+1)*w0f-w0*ii;
            if (ip_ratio > 0) {
                ip_ratio /= w0f;
                rel0 = rel1 = 0.0f;
                for (j = 1; j < JISU; j++) {
                    rel0 += f_coef[j*R-p]*re[i+j];
                    rel1 += f_coef[j*R-(p+1)]*re[i+j];
                }
                am[ii] = rel0*ip_ratio + rel1*(1.0f-ip_ratio);
                ii++;
            }
        }
    }

```

#### (S4) Inverse vector quantization of the enhancement layer:

```

target[0]=0.;
k = 1;
for (i = 0; i < 2; i++, k++)
    target[k] = cb4k[0][SE_shape3][i];
for (i = 0; i < 4; i++, k++)
    target[k] = cb4k[1][SE_shape4][i];
for (i = 0; i < 4; i++, k++)
    target[k] = cb4k[2][SE_shape5][i];
if(HVXCrate >= 4000){
    for (i = 0; i < 4; i++, k++)
        target[k] = cb4k[3][SE_shape6][i];
}
else{
    for (i = 0; i < 4; i++, k++)
        target[k] = 0.0f;
}
if (send > 14) {
    for (i = 15; i <= send; i++)
        target[i] = 0.0f;
}

for (i = 0; i <= send; i++)
    am[i] += target[i];

```

#### 2.5.3.4 Tables

See harmonic VQ table of Annex 2.E

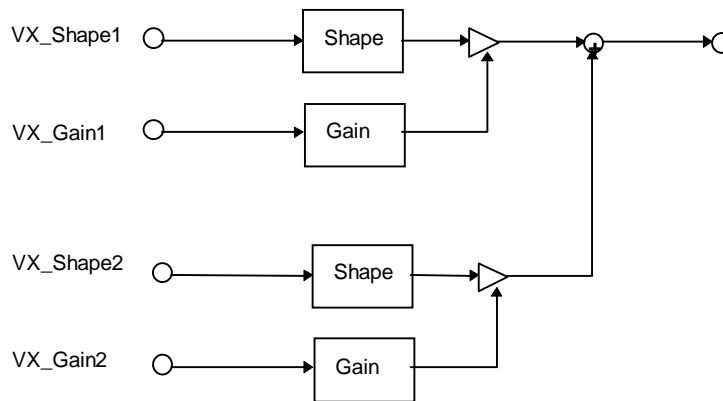
**2.5.4 Time domain decoder**

**2.5.4.1 Tool description**

For unvoiced segments of speech, a scheme that is similar to VXC (Vector Excitation Coding) is used. The time domain decoder generates an excitation waveform for the unvoiced portion by table look up using the transmitted indices. The shape vector and gain of the base layer are updated every 10 ms. The shape is scaled by multiplying each sample with the gain value. In the 2.0 kbps mode, only the output of the first stage (base layer) is used. In the 4.0 kbps and 3.7 kbps modes, the shape vector and the gain of the second stage (enhancement layer) are multiplied and added to the output of the first stage. The shape and gain of the enhancement layer are updated every 5 ms.

**Table 2.62 - Configuration of the time domain VQ**

1st stage	(80dimension 6bits shape + 4bits gain) x 2
2nd stage	(40dimension 5bits shape + 3bits gain) x 4



**Figure 2.7 - Time domain VQ decoder**

**2.5.4.2 Definitions**

**Definitions of constants**

*DimShape* : the first stage VXC frame length (=80)

*DimShape2* : the second stage VXC frame length (=40)

**Definition of variables**

HVXCrate : operating bit rate of the decoder

*res[i]* : VXC decoder output ( $0 \leq i < FRM$ )

*cbL0\_g[i]* : *i*-th entry of the first stage VXC gain codebook

*cbL0\_s[i][j]* : *j*-th component of *i*-th entry of the first stage VXC shape codebook

*cbL1\_g[i]* : *i*-th entry of the second stage VXC gain codebook

$cbL1\_s[i][j]$  :  $j$ -th component of  $i$ -th entry of the second stage VXC shape codebook

$VX\_gain1[i]$  : the index of the  $i$ -th subframe's VXC gain (base layer,  $i = 0,1$ )

$VX\_shape1[i]$  : the index of the  $i$ -th subframe's VXC shape (base layer,  $i = 0,1$ )

$VX\_gain2[i]$  : the index of the  $i$ -th subframe's VXC gain (enhancement layer,  $i = 0,1,2,3$ )

$VX\_shape2[i]$  : the index of the  $i$ -th subframe's VXC shape (enhancement layer,  $i = 0,1,2,3$ )

### 2.5.4.3 Decoding process

For the base layer:

```
for (i = 0; i < DimShape; i++)
  res[i] = cbL0_g[VX_gain[0]] * cbL0_s[VX_shape[0]][i];
for (i = 0; i < DimShape; i++)
  res[i + DimShape] = cbL0_g[VX_gain[1]] * cbL0_s[VX_shape[1]][i];
```

To add the enhancement layer:

```
if(HVXCrate >= 4000){ /* 4.0 kbps mode */
  for (i = 0; i < 4; i++){
    for (j = 0; j < DimShape2; j++){
      res[j+DimShape2*i] += cbL1_g[VX_gain2[i]]*cbL1_s[VX_shape2[i]][j];
    }
  }
}
else{ /* 3.7kbps mode */
  for (i = 0; i < 3; i++){
    for (j = 0; j < DimShape2; j++){
      res[j+DimShape2*i] += cbL1_g[VX_gain2[i]]*cbL1_s[VX_shape2[i]][j];
    }
  }
}
```

### 2.5.4.4 Tables

See stochastic codebook table of Annex 2.E.

## 2.5.5 Parameter interpolation for speed control

### 2.5.5.1 Tool description

The scheme for Speed Control is explained in this subclause. The decoder has a scheme for parameter interpolation to generate the input for the “Voiced Component Synthesizer” and “Unvoiced Component Synthesizer” at any arbitrary time instant. By this scheme, a sequence of parameters in modified intervals are computed and applied to the both of the synthesizers. In this way, decoder output in a modified time scale is obtained.

## 2.5.5.2 Definitions

### Definitions of constants

$FRM$  : frame interval (=160)

$P$  : LPC order (=10)

### Definitions of variables

"Parameter Interpolation" block computes the parameters in the modified time scale by interpolating the received parameters. The operation of this block is described below.

The operation of this block is basically linear interpolation and replacement of parameters.

Let us denote the arrays of original parameters as:

$pch[n]$  : pitch lag value at time index  $n$

$vuv[n]$  : V/UV index at time index  $n$

$lsp[n][i]$  : decoded LSPs at time index  $n$  ( $0 \leq i < P$ )

$send[n]$  : number of harmonics at time index  $n$

$am[n][i]$  : harmonic magnitudes at time index  $n$  ( $0 \leq i < send[n]$ )

$vex[n][i]$  : decoded VXC excitation signal at time index  $n$  ( $0 \leq i < FRM$ )

$param[n]$  : parameter at time index  $n$

and interpolated parameters as:

$mdf\_pch[m]$  : pitch lag value at time index  $m$

$mdf\_vuv[m]$  : V/UV index at time index  $m$

$mdf\_lsp[m][i]$  : decoded LSPs at time index  $m$  ( $0 \leq i < 10$ )

$mdf\_send[m]$  : number of harmonics at time index  $m$

$mdf\_am[m][i]$  : harmonic magnitudes at time index  $m$  ( $0 \leq i < send[m]$ )

$mdf\_vex[m][i]$  : decoded VXC excitation signal at time index  $m$  ( $0 \leq i < FRM$ )

$mdf\_param[m]$  : parameter at time index  $m$

where  $n$  and  $m$  are the time indices (frame number) before and after the time scale modification. The frame intervals are both 20 ms.

$spd$  : the ratio of speed change ( $0.5 \leq spd \leq 2.0$ )

$N_1$  : the duration of the original speech (total frame number)

$N_2$  : the duration of the speed controlled speech (total frame number)

$fr_0, fr_1$  : frame indices adjacent to the interpolation point

$left, right$  : interpolation ratio

$tmp\_vuv$  : temporal V/UV index

### 2.5.5.3 Speed control process

Let us define the ratio of speed change as  $spd$  :

$$spd = N_1/N_2 \quad (2.5.5.1)$$

where  $N_1$  is the duration of the original speech and  $N_2$  is the duration of the speed controlled speech. Therefore,

$$0 \leq n < N_1 \text{ and } 0 \leq m < N_2.$$

If the speed is controlled by the time scaling factor in the **speed** field of the AudioSource BIFS node, the speed change ratio is:

$$spd = 1 / speed$$

Basically, the time scale modified parameters are expressed as:

$$mdf\_param[m] = param[m \times spd] \quad (2.5.5.2)$$

where  $param$  are:  $pch, vuv, lsp$  and  $am$ . In general, however,  $m \times spd$  is not an integer.

We therefore define:

$$\begin{cases} fr_0 = \lfloor m \times spd \rfloor \\ fr_1 = fr_0 + 1 \end{cases} \quad (2.5.5.3)$$

to generate parameters at a time index  $m \times spd$  by linearly interpolating the parameters at time indices  $fr_0$  and  $fr_1$ . In order to execute linear interpolation, let us define:

$$\begin{cases} left = m \times spd - fr_0 \\ right = fr_1 - m \times spd \end{cases} \quad (2.5.5.4)$$

Then Eq.(2.5.5.2) can be approximated as:

$$mdf\_param[m] = param[fr_0] \times right + param[fr_1] \times left \quad (2.5.5.5)$$

where *param* are: *pch*, *vuv*, *lsp* and *am*.

For *lsp*[*n*][*i*] and *am*[*n*][*i*], this linear interpolation is applied with the index *i* being fixed.

As for the parameter *vex*:

*vex*[*n*][*i*] has excitation signals for UV frames by codebook look-up.

*FRM* samples from *vex*[*n*][*i*] centering around the time  $m \times spd$  are taken and the energy over the *FRM* samples are computed. Gaussian noise consisting of *FRM* samples is then generated and its norm is adjusted so that its energy be equal to that of *FRM* samples taken from *vex*[*n*][*i*]. This gain adjusted Gaussian noise sequence is used for *mdf\_vex*[*m*][*i*].

Principal operation of time scale modification can be expressed by the Eq.(2.5.5.5), however, the V/UV decisions at *fr*<sub>0</sub> and *fr*<sub>1</sub>, prior to the interpolation, have to be considered.

The interpolation and replacement strategies adapted to V/UV decisions are described below. In the explanation below, full voiced and mixed voiced (*vuv*[*n*] ≠ 0) are grouped as "Voiced", and only *vuv*[*n*] = 0 case is regarded as "Unvoiced". In the case of variable rate coding, "Background Noise" mode (*vuv*[*n*] = 1) is also treated as "Unvoiced".

- When V/UV decisions at *fr*<sub>0</sub> and *fr*<sub>1</sub> are Voiced - Voiced:

New V/UV index *mdf\_vuv*[*m*] is obtained as follows;

$$tmp\_vuv = vuv[fr_0] \times right + vuv[fr_1] \times left$$

if (*tmp\_vuv* > 2)

$$mdf\_vuv[m] = 3$$

else if (*tmp\_vuv* > 1)

$$mdf\_vuv[m] = 2$$

else if (*tmp\_vuv* > 0)

$$mdf\_vuv[m] = 1$$

New pitch lag value *mdf\_pch*[*m*] is obtained as follows;

if ( $0.57 < pch[fr_0] / pch[fr_1] \ \&\& \ pch[fr_0] / pch[fr_1] < 1.75$ )

$$mdf\_pch[m] = pch[fr_0] \times right + pch[fr_1] \times left$$

else

if (*left* < *right*)

$$mdf\_pch[m] = pch[fr_0]$$

else

$$mdf\_pch[m] = pch[fr_1]$$

All the other parameters are interpolated by Eq.(2.5.5.5).

- When V/UV decisions at  $fr_0$  and  $fr_1$  are Unvoiced - Unvoiced:  
All the parameters are interpolated by Eq.(2.5.5.5) except for  $mdf\_vex$ .  $mdf\_vex[m][i]$  is generated by Gaussian noise having the same energy as that of  $FRM$  samples taken from  $vex[n][i]$  centering around the time  $m \times spd$ .
- When V/UV decisions at  $fr_0$  and  $fr_1$  are Voiced - Unvoiced:  
If  $left < right$   
All the parameters at time  $fr_0$  are used instead of computing the parameters at  $m \times spd$ .  
If  $left \geq right$   
All the parameters at time  $fr_1$  are used instead of computing the parameters at  $m \times spd$ .  
 $mdf\_vex[m][i]$  is also generated by Gaussian noise having the same energy as that of  $FRM$  samples from  $vex[fr_1][i]$ . ( $0 \leq i < FRM$ )
- When V/UV decisions at  $fr_0$  and  $fr_1$  are Unvoiced - Voiced:  
If  $left < right$   
All the parameters at time  $fr_0$  are used instead of computing the parameters at  $m \times spd$ .  
 $mdf\_vex[m][i]$  is also generated by Gaussian noise having the same energy as that of  $FRM$  samples from  $vex[fr_0][i]$ . ( $0 \leq i < FRM$ )  
If  $left \geq right$   
All the parameters at time  $fr_1$  are used instead of computing the parameters at  $m \times spd$ .

In this manner, one obtain all the necessary parameters for the HVXC decoder. Just by applying these modified parameters,  $mdf\_param[m]$ , to “Voiced Component Synthesizer” and “Unvoiced Component Synthesizer” in the same way as usual (normal) decoding process, one obtain the time scale modified output.

Apparently, when  $N_2 < N_1$ , speed up decoding is executed, and when  $N_2 > N_1$  speed down decoding is executed. Power spectrum and pitch are not affected by this speed control, thus we can obtain good quality speech for the speed control factor of about  $0.5 < spd < 2.0$ .

### 2.5.6 Voiced component synthesizer

#### 2.5.6.1 Tool description

The voiced component synthesizer consists of the steps below;

- harmonic magnitudes modification
- harmonic excitation synthesis
- noise component addition
- LPC synthesis
- postfilter

An efficient harmonic excitation synthesis method is first used to obtain a periodic excitation waveform from harmonic magnitudes. By adding a noise component to the periodic waveform the voiced excitation signal is obtained, which is then fed into the LPC synthesis filter and postfilter to generate a voiced speech signal. The configuration of the postfilter is not normative and described in Annex 2.B.

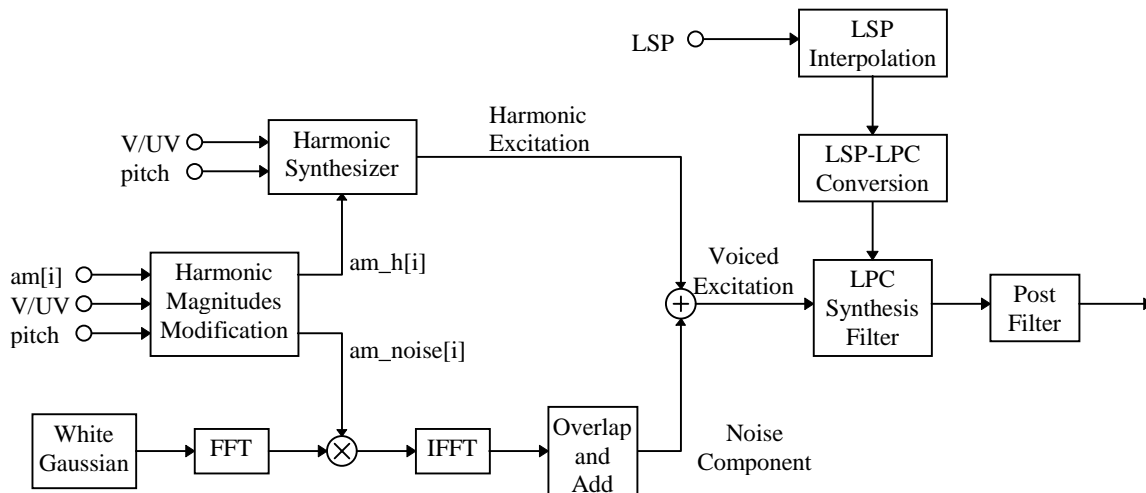


Figure 2.8 - Voiced component synthesizer

#### 2.5.6.2 Definitions

##### Definitions of constants

*PI* :  $\pi = 3.14159265358979\dots$

*FRM* : frame interval (=160)

*SAMPLE* : frame length of analysis (=256)



*WAVE\_LEN* : length of one pitch period waveform (=128)

*WDEVI* : threshold of the ratio of the fundamental frequency change (=0.1)

*LD\_LEN* : decode interval shifting for low delay mode (=20)

*RND\_MAX* : maximum number generated by random number generator (=0x7ffffff)

*SCALEFAC* : scale factor for the modified harmonic magnitudes for noise component generation (=10)

*c\_dis[i]* : trapezoid window for the harmonic synthesis of pitch discontinuous waveform shown in the Figure 2.9 ( $0 \leq i < FRM + LD\_LEN$ )

*ham[i]* : Hamming window ( $0 \leq i < SAMPLE$ )

*ham\_z[i]* : zero padded Hamming window ( $0 \leq i < 2 \times FRM$ )

*HAMLD* : the length of the Hamming window for low delay mode (=240)

*P* : LPC order (=10)

### Definitions of variables

*am2[i]* : harmonic magnitudes at the ending boundary of decode interval. They are obtained as *am[i]* in the subclause 2.5.3.3.

*am\_h[i]* : modified harmonic magnitudes at the ending boundary of the decode interval

*am\_noise[i]* : modified harmonic magnitudes for noise component generation at the ending boundary of the decode interval

*vuv2* : V/UV index at the ending boundary of the decode interval. It is obtained as **VUV** from the bitstream.

*vuv1* : V/UV index at the beginning boundary of the decode interval

*vuv0* : V/UV index at the beginning boundary of the previous decode interval

*pch2* : pitch lag value[sample] at the ending boundary of the decode interval. It is obtained as *pch* (pitch lag value in the current frame) in the subclause 2.5.3.3.

*pch1* : pitch lag value[sample] at the beginning boundary of the decode interval

*send2* : the number of harmonics at the ending boundary of the decode interval

*send1* : the number of harmonics at the beginning boundary of the decode interval

*w02* : the fundamental frequency at the ending boundary of the decode interval [rad/sample]

*w01* : the fundamental frequency at the beginning boundary of the decode interval [rad/sample]

*pha2[i]* : harmonic phase values at the ending boundary of the decode interval

*pha1[i]* : harmonic phase values at the beginning boundary of the decode interval

*ovsr2* : over sampling ratio at the ending boundary of the decode interval

*ovsr1* : over sampling ratio at the beginning boundary of the decode interval

*ovsrc* : linearly interpolated over sampling ratio

*wave2[i]* : one pitch period waveform generated from pitch and harmonic magnitudes at the ending boundary of the decode interval ( $0 \leq i < WAVE\_LEN$ )

*wave1[i]* : one pitch period waveform generated from pitch and harmonic magnitudes at the beginning boundary of the decode interval ( $0 \leq i < WAVE\_LEN$ )

*lp12*, *lp12r* : the length of over-sampled waveform needed to reconstruct a waveform of the decode interval (in case of continuous pitch transition)

*lp1*, *lp2*, *lp2r* : the length of over-sampled waveform needed to reconstruct a waveform of the decode interval (in case of discontinuous pitch transition)

*st* : offset for cyclic extension of *wave2[i]* at the beginning boundary of the decode interval

*iflat1*, *iflat2* : the length of the period where no parameter interpolation is done in over sampled domain (for low delay mode)

*out2[]* : cyclically extended waveform generated from *wave2[i]*

*out1[]* : cyclically extended waveform generated from *wave1[i]*

*out3[]* : weighted overlapped and added cyclically extended waveform within the decode interval generated from *out1[]* and *out2[]* (in case of continuous pitch transition)

*sv2[i]* : re-sampled waveform from *out2[]* (in case of discontinuous pitch transition,  $0 \leq i < FRM$ )

*sv1[i]* : re-sampled waveform from *out1[]* (in case of discontinuous pitch transition,  $0 \leq i < FRM$ )

*sv[i]* : generated voiced excitation signal ( $0 \leq i < FRM$ )

*ns[i]* : Gaussian noise with zero mean and unit variance ( $0 \leq i < SAMPLE$ )

*wns[i]* : Hamming windowed Gaussian noise ( $0 \leq i < SAMPLE$ )

*rms[i]* : spectral amplitude array ( $0 \leq i \leq SAMPLE / 2$ )

*ang[i]* : spectral phase array ( $0 \leq i \leq SAMPLE / 2$ )

*re[i]* : real part of the FFT coefficients ( $0 \leq i < SAMPLE$ )

*im[i]* : imaginary part of the FFT coefficients ( $0 \leq i < SAMPLE$ )

*w0* : fundamental frequency of the current frame, where  $2\pi$  is expressed as  $SAMPLE (= 256)$

*cns[i]* : the IFFT result of the current frame for noise component generation ( $0 \leq i < SAMPLE$ )

$cms\_z[i]$  : zero padded array to  $cms[]$  ( $0 \leq i < 2 \times FRM$ )

$cms\_z\_p[i]$  : the previous frame's  $cms\_z[]$  for overlap and add ( $0 \leq i < 2 \times FRM$ )

$add\_uv[i]$  : generated noise component at the decode interval ( $0 \leq i < FRM$ )

$lsp2[]$  : the de-quantized LSPs of the current frame obtained as  $qLsp[]$  in the subclause 2.5.2.3.

$lsp1[]$  : the de-quantized LSPs of the previous frame

$lspip[][]$  : the interpolated LSPs

$alpaip[][]$  : the linear predictive coefficients converted from the interpolated LSPs  $lspip[][]$

### Definitions of functions

$random()$  : random number generator that returns random numbers ranging from 0 to  $RND\_MAX$

$ceil(x)$  : a function that returns the least integer greater than or equal to  $x$

$floor(x)$  : a function that returns the greatest integer less than or equal to  $x$

### 2.5.6.3 Synthesis process

Voiced component signal is synthesized as shown below. The synthesis algorithm can be applied for both normal delay mode and low delay mode. For low delay mode, decode interval shifting,  $LD\_LEN$ , is used. The synthesized waveform covers from  $N = -160 + LD\_LEN$  [sample] to  $N = 0 + LD\_LEN$  [sample].  $N = 0$  represents the center of the current frame. If the frame shifting is 0 ( $LD\_LEN = 0$ ), the synthesized waveform is identical to that of the normal delay mode. This is shown in the figure below.

During the period from  $N = 0$  to  $N = LD\_LEN$ , pitch lag, harmonic magnitudes and LSP parameters are not interpolated and hold. If  $LD\_LEN = 0$ , decoder delay is 10[ms] and if  $LD\_LEN = 20$ , decoder delay is 7.5[ms].

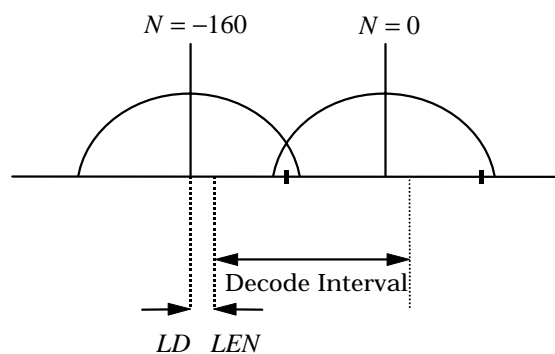


Figure 2.9 - Decode interval

**2.5.6.3.1 Harmonic magnitudes modification**

Harmonic magnitudes,  $am2[i]$ , are modified for harmonic excitation synthesis and noise component generation independently according to the V/UV index of the current frame,  $vuv2$ .

The modified harmonic magnitudes  $am\_h[i]$  for harmonic excitation synthesis and  $am\_noise[i]$  for noise component generation are obtained as described below.

When V/UV index,  $vuv2$ , is 0:

Do Nothing (just VXC decoder works).

When V/UV index,  $vuv2$ , is 1:

$$am\_h[i] = \begin{cases} am2[i] & (1 \leq i < send2 \times B\_TH1) \\ am2[i] \times AH1 & (send2 \times B\_TH1 \leq i \leq send2) \end{cases}$$

$$am\_noise[i] = \begin{cases} 0 & (1 \leq i < send2 \times B\_TH1) \\ SCALEFAC \times am2[i] \times AN1 & (send2 \times B\_TH1 \leq i \leq send2) \end{cases}$$

When V/UV index,  $vuv2$ , is 2:

$$am\_h[i] = \begin{cases} am2[i] & (1 \leq i < send2 \times B\_TH2) \\ am2[i] \times AH2 & (send2 \times B\_TH2 \leq i < send2 \times B\_TH2\_2) \\ am2[i] \times AH2\_2 & (send2 \times B\_TH2\_2 \leq i \leq send2) \end{cases}$$

$$am\_noise[i] = \begin{cases} 0 & (1 \leq i < send2 \times B\_TH2) \\ SCALEFAC \times am2[i] \times AN2 & (send2 \times B\_TH2 \leq i < send2 \times B\_TH2\_2) \\ SCALEFAC \times am2[i] \times AN2\_2 & (send2 \times B\_TH2\_2 \leq i \leq send2) \end{cases}$$

When V/UV index,  $vuv2$ , is 3:

$$am\_h[i] = \begin{cases} am2[i] & (1 \leq i < send2 \times B\_TH3) \\ am2[i] \times AH3 & (send2 \times B\_TH3 \leq i \leq send2) \end{cases}$$

$$am\_noise[i] = \begin{cases} 0 & (1 \leq i < send2 \times B\_TH3) \\ SCALEFAC \times am2[i] \times AN3 & (send2 \times B\_TH3 \leq i \leq send2) \end{cases}$$

**Table 2.63 - Constant values for harmonic magnitude modification**

**2 kbps:**

B_TH1	AN1	AH1	B_TH2	AN2	AH2	B_TH2_2	AN2_2	AH2_2	B_TH3	AN3	AH3
0.5	0.4	0.8	0.5	0.3	0.9	0.85	0.5	0.5	0.7	0.2	1.0

**4 kbps:**

B TH1	AN1	AH1	B TH2	AN2	AH2	B TH2 2	AN2 2	AH2 2	B TH3	AN3	AH3
0.5	0.3	0.9	0.5	0.3	0.9	0.85	0.5	0.5	0.8	0.2	1.0

**2.5.6.3.2 Harmonic excitation synthesis**

Harmonic excitation signal  $sv[i]$  ( $0 \leq i < FRM$ ) can be obtained by a fast synthesis method composed of an IFFT and sampling rate conversion.

First, using harmonic magnitudes and phase values, a waveform over one pitch period is generated by an IFFT. According to the pitch continuity, one of two different operations of cyclic extension and re-sampling of a waveform over one pitch period is selected and performed to obtain the harmonic excitation signal.

**Generation of a waveform over one pitch period**

The fundamental frequency at the beginning boundary of the decode interval,  $w01$ , is computed as

$$w01 = 2.0 * \text{PI} / \text{pch1};$$

The fundamental frequency at the ending boundary of the decode interval,  $w02$ , is computed as

$$w02 = 2.0 * \text{PI} / \text{pch2};$$

Harmonic phase values at the ending boundary of the decode interval,  $pha2[]$ , are computed from the harmonic phase values at the beginning boundary,  $pha1[]$ . When both of the V/UV index at the beginning of the current decode interval,  $vuv1$ , and the V/UV index at the beginning boundary of the previous decode interval,  $vuv0$ , are 0 (Unvoiced), harmonic phase values  $pha2[]$  are initialized using random phase values uniformly distributing between 0 to  $0.5\pi$ .

For normal delay mode :

```

if (vuv1 == 0 && vuv0 == 0) {
  for (i = 0; i < WAVE_LEN/2; i++)
    pha2[i] = 0.5*PI*(float)random()/(float)RND_MAX;
}
else {
  for (i = 0; i < WAVE_LEN/2; i++)
    pha2[i] = pha1[i] + 0.5*(w01+w02)*i*FRM;
}

```

For low delay mode:

```

if (vuv1 == 0 && vuv0 == 0) {
  for (i = 0; i < WAVE_LEN/2; i++)
    pha2[i] = 0.5*PI*(float)random()/(float)RND_MAX;
}
else {
  for (i = 0; i < WAVE_LEN/2; i++) {
    pha2[i] = pha1[i] + 0.5*(w01+w02)*i*(FRM-LD_LEN) + w01*i*LD_LEN;
  }
}

```

At the end boundary of the decode interval we have a spectrum with  $send2$  harmonics, whose magnitudes are  $am\_h[i]$  ( $1 \leq i \leq send2$ ) and phase values are  $pha2[i]$  ( $1 \leq i \leq send2$ ). Appending zeros to these arrays yields new arrays with  $WAVE\_LEN/2 (= 64)$  components ranging from 0 to  $\pi$ . If  $send2$  is greater than 63, first 64 values of  $am\_h[i]$  and  $pha2[i]$  are used. 128 point IFFT is applied to these arrays of magnitudes and phase values with the constraint that the results be real numbers. Now we have an over-sampled waveform over one pitch period,  $wave2[i]$  ( $0 \leq i < WAVE\_LEN$ ).

$WAVE\_LEN (= 128)$  samples are used to express the one pitch period of the waveform. Since the actual pitch lag value is  $pch2$ , the over-sampling rate  $ovsr2$  is

```
ovsr2 = WAVE_LEN/pch2;
```

Similarly the over-sampling rate  $ovsr1$  for the waveform over one pitch period at the beginning boundary of the decode interval,  $wave1[]$ , is

```
ovsr1 = WAVE_LEN/pch1;
```

### Pitch continuity check

When the ratio of the fundamental frequencies,  $|(w02 - w01)/w02|$ , is less than  $WDEVI (= 0.1)$ , it is regarded that pitch transition is continuous. In such a case, the fundamental frequencies and harmonic magnitudes are linearly interpolated between the beginning and the ending of the decode interval. Otherwise it is regarded that pitch transition is discontinuous and the fundamental frequencies and harmonic magnitudes are not linearly interpolated. In this case, independently synthesized periodic waveforms are added using appropriate weighting functions.

### Cyclic extension and re-sampling operation (continuous pitch transition)

Waveforms over one pitch period,  $wave1[]$  and  $wave2[]$ , are cyclically extended respectively to have sufficient length in the over over-sampled domain.

The length of over-sampled waveform needed to reconstruct a waveform of length  $FRM (= 160)$  at the original sampling rate (8 kHz) is at most  $lp12$ .

For normal delay mode:

```
lp12 = ceil(FRM*0.5*(over1+ovsr2));
lp12r = floor(FRM*0.5*(ovsr1+ovsr2)+0.5);
st = WAVE_LEN - (lp12r*WAVE_LEN);
for (i = 0; i < lp12; i++) {
    out1[i] = wave1[i*WAVE_LEN];
    out2[i] = wave2[(st+i)*WAVE_LEN];
}
```

For low delay mode:

```
lp12 = ceil((FRM-LD_LEN)*0.5*(over1+ovsr2)+LD_LEN*ovsr1);
lp12r = floor((FRM-LD_LEN)*0.5*(ovsr1+ovsr2)+LD_LEN*ovsr1+0.5);
st = WAVE_LEN - (lp12r*WAVE_LEN);
iflat1 = floor(ovsr1*LD_LEN+0.5);
iflat2 = floor(ovsr2*LD_LEN+0.5);
for (i = 0; i < lp12+iflat2+10; i++) {
    out1[i] = wave1[i*WAVE_LEN];
    out2[i] = wave2[(st+i)*WAVE_LEN];
}
```

These two waveforms,  $out1[]$  and  $out2[]$ , have the same “pseudo” pitch period ( $=WAVE\_LEN$  [sample]) and they are aligned. So simply adding these two waveforms using triangular windows produces the waveform  $out3[]$ .

For normal delay mode:

```
for (i = 0; i < lp12; i++)
    out3[i] = out1[i]*(float)(lp12-i)/(float)lp12 + out2[i]*(float)i/(float)lp12;
```

For low delay mode:

```
for (i = 0; i < iflat1; i++)
    out3[i] = out1[i];

for (i = iflat1; i < lp12; i++)
    out3[i] = out1[i]*(float)(lp12-i)/(float)(lp12-iflat1)
        + out2[i]*(float)(i-iflat1)/(float)(lp12-iflat1);

for (i = lp12; i < lp12+iflat2; i++)
    out3[i] = out2[i];
```

Finally,  $out3[]$  has to be re-sampled so that the resulting waveform can be expressed at the original sampling rate (8 kHz). This operation brings the waveform back from the “pseudo” pitch domain to the actual pitch domain. In principle, the re-sampling operation is just

$$sv[i] = out3[f(i)] \quad (0 \leq i < FRM)$$

where

$$f(i) = \int_0^i \left( ovsr1 \times \frac{FRM-t}{FRM} + ovsr2 \times \frac{t}{FRM} \right) dt.$$

The function  $f(i)$  maps the time index  $i$  of the original sampling rate (8 kHz) to the time index at the over-sampled rate under the condition that the fundamental frequencies,  $w01$  and  $w02$ , is linearly interpolated. Since  $f(i)$  does not return an integer value,  $sv[i]$  is obtained by linearly interpolating  $out3[\lfloor f(i) \rfloor]$  and  $out3[\lceil f(i) \rceil]$ .

For normal delay mode:

```
sv[0] = out3[0];
ffi = 0;
for (i = 1; i < FRM; i++) {
    ovsr1 = ovsr1*(float)(FRM-i)/(float)FRM + ovsr2*(float)i/(float)FRM;
    ffi += ovsr1;
    ffm = floor(ffi);
    ffip = ceil(ffi);
    if (ffm == ffip)
        sv[i] = out3[(int)ffip];
    else
        sv[i] = (ffi-ffm)*out3[(int)ffip] + (ffip-ffi)*out3[(int)ffm];
}
```

For low delay mode:

```
sv[0] = out3[iflat1];
ffi = 0;
for (i = 1; i < FRM; i++) {
    ovsr1 = ovsr1*(float)(FRM-LD_LEN-i)/(float)(FRM-LD_LEN)
        + ovsr2*(float)i/(float)(FRM-LD_LEN);
    ffi += ovsr1;
    ffm = floor(ffi);
    ffip = ceil(ffi);
```

```

if (ffim == ffip)
    sv[i] = out3[(int)ffip+iflat1];
else
    sv[i] = (ffi-ffim)*out3[(int)ffip+iflat1]
            + (ffip-ffi)*out3[(int)ffim+iflat1];
}

```

### Cyclic extension and re-sampling operation (discontinuous pitch transition)

Waveforms over one pitch period, *wave1[]* and *wave2[]* are cyclically extended to have sufficient length in the over-sampled domain. At both the beginning and ending boundary of the decode interval, cyclically extended waveforms, *out1[]* and *out2[]*, are obtained.

```

lp1 = ceil(FRM*ovsr1);
lp2 = ceil(FRM*ovsr2);
lp2r = floor(FRM*ovsr2+0.5);
st = WAVE_LEN - (lp2r*WAVE_LEN);

```

For normal delay mode:

```

for (i = 0; i < lp1; i++)
    out1[i] = wave1[i%WAVE_LEN];
for (i = 0; i < lp2; i++)
    out2[i] = wave2[(st+i)%WAVE_LEN];

```

For low delay mode:

```

iflat1 = floor(ovsr1*LD_LEN+0.5);
iflat2 = floor(ovsr2*LD_LEN+0.5);

for (i = 0; i < lp1+iflat1+10; i++)
    out1[i] = wave1[i%WAVE_LEN];
for (i = 0; i < lp2+iflat2+10; i++)
    out2[i] = wave2[(st+i)%WAVE_LEN];

```

where *lp1* and *lp2* are the length of over-sampled waveforms needed to reconstruct a waveform of length *FRM*(=160) at the original sampling rate (8 kHz).

These two waveforms, *out1[]* and *out2[]*, are re-sampled independently using the same linear interpolation method as in the case of "continuous pitch transition".

For normal delay mode:

```

sv1[0] = out1[0];
ffi = 0;
for (i = 1; i < FRM; i++) {
    ffi += ovsr1;
    ffim = floor(ffi);
    ffip = ceil(ffi);
    if (ffim == ffip)
        sv1[i] = out1[(int)ffip];
    else
        sv1[i] = (ffi-ffim)*out1[(int)ffip] + (ffip-ffi)*out1[(int)ffim];
}
sv2[0] = out2[0];
ffi = 0;
for (i = 1; i < FRM; i++) {
    ffi += ovsr2;
    ffim = floor(ffi);
    ffip = ceil(ffi);
    if (ffim == ffip)

```



```

    sv2[i] = out2[(int)ffip];
else
    sv2[i] = (ffi-ffim)*out2[(int)ffip] + (ffip-ffi)*out2[(int)ffim];
}

```

For low delay mode:

```

sv1[0] = out1[iflat1];
ffi = 0;
for (i = 1; i < FRM; i++) {
    ffi += ovsr1;
    ffim = floor(ffi);
    ffip = ceil(ffi);
    if (ffim == ffip)
        sv1[i] = out1[(int)ffip+iflat1];
    else
        sv1[i] = (ffi-ffim)*out1[(int)ffip+iflat1]
            + (ffip-ffi)*out1[(int)ffim+iflat1];
}
sv2[0] = out2[iflat2];
ffi = 0;
for (i = 1; i < FRM; i++) {
    ffi += ovsr2;
    ffim = floor(ffi);
    ffip = ceil(ffi);
    if (ffim == ffip)
        sv2[i] = out2[(int)ffip+iflat2];
    else
        sv2[i] = (ffi-ffim)*out2[(int)ffip+iflat2]
            + (ffip-ffi)*out2[(int)ffim+iflat2];
}

```

Using re-sampled waveforms at the original sampling rate (8 kHz), *sv1[]* and *sv2[]*, are then overlapped and added with the trapezoid window *c\_dis[]* shown in the Figure 2.10, where  $HM\_UP = HM\_DOWN = 60$ ,  $HM\_FLAT = 50$ .

```

for (i = 0; i < HM_FLAT; i++)
    c_dis[i] = 1.0f;
for (i = HM_FLAT; i < HM_FLAT+HM_DOWN; i++)
    c_dis[i] = (-i+(HM_FLAT+HM_DOWN))/(float)HM_DOWN;
for (i = HM_FLAT+HM_DOWN; i < FRM; i++)
    c_dis[i] = 0.0f;

```

For normal delay mode:

```

for (i = 0; i < FRM; i++)
    sv[i] = sv1[i]*c_dis[i] + sv2[i]*(1.0-c_dis[i]);

```

For low delay mode:

```

for (i = 0; i < FRM; i++)
    sv[i] = sv1[i]*c_dis[i+LD_LEN] + sv2[i]*(1.0-c_dis[i+LD_LEN]);
}

```

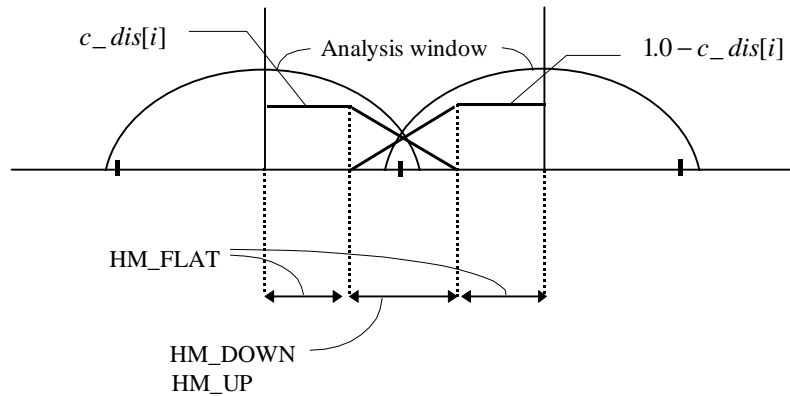


Figure 2.10 - Synthesis window for discontinuous pitch

### 2.5.6.3.3 Noise component generation

For the noise component generation for voiced excitation, white Gaussian noise is first generated. It is then colored and gain adjusted by the modified harmonic magnitudes,  $am\_noise[]$ , and weighted overlap and add is used to generate a continuous noise component in the time domain.

Hamming window of length  $SAMPLE(= 256)$ ,  $ham[i]$ , is first defined.

For normal delay mode,

```
for (i = 0; i < SAMPLE; i++)
    ham[i] = 0.54-0.46*cos(2.0*PI*i/(SAMPLE-1));
```

For low delay mode,

```
for (i = 0; i < (SAMPLE-HAMLD)/2; i++)
    ham[i] = 0.0;
for (i = (SAMPLE-HAMLD)/2; i < (SAMPLE+HAMLD)/2; i++)
    ham[i] = 0.54-0.46*cos(2.0*PI*i/(SAMPLE-1));
for (i = (SAMPLE+HAMLD)/2; i < SAMPLE; i++)
    ham[i] = 0.0;
```

Window  $ham[]$  is normalized to have a unit energy.

Let  $ns[i](0 \leq i < SAMPLE)$  be samples of white Gaussian noise with zero mean and unit variance. Hamming window  $ham[i]$  is then multiplied to  $ns[i]$  and  $wns[i]$  is obtained.

```
for (i = 0; i < SAMPLE; i++)
    wns[i] = ns[i]*ham[i];
```

256 point FFT of  $wns[i]$  is computed and spectral amplitude array  $rms[i](0 \leq i \leq SAMPLE/2)$  and spectral phase array  $ang[i](0 \leq i \leq SAMPLE/2)$  are computed as

```
for (i = 0; i <= SAMPLE/2; i++) {
    rms[i] = sqrt(re[i]*re[i]+im[i]*im[i]);
    ang[i] = atan2(im[i], re[i]);
}
```

where  $re[i](0 \leq i < SAMPLE)$  and  $im[i](0 \leq i < SAMPLE)$  are real part and imaginary part of the FFT coefficients respectively. Then spectral amplitude  $rms[]$  is colored and gain adjusted by the modified harmonic magnitudes,  $am\_noise[]$ . ( $w_0$  is the fundamental frequency of the current frame where  $w_0$  of value  $SAMPLE$  represents  $2\pi$ .)

```

for (i = 0; i <= send2; i++) {
  if (i == 0)
    lb = 0;
  else
    lb = ub+1;
  if (i == send2)
    ub = SAMPLE/2;
  else
    ub = floor((float)i*w0+w0/2.0+0.5);
  if (ub >= SAMPLE/2)
    ub = SAMPLE/2;
  bw = ub-lb+1;
  s = 0.0;
  for (j = lb; j <= ub; j++)
    s += rms[j]*rms[j];
  s = sqrt(s/(float)bw);
  for (j = lb; j <= ub; j++)
    rms[j] *= am_noise[i]/s;
}

```

256 point IFFT is computed with the colored and gain adjusted spectral amplitude array  $rms[]$  and the original spectral phase array  $ang[]$  with a constraint that the result be real numbers. Let the result of IFFT be  $cns[i]$  ( $0 \leq i < SAMPLE$ ).

It should be noted that when the current frame is "Unvoiced"

frame ( $vuv2 = 0$ ),  $cns[i] = 0.0$  ( $0 \leq i < SAMPLE$ ).

In order to generate noise component signal over the decode interval, weighted overlap and add with the previous frame's IFFT result is performed. The array  $cns\_z[i]$  ( $0 \leq i < 2 \times FRM$ ) shown below is obtained by padding ( $FRM - SAMPLE/2$ ) of zeros to both sides (beginning and ending) of  $cns[]$ .

```

for (i = 0; i < FRM-SAMPLE/2; i++)
  cns_z[i] = 0.0;
for (i = FRM-SAMPLE/2; i < FRM+SAMPLE/2; i++)
  cns_z[i] = cns[i-FRM+SAMPLE/2];
for (i = FRM+SAMPLE/2; i < 2*FRM; i++)
  cns_z[i] = 0.0;

```

In the same manner, zero padded Hamming window array  $ham\_z[i]$  ( $0 \leq i < 2 \times FRM$ ) is defined as:

```

for (i = 0; i < FRM-SAMPLE/2; i++)
  ham_z[i] = 0.0;
for (i = FRM-SAMPLE/2; i < FRM+SAMPLE/2; i++)
  ham_z[i] = ham[i-FRM+SAMPLE/2];
for (i = FRM+SAMPLE/2; i < 2*FRM; i++)
  ham_z[i] = 0.0;

```

Let us denote the  $cns\_z[]$  of the previous frame  $cns\_z\_p[]$ . Now noise component over the decode interval,  $add\_uv[i]$  ( $0 \leq i < FRM$ ), is obtained by combining  $cns\_z[]$  and  $cns\_z\_p[]$ .

For normal delay mode,

```

for (i = 0; i < FRM; i++)
  add_uv[i] = (cns_z_p[FRM+i]*ham_z[FRM+i]+cns_z[i]*ham_z[i])
    / (ham_z[FRM+i]*ham_z[FRM+i]+ham_z[i]*ham_z[i])

```

For low delay mode, noise component at the shifted decode interval is obtained as follows:

```

for (i = 0; i < FRM-LD_LEN; i++)
  add_uv[i] = (cns_z_p[FRM+i+LD_LEN]*ham_z[FRM+i+LD_LEN]+
              cns_z[i+LD_LEN]*ham_z[i+LD_LEN])
              / (ham_z[FRM+i+LD_LEN]*ham_z[FRM+i+LD_LEN]
                +ham_z[i+LD_LEN]*ham_z[i+LD_LEN]);
for (i = FRM-LD_LEN; i < FRM; i++)
  add_uv[i] = cns_z[i+LD_LEN]/ham_z[i+LD_LEN]);

```

Noise component *add\_uv*[] is added to harmonic excitation signal *sv*[] to make a voiced excitation signal:

```

for (i = 0; i < FRM; i++)
  sv[i] += add_uv[i];

```

#### 2.5.6.3.4 LPC synthesis

The voiced excitation signal obtained above, *sv*[*i*] ( $0 \leq i < FRM$ ), is fed into the LPC synthesis filter, whose coefficients are updated every 2.5 ms (=20 samples).

By linearly interpolating de-quantized LSPs of the previous and the current frame, *lsp1*[][] and *lsp2*[], the 8 sets of the interpolated LSPs, *lspip*[], are obtained.

```

for (i = 0; i < 8; i++)
  for (j = 0; j < P; j++)
    lspip[i][j] = (2.0*i+1.0)/16.0*lsp2[j] + (16.0-2.0*i-1.0)/16.0*lsp1[j];

```

For the low delay mode, since the decode interval is shifted by *LD\_LEN*(=20) samples (2.5 ms), the LSPs interpolation is carried out for the first 17.5 ms of the decode interval shown in Figure 2.9, and the LSPs of the current frame, *lsp2*[], is used for the last 2.5ms without interpolation.

```

for (i = 0; i < 7; i++)
  for (j = 0; j < P; j++)
    lspip[i][j] = (2.0*i+1.0)/14.0*lsp2[j] + (14.0-2.0*i-1.0)/14.0*lsp1[j];
for (j = 0; j < P; j++)
  lspip[7][j] = lsp2[j];

```

The 8 sets of the interpolated LSPs, *lspip*[], are converted to the linear predictive coefficients, *alpaip*[], respectively.

The transfer function of the *i* -th interval of 2.5 ms (20 samples) of the LPC synthesis filter is

$$H_i(z) = \frac{1}{\sum_{n=0}^P \text{alpaip}[i][n] z^{-n}} \quad (0 \leq i < 8)$$

Output of the LPC synthesis filter is fed into the postfilter described in subclause 2.B.1.3.1. The output signal is in the range from -32768 to 32767.

## 2.5.7 Unvoiced component synthesizer

### 2.5.7.1 Tool description

The unvoiced component synthesizer is composed of three steps, which are windowing the unvoiced excitation signal, LPC synthesis filter, and post filter operation. For unvoiced segments, the VXC (CELP) scheme is used.

### 2.5.7.2 Definitions

#### Definitions of constants

*FRM* : frame interval (=160)

*LD\_LEN* : decode interval shifting for low delay mode (=20)

*w\_celp\_up*[*i*] : the window from voiced frame to unvoiced frame ( $0 \leq i < FRM + LD\_LEN$ )

*w\_celp\_down*[*i*] : the window from unvoiced frame to voiced frame ( $0 \leq i < FRM + LD\_LEN$ )

*P* : LPC order (=10)

#### Definitions of variables

*qRes*[*i*] : decoded unvoiced excitation signal obtained as *res*[*i*] in subclause 2.5.4.3 ( $0 \leq i < FRM$ )

*old\_qRes*[*i*] : the last half of the decoded unvoiced excitation signal of the previous frame ( $0 \leq i < FRM / 2$ )

*suv*[*i*] : unvoiced excitation signal over the decode interval ( $0 \leq i < FRM$ )

*vuv2* : V/UV index of the current frame obtained as **VUV** in a bitstream

*vuv1* : V/UV index of the previous frame

*lsp2*[] : de-quantized LSPs of the current frame obtained as *qLsp*[] in the subclause 2.5.2.3

*lsp1*[] : de-quantized LSPs of the previous frame

*alpha2*[] : LPC coefficients converted from the LSPs *lsp2*[]

*alpha1*[] : LPC coefficients converted from the LSPs *lsp1*[]

### 2.5.7.3 Synthesis process

An unvoiced excitation signal at the decode interval, *suv*[], is generated from the decoded unvoiced excitation signal of the current frame, *qRes*[], and the last half of the decoded unvoiced excitation signal of the previous frame, *old\_qRes*[].

```
for (i = 0; i < FRM/2; i++) {
    suv[i] = old_qRes[i];
```

```

    suv[i+FRM/2] = qRes[i];
    old_qRes[i] = qRes[i+FRM/2];
}

```

For low delay mode, excitation signal of *LD\_LEN*(= 20) samples shifted version is used.

```

for (i = 0; i < FRM/2-LD_LEN; i++) {
    suv[i] = old_qRes[i+LD_LEN];
    suv[i+FRM/2] = qRes[i+LD_LEN];
    old_qRes[i] = qRes[i+FRM/2];
}
for (i = FRM/2-LD_LEN; i < FRM/2; i++) {
    suv[i] = qRes[i-FRM/2+LD_LEN];
    suv[i+FRM/2] = qRes[i+LD_LEN];
    old_qRes[i] = qRes[i+FRM/2];
}

```

The unvoiced excitation signal generated is windowed to be smoothly connected to a voiced frame. Figure 2.11 and Figure 2.12 shows the window shape for excitation waveform where *V/UV* is changed from unvoiced to voiced and from voiced to unvoiced respectively. The parameters in the figure are set as: *TD\_UP* = 30, *TD\_FLAT* = 50, *TD\_DOWN* = 30, *HM\_DOWN* = 60, *HM\_FLAT* = 50, *HM\_UP* = 60. These windows for an unvoiced frame are used only when an unvoiced frame is placed adjacent to voiced or mixed voiced frame.

```

for (i = 0; i < FRM-TD_UP-TD_FLAT; i++)
    w_celp_up[i] = 0.0;
for (i = FRM-TD_UP-TD_FLAT; i < FRM-TD_FLAT; i++)
    w_celp_up[i] = (float)(i-FRM+TD_UP+TD_FLAT)/(float)TD_UP;
for (i = FRM-TD_FLAT; i < FRM+LD_LEN; i++)
    w_celp_up[i] = 1.0;

for (i = 0; i < TD_FLAT; i++)
    w_celp_down[i] = 1.0;
for (i = TD_FLAT; i < TD_FLAT+TD_DOWN; i++)
    w_celp_down[i] = (float)(TD_FLAT+TD_DOWN-i)/(float)TD_DOWN;
for (i = TD_FLAT+TD_DOWN; i < FRM+LD_LEN; i++)
    w_celp_down[i] = 0.0;

```

For normal delay mode,

```

if (vuv1 != 0 && vuv2 != 0) {
    for (i = 0; i < FRM; i++)
        s_uv[i] = 0.0f;
}
else if (vuv1 != 0 && vuv2 == 0) {
    for (i = 0; i < FRM; i++)
        s_uv[i] *= w_celp_up[i];
}
else if (vuv1 == 0 && vuv2 != 0) {
    for (i = 0; i < FRM; i++)
        s_uv[i] *= w_celp_down[i];
}

```

For low delay mode, windowing position is shifted by *LD\_LEN* samples.

```

if (vuv1 != 0 && vuv2 != 0) {
    for (i = 0; i < FRM; i++)
        s_uv[i] = 0.0f;
}
else if (vuv1 != 0 && vuv2 == 0) {
    for (i = 0; i < FRM; i++)
        s_uv[i] *= w_celp_up[i+LD_LEN];
}
else if (vuv1 == 0 && vuv2 != 0) {
    for (i = 0; i < FRM; i++)

```

```

s_uv[i] *= w_celp_down[i+LD_LEN]
}

```

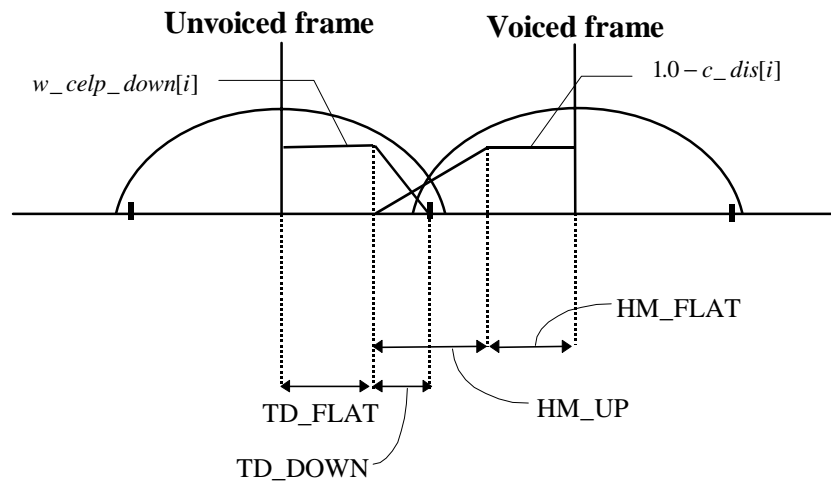


Figure 2.11 - Synthesis window for unvoiced/voiced

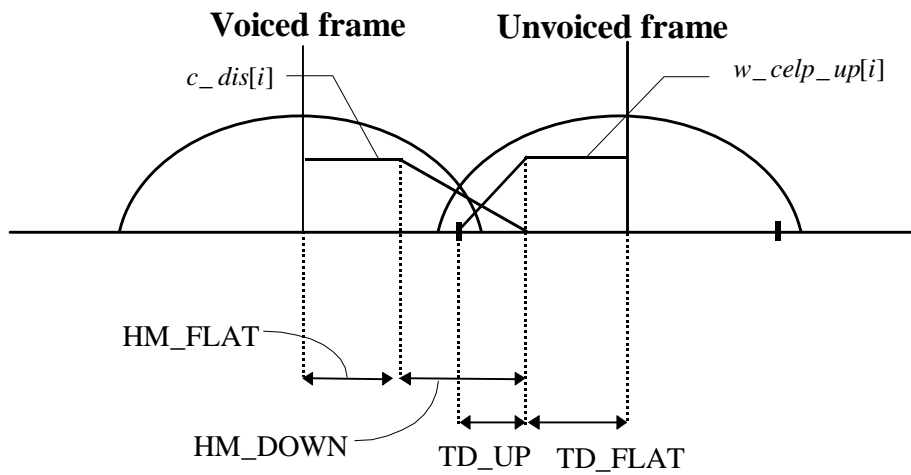


Figure 2.12 - Synthesis window for voiced/unvoiced

The first half of the unvoiced excitation signal,  $s_{uv}[i]$  ( $0 \leq i < FRM / 2$ ), is fed into the LPC synthesis filter

$$H_1(z) = \frac{1}{\sum_{n=0}^P \alpha_1[n] z^{-n}}$$

where  $\alpha1[]$  are linear predictive coefficients converted from the de-quantized LSPs of the previous frame,  $lsp1[]$ . When the second half of unvoiced excitation signal,  $suv[i]$  ( $FRM / 2 \leq i < FRM$ ), is fed into the LPC synthesis filter, the transfer function is switched to:

$$H_2(z) = \frac{1}{\sum_{n=0}^P \alpha2[n] z^{-n}}$$

where  $\alpha2[]$  are linear predictive coefficients converted from the de-quantized LSPs of the current frame,  $lsp2[]$ .

For low delay mode, since the decode interval is shifted by  $LD\_LEN$  samples, unvoiced excitation signal,  $suv[i]$  ( $0 \leq i < FRM / 2 - LD\_LEN$ ), is fed into the LPC synthesis filter  $H_1(z)$  and unvoiced excitation signal,  $suv[i]$  ( $FRM / 2 - LD\_LEN \leq i < FRM$ ), is fed into the LPC synthesis filter  $H_2(z)$ .

Output of the LPC synthesis filter is fed into the postfilter described in subclause 2.B.1.3.2.

The output signal is in the range from -32768 to 32767.

## 2.5.8 Variable rate decoder

### 2.5.8.1 Tool description

This subclause describes tools for variable rate decoding with HVXC core. This tool allows HVXC to operate at variable bit rates, where the average bit rate is reduced to 1.2 - 1.7 kbps for typical speech material. The major part of the algorithm is composed of "background noise decoding", where only the mode bits are received during the "background noise mode", and unvoiced frame is received with certain period of time for background noise generation.

### 2.5.8.2 Definitions

#### Definitions of constants

$BGN\_INTVL$ : maximum background noise interval (=8)

#### Definitions of variables

$idVUV$ : V/UV decision of the current frame

$prevLSP1$ : previously transmitted LSP vector

$prevLSP2$ : previously transmitted LSP vector before  $prevLSP1$

$bgnCnt$ : frame count of successive "Background Noise" frames

### 2.5.8.3 Decoding process

$idVUV$  is a parameter that has the result of V/UV decision and defined as;



$$idVUV = \begin{cases} 0 & \text{Unvoiced Speech} \\ 1 & \text{Background noise interval} \\ 2 & \text{Mixed voiced speech} \\ 3 & \text{Voiced speech} \end{cases}$$

Using the background noise detection method, variable rate coding is carried out based on fixed bit rate 2kbps HVXC.

**Table 2.64 - Bit allocations of the encoded parameters for the variable bit rate mode**

Mode (idVUV)	Background Noise (1)	UV (0)	MV (2),V (3)
V/UV	2 bit/20 msec	2 bit/20 msec	2 bit/20 msec
LSP	0 bit/20 msec	18 bit/20 msec	18 bit/20 msec
Excitation	0 bit/20 msec	8 bit/20 msec (gain only)	20 bit/20 msec (Pitch & harmonic spectral parameters)
Total	2 bit/20 msec  0.1 kbps	28 bit/20 msec  1.4 kbps	40 bit/20 msec  2.0 kbps

For "Mixed voiced speech" and "Voiced speech" ( $idVUV = 2,3$ ), the same decoding method as fixed bit rate mode is used.

In the decoder, two sets of LSP parameters,  $prevLSP1$  and  $prevLSP2$ , are hold where  $prevLSP1$  represents previously transmitted LSP parameters and  $prevLSP2$  represents previously transmitted LSP parameters before  $prevLSP1$ . For "Background Noise" frame ( $idVUV = 1$ ), VXC decoder is used in the same manner as UV frame, but no LSP parameters are sent. LSP parameters generated by linearly interpolating  $prevLSP1$  and  $prevLSP2$  are used for LPC synthesis, and the same gain index as the previous frame is used for excitation generation of VXC decoding. During the period of "Background Noise" frame, "Unvoiced speech (UV)" frame is inserted every ( $BGN\_INTVL + 1$ ) (=9) frames to transmit background noise parameters. This UV frame may or may not be a real UV frame of the beginning of speech bursts. Whether or not the frame is real UV is judged by the transmitted gain index. If the gain index is smaller than or equal to that of previous one+2, then this UV frame is regarded as "Background Noise" frame, and therefore the previously transmitted LSP vector (=  $prevLSP1$ ) is used to keep the smooth variation of LSP parameters, otherwise; currently transmitted LSPs are used as a real UV frame. The gain indices are sorted according to the magnitudes. If "Background Noise" mode is selected again, then linearly interpolated LSPs using  $prevLSP1$  and  $prevLSP2$  are used.

For both "Unvoiced speech" and "Background Noise" frame ( $idVUV = 0,1$ ), Gaussian noise with a unit energy is used for excitation of VXC decoding (in place of stochastic shape codevector for VXC decoding).

Figure 2.13 shows an example. Suppose that Frame #0 and Frame #1 are "Unvoiced speech" frame, and Frame #2...Frame #9 are "Background Noise" frame. During decoding of Frame #2...Frame #9,  $prevLSP1$  and  $prevLSP2$  are set as:  $prevLSP1 = LSP(1)$  and  $prevLSP2 = LSP(0)$  and LSP vector of Frame#  $i$ ,  $LSP(i)$  ( $2 \leq i \leq 9$ ), are generated as

$$LSP(i) = \frac{prevLSP2 \times (2 \times BGN\_INTVL - 2 \times bgnCnt - 1) + prevLSP1 \times (2 \times bgnCnt + 1)}{2 \times BGN\_INTVL}$$

where *BGN\_INTVL* is maximum background noise interval (=8) and *bgnCnt* is frame count of successive "Background Noise" frames.

In this example, *bgnCnt* = 0 for Frame #2, *bgnCnt* = 1 for Frame #3, ..., *bgnCnt* = 7 for Frame #9. As for gain index of VXC decoding during Frame #2...Frame #9, the gain index of Frame #1 is used. When parameters of Frame #10 are received, *prevLSP1* and *prevLSP2* are updated as: *prevLSP1* = *LSP(10)* and *prevLSP2* = *LSP(1)*. For decoding of Frame #10, gain index is first checked whether or not it is greater than "Frame #1's index value + 2". If yes, Frame #10 is decoded as usual UV frame; otherwise it is decoded as "Background Noise" frame and *LSP(1)* is used instead of *LSP(10)* while received gain index is used for both cases.

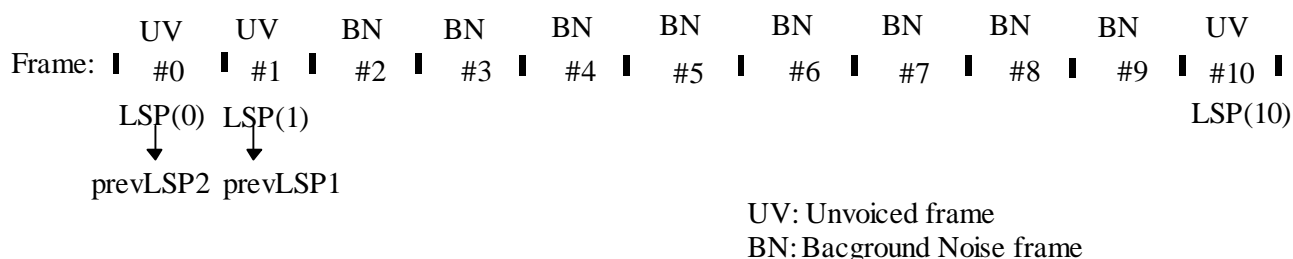


Figure 2.13 - Decoder parameter generation for background noise interval

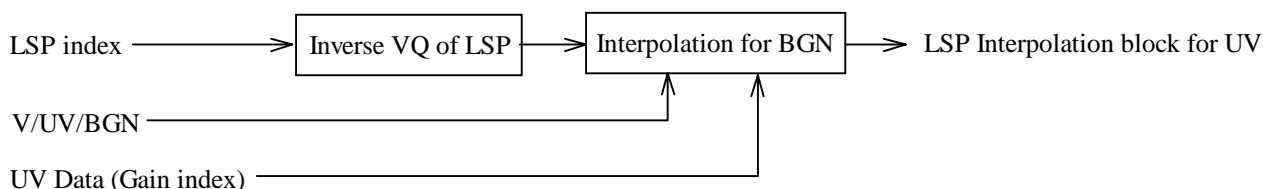


Figure 2.14 - Background noise decoder for variable rate

### 2.5.9 Extension of HVXC variable rate mode

#### 2.5.9.1 Tool description

In subclause 2.5.8, variable bitrate mode based on 2 kbit/s mode is described. Here the operation of the variable bitrate mode of 4.0 kbit/s maximum is described.

In the fixed bitrate mode, we have 2 bit V/UV decision that is:

VUV=3 : full voiced, VUV=2 : mixed voiced, VUV=1 : mixed voiced, VUV=0 : unvoiced.

When the operating mode is variable bitrate mode, VUV=1 indicates "Background noise" status instead of "mixed voiced". The current operating mode is defined by "HVXCconfig()" and decoder knows whether it's variable or fixed rate mode and can understand the meaning of VUV=1. In the "variable rate coding", bit assignment is varied depending on Voiced/unvoiced decision and bitrate saving is obtained mostly by reducing the bit assignment for Unvoiced speech (VUV=0) segment. When VUV=0 is selected, then it is checked whether the segment is real

“unvoiced speech” or “background noise” segments. If it is declared to be “background noise”, then VUV is changed to 1 and bit assignment to the frame is further reduced. During the “background noise” mode, only the mode bits or noise update frame is transmitted according to the change of the background noise characteristics. Using this variable rate mode, average bitrate is reduced to 56-85% of the fixed bitrate mode depending on the source items.

### 2.5.9.2 Definitions

#### Definitions of constants

NUM\_SUBF1: the number of subframes in one frame(=2)

NUM\_SHAPE\_L0: the number of codebook index(=64)

BGN\_INTVL: background noise update interval(=12)

#### Definitions of variables

prevLSP1: transmitted LSP parameters

prevLSP2: transmitted LSP parameters before prevLSP1

qLsp: LSP to be used for decoding operation of the current frame

bgnIntval: a counter which counts the number of consecutive background noise frames

rnd: a randomly generated integer value between -3 and 3

### 2.5.9.3 Transmission Payload

Transmission payloads with four different bitrates are used depending on V/UV decision and the result of background noise detection. VUV flag and UpdateFlag indicate the type of transmission payloads.

VUV is a parameter that has the result of V/UV decision and defined as;

$$VUV = \begin{cases} 0 & \text{Unvoiced speech} \\ 1 & \text{Background noise interval} \\ 2 & \text{Voiced speech 1} \\ 3 & \text{Voiced speech 2} \end{cases}$$

To indicate whether or not the frame marked “VUV=1” is noise update frame, a parameter “UpdateFlag” is introduced. UpdateFlag is used only when VUV=1.

$$\text{UpdateFlag} = \begin{cases} 0 & \text{not noise update frame} \\ 1 & \text{noise update frame} \end{cases}$$

If UpdateFlag is 0, the frame is not noise update frame, and if UpdateFlag is 1, the frame is noise update frame. The first frame of the “Background noise” mode is always classified as the noise update frame. In addition, if the gain or spectral envelope of the background noise frame is changed, a noise update frame is inserted.

At the noise update frame, the average of LSP parameters over the last 3 frames is computed and coded as LSP indices in the encoder. In the same manner, the average of Celp gain over the last 4 frames (8 subframes) is computed and coded as Celp gain index.

During the background noise interval (VUV=1), LSP parameters and excitation parameters are sent only when noise update frame is selected (UpdateFlag=1). Decoder output signals for background noise interval are generated using the LSP and excitation parameters transmitted at noise update frames.

If the current frame or the previous frame is "Background noise" mode, differential mode in LSP quantization is inhibited in the encoder, because LSP parameters are not sent during "Background noise" mode and inter frame coding is not possible.

Using the background noise detection method described above, variable rate coding is carried out based on fixed bitrate 4 kbit/s HVXC. The bitrate at each mode is shown below.

Mode(VUV)	Back Ground Noise(1)		UV(0)	V(2,3)
	UpdateFlag=0	UpdateFlag=1		
V/UV	2bit/20msec	2bit/20msec	2bit/20msec	2bit/20msec
UpdateFlag	1bit/20msec	1bit/20msec	0bit/20msec	0bit/20msec
LSP	0bit/20msec	18bit/20msec	18bit/20msec	26bit/20msec
Excitation		4bit/20msec (gain only)	20bit/20msec	52bit/20msec
Total	3bit/20msec 0.15 kbit/s	25bit/20msec 1.25 kbit/s	40bit/20msec 2.0 kbit/s	80bit/20msec 4.0 kbit/s

**2.5.9.4 Decoding Process**

In the decoder, voiced frame (VUV=2,3) is processed in the same manner as 4 kbit/s fixed bitrate mode, and unvoiced frame (VUV=0) is processed in the same manner as 2 kbit/s fixed bitrate mode. When the background noise mode is selected (VUV=1), decoder output signal is generated in the same manner as unvoiced speech at 2 kbit/s fixed bitrate mode. The decoder parameters for back ground noise interval are generated by using the parameters transmitted at noise update frames (VUV=1, UpdateFlag=1) and sometimes at preceding unvoiced frames (VUV=0). The subclauses below show how to generate the decoder parameters for back ground noise interval.

**2.5.9.4.1 LSP decoding**

In the decoder, two sets of previously transmitted LSP parameters, prevLSP1 and prevLSP2, are held.

prevLSP1: transmitted LSP parameters

prevLSP2: transmitted LSP parameters before prevLSP1

"Background noise" mode occurs only after "unvoiced" or "background noise" mode. When the "background noise" mode is selected, LSP parameters are transmitted only when the frame is "noise update frame" (UpdateFlag=1). If new LSP parameters are transmitted, prevLSP1 is copied to prevLSP2 and newly transmitted LSPs are copied to prevLSP1 regardless of VUV decision.

LSP parameters for each frame during the "background noise" mode are generated by the interpolation between prevLSP1 and prevLSP2 using the equation:

$$qLsp(i) = ratio \cdot prevLsp1(i) + (1 - ratio) \cdot prevLsp2(i) \quad i = 1..10 \quad (2.5.9.1)$$

where

$$ratio = \frac{2 \cdot (bgnIntval + rnd) + 1}{2 \cdot BGN\_INTVL} \quad (2.5.9.2)$$

qLsp(i) is the i-th LSP to be used for decoding operation of the current frame, prevLsp1(i) is the i-th LSP of prevLSP1, prevLsp2(i) is the i-th LSP of prevLSP2 (1 ≤ i ≤ 10). In this equation, bgnIntval is a counter which counts

the number of consecutive background noise frames, and is reset to 0 at the receipt of background noise update frame.  $BGN\_INTVL(=12)$  is a constant, and  $rnd$  is a randomly generated integer value between  $-3$  and  $3$ . If counter  $bgnIntvl$  reaches  $BGN\_INTVL$ ,  $bgnIntvl$  is set to  $BGN\_INTVL-1$ , and if the ratio obtained by the equation (2.5.9.2) is smaller than 0 or greater than 1, the value of  $rnd$  is set to 0 and ratio is recomputed.

### 2.5.9.5 Excitation generation

During the period of “background noise” mode, the Gain index ( $VX\_gain[0]$ ) transmitted in the noise update frame is used for all the subframes, the values of Shape index ( $VX\_Shape1[0,1]$ ) are randomly generated between 0 and  $NUM\_SHAPE\_L0-1$ . These excitation parameters are used with the interpolated LSP parameters as described above to generate the signals of background noise mode.

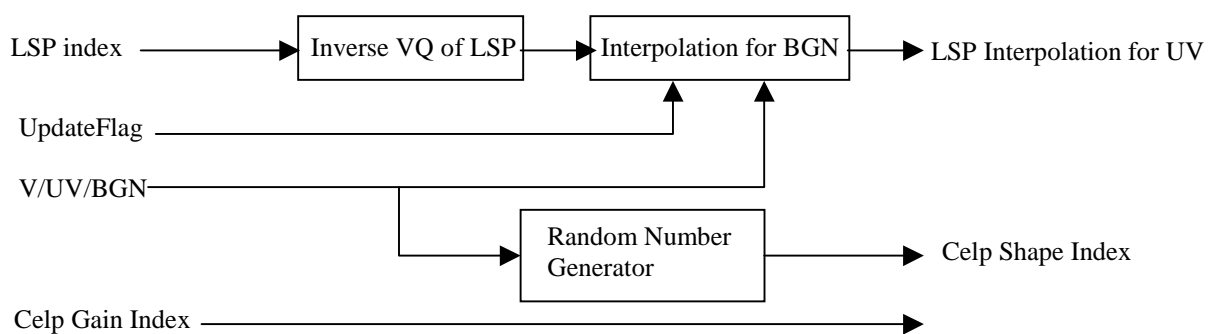


Figure 2.15 - Additional diagram for variable rate decoder

## Annex 2.A (informative)

### HVXC Encoder tools

#### 2.A.1 Overview of encoder tools

Figure 2.A.1 shows the overall structure of the encoder. Speech input at a sampling rate of 8kHz is formed into frames with a length and interval of 256 and 160 samples, respectively. LPC analysis is carried out using windowed input data over one frame. LPC residual signals are computed by the inverse filtering of input data using quantized and interpolated LSP parameters. The residual signals are then fed into the pitch and spectral magnitude estimation block, where the spectral envelopes for LPC residual are estimated in the same manner as in the MBE coder except that only a two-bit V/UV decision is used per frame. The spectral envelope for voiced segment is then vector quantized with weighted distortion measure. For unvoiced segment, a closed loop search for the vector excitation coding is carried out. Detailed configurations are described below.

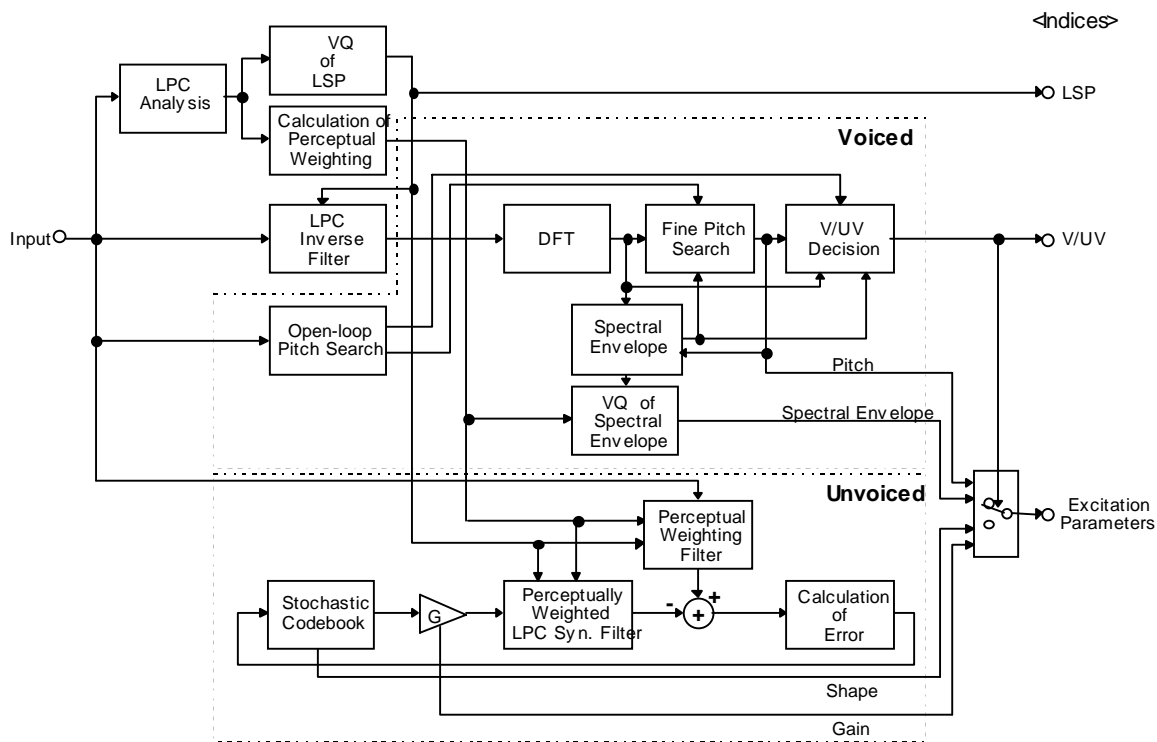


Figure 2.A.1 - Blockdiagram of the HVXC encoder

## 2.A.2 Normalization

### 2.A.2.1 Tool description

The normalization process is composed of three operations, that is, LPC analysis, LSP parameter quantization, and inverse filtering. These operations are described below.

### 2.A.2.2 Normalization process

#### 2.A.2.2.1 LPC analysis

10th order LPC coefficients are computed for every frame, using Hamming-windowed input signals by autocorrelation method.

#### 2.A.2.2.2 LSP quantization

The same LSP quantizer as that of the narrowband CELP is used.

LPC coefficients are first converted to Line Spectral Pair (LSP) parameters. LSP parameters are then quantized with a Vector Quantization (VQ). In case of the base layer, there are two methods for quantizing the LSPs as described in the decoding section; a two-stage VQ without interframe prediction, and a combination of the VQ and the inter-frame predictive VQ. At the encoding process, both methods are used to quantize the LSPs and one of them is selected by comparing the quantization error. The quantization error is calculated as a weighted euclidean distance.

In the case of the enhancement layer, a 10-dimensional vector quantizer, which has an 8 bit codebook, is added to the bottom of the current LSP quantizer scheme of the 2.0 kbps coder. The bit rate of LSPs is increased from 18 bits/20 msec to 26 bits/20 msec.

The encoding process of the base layer is as follows.

The weighting coefficients ( $w[i]$ ) are,

$$w[i] = \begin{cases} \frac{1}{lsp[0]} + \frac{1}{lsp[1] - lsp[0]} & (i = 0) \\ \frac{1}{lsp[i] - lsp[i-1]} + \frac{1}{lsp[i+1] - lsp[i]} & (0 < i < Np - 1) \\ \frac{1}{lsp[Np-1] - lsp[Np-2]} + \frac{1}{1.0 - lsp[Np-1]} & (i = Np - 1) \end{cases}$$

where  $Np$  is the LP analysis order and  $lsp[i]$ s are the converted LSPs.

```
w_fact = 1.;
for (i=0;i<4;i++) w[i] *= w_fact;
for (i=4;i<8;i++) {
    w_fact *= .694;
    w[i] *= w_fact;
}
for (i=8;i<10;i++) {
    w_fact *= .510;
    w[i] *= w_fact;
}
```

The first stage quantizer is the same for each quantization method. The LSPs are quantized by using a vector quantizer and corresponding index is stored in **LSP1**. In order to carry out delayed decision, plural indices are stored as candidates for the second stage. The quantization error in the first stage  $err1[]$  is given by:

$$err1[n] = \sum_{i=0}^{dim-1} \left\{ \left( lsp[sp+i] - lsp\_tbl[n][m][i] \right)^2 \cdot w[sp+i] \right\} \quad n = 0$$

where  $n$  is the split vector number,  $m$  is the index of the candidate split vector,  $sp$  is the starting LSP order of the  $n$ -th split vector and  $dim$  is the dimension of the  $n$ -th split vector. ( $lsp\_tbl[][][]$  is shown in Annex 2.E)

**Table 2.A.1 - Starting order and dimension of the first stage LSP vector**

Split vector number: n	Starting LSP order: sp	Dimension of the vector: dim
0	0	10

In the second stage, above-mentioned two quantization methods, which are two-split vector quantizer, are applied respectively. Total quantization errors in the second stage are calculated for all combinations of the first stage candidates and the second stage candidates and the one which has the minimum error is selected. As a result, indices of the first stage are determined and corresponding indices and signs for the second stage are stored in **LSP2** and **LSP3**. The flag which indicates the selected quantization method is also stored in **LSP4**. The quantization error in the second stage  $err2\_total$  is given by:

VQ without interframe prediction:

$$err2\_total = err2[0] + err2[1]$$

$$err2[n] = \sum_{i=0}^{dim-1} \left\{ \left( lsp\_res[sp+i] - sign[n] \cdot d\_tbl[n][m][i] \right)^2 \cdot w[sp+i] \right\} \quad n = 0,1$$

$$lsp\_res[sp+i] = lsp[sp+i] - lsp\_first[sp+i]$$

where  $lsp\_first[]$  is the quantized LSP vector of the first stage,  $n$  is the split vector number,  $m$  is the index of the candidate split vector,  $sp$  is the starting LSP order of the  $n$ -th split vector and  $dim$  is the dimension of the  $n$ -th split vector. ( $d\_tbl[][][]$  is shown in Annex 2.E.)

VQ with interframe prediction:

$$err2\_total = err2[0] + err2[1]$$

$$err2[n] = \sum_{i=0}^{dim-1} \left\{ \left( lsp\_pres[sp+i] - sign[n] \cdot pd\_tbl[n][m][i] \right)^2 \cdot w[sp+i] \right\} \quad n = 0,1$$

$$lsp\_pres[sp+i] = lsp[sp+i] - \left\{ (1 - ratio\_predict) \cdot lsp\_first[sp+i] + ratio\_predict \cdot lsp\_previous[sp+i] \right\}$$

where  $lsp\_first[]$  is the quantized LSP vector of the first stage,  $n$  is the split vector number,  $m$  is the index of the candidate split vector,  $sp$  is the starting LSP order of the  $n$ -th split vector,  $dim$  is the dimension of the  $n$ -th split vector and  $ratio\_predict=0.7$ . ( $pd\_tbl[][][]$  is shown in Annex 2.E.)



**Table 2.A.2 - Starting order and dimension of the second stage LSP vector**

Split vector number: n	Starting LSP order: sp	Dimension of the vector: dim
0	0	5
1	5	5

The quantized LSPs *lsp\_current[]* are stabilized in order to ensure stability of the LPC synthesis filter which is derived from the quantized LSPs. The quantized LSPs are arranged in ascending order, having a minimum distance between adjacent coefficients.

```

for (i=0;i<LPCORDER;i++) {
    if (lsp_current[i] < min_gap) lsp_current[i] = min_gap;
}
for (i=0;i<LPCORDER-1;i++) {
    if (lsp_current[i+1]-lsp_current[i] < min_gap) {
        lsp_current[i+1] = lsp_current[i]+min_gap;
    }
}
for (i=0;i<LPCORDER;i++) {
    if (lsp_current[i] > 1-min_gap) lsp_current[i] = 1-min_gap;
}
for (i=LPCORDER-1;i>0;i--) {
    if (lsp_current[i]-lsp_current[i-1] < min_gap) {
        lsp_current[i-1] = lsp_current[i]-min_gap;
    }
}

```

```

for (i=0;i<LPCORDER;i++){
    qLsp[i] = lsp_current[i];
}

```

where  $\text{min\_gap} = 4.0/256.0$

After the LSP encoding process, the current LSPs have to be stored in memory, since they are used for prediction at the next frame.

```

for (i=0;i<LPCORDER;i++) {
    lsp_previous[i] = lsp_current[i];
}

```

It must be noted that the stored LSPs *lsp\_previous[]* must be initialized as described below when the whole of the encoder is initialized.

```

for (i=0;i<LPCORDER;i++) {
    lsp_previous[i] = (i+1) / (LPCORDER+1);
}

```

The third stage for the enhancement layer (4.0 and 3.7 kbps) has 10-dimensional VQ structure. The error between the quantized version of the base layer and the original version is quantized and corresponding index is stored in **LSP5**.

After the quantization, the LSPs of the enhancement layer *qLsp[]* are stabilized again.

```

for (i = 0; i < 2; i++)
{
    if (qLsp[i + 1] - qLsp[i] < 0)
    {
        tmp = qLsp[i + 1];
    }
}

```

```

        qLsp[i + 1] = qLsp[i];
        qLsp[i] = tmp;
    }

    if (qLsp[i + 1] - qLsp[i] < THRSLD_L)
    {
        qLsp[i + 1] = qLsp[i] + THRSLD_L;
    }
}

for (i = 2; i < 6; i++)
{
    if (qLsp[i + 1] - qLsp[i] < THRSLD_M)
    {
        tmp = (qLsp[i + 1] + qLsp[i]) / 2.0;
        qLsp[i + 1] = tmp + THRSLD_M / 2.0;
        qLsp[i] = tmp - THRSLD_M / 2.0;
    }
}

for (i = 6; i < LPCORDER - 1; i++)
{
    if (qLsp[i + 1] - qLsp[i] < 0)
    {
        tmp = qLsp[i + 1];
        qLsp[i + 1] = qLsp[i];
        qLsp[i] = tmp;
    }

    if (qLsp[i + 1] - qLsp[i] < THRSLD_H)
    {
        qLsp[i] = qLsp[i + 1] - THRSLD_H;
    }
}

```

where, THRSLD\_L=0.020, THRSLD\_M=0.020 and THRSLD\_H=0.020

**Table 2.A.3 - Configuration of the multistage LSP VQ**

1st stage	10 LSP VQ	5bits
2nd stage	(5+5)LSP VQ	(7+5+1)bits
3rd stage	10LSP VQ	8bits

**2.A.2.2.3 LPC inverse filter**

LSPs are converted to alpha parameters to form a LPC inverse filter in a direct form. LPC residual signals are then computed by inverse filtering the input signal. Only the current frame's quantized LPC is used without any interpolation for the inverse filtering to compute the LPC residual signal. The residual signal is then 256 point Hamming windowed to compute the power spectrum. The transfer function of the LPC inverse filter is,

$$A(z) = \sum_{n=0}^P \alpha_n z^{-n}$$

where  $P=10$  and  $\alpha_0=1$ . LPC coefficients  $\alpha_n$  stay unchanged during the computation of residual samples of one frame length (256 samples).

## 2.A.3 Pitch estimation

### 2.A.3.1 Tool description

To obtain the first estimation of the pitch lag value, the autocorrelation values of the LPC residual signals are computed. Based on the lag values which give the peaks in autocorrelation, the open loop pitch is estimated. Pitch tracking is carried out in the process of pitch estimation so that the estimated pitch gets more reliable.

### 2.A.3.2 Pitch estimation process

In the low delay mode encoder, pitch tracking is conducted using only current and past frames to keep encoder delay 26 ms. When the normal delay mode is used, pitch tracking uses one frame ahead and the encoder delay becomes 46 ms.

### 2.A.3.3 Pitch tracking

For the low delay mode of HVXC, pitch tracking algorithm does not use the pitch value of the future (look ahead) frame. The pitch tracking algorithm operates based on reliable pitch "rbIPch", V/UV decision of the previous frame "prevVUV", and past/current pitch parameters.

The basic operation is as follows:

- When prevVUV != 0 and rbIPch !=0, pitch is tracked based on rbIPch. However, the pitch value of the previous frame has higher priority.
- When prevVUV = 0 and rbIPch !=0, pitch is tracked based on rbIPch.
- When prevVUV !=0 and rbIPch = 0, tracking is conducted based on the pitch of the previous frame.
- When prevVUV = 0 and rbIPch = 0, current pitch parameter is simply used.

prevVUV !=0 represents Voiced, and prevVUV = 0 represents Unvoiced status respectively.

By this strategy, pitch tracking is carried out without any look ahead. Source code of the pitch tracking is shown below.

```
typedef struct
{
    float pitch;      /* pitch */
    float prob;       /* 1st peak divided by 2nd peak of autocorrelation */
    float r0r;        /* 1st peak of autocorrelation - modified */
    float rawR0r;     /* 1st peak of autocorrelation - raw */
    float rawPitch;   /* pitch with no tracking */
}
NgbPrm;

static int NearPitch(
float p0,
float p1,
float ratio)
{
    return((p0 * (1.0 - ratio) < p1) && (p1 < p0 * (1.0 + ratio)));
}

static float TrackingPitch(
NgbPrm *crntPrm, /* current parameter */
NgbPrm *prevPrm, /* previous parameter */
int *scanlimit, /* Number of autocorrelation peaks */
float *ac, /* Autocorrelation */
int peakPos[PEAKMAX], /* Position of autocorrelation peaks */
int prevVUV) /* V/UV of pprevious frame */
```

```

{
float kimete;
float pitch;
int i;
static float prevRawp= 0.0;
int st0, st1, st2;
float stdPch;
static float rblPch = 0.0;
static float prevRblPch = 0.0;

rblPch = global_pitch;
if (prevVUV != 0 && rblPch != 0.0)
{
st0 = Ambiguous(prevPrm->pitch, rblPch, 0.11);
st1 = Ambiguous(crntPrm->pitch, rblPch, 0.11);
if (!(st0 || st1))
{
if (NearPitch(crntPrm->pitch, prevPrm->pitch, 0.2))
pitch = crntPrm->pitch;
else if (NearPitch(crntPrm->pitch, rblPch, 0.2))
pitch = crntPrm->pitch;
else if (NearPitch(prevPrm->pitch, rblPch, 0.2))
{
if (crntPrm->r0r > prevPrm->r0r && crntPrm->prob > prevPrm->prob)
pitch = crntPrm->pitch;
else
pitch = prevPrm->pitch;
}
else
pitch = GetStdPch2Elms(crntPrm, prevPrm, scanlimit, peakPos, ac);
}
else if (!st0)
{
if (NearPitch(prevPrm->pitch, crntPrm->pitch, 0.2))
pitch = crntPrm->pitch;
else if ((gpMax * 1.2 > crntPrm->pitch) &&
NearPitch(prevPrm->rawPitch, crntPrm->pitch, 0.2))
pitch = crntPrm->pitch;
else
pitch = prevPrm->pitch;
}
else if (!st1)
{
if ((crntPrm->rawPitch != crntPrm->pitch) &&
NearPitch(crntPrm->rawPitch, prevPrm->rawPitch, 0.2))
pitch = crntPrm->rawPitch;
else
pitch = crntPrm->pitch;
}
else
{
if (NearPitch(prevPrm->pitch, crntPrm->pitch, 0.2))
pitch = crntPrm->pitch;
else
pitch = rblPch;
}
}
else if (prevVUV == 0 && rblPch != 0.0)
{
st1 = Ambiguous(crntPrm->pitch, rblPch, 0.11);
if (!st1)
pitch = crntPrm->pitch;
else
pitch = rblPch;
}
else if (prevVUV != 0 && rblPch == 0.0)
{
st1 = Ambiguous(crntPrm->pitch, prevPrm->pitch, 0.11);

```

```

if (!st1)
    pitch = GetStdPch2Elms(crntPrm, prevPrm, scanlimit, peakPos, ac);
else
{
    if (prevPrm->r0r < crntPrm->r0r)
        pitch = crntPrm->pitch;
    else
        pitch = prevPrm->pitch;
}
}
else
pitch = crntPrm->pitch;

crntPrm->pitch = pitch;
prevRblPch = rblPch;
prevRawp = pitch;
return(pitch);
}

```

## 2.A.4 Harmonic magnitudes extraction

### 2.A.4.1 Tool description

Harmonic magnitudes extraction consists of two steps: fine pitch search and estimation of the spectral envelope. The operation of each step is described below.

### 2.A.4.2 Harmonic magnitudes extraction process

#### 2.A.4.2.1 Fine pitch search

Using the open loop integer pitch lag, the fractional pitch lag value is estimated here. The step size of the fraction is 0.25. This is carried out by minimizing the error between the synthesized spectrum and original spectrum. Here, pitch lag value and spectral harmonic magnitudes are estimated simultaneously. Estimated pitch lag value,  $pch$ , is transmitted to the decoder as **Pitch**.

$$Pitch = (int)(pch - 20.0)$$

#### 2.A.4.2.2 Estimation of spectral envelope

A 256 point DFT is applied to the LPC residual signals to obtain the original spectrum. Using the original spectrum, estimation of the spectral envelope is carried out in the part of the above mentioned fine pitch search. The spectral envelope is a set of spectral magnitudes estimated at each harmonic. This magnitude estimation is carried out by computing an optimal amplitude  $A_m$  using pre-defined basis function  $E(j)$ , and original spectrum  $X(j)$ . Let  $a_m$  and  $b_m$  be the indices of DFT coefficient of lower and higher boundary of the  $m$ -th band respectively, covering  $m$ -th harmonic band. The amplitude estimation error  $\epsilon_m$  is defined as:

$$\epsilon_m = \sum_{j=a_m}^{b_m} \left( |X(j)| - |A_m| |E(j)| \right)^2$$

Solving

$$\frac{\partial \epsilon_m}{\partial |A_m|} = 0$$

gives

$$|A_m| = \frac{\sum_{j=a_m}^{b_m} |X(j)| |E(j)|}{\sum_{j=a_m}^{b_m} |E(j)|^2}$$

This value is defined as spectral magnitude. Basis function  $E(j)$  can be obtained by DFT of 256 point Hamming window.

## 2.A.5 Perceptual weighting

### 2.A.5.1 Tool description

The frequency response of a perceptual weighting filter is computed which is for the use of weighted vector quantization of the harmonic spectral envelope. Here, a perceptual weighting filter is derived from linear predictive coefficients  $\alpha_n$ . The transfer function of the perceptual weighting filter is;

$$w(z) = \frac{\sum_{n=0}^P \alpha_n A^n z^{-n}}{\sum_{n=0}^P \alpha_n B^n z^{-n}}$$

where  $A = 0.9$   $B = 0.4$  could be used. In this tool, the frequency response of the LPC synthesis filter  $h(z)$  is also computed so that it could be incorporated where,

$$h(z) = \frac{1}{\sum_{n=0}^P \alpha_n z^{-n}}$$

In this tool, frequency response of  $w(z)h(z)$  is computed and outputted as an array \*per\_weight, which could be used as diagonal components of the weighting matrices  $WH$ .

## 2.A.6 Harmonic VQ encoder

### 2.A.6.1 Tool description

The harmonic VQ encoding process consists of two steps: dimension conversion and vector quantization of the spectral envelope vectors. The operations of each step are given below

## 2.A.6.2 Encoding process

### 2.A.6.2.1 Dimension converter

The number of points which composes the spectral envelope varies depending on pitch values, since the spectral envelope is a set of the estimates of the magnitudes at each harmonic. The number of harmonics ranges from about 9 to 70.

In order to vector quantize the spectral envelope, the coder has to convert them to a constant number for a fixed-dimension VQ. A band-limited interpolation is used for the sampling frequency conversion to obtain the fixed-dimension spectral vectors. The number of points, which represent the shape of the spectral envelope, should be modified without changing the shape. For this purpose, a dimension converter for a spectral envelope by a combination of low pass filter and 1st order linear interpolator is used. An FIR low pass filter with 7 sets of coefficients, each set consisting of 8 coefficients, is used for the first stage 8-times over-sampling. The 7 sets of the filter coefficients are obtained by grouping 8 every coefficients from a windowed sinc,  $coef[i]$ , with the offsets of 1 through 7, where

$$coef[i] = \frac{\sin \pi(i-32)/8}{\pi(i-32)/8} (0.5 - 0.5 \cos 2\pi i / 64) \quad 0 \leq i \leq 64$$

This FIR filtering allows decimated computation, in which only the points used at the next stage are computed. They are the left and right adjacent points of the final output of the dimension converter.

At the second over-sampling stage, 1st order linear interpolation is applied to obtain the necessary output points. In this way, we get fixed-dimension (= 44) spectral vectors.

### 2.A.6.2.2 Vector quantization

A fixed-dimension (= 44) spectral vector is then quantized. In the base layer, a two-stage vector quantization scheme is employed for the spectral shape together with a scalar quantizer for the gain. The weighted distortion measure  $D$  is used for the codebook search of both shape and gain.

$$D = \|\mathbf{WH}(\mathbf{x} - g(\mathbf{s}_1 + \mathbf{s}_2))\|^2$$

where  $x$  is a source vector,  $s_1$  is the output of Spectral Envelope (SE) shape1 codebook,  $s_2$  is the output of SE shape2 codebook, and  $g$  is the output of the SE gain codebook. Corresponding indices are denoted as **SE\_shape1**, **SE\_shape2** and **SE\_gain** respectively. The dimension of the shape codebooks is fixed (=44). The diagonal components of the matrices  $\mathbf{H}$  and  $\mathbf{W}$  are the magnitudes of the frequency response of the LPC synthesis filter and the perceptual weighting filter, respectively. For enhancement layer, additional vector quantizers are added to the bottom of the quantizer of the base layer. The 2.0kbps mode uses only the quantizers of the base layer.

For 4.0kbps mode, the quantized harmonic magnitudes with fixed dimension (=44) at the base layer is first converted to the dimension of the original harmonic vector, which varies depending on the pitch value. The difference between the quantized/dimension recovered harmonic vector and the original harmonic vector is computed. This difference is then quantized with a split VQ scheme composed of four vector quantizers at the enhancement layer. The corresponding indices are **SE\_shape3**, **SE\_shape4**, **SE\_shape5** and **SE\_shape6**.

Index **SE\_shape6** is not used when 3.7kbps mode is selected.

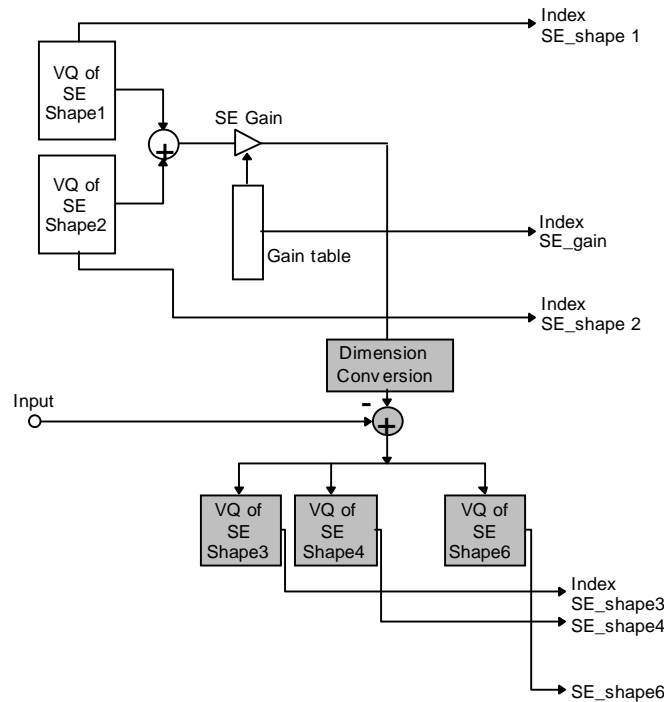


Figure 2.A.2 - Vector quantization of spectral envelope

## 2.A.7 V/UV decision

### 2.A.7.1 Tool description

A V/UV decision is made every 20 ms frame. The decision is made based on: similarity of the shape of the synthesized spectrum and original spectrum, signal power, maximum autocorrelation of LPC residual signals normalized by residual signal power, and number of zero crossing.

### 2.A.7.2 Encoding process

The V/UV decision is composed of three different modes, that is, Unvoiced, Mixed Voiced, and Fully Voiced.

In order to send out the information of the three modes, two bits are used for V/UV. First, a decision is made whether or not the current frame is Unvoiced. When this decision is Voiced, then the voicing strength is evaluated from the value of the normalized maximum peak of the autocorrelation of LPC residual signal. Let us denote the value  $r0r$ .

Shown below is the decision rule :

When the first decision is Unvoiced V/UV is 0

- Unvoiced



When the first decision is Voiced	V/UV is 1	for $r_{0r} < TH2$	- Mixed Voiced 1
	V/UV is 2	for $TH2 \leq r_{0r} < TH1$	- Mixed Voiced 2
	V/UV is 3	for $TH1 \leq r_{0r}$	- Fully Voiced

Example:

TH1=0.7

TH2=0.5

The number 0,1,2,3 is send out to the decoder as index **VUV** using the two bits.

## 2.A.8 Time domain encoder

### 2.A.8.1 Tool description

When a speech segment is unvoiced, Vector Excitation Coding (VXC) algorithm is used. Figure 2.A.3 shows the overall structures of the VXC encoder. The operation of the VXC is described below.

### 2.A.8.2 Encoding process

LPC analysis is carried out first, and LPC coefficients ( $\alpha$ ) are then converted to LSP parameters in the same manner as in the voiced case. LSP parameters are quantized, and quantized LSPs are converted to LPC coefficients ( $\hat{\alpha}$ ).

The perceptually weighted LPC synthesis filter  $H(z)$  is expressed as:

$$H(z) = A(z)W(z)$$

where  $A(z)$  is a transfer function of the LPC synthesis filter, and  $W(z)$  is a perceptual weighting filter derived from LPC coefficients.

Let  $x_w(n)$  be perceptually weighted input signal. Subtracting zero-input response  $z(n)$  from  $x_w(n)$ , we obtain reference signal,  $r(n)$ , for the analysis-by-synthesis procedure of the VXC. Optimal shape and gain vectors are searched using the distortion measure  $E$ :

$$E = \sum_{n=0}^{N-1} (r(n) - g \times \text{syn}(n))^2$$

where  $\text{syn}(n)$  is zero-state response of  $H(z)$ , driven by a excitation input by only a shape vector  $s(n)$ , which is an output of VX\_shape codebook.  $g$  is a gain, which is an output of VX\_gain codebook.  $N$  is vector dimension of VX\_shape codebook.

The codebook search process for the VXC consists of two steps, that are:

1. Search  $s(n)$  that maximize

$$E_s = \frac{\sum_{n=0}^{N-1} r(n) \times \text{syn}(n)}{\sqrt{\sum_{m=0}^{N-1} \text{syn}(m)^2}}$$

2. Search  $g$  that minimize

$$E_g = (g_{ref} - g)^2$$

where

$$g_{ref} = \frac{\sum_{n=0}^{N-1} r(n) \times \text{syn}(n)}{\sum_{m=0}^{N-1} \text{syn}(m)^2}$$

The quantization error  $e(n)$  is computed as:

$$e(n) = r(n) - g \times \text{syn}(n)$$

When the bit-rate is 4 kbps, one more stage is used for the quantization of unvoiced segments, and  $e(n)$  is used as reference input to the second stage VQ.

The operation of the second stage VQ is the same as that of the first stage VQ.

The 2.0 kbps coder uses 6bits shape and 4 bit gain codebooks for the unvoiced excitation for every 10 msec. The corresponding indices are **VX\_shape[i]**, **VX\_gain[i]** (i=0,1). The 4.0 kbps scheme adds 5 bits shape and 3 bits gain codebooks for every 5 msec at the bottom of the current quantizer. The corresponding indices are **VX\_shape2[i]**, **VX\_gain2[i]** (i=0,1,2,3). 3.7 kbps mode may use the same encoding procedure as that of 4.0kbps mode though VX\_shape2[3] and VX\_gain2[3] are not used in the decoder at 3.7kbps mode.

**Table 2.A.4 - Configuration of the VXC codebooks**

1st stage	(80dimension 6bits shape + 4bits gain) x 2
2nd stage	(40dimension 5bits shape + 3bits gain) x 4

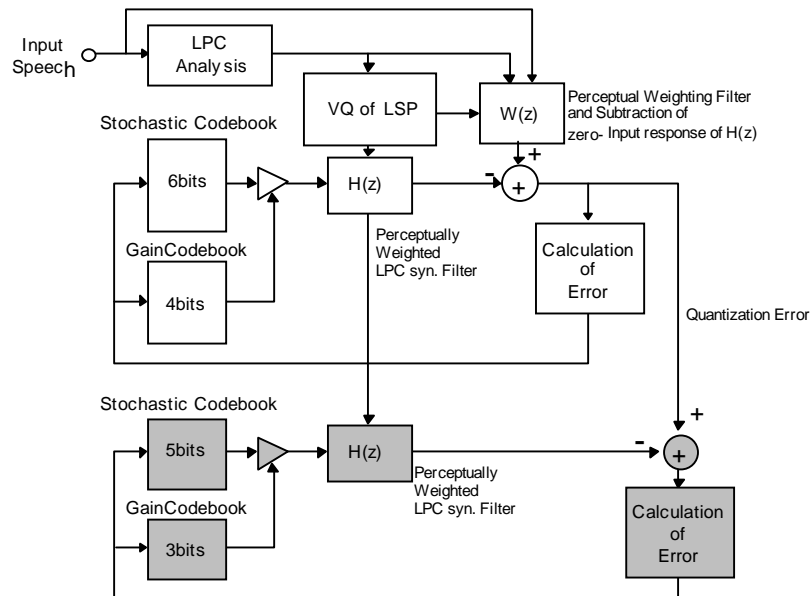


Figure 2.A.3 - Vector Excitation Coding (VXC) for unvoiced signals

### 2.A.9 Variable rate encoder

This subclause describes a tool for variable rate coding with the HVXC core. This tool allows HVXC to operate at variable bit rates. The major part of the algorithm is composed of "background noise interval detection", where only the mode bits are transmitted during the "background noise mode", and unvoiced frame is inserted with certain period of time to send parameters for background noise generation

In the encoding algorithm, minimum level tracker has temporal minimum level in order to adjust a threshold value by which a decision is made about whether or not the input segment is speech.

#### Minimum level tracking

Parameters are defined as follows:

lev: r.m.s. of a speech frame

vCont: number of continuous voiced frames

cdLev: candidate value of minimum level

prevLev: r.m.s. of a previous frame

gmlSetState: number of frames in which the candidate value is set

gmlResetState: number of frames in which the candidate value is not set after setting of minimum level

gml: minimum level

The minimum level is tracked according to the algorithm shown below.

```
if (vCont > 4)
```

```

{
    cdLev = 0.0;
    gmlSetState = 0;
    gmlResetState++;
}
else if (lev < MIN_GML)
{
    *gml = MIN_GML;
    gmlResetState = 0;
    gmlSetState = 0;
}
else if (*gml > lev)
{
    *gml = lev;
    gmlResetState = 0;
    gmlSetState = 0;
}
else if ((lev < 500.0 && cdLev * 0.70 < lev && lev < cdLev * 1.30) || lev < 100.0)
{
    if (gmlSetState > 6)
    {
        *gml = lev;
        gmlResetState = 0;
        gmlSetState = 0;
    }
    else
    {
        cdLev = lev;
        gmlResetState = 0;
        gmlSetState++;
    }
}
else if ((lev < 500.0 && prevLev * 0.70 < lev && lev < prevLev * 1.30) || lev < 100.0)
{
    cdLev = lev;
    gmlResetState = 0;
    gmlSetState++;
}
else if (gmlResetState > 40)
{
    *gml = MIN_GML;
    gmlResetState = 0;
    gmlSetState = 0;
}
else
{
    cdLev = 0.0;
    gmlSetState = 0;
    gmlResetState++;
}
}

```

As shown above, the minimum level is held during appropriate period of time and updated. The minimum level is always set higher than a predetermined value MIN\_GML.

**Background noise detection based on the minimum level**

The reference level *refLev* is computed as,

$$refLev = A \times \max(lev, refLev) + (1.0 - A) \times \min(lev, refLev) \quad (1)$$

Usually *A* is set to 0.75. Background noise detection is carried out using the *refLevel* derived from equation (1).

For the frames where "voiced" decision is made:

```
if (refLev < B *gml && contV < 2) {
    idVUV = 1;
}
```

For the frames where "unvoiced" decision is made:

```
if (refLev < B *gml) { /* condition-1 */
    if (bgnCnt < 3) {
        bgnCnt++;
    }
    else {
        if (bgnIntvl < 8) {
            idVUV=1;
            bgnIntvl++;
        }
        else {
            bgnIntvl=0;
        }
    }
}
else {
    bgnCnt=0;
}
```

where *B* is a constant. In this case we set *B*=2.0

Each parameter is defined below.

*countV* : number of consecutive voiced frames

*bgnCnt*: number of frames which satisfies the condition-1

*bgnIntvl*: number of frames where "Background noise" mode is declared

*idVUV* is a parameter that has the result of V/UV decision and defined as;

$$idVUV = \begin{cases} 0 & \text{Unvoiced speech} \\ 1 & \text{Background noise interval} \\ 2 & \text{Mixed voiced speech} \\ 3 & \text{Voiced speech} \end{cases}$$

If the current frame is declared to be "Voiced", it is checked whether or not the previous frame is "Voiced". If the previous frame is "Voiced", "backgroundnoise" mode is not selected; otherwise, "background noise" mode is selected.

If the current frame is declared to be "Unvoiced", "background noise" mode is selected only after the condition-1 is satisfied for four consecutive frames. When "background noise" mode is selected for consecutive N frames, then the last frame is replaced with "Unvoiced" mode in order to transmit speech parameters which represents characteristics of the time varying back ground noise.

At this moment, N is set to 9.

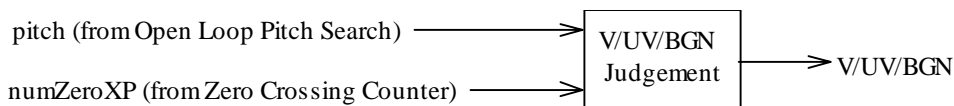
**Variable rate encoding:**

Using the background noise detection method described above, variable rate coding is carried out based on fixed bit rate 2kbps HVXC.

**Table 2.A.5 - Bit allocations of the encoded parameters for the variable bit rate mode**

Mode (idVUV)	Back Ground Noise (1)	UV (0)	MV (2), V (3)
V/UV	2bit/20msec	2bit/20msec	2bit/20msec
LSP	0bit/20msec	18bit/20msec	18bit/20msec
Excitation	0bit/20msec	8bit/20msec (gain only)	20bit/20msec
Total	2bit/20msec 0.1kbps	28bit/20msec 1.4kbps	40bit/20msec 2.0kbps

If the current frame or the previous frame is "Background noise" mode, differential mode in LSP quantization is inhibited in the encoder because LSP parameters are not sent during "Background noise" mode and inter frame coding is not possible.



**Figure 2.A.4 - Block diagram for variable rate encoding**

**2.A.10 Extension of HVXC variable rate encoder**

In subclause 2.A.9, the encoder operation of the variable rate mode of 2.0 kbit/s maximum is described. Here in this subclause one example of the implementation of HVXC variable rate mode encoder of 4.0 kbit/s maximum is described. Basically any kind of background noise/unvoiced speech decision algorithm could be used. The change of the signal level and the spectral envelope are used to detect background noise interval from unvoiced frames. During the noise interval, it is assumed that the signal level and the shape of the spectral envelope are stable. The log squared magnitude response at low frequency range of the spectral envelope is computed from LPC cepstrum coefficients. The log squared magnitude response of the current frame is compared with the average of the log magnitude response over several past frames. If the difference is small, it is assumed that the signal is stable. When these parameters show the signal condition is stable enough, the frame is classified as "background noise interval". If the signal characteristic is changed in certain range, it is assumed that the background noise characteristic is changed, and noise update frame is transmitted. If the signal characteristic is changed more than pre-defined range, then signal is assumed to be unvoiced speech.

**2.A.10.1 Definitions**

- stFlag: signal stability flag
- bgnCnt: background noise counter

BgnIntvl: background noise interval counter  
 UpdateFlag : background noise update flag

### 2.A.10.2 RMS computation

RMS of the input signal  $s[]$  is first computed to obtain minimum signal level ( $\text{min\_rms}$ ). If the RMS value is smaller than the predefined value that is the smallest possible speech level, and the deviation of RMS values of several frames are in certain range, then the minimum level is smoothly updated using the detected RMS value and the current  $\text{min\_rms}$ . The current RMS value " $\text{rms}$ " is then divided by current  $\text{min\_rms}$  to obtain the value " $\text{ratio}$ ".

$$\text{ratio} = \frac{\text{rms}}{\text{min\_rms}}$$

This value is used for signal stability detection as described in 2.A.10.4.

### 2.A.10.3 Spectral comparison

The spectral envelope is computed using Linear Prediction (LP) coefficients. LP coefficients are converted to cepstrum parameters  $C_L[]$ . From cepstrum parameters, log squared magnitude response  $\ln|H_L(e^{j\Omega})|^2$  is computed as:

$$\ln|H_L(e^{j\Omega})|^2 = 2 \cdot \sum_{m=0}^M C_L(m) \cos(\Omega m)$$

The average level of each of the frequency bands at the n-th frame is computed as:

$$\log \text{Amp}(n, i) = \frac{1}{\omega} \int_{\Omega_i}^{\Omega_{i+1}} \ln|H_L(e^{j\Omega})|^2 d\Omega = \frac{1}{\omega} \left[ 2 \sum_{m=1}^M \frac{1}{m} C_L(m) \sin(\Omega m) \right]_{\Omega_i}^{\Omega_{i+1}} \quad i=0\dots3$$

Here,  $\omega$  is the band width (=500Hz), and  $i$  is the band number where 4 bands between 0 and 2kHz are computed. From the obtained log squared magnitude values of the last 4 frames, averaged values for each band is computed as:

$$\text{aveAmp}(n, i) = \frac{1}{4} \sum_{j=1}^4 \log \text{Amp}(n - j, i)$$

Using these equations, The difference " $\text{wdif}$ " between the current level and averaged level for each of the frequency band  $i$  is computed as shown below.

$$\text{wdif} = \sqrt{\frac{1}{4} \sum_{i=0}^3 (\log \text{Amp}(n, i) - \text{aveAmp}(n, i))^2}$$

### 2.A.10.4 Signal stability detection

From the combination of  $\text{ratio}$  and  $\text{wdif}$ , signal stability flag " $\text{stFlag}$ " which has the following 3 states is computed. Naturally, the smaller these variables are, the more stable the signal state is.

$$\text{stFlag} = \begin{cases} 0 & \text{Almost stable} \\ 1 & \text{Unstable} \\ 2 & \text{A little stable} \end{cases}$$

### 2.A.10.5 Background noise detection

According to the variable stFlag and the V/UV decision, background noise is detected. The flowchart is shown in Figure 2.A.6 - 2.A.6. When the V/UV decision is voiced (VUV=2,3), the frame is classified as voiced regardless as the value of stFlag. When background noise is detected in unvoiced interval (VUV=0), VUV is set to 1. When background noise parameters have to be updated, "UpdateFlag" is set to 1. In order to secure stable operation of the background noise detection algorithm, the variables, background noise counter (bgnCnt) and background noise interval counter (bgnIntvl), are introduced. BGN\_CNT and BGN\_INTVL are predefined constants.

### 2.A.10.6 Transmitted parameters decision

VUV is a parameter that has the result of V/UV decision and defined as;

$$\text{VUV} = \begin{cases} 0 & \text{Unvoiced speech} \\ 1 & \text{Background noise interval} \\ 2 & \text{Voiced speech 1} \\ 3 & \text{Voiced speech 2} \end{cases}$$

In order to indicate whether or not the frame marked "VUV=1" is noise update frame, a parameter "UpdateFlag" is introduced. UpdateFlag below is used only when VUV=1.

$$\text{UpdateFlag} = \begin{cases} 0 & \text{not noise update frame} \\ 1 & \text{noise update frame} \end{cases}$$

At noise update frame, averaged LSP and Celp Gain parameters are computed.

As for LSP, raw LSPs are averaged around the last 3 frames.

$$\text{aveLsp}[n][i] = \frac{1}{3} \sum_{j=0}^2 \text{Lsp}[n-j][i], \quad i=1, \dots, \text{NP}$$

where  $\text{aveLsp}[n][i]$  denotes the averaged LSP at the n-th frame and  $\text{Lsp}[n][i]$  denotes the raw LSP at the n-th frame. NP is the order of LSP.  $\text{aveLsp}[n][i]$  is quantized and coded, where LSP coding by differential mode is inhibited.

As for Celp gain, raw Celp gains around the last 4 frames(8 subframes) are averaged.

$$\text{aveGain}[n] = \frac{1}{8} \sum_{j=0}^3 \sum_{k=0}^1 \text{Gain}[n-j][k]$$

where  $\text{aveGain}[n]$  is the averaged Celp gain at the n-th frame and  $\text{Gain}[n][k]$  is the raw Celp gain at the n-th frame and the k-th subframe. The averaged Celp gain is quantized and coded in the same manner as usual Celp gain.

According to VUV, the following parameters are transmitted.



Table 2.A.6 - coded parameters for 4kbit/s variable rate mode

Mode(VUV)	Back Ground Noise(1)		UV(0)	V(2,3)
	UpdateFlag=0	UpdateFlag=1		
V/UV	2bit/20msec	2bit/20msec	2bit/20msec	2bit/20msec
UpdateFlag	1bit/20msec	1bit/20msec	0bit/20msec	0bit/20msec
LSP	0bit/20msec	18bit/20msec	18bit/20msec	26bit/20msec
Excitation		4bit/20msec (gain only)	20bit/20msec	52bit/20msec
Total	3bit/20msec 0.15kbit/s	25bit/20msec 1.25kbit/ss	40bit/20msec 2.0kbit/ss	80bit/20msec 4.0kbit/s

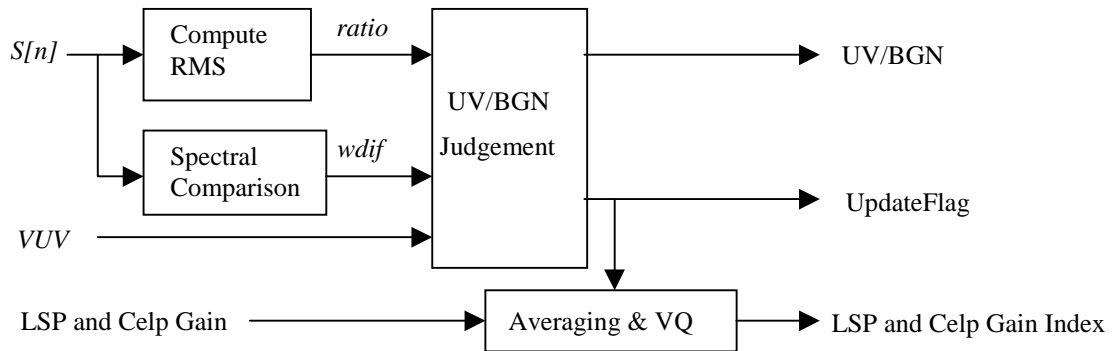


Figure 2.A.5 - Additional Blockdiagram for Encoder

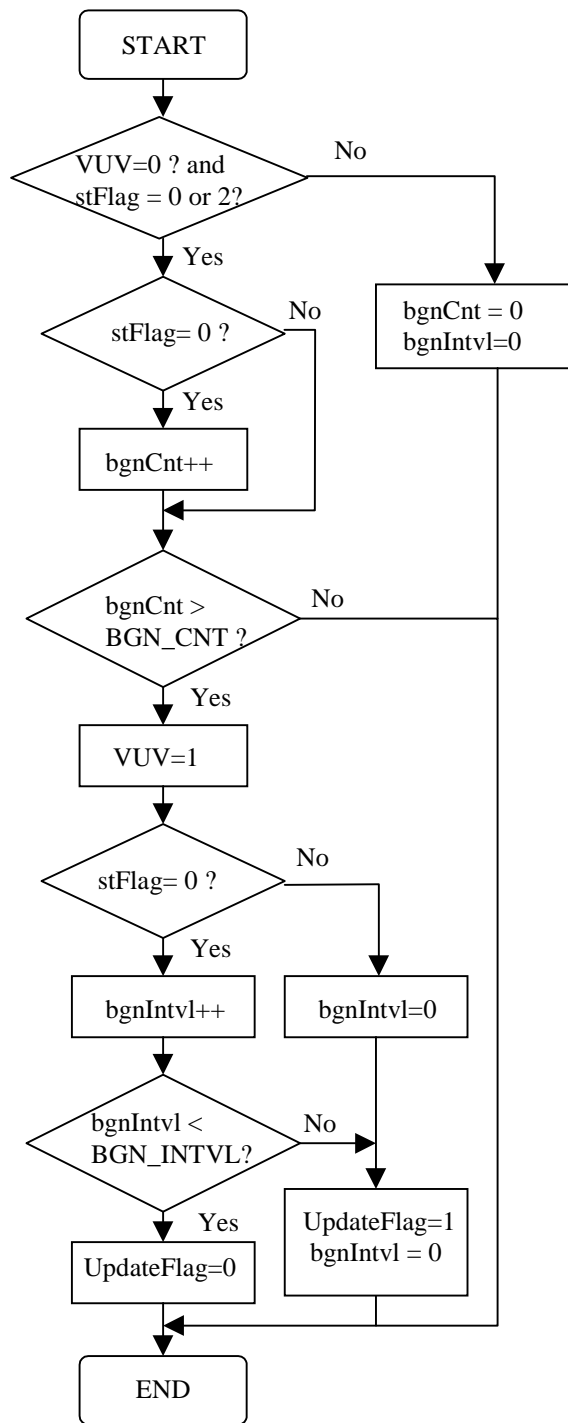


Figure 2.A.6 - Flowchart of noise detection

## Annex 2.B (informative)

### HVXC Decoder tools

#### 2.B.1 Postfilter

##### 2.B.1.1 Tool description

The basic operation of the post-filter is to enhance the spectral formants and suppress the spectral valleys. One can use a single postfilter after voiced and unvoiced synthesized speech are added. Alternatively, one can use independent postfilters for voiced and unvoiced speech respectively. Use of independent postfilters for voiced and unvoiced signals is recommended.

In the description below, each of the voiced speech and unvoiced speech obtained is fed into the independent postfilters.

The use of a postfilter is required, however, the configuration and constants of the postfilters described here is one example and not normative, and they could be modified.

##### 2.B.1.2 Definitions

$Pf_v(z)$ : Transfer function of spectral shaping filter for voiced speech.

$Pf_{uv}(z)$ : Transfer function of spectral shaping filter for unvoiced speech

$r_{adj}$ : gain adjustment factor for spectral shaping.

$S(n)$ : LPC synthesis filter  $H(z)$  output.

$S_{pf}(n)$ : Spectral shaping filter  $Pf_v(z)$  output.

$S'_{pf}(n)$ : Spectral shaped and gain adjusted output.

$S'_{pf-prev}(n)$ :  $S'_{pf}(n)$  computed using previous frame's  $\alpha_n$  and  $r_{adj}$ .

$vspeech(n)$ : Postfiltered voiced speech.

$uvspeech(n)$ : Postfiltered unvoiced speech.

##### 2.B.1.3 Processing

Each of the operation of the postfilter consists of three steps, that is, spectral shaping, gain adjustment and smoothing process.

###### 2.B.1.3.1 Voiced speech

**Spectral shaping:**

$$Pf_v(z) = \frac{\sum_{n=0}^P \alpha_n \gamma^n z^{-n}}{\sum_{n=0}^P \alpha_n \beta^n z^{-n}} (1 - \delta z^{-1})$$

Output of the LPC synthesis filter  $s(n)$  is first fed into the spectral shaping filter  $Pf_v(z)$ , where  $\alpha_n$  are linear predictive coefficients converted from de-quantized and linearly interpolated LSPs, which are updated every 2.5 ms.  $p = 10, \gamma = 0.5, \beta = 0.8, \delta = -0.15 \times \alpha_1$  where the value of  $\delta$  is limited to the range of  $0 \leq \delta \leq 0.5$ . When low delay decode mode is selected, the decode frame interval is shifted by 2.5 ms and interpolation of LSPs is carried out for the first 17.5ms of decode frame interval shown in Figure 2.9, and the latest LSPs are used for the last 2.5 ms without interpolation.

**Gain adjustment:**

The output of the spectral shaping filter,  $S_{pf}(n)$ , is then gain adjusted so that the frame gain of the input and output of the spectral shaping is unchanged. Gain adjustment is done once every 160 sample frame while LSPs are updated every 2.5ms. The gain adjustment factor  $r_{adj}$  is computed as follows:

$$r_{adj} = \sqrt{\frac{\sum_{n=0}^{159} \{s(n)\}^2}{\sum_{n=0}^{159} \{s_{pf}(n)\}^2}}$$

The spectrally shaped and gain adjusted output  $s_{pf}'(n)$  is obtained as:

$$s_{pf}'(n) = r_{adj} s_{pf}(n) \quad (0 \leq n \leq 159)$$

**Smoothing process:**

The spectrally shaped and gain adjusted output  $s_{pf}'(n)$  is then smoothed to avoid discontinuities due to parameter change at the beginning of each frame. Postfiltered voiced output  $vspeech(n)$  is obtained as follows

$$vspeech(n) = \begin{cases} \left(1 - \frac{n}{20}\right) s_{pf\_prev}'(n) + \frac{n}{20} s_{pf}'(n) & (0 \leq n \leq 19) \\ s_{pf}'(n) & (20 \leq n \leq 159) \end{cases}$$

where  $s_{pf\_prev}'(n)$  is a  $s_{pf}'(n)$  computed using previous frame's  $\alpha_n$  and  $r_{adj}$ .

### 2.B.1.3.2 Unvoiced speech

#### Spectral shaping:

$$Pf_{uv}(z) = \frac{\sum_{n=0}^P \alpha_n \gamma^n z^{-n}}{\sum_{n=0}^P \alpha_n \beta^n z^{-n}} (1 - \delta z^{-1})$$

Similarly the transfer function of the spectral shaping filter for unvoiced speech is given as  $Pf_{uv}(z)$ , where  $\alpha_n$  are linear predictive coefficients converted from de-quantized LSPs, which are updated every 20 ms. When normal decode mode is selected, LSP coefficients are updated at the middle of decode frame interval. If low delay decode mode is selected, decode frame interval is shifted by 2.5 ms and LSP update happens at 7.5 ms point from the beginning of the decode interval in Figure 2.9  $p=10, \gamma=0.5, \beta=0.8, \delta=0.1$ . The **gain adjustment** and the **smoothing process** are the same as in the voiced part described above, and produces postfiltered unvoiced speech  $uvspeech(n)$ .

The output of each of the postfilters,  $vspeech(n)$  and  $uvspeech(n)$  are added to generate postfiltered speech output.

## 2.B.2 Post processing

### 2.B.2.1 Tool description

The output of the postfilter is fed into the post processing part. Post processing is composed of three filters, those are, high pass filter, high frequency emphasis filter, and low pass filter. High pass filter is used to remove unnecessary low frequency components, high frequency emphasis is used to increase the brightness of the speech, and low pass filter is used to remove unnecessary high frequency components. The filter configurations and constants described here is one example and not normative, and they could be modified.

### 2.B.2.2 Definitions

$HPF(z)$ : Transfer function of high pass filter.

$Emp(z)$ : Transfer function of high frequency emphasis filter.

$LPF(z)$ : Transfer function of low pass filter.

### 2.B.2.3 Processing

The three filters below are applied to the output of the postfilter.

#### High Pass Filter:

$$HPF(z) = G_{inv} \frac{1 + A_1 z^{-1} + B_1 z^{-2}}{1 + C_1 z^{-1} + D_1 z^{-2}} \cdot \frac{1 + A_2 z^{-1} + B_2 z^{-2}}{1 + C_2 z^{-1} + D_2 z^{-2}}$$

High Frequency Emphasis:

$$Emp(z) = GG \frac{1 + AA z^{-1} + BB z^{-2}}{1 + CC z^{-1} + DD z^{-2}}$$

Low Pass Filter:

$$LPF(z) = GL \frac{1 + AL z^{-1} + BL z^{-2}}{1 + CL z^{-1} + DL z^{-2}}$$

2.B.2.4 Tables

Table 2.B.1 - Coefficients of the high pass filter

$G_{inv}$	1.100000000000000
$A_1$	-1.998066423746901
$B_1$	1.000000000000000
$C_1$	-1.962822436245804
$D_1$	0.9684991816600951
$A_2$	-1.999633313803449
$B_2$	0.999999999999999
$C_2$	-1.858097918647416
$D_2$	0.8654599838007603

Table 2.B.2 - Coefficients of the high frequency emphasis filter

$AA$	0.551543
$BB$	0.152100
$CC$	0.89
$DD$	0.198025
$GG$	1.226

Table 2.B.3 - Coefficients of the low pass filter

$AL$	$-2. * 1. * \cos((4.0/4.0) * \pi)$
$BL$	1.
$CL$	$-2. * 0.78 * \cos((3.55/4.0) * \pi)$
$DL$	$0.78 * 0.78$
$GL$	0.768

**Annex 2.C**  
(informative)

**System layer definitions**

***2.C.1 Random access point***

Random access point in HVXC bitstream can be set at any frame boundary point.

## Annex 2.D (informative)

### Example of EP tool setting and error concealment for HVXC

#### 2.D.1 Overview

This section describes one example of the implementation of EP (Error Protection) tool and error concealment method for HVXC. Some of perceptually important bits are protected by FEC (forward error correction) scheme and some are checked by CRC to judge whether or not erroneous bits are included. When CRC error occurs, error concealment is executed to reduce perceptible degradation.

It should be noted that error correction method and EP tool setting, error concealment algorithm described below are one example, and they should be modified depending on the actual channel conditions.

#### 2.D.2 EP tool setting

##### 2.D.2.1 Out-band Information Example for HVXC

- 2 kbit/s fixed rate

ESC(Error Sensitivity Category) instances of two adjacent frames are applied for EP classes directly:

Class 1: 22bit(fixed), 2frame concatenated, SRCPC code rate 8/16, 6 bit CRC

Class 2: 4 bit(fixed), SRCPC code rate 8/8, 1 bit CRC

Class 3: 4 bit(fixed), SRCPC code rate 8/8, 1 bit CRC

Class 4: 20bit(fixed), 2frame concatenated, SRCPC code rate 8/8, no CRC

1	/* number of predefined sets */
2	/* bit interleaving */
0	/* bitstuffing */
2	/* 2 frame concatenate */
4	/* number of classes */
0 0 0 1 0 0 2 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
22	/* bits used for class length (0 = until the end) */
8	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
6	/* crc length */
0 0 0 0 0 0 3 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4	/* bits used for class length (0 = until the end) */
0	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
1	/* crc length */
0 0 0 0 0 0 3 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4	/* bits used for class length (0 = until the end) */
0	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */



1	/* crc length */
0 0 0 1 0 0 3 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
10	/* bits used for class length (0 = until the end) */
0	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0	/* crc length */

- 4kbit/s fixed rate

ESC(Error Sensitivity Category) instances of two adjacent frames are applied for EP classes directly:

Class 1: 33bit(fixed), 2frame concatenated, SRCPC code rate 8/16, 6bit CRC

Class 2: 22bit(fixed), 2frame concatenated, SRCPC code rate 8/8, 6bit CRC

Class 3: 4bit(fixed), SRCPC code rate 8/8, 1bit CRC

Class 4: 4bit(fixed), SRCPC code rate 8/8, 1bit CRC

Class 5: 17bit(fixed), 2frame concatenated, SRCPC code rate 8/8, no CRC

1	/* number of predefined sets */
2	/* 1 bit interleaving */
0	/* bitstuffing */
2	/* 2 frame concatenate */
5	/* number of classes */
0 0 0 1 0 0 2 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
33	/* bits used for class length (0 = until the end) */
8	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
6	/* crc length */
0 0 0 1 0 0 3 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
22	/* bits used for class length (0 = until the end) */
0	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
6	/* crc length */
0 0 0 0 0 3 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4	/* bits used for class length (0 = until the end) */
0	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
1	/* crc length */
0 0 0 0 0 3 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
4	/* bits used for class length (0 = until the end) */
0	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0	/* crc length */
0 0 0 1 0 0 3 0	/* length_esc, srcpc_esc, crc_esc, concatenate, FEC type, No termination, interleave SW, class option */
17	/* bits used for class length (0 = until the end) */
0	/* puncture rate for srcpc 0 = 8/8 ... 24 = 32/8 */
0	/* crc length */

The Table 2.D.1 below shows a channel coded bit assignment for the use of the above EP tool settings.

**Table 2.D.1 - The channel coded bit assignment for the use of the EP tool**

	2kbit/s fixed rate	4kbit/s fixed rate
<b>Class I</b>		
Source coder bits	44 <sup>(*)</sup>	66 <sup>(*)</sup>
CRC parity	6	6
Code Rate	8/16	8/16
Class I total	100	144
<b>Class II</b>		
Source coder bits	4	44 <sup>(*)</sup>
CRC parity	1	6
Code Rate	8/8	8/8
Class II total	5	50
<b>Class III</b>		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class III total	5	5
<b>Class IV</b>		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class IV total	5	5
<b>Class V</b>		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class V total	5	5
<b>Class VI</b>		
Source coder bits	20 <sup>(*)</sup>	4
CRC parity	0	1
Code Rate	8/8	8/8
Class VI total	20	5
<b>Class VII</b>		
Source coder bits		34 <sup>(*)</sup>
CRC parity		0
Code Rate		8/8
Class VII total		34
Total Bit of All Classes	140	248
Bitrate	3.5 kbit/s	6.2 kbit/s

(\*1) 2 frame concatenated.

Class I:

CRC covers all the Class I bits, and Class I bits including CRC are protected by convolutional coding.

Class II-V(2kbit/s), II-VI(4kbit/s):

At least one CRC bits cover the source coder bits of these classes.

Class VI(2kbit/s), VII(4kbit/s) :

The source coder bits are not checked by CRC nor protected by any error correction scheme

### 2.D.3 Error concealment

When CRC error is detected, error concealment processing (bad frame masking) is carried out. An example of concealment method is described below.

A frame masking state of the current frame is updated based on the decoded CRC result of Class I. The state transition diagram is shown in Figure 2.D.1. The initial state is state=0. The arrow with a letter "1" denotes the transition for a bad frame, and that with a letter "0" a good frame.

#### 2.D.3.1 Parameter replacement

According to the state value, the following parameter replacement is done. In error free condition, state value becomes 0, and received source coder bits are used without any concealment processing.

##### 2.D.3.1.1 LSP parameters

At state=1..6, LSP parameters are replaced with those of previous ones.

When state=7, If LSP4=0 (LSP quantization mode without inter-frame prediction), then LSP parameters are calculated from all LSP indexes received in the current frame. If LSP4=1 (LSP quantization mode with inter-frame coding), then LSP parameters are calculated with the following method.

In this mode, LSP parameters from LSP1 index are interpolated with the previous LSPs.

$$LSP_{base}(n) = p \cdot LSP_{prev}(n) + (1-p)LSP_{1st}(n) \quad \text{for } n=1..10 \quad (1)$$

$LSP_{base}(n)$  is LSP parameters of the base layer,  $LSP_{prev}(n)$  is the previous LSPs,  $LSP_{1st}(n)$  is the decoded LSPs from the current LSP1 index, and  $p$  is the factor of interpolation.  $p$  is changed according to the number of previous CRC error frames of Class I bits as shown in Table 2.D.2. LSP indexes LSP2, LSP3 and LSP5 are not used and  $LSP_{base}(n)$  is used as current LSP parameters.

Table 2.D.2 - p factor

Frame	p
0	0.7
1	0.6
2	0.5
3	0.4
4	0.3
5	0.2
6	0.1
≥7	0.0

**2.D.3.1.2 Mute variable**

According to the “state” value, a variable “mute” is set to control output level of speech.

The “mute” value below is used.

In state=7, the average of 1.0 and “mute” value of the previous frame( = 0.5 ( 1.0 + previous “mute value” ) ) is used, but when this value is more than 0.8, “mute” value is replaced with 0.8.

**Table 2.D.3 - mute variable**

State	mute
0	1.000
1	0.800
2	0.700
3	0.500
4	0.250
5	0.125
6	0.000
7	Average/0.800

**2.D.3.1.3 Replacement and gain control of “voiced” parameters**

In state=1..6, spectrum parameter SE\_shape1, SE\_shape2, spectrum gain parameter SE\_gain, spectrum parameter for 4kbit/s codec SE\_shape3 .. SE\_shape6 are replaced with corresponding parameters of the previous frame. Also, to control volume of output speech, harmonic magnitude parameters of LPC residual signal “ $Am[0K 127]$ ” is gain controlled as shown in Eq.(1). In the equation,  $Am_{(org)}[i]$  is computed from the received spectrum parameters from the latest error free frame.

$$Am[i] = mute * Am_{(org)}[i] \quad \text{for } i=0..127 \quad (1)$$

If previous frame is unvoiced and current state is state=7, Eq.(1) is replaced with Eq.(2).

$$Am[i] = 0.6 * mute * Am_{(org)}[i] \quad \text{for } i=0..127 \quad (2)$$

As described before, SE\_shape1 and SE\_shape2 are individually protected by 1 bit CRC. In state=0 or 7, when CRC errors of these classes are detected at the same time, the quantized harmonic magnitudes with fixed dimension  $Am_{qnt}[1..44]$  are gain suppressed as shown in Eq.(3).

$$Am_{qnt}[i] = s[i] * Am_{qnt(org)}[i] \quad \text{for } i=1..44 \quad (3)$$

$s[i]$  is the factor for the gain suppression.

**Table 2.D.4 - factor for gain suppression ‘s[0..44]’**

$i$	1	2	3	4	5	6	7..44
$s[i]$	0.10	0.25	0.40	0.55	0.70	0.85	1.00

At 4kbit/s, SE\_shape4, SE\_shape5, and SE\_shape6 are checked by CRC as Class II bits. When CRC error is detected, the spectrum parameter of the enhancement layer is not used.

#### 2.D.3.1.4 Replacement and gain control of “unvoiced” parameters.

In state=1..6, stochastic codebook gain parameter  $VX\_gain1[0]$ ,  $VX\_gain1[1]$  are replaced with the  $VX\_gain1[1]$  from the latest error free frame. Also stochastic codebook gain parameter for 4kbit/s codec  $VX\_gain2[0]..VX\_gain2[3]$  are replaced with the  $VX\_gain2[3]$  from the latest error free frame.

Stochastic codebook shape parameter  $VX\_shape1[0]$ ,  $VX\_shape1[1]$ , and stochastic codebook shape parameter for 4kbit/s codec are generated from randomly generated index values.

Also, to control volume of output speech, LPC residual signal  $res[0K159]$  is gain controlled as shown in Eq.(4). In the equation,  $res_{(org)}[i]$  is computed from stochastic codebook parameters.

$$res[i] = mute * res_{(org)}[i] \quad (0 \leq i \leq 159) \quad (4)$$

#### 2.D.3.1.5 Frame Masking State Transitions

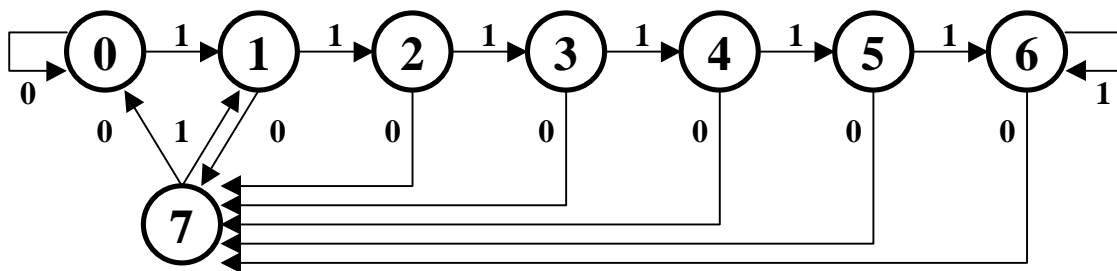


Figure 2.D.1 - Frame Masking State Transitions

## Annex 2.E (normative)

### VQ codebooks for HVXC

#### 2.E.1 List of the VQ codebooks

In this Annex, VQ codebook tables listed in the Table 2.E.1 are given. All the codeword values in the tables in this Annex must be divided by the values indicated as a factor for each of the tables before use.

Table 2.E.1 - List of the VQ codebooks

harmonic VQ table - 2k	CbAm
harmonic VQ table - 4k	CbAm4k
stochastic codebook table - 2k	CbCelp
stochastic codebook table - 4k	CbCelp4k
LSP quantizer table - 2k	CbLsp
LSP quantizer table - 4k	CbLsp4k

#### 2.E.2 CbAm

VQ codebook for harmonic spectral vector for 2kbps

Table 2.E.2 - Spectral gain codebook (base layer):  $g_0[]$   
 dim= 1 x 32 codewords  
 16 bits signed  
 factor =  $2^2$

index	Codeword	index	codeword
0	57	16	26628
1	98	17	22212
2	206	18	17853
3	152	19	20138
4	625	20	12642
5	480	21	13691
6	278	22	16103
7	364	23	14774
8	4544	24	5362
9	3787	25	6172
10	2460	26	8005
11	3077	27	7125
12	826	28	11519
13	1110	29	10688

14	1940	30	8954
15	1482	31	9856

**Table 2.E.3 - Spectral shape (base layer): cb0[16][44]  
dim=44 \* 16 codevectors  
16 bits signed  
factor =2^17**

index (SE_shape1)	codeword			
0	7710	12303	6176	1720
	5594	7400	2762	185
	2501	4264	3853	2958
	4090	6822	6583	4672
	3971	4733	3289	2402
	3414	4836	5483	5375
	5788	6150	6913	5797
	4675	4611	5268	3911
	2219	1691	2596	3258
	4269	5670	7460	8297
	6711	3040	2429	938
1	6406	10409	8027	7324
	9265	7299	2691	-89
	547	2846	4081	3718
	2848	4351	5958	8480
	9650	8783	5877	4566
	5500	7227	8808	8891
	7361	6209	5672	5526
	5247	5120	5262	3651
	1414	881	2038	3277
	3965	4049	4623	5491
	5508	6985	12000	11198
2	3235	5535	5477	5511
	6480	6626	5549	3676
	2540	3077	4530	4482
	3887	6011	6418	6445
	6392	7353	6602	4748
	4929	6353	6300	4968
	4259	4319	4881	6797
	11052	13332	15290	16039
	13107	9461	5263	2697
	2047	2080	3535	4610
	3164	-679	-831	700
3	4236	9106	11295	11039
	10359	9003	6981	4421
	2691	2005	2262	2279
	2289	4095	4052	3312
	3592	3944	2875	1864
	2101	3171	3253	3162
	2987	3429	3525	3078
	2896	2494	1813	920
	292	715	2401	3234

	4049 3711	4442 -395	4998 -154	5256 1377
4	13954 2890 2592 1614 5208 1548 5052 6135 473 4368 4248	8825 2461 3760 3113 4689 3179 5547 5499 1034 5192 536	-2191 2722 4872 3932 2860 4830 5793 3854 2835 6498 -645	957 2351 3592 5058 1190 4917 5944 2075 3291 6736 -861
5	2225 9443 -2831 2106 2840 2311 1998 1991 1596 1795 2887	5420 4811 434 2885 4028 2970 2021 2124 1733 1555 -736	8507 -1364 2964 2677 3783 3318 2407 2534 1779 2944 -1242	10359 -4304 3312 2142 2347 2793 2170 2117 1550 4183 -152
6	1326 16304 1521 4371 4107 3069 2611 4701 -17 4116 3392	4724 10623 5819 4060 5364 4169 3101 4975 219 3947 -1057	9967 3163 8273 3718 4891 4688 3915 3489 1874 4988 -2291	15539 -788 6952 3298 3022 3447 4441 1533 3136 5386 -1434
7	5676 4416 1590 4485 2936 2694 3967 3685 552 5873 4194	11275 1903 4552 3794 3365 4167 4379 2961 1424 5071 397	11435 429 7473 2766 2673 4494 4656 2096 3723 5288 -662	8100 -130 7119 2043 1994 4121 4373 1185 5183 5628 25
8	8006 5616 8212 9117 9115 7170 8279 9885 5138 10498 10706	14433 9749 4869 12166 9020 8236 8558 8427 5211 11176 7004	13555 15468 5584 11620 8153 8790 9550 7520 7231 12207 6441	7820 13912 7375 9748 7046 8369 9980 5986 9033 12670 4155



9	5036	10205	11528	9840	
	9263	8782	7973	7590	
	7427	7360	7549	6667	
	6528	9201	10278	10508	
	11114	10054	7920	6415	
	6869	8687	10127	11499	
	13521	16759	17481	14237	
	9991	7277	5499	4023	
	3069	3202	4544	5645	
	7247	9846	14884	22076	
	23893	17146	9262	2938	
	10	2928	6339	7501	7908
		8465	8757	7683	6864
6934		7277	8001	6882	
4939		6921	8146	8670	
9110		9483	8002	6317	
6217		6919	6816	6580	
6868		7313	7860	7728	
7708		7120	4417	2995	
5624		13956	22866	26244	
25377		20595	14318	8088	
3734		-843	-1401	-767	
11		2324	4896	6745	9373
	13292	15734	15021	11516	
	8153	6101	5412	4278	
	3763	6757	9367	11234	
	12719	13716	12301	10678	
	10558	11499	11287	9989	
	8202	7439	7656	8212	
	8505	8095	7800	6342	
	5458	5804	7203	8282	
	8899	9103	10506	11557	
	10899	8015	7049	5679	
	12	5291	9095	6217	2989
3371		6555	9455	10807	
10012		8879	8801	7892	
6204		8161	9912	10641	
9898		9877	8923	8066	
7580		7453	7666	7519	
7339		8115	8627	8928	
8720		7023	5735	4203	
3283		3492	4904	5389	
4974		4628	5865	7185	
9450		14948	24101	24286	
13		7485	7194	760	6246
	12284	4764	-576	2634	
	5063	4651	3470	2519	
	2409	4105	3939	3596	
	4844	5418	4188	3338	
	3544	4956	5411	5607	
	5167	5010	5619	5694	
	4935	4568	3990	2538	
	744	412	2351	3498	
	5146	6572	8301	8349	
	6779	2113	665	1038	
		3861	7854	10553	12074
	13784	13715	11321	7859	

14	6536	8311	12485	16123
	17146	16537	12534	9378
	8763	9137	8451	7949
	8435	9215	9124	8284
	7246	6931	7589	8877
	10426	10428	10085	8625
	7294	7266	8688	9873
	10573	10885	12186	13044
	11178	6062	4333	3668
15	5319	10458	11442	9126
	6605	4864	4486	5813
	8369	9902	9479	7280
	5596	7771	9447	10339
	12120	13645	13169	11839
	11237	10432	8377	6807
	5851	5981	6977	8020
	10188	11005	9799	7195
	4792	4291	5736	6737
	6776	6560	7849	9576
	8904	5430	4240	2797

Table 2.E.4 - Spectral shape (base layer): cb1[16][44]  
 dim=44 \* 16 codevectors  
 16 bits signed  
 factor = 2<sup>17</sup>

index (SE_shape2)	Codeword				
0	5073	9567	9861	7804	
	9063	9510	6902	4357	
	3660	4111	4709	6181	
	7125	6090	8049	9594	
	7528	5771	6747	7965	
	7779	6443	5107	5658	
	6264	6812	7159	7779	
	9055	8740	7823	8271	
	9104	9062	7726	6345	
	5475	5241	4112	4086	
	6450	10299	10485	7481	
	1	7272	13230	10938	6723
		11197	18250	15132	9622
9201		10191	10757	10474	
10315		8562	8277	9264	
9009		8390	8987	9757	
9670		9058	9393	10433	
11379		10843	9726	9623	
9375		9673	9576	10383	
11793		12855	14061	14168	
13653		14983	15013	13399	
12238		13728	11364	6705	
	2959	7446	13747	18610	
	18985	17157	15474	13007	

2	9416	6379	4671	5822	
	9248	10374	11050	10787	
	9149	8527	9056	9764	
	8961	7911	8203	9124	
	8923	8179	7701	7757	
	8100	8968	10546	12239	
	13814	13619	11804	9965	
	8487	7732	6518	5977	
	7981	11564	10534	7579	
	3	881	3834	9534	15687
19571		21127	21317	20686	
18354		16020	14101	14310	
15415		13359	12665	12287	
11855		11085	11885	12729	
12010		11130	10545	11096	
12206		12018	11505	11086	
10859		11481	12562	14110	
16114		16406	14734	13185	
12023		11632	10977	11301	
14069		18042	15926	11411	
4	7360	14622	15669	9368	
	2755	2083	5117	7701	
	7120	4817	3060	3642	
	4554	3001	2864	3714	
	4100	3934	4084	4683	
	4396	3489	2996	3929	
	5090	5027	4643	4134	
	3751	3700	4283	5958	
	7244	7072	5787	4815	
	4386	3571	1939	1097	
	2929	6701	7463	7336	
5	5196	10635	12117	10303	
	7333	5421	5336	6648	
	9173	12792	15868	15774	
	11876	5957	4484	5594	
	6152	5943	7013	8206	
	7946	6912	6904	8086	
	9500	9693	9573	9461	
	8868	9403	9563	10010	
	10884	10854	9298	7607	
	7178	7287	6657	5052	
	6361	10084	8921	5501	
6	3814	7967	12276	15441	
	13000	8461	5054	3396	
	3144	3838	5477	7326	
	8611	6356	5259	4945	
	4360	4073	5227	6583	
	6294	5237	5170	5842	
	6016	5587	5788	5732	
	5040	4746	5318	6409	
	7897	7905	6350	5237	
	4578	4646	4151	4475	
	8546	16193	17112	11921	
		3614	8494	13158	15420
		14353	11638	9963	11367
14001		15703	14012	11770	
10243		7114	6543	7277	

7	7620	7790	9649	11153
	10989	9370	8266	7678
	8114	8703	9417	10506
	10421	10014	9189	10121
	11809	12712	12751	12325
	10815	9566	8154	8222
	9942	12697	10582	6937
8	9640	16893	13829	7074
	5462	9678	15386	15346
	10141	7792	9380	12736
	13929	10195	8097	8979
	9992	9693	10282	11185
	11110	9595	8741	8421
	8601	8993	9737	9894
	10042	10655	11404	12775
	14831	15267	13130	11306
	10232	9384	8076	7109
8837	12760	11663	6908	
9	4431	9140	11976	11846
	10802	10726	11703	12710
	11853	10104	8574	9669
	11691	10610	10325	10593
	9775	9589	11402	14198
	15648	15642	15299	14194
	11693	8709	6792	6841
	7525	9517	12122	16429
	20201	21244	18251	13723
	8741	6187	5201	5710
8559	12380	10814	7065	
10	6417	13791	17968	17891
	13116	9753	9387	10587
	10112	9537	8292	9676
	11265	10567	10373	10581
	10035	9550	10655	11942
	11224	10169	9913	10739
	12516	13678	13764	13060
	11483	10871	11190	10454
	11006	11958	13540	17694
	22764	25085	23894	17793
12022	10893	8946	5785	
11	6643	13495	16708	14546
	9833	8439	12580	19060
	21155	17246	11261	8975
	10488	10761	11801	12682
	12643	11747	12055	12582
	12365	11183	10826	11197
	10922	10288	9831	10217
	10597	11134	11651	13620
	15546	15759	14419	13073
	11822	11093	10186	10018
11946	15324	13076	8063	
12	9320	13367	7272	3508
	4787	6638	6347	4309
	3669	4306	5158	5702
	6564	4750	4151	4588
	4773	4885	6161	6830
	6057	4794	4497	5135

	5719	5897	5270	4610
	4953	6288	6932	7984
	8446	7828	6074	5357
	4867	4268	2723	3372
	7136	12675	11294	5078
13	9271	12500	5475	5105
	13647	13655	7248	5595
	5206	5703	8553	11643
	10651	7301	6630	8221
	7870	7294	8693	9878
	9740	8182	8139	7921
	8812	9644	10152	10275
	10590	10528	9952	9091
	9880	10394	8606	7573
	6580	5574	3725	2508
	5452	13958	21519	22685
14	7366	14917	18246	15291
	9380	6686	7609	8991
	8434	6401	4936	7007
	11702	13856	13723	12350
	10270	8661	9110	10490
	10160	8741	8462	8722
	8443	8320	8454	8981
	9163	10173	10973	11715
	12739	12526	10981	9192
	7477	6617	5552	5244
	7887	14437	15124	10911
15	3647	8297	13029	15393
	13816	10428	7568	6298
	5446	5598	6989	10927
	14126	13634	13227	13814
	13809	13528	12929	11754
	9866	8138	8304	9576
	11156	12009	11476	10823
	10434	10355	11424	13811
	16041	16689	15293	12292
	8551	5943	4131	4170
	7446	13011	12150	8225

### 2.E.3 CbAm4k

VQ codebook for harmonic spectral vector quantization for 4kbps.

**Table 2.E.5 - Spectral shape (enhancement layer): cb4k[0][128][2]  
dim=2x128 codewords  
16 bits signed  
factor = 2<sup>6</sup>**

index (SE_shape3)	codeword

0	-5773	858
1	-1768	1201
2	644	1969
3	444	46
4	-2723	3150
5	-121	172
6	-4459	6812
7	-1193	2298
8	-11663	2137
9	-2386	-178
10	-3323	1768
11	-195	-15
12	-2449	567
13	45	-11
14	-7228	3785
15	-2010	1917
16	-1169	3581
17	417	284
18	-684	8757
19	-439	2544
20	-2127	14132
21	501	4365
22	-3449	25883
23	-3584	9653
24	-3401	917
25	-172	-463
26	-848	6067
27	380	1285
28	-4159	3823
29	-1099	810
30	-4328	16051
31	-2563	4871
32	970	382
33	388	-200
34	7754	1041
35	2254	1818
36	698	598
37	745	-435
38	1684	2655
39	842	137
40	-4839	2005
41	-50	627
42	1287	-141
43	701	-127
44	-169	1056
45	242	-411
46	-834	1453
47	18	93
48	4585	2897
49	1556	348
50	9446	6193
51	3441	976
52	2890	4293
53	1702	898
54	5292	11266
55	274	3043
56	305	827

57	528	-791
58	4109	29
59	1634	-459
60	-210	1695
61	-83	-244
62	2193	7152
63	1071	1264
64	5773	-858
65	1768	-1201
66	-644	-1969
67	-444	-46
68	2723	-3150
69	121	-172
70	4459	-6812
71	1193	-2298
72	11663	-2137
73	2386	178
74	3323	-1768
75	195	15
76	2449	-567
77	-45	11
78	7228	-3785
79	2010	-1917
80	1169	-3581
81	-417	-284
82	684	-8757
83	439	-2544
84	2127	-14132
85	-501	-4365
86	3449	-25883
87	3584	-9653
88	3401	-917
89	172	463
90	848	-6067
91	-380	-1285
92	4159	-3823
93	1099	-810
94	4328	-16051
95	2563	-4871
96	-970	-382
97	-388	200
98	-7754	-1041
99	-2254	-1818
100	-698	-598
101	-745	435
102	-1684	-2655
103	-842	-137
104	4839	-2005
105	50	-627
106	-1287	141
107	-701	127
108	169	-1056
109	-242	411
110	834	-1453
111	-18	-93
112	-4585	-2897
113	-1556	-348

114	-9446	-6193
115	-3441	-976
116	-2890	-4293
117	-1702	-898
118	-5292	-11266
119	-274	-3043
120	-305	-827
121	-528	791
122	-4109	-29
123	-1634	459
124	210	-1695
125	83	244
126	-2193	-7152
127	-1071	-1264

Table 2.E.6 - Spectral shape (enhancement layer): cb4k[1][1024][4]  
dim=4x1024 codewords  
16 bits signed  
factor = 2<sup>6</sup>

index (SE_shape4)	codeword			
0	-3708	4595	174	5466
1	-6642	6292	-2437	-442
2	-5553	3819	1853	8413
3	-425	627	-5619	3556
4	-7817	7075	-10600	2258
5	-3925	2833	72	1358
6	-11037	7197	-5288	14434
7	-2425	4350	-333	9257
8	-1515	-215	194	-90
9	-1555	2325	-2488	2620
10	-687	657	-3195	650
11	-3784	3980	-12574	-5902
12	-1194	1024	-2115	25
13	4326	4919	7640	-326
14	-793	2014	-1672	6939
15	-653	754	-2286	-318
16	-5673	1116	-8246	2246
17	-11688	8435	-16555	-2551
18	1247	1855	-4749	1838
19	1005	7635	-17838	475
20	-1301	1082	-2757	4124
21	-9574	1676	-8532	952
22	-2056	2257	-7359	4384
23	-2787	1039	-5899	11
24	-1070	4111	-1342	-1460
25	-1116	12475	-9995	-650
26	-1723	6725	-6020	-5427
27	1257	12863	-20202	-9518
28	-730	-2201	2676	-568
29	-426	2249	-2616	2826



30	-236	197	-921	-2407
31	-347	3829	-8218	-8722
32	-13058	10488	-8453	9753
33	-5056	2191	-4350	905
34	-1778	15731	-4754	10386
35	-2236	5103	-2860	3991
36	-7333	9303	-15598	16357
37	-6027	13542	-2855	7737
38	-61	12148	-12735	23167
39	1448	14926	-6518	13767
40	-8335	2500	1145	3713
41	-1633	-1142	-251	-2105
42	-3704	2818	-4206	5422
43	681	1046	-4699	-4321
44	-5111	8969	-10188	2045
45	-3247	1963	-1582	2605
46	-2955	7990	-14298	5719
47	-1580	4537	-825	6925
48	-4958	5233	-573	2004
49	-4890	14378	-10816	-4353
50	-2741	2600	-3685	4183
51	-5465	1216	-8887	-3905
52	-6944	5380	-5414	7765
53	-3224	4618	-1955	1816
54	335	10495	-15036	6311
55	-90	4485	-1280	3781
56	290	-398	727	-2585
57	1072	3908	-3961	-8898
58	-1920	1344	-7820	-2107
59	4502	8827	-17295	-6552
60	-967	1650	1808	-4294
61	-480	493	-750	-1257
62	971	368	-5922	5544
63	-462	2029	-8023	-2102
64	-782	470	269	853
65	-661	803	-223	123
66	-2119	266	635	4444
67	-630	-1052	-315	942
68	-3596	473	-2921	1915
69	-277	-356	759	478
70	-2282	2421	-5053	10636
71	-1276	818	-1150	2733
72	355	-532	1715	-1003
73	-53	-1318	1661	582
74	632	-2107	-217	-1528
75	211	-551	-2387	-1385
76	-616	-292	501	527
77	-2509	364	5987	2601
78	391	166	-236	-3245
79	438	-1049	1580	-160
80	-5314	325	1110	295
81	-4272	653	-7224	3198
82	-1	-133	-365	268
83	-1713	1072	-5118	-100
84	-198	-287	48	211
85	-2646	-1197	-1365	624
86	-673	1509	-1153	1719

87	-236	-1532	-859	319
88	2237	58	5065	-7544
89	-1467	312	-3129	-1035
90	3474	-1701	-1247	-1553
91	7069	813	-11959	-7397
92	-61	-242	58	-2288
93	-588	131	1017	1208
94	3938	4608	-446	-10269
95	873	54	-2588	-3892
96	-7524	2222	-2307	5325
97	-912	628	-939	696
98	3404	1697	-1102	10570
99	-1136	-638	-1377	3107
100	1671	2529	-3694	9330
101	-697	3208	1806	5118
102	3035	3843	-15529	23374
103	-3701	2884	-9284	8114
104	-1406	-309	-1186	730
105	62	-62	8	32
106	1637	-1305	-1739	2757
107	274	18	-1460	41
108	-2344	265	-1672	629
109	-1109	3991	1669	2016
110	410	923	-10336	6935
111	-403	148	-1632	3694
112	-1298	-807	252	1014
113	325	3999	-3187	-3229
114	1073	1368	-2383	4904
115	-1227	255	-1438	-666
116	-670	14	-2684	3137
117	-287	103	63	135
118	255	3490	-11611	9597
119	522	36	-2783	4220
120	1835	-2453	1409	-1043
121	218	-1024	-813	-2278
122	-54	186	-408	-884
123	3969	4248	-5097	-4668
124	26	-991	1669	-862
125	-19	-137	245	-54
126	1085	6265	-376	1538
127	-108	-130	-1726	-1060
128	-1121	635	-236	988
129	-1737	1161	771	1046
130	-1419	1019	-760	1417
131	-356	144	-410	270
132	-4517	438	-722	1664
133	-763	3156	5529	9680
134	-2411	1731	1934	8028
135	1668	-1040	1629	2868
136	-6824	4650	-315	-2845
137	-810	3890	5416	3951
138	-1088	-249	-388	-310
139	-1714	-705	-2398	-520
140	1243	8345	6597	302
141	595	12918	13592	9238
142	-15	-744	809	-3756
143	2603	2817	8832	702

144	-3589	2930	274	-1360
145	-11450	7157	-1170	-474
146	-486	138	-202	-726
147	-198	3273	-5179	2160
148	-649	-2464	410	1835
149	-3013	-221	-1093	2541
150	360	458	-597	-1915
151	-129	75	-75	152
152	-5599	10181	-1527	-13002
153	-5539	2837	-1770	-1665
154	-2093	1851	-4449	-5297
155	-34	7568	-8611	-3506
156	-958	2360	-1516	-4489
157	1209	2368	7398	1540
158	6201	5468	-1745	-7751
159	1834	204	-3621	-1843
160	-4192	2003	-1263	9486
161	48	101	-1465	1441
162	-1987	5262	1146	4680
163	339	445	-67	1228
164	-9438	9847	-3534	4707
165	-1770	2859	992	3176
166	6020	10262	-774	10647
167	-12	1028	-1248	10011
168	-2051	-420	-50	956
169	652	-684	1852	1829
170	-93	186	-430	1151
171	94	-261	-2394	2313
172	-2006	504	-1688	1855
173	213	1148	4752	7458
174	-814	230	-4472	6898
175	-165	95	-3559	2178
176	-1963	-2056	207	2212
177	-2769	5330	-1410	-1443
178	25	331	158	1048
179	-827	-650	-2209	622
180	-5239	33	-1597	9961
181	-972	-1168	-365	2043
182	-1400	1650	-4458	4091
183	1630	305	-3728	3573
184	-5091	1693	2809	-2600
185	-258	-485	-1732	264
186	580	-2735	-2014	2558
187	125	135	-11841	2969
188	-1858	-1064	-981	1674
189	-2773	74	-4578	1621
190	1280	3555	-3417	-2632
191	-2153	3308	-15960	2431
192	-4241	958	416	-865
193	-3128	-2675	-610	-858
194	1026	-2422	-211	-571
195	-228	-3161	1008	-2019
196	-20	-63	-54	-41
197	-986	-1518	1451	2295
198	9	1388	-29	-3287
199	-456	-443	-111	600
200	-2355	38	8709	-6915

201	-1567	-840	1919	-117
202	3597	48	-3335	-6013
203	1694	-2322	315	-1579
204	-180	1768	-1223	-4678
205	-5114	3261	8514	1950
206	6386	179	-1249	-16705
207	1122	813	-965	-5297
208	-10496	6615	-2131	-9755
209	-7880	1186	-407	393
210	-1599	20	-3765	-2851
211	-842	-123	476	-700
212	-900	493	-116	-3041
213	-1045	-407	-332	589
214	-747	5161	-3022	-10826
215	4573	1692	339	-870
216	-3881	3559	5075	-19570
217	-2652	1226	401	-4968
218	7038	1565	-1483	-12077
219	899	-918	-616	-2329
220	7936	3162	3507	-6532
221	3970	187	3887	1722
222	12721	12975	-942	-19702
223	12004	5659	2488	-4690
224	-352	266	-911	1398
225	-1089	-2159	-699	-243
226	2629	-586	-1150	2141
227	-129	-893	397	804
228	-2755	289	-1942	3435
229	-287	213	683	701
230	3615	4600	-6882	6219
231	-1395	23	-990	6559
232	-2218	-2573	606	-2765
233	-677	-3951	1530	86
234	130	-3203	-1032	-1364
235	-276	-3743	-919	457
236	-469	127	428	-45
237	-1603	387	1020	80
238	1750	1781	-1595	-3811
239	157	-203	-318	687
240	-2340	127	-2231	-2730
241	-526	67	38	658
242	2557	-1795	986	-113
243	680	-1638	-353	-266
244	-1100	13	-1727	1545
245	839	-2868	-84	593
246	-9	5786	-565	2024
247	-148	1	-83	655
248	-1465	226	-319	-5095
249	-1974	-2118	-1354	-1007
250	3150	2292	14	-2752
251	780	-2824	-2501	-139
252	958	5024	1323	252
253	11	-25	7	354
254	9236	13635	-293	-4887
255	1463	3887	-2164	1698
256	-708	3836	-6314	-3506
257	-1970	17471	-12464	-351

258	-869	208	-168	1846
259	-1071	5458	-2417	569
260	-1464	1135	-2463	1733
261	-1187	3095	-5791	1952
262	-4159	3332	-734	3910
263	2275	10821	395	4807
264	86	-624	-1147	-246
265	-1117	4502	-4199	-2764
266	166	52	-118	173
267	449	3424	-3234	696
268	-311	-1708	2741	-2083
269	-2785	4153	3701	2760
270	-1135	302	550	-1588
271	-1184	971	9	4240
272	-808	68	-1424	26
273	-5136	6187	-3183	-3564
274	-7	-188	-960	324
275	748	1252	-7581	1604
276	-14	-912	2205	-1789
277	-1405	751	-1377	554
278	-648	-514	-1042	836
279	143	953	-515	1550
280	-883	-725	612	-3834
281	-662	2368	-2651	1351
282	1726	4225	973	-1570
283	4675	12839	-9323	-2710
284	-3621	174	12424	-5116
285	-1433	-1693	1521	-2069
286	-267	1064	4460	-2164
287	-1065	3217	-2198	1709
288	-6483	74	-2695	5213
289	2023	3583	-5542	-1418
290	779	3571	-587	5003
291	8624	5214	-562	3138
292	-110	2848	-5952	6395
293	2292	3251	-2925	2462
294	1034	10647	-9901	8040
295	14588	11498	-1318	8601
296	413	2206	934	-3276
297	140	1706	-22	1385
298	206	1621	-209	604
299	5955	10295	72	-2099
300	19	1128	-1935	-7769
301	-1370	1144	2203	378
302	-2795	669	-3004	3822
303	7437	1868	-4126	905
304	-400	26	554	-3070
305	-3637	2283	-3713	-5056
306	-433	901	222	-712
307	-699	8416	770	-46
308	1113	2821	2692	-2759
309	-294	697	400	98
310	1524	1989	-3552	4000
311	5160	4928	2172	1541
312	-1064	1709	2875	-15351
313	-760	6388	-492	-3470
314	-2026	4468	-1561	-5057

315	11	19162	-6693	-6267
316	3885	2603	9824	-16992
317	985	1742	339	-6632
318	2117	350	1174	-5597
319	2698	11997	-4125	-5799
320	-232	304	-1121	142
321	-3770	4053	-6203	2270
322	-2	-2321	-597	1057
323	-451	681	-671	599
324	609	379	1198	1452
325	-3582	-989	1566	1552
326	531	1194	-1106	1851
327	1116	716	1651	2403
328	-81	-2175	1785	-429
329	-2400	529	1340	1498
330	3614	-1796	546	-359
331	1849	-1169	-37	562
332	-3540	-1102	2407	-391
333	-12764	592	7511	2433
334	-715	6994	3431	-7914
335	-2516	-1617	2003	-1330
336	-654	-705	256	57
337	-179	653	-1087	-743
338	3320	-318	676	253
339	-154	-473	-615	477
340	-735	-944	1342	-358
341	-503	-742	1803	1393
342	310	1132	4767	542
343	134	-4	288	12
344	1170	828	3958	-1735
345	-1877	636	180	724
346	14629	473	2203	-5314
347	4959	3220	-1851	716
348	-3032	2368	6663	-1038
349	-5137	1085	2171	6217
350	4773	7174	12228	-6629
351	1695	2016	4811	-791
352	-700	206	-691	999
353	-931	195	-509	254
354	-210	354	24	5106
355	1461	-303	460	1298
356	1767	322	1276	7437
357	-1073	77	3022	-246
358	-328	3909	-5164	13052
359	4962	1555	1346	4890
360	164	-825	148	-664
361	-712	1534	1882	100
362	1639	-1477	1551	719
363	458	1604	1689	681
364	-492	693	2036	-752
365	-2953	3422	12980	1546
366	-1684	1399	-743	2662
367	170	975	2754	3402
368	1001	-1123	965	625
369	-1298	-706	383	-913
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894	-1569	-4734	-2570	-621
895	-948	-2114	481	-471
896	230	404	911	496
897	-364	-6114	3405	-772
898	-219	1386	590	-392
899	-132	-409	12	-327
900	415	1352	-527	-839
901	-1262	-1249	-1092	-4365
902	316	137	-284	576
903	-1052	-813	-784	-682
904	1133	-708	59	802
905	1122	1	-1803	-2188
906	140	868	698	1287
907	-30	191	227	322
908	440	1237	-2506	-481
909	462	-222	-11632	-6914
910	682	-2262	3326	7404
911	-693	-1250	-69	-1324
912	837	670	-562	1320
913	3340	763	-15	-562
914	-20	429	138	-175
915	54	-303	643	-439
916	-692	1589	-1010	97
917	-215	-51	-496	-351
918	404	569	-3822	821
919	-526	-2893	-1705	-5612
920	1516	-262	-1503	3924
921	1227	468	-1129	-836
922	813	-138	17	123
923	209	-750	4825	3249
924	789	180	-1749	1877
925	359	-1774	-1935	-3252
926	94	-4214	-4910	1833
927	3882	-7957	5511	-3855
928	367	47	342	-1252
929	134	-229	149	240
930	-250	-394	833	-716
931	-2121	202	-266	-2311
932	4210	-116	748	-4067
933	-786	1697	-147	-2305
934	-5624	-1248	5200	-3876
935	-5066	-352	768	-9894
936	-204	382	154	238
937	184	879	95	465
938	1776	1346	649	43
939	539	-1756	4159	445
940	-374	683	626	1193
941	229	-1834	493	-2550



942	1266	-149	2937	-1089
943	-3233	-9153	9378	-3144
944	-1149	1498	-174	-44
945	351	-332	22	165
946	-2122	117	-617	-378
947	1944	-2443	3429	-403
948	1496	-19	-58	-2165
949	-3673	-231	1501	-5085
950	-699	-3509	-2043	-2589
951	-3559	-11798	3227	-5935
952	658	1097	445	2784
953	3450	-1745	2433	2001
954	6719	-3703	2081	-1694
955	7042	-9134	13671	2614
956	-364	-1059	-630	5065
957	228	-6003	3945	-3588
958	-847	-4540	6326	-742
959	-3283	-16010	17227	-1636
960	-285	362	-322	672
961	473	-199	786	215
962	369	-229	-798	1100
963	-182	2018	639	464
964	561	-111	-55	1666
965	-583	2159	-1414	-751
966	4419	-3446	-3790	7302
967	-39	-518	15	1495
968	1016	-990	-3025	1759
969	2307	200	-948	624
970	2763	-2359	-1216	7404
971	231	-752	-577	1353
972	752	-1261	-2326	10115
973	1315	-740	-5628	-1252
974	5387	-7958	-1691	19224
975	-1006	-5692	-4524	8230
976	2944	-1850	-2262	1848
977	1732	1352	-1341	-956
978	194	-1366	-3307	500
979	-64	293	-192	355
980	456	-847	-1121	1999
981	54	539	-525	-31
982	-2387	-9134	-9081	7328
983	2028	-3516	-2003	-215
984	9535	-708	-11389	7382
985	7260	-3968	-5869	-228
986	-1707	-1846	-4683	4601
987	953	262	-948	-96
988	234	-3920	-7662	8159
989	-198	-731	-1058	1634
990	-812	-11394	-10228	14268
991	1258	-3393	-4573	4991
992	-75	1149	-63	73
993	3042	3091	-903	-67
994	-2452	2662	-1675	-1564
995	-1612	3458	-53	543
996	-165	2659	-745	-2079
997	-285	1048	-448	-364
998	690	-394	1441	-2126

999	-1270	1203	-20	-1981
1000	319	987	-689	240
1001	-212	524	-354	-73
1002	2474	606	106	1153
1003	502	80	106	13
1004	1307	3030	-240	1958
1005	179	-2199	-1957	-1373
1006	4028	-935	2361	9619
1007	78	-92	2357	258
1008	-2251	942	-2698	-1414
1009	-183	2076	-890	-255
1010	-12708	-266	-3516	-1450
1011	-2618	2366	-1898	1164
1012	33	64	-635	-197
1013	-1921	3074	837	-1172
1014	-1510	-3343	-3822	-2642
1015	-328	-1840	568	-2628
1016	1948	-1732	-4074	774
1017	1101	-338	-457	-252
1018	-2653	-1423	-1556	-3652
1019	2851	-2299	4021	-778
1020	977	-702	-1439	1033
1021	-512	-190	1255	-1133
1022	-3095	-2366	-4297	3993
1023	-5603	-6801	3278	-4825

**Table 2.E.7 - Spectral shape (enhancement layer) cb4k[2][512][4]  
dim=4x512 codewords  
16 bits signed  
factor = 2<sup>5</sup>**

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4	1198	336	93	370
5	-803	459	-843	146
6	1259	3665	436	2673
7	-5678	2246	-1009	947
8	1069	-145	-185	764
9	-52	234	-2248	1307
10	54	723	-828	2248
11	-5190	2683	-3314	4210
12	1910	1707	-1557	4536
13	-3734	4448	-2791	1786
14	681	2683	-3198	10239}

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16	-821	1524	-2028	2000
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18	-5621	3750	-8129	-305
19	-1551	2159	-2029	226
20	-2125	-1176	765	-4
21	261	979	-764	237
22	-1468	1029	-1065	-1167
23	-823	1334	-409	-876
24	-1237	-1215	-1426	-445
25	7	3	117	-388
26	-1877	90	-1748	-2169
27	-451	1766	-647	201
28	1134	643	-735	-105
29	-2354	1324	-886	-2290
30	-604	2078	1214	2999
31	-4053	7276	-2774	306
32	-386	1110	-1488	91
33	670	24	-1432	173
34	-2221	1788	-9716	1702
35	-1343	2122	-3768	2392
36	40	126	-283	-57
37	865	-807	-576	-312
38	1103	2566	-4258	-64
39	-1057	-155	-1320	1680
40	309	31	1	-682
41	36	-1863	-722	170
42	-480	-113	-2137	-861
43	-906	864	-110	2023
44	967	-467	68	375
45	-274	433	-396	1233
46	1733	1607	1388	4135
47	-4658	3707	-2344	4721
48	-2776	4606	-4730	-867
49	1978	1217	-4881	-3162
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51	-2658	6355	-8561	700
52	-830	863	1077	-2990
53	472	22	-1019	-893
54	-2349	2474	-4673	-6953
55	496	1994	-2199	-2257
56	770	448	-3023	-3298
57	436	-876	-1691	-1622
58	1138	987	-5905	-7911
59	384	1468	-2813	-324
60	216	86	445	-259
61	-476	-148	-158	70
62	1318	178	-2396	-1602
63	-1541	1857	842	1390
64	1104	2194	-991	-1366
65	183	240	-55	-218
66	1676	612	-410	1323
67	2785	515	599	1015
68	-216	96	-849	461
69	281	375	-951	1161
70	167	2080	-4014	3066
71	-1276	-1228	-1647	620

72	-583	-785	-4	-450
73	-485	564	-641	-434
74	139	-150	59	149
75	-1418	-789	-536	1140
76	-320	625	-349	353
77	-1834	1808	-4797	61
78	-634	246	651	3934
79	-6263	1011	-976	3099
80	2044	2788	649	394
81	-15	399	-1583	-494
82	-2441	726	-1402	-99
83	-86	386	225	223
84	-737	1069	-79	-133
85	-2916	3617	-4112	1740
86	975	250	-599	445
87	-2853	1039	-199	550
88	-66	435	-158	-405
89	-1400	3094	-379	-982
90	-523	386	-40	107
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92	-809	2683	-2140	-3902
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95	-8049	4056	-672	-3038
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97	-285	3122	415	-518
98	693	3599	-1614	3600
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101	4315	362	-1755	958
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105	-327	-286	-359	-353
106	206	159	-367	227
107	-827	-1979	-13	-354
108	1294	-427	-997	-283
109	10	137	-742	-12
110	1515	785	-3089	2309
111	-547	1041	-1296	994
112	2724	7267	3470	-3280
113	4307	87	-3989	-2418
114	-454	516	-4247	-2829
115	-976	2313	-1135	1590
116	1965	3341	3746	-1796
117	119	922	-740	-523
118	316	1407	-2417	1308
119	1171	-1377	-1912	-257
120	1645	2575	816	-1268
121	562	-820	-1226	-605
122	1082	142	-1702	-2577
123	-284	342	279	-209
124	-411	-376	-97	-1217
125	-3046	2378	-627	-523
126	83	-66	-87	77
127	-1852	2106	1081	-487
128	232	762	1513	-252

129	-70	-1473	273	-736
130	1608	305	907	323
131	1307	562	446	1448
132	-1495	1065	7	-118
133	-200	-29	480	458
134	142	-122	409	743
135	-1785	3094	-27	3163
136	116	-210	157	-182
137	-418	-35	76	-299
138	47	863	67	218
139	-3150	2681	1193	896
140	99	429	-454	717
141	-2329	1558	1614	2031
142	81	2328	370	4929
143	-9217	4208	3606	8575
144	-56	1395	605	-1845
145	392	-53	-296	-1474
146	-2063	1886	-327	724
147	-95	359	805	97
148	-6063	1753	2742	-3384
149	-749	1189	1409	-1212
150	-679	137	629	-1369
151	-576	5069	3880	-729
152	-208	-1798	717	-1709
153	-1360	-1858	-194	-1201
154	1837	-312	-568	-944
155	-2331	-180	1320	-315
156	-620	511	-138	-2351
157	-646	533	2408	-1218
158	-1855	-1354	1500	160
159	-6498	2601	4574	-776
160	-989	5956	2366	2583
161	-453	1352	544	-776
162	-127	3113	-2159	286
163	-629	-616	-572	77
164	-241	2165	2393	-1884
165	-253	-111	63	-654
166	717	-28	-498	-27
167	-516	228	349	377
168	972	1692	428	2383
169	12	10	-40	-38
170	974	357	-392	-640
171	-1975	-1191	538	1088
172	47	436	304	-584
173	-202	575	229	716
174	107	424	1091	632
175	-4219	2280	3206	3278
176	5301	3994	-60	-2715
177	1568	1015	-309	-6296
178	-3493	1401	-3514	-2795
179	-737	1947	-1687	-1067
180	-5407	4499	3109	-8521
181	-1102	3285	263	-2446
182	-2427	596	964	-1298
183	-345	862	375	-97
184	519	602	792	-885
185	1069	-256	517	-1184

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187	40	-1048	-720	-1240
188	-1030	1415	4368	-2676
189	728	-62	337	-690
190	556	796	-230	-1395
191	-1759	1011	2868	1114
192	1371	5701	528	6
193	3030	1144	869	-29
194	3378	3010	2763	1943
195	4054	1751	3238	3947
196	215	1657	-1001	-282
197	124	47	84	-74
198	979	972	-557	905
199	2015	641	2513	929
200	-4	867	-54	-255
201	34	172	70	71
202	647	281	712	1043
203	1029	55	872	3129
204	1085	-638	-134	-793
205	483	2163	1615	1069
206	921	207	3274	198
207	-939	1034	6613	4879
208	3647	499	4586	-1013
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210	1472	-314	648	-478
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212	140	1736	2784	-3675
213	-109	674	-1718	-1344
214	-463	-907	-1230	-1868
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216	316	778	4421	-1091
217	691	15	587	376
218	3163	648	1817	-1144
219	83	308	367	-6
220	1493	2	1757	-2674
221	-1778	1130	90	-1162
222	-300	264	-286	-108
223	-1769	3654	1894	674
224	6094	13540	1477	2528
225	2028	192	1053	-4005
226	1446	8069	2503	4816
227	-323	2370	2421	1159
228	-347	6855	-1048	-3343
229	636	1544	-162	-597
230	3994	2764	-2044	557
231	510	345	-1363	774
232	211	4953	-1633	1087
233	378	559	-550	-81
234	-1555	4061	-93	1843
235	195	353	-69	82
236	-196	1292	-958	-1485
237	69	9	191	255
238	528	471	-25	-389
239	181	824	3944	1653
240	15684	9882	11257	-3356
241	14361	3408	1203	-12054
242	3316	1761	4754	-364

243	4565	344	-1059	-391
244	4585	7075	7707	-13837
245	1228	128	1479	-5643
246	-249	1882	-2060	-6610
247	1507	420	-230	-2383
248	2804	4819	5060	341
249	3754	2544	251	-940
250	1353	855	41	-5
251	972	36	140	-183
252	-3375	1698	1958	-5789
253	-607	522	82	-988
254	-100	551	42	-1312
255	-245	76	168	71
256	412	994	739	344
257	-236	1661	696	684
258	2190	-821	1694	-1341
259	707	-1508	-61	-1107
260	-1198	-336	-93	-370
261	803	-459	843	-146
262	-1259	-3665	-436	-2673
263	5678	-2246	1009	-947
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265	52	-234	2248	-1307
266	-54	-723	828	-2248
267	5190	-2683	3314	-4210
268	-1910	-1707	1557	-4536
269	3734	-4448	2791	-1786
270	-681	-2683	3198	-10239
271	20207	-14438	4021	-16989
272	821	-1524	2028	-2000
273	-46	-263	961	1384
274	5621	-3750	8129	305
275	1551	-2159	2029	-226
276	2125	1176	-765	4
277	-261	-979	764	-237
278	1468	-1029	1065	1167
279	823	-1334	409	876
280	1237	1215	1426	445
281	-7	-3	-117	388
282	1877	-90	1748	2169
283	451	-1766	647	-201
284	-1134	-643	735	105
285	2354	-1324	886	2290
286	604	-2078	-1214	-2999
287	4053	-7276	2774	-306
288	386	-1110	1488	-91
289	-670	-24	1432	-173
290	2221	-1788	9716	-1702
291	1343	-2122	3768	-2392
292	-40	-126	283	57
293	-865	807	576	312
294	-1103	-2566	4258	64
295	1057	155	1320	-1680
296	-309	-31	-1	682
297	-36	1863	722	-170
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299	906	-864	110	-2023

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303	4658	-3707	2344	-4721
304	2776	-4606	4730	867
305	-1978	-1217	4881	3162
306	4620	-1838	13993	7419
307	2658	-6355	8561	-700
308	830	-863	-1077	2990
309	-472	-22	1019	893
310	2349	-2474	4673	6953
311	-496	-1994	2199	2257
312	-770	-448	3023	3298
313	-436	876	1691	1622
314	-1138	-987	5905	7911
315	-384	-1468	2813	324
316	-216	-86	-445	259
317	476	148	158	-70
318	-1318	-178	2396	1602
319	1541	-1857	-842	-1390
320	-1104	-2194	991	1366
321	-183	-240	55	218
322	-1676	-612	410	-1323
323	-2785	-515	-599	-1015
324	216	-96	849	-461
325	-281	-375	951	-1161
326	-167	-2080	4014	-3066
327	1276	1228	1647	-620
328	583	785	4	450
329	485	-564	641	434
330	-139	150	-59	-149
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332	320	-625	349	-353
333	1834	-1808	4797	-61
334	634	-246	-651	-3934
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338	2441	-726	1402	99
339	86	-386	-225	-223
340	737	-1069	79	133
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343	2853	-1039	199	-550
344	66	-435	158	405
345	1400	-3094	379	982
346	523	-386	40	-107
347	2380	-4244	-1863	125
348	809	-2683	2140	3902
349	7338	-5424	3747	5313
350	3068	-375	431	-1535
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352	1059	-9447	2592	163
353	285	-3122	-415	518
354	-693	-3599	1614	-3600
355	452	-108	1904	-627
356	-1553	-1954	1475	-1034



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359	-2318	-294	6746	-2268
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362	-206	-159	367	-227
363	827	1979	13	354
364	-1294	427	997	283
365	-10	-137	742	12
366	-1515	-785	3089	-2309
367	547	-1041	1296	-994
368	-2724	-7267	-3470	3280
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370	454	-516	4247	2829
371	976	-2313	1135	-1590
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374	-316	-1407	2417	-1308
375	-1171	1377	1912	257
376	-1645	-2575	-816	1268
377	-562	820	1226	605
378	-1082	-142	1702	2577
379	284	-342	-279	209
380	411	376	97	1217
381	3046	-2378	627	523
382	-83	66	87	-77
383	1852	-2106	-1081	487
384	-232	-762	-1513	252
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386	-1608	-305	-907	-323
387	-1307	-562	-446	-1448
388	1495	-1065	-7	118
389	200	29	-480	-458
390	-142	122	-409	-743
391	1785	-3094	27	-3163
392	-116	210	-157	182
393	418	35	-76	299
394	-47	-863	-67	-218
395	3150	-2681	-1193	-896
396	-99	-429	454	-717
397	2329	-1558	-1614	-2031
398	-81	-2328	-370	-4929
399	9217	-4208	-3606	-8575
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401	-392	53	296	1474
402	2063	-1886	327	-724
403	95	-359	-805	-97
404	6063	-1753	-2742	3384
405	749	-1189	-1409	1212
406	679	-137	-629	1369
407	576	-5069	-3880	729
408	208	1798	-717	1709
409	1360	1858	194	1201
410	-1837	312	568	944
411	2331	180	-1320	315
412	620	-511	138	2351
413	646	-533	-2408	1218

414	1855	1354	-1500	-160
415	6498	-2601	-4574	776
416	989	-5956	-2366	-2583
417	453	-1352	-544	776
418	127	-3113	2159	-286
419	629	616	572	-77
420	241	-2165	-2393	1884
421	253	111	-63	654
422	-717	28	498	27
423	516	-228	-349	-377
424	-972	-1692	-428	-2383
425	-12	-10	40	38
426	-974	-357	392	640
427	1975	1191	-538	-1088
428	-47	-436	-304	584
429	202	-575	-229	-716
430	-107	-424	-1091	-632
431	4219	-2280	-3206	-3278
432	-5301	-3994	60	2715
433	-1568	-1015	309	6296
434	3493	-1401	3514	2795
435	737	-1947	1687	1067
436	5407	-4499	-3109	8521
437	1102	-3285	-263	2446
438	2427	-596	-964	1298
439	345	-862	-375	97
440	-519	-602	-792	885
441	-1069	256	-517	1184
442	-1374	-1187	3730	4497
443	-40	1048	720	1240
444	1030	-1415	-4368	2676
445	-728	62	-337	690
446	-556	-796	230	1395
447	1759	-1011	-2868	-1114
448	-1371	-5701	-528	-6
449	-3030	-1144	-869	29
450	-3378	-3010	-2763	-1943
451	-4054	-1751	-3238	-3947
452	-215	-1657	1001	282
453	-124	-47	-84	74
454	-979	-972	557	-905
455	-2015	-641	-2513	-929
456	4	-867	54	255
457	-34	-172	-70	-71
458	-647	-281	-712	-1043
459	-1029	-55	-872	-3129
460	-1085	638	134	793
461	-483	-2163	-1615	-1069
462	-921	-207	-3274	-198
463	939	-1034	-6613	-4879
464	-3647	-499	-4586	1013
465	-3299	-1829	1729	1373
466	-1472	314	-648	478
467	-3224	-865	236	-2468
468	-140	-1736	-2784	3675
469	109	-674	1718	1344
470	463	907	1230	1868

471	367	-494	-708	753
472	-316	-778	-4421	1091
473	-691	-15	-587	-376
474	-3163	-648	-1817	1144
475	-83	-308	-367	6
476	-1493	-2	-1757	2674
477	1778	-1130	-90	1162
478	300	-264	286	108
479	1769	-3654	-1894	-674
480	-6094	-13540	-1477	-2528
481	-2028	-192	-1053	4005
482	-1446	-8069	-2503	-4816
483	323	-2370	-2421	-1159
484	347	-6855	1048	3343
485	-636	-1544	162	597
486	-3994	-2764	2044	-557
487	-510	-345	1363	-774
488	-211	-4953	1633	-1087
489	-378	-559	550	81
490	1555	-4061	93	-1843
491	-195	-353	69	-82
492	196	-1292	958	1485
493	-69	-9	-191	-255
494	-528	-471	25	389
495	-181	-824	-3944	-1653
496	-15684	-9882	-11257	3356
497	-14361	-3408	-1203	12054
498	-3316	-1761	-4754	364
499	-4565	-344	1059	391
500	-4585	-7075	-7707	13837
501	-1228	-128	-1479	5643
502	249	-1882	2060	6610
503	-1507	-420	230	2383
504	-2804	-4819	-5060	-341
505	-3754	-2544	-251	940
506	-1353	-855	-41	5
507	-972	-36	-140	183
508	3375	-1698	-1958	5789
509	607	-522	-82	988
510	100	-551	-42	1312
511	245	-76	-168	-71

**Table 2.E.8 - Spectral shape (enhancement layer): cb4k[3][64][4]  
dim=4x64 codewords  
16 bits signed  
factor = 2<sup>6</sup>**

index (SE_shape6)	codeword			
0	-3768	947	-5129	3159
1	1856	645	-1585	744
2	-4337	2591	1792	1066

3	-4965	-846	-951	-897
4	-8691	3396	-5050	-2288
5	-2722	-2820	-2120	387
6	-1439	-2113	-366	-1372
7	-3042	265	1742	-5266
8	777	338	487	285
9	8269	4345	-3265	-4617
10	-1473	51	9119	7231
11	-132	126	126	136
12	-899	-707	-2724	-2446
13	1298	2196	-4573	-505
14	-1113	869	733	5361
15	654	-787	-1161	-806
16	-1026	9637	-5916	4157
17	2689	405	86	-1898
18	-6583	15055	4584	5480
19	-5119	3610	8620	-5558
20	-1484	1303	-485	-738
21	390	1243	-86	-628
22	-2005	3752	-2611	-3901
23	283	6610	6930	-15838
24	2401	5663	5252	3885
25	23296	5077	7584	-6529
26	-1723	3993	283	1313
27	558	-1877	2107	-2168
28	-1	379	-887	599
29	1464	5776	-374	-3465
30	38	-1486	-3389	1175
31	326	736	-98	-2403
32	3768	-947	5129	-3159
33	-1856	-645	1585	-744
34	4337	-2591	-1792	-1066
35	4965	846	951	897
36	8691	-3396	5050	2288
37	2722	2820	2120	-387
38	1439	2113	366	1372
39	3042	-265	-1742	5266
40	-777	-338	-487	-285
41	-8269	-4345	3265	4617
42	1473	-51	-9119	-7231
43	132	-126	-126	-136
44	899	707	2724	2446
45	-1298	-2196	4573	505
46	1113	-869	-733	-5361
47	-654	787	1161	806
48	1026	-9637	5916	-4157
49	-2689	-405	-86	1898
50	6583	-15055	-4584	-5480
51	5119	-3610	-8620	5558
52	1484	-1303	485	738
53	-390	-1243	86	628
54	2005	-3752	2611	3901
55	-283	-6610	-6930	15838
56	-2401	-5663	-5252	-3885
57	-23296	-5077	-7584	6529
58	1723	-3993	-283	-1313
59	-558	1877	-2107	2168

60	1	-379	887	-599
61	-1464	-5776	374	3465
62	-38	1486	3389	-1175
63	-326	-736	98	2403

**2.E.4 CbCelp**

VQ codebook for stochastic excitation vector for 2kbps

**Table 2.E.9 - 1st stage VXC gain codebook (base layer): cbL0\_g[16]**  
 dim=1 x 16 codewords  
 16 bits signed  
 factor = 2<sup>2</sup>

index (VX_gain1[])	codeword
0	65
1	164
2	325
3	764
4	1505
5	2069
6	2428
7	3257
8	4132
9	4687
10	5729
11	6294
12	7137
13	8037
14	11402
15	18768

**Table 2.E.10 - 1st stage VXC shape codebook(base layer): cbL0\_s[64][80]**  
 dim=80 x 64 codewords  
 16 bits signed  
 factor = 2<sup>15</sup>

index (VX_shape1[])	codeword										
0	0	0	0	1628	0	-9648	0	0	0	0	0
	0	0	0	0	0	1446	0	0	0	0	-886
	0	-5869	5431	0	0	0	0	0	-5434	423	0
	0	0	-10553	0	0	-1118	0	0	0	0	0
	0	0	0	0	-2550	0	0	0	0	0	0
	26	0	350	0	0	-339	5478	-541	0	0	0

	0	0	0	0	2070	-16335	9139	0	2613	-15972
	0	0	9563	0	3610	0	-116	-3282	0	0
1	0	0	0	1475	2107	0	755	0	0	-3955
	-2615	-965	-407	-708	1825	0	-5847	3465	420	-6981
	0	0	-2308	2285	-2949	0	0	6434	-7241	-3798
	3568	1179	0	4944	0	0	-4981	3712	-2667	-6447
	5312	-9873	0	3222	-6426	0	669	5199	0	0
	412	-2163	2657	0	0	1597	2207	2172	-7297	0
	5462	-971	-3483	749	0	2047	-3053	0	0	-11458
	-2981	0	-4517	3871	-931	4693	10075	2245	-2730	6256
2	0	-3703	4128	676	0	-874	4680	-4808	1978	-8956
	1205	8163	0	0	-466	-3607	7256	0	0	-617
	0	0	2287	0	1506	-1742	1919	0	986	5364
	-3468	0	493	4803	0	1028	550	2746	0	0
	116	6542	0	480	1829	-3433	999	-10662	-8148	-985
	-3087	4157	-309	3235	-10231	5358	-4104	-3156	0	4481
	-2203	0	5019	0	6958	0	0	-5297	0	-557
	-4389	0	5956	623	-3635	-3938	344	-2245	5708	1950
3	0	0	1073	-2434	-6808	2035	0	-3928	0	-2511
	-2238	3122	4694	-12171	0	0	-1316	0	552	0
	-1752	2655	693	-948	5612	3864	-196	1977	0	3197
	-2269	776	-2206	605	-901	-1997	-1395	502	-9523	2649
	3943	-2173	-647	325	-3185	-1515	199	-11174	-1384	-6662
	-1582	0	0	-3042	4265	4346	7434	5001	-789	1679
	1369	655	-6038	2194	-1661	0	5080	3424	-1932	98
	-2938	11836	6427	0	0	918	0	-1182	0	-1693
4	-4632	0	336	-2778	-4905	-3720	57	195	6479	4647
	6028	-3904	3466	0	3156	-1882	1249	-2516	-106	-770
	2434	-7902	7161	-1952	-8165	0	-7560	0	0	-5449
	0	2652	0	696	0	0	-2030	-1140	4057	-2308
	1796	-4008	0	0	0	-1746	0	1361	0	1440
	-8255	2610	-5339	-463	-2478	-4294	-3816	-4121	0	-6542
	0	-4338	-576	0	0	7433	0	5051	-1554	-8066
	7895	-2736	7067	289	2992	0	0	4541	11	0
5	0	0	0	0	0	0	-2440	0	2741	0
	0	0	4142	0	0	0	5335	0	10364	0
	0	0	0	-5110	0	0	0	13977	-90	-2187
	2840	0	10564	0	0	0	0	0	0	0
	0	4350	-3172	1911	2832	0	0	-4587	0	2496
	-4029	-309	3503	6785	0	676	235	0	0	0
	0	0	11368	0	-8166	0	0	0	0	-1013
	-5017	0	143	0	286	-10526	0	0	9233	0
6	2801	3312	0	3537	-2895	3251	-2901	-1217	0	-3317
	-2481	-2216	-626	1046	723	-514	0	-1160	0	7014
	0	-5144	-5483	2965	4999	-3799	-6616	-4771	-4315	-3333
	7797	-488	0	3845	1670	2468	5530	7787	0	-1978
	6397	-2176	0	522	-3652	652	-1728	2135	-3578	0
	-71	0	-11335	1286	-8933	-2911	-377	847	0	-1155
	3491	-1048	-1559	5575	4336	8164	0	0	0	-581
	0	-764	0	-372	-3988	-3748	0	9410	0	-2331
7	-5896	2103	0	0	0	0	0	0	0	0
	0	0	-1125	0	1197	0	0	1459	0	7113
	-4932	0	3371	1837	0	0	0	259	0	-17178
	0	1971	11477	0	-367	-9599	0	3890	3020	0
	0	-3923	0	0	0	0	0	-2635	0	0
	0	0	0	0	5266	-2257	0	2639	0	6862
	0	-2905	846	0	0	7696	-6590	-293	0	0

	0	8117	0	9712	3390	0	0	0	0	0
8	-1065	-5252	-2075	3686	3042	-2580	556	-5459	4931	-1895
	-3616	162	-392	5018	-1610	-101	2791	8132	38	-4170
	6590	5939	0	4939	0	2589	3788	-197	0	5150
	2915	-4902	0	-1952	-6332	-6494	-4614	5070	0	6238
	-1369	213	-3010	-579	745	2480	0	1121	-8112	0
	0	-7134	-4975	0	0	5318	1336	0	-516	-5017
	0	0	8462	0	0	564	0	4592	-9288	0
	-1438	1780	1410	0	-1208	-3160	1072	2857	5309	-3738
9	849	0	1435	-552	0	-8434	1680	1106	3764	-1977
	98	0	-5953	0	14849	4047	-3176	0	130	0
	3853	-3289	-5691	5670	1809	-1264	-329	0	-1706	0
	-852	-11144	3495	-3522	-2924	4615	-2256	960	-705	-2819
	266	-3377	-2016	0	0	0	1344	2383	7518	-971
	1613	1710	-3127	-2451	-2077	2992	-10137	0	-265	2079
	0	-8393	0	2089	446	-334	0	-2613	0	-153
	3342	0	4900	-1937	-7294	1668	0	1973	0	-235
10	0	-3054	1210	55	0	10118	3822	-1164	4682	1485
	-3077	1877	-5951	0	-3389	-3625	2156	0	-3513	0
	-4489	5203	4396	0	-1647	0	-443	-1805	0	4814
	7914	-696	0	-973	0	0	-2873	3341	2900	0
	4343	-5076	1170	638	-2612	-1071	0	0	-2392	0
	3912	-7665	2586	-322	-2691	-202	-10380	-9307	4649	3806
	-1057	1274	1739	0	-592	7202	-1524	5719	0	-789
	5937	7425	5298	-2991	0	1659	-4841	283	3365	-2322
11	0	0	-1134	0	3516	0	-4115	4584	1049	-731
	0	0	392	-5397	3288	0	4564	-11448	0	1214
	0	-2453	-867	0	2145	-1018	0	1303	0	1205
	-1938	-1729	371	0	0	1984	265	469	-4587	-3683
	-4090	0	-786	5759	-3493	7663	-685	2011	-1585	0
	5208	4688	2324	-2484	-983	0	-141	-9515	-2796	4548
	3400	-4490	2241	-173	321	5670	-176	-482	10008	-1300
	3679	5685	6846	6285	-7339	-60	4848	-1152	1883	-8428
12	5050	-2352	0	-2587	-33	-5587	-7823	3089	273	1542
	-1374	0	638	2094	2421	-1455	6703	4091	4090	-6905
	3113	0	-1907	-1358	0	-558	0	0	176	12935
	-2019	-4243	0	0	185	-2409	-412	0	-2999	541
	0	-4530	1876	0	0	-8879	-862	-1789	9535	-347
	-4275	3231	0	189	-3061	-3445	-1998	-782	1297	0
	3723	-109	2703	-1393	-3715	3781	-614	-6610	0	779
	3172	-6262	-1869	1864	-3723	9765	0	2667	3084	6209
13	0	0	-6233	0	4287	5387	2358	1427	-3202	4462
	-1146	-91	436	3305	-684	4828	3859	3954	10108	-167
	-7522	824	-6239	-1820	0	2383	3595	141	-5560	697
	-5941	-3208	-3586	-2679	3457	2050	0	-6246	-5370	-892
	0	6372	-2173	4247	0	-3670	4879	1486	447	-107
	-3250	2538	1010	-4783	-1306	-2882	-1413	0	2587	1033
	6844	-46	4332	0	0	-1890	2144	6195	-2670	4559
	-1876	0	2828	-9121	0	-2064	0	-8753	-412	-1143
14	0	0	0	9266	0	-4289	0	0	15630	0
	0	0	0	0	0	0	0	0	-3536	0
	0	0	0	0	0	1043	0	0	-1938	-3441
	0	0	0	0	1315	3492	6433	12535	-15578	0
	3885	0	0	6219	-5423	3477	0	0	0	0
	0	0	0	-2157	0	0	-5362	0	0	0
	1369	0	4980	0	0	0	-986	0	0	0
	6508	0	0	0	0	0	-6175	2022	0	0

15	0	1000	0	0	0	4115	-411	0	0	0
	0	0	0	0	4732	0	0	0	0	0
	-552	-1705	0	0	0	2493	4927	-1114	1317	0
	7065	0	0	0	4801	0	0	-5461	0	0
	0	0	0	21047	0	0	0	0	0	12979
	0	0	0	1009	0	0	-4331	0	0	4513
	0	0	5996	0	0	-763	0	5993	-1291	0
	-7022	8196	0	5302	0	-4124	-2552	0	0	0
16	0	0	0	0	-5383	-7389	0	0	0	0
	0	-13760	0	0	-3669	0	0	-1643	0	0
	0	2977	0	4044	0	0	0	0	0	6383
	13648	0	-6472	0	13024	4114	0	0	-1042	0
	0	0	0	0	0	0	3187	5351	0	0
	0	0	8936	0	0	0	0	0	4985	0
	1943	-8121	4195	0	0	-7224	0	-839	0	0
	0	0	241	-1024	4255	0	0	0	0	
17	0	-352	-9216	-2509	0	1151	0	1168	0	6570
	-830	0	0	0	-2908	8076	0	5816	5142	0
	7360	-713	-7775	-276	42	-885	0	6374	3988	0
	-1463	3099	0	1129	-296	1663	-9442	-1374	395	-2783
	0	0	0	0	-6757	-432	-2752	268	2439	27
	814	0	8202	1398	3575	5765	0	5918	-4320	5177
	-343	-3570	0	0	-362	0	-3256	9844	-3848	0
	-5975	781	-6308	5389	-564	-2132	0	1289	0	3434
18	11485	4693	-1464	0	1169	-373	-2657	-1288	-3197	0
	2976	819	-5121	0	290	-6066	3512	0	-4592	-3678
	-5017	7883	2549	-2245	0	0	-2861	0	0	3968
	0	-105	8171	0	-182	-5712	-1732	-2874	-51	-7855
	5330	-4002	0	2438	2085	128	-7971	647	1517	-1079
	-2742	8536	4959	-11	351	2243	314	3201	0	-4812
	4864	-3055	-693	-271	0	0	0	6266	-709	4732
	-6127	740	-3295	553	1130	0	4146	-6180	2141	3239
19	0	0	0	0	0	0	0	0	-6369	0
	0	0	-8103	-12722	0	0	0	0	0	-5372
	0	0	4509	0	8314	0	0	0	6881	0
	-2386	0	0	0	0	0	-985	0	0	0
	0	0	-1170	0	0	0	0	0	0	0
	14144	0	0	0	3394	5389	11441	0	0	0
	0	0	0	0	0	0	0	6490	0	0
	-11175	-7384	-944	-51	0	5487	-2815	0	0	0
20	0	0	7331	-30	772	-3345	-4	5323	0	0
	-3825	861	4892	-3041	0	0	-5795	4004	2822	-3616
	-3113	-1839	0	-3353	-2123	4453	2619	-2365	0	-573
	0	8823	4250	4137	11473	5642	0	4533	-1725	0
	0	1444	789	1955	0	-939	2634	5827	-2601	0
	3843	3417	-4439	8355	0	-1822	13	-1928	2517	6680
	-1708	0	0	0	2300	-1011	0	-3620	6077	-9616
	0	-2230	0	2865	1787	0	-2575	-475	-9613	1683
21	-6370	3044	-1614	5027	0	0	-188	-4008	4652	1786
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	203	508	-343	-5721	9816	8157	0	-511	-6950	1274
-2501	217	1725	8232	953	0	4249	-1549	1561	1298	
54	4120	2452	0	0	0	7557	253	-5271	4395	0
	-1383	-1664	448	0	6496	0	-8286	-6876	2217	2241
	5546	4159	-3094	-89	-1928	1582	2991	6757	0	-8154
	6394	-1902	3271	-2294	-5468	-8427	-110	-1715	445	-8136
	0	0	0	-7386	1552	-8494	0	0	-2300	689
	3332	3630	0	138	-1099	3992	913	-946	0	0
	0	3311	0	0	5438	-3781	4147	0	1997	7183
-2832	1887	3664	-251	-147	0	-2943	-1761	295	3906	
55	5107	0	0	-5554	0	7204	0	0	3108	-10958
	-2828	2865	4753	3109	5593	3208	1682	1097	4323	0
	0	0	-4268	0	2139	-1085	0	4253	-3418	-2898
	0	-2676	567	4796	-178	0	-1409	0	-6034	-36
	0	354	-6967	2796	-8976	-7085	0	-3037	-2837	0
	-4467	-2711	5297	1146	6279	2830	-5912	4656	-1005	2739
	0	320	-1205	6012	5827	-277	-2814	838	-3587	-1270
959	5659	851	2368	-3944	0	6922	0	-4475	0	
56	-2041	2004	-5807	7	0	-93	0	804	0	-2145
	3096	443	-1422	-4705	0	1012	1070	3094	3329	-3558
	5845	-2769	-6684	-13751	159	4829	0	0	-4992	630
	394	1387	-870	5289	0	-1702	-6045	2847	-638	0
	-120	-2919	-2205	-5628	2922	0	-3884	-1216	2743	8517
	0	0	0	2379	-3830	-3287	5517	0	3794	-1250
	0	5108	0	-6727	-3890	0	-8902	-2978	0	2671
-378	-2132	7835	-2072	7198	-3121	-3499	0	701	0	
57	1084	9811	2812	-307	1685	-1395	-2176	-874	-938	-2237
	0	0	-2626	1287	2104	1192	-5465	-1341	279	0
	-9960	-4426	3028	6230	8698	0	480	2983	-2306	-1797
	-215	4147	323	-455	4713	0	1598	12536	0	3033
	1293	-5039	4209	2357	-3772	-3194	1317	6097	97	0
-3311	0	-2859	3491	946	2342	2145	-2822	-2197	785	

	1301	6387	4577	0	-6077	2620	2301	-2787	5771	-359
	-3002	2689	2601	-5782	1777	-1149	0	902	2232	6885
58	1706	-3623	7171	-5205	717	-819	0	5399	3431	9984
	0	895	812	-2398	0	230	6871	0	-5805	0
	0	1062	-6539	-2553	3109	511	8288	-1209	-4089	-3174
	-1641	0	1375	-6902	-1479	-4840	-129	3638	0	5979
	0	0	-2057	-1653	-3548	4513	-1652	0	5528	0
	0	-2222	318	-1322	-4640	-8416	1588	0	-5503	6732
	0	0	-2958	3042	0	-3508	-9357	-2262	-4697	-4754
	-602	-455	-2598	0	-1652	-5161	1436	0	-1153	2628
59	0	0	4122	-2261	-2255	-5431	-165	-232	0	1523
	-2961	-8009	-1679	0	-2634	-5995	5033	-2076	-4643	-3632
	-10186	2828	-3728	0	3280	264	2782	-2747	4782	-4130
	-3962	-1547	2122	0	-3027	0	0	5551	-1812	3402
	1021	-2948	-5856	3754	-5926	0	832	4974	-1819	-3626
	-4830	-758	0	0	0	-7293	0	7381	1682	-2552
	-2267	3943	3057	77	0	7395	0	0	0	-8258
	1749	-5276	-911	1313	-6356	-101	-7532	-1735	0	0
60	-9072	1778	-3454	0	-7230	-4572	0	5294	-2407	-4007
	460	0	3073	-7235	2091	0	0	0	-530	-2369
	-345	906	5179	464	4748	-8088	5287	2219	-5722	1440
	3436	303	-765	-445	-3825	0	0	-1555	6168	-7294
	4325	161	2464	0	3557	-4776	-6953	-1438	0	-2214
	-3138	-8154	-3604	-226	0	4286	98	1317	0	0
	-6270	-883	1838	-4342	-1714	0	-543	7746	0	5677
	0	0	-605	-1272	6247	-4822	0	406	5400	0
61	1097	0	3924	8023	0	-2634	0	0	0	0
	6244	0	10975	2833	0	0	0	5461	0	-2082
	0	0	0	3415	9918	-1542	-4593	0	0	0
	0	-5352	0	0	8238	0	0	12856	0	0
	0	-3890	4495	0	0	0	0	2047	0	0
	0	0	5080	0	0	4531	0	0	0	12455
	-1336	-2662	3712	473	0	3681	-4013	0	0	2858
	0	-30	0	0	1423	0	0	0	9231	0
62	1128	0	-3372	-2295	3313	7195	-1280	0	-1432	-418
	0	2962	3011	-1226	-5502	-4299	-9680	-6982	-66	-6894
	0	1637	-3362	-5338	-1142	-467	-1384	2865	0	0
	-1018	-1847	-368	-11959	0	-7847	-4692	4576	3965	2424
	1024	-4240	1882	1208	0	3357	0	2708	0	1789
	5718	2094	-1961	-791	0	0	0	0	3306	-1323
	-2281	-940	6635	-4014	0	-4353	-2789	0	-4408	-3033
	-7680	-3426	-6043	-1490	-3924	-4250	2864	0	2128	-5098
63	0	0	3227	0	3958	-67	-9489	2570	2475	2381
	3205	4690	8091	0	2284	58	0	0	0	1318
	-5077	-2795	-4787	0	3518	707	515	-1896	10246	-563
	-6495	-3516	-5934	-4480	-976	-12457	0	2461	0	1943
	473	-5375	0	-3354	-620	-3095	2340	1644	-1973	0
	0	-3	593	0	689	0	-5169	2305	3561	-4592
	1724	-2567	2833	0	-7678	-3137	-636	0	896	3601
	1221	866	-6375	-3067	341	-945	5307	4781	-6805	2

## 2.E.5 CbCelp4k

VQ codebook for stochastic excitation vector for 4kbps

**Table 2.E.11 - 2nd stage VXC gain (enhancement layer) codebook cbL1\_g[8]**  
 dim=1 x 8 codewords  
 16 bits signed  
 factor = 2<sup>1</sup>

index (VX_gain2[])	codeword
0	29
1	351
2	1523
3	820
4	16956
5	2571
6	7670
7	4303

**Table 2.E.12 - 2nd stage VXC shape (enhancement layer) codebook cbL1\_s[32][40]**  
 dim=40 x 32 codewords  
 16 bits signed  
 factor = 2<sup>15</sup>

index (VX_shape2[])	codeword										
0	0	0	0	0	-3670	0	-3592	6877	0	0	0
	0	0	0	-1882	0	0	0	5892	0	0	5063
	0	0	0	0	0	11204	0	0	0	0	0
	-619	-6147	0	0	0	0	9657	8329	16851	-18053	0
1	0	0	0	-3118	0	0	-5467	0	9244	0	0
	2584	0	1675	0	5550	-786	5419	2660	0	0	15964
	-14118	-15419	0	0	0	0	-1030	0	0	0	0
	2789	1168	5227	0	0	0	0	-7551	0	0	-9213
2	1105	2945	7478	0	2906	542	-538	-1123	6571	0	-9293
	8215	5999	5572	0	1720	-2450	12025	3924	-8051	0	-2883
	2211	0	3123	-299	5563	2605	8868	2438	2353	0	6959
	-242	-675	-1992	-5682	-4249	12192	-4202	0	6805	0	-5750
3	54	9003	0	2103	0	266	-7156	-846	3992	0	0
	4324	5100	-6345	0	1619	-3191	0	6368	0	0	0
	0	-6450	-1637	-2510	-4786	-8345	-4638	-7092	-4275	0	4220
	16349	-4636	0	0	-10748	0	8287	0	0	0	-7619
4	0	0	0	0	0	0	0	0	0	0	0
	0	-30522	0	-6518	0	0	0	0	0	0	0

	0	0	0	2300	0	3129	0	0	0	0
	0	0	0	5230	0	0	0	7566	0	0
5	0	-532	0	6930	0	0	0	0	0	59
	0	-8218	0	0	1674	0	-24232	0	0	712
	0	0	0	0	0	0	0	0	0	0
	0	10185	-3778	9	-3862	-7654	0	0	-13262	0
6	0	0	0	0	-2098	0	0	-19805	0	0
	795	-10197	0	0	0	0	0	10192	-11014	0
	-1663	0	0	0	2713	0	0	0	0	0
	0	0	0	0	-18106	0	3056	0	0	0
7	0	-1230	0	0	4856	15371	0	-7639	0	16108
	7534	0	0	0	0	0	4774	0	3514	0
	0	0	-1939	0	0	-4330	0	0	-16240	0
	0	0	0	0	-8721	-6353	0	0	0	0
8	-7583	-949	0	1701	-4425	0	0	-8468	-7271	0
	1000	3659	-1614	5090	-6310	-5811	0	-535	13517	0
	0	3450	-1640	0	2814	-4112	-2567	6184	508	10094
	0	-1806	5543	-1061	-4553	0	-12870	-3778	10647	5770
9	-5533	-9268	0	5815	-456	0	0	3105	5292	8540
	2957	11480	-877	1426	-1279	4972	3974	5746	-614	15998
	-8941	404	-6202	0	0	3109	-4787	-3417	1754	0
	3235	-6437	7917	4443	-3728	725	-4565	0	-1519	0
10	0	0	0	0	0	0	0	15292	0	4112
	-3166	-131	20232	0	0	0	-7397	0	0	0
	0	15002	0	7723	0	0	0	5995	0	0
	0	3968	0	-3532	0	0	0	0	0	0
11	0	1009	0	-6702	-793	7142	419	7381	6528	1006
	-10093	10645	-11016	735	0	-7898	-3148	426	494	2241
	13739	7757	3838	-753	-4526	0	6304	0	0	-5553
	-1579	-2655	0	3192	0	0	-4345	-2671	-7876	0
12	0	-2889	-207	240	4040	5354	0	-12614	0	-4645
	3752	-14607	4174	-2397	1174	0	1403	3077	6548	0
	10126	0	-2811	0	2615	8341	6045	0	2943	0
	-341	0	10856	4636	5544	-11308	346	189	1374	1064
13	0	-17688	14233	7304	3965	1493	0	0	0	0
	-3907	-2200	0	0	1184	0	0	0	0	0
	0	-4015	0	0	0	0	0	0	0	0
	0	17747	0	0	-4870	0	10521	0	0	0
14	-1201	-7227	-8405	0	2758	4653	-3605	-4825	-4317	-7995
	938	-7482	190	4838	3207	0	-2449	-8396	1109	-10123
	-225	13603	5322	5193	6497	-920	4448	587	-122	-1543
	6005	-4309	1921	3173	7250	5770	0	-8388	0	0
15	0	-3983	0	-2562	19732	0	0	0	0	0
	0	0	-6400	0	0	-11621	2715	0	0	-1657
	0	6695	0	-2777	0	-11764	8241	0	-13969	0
	2647	0	0	0	0	2337	0	0	3062	0
16	0	0	0	0	3670	0	3592	-6877	0	0
	0	0	0	1882	0	0	0	-5892	0	-5063
	0	0	0	0	0	-11204	0	0	0	0
	619	6147	0	0	0	0	-9657	-8329	-16851	18053
17	0	0	0	3118	0	0	5467	0	-9244	0
	-2584	0	-1675	0	-5550	786	-5419	-2660	0	-15964
	14118	15419	0	0	0	0	1030	0	0	0
	-2789	-1168	-5227	0	0	0	0	7551	0	9213
18	-1105	-2945	-7478	0	-2906	-542	538	1123	-6571	9293
	-8215	-5999	-5572	0	-1720	2450	-12025	-3924	8051	2883
	-2211	0	-3123	299	-5563	-2605	-8868	-2438	-2353	-6959



	242	675	1992	5682	4249	-12192	4202	0	-6805	5750
19	-54	-9003	0	-2103	0	-266	7156	846	-3992	0
	-4324	-5100	6345	0	-1619	3191	0	-6368	0	0
	0	6450	1637	2510	4786	8345	4638	7092	4275	-4220
	-16349	4636	0	0	10748	0	-8287	0	0	7619
20	0	0	0	0	0	0	0	0	0	0
	0	30522	0	6518	0	0	0	0	0	0
	0	0	0	-2300	0	-3129	0	0	0	0
	0	0	0	-5230	0	0	0	-7566	0	0
21	0	532	0	-6930	0	0	0	0	0	-59
	0	8218	0	0	-1674	0	24232	0	0	-712
	0	0	0	0	0	0	0	0	0	0
	0	-10185	3778	-9	3862	7654	0	0	13262	0
22	0	0	0	0	2098	0	0	19805	0	0
	-795	10197	0	0	0	0	0	-10192	11014	0
	1663	0	0	0	-2713	0	0	0	0	0
	0	0	0	0	18106	0	-3056	0	0	0
23	0	1230	0	0	-4856	-15371	0	7639	0	-16108
	-7534	0	0	0	0	0	-4774	0	-3514	0
	0	0	1939	0	0	4330	0	0	16240	0
	0	0	0	0	8721	6353	0	0	0	0
24	7583	949	0	-1701	4425	0	0	8468	7271	0
	-1000	-3659	1614	-5090	6310	5811	0	535	-13517	0
	0	-3450	1640	0	-2814	4112	2567	-6184	-508	-10094
	0	1806	-5543	1061	4553	0	12870	3778	-10647	-5770
25	5533	9268	0	-5815	456	0	0	-3105	-5292	-8540
	-2957	-11480	877	-1426	1279	-4972	-3974	-5746	614	-15998
	8941	-404	6202	0	0	-3109	4787	3417	-1754	0
	-3235	6437	-7917	-4443	3728	-725	4565	0	1519	0
26	0	0	0	0	0	0	0	-15292	0	-4112
	3166	131	-20232	0	0	0	7397	0	0	0
	0	-15002	0	-7723	0	0	0	-5995	0	0
	0	-3968	0	3532	0	0	0	0	0	0
27	0	-1009	0	6702	793	-7142	-419	-7381	-6528	-1006
	10093	-10645	11016	-735	0	7898	3148	-426	-494	-2241
	-13739	-7757	-3838	753	4526	0	-6304	0	0	5553
	1579	2655	0	-3192	0	0	4345	2671	7876	0
28	0	2889	207	-240	-4040	-5354	0	12614	0	4645
	-3752	14607	-4174	2397	-1174	0	-1403	-3077	-6548	0
	-10126	0	2811	0	-2615	-8341	-6045	0	-2943	0
	341	0	-10856	-4636	-5544	11308	-346	-189	-1374	-1064
29	0	17688	-14233	-7304	-3965	-1493	0	0	0	0
	3907	2200	0	0	-1184	0	0	0	0	0
	0	4015	0	0	0	0	0	0	0	0
	0	-17747	0	0	4870	0	-10521	0	0	0
30	1201	7227	8405	0	-2758	-4653	3605	4825	4317	7995
	-938	7482	-190	-4838	-3207	0	2449	8396	-1109	10123
	225	-13603	-5322	-5193	-6497	920	-4448	-587	122	1543
	-6005	4309	-1921	-3173	-7250	-5770	0	8388	0	0
31	0	3983	0	2562	-19732	0	0	0	0	0
	0	0	6400	0	0	11621	-2715	0	0	1657
	0	-6695	0	2777	0	11764	-8241	0	13969	0
	-2647	0	0	0	0	-2337	0	0	-3062	0

2.E.6 CbLsp

Table 2.E.13 - LSP table for the first stage (base layer) lsp\_tbl[0][32][10]  
 dim=10 x 32 codevectors  
 16 bits signed  
 factor = 2<sup>15</sup>

index (LSP1)	codeword									
0	3375	4912	6549	9040	10942	13276	18798	21307	25336	28572
1	2974	4358	6129	8373	10084	12442	19149	23195	26224	28839
2	3077	4360	6147	9751	12072	14239	16729	19410	25495	28508
3	2919	4067	6571	9049	11301	13176	15546	21792	25596	28758
4	3935	6019	9204	12175	15469	18353	21731	24243	27476	29137
5	3746	5894	8346	10956	14067	16427	19980	22820	26182	28388
6	3138	4830	6689	9712	13093	14804	17683	20287	23614	27858
7	3660	5585	8408	11300	13434	15086	18394	21960	25381	27908
8	2546	3478	5301	11498	13855	15746	18993	21490	26264	28407
9	2415	3239	5561	12877	15707	17387	20352	22286	26337	28318
10	2887	5150	8479	11388	14513	17282	20858	23623	26958	29134
11	2229	4162	7949	11474	14612	16990	19987	22614	25791	28349
12	2086	3932	8403	12430	16065	19196	22252	24785	27464	29264
13	3539	5949	9973	13626	17153	20423	23144	25356	28016	29594
14	3748	6180	9973	12764	15130	17300	20631	23046	26361	28654
15	4005	6856	12057	14541	17344	19943	22286	24277	26824	28540
16	2959	4902	6772	9100	12970	14348	17631	22669	24817	27497
17	1659	3244	6690	9923	13244	16000	19579	22730	26119	28767
18	2116	2880	4959	10683	16288	18759	21936	24273	27098	29082
19	1623	2982	7079	10655	14022	17143	20470	23340	26641	29021
20	2954	4057	5645	9387	15193	16955	19540	22376	24759	27631
21	1829	2579	4149	9097	13989	16374	19707	22537	26415	29208
22	2395	3424	5074	8394	15219	18639	20682	22949	25314	27660
23	2155	2832	4495	7776	13914	18654	22078	24004	26849	28673
24	2853	4870	7173	10343	12233	14846	19653	22280	26468	28773
25	2952	4252	6548	8892	10909	16697	19529	21846	25191	27573
26	2438	3437	5374	7557	9440	15048	19085	22361	26340	29023
27	2745	3740	5868	8032	10298	16927	21146	23387	26806	28435
28	2212	3253	5397	9031	13431	15972	19009	20587	24238	28524
29	2327	3469	5224	7762	13556	15354	18303	22448	25218	28360
30	2114	3126	4926	9253	11770	14176	19094	21896	25847	28980
31	1877	2610	4300	7363	12346	16129	19777	22688	26019	29255

Table 2.E.14 - LSP table for the second stage of VQ (base layer) with inter-frame prediction  
 pd\_tbl[0][64][5]....Lower 5 LSPs  
 dim=5 x 64 codevectors  
 16 bits signed  
 factor = 2<sup>18</sup>

index (LSP2&0x3f)	codeword				
0	2981	1974	3673	2079	3154
1	467	1177	2624	4315	-2970
2	7204	-2351	-225	2548	1793

3	3151	1499	1654	7626	8950
4	1476	2753	-2614	5096	4202
5	2121	4158	5054	-4650	-4200
6	3688	5151	590	7476	-4205
7	-435	1111	1001	-257	3523
8	199	1958	-2865	-1864	4551
9	-2645	-3880	1767	6223	2994
10	2283	5489	4357	412	-7054
11	-1025	668	-181	4580	713
12	506	-401	-3597	8910	9823
13	-1358	-388	624	3479	5877
14	-501	-886	-506	10302	4008
15	-4779	3673	611	3201	3169
16	1979	3589	779	1408	7565
17	2129	6236	2383	5214	1688
18	5956	11047	-2496	1672	-1735
19	1969	1523	1908	7647	1473
20	5550	9285	7505	6436	5741
21	6357	13996	6349	-2076	-2341
22	7812	4527	1326	6386	2207
23	3594	4936	-2939	1523	288
24	755	1649	3576	-1211	-401
25	-239	-1389	-2181	765	7891
26	1617	852	3819	15383	92
27	792	-1261	2456	3584	1098
28	1575	1148	3979	7198	-8126
29	2071	1214	-574	1085	307
30	3397	501	4493	1132	-3308
31	-844	794	6753	8625	-1594
32	440	-1832	3555	-1777	2459
33	797	-1148	-936	1547	3481
34	3329	477	-2144	5308	-2666
35	149	-4244	8061	3492	739
36	855	2485	7891	-6944	404
37	4160	6139	-4802	-5122	3500
38	637	1712	6758	7673	4768
39	-63	-2758	1751	8480	-2330
40	-2645	-3741	2750	-225	7649
41	-2021	-4108	6286	11770	7086
42	-8	3906	1311	1583	-1916
43	-3064	-2095	-1437	4685	10533
44	-1667	6748	8748	2755	-1358
45	-60	994	5023	2697	9461
46	-2593	-121	4202	2273	771
47	-4401	-724	12926	3948	3434
48	1274	4462	894	-2286	1214
49	4538	9141	-1080	962	6478
50	5405	1950	1001	-4055	551
51	1395	4590	16809	6637	1025
52	4538	6252	6019	-252	2855
53	8968	5665	1282	-199	-2789
54	3869	3712	6493	4913	-933
55	4045	2713	438	304	-3185
56	-1114	3025	6289	-367	3639
57	907	-1691	-3196	5450	2254
58	1311	2721	-3743	551	13002
59	-1077	10402	2066	-398	907

60	1211	1067	1772	-5211	5101
61	3953	66	-3337	-1368	5204
62	744	629	1077	-2087	11838
63	-304	-299	8247	-414	-3557

**Table 2.E.15 - LSP table for the second stage of VQ (base layer) with inter-frame prediction**  
**pd\_tbl[1][16][5]....Higher 5 LSPs**  
**dim=5 x 16 codevectors**  
**16 bits signed**  
**factor = 2<sup>18</sup>**

index (LSP3&0x0f))	codeword				
0	9524	346	-2981	267	-336
1	2629	-5468	-1733	7104	2469
2	2336	3321	-2915	6391	2404
3	-3961	-370	1623	9820	4352
4	9849	6079	3437	7157	2553
5	3586	755	1937	-1298	2530
6	5240	8829	-624	-1243	-464
7	-2223	3743	-31	1688	5419
8	6027	-2265	2632	3738	-1646
9	-792	-3861	5951	2705	-160
10	671	2637	1455	1216	-2724
11	-4336	2679	7699	1557	-186
12	9484	2750	3806	-2873	-4066
13	3054	4257	8559	6189	1531
14	1922	4716	7917	-3945	-2375
15	-881	10551	3992	3806	2097

**Table 2.E.16 - LSP table for the second stage of VQ (base layer) without inter-frame prediction**  
**d\_tbl[0][64][5] ...Lower 5 LSPs**  
**dim=5 x 64 codevectors**  
**16 bits signed**  
**factor = 2<sup>18</sup>**

index (LSP2&0x3f))	codeword				
0	-223	1541	5665	3589	4611
1	-1733	5350	5348	-2451	2902
2	4365	1696	-1562	10659	-2362
3	2954	3694	8486	1274	1578
4	4567	7319	2747	1699	8294
5	-47	2110	650	2165	983
6	7083	9013	-2564	-8268	3416
7	3979	278	4509	2923	-1523

8	-2687	-4168	10231	5379	2732
9	-1269	-2312	15862	4470	-5888
10	1927	383	9885	10635	102
11	3358	5584	194	4881	-9241
12	-29	-81	-454	3785	7969
13	3743	6973	7581	-6653	637
14	3510	3856	8617	4687	-6399
15	10040	9558	5277	2257	13
16	-3358	2847	9891	3261	-1167
17	-6430	-1893	9864	11762	6879
18	-191	1827	2063	7943	-3277
19	-1903	-3143	5510	-446	7222
20	-16	-2212	-3951	10250	4797
21	-4687	548	2074	7198	3906
22	2074	1735	-1153	10121	12945
23	-377	-419	6189	10315	-8486
24	-7073	1169	3332	2551	9020
25	-4939	-8029	17592	12255	703
26	-2564	6800	2024	4650	71
27	-1458	6008	13668	5980	5924
28	-3209	-5445	-3395	14132	7644
29	-687	7188	8622	-2095	-6593
30	-912	2008	-5607	6719	3109
31	3513	6559	6921	8247	1853
32	4669	17485	10142	-1971	-2771
33	3135	5544	-954	-9437	8593
34	10533	126	-6483	-3484	419
35	8284	3707	-1059	3534	-2933
36	10785	12945	-1140	-8242	-8908
37	5337	-89	-3880	-404	8646
38	6491	6700	-1067	-2202	1523
39	2485	-1591	3112	-2773	4467
40	1725	2511	7398	1536	11975
41	3169	5062	886	6984	3715
42	6868	4021	3720	-3122	-6226
43	8420	8389	-8045	734	322
44	5730	2192	1583	2847	5028
45	6637	-4359	1064	6286	2658
46	1316	2881	3332	-2060	-3492
47	3445	6693	2477	328	-1851
48	448	-3172	4742	3854	760
49	-1769	-1250	-6430	2102	13361
50	4126	666	-2931	336	477
51	3101	3133	1699	-4653	3374
52	5838	996	-8074	6229	3678
53	328	294	8632	-553	-4315
54	844	-147	4902	10163	7196
55	-2157	821	-881	-2902	7178
56	-2763	-5652	1966	6113	8562
57	-1442	-4297	1481	8989	385
58	1667	2842	-637	-3439	13508
59	3287	4981	-8630	2231	10179
60	338	2173	10124	-7749	3300
61	1201	5500	-3513	747	3932
62	-3038	12727	2823	-4664	2621
63	-1376	-1489	634	15435	-1937

**Table 2.E.17 - LSP table for the second stage of VQ (base layer) without inter-frame prediction  
d\_tbl[1][16][5] ...Higher 5 LSPs  
dim=5 x 16 codevectors  
16 bits signed  
factor = 2<sup>18</sup>**

index (LSP3&0x0f)	codeword				
0	9875	-2490	3812	946	-2427
1	3125	-3028	7267	8889	954
2	11600	5883	2380	-3463	-4367
3	2878	8431	9744	-3673	-4653
4	3400	1762	-202	-380	-1056
5	-6367	2517	7846	2265	-58
6	8800	10082	7710	7075	2535
7	2302	6040	3728	6978	1756
8	2412	-3458	-2760	10473	4608
9	-7044	-2173	3982	11235	7959
10	11136	1680	-4422	5780	2074
11	1387	965	4569	802	2606
12	-1717	5264	-3919	6189	5738
13	-2530	8092	12281	9626	4338
14	5219	9477	-2813	-852	1308
15	-4401	12027	3458	-1256	1859

**2.E.7 CbLsp4k**

**Table 2.E.18 - VQ codebook for LSP quantization of enhancement layer for 4kbps  
vqLsp[256][10].  
dim=10 x 256 codevectors  
16 bits signed  
factor = 2<sup>18</sup>**

index (LSP5)	codeword									
0	-1488	1955	6590	-1047	-422	-567	-311	46	-916	-397
1	445	1443	1785	6945	2534	806	2409	1184	707	4339
2	553	2225	250	390	3316	4497	-67	-2833	-169	3250
3	1219	1234	1126	639	1238	1503	5514	4088	258	4326
4	-716	-923	1856	-2189	1003	-851	-1693	2519	-3541	-1953
5	3885	-4033	-1171	1778	1840	-201	423	-254	-1660	110
6	532	-674	26	-1549	531	896	-1359	-2189	-817	2616
7	1160	-652	-444	879	-264	1034	1506	-637	-759	5774
8	-284	-767	2894	163	463	532	-1775	-1743	1438	3535

9	14	1248	1582	1472	7825	-41	-1292	-738	638	2979
10	1723	-1233	-729	-2381	2160	951	-366	2293	2087	2766
11	633	-1517	-1386	-104	4742	1236	-553	2077	-1963	-19
12	-889	-1170	1810	-2026	1300	1412	2235	-2597	-365	-73
13	-315	-1085	1493	-804	4255	581	-384	-1215	4135	1961
14	-367	-1149	843	-3832	-751	1773	-141	1380	2936	1253
15	-1436	-2531	-1351	2589	3187	548	-322	1020	2293	235
16	-3201	-3223	7383	297	-543	651	-678	-323	1849	103
17	279	-953	-88	4524	1394	985	-368	-162	3737	869
18	140	1243	1575	1246	1088	12797	-2644	1357	1423	3826
19	276	1147	-496	2601	816	5761	-409	2353	180	3808
20	-1839	850	1755	987	-1721	-136	-1827	1149	1978	2236
21	-1593	-989	2006	1295	-178	383	370	2375	2045	-1645
22	676	706	442	-406	582	3248	-2396	-1511	102	-834
23	958	743	-1318	380	143	1849	618	2813	646	-691
24	-1454	-1321	534	-226	-169	498	-728	-3274	6877	4855
25	-2038	907	756	503	2000	-682	-511	-1216	455	2455
26	654	-1055	80	-932	1317	7334	-623	-987	2224	1816
27	-1838	934	-1394	1145	1012	1086	30	-1165	3178	5932
28	-650	-1000	439	28	855	1335	309	3629	8164	22246
29	-74	624	981	-277	2096	949	-252	1934	2831	1794
30	-390	-824	10	-502	180	1024	-3594	1312	453	13048
31	-441	-689	-748	339	1238	1602	-256	1273	4056	1139
32	-488	847	1551	-1291	-1350	-2360	1121	1975	2557	1389
33	-28	-1340	-598	4263	267	-583	339	2073	-5695	-1087
34	604	-1471	-121	-810	-204	556	8401	9515	-1066	529
35	-990	1082	-143	2570	2955	846	3306	2630	-2567	-953
36	612	-870	816	-2482	-2483	-141	219	-1016	-8297	-4247
37	1846	-970	-555	1981	70	36	-506	-3256	-4191	-3182
38	277	-1461	1531	-1314	-77	-841	5243	-549	-3285	-835
39	-911	-775	878	110	-495	1006	210	728	-3255	7673
40	-435	682	1077	223	1909	-2036	2401	-1488	170	146
41	-1198	-729	200	1756	657	-562	2930	-1896	-760	569
42	-402	-833	-1728	-1750	-410	1849	1699	-567	594	-1553
43	-2410	-992	-784	-55	181	206	2038	-1294	-5254	1128
44	183	-1304	-251	-43	-1774	588	3407	-6621	-2567	-811
45	-968	-765	530	-424	-1	916	945	-1418	-215	-11098
46	-387	-813	-507	-3149	-587	-334	2211	953	-177	2276
47	-4723	-2206	387	-2062	1889	242	-460	295	-50	1006
48	148	-1071	5072	591	-2189	271	-891	1074	1059	-269
49	610	-1281	1949	2982	-300	2178	1236	-1529	1117	2360
50	-964	-1344	808	772	218	4011	1237	11201	5525	6017
51	-1284	-902	-1566	935	1881	2420	2133	1300	-508	-605
52	253	-1230	1713	-143	-4161	-46	972	-3265	2170	-1523
53	293	-985	224	1218	-1262	392	-21	-1539	3888	-4842
54	1135	777	142	-1582	-2476	2111	752	-1979	-819	-3038
55	-1636	-3260	-2040	1427	-1727	1199	704	-383	-1769	-1835
56	84	997	466	1081	-699	-424	-1982	-4373	5391	-2342
57	927	1404	173	1332	715	1537	4062	-3081	-2540	265
58	-469	986	610	-1306	-366	2774	955	1021	1785	4607
59	-45	811	-1953	1637	1037	126	376	-1108	-3463	339
60	-466	1024	-256	-234	-559	-601	2253	-4365	7628	2666
61	250	-820	-1061	-430	-1	-1248	2152	-2411	3407	-9281
62	-923	746	-980	-318	-1947	-611	-1862	-41	2343	4071
63	-1280	-879	-1667	1035	-453	-426	-725	-3454	674	1283
64	2634	1214	3448	394	-462	228	382	-1273	-558	-1341
65	1627	814	332	1085	527	977	-1435	-625	-2965	860

66	656	770	389	-1721	1732	761	140	-1008	-3829	-5355
67	-206	2713	263	421	-59	-440	477	593	616	576
68	595	-700	-107	-1406	-1509	-2131	-50	341	143	4214
69	2611	-1155	-866	1679	-1	1544	-692	2680	893	-1559
70	758	793	-1172	-1539	-2007	-646	323	30	2017	4193
71	2312	981	-173	653	-1087	-1344	2133	-650	28	968
72	-318	999	2863	-1696	307	681	-926	-1649	-39	-781
73	1284	799	891	-801	1564	-2095	-1583	848	-1212	-327
74	1808	1162	66	-4747	3304	1008	-173	-423	-1215	1181
75	2641	1069	-902	232	2953	688	522	268	293	175
76	797	-1150	1524	-4061	3382	-287	-484	-45	-3144	-3137
77	470	-666	-106	-1850	1222	-829	-293	1278	1657	-5862
78	-1069	-1820	-3524	-5746	2775	-406	-445	-44	1401	1840
79	1027	831	-346	950	1179	-982	1914	222	1854	2065
80	-129	733	1895	58	-1454	746	1360	1403	-1829	-3023
81	300	901	1115	2174	-863	-1753	632	6038	-2705	-874
82	-722	886	485	-1167	-214	1057	-3351	2192	-5680	-3009
83	-824	5704	-1072	2166	346	-335	528	304	-376	-858
84	1073	1305	1095	269	-3150	618	-826	2718	2248	6336
85	979	-1038	-1193	-271	-1048	-620	29	5764	4654	2798
86	201	-729	-2020	-1455	-100	-444	-1882	1077	-3250	-928
87	-571	950	-1500	130	-763	1072	-1895	4771	-1534	-3396
88	-158	875	465	-1906	441	72	-1897	-3564	1026	4080
89	-20	982	-334	-101	2873	-544	-2399	-1914	-1003	-1133
90	-1045	729	-1377	-2187	874	2318	-872	-52	-1335	-1465
91	-697	2235	-2466	-177	525	1147	1289	-2700	-424	86
92	697	1379	1799	642	14	-1276	-215	2441	16386	10749
93	-554	-794	-418	-40	1042	-382	-1777	7380	618	-1064
94	1822	1848	-3881	-582	-1192	1700	-540	639	4302	3165
95	166	793	-1864	-171	1504	-175	-1231	-792	3369	1618
96	87	1121	-1578	-1136	-2467	-2734	-7351	-1152	-2241	-8569
97	855	970	-420	301	36	582	-8104	426	-8356	-9449
98	952	1536	432	-2641	-115	-1466	5846	233	-1257	-2578
99	-2224	1399	-1103	-450	1763	-998	1686	1920	-736	-47
100	-50	1189	-1158	-1160	-3325	-15702	-2440	-887	-4671	-1783
101	183	-828	-1639	626	237	395	-2563	167	-1248	-11786
102	-87	841	-985	605	-2211	-1697	5499	238	127	4349
103	-1165	1051	-848	-402	-1601	-332	2467	252	-3375	847
104	253	-1060	417	413	68	-1570	-3920	-10768	1361	-5063
105	-166	-1070	-193	-170	111	-740	-3604	-2427	-11876	-5145
106	-10	616	-659	-1902	294	244	2431	-33	1045	-1334
107	-1057	-726	-302	-1061	1595	-1635	-514	-511	-2395	266
108	-940	-1624	-582	-786	-468	-4660	-1003	-15749	-6529	-2107
109	97	-1067	63	-165	-1211	-1610	224	-1875	-3150	-19091
110	217	-1432	-2153	-1075	833	-1807	3821	-1762	-1981	-121
111	209	1024	-535	-1537	1524	-3995	3431	-1481	2876	-237
112	30	-1606	551	1401	-4167	-1196	-2925	613	-2783	-3431
113	-221	1002	-928	526	-1723	1525	-2327	-2027	-2499	-607
114	-2516	1081	-673	1540	33	-33	-1258	1336	-2577	-223
115	-5703	2336	-1985	1779	197	370	931	-409	540	-540
116	438	1551	-1680	-902	-8699	-284	-91	-1765	-2392	-1349
117	581	1066	-308	265	-1940	72	-584	-344	564	-10666
118	-461	-1095	-3661	-587	-3356	-1699	647	-1080	2132	24
119	-2050	1114	-2993	-786	-1373	-398	258	-373	3718	-35
120	741	1008	-744	-1369	-1576	-2146	-7186	-21217	-308	-2551
121	243	949	-942	-268	122	817	1439	-2225	-15141	-4756
122	-1657	788	16	-902	-1171	1864	324	-2561	397	-3813



123	-819	706	-1349	-35	-290	-2124	-637	-2343	-1058	-890
124	-956	-2053	-1641	-1195	-1850	-4251	-10855	-15001	3517	-3817
125	-417	1287	964	-355	359	-935	-2955	-1426	289	-14426
126	1541	-786	-2131	-433	-802	-932	-2248	-2316	4924	96
127	202	1178	-2964	833	-995	-5359	69	-4858	1256	-4349
128	1488	-1955	-6590	1047	422	567	311	-46	916	397
129	-445	-1443	-1785	-6945	-2534	-806	-2409	-1184	-707	-4339
130	-553	-2225	-250	-390	-3316	-4497	67	2833	169	-3250
131	-1219	-1234	-1126	-639	-1238	-1503	-5514	-4088	-258	-4326
132	716	923	-1856	2189	-1003	851	1693	-2519	3541	1953
133	-3885	4033	1171	-1778	-1840	201	-423	254	1660	-110
134	-532	674	-26	1549	-531	-896	1359	2189	817	-2616
135	-1160	652	444	-879	264	-1034	-1506	637	759	-5774
136	284	767	-2894	-163	-463	-532	1775	1743	-1438	-3535
137	-14	-1248	-1582	-1472	-7825	41	1292	738	-638	-2979
138	-1723	1233	729	2381	-2160	-951	366	-2293	-2087	-2766
139	-633	1517	1386	104	-4742	-1236	553	-2077	1963	19
140	889	1170	-1810	2026	-1300	-1412	-2235	2597	365	73
141	315	1085	-1493	804	-4255	-581	384	1215	-4135	-1961
142	367	1149	-843	3832	751	-1773	141	-1380	-2936	-1253
143	1436	2531	1351	-2589	-3187	-548	322	-1020	-2293	-235
144	3201	3223	-7383	-297	543	-651	678	323	-1849	-103
145	-279	953	88	-4524	-1394	-985	368	162	-3737	-869
146	-140	-1243	-1575	-1246	-1088	-12797	2644	-1357	-1423	-3826
147	-276	-1147	496	-2601	-816	-5761	409	-2353	-180	-3808
148	1839	-850	-1755	-987	1721	136	1827	-1149	-1978	-2236
149	1593	989	-2006	-1295	178	-383	-370	-2375	-2045	1645
150	-676	-706	-442	406	-582	-3248	2396	1511	-102	834
151	-958	-743	1318	-380	-143	-1849	-618	-2813	-646	691
152	1454	1321	-534	226	169	-498	728	3274	-6877	-4855
153	2038	-907	-756	-503	-2000	682	511	1216	-455	-2455
154	-654	1055	-80	932	-1317	-7334	623	987	-2224	-1816
155	1838	-934	1394	-1145	-1012	-1086	-30	1165	-3178	-5932
156	650	1000	-439	-28	-855	-1335	-309	-3629	-8164	-22246
157	74	-624	-981	277	-2096	-949	252	-1934	-2831	-1794
158	390	824	-10	502	-180	-1024	3594	-1312	-453	-13048
159	441	689	748	-339	-1238	-1602	256	-1273	-4056	-1139
160	488	-847	-1551	1291	1350	2360	-1121	-1975	-2557	-1389
161	28	1340	598	-4263	-267	583	-339	-2073	5695	1087
162	-604	1471	121	810	204	-556	-8401	-9515	1066	-529
163	990	-1082	143	-2570	-2955	-846	-3306	-2630	2567	953
164	-612	870	-816	2482	2483	141	-219	1016	8297	4247
165	-1846	970	555	-1981	-70	-36	506	3256	4191	3182
166	-277	1461	-1531	1314	77	841	-5243	549	3285	835
167	911	775	-878	-110	495	-1006	-210	-728	3255	-7673
168	435	-682	-1077	-223	-1909	2036	-2401	1488	-170	-146
169	1198	729	-200	-1756	-657	562	-2930	1896	760	-569
170	402	833	1728	1750	410	-1849	-1699	567	-594	1553
171	2410	992	784	55	-181	-206	-2038	1294	5254	-1128
172	-183	1304	251	43	1774	-588	-3407	6621	2567	811
173	968	765	-530	424	1	-916	-945	1418	215	11098
174	387	813	507	3149	587	334	-2211	-953	177	-2276
175	4723	2206	-387	2062	-1889	-242	460	-295	50	-1006
176	-148	1071	-5072	-591	2189	-271	891	-1074	-1059	269
177	-610	1281	-1949	-2982	300	-2178	-1236	1529	-1117	-2360
178	964	1344	-808	-772	-218	-4011	-1237	-11201	-5525	-6017
179	1284	902	1566	-935	-1881	-2420	-2133	-1300	508	605

180	-253	1230	-1713	143	4161	46	-972	3265	-2170	1523
181	-293	985	-224	-1218	1262	-392	21	1539	-3888	4842
182	-1135	-777	-142	1582	2476	-2111	-752	1979	819	3038
183	1636	3260	2040	-1427	1727	-1199	-704	383	1769	1835
184	-84	-997	-466	-1081	699	424	1982	4373	-5391	2342
185	-927	-1404	-173	-1332	-715	-1537	-4062	3081	2540	-265
186	469	-986	-610	1306	366	-2774	-955	-1021	-1785	-4607
187	45	-811	1953	-1637	-1037	-126	-376	1108	3463	-339
188	466	-1024	256	234	559	601	-2253	4365	-7628	-2666
189	-250	820	1061	430	1	1248	-2152	2411	-3407	9281
190	923	-746	980	318	1947	611	1862	41	-2343	-4071
191	1280	879	1667	-1035	453	426	725	3454	-674	-1283
192	-2634	-1214	-3448	-394	462	-228	-382	1273	558	1341
193	-1627	-814	-332	-1085	-527	-977	1435	625	2965	-860
194	-656	-770	-389	1721	-1732	-761	-140	1008	3829	5355
195	206	-2713	-263	-421	59	440	-477	-593	-616	-576
196	-595	700	107	1406	1509	2131	50	-341	-143	-4214
197	-2611	1155	866	-1679	1	-1544	692	-2680	-893	1559
198	-758	-793	1172	1539	2007	646	-323	-30	-2017	-4193
199	-2312	-981	173	-653	1087	1344	-2133	650	-28	-968
200	318	-999	-2863	1696	-307	-681	926	1649	39	781
201	-1284	-799	-891	801	-1564	2095	1583	-848	1212	327
202	-1808	-1162	-66	4747	-3304	-1008	173	423	1215	-1181
203	-2641	-1069	902	-232	-2953	-688	-522	-268	-293	-175
204	-797	1150	-1524	4061	-3382	287	484	45	3144	3137
205	-470	666	106	1850	-1222	829	293	-1278	-1657	5862
206	1069	1820	3524	5746	-2775	406	445	44	-1401	-1840
207	-1027	-831	346	-950	-1179	982	-1914	-222	-1854	-2065
208	129	-733	-1895	-58	1454	-746	-1360	-1403	1829	3023
209	-300	-901	-1115	-2174	863	1753	-632	-6038	2705	874
210	722	-886	-485	1167	214	-1057	3351	-2192	5680	3009
211	824	-5704	1072	-2166	-346	335	-528	-304	376	858
212	-1073	-1305	-1095	-269	3150	-618	826	-2718	-2248	-6336
213	-979	1038	1193	271	1048	620	-29	-5764	-4654	-2798
214	-201	729	2020	1455	100	444	1882	-1077	3250	928
215	571	-950	1500	-130	763	-1072	1895	-4771	1534	3396
216	158	-875	-465	1906	-441	-72	1897	3564	-1026	-4080
217	20	-982	334	101	-2873	544	2399	1914	1003	1133
218	1045	-729	1377	2187	-874	-2318	872	52	1335	1465
219	697	-2235	2466	177	-525	-1147	-1289	2700	424	-86
220	-697	-1379	-1799	-642	-14	1276	215	-2441	-16386	-10749
221	554	794	418	40	-1042	382	1777	-7380	-618	1064
222	-1822	-1848	3881	582	1192	-1700	540	-639	-4302	-3165
223	-166	-793	1864	171	-1504	175	1231	792	-3369	-1618
224	-87	-1121	1578	1136	2467	2734	7351	1152	2241	8569
225	-855	-970	420	-301	-36	-582	8104	-426	8356	9449
226	-952	-1536	-432	2641	115	1466	-5846	-233	1257	2578
227	2224	-1399	1103	450	-1763	998	-1686	-1920	736	47
228	50	-1189	1158	1160	3325	15702	2440	887	4671	1783
229	-183	828	1639	-626	-237	-395	2563	-167	1248	11786
230	87	-841	985	-605	2211	1697	-5499	-238	-127	-4349
231	1165	-1051	848	402	1601	332	-2467	-252	3375	-847
232	-253	1060	-417	-413	-68	1570	3920	10768	-1361	5063
233	166	1070	193	170	-111	740	3604	2427	11876	5145
234	10	-616	659	1902	-294	-244	-2431	33	-1045	1334
235	1057	726	302	1061	-1595	1635	514	511	2395	-266
236	940	1624	582	786	468	4660	1003	15749	6529	2107

237	-97	1067	-63	165	1211	1610	-224	1875	3150	19091
238	-217	1432	2153	1075	-833	1807	-3821	1762	1981	121
239	-209	-1024	535	1537	-1524	3995	-3431	1481	-2876	237
240	-30	1606	-551	-1401	4167	1196	2925	-613	2783	3431
241	221	-1002	928	-526	1723	-1525	2327	2027	2499	607
242	2516	-1081	673	-1540	-33	33	1258	-1336	2577	223
243	5703	-2336	1985	-1779	-197	-370	-931	409	-540	540
244	-438	-1551	1680	902	8699	284	91	1765	2392	1349
245	-581	-1066	308	-265	1940	-72	584	344	-564	10666
246	461	1095	3661	587	3356	1699	-647	1080	-2132	-24
247	2050	-1114	2993	786	1373	398	-258	373	-3718	35
248	-741	-1008	744	1369	1576	2146	7186	21217	308	2551
249	-243	-949	942	268	-122	-817	-1439	2225	15141	4756
250	1657	-788	-16	902	1171	-1864	-324	2561	-397	3813
251	819	-706	1349	35	290	2124	637	2343	1058	890
252	956	2053	1641	1195	1850	4251	10855	15001	-3517	3817
253	417	-1287	-964	355	-359	935	2955	1426	-289	14426
254	-1541	786	2131	433	802	932	2248	2316	-4924	-96
255	-202	-1178	2964	-833	995	5359	-69	4858	-1256	4349

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## Subpart 3: Speech Coding - CELP

### 3.1 Scope

#### 3.1.1 General description of the CELP decoder

This subclause provides a brief overview of the CELP (Code Excited Linear Prediction) decoder. A basic block diagram of the CELP decoder is given in Figure 3.1.

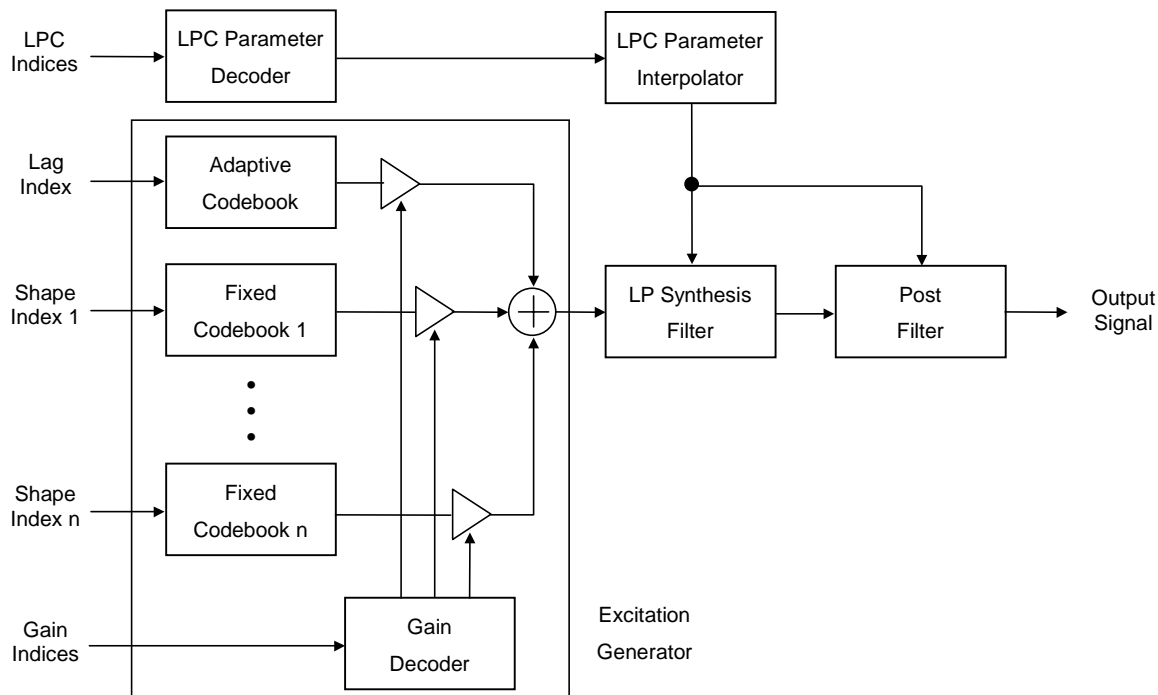


Figure 3.1 - Block diagram of a CELP decoder

The CELP decoder primarily consists of an excitation generator and a synthesis filter. Additionally, CELP decoders often include a post-filter. The excitation generator has an adaptive codebook to model periodic components, fixed codebooks to model random components and a gain decoder to represent a speech signal level. Indices for the codebooks and gains are provided by the encoder. The codebook indices (pitch-lag index for the adaptive codebook and shape index for the fixed codebook) and gain indices (adaptive and fixed codebook gains) are used to generate the excitation signal. It is then filtered by the linear predictive synthesis filter (LP synthesis filter). Filter coefficients are reconstructed using the LPC indices, then are interpolated with the filter coefficients of successive analysis frames. Finally, a post-filter can optionally be applied in order to enhance the speech quality.

#### 3.1.2 Functionality of MPEG-4 CELP

MPEG-4 CELP is a generic coding algorithm with new functionalities. Conventional CELP coders offer compression at a single bitrate and are optimized for specific applications. Compression is one of the functions provided by MPEG-4 CELP, enabling the use of one basic coder for various applications. It provides scalability in bitrate and bandwidth, as well as the ability to generate bitstreams at arbitrary bitrates. The MPEG-4 CELP coder supports two sampling rates, namely, 8 and 16 kHz. The associated bandwidths are 100 – 3400 Hz for 8 kHz sampling rate and 50 – 7000 Hz for 16 kHz sampling rate. Furthermore, silence compression and error resilient bitstream reordering are newly adopted.

### 3.1.2.1 Configuration of the MPEG-4 CELP coder

Two different tools can be used to generate the excitation signal. These are the Multi-Pulse Excitation (MPE) tool or the Regular-Pulse Excitation (RPE) tool. MPE is used for speech sampled at 8 kHz or 16 kHz. RPE is only used for speech sampled at 16 kHz. The two possible coding modes are summarized in Table 3.1.

**Table 3.1 - Coding modes in the MPEG-4 CELP coder**

Coding Mode	Excitation tool	Sampling rate
I	RPE	16 kHz
II	MPE	8, 16 kHz

### 3.1.2.2 Features of the MPEG-4 CELP coder

The MPEG-4 CELP coder offers the following functionality, depending on the coding mode.

**Table 3.2 - Functionality of the MPEG-4 CELP coder**

Coding Mode	Functionality
I	Multiple bitrates, FineRate Control
II	Multiple bitrates, Bitrate Scalability, Bandwidth Scalability, FineRate Control

For both coding modes, silence compression and error resilient bitstream reordering are available.

**Multiple bitrates:** The available bitrates depend on the coding mode and the sampling rate. The following fixed bitrates are supported:

**Table 3.3 - Fixed bitrates for the mode I coder**

Bitrates for the 16 kHz sampling rate (bit/s)
14400, 16000, 18667, 22533

**Table 3.4 - Fixed bitrates for the mode II coder**

Bitrates for the 8 kHz sampling rate (bit/s)	Bitrates for the 16 kHz sampling rate (bit/s)
3850, 4250, 4650, 4900, 5200, 5500, 5700, 6000, 6200, 6300, 6600, 6900, 7100, 7300, 7700, 8300, 8700, 9100, 9500, 9900, 10300, 10500, 10700, 11000, 11400, 11800, 12000, 12200	10900, 11500, 12100, 12700, 13300, 13900, 14300, 14700, 15900, 17100, 17900, 18700, 19500, 20300, 21100, 13600, 14200, 14800, 15400, 16000, 16600, 17000, 17400, 18600, 19800, 20600, 21400, 22200, 23000, 23800

During non-active frames, the silence compression tool is used and the CELP coder operates at bitrates shown in Table 3.5. The bitrate depends on the coding mode, the sampling rate and the frame length.

**Table 3.5 - Bitrates for the silence compression tool**

Coding mode	Sampling rate [kHz]	Band width scalability	Frame length [ms]	Bitrate [bit/s]		
				TX_flag	HR-SID	LR-SID
I (RPE)	16	-	15	133	2533	400
			10	200	3800	600
II (MPE)	8	On, Off	40	50	525	150
			30	67	700	200
			20	100	1050	300
			10	200	2100	600
	16	Off	20	100	1900	300
			10	200	3800	600
		On	40	50	1050	150
			30	67	1400	200
			20	100	2100	300
			10	200	4200	600

**Fine Rate Control:** Enables fine step bitrate control (permitting variable bitrate operation). This is achieved purely by controlling the transmission rate of the LPC parameters using a combinations of the two bitstream elements **interpolation\_flag** and **LPC\_present** flag. Using FineRate Control it is possible to vary the ratio of LPC-frames to total frames between 50% and 100%. This enables the bitrate to be decreased with respect to the anchor bitrate, as defined in the Semantics.

**Bitrate Scalability:** Bitrate scalability is provided by adding enhancement layers. Enhancement layers can be added with a step of 2000 bit/s for signals sampled at 8 kHz or 4000 bit/s for signals sampled at 16 kHz. A maximum of three enhancement layers may be combined with any bitrate chosen from Table 3.4.

**Bandwidth Scalability:** Bandwidth scalability to cover both sampling rates is achieved by incorporating a bandwidth extension tool in the CELP coder. This is an enhancement tool, supported in Mode II, which may be added if scalability from the 8 kHz sampling rate to the 16 kHz sampling rate is required. A complete coder with bandwidth scalability consists of a core CELP coder for the 8 kHz sampling rate and the bandwidth extension tool to provide a single layer of scalability. The core CELP coder for the 8 kHz sampling rate can comprise several layers. It should be noted that an 8 kHz sampling rate coder with this tool is not the same as a 16 kHz sampling rate coder. Both configurations (8 kHz sampling rate coder with bandwidth scalability and 16 kHz sampling rate coder) offer greater intelligibility and naturalness of decoded speech than does the 8 kHz coder alone because they expand the bandwidth to 7 kHz. The additional bitrate required for the bandwidth scalability tool can be selected from 4 discrete steps for each core layer bitrate as shown in Table 3.6.

**Table 3.6 - Bitrates for the bandwidth scalable mode**

Bitrate of the core layer (bit/s)	Additional bitrate (bit/s)
3850 - 4650	+9200, +10400, +11600, +12400
4900 - 5500	+9467, +10667, +11867, +12667
5700 - 10700	+10000, +11200, +12400, +13200
11000 - 12200	+11600, +12800, +14000, +14800

**Silence compression:** The silence compression tool can be used to reduce the bitrate for input signals with little voice activity. For such non-active periods, the decoder substitutes the regular excitation signal with synthetically generated noise. For voice active periods, the regular speech synthesis process is always used. The silence compression tool is available when the ER-CELP object type is used.

**Error resilient bitstream reordering:** Error resilient bitstream reordering allows the effective use of advanced channel coding techniques like unequal error protection (UEP). The basic idea is to rearrange the audio frame content depending on its error sensitivity in one or more instances belonging to different error sensitivity categories

(ESC). This rearrangement works either data element-wise or even bit-wise. An error resilient bitstream frame is build by concatenating these instances. This functionality is available when the ER-CELP object type is used.

### 3.1.2.3 Algorithmic delay of MPEG-4 CELP modes

The algorithmic delay of the CELP coder comes from the frame length and an additional look ahead length. The frame length depends on the coding mode and the bitrate. The look ahead length, which is an informative parameter, also depends on the coding mode. The delays presented below are applicable to the modes where FineRate Control is off. When FineRate Control is on, additional one-frame delay is introduced. Bandwidth scalability in the mode II coder requires an additional look ahead of 5 ms due to down-sampling.

**Table 3.7 - Delay and frame length for the mode I coder of the 16 kHz sampling rate**

Bitrate for Mode I (bit/s)	Delay (ms)	Frame Length (ms)
14400	26.25	15
16000	18.75	10
18667	26.56	15
22533	26.75	15

**Table 3.8 - Delay and frame length for the mode II coder of the 8 kHz sampling rate**

Bitrate for Mode II (bit/s)	Delay (ms)	Frame Length (ms)
3850, 4250, 4650	45	40
4900, 5200, 5500, 6200	35	30
5700, 6000, 6300, 6600, 6900, 7100, 7300, 7700, 8300, 8700, 9100, 9500, 9900, 10300, 10500, 10700	25	20
11000, 11400, 11800, 12000, 12200	15	10

**Table 3.9 - Delay and frame length for the mode II coder of the 16 kHz sampling rate**

Bitrate for Mode II (bit/s)	Delay (ms)	Frame Length (ms)
10900, 11500, 12100, 12700, 13300, 13900, 14300, 14700, 15900, 17100, 17900, 18700, 19500, 20300, 21100	25	20
13600, 14200, 14800, 15400, 16000, 16600, 17000, 17400, 18600, 19800, 20600, 21400, 22200, 23000, 23800	15	10

In case silence compression is used, the algorithmic delay is the same as without silence compression, since the same frame length and the same additional look-ahead length are used.

## 3.2 Definitions

**3.2.1 adaptive codebook:** An approach to encode the long-term periodicity of the signal. The entries of the codebook consists of overlapping segments of past excitations.

**3.2.2 bandwidth scalability:** The possibility to change the bandwidth of the signal during transmission.

**3.2.3 bitrate scalability:** The possibility to transmit a subset of the bitstream and still decode the bitstream with the same decoder.

**3.2.4 CELP:** Code Excited Linear Prediction

**3.2.5 demultiplexing:** Splitting one bitstream into several.

**3.2.6 excitation:** The excitation signal represents the input to the LPC module. The signal consists of contributions that cannot be covered by the LPC model.

**3.2.7 enhancement layer(s):** The part(s) of the bitstream that is possible to drop in a transmission and still decode the bitstream.



- 3.2.8 fine rate control:** The possibility to change the bitrate by, under some circumstances, skipping transmission of the LPC indices.
- 3.2.9 fixed codebook:** The fixed codebook contains excitation vectors for the speech synthesis filter. The contents of the codebook are non-adaptive (i.e. fixed).
- 3.2.10 index:** Number indicating the quantized value(s).
- 3.2.11 LPC:** Linear Predictive Coding.
- 3.2.12 LSP:** Line Spectral Pairs.
- 3.2.13 MPE:** Multi Pulse Excitation.
- 3.2.14 multiplexing:** Combining several bitstreams into one.
- 3.2.15 post-filter:** This filter is applied to the output of the synthesis filter to enhance the perceptual quality of the reconstructed speech.
- 3.2.16 RPE:** Regular Pulse Excitation.
- 3.2.17 unvoiced frame:** Frame containing unvoiced speech which looks like random noise with no periodicity.
- 3.2.18 variable bitrate:** The ability to permit a variable number of bits corresponding to one coded frame.
- 3.2.19 vector quantizer:** Tool that quantizes several values to one index.
- 3.2.20 voiced frame:** A voiced speech segment is known by its relatively high energy content, but more importantly it contains periodicity which is called the pitch of the voiced speech.

### 3.3 Bitstream syntax

#### 3.3.1 CELP object type

##### 3.3.1.1 Header syntax

##### CelpSpecificConfig()

The following CelpSpecificConfig() is required for the CELP object type:

**Table 3.10 - Syntax of CelpSpecificConfig ()**

Syntax	No. of bits	Mnemonic
CelpSpecificConfig (uint(4) samplingFrequencyIndex)		
{		
<b>isBaseLayer</b>	<b>1</b>	<b>uimsbf</b>
if (isBaseLayer)		
{		
CelpHeader (samplingFrequencyIndex)		
}		
else		
{		
<b>isBWSLayer</b>	<b>1</b>	<b>uimsbf</b>
if (isBWSLayer)		
{		
CelpBWSenhHeader ()		
}		
else		
{		
<b>CELP-BRS-id</b>	<b>2</b>	<b>uimsbf</b>
}		
}		
}		

Table 3.11 - Syntax of CelpHeader()

Syntax	No. of bits	Mnemonic
CelpHeader (samplingFrequencyIndex)		
{		
<b>ExcitationMode</b>	<b>1</b>	<b>uimsbf</b>
<b>SampleRateMode</b>	<b>1</b>	<b>uimsbf</b>
<b>FineRateControl</b>	<b>1</b>	<b>uimsbf</b>
if (ExcitationMode == RPE) {		
<b>RPE_Configuration</b>	<b>3</b>	<b>uimsbf</b>
}		
if (ExcitationMode == MPE) {		
<b>MPE_Configuration</b>	<b>5</b>	<b>uimsbf</b>
<b>NumEnhLayers</b>	<b>2</b>	<b>uimsbf</b>
<b>BandwidthScalabilityMode</b>	<b>1</b>	<b>uimsbf</b>
}		
}		

Table 3.12 - Syntax of CelpBWSenhHeader()

Syntax	No. of bits	Mnemonic
CelpBWSenhHeader ()		
{		
<b>BWS_configuration</b>	<b>2</b>	<b>uimsbf</b>
}		

### 3.3.1.2 Frame syntax

#### Transmission of CELP bitstreams

Each layer of an MPEG-4 CELP audio bitstream is transmitted in an Elementary Stream. In slPacketPayload, the following dynamic data for CELP Audio has to be included:

#### CELP Base Layer -- Access Unit payload

```
slPacketPayload
{
  CelpBaseFrame();
}
```

#### CELP Enhancement Layer -- Access Unit payload

To parse and decode the CELP enhancement layer, information decoded from the CELP base layer is required. For the bitrate scalable mode, the following data for the CELP enhancement layer has to be included:

```
slPacketPayload
{
  CelpBRSenhFrame();
}
```

For the bandwidth scalable mode, the following data for the CELP enhancement layer has to be included:

```
slPacketPayload
{
    CelpBWSenhFrame();
}
```

In case bitrate scalability and bandwidth scalability are both used simultaneously, first all bitrate enhancement layers have to be conveyed prior to the bandwidth scalability layer.

**Table 3.13 - Syntax of CelpBaseFrame()**

Syntax	No. of bits	Mnemonic
CelpBaseFrame() { Celp_LPC() if (ExcitationMode==MPE) { MPE_frame() } if ((ExcitationMode==RPE)&&(SampleRateMode==16kHz)) { RPE_frame() } }		

**Table 3.14 - Syntax of CelpBRSenhFrame()**

Syntax	No. of bits	Mnemonic
CelpBRSenhFrame() { for (subframe=0; subframe<nrof_subframes; subframe++) { <b>shape_enh_positions [subframe][enh_layer]</b> <b>shape_enh_signs [subframe][enh_layer]</b> <b>gain_enh_index [subframe][enh_layer]</b> } }	   <b>4, 12</b> <b>2, 4</b> <b>4</b>	   <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

**Table 3.15 - Syntax of CelpBWSenhFrame()**

Syntax	No. of bits	Mnemonic
CelpBWSenhFrame() { BandScalable_LSP() for (subframe=0; subframe<nrof_subframe_bws; subframe++) { <b>shape_bws_delay [subframe]</b> <b>shape_bws_positions [subframe]</b> <b>shape_bws_signs [subframe]</b> <b>gain_bws_index [subframe]</b> } }	   <b>3</b> <b>22, 26, 30, 32</b> <b>6, 8, 10, 12</b> <b>11</b>	   <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

## 3.3.1.2.1 LPC syntax

Table 3.16 - Syntax of Celp\_LPC()

Syntax	No. of bits	Mnemonic
Celp_LPC() { if (FineRateControl == ON){ <b>interpolation_flag</b> <b>LPC_Present</b> if (LPC_Present == YES) { LSP_VQ() } } else { LSP_VQ() } }	1 1	<b>uimsbf</b> <b>uimsbf</b>

Table 3.17 - Syntax of LSP\_VQ()

Syntax	No. of bits	Mnemonic
LSP_VQ() { if (SampleRateMode == 8kHz) { NarrowBand_LSP() } else { WideBand_LSP() } }		

Table 3.18 - Syntax of NarrowBand\_LSP()

Syntax	No. of bits	Mnemonic
NarrowBand_LSP() { <b>lpc_indices [0]</b> <b>lpc_indices [1]</b> <b>lpc_indices [2]</b> <b>lpc_indices [3]</b> <b>lpc_indices [4]</b> }	4 4 7 6 1	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 3.19 - Syntax of BandScalable\_LSP()

Syntax	No. of bits	Mnemonic
BandScalable_LSP() {		
lpc_indices [5]	4	uimsbf
lpc_indices [6]	7	uimsbf
lpc_indices [7]	4	uimsbf
lpc_indices [8]	6	uimsbf
lpc_indices [9]	7	uimsbf
lpc_indices [10]	4	uimsbf
}		

Table 3.20 - Syntax of WideBand\_LSP()

Syntax	No. of bits	Mnemonic
WideBand_LSP() {		
lpc_indices [0]	5	uimsbf
lpc_indices [1]	5	uimsbf
lpc_indices [2]	7	uimsbf
lpc_indices [3]	7	uimsbf
lpc_indices [4]	1	uimsbf
lpc_indices [5]	4	uimsbf
lpc_indices [6]	4	uimsbf
lpc_indices [7]	7	uimsbf
lpc_indices [8]	5	uimsbf
lpc_indices [9]	1	uimsbf
}		

## 3.3.1.2.2 Excitation syntax

Table 3.21 - Syntax of RPE\_frame()

Syntax	No. of bits	Mnemonic
RPE_frame() {		
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
shape_delay [subframe]	8	uimsbf
shape_index [subframe]	11,12	uimsbf
gain_indices [0][subframe]	6	uimsbf
gain_indices [1][subframe]	3,5	uimsbf
}		
}		

Table 3.22 - Syntax of MPE\_frame()

Syntax	No. of bits	Mnemonic
MPE_frame() {		
<b>signal_mode</b>	<b>2</b>	<b>uimsbf</b>
<b>rms_index</b>	<b>6</b>	<b>uimsbf</b>
for (subframe=0; subframe<nrof_subframes; subframe++) {		
<b>shape_delay [subframe]</b>	<b>8, 9</b>	<b>uimsbf</b>
<b>shape_positions [subframe]</b>	<b>14 ... 32</b>	<b>uimsbf</b>
<b>shape_signs [subframe]</b>	<b>3 ... 12</b>	<b>uimsbf</b>
<b>gain_index [subframe]</b>	<b>6, 7</b>	<b>uimsbf</b>
}		
}		

### 3.3.2 ER-CELP object type

#### 3.3.2.1 Header syntax

##### ErrorResilientCelpSpecificConfig()

The following ErrorResilientCelpSpecificConfig () is required for the ER-CELP object type:

Table 3.23 - Syntax of ErrorResilientCelpSpecificConfig ()

Syntax	No. of bits	Mnemonic
ErrorResilientCelpSpecificConfig (uint(4) samplingFrequencyIndex) {		
<b>isBaseLayer</b>	<b>1</b>	<b>uimsbf</b>
if (isBaseLayer)		
{		
ER_SC_CelpHeader (samplingFrequencyIndex)		
}		
else		
{		
<b>isBWSLayer</b>	<b>1</b>	<b>uimsbf</b>
if (isBWSLayer)		
{		
CelpBWSenhHeader ()		
}		
else		
{		
<b>CELP-BRS-id</b>	<b>2</b>	<b>uimsbf</b>
}		
}		
}		
}		

**Table 3.24 - Syntax of ER\_SC\_CelpHeader()**

Syntax	No. of bits	Mnemonic
ER_SC_CelpHeader (samplingFrequencyIndex)		
{		
<b>ExcitationMode;</b>	<b>1</b>	<b>uimsbf</b>
<b>SampleRateMode;</b>	<b>1</b>	<b>uimsbf</b>
<b>FineRateControl;</b>	<b>1</b>	<b>uimsbf</b>
<b>SilenceCompression;</b>	<b>1</b>	<b>uimsbf</b>
if (ExcitationMode == RPE) {		
<b>RPE_Configuration;</b>	<b>3</b>	<b>uimsbf</b>
}		
if (ExcitationMode == MPE) {		
<b>MPE_Configuration;</b>	<b>5</b>	<b>uimsbf</b>
<b>NumEnhLayers;</b>	<b>2</b>	<b>uimsbf</b>
<b>BandwidthScalabilityMode</b>	<b>1</b>	<b>uimsbf</b>
}		
}		

**3.3.2.2 Frame syntax**

In order to describe the bit error sensitivity of bitstream elements, error sensitivity categories (ESC) are introduced. To describe single bits of elements, the following notation is used.

**gain, x-y**

Denotes bit x to bit y of element gain, whereby x is transmitted first. The LSB is bit zero and the MSB of an element that consist of N bit is N-1. The MSB is always the first bit in the bitstream.

The following syntax is a replacement for CelpBaseFrame. The syntax for enhancement layer for bitrate and bandwidth scalability is not affected.

**Transmission of CELP bitstreams**

The payload data for the ER CELP object is transmitted as slPacketPayload payload in the base layer and the optional enhancement layer Elementary Stream.

**Error Resilient CELP Base Layer -- Access Unit payload**

```
slPacketPayload
{
    ER_SC_CelpBaseFrame();
}
```

**Error Resilient CELP Enhancement Layer -- Access Unit payload**

To parse and decode the Error Resilient CELP enhancement layers, information decoded from the Error Resilient CELP base layer is required. For the bitrate scalable mode, the following data for the Error Resilient CELP enhancement layers has to be included:

```

slPacketPayload
{
    ER_SC_CelpBRSenhFrame();
}

```

For the bandwidth scalable mode, the following data for the Error Resilient CELP enhancement layer has to be included:

```

slPacketPayload
{
    ER_SC_CelpBWSenhFrame();
}

```

### 3.3.2.2.1 CELP base layer

**Table 3.25 - Syntax of ER\_SC\_CelpBaseFrame()**

Syntax	No. of bits	Mnemonic
<pre> ER_SC_CelpBaseFrame() {     if (SilenceCompression == OFF) {         ER_CelpBaseFrame();     } else {         SC_VoiceActivity_ESC0();         if (TX_flag == 1) {             ER_CelpBaseFrame();         } else if (TX_flag == 2) {             SID_LSP_VQ_ESC0();             SID_Frame_ESC0();         } else if (TX_flag == 3) {             SID_Frame_ESC0();         }     } } </pre>		

**Table 3.26 - Syntax of SC\_VoiceActivity\_ESC0 ()**

Syntax	No. of bits	Mnemonic
<pre> SC_VoiceActivity_ESC0 () {     <b>TX_flag;</b> } </pre>	<b>2</b>	<b>uimsbf</b>



**Table 3.27 - Syntax of ER\_CelpBaseFrame ()**

Syntax	No. of bits	Mnemonic
<pre> ER_CelpBaseFrame() {   if (ExcitationMode==MPE) {     if (SampleRateMode == 8kHz) {       MPE_NarrowBand_ESC0();       MPE_NarrowBand_ESC1();       MPE_NarrowBand_ESC2();       MPE_NarrowBand_ESC3();       MPE_NarrowBand_ESC4();     }     if (SampleRateMode == 16kHz) {       MPE_WideBand_ESC0();       MPE_WideBand_ESC1();       MPE_WideBand_ESC2();       MPE_WideBand_ESC3();       MPE_WideBand_ESC4();     }   }    if ((ExcitationMode==RPE) &amp;&amp; (SampleRateMode==16kHz)) {     RPE_WideBand_ESC0();     RPE_WideBand_ESC1();     RPE_WideBand_ESC2();     RPE_WideBand_ESC3();     RPE_WideBand_ESC4();   } }         </pre>		

**3.3.2.2.1.1 MPE narrowband syntax**

**Table 3.28 - Syntax of MPE\_NarrowBand\_ESC0()**

Syntax	No. of bits	Mnemonic
<pre> MPE_NarrowBand_ESC0() {   if (FineRateControl == ON) {     <b>interpolation_flag;</b>     <b>LPC_Present;</b>   }   <b>rms_index, 5-4;</b>   for (subframe = 0; subframe &lt; nrof_subframes; subframe++) {     <b>shape_delay[subframe], 7;</b>   } }         </pre>	  <b>1</b> <b>1</b>  <b>2</b>  <b>1</b>	  <b>uimsbf</b> <b>uimsbf</b>  <b>uimsbf</b>  <b>uimsbf</b>

Table 3.29 - Syntax of MPE\_NarrowBand\_ESC1()

Syntax	No. of bits	Mnemonic
MPE_NarrowBand_ESC1() {		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [0], 1-0;	2	uimsbf
lpc_indices [1], 0;	1	uimsbf
}		
} else {		
lpc_indices [0], 1-0;	2	uimsbf
lpc_indices [1], 0;	1	uimsbf
}		
<b>signal_mode;</b>	2	uimsbf
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
<b>shape_delay</b> [subframe], 6-5;	2	uimsbf
}		
}		

Table 3.30 - Syntax of MPE\_NarrowBand\_ESC2()

Syntax	No. of bits	Mnemonic
MPE_NarrowBand_ESC2() {		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [2], 6;	1	uimsbf
lpc_indices [2], 0;	1	uimsbf
lpc_indices [4];	1	uimsbf
}		
} else {		
lpc_indices [2], 6;	1	uimsbf
lpc_indices [2], 0;	1	uimsbf
lpc_indices [4];	1	uimsbf
}		
<b>rms_index, 3</b>	1	uimsbf
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
<b>shape_delay</b> [subframe], 4-3;	2	uimsbf
<b>gain_index</b> [subframe], 1-0;	2	uimsbf
}		
}		

Table 3.31 - Syntax of MPE\_NarrowBand\_ESC3()

Syntax	No. of bits	Mnemonic
MPE_NarrowBand_ESC3() { if (FineRateControl == ON) { if (LPC_Present == YES) { lpc_indices [0], 3-2; lpc_indices [1], 2-1; lpc_indices [2], 5-1; } } else { lpc_indices [0], 3-2; lpc_indices [1], 2-1; lpc_indices [2], 5-1; } for (subframe = 0; subframe < nrof_subframes; subframe++) { shape_delay[subframe], 2-0; shape_signs[subframe]; gain_index[subframe], 2; } }	2 2 5  2 2 5  3 3 ... 12 1	uimsbf uimsbf uimsbf  uimsbf uimsbf uimsbf  uimsbf uimsbf uimsbf

Table 3.32 - Syntax of MPE\_NarrowBand\_ESC4()

Syntax	No. of bits	Mnemonic
MPE_NarrowBand_ESC4() { if (FineRateControl == ON) { if (LPC_Present == YES) { lpc_indices [1], 3; lpc_indices [3]; } } else { lpc_indices [1], 3; lpc_indices [3]; } rms_index, 2-0 for (subframe = 0; subframe < nrof_subframes; subframe++) { shape_positions[subframe];  gain_index[subframe], 5-3; } }	1 6  1 6 3 13 ... 32 3	uimsbf uimsbf  uimsbf uimsbf uimsbf uimsbf uimsbf

## 3.3.2.2.1.2 MPE wideband syntax

Table 3.33 - Syntax of MPE\_WideBand\_ESC0()

Syntax	No. of bits	Mnemonic
MPE_WideBand_ESC0() {		
if (FineRateControl == ON) {		
<b>interpolation_flag;</b>	1	uimsbf
<b>LPC_Present;</b>	1	uimsbf
if (LPC_Present == YES) {		
<b>lpc_indices [0];</b>	5	uimsbf
<b>lpc_indices [1], 1-0;</b>	2	uimsbf
<b>lpc_indices [2], 6;</b>	1	uimsbf
<b>lpc_indices [2], 4-0;</b>	5	uimsbf
<b>lpc_indices [4];</b>	1	uimsbf
<b>lpc_indices [5], 0;</b>	1	uimsbf
}		
} else {		
<b>lpc_indices [0];</b>	5	uimsbf
<b>lpc_indices [1], 1-0;</b>	2	uimsbf
<b>lpc_indices [2], 6;</b>	1	uimsbf
<b>lpc_indices [2], 4-0;</b>	5	uimsbf
<b>lpc_indices [4];</b>	1	uimsbf
<b>lpc_indices [5], 0;</b>	1	uimsbf
}		
<b>rms_index, 4-5;</b>	2	uimsbf
}		

Table 3.34 - Syntax of MPE\_WideBand\_ESC1()

Syntax	No. of bits	Mnemonic
MPE_WideBand_ESC1() {		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
<b>lpc_indices [1], 3-2;</b>	2	uimsbf
<b>lpc_indices [2], 5;</b>	1	uimsbf
<b>lpc_indices [5], 1;</b>	1	uimsbf
<b>lpc_indices [6], 1-0;</b>	2	uimsbf
}		
} else {		
<b>lpc_indices [1], 3-2;</b>	2	uimsbf
<b>lpc_indices [2], 5;</b>	1	uimsbf
<b>lpc_indices [5], 1;</b>	1	uimsbf
<b>lpc_indices [6], 1-0;</b>	2	uimsbf
}		
<b>signal_mode;</b>	2	uimsbf
for (subframe=0; subframe < nrof_subframes; subframe++) {		
<b>shape_delay[subframe], 8-6;</b>	3	uimsbf
}		
}		

Table 3.35 - Syntax of MPE\_WideBand\_ESC2()

Syntax	No. of bits	Mnemonic
MPE_WideBand_ESC2()		
{		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [1], 4;	1	uimsbf
lpc_indices [3], 6;	1	uimsbf
lpc_indices [3], 1;	1	uimsbf
lpc_indices [5], 2;	1	uimsbf
lpc_indices [6], 3;	1	uimsbf
lpc_indices [7], 6;	1	uimsbf
lpc_indices [7], 4;	1	uimsbf
lpc_indices [7], 1-0;	2	uimsbf
lpc_indices [9];	1	uimsbf
}		
} else {		
lpc_indices [1], 4;	1	uimsbf
lpc_indices [3], 6;	1	uimsbf
lpc_indices [3], 1;	1	uimsbf
lpc_indices [5], 2;	1	uimsbf
lpc_indices [6], 3;	1	uimsbf
lpc_indices [7], 6;	1	uimsbf
lpc_indices [7], 4;	1	uimsbf
lpc_indices [7], 1-0;	2	uimsbf
lpc_indices [9];	1	uimsbf
}		
rms_index, 3;	1	uimsbf
for (subframe=0; subframe < nrof_subframes; subframe++) {		
shape_delay[subframe], 5-4;	2	uimsbf
gain_index[subframe], 1-0;	2	uimsbf
}		
}		

Table 3.36 - Syntax of MPE\_WideBand\_ESC3()

Syntax	No. of bits	Mnemonic
MPE_WideBand_ESC3() {		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [3], 4-2;	3	uimsbf
lpc_indices [3], 0;	1	uimsbf
lpc_indices [5], 3;	1	uimsbf
lpc_indices [6], 2;	1	uimsbf
lpc_indices [7], 5;	1	uimsbf
lpc_indices [7], 3-2;	2	uimsbf
lpc_indices [8], 4-1;	4	uimsbf
}		
} else {		
lpc_indices [3], 4-2;	3	uimsbf
lpc_indices [3], 0;	1	uimsbf
lpc_indices [5], 3;	1	uimsbf
lpc_indices [6], 2;	1	uimsbf
lpc_indices [7], 5;	1	uimsbf
lpc_indices [7], 3-2;	2	uimsbf
lpc_indices [8], 4-1;	4	uimsbf
}		
for (subframe=0; subframe < nrof_subframes; subframe++) {		
shape_delay[subframe], 3-2;	2	uimsbf
shape_signs[subframe];	3 ... 12	uimsbf
gain_index[subframe], 2;	1	uimsbf
}		
}		

Table 3.37 - Syntax of MPE\_WideBand\_ESC4()

Syntax	No. of bits	Mnemonic
MPE_WideBand_ESC4() {		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [3], 5;	1	uimsbf
lpc_indices [8], 0;	1	uimsbf
}		
} else {		
lpc_indices [3], 5;	1	uimsbf
lpc_indices [8], 0;	1	uimsbf
}		
rms_index, 2-0;	3	uimsbf
for (subframe=0; subframe < nrof_subframes; subframe++) {		
shape_delay[subframe], 1-0;	2	uimsbf
shape_positions[subframe];	14 ... 32	uimsbf
gain_index[subframe], 6-3;	4	uimsbf
}		
}		

3.3.2.2.1.3 RPE wideband syntax

Table 3.38 - Syntax of RPE\_WideBand\_ESC0()

Syntax	No. of bits	Mnemonic
RPE_WideBand_ESC0() {		
if (FineRateControl == ON){		
<b>interpolation_flag;</b>	1	<b>uimsbf</b>
<b>LPC_Present;</b>	1	<b>uimsbf</b>
if (LPC_Present == YES) {		
<b>lpc_indices [0];</b>	5	<b>uimsbf</b>
<b>lpc_indices [1], 1-0;</b>	2	<b>uimsbf</b>
<b>lpc_indices [2], 6;</b>	1	<b>uimsbf</b>
<b>lpc_indices [2], 4-0;</b>	5	<b>uimsbf</b>
<b>lpc_indices [4];</b>	1	<b>uimsbf</b>
<b>lpc_indices [5], 0;</b>	1	<b>uimsbf</b>
}		
} else {		
<b>lpc_indices [0];</b>	5	<b>uimsbf</b>
<b>lpc_indices [1], 1-0;</b>	2	<b>uimsbf</b>
<b>lpc_indices [2], 6;</b>	1	<b>uimsbf</b>
<b>lpc_indices [2], 4-0;</b>	5	<b>uimsbf</b>
<b>lpc_indices [4];</b>	1	<b>uimsbf</b>
<b>lpc_indices [5], 0;</b>	1	<b>uimsbf</b>
}		
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
<b>gain_indices[0][subframe], 5-3;</b>	3	<b>uimsbf</b>
if (subframe == 0) {		
<b>gain_indices[1][subframe], 4-3;</b>	2	<b>uimsbf</b>
} else{		
<b>gain_indices[1][subframe], 2;</b>	1	<b>uimsbf</b>
}		
}		
}		

Table 3.39 - Syntax of RPE\_WideBand\_ESC1()

Syntax	No. of bits	Mnemonic
RPE_WideBand_ESC1()		
{		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [1], 3-2;	2	uimsbf
lpc_indices [2], 5;	1	uimsbf
lpc_indices [5], 1;	1	uimsbf
lpc_indices [6], 1-0;	2	uimsbf
}		
} else {		
lpc_indices [1], 3-2;	2	uimsbf
lpc_indices [2], 5;	1	uimsbf
lpc_indices [5], 1;	1	uimsbf
lpc_indices [6], 1-0;	2	uimsbf
}		
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
shape_delay[subframe], 7-5;	3	uimsbf
}		
}		



Table 3.40 - Syntax of RPE\_WideBand\_ESC2()

Syntax	No. of bits	Mnemonic
RPE_WideBand_ESC2()		
{		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [1], 4;	1	uimsbf
lpc_indices [3], 6;	1	uimsbf
lpc_indices [3], 1;	1	uimsbf
lpc_indices [5], 2;	1	uimsbf
lpc_indices [6], 3;	1	uimsbf
lpc_indices [7], 6;	1	uimsbf
lpc_indices [7], 4;	1	uimsbf
lpc_indices [7], 1-0;	2	uimsbf
lpc_indices [9];	1	uimsbf
}		
} else {		
lpc_indices [1], 4;	1	uimsbf
lpc_indices [3], 6;	1	uimsbf
lpc_indices [3], 1;	1	uimsbf
lpc_indices [5], 2;	1	uimsbf
lpc_indices [6], 3;	1	uimsbf
lpc_indices [7], 6;	1	uimsbf
lpc_indices [7], 4;	1	uimsbf
lpc_indices [7], 1-0;	2	uimsbf
lpc_indices [9];	1	uimsbf
}		
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
shape_delay[subframe], 4-3;	2	uimsbf
gain_index[0][subframe], 2;	1	uimsbf
if (subframe == 0) {		
gain_indices[1][subframe], 2;	1	uimsbf
} else{		
gain_indices[1][subframe], 1;	1	uimsbf
}		
}		
}		

Table 3.41 - Syntax of RPE\_WideBand\_ESC3()

Syntax	No. of bits	Mnemonic
RPE_WideBand_ESC3()		
{		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [3], 4-2;	3	uimsbf
lpc_indices [3], 0;	1	uimsbf
lpc_indices [5], 3;	1	uimsbf
lpc_indices [6], 2;	1	uimsbf
lpc_indices [7], 5;	1	uimsbf
lpc_indices [7], 3-2;	2	uimsbf
lpc_indices [8], 4-1;	4	uimsbf
}		
} else {		
lpc_indices [3], 4-2;	3	uimsbf
lpc_indices [3], 0;	1	uimsbf
lpc_indices [5], 3;	1	uimsbf
lpc_indices [6], 2;	1	uimsbf
lpc_indices [7], 5;	1	uimsbf
lpc_indices [7], 3-2;	2	uimsbf
lpc_indices [8], 4-1;	4	uimsbf
}		
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
shape_delay[subframe], 2-1;	2	uimsbf
}		
}		

Table 3.42 - Syntax of RPE\_WideBand\_ESC4()

Syntax	No. of bits	Mnemonic
RPE_WideBand_ESC4() {		
if (FineRateControl == ON) {		
if (LPC_Present == YES) {		
lpc_indices [3], 5;	1	uimsbf
lpc_indices [8], 0;	1	uimsbf
}		
} else {		
lpc_indices [3], 5;	1	uimsbf
lpc_indices [8], 0;	1	uimsbf
}		
for (subframe = 0; subframe < nrof_subframes; subframe++) {		
shape_delay[subframe], 0;	1	uimsbf
shape_index[subframe];	11, 12	uimsbf
gain_indices[0][subframe], 1-0;	2	uimsbf
if (subframe == 0) {		
gain_indices[1][subframe], 1-0;	2	uimsbf
} else{		
gain_indices[1][subframe], 0;	1	uimsbf
}		
}		
}		

3.3.2.2.2 CELP enhancement layers

Table 3.43 - Syntax of ER\_SC\_CelpBRSenhFrame()

Syntax	No. of bits	Mnemonic
ER_SC_CelpBRSenhFrame() {		
if (SilenceCompression == OFF) {		
CelpBRSenhFrame();		
} else if (TX_flag == 1) {		
CelpBRSenhFrame();		
}		
}		

Table 3.44 - Syntax of ER\_SC\_CelpBWSenhFrame()

Syntax	No. of bits	Mnemonic
ER_SC_CelpBWSenhFrame() { if (SilenceCompression == OFF) { CelpBWSenhFrame(); } else { if (TX_flag == 1) { CelpBWSenhFrame(); } if (TX_flag == 2) { SID_BandScalable_LSP(); } } }		

## 3.3.2.2.3 Syntax elements for non-active frames

Table 3.45 - Syntax of SID\_LSP\_VQ\_ESC0 ()

Syntax	No. of bits	Mnemonic
SID_LSP_VQ_ESC0() { if (SampleRateMode == 8kHz) { SID_NarrowBand_LSP(); } else { SID_WideBand_LSP(); } }		

Table 3.46 - Syntax of SID\_NarrowBand\_LSP()

Syntax	No. of bits	Mnemonic
SID_NarrowBand_LSP() { <b>SID_lpc_indices [0];</b> <b>SID_lpc_indices [1];</b> <b>SID_lpc_indices [2];</b> }	   <b>4</b> <b>4</b> <b>7</b>	   <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 3.47 - Syntax of SID\_BandScalable\_LSP()

Syntax	No. of bits	Mnemonic
SID_BandScalable_LSP() { <b>SID_lpc_indices [3];</b> <b>SID_lpc_indices [4];</b> <b>SID_lpc_indices [5];</b> <b>SID_lpc_indices [6];</b> }	   <b>4</b> <b>7</b> <b>4</b> <b>6</b>	   <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

**Table 3.48 - Syntax of SID\_WideBand\_LSP()**

Syntax	No. of bits	Mnemonic
SID_WideBand_LSP() {		
<b>SID_lpc_indices [0];</b>	<b>5</b>	<b>uimsbf</b>
<b>SID_lpc_indices [1];</b>	<b>5</b>	<b>uimsbf</b>
<b>SID_lpc_indices [2];</b>	<b>7</b>	<b>uimsbf</b>
<b>SID_lpc_indices [3];</b>	<b>7</b>	<b>uimsbf</b>
<b>SID_lpc_indices [4];</b>	<b>4</b>	<b>uimsbf</b>
<b>SID_lpc_indices [5];</b>	<b>4</b>	<b>uimsbf</b>
}		

**Table 3.49 - Syntax of SID\_Frame\_ESC0 ()**

Syntax	No. of bits	Mnemonic
SID_Frame_ESC0() {		
<b>SID_rms_index;</b>	<b>6</b>	<b>uimsbf</b>
}		

### 3.4 Semantics

This subclause describes the semantics of the syntactic elements. Bitstream elements are shown in **bold**-face and the help variables that appear in the syntax and are needed to extract the bitstream elements are shown in *italics*.

#### 3.4.1 Header semantics

**isBaseLayer** A one-bit identifier representing whether the corresponding layer is the base layer (1) or an bandwidth scalable or bitrate scalable enhancement layer (0).

**isBWSLayer** A one-bit identifier representing whether the corresponding layer is the bandwidth scalable enhancement layer (1) or the bitrate scalable enhancement layer (0).

**CELP-BRS-id** A two-bit identifier representing the order of the bitrate scalable enhancement layers, where the first enhancement layer has the value of '1'. The value of '0' should not be used.

**ExcitationMode** A one-bit identifier representing whether the Multi-Pulse Excitation tool or the Regular-Pulse Excitation tool is used.

**Table 3.50 - Description of ExcitationMode**

ExcitationMode	ExcitationID	Description
0	MPE	MPE tool is used
1	RPE	RPE tool is used

**SampleRateMode** A one-bit identifier representing the sampling rate. Two sampling rates are supported.

**Table 3.51 - Description of SampleRateMode**

SampleRateMode	SampleRateID	Description
0	8kHz	8 kHz Sampling rate
1	16kHz	16 kHz Sampling rate

**FineRateControl** A one-bit flag indicating whether fine rate control in very fine steps is enabled or disabled.

**Table 3.52 - Description of FineRateControl**

FineRateControl	RateControlID	Description
0	OFF	Fine-rate control is disabled
1	ON	Fine-rate control is enabled

FineRate Control enables the bitrate to be decreased with respect to its anchor bitrate. When transmitting the LPC parameters in every frame, the anchor bitrate will be obtained. The lowest bitrate possible bitrate for each configuration can be obtained by transmitting the LPC parameters in 50% of the frames.

**SilenceCompression** A one bit identifier indicating whether Silence Compression is used or not.

SilenceCompression	SilenceCompressionID	Description
0	SC_OFF	SilenceCompression is disabled
1	SC_ON	SilenceCompression is enabled

**RPE\_Configuration** This is a 3-bit identifier which configures the MPEG-4 CELP coder using the Regular-Pulse Excitation tool. This parameter directly determines the set of allowed bitrates (Table 3.53) and the number of subframes in a CELP frame (Table 3.54).

**Table 3.53 - Rate allocation for the 16 kHz mode I coder**

RPE_Configuration	Fixed bitrate FineRate Control OFF (bit/s)	Min. bitrate FineRate Control ON, 50% LPC (bit/s)	Max. bitrate FineRate Control ON, 100% LPC (bit/s)
0	14400	13000	14533
1	16000	13900	16200
2	18667	17267	18800
3	22533	21133	22667
4 ... 7	Reserved		

**MPE\_Configuration** This is a 5-bit field that configures the MPEG-4 CELP coder using the Multi-Pulse Excitation tool. This parameter determines the variables *nrof\_subframes* and *nrof\_subframes\_bws*. This parameter also specifies the number of bits for **shape\_positions[i]**, **shape\_signs[i]**, **shape\_enh\_positions[i][j]** and **shape\_enh\_signs[i][j]**.

*nrof\_subframes* is a help parameter, specifying the number of subframes in a CELP frame, and is used to signal how many times the excitation parameters must be read. For the Regular-Pulse Excitation tool operating at the sampling rate of 16 kHz, this variable is dependent on the **RPE\_Configuration** as follows:

**Table 3.54 - Definition of nrof\_subframes for the 16 kHz mode I coder**

RPE_Configuration	nrof_subframes
0	6
1	4
2	8
3	10
4 ... 7	Reserved

For the Multi-Pulse Excitation tool, it is derived from the **MPE\_Configuration** depending on the sampling rate as follows:

**Table 3.55 - Definition of nrof\_subframes for the 8 kHz mode II coder**

MPE_Configuration	nrof_subframes
0,1,2	4
3,4,5	3
6 ... 12	2
13 ... 21	4
22 ... 26	2
27	4
28 ... 31	reserved

**Table 3.56 - Definition of nrof\_subframes for the 16 kHz mode II coder**

MPE_Configuration	nrof_subframes
0 ... 6	4
8 ... 15	8
16 ... 22	2
24 ... 31	4
7, 23	reserved

**NumEnhLayers** This is a two-bit field specifying the number of enhancement layers that are used.

**Table 3.57 - Definition of nrof\_enh\_layers**

NumEnhLayers	nrof_enh_layers
0	0
1	1
2	2
3	3

**BandwidthScalabilityMode** This is a one-bit identifier that indicates whether bandwidth scalability is enabled. This mode is only valid when **ExcitationMode** = MPE.

**Table 3.58 - Description of BandwidthScalabilityMode**

BandwidthScalabilityMode	ScalableID	Description
0	OFF	Bandwidth scalability is disabled
1	ON	Bandwidth scalability is enabled

**BWS\_Configuration** This is a two-bit field that configures the bandwidth extension tool. This identifier is only valid when **BandwidthScalabilityMode** = ON. This parameter specifies the number of bits for **shape\_bws\_positions[i]**, **shape\_bws\_signs[i]**.

**nrof\_subframes\_bws** This parameter, which is a help variable, represents the number of subframes in the bandwidth extension tool and is derived from the MPE\_Configuration as follows:

**Table 3.59 - Definition of nrof\_subframes\_bws**

MPE_Configuration	nrof_subframes_bws
0,1,2	8
3,4,5	6
6 ... 12	4
13 ... 21	4
22 ... 26	2
27	not valid
28 ... 31	reserved

### 3.4.2 Frame semantics

**interpolation\_flag** This is a one-bit flag. When set, it indicates that the LPC parameters for the current frame must be derived using interpolation.

**Table 3.60 - Description of interpolation\_flag**

interpolation_flag	InterpolationID	Description
0	OFF	LPC coefficients of the frame do not have to be interpolated
1	ON	LPC coefficients of the frame must be retrieved by interpolation

**LPC\_Present** This bit indicates whether LPC parameters are attached to the current frame. These LPC parameters are either of the current frame or the next frame.

**Table 3.61 - Description of LPC\_Present**

LPC_Present	LPCID	Description
0	NO	Frame does not carry LPC data
1	YES	Frame carries LPC data

Together, the **interpolation\_flag** and the **LPC\_Present** flag describe how the LPC parameters are to be derived.



**Table 3.62 - LPC decoding process described by interpolation\_flag and LPC\_Present flag**

interpolation_flag	LPC_Present	Description
1	1	LPC Parameters of the current frame must be extracted using interpolation. The current frame carries LPC Parameters belonging to the next frame.
1	0	RESERVED
0	1	LPC Parameters of the current frame are present in the current frame.
0	0	LPC Parameters of the previous frame must be used in the current frame.

**lpc\_indices[]** These are multi-bit fields representing LPC coefficients. These contain information needed to extract the LSP coefficients. The exact extraction procedure is described in the Decoding Process.

**shape\_delay[subframe]** This bit field represents the adaptive codebook lag. The decoding of this field depends on **ExcitationMode** and **SampleRateMode**.

**Table 3.63 - Number of bits for shape\_delay[]**

ExcitationMode	SampleRateMode	shape_delay[] (bits)
RPE	16 kHz	8
MPE	8 kHz	8
MPE	16 kHz	9

**shape\_index[subframe]** This index contains information needed to extract the fixed codebook contribution from the regular pulse codebook. The number of bits consumed by this field depends on the bitrate (derived from the **RPE\_configuration**).

**Table 3.64 - Number of bits for shape\_index[]**

RPE_Configuration	number of bits representing shape_index[]
0	11
1	11
2	12
3	12
4 ... 7	Reserved

**gain\_indices[0][subframe]** These bit fields specify the adaptive codebook gain in the RPE tool using 6 bits. It is read from the bitstream for every subframe.

**gain\_indices[1][subframe]** These bit fields specify the fixed codebook gain in the RPE tool. It is read from the bitstream for every subframe. The number of bits read to represent this field depends on the subframe number. For the first subframe this is 5 bits, while for the remaining subframes it is 3 bits.

**gain\_index[subframe]** This 6 or 7-bit field represents the gains for the adaptive codebook and the multi-pulse excitation for the 8 kHz or 16 kHz sampling rate, respectively.

**gain\_enh\_index[subframe]** This 4-bit field represents the gain for the enhancement multi-pulse excitation in the CELP coder at 8 kHz.

**gain\_bws\_index[subframe]** This 11-bit field represents the gains for the adaptive codebook and two multi-pulse excitation in the bandwidth extension tool.

**signal\_mode** This 2-bit field represents the type of signal. This information is used in the MPE tool. The gain codebooks are switched depending on this information.

**Table 3.65 - Description of signal\_mode**

signal_mode	Description
0	Unvoiced
1,2,3	Voiced

**rms\_index** This parameter indicates the rms level of the frame. This information is only utilized in the MPE tool.

**shape\_positions[subframe], shape\_signs[subframe]** These bit fields represent the pulse positions and the pulse signs for the multi-pulse excitation. The length of the bit field is dependent on **MPE\_Configurations**.

**Table 3.66 - Definitions of shape\_positions[] and shape\_signs[] for speech sampled at 8 kHz**

MPE_Configuration	shape_positions[] (bits)	shape_signs[] (bits)
0	14	3
1	17	4
2	20	5
3	20	5
4	22	6
5	24	7
6	22	6
7	24	7
8	26	8
9	28	9
10	30	10
11	31	11
12	32	12
13	13	4
14	15	5
15	16	6
16	17	7
17	18	8
18	19	9
19	20	10
20	20	11
21	20	12
22	18	8
23	19	9
24	20	10
25	20	11
26	20	12
27	19	6
28 ... 31	reserved	

**Table 3.67 - Definitions of shape\_positions[] and shape\_signs[] for speech sampled at 16 kHz**

MPE_Configuration	shape_positions[] (bits)	shape_signs[] (bits)
0, 16	20	5
1, 17	22	6
2, 18	24	7
3, 19	26	8
4, 20	28	9
5, 21	30	10
6, 22	31	11
7, 23	reserved	reserved
8, 24	11	3
9, 25	13	4
10, 26	15	5
11, 27	16	6
12, 28	17	7
13, 29	18	8
14, 30	19	9
15, 31	20	10

**shape\_enh\_positions[subframe][], shape\_enh\_signs[subframe][]** These bit fields represent the pulse positions and the pulse signs for the multi-pulse excitation in each enhancement layer. The length of the bit field is dependent on **MPE\_Configuration**.

**Table 3.68 - Definition of shape\_enh\_positions[][] and shape\_enh\_signs[][] for speech sampled at 8 kHz**

MPE_Configuration	shape_enh_positions[][] (bits)	shape_enh_signs[][] (bits)
0 ... 12	12	4
13 ... 26	4	2
27	not valid	
28 ... 31	reserved	

**Table 3.69 - Definition of shape\_enh\_positions[][] and shape\_enh\_signs[][] for speech sampled at 16 kHz**

MPE_Configuration	shape_enh_positions[][] (bits)	shape_enh_signs[][] (bits)
0 ... 6, 16 ... 22	12	4
8 ... 15, 24 ... 31	4	2
7, 23	reserved	

**shape\_bws\_delay[subframe]** This 3-bit field is utilized in decoding the adaptive codebook for the bandwidth extension tool. This value indicates the differential lag from the lag described in **shape\_delay[]**.

**shape\_bws\_positions[subframe], shape\_bws\_signs[subframe]** These fields represent the pulse positions and the pulse signs for the multi-pulse excitation in the bandwidth extension tool. The length of the bit field is dependent on **BWS\_Configuration**.

**Table 3.70 - Definition of shape\_bws\_positions[] and shape\_bws\_signs[]**

BWS_Configuration	shape_bws_positions[] (bits)	shape_bws_signs[] (bits)
0	22	6
1	26	8
2	30	10
3	32	12

**TX\_flag**

Two-bit field indicating the transmission mode.

**Table 3.71 - Definition of TX\_flag**

TX_flag	Transmission mode
0	Non-active frame. No frame energy or LPC indices are transmitted.
1	Active frame.
2	Non-active frame. Frame energy and LPC indices are transmitted.
3	Non-active frame. Only frame energy is transmitted, no LPC indices are transmitted.

**SID\_rms\_index**

6-bit field indicating the energy of the frame.

**SID\_lpc\_indices**

These are multi-bit fields representing LPC coefficients for the non-active LPC frames (TX\_flag = 2). These contain information needed to extract the LSP coefficients. Bitstream semantics for the SID\_lpc\_indices are shown in Table 3.72.

**Table 3.72 - Bitstream semantics for the SID\_lpc\_indices**

Coding mode	Sampling rate [ kHz]	Band width scalability	Parameter	Description		
I	16	Off	SID_lpc_indices[0]	0-4 <sup>th</sup> LSPs of the 1st stage VQ		
			SID_lpc_indices[1]	5-9 <sup>th</sup> LSPs of the 1st stage VQ		
			SID_lpc_indices[2]	10-14 <sup>th</sup> LSPs of the 1st stage VQ		
			SID_lpc_indices[3]	15-19 <sup>th</sup> LSPs of the 1st stage VQ		
			SID_lpc_indices[4]	0-4 <sup>th</sup> LSPs of the 2nd stage VQ		
			SID_lpc_indices[5]	5-9 <sup>th</sup> LSPs of the 2nd stage VQ		
II	8	On, Off	SID_lpc_indices[0]	0-4 <sup>th</sup> LSPs of the 1st stage VQ		
			SID_lpc_indices[1]	5-9 <sup>th</sup> LSPs of the 1st stage VQ		
			SID_lpc_indices[2]	0-4 <sup>th</sup> LSPs of the 2nd stage VQ		
			16	On	SID_lpc_indices[3]	0-9 <sup>th</sup> LSPs of 1st stage VQ
					SID_lpc_indices[4]	10-19 <sup>th</sup> LSPs of 1st stage VQ
					SID_lpc_indices[5]	0-4 <sup>th</sup> LSPs of 2nd stage VQ
	SID_lpc_indices[6]	5-9 <sup>th</sup> LSPs of 2nd stage VQ				
	Off	SID_lpc_indices[0]			0-4 <sup>th</sup> LSPs of 1st stage VQ	
		SID_lpc_indices[1]			5-9 <sup>th</sup> LSPs of 1st stage VQ	
		SID_lpc_indices[2]	10-14 <sup>th</sup> LSPs of 1st stage VQ			
		SID_lpc_indices[3]	15-19 <sup>th</sup> LSPs of 1st stage VQ			
		SID_lpc_indices[4]	0-4 <sup>th</sup> LSPs of 2nd stage VQ			
		SID_lpc_indices[5]	5-9 <sup>th</sup> LSPs of 2nd stage VQ			

### 3.5 MPEG-4 CELP Decoder tools

This subclause provides a brief description of the functionality, parameter definition and the decoding processes of the tools supported by the MPEG-4 CELP core. The description of each tool comprises three parts and an optional part containing tables used by the tool:

1. Tool description: a short description of the functionality of the tool is given together with its interface.
2. Definitions: the input and output parameters as well as help elements of the tool are described here. Each element is either bold or italic. Bold names indicate that the element is read from the bitstream, italic names indicate auxiliary elements. If elements are already used by another tool, a reference to the previous definition is given.
3. Decoding process: The decoding process is explained here in detail with the aid of mathematical equations and pseudo-C code.
4. Tables: this optional fourth part contains tables that are used by the tool.

#### 3.5.1 General Introduction to the MPEG-4 CELP decoder tool-set

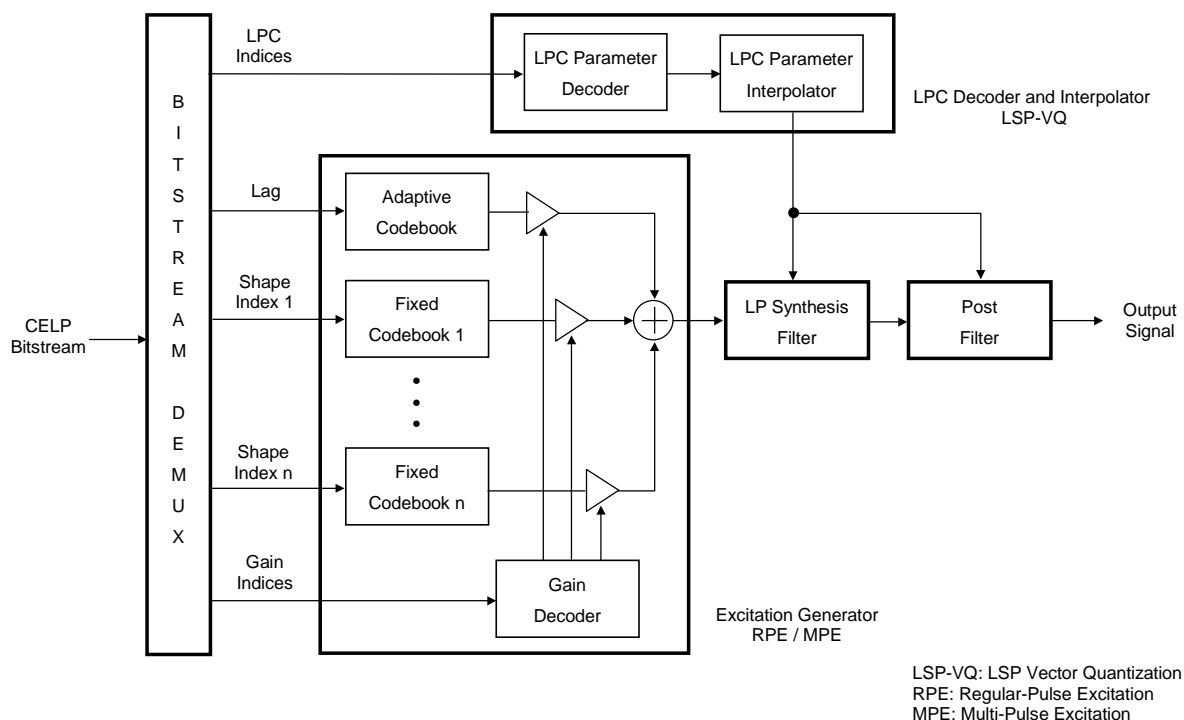


Figure 3.2 - Block diagram illustrating the tools used in the MPEG-4 CELP coder

Figure 3.2 illustrates the MPEG-4 CELP decoder. It operates at a sampling rate of 8 or 16 kHz. One or more individual blocks have been grouped together (outlined in bold), forming the tools available for MPEG-4 CELP decoding. The following tools are supported:

- CELP bitstream demultiplexer
- CELP LPC decoder and interpolator
  - Narrowband LSP-VQ Decoding Tool
  - Wideband LSP-VQ Decoding Tool
  - Bandwidth Scalable LSP-VQ Decoding Tool
- CELP excitation generator
  - Regular-Pulse Excitation Generation Tool
  - Multi-Pulse Excitation Generation Tool
  - Bitrate Scalable Multi-Pulse Excitation Generation Tool
  - Bandwidth Scalable Multi-Pulse Excitation Generation Tool
- CELP LPC synthesis filter
- CELP post-processor (Informative Tool)

The decoding is performed on a frame basis and each frame is divided into subframes. A frame in the bitstream is demultiplexed by the CELP bitstream demultiplexer module. The parameters that are extracted from the bitstream are header information, codes representing LPC coefficients of the frame and the excitation parameters for each subframe. These codes are decoded and interpolated for each subframe by the CELP LPC decoder and interpolator module. For each subframe, the excitation parameters are used to generate the excitation signal using the CELP excitation generator module. The CELP LPC synthesis filter module reconstructs the speech signal on a subframe basis from the interpolated LPC coefficients and the generated excitation signal. Enhancement of the synthesized signal is obtained by the optional CELP post-processor module.

In order to realize Bitrate Scalability, the Bitrate Scalable Multi-Pulse Excitation (MPE) Generation tool is utilized to generate the excitation signal. The Bitrate Scalable MPE Generation tool is realized by adding the enhancement excitation decoding tool to the MPE generation tool in order to enhance the quality of the excitation signal.

The Bandwidth Scalable CELP decoder is realized using both the Bandwidth Scalable LSP-VQ tool and the Bandwidth Scalable MPE Generation tool. These scalable tools are utilized to expand the bandwidth of the decoded signal from 3.4 kHz to 7 kHz.

### 3.5.2 AAC/CELP scalable configuration

When the Narrowband CELP decoder is used as a “core coder” in the AAC/CELP scalable configuration (see the T/F part) for large-step scalability, the post-filter is switched off. In case the Narrowband CELP decoder operates as a core decoder in a scalable configuration, this decoder may decode the CELP audio object at a sampling rate other than 8 kHz. The sampling frequency of the CELP audio object is specified by **samplingFrequencyIndex** in **AudioSpecificInfo** (See subpart 1). The **SampleRateMode** flag in the CELP header controlling the decoder operation, should indicate 8 kHz. In case the **samplingFrequencyIndex** indicates a sampling rate other than 8000 Hz, the bitrate, the frame length and delay are changed accordingly.

### 3.5.3 Helping variables

Although each tool has a description of the variables it uses, provided in this subclause are the most commonly used variables shared by multiple tools.

*frame\_size*: This field indicates the number of samples in a frame. The decoder outputs a frame with *frame\_size* samples.

*nrof\_subframes*: A frame is built up of a number of subframes. The number of subframes is specified in this field.

*sbfrm\_size*: A subframe consists of a number of samples, which is indicated by this field. The number of samples in a frame must always be equal to the sum of the number of samples in the subframes. Accordingly, the following relation must always hold

$$frame\_size = nrof\_subframes * sbfrm\_size$$

These three parameters depend on the coding mode and the bitrate settings as tabulated in Table 3.73 for Mode I, and Table 3.74 and Table 3.75 for Mode II.

**Table 3.73 - CELP coder (Mode I) configuration for 16 kHz sampling rate**

RPE_Configuration	Frame_size (#samples)	nrof_subframes	sbfrm_size (#samples)
0	240	6	40
1	160	4	40
2	240	8	30
3	240	10	24
4 ... 7	Reserved		

**Table 3.74 - CELP coder (Mode II) configuration for 8 kHz sampling rate**

MPE_Configuration	Frame_size (#samples)	nrof_subframes	sbfrm_size (#samples)
0,1,2	320	4	80
3,4,5	240	3	80
6 ... 12	160	2	80
13 ... 21	160	4	40
22 ... 26	80	2	40
27	240	4	60
28 ... 31	Reserved		

**Table 3.75 - CELP coder (Mode II) configuration for 16 kHz sampling rate**

MPE_Configuration	Frame_size (#samples)	nrof_subframes	sbfrm_size (#samples)
0, ..., 6	320	4	80
8, ..., 15	320	8	40
16, ..., 22	160	2	80
24, ..., 31	160	4	40
7, 23	Reserved		

*lpc\_order*. This field indicates the number of coefficients used for Linear Prediction. The value of this field is 20 for a sampling rate of 16 kHz and 10 for 8 kHz.

*num\_lpc\_indices*. This parameter specifies the number of indices containing LPC information that must be read from a bitstream. This is not equal to the LPC-order. The *num\_lpc\_indices* is 5 in the 8 kHz mode and an additional 6 for the bandwidth extension tool. For a sampling rate of 16 kHz in the Vector Quantizer, this value is 10.

### 3.5.4 Bitstream elements for the MPEG-4 CELP decoder tool-set

Descriptions of all the bitstream variables are listed in subclause 3.4.

### 3.5.5 CELP bitstream demultiplexer

#### 3.5.5.1 Tool description

The CELP bitstream demultiplexer tool extracts a CELP frame from the received bitstream.

#### 3.5.5.2 Definitions

All the bitstream elements and the associated help variables have been defined in the subclause 3.4.

#### 3.5.5.3 Decoding process

The decoding of bitstream elements is in accordance with the Syntax described in the subclause 3.3.

### 3.5.6 CELP LPC decoder and interpolator

The CELP LPC decoder and interpolator has two functions:

1. Retrieves LPC coefficients from the **lpc\_indices[]**.
2. Interpolates the retrieved LPC coefficients for every subframe.

Depending on the quantization mode, the **lpc\_indices[]** contain information needed to retrieve LSP Coefficients. In the LSP Decoding process, there are three types of decoding processes, namely the Narrowband LSP Decoding process, the Wideband LSP Decoding process and the Bandwidth Scalable LSP Decoding process. The outputs of this process are the filter coefficients, called LPC coefficients, or a-parameters that can be used in the direct-form filter.

**Table 3.76 - LPC decoding tools**

Coding Mode	Sampling Rate	Tool
Mode I	16 kHz	Wideband LSP-VQ
Mode II	8 kHz	Narrowband LSP-VQ
	16 kHz	Wideband LSP-VQ
	8/16 kHz (BWS)	Bandwidth Scalable (BWS) LSP-VQ

#### 3.5.6.1 Narrowband LSP-VQ decoding tool

##### 3.5.6.1.1 Tool description

The Narrowband LSP Decoding tool has a two-stage VQ structure. The quantized LSPs are reproduced by adding LSPs decoded in the first stage and in the second stage. In the second stage there are two decoding processes, with or without interframe prediction, and the decoding process is selected according to the **lpc\_indices[4]** flag. Then the decoded LSPs are interpolated and converted to LPC coefficients.



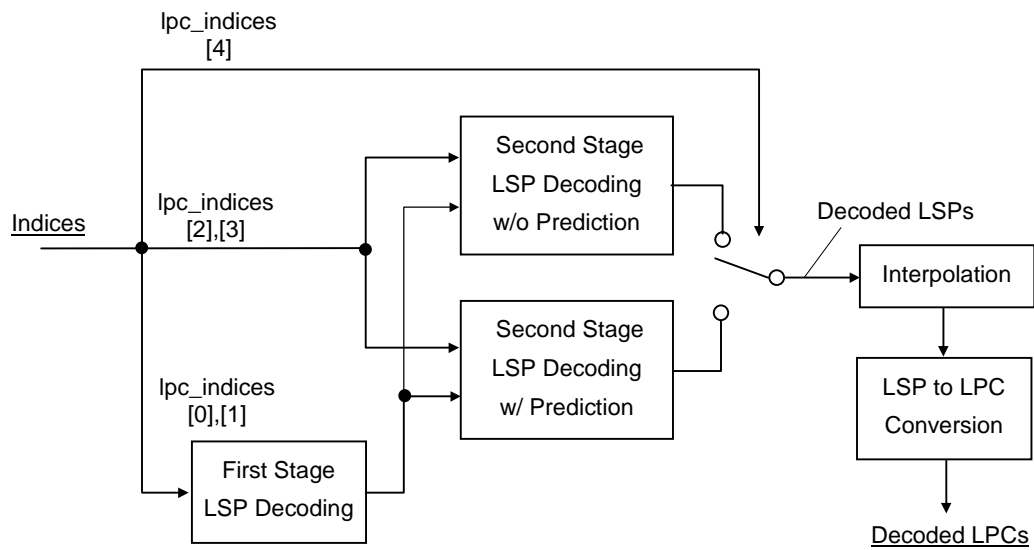


Figure 3.3 - Structure of the narrowband LSP decoder

### 3.5.6.1.2 Definitions

#### Input

**lpc\_indices[]:** The dimension of this array is *num\_lpc\_indices* and contains the packed lpc indices.

#### Output

**int\_Qlpc\_coefficients[]:** This array contains the LPC coefficients for each subframe. The LPC coefficients are decoded and interpolated as described in the decoding process. The LPC coefficients are stacked one after the other in blocks of *lpc\_order*. Thus, the dimension of the array is *lpc\_order \* nrof\_subframes*.

#### Configuration

**lpc\_order:** This field indicates the order of LPC being used.

**num\_lpc\_indices:** This field contains the number of packed LPC codes. For the Narrowband LSP decoding process, *num\_lpc\_indices* is set to 5.

**nrof\_subframes:** This field contains the number of subframes.

The following are help elements used in the Narrowband LSP decoding process:

- lsp\_tbl[][][]* look-up tables for the first stage decoding process
- d\_tbl[][][]* look-up tables for the second stage decoding process of the VQ without interframe prediction
- pd\_tbl[][][]* look-up tables for the second stage decoding process of the VQ with interframe prediction.
- dim[][]* dimensions for the split vector quantization

<i>sign</i>	sign of code vector for the second stage decoding process
<i>idx</i>	unpacked index for the second stage decoding process
<i>lsp_first[]</i>	the LSPs decoded at the first stage decoding process
<i>lsp_previous[]</i>	the LSPs decoded at the previous frame
<i>lsp_predict[]</i>	the LSPs predicted from <i>lsp_previous[]</i> and <i>lsp_first[]</i>
<i>lsp_current[]</i>	the LSPs decoded at the current frame
<i>lsp_subframe[][]</i>	the LSPs interpolated at each subframe
<i>ratio_predict</i>	prediction ratio for predicting <i>lsp_predict[]</i>
<i>ratio_sub</i>	interpolation ratio for calculating <i>lsp_subframe[][]</i>
<i>min_gap</i>	the minimum distance between adjacent LSPs
<i>Convert2lpc()</i>	function for converting LSPs to LPCs

### 3.5.6.1.3 Decoding process

The LSP decoding process for retrieving interpolated LPC coefficients for each subframe is described below.

#### 3.5.6.1.3.1 Converting indices to LSPs

The LSPs of the current frame (*lsp\_current[]*), which are coded by split and two-stage vector quantization, are decoded with a two-stage decoding process. The dimension of each vector is described in Table 3.77 and Table 3.78. The **lpc\_indices[0],[1]** and **lpc\_indices[2],[3]** represent indices for the first and the second stage respectively.

**Table 3.77 - Dimension of the first stage LSP vector**

Split Vector Index: i	Vector Dimension: dim[0][i]
0	5
1	5

**Table 3.78 - Dimension of the second stage LSP vector**

Split Vector Index: i	Vector Dimension: dim[1][i]
0	5
1	5

In the first stage, the LSP vector of the first stage *lsp\_first[]* is decoded by looking up the table *lsp\_tbl[][][]*. (The table *lsp\_tbl[][][]* is shown in Annex 3.C)

```

for(i = 0; i < dim[0][0]; i++)
{
    lsp_first[i] = lsp_tbl[0][lpc_indices[0]][i];
}

```

```

for(i = 0; i < dim[0][1]; i++)
{
    lsp_first[dim[0][0]+i] = lsp_tbl[1][lpc_indices[1]][i];
}

```

In the second stage, there are two types of decoding processes, namely, the decoding process of VQ without interframe prediction and of VQ with interframe prediction. The **lpc\_indices[4]** indicates which process should be selected.

**Table 3.79 - Decoding process for the second stage**

LPC Index: lpc_indices[4]	Decoding process
0	VQ without interframe prediction
1	VQ with interframe prediction

Decoding process of VQ without interframe prediction

In order to obtain LSPs of the current frame *lsp\_current[]*, the decoded vectors in the second stage are added to the decoded first stage LSP vector *lsp\_first[]*. The MSB of the **lpc\_indices[]** represents the sign of the decoded vector, and the remaining bits represent the index for the table *d\_tbl[][][]*. (The table *d\_tbl[][][]* is shown in Annex 3.C)

```

sign = lpc_indices[2] >> 6;
idx = lpc_indices[2] & 0x3f;
if (sign==0)
{
    for (i = 0; i < dim[1][0]; i++)
    {
        lsp_current[i] = lsp_first[i] + d_tbl[0][idx][i];
    }
}
else
{
    for (i = 0; i < dim[1][0]; i++)
    {
        lsp_current[i] = lsp_first[i] - d_tbl[0][idx][i];
    }
}
sign = lpc_indices[3] >> 5;
idx = lpc_indices[3] & 0x1f;
if (sign==0)
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current[dim[1][0]+i] = lsp_first[dim[1][0]+i] + d_tbl[1][idx][i];
    }
}
else
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current[dim[1][0]+i] = lsp_first[dim[1][0]+i] - d_tbl[1][idx][i];
    }
}
}

```

Decoding process of VQ with interframe prediction

In order to obtain LSPs of the current frame *lsp\_current[]*, the decoded vectors of the second stage are added to the LSP vector *lsp\_predict[]*, which are predicted from the decoded LSPs of the previous frame *lsp\_previous[]* and the decoded first stage LSP vector *lsp\_first[]*. In the same way as the decoding process of VQ without interframe prediction, the MSB of the **lpc\_indices[]** represents the sign of the decoded vector, and the remaining bits represent the index for the table *pd\_tbl[][][]*. (The table *pd\_tbl[][][]* is shown in Annex 3.C)

```
for (i = 0; i < lpc_order; i++)
{
    lsp_predict[i]=(1-ratio_predict)*lsp_first[i]
                +ratio_predict*lsp_previous[i]
}
```

where *ratio\_predict* = 0.5

```
sign = lpc_indices[2] >> 6;
idx = lpc_indices[2] & 0x3f;
if (sign==0)
{
    for (i = 0; i < dim[1][0]; i++)
    {
        lsp_current[i] = lsp_predict[i] + pd_tbl[0][idx][i];
    }
}
else
{
    for (i = 0; i < dim[1][0]; i++)
    {
        lsp_current[i] = lsp_predict[i] - pd_tbl[0][idx][i];
    }
}
sign = lpc_indices[3] >> 5;
idx = lpc_indices[3] & 0x1f;
if(sign==0)
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current[dim[1][0]+i] = lsp_predict[dim[1][0]+i] + pd_tbl[1][idx][i];
    }
}
else
{
    for (i = 0; i < dim[1][1]; i++)
    {
        lsp_current[dim[1][0]+i] = lsp_predict[dim[1][0]+i] - pd_tbl[1][idx][i];
    }
}
```

**3.5.6.1.3.2 Stabilization of LSPs**

The decoded LSPs *lsp\_current[]* are stabilized in order to ensure stability of the LPC synthesis filter, which is derived from the decoded LSPs. The decoded LSPs are arranged in ascending order, having a distance of at least *min\_gap* between adjacent coefficients.

```

for(i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] < min_gap)
    {
        lsp_current[i] = min_gap;
    }
}
for (i = 0; i < lpc_order-1; i++)
{
    if (lsp_current[i+1]-lsp_current[i] < min_gap)
    {
        lsp_current[i+1] = lsp_current[i]+min_gap;
    }
}
for (i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] > 1-min_gap)
    {
        lsp_current[i] = 1-min_gap;
    }
}
for (i = lpc_order-1; i > 0; i--)
{
    if (lsp_current[i]-lsp_current[i-1] < min_gap)
    {
        lsp_current[i-1] = lsp_current[i]-min_gap;
    }
}
}

```

where  $\text{min\_gap} = 2.0/256.0$

### 3.5.6.1.3.3 Interpolation of LSPs

The decoded LSPs *lsp\_current[]* are interpolated linearly at each subframe using the LSPs of the previous frame *lsp\_previous[]*.

```

for (n = 0; n < nrof_subframes; n++)
{
    ratio_sub=(n+1)/nrof_subframes
    for(i = 0; i < lpc_order; i++)
    {
        lsp_subframe[n][i] = ((1-ratio_sub)*lsp_previous[i]+ratio_sub*lsp_current[i]);
    }
}

```

### 3.5.6.1.3.4 LSP to LPC conversion

The interpolated LSPs are converted to the LPCs using the auxiliary function *Convert2lpc()*.

```

for (n = 0; n < nrof_subframes; n++)
{
    Convert2lpc (lpc_order, lsp_subframe[n], int_Qlpc_coefficients + n*lpc_order);
}

```

The LSP to LPC conversion is described below. The input LSPs must be normalized in the range of zero to  $\pi$ .

```

tmp_lpc[0] = 1.0;
for (i = 1; i < lpc_order + 1; i++) tmp_lpc[i] = 0.0;
for (i = 0; i < (lpc_order+1)*2; i++) w[i] = 0.0;

for (j = 0; j < lpc_order + 1; j++)
{
    xin1 = tmp_lpc[j];
    xin2 = tmp_lpc[j];
    for (i = 0; i < (lpc_order >> 1); i++)
    {
        n1 = i*4;
        n2 = n1+1;
        n3 = n2+1;
        n4 = n3+1;
        xout1 = -2. * cos(lsp_coefficients[i*2+0]) * w[n1] + w[n2] + xin1;
        xout2 = -2. * cos(lsp_coefficients[i*2+1]) * w[n3] + w[n4] + xin2;
        w[n2] = w[n1];
        w[n1] = xin1;
        w[n4] = w[n3];
        w[n3] = xin2;
        xin1 = xout1;
        xin2 = xout2;
    }

    xout1 = xin1 + w[n4+1];
    xout2 = xin2 - w[n4+2];
    tmp_lpc[j] = 0.5 * (xout1 + xout2);

    w[n4+1] = xin1;
    w[n4+2] = xin2;
}

for (i = 0; i < lpc_order; i++)
{
    lsp_coefficients[i] = tmp_lpc[i+1];
}

```

### 3.5.6.1.3.5 Storing the coefficients

After calculation of the LPC coefficients, the current LSPs must be stored in memory, since they are used for interpolation at the next frame.

```

for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = lsp_current[i];
}

```

It should be noted that the stored LSPs *lsp\_previous[]* must be initialized as described below when the entire decoder is initialized.

```

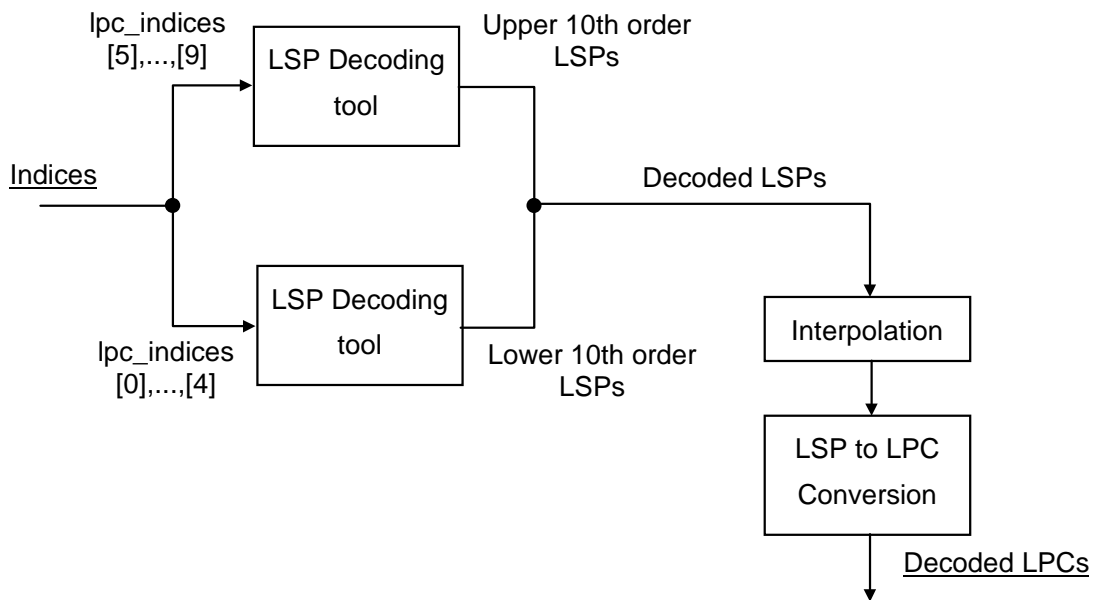
for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = (i+1) / (lpc_order+1);
}

```

**3.5.6.2 Wideband LSP-VQ decoding tool**

**3.5.6.2.1 Tool description**

The Wideband LSP decoding process is based on the Narrowband LSP Decoding process. As described in Figure 3.4, the Wideband LSP Decoding process consists of two LSP decoding blocks connected in parallel. Each of the decoding blocks is identical to the Narrowband LSP Decoding process and decodes the lower and the upper part of the LSPs respectively. The decoded LSPs are combined and output as one set of the parameters.



**Figure 3.4 - Structure of the wideband LSP decoder**

**3.5.6.2.2 Definitions**

Input

**lpc\_indices[]**: The dimension of this array is *num\_lpc\_indices* and contains the packed lpc indices.

Output

**int\_Qlpc\_coefficients[]**: This array contains the LPC coefficients for each subframe. The LPC coefficients are decoded and interpolated as described in the decoding process. The LPC coefficients are stacked one after the other in blocks of *lpc\_order*. Thus, the dimension of the array is *lpc\_order \* nrof\_subframes*.

## Configuration

*lpc\_order*: This field indicates the order of LPC being used.

*num\_lpc\_indices*: This field contains the number of packed LPC codes. For the Wideband LSP Decoding process, *num\_lpc\_indices* is set to 10.

*nrof\_subframes*: This field contains the number of subframes.

The following are help elements used in the Wideband LSP Decoding process:

<i>lsp_tbl[][][]</i>	look-up tables for the first stage decoding process
<i>d_tbl[][][]</i>	look-up tables for the second stage decoding process of the VQ without interframe prediction
<i>pd_tbl[][][]</i>	look-up tables for the second stage decoding process of the VQ with interframe prediction.
<i>dim[][]</i>	dimensions for the split vector quantization
<i>sign</i>	sign of code vector for the second stage decoding process
<i>idx</i>	unpacked index for the second stage decoding process
<i>lsp_first[]</i>	the LSPs decoded at the first stage decoding process
<i>lsp_previous[]</i>	the LSPs decoded at the previous frame
<i>lsp_predict[]</i>	the LSPs predicted from <i>lsp_previous[]</i> and <i>lsp_first[]</i>
<i>lsp_current[]</i>	the LSPs decoded at the current frame
<i>lsp_current_lower[]</i>	the lower part of the current LSPs
<i>lsp_current_upper[]</i>	the upper part of the current LSPs
<i>lsp_subframe[][]</i>	the LSPs interpolated at each subframe
<i>ratio_predict</i>	prediction ratio for predicting <i>lsp_predict[]</i>
<i>ratio_sub</i>	interpolation ratio for calculating <i>lsp_subframe[][]</i>
<i>min_gap</i>	the minimum distance between adjacent LSPs
<i>Convert2lpc()</i>	function for converting LSPs to LPCs

### 3.5.6.2.3 Decoding process

The LSP decoding process for retrieving interpolated LPC coefficients for each subframe is described below.

#### 3.5.6.2.3.1 Converting indices to LSPs

Using the same manner as the Narrowband LSP Decoding process, The LSPs of the current frame (*lsp\_current[]*), which are coded by split and two-stage vector quantization, are decoded with a two-stage decoding process.



Firstly, the lower part of the current LSPs *lsp\_current\_lower[]* are decoded. The dimension of each vector is described in Table 3.80 and Table 3.81. The **lpc\_indices[0],[1]** and **lpc\_indices[2],[3]** represent indices for the first and the second stage respectively.

**Table 3.80 - Dimension of the first stage LSP vector**

Split Vector Index: i	Vector Dimension: dim[0][i]
0	5
1	5

**Table 3.81 - Dimension of the second stage LSP vector**

Split Vector Index: i	Vector Dimension: dim[1][i]
0	5
1	5

In the first stage, the LSP vector of the first stage *lsp\_first[]* is decoded by looking up the table *lsp\_tbl[][][]*. (The table *lsp\_tbl[][][]* is shown in Annex 3.C)

```

for(i = 0; i < dim[0][0]; i++)
{
    lsp_first[i] = lsp_tbl[0][lpc_indices[0]][i];
}

for(i = 0; i < dim[0][1]; i++)
{
    lsp_first[dim[0][0]+i] = lsp_tbl[1][lpc_indices[1]][i];
}
    
```

In the second stage, the **lpc\_indices[4]** indicates which process should be selected.

**Table 3.82 - Decoding process for the second stage**

LPC Index: lpc_indices[4]	Decoding process
0	VQ without interframe prediction
1	VQ with interframe prediction

Decoding process of VQ without interframe prediction

In order to obtain LSPs of the current frame *lsp\_current\_lower[]*, the decoded vectors in the second stage are added to the decoded first stage LSP vector *lsp\_first\_lower[]*. The MSB of the **lpc\_indices[]** represents the sign of the decoded vector, and the remaining bits represent the index for the table *d\_tbl[][][]*. (The table *d\_tbl[][][]* is shown in Annex 3.C)

```

sign = lpc_indices[2] >> 6;
idx = lpc_indices[2] & 0x3f;
if (sign == 0)
{
    for(i = 0; i < dim[1][0]; i++)
    {
    
```

```

        lsp_current_lower[i] = lsp_first[i] + d_tbl[0][idx][i];
    }
}
else
{
    for(i = 0; i < dim[1][0]; i++)
    {
        lsp_current_lower[i] = lsp_first[i] - d_tbl[0][idx][i];
    }
}
sign = lpc_indices[3] >> 6;
idx = lpc_indices[3] & 0x3f;
if (sign == 0)
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current_lower[dim[1][0]+i] = lsp_first[dim[1][0]+i] + d_tbl[1][idx][i];
    }
}
else
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current_lower[dim[1][0]+i] = lsp_first[dim[1][0]+i] - d_tbl[1][idx][i];
    }
}
}

```

#### Decoding process of VQ with interframe prediction

In order to obtain LSPs of the current frame *lsp\_current\_lower[]*, the decoded vectors of the second stage are added to the LSP vector *lsp\_predict[]*, which are predicted from the decoded LSPs of the previous frame *lsp\_previous[]* and the decoded first stage LSP vector *lsp\_first[]*. In the same way as the decoding process of VQ without interframe prediction, the MSB of the **lpc\_indices[]** represents the sign of the decoded vector, and the remaining bits represent the index for the table *pd\_tbl[][][]*. (The table *pd\_tbl[][][]* is shown in Annex 3.C)

```

for(i = 0; i < lpc_order/2; i++)
{
    lsp_predict[i] = (1-ratio_predict)*lsp_first[i] + ratio_predict*lsp_previous[i];
}

```

where *ratio\_predict* = 0.5

```

sign = lpc_indices[2] >> 6;
idx = lpc_indices[2] & 0x3f;
if (sign == 0)
{
    for(i = 0; i < dim[1][0]; i++)
    {
        lsp_current_lower[i] = lsp_predict[i] + pd_tbl[0][idx][i];
    }
}
else
{
    for(i = 0; i < dim[1][0]; i++)
    {

```

```

    lsp_current_lower[i] = lsp_predict[i] - pd_tbl[0][idx][i];
}
}
sign = lpc_indices[3] >> 6;
idx = lpc_indices[3] & 0x3f;
if (sign == 0)
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current_lower[dim[1][0]+i] = lsp_predict[dim[1][0]+i] + pd_tbl[1][idx][i];
    }
}
else
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current_lower[dim[1][0]+i] = lsp_predict[dim[1][0]+i] - pd_tbl[1][idx][i];
    }
}
}

```

Next, the upper part of the current LSPs are decoded in the same manner as the decoding process of the lower part. The dimension of each vector is described in Table 3.83 and Table 3.84. The **lpc\_indices[5],[6]** and **lpc\_indices[7],[8]** represent indices for the first and the second stage respectively.

**Table 3.83 - Dimension of the first stage LSP vector**

Split Vector Index: i	Vector Dimension: dim[0][i]
0	5
1	5

**Table 3.84 - Dimension of the second stage LSP vector**

Split Vector Index: i	Vector Dimension: dim[1][i]
0	5
1	5

```

for(i = 0; i < dim[0][0]; i++)
{
    lsp_first[i] = lsp_tbl[0][lpc_indices[5]][i];
}

for(i = 0; i < dim[0][1]; i++)
{
    lsp_first[dim[0][0]+i] = lsp_tbl[1][lpc_indices[6]][i];
}

```

In the second stage, the **lpc\_indices[9]** indicates which process should be selected.

**Table 3.85 - Decoding process for the second stage**

LPC Index: lpc_indices[9]	Decoding process
0	VQ without interframe prediction
1	VQ with interframe prediction

Decoding process of VQ without interframe prediction

```

sign = lpc_indices[7] >> 6;
idx = lpc_indices[7] & 0x3f;
if (sign==0)
{
  for(i = 0; i < dim[1][0]; i++)
  {
    lsp_current_upper[i] = lsp_first[i] + d_tbl[0][idx][i];
  }
}
else
{
  for(i = 0; i < dim[1][0]; i++)
  {
    lsp_current_upper[i] = lsp_first[i] - d_tbl[0][idx][i];
  }
}
sign = lpc_indices[8] >> 4;
idx = lpc_indices[8] & 0x0f;
if (sign==0)
{
  for (i = 0; i < dim[1][1]; i++)
  {
    lsp_current_upper[dim[1][0]+i] = lsp_first[dim[1][0]+i] + d_tbl[1][idx][i];
  }
}
else
{
  for(i = 0; i < dim[1][1]; i++)
  {
    lsp_current_upper[dim[1][0]+i] = lsp_first[dim[1][0]+i] - d_tbl[1][idx][i];
  }
}

```

Decoding process of VQ with interframe prediction

```

for (i = 0; i < lpc_order/2; i++)
{
  lsp_predict[i]=(1-ratio_predict)*lsp_first[i] +ratio_predict*lsp_previous[lpc_order/2+i];
}

```

where *ratio\_predict* = 0.5

```

sign = lpc_indices[7] >> 6;
idx = lpc_indices[7] & 0x3f;
if (sign==0)
{

```

```

    for (i = 0; i < dim[1][0]; i++)
    {
        lsp_current_upper[i] = lsp_predict[i] + pd_tbl[0][idx][i];
    }
}
else
{
    for(i = 0; i < dim[1][0]; i++)
    {
        lsp_current_upper[i] = lsp_predict[i] - pd_tbl[0][idx][i];
    }
}
sign = lpc_indices[8] >> 4;
idx = lpc_indices[8] & 0x0f;
if(sign==0)
{
    for (i = 0; i < dim[1][1]; i++)
    {
        lsp_current_upper[dim[1][0]+i] = lsp_predict[dim[1][0]+i] + pd_tbl[1][idx][i];
    }
}
else
{
    for(i = 0; i < dim[1][1]; i++)
    {
        lsp_current_upper[dim[1][0]+i] = lsp_predict[dim[1][0]+i] - pd_tbl[1][idx][i];
    }
}

```

Finally, the decoded LSPs *lsp\_current\_lower[]* and *lsp\_current\_upper[]* are combined and stored to the array *lsp\_current[]*.

```

for (i = 0; i < lpc_order/2; i++)
{
    lsp_current[i] = lsp_current_lower[i];
}
for (i = 0; i < lpc_order/2; i++)
{
    lsp_current[lpc_order/2 + i] = lsp_current_upper[i];
}

```

#### 3.5.6.2.4 Stabilization of LSPs

The decoded LSPs *lsp\_current[]* are stabilized in order to ensure stability of the LPC synthesis filter, which is derived from the decoded LSPs. The decoded LSPs are arranged in ascending order, having a distance of at least *min\_gap* between adjacent coefficients.

```

for(i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] < min_gap)
    {
        lsp_current[i] = min_gap;
    }
}

```

```

for(i = 0; i < lpc_order - 1; i++)
{
    if (lsp_current[i+1]-lsp_current[i] < min_gap)
    {
        lsp_current[i+1] = lsp_current[i]+min_gap;
    }
}

for (i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] > 1-min_gap)
    {
        lsp_current[i] = 1-min_gap;
    }
}

for (i = lpc_order - 1; i > 0; i--)
{
    if (lsp_current[i]-lsp_current[i-1] < min_gap)
    {
        lsp_current[i-1] = lsp_current[i]-min_gap;
    }
}

```

where  $lpc\_order = 20$  and  $min\_gap = 1.0/256.0$

### 3.5.6.2.5 Interpolation of LSPs

The decoded LSPs *lsp\_current[]* are interpolated linearly at each subframe using the LSPs of the previous frame *lsp\_previous[]*.

```

for (n = 0; n < nrof_subframes; n++)
{
    ratio_sub=(n+1)/nrof_subframes
    for(i = 0; i < lpc_order; i++)
    {
        lsp_subframe[n][i] = ((1-ratio_sub)*lsp_previous[i]+ratio_sub*lsp_current[i]);
    }
}

```

### 3.5.6.2.6 LSP to LPC conversion

The interpolated LSPs are converted to the LPCs using the auxiliary function *Convert2lpc()*.

```

for (n = 0; n < nrof_subframes; n++)
{
    Convert2lpc (lpc_order, lsp_subframe[n], int_Qlpc_coefficients + n*lpc_order);
}

```

### 3.5.6.2.7 Storing the coefficients

After calculation of the LPC coefficients, the current LSPs must be stored in memory, since they are used for interpolation at the next frame.

```

for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = lsp_current[i];
}

```

It should be noted that the stored LSPs *lsp\_previous[]* must be initialized as described below when the entire decoder is initialized.

```

for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = (i+1) / (lpc_order+1);
}

```

### 3.5.6.3 Bandwidth scalable LSP-VQ decoding tool

#### 3.5.6.3.1 Tool description

The Bandwidth Scalable LSP-VQ decoding tool is utilized to add bandwidth scalability to the Mode II coder. In the Bandwidth Scalable LSP-VQ decoding tool, the bandwidth extension tool is connected with the Narrowband LSP-VQ decoding tool as shown in Figure 3.5. The LPC coefficients for each subframes are reconstructed by retrieving *lpc\_indices[]* and converting the decoded LSPs in the narrowband mode.

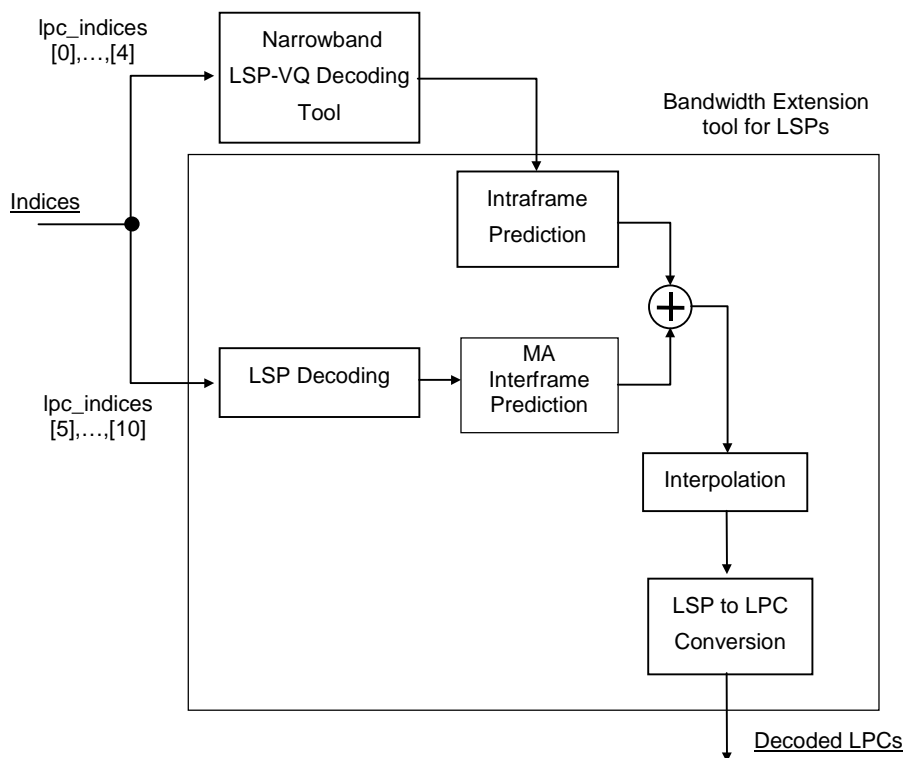


Figure 3.5 - Blockdiagram of the bandwidth scalable LSP-VQ decoding tool

### 3.5.6.3.2 Definitions

#### Input

**lpc\_indices[]**: The dimension of this array is *num\_lpc\_indices* and contains the packed lpc indices.

*lsp\_current[]*: This array contains the decoded LSP parameters, which are normalized in the range of zero to  $\pi$ , at the Narrowband LSP Decoding tool. These parameters are obtained as intermediate parameters in the Narrowband LSP Decoding process described in subclause 3.5.6.1 and forwarded to the Bandwidth Scalable LSP Decoding tool.

#### Output

*int\_Qlpc\_coefficients[]*: This array contains the LPC coefficients for each subframe. The LPC coefficients are decoded and interpolated as described in the decoding process. The LPC coefficients are stacked one after the other in blocks of *lpc\_order*. Thus, the dimension of the array is *lpc\_order \* nrof\_subframes*.

#### Configuration

*lpc\_order*: This field indicates the order of LPC being used.

*num\_lpc\_indices*: This field contains the number of packed LPC codes. For the Bandwidth Scalable LSP Decoding process, *num\_lpc\_indices* is set to 11.

*nrof\_subframes*: This field contains the number of subframes.

*nrof\_subframes\_bws*: This parameter, which is a help variable, represents the number of subframes in the bandwidth extension tool.

The following help elements are used in the decoding procedure.

<i>lsp_bws_tb[][][]</i>	look-up tables for bandwidth scalable LSP decoding process
<i>lsp_bws_buf[][]</i>	buffer containing LSP prediction residual
<i>bws_ma_prdct[][]</i>	prediction coefficients for moving average prediction
<i>bws_nw_prdct[]</i>	prediction coefficients for conversion for LSPs from narrowband to wideband
<i>lsp_current[]</i>	the decoded LSPs in the narrowband CELP coder
<i>lsp_bws_current[]</i>	the decoded LSPs at the current frame in the bandwidth extension tool
<i>lsp_bws_previous[]</i>	the decoded LSPs at the previous frame in the bandwidth extension tool
<i>lpc_order_bws</i>	the order of LPC in the bandwidth extension tool (=20). This is twice the <i>lpc_order</i> in narrowband CELP.



### 3.5.6.3.3 Decoding process

The LPC coefficients at each subframe are reproduced using **lpc\_indices[5]**, ... , **lpc\_indices[10]** and the current LSPs (*lsp\_current[]*) for the narrowband CELP coder. The decoding procedure consists of three steps, decoding LSPs at last subframe, interpolation for the other subframes and conversion of LSPs to LPC coefficients.

#### Decoding process of the bandwidth extension tool for LSP parameters

The intraframe prediction module produces estimated LSP parameters by converting the decoded LSP parameters (*lsp\_current[]*) at the Narrowband LSP-VQ decoding tool. The estimated residual LSP parameters are decoded from **lpc\_indices[]** by means of the moving average (MA) interframe prediction.

The LSPs at the last subframe (*lsp\_bws\_current[]*) for the bandwidth extension tool are reconstructed by table look up and prediction as follows:

```

for (i = 0; i < 5; i++)
{
    lsp_bws_buf[0][i] = lsp_bws_tbl[0][lpc_indices[5]][i]+lsp_bws_tbl[2][lpc_indices[7]][i]
}

for (i = 5; i < 10; i++)
{
    lsp_bws_buf[0][i] = lsp_bws_tbl[0][lpc_indices[5]][i]+lsp_bws_tbl[3][lpc_indices[8]][i-5]
}

for (i = 10; i < 15; i++)
{
    lsp_bws_buf[0][i] = lsp_bws_tbl[1][lpc_indices[6]][i-10]+lsp_bws_tbl[4][lpc_indices[9]][i-10]
}

for (i = 15; i < 20; i++)
{
    lsp_bws_buf[0][i] = lsp_bws_tbl[1][lpc_indices[6]][i-10]+lsp_bws_tbl[5][lpc_indices[10]][i-15]
}

for (n = 0; n <= 2; n++)
{
    for (i = 0; i < 20; i++)
    {
        lsp_bws_current[i] += bws_ma_prdct[n][i]*lsp_bws_buf[n][i]
    }
}

for (i = 0; i < 10; i++)
{
    lsp_bws_current[i] += bws_nw_prdct[i]*lsp_current[i]
}

```

where *lsp\_bws\_tbl[][][]* are LSP codebooks listed in Annex 3.C. *bws\_ma\_prdct[][]* and *bws\_nw\_prdct[]* are prediction coefficients for moving average interframe prediction and intraframe prediction, respectively. *lsp\_bws\_buf[][]* is a buffer containing LSP prediction residual at the current frame and previous two ones. This buffer is shifted for the next frame operation as follows:

```

for (n = 2; n > 0; n--)
{
    for (i = 0; i < 20; i++)
    {

```

```

        lsp_bws_buf[n][i] = lsp_bws_buf[n-1][i];
    }
}

```

The decoded LSPs *lsp\_bws\_current[]* are stabilized in order to ensure stability of the LPC synthesis filter, which is derived from the decoded LSPs. The decoded LSPs are arranged in ascending order, having a distance of at least *min\_gap* between adjacent coefficients.

```

for (i = 0; i < lpc_order_bws; i++)
{
    if (lsp_bws_current[i] < 0.0 || lsp_bws_current[i] > PAI)
    {
        lsp_bws_current[i] = 0.05 + PAI * i / lpc_order_bws;
    }
}

for (i = (lpc_order_bws-1); i > 0; i--)
{
    for (j = 0; j < i; j++)
    {
        if (lsp_bws_current[j] + min_gap > lsp_bws_current[j+1])
        {
            tmp = 0.5 * (lsp_bws_current[j] + lsp_bws_current[j+1]);
            lsp_bws_current[j] = tmp - 0.51 * min_gap;
            lsp_bws_current[j+1] = tmp + 0.51 * min_gap;
        }
    }
}

```

where PAI = 3.141592 and min\_gap = 0.028.

### Interpolation of the LSP parameters

The decoded LSPs are interpolated linearly at each subframe.

```

for (n = 0; n < nrof_subframes_bws; n++)
{
    ratio_sub = (n+1)/nrof_subframes_bws;
    for(i = 0; i < 2*lpc_order; i++)
    {
        lsp_bws_subframe[n][i] = ((1-ratio_sub)*lsp_bws_previous[i]
            + ratio_sub*lsp_bws_current[i])
    }
}

for (i = 0; i < 2*lpc_order; i++)
{
    lsp_bws_previous[i] = lsp_bws_subframe[nrof_subframes_bws-1][i]
}

```

### Conversion of LSPs to LPC coefficients

The interpolated LSPs are converted to the LPC coefficients at each subframes.

```

for (n = 0; n < nrof_subframes_bws; n++)
{
    Convert2lpc (lpc_order_bws, lsp_bws_subframe[n],
                &int_Qlpc_coefficients[n*lpc_order_bws]);
}

```

### 3.5.6.4 Fine rate control in the LSP decoding tool

#### 3.5.6.4.1 Tool description

Fine Rate Control is available with the LSP Decoding tool. If the FRC is used, two additional parameters, **interpolation\_flag** and **LPC\_present**, are included in the input parameters. The Narrowband LSP Decoding process is carried out if the **lpc\_indices[]** are present in the current frame (**LPC\_present** = YES).

#### 3.5.6.4.2 Definitions

##### Input

**lpc\_indices[]**: The dimension of this array is *num\_lpc\_indices* and contains the LPC indices.

**interpolation\_flag**: This is a one-bit flag. When set, the flag indicates that the frame under consideration is an incomplete frame, i.e. the frame does not carry the LPC coefficients of the current speech frame, but only its excitation parameters (adaptive and fixed codebook parameters). The LPC coefficients for the speech frame under consideration should be obtained using interpolation of the LPC coefficients of the adjacent frames.

1 LPC coefficients of the frame must be retrieved by interpolation

0 LPC coefficients of the frame do not have to be interpolated.

For maintaining good subjective quality, there may never be more than one frame in succession without the LPC information, i.e. the **interpolation\_flag** may not have the value 1 in two successive CelpFrames.

**LPC\_Present** This field indicates the presence of LPC parameters in the speech frame under consideration. These LPC coefficients are either of the current speech frame or those of a subsequent frame. When used in combination with the **interpolation\_flag**, the two parameters completely describe how the LPC coefficients of the current frame are derived. If the interpolation flag is set, the LPC coefficients of the current frame are calculated by using the LPC coefficients of the previous and next frame. Generally, this would mean that the decoding of the current frame must be delayed by one frame. To avoid this additional delay in the decoder, the LPC coefficients of the next frame are enclosed in the current frame. In this case, the **LPC\_Present** flag is set. Since the LPC coefficients of the next frame are already present in the current frame, the next frame will not contain LPC information.

- If the **interpolation\_flag** is "1" and the **LPC\_Present** is "1", then (a) the LPC parameters of the current frame are derived using the LPC parameters of the previous frame and that of the next frame and (b) the current frame (frame under consideration) carries LPC parameters of a subsequent frame, but not those of the frame under consideration. The LPC coefficients of the frame under consideration are obtained by interpolation of the previously received LPC parameters and the LPC parameters received in the frame under consideration.
- If the **interpolation\_flag** is "0" and the **LPC\_Present** is "0", the LPC parameters to be used with the frame under consideration are those received in the previous frame.
- When **interpolation\_flag** is "0" and the **LPC\_Present** is "1", then the current frame is a complete frame and the LPC parameters received in the current frame belong to the current frame.

Such a construction is chosen so as to minimize the delay when the decoder begins reconstructing the frame, the LPC coefficients of which are obtained using interpolation without having to wait for the next frame to arrive. Secondly, such a combination makes it possible to decode the bitstream from any point (random access). In the fixed bitrate configuration, these two flags exhibit a fixed pattern

01, 01, 01, 01, 01, 01, ... The string 01 is repeated

11, 00, 11, 00, 11, 00, ... The string 11, 00 is repeated (Fixed bitrate achieved over two frames)

For variable bitrate (when **FineRateControl** = ON), the string will, generally, not exhibit a fixed pattern.

#### Output

*int\_Qlpc\_coefficients[]*: This array contains the LPC coefficients for each subframe. The LPC coefficients are decoded and interpolated as described in the decoding process.

#### 3.5.6.4.3 Decoding process

If the **interpolation\_flag** is set, the LSPs decoded in the current frame belong to the next frame and LSPs for the current frame are calculated by linearly interpolating the LSPs of the adjacent frames.

If the **interpolation\_flag** is not set and the **lpc\_indices[]** are present in the current frame, the computed coefficients belong to the current frame and no interpolation need be performed.

If the **interpolation\_flag** is not set and the **lpc\_indices[]** are not present in the current frame, the LSPs for the current frame are already received in the previous frame. Therefore the LSPs received in the previous frame are copied and used in the current frame.

#### 3.5.7 CELP excitation generator

The CELP excitation generator generates the excitation signal for one subframe from the received indices using either regular pulse excitation (RPE) process or a multi-pulse excitation (MPE) process depending on the coding mode.

**Table 3.86 - Excitation generation tools**

Coding Mode	Sampling Rate	Tool
Mode I	16 kHz	RPE
Mode II	8, 16 kHz 8, 16 kHz 8/16 kHz (BWS)	MPE Bitrate Scalable MPE Bandwidth Scalable MPE

#### 3.5.7.1 Regular pulse excitation generation tool

##### 3.5.7.1.1 Tool description

Figure 3.6 illustrates the excitation generation process used for the 16 kHz sampling rate. The excitation signal is constructed from a periodic component (adaptive codebook contribution) and non periodic component (RPE contribution) scaled by their respective gains. Using the **shape\_delay[sub\_frame]** and the **gain\_indices[0][sub\_frame]**, the adaptive codebook contribution is computed. The RPE contribution is computed by using **shape\_index[sub\_frame]** and **gain\_indices[1][sub\_frame]**. For clarity, the indexing based on *sub\_frame* is dropped. It is noted that the excitation generation process is repeated every subframe.

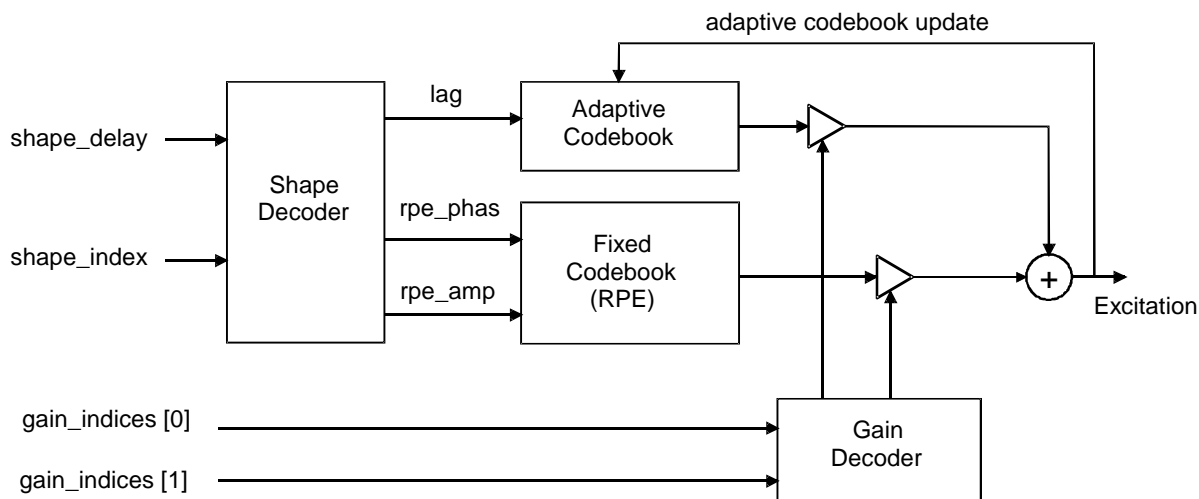


Figure 3.6 - Regular pulse excitation generator

### 3.5.7.1.2 Definitions

#### Input

**shape\_delay[]**: This array is of dimension *nrof\_subframes* and contains the adaptive codebook lag.

**shape\_index[]**: This array is of dimension *nrof\_subframes* and contains the RPE codebook index.

**gain\_indices[0][]**: This array is of dimension *nrof\_subframes* and contains the adaptive codebook gain index.

**gain\_indices[1][]**: This array is of dimension *nrof\_subframes* and contains the RPE codebook gain.

#### Output

**excitation[]**: This array is of dimension *sbfrm\_size* and contains the excitation signal. This signal is reconstructed from shape and gain vectors using the adaptive and fixed codebooks.

**lag**: This field contains the decoded lag (pitch period) for adaptive codevector.

**adaptive\_gain**: This field contains the decoded gain for adaptive codevector.

#### Configuration

**sbfrm\_size**: This field indicates the number of samples in the subframe.

**nrof\_subframes**: This field indicates the number of subframes.

The additional elements used in the RPE excitation mode are as follows:

<i>tbl_cba_gain[]</i>	look-up-table for adaptive codebook gain
<i>tbl_cbf_gain[]</i>	look-up-table for fixed codebook gain
<i>tbl_cbf_gain_dif []</i>	look-up-table for fixed codebook gain difference
<i>cba[]</i>	adaptive codebook

*prev\_Gf* fixed codebook gain of the previous subframe

### 3.5.7.1.3 Decoding process

#### 3.5.7.1.3.1 Shape decoder

This block describes the extraction of Adaptive codebook lag and the RPE parameters. The adaptive codebook lag is derived from the **shape\_delay** in the following way:

```
lag = Lmax - Lmin - shape_delay
```

where Lmax and Lmin are the maximum and minimum lag, respectively, equal to 295 and 40. The decimation factor D and number of pulses  $N_p$  are tabulated below. The RPE parameters, namely the RPE phase (*rpe\_phase*) and the RPE amplitudes (*rpe\_amps*) are derived as follows:

```
rpe_index = shape_index;
rpe_phase = rpe_index MOD D
rpe_index = rpe_index / D;
for (n = Np - 1; n >= 0; n--)
{
    rpe_amps[n] = (rpe_index MOD 3) - 1;
    rpe_index = rpe_index / 3;
}
```

#### 3.5.7.1.3.2 Gain decoder

This block derives the scalar quantized adaptive and fixed codebook gains from the gain indices. The adaptive codebook gain *G<sub>a</sub>* is determined by a table look-up on *cba\_gain[]* (Table 3.88):

```
if (gain_indices[0] > 31)
{
    Ga = -1 * cba_gain[((64 - gain_indices[0]) - 1)];
}
else
{
    Ga = cba_gain[gain_indices[0]];
}
```

The decoding of the fixed codebook gain depends on the subframe under consideration. The gain vector is used for retrieving the fixed codebook gain *G<sub>f</sub>*. For the first subframe in a frame the gain is decoded with:

```
Gf = cbf_gain[gain_indices[1]];
```

where *cbf\_gain[]* is provided in Table 3.89. For all later subframes the gain is decoded using *cbf\_gain\_dif[]* provided in Table 3.90:

```
Gf = cbf_gain_dif[gain_indices[1]] * prev_Gf;
```

where *prev\_Gf* is the decoded gain of the previous subframes.

**3.5.7.1.3.3 Adaptive codebook generation**

First the generation of the excitation signal due to the adaptive codebook will be described. The excitation for the adaptive codebook  $ya[n]$  is computed using the shape vector:

```
for (n = 0; n < sbfrm_size; n++)
{
    ya[n] = cba[lag + n];
}
```

**3.5.7.1.3.4 Fixed codebook generation**

The excitation for the fixed codebook is computed in two steps. To generate the fixed codebook, we also need two extra parameters, namely  $D$  (the pulse spacing or the decimation factor) and  $Np$ , the number of pulses.

These values are dependent on the bitrate and are tabulated in Table 3.87.

**Table 3.87 - Allocation of pulse spacing and number of pulses in RPE**

RPE_Configuration	D	Np
0	8	5
1	8	5
2	5	6
3	4	6
4 ... 7	Reserved	

Once the phase and amplitudes are known, the amplitudes are placed on a regular grid. The amplitudes can have 3 different values: -1, 0 and 1. The excitation for the fixed codebook,  $yf$ , is computed using the RPE-formula:

```
for (n = 0; n < sbfrm_size; n++)
{
    yf[n] = 0;
}
for (n = 0; n < Np; n++)
{
    yf[phase + D*n] = amp[n];
}
```

**3.5.7.1.3.5 Excitation generation**

The excitation is the sum of adaptive and fixed excitation multiplied by their corresponding gains:

```
for (n = 0; n < sbfrm_size; n++)
{
    excitation[n] = Ga * ya[n] + Gf * yf[n];
}
```

Finally, the adaptive codebook has to be updated using the excitation. This is done by shifting the adaptive codebook by  $sbfrm\_size$  entries and filling the empty entries with the calculated excitation.

```

for (n = sbfrm_size; n < Lmax; n++)
{
    cba[n-sbfrm_size] = cba[n];
}
for (n = 0; n < sbfrm_size; n++)
{
    cba[Lmax-1-n] = excitation[sbfrm_size - 1 - n];
}

```

The adaptive codebook is initialized by filling it with zeros.

**Table 3.88 - Representation table for adaptive codebook gain**

Scale factor:  $2^{17}$

Index	cba_gain	Index	cba_gain
0	12386	3	47147
1	27800	4	54827
2	38483	5	61669

Scale factor:  $2^{15}$

Index	cba_gain	Index	cba_gain
6	16993	15	30284
7	18520	16	31877
8	20021	17	33607
9	21493	18	35550
10	22938	19	37808
11	24396	20	40613
12	25834	21	44122
13	27292	22	48847
14	28767	23	55745

Scale factor:  $2^8$

Index	cba_gain	Index	cba_gain
24	528	28	5553
25	727	29	11107
26	1388	30	22214
27	2777	31	44426

**Table 3.89 - Representation table for fixed codebook gain**

Scale factor:  $2^{12}$

Index	cbf_gain	Index	cbf_gain
0	8192	4	26573
1	11578	5	34324
2	14877	6	44160
3	19909	7	55564



Scale factor: 2<sup>9</sup>

Index	cbf_gain
8	8739
9	10823
10	13458
11	16451
12	19978
13	24039

Index	cbf_gain
14	28624
15	33682
16	39118
17	45372
18	52626
19	60243

Scale factor: 2<sup>5</sup>

Index	cbf_gain
20	4327
21	4954
22	5700
23	6551
24	7649
25	8977

Index	cbf_gain
26	10689
27	13066
28	16183
29	21291
30	32832
31	reserved

**Table 3.90 - Representation table for differential fixed codebook gain**

Scale factor: 2<sup>13</sup>

Index	cbf_gain_dif
0	819
1	3205
2	5418
3	7335

Index	cbf_gain_dif
4	9544
5	12763
6	22118
7	53248

### 3.5.7.2 Multi-pulse excitation generation tool

#### 3.5.7.2.1 Tool description

Figure 3.7 shows the block diagram of the Multi-Pulse Excitation Generation tool. For clarity, the indexing for each subframe is dropped. It is noted that the excitation generation process is repeated every subframe. The excitation signal is constructed from a periodic component (adaptive codebook vector) and non-periodic component (fixed codebook vector) scaled by their respective gains. Both the adaptive and the fixed codebook vectors are decoded from **shape\_delay**, **shape\_positions** and **shape\_signs**. The gains are decoded from three types of indices, **signal\_mode**, **rms\_index** and **gain\_index**.

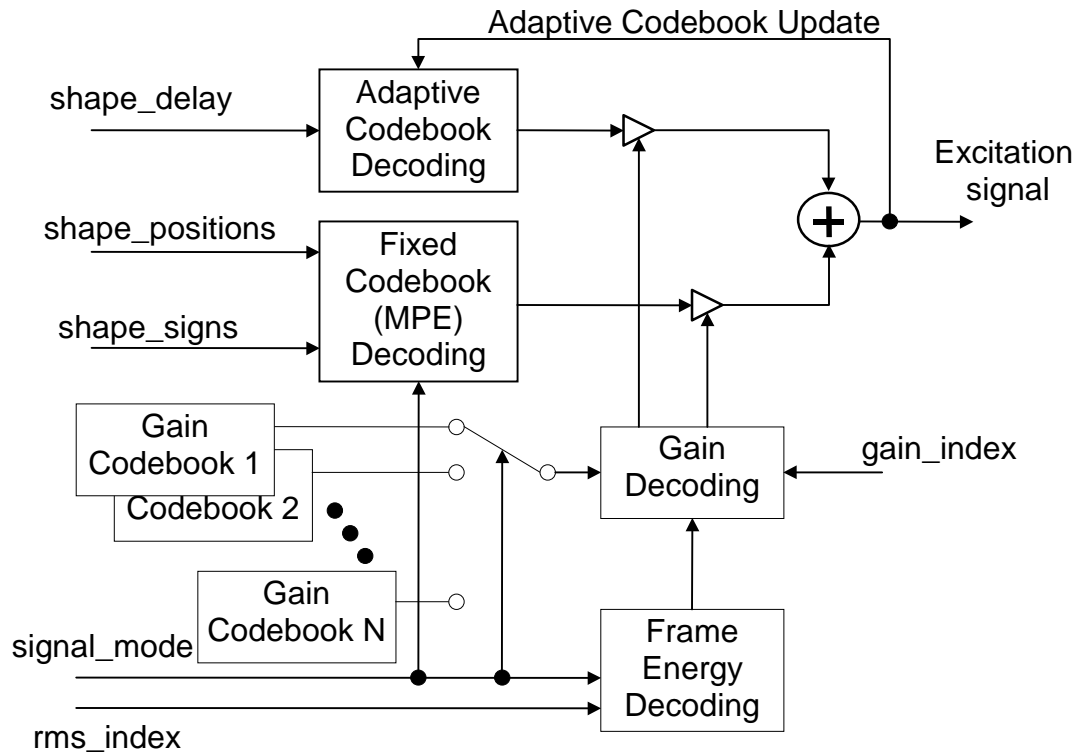


Figure 3.7 - Multi-pulse excitation generator

### 3.5.7.2.2 Definitions

#### Input

**shape\_delay[]**: This array is of dimension *nrof\_subframes* and contains the adaptive codebook lag.

**shape\_positions[]**: This array is of dimension *nrof\_subframes* and contains the pulse positions.

**shape\_signs[]**: This array is of dimension *nrof\_subframes* and contains the pulse signs.

**gain\_index[]**: This array is of dimension *nrof\_subframes* and contains the adaptive codebook gain index and the fixed codebook gain index.

**rms\_index**: This field defines the index for the signal power.

**signal\_mode**: This field contains the voiced/unvoiced flag.

**int\_Qlpc\_coefficients[]**: This is an array of dimension *lpc\_order* and contains the quantized and interpolated LPC coefficients of one subframe.

#### Output

**excitation[]**: This array is of dimension *sbfrm\_size* and contains the excitation signal. This signal is reconstructed from shape and gain vectors using the adaptive and fixed codebooks.

**acb\_delay**: This field contains the decoded lag for the adaptive codevector.

*adaptive\_gain*: This field contains the decoded gain for the adaptive codevector.

Configuration

*lpc\_order*: This field indicates the order of LPC that is used.

*sbfrm\_size*: This field indicates the number of samples in the subframe.

*nrof\_subframes*: This field indicates the number of subframes.

The additional elements used in the MPE tool are as follows:

<i>pacb[]</i>	look-up-table of excitation for the periodic component
<i>pos_tbl[][]</i>	look-up-table of excitation for the non-periodic component
<i>acb[]</i>	decoded excitation signal as the periodic component
<i>fcb[]</i>	decoded excitation signal as the non-periodic component
<i>ga</i>	decoded gain for the periodic component
<i>gf</i>	decoded gain for the non-periodic component
<i>qxnrm[]</i>	decoded root mean squares of the speech signal
<i>par[]</i>	reflection coefficients converted from the <i>int_Qlpc_coefficients[]</i>
<i>acb_energy</i>	energy of <i>acb[]</i>
<i>fcb_energy</i>	energy of <i>fcb[]</i>
<i>acb_delay</i>	Integer part of the pitch lag
<i>acb_frac</i>	Fractional part of the pitch lag

**3.5.7.2.3 Decoding process**

**3.5.7.2.3.1 Decoding of signal\_mode**

**Signal\_mode** represents one of four modes for each frame. Modes 0 and 1 correspond to unvoiced and transition frames, respectively. Modes 2 and 3 correspond to voiced frames, and the latter indicates higher pitch periodicity than the former. This information is utilized in decoding the frame energy, the multi-pulse excitation and gains.

**3.5.7.2.3.2 Decoding of frame energy**

The root mean square (rms) value at last subframe is reconstructed using **signal\_mode** and **rms\_index**. The rms value is decoded in the  $\mu$ -law scale.  $\mu$ -law parameters are dependent on **signal\_mode**. The rms values of other subframes are obtained by linear-interpolating the decoded rms values at the last subframe of the current and the previous frames. The quantized rms values are used for the gain decoding process.

```
delt = 1.0 / 64;  
aa = 1.0 / log10(1.0 + mu_law);  
bb = rms_max / mu_law;
```

```

pwk = aa * log10(1.0 + pqxnorm / bb);
qwk = delt*(rms_index+1);
for (i = 0; i < n_subframes; i++ )
{
    nwk = (qwk-pwk)*(i+1)/n_subframes + pwk;
    qxnorm[i] = bb * (pow((double)10.0, (nwk/aa)) - 1.0);
}
pqxnorm = qxnorm [n_subframes-1];

```

Table 3.91 - Value of rms\_max and mu\_law

Signal_mode	rms_max	mu_law
0	7932	1024
1, 2, 3	15864	512

### 3.5.7.2.3.3 Decoding of the adaptive codebook vector

The integer and the fractional parts of the pitch delay are obtained from **shape\_delay**. The mapping tables between **shape\_delay** and the two parts of the pitch delay are shown in Table 3.92 for the 8 kHz sampling rate and Table 3.93 for the 16 kHz sampling rate. The adaptive codebook vector  $acb[n]$  is computed by interpolating the past excitation signal  $pacb[n]$  at the decoded integer delay  $acb\_delay$  and fraction  $acb\_frac$ . The interpolation is done using an FIR filter  $int\_fil[k]$  based on a Hamming-windowed sinc function. If the value of **shape\_delay** is 255 for the 8 kHz sampling rate or 511 for the 16 kHz sampling rate, the output  $acb[]$  consists of all zero-valued samples. For the other indices, the output  $acb[]$  is produced by the following procedure:

```

for (n = 0; n < sbfrm_size;)
{
    tt += acb_frac;
    kt = acb_delay + tt / 6;
    tt = tt % 6;
    for (i = 0; i < kt && n < sbfrm_size; i++, n++)
    {
        for (k = -iftap; k <= iftap; k++)
        {
            kk = (k+1) * 6 - tt;
            acb[n] += int_fil[abs(kk)] * pacb[pacb_size-(kt-i+k+1)];
            pacb[pacb_size+n] = acb[n];
        }
    }
}

```

where the filter coefficients for the 8 kHz and 16 kHz sampling rates are listed in Annex 3.C. The help parameters  $iftap$  and  $pacb\_size$  are dependent on the sampling rate.

Table 3.92 - The mapping table between the shape\_delay and the pitch delay for 8 kHz sampling rate

shape_delay	acb_delay	acb_frac
0 – 161	$\text{shape\_delay}/3+17$	$(2*\text{shape\_delay})\%6$
162 – 199	$(\text{shape\_delay}-162)/2+71$	$(3*(\text{shape\_delay}-162))\%6$
200 – 254	$\text{shape\_delay}-200+90$	0
255	0	0

**Table 3.93 - The mapping table between the shape\_delay and the pitch delay for 16 kHz sampling rate**

shape_delay	acb_delay	acb_frac
0 – 215	shape_delay/3+20	(2*shape_delay)%6
216 – 397	(shape_delay- 216)/2+92	(3*(shape_delay-216))%6
398 – 510	shape_delay-398+183	0
511	0	0

**Table 3.94 - The number of interpolation filter taps and adaptive codebook size**

Sampling Rate	iftap	pacb_size
8 kHz	6	150
16 kHz	11	306

**3.5.7.2.3.4 Decoding of the fixed codebook vector**

The fixed codebook vector contains several non-zero pulses and is represented by the pulse position and pulse amplitudes. The pulse positions *pul\_pos[i]* are extracted from **shape\_positions**. The pulse amplitudes *pul\_amp[i]* are obtained from **shape\_signs**.

```

for (i = num_pulse-1, k = 0; i >= 0; i--)
{
  for ( j = 0; j < num_bit_pos[i]; j++ )
  {
    pos_idx[i] |= ((shape_positions>>k) & 0x1) << j;
    k++;
  }
  pul_amp[i] = 1.0;
  if (((shape_signs>>(num_pulse-1-i)) & 0x1) == 1)
  {
    pul_amp[i] = -1.0;
  }
  pul_pos[i] = pos_tbl[i][pos_idx[i]];
}

```

**Table 3.95 - The number of pulses for 8 kHz sampling rate**

MPE_Con figuration	num_pulse	MPE_Con figuration	num_pulse	MPE_Con figuration	num_pulse
0	3	10	10	20	11
1	4	11	11	21	12
2	5	12	12	22	8
3	5	13	4	23	9
4	6	14	5	24	10
5	7	15	6	25	11
6	6	16	7	26	12
7	7	17	8	27 ... 31	reserved
8	8	18	9		
9	9	19	10		

Table 3.96 - The number of pulses for 16 kHz sampling rate

MPE_Configuration	num_pulse	MPE_Configuration	num_pulse
0, 16	5	8, 24	3
1, 17	6	9, 25	4
2, 18	7	10, 26	5
3, 19	8	11, 27	6
4, 20	9	12, 28	7
5, 21	10	13, 29	8
6, 22	11	14, 30	9
7, 23	reserved	15, 31	10

where *num\_pulse* is the number of pulses and is set from **MPE\_Configuration** depending on the sampling rate. *num\_bit\_pos[i]* is the number of bits for encoding the *i*-th pulse position. *pos\_tbl[i][j]* is the restriction table, which indicates the possible positions for each pulse. The table indicates the possible positions for each pulse. The table is set up in accordance with the combination of *subfrm\_size*, *num\_pulse* and *num\_bit\_pos[]* as follows:

```

step = subfrm_size / min_num_bit_pos;
for (i = 0; i < num_pulse; i++)
{
    m = 1 << (num_bit_pos[i]-min_num_bit_pos);
    for (j = 0, k = 0; k < m)
    {
        ch[j] = i;
        k++;
        j += (long)((float)step/m + 0.5);
        j = j % step;
    }
}

for (i = 0; i < num_pulse; i++)
{
    for (l = 0, k = 0; k < step; k++)
    {
        if (i == ch[k])
        {
            for (j = 0; j < min_num_bit_pos; j++)
            {
                pos_tbl[i][l++] = k + step * j;
            }
        }
    }
}

```

For example, the pulse position table used in 6 kbit/s Mode II coder for the 8 kHz sampling rate is shown in Table 3.97.

Table 3.97 - The pulse position table for 6 kbit/s coder

Pulse number: l	Num_bit_pos[i]	Pulse positions: pos_tbl[i][j]
0	4	0,5,10,15,20,25,30,35, 40,45,50,55,60,65,70,75
1	4	1,6,11,16,21,26,31,36, 41,46,51,56,61,66,71,76
2	4	2,7,12,17,22,27,32,37 42,47,52,57,62,67,72,77
3	3	3,13,23,33,43,53,63,73
4	3	4,14,24,34,44,54,64,74
5	3	8,18,28,38,48,58,68,78
6	3	9,19,29,39,49,59,69,79

The fixed codebook vector  $fc_b[n]$  is computed from  $pul\_pos[i]$  and  $pul\_amp[i]$  as follows:

```
for (n = 0; n < sbfrm_size; n++)
{
    fcb[n] = 0.0;
}
for (i = 0; i < num_pulse; i++)
{
    fcb[pul_pos[i]] = pul_amp[i];
}
```

If the integer delay  $acb\_delay$  is less than the subframe size  $sbfrm\_size$ , the fixed codebook vector signal  $fc_b[n]$  is modified by zero-state-combfiltering as follows:

```
for (n = 0; n < sbfrm_size; n++)
{
    if ( n - acb_delayt >= 0 )
    {
        ix = fcb[n - acb_delay];
    }
    else
    {
        ix = 0.0;
    }
    fcb[n] += cga[signal_mode] * ix;
}
```

where  $cga[4] = \{0.0, 0.0, 0.6, 0.8\}$  are the gains of the comb-filter.

### 3.5.7.2.3.5 Decoding of the adaptive and the fixed codebook gains

The **gain\_index** is converted to the normalized gains  $nga$ ,  $ngf$  for the adaptive and the fixed codebook vectors by looking up the gain table. The gain table is changed in accordance with **signal\_mode**,  $sbfrm\_size$  and sampling rate. The gain tables for this tool are given in Annex 3.C. The adaptive codebook gain  $ga$  and the fixed codebook gain  $gf$  are calculated as follows:

```
ga = nga * sqrt (norm / acb_energy);
gf = ngf * sqrt (norm / fcb_energy);
```

where *acb\_energy* and *fcb\_energy* are the energies for the adaptive codebook and fixed codebook vectors, respectively. The *norm* is the normalization factor.

```
norm = (qxnorm*subfrm_size)*(qxnorm*subfrm_size);
for(i = 0; i < lpc_order; i++)
{
    norm *= (1 - par[i] * par[i]);
}
```

where *par[]* are the reflection coefficients and are calculated from the LP coefficients *int\_Qlpc\_coefficients[]*. *qxnorm* is the quantized subframe energy, which is decoded from the **rms\_indx** (see subclause 3.5.7.2.3.2).

### 3.5.7.2.3.6 Excitation signal generation

The excitation signal (*excitation[]*) is calculated by summing *acb[]* and *fcb[]* scaled by *ga* and *gf* respectively.

```
for (i = 0; i < sbfrm_size; i++)
{
    excitation[i] = ga*acb[i] + gf*fcb[i]
}
```

### 3.5.7.2.3.7 Updating the adaptive codebook

The adaptive codebook is updated for the decoding process at the next frame by the generated excitation signal *excitation[]* as follows:

```
for(i = 0; i < pacb_size - sbfrm_size; i++)
{
    pacb[i] = pacb[sbfrm_size+i];
}

for(i = 0; i < sbfrm_size; i++)
{
    pacb[pacb_size-sbfrm_size+i] = excitation[i];
}
```

## 3.5.7.3 Bitrate scalable multi-pulse excitation generation tool

### 3.5.7.3.1 Tool description

The bitrate scalable decoder is realized using the bitrate scalable Multi-Pulse Excitation tool, which consists of the Multi-Pulse Excitation tool and the enhancement excitation decoding tool. This scalability enhancement is allowed only for the Mode II coder. The enhancement excitation signal is reconstructed by retrieving **shape\_enh\_positions**, **shape\_enh\_signs**, **gain\_enh\_index** and utilizing the decoded multi-pulse excitation signal in the Multi-Pulse Excitation tool.



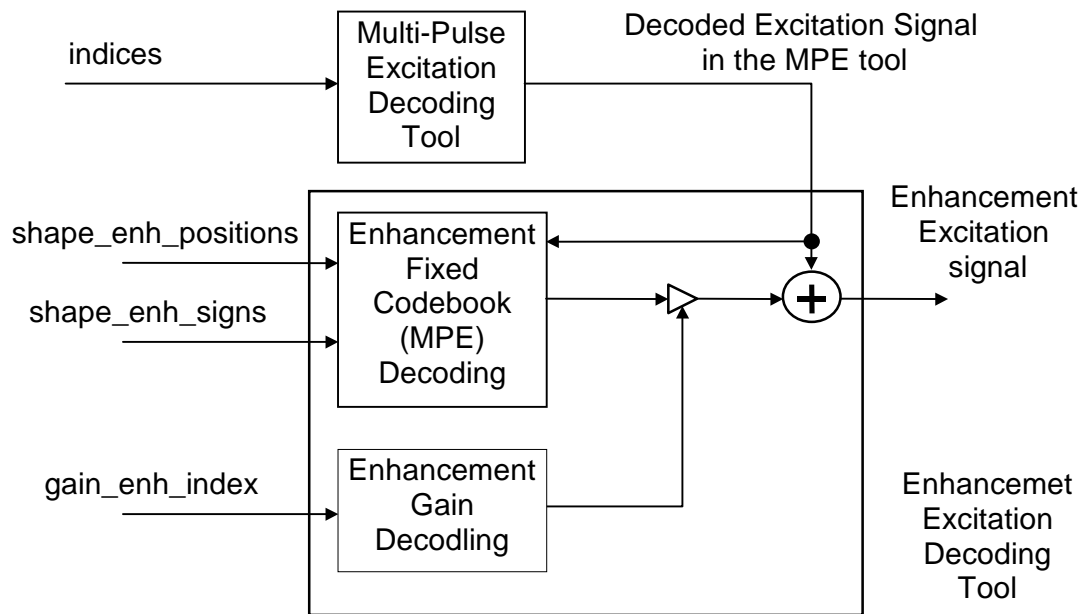


Figure 3.8 - Bitrate scalable multi-pulse excitation generation process

### 3.5.7.3.2 Definitions

#### Input

**shape\_delay[]**: This array is of dimension *nrof\_subframes* and contains the adaptive codebook lag.

**shape\_positions[]**: This array is of dimension *nrof\_subframes* and contains the pulse positions.

**shape\_signs[]**: This array is of dimension *nrof\_subframes* and contains the pulse signs.

**shape\_enh\_positions[]**: This array is of dimension *nrof\_subframes* and contains the pulse positions.

**shape\_enh\_signs[]**: This array is of dimension *nrof\_subframes* and contains the pulse signs.

**gain\_enh\_index[]**: This array is of dimension *nrof\_subframes* and contains the adaptive codebook gain index and the fixed codebook gain index.

**int\_Qlpc\_coefficients[]**: This is an array of dimension *lpc\_order* and contains the quantized and interpolated LPC coefficients of one subframe.

#### Output

**enh\_excitation[]**: This array is of dimension *sbfrm\_size* and contains the enhancement excitation signal. This signal is reconstructed from shape and gain vectors using the fixed codebook.

#### Configuration

**lpc\_order**: This field indicates the order of LPC that is used.

*sbfrm\_size*: This field indicates the number of samples in the subframe.

*nrof\_subframes*: This field indicates the number of subframes.

The additional elements used in the Bitrate Scalable MPE tool are as follows:

<i>pos_tbl[][]</i>	look-up-table of excitation for the non-periodic component
<i>enh_fcb[]</i>	decoded excitation signal as the non-periodic component
<i>ge</i>	decoded gain for the non-periodic component
<i>enh_fcb_energy</i>	energy of <i>enh_fcb[]</i>

### 3.5.7.3.3 Decoding process

#### 3.5.7.3.3.1 Decoding of the enhancement fixed codebook vector

The enhancement fixed codebook vector also consists of several non-zero pulses. The pulse positions and amplitudes are extracted from **shape\_enh\_positions**, **shape\_enh\_signs** by the same decoding algorithm as the fixed codebook. The enhancement fixed codebook vectors *enh\_fcb[n]* is computed from *pul\_pos[i]* and *pul\_amp[i]* as follows:

```

for (i = num_pulse_enh-1, k = 0; i >= 0; i--)
{
  for (j = 0; j < num_bit_pos[i]; j++)
  {
    pos_idx[i] |= ((shape_enh_positions >> k) & 0x1) << j;
    k++;
  }
  pul_amp[i] = 1.0;
  if (((shape_enh_signs >> (num_pulse_enh-1-i)) & 0x1) == 1)
  {
    pul_amp[i] = -1.0;
  }
  pul_pos[i] = pos_tbl[i][pos_idx[i]];
}

for (n = 0; n < sbfrm_size; n++)
{
  enh_fcb[n] = 0.0;
}
for (i = 0; i < num_pulse_enh; i++)
{
  enh_fcb[pul_pos[i]] = pul_amp[i];
}

```

**Table 3.98 - Definition of num\_pulse\_enh**

<i>sbfrm_size</i>	<i>num_pulse_enh</i>
40	2
80	4

The pulse position table used in the enhancement tool is adaptively controlled so that the enhancement pulses stand at different positions from those of the pulses selected in the core excitation. The pulse position table is temporarily generated by the same decoding algorithm as the fixed codebook. The temporary pulse position table is modified as follows,

```

for (n = 0; n < num_enh; n++)
{
  for (i = num[n]-1, k = 0; i >= 0; i--)
  {
    pul_loc = 0;
    for (j = 0; j < bit_pos[i]; j++)
    {
      pul_loc |= ((idx[n]>>k)&0x1)<<j;
      k++;
    }
    pul_loc = chn_pos[i*len+pul_loc];
    for (l = 0; l < 10; l++)
    {
      for (m = 0; m < (1 << bit_pos_org[l]); m++)
      {
        if (pul_loc == chn_pos_org[l*len+m])
        {
          chn_ctr[l]++;
          break;
        }
      }
    }
  }
}

for (i = 0; i < 10; i++ )
{
  ctr_tmp[i] = chn_ctr[i];
}
for (i = 0; i < num[n+1]; i++ )
{
  min_ctr = len;
  for ( j = 0; j < 10; j++ )
  {
    if ( ctr_tmp[j] < min_ctr )
    {
      min_ctr = ctr_tmp[j];
      min_chn = j;
    }
  }
  ctr_tmp[min_chn] = len;
  bit_pos[i] = bit_pos_org[min_chn];
  for ( j = 0; j < (1<<bit_pos_org[min_chn]); j++ )
  {
    chn_pos[i*len+j] = chn_pos_org[min_chn*len+j];
  }
}
}

for (i = 0; i < num[num_enh]; i++)
{
  bit[i] = bit_pos[i];
  for (j = 0; j < (1<<bit[i]); j++)

```

```

{
    pos_tbl[i*len+j] = chn_pos[i*len+j];
}
}

```

The temporary pulse position tables used in the enhancement excitation tool is shown in Table 3.99 and Table 3.100. The table changes depending on the subframe length.

**Table 3.99 - The temporary pulse position table for 80-sample subframe**

Pulse number: i	bit_pos_org[i]	Pulse positions: chn_pos_org[i][j]
0	3	0,10,20,30,40,50,60,70
1	3	1,11,21,31,41,51,61,71
2	3	2,12,22,32,42,52,62,72
3	3	3,13,23,33,43,53,63,73
4	3	4,14,24,34,44,54,64,74
5	3	5,15,25,35,45,55,65,75
6	3	6,16,26,36,46,56,66,76
7	3	7,17,27,37,47,57,67,77
8	3	8,18,28,38,48,58,68,78
9	3	9,19,29,39,49,59,69,79

**Table 3.100 - The temporary pulse position table for 40-sample subframe**

Pulse number: i	bit_pos_org[i]	Pulse positions: chn_pos_org[i][j]
0	2	0,10,20,30
1	2	1,11,21,31
2	2	2,12,22,32
3	2	3,13,23,33
4	2	4,14,24,34
5	2	5,15,25,35
6	2	6,16,26,36
7	2	7,17,27,37
8	2	8,18,28,38
9	2	9,19,29,39

### 3.5.7.3.3.2 Decoding of the enhancement fixed codebook gain

The **gain\_enh\_index** is converted to the normalized gain *nge* for the enhancement fixed codebook vector by looking up the gain table. The gain table is changed in accordance with **signal\_mode** and is given in Annex 3.C. The enhancement fixed codebook gain *ge* are calculated as follows:

$$ge = nge * \text{sqrt} (\text{norm} / \text{enh\_fcb\_energy});$$

where the *enh\_fcb\_energy* is the energy for the enhancement fixed codebook vector. The *norm* is the normalization factor.

### 3.5.7.3.3.3 Enhanced excitation signal generation

The enhanced excitation signal (*enh\_excitation[l]*) is calculated by adding the enhancement codebook vector to excitation signals.

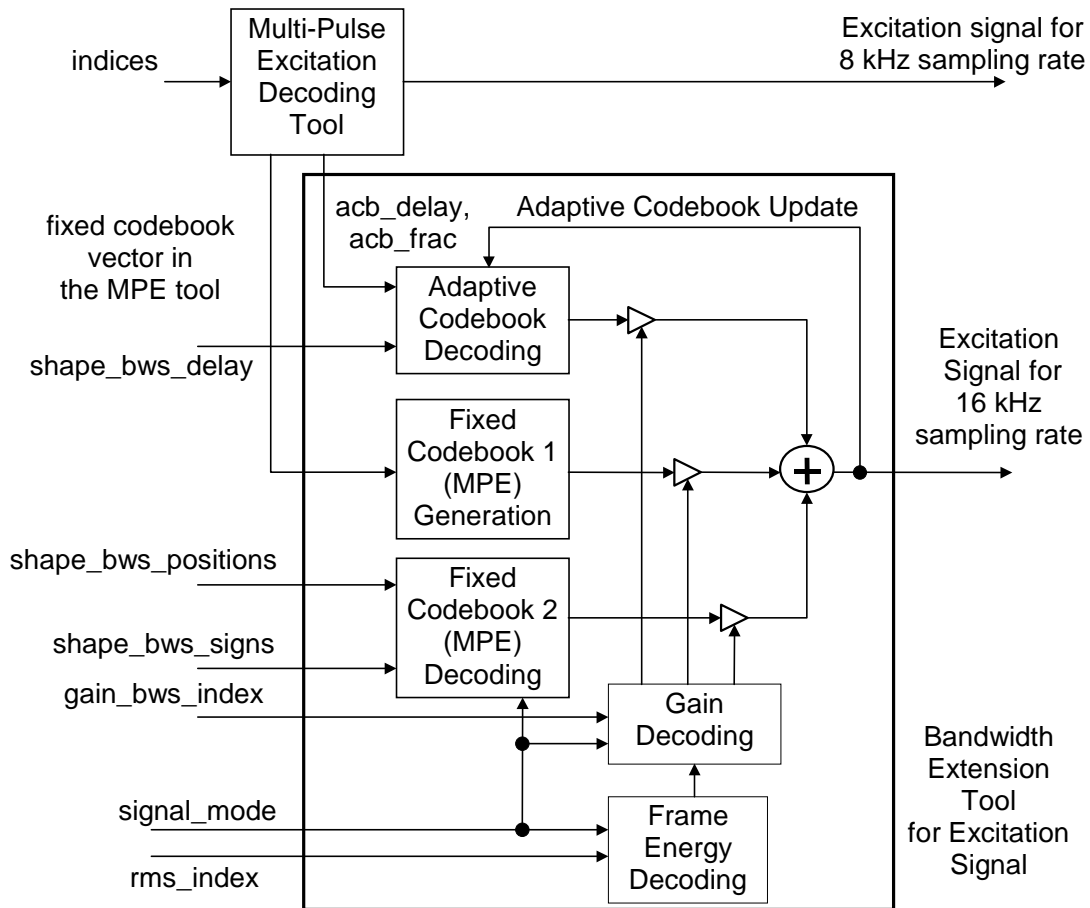
```

for (i = 0; i < sbfrm_size; i++)
{
    enh_excitation[i] = excitation[i] + ge*enh_fcb[i]
}
    
```

**3.5.7.4 Bandwidth scalable multi-pulse excitation generation tool**

**3.5.7.4.1 Tool description**

The excitation decoding process in the bandwidth scalable decoder is achieved by the bandwidth scalable Multi-Pulse Excitation tool, which consists of the Multi-Pulse Excitation tool and the bandwidth extension tool as shown in Figure 3.9. This scalability enhancement is allowed only for the Mode II coder. The scalable excitation signal is reconstructed by retrieving **shape\_bws\_positions**, **shape\_bws\_signs** and **gain\_bws\_index** and utilizing the decoded outputs for the 8 kHz sampling rate, namely the integer and the fractional parts of the pitch delay and the fixed codebook vector. These outputs are generated at the MPE decoding tool (subclause 3.5.7.2) or the bitrate scalable MPE decoding tool (subclause 3.5.7.3).



**Figure 3.9 - Bandwidth scalable multi-pulse excitation generation process**

### 3.5.7.4.2 Definitions

#### Input

**shape\_bws\_delay[]**: This array is of dimension *nrof\_subframes\_bws* and contains the adaptive codebook lag.

**shape\_bws\_positions[]**: This array is of dimension *nrof\_subframes\_bws* and contains the pulse positions.

**shape\_bws\_signs[]**: This array is of dimension *nrof\_subframes\_bws* and contains the pulse signs.

**gain\_bws\_index[]**: This array is of dimension *nrof\_subframes\_bws* and contains the adaptive codebook gain index and the fixed codebook gain index.

**rms\_index**: This field defines the index for the signal power.

**signal\_mode**: This field contains the voiced/unvoiced flag.

**int\_Qlpc\_coefficients[]**: This is an array of dimension *lpc\_order* and contain the quantized and interpolated LPC coefficients of one subframe.

#### Output

**excitation[]**: This array is of dimension *sbrfm\_size* and contains the excitation signal. This signal is reconstructed from shape and gain vectors using the adaptive and fixed codebooks.

**acb\_delay**: This field contains the decoded lag for the adaptive codevector.

**adaptive\_gain**: This field contains the decoded gain for the adaptive codevector.

#### Configuration

**lpc\_order**: This field indicates the order of LPC that is used.

**sbrfm\_size**: This field indicates the number of samples in the subframe in the bandwidth extension tool.

**nrof\_subframes\_bws**: This field indicates the number of subframes in the bandwidth extension tool.

The additional elements used in the Bandwidth Scalable MPE tool are as follows:

<b>pacb[]</b>	look-up-table of excitation for the periodic component
<b>pos_tbl[][]</b>	look-up-table of excitation for the non-periodic component
<b>acb[]</b>	decoded excitation signal as the periodic component
<b>fcb1[]</b>	decoded excitation1 signal as the non-periodic component
<b>fcb2[]</b>	decoded excitation2 signal as the non-periodic component
<b>ga</b>	decoded gain for the periodic component
<b>gn</b>	decoded gain for the non-periodic component
<b>gf</b>	decoded gain for the non-periodic component
<b>qxnorm[]</b>	decoded root mean squares of the speech signal

<i>par</i> []	reflection coefficients converted from the <i>int_Qlpc_coefficients</i> []
<i>acb_energy</i>	energy of <i>acb</i> []
<i>fcb1_energy</i>	energy of <i>fcb1</i> []
<i>fcb2_energy</i>	energy of <i>fcb2</i> []
<i>acb_delay_wb</i>	Integer part of the pitch lag
<i>acb_frac_wb</i>	Fractional part of the pitch lag

### 3.5.7.4.3 Decoding process

For the Mode II decoder with bandwidth scalability, the excitation signal at 16 kHz sampling is constructed from a periodic component (adaptive codebook vector) and two non-periodic components (fixed codebook vector 1 and 2) scaled by their respective gains.

#### 3.5.7.4.3.1 Decoding of signal\_mode

**Signal\_mode** is also utilized in decoding the frame energy, the multi-pulse excitation and gains in this decoding process.

#### 3.5.7.4.3.2 Decoding of frame energy

The decoding procedure is same as the MPE decoding tool.

#### 3.5.7.4.3.3 Decoding of the adaptive codebook vector

The integer and the fractional parts of the pitch delay are obtained from **shape\_delay** and **shape\_bws\_delay**.

The *acb\_delay* and *acb\_frac* in the 8 kHz sampling rate are decoded at the MPE tool and are fed to the bandwidth extension tool for the excitation signal. The 8 kHz sampling rate parameters are converted to the 16 kHz sampling rate parameters *acb\_delay\_wb*, *acb\_frac\_wb* in 16 kHz sampling rate as follows:

```
op_delay_wb = 2 * acb_delay
if (acb_frac != 0)
{
    op_delay_wb ++
}

if (op_delay_wb == 0)
{
    op_idx_wb = 778
}
else
{
    op_idx_wb = (op_delay_wb - 32) * 3 + 2
}

st_idx_wb = op_idx_wb - 4
if (st_idx_wb < 0)
{
```

```

    st_idx_wb = 0
}
if ((st_idx_wb + 7) >= 778)
{
    st_idx_wb = 778 - 8
}

if (op_idx_wb == 778)
{
    acb_idx_wb = 778
}
else
{
    acb_idx_wb = st_idx_wb + shape_bws_delay
}

```

The mapping table between `acb_idx_wb` and the pitch delay parameters `acb_delay_wb`, `acb_frac_wb` is shown in Table 3.101. The adaptive codebook vector  $acb[n]$  is computed by interpolating the past excitation signal  $pacb[n]$  at the decoded integer delay `acb_delay_wb` and fraction `acb_frac_wb`. The interpolation is done using an FIR filter  $int\_fil[k]$ ,  $k=0,\dots,66$  based on a Hamming windowed sinc function. If the value of **shape\_delay** is 255 or the value of **shape\_bws\_delay** is 768, the output  $acb[]$  consists of all zero-valued samples. For other combinations of the indices, the output  $acb[]$  is produced by the following procedure:

```

for (n = 0; n < sbfrm_size;)
{
    tt += acb_frac_wb;
    kt = acb_delay_wb + tt / 6;
    tt = tt % 6;
    for (i = 0; i < kt && n < sbfrm_size; i++, n++)
    {
        for (k = -11; k <= 11; k++)
        {
            kk = (k+1) * 6 - tt;
            acb[n] += int_fil [abs(kk)] * pacb [306-(kt-i+k)];
            pacb[306+n] = acb[n];
        }
    }
}

```

where the filter coefficients for the 16 kHz sampling rate are listed in Annex 3.C.

**Table 3.101 - The mapping table between the shape\_delay and the pitch delay**

acb_idx_wb	acb_delay_wb	acb_frac_wb
0, 1	32	$(2^{*}(\text{acb\_idx\_wb}+1))\%6$
2 - 777	$32+2^{*}(\text{acb\_idx\_wb}-2)/6$	$(2^{*}(\text{acb\_idx\_wb}-2))\%6$
778	0	0

#### 3.5.7.4.3.4 Decoding of the fixed codebook vector 1

The fixed codebook vector 1  $fc1[n]$  is obtained by sampling-rate conversion of the fixed codebook vector  $nb\_fc1[n]$  used in the MPE tool or the bitrate scalable MPE tool as follows:



```

for (n = 0; n < sbfrm_size/2; n++)
{
    fcb1[2*n] = nb_fcb[n]
    fcb1[2*n+1] = 0
}

```

**3.5.7.4.3.5 Decoding of the fixed codebook vector 2**

The fixed codebook vector 2 contains several non-zero pulses and is represented by the pulse position and pulse amplitudes. The pulse positions *pul\_pos[i]* are extracted from **shape\_bws\_positions**. The pulse amplitudes *pul\_amp[i]* are obtained from **shape\_bws\_signs**.

```

for (i = num_pulse_bws - 1, k = 0; i >= 0; i-- )
{
    for (j = 0; j < num_bit_pos[i]; j++)
    {
        pos_idx[i] |= ((shape_bws_positions >> k) & 0x1) << j;
        k++;
    }
    pul_amp[i] = 1.0;

    if (((shape_bws_signs >> (num_pulse_bws-1-i)) & 0x1) == 1)
    {
        pul_amp[i] = -1.0;
    }
    pul_pos[i] = pos_tbl[i][pos_idx[i]];
}

```

where *num\_pulse\_bws* is the number of pulses and is one of 6, 8, 10, 12. The choice is dependent on *BWS\_configuration*. *num\_bit\_pos[i]* is the number of bits for encoding the *i*-th pulse position. *pos\_tbl[i][j]* is the restriction table, which indicates the possible positions for each pulse. The table indicates the possible positions for each pulse.

**Table 3.102 - Definition of num\_pulse\_bws**

<b>BWS_configuration</b>	<i>num_pulse_bws</i>
0	6
1	8
2	10
3	12

The pulse position tables for each number of pulses are also set up in accordance with the combination of *subfrm\_size*, *num\_pulse* and *num\_bit\_pos[]* by the same procedure as the MPE tool.

The fixed codebook vector *fc2[n]* is computed from *pul\_pos[i]* and *pul\_amp[i]* as follows:

```

for (n = 0; n < sbfrm_size; n++)
{
    fcb2[n] = 0.0;
}
for (i = 0; i < num_pulse_bws; i++)
{
    fcb2[pul_pos[i]] = pul_amp[i];
}

```

If the integer delay *acb\_delay\_wb* is less than the subframe size *sbfrm\_size*, the fixed codebook vector signal *fcb2[n]* is modified by zero-state-comb filtering as follows:

```

for (n = 0; n < sbfrm_size; n++)
{
    if (n - acb_delayt >= 0)
    {
        ix = fcb2[n - acb_delay];
    }
    else
    {
        ix = 0.0;
    }
    fcb2[n] += cga[signal_mode] * ix;
}

```

where  $cga[4] = \{0.0, 0.0, 0.6, 0.8\}$  are the gains of the comb-filter.

### 3.5.7.4.3.6 Decoding of the adaptive and the fixed codebook gains

The *gain\_bws\_index* is converted to two indices.

```

qga_idx = gain_bws_index >> 7;
qgc_idx = gain_bws_index - (gain_bws_index << 7);

```

The *qga\_idx* is converted to the normalized gains *nga*, *ngf* for the adaptive and the fixed codebook vector 2 by looking up the gain table. The *qgc\_idx* is converted to the normalized gain *ngn* for the fixed codebook vector 1 by looking up the gain table. The gain table is changed in accordance with the **signal\_mode** and is given in Annex 3.C. The adaptive codebook gain *ga*, the fixed codebook 1 gain *gn* and the fixed codebook 2 gain *gf* are calculated as follows:

```

ga = nga * sqrt( norm / acb_energy );
gn = ngn * sqrt( norm / fcb1_energy );
gf = ngf * sqrt( norm / fcb2_energy );

```

where the *acb\_energy*, *fcb2\_energy* and *fcb1\_energy* are the energies for the adaptive codebook and the two fixed codebook vectors. The *norm* is the normalization factor.

```

norm = (qxnrm*subfrm_size) * (qxnrm*subfrm_size);
for (i = 0; i < lpc_order; i++)
{
    norm *= (1 - par[i] * par[i]);
}

```

where  $par[]$  are the reflection coefficients and are calculated from the LP coefficients  $int\_Qlpc\_coefficients[]$ . The  $qnorm$  is the quantized subframe energy and is decoded from the  $rms\_indx$ .

#### 3.5.7.4.3.7 Excitation signal generation

The excitation signal ( $excitation[]$ ) is calculated by summing  $acb[]$ ,  $fcbl[]$  and  $fcbl2[n]$  scaled by  $ga$ ,  $gn$  and  $gf$  respectively.

```
for (i = 0; i < sbfrm_size; i++)
{
    excitation[i] = ga*acb[i] + gn*fcbl[i] + gf*fcbl2[i]
}
```

#### 3.5.7.4.3.8 Updating the adaptive codebook

The adaptive codebook is updated for the decoding process at the next frame by the generated excitation signal  $excitation[]$  as follows:

```
for (i = 0; i < 306-sbfrm_size; i++)
{
    pacb[i] = pacb[sbfrm_size+i];
}

for (i = 0; i < sbfrm_size; i++)
{
    pacb[306-sbfrm_size+i] = excitation[i];
}
```

### 3.5.8 CELP LPC synthesis filter

#### 3.5.8.1 Tool description

The CELP LPC synthesis filter constructs the synthesized signal from the LPC coefficients and the excitation signal for each subframe.

#### 3.5.8.2 Definitions

##### Input

$excitation[]$ : This array contains the excitation signal for one subframe.

$int\_Qlpc\_coefficients[]$ : This array of dimension  $lpc\_order$  contains the quantized and interpolated LPC coefficients.

##### Output

$synth\_signal[]$ : The excitation signal,  $excitation[]$ , is fed through the synthesis filter using the LPC coefficients of  $int\_Qlpc\_coefficients[]$ . The dimension of this array is  $lpc\_order$ .

## Configuration

*lpc\_order*: This field contains the order of LPC used.

*sbfrm\_size*: This field contains the number of samples in the subframe.

### 3.5.8.3 Decoding process

Using the interpolated LPC coefficients of one subframe, the excitation signal is fed through the following filter:

$$H_s(z) = \frac{1}{A(z)} = \frac{1}{1 - \sum_{k=1}^{lpc\_order} a_k \cdot z^{-k}}$$

where  $A(z)$  is the LPC inverse filter using the quantized LPC coefficients. The coefficient  $a_k$  is the  $k$ -th LPC coefficient (*int\_Qlpc\_coefficients*[ $k-1$ ]). The output of the inverse filter is the reconstructed speech. The LPC order is fixed to 10 and 20 for the 8 kHz and 16 kHz sampling rate, respectively.

The following algorithm is an implementation of the above filter.

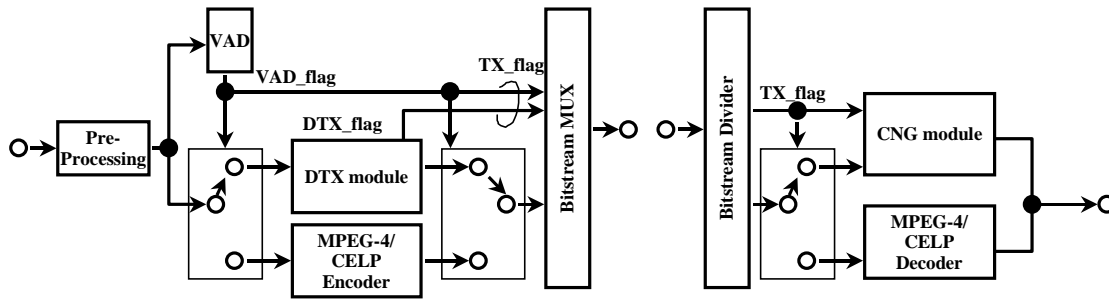
```
for(n = 0; n < sbfrm_size; n++)
{
    tmp = excitation[n];
    for(k = 0; k < lpc_order; k++)
    {
        tmp = tmp + Filter_states[k] * int_Qlpc_coefficients[k];
    }
    synth_signal[n] = tmp;
    for(k = lpc_order-1; k > 0; k--)
    {
        Filter_states[k] = Filter_states[k-1];
    }
    Filter_states[0] = synth_signal[n];
}
```

The array *Filter\_states* is initially set to zero.

## 3.5.9 CELP silence compression tool

### 3.5.9.1 Tool description

The silence compression tool comprises a Voice Activity Detection (VAD) module, a Discontinuous Transmission (DTX) unit and a Comfort Noise Generator (CNG) module. The tool encodes/decodes the input signal at a lower bitrates during the non-active (silent) frames. During the active-voice (speech) frames, MPEG-4 CELP encoding and decoding are used. A block diagram of the CELP codec system with the silence compression tool is depicted in Figure 3.10.



**Figure 3.10 - Block diagram of the codec system with the silence compression tool**

At the transmission side, the DTX module encodes the input speech during the non-active frames. During the active-voice frames, the MPEG-4 CELP encoder is used. The voice activity flag (*VAD\_flag*) indicating a non-active frame (*VAD\_flag*=0) or an active-voice frame (*VAD\_flag*=1) is determined from the input speech by the VAD module. During the non-active frames, the DTX module detects frames where the input characteristics change (*DTX\_flag*=1 and 2: Change, *DTX\_flag*=0: No Change). When a change is detected, the DTX module encodes the input speech to generate a SID (Silence Insertion Descriptor) information. The *VAD\_flag* and the *DTX\_flag* are sent together as a *TX\_flag* to the decoder to keep synchronization between the encoder and the decoder.

At the receive side, the CNG module generates comfort noise based on the SID information during the non-active frames. During the active-voice frames, the MPEG-4 CELP decoder is used instead. Either the CNG module or the MPEG-4 CELP decoder is selected according to the *TX\_flag*.

The SID information and the *TX\_flag* are transmitted only when a change of the input characteristics is detected. Otherwise only the *TX\_flag* is transmitted during non-active frames.

### 3.5.9.2 Definitions

*CNG*: Comfort Noise Generation

*Coding mode*: “I” for the RPE and “II” for the MPE (see Table 3.1)

*DTX*: Discontinuous Transmission

*LP*: Linear Prediction

*LPCs*: LP Coefficients

*MPE*: Multi-Pulse Excitation

*MPE\_Configuration*: see subclause 3.4

*RMS*: root mean square

*RPE*: Regular-Pulse Excitation

*RPE\_Configuration*: see subclause 3.4

*SID*: Silence Insertion Descriptor

*SID frame*: frame where the SID information is sent/received

*signal mode*: mode determined based on the average pitch prediction gain (see subclause 3.4)

*VAD*: Voice Activity Detection

### 3.5.9.3 Decoding process

#### 3.5.9.3.1 Transmission payload

There are four types of transmission payloads depending on the VAD/DTX decision. A TX\_flag indicates the type of transmission payload and is determined by the VAD\_flag and the DTX\_flag as shown in Table 3.103. When the TX\_flag indicates a active-voice frame (TX\_flag=1), information generated by the MPEG-4 CELP encoder and the TX\_flag are transmitted. When the TX\_flag indicates a transition frame between an active-voice frame and a non-active frame, or a non-active frame in which the spectral characteristics of the input signal changes (TX\_flag=2), High-Rate (HR) SID information and the TX\_flag are transmitted to update the CNG parameters. When the TX\_flag indicates a non-active frame in which the frame power of the input signal changes (TX\_flag=3), Low-Rate (LR) SID information and the TX\_flag are transmitted. Other non-active frames are categorized into the fourth type of the TX\_flag (TX\_flag=0). In this case, only the TX\_flag is transmitted. Examples of the TX\_flag change according to the VAD\_flag and the DTX\_flag are shown in Table 3.104.

**Table 3.103 - Relation among flags for the silence compression tool**

Flags	Active-voice	Non-active		
VAD_flag	1	0		
DTX_flag	-	0	1	2
TX_flag	1	0	2	3

**Table 3.104 - Examples of the TX\_flag change according to the VAD\_flag and the DTX\_flag**

Frame #	...k-5	k-4	k-3	k-2	k-1	k	k+1	k+2	k+3	k+4	k+5	k+6	k+7
VAD_flag	...1	1	1	1	0	0	0	0	0	0	0	0	0...
DTX_flag	...-	-	-	-	1	0	0	0	1	0	0	0	2...
TX_flag	...1	1	1	1	2	0	0	0	2	0	0	0	3...

active-voice period	non-active period
---------------------	-------------------

#### 3.5.9.3.2 LSP transmission

In case the silence compression tool is used in combination with FineRate Control enabled, a CELP frame with LPC\_present = 1 and interpolation\_flag = 0 must be transmitted in the first voice-active frame following a non-active frame. Voice-active frames are signaled by TX\_flag = 1, non-active frames are signaled by TX\_flag = 0, 2 or 3.

#### 3.5.9.3.3 CNG module

Figure 3.11 depicts a structure of the CNG module that generates the comfort noise. The comfort noise is generated by filtering an excitation with an LP synthesis filter in a similar manner as with voice-active speech signals. The post filter may be used to improve the coding quality. The excitation is given by adding a multi-pulse excitation or a regular pulse excitation and a random excitation, scaled by their corresponding gains. The excitations are generated based on a random sequence independent of the SID information. The coefficients for the LP synthesis filter and gains are calculated from LSPs and a RMS value (frame energy) respectively, which are received as the SID information. The LSPs and the RMS are smoothed in order to improve the coding quality for the noisy input speech. The CNG module uses the same frame and subframe sizes as those in active speech frames. Processing in each part is described in the following subclauses.

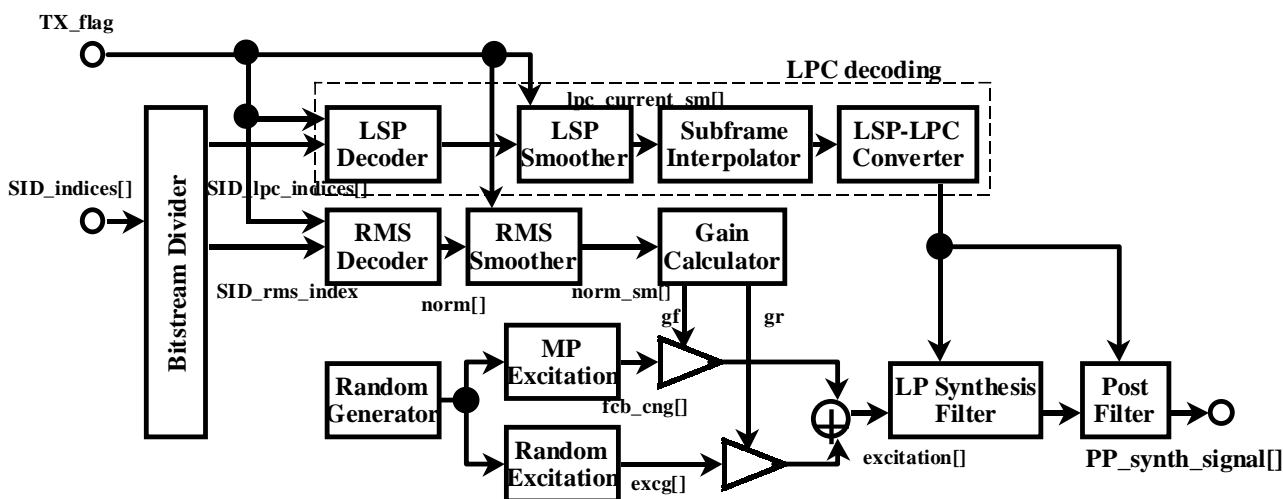


Figure 3.11 - CNG module

3.5.9.3.3.1 Definitions

Input

- TX\_flag This field contains the transmission mode.
- SID\_lpc\_indices[] This array contains the packed LP indices. The dimension is 3, 5 or 6 (see Table 3.72).
- SID\_rms\_index This field contains the RMS index.

Output

- PP\_synth\_signal[] This array contains the post-filtered (enhanced) speech signal. The dimension is *sbfrm\_size*. The following are help elements used in the CNG module:

- lpc\_order*: the order of LP
- sbfrm\_size*: the number of samples in a subframe
- n\_subframe*: the number of subframes in a frame
- int\_Q\_lpc\_coefficients[]*: interpolated LPCs (see subclause 3.5.8.2).

3.5.9.3.3.2 LSP decoder

LSP *lpc\_current[]* is decoded from the LSP indices *SID\_lpc\_indices[]*. The decoding process is identical to that described in subclause 3.5.6 with the following exceptions:

- (1) A sub-set of the *lpc\_indices[]* is transmitted to the decoder. A relation between the transmitted LSP indices *SID\_lpc\_indices[]* and the LSP indices for MPEG-4 CELP, *lpc\_indices[]* is shown in Table 3.105.
- (2) The decoding process is not carried out for the untransmitted indices.

Table 3.105 - LSP index relation between the silence compression tool and MPEG-4 CELP

Coding mode	Sampling rate [kHz]	Band width scalability	Silence compression tool	MPEG-4 CELP
I (RPE)	16	Off	SID_lpc_indices[0]	lpc_indices[0]
			SID_lpc_indices[1]	lpc_indices[1]
			SID_lpc_indices[2]	lpc_indices[2]
			SID_lpc_indices[3]	lpc_indices[3]
			SID_lpc_indices[4]	lpc_indices[5]
			SID_lpc_indices[5]	lpc_indices[6]
II (MPE)	8	On, Off	SID_lpc_indices[0]	lpc_indices[0]
			SID_lpc_indices[1]	lpc_indices[1]
			SID_lpc_indices[2]	lpc_indices[2]
	16	On	SID_lpc_indices[3]	lpc_indices[5]
			SID_lpc_indices[4]	lpc_indices[6]
			SID_lpc_indices[5]	lpc_indices[7]
			SID_lpc_indices[6]	lpc_indices[8]
		Off	SID_lpc_indices[0]	lpc_indices[0]
			SID_lpc_indices[1]	lpc_indices[1]
			SID_lpc_indices[2]	lpc_indices[2]
			SID_lpc_indices[3]	lpc_indices[3]
			SID_lpc_indices[4]	lpc_indices[5]
			SID_lpc_indices[5]	lpc_indices[6]

### 3.5.9.3.3.3 LSP smoother

Smoothed LSPs  $lsp\_current\_sm[i]$  are updated using the decoded LSPs  $lsp\_current[i]$  in each frame as:

$$lsp\_current\_sm[i] = \begin{cases} 0.875 lsp\_current\_sm[i] + 0.125 lsp\_current[i], & TX\_flag = 2 \\ 0.875 lsp\_current\_sm[i] + 0.125 lsp\_current\_sid[i], & TX\_flag = 0 \text{ or } 3 \end{cases}$$

where  $i=0, \dots, lpc\_order-1$  and  $lsp\_current\_sid[i]$  is  $lsp\_current[i]$  in the last SID frame. At the beginning of every non-active period,  $lsp\_current\_sm[i]$  is initialized with  $lsp\_current[i]$  at the end of the previous active-voice period.

### 3.5.9.3.3.4 LSP interpolation and LSP-LPC conversion

LPCs for the LP synthesis,  $int\_Q\_lpc\_coefficients[i]$  are calculated from the smoothed LSPs  $lpc\_current\_sm[i]$  using the LSP interpolation with the stabilization and the LSP-to-LPC conversion. These processes are described in subclause 3.5.6. A common buffer for the previous frame  $lsp\_previous[i]$  is used in both the silence compression tool and in CELP coding for active speech frames.

### 3.5.9.3.3.5 RMS decoder

The RMS of the input speech  $qxnorm$  in each subframe is reconstructed using  $SID\_rms\_index$  in the same process as described in subclause 3.5.7.2.3.2 with the exception that the  $\mu$ -law parameters are independent of the signal mode and set as  $rms\_max=7932$  and  $mu\_law=1024$ .

The reconstructed RMS of the input speech is converted into the RMS of the excitation signal ( $norm$ ) using the reflection coefficients  $par[i]$  as follows and used for the gain calculation:

```
norm = (qxnorm*subfrm_size)*(qxnorm*subfrm_size);
for (i = 0; i < lpc_order; i++)
{
```



```

    norm *= (1 - par[i] * par[i]) *  $\alpha_s$ ;
}

```

where  $par[]$  is calculated from the LPCs  $int\_Qlpc\_coefficients[]$  and a scaling factor  $\alpha_s$  is 0.8.

### 3.5.9.3.3.6 RMS smoother

A smoothed RMS  $norm\_sm$  is updated using  $norm$  in each subframe as follows:

$$norm\_sm = \begin{cases} 0.875 norm\_sm + 0.125 norm[subnum] & \text{for } TX\_flag = 2 \text{ or } 3 \\ 0.875 norm\_sm + 0.125 norm\_sid & \text{for } TX\_flag = 0 \end{cases}$$

where  $subnum$  is the current subframe number ranging from 0 to  $n\_subframe-1$  and  $norm\_sid$  is  $norm[n\_subframe-1]$  in the last SID frame. In the first frame of every non-active period,  $norm\_sm$  is set to  $norm$ . During the first 40 msec of the non-active period,  $norm\_sm$  is initialized with  $norm[subnum]$ , when  $TX\_flag = 2$  or 3 and

$$|20\log_{10}norm\_sid - 20\log_{10}norm[n\_subframe - 1]| > 6 \text{ dB}.$$

### 3.5.9.3.3.7 CNG excitation generation

The CNG excitation signal  $excitation[]$  is computed from a multi-pulse excitation signal and a random excitation signal as follows:

```

for (i = 0; i < sbfrm_size; i++)
{
    excitation[i] = gf * fcb_cng[i] + gr * excg[i];
}

```

where  $fcb\_cng[]$  and  $excg[]$  are respectively the multi-pulse (MP) or regular pulse (RP) excitation signal and a random excitation signal.  $gf$  and  $gr$  are their corresponding gains. The random excitation signal, positions and signs of pulses for the MP/RP excitation are sequentially produced from a random sequence in each subframe. In order to synchronise the random number CNG generator in the encoder and the decoder, the random excitation signal  $excg[]$  for a given subframe must be computed prior to the MP/RP Excitation generation  $fcb\_cng[]$  for that subframe.

#### 3.5.9.3.3.7.1 Random sequence

Random sequence are generated by the following function and are used for generation of the multi-pulse and the random excitation signals:

```

short Random (*seed)
{
    *seed = (short) ((int)(*seed * 31821 + 13849));
    return (*seed);
}

```

with an initial seed value of 21845 and commonly used for both excitations. This generator has a periodic cycle of 16 bits. The seed is initialized with 21845 at the beginning of every non-active period.

#### 3.5.9.3.3.7.2 Generation of the random excitation

The random excitation signal of each subframe is a Gaussian random sequence, which is generated as follows:

```

for (i = 0; i < sbfrm_size; i++)
{
    excg[i] = Gauss (seed);
}

```

where

```

float Gauss (short *seed)
{
    temp = 0;
    for (i = 0; i < 12; i++)
    {
        temp += (float) Random (seed);
    }
    temp /= (2 * 32768);
    return (temp);
}

```

### 3.5.9.3.3.7.3 Multi-pulse excitation generation

In case MPE is used for coding of the voice-active frames, a multi-pulse excitation signal is generated for each subframe by randomly selecting the positions and signs of pulses. Multi-pulse structures of MPEG-4 version 1 CELP with MPE\_Configuration = 24 and 31 are used for the sampling rate of 8 and 16 kbit/s, respectively. Positions and signs of 10 pulses are generated in a 40-sample vector. For subframe size of 80 samples, it is carried out twice to generate 20 pulses in an 80-sample vector. Indices of the positions and signs, mp\_pos\_idx and mp\_sign\_idx, are generated in each subframe as follows:

```

if (subframe size is 40 samples)
{
    setRandomBits (&mp_pos_idx, 20, seed);
    setRandomBits (&mp_sgn_idx, 10, seed);
}

if (subframe size is 80 samples)
{
    setRandomBits (&mp_pos_idx_1st_half, 20, seed);
    setRandomBits (&mp_sgn_idx_1st_half, 10, seed);
    setRandomBits (&mp_pos_idx_2nd_half, 20, seed);
    setRandomBits (&mp_sgn_idx_2nd_half, 10, seed);
}

```

where mp\_pos\_idx\_1st\_half and mp\_sgn\_idx\_1st\_half are indices of the positions and signs of the first half of the subframe and mp\_pos\_idx\_2nd\_half and 20 mp\_sgn\_idx\_2nd\_half are indices for the second half. The function setRandomBits() is defined in subclause 3.5.9.3.3.7.5.

### 3.5.9.3.3.7.4 Regular Pulse Excitation

In case RPE is used for coding of the voice-active frames, a regular pulse excitation signal is generated for each subframe. In case of non-active frames, the content of the adaptive codebook is initialised with zero. For the non-active frames only the fixed codebook is used. The fixed codebook excitation signal is generated by using a random shape\_index as input for the RPE decoding process.

```

setRandomBits (&shape_index, n_bits, seed);

```

```

rpe_index = shape_index;
rpe_phase = rpe_index % D;
rpe_index = rpe_index / D;

for (n = Np - 1; n >= 0; n--)
{
    rpe_amps [n] = (rpe_index % 3) - 1;
    rpe_index = rpe_index / 3;
}

for (n = 0; n < sbfrm_size; n++)
{
    fcb_cng[n] = 0.0F;
}

for (n = 0; n < Np; n++)
{
    fcb_cng[rpe_phase + D*n] = gn[RPE_Configuration] * (float)(rpe_amps [n]);
}

```

nbits is set to 11 for RPE\_Configurations 0 and 1 and set to 12 for RPE\_configurations 2 and 3. D is the decimation factor, Np is the number of pulses per subframe and sbfrm\_size is the number of samples per subframe as defined in MPEG-4 CELP version 1. The normalising factor gn is defined in Table 3.106.

**Table 3.106 - Normalizing factor gn for RPE**

RPE_Configuration	gn [ ]
0	56756 / 32768
1	56756 / 32738
2	44869 / 32768
3	40132 / 32768

**3.5.9.3.3.7.5 Random index generator function**

A random index generator function setRandomBits() is defined for MPE and RPE as follows:

```

void setRandomBits (long *l, int n, short *seed)
{
    *l = 0xffff & Random(seed);
    if (n > 16)
    {
        *l |= (0xffff & Random(seed)) << 16;
    }

    if (n < 32)
    {
        *l &= ((unsigned long)1 << n) - 1;
    }
}

```

**3.5.9.3.3.7.6 Gain calculation**

Gains *gf* and *gr* are calculated from the smoothed RMS of the excitation, *norm\_sm* as follows:

$$gf = \alpha \cdot norm\_sm / \sqrt{\sum_{i=0}^{sbfm\_size-1} fcb\_cng(i)^2} / sbfrm\_size .$$

$$gr = [-\alpha \cdot A_3 + \sqrt{\alpha^2 \cdot A_3^2 - (\alpha^2 - 1)A_1A_2}] / A_2$$

where  $\alpha = 0.6$  and

$$A_1 = \sum_{i=0}^{sbfm\_size-1} gf^2 fcb\_cng[i]^2$$

$$A_2 = \sum_{i=0}^{sbfm\_size-1} excg[i]^2$$

$$A_3 = \sum_{i=0}^{sbfm\_size-1} gf \cdot fcb\_cng[i] \cdot excg[i] .$$

#### 3.5.9.3.3.8 LP Synthesis filter

The synthesis filter is identical to LP synthesis filter in MPEG-4 CELP described in subclause 3.5.8.

#### 3.5.9.3.3.9 Memory update

Since the encoder and decoder need to be kept synchronized during non-active periods, the excitation generation is performed on both encoder and decoder sides to update the corresponding buffers for the LP synthesis. The adaptive codebook is not used and is initialized with zero during non-active frames.

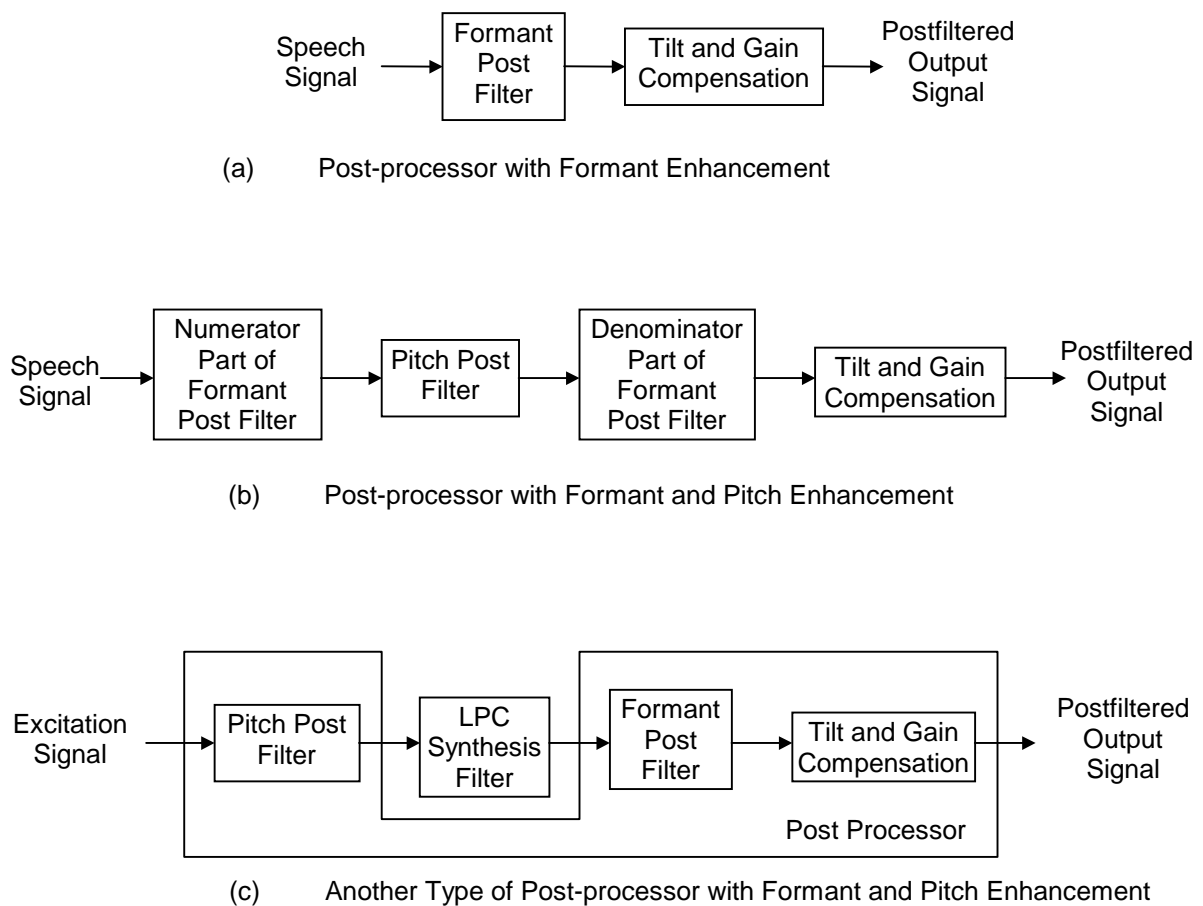
## Annex 3.A (informative)

### MPEG-4 CELP decoder tools

#### 3.A.1 CELP post-processor

##### 3.A.1.1 Tool description

The CELP post-processor enhances the reconstructed speech signal generated by the synthesis filter for one subframe. The post-filtering tools comprise a formant post-filter and a tilt compensation post-filter. Some examples on implementation of the post-processor are shown in Figure 3.A.1.



**Figure 3.A.1 - Examples of post-processor**

### 3.A.1.2 Definitions

#### Input

*synth\_signal[]*: This array contains the reconstructed speech signal.

*int\_Qlpc\_coefficients[]*: This array contains the LPC coefficients for each subframe.

*acb\_delay*: This field indicates the pitch delay, which is used for the pitch post-filter. If the pitch post-filter is not needed, *acb\_delay* must be set to a value less than 10.

*adaptive\_gain*: This field indicates the gain coefficient for the periodic component of the excitation signal. This gain coefficient is used for the pitch post-filter. If the pitch post-filter is not needed, *adaptive\_gain* must be set to a value less than 0.4.

#### Output

*PP\_synth\_signal[]*: This array contains the post-filtered (enhanced) speech signal. The dimension of this array is *sbfrm\_size*.

#### Configuration

*lpc\_order*: This field indicates the order of LPC that is used.

*sbfrm\_size*: This field indicates the number of samples in a subframe.

### 3.A.1.3 Decoding process

The decoding procedure is composed of post-filtering and adaptive gain control. The post-filter  $H_{fpt}(z)$  is the cascade of three filters: a formant post-filter  $H_f(z)$ , a optional pitch post-filter  $H_p(z)$ , and a tilt compensation filter  $H_t(z)$ :

$$H_{fpt}(z) = H_f(z) \cdot H_p(z) \cdot H_t(z)$$

The formant post-filter is given by

$$H_f(z) = \frac{\hat{A}(z/\gamma_n)}{\hat{A}(z/\gamma_d)} = \frac{1 - \sum_{i=1}^{lpc\_order} \gamma_n^i a_i z^{-i}}{1 - \sum_{i=1}^{lpc\_order} \gamma_d^i a_i' z^{-i}},$$

where  $\hat{A}(z)$  is the LPC inverse filter and the factors  $\gamma_n$  and  $\gamma_d$  control the degree of formant post-filtering.  $\gamma_n$  and  $\gamma_d$  are set to 0.65 and 0.75, respectively.

The pitch post-filter is given by

$$H_p(z) = \frac{1}{1 + \gamma_p g_a z^{-acb\_delay}},$$

$$g_a = \frac{\sum_{n=0}^{sbfm\_size-1} s_r(n)s_r(n - acb\_delay)}{\sum_{n=0}^{sbfm\_size-1} s_r^2(n - acb\_delay)},$$

where  $s_r(n)$  is the residual signal produced by filtering the input signal through the numerator part of the formant post-filter. The gain coefficient  $g_a$  is bounded by 1. The factor  $\gamma_p$  controls the degree of pitch post-filtering and has the value of 0.5. The pitch post-filter is applied only if the gain is more than 0.4 and the pitch delay is more than 10. In the 16 kHz sampling rate for the Mode I coder, the pitch post-filter is not applied.

The filter  $H_t(z)$  compensates the high-frequency tilt and is given by

$$H_t(z) = 1 - \gamma_t z^{-1},$$

where  $\gamma_t$  is a tilt factor of 0.3.

The synthesized signal ( $synth\_signal[]$ ) is filtered through the numerator part of the formant post-filter  $\hat{A}(z/\gamma_n)$  to produce the residual signal. In the Mode II coder, the residual signal is then filtered through the pitch post-filter  $H_p(z)$ , whereas in the Mode I coder no pitch post-filter is used. The pitch post-filtered residual is fed through the denominator part of the formant post-filter  $\hat{A}(z/\gamma_d)$ . The output signal of the formant and pitch post-filters is passed through the tilt compensation filter  $H_t(z)$  to generate the post-filtered synthesized signal that is not yet gain-compensated.

Adaptive gain control compensates gain differences between the synthesized signal  $s(n)$  and the post-filtered signal  $s_p(n)$ . The gain control factor  $G$  for the current subframe is computed by

$$G = \sqrt{\frac{\sum_{n=0}^{sbfm\_size-1} s^2(n)}{\sum_{n=0}^{sbfm\_size-1} s_p^2(n)}}.$$

The gain-scaled post-filtered signal  $s'_p(n)$  is given by

$$s'(n) = g(n)s_p(n), \quad 0 \leq n \leq sbfrm\_size - 1,$$

where  $g(n)$  is updated on a sample-by-sample basis and given by

$$g(n) = 0.95g(n-1) + 0.05G, \quad 0 \leq n \leq sbfrm\_size - 1.$$

The initial value of  $g(-1) = 0.0$  is used. Then for each new subframe,  $g(-1)$  is set equal to  $g(N-1)$  of the previous subframe.

## Annex 3.B (informative)

### MPEG-4 CELP encoder tools

#### 3.B.1 General Introduction to the MPEG-4 CELP encoder tool-set

This Annex provides a brief description of the functionality, parameter definition and the encoding processes of the tools supported by the MPEG-4 CELP core. The description of each tool comprises up to four parts: tool description, definitions, encoding process and tables.

The following encoder tools are supported:

- CELP preprocessing
- CELP LPC analysis
- CELP LPC quantizer and interpolator
  - Vector Quantizer
  - Bandwidth Scalable encoder
- CELP LPC analysis filter
- CELP weighting module
- CELP excitation analysis
  - regular pulse excitation
  - multi-pulse excitation
- CELP bitstream multiplexer

The encoding is performed on frame basis and each frame is divided in subframes. The CELP excitation analysis tool is used every subframe, while the other tools are used every frame.

#### 3.B.2 Helping variables

For each encoder tool a description of the variables it uses is given. In this subclause the most commonly used variables are provided, which are shared by multiple tools.

*frame\_size*: This field indicates the number of samples in a frame. The decoder outputs a frame with *frame\_size* samples.

*nrof\_subframes*: A frame is built up of a number of subframes. The number of subframes is specified in this field.

*sbfm\_size*: A subframe consists of a number of samples, which is indicated by this field. The number of samples in a frame must always be equal to the sum of the number of samples in the subframes. So, the following relation must always hold

$$frame\_size = nrof\_subframes * sbfm\_size$$

These three parameters depend on the sampling rate and the bitrate settings as tabulated in Table 3.73, for the Mode I coder and Table 3.74, for the Mode II coder.

*lpc\_order*: This field indicates the number of coefficients used for Linear Prediction. By default, the value of this field is 20 for a sampling rate of 16 kHz and 10 for 8 kHz.



*num\_lpc\_indices*. This parameter specifies the number of indices containing LPC information, that must be written to the bitstream. This is not equal to the LPC-order. The *num\_lpc\_indices* is 5 in the 8 kHz mode and an additional 6 for the band scalable layer.

*n\_lpc\_analysis*: This field indicates how often LPC analysis is performed per frame. It is possible to perform several LPC analyses per frame with a varying window size and offset. For 16 kHz sampling rate, the value of this field is 1, indicating that LPC analysis is only performed once. For 8 kHz sampling rate, the value of this field is determined by the ratio *sbfrm\_size*/80.

*window\_offsets*[]): This array contains the offsets of the LPC analysis windows and its dimension is *n\_lpc\_analysis*.

*window\_sizes*[]): This array contains the window sizes for the LPC analysis. Since the LPC analysis is performed *n\_lpc\_analysis* times, the dimension of this array is *n\_lpc\_analysis*.

**Table 3.B.1 - Window size and offset for the 16 kHz mode I coder**

RPE_Configuration	Window_sizes[] (samples)	Window_offsets[] (samples)
0	400	280
1	320	160
2	400	280
3	400	280
4...7	Reserved	

**Table 3.B.2 - Window size and offset for the 8 kHz mode II coder**

MPE_Configuration	Window_sizes[] (samples)	Window_offsets[] (samples)
0,1,2	200	0, 80, 160, 240
3,4,5	200	0, 80, 160
6 ... 12	200	0, 80
13 ... 21	200	0, 80
22 ... 26	200	0

**Table 3.B.3 - Window size and offset for the 16 kHz mode II coder**

MPE_Configuration	Window_sizes[] (samples)	Window_offsets[] (samples)
0 ... 6, 8 ... 15	320	0, 80, 160, 240
16 ... 22, 24 ... 31	320	0, 80

*windows*[]): This array contains the window for each analysis, so the length of this array is the sum of *window\_sizes* times *n\_lpc\_analysis*. For the Mode I coder a squared Hamming window is used:

```

for (x = 0; x < window_sizes[i]; x++)
{
    window[i][x] = (0.54 - 0.46 * cos(2 * pi * (x/window_sizes[i])));
    window[i][x] = window[i][x] * window[i][x];
}

```

For the Mode II coder, a hybrid window is used. The window consists of two parts; the half of a Hamming window and a quarter of the cosine function, given by:

```

for (n = 0; n < nlb+len_lpcana/2; n++)
{
    Hw(n) = 0.54 - 0.46 * cos( 2 * PI * n / (2*(nlb+len_lpcana/2) - 1));
}

for (n = nlb+len_lpcana/2; n < window_sizes[i]; n++ )
{
    Hw(n) = cos ( 2 * PI * (n - (nlb+len_lpcana/2)) / (4*(nla+len_lpcana/2)-1));
}

```

*nlb* samples of the past analysis frame, *len\_lpcana* samples of the present analysis frame, and *nla* samples of the future frame form one block for windowing.

**Table 3.B.4 - Window parameters for the mode II coder**

Sampling rate (kHz)	<i>nla</i>	<i>nlb</i>	<i>Len_lpc_ana</i>
8	40	80	80
16	80	160	80

*gamma\_be[]*: This array is of size *lpc\_order* and contains the weights for applying bandwidth expansion on the LPC coefficients. This information is used only for the mode I coder.

```

gamma_be[0] = GAMMA;
for (x = 1; x < lpc_order; x++)
{
    gamma_be[x] = GAMMA * gamma_be[x-1];
}

```

The value of GAMMA is 0.9883 for the RPE tool.

*n\_lag\_candidates*: This field contains the number of pitch candidates. This information is used only for 16 kHz sampling rate, the value of this field is 15.

*max\_pitch\_frequency*: This field contains the maximum frequency of the pitch (lag). For 16 kHz sampling rate, this field has the value 0.025, since the minimum lag is 40. For 8 kHz sampling rate, this field has the value 0.05882, since the minimum lag is 17.

*min\_pitch\_frequency*: This field contains the minimum frequency of the pitch (lag). The value of this field for 16 kHz sampling rate is 3.3898E-3, since the maximum lag is 295. For 8 kHz sampling rate, this field has the value 6.944E-3, since the maximum lag is 144.

### 3.B.3 Bitstream elements for the MPEG-4 CELP encoder tool-set

See subclause 3.5.4.

### 3.B.4 CELP preprocessing

#### 3.B.4.1 Tool description

The CELP preprocessing tool produces a DC free speech signal.

#### 3.B.4.2 Definitions

##### Input

*s*[]): This is an array of dimension *frame\_size* and contains input speech samples.

##### Output

*pp\_s* []: This is an array of length *frame\_size* and contains the DC free speech samples.

##### Input/Output

*prev\_x*, *prev\_y*: memory of the preprocessing filter

##### Configuration

*frame\_size*: This field indicates the number of samples in the input signal.

#### 3.B.4.3 Encoding process

This block removes the DC element from the input signal, *s*[*n*]. It is first order recursive filter of the form:

$$H_{pre}(z) = \frac{1 - z^{-1}}{1 - 0.99 \cdot z^{-1}}$$

An implementation of this filter is given below:

```
for (n = 0; n < frame_size; n++)
{
    pp_s[n] = s[n] - prev_x + 0.99 * prev_y;
    prev_x = s[n];
    prev_y = pp_s[n];
}
```

The input/output filter states *prev\_x* and *prev\_y* are initialized to zero.

### 3.B.5 CELP LPC analysis

#### 3.B.5.1 Tool description

The CELP LPC Analysis tool estimates the short-term spectrum. The LPC analysis is performed on the preprocessed speech signal, *pp\_s*[]. The order of linear prediction is specified by the parameter *lpc\_order*. A window, with size specified in *window\_size*[], is used to weight the preprocessed speech. The parameter, *window\_offset*[], is provided to specify the offset for each window.

### 3.B.5.2 Definitions

#### Input

*PP\_InputSignal[]*: This array contains the preprocessed speech signal. This dimension is *frame\_size*.

#### Output

*lpc\_coefficients[]*: This array contains the computed LPC coefficients and has size *lpc\_order*.

*first\_order\_lpc\_par*: This output field contains the LPC coefficient for 1st-order fit. This parameter is used for the preselection in the adaptive codebook search.

#### Configuration

*frame\_size*: This field denotes the number of samples in frame.

*window\_offset[]*: This array contains the offset of the window.

*window\_size[]*: This array contains the LPC analysis-window size.

*windows[]*: This array contains the windows used to weight the speech signal.

*gamma\_be[]*: This array contains the gamma coefficients that are used for bandwidth expansion of the LPC coefficients.

*lpc\_order*: This field indicates the order of LPC.

*n\_lpc\_analysis*: This field denotes the number of LPC analysis.

### 3.B.5.3 Encoding process

The Linear Prediction analysis is performed *n\_lpc\_analysis* times, each with a different window size and offset as specified in the array's *window\_size* and *window\_offset*. Each time the input signal *PP\_InputSignal* is weighted, *sw[n]*.

By means of the weighted signal the autocorrelation coefficients are derived using:

$$acf[k] = \sum_{n=0}^{frame\_size-k-1} sw[n] \cdot sw[n+k], 0 \leq k \leq lpc\_order$$

There are *lpc\_order+1* autocorrelation coefficients. For the Mode II coder, a bandwidth expansion and a white-noise correction is applied by modifying the autocorrelation coefficients as follows:

```
for (k = 0; k < lpc_order; k++)
{
    acf[k] *= lag_win[k]
}
```

where *lag\_win[]* is the coefficients for the bandwidth expansion and is shown in Annex 3.D.

The LPC coefficients are computed using the conventional Levinson-Durbin recursion. The first LPC coefficient is assigned to `first_order_lpc_par`. For Mode I coder, bandwidth expansion is applied on the LPC coefficients using array `gamma_be`. For each LPC analysis, the calculated `lpc_coefficients` are stacked, resulting in  $n\_lpc\_analysis * lpc\_order$  coefficients.

### 3.B.6 CELP LPC quantizer and interpolator

The LPC coefficients are quantized by using one of three quantizers, Narrowband LSP Quantization tool, Wideband LSP Quantization tool or Bandwidth Scalable LSP Quantization tool.

#### 3.B.6.1 Narrowband LSP quantization tool

##### 3.B.6.1.1 Tool description

The Narrowband LSP Quantization tool quantizes the LPC coefficients as LSP parameters using a two-stage and split vector quantization technique.

##### 3.B.6.1.2 Definitions

###### Input

`lpc_coefficients[]`: This is an array of dimension `lpc_order` and contains the current unquantized LPC coefficients.

###### Output

`int_Qlpc_coefficients[]`: This is an array of length  $nrof\_subframes * lpc\_order$  and contains the interpolated and quantized LPC coefficients for each subframe. The LPC coefficients for each subframe are stacked one after the other, resulting in an  $nrof\_subframes * lpc\_order$  array.

`lpc_indices[]`: This is an array of dimension `num_lpc_indices` and contains the packed lpc indices that are written to the bitstream.

###### Configuration

`lpc_order`: This field contains the order of LPC.

`num_lpc_indices`: This field indicates the number of packed LPC codes.

`n_lpc_analysis`: This field contains the number of LPC parameters.

`nrof_subframes`: This field contains the number of subframes.

##### 3.B.6.1.3 Encoding process

The LPCs are converted into LSPs and quantized. As described in the decoding process, there are two methods for quantizing the LSPs; a two-stage VQ without interframe prediction, and a combination of the VQ and the interframe predictive VQ. At the encoding process, both methods are tried to quantize the LSPs and which method should be applied is determined by comparing the quantization error. The quantization error is calculated as a weighted Euclidean distance. The weighting coefficients ( $w[l]$ ) are,

$$w[i] = \begin{cases} \frac{1}{lsp[0]} + \frac{1}{lsp[1] - lsp[0]} & (i = 0) \\ \frac{1}{lsp[i] - lsp[i-1]} + \frac{1}{lsp[i+1] - lsp[i]} & (0 < i < Np - 1) \\ \frac{1}{lsp[Np-1] - lsp[Np-2]} + \frac{1}{1.0 - lsp[Np-1]} & (i = Np - 1) \end{cases}$$

where  $Np$  is the LP analysis order  $lpc\_order$  and  $lsp[]$  is the LSPs converted from the LPCs.

The first stage quantizer is the same for each quantization method. The LSPs are quantized by using a two-split vector quantizer and corresponding indices are stored in **lpc\_indices[0]** and **lpc\_indices[1]**. In order to carry out delayed decision, two indices, namely the best and the second best, are stored as candidates for the second stage. The quantization error in the first stage  $err1[]$  is given by:

$$err1[n] = \sum_{i=0}^{\dim-1} \left\{ (lsp[sp+i] - lsp\_tbl[n][m][i])^2 \cdot w[sp+i] \right\} \quad n = 0,1$$

where  $n$  is the split vector number,  $m$  is the index of the candidate split vector,  $sp$  is the starting LSP order of the  $n$ -th split vector and  $\dim$  is the dimension of the  $n$ -th split vector. ( $lsp\_tbl[][][]$  is shown in Annex 3.C)

**Table 3.B.5 - Starting order and dimension of the first stage LSP vector**

Split vector number: n	Starting LSP order: sp	Dimension of the vector: dim
0	0	5
1	5	5

In the second stage, the above-mentioned two quantization methods, which are also two-split vector quantizers, are applied respectively. Total quantization errors in the second stage are calculated for all combinations of the first stage candidates and the second stage candidates and the one that has the minimum error is selected. As a result, indices of the first stage are determined and corresponding indices and signs for the second stage are stored in **lpc\_indices[2]** and **lpc\_indices[3]**. The flag that indicates the selected quantization method is also stored in **lpc\_indices[4]**. The quantization error in the second stage  $err2\_total$  is given by:

VQ without interframe prediction:

$$err2\_total = err2[0] + err2[1]$$

$$err2[n] = \sum_{i=0}^{\dim-1} \left\{ (lsp\_res[sp+i] - sign[n] \cdot d\_tbl[n][m][i])^2 \cdot w[sp+i] \right\} \quad n = 0,1$$

$$lsp\_res[sp+i] = lsp[sp+i] - lsp\_first[sp+i]$$

where  $lsp\_first[]$  is the quantized LSP vector of the first stage,  $n$  is the split vector number,  $m$  is the index of the candidate split vector,  $sp$  is the starting LSP order of the  $n$ -th split vector and  $\dim$  is the dimension of the  $n$ -th split vector. ( $d\_tbl[][][]$  is shown in Annex 3.C)

VQ with interframe prediction:

$$err2\_total = err2[0] + err2[1]$$

$$err2[n] = \sum_{i=0}^{dim-1} \left\{ \left( lsp\_pres[sp+i] - sign[n] \cdot pd\_tbl[n][m][i] \right)^2 \cdot w[sp+i] \right\} \quad n = 0,1$$

$$lsp\_pres[sp+i] = lsp[sp+i] - \left\{ (1 - ratio\_predict) \cdot lsp\_first[sp+i] + ratio\_predict \cdot lsp\_previous[sp+i] \right\}$$

where *lsp\_first[]* is the quantized LSP vector of the first stage, *n* is the split vector number, *m* is the index of the candidate split vector, *sp* is the starting LSP order of the *n*-th split vector, *dim* is the dimension of the *n*-th split vector and *ratio\_predict*=0.5. (*pd\_tbl[][][]*) is shown in Annex 3.C)

**Table 3.B.6 - Starting order and dimension of the second stage LSP vector**

Split vector number: n	Starting LSP order: sp	Dimension of the vector: dim
0	0	5
1	5	5

The quantized LSPs *lsp\_current[]* are stabilized in order to ensure stability of the LPC synthesis filter, which is derived from the quantized LSPs. The quantized LSPs are arranged in order, having a distance of at least *min\_gap* between adjacent coefficients.

```

for (i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] < min_gap)
    {
        lsp_current[i] = min_gap;
    }
}

for (i = 0; i < lpc_order - 1; i++)
{
    if (lsp_current[i+1] - lsp_current[i] < min_gap)
    {
        lsp_current[i+1] = lsp_current[i]+min_gap;
    }
}

for(i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] > 1-min_gap)
    {
        lsp_current[i] = 1-min_gap;
    }
}

```

```

for (i = lpc_order - 1; i > 0; i--)
{
    if (lsp_current[i]-lsp_current[i-1] < min_gap)
    {
        lsp_current[i-1] = lsp_current[i]-min_gap;
    }
}

```

where  $\text{min\_gap} = 2.0/256.0$

After the quantization process, the quantized LSPs are interpolated linearly at each subframe.

```

for (n = 0; n < nrof_subframes; n++)
{
    ratio_sub=(n+1)/nrof_subframes
    for(i = 0; i < lpc_order; i++)
    {
        lsp_subframe[n][i]=((1-ratio_sub)*lsp_previous[i] + ratio_sub*lsp_current[i])
    }
}

```

The interpolated LSPs are converted to the LPCs using the auxiliary function *Convert2lpc()*.

```

for(n = 0; n < nrof_subframes; n++)
{
    Convert2lpc(lpc_order, lsp_subframe[n], int_Qlpc_coefficients + n*lpc_order);
}

```

After calculation of the LPC coefficients, the current LSPs must be stored in memory, since they are used for interpolation at the next frame.

```

for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = lsp_current[i];
}

```

It should be noted that the stored LSPs *lsp\_previous[]* must be initialized as described below when the entire encoder is initialized.

```

for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = (i+1) / (lpc_order+1);
}

```

### 3.B.6.2 Wideband LSP Quantization Tool

#### 3.B.6.2.1 Tool description

The Wideband LSP Quantization tool quantizes the LPC coefficients as LSP parameters using a two-stage and split vector quantization technique.



**3.B.6.2.2 Definitions**

Input

*lpc\_coefficients*[]): This is an array of dimension *lpc\_order* and contains the current unquantized LPC coefficients.

Output

*int\_Qlpc\_coefficients*[]): This is an array of length *nrof\_subframes* \* *lpc\_order* and contains the interpolated and quantized LPC coefficients for each subframe. The LPC coefficients for each subframe are stacked one after the other, resulting in a *nrof\_subframes* \* *lpc\_order* array.

**lpc\_indices**[]): This is an array of dimension *num\_lpc\_indices* and contains the packed lpc indices that are written to the bitstream.

Configuration

*lpc\_order*: This field contains the order of LPC.

*num\_lpc\_indices*: This field indicates the number of packed LPC codes.

*n\_lpc\_analysis*: This field contains the number of LPC parameters.

*nrof\_subframes*: This field contains the number of subframes.

**3.B.6.2.3 Encoding process**

The quantization scheme is based on the Narrowband LSP Quantization. The quantizer consists of two quantization blocks connected in parallel, each of which is identical to the Narrowband LSP Quantization tool. The input LSPs are divided into two parts, namely the lower part and the upper part, then the divided LSPs are input to the quantization blocks respectively.

First, the lower part is quantized in the same manner as the Narrowband LSP Quantization. The starting order and the dimension of the LSP vectors are described in Table 3.B.7 and

Table 3.B.8. The quantized LSPs are stored in the array *lsp\_current\_lower*[].

**Table 3.B.7 - Starting order and dimension of the first stage LSP vector**

Split vector number: n	Starting order of the lower LSP order: sp	Dimension of the vector: dim
0	0	5
1	5	5

**Table 3.B.8 - Starting order and dimension of the second stage LSP vector**

Split vector number: n	Starting order of the lower LSP order: sp	Dimension of the vector: dim
0	0	5
1	5	5

Next, the upper part is quantized in the same manner. The starting order and the dimension of the LSP vectors are described in Table 3.B.9 and

Table 3.B.10. The quantized LSPs are stored in the array *lsp\_current\_upper[]*. In the quantization of the upper part, indices of the first stage are stored in **lpc\_indices[5]** and **lpc\_indices[6]**, and indices and signs for the second stage are stored in **lpc\_indices[7]** and **lpc\_indices[8]**. The flag that indicates the selected quantization method is also stored in **lpc\_indices[9]**.

**Table 3.B.9 - Starting order and dimension of the first stage LSP vector**

Split vector number: n	Starting order of the upper LSP order: sp	Dimension of the vector: dim
0	0	5
1	5	5

**Table 3.B.10 - Starting order and dimension of the second stage LSP vector**

Split vector number: n	Starting order of the upper LSP order: sp	Dimension of the vector: dim
0	0	5
1	5	5

Finally, the decoded LSPs *lsp\_current\_lower[]* and *lsp\_current\_upper[]* are combined and stored in the array *lsp\_current[]*.

```

for (i = 0; i < lpc_order/2; i++)
{
    lsp_current[i] = lsp_current_lower[i];
}
for (i = 0; i < lpc_order/2; i++)
{
    lsp_current[lpc_order/2+i] = lsp_current_upper[i];
}

```

The quantized LSPs *lsp\_current[]* are stabilized in order to ensure stability of the LPC synthesis filter, which is derived from the quantized LSPs. The quantized LSPs are arranged in ascending order, having a distance of at least *min\_gap* between adjacent coefficients.

```

for (i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] < min_gap)
    {
        lsp_current[i] = min_gap;
    }
}

for (i = 0; i < lpc_order - 1; i++)
{
    if (lsp_current[i+1] - lsp_current[i] < min_gap)
    {
        lsp_current[i+1] = lsp_current[i]+min_gap;
    }
}

```

```

for(i = 0; i < lpc_order; i++)
{
    if (lsp_current[i] > 1-min_gap)
    {
        lsp_current[i] = 1-min_gap;
    }
}

for (i = lpc_order - 1; i > 0; i--)
{
    if (lsp_current[i]-lsp_current[i-1] < min_gap)
    {
        lsp_current[i-1] = lsp_current[i]-min_gap;
    }
}

```

where  $lpc\_order = 20$  and  $min\_gap = 1.0/256.0$

After the quantization process, the quantized LSPs are interpolated linearly at each subframe.

```

for (n = 0; n < nrof_subframes; n++)
{
    ratio_sub=(n+1)/nrof_subframes
    for(i = 0; i < lpc_order; i++)
    {
        lsp_subframe[n][i]=((1-ratio_sub)*lsp_previous[i] + ratio_sub*lsp_current[i])
    }
}

```

The interpolated LSPs are converted to the LPCs using the auxiliary function *Convert2lpc()*.

```

for(n = 0; n < nrof_subframes; n++)
{
    Convert2lpc(lpc_order, lsp_subframe[n], int_Qlpc_coefficients + n*lpc_order);
}

```

After calculation of the LPC coefficients, the current LSPs must be stored in memory, since they are used for interpolation at the next frame.

```

for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = lsp_current[i];
}

```

It should be noted that the stored LSPs *lsp\_previous[]* must be initialized as described below when the entire encoder is initialized.

```

for (i = 0; i < lpc_order; i++)
{
    lsp_previous[i] = (i+1) / (lpc_order+1);
}

```

### 3.B.6.3 Bandwidth scalable LSP quantization tool

#### 3.B.6.3.1 Tool description

The Bandwidth Scalable LSP Quantization tool quantizes the input 16 kHz sampling rate LSPs using vector quantization scheme with intraframe and interframe prediction.

#### 3.B.6.3.2 Definitions

##### Input

*lpc\_coefficients[]*: This is an array of dimension *lpc\_order* and contains the current unquantized LPC coefficients.

*lsp\_current[]*: This array contains the decoded LSP parameters, which are normalized in the range of zero to  $\pi$ , at the Narrowband LSP Quantization tool. These parameters are obtained as intermediate parameters in the Narrowband LSP Quantization process and forwarded to the Bandwidth Scalable LSP Decoding tool.

##### Output

*int\_Qlpc\_coefficients[]*: This is an array of length *nrof\_subframes* \* *lpc\_order* and contains the interpolated and quantized LPC coefficients for each subframe. The LPC coefficients for each subframe are stacked one after the other, resulting in an *nrof\_subframes* \* *lpc\_order* array.

**lpc\_indices[]**: This is an array of dimension *num\_lpc\_indices* and contains the packed lpc indices that are written to the bitstream.

##### Configuration

*lpc\_order*: This field contains the order of LPC.

*num\_lpc\_indices*: This field indicates the number of packed LPC codes.

*n\_lpc\_analysis*: This field contains the number of LPC parameters.

*nrof\_subframes*: This field contains the number of subframes.

*nrof\_subframes\_bws*: This parameter, which is a help variable, represents the number of subframes in the bandwidth scalable layer.

#### 3.B.6.3.3 Encoding process

The input 16 kHz sampling rate LSPs (*input\_lsp[]*) are vector-quantized with intraframe and interframe prediction approaches. The intraframe prediction module produces estimated LSPs by converting the quantized LSPs obtained in the 8 kHz sampling rate CELP coder. Furthermore, an interframe moving average predictive VQ is also employed for more accurate prediction. The prediction residual LSPs (*err\_lsp[]*) are computed as follows:

```
for (i = 0; i < 20; i++)
{
    err_lsp[i] = (input_lsp[i] - pred_lsp[i]) / bws_ma_prdct[0][i]
}
```

```

for (n = 1; n <= 2; n++)
{
    for (i = 0; i < 20; i++)
    {
        pred_lsp[i] += bws_ma_prdct[n][i]*lsp_bws_buf[n][i]
    }
}

for (i = 0; i < 10; i++)
{
    pred_lsp[i] += bws_nw_prdct[i]*lsp_current[i]
}

```

where *pred\_lsp[]* contains the predicted LSPs. *bws\_ma\_prdct[][]* and *bws\_nw\_prdct[]* are prediction coefficients for moving average interframe prediction and intraframe prediction, respectively. *lsp\_bws\_buf[][]* is a buffer containing LSP prediction residual at the previous two frames.

Then, *err\_lsp[]* are vector-quantized using a two-stage and split vector quantization scheme and corresponding indices are stored in **lpc\_indices[5]**, ... , **lpc\_indices[10]**. The buffer *lsp\_bws\_buf[][]* is shifted for the next frame operation as follows:

```

for (i = 0; i < 20; i++)
{
    lsp_bws_buf[0][i] = err_lsp[i]
}

for (n = 2; n > 0; n--)
{
    for (i = 0; i < 20; i++)
    {
        lsp_bws_buf[n][i] = lsp_bws_buf[n-1][i];
    }
}

```

After the quantization process, the quantized LSPs (*lsp\_bws\_current[]*) are interpolated linearly at each subframe.

```

for (n = 0; n < nrof_subframes_bws; n++)
{
    ratio_sub = (n+1)/nrof_subframes_bws
    for(i = 0; i < 2*lpc_order; i++)
    {
        lsp_bws_subframe[n][i]=((1-ratio_sub)*lsp_bws_previous[i]
                                + ratio_sub*lsp_bws_current[i])
    }
}

for (i = 0; i < 2*lpc_order; i++)
{
    lsp_bws_previous[i] = lsp_bws_subframe[nrof_subframes_bws-1][i]
}

```

The interpolated LSPs are converted to the LPC coefficients at each subframes.

```

for (n = 0; n < nrof_subframes_bws; n++)
{
    Convert2lpc (lpc_order_bws, lsp_bws_subframe[n],
                &int_Qlpc_coefficients[n*lpc_order_bws]);
}

```

### 3.B.6.4 Fine rate control in the LSP quantization tool

#### 3.B.6.4.1 Tool description

Fine Rate Control (FRC) is available with the LSP Quantization tool. If the FRC is used, the array *lpc\_coefficients[]* contains the LPC coefficients of the next frame which is the frame following the frame currently processed. The decision is made whether or not the LSPs of the currently processed frame are transmitted to the decoder. According to the decision, the **interpolation\_flag** and **LPC\_present** are set to 1 or 0.

#### 3.B.6.4.2 Definitions

##### Input

*lpc\_coefficients[]*: This is an array of dimension *lpc\_order* and contains the current unquantized LPC coefficients.

##### Output

*int\_Qlpc\_coefficients[]*: This is an array of length *nrof\_subframes* \* *lpc\_order* and contains the interpolated and quantized LPC coefficients for each subframe. The LPC coefficients for each subframe are stacked one after the other, resulting in an *nrof\_subframes* \* *lpc\_order* array.

**lpc\_indices[]**: This is an array of dimension *num\_lpc\_indices* and contains the packed lpc indices that are written to the bitstream.

**interpolation\_flag**: This field indicates if interpolation on the LPC coefficients between frames is done. If the interpolation flag is set, the current LPC coefficients are computed from the previous and next LPC coefficients.

**LPC\_Present**: This flag indicates whether the current frame is complete or incomplete.

#### 3.B.6.4.3 Encoding process

The decision whether or not the LPC coefficients of the speech frame under analysis are transmitted to the decoder depends on the amount of change *d*, between the spectrum of the current frame and the spectrum of the adjacent frames. If *d* is greater than a particular threshold then the coefficients are transmitted to the decoder. This threshold is further made dependent on the desired bitrate setting as follows: The threshold is raised if the actual bitrate is higher than the desired bitrate setting and is lowered otherwise.

### 3.B.7 CELP LPC analysis filter

#### 3.B.7.1 Tool description

The CELP LPC Analysis filter tool feeds the input signals through a filter with LPC coefficients and returns the residual signal.

### 3.B.7.2 Definitions

#### Input

*PP\_InputSignal[]*: This array has dimension *sbfrm\_size* and contains the input signal.

*int\_Qlpc\_coefficients[]*: This array has dimension *lpc\_order* and contains the LPC coefficients.

#### Output

*lpc\_residual[]*: This array has dimension *sbfrm\_size* and contains the LPC filtered residual signal.

#### Configuration

*lpc\_order*: This field indicates the order of LPC.

*sbfrm\_size*: This field indicates the number of samples in one subframe.

### 3.B.7.3 Encoding process

The input signal is filtered using the filter coefficients.

```
for (k = 0; k < sbfrm_size; k++)
{
    tmp = PP_InputSignal[k];
    for (j = 0; j < lpc_order; j++)
    {
        tmp = tmp - int_Qlpc_coefficients[j] * Filter_States[j];
    }
    lpc_residual[k] = tmp;
    update_Filter_States;
}
```

The initial filter states are set to zero.

## 3.B.8 CELP weighting module

### 3.B.8.1 Tool description

The CELP weighting module computes the weights to be applied on the LPC coefficients.

### 3.B.8.2 Definitions

#### Input

*lpc\_coefficients[]*: This is an array of dimension *lpc\_order*, containing the LPC Coefficients.

*gamma\_num*: This field contains the weighting factor of the numerator.

*gamma\_den*: This field contains the weighting factor of the denominator.

#### Output

*Wnum\_coeff[]*: This is an array of dimension *Wnum\_order*, containing the weighted numerator coefficients.

*Wden\_coeff[]*: This array has dimension *Wden\_order* and contains the weighted denominator coefficients.

### Configuration

*lpc\_order*: This field contains the order of the LPC.

### 3.B.8.3 Encoding process

The weighted coefficients of the numerator are computed by:

```
for (k = 0; k < Wnum_order; k++)
{
    Wnum_coeff[k] = lpc_coefficients[k] * gamma_numk+1
}
```

The weighted coefficients of the denominator are computed by:

```
for (k=0; k < Wden_order; k++)
{
    Wden_coeff[k] = lpc_coefficients[k] * gamma_denk+1
}
```

### 3.B.9 CELP excitation analysis

The CELP excitation analysis computes the shape and gain vectors as well as the decoded synthesized speech signal. For excitation analysis modules, Regular Pulse Excitation (RPE) and Multi Pulse Excitation (MPE) are defined.

#### 3.B.9.1 Regular pulse excitation

##### 3.B.9.1.1 Tool description

For Regular Pulse Excitation, the output of the excitation analysis are **shape\_delay[]**, **shape\_index[]** and **gain\_indices[]**. The shape and gain indices are generated every subframe. The vector **shape\_delay[]** contains the adaptive codebook lag for each subframe, while the vector **shape\_index[]** contains the fixed codebook index. The vector **gain\_indices[0][]** contains the adaptive codebook gain for every subframe. The fixed codebook gain for every subframe is stored in vector **gain\_indices[1][]**.

##### 3.B.9.1.2 Definitions

#### Input

*PP\_inputSignal[]*: This array contains the preprocessed input signal and is of dimension *sfrm\_size*.

*lpc\_residual[]*: This contains the lpc residual signal.

*int\_Qlpc\_coefficients[]*: This array contains the interpolated and quantized LPC coefficients.

*Wnum\_coeff[]*: This array contains the weighting filter coefficients of the numerator.



*Wden\_coeff*[]): This array contains the weighting filter coefficients of the denominator.

*first\_order\_lpc\_par*: This field indicates the first LPC coefficient.

*lag\_candidates*[]): This array contains the lag candidates.

**signal\_mode**: This field contains the voiced/unvoiced flag .

**rms\_index**: This field defines the index for the signal power.

### Output

**shape\_delay**[]): This array is of dimension *nrof\_subframes*. It contains the codebook lag for the adaptive codebooks.

**shape\_index**[]): This array is of dimension *nrof\_subframes*. It contains the shape indices for the fixed codebooks.

**gain\_indices**[][]]: This array is of dimension 2 \* *nrof\_subframes*. It contains the gain indices for the adaptive and fixed codebooks.

*decoded\_excitation*[]): This array is of length *sbfrm\_size* and contains the synthesized speech signal.

### Configuration

*n\_lag\_candidates*: This field indicates the number of lag candidates.

*frame\_size*: This field indicates the number of samples in one frame.

*sbfrm\_size*: This field indicates the number of samples in one subframe.

*nrof\_subframes*: This field indicates the number of subframes in one frame.

*lpc\_order*: This field indicates the order of LPC.

### 3.B.9.1.3 Encoding process

All blocks are performed on a once per subframe basis. For each subframe the following steps are performed:

- perceptual weighting
- adaptive codebook search preselection
- adaptive codebook search
- fixed codebook search preselection
- fixed codebook search
- simulation of decoder

Next, each step will be described in detail. For convenience, *aq*[] is used instead of *int\_Qlpc\_coefficients*[], so the *aq*[] contains the quantized LPC coefficients for the subframe under consideration.

Perceptual weighting is performed by filtering the input signal  $PP\_InputSignal[]$  by the following filter:

$$W(z) = \frac{A(z)}{A(z/\gamma)} = \frac{1 - \sum_{k=0}^{lpc\_order-1} aq[k] \cdot z^{-k-1}}{1 - \sum_{k=0}^{lpc\_order-1} aq[k] \cdot \gamma^{k+1} \cdot z^{-k-1}}$$

with  $\gamma = 0.8$ . The resulting signal is called  $ws[n]$ .

The zero-input response  $z[n]$ , is determined by computing the response of  $S(z)$  to a zero-valued input signal.

$$S(z) = \frac{1}{A(z/\gamma)} = \frac{1}{1 - \sum_{k=0}^{lpc\_order-1} aq[k] \cdot \gamma^{k+1} \cdot z^{-k-1}}$$

where  $\gamma = 0.8$ .

The weighted target signal  $t[n]$  is obtained by subtracting  $z[n]$  from  $ws[n]$ :

```
for (n = 0; n < Nm; n++)
{
    t[n] = ws[n] - z[n];
}
```

The impulse response  $h[n]$  is calculated as follows:

```
for (k = 0; k < lpc_order; k++)
{
    tmp_states[k] = 0;
}
tmp = 1.0;
for (n = 0; n < sbfrm_size; n++)
{
    g = 0.8;
    for (k = 0; k < lpc_order; k++)
    {
        tmp = tmp + aq[k][s] * g * tmp_states[k];
        g = 0.8 * g;
    }
    h[n] = tmp;
    for (k = lpc_order-1; k > 0; k--)
    {
        tmp_states[k] = tmp_states[k-1];
    }
    tmp_states[0] = tmp;
}
```

After the above mentioned computations, preselection on the adaptive codebook is performed. The adaptive codebook contains 256 codebook sequences of which 5 are preselected. Preselection is achieved by evaluating the term (for  $0 \leq l < Lm$  and  $0 \leq i < nrof\_subframes$ ):

$$rap[i \cdot Lm + l] = \frac{\left( \sum_{n=0}^{sbfrm\_size-1} ca[Lmin + i \cdot sbfrm\_size + l \cdot 3 - n - 1] \cdot ta[n] \right)^2}{Ea[i \cdot Lm + l]}$$

where:

$$Lm = 1 + \left\lfloor \frac{sbfrm\_size - 1}{3} \right\rfloor$$

and  $ca[Lmin + i \cdot sbfrm\_size + 3 \cdot l - n - 1]$  and  $ta[n]$  representing the codebook sequence at delay  $(Lmin + i \cdot sbfrm\_size + 3 \cdot l)$  and the 'backward-filtered' target signal  $t[n]$ , respectively.

Backward filtering comprises the time-reversal of  $t[n]$ , filtering by  $S(z)$  and time-reversal again.  $Lmin$  is the minimum lag in samples. The value of  $Lmin$  is 40.

$Ea[i \cdot Lm + l]$  is the energy of that codebook sequence filtered by a reduced complexity synthesis filter  $S_p(z)$  which is a first-order estimation of the LPC synthesis filter  $S(z)$ :

$$S_p(z) = \frac{1}{A_p(z/\gamma)} = \frac{1}{1 - a \cdot \gamma \cdot z^{-1}}$$

The first LPC coefficient of a frame is taken for  $a$ . Depending on the number of subframe, which is under consideration, the first LPC coefficient of the current or the previous is taken using the following:

```

if (subframe_number < nr_subframes/2)
{
    a = prev_a;
}
else
{
    a = cur_a;
}

```

After evaluation of  $rap[]$  the 5 maximum sequences are selected together with their 2 neighbor sequences resulting in 15 candidates on which full search is applied. The indices of the preselected sequences and their neighbors are stored in  $ia[r]$  ( $0 \leq r < 15$ ).

The adaptive codebook search minimizes the mean-square weighted error between the original and reconstructed speech. This is achieved by searching for the index  $r$  maximizing the term

$$ra[r] = \frac{\left( \sum_{n=0}^{sbfrm\_size-1} t[n] \cdot y[r][n] \right)^2}{\sum_{n=0}^{sbfrm\_size-1} y^2[r][n]}$$

The signal  $t[n]$  represents the weighted target signal and  $y[r][n]$  is the convolution of the sequence  $ca[ia[r]-n-1]$  with the impulse response  $h[n]$ . The index  $(ia[r]-Lmin)$  of the maximum is referred as **shape\_delay**[subframe].

After determining the index the gain factor is computed according to

$$g = \frac{\sum_{n=0}^{sbfm\_size-1} t[n] \cdot y'[n]}{\sum_{n=0}^{sbfm\_size-1} y'^2[n]}$$

with  $y'[n]$  equal to the convolution of  $ca[l+Lmin-n-1]$  with  $h[n]$ . The gain factor is quantized by a non-uniform quantizer:

```

for (j = 0; abs(g) > cba_gain_quant[j] && j < 31; j ++);

if (g < 0)
{
    Ga = -cba_gain[j];
    gain_indices[0] = (((-j - 1) & 63));
}
else
{
    Ga = cba_gain[j];
    Gain_indices[0][subframe] = j;
}

```

The quantized gain factor is referred as  $G_a$ . With **shape\_delay[]** and  $G_a$  the contribution  $p[n]$  of the adaptive codebook is computed according to

$$p[n] = G_a \cdot y'[n]$$

The contribution  $p[n]$  of the adaptive codebook is subtracted from the weighted target signal  $t[n]$  to obtain the residual signal  $e[n]$ :

$$e[n] = t[n] - p[n], 0 \leq n < sbfrm\_size$$

The residual signal  $e[n]$  is 'backward-filtered' to obtain  $tf[n]$ .

The fixed codebook search also consists of a preselection and a selection phase. An RPE-codebook is used for the fixed codebook excitation. Each codebook vector has  $sbfm\_size$  pulses of which  $N_p$  pulses may have an amplitude of +1, 0 or -1. These  $N_p$  pulses are positioned on a regular grid defined by the phase  $p$  and the pulse spacing  $D$  such that the grid positions are at  $p + D/l$  where  $l$  is between 0 and  $N_p$ . The leaving  $(sbfm\_size - N_p)$  pulses are zero.  $D$  and  $N_p$  are dependent on the bitrate setting as is given in Table 3.87.

To reduce complexity a local RPE-codebook is generated for each subframe containing a subset of 16 entries. All vectors of this local RPE-codebook have the same phase  $P$  which computed as follows:

```

max = 0;
for (p = 0; p < D; p++)
{
    tmp_max = 0;
    for (l = 0; l < Np-1; l++)
    {
        tmp_max += abs(tf[p+D * l]);
    }
    if (tmp_max > max)
    {
        max = tmp_max;
        P = p;
    }
}

```

Having determined the phase P it is required that the amplitude of pulse l,  $0 \leq l < N_p$ , is either zero or equal to the sign of the corresponding sample of  $tf[n]$ . The sign is stored in  $amp[l]$  as follows:

```

for (l = 0; l < Np; l++)
{
    amp[l] = sign (tf[P + D * l]);
}

```

where sign is +1 for values greater than 0 and -1 otherwise.

For  $N_p-4$  pulses the possibility of zero-amplitude is excluded a-priori at those positions where  $tf[n]$  is maximal. Therefore, array  $pos[]$  is used, which has the following semantic:

- $pos[l]=0$  indicates that zero-amplitude possibility is included for pulse l.
- $pos[l]=1$  indicates that zero-amplitude possibility is excluded.

The following algorithm is used to determine the array  $pos[]$ :

1. initialize  $pos[]$  with zeros,
2. Determine n such that  $abs(tf[P + D * n]) \geq abs(tf[P + D * i])$  for all i not equal to n
3.  $pos[n] = 1$
4.  $tf[P + D * n] = 0$
5. Proceed with step 2 until  $N_p-4$  positions are fixed

Based on P,  $amp[]$  and  $pos$  the local RPE-codebook  $cf[k][n]$  is generated, using the following algorithm:

```

for(k = 0; k < 16; k++)
{
  for(n = 0; n < sbfrm_size; n++)
  {
    cf[k][n] = 0;
  }
}
for (l = 0; l < Np; l++)
{
  cf[0][P+D*l]=amp[l];
}

m = 1;
for (l = 0; l < Np; l++)
{
  if (pos[l]==0)
  {
    c = m;
    for (q = 0; q < c; q++)
    {
      for (n = 0; n < Np; n++)
      {
        cf[m][P+D*n] = cf[q][P+D*n];
      }
      cf[m++][P+D*l] = 0;
    }
  }
}
}

```

Based on the local RPE-codebook, 5 vectors out of 16 are preselected for the closed-loop search. Preselection is achieved by evaluating the term for  $0 \leq k < 16$ :

$$rfp[k] = \frac{\left( \sum_{n=0}^{sbfrm\_size-1} cf[k][n] \cdot tf[n] \right)^2}{Ef[l]}$$

with  $cf[k][n]$  representing a vector of the local RPE-codebook.  $Ef[k]$  is the energy of that codebook vector filtered by the reduced complexity synthesis filter  $S_p(z)$ . The indices of the preselected vectors are stored in  $if[r]$ .

Using these preselected indices, a closed-loop fixed codebook search is performed. By the closed-loop fixed codebook search it is searched for the index  $r$  maximizing the term

$$rf[r] = 2 \cdot Gf \cdot \sum_{n=0}^{sbfrm\_size-1} e[n] \cdot y[r][n] - Gf^2 \cdot \sum_{n=0}^{sbfrm\_size-1} y^2[r][n]$$

The signal  $e[n]$  represents the residual signal of the adaptive codebook search and  $y[r][n]$  is the convolution of  $cf[if[r]][n]$  with the impulse response  $h[n]$ .  $Gf$  is the quantized gain factor  $g$  which is determined according to

$$g = \frac{\sum_{n=0}^{sbfrm\_size-1} e[n] \cdot y[r][n]}{\sum_{n=0}^{sbfrm\_size-1} y^2[r][n]}$$

The quantization process of the fixed codebook gain is given below, where  $G_p$  is the previous quantized fixed codebook gain.

```

if (first subframe)
{
  for (m = 0; g > cbf_gain_quant[m] && m < 30; m ++);
  Gf = cbf_gain[m];
}
else
{
  g = g / Gp;
  for (m = 0; g > cbf_gain_quant_dif[m] && m < 7; m ++);
  g = Gp * cbf_gain_diff[m];
}
gain_indices[0] = m;

```

The representation tables `cbf_gain` and `cbf_gain_dif` are given in Table 3.89 and Table 3.90, whereas the quantization tables `cbf_gain_quant` and `cbf_gain_quant_dif` are given below.

Finally, the decoder is simulated by performing the following steps:

- the excitation of the fixed codebook is computed
- the excitation of the fixed and adaptive codebook are summed
- the memory of the adaptive codebook is updated
- the memory of the filters is updated

A detailed description of these steps is given in the decoder.

**Tables used for gain quantization**

**Table 3.B.11 - Quantization table for adaptive codebook gain**

Index	Cba_gain_quant	Index	cba_gain_quant
0	0.1622	16	0.9989
1	0.2542	17	1.0539
2	0.3285	18	1.1183
3	0.3900	19	1.1933
4	0.4457	20	1.2877
5	0.4952	21	1.4136
6	0.5425	22	1.5842
7	0.5887	23	1.8559
8	0.6341	24	2.3603
9	0.6783	25	3.8348
10	0.7227	26	7.6697
11	0.7664	27	15.339
12	0.8104	28	30.679
13	0.8556	29	61.357
14	0.9005	30	122.71
15	0.9487	31	245.43

**Table 3.B.12 - Quantization table for fixed codebook gain**

Index	Cbf_gain_quant	Index	cbf_gain_quant
0	2.4726	16	82.3374
1	3.1895	17	95.3755
2	4.2182	18	109.8997
3	5.6228	19	126.3037
4	7.3781	20	144.3995
5	9.5300	21	165.5142
6	12.1013	22	190.9742
7	15.2262	23	220.6299
8	19.0319	24	258.2699
9	23.6342	25	305.5086
10	29.1562	26	368.5894
11	35.3606	27	453.5156
12	42.8301	28	573.6164
13	51.1987	29	801.6422
14	60.6440	30	9999.9
15	70.9884	31	-----

**Table 3.B.13 - Quantization table for differential fixed codebook gain**

Index	cbf_gain_quant_dif	Index	cbf_gain_quant_dif
0	0.2500	4	1.3356
1	0.5378	5	1.8869
2	0.7795	6	4.2000
3	1.0230	7	9999.9

### 3.B.9.2 Multi-pulse excitation

#### 3.B.9.2.1 Tool description

For the CELP coder based on Multi Pulse Excitation, the output of the excitation analysis are **signal\_mode**, **rms\_index**, **shape\_delay[]**, **shape\_positions[]**, **shape\_signs[]** and **gain\_index[]**. The shape and gain indices are generated every subframe. The vector **shape\_delay[]** contains the adaptive codebook lag for each subframe, while the vectors **shape\_positions[]** and **shape\_signs[]** contain the pulse positions and signs respectively. The adaptive codebook gain and multi-pulse gain for each subframe are vector quantized and stored in the vector **gain\_index[]**.

#### 3.B.9.2.2 Definitions

##### Input

*PP\_inputSignal[]*: This array contains the preprocessed input signal and is of dimension *sfrm\_size*.

*int\_Qlpc\_coefficients[]*: This array contains the interpolated and quantized LPC coefficients.

*Wnum\_coeff[]*: This array contains the weighting filter coefficients of the numerator.

*Wden\_coeff[]*: This array contains the weighting filter coefficients of the denominator.



### Output

**signal\_mode:** This field contains the voiced/unvoiced flag .

**rms\_index:** This field defines the index for the signal power.

**shape\_delay[]:** This array is of dimension *nrof\_subframes*. It contains the codebook lag for the adaptive and fixed codebooks.

**shape\_positions[]:** This array is of dimension *nrof\_subframes*. It contains the pulse positions.

**shape\_signs[]:** This array is of dimension *nrof\_subframes*. It contains the pulse signs.

**gain\_index[]:** This array is of dimension *nrof\_subframes*. It contains the vector quantized gains for the adaptive codebook and the multi-pulse.

### Configuration

*frame\_size:* This field indicates the number of samples in one frame.

*sbfrm\_size:* This field indicates the number of samples in one subframe.

*nrof\_subframes:* This field indicates the number of subframes in one frame.

*lpc\_order:* This field indicates the order of LPC.

#### **3.B.9.2.3 Encoding process**

For the Mode II coder, the excitation signal is extracted and encoded on an analysis-by-synthesis basis with three steps, the adaptive codebook search for a periodic component, the fixed codebook search for a non-periodic component and the quantization of the gains for each component. The target signal is obtained by subtracting the zero-input response of the synthesis and perceptual weighting filters from the weighted input speech signal.

#### **Frame Energy Quantization**

The root mean square (rms) value of subframe input samples is calculated at the last subframe. The rms value is scalar-quantized in the  $\mu$ -law scale. The rms values of other subframes are obtained by linear-interpolating the quantized rms values at the last subframe of the current and the previous frames. The quantized rms values are used for gain normalization.

#### **Open-loop Pitch Estimation and Mode Decision**

The open-loop pitch estimation and mode decision procedures are jointly performed every analysis interval of 10 ms using the perceptually weighted signal. The pitch estimate is selected to minimize the square error between the current and previous blocks of the weighted input samples. A pitch prediction gain is calculated for each analysis interval. Every frame is classified into one of the four modes based on the average pitch prediction gain. Modes 0 and 1 correspond to unvoiced and transition frames, respectively. Modes 2 and 3 correspond to voiced frames, and the latter has higher periodicity than the former. If the subframe length is 5 ms, the same estimated pitch is used in two subframes for a closed-loop pitch search.

#### **Encoding of Excitation Signal**

The excitation signal is represented by a linear combination of the adaptive codevector and the fixed codevector scaled by their respective gains. Each component of the excitation signal is successively chosen by an analysis-by-synthesis search procedure so that the perceptually weighted error between the input signal and the reconstructed

signal is minimized. The adaptive codevector parameters are a closed-loop delay and a gain. The closed-loop delay is selected in the range around the estimated open-loop delay. The adaptive codevector is generated from a block of the past excitation signal samples with the selected closed-loop delay. The fixed codevector contains several non-zero pulses. The pulse positions are restricted in an algebraic structure. The restriction table of the pulse positions are set up from the parameters. In order to improve the performance, after determining plural sets of pulse position candidates, a combination search between the pulse position candidates and the pulse amplitude index is carried out. The gains for both the adaptive and the multi-pulse codevectors are normalized and vector-quantized. The normalization factor is calculated from the quantized LP coefficients, the quantized subframe rms and an excitation signal rms. The gain codebook is changed in accordance with the mode.

### Adaptive Codebook Search

For each subframe the optimal delay is determined through closed-loop analysis so that the mean-square error between the target signal  $x(n)$  and the weighted synthesized signal  $y_t(n)$  of the adaptive codevector is minimized.

$$E_t = \sum_{n=0}^{N-1} (x(n) - g_t y_t(n))^2, \quad t_{op} - 8 < t < t_{op} + 8$$

where  $t_{op}$  is the open-loop delay determined in the open-loop pitch analysis.  $g_t$  is the optimal gain as follows:

$$g_t = \frac{\sum_{n=0}^{N-1} x(n)y_t(n)}{\sum_{n=0}^{N-1} y_t^2(n)}$$

The optimal pitch delay is encoded with 8 bits based on the relationship between the shape\_indices[0] and the delay (see Decoding process).

### Fixed Codebook Search

The fixed codebook vector is represented by the pulse position and pulse amplitudes. The pulse positions and amplitudes are searched to minimize the mean-square error between the target signal and the weighted synthesized signal.

$$E_k = \sum_{n=0}^{N-1} (x(n) - g_t y_t(n) - g_k z_k(n))^2$$

where  $g_k$  is the optimal gain as follows:

$$g_k = \frac{\sum_{n=0}^{N-1} (x(n) - g_t y_t(n))z_k(n)}{\sum_{n=0}^{N-1} z_k^2(n)}$$

where  $k$  indicates the possible combination of the shape\_indices[1] and the shape\_indices[2] (see Decoding process). The fixed codebook vector is constructed from the pulse positions and the pulse amplitudes (see

Decoding process). The weighted signal  $z_k(n)$  is computed by filtering the fixed codebook vector through the LP synthesis filter  $1/A(z)$  and the perceptual weighting filter  $W(z)$ .

### Gain Quantization

Gains for the adaptive codevector and the fixed codevector are normalized by the prediction residual energy  $rs$  and scalar-quantized. The residual energy  $rs$  is calculated based on frame energy and reflection coefficients. The frame energy is calculated every subframe as a root mean square (RMS) value and quantized in the  $\mu$ -law domain. The reflection coefficients  $k(i)$  are converted from the interpolated LPCs  $int\_Qlpc\_coefficients[]$ . Consequently, the prediction residual energy ( $rs$ ) of a subframe is

$$rs = sbfrm\_len \cdot q\_amp^2 \cdot \prod_{i=1}^{N_p} (1 - k^2(i)),$$

where  $N_p$  is the LP analysis order,  $q\_amp$  is the quantized RMS amplitude and  $sbfrm\_len$  is the subframe length.

The gain quantization is carried out by finding the pair of gain indices that minimize the error between the target and the weighted synthesized signal. The error  $err_g$  due to gain quantization is given by:

$$err_g = \sum_{i=0}^{sbfrm\_len-1} \left( \sqrt{\frac{rs}{pow_{AC}}} \cdot AdaptGainCB[m] \cdot ac\_syn(i) + \sqrt{\frac{rs}{pow_{SC}}} \cdot FixedGainCB[n] \cdot sc\_syn(i) - targ(i) \right)^2$$

$$pow_{ac} = \sum_{i=0}^{sbfrm\_len-1} ac\_ex^2(i)$$

$$pow_{sc} = \sum_{i=0}^{sbfrm\_len-1} sc\_ex^2(i)$$

where  $m$  and  $n$  are indices,  $ac\_ex(i)$  is the selected adaptive codevector,  $ac\_syn(i)$  is the signal which is synthesized from  $ac\_ex(i)$  and perceptually weighted,  $sc\_ex(i)$  is the selected stochastic codevector and  $sc\_syn(i)$  is the signal which is synthesized from  $sc\_ex(i)$  and perceptually weighted.

The index pair of  $n$  and  $m$  that minimize the error  $err_g$  is selected and is stored in  $gain\_indices[]$ . Then the quantized adaptive gain  $qacg$ , the quantized stochastic  $qscg$  and the excitation signal  $decoded\_excitation(i)$  are calculated as follows.

$$decoded\_excitation(i) = qacg \cdot ac\_ex(i) + qscg \cdot sc\_ex(i) \quad (0 \leq i \leq sbfrm\_len - 1)$$

$$qacg = \sqrt{\frac{rs}{pow_{ac}}} \cdot AdaptGainCB(m)$$

$$qscg = \sqrt{\frac{rs}{pow_{sc}}} \cdot FixedGainCB(n)$$

### 3.B.9.3 Bitrate scalable multi-pulse excitation

#### 3.B.9.3.1 Tool description

The output of the enhancement excitation analysis are **shape\_enh\_positions[]**, **shape\_enh\_signs[]** and **gain\_enh\_index[]**. The shape and gain indices are generated every subframe. The vectors **shape\_enh\_positions[]** and **shape\_enh\_signs[]** contain the enhancement pulse positions and signs respectively. The enhancement multi-pulse gain for each subframe is scalar quantized and stored in the vector **gain\_index[]**.

#### 3.B.9.3.2 Definitions

##### Input

*PP\_inputSignal[]*: This array contains the preprocessed input signal and is of dimension *sfrm\_size*.

*int\_Qlpc\_coefficients[]*: This array contains the interpolated and quantized LPC coefficients.

*Wnum\_coeff[]*: This array contains the weighting filter coefficients of the numerator.

*Wden\_coeff[]*: This array contains the weighting filter coefficients of the denominator.

##### Output

**shape\_enh\_positions[]**: This array is of dimension *nrof\_subframes*. It contains the pulse positions.

**shape\_enh\_signs[]**: This array is of dimension *nrof\_subframes*. It contains the pulse signs.

**gain\_enh\_index[]**: This array is of dimension *nrof\_subframes*. It contains the vector quantized gains for the adaptive codebook and the multi-pulse.

##### Configuration

*n\_lag\_candidates*: This field indicates the number of lag candidates.

*frame\_size*: This field indicates the number of samples in one frame.

*sbfrm\_size*: This field indicates the number of samples in one subframe.

*nrof\_subframes*: This field indicates the number of subframes in one frame.

*lpc\_order*: This field indicates the order of LPC.

#### 3.B.9.3.3 Encoding process

The fixed codebook vector is represented by the pulse position and pulse amplitudes. The pulse positions and amplitudes are searched to minimize the mean-square error between the target signal and the weighted synthesized signal.

$$E_k = \sum_{n=0}^{N-1} (x(n) - g_t y_t(n) - g_k z_k(n))^2$$

where  $g_t$  is the optimal gain as follows:

$$g_k = \frac{\sum_{n=0}^{N-1} (x(n) - g_t y_t(n)) z_k(n)}{\sum_{n=0}^{N-1} z_k^2(n)}$$

where  $k$  indicates the possible combination of the `shape_enh_positions` and the `shape_enh_signs` (see Decoding process). The fixed codebook vector is constructed from the pulse positions and the pulse amplitudes (see Decoding process). The weighted signal  $z_k(n)$  is computed by filtering the fixed codebook vector through the LP synthesis filter  $1/A(z)$  and the perceptual weighting filter  $W(z)$ .

### 3.B.9.4 Multi-pulse excitation for the bandwidth extension tool

#### 3.B.9.4.1 Tool description

The output of the excitation analysis in the bandwidth extension tool are `shape_bws_delay[]`, `shape_bws_positions[]`, `shape_bws_signs[]` and `gain_bws_index[]`. The shape and gain indices are generated every subframe. The vector `shape_bws_delay[]` contains the adaptive codebook lag for each subframe, while the vectors `shape_bws_positions[]` and `shape_bws_signs[]` contain the pulse positions and signs respectively. The adaptive codebook gain and multi-pulse gain for each subframe are vector quantized and stored in the vector `gain_bws_index[]`.

#### 3.B.9.4.2 Definitions

##### Input

`PP_inputSignal[]`: This array contains the preprocessed input signal and is of dimension `sfrm_size`.

`int_Qlpc_coefficients[]`: This array contains the interpolated and quantized LPC coefficients.

`Wnum_coeff[]`: This array contains the weighting filter coefficients of the numerator.

`Wden_coeff[]`: This array contains the weighting filter coefficients of the denominator.

##### Output

`shape_bws_delay[]`: This array is of dimension `nrof_subframes_bws`. It contains the codebook lag for the adaptive and fixed codebooks.

`shape_bws_positions[]`: This array is of dimension `nrof_subframes_bws`. It contains the pulse positions.

`shape_bws_signs[]`: This array is of dimension `nrof_subframes_bws`. It contains the pulse signs.

`gain_bws_index[]`: This array is of dimension `nrof_subframes_bws`. It contains the vector quantized gains for the adaptive codebook and the multi-pulse.

##### Configuration

`n_lag_candidates`: This field indicates the number of lag candidates.

`frame_size`: This field indicates the number of samples in one frame.

`sbfrm_size`: This field indicates the number of samples in one subframe in the bandwidth extension tool.

*nrof\_subframes\_bws*: This field indicates the number of subframes in the bandwidth extension tool.

*lpc\_order*: This field indicates the order of LPC.

### 3.B.9.4.3 Encoding process

For the bandwidth extension tool, the excitation signal is extracted and encoded on an analysis-by-synthesis basis with three steps, the adaptive codebook search for a periodic component, the fixed codebook search for a non-periodic component and the quantization of the gains for each component. The target signal is obtained by subtracting the zero-input response of the synthesis and perceptual weighting filters from the weighted input speech signal.

#### Frame Energy Quantization

The same root mean square (rms) value as the 8 kHz sampling rate CELP is used.

#### Open-loop Pitch Estimation and Mode Decision

The open-loop pitch estimation is done by converting the 8 kHz sampling rate pitch delay (see Decoding process). The same mode as the 8 kHz sampling rate CELP is used.

#### Encoding of Excitation Signal

The excitation signal is represented by a linear combination of the adaptive codevector and the two fixed codevectors scaled by their respective gains. The pitch delay is decided in the range around the open-loop estimated pitch delay. One of the two fixed codevectors is obtained by sampling-rate expansion of the fixed codevector used in the 8 kHz sampling rate coder. The other fixed codevector is determined by an analysis-by-synthesis search procedure.

#### Adaptive Codebook Search

For each subframe the optimal delay is determined through closed-loop analysis so that the mean-square error between the target signal  $x(n)$  and the weighted synthesized signal  $y_t(n)$  of the adaptive codevector is minimized.

$$E_t = \sum_{n=0}^{N-1} (x(n) - g_t y_t(n))^2, \quad t_{op} - 8 < t < t_{op} + 8$$

where  $t_{op}$  is the open-loop delay determined in the open-loop pitch analysis.  $g_t$  is the optimal gain as follows:

$$g_t = \frac{\sum_{n=0}^{N-1} x(n)y_t(n)}{\sum_{n=0}^{N-1} y_t^2(n)}$$

The difference between the optimal pitch delay and the open-loop pitch delay is encoded with 3 bits based on the relationship between the *shape\_bws\_delay* and the differential delay (see Decoding process).

### Fixed Codebook 1 Search

The fixed codevector 1 is obtained by sampling-rate expansion of the fixed codevector used in the 8 kHz sampling rate coder (see Decoding Process).

### Fixed Codebook 2 Search

The fixed codebook vector is represented by the pulse position and pulse amplitudes. The pulse positions and amplitudes are searched to minimize the mean-square error between the target signal and the weighted synthesized signal.

$$E_k = \sum_{n=0}^{N-1} (x(n) - g_t y_t(n) - g_k z_k(n))^2$$

where  $g_t$  is the optimal gain as follows:

$$g_k = \frac{\sum_{n=0}^{N-1} (x(n) - g_t y_t(n)) z_k(n)}{\sum_{n=0}^{N-1} z_k^2(n)}$$

where  $k$  indicates the possible combination of the `shape_bws_positions` and the `shape_bws_signs` (see Decoding process). The fixed codebook vector is constructed from the pulse positions and the pulse amplitudes (see Decoding process). The weighted signal  $z_k(n)$  is computed by filtering the fixed codebook vector through the LP synthesis filter  $1/A(z)$  and the perceptual weighting filter  $W(z)$ .

### Gain Quantization

Gains for the adaptive codevector and the two fixed codevectors are normalized by the prediction residual energy  $rs$  and quantized. The residual energy  $rs$  is calculated based on frame energy and reflection coefficients. The frame energy is calculated every subframe as a root mean square (RMS) value and quantized in the  $\mu$ -law domain. The reflection coefficients  $k(i)$  are converted from the interpolated LPCs `int_Qlpc_coefficients[]`. Consequently, the prediction residual energy ( $rs$ ) of a subframe is

$$rs = sbfrm\_len \cdot q\_amp^2 \cdot \prod_{i=1}^{N_p} (1 - k^2(i)),$$

where  $N_p$  is the LP analysis order,  $q\_amp$  is the quantized RMS amplitude and  $sbfrm\_len$  is the subframe length.

The gains for the adaptive codevector and the fixed codevector 2 are vector quantized. The gain for the fixed codevector 1 is scalar quantized. These quantization operations are achieved by minimizing the perceptually weighted distortion (closed-loop).

## 3.B.10 CELP bitstream multiplexer

### 3.B.10.1 Tool description

The tool CELP bitstream multiplexer multiplexes a frame into the bitstream.

### 3.B.10.2 Definitions

All the bitstream elements and the helping variables have been defined in subclause 3.5.3 and subclause 3.5.4.

### 3.B.10.3 Encoding process

The parameters are encoded into a bitstream according to the syntax described in subclause 3.3.

## 3.B.11 CELP silence compression tool

### 3.B.11.1 VAD module

The VAD module makes a decision whether a frame is a non-active-voice frame or an active-voice frame based on how much the characteristics of the input signal change. The characteristics are represented by four parameters: the full-band and the low-band energies, the LSPs, and the zero-crossing rate of the input signal frame. A temporal decision is made in every 80 samples and the final frame decision is made based on the temporal decisions with a hangover constraint. For the wideband mode, the characteristics parameters are calculated from the input speech down-sampled from 16 kHz to 8 kHz.

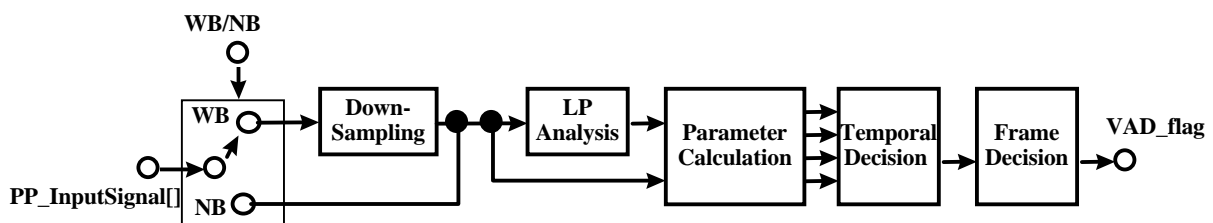


Figure 3.B.1 - VAD module

#### 3.B.11.1.1 Definitions

Input

**PP\_InputSignal[]** This array contains the pre-processed speech signal. The dimension is *frame\_size*.

Output

**VAD\_flag** This field contains the VAD flag (see Table 3.103).

The following are help elements used in the VAD module:

*lpc\_order*: the order of LP

*sfrm\_size*: the number of samples in a subframe

*frame\_size*: the number of samples in a frame

*n\_subframe*: the number of subframes in a frame



### 3.B.11.1.2 Down-sampling for the wideband

A signal  $s\_vad[]$  used in the VAD module is generated by preprocessing the input signal in the same manner as that described in subclause 3.B.4. When the sampling rate is  $16\text{ kHz}$ , the input signal is down-sampled to  $8\text{ kHz}$  after the preprocessing.

### 3.B.11.1.3 Parameter calculation

The LSPs of the input signal,  $lsp[]$  are calculated from the LPCs  $lpc\_coefficients[]$ , which is given by the MPEG-4 CELP weighting module described in subclause 3.B.8. A full-band energy  $P$ , a low-band energy  $P_l$  between 0 and 1 kHz, and a zero crossing rate  $Z$  are calculated as follows:

$$P = 10 \log_{10} R[0]$$

$$P_l = 10 \log_{10} \mathbf{h}^T \mathbf{R} \mathbf{h}$$

$$Z = \frac{1}{80} \sum_{i=0}^{80} |\text{sign}[s\_vad[i]] - \text{sign}[s\_vad[i-1]]| ,$$

where  $\mathbf{h}$  is an impulse response vector of the FIR filter with a cutoff frequency of  $1\text{ kHz}$ .  $R[0]$  is the first autocorrelation coefficient, and  $\mathbf{R}$  is a Toeplitz autocorrelation matrix with autocorrelation coefficients in each diagonal. These parameters are calculated every  $10\text{ msec}$ . Their averages are updated as follows:

$$\bar{P} = a \bar{P} + (1 - a) P$$

$$\bar{P}_l = b \bar{P}_l + (1 - b) P_l$$

$$\bar{Z} = c \bar{Z} + (1 - c) Z$$

$$\bar{L}lsp[i] = d \bar{L}lsp[i] + (1 - d) Lsp[i] , i=1, \dots, lpc\_order$$

where  $a=0.995$ ,  $b=0.995$ ,  $c=0.998$  and  $d=0.75$ . The following differential parameters are evaluated to make a temporal decision whether the input signal is characterized as an active-voice or a non-active-voice every  $80\text{ samples}$ :

$$\Delta P = \bar{P} - P$$

$$\Delta P_l = \bar{P}_l - P_l$$

$$\Delta Z = \bar{Z} - Z$$

$$\Delta Lsp = \sum_{i=1}^{lpc\_order} [\bar{L}lsp[i] - Lsp[i]]^2 , i=1, \dots, lpc\_order.$$

### 3.B.11.1.4 Temporal voice activity decision

If any of the following inequalities is satisfied, the temporal voice activity flag,  $vad\_flag\_sub[i]$  for  $i=0, \dots, \frac{frm\_size[samples]}{80[samples]}$ , is set to 1.

```

if(
     $\Delta L_{sp} > 0.0009$  or
     $\Delta L_{sp} > 0.00175 * \Delta Z + 0.00085$  or
     $\Delta L_{sp} > -0.00455 * \Delta Z + 0.00116$  or
     $\Delta P < -0.47$  or
     $\Delta P < -2.5 * \Delta Z - 0.5$  or
     $\Delta P < 2.0 * \Delta Z - 0.6$  or
     $\Delta P < 2.5 * \Delta Z - 0.7$  or
     $\Delta P < -2.91 * \Delta Z - 0.482$  or
     $\Delta P < 880.0 * \Delta L_{sp} - 1.22$  or
     $\Delta P_1 < 1400.0 * \Delta L_{sp} - 1.55$  or
     $\Delta P_1 > 0.929 * \Delta P + 0.114$  or
     $\Delta P_1 < -1.5 * \Delta P - 0.9$  or
     $\Delta P_1 < 0.714 * \Delta P - 0.214$ 
)
{
    vad_flag_sub[i] = 1;
}else{
    vad_flag_sub[i] = 0;
}

```

### 3.B.11.1.5 Frame voice activity decision

A frame voice activity flag, *vad\_flag* is determined based on the temporal flag *vad\_flag\_sub*[], which is made in the corresponding frame as follows:

$$vad\_flag = vad\_flag\_sub[0] \cup vad\_flag\_sub[1] \cup \dots \cup vad\_flag\_sub[L-1],$$

where “ $\cup$ ” stands for a logical OR.

### 3.B.11.1.6 Hangover

A hangover is applied to the final VAD decision *VAD\_flag* after the switch from an active-voice frame to a non-active-voice frame:

$$VAD\_flag = \begin{cases} 1, & \text{first 80 msec after the active-voice period (vad\_flag = 1)} \\ vad\_flag, & \text{otherwise} \end{cases}$$

When the active-voice period (*vad\_flag*=1) is shorter than 80 msec, *VAD\_flag* is always equal to *vad\_flag*.

### 3.B.11.2 DTX module

Figure 3.B.2 shows the structure of the DTX module. The DTX module detects frames in which the input characteristics change during the non-active-voice frames. In the first frame during each non-active-voice period and in the frame where the change is detected, the DTX module extracts the parameters; the frame energy and the LSPs of the input speech, and encodes these parameters. There are three DTX modes; DTX\_flag=0, 1 and 2 depending on what information is transmitted. When a change in the LSPs is detected (DTX\_flag=1), the encoded LSP and RMS parameters and the TX\_flag are transmitted as HR-SID information. When a change of the RMS (the

frame energy) is detected (DTX\_flag=2), only the encoded RMS parameter and the TX\_flag are transmitted as LR-SID information. Otherwise, only the TX\_flag is transmitted.

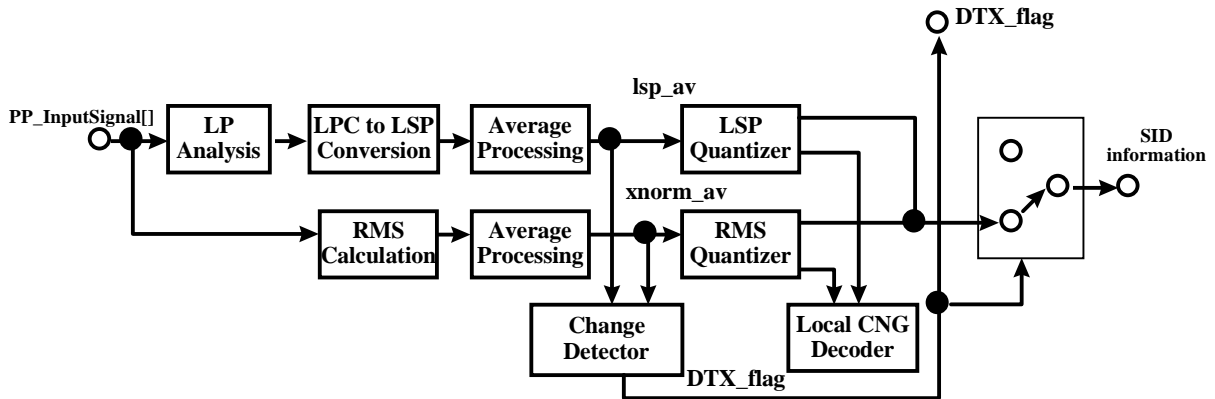


Figure 3.B.2 - DTX module

3.B.11.2.1 Definitions

Input

PP\_InputSignal[] This array contains the pre-processed speech signal. The dimension is *frame\_size*.

TX\_flag This field contains the transmission mode (see Table 3.103).

Output

VAD\_flag This field contains the VAD flag (see Table 3.103).

The following are help elements used in the DTX module:

*lpc\_order*: the order of LP

*frame\_size*: the number of samples in a frame

*n\_subframe*: the number of subframes in a frame

3.B.11.2.2 LP analysis

The same LP analysis and LPC-to-LSP conversion as those described in subclause 3.B.5 is used. This gives unquantized LSPs *lsp[][]*.

3.B.11.2.3 Averaging of the LSP

An average LSP *lsp\_av[]* is calculated from the unquantized LSPs *lsp[][]* as follows:

$$lsp\_av[i] = (1/L) \sum_{j=0}^{L-1} lsp[j][i], \quad i=0, \dots, lpc\_order-1.$$

where *lsp[j][i]* are unquantized LSPs in the *j*-th most recent frame, which is calculated from the unquantized LPCs *lpc\_coefficients[]* described in subclause 3.B.6. *L* is the number of frames per 80 msec.

### 3.B.11.2.4 RMS calculation

A RMS of the input signal is calculated in an identical manner to that described in subclause 3.B.9. This calculation gives unquantized RMS  $xnorm[]$  in each subframe.

### 3.B.11.2.5 Averaging of the RMS

An average RMS of the input signal,  $xnorm\_av[]$  is computed from the unquantized RMS  $xnorm[]$  as follows:

$$xnorm\_av = (1/M) \sum_{j=0}^{M-1} xnorm[j]$$

where  $M$  is the number of subframes per 80 msec. Also,  $xnorm\_av\_end$  is  $xnorm\_av[n\_subframe-1]$ , where  $n\_subframe$  is the number of subframes in a frame.

### 3.B.11.2.6 Detection of the characteristics change

Any change in the characteristics is detected based on the variations in the frame energy and the spectrum computed from the input signal as follows:

$$DTX\_flag = 0$$

$$\text{if} ( |20\log_{10} xnorm\_sid - 20\log_{10} xnorm\_end| > D_{xnorm} \text{ dB} ) DTX\_flag=2$$

$$\text{if} ( \sum_{i=1}^{lpc\_order} [lsp\_sid[i] - lsp\_av[i]]^2 > D_{lsp} ) DTX\_flag=1,$$

where LSPs are normalized to a range from 0 to 1. A threshold  $D_{xnorm}$  is switched according to  $xnorm\_sid$  as shown in Table 3.B.14. A threshold  $D_{lsp}$  is changed according to the sampling rate; 0.002 for 8 kHz and 0.0015 for 16 kHz.  $lsp\_av\_sid$  and  $xnorm\_sid$  are  $lsp\_av$  and  $xnorm\_av\_end$  of the last SID frame, respectively. There is a minimum period where the detection is not made. The length of the period is normally 20 msec, but it is 0 msec during the first 40 msec of the non-active-voice period.

**Table 3.B.14 - Relationship between  $D_{xnorm}$  and  $xnorm\_sid$**

$20\log_{10} xnorm\_sid$ [dB]	$D_{xnorm}$ [dB]
< 1.0:	6.0
1.0 ~ 5.0	4.0
5.0 ~ 9.0	3.0
9.0 ~ 12.0	2.5
12.0 <	2.0

### 3.B.11.2.7 Parameter encoding

The average LSP  $lsp\_av[]$  is encoded using the same process as that described in subclause 3.B.6 with the exceptions described in subclause 11.5.2 (LSP decoder). The encoding of the average RMS  $xnorm\_av[]$  is identical to that described in subclause 3.B.9, with the exceptions that the  $\mu$ -law parameters are independent of the signal mode and are set as  $rms\_max=7932$  and  $mu\_law=1024$ .

**3.B.11.2.8 Local CNG decoder**

Local CNG decoding is performed to update buffers for the LP synthesis filter. The processing is identical to the decoding process in the decoder.

## Annex 3.C (normative)

### Tables

#### 3.C.1 LSP VQ tables and gain VQ tables for 8 kHz sampling rate

Table 3.C.1 - LSP table for the first stage ( $\text{lsp\_tbl}[0][\text{Index}][\ ]$ )

scale\_factor =  $2^{15}$

Index	Codeword				
	0	1	2	3	4
0	2465	3900	5664	7874	9529
1	2430	4275	6343	8750	11411
2	1911	2913	4284	6757	12988
3	2978	5317	8371	11095	14258
4	1878	2844	4285	6524	8539
5	2382	3849	5636	8795	13807
6	1171	1944	4453	9571	12822
7	3212	6003	9409	12522	15568
8	1755	2837	4879	8370	11087
9	2987	5164	7422	9876	12412
10	1895	2870	4047	7313	15485
11	1749	3596	7928	12060	15520
12	1545	2253	3430	5844	11194
13	1816	3463	6818	10315	13176
14	1919	2912	4709	10681	15270
15	3405	6578	11374	14885	17519

Table 3.C.2 - LSP table for the first stage ( $\text{lsp\_tbl}[1][\text{Index}][\ ]$ )

scale\_factor =  $2^{15}$

Index	Codeword				
	0	1	2	3	4
0	16725	19702	22552	25746	27870
1	14891	18400	21563	24768	29116
2	18815	21093	23260	25905	28001
3	14648	18159	21499	24676	27025
4	16274	19321	22413	26092	29415
5	11847	19451	22070	25544	28584
6	20171	22959	25232	27632	29514
7	12579	16636	20917	23824	26623
8	16685	18912	21352	24430	27213
9	14427	18039	20139	26405	28635
10	17223	20497	23557	26856	29546
11	14627	17482	19981	22769	26192
12	14411	19945	22550	26869	29219
13	11882	16454	21931	25698	28836
14	18194	21846	24892	27871	29977
15	13161	16571	19143	23542	28717

Table 3.C.3 - LSP table for the second stage of VQ without interframe prediction (d\_tbl[0][Index][i])

scale\_factor = 2<sup>18</sup>

Index	Codeword				
	0	1	2	3	4
0	-1126	-109	2708	-1644	4815
1	-2453	2636	8784	5571	-1482
2	-1303	5382	2696	7113	-4855
3	2833	-70	915	1226	-223
4	3977	1246	-3288	-719	6271
5	3881	5460	5880	3784	-2910
6	-465	443	-2027	4891	2781
7	7186	8149	-6311	785	766
8	3509	4478	5088	354	3239
9	-2612	-1774	8384	2234	3546
10	455	-565	17	10318	-2947
11	3618	6966	7303	-7657	4683
12	-3251	-3354	-2990	6106	10100
13	3272	4496	5938	-3011	-3413
14	-1571	-2953	-4055	10707	3887
15	6610	11258	-11073	-1875	9146
16	661	5993	-567	-2306	8104
17	-2670	8858	6054	-96	1956
18	-4109	-3480	3662	11658	-1416
19	-200	425	3682	4465	-3850
10	6475	-1500	5350	6261	570
21	3196	3229	8497	2406	-8489
22	751	3263	3153	6347	2053
23	4730	7170	-728	3346	-5906
24	1520	1650	1984	5703	7990
25	-4481	-4064	15246	6561	-4680
26	1745	3676	7627	12515	177
27	2901	2443	2531	-5361	2756
28	988	-4549	-1624	404	9058
29	6914	11203	8517	683	-1274
30	-3178	-6034	10596	20414	12845
31	6578	8953	-2591	-7687	6476
32	4211	104	3424	-301	8485
33	-2034	1585	11201	-5153	1532
34	1639	2492	5533	10513	-8746
35	2358	5574	-808	-495	-316
36	3080	3666	-5363	6444	6223
37	9041	1554	2776	1534	-8749
38	-1962	-3634	1107	5018	4525
39	6003	9151	552	2144	5896
40	3118	4920	10734	3848	3150
41	-4584	-3459	4470	988	10814
42	4337	2406	-5712	8307	-3022
43	-675	3459	7994	-2264	9644
44	-1961	960	-1511	1664	8638
45	5874	10566	6805	-12232	-4384
46	1303	1031	231	12526	3738
47	11179	1372	-10700	1154	9015
48	573	758	906	-7236	10705
49	-5865	14935	10133	563	-8644

50	-2600	-2868	6607	8258	4025
51	244	-983	7562	-440	-2133
52	6886	-5508	-2676	5119	4081
53	-1360	4510	12050	-1224	-5202
54	-6719	3411	2672	5165	3416
55	6765	10685	-3144	-5258	-5219
56	891	4010	6076	11065	12632
57	-6906	-6860	13524	3954	4216
58	6686	6774	894	9019	380
59	11611	2335	203	-1753	1440
60	1729	725	-5941	624	13626
61	9827	13514	8008	-2229	-11638
62	-2343	-7659	-7405	15476	14625
63	16104	22055	-1580	110	3939

Table 3.C.4 - LSP table for the second stage of VQ without interframe prediction ( $d\_tbl[1][Index]$ )scale\_factor =  $2^{18}$ 

Index	Codeword				
	0	1	2	3	4
0	-784	4977	10919	2178	-7243
1	-1182	-3519	5460	3383	6735
2	2546	6242	2843	2291	6144
3	6510	-2312	3823	-3920	5884
4	-166	1153	4178	2694	-24
5	-6243	6349	-1005	7150	2470
6	2245	6605	2211	867	-3388
7	326	341	-2498	2721	5056
8	7981	2249	3442	4998	2602
9	-6443	-87	11399	890	2895
10	3935	1676	-5725	10571	2817
11	1083	-1790	-950	1243	15407
12	5676	7745	-1065	-6090	5950
13	-383	7422	7295	8602	2479
14	6200	768	8053	-2876	-2433
15	2730	-6624	2061	11171	923
16	-496	1617	4072	10057	-7194
17	4849	-5669	11789	3548	-816
18	-911	8873	-5233	1736	8291
19	-3367	4160	5035	-4114	7573
20	5630	810	-1134	573	-292
21	-6603	-703	5740	10320	2070
22	2973	12420	4445	-4076	-5042
23	10283	622	-7195	1350	7339
24	10608	-1538	1579	5882	-6505
25	-8052	10656	7008	-292	1287
26	4279	8612	-5498	5182	-3520
27	-5223	-2292	523	9511	13373
28	11418	5020	422	-4211	-911
29	8496	7600	10246	12203	12798
30	2299	5319	13330	-8096	1792
31	7693	-7706	-249	4207	6541



Table 3.C.5 - LSP table for the second stage of VQ with interframe prediction (pd\_tbl[0][Index][I])  
 scale\_factor = 2<sup>19</sup>

Index	Codeword				
	0	1	2	3	4
0	2532	3998	7136	9485	5083
1	-2097	5173	3711	4349	-2269
2	3145	936	-3045	4502	3310
3	-2425	-3497	2046	7254	4988
4	-1978	-2428	-5680	11862	7647
5	4003	306	4245	-2666	1903
6	1961	-2177	1126	120	9677
7	-1550	6306	8296	-4272	1462
8	1651	1603	5298	10637	-10274
9	3951	10014	1254	4008	-10934
10	6296	9279	-7936	1487	3241
11	-2658	-794	12908	609	-3544
12	631	1808	4942	10243	14943
13	6384	8886	7088	-543	7770
14	-707	482	6315	-6654	11595
15	-11649	12236	13527	9612	-2440
16	4684	5285	1631	18867	6223
17	3528	4132	7496	-15	-7553
18	2181	360	-3133	12049	-1862
19	-2449	-1085	9723	2395	6722
20	-560	-3142	-6080	28195	10042
21	4649	5443	-54	-8917	3778
22	-2700	-4856	-5676	10230	23087
23	4181	6107	16421	-6980	887
24	-4017	-2248	10625	12374	-2242
25	12560	13793	6109	843	-11284
26	16905	2231	-1388	-3055	2074
27	-2914	-1769	27848	7691	-3565
28	2504	5855	4073	3666	25794
29	16701	15278	9923	8078	1800
30	4846	4518	-3418	-5273	13037
31	4344	7195	13937	5995	-1118
32	6069	9436	4428	9661	-1595
33	2587	-1877	4404	5849	-2742
34	-2951	10357	-5009	10431	-1785
35	9470	-8852	10152	4690	-1396
36	5616	1774	-12799	12062	7965
37	-1384	6827	514	732	6505
38	-2066	990	-5485	2642	14029
39	6054	9511	5459	-6155	-3512
40	3894	5517	7460	19731	-12932
41	6945	6225	-1180	807	-3202
42	8980	9279	-16470	-2140	14198
43	-6027	-2368	20039	-3642	6502
44	6729	5214	-821	8497	9492
45	3454	8769	17467	847	14653
46	4004	7837	6595	-14235	9321
47	-3091	23678	7283	-149	8
48	4511	6737	14600	21923	4316
49	2426	8975	16750	1587	-14478
50	12018	6049	-5398	11537	-6297
51	-12145	8219	5436	-1020	10428

52	-3522	-6973	2794	16587	9151
53	9935	16017	-3390	-8287	1157
54	-5035	-5864	3814	2300	16845
55	7871	13250	13574	-14814	-5398
56	-1223	-1589	2711	21386	-3234
57	24577	29116	4713	-6772	-11585
58	10902	-8650	-4718	3868	10261
59	-6979	-10767	20809	14282	10494
60	3887	7227	11996	22138	24134
61	10278	16482	-939	5802	13611
62	924	-278	-2632	-11433	23190
63	6606	19360	25601	8376	4412

Table 3.C.6 - LSP table for the second stage of VQ with interframe prediction (pd\_tbl[1][Index][I])

scale\_factor = 2<sup>19</sup>

Index	Codeword				
	0	1	2	3	4
0	6216	-10079	2425	14328	263
1	444	660	4677	5548	1890
2	-9634	3530	-1050	15523	4278
3	-7425	10943	-3973	378	14153
4	1564	4930	8272	23911	-7335
5	-3677	13414	8730	-1472	-6113
6	-8226	4572	12153	-2383	10206
7	5960	5150	-8113	11969	176
8	9559	1543	-1145	1563	-2662
9	19979	1828	9044	10797	90
10	-6465	-6562	1897	5629	15698
11	5548	6185	4786	-5789	4433
12	2335	-3422	15593	5129	-6218
13	-250	3805	20165	-10762	2234
14	4761	8374	7904	10727	14375
15	2897	14078	-5945	-3216	1636
16	20354	-10762	-5133	5894	2845
17	8021	-7160	10153	-3581	7498
18	1747	-3572	-8747	21621	14054
19	7631	1608	-8888	2778	12518
20	10893	13976	3339	6689	-11601
21	7958	24421	15064	-14485	795
22	-19013	20319	15065	12309	6725
23	-3356	14533	498	9132	-1070
24	22372	6674	-5593	-7807	2231
25	17496	3465	14129	-6401	-7937
26	10957	-11012	5450	8484	19068
27	8076	4615	132	-13695	17504
28	-7926	-6788	19762	13593	5532
29	3638	11875	23191	6498	416
30	10807	25123	27312	12296	21037
31	11303	22562	-3922	6445	9484

**Table 3.C.7 - Gain VQ tables for narrowband CELP coder (subframe 10 ms)**

scale\_factor = 2<sup>14</sup>

Index	Mode 0		Mode 1		Mode 2		Mode 3	
	Codeword		Codeword		Codeword		Codeword	
	0	1	0	1	0	1	0	1
0	96	281	368	936	1306	2931	3022	6368
1	845	8737	5856	5202	7476	4118	9583	2479
2	871	6180	986	9557	5780	4373	7846	2595
3	4506	9494	6509	10788	12130	3716	11850	2593
4	709	4694	967	7335	5905	12020	1480	10720
5	706	10947	8873	6200	9709	3898	12085	5902
6	3881	6269	1282	15500	8719	6102	9043	4222
7	1279	16019	10471	17218	14796	9719	14434	3000
8	716	3313	1125	5115	1100	9653	6162	3034
9	3065	9462	6809	8187	9908	6467	11563	4303
10	3005	8360	4195	9873	3435	8728	10165	4356
11	7511	9871	8713	9682	15981	5240	12613	3700
12	2311	4285	3743	7146	2705	17876	7575	5897
13	3107	12901	11690	6429	11973	5831	14900	5233
14	5864	8939	2797	22418	9299	8816	9774	8479
15	5868	19081	19571	9804	23501	8763	17281	6495
16	557	2243	772	3424	1928	6882	6070	5420
17	2011	9153	7457	5735	9521	5312	10837	4086
18	1825	7239	925	12834	7829	5600	9652	3576
19	5558	11154	7986	12868	13957	4168	12966	2753
20	1952	6072	2791	9451	9505	11808	4916	12059
21	594	12315	10061	7133	10872	5904	12947	6324
22	5426	6526	4846	15527	8007	8422	9093	5935
23	3158	22812	14455	15456	13886	14573	15696	4124
24	2101	2949	3656	3962	1060	13149	7980	4211
25	2963	11141	8574	8122	11217	6998	12292	4651
26	4469	7696	4343	12456	5831	7830	10306	5719
27	10380	14515	12104	10043	17088	8137	13752	4874
28	3604	4931	5513	8467	8486	19002	7582	9357
29	5220	13359	14517	8918	13544	7879	14949	8191
30	7338	7999	7715	21994	11773	8537	11595	9334
31	9446	26080	20331	24179	22611	14536	23132	8850
32	493	1250	1037	2067	3707	4104	4637	3690
33	760	9893	5958	6948	8662	4462	10800	2566
34	764	7523	1116	11096	6600	5919	8773	3183
35	4708	10373	7971	10992	12729	4786	11969	3522
36	1639	5099	2369	7838	7255	14492	3708	9061
37	1750	11533	10381	5297	10847	3952	12039	7252
38	3557	7381	2307	18621	8907	7296	9639	4995
39	2017	18755	14314	20305	16349	12224	14379	4154
40	1426	3783	2609	5638	2162	10979	7052	3907
41	3637	10224	7307	9336	10293	7768	11653	5121
42	3775	8693	4999	10883	4576	10169	10873	5023
43	8861	11649	10093	10929	18764	6195	13094	4351
44	2766	5362	4448	8137	3795	25643	8457	7271
45	4064	15446	14209	6152	13187	6012	15496	6400
46	6091	9979	3392	29021	10528	9754	10810	7304
47	9807	19395	26110	13673	28108	13457	19747	6402
48	1323	2190	2146	3452	4667	6562	6163	7827
49	2185	10203	7695	7195	10418	5010	11242	3436
50	2308	7885	2734	13459	7471	7123	10368	3358

51	6892	11954	9775	13841	14328	5757	13550	3585
52	2820	6600	2876	11267	11101	13711	7341	20130
53	1252	13617	11657	8071	11569	4879	13531	7617
54	5622	7800	7343	16406	7996	10345	9750	6737
55	2966	28962	19660	15528	17861	17087	17267	4304
56	3336	3610	4366	5732	3443	13095	8562	5111
57	4297	11656	10011	8757	12345	7093	12939	5312
58	4812	8669	6094	12674	6487	9379	11206	6105
59	15465	16210	12367	12693	19647	10249	14038	6109
60	4959	5156	5834	9531	13310	23800	9634	11944
61	7145	14652	16105	12086	15149	7295	17026	9771
62	10026	9255	12981	28541	12619	10600	13830	10845
63	19180	26086	27798	24140	25784	23526	22790	15754

**Table 3.C.8 - Gain VQ tables for narrowband CELP coder (subframe 5 ms)**

scale\_factor = 2<sup>14</sup>

Index	Mode 0		Mode 1		Mode 2		Mode 3	
	Codeword		Codeword		Codeword		Codeword	
	0	1	0	1	0	1	0	1
0	391	1069	961	2363	1259	3029	1668	2677
1	5628	8023	8086	10164	7429	3747	9023	1963
2	829	8075	1156	9804	1329	10359	7434	2339
3	4140	12001	12362	4422	12439	2804	13857	2348
4	2737	7627	4699	4094	5954	3003	2904	5219
5	6954	10082	7567	4483	10677	3109	12795	2244
6	605	11822	1718	15709	8353	8941	10352	4793
7	1641	23690	19114	6013	17744	4540	18047	3357
8	915	5938	1056	6890	1308	6536	5974	2422
9	6832	11666	11421	10629	10975	4609	10554	2002
10	2738	9942	3921	9479	3023	17711	7977	6447
11	5245	16314	14333	7447	14874	3557	16346	2783
12	4451	4950	5972	9629	7137	5548	6258	10555
13	9248	11696	10037	7031	12123	6395	13598	3833
14	1108	14505	2485	24553	13258	7967	12066	4764
15	9896	17912	19492	12937	25939	9249	23084	5066
16	1237	4207	2899	3481	3041	2522	4664	2572
17	5410	9776	7877	13160	8970	3281	10241	3399
18	580	10144	1026	12705	5216	13404	9338	3486
19	4300	13531	14297	4156	12858	3999	14940	2521
20	4145	7719	6914	6858	4521	5187	2490	9940
21	9812	9792	10276	4339	12021	4928	12790	3475
22	2725	11800	7859	16648	10923	12399	11570	3269
23	7439	23129	23881	7023	23428	5987	18230	5831
24	2542	5383	4367	6669	4047	10576	7302	4612
25	6683	13086	12151	16983	10998	7464	11722	2116
26	4040	9861	5580	12745	5804	25533	11106	6224
27	9643	14815	15398	10431	16073	5571	15755	4799
28	7251	6897	8435	7460	9627	5713	4223	20978
29	13087	12577	12206	7124	14679	5729	14597	4329
30	1583	17435	11352	23655	17201	9850	13704	6495
31	17862	21631	21928	18475	23768	15557	28103	5726
32	1086	2604	1268	4528	2766	4553	3367	2373
33	6469	8905	9508	11003	8530	4977	9815	2508
34	1789	9034	2644	11119	1703	13131	8312	2981
35	5401	12323	13316	5662	13683	3154	14272	3294

36	3326	8775	5811	5419	5951	4597	5456	6165
37	8041	10614	8917	5781	11722	3697	13293	2960
38	1419	12776	2697	19079	10012	9407	11288	4662
39	4159	28738	19686	9274	20011	5339	19885	4449
40	1901	6811	2751	7956	3462	7679	6681	3515
41	8160	12688	13169	12490	10860	5906	11013	2739
42	3468	10893	4638	11131	8584	17996	9664	7087
43	6930	18050	16591	8026	16056	3835	16828	4183
44	5196	6583	6736	11176	8007	6954	9638	13184
45	10702	12680	10952	8527	13506	6436	13697	4894
46	3165	15136	5373	28942	14178	10526	12509	5786
47	12581	20934	24291	12982	29840	8673	25353	8792
48	2697	3664	2949	5624	4445	3303	5300	3932
49	5964	10859	10292	13748	9922	4299	10959	3752
50	1801	10841	3357	13330	7808	11753	9382	4964
51	5642	14198	16184	5465	14082	4498	15360	3579
52	4643	8881	7335	8619	5797	6905	2593	14049
53	11948	8660	11005	5798	13093	5209	12874	4452
54	2959	13080	7186	20738	14391	15928	12077	3842
55	11554	27975	29648	11305	21401	8985	20987	8017
56	3619	6425	5338	8052	6216	9336	8459	4302
57	7612	14974	16002	19976	11916	9009	12211	2850
58	4763	10908	5258	15290	16067	25296	11966	8220
59	12680	16015	16309	14492	17912	6929	16506	6487
60	8164	8664	9310	8810	9607	7222	9287	25100
61	18715	14800	12917	9000	15344	7661	14840	5640
62	3702	19285	17523	27956	19372	13134	14961	9721
63	21995	27675	27604	24296	29648	18568	30284	10468

Table 3.C.9 - Filter coefficients for adaptive codebook interpolation (for 8 kHz sampling rate)

scale\_factor = 2<sup>15</sup>

Coefficients.	Value
0	32767
1	31236
2	26910
3	20534
4	13174
5	5989
6	0
7	-4097
8	-6045
9	-6016
10	-4528
11	-2286
12	0
13	1768
14	2700
15	2750
16	2100
17	1068
18	0
19	-822
20	-1246
21	-1253
22	-942

23	-469
24	0
25	346
26	509
27	498
28	363
29	176
30	0
31	-123
32	-181
33	-180
34	-138
35	-72
36	0

### 3.C.2 LSP VQ tables and gain VQ tables for the 16 kHz sampling rate

Table 3.C.10 - Table for the first stage, lower part of the wideband VQ ( $\text{lsp\_tbl}[0][\text{Index}][i]$ )

scale\_factor =  $2^{15}$

Index	Codeword				
	0	1	2	3	4
0	1143	1850	2663	3727	5954
1	1190	1963	2858	4886	6208
2	773	1128	1666	2979	6276
3	1443	2587	4221	5995	7901
4	1174	1842	2711	3795	4538
5	584	1103	2971	4905	6525
6	715	1124	1896	4198	6235
7	2006	3414	5142	6644	8172
8	1295	2233	3107	4179	5193
9	1400	2383	3809	5381	7268
10	1009	1540	2200	3365	6642
11	729	1567	3993	5900	7466
12	827	1284	2127	3769	5157
13	1066	1763	2925	5187	7259
14	1146	1767	2500	4182	7030
15	1999	3874	6449	8025	9364
16	858	1641	3267	4316	5789
17	1361	2327	3246	4400	6554
18	881	1269	1925	2947	5442
19	1696	3106	4766	6030	7295
20	990	1518	2218	3231	4121
21	942	1917	3716	5212	6448
22	525	871	2059	5076	6670
23	1198	2899	4793	6665	8389
24	1528	2668	3599	4566	5630
25	1501	2692	4132	5353	6512
26	904	1378	1924	3294	7600
27	722	1671	4608	6828	8449
28	984	1561	2432	4460	5426
29	748	1182	2763	6170	7946
30	926	1387	2016	4515	7881
31	1564	3114	5408	7424	9031

**Table 3.C.11 - Table for the first stage, lower part of the wideband VQ (lsp\_tbl[1][Index[]])**  
 scale\_factor = 2<sup>15</sup>

Index	Codeword				
	0	1	2	3	4
0	7472	8934	10015	11175	13217
1	8233	9479	10725	12434	14154
2	7335	8745	10993	12330	13616
3	9920	11062	12168	13485	14666
4	6489	9325	10463	11799	13238
5	8528	9865	11277	12892	14502
6	7269	9858	11109	12483	14120
7	10116	11670	13053	14259	15501
8	6483	8177	9776	10950	12794
9	9462	10452	11420	12647	13795
10	7358	8940	9940	12541	13866
11	9136	10921	12617	14312	15907
12	5832	8075	11126	12420	13807
13	8475	10059	11736	13415	15033
14	6595	9134	10637	12879	14612
15	9831	11871	13624	15369	16869
16	7177	8241	10290	11834	12840
17	8737	9749	11079	12313	13420
18	8125	9201	10510	11858	13194
19	8977	10622	12235	13805	15243
20	6082	8849	10203	11627	14131
21	7740	9703	11425	13202	14809
22	7728	9142	10867	12826	14627
23	10868	12391	13674	14779	15831
24	6481	7886	8959	11538	13726
25	8873	10533	11925	13167	14460
26	7523	9139	10338	11718	14253
27	9226	11198	12943	14965	16628
28	6355	7787	10010	12579	13995
29	8071	10160	11937	14095	15765
30	5708	9623	11333	12828	14479
31	10737	12670	14292	15797	17066

**Table 3.C.12 - Table for the second stage without interframe prediction, lower part of the wideband VQ (d\_tbl[0][Index[]])**

scale\_factor = 2<sup>19</sup>

Index	Codeword				
	0	1	2	3	4
0	-2586	3968	2886	204	5966
1	1012	1572	-2381	4464	887
2	1167	1363	3292	853	1003
3	2042	3034	7092	3956	5132
4	4979	4760	-9300	-134	10831
5	-2905	-2776	674	2514	7577
6	3569	4039	2941	-3478	-1578
7	-1710	-548	3294	5376	3153
8	2186	3100	660	-1778	6145
9	-957	-795	3033	11341	-3589
10	4970	-4147	6490	4298	-572
11	417	7710	3470	273	-1453

12	-7	1046	-4507	2892	6555
13	-714	-1568	-2967	7020	4636
14	3772	3302	451	2446	-4140
15	2578	4384	9710	5358	-1217
16	-2532	-4202	7133	1880	5630
17	4804	5348	-3984	7228	3642
18	1111	3491	-1114	-430	1191
19	2678	3617	7802	-1392	-192
20	3806	8452	-2869	-3324	1352
21	3318	-6471	-1011	3546	7725
22	-3433	9091	3886	-6341	3227
23	2265	2960	3294	6694	-768
24	591	-253	91	-2479	9720
25	817	1668	4829	2207	-5378
26	6623	-2303	-3276	7908	1206
27	-4181	2006	12396	-134	3092
28	1687	1536	1396	4260	7281
29	-1745	-5444	996	9423	3266
30	6297	6104	5194	-160	-4582
31	-2136	-1987	7585	5756	-1961
32	738	1096	5525	-4589	4666
33	-645	-1470	265	5541	-1617
34	-2338	-431	7307	-347	-301
35	5627	6740	3288	2521	2080
36	10415	8027	-9823	-2155	3777
37	1325	-1970	-95	350	4293
38	3441	4597	5252	-8180	46
39	-6571	1403	-214	6608	4145
40	5454	3212	-1819	-6519	5840
41	1182	419	-1725	9004	-5253
42	7910	-2008	3452	-1016	4758
43	-3129	5539	2500	5557	-535
44	-349	9416	-3637	930	7559
45	1518	-2084	-9645	11648	4557
46	10192	415	-1341	1486	-5110
47	-2889	-1897	16428	3547	-5875
48	-3242	-265	6994	-2384	10971
49	5179	8334	-3396	4486	-3902
50	6665	2127	-3112	-489	1744
51	2773	6348	11843	-2666	-5595
52	8771	10848	112	-4867	-2993
53	-2051	-2805	-5007	3121	13573
54	4260	7545	5957	-4577	6355
55	1229	1851	2712	10833	3892
56	2501	4189	-323	-8917	13459
57	1756	3967	4998	7420	-8324
58	11279	-2442	-5259	2201	6640
59	-9527	9310	9443	2568	-1220
60	2819	5071	951	2904	13584
61	-3158	-4143	2986	14946	13769
62	-241	15823	5566	-2882	-8545
63	-4884	-7481	12990	9516	2628



**Table 3.C.13 - Table for the second stage without interframe prediction, lower part of the wideband VQ  
(d\_tbl[1][Index[]])**

scale\_factor = 2<sup>19</sup>

Index	Codeword					
	0	1	2	3	4	
0	-2443	7932	-738	6162	-4136	
1	-677	2914	7398	3994	-3822	
2	-880	11860	48	-2227	3747	
3	4653	-2433	-645	1289	5774	
4	-5992	6932	4819	-4768	8843	
5	4269	2134	4339	-913	-4016	
6	263	2727	191	-150	822	
7	3804	4965	330	3122	-3902	
8	8235	-4048	447	3751	-2411	
9	-10260	3203	8066	6308	472	
10	33	2151	-404	6877	6237	
11	6166	884	1015	-712	240	
12	204	-3409	1364	5262	1605	
13	1922	-2206	5780	510	1024	
14	1655	4433	8847	-1457	4321	
15	5754	7462	709	-1632	252	
16	3275	830	-4011	5372	166	
17	3562	7869	5134	71	-10033	
18	-2276	8828	-7269	3063	4892	
19	9444	-1435	-8008	3165	6199	
20	-2110	3531	8135	5994	6521	
21	11605	-5656	1614	-2073	3288	
22	2976	3726	3134	3191	2662	
23	10317	4135	-865	4701	3892	
24	2399	469	2879	9433	-256	
25	-2899	7059	5111	-1257	-1483	
26	-643	-1379	-6800	4876	10284	
27	3759	4372	-4707	714	6033	
28	-3760	-4921	2991	3895	7671	
29	71	-7924	6377	8260	1392	
30	7179	3685	2958	-4074	7007	
31	11439	2351	2992	-1922	-6670	
32	2341	8850	-929	8357	1839	
33	-2267	6105	13450	-523	-4823	
34	1194	6805	-3370	-6589	7559	
35	5727	-6273	160	4685	10448	
36	-4980	-363	9224	-1874	4153	
37	4819	-1241	10584	-1704	-4963	
38	-3671	1662	2849	2825	1633	
39	7957	1782	-753	8559	-7203	
40	5991	-4262	7252	5705	-6894	
41	-6883	5544	-1600	9315	2474	
42	-1126	635	919	-748	7096	
43	8914	1872	-3524	-5301	3577	
44	-5824	-2532	1377	10418	2894	
45	3915	-5883	8429	374	5885	
46	2308	7252	4975	-7637	1080	
47	4649	10765	-5746	1833	-2783	
48	4233	407	-10181	10870	1626	
49	6270	10391	8420	2016	-1259	
50	-6408	5141	-729	1832	8457	

51	5277	-5990	-2125	10813	4036
52	3332	5508	1806	2232	12391
53	7024	-1108	7624	-7499	1621
54	8546	280	6937	3987	1828
55	10174	1977	-4947	1058	-2864
56	-2769	315	4868	12245	-6828
57	-7240	11848	8060	3016	-3061
58	6405	-892	-5941	-3827	13649
59	9181	8383	-9470	-2700	6218
60	-5669	-1140	3997	10502	15094
61	-2045	-3486	12851	5447	-72
62	810	-2441	4093	-5399	11607
63	11810	9801	-1199	-7897	-3987

**Table 3.C.14 - Table for the second stage with interframe prediction, lower part of the wideband VQ  
(pd\_tbl[0][Index][i])**

scale\_factor = 2<sup>20</sup>

Index	Codeword				
	0	1	2	3	4
0	-423	8162	3591	1027	-1393
1	1510	-976	2127	-1862	5588
2	2712	1163	819	1966	-1098
3	1486	112	-2991	9568	-622
4	1310	1001	4839	7098	-5310
5	-1708	-4724	-1417	5607	9148
6	11270	12024	-9823	-2764	5319
7	-752	-1163	2297	5558	2639
8	3631	5562	10288	-3185	6849
9	-593	5589	3305	-8302	17200
10	6747	228	-4789	2016	3751
11	4539	5167	1948	9513	983
12	3071	22303	16697	3399	2876
13	2795	5989	1847	3168	6673
14	6878	6170	-824	2331	-7044
15	-2295	-6294	14996	1112	2109
16	2489	5774	5387	-8274	422
17	3293	6658	-3129	-4113	5510
18	664	18193	-504	9089	-5500
19	7492	6943	-7597	12275	-4567
20	2269	5895	7606	115	-7688
21	1370	1789	-6680	7488	7975
22	9523	12672	5301	-562	8445
23	1098	394	7784	-681	-368
24	-6107	6257	5386	-2910	5967
25	2687	3503	3902	2818	15370
26	7294	-7244	3126	1692	7991
27	55	447	153	16418	-8055
28	4039	4046	16811	2263	-6952
29	1351	1697	2436	10875	9000
30	5744	13856	-963	-4810	-3509
31	-4506	-709	5657	3139	7615
32	7301	2845	3243	-3558	-1997
33	-2526	-3081	6945	-4163	12184
34	2301	7126	-4694	3079	928
35	-1226	-3968	-3739	17350	7246

36	4379	5219	5428	9210	-12915
37	-1230	759	-3616	-536	11064
38	1094	8486	-12784	-556	15950
39	-2749	-4054	5048	12674	283
40	-2930	4167	14576	-11640	1847
41	4466	-561	-3958	-6857	19397
42	16820	3128	-3428	-6008	2652
43	2865	5115	8956	15768	-1583
44	-7399	25928	2084	-1257	3236
45	4036	4005	10262	6870	4940
46	11239	10153	6019	6280	-1579
47	-599	-6625	21393	13356	-4515
48	4022	12032	1243	-16378	5832
49	2807	2194	1544	-7964	8298
50	-7914	7848	-476	9582	1789
51	16219	2978	-4132	12303	5813
52	5665	8958	11103	-7796	-6245
53	3759	-5038	-9499	8712	16729
54	9512	16283	-1075	10111	12046
55	-5294	3076	10980	5078	-2573
56	-9705	8933	17367	5422	10101
57	-4006	-2796	-530	5854	18467
58	10783	-5541	5154	7238	-3177
59	2381	1238	2707	24266	2222
60	-8717	14128	16220	7269	-13329
61	2432	4218	6801	15890	16440
62	22723	16656	2318	-4151	-12759
63	-4697	-7096	9352	11102	11601

Table 3.C.15 - Table for the second stage with the interframe prediction, lower part of the wideband VQ  
(pd\_tbl[1][Index][])

scale\_factor = 2^20

Index	Codeword				
	0	1	2	3	4
0	9444	1828	-1379	664	1434
1	4316	-2360	7487	2691	232
2	-11548	800	13052	5765	8617
3	641	2689	-924	8929	-203
4	4737	814	2443	-6190	9333
5	-3842	-3076	909	7521	8486
6	1278	7620	3374	2521	-7664
7	-1781	4195	2105	-819	3093
8	15423	-3892	-9834	10664	448
9	9649	-731	3882	11712	7743
10	9704	3535	14839	4161	-9247
11	-5719	-2397	7548	9921	-2687
12	6872	11224	1951	-5952	-575
13	-11306	6686	-2318	8593	12745
14	-8832	10229	3631	5130	-156
15	2783	-624	-6826	3757	11469
16	8302	7502	-5167	6202	-7792
17	6088	-5881	94	12905	-2917
18	-113	-6158	15625	-3577	6335
19	2693	3359	-10442	15397	3406
20	13212	3987	-4853	-9899	13329

21	9749	-10490	2505	2257	9095
22	-1472	7684	11857	17536	-1232
23	1323	13374	-1514	-11289	12611
24	1977	-11506	-5158	16140	11268
25	13178	-4523	8932	-9732	5223
26	-3782	6838	11830	-3381	-598
27	-13884	-850	3082	23106	4970
28	19156	4585	235	-2255	-7658
29	4708	11415	1083	5521	11720
30	5705	12002	9256	5396	1247
31	953	10260	-6028	2010	1594
32	12552	-4987	5014	3963	-5039
33	6338	-10086	16190	8226	-4857
34	-8543	4389	3591	-3744	12922
35	-4168	13190	-3595	13328	-857
36	1596	6948	13535	-8044	11255
37	-2168	-13544	9163	11309	6468
38	-701	19627	8271	1945	-13273
39	473	3278	8118	4081	9567
40	14675	6431	-12422	276	4922
41	17014	5380	10270	1179	8012
42	7791	2301	9203	-6660	-3391
43	-6446	2897	19837	5364	-8208
44	16528	17154	767	-17143	-891
45	-12809	-2839	-419	10518	27038
46	-15472	21829	13893	7094	3026
47	-2340	10375	-10220	1851	13771
48	15023	9591	-166	12676	107
49	5362	-5174	6927	27166	-3086
50	6180	2204	18574	10349	9697
51	2408	8231	1457	21820	11880
52	1460	-408	-1465	-6386	27823
53	-427	-8411	6757	1010	18652
54	1356	5111	6358	15292	-14303
55	-2694	20171	5540	-3881	4060
56	11474	1094	-9006	9698	21887
57	23666	-7657	-1296	-414	5498
58	3539	11764	23025	-12228	-1961
59	-10130	-6581	24949	21344	10365
60	15803	16471	5757	352	-12586
61	-1231	13685	11266	7257	22928
62	7389	25109	10922	15030	2091
63	7499	21420	-7188	2244	1319

**Table 3.C.16 - Table for the first stage, upper part of the wideband VQ (Isp\_tbl[0][Index][])**  
 scale\_factor = 2<sup>15</sup>

Index	Codeword				
	0	1	2	3	4
0	14931	17260	19135	20731	22466
1	16130	17759	19272	20469	21832
2	14608	16251	18746	20647	22478
3	16753	18362	20121	21770	23327
4	15552	16841	18257	20245	22266
5	16539	18104	19687	21273	22868
6	15101	16611	18049	19528	21357

7	17953	19286	20727	22128	23608
8	15756	17140	18998	20941	22707
9	16502	17762	18956	20549	22556
10	14473	15588	17705	20283	22335
11	17433	18714	19989	21305	22774
12	15968	17244	18508	19837	21410
13	15937	17691	19538	21274	22907
14	14948	15926	17038	19218	21851
15	18442	19887	21423	22906	24280

**Table 3.C.17 - Table for the first stage, upper part of the wideband VQ ( $lsp\_tbl[1][Index][[]]$ )**

scale\_factor =  $2^{15}$

Index	Codeword				
	0	1	2	3	4
0	23968	25814	27338	28667	30395
1	24526	26064	27151	28531	29485
2	24511	26318	27711	29022	30694
3	23743	25259	27101	29078	30925
4	23698	25736	27594	29289	30968
5	23930	25559	26778	28291	29305
6	25175	26565	27871	29456	31049
7	22685	24448	26843	28939	30843
8	24480	25808	27105	28900	30888
9	25107	26620	27630	28864	29742
10	24549	26242	27882	29551	31152
11	22947	25325	27402	29152	30891
12	24232	25919	27608	29356	31064
13	23232	24879	26331	28014	29310
14	25418	27091	28522	29944	31227
15	23582	25139	26655	28449	30535

**Table 3.C.18 - Table for the second stage without interframe prediction, upper part of the wideband VQ ( $d\_tbl[0][Index][[]]$ )**

scale\_factor =  $2^{19}$

Index	Codeword				
	0	1	2	3	4
0	-2560	2297	6975	798	-1926
1	-2735	3872	-4564	8120	5746
2	4373	947	-955	-914	1315
3	-908	4401	2667	-1287	1377
4	-2268	5853	10025	-6673	-163
5	1631	-28	2810	1904	-776
6	4776	6142	-2947	949	1415
7	-2833	-2024	-1112	4600	7474
8	-3864	2739	1593	4064	2570
9	6155	-1553	5820	4596	3997
10	9695	1441	-7817	1020	3792
11	-7730	2722	1634	1598	10322
12	3962	312	9630	2830	-2605
13	-2899	2110	2715	8973	-2241
14	12594	3871	-1360	-3011	-1569
15	-320	2494	3735	6371	9300

16	-757	3321	8918	3096	3193
17	2877	2783	2616	3357	3051
18	7251	-1824	3237	-1791	-634
19	-1369	9286	-1718	2204	129
20	4206	8370	4473	-668	4288
21	-2354	1560	6156	-856	8537
22	8254	4987	-2811	-8196	8943
23	3138	-3705	-698	2210	5013
24	-6427	-949	6278	3290	2970
25	8151	2254	-2434	6749	-5431
26	10346	-3913	-5767	8392	2997
27	924	1305	-2580	-1451	7512
28	2445	6006	2659	5765	-2794
29	-925	-3269	4200	6055	1733
30	5123	4744	7229	-3713	-3367
31	-6848	-4226	1631	11032	8429
32	-4902	2443	11540	4342	-6701
33	3104	1331	-2367	7153	1612
34	4133	2777	577	-437	-2305
35	2507	7589	2359	-615	-3543
36	-2694	9975	2067	-5305	6002
37	-984	-139	1242	1991	2742
38	6947	9379	-9670	-2294	2472
39	2133	-6472	35	10186	4359
40	-7413	6480	4407	2173	1195
41	6511	-5078	1092	4486	-556
42	11374	-5247	-2568	-352	6938
43	-631	8186	-5089	571	10441
44	8932	-2634	5515	968	-7191
45	-6319	-2454	6844	14208	63
46	9519	4144	3772	2319	-76
47	7155	-182	1678	11743	12984
48	1535	6830	7371	8848	3087
49	7643	2726	-1115	3775	7386
50	5540	1143	5004	-5812	4505
51	-1943	11191	6790	166	-1802
52	12402	12273	2874	-1611	2355
53	-976	-4154	8899	-646	3132
54	4645	7836	-96	-7035	547
55	387	-7025	3243	1364	9505
56	-12238	2499	6829	6815	5606
57	5210	-2761	4878	11656	-3536
58	3995	-5308	-7682	5622	15238
59	3851	777	-876	-3333	15812
60	3980	3863	6117	4229	-9652
61	408	-9600	9596	6683	2041
62	11040	9818	2056	-4210	-9529
63	-6364	-3140	3061	12611	19828

Table 3.C.19 - Table for the second stage without interframe prediction, upper part of the wideband VQ (d\_tbl[1][Index][])

scale\_factor = 2<sup>19</sup>

Index	Codeword				
	0	1	2	3	4
0	1422	1178	933	494	245

1	2062	-50	1059	-1613	8136
2	6145	-3291	537	1902	636
3	5415	1583	3075	-34	-5337
4	3988	3443	4127	3416	2860
5	-1029	557	-693	4690	2872
6	5403	1770	-3950	1666	1720
7	-4164	3395	5114	1181	2743
8	571	6317	1269	-501	-1899
9	-1421	5826	-3067	449	6809
10	-194	-3079	5512	3052	1681
11	-66	233	2800	7851	-3319
12	6361	3473	-317	-4739	430
13	-4976	-1191	2889	10022	8119
14	3194	-4129	-3960	7152	6974
15	-3360	5872	10195	3113	-5435

**Table 3.C.20 - Table for the second stage with interframe prediction, upper part of the wideband VQ  
(pd\_tbl[0][Index][])  
scale\_factor = 2^20**

Index	Codeword				
	0	1	2	3	4
0	-1203	944	1720	3038	3535
1	5458	5668	869	7066	-6563
2	3802	4199	2277	-1104	839
3	1430	-1000	-4780	1722	12432
4	-3481	4866	15357	-7220	5939
5	-39	-2956	9749	3445	1807
6	7477	17963	-7474	-6183	-826
7	4002	5024	-6877	3408	2368
8	-8652	7341	-6437	11935	8628
9	-10816	6463	16189	4581	485
10	17072	2029	7080	4366	7505
11	7739	8418	-3875	-10078	9607
12	10534	-3599	12877	-102	-3909
13	-7772	-6321	8296	3489	14360
14	14615	2173	-10507	590	4864
15	-1143	-5842	158	11118	3919
16	5949	5520	2089	7716	7633
17	-2745	-3749	6564	27658	945
18	11687	-4594	216	-6389	7029
19	11556	444	-13192	-4181	21192
20	-873	7995	10019	-2129	-5377
21	-369	-7926	20397	8882	-7803
22	607	15613	8257	12527	3173
23	4507	-3390	-12943	14817	7604
24	-4219	10154	2618	-1573	2973
25	-1579	1587	8487	10408	-3916
26	8905	-8575	-663	8312	4636
27	3444	17398	4070	2	10376
28	2063	7578	24687	4928	-5119
29	-245	3940	9563	14485	11203
30	12855	7088	695	-5315	-4689
31	-10581	-7648	-1577	16167	14056
32	-3105	10468	-1419	6311	-852
33	14162	4204	-7633	11235	-5516

34	9829	-1206	427	868	-594
35	7368	-7729	854	5349	19156
36	865	1591	5717	-6359	13827
37	5506	-9137	11190	-1984	8665
38	2610	21044	4984	294	-8907
39	5334	14543	-12802	6947	8133
40	-4716	10487	-8174	-1002	16152
41	-14240	1302	8133	16156	3859
42	14699	2379	8972	19069	3086
43	-2564	16674	3571	-14974	9552
44	12229	3721	8738	4077	-13926
45	-9855	6261	4855	1123	13212
46	24316	9325	-5634	-5552	5707
47	926	3508	-4145	18955	-1468
48	6134	7499	13623	3666	5145
49	8844	-7849	4340	12907	-5543
50	9855	6391	10506	-11434	1879
51	362	-983	-13635	6571	30326
52	3270	15666	19579	-10066	-10388
53	1502	-10344	15691	14109	6226
54	13715	15044	2230	6362	1724
55	15769	1574	-8019	16234	14189
56	-14732	20228	5382	1847	1919
57	-749	6117	11215	16353	-16089
58	23579	-7536	-1287	6310	-1700
59	14786	17464	1283	7296	20111
60	14265	20437	23019	9216	7590
61	712	7336	11183	11507	27807
62	25661	16635	5441	-2706	-6497
63	-492	-7793	-2651	31751	22163

**Table 3.C.21 - Table for the second stage with interframe prediction, upper part of the wideband VQ  
(pd\_tbl[1][Index[]])**

scale\_factor = 2<sup>20</sup>

Index	Codeword				
	0	1	2	3	4
0	6381	-7650	4834	7646	932
1	-4305	8028	-987	6332	2983
2	-5563	737	7443	2389	1608
3	1154	3384	5233	10481	-7363
4	14733	4933	2231	1558	-679
5	-3648	10342	3735	-4610	10841
6	71	-2562	-1188	4491	13847
7	4948	10669	3642	-4753	-2807
8	4454	-5104	9349	-5458	8376
9	2866	1850	1176	764	635
10	-7310	251	9250	17609	8502
11	-4875	17621	12117	3826	-3023
12	17912	-5764	-7648	-132	8401
13	6281	11385	-8672	-577	4955
14	4289	-2614	-7635	14525	4348
15	2977	1457	15505	5	-7012



**Table 3.C.22 - Gain VQ tables for wideband CELP coder (subframe 5 ms)**  
 scale\_factor = 2<sup>14</sup>

Index	Mode 0		Mode 1		Mode 2		Mode 3	
	Codeword		Codeword		Codeword		Codeword	
	0	1	0	1	0	1	0	1
0	599	1298	921	1797	1035	4241	3783	2563
1	3784	8406	4936	4301	7553	3257	10037	3498
2	581	7957	864	7173	2967	3207	6451	1634
3	4942	10321	3850	13075	10901	3097	11975	1865
4	1300	6724	2314	7483	633	9042	6235	3533
5	6920	7215	7528	7618	12395	6988	11098	3753
6	513	10942	4555	10181	6634	6057	9183	3336
7	1201	14544	12644	8988	16328	3625	14991	2517
8	2061	5070	2986	4312	3984	7127	5117	1456
9	3884	9804	7037	4080	9791	5902	10326	1756
10	2461	9152	1032	10962	6958	4985	8423	1702
11	4957	14317	7835	16139	12333	3345	13414	2265
12	3262	7068	5031	7939	2677	15607	6757	5391
13	7729	8838	10302	9184	14873	9446	12594	5533
14	2628	12220	6918	12161	8031	6654	10097	4761
15	5519	17797	10732	18523	20272	3969	18268	4229
16	578	4716	957	5617	3512	4340	2775	5249
17	4880	9642	6512	6810	9787	3094	10657	2882
18	583	9435	2089	8670	5721	2457	8336	2738
19	6077	11871	2134	16662	11813	5534	13085	3145
20	2025	8112	3683	9109	3648	9315	7153	3994
21	6078	10223	6776	10350	11161	9303	11618	3622
22	2063	11531	5198	11228	7688	9861	8988	5010
23	1356	21967	20897	8812	18483	7804	17195	5592
24	3273	5001	3963	6995	5995	4324	6806	3000
25	4036	10986	11311	4569	11124	5030	11677	2654
26	2718	10100	3197	11103	8130	3983	9377	1700
27	8055	16469	2497	29313	13682	6394	14588	4414
28	5235	6729	5934	8224	8616	14962	7609	7663
29	8609	11771	13311	13135	19242	11679	12796	6707
30	4352	12357	9255	12338	9999	8627	11878	4405
31	11833	21206	15036	28428	23752	7325	20724	3765
32	1790	4058	2329	3522	1095	6554	5370	3292
33	4381	8633	5799	5809	9046	3700	10525	3564
34	1587	9070	712	9386	5436	3537	7507	1546
35	5464	11198	5621	15640	11831	4584	12685	1954
36	2114	7225	3309	8079	1663	10700	5032	5233
37	6342	9273	8588	8857	12567	8603	11014	4705
38	1049	12492	5986	9462	7326	8023	9579	3823
39	3465	18246	15150	6736	18316	4608	16558	3172
40	2105	6254	4042	6080	5767	8387	5723	2219
41	4167	10407	9659	5438	9854	6755	11015	1839
42	3145	9183	1125	13013	8103	5217	9185	2491
43	7151	14299	5957	21510	15111	4637	13715	3115
44	3977	7482	4620	8888	2714	23350	8086	5592
45	8389	10349	11301	10090	16607	11483	13967	5190
46	3110	14265	6781	14406	8558	7235	10273	5808
47	7255	21228	15508	21007	25117	4387	21764	6654
48	1433	5237	2623	5611	4727	6014	3007	10841
49	5360	9203	7340	6003	9910	4686	11079	2581
50	1704	10340	2353	10036	6966	4041	8503	3675

51	6913	11744	2367	22911	13097	5213	13125	4655
52	2805	8173	3744	10150	5318	12239	7923	4231
53	7179	10672	7985	10213	12904	11892	12798	3945
54	3314	11295	5753	11999	8800	11173	9491	6396
55	3660	28526	21722	13636	20381	6825	18735	7428
56	3566	6378	4921	6642	5664	6616	7668	2787
57	4754	11395	10095	7628	10784	6714	12416	2964
58	3440	10239	4533	11966	8944	5064	10021	2579
59	13254	17336	8867	24609	15539	6973	14814	5391
60	5003	8520	6950	9494	14154	17320	10893	8085
61	11698	12315	14671	15807	18446	20359	15172	8048
62	5358	12793	10619	13654	10920	7946	11769	6293
63	13834	29363	23354	21673	24703	15306	29030	6036
64	1217	3385	970	3773	2118	5206	4362	4087
65	3884	9153	6109	4402	8589	2983	10093	4046
66	1233	8428	1157	8314	4418	3302	7110	2223
67	5514	10509	4160	14769	11371	3970	12164	2471
68	1446	7421	3142	7222	2312	8066	6436	4471
69	6677	8470	8489	7821	13225	7445	11390	4214
70	1339	11251	5135	10444	7225	6896	9699	3222
71	1586	17391	13607	10505	16576	5367	15330	3342
72	2617	5358	3718	4905	4923	7648	4838	2611
73	4400	9853	8193	4698	10381	5612	10563	2324
74	2540	9677	2080	11469	7508	5739	8802	2189
75	5796	15608	8973	17349	13874	3817	14129	2471
76	3361	7818	5350	8771	5746	16868	7281	6389
77	9229	9006	10300	11272	16274	8794	13319	5727
78	3283	12954	8018	11970	9116	6473	10556	5058
79	7833	18892	12609	19254	21761	5505	19430	5621
80	736	5974	1800	6476	4088	5205	4194	6743
81	5356	9852	6698	7888	10437	3967	10904	3276
82	1098	10008	2877	8999	6536	3172	8778	3014
83	6480	12689	3018	19568	12560	5915	13384	3764
84	2334	8655	4221	9541	5040	9895	7468	4847
85	6612	10298	7310	11011	11744	10640	12132	3724
86	2724	11412	6218	11104	8850	9212	9534	5349
87	3089	24397	18307	11421	18131	9618	16750	7371
88	4095	5433	4410	7552	6247	5217	7326	3299
89	4110	11619	12974	5577	11048	5901	11946	3120
90	2826	10652	3268	12117	8630	4487	9801	2108
91	10339	16358	7854	30218	14324	7749	15442	4383
92	5729	7590	6501	8819	11463	15415	8741	8959
93	9284	13013	15291	12012	21497	13238	13494	8009
94	4363	13274	9151	14161	10102	10112	12006	5155
95	12772	25369	20681	29050	24798	10061	24790	3779
96	2540	4393	1865	4907	3107	6252	5651	4337
97	4601	9188	6652	5401	9552	4135	10617	4209
98	1906	9616	1543	9909	4984	4537	7929	2192
99	6196	11030	5841	18197	12631	4421	12787	2581
100	2652	7549	4054	8309	2985	12035	5928	6327
101	7174	9672	9234	9576	13356	9540	11438	5050
102	2112	13108	5889	10320	8105	8329	9540	4472
103	4464	20881	16440	8650	18603	6196	16418	4537
104	2765	6423	4756	5557	6585	9426	6273	2626
105	4730	10792	9195	6994	10171	7491	11462	2165
106	3263	9696	2380	13715	8574	5861	9549	2705
107	8523	14450	8870	21111	14774	5944	14338	3508

108	4606	7810	5135	9562	7565	20550	8669	6361
109	9922	10880	11802	11997	17612	15318	14223	6533
110	3950	15507	7771	13519	9210	7896	11060	5833
111	7844	24825	18328	22318	28486	6690	24442	8629
112	1831	5906	3151	6223	5524	5478	5793	9717
113	5887	9674	8244	6383	10514	4767	11358	3086
114	2181	10586	3024	10188	7608	4508	9006	4056
115	7632	12635	4519	25658	13976	5215	13784	4333
116	3134	8648	4096	11021	7163	12105	8454	4613
117	7613	11318	8930	10905	14556	12567	12439	4499
118	3485	12027	6291	13058	10252	12315	10265	7043
119	7791	29772	27639	14102	21184	9147	19895	10141
120	4323	6777	5660	7239	6410	7279	7963	3426
121	5219	11927	11532	7212	11596	6692	12610	3439
122	3547	10753	5058	13115	9448	5297	10245	2992
123	16531	19141	11854	23948	16924	7041	15764	5995
124	5738	8568	7630	8963	12211	24436	11009	10871
125	11457	14382	17618	16423	23208	23197	15748	10756
126	5953	13714	11619	15720	11784	7975	12010	7860
127	20557	25673	27408	24006	29628	14761	28518	13371

Table 3.C.23 - Gain VQ tables for wideband CELP coder (subframe 2.5 ms)

scale\_factor =  $2^{14}$ 

Index	Mode 0		Mode 1		Mode 2		Mode 3	
	Codeword		Codeword		Codeword		Codeword	
	0	1	0	1	0	1	0	1
0	623	1653	703	1600	925	1765	2183	1360
1	4977	7329	6999	7720	10136	2637	11461	1963
2	666	8305	773	4726	7284	2854	8235	1604
3	6899	10324	9384	5419	17693	3408	15241	1764
4	836	5878	5366	3996	3552	3626	6278	1420
5	8041	6466	1915	16812	13092	3610	13663	1633
6	758	10452	5218	9537	1252	8172	9952	1789
7	1744	17965	19240	5324	23891	4297	22255	2789
8	549	3524	3271	1975	930	4302	4727	1322
9	5259	9455	9741	10111	11475	7817	12848	3913
10	3357	7053	1016	7984	8913	5660	8032	5330
11	6477	14339	12019	10014	17203	6961	18901	2415
12	3465	4971	4268	6354	6085	3225	2755	4856
13	9498	11637	11057	19287	12780	7358	16607	5271
14	1162	13163	946	11254	1815	18430	9771	3962
15	14727	17847	18993	13700	21501	9618	24263	3548
16	2233	2742	2075	2655	2302	1585	3591	1370
17	6369	8148	9499	7235	10910	6438	12842	1755
18	2536	8587	2638	6053	8442	4614	8667	3083
19	7455	12035	12796	8078	20904	3754	16896	2250
20	1963	6182	7161	3209	4850	2132	7267	1518
21	9478	7743	2041	23208	15193	3309	14301	3564
22	3258	10620	6277	11342	1335	11752	11027	4360
23	2613	24379	24992	6118	30539	5101	21581	5803
24	2534	4510	3613	4377	5072	5039	5165	2753
25	4889	11162	7034	14112	10023	13225	14433	5337
26	3929	8618	3149	9271	4564	10549	2408	10035
27	10225	15484	14716	11416	17995	11869	18893	4618
28	5476	5282	5370	6841	7577	5740	7027	3594
29	12164	9762	15173	24543	14943	9463	18019	9866
30	3904	13407	4491	10826	14608	19852	10090	8292
31	16937	23095	29120	18723	25794	13415	28540	4549
32	719	2541	779	3070	2606	3061	3110	2077
33	5234	8582	8205	9287	11966	2871	12052	2613
34	1807	9290	859	6469	8410	2312	9039	1675
35	8211	9973	12740	5245	18666	5720	16124	2967
36	1498	7350	6175	5725	4664	3660	6560	2780
37	8017	8770	6764	16570	14048	4666	14753	2756
38	2227	10852	6582	9774	3404	7347	10781	2531
39	7023	18444	19832	10216	26317	4819	23634	5631
40	711	4714	4478	3309	3108	5525	5485	1551
41	5837	10768	10060	12511	13335	10213	13281	6175
42	4032	7748	2029	9324	10184	8061	9102	6383
43	8018	15511	12951	14163	19605	7670	20401	2633
44	4517	6184	4033	8206	6730	5149	5143	4680
45	12431	12041	14750	19319	14925	6376	17882	5722
46	1360	15172	1150	13784	2672	25528	10689	6099
47	21735	11850	21933	18982	25088	8345	25872	5708
48	3146	3669	2299	4178	3707	1495	4155	2796
49	6623	9260	11213	6767	12884	4993	13446	3316
50	2867	9706	2324	8065	10124	4932	10007	2862

51	9061	13290	14623	7410	21285	6916	17382	3230
52	2592	7688	7695	6137	5774	1902	7730	3213
53	10125	9396	5796	29402	16270	5653	15271	3473
54	3444	11804	7552	11915	5529	13556	11725	4997
55	9279	26510	25370	14528	28811	10300	23781	9389
56	2900	6049	4537	4853	5186	6805	5895	2900
57	6087	12216	9353	15797	10698	17281	15594	8576
58	4428	9889	3412	10687	8572	9470	4753	15856
59	9824	19814	15614	13667	21070	13055	20099	5645
60	6083	6650	5778	7953	7253	8188	6322	5956
61	14453	13532	19960	23752	13521	15297	18278	20753
62	4923	14800	4816	13435	19660	21913	12968	8969
63	24149	25073	26801	27969	26157	19536	29872	7509
64	1343	1958	1433	1972	1599	2840	1908	2712
65	5680	7722	7363	8708	10742	3638	12110	1597
66	845	9291	1692	5314	7933	3554	8326	2497
67	7852	10939	11043	4444	19097	3850	16095	1979
68	649	7044	6131	4275	4117	4580	6680	1974
69	8020	7694	4401	18565	14022	3151	14425	1890
70	1021	11649	5658	10466	3031	9278	10759	1586
71	4159	19643	21037	7928	24314	6271	22615	4297
72	1259	4010	3534	3208	1480	6154	4884	2127
73	5980	9889	10760	11043	12038	9471	13691	4463
74	4221	6962	791	9418	9774	6331	8909	4764
75	7905	13873	12597	11680	18157	8325	19421	3347
76	4371	5233	4516	7380	6633	3933	4270	5976
77	10680	12405	11586	22535	14104	7856	17412	4387
78	2586	13391	2157	12058	5314	19783	10044	4979
79	19007	19045	19934	16982	23597	10993	26254	3792
80	1980	3682	2644	3236	3174	2232	4178	1871
81	7128	8386	9236	8718	12090	5940	13302	2392
82	3176	8928	3308	6841	9449	4230	9172	3544
83	8469	12171	13966	9660	22246	5133	17895	2222
84	2427	6886	8753	3815	5250	2969	7755	2189
85	11337	7698	6214	23362	16354	3701	14966	4403
86	4069	10808	6373	12770	2903	14044	12013	3726
87	4726	29112	27001	8663	31096	8058	22199	7899
88	2566	5327	3626	5575	5905	5607	5405	3593
89	5277	11907	8489	13693	12120	12563	15585	5790
90	4537	9036	4155	9495	6506	10300	5333	8248
91	12014	17514	16827	10952	19001	15802	20115	4200
92	6832	5489	6418	6897	8057	6793	7221	4629
93	14981	10741	15658	29992	16610	10120	20225	11398
94	5273	13231	5049	11959	15893	27203	10868	11634
95	20157	28819	30803	23245	29259	15955	31248	4920
96	1319	2820	1504	3509	2374	4523	3315	3170
97	5901	8853	8577	10477	11889	4263	12697	2808
98	2006	10065	2016	6954	9045	3286	9333	2509
99	9150	10509	15152	4593	20107	5650	16662	3594
100	1775	8200	7212	4999	5640	4202	7201	2539
101	8961	9029	8028	19624	15144	4868	15429	2582
102	2385	11910	7351	10473	5328	8469	11381	3089
103	7377	22105	22295	12494	27918	6680	24775	7065
104	1734	5092	5147	2434	4266	6026	5886	2154
105	6678	11294	11181	13566	14762	12104	14331	7187
106	4702	8097	2271	10465	10433	10178	7955	9288
107	9089	17272	13586	16661	19612	10085	21153	3886

108	5263	6481	4910	8463	7461	4469	6051	4688
109	12274	14408	17041	20188	15871	7869	18924	6832
110	3309	15633	3010	14167	8007	29735	11824	6891
111	25017	18875	24828	19876	26724	9599	27400	7572
112	4402	3892	2859	4974	4059	2583	4363	3797
113	7410	9475	10800	8629	13674	6069	13961	2589
114	3733	9648	3238	7996	11064	5044	10569	3542
115	10146	13762	16439	8127	22951	7729	18249	3540
116	3305	8002	8169	7370	6582	2320	8174	4007
117	10831	10739	10658	28201	17431	5055	15913	4137
118	4326	12211	8946	11821	7740	15831	12650	5130
119	13330	28821	30514	12153	31223	12105	24856	13314
120	3668	6089	5168	5596	6491	6879	6295	3676
121	6832	12875	11177	16352	10034	22914	16956	7306
122	5036	10302	3554	12262	8204	11735	7943	19699
123	12133	22306	16900	16123	22923	16432	20369	7676
124	6891	7251	6187	8862	8955	7639	7342	6851
125	17114	15040	21821	28616	16435	14789	27140	20279
126	5790	16427	4909	15182	24024	27601	14456	13292
127	29293	29716	31068	29537	30256	23347	30549	11980

Table 3.C.24 - Filter coefficients for adaptive codebook interpolation (for 16 kHz sampling rate)

scale\_factor = 2<sup>15</sup>

Coefficients	Value
0	32767
1	31275
2	27042
3	20763
4	13437
5	6177
6	0
7	-4356
8	-6550
9	-6664
10	-5142
11	-2668
12	0
13	2202
14	3490
15	3703
16	2957
17	1578
18	0
19	-1357
20	-2185
21	-2349
22	-1896
23	-1020
24	0
25	890
26	1439
27	1552
28	1255
29	677
30	0

31	-588
32	-951
33	-1023
34	-825
35	-443
36	0
37	383
38	616
39	659
40	528
41	282
42	0
43	-239
44	-381
45	-404
46	-321
47	-169
48	0
49	142
50	224
51	236
52	186
53	98
54	0
55	-80
56	-126
57	-132
58	-105
59	-55
60	0
61	48
62	77
63	84
64	70
65	39
66	0

### 3.C.3 Gain tables for the bitrate scalable tool

Table 3.C.25 - Enhanced gain tables for bitrate scalable MPE tool (nge)

scale\_factor = 2<sup>14</sup>

Index	Mode 0	Mode 1	Mode 2	Mode 3
	Codeword	Codeword	Codeword	Codeword
0	460	3342	4827	9045
1	5049	3231	2242	1739
2	3328	1482	435	883
3	8745	5359	4010	4160
4	2155	135	1187	3582
5	6211	4113	3016	2736
6	4199	2452	1450	680
7	15087	8599	5760	7686
8	1199	1005	2490	5747
9	5562	3650	2617	2216

10	3775	2015	995	0
11	11112	6358	4691	5391
12	2832	848	241	2087
13	7228	4629	3477	3351
14	4615	2838	1854	1225
15	23253	15013	7969	11417

### 3.C.4 LSP VQ tables and gain VQ tables for the bandwidth scalable tool

**Table 3.C.26 - LSP VQ tables for the bandwidth scalable tool (lsp\_bws\_tbl[0])**

scale\_factor =  $2^{13}$

Index	Codeword									
	0	1	2	3	4	5	6	7	8	9
0	-41	-58	-95	-284	-293	-417	-687	-734	-862	-1824
1	-78	-159	-181	-258	-225	-155	19	48	237	104
2	-68	-60	-140	-243	-236	-1003	-265	-489	-647	-792
3	-3	70	117	128	110	180	490	363	806	749
4	-50	-46	-74	-272	-225	-372	-497	-498	-1380	-1000
5	-4	46	59	46	82	17	111	23	66	629
6	-52	-87	-125	-234	-261	-408	-430	-358	-309	-165
7	34	144	228	389	413	363	921	581	1211	1604
8	-36	-57	-114	-255	-261	-520	-518	-1228	-501	-901
9	-6	56	104	117	156	194	198	367	191	78
10	-19	-39	-45	-110	-102	-168	-159	-267	-345	-579
11	43	228	387	391	422	601	644	1149	851	856
12	-28	-48	-70	-208	-222	-277	-396	-499	-563	-1032
13	23	118	249	214	470	357	396	664	266	785
14	-4	40	27	1	41	-17	8	2	-70	-177
15	29	181	406	504	654	829	1167	1610	1555	1759

**Table 3.C.27 - LSP VQ tables for the bandwidth scalable tool (lsp\_bws\_tbl[1])**

scale\_factor =  $2^{13}$

Index	Codeword									
	0	1	2	3	4	5	6	7	8	9
0	10907	11766	12650	13565	14862	16624	18655	20941	22376	23585
1	11684	12595	13708	14900	16640	18214	19709	20667	21539	22525
2	11557	12621	13445	14584	15623	16358	17467	19933	22077	23331
3	12453	14102	15206	16263	17778	19035	20062	21238	22586	23799
4	10069	10772	12097	15128	17076	18536	20125	21164	22354	23609
5	11437	12290	14402	15497	16507	18259	20110	21510	22548	23866
6	10428	11550	13785	15584	17082	18505	19972	21362	22629	24040
7	13519	14437	15489	16474	17504	18223	19346	20260	21700	22671
8	11294	12217	13323	15370	16935	17819	19016	20873	22499	23702
9	12175	13269	14532	15637	16921	18040	19261	20229	21464	22346
10	12340	13277	14078	14979	15869	16793	19200	21345	22628	23859
11	13040	14209	15418	16742	17867	19029	20399	21350	22383	24253
12	10575	11293	11743	12973	15928	17866	19506	20907	22281	23775
13	12774	13427	14128	15005	16614	18510	20037	21408	22641	23998
14	11383	12366	13514	14730	16354	17611	18467	19590	21632	23199
15	14219	15088	15945	16742	17772	18822	20169	21096	22383	23076
16	11660	12545	13516	14524	15328	16601	19698	21415	22499	23376



17	12097	13307	14751	16156	17703	18738	19906	20757	21767	22627
18	11458	12811	14633	15588	16456	17350	18823	21007	22409	23592
19	12713	14672	15607	16709	17968	19510	20504	21681	22778	24083
20	10854	12003	13319	14259	16308	18370	19876	20997	22332	23913
21	11693	13666	14594	15448	17036	18421	19975	21474	22600	23992
22	11212	12873	14108	15210	17179	18468	19742	21081	22603	24048
23	13337	14268	15483	16817	18358	19295	20151	20896	21939	22724
24	11315	12649	13823	15061	17013	18564	21079	21928	22707	23308
25	12643	14086	15445	16473	17672	18695	19858	20768	21806	22616
26	13027	13721	14586	15957	17748	18844	19873	20745	21921	22785
27	13017	14220	15862	17090	18039	19147	20194	21511	22870	24079
28	11874	12624	13624	16234	17876	18853	19847	20702	21800	23113
29	12235	13632	15246	16266	17735	19063	20434	21858	22985	24491
30	11772	13540	15444	16547	17618	18357	19526	20418	21695	23442
31	14176	15233	16608	17791	18928	19646	20504	21269	22417	23112
32	11555	12468	13325	14186	16130	17528	18871	21143	22702	23983
33	12263	13275	14213	15361	16544	17688	19627	20733	22008	23127
34	12268	13260	14245	15156	16011	17040	18445	19590	21135	22502
35	12687	14062	15575	17189	18508	19599	20602	21568	22530	23829
36	11011	11731	12574	15253	17566	18568	19606	20538	21687	23153
37	12277	13236	14189	15629	17473	18858	20305	21733	22853	24353
38	10804	11753	13556	16294	18124	19224	20416	21279	22163	23291
39	13723	14568	15407	16425	17685	18831	20199	21009	22044	22759
40	12281	13269	14281	15760	17026	17823	19116	21239	22708	24006
41	13027	13935	14995	15941	16983	18139	19731	20719	21866	22601
42	12950	13976	14895	15815	16656	17439	18754	20665	22392	23447
43	13209	14239	15546	16814	18223	19333	20724	21848	22819	23543
44	11301	12020	12825	14070	17862	19418	20221	21125	22038	22948
45	12599	13411	14686	16302	17593	18998	20409	21827	22938	24441
46	11079	12232	14712	16017	17347	18206	19365	20374	21459	22870
47	14605	15535	16539	17338	18225	18959	19995	20885	22150	22882
48	11980	12961	13916	15070	17327	18500	19397	20507	22224	23689
49	12635	13608	14486	16291	17879	19395	20475	21406	22526	23907
50	12884	13800	14866	16695	17782	18659	19504	20597	22548	23877
51	13210	14328	15395	17103	18408	19454	20589	21687	22862	24303
52	11537	12323	13228	15167	17128	18898	20139	21372	22612	24073
53	11630	13730	14976	16399	17819	18939	20366	21637	22786	24166
54	11600	12937	14730	16090	17547	18924	20322	21772	22966	24445
55	14038	14990	16155	17219	18431	19453	20524	21208	22098	22768
56	12377	13483	14329	15311	17352	19158	20556	21374	22232	23017
57	13196	14174	15129	16082	17595	18849	20034	21032	22409	23146
58	13219	14289	14973	15687	16825	18633	20054	21364	22683	24070
59	13596	14613	15630	16771	18060	19138	20351	21647	22857	24294
60	11811	12929	14541	16053	17968	19782	20991	21708	22547	23559
61	12973	13958	15130	16734	18149	19201	20104	21750	22739	24049
62	11980	13767	15252	16959	17983	18928	20056	21157	22516	24032
63	14560	15534	16855	17972	19259	20100	21174	21854	22739	23280
64	11124	11915	13006	14838	15996	17157	19806	21653	22694	23721
65	11865	12936	14494	15839	17081	18236	19854	21004	22171	23000
66	12039	13102	13970	15107	16409	17327	18234	20445	22402	23722
67	12559	14111	15651	16731	17889	18967	20595	21688	22853	23943
68	10907	11714	12655	15055	17189	18884	20439	21654	22816	24213
69	12157	13171	14850	15851	16713	18603	20057	21359	22651	24068
70	11268	12146	14053	15983	17479	19054	20327	21583	22828	24213
71	13654	14819	16087	16993	17916	18623	19605	20522	22094	22989
72	11886	12696	13600	15575	16743	17762	19749	21490	22764	24053
73	12677	13744	15118	16219	17267	17986	19001	19888	21173	22142

74	12500	13587	14603	15564	16224	17338	19760	21668	23066	24193
75	13212	14183	15512	16436	17901	19524	20588	21663	22910	24196
76	11461	12178	12953	13883	15708	18666	20491	21381	22214	23141
77	12875	13648	14490	15698	17226	18289	20227	21858	23229	24271
78	11826	13288	14501	15868	17225	18048	18687	19656	22147	23559
79	14347	15334	16282	17307	18463	19509	20813	21608	22756	23358
80	12117	13159	14143	15321	16246	17682	20680	21755	22534	23266
81	12721	13508	15211	16121	17393	18360	19910	20824	22537	23381
82	11596	13812	15102	16106	17306	18166	19305	21120	22670	24020
83	12945	14571	15957	16990	18222	19348	20552	21777	23226	24419
84	11897	12862	14031	14917	16045	18237	19632	20995	22521	24045
85	12405	13813	14920	15812	17195	19245	20361	21635	22729	24025
86	10722	12804	14726	16072	17480	18862	20393	21364	22572	24020
87	13214	14506	16008	17050	18241	19275	20448	21361	22422	23196
88	12007	13171	14371	16289	17532	18732	21021	22062	22833	23501
89	13127	14043	15633	16744	17713	18819	20088	21178	22566	23242
90	12813	13993	15104	16112	17035	18061	20544	21715	22650	23342
91	13123	14793	15903	17038	18155	19189	20731	21785	22759	24035
92	12097	13086	14358	16610	18342	19402	20391	21148	22105	23037
93	12632	13900	15229	16543	17863	19170	20465	21966	22979	24529
94	11829	13441	15356	16337	17497	18756	20500	21414	22338	23098
95	14162	15325	16455	17659	18882	20060	21309	22418	23607	24472
96	12045	12847	13603	14450	16622	18426	20131	21616	22662	23924
97	12536	13617	14648	15575	16898	18591	19915	20970	22325	23043
98	12758	13636	14608	15798	16991	18225	19289	20383	21974	23815
99	12937	13718	15908	16869	18273	19405	20532	22671	23228	24035
100	11437	12259	13419	16195	18560	19580	20752	21828	23392	24403
101	12209	13019	15003	16484	17833	19177	20337	21436	22667	24075
102	11304	12537	14991	17148	18293	19139	20258	21199	22254	23559
103	13731	14744	15748	16970	18119	19390	20420	21416	22704	23536
104	12407	13326	14216	16176	17406	18491	19962	21156	22586	23850
105	13545	14580	15519	16456	17282	18164	19486	21053	22615	23852
106	12828	13959	15004	16154	17103	18199	18975	21215	23554	24506
107	13157	14462	15596	16821	18281	19763	20964	22012	23087	24329
108	11930	12741	13600	15086	17889	19365	20485	21511	22374	23246
109	13107	13994	14770	16370	17623	18984	20429	21574	23259	24282
110	11668	12599	14730	16358	17495	18417	19686	20984	22445	23896
111	15211	16134	17070	17965	18948	19620	20531	21290	22452	23034
112	12028	13013	13994	15115	16462	18398	20611	22476	23786	24570
113	12754	13834	15219	16509	18221	19449	20686	21448	22429	23138
114	12689	13824	15334	16647	17611	18465	19999	21569	22733	24159
115	13192	14300	16110	17226	18366	19584	20642	21645	22743	24179
116	11945	12739	13931	15701	17380	19109	20298	21508	22805	24138
117	12440	13777	15246	16751	18197	19637	20724	21662	22857	24211
118	11754	13477	14683	15761	17770	19081	20010	20890	22185	23886
119	13513	14663	15961	17337	18879	20081	21087	21778	22742	23409
120	12689	13744	14688	15645	17441	19107	21083	22232	23384	24243
121	13568	14439	15227	16313	17796	19363	20657	21617	22614	23544
122	13260	14617	15420	16243	17466	19002	20515	22061	23493	24525
123	13649	14770	15936	17149	18350	19477	20664	21918	23003	24228
124	12357	13518	14911	16697	18469	19951	21383	22319	23439	24360
125	13132	14203	15467	16836	18003	19226	20614	22038	23058	24481
126	12456	13734	15927	17116	18128	19069	20126	20945	21992	23021
127	15200	16311	17435	18492	19609	20520	21689	22532	23451	24216

**Table 3.C.28 - LSP VQ tables for the bandwidth scalable tool (Isp\_bws\_tbl[2])**

scale\_factor = 2<sup>13</sup>

Index	Codeword				
	0	1	2	3	4
0	-22	-5	-91	-439	36
1	-86	-180	-160	-22	116
2	19	-16	-10	-62	-383
3	-29	-32	67	-98	-32
4	29	54	-48	-89	222
5	-33	-132	61	158	-99
6	40	0	-121	106	-3
7	62	148	8	-85	-75
8	11	115	193	-72	122
9	-42	-15	64	127	160
10	26	89	129	154	-148
11	-18	66	432	178	101
12	-4	13	72	142	490
13	-21	-51	88	525	69
14	94	115	86	177	155
15	53	459	137	126	33

**Table 3.C.29 - LSP VQ tables for the bandwidth scalable tool (Isp\_bws\_tbl[3])**

scale\_factor = 2<sup>13</sup>

Index	Codeword				
	0	1	2	3	4
0	-274	-22	-435	135	-32
1	-61	-196	-282	-120	-75
2	-278	-398	105	-256	93
3	-398	-149	-392	-711	-473
4	-137	-85	-339	-351	285
5	14	-739	31	25	-160
6	-243	-109	-43	18	-169
7	-137	138	-257	-4	-204
8	-210	-25	-89	-202	61
9	494	285	-188	-275	-241
10	113	-400	-90	-301	115
11	-613	83	-80	-30	-12
12	29	-24	-65	-658	20
13	264	-111	-162	-295	-174
14	-319	-365	238	165	-236
15	-233	212	77	-100	-421
16	-180	-319	-126	142	235
17	128	-297	-472	131	-44
18	-25	-203	72	-219	-248
19	21	137	-129	-155	213
20	185	-113	-305	-84	304
21	69	-33	-280	-5	-338
22	-202	119	-86	298	-385
23	80	101	-460	-227	-42
24	96	132	-481	233	-113
25	139	123	-101	-183	-732
26	42	-191	-18	14	5
27	-235	160	111	3	244
28	-38	99	71	-4	-58

29	691	-152	-245	69	-42
30	6	-119	447	9	-247
31	-154	413	-234	-209	11
32	-76	-218	-204	228	-249
33	186	41	-114	-29	-75
34	-100	-155	358	13	171
35	-28	60	-105	-265	-216
36	-115	149	-268	142	400
37	242	-269	9	23	-347
38	-7	-63	54	60	-410
39	124	331	-116	57	-347
40	-30	-5	-248	103	105
41	73	213	123	-388	48
42	170	-430	143	74	371
43	-155	326	-95	239	40
44	-11	-87	95	-208	333
45	301	10	67	-183	236
46	91	-275	157	346	-81
47	-231	48	294	-269	-92
48	-19	-64	-291	588	29
49	255	-84	-178	278	150
50	185	-192	303	-272	9
51	-61	292	50	-279	685
52	61	20	40	136	317
53	220	-33	-150	317	-415
54	74	181	234	231	-329
55	221	283	-199	72	156
56	103	127	-40	275	-53
57	167	116	185	-192	-325
58	198	-43	197	106	35
59	-142	-23	14	292	85
60	120	270	198	2	127
61	405	132	99	108	-187
62	-198	121	359	243	-47
63	-107	511	244	-46	-107

Table 3.C.30 - LSP VQ tables for the bandwidth scalable tool (lsp\_bws\_tbl[4])

scale\_factor = 2<sup>13</sup>

Index	Codeword				
	0	1	2	3	4
0	97	-370	-55	31	-621
1	-692	315	193	-429	-730
2	-742	351	-448	-248	181
3	-223	-313	94	-619	-108
4	-104	-304	-774	-46	348
5	-415	-27	-405	-23	-113
6	-601	335	33	-3	-236
7	-970	-240	175	-132	-278
8	-408	-152	-46	-192	5
9	724	458	54	-493	-553
10	-395	233	-31	24	233
11	-404	-109	409	146	-609
12	207	-117	-364	-528	-616
13	208	141	0	-286	-420
14	-155	185	-441	167	-463

15	26	12	-389	616	-108
16	32	-171	-389	78	41
17	-346	491	-85	-545	164
18	-137	364	-33	228	-25
19	355	-270	-159	-337	-120
20	130	-186	-636	-1034	27
21	264	-381	-69	147	-126
22	-249	-826	454	256	-253
23	-417	-725	-608	630	138
24	-237	-267	-174	247	141
25	-373	147	338	-252	86
26	-124	20	86	-354	314
27	59	-468	176	-220	-95
28	400	181	-78	120	-714
29	196	347	109	15	-279
30	-1156	-675	408	435	264
31	193	-22	-75	285	72
32	96	-12	172	162	-329
33	-54	602	376	-129	-746
34	-309	419	-27	178	704
35	137	315	219	-197	43
36	62	-621	-232	21	249
37	269	51	-528	-253	76
38	-735	475	711	182	-296
39	-613	251	-212	474	14
40	13	-236	-57	-88	191
41	307	-44	517	121	-658
42	-269	-127	196	158	365
43	-169	-325	194	130	-208
44	185	126	-79	-571	39
45	462	389	51	213	292
46	-200	237	389	138	-80
47	145	-236	-599	669	814
48	593	-78	-32	530	-11
49	226	496	210	-440	442
50	-599	287	514	339	329
51	530	-83	-172	10	94
52	769	305	-85	-391	430
53	370	284	-179	364	-226
54	-48	-42	626	-103	314
55	-234	-206	58	856	432
56	106	197	136	126	90
57	31	-110	773	315	-201
58	167	7	264	-74	738
59	477	65	401	74	144
60	692	573	311	-12	-144
61	116	569	720	422	192
62	-174	-698	939	300	508
63	392	106	34	684	563
64	-374	-701	41	-235	-651
65	-440	28	-202	-527	-401
66	-730	-180	-223	156	322
67	153	-242	252	-389	-633
68	-726	-567	-587	-558	240
69	-89	-380	-679	-263	-476
70	-1226	531	195	-86	93
71	-539	-403	-120	292	-259

72	-201	-476	-212	-102	-164
73	259	161	-71	-674	-1205
74	-665	-14	118	-378	563
75	-331	-306	598	-210	-287
76	-732	-870	-1259	-2795	-3315
77	-14	-120	-172	-200	-235
78	-298	29	-22	-39	-538
79	-82	-425	-477	375	-342
80	394	-264	-699	175	-118
81	-121	164	-398	-408	-69
82	-114	319	-416	106	144
83	-45	-292	-298	-461	142
84	-24	71	-321	-729	644
85	-134	-87	-164	219	-216
86	-367	-553	241	-152	187
87	-28	-514	161	408	103
88	-498	-80	210	153	-15
89	-180	245	-34	-210	-91
90	146	-406	236	-472	418
91	350	-373	457	24	-188
92	278	-80	-348	81	-450
93	-71	455	-196	-147	-422
94	-451	-334	642	569	-19
95	46	25	-124	206	416
96	-148	-13	225	-206	-260
97	-87	547	244	-440	-279
98	339	246	-263	-78	816
99	345	-17	333	-422	7
100	287	-316	180	198	338
101	133	170	-253	32	-98
102	-356	826	285	4	70
103	-39	433	17	563	234
104	-252	-39	-321	-190	378
105	620	-52	99	19	-331
106	-198	-408	-72	-1	702
107	158	-297	113	509	-376
108	583	241	-406	-798	-85
109	86	268	-177	-184	284
110	-228	228	72	417	-390
111	-309	73	-463	455	418
112	193	-359	-363	643	229
113	123	640	-113	-54	101
114	-60	363	268	16	386
115	241	-18	57	-68	-77
116	359	-218	-373	-235	507
117	-157	-18	101	482	25
118	24	-142	308	26	71
119	75	-2	382	375	269
120	-144	3	-10	16	18
121	125	200	545	-80	-266
122	288	20	51	-153	329
123	299	127	282	424	-132
124	481	271	-173	-188	-100
125	66	600	367	296	-318
126	247	-286	555	781	101
127	453	136	-466	306	279

**Table 3.C.31 - LSP VQ tables for the bandwidth scalable tool (lsp\_bws\_tbl[5])**

scale\_factor = 2<sup>13</sup>

Index	Codeword				
	0	1	2	3	4
0	-231	-647	-207	108	-41
1	8	-228	-481	-388	60
2	182	-257	245	105	-240
3	-383	-213	-24	-290	-364
4	-494	-67	-22	106	177
5	238	149	-131	-415	-434
6	-130	370	-149	-166	239
7	-294	344	288	-86	-347
8	92	-234	-394	307	318
9	546	22	-210	-92	125
10	5	46	-196	176	-154
11	25	-93	187	-260	199
12	-128	-329	318	349	351
13	422	469	252	-33	-133
14	238	174	114	339	316
15	-172	311	472	307	46

**Table 3.C.32 - LSP VQ tables for the bandwidth scalable tool (bws\_ma\_prdct)**

scale\_factor = 2<sup>15</sup>

Codeword	Order		
	0	1	2
0	32307	6025	3826
1	29035	5743	3609
2	32346	4386	2552
3	29547	5161	2645
4	27530	6708	4095
5	28237	8367	4965
6	27312	8564	4803
7	24644	8340	4905
8	27090	10058	5834
9	23277	9490	5693
10	20474	8283	4014
11	21357	7799	3607
12	22134	7214	3424
13	22373	6960	3431
14	21710	7190	3887
15	20912	7440	4427
16	20712	7440	4634
17	20797	7242	4759
18	20987	6867	4921
19	23026	6144	3612

**Table 3.C.33 - LSP VQ tables for the bandwidth scalable tool (bws\_nw\_prdct)**

scale\_factor = 2<sup>15</sup>

Codeword	Order
0	16438
1	16363
2	16258

3	16625
4	16492
5	16832
6	16688
7	16911
8	16822
9	17241

Table 3.C.34 - Gain VQ tables for bandwidth extension tool

scale\_factor = 2<sup>13</sup> for Codeword 0, 2<sup>10</sup> for Codeword 1

Index	Mode 0		Mode 1		Mode 2		Mode 3	
	Codeword		Codeword		Codeword		Codeword	
	0	1	0	1	0	1	0	1
0	29	10	72	16	36	4	22	5
1	155	1313	1527	428	3773	164	4836	118
2	139	691	66	470	2280	147	3203	107
3	4311	1706	3693	442	184	789	4417	258
4	52	453	654	407	232	429	2521	193
5	7966	1055	2237	548	5995	164	6309	166
6	437	1062	1102	572	4182	262	3493	134
7	1482	4291	214	884	2085	2647	10201	139
8	267	245	431	290	503	214	1577	128
9	1721	1509	2608	456	4927	167	5443	115
10	3405	424	311	573	2878	157	3365	287
11	1202	2863	4799	447	1005	1218	5688	222
12	2032	319	1555	501	2604	430	3410	545
13	6916	1810	3188	651	6883	193	7856	112
14	3862	990	2265	602	4740	540	4120	104
15	3565	7136	3853	1489	371	6591	12458	88
16	332	72	288	141	428	57	591	312
17	335	2036	2305	494	4755	255	5426	186
18	2385	725	448	490	2389	273	2962	204
19	4815	2630	3789	598	493	956	4896	349
20	1120	468	886	510	1678	392	2300	461
21	6648	1551	2753	570	6457	382	6572	202
22	2272	999	1365	636	4938	405	4062	211
23	1474	5520	2073	903	4991	3408	11982	58
24	355	341	1800	274	1692	161	2375	89
25	3714	1293	3019	522	5496	256	5844	122
26	3851	738	332	689	3285	269	3997	341
27	9201	3462	6836	583	4511	1297	5709	313
28	2015	515	1746	553	3331	447	5663	735
29	12655	2009	3902	684	8027	246	7292	390
30	5839	949	1904	752	6078	868	4790	167
31	4604	13921	1932	2359	7119	16128	8754	232
32	326	36	188	84	152	32	131	135
33	628	1564	2057	427	4486	161	5023	164
34	1278	660	453	597	2706	233	3139	156
35	3767	2135	3900	560	596	855	4638	295
36	26	568	1068	447	982	281	2449	314
37	10488	1524	2500	574	6249	291	7008	161
38	828	949	1469	581	4458	377	3655	182
39	9425	4374	579	1221	8901	2654	11389	95
40	1130	264	948	316	1261	213	2040	131
41	2480	1676	3062	465	5376	160	5664	182



42	3932	542	767	621	3666	230	3805	265
43	4684	3644	4726	600	283	1452	6126	294
44	2514	389	1924	503	2226	616	4259	1073
45	7858	2378	2984	685	7234	309	7348	263
46	3288	1131	2576	645	6374	688	4494	116
47	1949	9019	9106	1394	21348	8971	13310	134
48	244	158	757	179	636	114	1699	319
49	1465	2347	2542	508	5070	285	5198	272
50	2440	860	715	546	2894	324	3297	225
51	5127	3083	4130	622	2433	977	5217	424
52	1409	548	1327	533	845	570	3136	424
53	9384	1804	2884	620	7343	531	6585	275
54	1700	1179	1845	645	5696	534	4361	169
55	14524	5593	2195	1168	2014	4582	11911	95
56	949	352	2252	363	1713	239	2886	97
57	5212	1499	3302	571	5775	319	6061	201
58	5362	700	1089	710	3690	354	4434	372
59	14442	3913	7643	1058	9267	1586	5986	489
60	2555	625	1883	593	3502	555	8981	919
61	11609	2905	4804	809	9577	442	7210	513
62	5427	1277	2983	771	6900	1105	5013	231
63	0	31787	5778	4721	22789	22049	9665	362
64	89	29	290	46	71	16	316	37
65	1238	1345	1710	468	4123	199	5119	119
66	697	792	21	534	2438	200	3639	79
67	4938	1937	4056	507	368	858	4687	236
68	645	496	795	472	237	513	2681	242
69	8986	1283	2267	575	6402	207	6626	137
70	259	1138	1178	603	4408	299	3826	136
71	4997	4849	905	969	3979	2327	11173	47
72	682	279	367	359	443	271	1777	208
73	3340	1543	2820	495	5127	220	5597	147
74	3153	528	332	619	3270	195	3587	338
75	1141	3398	5130	537	2115	1347	5898	255
76	1869	409	1713	518	2826	517	4038	639
77	7180	2034	3458	655	7379	224	8178	158
78	4736	998	2287	639	5138	669	4157	148
79	14840	7980	6170	1582	11237	7536	12675	100
80	680	106	344	215	178	109	1118	505
81	1687	2022	2401	529	4776	322	5365	228
82	3014	779	558	529	2431	342	3013	265
83	7087	2675	3723	635	114	1132	5295	332
84	1651	469	1090	530	2112	424	2150	714
85	7959	1618	3005	570	7028	406	6909	226
86	2427	1091	1535	670	5424	426	4303	215
87	6492	5822	2768	1003	7010	3938	12193	55
88	656	406	2464	234	2067	216	2601	123
89	4191	1420	3253	538	5953	240	6124	134
90	4479	834	559	767	3484	305	4300	315
91	13038	3429	6516	773	5118	1514	5794	382
92	2478	543	1976	566	3806	441	7203	762
93	14216	2430	4054	748	8929	295	7748	341
94	6018	1114	2372	775	7857	891	4868	202
95	3475	20505	5337	2943	3759	14544	8590	358
96	397	42	796	97	344	28	1165	191
97	382	1663	2101	473	4510	220	5287	153
98	1702	773	325	651	2964	253	3369	179

99	5845	2257	4305	566	1340	816	4914	278
100	829	587	1305	483	981	371	2863	340
101	13478	1351	2613	607	6686	292	7459	185
102	1678	933	1657	612	4322	466	3916	182
103	10888	4993	207	1579	12791	3031	11089	182
104	1585	218	1274	360	1448	299	2276	167
105	2991	1866	3427	501	5609	196	5898	175
106	4919	539	1015	651	3868	281	4101	269
107	8233	3965	5274	687	1575	1684	6346	355
108	2585	462	2070	527	2962	713	4180	2808
109	10285	2523	3345	729	8010	388	7974	250
110	4196	1172	2521	690	4466	935	4597	151
111	16560	10713	11683	1915	8917	11836	13208	206
112	791	196	1199	220	1020	154	2152	248
113	3636	2419	2675	539	5337	334	5511	269
114	3207	912	814	573	3214	373	3602	229
115	8462	3005	4503	673	2528	1076	5061	562
116	1899	621	1506	549	1531	520	3877	422
117	9930	2086	3203	606	8439	576	6895	323
118	2647	1322	2063	689	6391	498	4560	199
119	12868	6493	4074	1172	10597	4788	12074	108
120	1286	408	3429	341	1963	301	2836	155
121	5676	1686	3525	577	5940	399	6269	232
122	7086	803	1352	804	4069	361	4552	459
123	14891	4520	10736	841	6723	1888	6633	430
124	3296	639	2029	620	3979	632	12932	1373
125	15411	3038	5187	987	11638	607	8281	562
126	7025	1326	3450	879	9588	1258	5173	197
127	1466	336	8587	10865	0	26530	11167	457

Table 3.C.35 - Gain tables for excitation codebook 1 of bandwidth extension tool

scale\_factor = 2<sup>12</sup>

Index	Mode 0	Mode 1	Mode 2	Mode 3
	Codeword	Codeword	Codeword	Codeword
0	-9890	-334	-1119	-6181
1	-3715	429	-55	-2339
2	-1164	1003	557	-996
3	180	1442	1137	-261
4	1124	1802	1625	243
5	1717	2116	2059	708
6	2378	2403	2470	1157
7	3013	2684	2915	1641
8	3519	3001	3408	2180
9	4213	3349	3914	2802
10	5112	3750	4541	3361
11	6184	4275	5297	4052
12	8150	5030	6450	5008
13	11196	6491	8837	6691
14	16806	9366	14863	10432
15	26264	27589	31529	22189

## Annex 3.D (informative)

### Tables

#### 3.D.1 Bandwidth expansion tables in LPC analysis of the mode II coder

**Table 3.D.1 - Bandwidth expansion tables in LPC analysis**

For 8 kHz sampling rate		For 16 kHz sampling rate	
0	1.0001000	0	1.0000100
1	0.9988903	1	0.9997225
2	0.9955685	2	0.9988903
3	0.9900568	3	0.9975049
4	0.9823916	4	0.9955685
5	0.9726235	5	0.9930844
6	0.9608164	6	0.9900568
7	0.9470474	7	0.9864905
8	0.9314049	8	0.9823916
9	0.9139889	9	0.9777667
10	0.8949091	10	0.9726235
		11	0.9669703
		12	0.9608164
		13	0.9541719
		14	0.9470474
		15	0.9394543
		16	0.9314049
		17	0.9229120
		18	0.9139889
		19	0.9046499
		20	0.8949091

#### 3.D.2 Downsampling filter coefficients for the bandwidth scalable tool

**Table 3.D.2 - Downsampling filter coefficients**

0	0.487498	25	0.006314	50	0.003645	75	0.001263
1	0.318007	26	0.009227	51	0.001586	76	-0.000190
2	0.012479	27	-0.005043	52	-0.003242	77	-0.001148
3	-0.105197	28	-0.008776	53	-0.001706	78	0.000159
4	-0.012414	29	0.003938	54	0.002858	79	0.005156
5	0.062159	30	0.008312	55	0.001785	80	-0.000092
6	0.012306	31	-0.002977	56	-0.002497		
7	-0.043381	32	-0.007836	57	-0.001830		
8	-0.012155	33	0.002141	58	0.002158		
9	0.032701	34	0.007355	59	0.001844		
10	0.011965	35	-0.001414	60	-0.001842		
11	-0.025712	36	-0.006869	61	-0.001831		
12	-0.011735	37	0.000785	62	0.001551		
13	0.020722	38	0.006383	63	0.001794		

14	0.011468	39	-0.000241	64	-0.001284
15	-0.016942	40	-0.005902	65	-0.001737
16	-0.011167	41	-0.000222	66	0.001042
17	0.013954	42	0.005428	67	0.001663
18	0.010830	43	0.000616	68	-0.000825
19	-0.011527	44	-0.004961	69	-0.001576
20	-0.010467	45	-0.000942	70	0.000632
21	0.009494	46	0.004508	71	0.001478
22	1.01E-02	47	0.001210	72	-0.000462
23	-7.78E-03	48	-0.004068	73	-0.001373
24	-9.66E-03	49	-0.001423	74	0.000315

## Annex 3.E (informative)

### Example of a simple CELP transport stream

Two types of streams, header streams and raw data streams, are part of the MPEG-4 CELP coder syntax. The header streams contain configuration information necessary for the decoding process and parsing of the raw data streams. The raw data streams contain the coded audio information. A transport layer handles the delivery of these streams. The multiplexed bitstream of the header and the raw data stream is shown in the following pseudo code example:

```

CelpHeader()
if (BandwidthScalabilityMode==ON)
{
    CelpBWSenhHeader()
}
while(1)
{
    CelpBaseFrame()
    for (i = 0; i < nrof_enh_layers; i++)
    {
        CelpBRSenhFrame()
    }
    if (BandwidthScalabilityMode==ON)
    {
        CelpBWSenhFrame()
    }
}

```

The header information is attached at the beginning of the bitstream. Its update is necessary only when changes are made in the configuration of the bitstream. The raw data information follows the header stream and is transmitted frame by frame. In a scalable bitstream, two types of information, namely base information and enhancement information, are transmitted every frame.

In order to control transmission of the enhancement information, which is in `CelpBrsenhFrame()` or `CelpBwsenhFrame()`, frame by frame, the decoder requires additional parameters which indicate whether the enhancement information is available for decoding at every frame. The parameters for bitrate scalability and bandwidth scalability are *dec\_enh\_layers* and *dec\_bws\_mode*, respectively. The following procedure is applied to parse the bitstream.

```

CelpHeader()
if (BandwidthScalabilityMode==ON)
{
    CelpBWSenhHeader()
}
while(1)
{
    CelpBaseFrame()
    for (i = 0; i < dec_enh_layers; i++ )
    {
        CelpBRSenhFrame()
    }
}

```

```

if (BandwidthScalabilityMode==ON && dec_bws_mode==ON)
{
    CelpBWSenhFrame()
}
}

```

where *dec\_enh\_layers* is the number of available enhancement layers in the current frame and is equal to or less than the value of *nrof\_enh\_layers* in the header information. *dec\_bws\_mode* is a one-bit parameter that indicates whether bandwidth scalability is enabled in the current frame as follows:

**Table 3.E.1 - Definition of *dec\_bws\_mode***

<i>dec_bws_mode</i>	ScalableID	Description
0	OFF	Bandwidth scalability is disabled
1	ON	Bandwidth scalability is enabled

## **Annex 3.F** (informative)

### **Random access points**

Random access to the CELP bitstream is available at every frame. However, the decoder should be initialised before decoding the first frame. If FineRate Control is enabled the LPC filter coefficients should be initialized to zero as no LPC parameters might be present in the first frame being parsed.

## Contents for Subpart 4

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## Subpart 4: General Audio Coding (GA) – AAC, TwinVQ, BSAC

### 4.1 Scope

The General Audio (GA) coding subpart of MPEG-4 Audio is mainly intended to be used for generic audio coding at all but the lowest bitrates. Typically, GA encoding is used for complex music material in mono from 6 kbit/s per channel and for stereo signals from 12 kbit/s per stereo signal up to broadcast quality audio at 64 kbit/s or more per channel. MPEG-4 coded material can be represented either by a single set of data, like in MPEG-1 and MPEG-2 Audio, or by several subsets which allow the decoding at different quality levels, depending on the number of subsets being available at the decoder side (bitrate scalability).

MPEG-2 Advanced Audio Coding (AAC) syntax (including support for multi-channel audio) is fully supported by MPEG-4 Audio GA coding. All the features and possibilities of the MPEG-2 AAC standard also apply to MPEG-4. AAC has been tested to allow for ITU-R 'indistinguishable' quality according to [4] at data rates of 320 kb/s for five full-bandwidth channel audio signals. In MPEG-4 the tools derived from MPEG-2 AAC are available together with other MPEG-4 GA coding tools which provide additional functionalities, like bit rate scalability and improved coding efficiency at very low bit rates. Bit rate scalability is either achieved with only GA coding tools, or by using a combination with an external (non-GA, e.g. CELP) core coder.

MPEG-4 GA coding is not restricted to some fixed bitrates but supports a wide range of bitrates and variable rate coding. While efficient mono, stereo and multi-channel coding is possible using extended, MPEG-2 AAC derived tools, the document also provides extensions to this tool set which allow mono/stereo scalability, where a mono signal can be extracted by decoding only subsets of the encoded stereo stream.

#### 4.1.1 Technical Overview

##### 4.1.1.1 Encoder and Decoder Block Diagrams

The block diagrams of the GA encoder and decoder reflect the structure of MPEG-4 GA coding. In general, there are the MPEG-2 AAC related tools with MPEG-4 add-ons for some of them and the tools related to the TwinVQ quantization and coding. The TwinVQ is an alternative module for the AAC-type quantization and it is based on an interleaved vector quantization and LPC (Linear Predictive Coding) spectral estimation. It operates from 6 kbit/s/ch and is recommended to be used below 16 kbit/s/ch with constant bitrate.

The basic structure of the MPEG-4 GA system is shown in Figure 4.1 and Figure 4.2. The data flow in this diagram is from left to right, top to bottom. The functions of the decoder are to find the description of the quantized audio spectra in the bitstream, decode the quantized values and other reconstruction information, reconstruct the quantized spectra, process the reconstructed spectra through whatever tools are active in the bitstream in order to arrive at the actual signal spectra as described by the input bitstream, and finally convert the frequency domain spectra to the time domain, with or without an optional gain control tool. Following the initial reconstruction and scaling of the spectrum reconstruction, there are many optional tools that modify one or more of the spectra in order to provide more efficient coding. For each of the optional tools that operate in the spectral domain, the option to "pass through" is retained, and in all cases where a spectral operation is omitted, the spectra at its input are passed directly through the tool without modification.

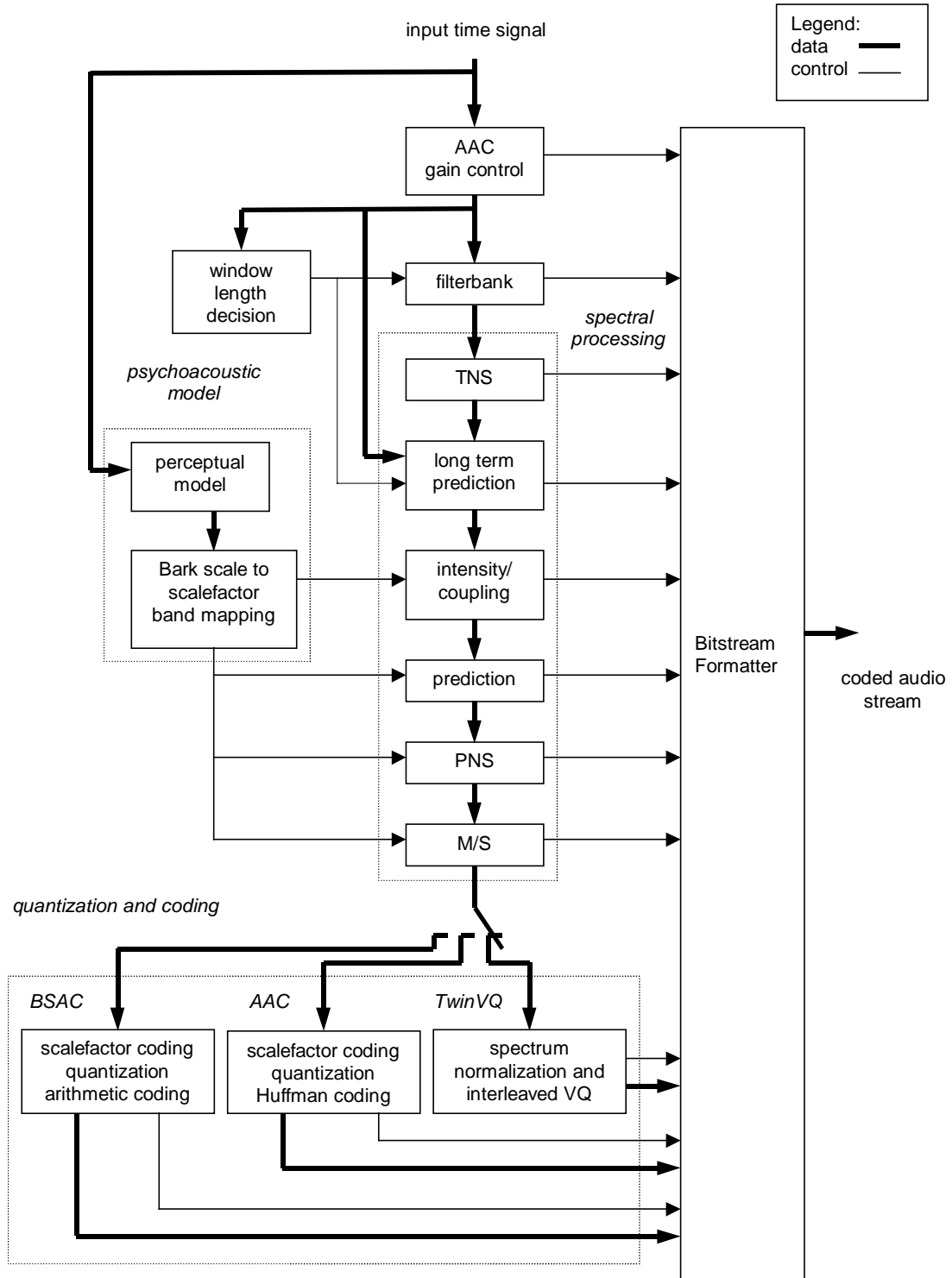


Figure 4.1 – Block diagram GA non scalable encoder

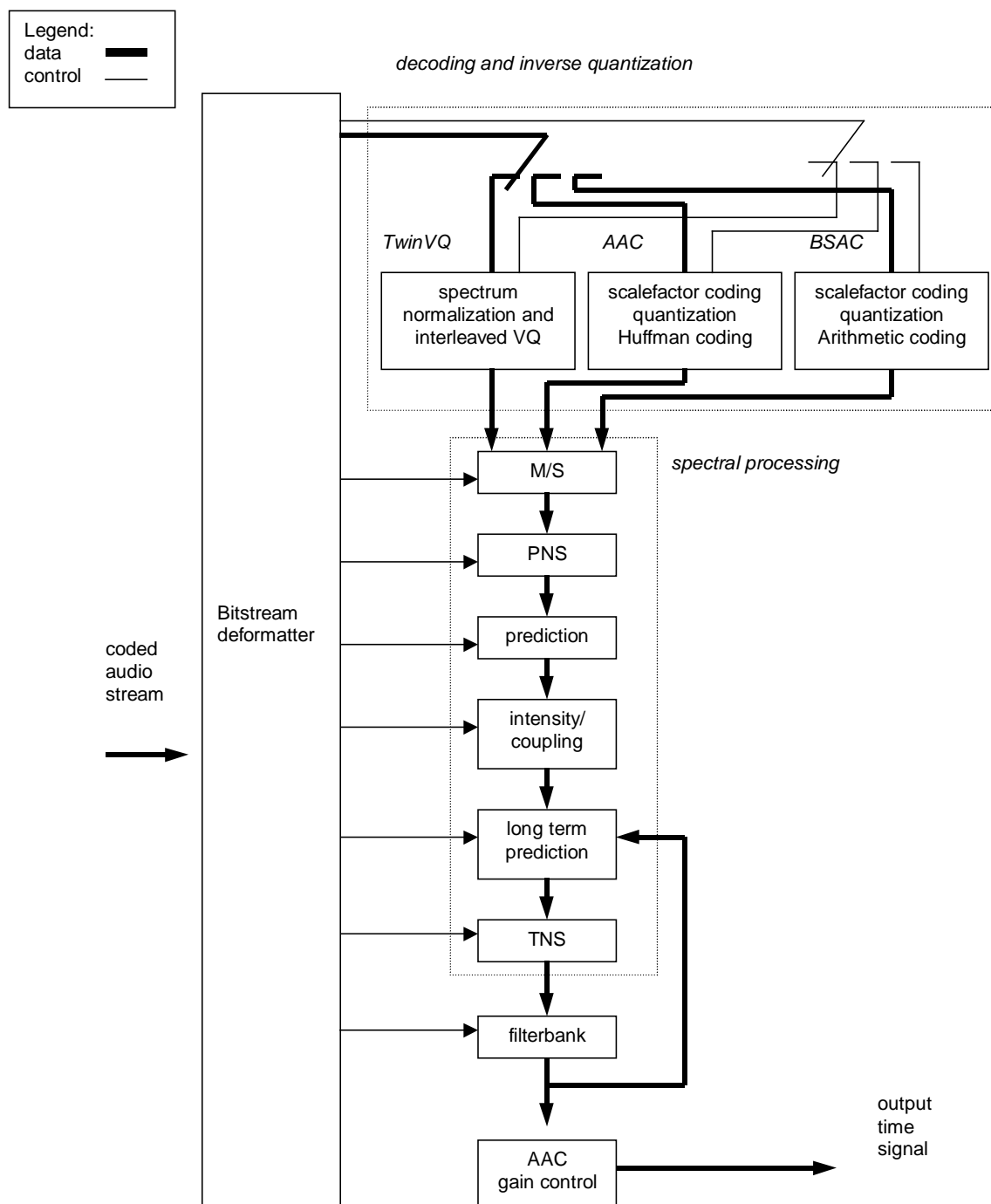


Figure 4.2 – Block diagram of the GA non scalable decoder

#### 4.1.1.2 Overview of the Encoder and Decoder Tools

The input to the bitstream demultiplexer tool is the MPEG-4 GA bitstream. The demultiplexer separates the bitstream into the parts for each tool, and provides each of the tools with the bitstream information related to that tool.

The outputs from the bitstream demultiplexer tool are:

- The quantized (and optionally noiselessly coded) spectra represented by either

## ISO/IEC 14496-3:2001(E)

- the sectioning information and the noiselessly coded spectra (AAC) or
- a set of indices of code vectors (TwinVQ) or
- the arithmetic model information and the noiselessly coded spectra (BSAC)
- The M/S decision information (optional)
- The predictor side information (optional)
- The perceptual noise substitution (PNS) information (optional)
- The intensity stereo control information and coupling channel control information (both optional)
- The temporal noise shaping (TNS) information (optional)
- The filterbank control information
- The gain control information (optional)
- Bitrate scalability related side information (optional)

The AAC noiseless decoding tool takes information from the bitstream demultiplexer, parses that information, decodes the Huffman coded data, and reconstructs the quantized spectra and the Huffman and DPCM coded scalefactors.

The inputs to the noiseless decoding tool are:

- The sectioning information for the noiselessly coded spectra
- The noiselessly coded spectra

The outputs of the noiseless decoding tool are:

- The decoded integer representation of the scalefactors:
- The quantized values for the spectra

The inverse quantizer tool takes the quantized values for the spectra, and converts the integer values to the non-scaled, reconstructed spectra. This quantizer is a non-uniform quantizer.

The input to the Inverse Quantizer tool is:

- The quantized values for the spectra

The output of the inverse quantizer tool is:

- The un-scaled, inversely quantized spectra

The scalefactor tool converts the integer representation of the scalefactors to the actual values, and multiplies the un-scaled inversely quantized spectra by the relevant scalefactors.

The inputs to the scalefactors tool are:

- The decoded integer representation of the scalefactors
- The un-scaled, inversely quantized spectra

The output from the scalefactors tool is:

- The scaled, inversely quantized spectra

The M/S tool converts spectra pairs from Mid/Side to Left/Right under control of the M/S decision information, improving stereo imaging quality and sometimes providing coding efficiency.

The inputs to the M/S tool are:

- The M/S decision information
- The scaled, inversely quantized spectra related to pairs of channels

The output from the M/S tool is:

- The scaled, inversely quantized spectra related to pairs of channels, after M/S decoding

Note: The scaled, inversely quantized spectra of individually coded channels are not processed by the M/S block, rather they are passed directly through the block without modification. If the M/S block is not active, all spectra are passed through this block unmodified.

The prediction tool reverses the prediction process carried out at the encoder. This prediction process re-inserts the redundancy that was extracted by the prediction tool at the encoder, under the control of the predictor state information. This tool is implemented as a second order backward adaptive predictor.

The inputs to the prediction tool are:

- The predictor state information

- The predictor side information
- The scaled, inversely quantized spectra

The output from the prediction tool is:

- The scaled, inversely quantized spectra, after prediction is applied.

Note: If the prediction is disabled, the scaled, inversely quantized spectra are passed directly through the block without modification.

Alternatively, there is a forward adaptive long term prediction tool provided.

The inputs to the long term prediction tool are:

- The reconstructed time domain output of the decoder
- The scaled, inversely quantized spectra

The output from the long term prediction tool is:

- The scaled, inversely quantized spectra, after prediction is applied.

Note: If the prediction is disabled, the scaled, inversely quantized spectra are passed directly through the block without modification.

The perceptual noise substitution (PNS) tool implements noise substitution decoding on channel spectra by providing an efficient representation for noise-like signal components.

The inputs to the perceptual noise substitution tool are:

- The inversely quantized spectra
- The perceptual noise substitution control information

The output from the perceptual noise substitution tool is:

- The inversely quantized spectra

Note: If either part of this block is disabled, the scaled, inversely quantized spectra are passed directly through this part without modification. If the perceptual noise substitution block is not active, all spectra are passed through this block unmodified.

The intensity stereo / coupling tool implements intensity stereo decoding on pairs of spectra. In addition, it adds the relevant data from a dependently switched coupling channel to the spectra at this point, as directed by the coupling control information.

The inputs to the intensity stereo / coupling tool are:

- The inversely quantized spectra
- The intensity stereo control information and coupling control information

The output from the intensity stereo / coupling tool is:

- The inversely quantized spectra after intensity and coupling channel decoding.

Note: If either part of this block is disabled, the scaled, inversely quantized spectra are passed directly through this part without modification. The intensity stereo tool and M/S tools are arranged so that the operation of M/S and Intensity stereo are mutually exclusive on any given scalefactor band and group of one pair of spectra.

The temporal noise shaping (TNS) tool implements a control of the fine time structure of the coding noise. In the encoder, the TNS process has flattened the temporal envelope of the signal to which it has been applied. In the decoder, the inverse process is used to restore the actual temporal envelope(s), under control of the TNS information. This is done by applying a filtering process to parts of the spectral data.

The inputs to the TNS tool are:

- The inversely quantized spectra
- The TNS information

The output from the TNS block is:

- The inversely quantized spectra

Note: If this block is disabled, the inversely quantized spectra are passed through without modification.

The filterbank tool applies the inverse of the frequency mapping that was carried out in the encoder, as indicated by the filterbank control information and the presence or absence of gain control information. An inverse modified discrete cosine transform (IMDCT) is used for the filterbank tool. If the gain control tool is not used, the IMDCT

input consists of either 1024 or 128 (depending on **window\_sequence** spectral coefficients (if **frameLengthFlag** is set to '0'), or of 960 or 120 spectral coefficients (if **frameLengthFlag** is set to '1'), respectively. If the gain control tool is used, the filterbank tool is configured to use four sets of either 256 or 32 coefficients, depending of the value of **window\_sequence**.

The inputs to the filterbank tool are:

- The inversely quantized spectra
- The filterbank control information

The output(s) from the filterbank tool is (are):

- The time domain reconstructed audio signal(s).

Two alternative, but very similar versions of this tool are available. The version with a frame length of 960 samples, which is not available in ISO/IEC 13818-7, allows for an integer frame length. For example at 48 kHz sampling rate, the frame length is exactly 20 ms with this version. This is especially useful for the CELP/AAC bitrate scalability combinations, where this allows the construction of combined CELP layer frames, which have a length of a multiple of 10 ms, and AAC enhancement layer frames. However, this feature can be used for configurations with only AAC or TwinVQ coding as well.

When present, the gain control tool applies a separate time domain gain control to each of 4 frequency bands that have been created by the gain control PQF filterbank in the encoder. Then, it assembles the 4 frequency bands and reconstructs the time waveform through the gain control tool's filterbank.

The inputs to the gain control tool are:

- The time domain reconstructed audio signal(s)
- The gain control information

The output(s) from the gain control tool is (are):

- The time domain reconstructed audio signal(s)

If the gain control tool is not active, the time domain reconstructed audio signal(s) are passed directly from the filterbank tool to the output of the decoder. This tool is used for the scalable sampling rate (SSR) audio object type only.

The spectrum normalization tool converts the reconstructed flat spectra to the actual values at the decoder. The spectral envelope is specified by LPC coefficients, a Bark scale envelope, periodic peak components, and gain.

The input to the spectral normalization tool are

- The reconstructed flat spectra
- The information of LPC coefficients, a Bark scale envelope, periodic peak components and gain

The output from the spectral normalization tool is

- The reconstructed actual spectra

The interleaved VQ tool converts the vector index to the flattened spectra at the TwinVQ decoder by means of table look-up of the codebook and inverse interleaving of the spectra. Quantization noise is minimized by a weighted distortion measure at the encoder instead of an adaptive bit allocation. This is an alternative to the AAC quantization tool.

The input to the interleaved VQ tool is:

- A set of indices of the code vector.

The output from the TwinVQ tool is:

- The reconstructed flattened spectra

The Frequency Selective Switch (FSS) tool is used to control the combination of the AAC coding layer with both, TwinVQ, and CELP coding layer, if these are used as base layer coder in scalable configurations. In a second function this tool is applied to control the combination of mono and stereo coding layer in scalable configurations where both mono, and stereo coding layer are used to code a stereo input signal.

The Up-sampling Filter tool adapts the sampling rate of a CELP core coder, which can be used as base layer coder in scalable configurations, to the sampling rate of the AAC extension layer.

The input to the Upsampling Filter tool is:

- The output of a CELP core coder running at a lower sampling rate than the AAC extension layer

The output from the Up-sampling Filter tool is:

- The up-sampled CELP core coder output, matching the sampling rate of the AAC extension layer, transformed into the frequency domain with exactly the same frequency and time resolution as the AAC extension layer.

The BSAC noiseless decoding tool takes information from the bitstream demultiplexer, parses that information, decodes the Arithmetic coded data, and reconstructs the quantized spectra and the Arithmetic coded scalefactors. The BSAC noiseless coding module is an alternative to the AAC coding module. The BSAC noiseless coding is used to make the bitstream scalable and error resilient and further reduce the redundancy of the scalefactors and the quantized spectrum.

The inputs to the BSAC decoding tool are

- The Arithmetic model information for the noiselessly coded spectra
- The noiselessly coded bit-sliced data

The outputs from the BSAC decoding tool are

- The decoded integer representation of the scalefactors
- The quantized value for the spectra

The virtual codebooks (VCB11) tool can extend the part of the bitstream demultiplexer that decodes the sectioning information. The VCB11 tool gives the opportunity to detect serious errors within the spectral data of an MPEG-4 AAC bitstream.

The input of the VCB11 tool is:

- The encoded section data using virtual codebooks

The output of the VCB11 tool is:

- The decoded sectioning information

The reversible variable length coding (RVLC) tool can replace the part of the noiseless coding tool that decodes the Huffman and DPCM coded scalefactors. The RVLC tool is used to increase the error resilience for the scalefactor data within an MPEG-4 AAC bitstream.

The input of the RVLC tool is:

- The noiselessly coded scalefactors using RVLC

The output of the RVLC tool is:

- The decoded integer representation of the scalefactors

The Huffman codeword reordering (HCR) tool can extend the part of the noiseless coding tool that decodes the Huffman coded spectral data. The HCR tool is used to increase the error resilience for the spectral data within an MPEG-4 AAC bitstream.

The input of the HCR tool is:

- The sectioning information for the noiselessly coded spectra
- The noiselessly coded spectral data in an error resilient reordered manner
- The length of the longest codeword within spectral\_data
- The length of spectral\_data

The output of the HCR tool is:

- The quantized value of the spectra

## 4.2 Normative references

ISO/IEC 11172-3:1993, *Information technology - Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s - Part 3: Audio*

ITU-T Rec.H.222.0 (1995) | ISO/IEC 13818-1:2000, *Information technology - Generic coding of moving pictures and associated audio information: Systems*

ISO/IEC 13818-3:1998, *Information technology - Generic coding of moving pictures and associated audio information - Part 3: Audio*



## 4.3 GA-specific definitions

- 4.3.1. **alias**: Mirrored spectral component resulting from sampling
- 4.3.2. **analysis filterbank**: Filterbank in the encoder that transforms a broadband PCM audio signal into a set of spectral coefficients.
- 4.3.3. **ancillary data**: Part of the bitstream that might be used for transmission of ancillary data.
- 4.3.4. **Bark**: The Bark is the standard unit corresponding to one critical band width of human hearing.
- 4.3.5. **block companding**: Normalizing of the digital representation of an audio signal within a certain time period.
- 4.3.6. **BSAC**: Bit Sliced Arithmetic Coding
- 4.3.7. **centre channel**: An audio presentation channel used to stabilize the central component of the frontal stereo image.
- 4.3.8. **core coder**: The term core coder is used to denote a base layer coder in certain scalability configurations. A core coder does not code the spectral samples of the MDCT filterbank of the subsequent AAC coding layers, but operates on a time domain signal. The output of the core decoder has to be up-sampled and transformed into the spectral domain, before it can be combined with the output of the AAC coding layers. Within the MPEG-4 Audio standard only the MPEG-4 CELP coder is a valid core coder. However, in principal, also another AAC coding layer, operating at a lower sampling rate, could be used on the time domain signal, and then combined with the other coding layer in exactly the same way as described for the CELP coder, and would therefore be called a core coder.
- 4.3.9. **critical band**: This unit of bandwidth represents the standard unit of bandwidth expressed in human auditory terms, corresponding to a fixed length on the human cochlea. It is approximately equal to 100 Hz at low frequencies and 1/3 octave at higher frequencies, above approximately 700 Hz.
- 4.3.10. **discrete cosine transform; DCT**: Either the forward discrete cosine transform or the inverse discrete cosine transform. The DCT is an invertible, discrete orthogonal transformation.
- 4.3.11. **filterbank**: A set of band-pass filters covering the entire audio frequency range.
- 4.3.12. **FSS**: Frequency Selective Switch. Module which selects one of two input signals independently in each scalefactor band.
- 4.3.13. **hybrid filterbank**: A serial combination of subband filterbank and MDCT. Used in MPEG-1 and MPEG-2 Audio.
- 4.3.14. **IDCT**: Inverse Discrete Cosine Transform.
- 4.3.15. **IMDCT**: Inverse Modified Discrete Cosine Transform.
- 4.3.16. **intensity stereo**: A method of exploiting stereo irrelevance or redundancy in stereophonic audio programmes based on retaining at high frequencies only the energy envelope of the right and left channels.
- 4.3.17. **joint stereo coding**: Any method that exploits stereophonic irrelevance or stereophonic redundancy.
- 4.3.18. **joint stereo mode**: A mode of the audio coding algorithm using joint stereo coding.
- 4.3.19. **main audio channels**: All `single_channel_elements` (see subclause 4.5.2.1) or `channel_pair_elements` (see subclause 4.5.2.1) in one program.
- 4.3.20. **mapping**: Conversion of an audio signal from time to frequency domain by subband filtering and/or by MDCT.
- 4.3.21. **masking**: A property of the human auditory system by which an audio signal cannot be perceived in the presence of another audio signal.
- 4.3.22. **masking threshold**: A function in frequency and time below which an audio signal cannot be perceived by the human auditory system.

- 4.3.23. **modified discrete cosine transform (MDCT)**: A transform which has the property of time domain aliasing cancellation. An analytical expression of the MDCT can be found in subclause 4.6.15.3.3.
- 4.3.24. **M/S stereo**: A method of removing imaging artifacts as well as exploiting stereo irrelevance or redundancy in stereophonic audio programs based on coding the sum and difference signal instead of the left and right channels.
- 4.3.25. **non-tonal component**: A noise-like component of an audio signal.
- 4.3.26. **PNS**: Perceptual Noise Substitution.
- 4.3.27. **polyphase filterbank**: A set of equal bandwidth filters with special phase interrelationships, allowing an efficient implementation of the filterbank.
- 4.3.28. **program**: A set of main audio channels, `coupling_channel_elements` (see subclause 4.5.2.1), `lfe_channel_elements` (see subclause 4.5.2.1), and associated data streams intended to be decoded and played back simultaneously. A program may be defined by default, or specifically by a `program_configuration_element` (see subclause 4.5.1.2). A given `single_channel_element` (see subclause 4.5.2.1), `channel_pair_element` (see subclause 4.5.2.1), `coupling_channel_element`, `lfe_channel_element` or data channel may accompany one or more programs in any given bitstream.
- 4.3.29. **psychoacoustic model**: A mathematical model of the masking behaviour of the human auditory system.
- 4.3.30. **SBA**: Segmented Binary Arithmetic Coding which is the error resilient tool for BSAC
- 4.3.31. **scalefactor**: Factor by which a set of values is scaled before quantization.
- 4.3.32. **scalefactor band**: A set of spectral coefficients which are scaled by one scalefactor.
- 4.3.33. **scalefactor index**: A numerical code for a scalefactor.
- 4.3.34. **spectral coefficients**: Discrete frequency domain data output from the analysis filterbank.
- 4.3.35. **spreading function**: A function that describes the frequency spread of masking effects.
- 4.3.36. **stereo-irrelevant**: A portion of a stereophonic audio signal which does not contribute to spatial perception.
- 4.3.37. **synthesis filterbank**: Filterbank in the decoder that reconstructs a PCM audio signal from subband samples.
- 4.3.38. **TNS**: Temporal Noise Shaping
- 4.3.39. **tonal component**: A sinusoid-like component of an audio signal.

## 4.4 Syntax

### 4.4.1 Decoder configuration (GASpecificConfig)

Table 4.1 – Syntax of GASpecificConfig()

Syntax	No. of bits	Mnemonic
GASpecificConfig ( samplingFrequencyIndex, channelConfiguration, audioObjectType )		
{		
<b>frameLengthFlag;</b>	<b>1</b>	<b>bslbf</b>
<b>dependsOnCoreCoder;</b>	<b>1</b>	<b>bslbf</b>
if ( dependsOnCoreCoder ) {		
<b>coreCoderDelay;</b>	<b>14</b>	<b>uimsbf</b>
}		
<b>extensionFlag;</b>	<b>1</b>	<b>bslbf</b>
if ( ! channelConfiguration ) {		
program_config_element ();		
}		
if ((audioObjectType == 6)    (audioObjectType == 20)) {		
<b>layerNr;</b>	<b>3</b>	<b>uimsbf</b>
}		
if ( extensionFlag ) {		
if ( audioObjectType == 22 ) {		
<b>numOfSubFrame;</b>	<b>5</b>	<b>bslbf</b>
<b>layer_length;</b>	<b>11</b>	<b>bslbf</b>
}		
if ( audioObjectType == 17    audioObjectType == 19		
audioObjectType == 20    audioObjectType == 23 ) {		
<b>aacSectionDataResilienceFlag;</b>	<b>1</b>	<b>bslbf</b>
<b>aacScalefactorDataResilienceFlag;</b>	<b>1</b>	<b>bslbf</b>
<b>aacSpectralDataResilienceFlag;</b>	<b>1</b>	<b>bslbf</b>
}		
<b>extensionFlag3;</b>	<b>1</b>	<b>bslbf</b>
if ( extensionFlag3 ) {		
/* tbd in version 3 */		
}		
}		

#### 4.4.1.1 Program config element

Table 4.2 – Syntax of program\_config\_element()

Syntax	No. of bits	Mnemonic
Program_config_element()		
{		
<b>element_instance_tag</b>	<b>4</b>	<b>uimsbf</b>
<b>object_type</b>	<b>2</b>	<b>uimsbf</b>
<b>sampling_frequency_index</b>	<b>4</b>	<b>uimsbf</b>
<b>num_front_channel_elements</b>	<b>4</b>	<b>uimsbf</b>
<b>num_side_channel_elements</b>	<b>4</b>	<b>uimsbf</b>
<b>num_back_channel_elements</b>	<b>4</b>	<b>uimsbf</b>
<b>num_lfe_channel_elements</b>	<b>2</b>	<b>uimsbf</b>
<b>num_assoc_data_elements</b>	<b>3</b>	<b>uimsbf</b>
<b>num_valid_cc_elements</b>	<b>4</b>	<b>uimsbf</b>

<b>mono_mixdown_present</b>	1	uimsbf
if ( mono_mixdown_present == 1 )		
<b>mono_mixdown_element_number</b>	4	uimsbf
<b>stereo_mixdown_present</b>	1	uimsbf
if ( stereo_mixdown_present == 1 )		
<b>stereo_mixdown_element_number</b>	4	uimsbf
<b>matrix_mixdown_idx_present</b>	1	uimsbf
if ( matrix_mixdown_idx_present == 1 ) {		
<b>matrix_mixdown_idx</b>	2	uimsbf
<b>pseudo_surround_enable</b>	1	uimsbf
}		
for ( i = 0; i < num_front_channel_elements; i++) {		
<b>front_element_is_cpe[i];</b>	1	bslbf
<b>front_element_tag_select[i];</b>	4	uimsbf
}		
for ( i = 0; i < num_side_channel_elements; i++) {		
<b>side_element_is_cpe[i];</b>	1	bslbf
<b>side_element_tag_select[i];</b>	4	uimsbf
}		
for ( i = 0; i < num_back_channel_elements; i++) {		
<b>back_element_is_cpe[i];</b>	1	bslbf
<b>back_element_tag_select[i];</b>	4	uimsbf
}		
for ( i = 0; i < num_lfe_channel_elements; i++)		
<b>lfe_element_tag_select[i];</b>	4	uimsbf
for ( i = 0; i < num_assoc_data_elements; i++)		
<b>assoc_data_element_tag_select[i];</b>	4	uimsbf
for ( i = 0; i < num_valid_cc_elements; i++) {		
<b>cc_element_is_ind_sw[i];</b>	1	uimsbf
<b>valid_cc_element_tag_select[i];</b>	4	uimsbf
}		
byte_alignment()		
<b>comment_field_bytes</b>	8	uimsbf
for ( i = 0; i < comment_field_bytes; i++)		
<b>comment_field_data[i];</b>	8	uimsbf
}		
}		

## 4.4.2 GA Bitstream Payloads

### 4.4.2.1 Payloads for the audio object types AAC main, AAC SSR, AAC LC and AAC LTP

Table 4.3 – Syntax of top level payload for audio object types AAC Main, SSR, LC, and LTP (raw\_data\_block())

Syntax	No. of bits	Mnemonic
Raw_data_block()		
{		
while( (id = <b>id_syn_ele</b> ) != ID_END ){	3	uimsbf
switch (id) {		
case ID_SCE: single_channel_element()		
break;		
case ID_CPE: channel_pair_element()		
break;		
case ID_CCE: coupling_channel_element()		
break;		
case ID_LFE: lfe_channel_element()		
break;		
}		
}		
}		

```

        case ID_DSE: data_stream_element()
            break;
        case ID_PCE: program_config_element()
            break;
        case ID_FIL: fill_element()
            break;
    }
}
}

```

**Table 4.4 – Syntax of single\_channel\_element()**

Syntax	No. of bits	Mnemonic
single_channel_element() { <b>element_instance_tag</b> individual_channel_stream(0,0) }	<b>4</b>	<b>uimsbf</b>

**Table 4.5 – Syntax of channel\_pair\_element()**

Syntax	No. of bits	Mnemonic
channel_pair_element() { <b>element_instance_tag</b>  <b>common_window</b> if(common_window) { ics_info() <b>ms_mask_present</b> if( ms_mask_present == 1 ) { for( g=0; g < num_window_groups; g++ ) { for( sfb=0; sfb < max_sfb; sfb++ ) { <b>ms_used[g][sfb]</b> } } } } individual_channel_stream(common_window,0) individual_channel_stream(common_window,0) }	<b>4</b>  <b>1</b>  <b>2</b>  <b>1</b>	<b>uimsbf</b>  <b>uimsbf</b>  <b>uimsbf</b>  <b>uimsbf</b>

**Table 4.6 – Syntax of ics\_info()**

Syntax	No. of bits	Mnemonic
ics_info() { <b>ics_reserved_bit</b> ; <b>window_sequence</b> ; <b>window_shape</b> ; if( window_sequence == EIGHT_SHORT_SEQUENCE ) { <b>max_sfb</b> ; <b>scale_factor_grouping</b> ; } else {	<b>1</b> <b>2</b> <b>1</b> <b>4</b> <b>7</b>	<b>bslbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

<b>max_sfb;</b>	<b>6</b>	<b>uimsbf</b>
<b>predictor_data_present;</b>	<b>1</b>	<b>uimsbf</b>
if (predictor_data_present) {		
if (audioObjectType == 1) {		
<b>predictor_reset;</b>	<b>1</b>	<b>uimsbf</b>
if (predictor_reset) {		
<b>predictor_reset_group_number;</b>	<b>5</b>	<b>uimsbf</b>
}		
for ( sfb = 0; sfb < min ( max_sfb,		
<b>PRED_SFB_MAX ); sfb++ ) {</b>		
<b>prediction_used[sfb];</b>	<b>1</b>	<b>uimsbf</b>
}		
}		
else {		
<b>ltp_data_present;</b>	<b>1</b>	<b>uimsbf</b>
if (ltp_data_present) {		
ltp_data();		
}		
if (common_window) {		
<b>ltp_data_present;</b>	<b>1</b>	<b>uimsbf</b>
if (ltp_data_present) {		
ltp_data();		
}		
}		
}		
}		

Table 4.7 – Syntax of pulse\_data()

Syntax	No. of bits	Mnemonic
pulse_data() {		
<b>number_pulse</b>	<b>2</b>	<b>uimsbf</b>
<b>pulse_start_sfb</b>	<b>6</b>	<b>uimsbf</b>
for (i=0; i<number_pulse+1; i++) {		
<b>pulse_offset[i]</b>	<b>5</b>	<b>uimsbf</b>
<b>pulse_amp[i]</b>	<b>4</b>	<b>uimsbf</b>
}		
}		

Table 4.8 – Syntax of coupling\_channel\_element()

Syntax	No. of bits	Mnemonic
coupling_channel_element()		
{		
<b>element_instance_tag</b>	<b>4</b>	<b>uimsbf</b>
<b>ind_sw_cce_flag</b>	<b>1</b>	<b>uimsbf</b>
<b>num_coupled_elements</b>	<b>3</b>	<b>uimsbf</b>
num_gain_element_lists = 0		
for (c=0; c<num_coupled_elements+1; c++) {		
num_gain_element_lists++		
<b>cc_target_is_cpe[c]</b>	<b>1</b>	<b>uimsbf</b>
<b>cc_target_tag_select[c]</b>	<b>4</b>	<b>uimsbf</b>
if ( cc_target_is_cpe[c] ) {		
<b>cc_l[c]</b>	<b>1</b>	<b>uimsbf</b>
<b>cc_r[c]</b>	<b>1</b>	<b>uimsbf</b>
}		
}		

<pre>                 if (cc_l[c] &amp;&amp; cc_r[c] )                     num_gain_element_lists++             }         } <b>cc_domain</b> <b>gain_element_sign</b> <b>gain_element_scale</b> individual_channel_stream(0,0) for ( c=1; c&lt;num_gain_element_lists; c++ ) {     if ( ind_sw_cce_flag ) {         cge = 1     } else {         <b>common_gain_element_present</b>[c]         cge = common_gain_element_present[c]     }     if ( cge )         <b>hcod_sf</b>[common_gain_element[c]]     else {         for (g=0; g&lt;num_window_groups; g++) {             for (sfb=0; sfb&lt;max_sfb; sfb++) {                 if ( sfb_cb[g][sfb] != ZERO_HCB )                     <b>hcod_sf</b>[dpcm_gain_element[c][g][sfb]]             }         }     } } </pre>	<p>1</p> <p>1</p> <p>2</p> <p>1</p> <p>1..19</p> <p>1..19</p>	<p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p>
--	---	---

Table 4.9 – Syntax of lfe\_channel\_element()

Syntax	No. of bits	Mnemonic
lfe_channel_element()		
{		
<b>element_instance_tag</b>	4	<b>uimsbf</b>
individual_channel_stream(0,0)		
}		

Table 4.10 – Syntax of data\_stream\_element()

Syntax	No. of bits	Mnemonic
data_stream_element()		
{		
<b>element_instance_tag</b>	4	<b>uimsbf</b>
<b>data_byte_align_flag</b>	1	<b>uimsbf</b>
cnt = <b>count</b>	8	<b>uimsbf</b>
if (cnt == 255)		
cnt += <b>esc_count</b> ;	8	<b>uimsbf</b>
if (data_byte_align_flag)		
byte_alignment()		
for (i=0; i<cnt; i++)		
<b>data_stream_byte</b> [element_instance_tag][i];	8	<b>uimsbf</b>
}		

Table 4.11 – Syntax of fill\_element()

Syntax	No. of bits	Mnemonic

Table 4.12 – Syntax of gain\_control\_data()

Syntax	No. of bits	Mnemonic
gain_control_data()		
{		
<b>max_band</b>	<b>2</b>	<b>uimsbf</b>
if (window_sequence == ONLY_LONG_SEQUENCE) {		
for (bd=1; bd<=max_band; bd++) {		
for (wd=0; wd<1; wd++) {		
adjust_num[bd][wd]	<b>3</b>	<b>uimsbf</b>
for (ad=0; ad<adjust_num[bd][wd]; ad++) {		
<b>alevcode</b> [bd][wd][ad]	<b>4</b>	<b>uimsbf</b>
<b>alocode</b> [bd][wd][ad]	<b>5</b>	<b>uimsbf</b>
}		
}		
}		
}		
else if (window_sequence == LONG_START_SEQUENCE) {		
for (bd=1; bd<=max_band; bd++) {		
for (wd=0; wd<2; wd++) {		
<b>adjust_num</b> [bd][wd]	<b>3</b>	<b>uimsbf</b>
for (ad=0; ad<adjust_num[bd][wd]; ad++) {		
<b>alevcode</b> [bd][wd][ad]	<b>4</b>	<b>uimsbf</b>
if (wd == 0)		
<b>alocode</b> [bd][wd][ad]	<b>4</b>	<b>uimsbf</b>
else		
<b>alocode</b> [bd][wd][ad]	<b>2</b>	<b>uimsbf</b>
}		
}		
}		
}		
else if (window_sequence == EIGHT_SHORT_SEQUENCE) {		
for (bd=1; bd<=max_band; bd++) {		
for(wd=0; wd<8; wd++) {		
<b>adjust_num</b> [bd][wd]	<b>3</b>	<b>uimsbf</b>
for (ad=0; ad<adjust_num[bd][wd]; ad++) {		
<b>alevcode</b> [bd][wd][ad]	<b>4</b>	<b>uimsbf</b>
<b>alocode</b> [bd][wd][ad]	<b>2</b>	<b>uimsbf</b>
}		
}		
}		
}		
else if (window_sequence == LONG_STOP_SEQUENCE) {		
for (bd=1; bd<=max_band; bd++) {		
for(wd=0; wd<2; wd++) {		
<b>adjust_num</b> [bd][wd]	<b>3</b>	<b>uimsbf</b>



<pre>                 for (ad=0; ad&lt;adjust_num[bd][wd]; ad++) {                     <b>alevcode</b>[bd][wd][ad]                     if (wd == 0)                         <b>alocode</b>[bd][wd][ad]                     else                         <b>alocode</b>[bd][wd][ad]                 }             }         }     } </pre>	<p>4</p> <p>4</p> <p>5</p>	<p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p>
---	----------------------------	--

4.4.2.2 Payloads for the audio object type AAC scalable

**Table 4.13 – Syntax of the ASME top level payload for the audio object type AAC scalable (aac\_scalable\_main\_element)**

Syntax	No. of bits	Mnemonic
<pre> aac_scalable_main_element() {     aac_scalable_main_header()     for (ch=0; ch&lt;(this_layer_stereo ? 2:1); ch++){         individual_channel_stream(1, 1)     }     cnt = bits_to_decode() / 8     while (cnt &gt;= 1) {         cnt -= extension_payload(cnt)     } } </pre>		

**Table 4.14 – Syntax of the ASEE top level payload for the audio object type AAC scalable (aac\_scalable\_extension\_element)**

Syntax	No. of bits	Mnemonic
<pre> aac_scalable_extension_element() {     aac_scalable_extension_header()     for (ch=0; ch&lt;(this_layer_stereo ? 2:1); ch++){         individual_channel_stream(1, 1)     }     cnt = bits_to_decode() / 8     while (cnt &gt;= 1) {         cnt -= extension_payload(cnt)     } } </pre>		

**Table 4.15 – Syntax of aac\_scalable\_main\_header()**

Syntax	No. of bits	Mnemonic
<pre> aac_scalable_main_header() {     <b>ics_reserved_bit</b>     If ( tvq_layer_present == 0 ) {         <b>window_sequence</b>     } } </pre>	<p>1</p> <p>2</p>	<p><b>bslbf</b></p> <p><b>uimsbf</b></p>

<b>window_shape</b>	1	<b>uimsbf</b>
}		
if( window_sequence == EIGHT_SHORT_SEQUENCE ) {		
<b>max_sfb</b>	4	<b>uimsbf</b>
<b>scale_factor_grouping</b>	7	<b>uimsbf</b>
} else {		
<b>max_sfb</b>	6	<b>uimsbf</b>
}		
if( this_layer_stereo ) {		
<b>ms_mask_present</b>	2	<b>bslbf</b>
if( ms_mask_present == 1 ) {		
ms_data()		
}		
}		
if( mono_stereo_flag && ( core_flag		
(tvq_layer_present && tvq_mono_tns==0) ) )		
<b>tns_channel_mono_layer</b>	1	<b>bslbf</b>
for( ch=0; ch< (this_layer_stereo ? 2:1 ); ch++ ) {		
if ( !tvq_layer_present    (tns_aac_tvq_en[ch] == 1) ) {		
<b>tns_data_present</b>	1	<b>bslbf</b>
if( tns_data_present )		
tns_data()		
}		
if( core_flag    tvq_layer_present ) {		
if((ch==0)    ((ch==1) && (core_stereo    tvq_stereo) )		
diff_control_data()		
if( mono_stereo_flag		
diff_control_data_lr()		
} else {		
<b>ltp_data_present</b>	1	<b>bslbf</b>
if( ltp_data_present ) {		
ltp_data ()		
}		
}		
}		
}		

Table 4.16 – Syntax of aac\_scalable\_extension\_header()

Syntax	No. of bits	Mnemonic
aac_scalable_extension_header()		
{		
if( window_sequence == EIGHT_SHORT_SEQUENCE ) {		
<b>max_sfb</b>	4	<b>bslbf</b>
} else {		
<b>max_sfb</b>	6	<b>bslbf</b>
}		
if( this_layer_stereo ) {		
<b>ms_mask_present</b>	2	<b>bslbf</b>
if( ms_mask_present == 1 ) {		
ms_data()		
}		
}		
if( mono_stereo_flag ) {		
for( ch=0; ch<2; ch++ ) {		

```

        tns_data_present          1          bslbf
        if( tns_data_present)
            tns_data()
    }
}
if ( ( mono_layer_flag ) && (this_layer_stereo) ){
    for( ch=0; ch<2; ch++ ) {
        diff_control_data_lr()
    }
}
}

```

**Table 4.17 – Syntax of diff\_control\_data()**

Syntax	No. of bits	Mnemonic
diff_control_data() { if(window_sequence == EIGHT_SHORT_SEQUENCE) for( win=0; win<8; w++ ) <b>diff_control[win][0]</b>	<b>1</b>	<b>bslbf</b>
else for( dc_group=0; dc_group<no_of_dc_groups; dc_group++ ) <b>diff_control[0][dc_group]</b>	<b>2..5</b>	<b>bslbf</b>
}		

**Table 4.18 – Syntax of diff\_control\_data\_lr()**

Syntax	No. of bits	Mnemonic
diff_control_data_lr() { if( window_sequence != EIGHT_SHORT_SEQUENCE ) { for( sfb=last_max_sfb_ms; sfb<min(last_mono_max_sfb;max_sfb); sfb++ ) if( !ms_used[0][sfb]) <b>diff_control_lr[0][sfb]</b>	<b>1</b>	<b>bslbf</b>
} else { if( last_max_sfb_ms==0) /* only in the first stereo layer*/ for( win=0; win<8; win++ ) <b>diff_control_lr[win][0]</b>	<b>1</b>	<b>bslbf</b>
} }		

**4.4.2.3 Payloads for the audio object types ER AAC LC, ER AAC LTP and ER AAC LD**

**Table 4.19 – Syntax of top level payload for audio object types ER AAC LC, ER AAC LTP and ER AAC LD (er\_raw\_data\_block())**

Syntax	No. of bits	Mnemonic
er_raw_data_block() { if ( channelConfiguration == 0 ) { /* reserved */ } if ( channelConfiguration == 1 ) {		

```

    single_channel_element ();
}
if ( channelConfiguration == 2 ) {
    channel_pair_element ();
}
if ( channelConfiguration == 3 ) {
    single_channel_element ();
    channel_pair_element ();
}
if ( channelConfiguration == 4 ) {
    single_channel_element ();
    channel_pair_element ();
    single_channel_element ();
}
if ( channelConfiguration == 5 ) {
    single_channel_element ();
    channel_pair_element ();
    channel_pair_element ();
}
if ( channelConfiguration == 6 ) {
    single_channel_element ();
    channel_pair_element ();
    channel_pair_element ();
    lfe_channel_element ();
}
if ( channelConfiguration == 7 ) {
    single_channel_element ();
    channel_pair_element ();
    channel_pair_element ();
    channel_pair_element ();
    lfe_channel_element ();
}
if ( channelConfiguration >= 8 ) {
    /* reserved */
}
cnt = bits_to_decode() / 8;
while ( cnt >= 1 ) {
    cnt -= extension_payload(cnt);
}
}

```

#### 4.4.2.4 Payloads for the audio object type Twin\_VQ

**Table 4.20 – Syntax of the TSME top level payload for the audio object type Twin\_VQ (tvq\_scalable\_main\_element)**

Syntax	No. of bits	Mnemonic
tvq_scalable_main_element() { tvq_scalable_main_header() vq_single_element(0) }		

**Table 4.21 – Syntax of the TSEE top level payload for the audio object type Twin\_VQ (tvq\_scalable\_extension\_element)**

Syntax	No. of bits	Mnemonic

```

tvq_scalable_extension_element()
{
    tvq_scalable_extension_header()
    vq_single_element(layer)
}
    
```

**Table 4.22 – Syntax of tvq\_scalable\_main\_header()**

Syntax	No. of bits	Mnemonic
tvq_scalable_main_header()		
{		
<b>Window_sequence</b>	<b>2</b>	<b>bslbf</b>
<b>window_shape</b>	<b>1</b>	<b>bslbf</b>
if( this_layer_stereo ) {		
<b>ms_mask_present</b>	<b>2</b>	<b>bslbf</b>
if( ms_mask_present == 1 ) {		
if ( window_sequence ==		
EIGHT_SHORT_SEQUENCE )		
<b>scale_factor_grouping</b>	<b>7</b>	<b>bslbf</b>
ms_data()		
}		
}		
for( ch=0; ch< (this_layer_stereo ? 2:1); ch++ ) {		
<b>ltp_data_present</b>	<b>1</b>	<b>bslbf</b>
if (ltp_data_present)		
ltp_data ()		
<b>tns_data_present</b>	<b>1</b>	<b>bslbf</b>
if( tns_data_present )		
tns_data()		
}		
}		

**Table 4.23 – Syntax of tvq\_scalable\_extension\_header()**

Syntax	No. of bits	Mnemonic
tvq_scalable_extension_header()		
{		
if( this_layer_stereo ) {		
<b>ms_mask_present</b>	<b>2</b>	<b>bslbf</b>
if( ms_mask_present == 1 ) {		
ms_data()		
}		
}		
}		

**Table 4.24 – Syntax of vq\_single\_element**

Syntax	No. of bits	Mnemonic
vq_single_element( lyr)		
{		
if (lyr == 0)		
<b>bandlimit_present</b>	<b>1</b>	<b>uimsbf</b>

if (window_sequence != EIGHT_SHORT_SEQUENCE && lyr == 0){ <b>ppc_present</b>	1	uimsbf
<b>postprocess_present</b>	1	uimsbf
} if (lyr >= 1) for (i_ch=0; i_ch<n_ch; i_ch++){ <b>fb_shift[i_ch]</b>	2	uimsbf
} if (lyr == 0 && <b>bandlimit_present</b> ){ for (i_ch=0; i_ch<n_ch; i_ch++){ <b>index_blim_h[i_ch]</b>	2	uimsbf
<b>index_blim_l[i_ch]</b>	1	uimsbf
} } if ( <b>ppc_present</b> ){ for (idiv=0; idiv<N_DIV_P; idiv++){ <b>index_shape0_p[idiv]</b>	7	uimsbf
<b>index_shape1_p[idiv]</b>	7	uimsbf
} for (i_ch=0; i_ch<n_ch; i_ch++){ <b>index_pit[i_ch]</b>	8	uimsbf
<b>index_pgain[i_ch]</b>	7	uimsbf
} } for (idiv=0; idiv<N_DIV; idiv++){ <b>index_shape0[idiv]</b>	5/6	uimsbf
<b>index_shape1[idiv]</b>	5/6	uimsbf
} for (i_ch=0; i_ch<n_ch; i_ch++){ for (isb=0; isb<N_SF; isb++){ for (ifdiv=0; ifdiv<FW_N_DIV; ifdiv++){ <b>index_env[i_ch][isb][ifdiv]</b>	0,6	uimsbf
} } } for (i_ch=0; i_ch<n_ch; i_ch++){ for (isbm=0; isbm<N_SF; isbm++){ <b>index_fw_alf[i_ch][isbm]</b>	0,1	uimsbf
} } for (i_ch=0; i_ch<n_ch; i_ch++){ <b>index_gain[i_ch]</b>	8..9	uimsbf
if (N_SF[b_type]>1){ for (isbm=0; isbm<N_SF[b_type]; isbm++){ <b>index_gain_sb[i_ch][isbm]</b>	4	uimsbf
} } } for (i_ch=0; i_ch<n_ch; i_ch++){ <b>index_lsp0[i_ch]</b>	1	uimsbf
<b>index_lsp1[i_ch]</b>	6	uimsbf
for (isplt=0; isplt<LSP_SPLIT; isplt++){ <b>index_lsp2[i_ch][isplt]</b>	4	uimsbf
} } }		

4.4.2.5 Payloads for the audio object type ER TwinVQ

Table 4.25 – Syntax of ER TwinVQ object type (base)

Syntax	No. of bits	Mnemonic
<pre>tvq_scalable_main_element() {     Error_Sensitivity_Category1();     Error_Sensitivity_Category2(); }</pre>		

Table 4.26 – Syntax of ER TwinVQ object type (enhancement)

Syntax	No. of bits	Mnemonic
<pre>tvq_scalable_extension_element() {     Error_Sensitivity_Category3();     Error_Sensitivity_Category4(); }</pre>		

Table 4.27 – Syntax of Error\_Sensitivity\_Category1()

Syntax	No. of bits	Mnemonic
<pre>Error_Sensitivity_Category1() {     <b>window_sequence;</b>     <b>window_shape;</b>     if( this_layer_stereo) {         <b>ms_mask_present;</b>         if( ms_mask_present == 1) {             if ( window_sequence ==                 EIGHT_SHORT_SEQUENCE )                 <b>scale_factor_grouping;</b>             ms_data();         }     }     for( ch=0; ch&lt; (this_layer_stereo ? 2:1); ch++ ) {         <b>ltp_data_present;</b>         if (ltp_data_present)             ltp_data ();          <b>tns_data_present;</b>         if( tns_data_present )             tns_data();     }      <b>bandlimit_present;</b>     if (window_sequence != EIGHT_SHORT_SEQUENCE){         <b>ppc_present;</b>         <b>postprocess_present;</b>     }     if (bandlimit_present){         for (i_ch=0; i_ch&lt;n_ch; i_ch++){             <b>index_blim_h[i_ch];</b> </pre>	<p>2</p> <p>1</p> <p>2</p> <p>7</p> <p>1</p> <p>1</p> <p>1</p> <p>1</p> <p>1</p> <p>1</p> <p>2</p>	<p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p>

<pre>                 index_blim_l[i_ch];             }         }         if (ppc_present){             for (idiv=0; idiv&lt;N_DIV_P; idiv++){                 index_shape0_p[idiv];                 index_shape1_p[idiv];             }             for (i_ch=0; i_ch&lt;n_ch; i_ch++){                 index_pit[i_ch];                 index_pgain[i_ch];             }         }          for (i_ch=0; i_ch&lt;n_ch; i_ch++){             index_gain[i_ch];             if (N_SF[b_type]&gt;1){                 for (isbm=0; isbm&lt;N_SF[b_type]; isbm++){                     index_gain_sb[i_ch][isbm];                 }             }         }          for (i_ch=0; i_ch&lt;n_ch; i_ch++){             index_lsp0[i_ch];             index_lsp1[i_ch];             for (isplt=0; isplt&lt;LSP_SPLIT; isplt++){                 index_lsp2[i_ch][isplt];             }         }          for (i_ch=0; i_ch&lt;n_ch; i_ch++){             for (isb=0; isb&lt;N_SF; isb++){                 for (ifdiv=0; ifdiv&lt;FW_N_DIV; ifdiv++){                     index_env[i_ch][isb][ifdiv];                 }             }         }          for (i_ch=0; i_ch&lt;n_ch; i_ch++){             for (isbm=0; isbm&lt;N_SF; isbm++){                 index_fw_alf[i_ch][isbm];             }         }     } </pre>	<p>1</p> <p>7</p> <p>7</p> <p>8</p> <p>7</p> <p>9</p> <p>4</p> <p>1</p> <p>6</p> <p>4</p> <p>0,6</p> <p>0,1</p>	<p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p>
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Table 4.28 – Syntax of Error\_Sensitivity\_Category2()

Syntax	No. of bits	Mnemonic
<pre> Error_Sensitivity_Category2() {     for (idiv=0; idiv&lt;N_DIV; idiv++){         index_shape0[idiv];         index_shape1[idiv];     } } </pre>	<p>5/6</p> <p>5/6</p>	<p>uimsbf</p> <p>uimsbf</p>



Table 4.29 – Syntax of Error\_Sensitivity\_Category3()

Syntax	No. of bits	Mnemonic
<pre> Error_Sensitivity_Category3() {   if( this_layer_stereo ) {     <b>ms_mask_present;</b>     if( ms_mask_present == 1 ) {       ms_data();     }   }    for (i_ch=0; i_ch&lt;n_ch; i_ch++){     <b>fb_shift[i_ch];</b>   }    for (i_ch=0; i_ch&lt;n_ch; i_ch++){     <b>index_gain[i_ch];</b>     if (N_SF[b_type]&gt;1){       for (isbm=0; isbm&lt;N_SF[b_type]; isbm++){         <b>index_gain_sb[i_ch][isbm];</b>       }     }   }    for (i_ch=0; i_ch&lt;n_ch; i_ch++){     <b>index_lsp0[i_ch];</b>     <b>index_lsp1[i_ch]</b>     for (isplt=0; isplt&lt;LSP_SPLIT; isplt++){       <b>index_lsp2[i_ch][isplt];</b>     }   }    for (i_ch=0; i_ch&lt;n_ch; i_ch++){     for (isb=0; isb&lt;N_SF; isb++){       for (ifdiv=0; ifdiv&lt;FW_N_DIV; ifdiv++){         <b>index_env[i_ch][isb][ifdiv];</b>       }     }   }    for (i_ch=0; i_ch&lt;n_ch; i_ch++){     for (isbm=0; isbm&lt;N_SF; isbm++){       <b>index_fw_alf[i_ch][isbm];</b>     }   } } </pre>	<p>2</p> <p>2</p> <p>8</p> <p>4</p> <p>1</p> <p>6</p> <p>4</p> <p>0,6</p> <p>0,1</p>	<p><b>bslbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p> <p><b>uimsbf</b></p>

Table 4.30 – Syntax of Error\_Sensitivity\_Category4()

Syntax	No. of bits	Mnemonic
<pre> Error_Sensitivity_Category4() {   for (idiv=0; idiv&lt;N_DIV; idiv++){     <b>index_shape0[idiv];</b>     <b>index_shape1[idiv];</b>   } } </pre>	<p>5/6</p> <p>5/6</p>	<p><b>uimsbf</b></p> <p><b>uimsbf</b></p>

## 4.4.2.6 Payloads for the audio object type ER BSAC

Table 4.31 – Syntax of top level payload for audio object type ER BSAC (bsac\_payload())

Syntax	No. of bits	Mnemonic
<pre>bsac_payload(layer) {     for (frm=0; frm&lt;numOfSubFrame; frm++) {         bsac_lstep_element(frm, layer);     } } /* bsac_lstep_element(frm, layer) should be mapped to the fine grain audio data, bsac_raw_data_block(), for the actual decoding. See subclause "Decoding of payload for audio object type ER BSAC" for more detailed description.*/ }</pre>		

Table 4.32 – Syntax of bsac\_lstep\_element()

Syntax	No. of bits	Mnemonic
<pre>bsac_lstep_element(frm, layer) {     offset=LayerStartByte[frm][layer];     for(i=0;i&lt;LayerLength[frm][layer];i++)         bsac_stream_byte[frm][offset+i]; } /* bsac_stream_byte should be mapped to the fine grain audio data, bsac_raw_data_block(), for the actual decoding. See subclause "Decoding of payload for audio object type ER BSAC" for more detailed description. */ }</pre>	8	uimsbf

Table 4.33 – bsac\_raw\_data\_block()

Syntax	No. of bits	Mnemonic
<pre>bsac_raw_data_block() {     bsac_base_element();     layer=slayer_size;     while(data_available() &amp;&amp; layer&lt;(top_layer+slayer_size)) {         bsac_layer_element(nch, layer);         layer++;     }     byte_alignment(); }</pre>		

Table 4.34 – Syntax of bsac\_base\_element()

Syntax	No. of bits	Mnemonic
bsac_base_element() { <b>frame_length</b> ; bsac_header(); general_header(); byte_alignment(); for (slayer = 0; slayer < slayer_size; slayer++) bsac_layer_element(slayer); }	11	uimbf

Table 4.35 – Syntax of bsac\_header()

Syntax	No. of bits	Mnemonic
bsac_header() { <b>header_length</b> ; <b>sba_mode</b> ; <b>top_layer</b> ; <b>base_snf_thr</b> ;  for(ch=0;ch<nch;ch++) <b>max_scalefactor[ch]</b> ;  <b>base_band</b> ;  for(ch=0;ch<nch;ch++) { <b>cband_si_type[ch]</b> ;  <b>base_scf_model[ch]</b> ; <b>enh_scf_model[ch]</b> ; <b>max_sfb_si_len[ch]</b> ; } }	4 1 6 2  8  5  5 3 3 4	uimbf uimbf uimbf uimbf  uimbf  uimbf  uimbf uimbf uimbf

Table 4.36 – Syntax of general\_header()

Syntax	No. of bits	Mnemonic
general_header () { <b>reserved_bit</b> ; <b>window_sequence</b> ; <b>window_shape</b> ; if( window_sequence == EIGHT_SHORT_SEQUENCE ) { <b>max_sfb</b> ; <b>scale_factor_grouping</b> ; } else { <b>max_sfb</b> ; }  <b>pns_data_present</b> ; if (pns_data_present) <b>pns_start_sfb</b> ;  if( nch == 2 ) <b>ms_mask_present</b> ; }	1 2 1 4 7 6 1 6 2	bslbf uimsbf uimsbf uimsbf uimsbf uimsbf uimbf uimbf bslbf

<pre> for( ch=0 ch&lt; nch; ch++ ) {   <b>tns_data_present[ch];</b>   if( tns_data_present[ch] )     tns_data();    <b>ltp_data_present[ch];</b>   if( ltp_data_present[ch] )     ltp_data(last_max_sfb, max_sfb); } </pre>	1	bslbf
<pre> </pre>	1	bslbf

Table 4.37 – Syntax of bsac\_layer\_element()

Syntax	No. of bits	Mnemonic
<pre> bsac_layer_element(layer) {   layer_cband_si(layer);   layer_sfb_si(layer);    bsac_layer_spectra (layer);   if (!sba_mode) {     bsac_lower_spectra (layer)   }   else if (terminal_layer[layer]) {     bsac_lower_spectra (layer);     bsac_higher_spectra (layer);   } } </pre>		

Table 4.38 – Syntax of layer\_cband\_si()

Syntax	No. of bits	Mnemonic
<pre> layer_cband_si (layer) {   g = layer_group[layer];   for (ch=0; ch&lt;nch; ch++) {     for(cband=layer_start_cband[g][layer];        cband&lt;layer_end_cband[g][layer]; cband++) {       <b>acode_cband_si[ch][g][cband];</b>     }   } } </pre>	1..14	bslbf

Table 4.39 – Syntax of layer\_sfb\_si()

Syntax	No. of bits	Mnemonic
<pre> layer_sfb_si (layer) {   g = layer_group[layer];   for (ch=0; ch&lt;nch; ch++)   for(sfb=layer_start_sfb[layer];sfb&lt;layer_end_sfb[layer];sfb++) {     if (nch==1) {       if(pns_data_present &amp;&amp; sfb &gt;= pns_start_sfb) {         <b>acode_noise_flag[g][sfb];</b>       }     }   } } </pre>	1	bslbf

<pre> } else if (stereo_side_info_coded[g][sfb]==0) {   if (ms_mask_present !=2 ) {     if (ms_mask_present==1) {       <b>acode_ms_used[g][sfb];</b>       pns_data_present = 0;     } else if (ms_mask_present==3) {       <b>acode_stereo_info[g][sfb];</b>     }     if(pns_data_present &amp;&amp; sfb&gt;=pns_start_sfb) {       <b>acode_noise_flag_l[g][sfb];</b>       <b>acode_noise_flag_r[g][sfb];</b>       if(ms_mask_present==3 &amp;&amp; stereo_info==3)         {           if(noise_flag_l &amp;&amp; noise_flag_r){             <b>acode_noise_mode[g][sfb];</b>           }         }     }     stereo_side_info_coded[g][sfb] = 1;   }   if (noise_flag[ch][g][sfb]) {     if (noise_pcm_flag[ch]==1) {       <b>acode_max_noise_energy[ch];</b>       noise_pcm_flag[ch] = 0;     }     <b>acode_dpcm_noise_energy_index[ch][g][sfb];</b>   } else if (stereo_info[g][sfb]&gt;=2 &amp;&amp; ch==1) {     <b>acode_is_position_index[g][sfb];</b>   } else {     <b>acode_scf_index[ch][g][sfb];</b>   } } </pre>	<p><b>1</b></p> <p><b>0..4</b></p> <p><b>1</b></p> <p><b>1</b></p> <p><b>2</b></p> <p><b>9</b></p> <p><b>0..14</b></p> <p><b>0..14</b></p> <p><b>1..14</b></p>	<p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p> <p><b>bslbf</b></p>
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**Table 4.40 – Syntax of bsac\_layer\_spectra()**

Syntax	No. of bits	Mnemonic
<pre> bsac_layer_spectra(layer) {   g = layer_group[layer];   start_index[g] = layer_start_index[layer];   end_index[g] = layer_end_index[layer];   if (layer &lt; slayer_size)     thr_snf = base_snf_thr;   else     thr_snf = 0;   bsac_spectral_data (g, g+1, thr_snf, cur_snf); } </pre>		

Table 4.41 – Syntax of `bsac_lower_spectra()`

Syntax	No. of bits	Mnemonic
<pre> bsac_lower_spectra(layer) {   for (g = 0; g &lt; num_window_groups; g++) {     start_index[g] = 0;     end_index[g] = 0;   }   for (play = 0; play &lt; layer; play++) {     end_index[-layer_group[play]] = layer_end_index[play];   }   bsac_spectral_data (0, num_window_groups, 0, unc_snf); } </pre>		

Table 4.42 – Syntax of `bsac_higher_spectra()`

Syntax	No. of bits	Mnemonic
<pre> bsac_higher_spectra(layer) {   for (nlay=layer+1;nlay &lt; top_layer+slayer_size;nlay++) {     g = layer_group[nlay];     start_index[g] = layer_start_index[nlay];     end_index[g] = layer_end_index[nlay];     bsac_spectral_data (g, g+1, 0, unc_snf);   } } </pre>		

Table 4.43 – Syntax of `bsac_spectral_data ()`

Syntax	No. of bits	Mnemonic
<pre> bsac_spectral_data(start_g, end_g, thr_snf, cur_snf) {   if (layer_data_available()) return;    for (snf=maxsnf; snf&gt;thr_snf; snf--)   for (g = start_g; g &lt; end_g; g++)   for (i=start_index[g];i&lt;end_index[g]; i++)   for(ch=0;ch&lt;nch;ch++) {     if ( cur_snf[ch][g][i]&lt;snf) continue;      if (!sample[ch][g][i]    sign_is_coded[ch] [g][i])       <b>acod_sliced_bit[ch][g][i][snf];</b> </pre>	<b>0..6</b>	<b>bslbf</b>
<pre>     if (sample[ch][g][i] &amp;&amp; !sign_is_coded[ch] [g][i]) {       if (layer_data_available()) return;       <b>acod_sign[ch][g][i];</b>       sign_is_coded[ch][g][i] = 1;     }     cur_snf[ch][g][i]--;     if (layer_data_available()) return;   } } </pre>	<b>1</b>	<b>bslbf</b>

4.4.2.7 Subsidiary payloads

Table 4.44 – Syntax of individual\_channel\_stream()

Syntax	No. of bits	Mnemonic
individual_channel_stream ( common_window, scale_flag ) { <b>global_gain;</b> if ( ! common_window && ! scale_flag ) { ics_info (); } section_data (); scale_factor_data (); if ( ! scale_flag ) { <b>pulse_data_present;</b> if ( pulse_data_present ) { pulse_data (); } <b>tns_data_present;</b> if ( tns_data_present ) { tns_data (); } <b>gain_control_data_present;</b> if ( gain_control_data_present ) { gain_control_data (); } } if ( ! aacSpectralDataResilienceFlag ) { spectral_data (); } else { <b>length_of_reordered_spectral_data;</b> <b>length_of_longest_codeword;</b> reordered_spectral_data (); } }	8	<b>uimsbf</b>
	1	<b>uimsbf</b>
	1	<b>uimsbf</b>
	1	<b>uimsbf</b>
	14	<b>uimsbf</b>
	6	<b>uimsbf</b>

Table 4.45 – Syntax of reordered\_spectral\_data ()

Syntax	No. of bits	Mnemonic
reordered_spectral_data () { /* complex reordering, see tool description of Huffman codeword reordering (subclause 4.6.16.3) */ }		

Table 4.46 – Syntax of section\_data()

Syntax	No. of bits	Mnemonic
section_data() { if ( window_sequence == EIGHT_SHORT_SEQUENCE ) { sect_esc_val = ( 1 << 3 ) - 1; } else { sect_esc_val = ( 1 << 5 ) - 1; } }		

for ( g = 0; g < num_window_groups; g++ ) {		
k = 0;		
i = 0;		
while ( k < max_sfb ) {		
if ( aacSectionDataResilienceFlag )		
<b>sect_cb[g][i];</b>	<b>5</b>	<b>uimsbf</b>
}		
else {		
<b>sect_cb[g][i];</b>	<b>4</b>	<b>uimsbf</b>
}		
sect_len = 0;		
if ( ! aacSectionDataResilienceFlag		
sect_cb < 11    ( sect_cb > 11 && sect_cb < 16 ) ) {		
while ( <b>sect_len_incr</b> == sect_esc_val ) {	<b>3/5</b>	<b>uimsbf</b>
sect_len += sect_esc_val;		
}		
}		
else {		
sect_len_incr = 1;		
}		
sect_len += sect_len_incr;		
sect_start[g][i] = k;		
sect_end[g][i] = k + sect_len;		
for ( sfb = k; sfb < k + sect_len; sfb++ ) {		
sfb_cb[g][sfb] = sect_cb[g][i];		
}		
k += sect_len;		
i++;		
}		
num_sec[g] = i;		
}		

Table 4.47 – Syntax of scale\_factor\_data()

Syntax	No. of bits	Mnemonic
scale_factor_data()		
{		
if ( ! aacScalefactorDataResilienceFlag ) {		
noise_pcm_flag = 1;		
for ( g = 0; g < num_window_groups; g++ ) {		
for ( sfb = 0; sfb < max_sfb; sfb++ ) {		
if ( sfb_cb[g][sfb] != ZERO_HCB ) {		
if ( is_intensity ( g, sfb ) ) {		
<b>hcod_sf[dpcm_is_position[g][sfb]];</b>	<b>1..19</b>	<b>vlclbf</b>
}		
else {		
if ( is_noise(g, sfb) ) {		
if ( noise_pcm_flag ) {		
noise_pcm_flag = 0;		
<b>dpcm_noise_nrg[g][sfb];</b>	<b>9</b>	<b>uimsbf</b>
}		
else {		
<b>hcod_sf[dpcm_noise_nrg[g][sfb]];</b>	<b>1..19</b>	<b>vlclbf</b>
}		
}		
else {		
<b>hcod_sf[dpcm_sf[g][sfb]];</b>	<b>1..19</b>	<b>vlclbf</b>
}		
}		
}		
}		
}		



<pre>                 }             }         }     } else {     intensity_used = 0;     noise_used = 0; <b>sf_concealment;</b> <b>rev_global_gain;</b> <b>length_of_rvlc_sf;</b>     for ( g = 0; g &lt; num_window_groups; g++ ) {         for ( sfb=0; sfb &lt; max_sfb; sfb++ ) {             if ( sfb_cb[g][sfb] != ZERO_HCB ) {                 if ( is_intensity ( g, sfb ) ) {                     intensity_used = 1;                     <b>rvlc_cod_sf[dpcm_is_position[g][sfb]];</b>                 }                 else {                     if ( is_noise(g,sfb) ) {                         if ( ! noise_used ) {                             noise_used = 1;                             <b>dpcm_noise_nrg[g][sfb];</b>                         }                         else {                             <b>rvlc_cod_sf[dpcm_noise_nrg[g][sfb]];</b>                         }                     }                     else {                         <b>rvlc_cod_sf[dpcm_sf[g][sfb]];</b>                     }                 }             }         }     } } if ( intensity_used ) {     <b>rvlc_cod_sf[dpcm_is_last_position];</b> } <b>sf_escapes_present;</b> if ( sf_escapes_present ) {     <b>length_of_rvlc_escapes;</b>     for ( g = 0; g &lt; num_window_groups; g++ ) {         for ( sfb = 0; sfb &lt; max_sfb; sfb++ ) {             if ( sfb_cb[g][sfb] != ZERO_HCB ) {                 if ( is_intensity ( g, sfb ) &amp;&amp;                     dpcm_is_position[g][sfb] == ESC_FLAG ) {                     <b>rvlc_esc_sf[dpcm_is_position[g][sfb]];</b>                 }                 else {                     if ( is_noise ( g, sfb ) &amp;&amp;                         dpcm_noise_nrg[g][sfb] == ESC_FLAG ) {                         <b>rvlc_esc_sf[dpcm_noise_nrg[g][sfb]];</b>                     }                     else {                         if ( dpcm_sf[g][sfb] == ESC_FLAG ) {                             <b>rvlc_esc_sf[dpcm_sf[g][sfb]];</b>                         }                     }                 }             }         }     } } </pre>	<p>1</p> <p>8</p> <p>11/9</p> <p>1..9</p> <p>9</p> <p>1..9</p> <p>1..9</p> <p>1..9</p> <p>1..9</p> <p>1</p> <p>8</p> <p>2..20</p> <p>2..20</p> <p>2..20</p>	<p>uimsbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>vlclbf</p> <p>uimsbf</p> <p>vlclbf</p> <p>vlclbf</p> <p>vlclbf</p> <p>vlclbf</p> <p>uimsbf</p> <p>uimsbf</p> <p>vlclbf</p> <p>vlclbf</p> <p>vlclbf</p>
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<pre>         }       }       if ( intensity_used &amp;&amp;           dpcm_is_position[g][sfb] == ESC_FLAG ) {           <b>rvlc_esc_sf[dpcm_is_last_position];</b>       }     }     if ( noise_used ) {       <b>dpcm_noise_last_position;</b>     }   } } </pre>	2..20	vlclbf
<pre>           <b>dpcm_noise_last_position;</b>         }       }     }   } } </pre>	9	uimsbf

Table 4.48 – Syntax of tns\_data()

Syntax	No. of bits	Mnemonic
tns_data()		
{		
for (w=0; w<num_windows; w++) {		
<b>n_filt[w]</b>	1..2	uimsbf
if (n_filt[w])		
<b>coef_res[w]</b>	1	uimsbf
for (filt=0; filt<n_filt[w]; filt++) {		
<b>length[w][filt]</b>	{4;6}	uimsbf
<b>order[w][filt]</b>	{3;5}	uimsbf
if (order[w][filt]) {		
<b>direction[w][filt]</b>	1	uimsbf
<b>coef_compress[w][filt]</b>	1	uimsbf
for (i=0; i<order[w][filt]; i++)		
<b>coef[w][filt][i]</b>	2..4	uimsbf
}		
}		
}		
}		

Table 4.49 – Syntax of ltp\_data()

Syntax	No. of bits	Mnemonic
ltp_data()		
{		
if ( AudioObjectType == ER AAC LD ) {		
<b>ltp_lag_update;</b>	1	uimsbf
if ( ltp_lag_update ) {		
<b>ltp_lag;</b>	10	uimsbf
} else {		
ltp_lag = ltp_prev_lag;		
}		
<b>ltp_coef;</b>	3	uimsbf
for ( sfb = 0; sfb < max_sfb; sfb++ ) {		
<b>ltp_long_used[sfb];</b>	1	uimsbf
}		
}		
else {		
<b>ltp_lag</b>	11	uimsbf
<b>ltp_coef</b>	3	uimsbf
if(window_sequence==EIGHT_SHORT_SEQUENCE) {		

for (w=0; w<num_windows; w++ ) {		
<b>ltp_short_used[w]</b>	<b>1</b>	<b>uimsbf</b>
if (ltp_short_used [w]) {		
<b>ltp_short_lag_present[w]</b>	<b>1</b>	<b>uimsbf</b>
if (ltp_short_lag_present[w]) {		
<b>ltp_short_lag[w]</b>	<b>4</b>	<b>uimsbf</b>
}		
}		
}		
} else {		
for ( sfb=0; sfb<max_sfb; sfb++ ) {		
<b>ltp_long_used[sfb]</b>	<b>1</b>	<b>uimsbf</b>
}		
}		
}		

Table 4.50 – Syntax of spectral\_data()

Syntax	No. of bits	Mnemonic
Spectral_data()		
{		
for( g=0; g<num_window_groups; g++ ) {		
for (i=0; i<num_sec[g]; i++) {		
if (sect_cb[g][i] != ZERO_HCB &&		
sect_cb[g][i] != NOISE_HCB &&		
sect_cb[g][i] != INTENSITY_HCB &&		
sect_cb[g][i] != INTENSITY_HCB2 ) {		
for (k=sect_sfb_offset[g][sect_start[g][i]];		
k < sect_sfb_offset[g][sect_end[g][i]]; ) {		
if (sect_cb[g][i]<FIRST_PAIR_HCB) {		
<b>hcod</b> [sect_cb[g][i][w][x][y][z]	<b>1.16</b>	<b>bslbf</b>
if( unsigned_cb[sect_cb[g][i]] )		
<b>quad_sign_bits</b>	<b>0.4</b>	<b>bslbf</b>
k += QUAD_LEN		
}		
else {		
<b>hcod</b> [sect_cb[g][i][y][z]	<b>1.15</b>	<b>bslbf</b>
if( unsigned_cb[sect_cb[g][i]] )		
<b>pair_sign_bits</b>	<b>0.2</b>	<b>bslbf</b>
k += PAIR_LEN		
if (sect_cb[g][i]==ESC_HCB) {		
if (y==ESC_FLAG)		
<b>hcod_esc_y</b>	<b>5.21</b>	<b>bslbf</b>
if (z==ESC_FLAG)		
<b>hcod_esc_z</b>	<b>5.21</b>	<b>bslbf</b>
}		
}		
}		
}		
}		

Table 4.51 – Syntax of extension\_payload()

extension_payload(cnt)		
{		
<b>extension_type</b>	4	uimsbf
switch( extension_type ) {		
case EXT_DYNAMIC_RANGE:		
n = dynamic_range_info();		
return n;		
case EXT_FILL_DATA:		
<b>fill_nibble</b> /* must be '0000' */	4	uimsbf
for (i=0; i<cnt-1; i++)		
<b>fill_byte[i]</b> /* must be '10100101' */	8	uimsbf
return cnt		
case default:		
for (i=0; i<8*(cnt-1)+4; i++)		
<b>other_bits[i]</b>	1	uimsbf
return cnt		
}		
}		

Table 4.52 – Syntax of dynamic\_range\_info()

Syntax	No. of bits	Mnemonic
dynamic_range_info()		
{		
n = 1		
drc_num_bands = 1		
<b>pce_tag_present</b>	1	uimsbf
if (pce_tag_present == 1) {		
<b>pce_instance_tag</b>	4	uimsbf
<b>drc_tag_reserved_bits</b>	4	uimsbf
n++		
}		
<b>excluded_chns_present</b>	1	uimsbf
if (excluded_chns_present == 1) {		
n += excluded_channels()		
}		
<b>drc_bands_present</b>	1	uimsbf
if (drc_bands_present == 1) {		
<b>drc_band_incr</b>	4	uimsbf
<b>drc_bands_reserved_bits</b>	4	uimsbf
n++		
drc_num_bands = drc_num_bands + drc_band_incr		
for (i=0; i<drc_num_bands; i++) {		
<b>drc_band_top[i]</b>	8	uimsbf
n++		
}		
}		
<b>prog_ref_level_present</b>	1	uimsbf
if (prog_ref_level_present == 1) {		
<b>prog_ref_level</b>	7	uimsbf
<b>prog_ref_level_reserved_bits</b>	1	uimsbf
n++		
}		
for (i=0; i<drc_num_bands; i++) {		
<b>dyn_rng_sgn[i]</b>	1	uimsbf
<b>dyn_rng_ctl[i]</b>	7	uimsbf

```

        n++
    }
    return n
}

```

**Table 4.53 – Syntax of excluded\_channels()**

Syntax	No. of bits	Mnemonic
<pre> Excluded_channels( ) {     n = 0     num_excl_chan = 7     for (i=0; i&lt;7; i++)         <b>exclude_mask[ i ]</b>     n++     while (<b>additional_excluded_chns[n-1] == 1</b>) {         for (i= num_excl_chan; i&lt; num_excl_chan+7; i++)             <b>exclude_mask[ i ]</b>         n++         num_excl_chan += 7     }     return n } </pre>	1	<b>uimsbf</b>
	1	<b>uimsbf</b>
	1	<b>uimsbf</b>

**Table 4.54 – Syntax of ms\_data()**

Syntax	No. of bits	Mnemonic
<pre> ms_data() {     for( g=0; g&lt;num_window_groups; g++ ) {         for( sfb=last_max_sfb_ms; sfb&lt;max_sfb; sfb++) {             <b>ms_used[g][sfb];</b>         }     } } </pre>	1	<b>bslbf</b>

## 4.5 General information

### 4.5.1 Decoding of the GA specific configuration

#### 4.5.1.1 GASpecificConfig()

The call parameters ‘samplingFrequencyIndex’, ‘channelConfiguration’, ‘audioObjectType’ are passed down from the audio specific configuration element defined in subpart 1. The information contained in these parameters is mandatory for the decoding process.

If the sampling rate is not one of the rates listed in the right column in Table 4.55, the sampling frequency dependent tables (code tables, scale factor band tables etc.) must be deduced in order for the bit stream to be parsed. Since a given sampling frequency is associated with only one sampling frequency table, and since maximum flexibility is desired in the range of possible sampling frequencies, the following table shall be used to associate an implied sampling frequency with the desired sampling frequency dependent tables. However, there is one exception to this rule, which is described in subclause 4.6.14.1 for Table 4.112.

Table 4.55 – Sampling frequency mapping

Frequency range (in Hz)	Use tables for sampling frequency (in Hz)
$f \geq 92017$	96000
$92017 > f \geq 75132$	88200
$75132 > f \geq 55426$	64000
$55426 > f \geq 46009$	48000
$46009 > f \geq 37566$	44100
$37566 > f \geq 27713$	32000
$27713 > f \geq 23004$	24000
$23004 > f \geq 18783$	22050
$18783 > f \geq 13856$	16000
$13856 > f \geq 11502$	12000
$11502 > f \geq 9391$	11025
$9391 > f$	8000

If a certain sampling frequency dependent table stated in the right column of Table 4.55 is not defined, the nearest defined table shall be used.

<b>frameLengthFlag</b>	The window length of the IMDCT: if set to 0 a 1024 lines IMDCT is used and frameLength is set to 1024, if set to 1 a 960 line IMDCT is used and frameLength is set to 960.
<b>DependsOnCoreCoder</b>	Set to 1 if a coder at a different sampling rate is used as a core coder in a scalable bitstream.
<b>CoreCoderDelay</b>	The delay in samples that has to be applied to the up-sampled core decoder output, before the MDCT calculation. To save memory it is also possible to delay the bitstream by the appropriate number of core frames instead. In that case it could be necessary to decode a core frame more than once.
<b>extensionFlag:</b>	Shall be '0' for audio object types 1, 2, 3, 4, 6, 7. Shall be '1' for audio object types 17, 19, 20, 21, 22, 23.
<b>layerNr:</b>	A 3-bit field indicating the AAC layer number in a scalable configuration. The first AAC layer is signaled by a value of 0.
<b>numOfSubFrame</b>	A 5-bit unsigned integer value representing the number of the sub-frames which are grouped and transmitted in a super-frame.
<b>layer_length</b>	An 11-bit unsigned integer value representing the average length of the large-step layers in bytes.
<b>aacSectionDataResilienceFlag</b>	This flag signals a different coding scheme of AAC section data. If codebook 11 is used, this scheme transmits additional information about the maximum absolute value for spectral lines. This allows error detection of spectral lines that are larger than this value.
<b>aacScalefactorDataResilienceFlag</b>	This flag signals a different coding scheme of the AAC scalefactor data, that is more resilient against errors as the original one.
<b>aacSpectralDataResilienceFlag</b>	This flag signals a different coding scheme (HCR) of the AAC spectral data, that is more resilient against errors as the original one
<b>extensionFlag3</b>	Extension flag for the future use. Shall be '0'.

#### Restrictions:

program\_config\_element() shall be used only for the audio object types AAC main, AAC SSR, AAC LC and AAC LTP.

An MPEG-4 Audio decoder is only required to follow the Program Configuration Element in GASpecificConfig(). The decoder shall ignore any Program Configuration Elements that may occur in raw data blocks. PCEs transmitted in raw data blocks cannot be used to convey decoder configuration information.

4.5.1.2 Program Config Element (PCE)

**object\_type** The two-bit object type index from Table 4.56. This field replaces the profile field of the PCE in ISO/IEC 13818-7 in a backward compatible way.

**Table 4.56 – Object type index**

Index	object type
0	AAC Main
1	AAC LC
2	AAC SSR
3	AAC LTP

**sampling\_frequency\_index** Indicates the sampling rate of the program (and all other programs in this bitstream). See definition in subpart 1.

**num\_front\_channel\_elements** The number of audio syntactic elements in the front channels, front center to back center, symmetrically by left and right, or alternating by left and right in the case of single channel elements

**num\_side\_channel\_elements** Number of elements to the side as above

**num\_back\_channel\_elements** As number of side and front channel elements, for back channels

**num\_lfe\_channel\_elements** number of LFE channel elements associated with this program

**num\_assoc\_data\_elements** The number of associated data elements for this program

**num\_valid\_cc\_elements** The number of CCE's that can add to the audio data for this program

**mono\_mixdown\_present** One bit, indicating the presence of the mono mixdown element

**mono\_mixdown\_element\_number** The number of a specified SCE that is the mono mixdown

**stereo\_mixdown\_present** One bit, indicating that there is a stereo mixdown present

**stereo\_mixdown\_element\_number** The number of a specified CPE that is the stereo mixdown element

**matrix\_mixdown\_idx\_present** One bit, indicating the presence of matrix mixdown information

**matrix\_mixdown\_idx** Two bit field, specifying the index of the surround mixdown coefficient

**pseudo\_surround\_enable** One bit, indicating the possibility of mixdown for pseudo surround reproduction

**front\_element\_is\_cpe** indicates whether a SCE or a CPE is addressed as a front element

'0' selects an SCE

'1' selects an CPE

The instance of the SCE or CPE addressed is given by

**front\_element\_tag\_select**

the instance\_tag of the SCE/CPE addressed as a front element

**front\_element\_tag\_select**

**side\_element\_is\_cpe** see **front\_element\_is\_cpe**, but for side elements

**side\_element\_tag\_select** see **front\_element\_tag\_select**, but for side elements

**back\_element\_is\_cpe** see **front\_element\_is\_cpe**, but for back elements

**back\_element\_tag\_select** see **front\_element\_tag\_select**, but for back elements

**lfe\_element\_tag\_select** instance\_tag of the LFE addressed

**assoc\_data\_element\_tag\_select** instance\_tag of the DSE addressed

**valid\_cc\_element\_tag\_select** instance\_tag of the CCE addressed

**cc\_element\_is\_ind\_sw** One bit, indicating that the corresponding CCE is an independently switched coupling channel

**comment\_field\_bytes** The length, in bytes, of the following comment field

**comment\_field\_data** The data in the comment field

SCE or CPE elements within the PCE are addressed with two syntax elements. First, an is\_cpe syntax element selects whether a SCE or CPE is addressed. Second, a tag\_select syntax element selects the instance\_tag of a SCE/CPE. LFE, CCE and DSE elements are directly addressed with their instance\_tag.

4.5.1.2.1 Implicit and defined channel configurations

The AAC audio syntax provides two ways to convey the mapping of channels within a set of syntactic elements to physical locations of speakers. The first way is a default mapping based on the specific set of syntactic elements received and the order in which they are received. The most common mappings are further defined in subpart 1,

Table 1.11. If MPEG-4 Audio is used together with the MPEG-4 Systems audio compositor only these mappings shall be used. If a mapping shown in subpart 1, Table 1.11 is not used, the following methods describe the default determination of channel mapping:

1) Any number of SCEs may appear (as long as permitted by other constraints, for example profile, level). If this number of SCEs is odd, then the first SCE represents the front center channel, and the other SCEs represent L/R pairs of channels, proceeding from center front outwards and back to center rear.

If the number of SCEs is even, then the SCEs are assigned as pairs as center-front L/R, in pairs proceeding out and back from center front toward center back.

2) Any number of CPEs or *PAIRS* of SCEs may appear. Each CPE or pair of SCEs represents one L/R pair, proceeding from where the first sets of SCEs left off, pairwise until reaching either center back pair.

3) Any number of SCEs may appear. If this number is even, allocating pairs of SCEs Left/Right, from 2), back to center back. If this number is odd, allocated as L/R pairs, except for the final SCE, which is assigned to center back.

4) Any number of LFEs may appear. No speaker mapping is defined in case of multiple LFEs.

In case of this default (or implicit) mapping the number and order of SCEs, CPEs and LFEs and the resulting configuration may not change within the bitstream without sending a `program_config_element`, i.e. an implicit reconfiguration is not allowed.

Other audio syntactic elements that do not imply additional output speakers, such as `coupling_channel_element`, may follow the listed set of syntactic elements. Obviously non-audio syntactic elements may be received in addition and in any order relative to the listed syntactic elements.

If reliable mapping of channel set to speaker geometry is a concern, then it is recommended that an implicit mapping from subpart 1, Table 1.11, or a program configuration element be used.

For more complicated configurations a **Program Configuration Element (PCE)** is defined. The same restrictions apply with respect to the PCE as defined in ISO/IEC 13818-7. However, an MPEG-4 decoder is only required to parse PCEs in `raw_data_blocks()`, without interpreting them. Only the PCE provided within `GASpecificConfig()` describes the decoder configuration for the elementary stream under consideration. This implies that only one program can be configured at a certain time.

## 4.5.2 Decoding of the GA bitstream payloads

### 4.5.2.1 Top Level Payloads for the audio object types AAC main, AAC SSR, AAC LC and AAC LTP

#### 4.5.2.1.1 Definitions

`raw_data_block()` block of raw data that contains audio data for a time period of 1024 or 960 samples, related information and other data. There are 8 syntactic elements, identified as syntactic element `id_syn_ele`. The audio elements in one raw data block must have one and only one sampling rate. In the raw data block, several instances of the same `id_syn_ele` may occur, but each such instance of an `id_syn_ele` except for a `data_stream_element` must have a different 4-bit `element_instance_tag`. Therefore, in one raw data block, there can be from 0 to at most 16 of any `id_syn_ele`. The exceptions to this are the `data_stream_element`, the `fill_element` and the terminator element. If multiple `data_stream_element`s occur which have unique `element_instance_tags` then they are part of distinct data streams. If multiple `data_stream_element`s occur which have the same `element_instance_tag` then they are part of the same data stream. The `fill_element` has no `element_instance_tag` (since the content does not require subsequent reference) and can occur any number of times.



The terminator element has no `element_instance_tag` and must occur exactly once, as it marks the end of the `raw_data_block`.

`id_syn_ele` a bitstream element that identifies one of the following syntactic elements:

**Table 4.57 – Syntactic elements**

Syntactic Element	ID name	encoding	Abbreviatio
single_channel_element	ID_SCE	0x0	SCE
channel_pair_element	ID_CPE	0x1	CPE
coupling_channel_element	ID_CCE	0x2	CCE
lfe_channel_element	ID_LFE	0x3	LFE
data_stream_element	ID_DSE	0x4	DSE
program_config_element	ID_PCE	0x5	PCE
fill_element	ID_FIL	0x6	FIL
terminator	ID_END	0x7	TERM

- `single_channel_element()` abbreviation SCE. Syntactic element of the bitstream containing coded data for a single audio channel. A `single_channel_element()` basically consists of an `individual_channel_stream()`. There may be up to 16 such elements per raw data block, each one must have a unique `element_instance_tag`.
- `channel_pair_element()` abbreviation CPE. Syntactic element of the bitstream containing data for a pair of channels. A `channel_pair_element` consists of two `individual_channel_streams` and additional joint channel coding information. The two channels may share common side information. The `channel_pair_element` has the same restrictions as the single channel element as far as `element_instance_tag`, and number of occurrences.
- `coupling_channel_element()` Abbreviation CCE. Syntactic element that contains audio data for a coupling channel. A coupling channel represents the information for multi-channel intensity for one block, or alternately for dialogue for multilingual programming. The rules for number of `coupling_channel_elements` and instance tags are as for `single_channel_element`.
- `lfe_channel_element()` Abbreviation LFE. Syntactic element that contains a low sampling frequency enhancement channel. The rules for the number of `lfe_channel_elements` and instance tags are as for `single_channel_elements`.
- `program_config_element()` Abbreviation PCE. Syntactic element that contains program configuration data. The rules for the number of `program_config_elements` and element instance tags are the same as for `single_channel_elements`. PCEs must come before all other syntactic elements in a `raw_data_block()`.
- `fill_element()` Abbreviation FIL. Syntactic element that contains fill data. There may be any number of fill elements, that can come in any order in the raw data block.
- `data_stream_element()` Abbreviation DSE. Syntactic element that contains data. Again, there are 16 `element_instance_tags`. There is, however, no restriction on the number of `data_stream_elements` with any one instance tag, as a single data stream may continue across multiple `data_stream_elements` with the same instance tag.
- `terminator (ID_END)` The terminator `id_syn_ele ID_END` indicates the end of a raw data block. There must be one and only one terminator per raw data block.
- element\_instance\_tag** unique instance tag for syntactic elements other than terminator element and fill element. All syntactic elements containing instance tags may occur more than once, but, except for `data_stream_elements`, must have an unique `element_instance_tag` in each `raw_data_block`. This tag is also used to reference audio syntactic elements in a `coupling_channel_element`, and `single_channel_elements`, `channel_pair_elements`, `lfe_channel_elements`, `data_channel_elements`, and `coupling_channel_elements` inside a `program_config_element`, and provides the possibility of up to 16 independent `program_config_elements`.
- `audio_channel_element` generic term for `single_channel_element`, `channel_pair_element`, `coupling_channel_element` and `lfe_channel_element`.
- common\_window** a flag indicating whether the two `individual_channel_streams` share a common `ics_info` or not. In case of sharing, the `ics_info` is part of the `channel_pair_element` and must be used for both channels. Otherwise, the `ics_info` is part of each `individual_channel_stream`.

#### 4.5.2.1.2 Decoding process

Assuming that the start of a `raw_data_block` is known, it can be decoded without any additional „transport-level“ information except the sampling rate (needed for the selection of `sfb` and other tables) and produces 1024 or 960 audio samples per output channel.

Examples of the simplest possible bitstreams are

bitstream segment	output signal
<SCE><TERM><SCE><TERM>...	mono signal
<CPE><TERM><CPE><TERM>...	stereo signal
<SCE><CPE><CPE><LFE><TERM><SCE><CPE><CPE><LFE><TERM>...	5.1 channel signal

where angle brackets (< >) are used to delimit syntactic elements. For the mono signal each SCE must have the same value in its `element_instance_tag`, and similarly, for the stereo signal each CPE must have the same value in its `element_instance_tag`. For the 5.1 channel signal each SCE must have the same value in its `element_instance_tag`, each CPE associated with the front channel pair must have the same value in its `element_instance_tag`, and each CPE associated with the back channel pair must have the same value in its `element_instance_tag`.

If these bitstreams are to be transmitted over a constant rate channel then they might include a `fill_element` to adjust the instantaneous bit rate. In this case an example of a coded stereo signal is

<CPE><FIL><TERM><CPE><FIL><TERM>...

If the bitstreams are to carry ancillary data and run over a constant rate channel then an example of a coded stereo signal is

<CPE><DSE><FIL><TERM><CPE><DSE><FIL><TERM>...

All `data_stream_elements` have the same `element_instance_tag` if they are part of the same data stream.

A `single_channel_element` is composed of an `element_instance_tag` and an `individual_channel_stream`. In this case `ics_info` is always located in the `individual_channel_stream`.

A `channel_pair_element` begins with an `element_instance_tag` and `common_window` flag. If the `common_window` equals '1', then `ics_info` is shared amongst the two `individual_channel_stream` elements and the MS information is transmitted. If `common_window` equals '0', then there is an `ics_info` within each `individual_channel_stream` and there is no MS information.

#### 4.5.2.1.3 Matrix-mixdown Method

##### 4.5.2.1.3.1 Description

The matrix-mixdown method applies only for mixing a 3-front/2-back speaker configuration, 5-channel program, down to a stereo or a mono program. It is not applicable to any program with other than the 3/2 configuration and only decoders capable of decoding a 3/2 configuration must be able to decode this mode.

##### 4.5.2.1.3.2 Definitions

- matrix\_mixdown\_idx\_present** One bit indicating that a stereo matrix coefficient index is present (see Table 4.58). For all configurations other than the 3/2 format this bit must be zero.
- matrix\_mixdown\_idx** A two bit field that indicates that the coefficient to be used in the 5-channel to 2-channel matrix-mixdown. Possible matrix coefficients are listed in Table 4.58.
- pseudo\_surround\_enable** One bit indicating that pseudo surround decoding is possible.

**4.5.2.1.3.3 Matrix-mixdown process**

A derived stereo signal can be generated within a matrix-mixdown decoder by use of one of the two following sets of equations.

Set 1:

$$L' = \frac{1}{1 + 1/\sqrt{2} + A} \cdot [L + C/\sqrt{2} + A \cdot L_S]$$

$$R' = \frac{1}{1 + 1/\sqrt{2} + A} \cdot [R + C/\sqrt{2} + A \cdot R_S]$$

Set 2:

$$L' = \frac{1}{1 + 1/\sqrt{2} + 2 \cdot A} \cdot [L + C/\sqrt{2} - A \cdot (L_S + R_S)]$$

$$R' = \frac{1}{1 + 1/\sqrt{2} + 2 \cdot A} \cdot [R + C/\sqrt{2} + A \cdot (L_S + R_S)]$$

Where L, C, R, LS and RS are the source signals, L' and R' are the derived stereo signals and A is the matrix coefficient indicated by matrix\_mixdown\_idx. LFE channels are omitted from the mixdown.

If pseudo\_surround\_enable is not set, then only set 1 should be used. If pseudo\_surround\_enable is set, then either set 1 or set 2 equations can be used, depending on whether the receiver has facilities to invoke some form of surround synthesis.

As further information it should be noted that one can derive a mono signal using the following equation:

$$M = \frac{1}{3 + 2 \cdot A} \cdot [L + C + R + A \cdot (L_S + R_S)]$$

**4.5.2.1.3.4 Advisory**

The matrix-mixdown provision enables a mode of operation which may be beneficial in some circumstances. However, it is advised that this method should not be used. The psychoacoustic principles on which the audio coding is based are violated by this form of post-processing, and a perceptually faithful reconstruction of the signal cannot be guaranteed. The preferred method is to use the stereo or mono mixdown channels in the AAC syntax to provide stereo or mono programming which is specifically created by conventional studio mixing prior to bitrate reduction.

The stereo and mono mixdown channels additionally enable the content provider to separately optimize the stereo and multichannel program mixes - this is not possible by using the matrix-mixdown method.

It is additionally relevant to note that, due to the algorithms used for the multichannel and stereo mixdown coding, a better combination of quality and bitrate is usually provided by use of the stereo mixdown channels than can be provided by the matrix-mixdown process.

**4.5.2.1.3.5 Tables**

**Table 4.58 – Matrix-mixdown coefficients**

matrix_mixdown_idx	A
0	$1/\sqrt{2}$
1	$1/2$

2	$1/(2\sqrt{2})$
3	0

#### 4.5.2.1.4 Low Frequency Enhancement (LFE) channel element

##### 4.5.2.1.4.1 General

In order to maintain a regular structure in the decoder, the `lfe_channel_element` is defined as a standard `individual_channel_stream(0)` element, i.e. equal to a `single_channel_element`. Thus, decoding can be done using the standard procedure for decoding a `single_channel_element`.

In order to accommodate a more bitrate and hardware efficient implementation of the LFE decoder, however, several restrictions apply to the options used for the encoding of this element:

- The `window_shape` field is always set to 0, i.e. sine window
- The `window_sequence` field is always set to 0 (`ONLY_LONG_SEQUENCE`)
- The index of the highest non-zero spectral coefficient present in the element is 12
- No Temporal Noise Shaping is used, i.e. `tns_data_present` is set to 0
- No prediction is used, i.e. `predictor_data_present` is set to 0

##### 4.5.2.1.5 Data Stream Element (DSE)

Bitstream elements:

<b>data_byte_align_flag</b>	One bit indicating that a byte alignment is performed within the data stream element
<b>count</b>	Initial value for length of data stream
<b>esc_count</b>	Incremental value of length of data or padding element
<b>data_stream_byte</b>	A data stream byte extracted from bitstream

A data element contains any additional data, e.g. auxiliary information, that is not part of the audio information itself. Any number of data elements with the same `element_instance_tag` or up to 16 data elements with different `element_instance_tags` are possible. The decoding process of the data element is described in this subclause.

##### Decoding process:

The first syntactic element to be read is the 1 bit **data\_byte\_align\_flag**. Next is the 8 bit value **count**. It contains the initial byte-length of the data stream. If **count** equals 255, its value is incremented by a second 8 bit value, **esc\_count**, this final value represents the number of bytes in the data stream element. If **data\_byte\_align\_flag** is set, a byte alignment is performed. The bytes of the data stream follow.

##### 4.5.2.1.6 Fill element (FIL)

Bitstream elements:

<b>count</b>	Initial value for length of fill data
<b>esc_count</b>	Incremental value of length of fill data
<b>extension_type</b>	Four bit field indicating the type of fill element content
<b>fill_nibble</b>	Four bit field for fill
<b>fill_byte</b>	Byte to be discarded by the decoder
<b>other_bits</b>	Bits to be discarded by the decoder

Fill elements have to be added to the bitstream if the total bits for all audio data together with all additional data is lower than the minimum allowed number of bits in this frame necessary to reach the target bitrate. Dynamic Range

Control (DRC) bits must be added to the fill element whenever the encoder wishes to include DRC information. Under normal conditions fill bits are avoided and free bits are used to fill up the bit reservoir. Only if the bit reservoir is full, fill bits are written. Any number of fill elements are allowed.

Fill elements containing an extension\_payload with a extension\_type of EXT\_DYNAMIC\_RANGE (see below) are reserved for dynamic range information. In this case, the fill\_element count field must be set equal to the total length, in bytes, of all dynamic range information plus the extension\_type field.

The following symbolic abbreviations for values of the extension\_type field are defined currently:

**Table 4.59 – Values of the extension\_type field**

Symbol	Value of extension_type	Purpose
EXT_FILL	'0000'	Bitstream filler
EXT_FILL_DATA	'0001'	Bitstream data as filler
EXT_DYNAMIC_RANGE	'1011'	Dynamic range control
-	all other values	reserved

The 'reserved' values can be used for further extension of the syntax in a compatible way.

Note that fill\_nibble is normatively defined to be '0000' and fill\_byte is normatively defined to be '10100101' (to ensure that self-clocked data streams, such as radio modems, can perform reliable clock recovery).

**Decoding process:**

The syntactic element **count** gives the initial value of the length of the fill data. In the same way as for the data element this value is incremented with the value of **esc\_count** if **count** equals 15. The resulting number gives the number of bytes to be read.

**4.5.2.2 Payloads for the audio object type AAC scalable**

**4.5.2.2.1 Definitions**

- aac\_scalable\_main\_element() abbreviation ASME; syntactic element of the bitstream containing coded data for the first AAC coding layer in a scalable configuration. This type of syntax can be used also for single coding layer (non-scalable) applications. In this case exactly one ASME contains all the coded information. An ASME consists of an aac\_scalable\_main\_header(), and for each coded audio output channel, one individual\_channel\_stream(). The maximum number of such elements is limited to 1 for each audio object.
- aac\_scalable\_main\_header() contains all side information required for the first AAC coding layer, with the exception of the side information which is transmitted within the individual channel streams. The ics\_info() sub-block of AAC is not used here. Instead the information, normally carried in ics\_info(), is contained directly in this element.
- aac\_scalable\_extension\_element() abbreviation ASEE; syntactic element of the bitstream containing coded data for all other than the first AAC coding layer in a scalable configuration. An ASEE consists of an aac\_scalable\_extension\_header(), and for each coded audio output channel, one individual\_channel\_stream(). The maximum number of such elements is limited to 7 for one audio object.
- aac\_scalable\_extension\_header() contains all side information required for an AAC extension layer, with the exception of the side information that is transmitted within the individual channel streams. Extension in this context means, that this is not the first AAC coding layer.
- bits\_to\_decode() a helper function; returns the number of bits not yet decoded in the current ASME, or ASEE, respectively, if the length of these elements has been

	signaled by a system/transport layer. If the length of these elements is unknown, <code>bits_to_decode()</code> returns 0.
<b>diff_control[w][dc_group]</b>	for each window this is the FSS control information for one <code>dc_group</code> . If the window type is not <code>SHORT_WINDOW</code> . <b>diff_control[w][dc_group]</b> is huffman encoded using Table 4.111 in the bitstream.
<b>diff_control_lr[w][sfb]</b>	element used in a mono GA / stereo GA configuration, to control the interaction of the M-channel with the L and R channel.

**Help elements:**

<i>this_layer_stereo</i>	set to '1', if the current layer is a stereo layer; set to '0' otherwise.
<i>mono_layer_flag</i>	set to '1', if there is any mono layer; set to '0' otherwise.
<i>mono_stereo_flag</i>	set to '1', if there is at least one mono layer and this is the first stereo layer; set to '0' otherwise.
<i>last_max_sfb</i>	<code>max_sfb</code> of the previous coding layer. Set to '4* no_of_dc_groups', if the previous layer is running at a different sampling rate, or is a non GA coder.
<i>last_max_sfb_ms</i>	<code>max_sfb</code> of the previous stereo layer; set to '0', if the previous layer is a mono layer
<i>last_mono_max_sfb</i>	<code>max_sfb</code> of the highest mono layer.
<i>core_flag</i>	set to '1', if a core coder is present. Within the MPEG-4 Audio standard the only valid core coder is MPEG-4 Celp, although the mechanisms could be used e.g. for other waveform coders or an AAC coder operating at a lower sampling rate than the AAC extension layer; set to '0' if there is no core coder.
<i>tvq_layer_present</i>	set to '1', if a TwinVQ coder is present in the previous layer; set to '0' otherwise.
<i>tvq_tns_present</i>	set to '1', if <code>tns_present</code> is set to 1 in the previous TwinVQ layer; set to '0' otherwise.
<i>tvq_stereo</i>	set to '1', if the previous TwinVQ layer is stereo; set to '0' otherwise.
<i>tvq_mono_tns</i>	set to '1', if there is a tvq mono layer present that uses <code>tns</code> ; set to 0 if there is a tvq mono layer present that does not use <code>tns</code> .
<i>tvq_stereo_tns_left</i>	set to '1', if there is a tvq stereo layer present that uses <code>tns</code> in the left channel; set to 0 if there is a tvq stereo layer present that does not use <code>tns</code> in the left channel.
<i>tvq_stereo_tns_right</i>	set to '1', if there is a tvq stereo layer present that uses <code>tns</code> in the right channel; set to 0 if there is a tvq stereo layer present that does not use <code>tns</code> in the right channel.
<i>tns_aac_tvq_en[ch]</i>	set to '1' if a <code>tns_data_present</code> bit is transmitted in an ASME, if a TwinVQ layer is present (see Table 4.63).

**4.5.2.2.2 Decoding process**

Assuming that the start of an **ASME** or **A SEE** is known, it can be decoded with the additional information given by the `GASpecificConfig()`. 1024 or 960 audio samples per output channel are produced for each element.

The **ASME** or **A SEE** support encoding for both, constant rate, and variable rate channels. In each case the structure of the bitstream and the operation of the decoder are identical.

In the decoder, the output of the ASME and the output of all the available ASEE, where the previous coding layer output is available, has to be combined to form the output signal, giving the maximum audio quality. If there is an intermediate layer missing, e.g. due to transmission errors, the information in the subsequent layers can not be used. Furthermore, if there is a CELP core coder, or one or more TwinVQ layer signaled for a scalable audio object, the output of these coders has to be combined with the AAC coding layers.

A list of the valid combinations of different coding schemes is given in the subsequent subclause. The methods how to combine the individual coding layers are described in subclauses 4.5.2.2.4, 4.5.2.2.5, and 4.5.2.2.6.

**4.5.2.2.3 Valid combinations of AAC with either TwinVQ or CELP**

The AAC scalable coder in combination with a TwinVq or CELP base layer coder provides one way of achieving bit rate scalability. It is based on the calculation of a difference signal between the output signal of a base layer coder and the original input signal. At least one AAC extension layer is used. Joint stereo coding and mixed mono/stereo configurations are possible. All AAC joint stereo modes are available in the combined coder.

Three major classes of scalable configurations with AAC exist, depending on the coder types used:

1. AAC layers only
2. Narrow-band CELP base layer plus AAC
3. TwinVQ base layer plus AAC

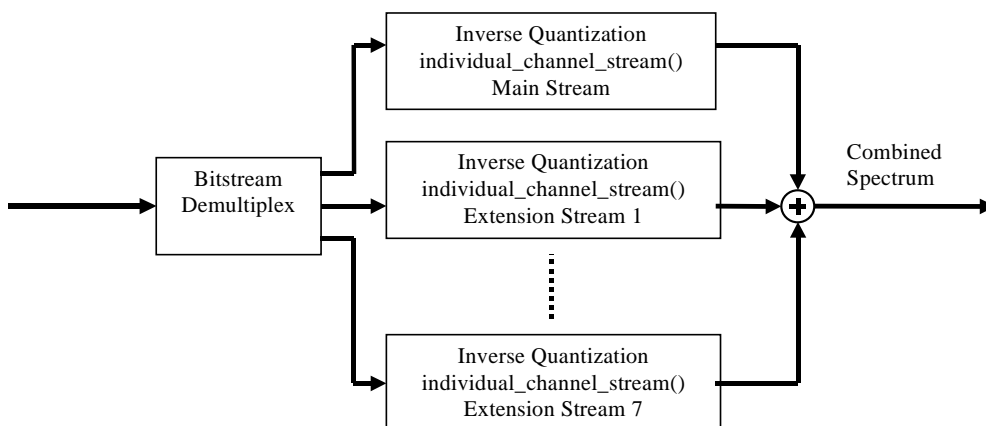
AAC or TwinVQ coding layer can be either coded in mono, or stereo/joint stereo. The following table summarizes the possible transitions between two layers with respect to the coder combinations and the channel configurations for each of the three coder types in mono and stereo.

**Table 4.60 – Valid scalable combinations**

Layer N	Layer N+1				
	NB CELP mono	TwinVQ mono	TwinVQ stereo	AAC mono	AAC stereo
Narrow Band CELP mono	X			X	X
TwinVQ mono		X		X	X
TwinVQ stereo			X		X
AAC mono				X	X
AAC stereo					X

**4.5.2.2.4 Decoding of AAC only combinations**

AAC only combinations basically rely on the difference encoding of the spectrum in the **Scalable Inverse AAC Quantization Module (SIAQ)** as shown in Figure 4.3 below. In the SIAQ module, after the inverse quantization, the reconstructed spectra of all individual\_channel\_streams are added. The number of layers is restricted to one AAC main layer and up to 7 AAC extension layers. The bitrate of the main layer and the additional layers can be any bit rate possible for AAC. However, the bitstream buffer restrictions given in subclause 4.5.2.7 apply. SIAQ is also a main building block for the CELP and TwinVQ combinations with AAC. (Exceptions from this are given in subclause 4.5.2.2.7 “Combining AAC layers, if PNS, MS or Intensity tools are used in a particular scale factor band”)



**Figure 4.3 – Scalable Inverse AAC Quantization Module (SIAQ)**

Three audio channel configurations of a scalable coder, based on AAC only, are possible:

AAC-Only-M:	AAC mono main layer 0 to 7 AAC mono extension layers	plus
AAC-Only-S:	AAC stereo main layer 0 to 7 AAC stereo extension layers	plus
AAC-Only-M/S:	AAC mono main layer 0 to 7 AAC mono extension layers 1 to 7 AAC stereo extension layers	plus plus ( total number of layer <= 8 )

Figure 4.4 shows the block diagram of the decoder for mode AAC-Only-M/S, which is a superset of both of the other modes. In this mode three SIAQ modules are present. The first one for the Left (L), or Mid (M), or Intensity (I) audio channel, respectively, the second for the Right (R), or Side (S), audio channel, and the third one for the mono layer (M) which, in general, is a down-mix of Left and Right which was generated in the encoder according to  $M = 0.5 * (L+R)$ . For all scale factor bands where M/S coding is selected, the M-Signal is calculated by adding  $M''$  and  $M'$  (the restrictions given in subclause 4.5.2.2.7 have to be followed which prohibit the addition under certain circumstances). For all bands where L/R coding is selected L and R are generated from  $L'$ ,  $R'$ , and  $2.0 * M''$ , using **diff\_control\_lr** to control the corresponding FSS unit. M, S, L and R are then fed into the AAC inverse joint-stereo processing module which produces the L and R spectra. After processing in the serial TNS filter combination, as described in subclauses 4.6.9 to 4.6.9.5, the time signals can be calculated via the IMDCT filterbank.

To decode mode AAC-Only-S both FSS modules and the Inverse TNS-M filters are by-passed. The  $M''$ -channel SIAQ is not present in this mode. For the decoding of mode AAC-Only-M the inverse joint stereo processing module and the Right/Side SIAQ module, as well as the corresponding TNS/IMDCT combination are not required. Only one inverse TNS filter is applied to the spectrum before the IMDCT.

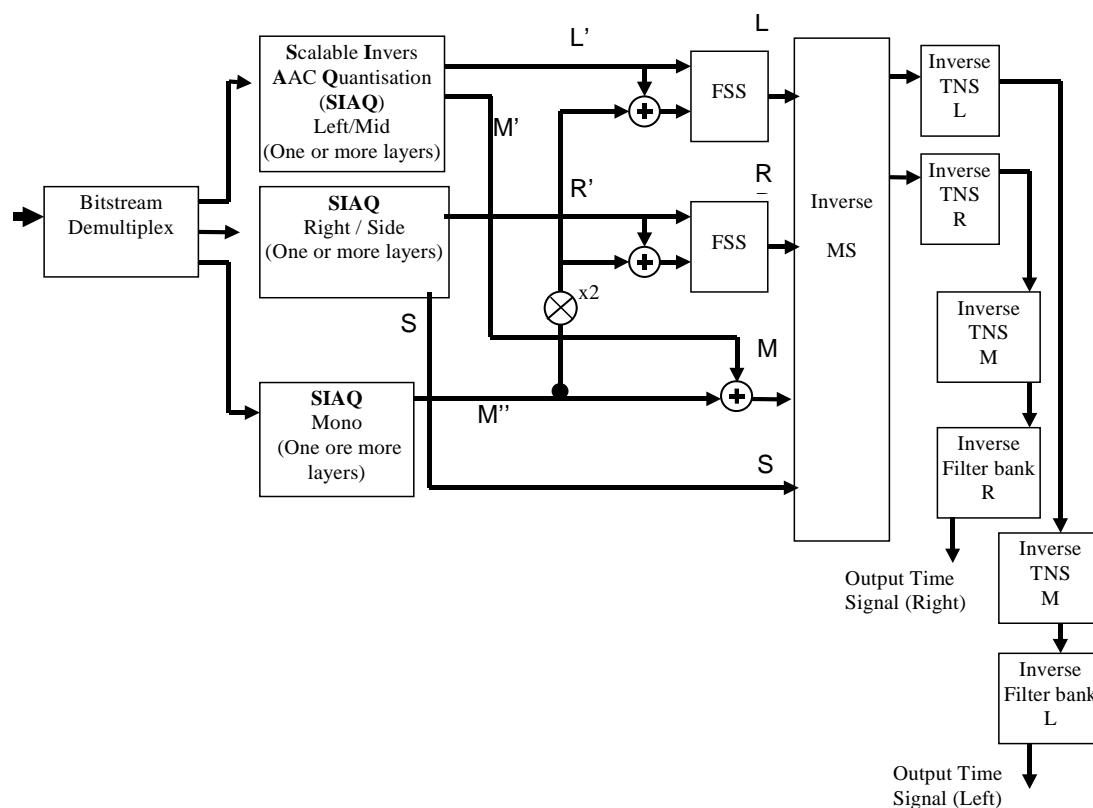


Figure 4.4 – AAC mono / AAC stereo configuration (AAC-Only-M/S)



4.5.2.2.5 Decoding of CELP / AAC combinations

This combination is primarily used to add a very low bit rate base layer in a scalable system. While in the TwinVQ/AAC combinations (see subclause 4.5.2.2.6) all coding layers work in the frequency domain, the CELP coding layer (called core coder subsequently) is a time domain coder which, in general, works with a different sampling rate. Using the up-sampling tool, which is described in subclause 4.6.15, and a FSS module (see subclause 4.6.14), the combination makes the coding gain of the CELP core coder available for the subsequent extension layers. The ratio of the sampling rate of the core coder and the sampling rate of the AAC coder always is an integer, to make the use of the upsampling tool defined in subclause 4.6.15 possible.

4.5.2.2.5.1 CELP post-filter

The post-filter of the CELP core coder has to be switched off for the output signal which is then combined with the AAC enhancement stages in the scalable coder. If only the CELP coder output is desired, the post-filter can be used to generate this output.

4.5.2.2.5.2 Frame length adaptation / super-frames

Since the AAC frame length may not necessarily be a multiple of the core frame length, the least common multiple of these two lengths can be very large. To avoid such situations an alternative AAC frame length of 960 samples, instead of 1024, is available. These frame length allows an easy integration of the MPEG-4 CELP core and AAC by constructing super-frames of a reasonable size. Some combinations of CELP-coder and GA-coder sampling rates require different numbers of core coder frames and AAC frames to build common super-frames. Table 4.61 gives an overview of the length of an AAC frame with 960 samples at various sampling rates and the length of the smallest possible super-frames for each sampling rate for all of the MPEG-4 CELP core frame length options (40, 30, 20, and 10 ms; see subpart 3, Table 3.8). Also listed in the table is the number of AAC and CELP coder frames in these smallest possible super-frames. These superframes do not exist in MPEG-4 Systems, as only single frames of each coding layer are addressed by MPEG-4 systems, making different solutions possible. However, the MPEG-4 Audio decoder is only required to support these minimum frame length superframes. Note that it is not necessary to buffer the whole super-frame. The decoder can start to produce the combined output samples whenever a sufficient number of up-sampled CELP core output samples and a full AAC layer frame is available.

In general, the CELP core signal has to be delayed after the zero-insertion in the up-sampling tool (see subclause 4.6.15) before MDCT filterbank. The value „coreCoderDelay“ given in the GASpecificConfig() of the first AAC coding layer is the number of samples, which the core signal has to be delayed. The encoder has the choice between optimizing the total system delay for the Celp core stream (no additional delay added to the CELP coder in the encoder), or to adjust the signal of the core and AAC layer in a way so that there is no delay in the scalable decoder necessary, or any value in between those two extremes. In any case the Composition Time Stamp (if used, see ISO/IEC 14496-1) of the first Core and its corresponding first AAC frame in a superframe (see above) has to be identical. However, if only the output of the Celp core presented to the compositor and a minimum delay is desired (eg. for 2-way communication) the receiving terminal can subtract the time *td* that corresponds to **CoreCoderDelay** from the Composition Time. This is only true if there is no associated video stream for which synchronous composition has to be assured. *td* is calculated according to:

$$td = \text{CoreCoderDelay} / fs$$

where *fs* is the sampling rate of the AAC layer(s).

Table 4.61 – AAC frame length for 960 samples per frame and super-frame length of AAC/CELP combinations

Sampling rate (kHz):	96	64	48	32	24	16	8
AAC Frame length (ms)	10	15	20	30	40	60	120
Super-Frame length (40 ms core frame) (ms)	40	120	40	120	40	120	120
AAC / CELP frames per super-frame (40 ms)	4 / 1	8 / 3	2 / 1	4 / 3	1 / 1	2 / 3	1 / 3
Super-Frame length (30 ms core frame) (ms)	30	30	60	30	120	60	120
AAC / CELP frames per super-frame (30 ms)	3 / 1	2 / 1	3 / 2	1 / 1	3 / 4	1 / 2	1 / 4
Super-Frame length (20 ms core frame) (ms)	20	60	20	60	40	60	120

AAC / CELP frames per super-frame (20 ms)	2 / 1	4 / 3	1 / 1	2 / 3	1 / 2	1 / 3	1 / 6
Super-Frame length (10 ms core frame) (ms)	10	30	20	30	40	60	120
AAC / CELP frames per super-frame (10 ms)	1 / 1	2 / 3	1 / 2	1 / 3	1 / 4	1 / 6	1 / 12

#### 4.5.2.2.5.3 CELP core coder with AAC running at 88.2 kHz, 44.1 kHz, or 22.05 kHz sampling rate

AAC frames using sampling rates of 88.2 kHz, 44.1 kHz, or 22.05 kHz can be achieved by adjusting the sampling rate of the CELP core coder such, that an integer ratio between these two sampling rate is achieved. Table 4.112 shows the mapping of the AAC sampling rates to the CELP core coder sampling rates. The CELP core coder runs with the sampling rate listed in this table. The CELP decoding process is completely identical to the methods defined for a sampling rate of 8 kHz for the narrow band CELP coder. Table 4.62 shows the super frame parameters for the AAC sampling rates 88.2 kHz, 44.1 kHz, and 22.05 kHz.

**Table 4.62 – Super-frame parameters of AAC/CELP combinations at AAC sampling rates of 88.2 kHz, 44.1 kHz and 22.05 kHz**

Sampling rate AAC (kHz)	88.2	44.1	22.05
Sampling rate CELP (Hz)	7350	7350	7350
AAC Frame length (ms)	10.884	21.768	43.537
Super-frame length (43.537 ms core frame) (ms)	43.537	43.537	43.537
AAC / CELP frames per super-frame (43.537 ms)	4/1	2 / 1	1 / 1
Super-frame length (32.653 ms core frame) (ms)	32.653	65.306	130.612
AAC / CELP frames per super-frame (30 ms)	3/1	3 / 2	3 / 4
Super-frame length (21.768 ms core frame) (ms)	21.768	21.768	43.537
AAC / CELP frames per super-frame (20 ms)	2/1	1 / 1	1 / 2
Super-frame length (10.884ms core frame) (ms)	10.884	21.768	43.537
AAC / CELP frames per super-frame (10 ms)	1/1	1 / 2	1 / 4

#### 4.5.2.2.5.4 CELP / AAC Configurations

Several CELP/AAC combinations, which differ in the number of audio channels in each layer, are defined:

CELP-AAC-M	CELP mono layer AAC mono main layer 0 to 7 AAC mono extension layer	plus plus
CELP-AAC-MS	CELP mono layer AAC mono main layer 0 to 7 AAC extension layer 0 to 7 AAC stereo extension layer	plus plus plus (total number of AAC layer <= 8)
CELP-AAC-S	CELP mono layer AAC stereo main layer 0 to 7 AAC stereo extension layer	plus plus

Note that the CELP coder also has an intrinsic scalability function (see subpart 3). This means that the CELP coding layer itself might consist of several narrow-band CELP coding layers.

Figure 4.5 below shows the decoder block diagram of the CELP-AAC-M configuration.:

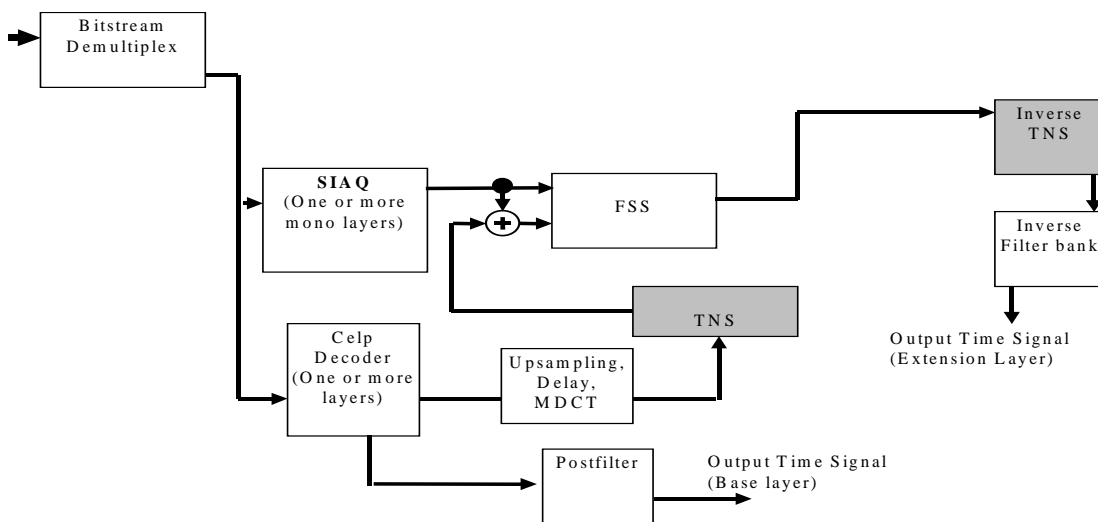


Figure 4.5 – Block diagram of the CelP + AAC mono combination

The CELP core is decoded and the output is up-sampled, delayed and transformed into the frequency domain as described in subclause 4.6.15. This signal is then combined with the reconstructed AAC layer spectrum in the Frequency Selective Switch (FSS, see subclause 4.6.14) tool to get the complete output spectrum. If TNS is used the encoder TNS filter is applied to the spectrum of the core coder. The normal decoder TNS-filter is then applied before the calculation of the inverse IMDCT Filterbank as usual.

Figure 4.6 shows mode CELP-AAC-S, where a mono CELP core is combined with a stereo AAC layer. The TNS filter is applied to the MDCT coefficients calculated from the upsampled CELP decoder output. The inverse TNS filter is needed only for the enhancement layer output.

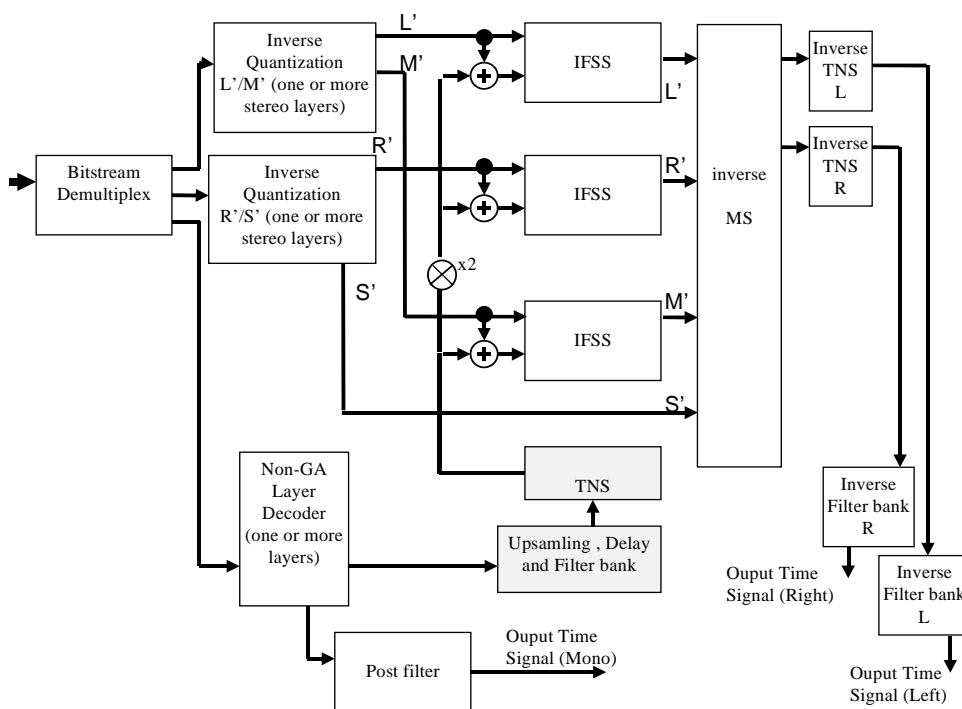


Figure 4.6 – Decoding of a non-GA (i.e. CELP) mono plus AAC stereo combination

Figure 4.7 shows mode CELP-AAC-MS, where a non-GA (i.e. CELP) mono layer, and at least one AAC mono layer, plus at least one AAC stereo layer is combined. The TNS filter for the Mid signal is applied to the MDCT coefficients reconstructed from the CELP core. Both inverse TNS filter, the inverse filter for the M signal, and the inverse TNS filter for the L/R signal are applied to the final layer output as described in subclause 4.6.9 to 4.6.9.5.

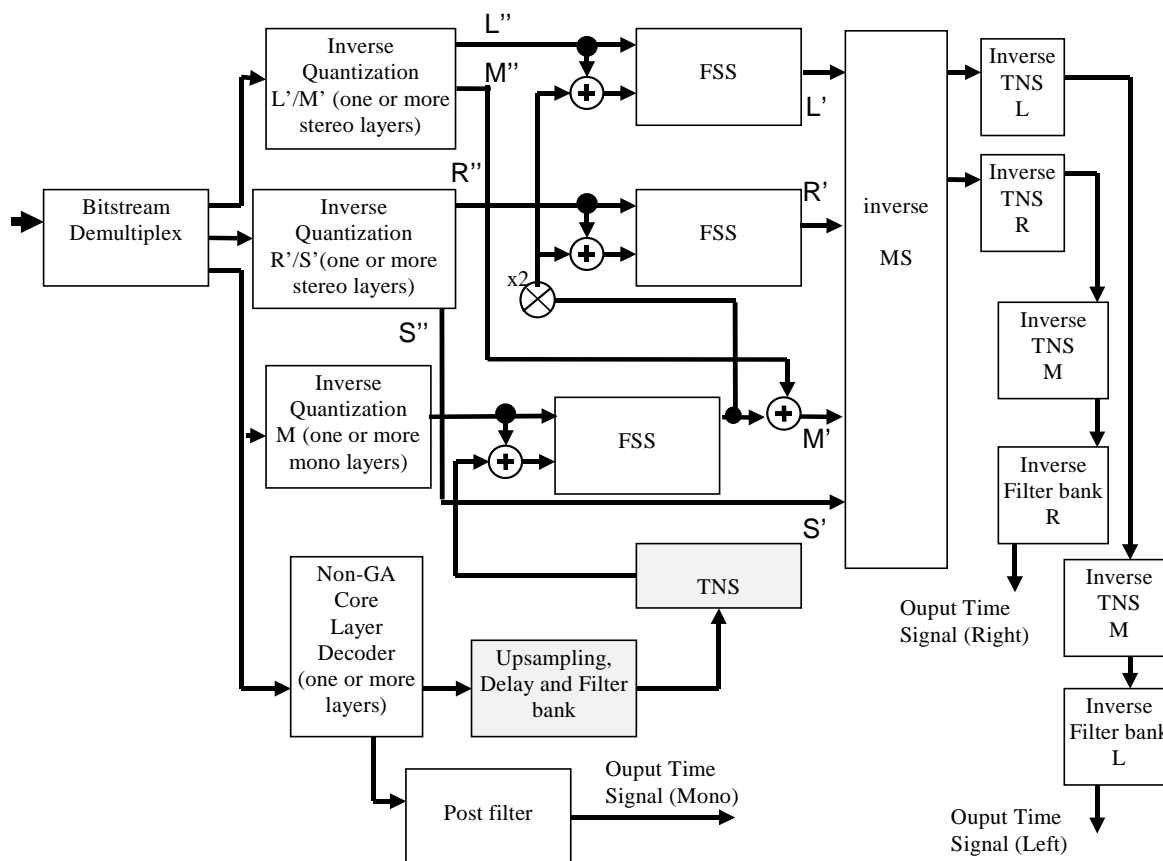


Figure 4.7 – Decoding of a Celp(mono) + AAC(mono) + AAC(stereo) configuration.

#### 4.5.2.2.5.5 Alternative core coder

Within MPEG-4, only the MPEG-4 narrow-band CELP coder is available as a core coder. However, there is no principal limitation as to which core coder can be used, although, in general, the core coder should encode the waveform of the input signal, to allow a useful difference signal to be calculated. Especially CELP coder with a frame length of a multiple of 10 ms can easily be integrated.

#### 4.5.2.2.6 Decoding of TwinVQ / AAC combinations

It is possible to construct a scalable coder by combining TwinVQ and AAC. This configuration provides the advantages of both coders: TwinVQ can reduce the averaged distortion at the low bit rate range and AAC is good at controlling the quantization noise for high quality coding. The scalable bitstream is first demultiplexed and the spectral information of the TwinVQ and the AAC layer are extracted, as well as the side information. After the AAC inverse quantization, the addition of the two spectra is performed considering the FSS control information. Finally, time domain signal is generated through IMDCT.

One advantage of the combined TwinVQ - AAC audio coding is that both quantization schemes operate on the same signal space, i.e. MDCT coefficients. Also, both coders share the same sampling rate. Therefore no

"interfacing" is necessary to convert between the internal representations of the audio data of the two coders, like for the CELP coder, where an additional up-sampling filterbank has to be calculated.

**4.5.2.2.6.1 Combination with the AAC tools**

The AAC oriented tools such as TNS, LTP and joint stereo coding can be combined with TwinVQ and the scalable coder. LTP is applied only to the base layer and other tools are commonly used for both TwinVQ layer and AAC layer. There are number of possibilities how to combine these tools. The combinations of TwinVQ with TNS are shown in Figure 4.8 to Figure 4.15. The transmission of `tns_data_present` and `tns_data()` in the first AAC layer depends on the actually used configuration. The following table shows the conditions when the TNS related bitstream elements are transmitted, and when they are not.

**Table 4.63 – Values of helper element `tns_aac_tvq_en[]` depending on `tvq_mono_tns`, `tvq_stereo_tns_left` and `tvq_stereo_tns_right`.**

	<code>tns_aac_tvq_en[0]</code>	<code>tns_aac_tvq_en[1]</code>
<code>(tvq_mono_tns==1) &amp;&amp; (this_layer_stereo == 1)</code>	1	1
<code>(tvq_mono_tns==1) &amp;&amp; (this_layer_stereo == 0)</code>	0	X
<code>(tvq_mono_tns==0)</code>	1	1
<code>(tvq_stereo_tns_left==1) &amp;&amp; (tvq_stereo_tns_right==1)</code>	0	0
<code>(tvq_stereo_tns_left==1) &amp;&amp; (tvq_stereo_tns_right==0)</code>	0	1
<code>(tvq_stereo_tns_left==0) &amp;&amp; (tvq_stereo_tns_right==1)</code>	1	0
<code>(tvq_stereo_tns_left==0) &amp;&amp; (tvq_stereo_tns_right==0)</code>	1	1

**4.5.2.2.6.2 max\_sfb and scale\_factor\_grouping for TwinVQ**

The scalefactor band tables and the parameter `max_sfb` are necessary for most of the AAC related tools, such as LTP, TNS, FSS (`diff_control`) and joint stereo coding. If TwinVQ is combined with these tools, the identical scalefactor bands defined in Table 4.73 – Table 4.91 of subclause 4.5.4 are used. On the other hand, TwinVQ normally does not use the parameter `max_sfb`. However, it can be calculated from the frequency range that is used in the interleaved vector quantization as shown in the following pseudo-code:

```
tvq_main_element :
    max_line = (int) (frame_length/num_windows * qsample);
    for(iw=0, i_sfb=0; ((iw<num_swb) && (swb_offset[iw] < max_line)); iw++){
        i_sfb = iw+1;
    }
    max_sfb = i_sfb;
```

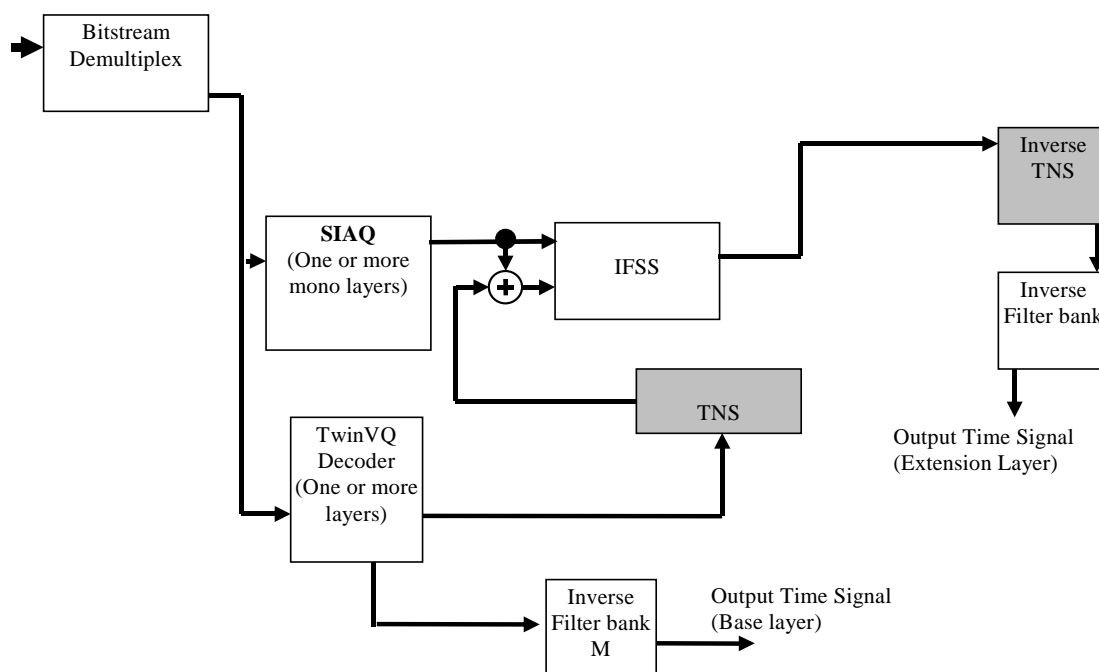
```
tvq_extension_element:
    max_line = (int) (frame_length /num_windows* (qsample+bias*3));
    for(iw=0, i_sfb=0; ((iw<num_swb) && (swb_offset[iw] < max_line)); iw++){
        i_sfb = iw+1;
    }
    max_sfb[lvr] = i_sfb;
```

The parameter, `max_line` is a highest frequency line that the interleave VQ quantizes. `Frame_length` is either 1024 or 960. In case of `EIGHT_SHORT_SEQUENCES`, `num_windows` is 8, otherwise it is 1. The values of `qsample` and `bias` are defined in the section of adaptive active band selection (subclause 4.6.5.4), and depend only on the sampling frequency and the bitrate. The tables of `swb_offset` are listed in subclause 4.5.4. Note that the parameter of `scale_factor_grouping` is transmitted in the TwinVQ element if it is needed for stereo joint coding with TwinVQ.

**4.5.2.2.6.3 Audio channel configurations**

TVQ-AAC-MM	TwinVQ mono layer AAC mono main layer 0 to 7 AAC mono extension layer	plus plus
TVQ-AAC-MMS	TwinVQ mono layer AAC mono main layer 0 to 7 AAC mono extension layer 0 to 7 AAC stereo extension layer	plus plus plus (total number of AAC layer <= 8)
TVQ-AAC-MS	TwinVQ mono layer AAC stereo main layer 0 to 7 AAC stereo extension layer	plus plus
TVQ-AAC-SS	TwinVQ stereo layer AAC stereo main layer 0 to 7 AAC stereo extension layer	plus plus

Figure 4.8 shows the case where TwinVQ is mono, and TwinVQ does not use TNS, where the TNS filter is applied to the reconstructed MDCT coefficients of the TwinVQ decoder. The inverse TNS filters for the L- and R-channel are needed only for the extension layer output.



**Figure 4.8 – Decoding of a TwinVQ (mono) + AAC (mono) configuration, TwinVQ does not use TNS**

Figure 4.9 shows the case where TwinVQ is mono, and uses TNS, where the inverse TNS filter is applied for both, for the TwinVQ only output, and the combined TwinVQ-AAC output signal.

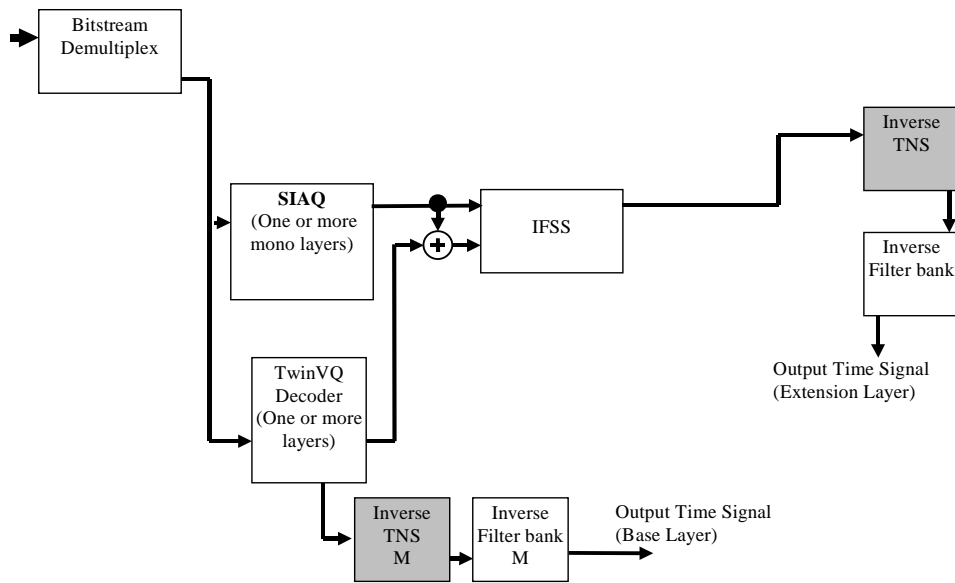


Figure 4.9 – Decoding of a TwinVQ (mono) + AAC (mono) configuration, TwinVQ does use TNS

Figure 4.10 shows the case where TwinVQ is mono, and uses TNS, and at least one AAC mono and one AAC stereo layer is present, where the both the middle, and the left/right inverse TNS filter is applied to the left and right spectrum before calculating the IMDCT.

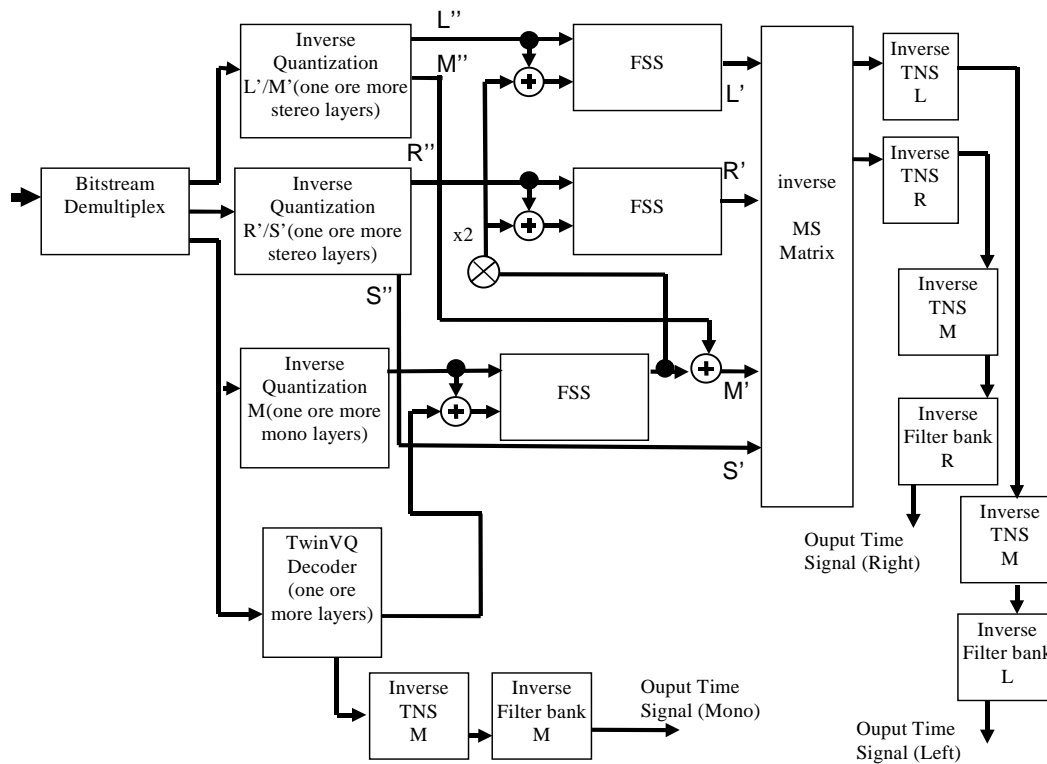


Figure 4.10 – Decoding of a TwinVQ (mono) AAC (mono) and AAC (stereo) configuration, TwinVQ does use TNS

Figure 4.11 shows the case where TwinVQ is mono and does **not** use TNS, and at least one AAC mono and one AAC stereo layer is present, where both, the middle, and the left/right channel inverse TNS filter is applied to the left and right spectrum before calculating the IMDCT.

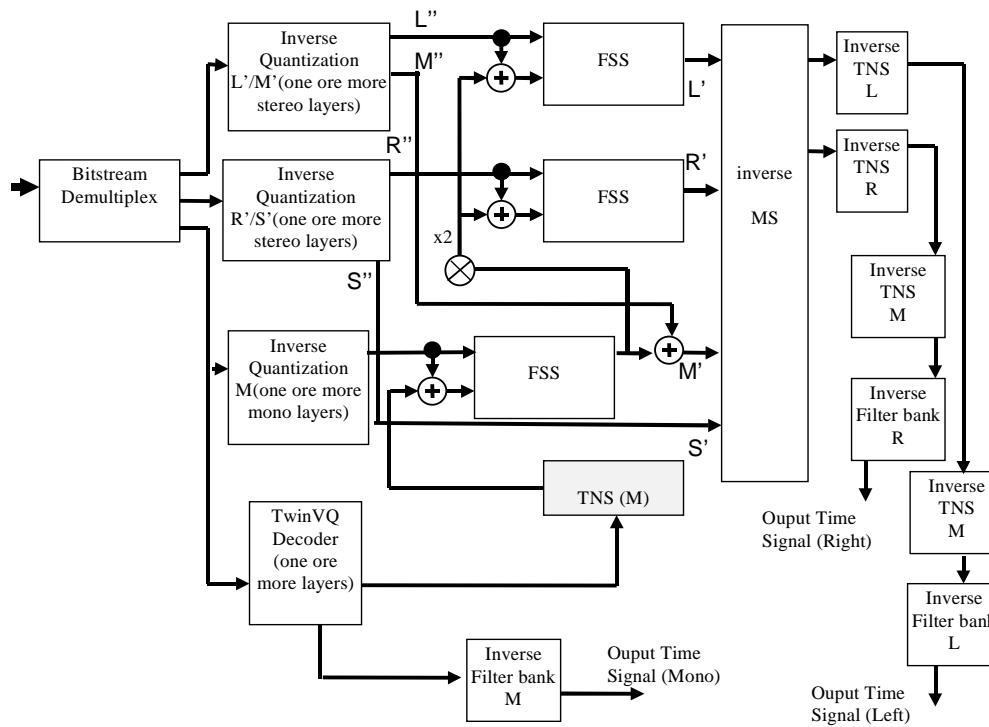


Figure 4.11 – Decoding of a TwinVQ (mono) + AAC (mono) + AAC (stereo) configuration, if TNS is NOT used in the TwinVQ layer.

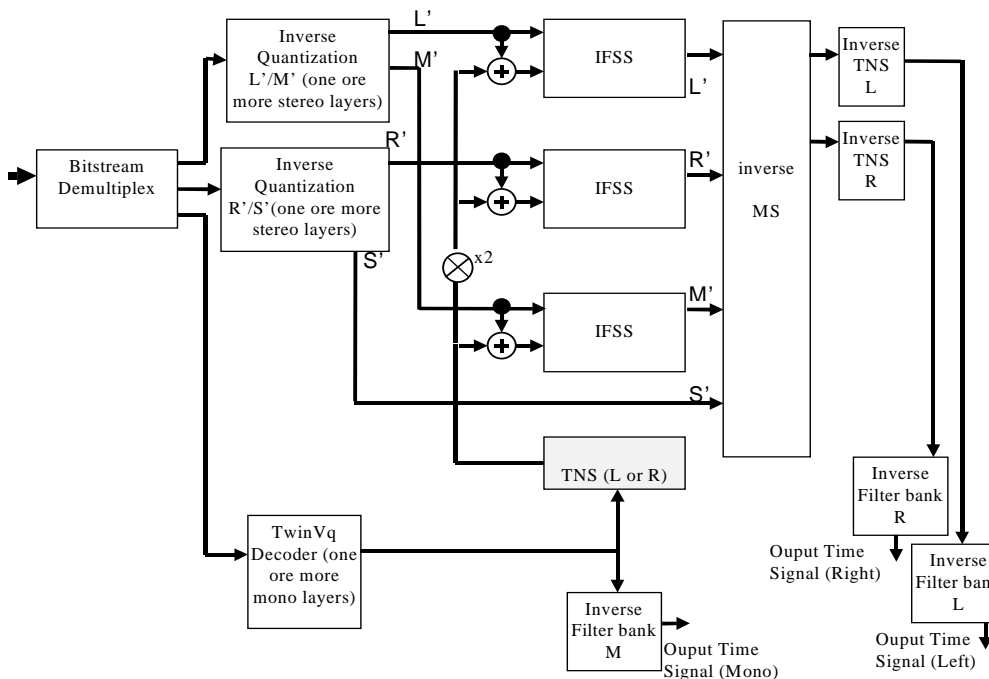


Figure 4.12 – Decoding of a TwinVQ (mono) + AAC (stereo) configuration, if TNS is NOT used in the TwinVQ layer.



Figure 4.13 shows the case that TwinVQ is stereo and TwinVQ does not use TNS. Here a TNS filter is applied to the reconstructed MDCT coefficients of the TwinVQ decoder. The inverse TNS filter is needed only for the enhancement layer output.

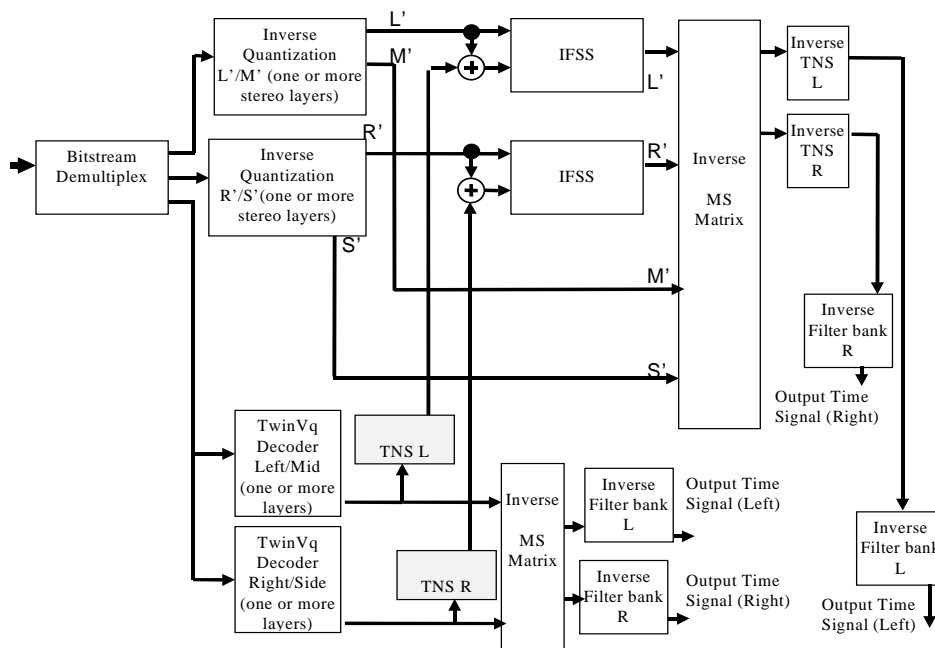


Figure 4.13 – Decoding of a TwinVQ (stereo) +AAC (stereo) configuration, if TNS is NOT used in the TwinVQ layer.

Figure 4.14 shows the case where TwinVQ is stereo, and TwinVQ uses TNS, where the common inverse TNS filter is applied to both, the base layer, and the enhancement layer output.

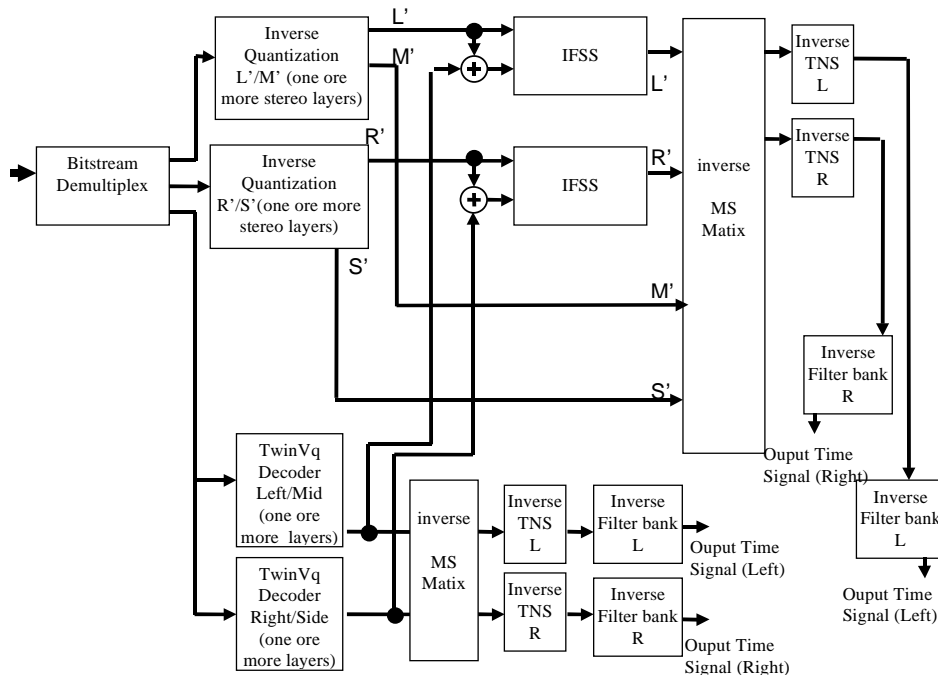


Figure 4.14 – Decoding of a TwinVQ (stereo) + AAC (stereo) configuration, if TNS is used in a TwinVQ layer.

Figure 4.15 shows the case where TwinVQ is mono, and TwinVQ uses TNS, where the common inverse TNS filter is applied to both, the base layer, and the enhancement layer output.

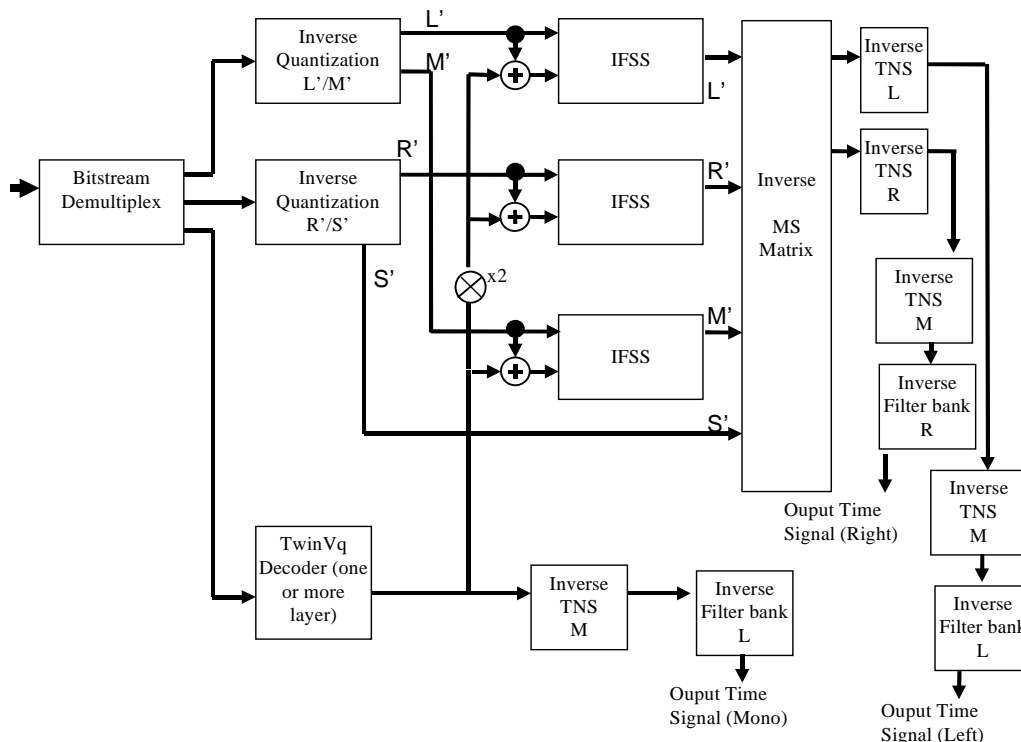


Figure 4.15 – Decoding of a TwinVQ (mono) + AAC (stereo) configuration, if TNS is used in the first TwinVQ layer.

4.5.2.2.7 Combining AAC layers, if PNS, MS or Intensity tools are used in a particular scale factor band

The following tables specify the output spectrum of a particular scale factor band of the combined layers N and N+1 for various combinations of the PNS, Intensity, and MS coding tools in layer N and layer N+1 for different layer combinations:

Mono-mono layer combination:

Tool used in Layer N	Tool used in Layer N+1	Output of the combined Layer:
No Tool	No Tool	Sum of Layer N and Layer N+1
No Tool	PNS	Invalid combination
PNS	No Tool	see subclause 4.6.13.6
PNS	PNS	Layer N+1

Stereo-stereo layer combination:

Tool used in Layer N	Tool used in Layer N+1	Output of the combined Layer:
No Tool or MS	No Tool or MS	Sum of Layer N and Layer N+1
No Tool or MS	PNS	Invalid combination
No Tool or MS	Intensity	Invalid combination
No Tool or MS	PNS & Intensity	Invalid combination
PNS	No Tool	see subclause 4.6.13.6
PNS	MS	Layer N+1
PNS	Intensity	Layer N+1

PNS	PNS	Layer N+1
PNS	PNS & Intensity	Layer N+1
Intensity	No Tool or MS	Layer N+1
Intensity	PNS	Invalid combination
Intensity	Intensity	Sum of Layer N and Layer N+1, only M/L-channel; Take Positions from Layer N+1
Intensity	PNS & Intensity	Invalid combination
PNS & Intensity	No Tool or MS	Invalid combination
PNS & Intensity	PNS	Invalid combination
PNS & Intensity	Intensity	Layer N+1
PNS & Intensity	PNS & Intensity	Layer N+1

**Mono-stereo layer combination:**

Tool used in Layer N	Tool used in Layer N+1	Output of the combined Layer:
No Tool	No Tool	Sum of Layer N and Layer N+1 (FSS-Tool)
No Tool	MS	Sum of Layer N and Layer N+1
No Tool	PNS	Invalid combination
No Tool	Intensity	Layer N+1
No Tool	PNS & Intensity	Layer N+1
PNS	No Tool	see subclause 4.6.13.6
PNS	MS	Layer N+1
PNS	Intensity	Layer N+1
PNS	PNS	Layer N+1
PNS	PNS & Intensity	Layer N+1

**4.5.2.3 Decoding of an individual\_channel\_stream (ICS) and ics\_info**

The Individual Channel Stream (ICS) elements are sub-elements of both, the MPEG-2 AAC style audio object types (AAC Main, LC, SSR, LTP), and the scalable AAC audio object type (AAC Scalable).

**4.5.2.3.1 Definitions**

**Bit stream elements:**

- `individual_channel_stream()` contains data necessary to decode one channel
- `ics_info()` contains side information necessary to decode an individual\_channel\_stream for SCE and CPE elements. The individual\_channel\_streams of a channel\_pair\_element may share one common ics\_info
- `ics_reserved_bit` bit reserved for future use
- `window_sequence` indicates the sequence of windows as defined in Table 4.72.
- `window_shape` A 1 bit field that determines what window is used for the right hand part of this analysis window
- `max_sfb` number of scalefactor bands transmitted per group either in ics\_info(), aac\_scalable\_main\_element(), or aac\_scalable\_extension\_element(). For TwinVQ elements max\_sfb is not transmitted directly, but is calculated in the decoder from other transmitted information as specified in subclause 4.5.2.2.6.1.
- `scale_factor_grouping` a bit field that contains information about grouping of short spectral data

**Help elements:**

- `scalefactor window band` term for scalefactor bands within a window, given in Table 4.73 to Table 4.91.
- `scalefactor band` term for scalefactor band within a group. In case of EIGHT\_SHORT\_SEQUENCE and grouping a scalefactor band may contain several scalefactor window bands

	of corresponding frequency. For all other window_sequences scalefactor bands and scalefactor window bands are identical.
g	group index
win	window index within group
sfb	scalefactor band index within group
swb	scalefactor window band index within window
bin	coefficient index
num_window_groups	number of groups of windows which share one set of scalefactors
window_group_length[g]	number of windows in each group.
bit_set(bit_field,bit_num)	function that returns the value of bit number bit_num of a bit_field (most right bit is bit 0)
num_windows	number of windows of the actual window sequence
num_swb_long_window	number of scalefactor bands for long windows. This number has to be selected depending on the sampling frequency. See subclause 4.5.4.
num_swb_short_window	number of scalefactor window bands for short windows. This number has to be selected depending on the sampling frequency. See subclause 4.5.4.
num_swb	number of scalefactor window bands for short windows in case of EIGHT_SHORT_SEQUENCE, number of scalefactor window bands for long windows otherwise
swb_offset_long_window[swb]	table containing the index of the lowest spectral coefficient of scalefactor band sfb for long windows. This table has to be selected depending on the sampling frequency.
swb_offset_short_window[swb]	table containing the index of the lowest spectral coefficient of scalefactor band sfb for short windows. This table has to be selected depending on the sampling frequency.
swb_offset[swb]	table containing the index of the lowest spectral coefficient of scalefactor band sfb for short windows in case of EIGHT_SHORT_SEQUENCE, otherwise for long windows
sect_sfb_offset[g][section]	table that gives the number of the start coefficient for the section_data() within a group. This offset depends on the window_sequence and scale_factor_grouping.
sampling_frequency_index	see subclause 1.6.3.3 of subpart 1.

#### 4.5.2.3.2 Decoding process

##### Decoding an individual\_channel\_stream (ICS)

In the individual\_channel\_stream, the order of decoding is:

- Get global\_gain
- Get ics\_info (parse bitstream if common information is not present)
- Get section\_data, if present
- Get scale\_factor\_data, if present
- Get pulse\_data if present
- Get tns\_data, if present
- Get gain\_control\_data, if present

If the HCR tool is not used:

- Get spectral\_data, if present

If the HCR tool is used:

- Get length\_of\_reordered\_spectral\_data, if present
- Get length\_of\_longest\_codeword, if present
- Get reordered\_spectral\_data, if present

The process of recovering pulse\_data is described in subclause 4.6.3, tns\_data in subclause 4.6.9, and gain\_control data in subclause 4.6.12. An overview of how to decode ics\_info section\_data, scalefactor\_data, and spectral\_data will be in the following paragraphs.

##### Recovering ics\_info()

For `single_channel_elements` `ics_info` is always located immediately after the `global_gain` in the `individual_channel_stream`. For a channel pair element there are two possible locations for the `ics_info`.

In case of `channel_pair_element` if the `common_window` flag is set to 1 both channels share the same `ics_info()` (i.e. both have same `window_sequence`, same `window_shape`, same `scale_factor_grouping`, same `max_sfb` etc.). Otherwise (i.e. `common_window` is set to 0) there is an `ics_info` immediately after the `global_gain` for each of the two `individual_channel_stream()`.

The `ics_info()` carries the window information associated with an ICS and thus permits channels in a `channel_pair` to switch separately if desired. In addition it carries the `max_sfb` which places an upper limit on the number of `ms_used[]` and `predictor_used[]` bits that must be transmitted. If the `window_sequence` is `EIGHT_SHORT_SEQUENCE` then `scale_factor_grouping` is transmitted. If a set of short windows form a group then they share scalefactors as well as intensity stereo positions and PNS information and have their spectral coefficients interleaved. The first short window is always a new group so no grouping bit is transmitted. Subsequent short windows are in the same group if the associated grouping bit is 1. A new group is started if the associated grouping bit is 0. It is assumed that grouped short windows have similar signal statistics. Hence their spectra are interleaved to place correlated coefficients next to each other. The manner of interleaving is indicated in Figure 4.22. The `ics_info()` also carries the `prediction_data()` for the individual channel or channel pair (see subclause 4.6.6 and 4.6.7).

### Recovering section\_data()

In the ICS, the information about one long window, or eight short windows, is recovered. The `section_data()` is the first field to be decoded, and describes the Huffman codes that apply to the scalefactor bands in the ICS. The form of the section data is:

**sect\_cb** The codebook for the section  
and  
**sect\_len** The length of the section.

This length is recovered by reading the bitstream sequentially for a section length, adding the escape value to the total length of the section until a non-escape value is found, which is added to establish the total length of the section. This process is clearly explained in the C-like syntax description. If the `aacSectionDataResilienceFlag` is set, **sect\_len\_incr** is not transmitted but is set to one per default in case the codebook for a section is 11 or in the range of 16 and 31. Note that within each group the sections must delineate the scalefactor bands from zero to **max\_sfb** so that the first section within each group starts at bands zero and the last section within each group ends at **max\_sfb**.

The sectioning data describes the codebook, and then the length of the section using that codebook, starting from the first scalefactor band and continuing until the total number of scalefactor bands is reached.

After this description is provided, all scalefactors and spectral data corresponding to codebook zero are zeroed, and no values corresponding to these scalefactors or spectral data will be transmitted. When scanning for scalefactor data it is important to note that scalefactors for any scalefactor bands whose Huffman codebook is zero will be omitted. Similarly, all spectral data associated with Huffman codebook zero are omitted.

In addition spectral data associated with the scalefactor bands that have an intensity codebook will not be transmitted, but intensity stereo positions will be transmitted in place of the scalefactors, as described in section 4.6.8.2.

### scalefactor\_data() parsing and decoding

For each scalefactor band that is not in a section coded with the zero codebook (`ZERO_HCB`), a scalefactor is transmitted. These will be denoted as 'active' scalefactor bands and the associated scalefactors as active scalefactors. Global gain, the first bitstream element in an ICS, is typically the value of the first active scalefactor. All scalefactors ( and also the stereo positions and pns energies) are transmitted using Huffman coded DPCM relative to the previous active scalefactor (respectively previous stereo position or previous pns energy, see subclause 4.6.2 and 4.6.3). The first active scalefactor is differentially coded relative to the global gain. Note that it is not illegal, merely inefficient, to provide a `global_gain` that is different from the first active scalefactor and then a

non-zero DPCM value for the first scalefactor DPCM value. If any intensity stereo positions are received interspersed with the DPCM scalefactor elements, they are sent to the intensity stereo module, and are not involved in the DPCM coding of scalefactor values. This also applies for PNS energies. The value of the first active scalefactor is usually transmitted as the `global_gain` with the first DPCM scalefactor having a zero value. Once the scalefactors are decoded, the actual values are found via a power function.

### **spectral\_data() parsing and decoding**

The spectral data is recovered as the last part of the parsing of an ICS. It consists of all the non-zeroed coefficients remaining in the spectrum or spectra, ordered as described in the `ICS_info`. For each non-zero, non-intensity codebook, the data are recovered via Huffman decoding in quads or pairs, as indicated in the noiseless coding tool (see subclause 4.6.9). If the spectral data is associated with an unsigned Huffman codebook, the necessary sign bits follow the Huffman codeword. In the case of the ESCAPE codebook, if any escape value is received, a corresponding escape sequence will appear after that Huffman code. There may be zero, one or two escape sequences for each codeword in the ESCAPE codebook, as indicated by the presence of escape values in that decoded codeword. For each section the Huffman decoding continues until all the spectral values in that section have been decoded. Once all sections have been decoded, the data is multiplied by the decoded scalefactors and deinterleaved if necessary.

If the HCR tool is used, spectral data does not consist of consecutive codewords anymore. Concerning HCR, the whole data necessary to decode two or four lines are referred as codeword. This includes Huffman codeword, sign bits, and escape sequences.

#### **4.5.2.3.3 Windows and window sequences**

Quantization and coding is done in the frequency domain. For this purpose, the time signal is mapped into the frequency domain in the encoder. The decoder performs the inverse mapping as described in subclause 4.6.11. Depending on the signal, the coder may change the time/frequency resolution by using two different windows: `LONG_WINDOW` and `SHORT_WINDOW`. To switch between windows, the transition windows `LONG_START_WINDOW` and `LONG_STOP_WINDOW` are used. Table 4.71 lists the windows, specifies the corresponding transform length and shows the shape of the windows schematically. Two transform lengths are used: 1024 (or 960) (referred to as long transform) and 128 (or 120) coefficients (referred to as short transform).

Window sequences are composed of windows in a way that a `raw_data_block` always contains data representing 1024 (or 960) output samples. The bitstream element **window\_sequence** indicates the window sequence that is actually used. lists how the window sequences are composed of individual windows. Refer to subclause 4.6.11 for more detailed information about the transform and the windows.

#### **4.5.2.3.4 Scalefactor bands and grouping**

Many tools of the decoder perform operations on groups of consecutive spectral values called scalefactor bands (abbreviation 'sfb'). The width of the scalefactor bands is built in imitation of the critical bands of the human auditory system. For that reason the number of scalefactor bands in a spectrum and their width depend on the transform length and the sampling frequency. Table 4.73 to Table 4.91 list the offset to the beginning of each scalefactor band on the transform lengths 1024 (960) and 128 (120) and on the sampling frequencies.

To reduce the amount of side information in case of sequences which contain `SHORT_WINDOWS`, consecutive `SHORT_WINDOWS` may be grouped (see Figure 4.20). The information about the grouping is contained in the **scale\_factor\_grouping** bitstream element. Grouping means that only one set of scalefactors is transmitted for all grouped windows as if there was only one window. The scalefactors are then applied to the corresponding spectral data in all grouped windows. To increase the efficiency of the noiseless coding (see subclause 4.6.3), the spectral data of a group is transmitted in an interleaved order given in subclause 4.5.2.3.5. The interleaving is done on a scalefactor band by scalefactor band basis, so that the spectral data can be grouped to form a virtual scalefactor band to which the common scalefactor can be applied. Within this document the expression 'scalefactor band' (abbreviation 'sfb') denotes these virtual scalefactor bands. If the scalefactor bands of the single windows are referred to, the expression 'scalefactor window band' (abbreviation 'swb') is used. Due to its influence on the scalefactor bands, grouping affects the meaning of `section_data` (see subclause 4.6.3), the order of spectral data (see subclause 4.5.2.3.5), and the total number of scalefactor bands. For a `LONG_WINDOW` scalefactor bands and scalefactor window bands are identical since there is only one group with only one window.

To reduce the amount of information needed for the transmission of side information specific to each scalefactor band, the bitstream element **max\_sfb** is transmitted. Its value is one greater than the highest active scalefactor band in all groups. **max\_sfb** has influence on the interpretation of section data (see subclause 4.6.3), the transmission of scalefactors (see subclause 4.6.3 and 4.6.2), the transmission of predictor data (see subclause 4.6.6 and 4.6.7), the FSS control information in the mono-stereo coding modes (see subclause 4.6.14 and 4.5.2.2), and the transmission of the **ms\_mask** (see subclause 4.6.8.1).

Since scalefactor bands are a basic element of the coding algorithm, some help variables and arrays are needed to describe the decoding process in all tools using scalefactor bands. These help variables depend on **sampling\_frequency**, **window\_sequence**, **scalefactor\_grouping** and **max\_sfb** and must be built up for each **raw\_data\_block**. The pseudo code shown below describes

- how to determine the number of windows in a **window\_sequence** named *num\_windows*
- how to determine the number of **window\_groups** named *num\_window\_groups*
- how to determine the number of windows in each group named *window\_group\_length[g]*
- how to determine the total number of scalefactor window bands named *num\_swb* for the actual window type
- how to determine *swb\_offset[swb]*, the offset of the first coefficient in scalefactor window band named *swb* of the window actually used
- how to determine *sect\_sfb\_offset[g][section]*, the offset of the first coefficient in section named *section*. This offset depends on **window\_sequence** and **scale\_factor\_grouping** and is needed to decode the *spectral\_data()*.

A long transform window is always described as a **window\_group** containing a single window. Since the number of scalefactor bands and their width depend on the sampling frequency, the affected variables are indexed with **sampling\_frequency\_index** to select the appropriate table.

```

fs_index = sampling_frequency_index;
switch( window_sequence ) {
  case ONLY_LONG_SEQUENCE:
  case LONG_START_SEQUENCE:
  case LONG_STOP_SEQUENCE:
    num_windows = 1;
    num_window_groups = 1;
    window_group_length[num_window_groups-1] = 1;
    num_swb = num_swb_long_window[fs_index];
    /* preparation of sect_sfb_offset for long blocks */
    /* also copy the last value! */
    for( i=0; i< max_sfb + 1; i++ ) {
      sect_sfb_offset[0][i] = swb_offset_long_window[fs_index][i];
      swb_offset[i] = swb_offset_long_window[fs_index][i];
    }
    break;
  case EIGHT_SHORT_SEQUENCE:
    num_windows = 8;
    num_window_groups = 1;
    window_group_length[num_window_groups-1] = 1;
    num_swb = num_swb_short_window[fs_index];
    for( i=0; i< num_swb_short_window[fs_index] + 1; i++ )
      swb_offset[i] = swb_offset_short_window[fs_index][i];
    for( i=0; i< num_windows-1; i++ ) {
      if( bit_set(scale_factor_grouping,6-i) == 0 ) {
        num_window_groups += 1;
        window_group_length[num_window_groups-1] = 1;
      }
      else {
        window_group_length[num_window_groups-1] += 1;
      }
    }
}
/* preparation of sect_sfb_offset for short blocks */
for( g=0; g<num_window_groups; g++ ) {
  sect_sfb = 0;
  offset = 0;
}

```

```

    for( i=0; i< max_sfb; i++ ) {
        width = swb_offset_short_window[fs_index][i+1] -
                swb_offset_short_window[fs_index][i];
        width *= window_group_length[g];
        sect_sfb_offset[g][sect_sfb++] = offset;
        offset += width;
    }
    sect_sfb_offset[g][sect_sfb] = offset;
}
break;
default:
break;
}

```

#### 4.5.2.3.5 Order of spectral coefficients in spectral\_data

If the Huffman codeword reordering (HCR) tool is used subclause 4.6.16.3.3 applies. Otherwise the spectral coefficients are ordered as follows.

For ONLY\_LONG\_SEQUENCE windows (`num_window_groups = 1`, `window_group_length[0] = 1`) the spectral data is in ascending spectral order, as shown in Figure 4.21.

For the EIGHT\_SHORT\_SEQUENCE window, the spectral order depends on the grouping in the following manner:

- Groups are ordered sequentially
- Within a group, a scalefactor band consists of the spectral data of all grouped SHORT\_WINDOWS for the associated scalefactor window band. To clarify via example, the length of a group is in the range of one to eight SHORT\_WINDOWS.
  - If there are eight groups each with length one (`num_window_groups = 8`, `window_group_length[0..7] = 1`), the result is a sequence of eight spectra, each in ascending spectral order.
  - If there is only one group with length eight (`num_window_groups = 1`, `window_group_length[0] = 8`), the result is that spectral data of all eight SHORT\_WINDOWS is interleaved by scalefactor window bands.
  - Figure 4.22 shows the spectral ordering for an EIGHT\_SHORT\_SEQUENCE with grouping of SHORT\_WINDOWS according to Figure 4.20 (`num_window_groups = 4`).
- Within a scalefactor window band, the coefficients are in ascending spectral order.

#### 4.5.2.3.6 Output word length

The global gain for each audio channel is scaled such that the integer part of the output of the IMDCT can be used directly as a 16-bit PCM audio output to a digital-to-analog (D/A) converter. This is the default mode of operation and will result in correct audio levels. If the decoder has a D/A converter with a resolution greater than 16-bit then the output of the IMDCT can be scaled up such that the appropriate number of fractional bits are included to form the desired D/A word size. In this case the level of the converter output would be matched to that of a 16-bit D/A, but would have the advantage of greater signal dynamic range and lower converter noise floor. Similarly, shorter D/A word lengths can be accommodated.

#### 4.5.2.4 Payloads for the audio object types ER AAC LC, ER AAC LTP, ER AAC LD and ER AAC scalable

For AAC, two kinds of bitstream payload syntax are available: scalable and multichannel. The following changes have to be applied to them to obtain their error resilient counterparts:

- **Multichannel AAC:** The syntax of the top-level payload has been modified. `raw_data_block()` is replaced by `er_raw_data_block()` as described in Table 4.19. Please note, that due to this modification `coupling_channel_element()`, `data_stream_element()`, `program_config_element()`, and `fill_element()` are not supported within the error resilient bitstream syntax.
- **Scalable AAC:** The syntax of `aac_scalable_main_element()` is not changed for error resilience.

No other changes regarding syntax occur.



Data elements are subdivided into different categories depending on its error sensitivity and collected in instances of these categories.

Depending on epConfig, there are several ways to obtain these instances on decoder site as described in subclause 1.6.3.5 of subpart 1.

Both, the order of these instances within an error resilient AAC frame and the order of data elements inside these instances, are described within the next section. If separate access units are used, the dependency structure between elementary streams has to be set up according to the order of instances.

**4.5.2.4.1 Error sensitivity category assignment**

The following table gives an overview about the error sensitive categories used for AAC (channel\_pair\_element() = CPE, individual\_channel\_stream() = ICS, extension\_payload() = EPL):

**Table 4.64 – Error sensitivity category assignment**

category	payload	mandatory	leads / may lead to one instance per	description
0	main	yes	CPE	commonly used side information
1	main	yes	ICS	channel dependent side information
2	main	no	ICS	error resilient scale factor data
3	main	no	ICS	TNS data
4	main	yes	ICS	spectral data
5	extended	no	EPL	extension type
6	extended	no	EPL	DRC data
7	extended	no	EPL	bit stuffing

Table 4.92 shows the category assignment for the main payload (supported elements are SCE, LFE, and CPE). Within this table “-“ means that this data element does not occur within this configuration.

Table 4.93 shows the category assignment for the extended payload.

**4.5.2.4.2 Category instances and its dependency structure**

The subdivision into instances is done on a frame basis, in case of scalable syntax in addition on a layer basis.

The order of instances within the error resilient AAC frame/layer as well as the dependency structure in case of several elementary streams is assigned according to the following rules:

**Table 4.65 – Dependency structure within the ER AAC payload**

hierarchy level	error resilient multi-channel syntax	error resilience scalable syntax
frame / layer	base payload followed by extension payload	
base payload	order of syntactic elements follows order stated in Table 4.19	commonly used side information followed by ICSs
extended payload	no rule regarding the order of multiple EPLs is given, the kind of extension payload can be identified by extension_type	
syntactic element in base payload	commonly used side information followed by ICSs	-
ICS order in case of stereo	left channel followed by right channel	
ICS / EPL	dependency structure according to category numbers	

Figure 4.23 shows an example for the error resilient multi-channel syntax. The order of data elements inside each instance is based on the syntax description, i. e. the logical order is kept.

Note: If the reversible variable length coding (RVLC) tool is not used (`aacScalefactorDataResilienceFlag == 0`), the decoder does not expect instances of error sensitivity category 2, i. e. no access unit shall contain those instances.

#### 4.5.2.5 Payloads for the audio object types TwinVQ and ER TwinVq

##### 4.5.2.5.1 Definitions

<code>tvq_scalable_main_element()</code>	data element to decode core TwinVQ data.
<code>tvq_scalable_extension_element()</code>	data element to decode scalable TwinVQ data.
<code>vq_single_element()</code>	A data block containing one frame of syntax elements for TwinVQ base layer element and TwinVQ enhancement layer element.
<b>ppc_present</b>	indicates the activation of periodic peak components coding.
<b>postprocess_present</b>	indicates the activation of the postprocessor in the spectral normalization
<b>bandlimit_present</b>	indicates the activation of the bandwidth control of MDCT coefficients
<b>fb_shift</b>	indicates location of active frequency band in enhancement layer
<b>index_blim_h</b>	indicates the upper boundary for the bandwidth control process of the spectrum normalization tool.
<b>index_blim_l</b>	indicates the lower boundary for the bandwidth control process of the spectrum normalization tool
<b>index_shape0</b>	indicates the codevector number of the shape codebook 1 and polarity of the codevector for the interleaved vector quantization tool.
<b>index_shape1</b>	indicates the codevector number of the shape codebook 1 of the interleaved vector quantization tool.
<b>index_env</b>	indicates the codevector number of the Bark-scale envelope codebook of the spectrum normalization tool.
<b>index_fw_alf</b>	indicates the prediction switch of the Bark-scale envelope coding of the spectrum normalization tool.
<b>index_gain</b>	indicates the gain factor of the spectrum normalization tool.
<b>index_gain_sb</b>	indicates the sub-block gain factor of the spectrum normalization tool.
<b>index_lsp0</b>	indicates LSP MA prediction switch of the spectrum normalization tool.
<b>index_lsp1</b>	indicates codevector number of the first-stage LSP VQ in the spectrum normalization tool.
<b>index_lsp2</b>	indicates codevector number of the second-stage LSP VQ in the spectrum normalization tool.
<b>index_shape0_p</b>	indicates the codevector number of codebook 0 of the periodic peak component coding in the spectrum normalization tool.
<b>index_shape1_p</b>	indicates the codevector number of codebook 1 of the periodic peak component coding in the spectrum normalization tool.
<b>index_pit</b>	indicates the base frequency of the periodic peak component in the spectrum normalization tool.
<b>index_pgain</b>	indicates the gain factor of the periodic peak component in the spectrum normalization tool.

##### 4.5.2.5.2 Parameter setting

Following parameters are used for decoding of TwinVQ base layer element and the TwinVQ enhancement layer element:

<code>N_CH</code>	number of channels defined by the system layer
<code>N_DIV</code>	number of sub-vector division for interleaved vector quantization It is calculated according to the decoder status.
<code>N_SF</code>	number of filterbank subblocks in a frame
<code>FW_N_DIV</code>	number of codebook division for the Bark-scale envelop quantization
<code>LSP_SPLIT</code>	number of subvectors for LSP VQ.
<code>N_DIV_P</code>	number of sub-vector division for periodic peak component coding

These parameters are also referenced from the interleaved vector quantization tool and the spectrum normalization tool. The following parameters have constant values as shown below:

N\_SF (SHORT) = 8  
 N\_SF (LONG) = 1  
 FW\_N\_DIV = 7  
 LSP\_SPLIT = 3  
 N\_DIV\_P = 2\*N\_CH

4.5.2.5.3 Bit allocation

4.5.2.5.3.1 Spectrum normalization tool

For syntax elements listed below, the number of bits is set according to the parameters **lyr** and **window\_sequence**:

Table 4.66 – Bit allocation of syntax elements

Name of variables	Name of number of bits	lyr = 0 LONG	lyr = 0 SHORT	lyr >= 1 LONG	lyr >= 1 SHORT	Times per frame
<b>fb_shift</b>	-	0	0	2	2	N_CH
<b>index_blim_h</b>	-	2/0	2/0	0	0	N_CH
<b>index_blim_l</b>	-	1/0	1/0	0	0	N_CH
<b>index_env</b>	FW_N_BIT	6	0	6	0	FW_N_DIV*N_CH
<b>index_fw_alf</b>	-	1	0	1	0	N_CH
<b>index_gain</b>	GAIN_BIT	9	9	8	8	N_CH
<b>index_gain_sb</b>	SUB_GAIN_BIT	0	4	0	4	N_SF*N_CH
<b>index_lsp0</b>	LSP0_BIT	1	1	1	1	N_CH
<b>index_lsp1</b>	LSP1_BIT	6	6	6	6	N_CH
<b>index_lsp2</b>	LSP2_BIT	4	4	4	4	LSP_SPLIT*N_CH
<b>index_shape0_p</b>	MAXBIT_P+1	7/0	0	0	0	N_DIV_P
<b>index_shape1_p</b>	MAXBIT_P+1	7/0	0	0	0	N_DIV_P
<b>index_pit</b>	BASF_BIT	8/0	0	0	0	N_CH
<b>index_pgain</b>	PGAIN_BIT	7/0	0	0	0	N_CH

4.5.2.5.3.2 Interleaved VQ tool

Parameter N\_DIV, Number of bits of shape code index 0, bits0, and number of bits of shape code index 1, bits1, are calculated according to the following procedures:

First, number of bits for side information, bits\_for\_side\_information is calculated as follows:

$$\text{bits\_for\_side\_information} = \text{WS\_TBIT} + \text{OPT\_TBIT} + \text{LSP\_TBIT} + \text{GAIN\_TBIT} + \text{FW\_TBIT} + \text{PIT\_TBIT} + \text{BAND\_TBIT} + \text{used\_bits}$$

where

WS\_TBIT is number of bits for the flags for window sequence, window shape, joint stereo coding (ms\_present), LTP, and TNS tools,

OPT\_BIT is number of bits for the flags for bandlimit, ppc, postprocess and active band selection.

Note that If ppc\_present is activated, 43 bits per channel (PIT\_TBIT), and bandlimit\_present is activated 3 bits per channel (BAND\_TBIT) are consumed.

used\_bit is number of bits used by tools other than spectrum normalization tool such as joint stereo tools, LTP tools and TNS tools.

LSP\_TBIT, GAIN\_TBIT, FW\_TBIT, and PIT\_TBIT is number of bits for optional information, lsp coding, gain coding, Bark-scale envelope coding, and periodic peak components coding respectively. They are set as follows:

$$\text{LSP\_TBIT} = (\text{LSP\_BIT0} + \text{LSP\_BIT1} + (\text{LSP\_BIT2} * \text{LSP\_SPLIT})) * \text{N\_CH};$$

if (FW\_N\_BIT > 0){

```

    FW_TBIT = ((FW_N_BIT * FW_N_DIV + 1) * N_SF) * N_CH;
}
else{
    FW_TBIT = 0;
}

switch(window_sequence){
    case EIGHT_SHORT_SEQUENCE:
        GAIN_TBIT = (GAIN_BIT + SUB_GAIN_BIT * N_SF) * N_CH;
        PIT_TBIT = 0;
        break;

    default:
        GAIN_TBIT = GAIN_BIT * N_CH;
        if (ppc_present == TRUE)
            PIT_TBIT = (MAXBIT_P+1)*N_DIV_P*2+ (BASF_BIT + PGAIN_BIT) * N_CH;

        else
            PIT_TBIT = 0;
        break;
}

```

Numbers of bits for side Information are listed in the following table.

**Table 4.67 – Bit allocation of side information**

Scalable layer lyr Window_sequence	0 long	0 long	0 short	0 short	>=1 long	>=1 long	>=1 short	>=1 short
N_CH	1	2	1	2	1	2	1	2
WS_TBIT	5	9	5	9	0	2	0	2
OPT_TBIT	3	3	1	1	2	4	2	4
LSP_TBIT	19	38	19	38	19	38	19	38
GAIN_TBIT	9	18	41	82	8	16	40	80
FW_TBIT	43	86	0	0	43	86	0	0
BAND_TBIT	3/0	6/0	3/0	6/0	0	0	0	0
PIT_TBIT	43/0	86/0	0	0	0	0	0	0

Number of available bits, bits\_available\_vq is calculated as follows:

bits\_available\_vq =

$$(\text{int})(((\text{FRAME\_SIZE} * \text{bitrate}/\text{sampling\_frequency})/8+0.5)*8) - \text{bits\_for\_side\_information},$$

where bitrate is given by a system parameter in [bit/s] and sampling frequency is given in the right column of Table 4.68

Finally, number of shape sub-vector, N\_DIV and number of bits for shape code indexes, bits0 and bits1, are calculated as follows:

$$N\_DIV = ((\text{int})((\text{bits\_available\_vq} + \text{MAXBIT} * 2 - 1) / (\text{MAXBIT} * 2)));$$

$$\text{bits} = (\text{bits\_available\_vq} + N\_DIV - 1 - \text{idiv}) / N\_DIV;$$

$$\text{bits0} = (\text{int})(\text{bits} + 1) / 2;$$

$$\text{bits1} = (\text{int})\text{bits} / 2;$$

where MAXBIT is maximum number of shape code bit. The MAXBIT is always set to 6.

#### 4.5.2.5.3.3 Scalable coder by means of TwinVQ

Scalable coder can be constructed by cascading the TwinVQ core and the TwinVQ extension (enhancement) quantizers. The decoding process is straight forward. MDCT coefficients are reconstructed by using the demultiplexed bitstream for each layer. Each enhancement layer covers the different part of the coefficients whose regions are adaptively specified by the shift parameters. Summation of these reconstructed coefficients is

performed. Finally, an IMDCT filter bank maps back the spectrum (MDCT coefficients) into a time domain signal using a synthesis window and overlap / add techniques. If one wishes to decode only the core MDCT coefficients, the decoded core coefficients have to be passed on directly to the filter bank. The core layer of this scalable coder can be combined with the AAC tools such as TNS, LTP and joint stereo coding. This coder, however, does neither support the mono-stereo cascading coder nor the FSS control.

If in `tvq_scalable_element` the following condition is true:

```
(ms_mask == 1) and
(this_layer_stereo == 1) and
(window_sequence == EIGHT_SHORT_SEQUENCE) and
there is no scale_factor_grouping information in the previous layer(s)
```

then

`num_window_group` shall be set to '1' (`scale_factor_grouping = 0x7F`).

#### 4.5.2.6 Payloads for the audio object type ER BSAC

##### 4.5.2.6.1 Decoding of payload for audio object type ER BSAC (`bsac_payload()`)

Fine grain scalability would create large overhead if one would try to transmit fine grain layers over multiple elementary streams (ES). So, in order to reduce overhead and implement the fine grain scalability efficiently in current MPEG-4 system, the server can organize the fine grain audio data into the payload by dividing the fine grain audio data into the large-step layers and concatenating the large step layers of the several sub-frames. Then the payload is transmitted over ES.

So, the payload transmitted over ES requires the rearrangement process for the actual decoding.

##### 4.5.2.6.1.1 Definitions

`bsac_payload(layer)` Sequence of `bsac_istep_element()`s. Syntactic element of the payload transmitted over  $lay^{th}$  layer ES. A `bsac_payload(layer)` basically consists of several  $lay^{th}$  layer bitstream, `bsac_istep_element()` of several sub-frames.

`bsac_istep_element(frame, layer)` Syntactic element for the  $lay^{th}$  large-step layer bitstream of  $frame^{th}$  sub-frame.

`bsac_stream_byte[frame][offset+i]` (`offset+i`)-th byte which is extracted from the payload. After `bsac` stream bytes are extracted from all the payloads that have been transmitted to the receiver, these data are concatenated and saved in the array `bsac_stream_byte[frame]` which is the bitstream of  $frame^{th}$  sub-frame. Then, we proceed to decode the concatenated stream, `bsac_stream_byte[frame]` using the syntax of the BSAC fine grain scalability.

##### Help elements:

`data_available()` function that returns "1" as long as data is available, otherwise "0"

`LayerStartByte[frame][layer]` Start position of  $lay^{th}$  large-step layer in bytes which is located on  $frame^{th}$  sub-frame's bitstream. See sub-clause 4.5.2.6.1.2 for the calculation process of this value.

`LayerLength[frame][layer]` Length of the large-step layer in bytes which is located on the payload of  $lay^{th}$  layer ES and concatenated to  $frame^{th}$  sub-frame's bitstream. See sub-clause 4.5.2.6.1.2 for the calculation process of this value.

`LayerOffset[frame][layer]` Start position of the large-step layer of  $frame^{th}$  frame in bytes which is located on the payload of  $lay^{th}$  layer ES. See sub-clause 4.5.2.6.1.2 for the calculation process of this value.

`frm` index of frame in which `bsac` stream bytes are saved.

`lay` index of the large-step layer over which the fine granule audio data is transmitted.

`numOfSubFrame` Number of the sub-frames which are grouped and transmitted in a super-frame in order to reduce the transmission overhead.

`layer_length` Average length of the large-step layers in bytes which are assembled in a payload.

`numOfLayer` number of the large-step layers which the fine grain audio data is divided into.

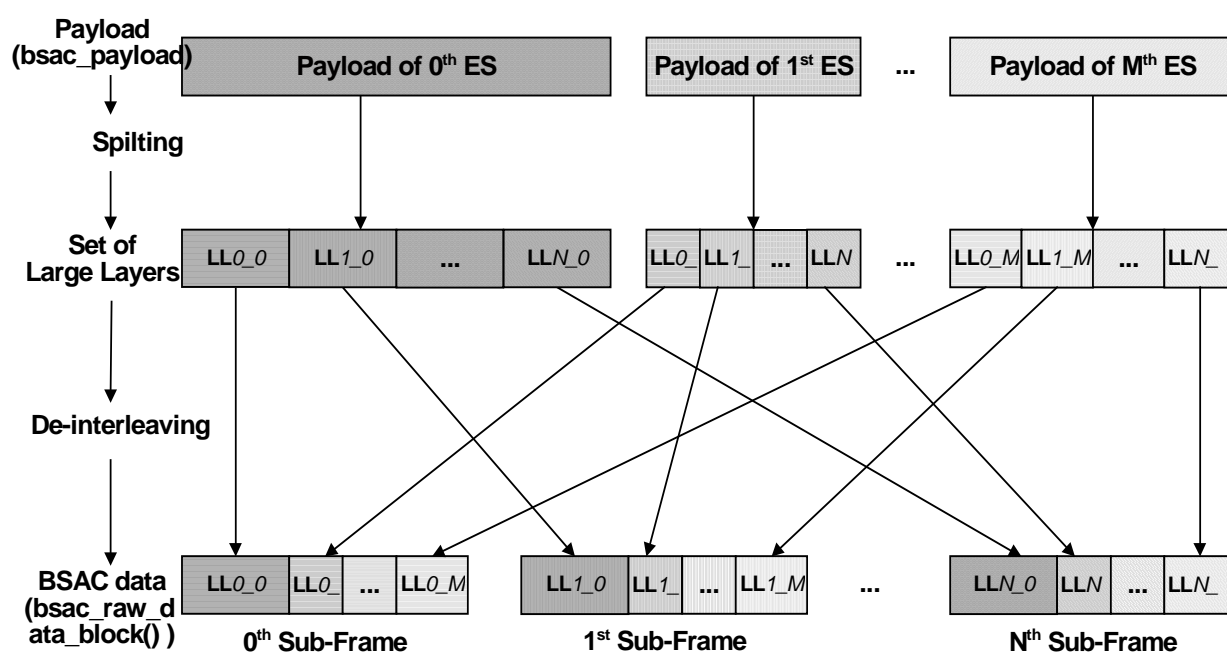
#### 4.5.2.6.1.2 Decoding process

On the sync layer (SL) of MPEG-4 system, an elementary stream is packetized into access units or parts thereof. Such a packet is called SL packet. Access Unit(AU)s are the only semantic entities at the sync layer (SL) of MPEG-4 system that need to be preserved from end to end. AUs are used as the basic unit for synchronization which are made up of one or more SL packets.

The dynamic data for the BSAC is transmitted as SL\_Packet payload in the base layer Elementary Stream(ES) and the enhancement layer ESs. The dynamic data is made up of the large-step layers of one or more subsequent sub-frames.

When the SL packets of an AU arrives in the receiver, a sequence of packet is mapped into a payload which is split into the large step layers,  $bsac\_lstep\_layer(frm, lay)$  for the subsequent sub-frames. And the split layers should be concatenated with the large-step layers which are transmitted over the other ES.

In the receiver, BSAC data is reconstructed from the payloads as shown in Figure 4.16.



where,  $LL_{i_k}$  is the k-th large-step layer of the i-th sub-frame

(M+1) is the number of the large-step layer to be transmitted (numOfLayer)

(N+1) is the number of the sub-frame to be grouped in an AU (numOfSubFrame)

Figure 4.16 – Reconstruction of BSAC data

The large-step layers are split from a payload of k<sup>th</sup> layer ES that is organized as shown in Figure 4.17.

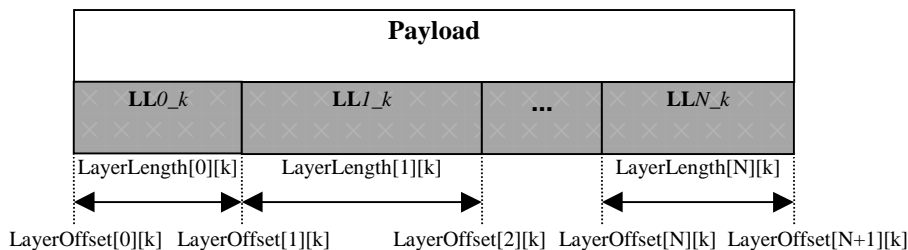


Figure 4.17 – Structure of the payload

The split large-step layers are deinterleaved and concatenated to map the entire fine grain BSAC data. And then, decode the concatenated bitstreams using the syntax (`bsac_raw_data_block()`) for fine grain scalability to make the reconstructed signal.

Some help variables and arrays are needed to describe the re-arranging process of the payload transmitted over ES. These help variables depend on layer, `numOfLayer`, `numOfSubFrame`, `layer_length` and `frame_length` and must be built up for mapping `bsac_raw_data_block()` of each sub-frame from the payloads. The pseudo code shown below describes

- how to calculate  $LayerLength[i][k]$ , the length of the large-step layer which is located on the fine granule audio data, `bsac_raw_data_block()` of  $i^{th}$  sub-frame.
- how to calculate  $LayerOffset[i][k]$  which indicates the start position of the large-step layer of  $i^{th}$  frame which is located on the payload of the  $k^{th}$  ES (`bsac_payload()`)
- how to calculate  $LayerStartByte[i][k]$  which indicates the start position of the large-step layer which is located on the fine granule audio data, `bsac_raw_data_block()` of  $i^{th}$  sub-frame

```

for (k = 0; k < numOfLayer; k++) {
    LayerStartByte[0][k] = 0;
    for (i = 0; i < numOfSubFrame; i++) {
        if (k == (numOfLayer-1) ) {
            LayerEndByte[i][k] = frame_length[i];
        } else {
            LayerEndByte[i][k]=LayerStartByte[i][k] + layer_length[k];
            if (frame_length[i] < LayerEndByte[i][k])
                LayerEndByte [i][k] = frame_length[i];
        }
        LayerStartByte[i+1][k] = LayerEndByte[i][k];
        LayerLength[i][k] = LayerEndByte[i][k] - LayerStartByte[i][k];
    }
}
for (k = 0; k < numOfLayer; k++) {
    LayerOffset[0][k] = 0;
    for (i = 0; i < numOfSubFrame; i++) {
        LayerOffset[i+1][k] = LayerOffset[i][k] + LayerLength[i][k];
    }
}

```

Where,  $frame\_length[i]$  is the length of  $i^{th}$  frame's bitstream which is obtained from the syntax element `frame_length` and  $layer\_length[i]$  is the average length of the large-step layers in the payload of  $i^{th}$  layer ES and is obtained from `AudioDecoderSpecificInfo`.

#### 4.5.2.6.2 Decoding of a `bsac_raw_data_block()`

##### 4.5.2.6.2.1 Definitions

Bitstream elements:

<code>bsac_raw_data_block()</code>	block of raw data that contains coded audio data, related information and other data. A <code>bsac_raw_data_block()</code> basically consists of <code>bsac_base_element()</code> and several <code>bsac_layer_element()</code> .
<code>bsac_base_element()</code>	Syntactic element of the base layer bitstream containing coded audio data, related information and other data.
<b>frame_length</b>	the length of the frame including headers in bytes.
<code>bsac_header()</code>	contains general information used for BSAC.
<b>header_length</b>	the length of the headers including <code>frame_length</code> , <code>bsac_header()</code> and <code>general_header()</code> in bytes. The actual length is $(\text{header\_length}+7)$ bytes. However if <code>header_length</code> is 0, it represents that the actual length is smaller than or equal to 7 bytes. And if <code>header_length</code> is 15, it represents that the actual length is larger than or equal to $(15+7)$ bytes and should be calculated through the decoding of the headers .
<b>sba_mode</b>	indicates that the segmented binary arithmetic coding (SBA) scheme is used if this element is 1. Otherwise the general binary arithmetic coding scheme is used.
<b>top_layer</b>	top scalability layer index
<b>base_snf_thr</b>	significance threshold used for coding the bit-sliced data of the base layer.
<b>base_band</b>	indicates the maximum spectral line of the base layer. If the <code>window_sequence</code> is <code>SHORT_WINDOW</code> , $4*\text{base\_band}$ is the maximum spectral line. Otherwise, $32*\text{base\_band}$ is the maximum spectral line.
<b>max_scalefactor[ch]</b>	the maximum value of the scalefactors
<b>cband_si_type[ch]</b>	the type of the coding band side-information(si). Using this element, the largest value of <code>cband_si</code> 's and the arithmetic model for decoding <code>cband_si</code> can be set as shown in Table 4.A.29.
<b>base_scf_model[ch]</b>	the arithmetic model for decoding the scalefactors in the base layer.
<b>enh_scf_model[ch]</b>	the arithmetic model used for decoding the scalefactors in the other enhancement layers.
<b>max_sfb_si_len[ch]</b>	maximum length which can be used per channel for coding the scalefactor-band side information including scalefactor and stereo-related information within a scalefactor band. This value has a offset(5). The actual maximum length is $(\text{max\_sfb\_si\_len}+5)$ . This value is used for determining the bitstream size of each layer.
<code>general_header()</code>	contains header data for the General Audio Coding
<b>reserved_bit</b>	bit reserved for future use
<b>window_sequence</b>	indicates the sequence of windows. See ISO/IEC 14496-3 General Audio Coding.
<b>window_shape</b>	A 1 bit field that determines what window is used for the trailing part of this analysis window
<b>max_sfb</b>	number of scalefactor bands transmitted per group
<b>scale_factor_grouping</b>	A bit field that contains information about grouping of short spectral data
<b>pns_data_present</b>	the flag indicating whether the perceptual noise substitution(pns) will be used (1) or not (0).
<b>pns_start_sfb</b>	the scalefactor band from which the pcns tool is started.
<b>ms_mask_present</b>	this two bit field (see ) indicates that the stereo mask is 00 Independent 01 1 bit mask of <code>ms_used</code> is located in the layer sfb side information part ( <code>layer_sfb_si()</code> ). 10 All <code>ms_used</code> are ones 11 2 bit mask of <code>stereo_info</code> is located in the layer sfb side information part <code>layer_sfb_si()</code> .
<code>layer_cband_si()</code>	contains the coding band side information necessary for Arithmetic encoding/decoding of the bit-sliced data within a coding band.
<code>layer_sfb_si()</code>	contains the side information of a scalefactor band such as the stereo-, the pns and the scalefactor information. <code>bsac_layer_element()</code> Syntactic element of the enhancement layer bitstream containing coded audio data for a time period of 1024(960) samples, related information and other data.
<code>bsac_layer_spectra()</code>	contains the arithmetic coded audio data of the quantized spectral coefficients which are newly added to each layer. See subclause 4.5.2.6.2.5 for the new spectral coefficients.
<code>bsac_lower_spectra()</code>	contains the arithmetic coded audio data of the quantized spectral coefficients which are lower than the spectra added to each layer.



`bsac_higher_spectra()` contains the arithmetic coded audio data of the quantized spectral coefficients which are higher than the spectra added to each layer.  
`bsac_spectral_data()` contains the arithmetic coded audio data of the quantized spectral coefficients.

**Help elements:**

`data_available()` function that returns “1” as long as bitstream is available, otherwise “0”  
`nch` a bitstream element that identifies the number of the channel.  
`scalefactor window band` term for scalefactor bands within a window. See ISO/IEC 14496-3 General Audio Coding.  
`scalefactor band` term for scalefactor band within a group. In case of `EIGHT_SHORT_SEQUENCE` and grouping a scalefactor band may contain several scalefactor window bands of corresponding frequency. For all other window\_sequences scalefactor bands and scalefactor window bands are identical.  
`g` group index  
`win` window index within group  
`sfb` scalefactor band index within group  
`swb` scalefactor window band index within window  
`num_window_groups` number of groups of windows which share one set of scalefactors. See subclause 4.5.2.6.2.4  
`window_group_length[g]` number of windows in each group. See subclause 4.5.2.6.2.4  
`bit_set(bit_field,bit_num)` function that returns the value of bit number `bit_num` of a `bit_field` (most right bit is bit 0)  
`num_windows` number of windows of the actual window sequence. See subclause 4.5.2.6.2.4  
`num_swb_long_window` number of scalefactor bands for long windows. This number has to be selected depending on the sampling frequency. See ISO/IEC 14496-3 General Audio Coding.  
`num_swb_short_window` number of scalefactor window bands for short windows. This number has to be selected depending on the sampling frequency. See ISO/IEC 14496-3 General Audio Coding.  
`num_swb` number of scalefactor window bands for shortwindows in case of `EIGHT_SHORT_SEQUENCE`, number of scalefactor window bands for long windows otherwise. See subclause 4.5.2.6.2.4  
`swb_offset_long_window[swb]` table containing the index of the lowest spectral coefficient of scalefactor band `sfb` for long windows. This table has to be selected depending on the sampling frequency. See ISO/IEC 14496-3 General Audio Coding.  
`swb_offset_short_window[swb]` table containing the index of the lowest spectral coefficient of scalefactor band `sfb` for short windows. This table has to be selected depending on the sampling frequency. See ISO/IEC 14496-3 General Audio Coding.  
`swb_offset[g][swb]` table containing the index of the lowest spectral coefficient of scalefactor band `sfb` for short windows in case of `EIGHT_SHORT_SEQUENCE`, otherwise for long windows. See subclause 4.5.2.6.2.4  
`layer_group[layer]` indicates the group index of the spectral data to be added newly in the scalability layer  
`layer_start_sfb[layer]` indicates the index of the lowest scalefactor band index to be added newly in the scalability layer  
`layer_end_sfb[layer]` indicates the highest scalefactor band index to be added newly in the scalability layer  
`layer_start_cband[layer]` indicates the lowest coding band index to be added newly in the scalability layer  
`layer_end_cband[layer]` indicates the highest coding band index to be added newly in the scalability layer  
`layer_start_index[layer]` indicates the index of the lowest spectral component to be added newly in the scalability layer  
`layer_end_index[layer]` indicates the index of the highest spectral component to be added newly in the scalability layer  
`start_index[g]` indicates the index of the lowest spectral component to be coded in the group `g`  
`end_index[g]` indicates the index of the highest spectral component to be coded in the group `g`  
`layer_data_available()` function that returns “1” as long as each layer’s bitstream is available, otherwise “0”. In other words, this function indicates whether the remaining bitstream of each layer is available or not.

*terminal\_layer[layer]* indicates whether a layer is the terminal layer of a segment which is made up of one or more scalability layers. If the segmented binary arithmetic coding is not activated, all these values are always set to 0 except that of the top layer. Otherwise, these values are defined as described in subclause 4.6.4.6.3.

#### 4.5.2.6.2.2 Decoding process

##### 4.5.2.6.2.2.1 Zero stuffing

In order to do the arithmetic decoding perfectly, 32-bit zero value should be concatenated to the bitstream. In case of the SBA mode, the bitstream of a frame is split into the several segments. So, zero value should be concatenated to all segments. For the detailed description, see subclause 4.6.4.6.3. However, in case of the non-SBA mode, one zero stuffing is good enough since the bitstream of a frame is not split.

##### 4.5.2.6.2.2.2 bsac\_raw\_data\_block

A total BSAC stream, *bsac\_raw\_data\_block* has the layered structure. First, *bsac\_base\_element* is parsed and decoded which is the bitstream for base scalability layer. Then, *bsac\_layer\_element* for the next enhancement layer is parsed and decoded. *bsac\_layer\_element* decoding routine is repeated while the decoded bitstream data is available and layer is smaller than or equal to the top layer, **top\_layer**.

##### 4.5.2.6.2.2.3 bsac\_base\_element

A *bsac\_base\_element* is made up of **frame\_length**, *bsac\_header*, *general\_header* and *bsac\_layer\_element*().

First, **frame\_length** is parsed from syntax. It represents the length of the frame including headers in bytes.

The syntax elements for the base layer are parsed which are composed of a *bsac\_header*(), a *general\_header*(), a *layer\_cband\_si*(), *layer\_sfb\_si*() and *bsac\_layer\_element*. *bsac\_base\_element* has several *bsac\_layer\_element* because the base layer is split into the several sub-layers for the error resilience of the base layer. The number of the sub-layers, *s\_layer\_size* is calculated using the group index and the coding band as shown in subclause 4.5.2.6.2.5.

##### 4.5.2.6.2.2.4 Recovering a bsac\_header

BSAC provides a 1-kits/sec/ch fine grain scalability which has the layered structure, one base layer and several enhancement layers. Base layer contains the general side information for all the layers, the specific side information for the base layer and the audio data. The general side information is transmitted in the syntax of *bsac\_header*() and *general\_header*() .

*bsac\_header* consists of **top\_layer**, **header\_length**, **sba\_mode**, **base\_band**, **max\_scalefactor**, **cband\_si\_type**, **base\_scf\_model** and **enh\_scf\_model**. All the bitstream elements are included in the form of the unsigned integer.

First, 4 bit **header\_length** is parsed which represents the length of the headers including *frame\_length*, *bsac\_header* and *general\_header* in bytes. The length of the headers is  $(\text{header\_length}+7)*8$ . Next, 1bit **sba\_mode** is parsed which represents whether the segmented binary arithmetic coding (SBA) is used or the binary arithmetic coding is used.

Next, 6 bit **top\_layer** is parsed which represents the top scalability layer index to be encoded. Next, 2 bit **base\_snf\_thr** is parsed which represents the significance threshold used for coding the bit-sliced data of the base layer.

Next, 8 bit **max\_scalefactor** is parsed which represents the maximum value of the scalefactors. If the number of the channel is not 1, this value is parsed one more.

Next, 5-bit **base\_band** is parsed which represents minimum spectral line which is coded in the base layer. If the window sequence is SHORT\_WINDOW,  $4*\text{base\_band}$  indicates the minimum spectral line. Otherwise  $32*\text{base\_band}$  indicates the minimum spectral line.

And, 5 bit **cband\_si\_type** is parsed which represents the arithmetic model of **cband\_si** and the largest **cband\_si** which can be decoded as shown in Table 4.A.29. 3 bit **base\_scf\_model** and **enh\_scf\_model** are parsed which represent the arithmetic model table for the scalefactors of the base layer and the other enhancement layers, respectively. Next, 4 bit **max\_sfb\_si\_len** is parsed which represents the maximum length of the scalefactor band side information to be able to be used in each layer. The maximum length is  $(\text{max\_sfb\_si\_len}+5)$ .

#### 4.5.2.6.2.2.5 Recovering a general\_header

The order for decoding the syntax of a **bsac\_header** is:

- get reserved\_bit
- get window\_sequence
- get window\_shape
- get max\_sfb
- get scale\_factor\_grouping if the window\_sequence is EIGHT\_SHORT\_SEQUENCE
- get pns\_present
- get pns\_start\_sfb if present
- get ms\_mask\_present flag if the number of the channel is 2
- get tns\_data\_present
- get TNS data if present
- get ltp\_data\_present
- get ltp data if present

If the number of the channel is not 1, the decoding of another channel is done as follows:

- get tns\_data\_present
- get TNS data if present
- get ltp\_data\_present
- get ltp data if present

The process of recovering **tns\_data** and **ltp\_data** is described in ISO/IEC 14496-3 General Audio Coding.

#### 4.5.2.6.2.2.6 bsac\_layer\_element

A **bsac\_layer\_element** is an enhancement layer bitstream and composed of **layer\_cband\_si()**, **layer\_sfb\_si()**, **bsac\_layer\_spectra()**, **bsac\_lower\_spectra()** and **bsac\_higher\_spectra()**. Decoding process of **bsac\_layer\_element** is as follows:

- Decode **layer\_cband\_si**
- Decode **layer\_sfb\_si**
- Decode **bsac\_layer\_spectra**
- Decode **bsac\_lower\_spectra**
- Decode **bsac\_higher\_spectra**

#### 4.5.2.6.2.2.7 Decoding of coding band side information (layer\_cband\_si)

The spectral coefficients are divided into coding bands which contain 32 quantized spectral coefficients for the noiseless coding. Coding bands(abbreviation 'cband') are the basic units used for the noiseless coding.

**cband\_si** represents the MSB plane and the probability table of the sliced bits within a coding band as shown in Table 4.A.31. Using this **cband\_si**, the bit-sliced data of each coding band are arithmetic-coded.

**cband\_si** is arithmetic\_coded with the model which is given in the syntax element **cband\_si\_type** as shown in Table 4.A.29.

An overview of how to decode **cband\_si** will be given in subclause 4.6.4.5.

#### 4.5.2.6.2.2.8 Decoding of the scalefactor band side information (layer\_sfb\_si)

An overview of how to decode **layer\_sfb\_si** will be given here. **layer\_sfb\_si** is made up of as follows:

Decoding of stereo\_info, ms\_used or noise\_flag.  
Decoding of scalefactors

#### 4.5.2.6.2.2.9 Decoding of stereo\_info, noise\_flag or ms\_used

Decoding process of stereo\_info, noise\_flag or ms\_used is depended on pns\_data\_present, number of channel, ms\_mask\_present.

If pns data is not present, decoding process is as follows:

If ms\_mask\_present is 0, arithmetic decoding of stereo\_info or ms\_used is not needed.

If ms\_mask\_present is 2, all ms\_used values are ones in this case. So, M/S stereo processing of AAC is done at all scalefactor band.

If ms\_mask\_present is 1, 1 bit mask of max\_sfb bands of ms\_used is conveyed in this case. So, ms\_used is arithmetic decoded. M/S stereo processing of AAC is done according to the decoded ms\_used.

If ms\_mask\_present is 3, stereo\_info is arithmetic decoded. stereo\_info is two-bit flag per scalefactor band indicating the M/S coding or Intensity coding mode. If stereo\_info is not 0, M/S stereo or intensity stereo of AAC is done with these decoded data.

If pns data is present and the number of channel is 1, decoding process is as follows:

If the number of channel is 1 and pns data is present, noise flag of the scalefactor bands between **pns\_start\_sfb** to **max\_sfb** is arithmetic decoded. Perceptual noise substitution is done according to the decoded noise flag.

If pns data is present and the number of channel is 2, decoding process is as follows:

If ms\_mask\_present is 0, noise flag for pns is arithmetic decoded. Perceptual noise substitution of independent mode is done according to the decoded noise flag.

If ms\_mask\_present is 2, all ms\_used values are ones in this case. So, M/S stereo processing of AAC is done at all scalefactor band. However, there is no pns processing regardless of pns\_data\_present flag

If ms\_mask\_present is 1, 1 bit mask of max\_sfb bands of ms\_used is conveyed in this case. So, ms\_used is arithmetic decoded. M/S stereo processing of AAC is done according to the decoded ms\_used. However, there is no pns processing regardless of pns\_data\_present flag

If ms\_mask\_present is 3, stereo\_info is arithmetic decoded. If stereo\_info is 1 or 2, M/S stereo or intensity stereo processing of AAC is done with these decoded data and there is no pns processing. If stereo\_info is 3 and scalefactor band is smaller than pns\_start\_sfb, out\_of\_phase intensity stereo processing is done. If stereo\_info is 3 and scalefactor band is larger than or equal to pns\_start\_sfb, noise flag for pns is arithmetic decoded. And then if the both noise flags of two channel are 1, noise substitution mode is arithmetic decoded. The perceptual noise is substituted or out\_of\_phase intensity stereo processing is done according to the substitution mode. Otherwise, the perceptual noise is substituted only if noise flag is 1.

The detailed description of how to decode this side information will be given in subclause 4.6.4.3.

#### 4.5.2.6.2.2.10 Decoding of scalefactors

The spectral coefficients are divided into scalefactor bands that contain a multiple of 4 quantized spectral coefficients. Each scalefactor band has a scalefactor. For all scalefactors the difference to the maximum scalefactor value, **max\_scalefactor** is arithmetic-coded using the arithmetic model given in Table 4.A.30. The arithmetic model necessary for coding the differential scalefactors in the base layer is given as a 3-bit unsigned integer in the bitstream element, **base\_scf\_model**. The arithmetic model necessary for coding the differential scalefactors in the other enhancement layers is given as a 3-bit unsigned integer in the bitstream element, **enh\_scf\_model**. The maximum scalefactor value is given explicitly as a 8 bit PCM in the bitstream element **max\_scalefactor**.

The detailed description of how to decode this side information will be given in subclause 4.6.4.4.

#### 4.5.2.6.2.2.11 Bit-Sliced spectral data

In BSAC encoder, the absolute values of quantized spectral coefficients is mapped into a bit-sliced sequence. These sliced bits are the symbols of the arithmetic coding. Every sliced bits are binary arithmetic coded with the proper probability (arithmetic model) from the lowest-frequency coefficient to the highest-frequency coefficient of the scalability layer, starting the Most Significant Bit(MSB) plane and progressing to the Least Significant Bit(LSB)

plane. The arithmetic coding of the sign bits associated with non-zero coefficient follows that of the sliced bit when the sliced bit is 1 for the first time.

The probability value should be defined in order to arithmetic-code the symbols (the sliced bits). Binary probability table is made up of probability values of the symbol "0". First of all, probability table is selected using `cband_si` as shown Table 4.A.31. The probability value is selected among the several values in the selected table according to the context such as the remaining available bit size and the sliced bits of successive non-overlapping 4 spectral data.

For the case of multiple windows per block, the concatenated and possibly grouped and interleaved set of spectral coefficients is treated as a single set of coefficients that progress from low to high as described in subclause 4.5.2.6.2.6. This set of spectral coefficients needs to be de-interleaved after they are decoded. The set of bit-sliced sequence is divided into coding bands. The probability table index used for encoding the bit-sliced data within each coding band is included in the bitstream element `cband_si` and transmitted starting from the lowest frequency coding band and progressing to the highest frequency coding band. The spectral information for all scalefactor bands equal to or greater than `max_sfb` is set to zero.

#### 4.5.2.6.2.2.12 Decoding the sliced bits of the spectral data

The spectral bandwidth is increased in proportion to the scalability layer. So, the new spectral data is added to each layer. First of all, these new spectral data are coded in each layer (`bsac_layer_spectra()`). The coding process is continued until the data of each layer is not available or all the sliced bits of the new spectra are coded. The length of the available bitstream (`available_len[]`) is initialized at the beginning of each layer as described in subclause 4.5.2.6.2.5. The estimated length of the codeword (`est_cw_len`) to be decoded is calculated from the arithmetic decoding process as described in subclause 4.5.2.6.2.7. After the arithmetic decoding of a symbol, the length of the available bitstream should be updated by subtracting the estimated codeword length from it. We can detect whether the remaining bitstream of each layer is available or not by checking the array `available_len[]`.

From the lowest layer to the top layer, the new spectra are arithmetic-coded layer-by-layer in the above first process (`bsac_layer_spectra()`). Some sliced bits cannot be coded for lack of the codeword allocated to the layer. After the first coding process is finished, the current significances (`cur_snf`) are saved for the secondary coding processes (`bsac_lower_spectra()` and `bsac_higher_spectra()`). The sliced bits which remain uncoded is coded using the saved significances (`unc_snf`) in the secondary coding process.

If there remains the available codewords after the first coding, the next symbol to be decoded with these redundant codewords depends on whether the segmented binary arithmetic coding (SBA) mode is active.

In case of the non-SBA mode, the uncoded symbols of the lower spectra in the layers than the current layer are coded in the secondary coding process (`bsac_lower_spectra()`).

In case of the SBA mode for the error resilience, the next symbol is dependant upon whether the layer is the terminal layer of a segment or not. If the layer is not a terminal of the segment, the spectral data of the next layer (`bsac_layer_spectra(layer+1)`) should be decoded. That is to say, the redundant length of the layer is added to the available bitstream length (`available_len[layer+1]`) of the next layer in the first coding process.

If the layer is a terminal of the segment, the uncoded symbols of the lower spectra in the layers than the current layer are coded in the secondary coding process (`bsac_lower_spectra()`). The uncoded symbol of the spectra in the layers higher than the current layer are coded in the secondary coding process (`bsac_higher_spectra()`) if the codeword of the layer is available in spite of having coded the lower spectra. And the remaining symbols are continuously coded in the layers whose codeword is available, starting from the lowest layer and progressing to the top layer.

If there are the redundant bits after the secondary coding, the size of the redundant bits is added to the available bitstream length (`available_len[layer+1]`) of the next layer and the redundant bits are used in the first coding of the next layer.

#### 4.5.2.6.2.2.13 Reconstruction of the decoded sample from bit-sliced data

In order to reconstruct the spectral data, a bit-sliced sequence that has been decoded should be mapped into quantized spectral values. An arithmetic decoded symbol is a sliced bit. A decoded symbol is translated to the bit values of quantized spectral coefficients, as specified in the following pseudo C code:

```
snf = the significance of the symbol (the sliced bit) to be decoded;
sliced_bit[ch][g][i][snf] = the decoded symbol (the sliced bits of the quantized spectrum);
sample[ch][g][i] = buffer for quantized spectral coefficients to be reconstructed;
scaled_bit = sliced_bit[ch][g][i][snf] << (snf-1);
if (sample[ch][g][i] < 0)
    sample[ch][g][i] -= scaled_bit;
else
    sample[ch][g][i] += scaled_bit;
```

And if the sign bit of the decoded sample is 1, the decoded sample `sample[ch][g][i]` has the negative value as follows :

```
if (sample[ch][g][i] != 0) {
    if (sign_bit[ch][g][i] == 1) sample[ch][g][i] = -sample[ch][g][i];
}
```

#### 4.5.2.6.2.3 Windows and window sequences for BSAC

Quantization and coding is done in the frequency domain. For this purpose, the time signal is mapped into the frequency domain in the encoder. Depending on the signal, the coder may change the time/frequency resolution by using two different windows: `LONG_WINDOW` and `SHORT_WINDOW`. To switch between windows, the transition windows `LONG_START_WINDOW` and `LONG_STOP_WINDOW` are used. Refer to ISO/IEC 14496-3 General Audio Coding for more detailed information about the transform and the windows since BSAC has the same transform and windows with AAC.

#### 4.5.2.6.2.4 Scalefactor bands, grouping and coding bands for BSAC

Many tools of the AAC/BSAC decoder perform operations on groups of consecutive spectral values called scalefactor bands (abbreviation “sfb”). The width of the scalefactor bands is built in imitation of the critical bands of the human auditory system. For that reason the number of scalefactor bands in a spectrum and their width depend on the transform length and the sampling frequency. Refer to ISO/IEC 14496-3 General Audio Coding for more detailed information about the scalefactor bands and grouping because BSAC has the same process with AAC.

BSAC decoding tool performs operations on groups of consecutive spectral values called coding bands (abbreviation “cband”). To increase the efficiency of the noiseless coding, the width of the coding bands is fixed as 32 irrespective of the transform length and the sampling frequency. In case of sequences which contain `LONG_WINDOW`, 32 spectral data are simply grouped into a coding band. Since the spectral data within a group are interleaved in an ascending spectral order in case of `SHORT_WINDOW`, the interleaved spectral data are grouped into a coding band. Each spectral index within a group is mapped into a coding band with a mapping function,  $cband = \text{spectral\_index}/32$ .

Since scalefactor bands and coding bands are a basic element of the BSAC coding algorithm, some help variables and arrays are needed to describe the decoding process in all tools using scalefactor bands and coding bands. These help variables must be defined for BSAC decoding. These help variables depend on `sampling_frequency`, **window\_sequence**, **scalefactor\_grouping** and **max\_sfb** and must be built up for each `bsac_raw_data_block`. The pseudo code shown below describes

- how to determine the number of windows in a `window_sequence` *num\_windows*
- how to determine the number of window groups *num\_window\_groups*
- how to determine the number of windows in each group *window\_group\_length[g]*
- how to determine the total number of scalefactor window bands *num\_swb* for the actual window type
- how to determine *swb\_offset[g][swb]*, the offset of the first coefficient in scalefactor window band *swb* of the window actually used

A long transform window is always described as a window\_group containing a single window. Since the number of scalefactor bands and their width depend on the sampling frequency, the affected variables are indexed with sampling\_frequency\_index to select the appropriate table.

```

fs_index = sampling_frequency_index;
switch( window_sequence ) {
  case ONLY_LONG_SEQUENCE:
  case LONG_START_SEQUENCE:
  case LONG_STOP_SEQUENCE:
    num_windows = 1;
    num_window_groups = 1;
    window_group_length[num_window_groups-1] = 1;
    num_swb = num_swb_long_window[fs_index];
    for( sfb=0; sfb< max_sfb+1; sfb++ ) {
      swb_offset[0][sfb] = swb_offset_long_window[fs_index][sfb];
    }
    break;
  case EIGHT_SHORT_SEQUENCE:
    num_windows = 8;
    num_window_groups = 1;
    window_group_length[num_window_groups-1] = 1;
    num_swb = num_swb_short_window[fs_index];
    for( i=0; i< num_windows-1; i++ ) {
      if( bit_set(scale_factor_grouping,6-i) == 0 ) {
        num_window_groups += 1;
        window_group_length[num_window_groups-1] = 1;
      }
      else {
        window_group_length[num_window_groups-1] += 1;
      }
    }
    for( g = 0; g < num_window_groups; g++ )
      swb_offset[g][0] = 0;

    for( sfb = 0; sfb < max_sfb; sfb++ ) {
      for( g = 0; g < num_window_groups; g++ ) {
        swb_offset[g][sfb] = swb_offset_short_window[fs_index][sfb];
        swb_offset[g][sfb] = swb_offset[g][sfb] * window_group_length[g];
      }
    }
    break;
  default:
    break;
}

```

#### 4.5.2.6.2.5 BSAC fine grain scalability layer

BSAC provides a 1-kits/sec/ch fine grain scalability which has the layered bitstream, one BSAC base layer and various enhancement layers. BSAC base layer is made up of the general side information for all the fine grain layers, the specific side information for only the base layer and the audio data. BSAC enhancement layers contain the layer side information and the audio data.

BSAC scalable coding scheme has the scalable band-limit according to the fine grain layer. First of all, the base band-limit is set. The base band-limit depends on the signal to be encoded and is in the syntax element, **base\_band**. The actually limited spectral line is 4\*base\_band if the window sequence is SHORT\_WINDOW. Otherwise, the limited spectral line is 32\*base\_band. In order to provide the fine grain scalability, BSAC extends the band-limit according to the fine grain layer. The band limit of each layer depends on the base band-limit, the transform lengths 1024(960) and 128(120) and the sampling frequencies. The spectral band is extended more and more as the number of the enhancement layer is increased. So, the new spectral components are added to each layer.

Some help variables and arrays are needed to describe the bit-sliced decoding process of the side information and spectral data in each BSAC fine grain layer. These help variables depend on `sampling_frequency`, `layer`, `nch`, `frame_length`, `top_layer`, `window_sequence` and `max_sfb` and must be built up for each `bsac_layer_element`. The pseudo code shown below describes

- how to determine `slayer_size`, the number of the sub-layers which the base layer is split into.

```

slayer_size = 0;
for ( g = 0; g < num_window_groups; g++) {
  if (window_sequence == EIGHT_SHORT_SEQUENCE) {
    end_index[g] = base_band * 4 * window_group_length[g];
    if (fs==44100 || fs==48000) {
      if (end_index[g]%32>=16)
        end_index[g] = (int)(end_index[g]/32)*32 + 20;
      else if ( end_index[g]%32 >= 4)
        end_index[g] = (int)(end_index[g]/32)*32 + 8;
    }
    else if (fs==22050 || fs==24000 || fs==32000) end_index[g] = (int)(end_index[g]/16)*16;
    else if (fs==11025 || fs==12000 || fs==16000) end_index[g] = (int)(end_index[g]/32)*32;
    else end_index[g] = (int)(end_index[g]/64)*64;
    end_cband[g] = (end_index[g] + 31) / 32;
  }
  else
    end_cband[g] = base_band;
  slayer_size += end_cband[g];
}

```

- how to determine `layer_group[]`, the group index of the spectral components to be added newly in the scalability layer

```

layer = 0
for ( g = 0; g < num_window_groups; g++)
  for ( w = 0; w < window_group_length[g]; w++)
    layer_group[layer++] = g;
for (layer = slayer_size+8; layer < (top_layer+slayer_size); layer++)
  layer_group[layer] = layer_group[layer-8];

```

- how to determine `layer_end_index[]`, the end offset of the spectral components to be added newly in each scalability layer
- how to determine `layer_end_cband[]`, the end coding band to be added newly in each scalability layer
- how to determine `layer_start_index[]`, the start offset of the spectral components to be added newly in each scalability layer
- how to determine `layer_start_cband[]`, the start coding band to be added newly in each scalability layer

```

layer = 0;
for ( g = 0; g < num_window_groups; g++) {
  for (cband = 0; cband < end_cband[g]; cband++) {
    layer_start_cband[layer] = cband;
    end_cband[g] = layer_end_cband[layer] = cband+1;
    layer_start_index[layer] = cband * 32;
    end_index[g] = layer_end_index[layer++] = (cband+1) * 32;
  }
  if (window_sequence == EIGHT_SHORT_SEQUENCE)
    last_index[g] = swb_offset_short_window[max_sfb] * window_group_length[g];
  else
    last_index[g] = swb_offset_long_window[max_sfb];
}

for (layer = slayer_size; layer < (top_layer+slayer_size); layer++) {
  g = layer_group[layer];
  layer_start_index[layer] = end_index[g];
  if (fs==44100 || fs==48000) {

```



```

    if (end_index[g]%32==0)
        end_index[g] += 8;
    else
        end_index[g] += 12;
}
else if (fs==22050 || fs==24000 || fs==32000)
    end_index[g] += 16;
else if (fs==11025 || fs==12000 || fs==16000)
    end_index[g] += 32;
else
    end_index[g] += 64;
if ( end_index[g] > last_index[g] )
    end_index[g] = last_index[g];
layer_end_index[layer] = end_index[g];
layer_start_cband[g] = end_cband[g];
end_cband[g] = layer_end_cband[layer] = (end_index[g] + 31) / 32;
}

```

where, fs is the sampling frequency.

- how to determine *layer\_end\_sfb[]*, the end scalefactor band to be added newly in each scalability layer
- how to determine *layer\_start\_sfb[]*, the start scalefactor band to be added newly in each scalability layer

```

for (g = 0; g < num_window_groups; g++)
    end_sfb[g] = 0;
for (layer = 0; layer < (top_layer+slayer_size); layer++) {
    g = layer_group[layer];
    layer_start_sfb[layer] = end_sfb[g];
    layer_end_sfb[layer] = max_sfb;
    for (sfb = 0; sfb < max_sfb; sfb++) {
        if ( layer_end_index[layer] <= swb_offset_short_window[sfb] * window_group_length[g] ) {
            layer_end_sfb[layer] = sfb + 1;
            break;
        }
    }
    end_sfb[g] = layer_end_sfb[layer];
}
}

```

- how to determine *available\_len[i]*, the available maximum size of the bitstream of the *i*-th layer. If the arithmetic coding was initialized at the beginning of the layer, 1 should subtracted from *available\_len[i]* since the additional 1 bit is required at the arithmetic coding termination. The maximum length of the 0<sup>th</sup> coding band side information(*max\_cband0\_si\_len*) is defined as 11.

```

for (layer = 0; layer <(top_layer+slayer_size); layer++) {
    layer_si_maxlen[layer] = 0;
    for (cband = layer_start_cband[layer]; cband < layer_end_cband[layer]; cband++) {
        for (ch=0; ch <nch; ch++) {
            if (cband == 0)
                layer_si_maxlen[layer] += max_cband0_si_len;
            else
                layer_si_maxlen[layer] += max_cband_si_len[cband_si_type[ch]];
        }
    }
    for (sfb = layer_start_sfb[layer]; sfb < layer_end_sfb[layer]; sfb++)
        for (ch = 0; ch < nch; ch++)
            layer_si_maxlen[layer] += max_sfb_si_len[ch] + 5;
}
}

```

```

for (layer = slayer_size; layer <= (top_layer + slayer_size); layer++) {
    layer_bitrate = nch * ( (layer-slayer_size) * 1000 + 16000);
    layer_bit_offset[layer] = layer_bitrate * BLOCK_SIZE_SAMPLES_IN_FRAME;
    layer_bit_offset[layer] = (int)(layer_bit_offset[layer] / SAMPLING_FREQUENCY / 8) * 8;
}

```

```

    if (layer_bit_offset[layer] > frame_length*8)
        layer_bit_offset[layer] = frame_length*8;
}

for (layer = (top_layer + slayer_size - 1); layer >= slayer_size; layer--) {
    bit_offset = layer_bit_offset[layer+1] - layer_si_maxlen[layer]
    if ( bit_offset < layer_bit_offset[layer] )
        layer_bit_offset[layer] = bit_offset
}

for (layer = slayer_size - 1; slayer_size >= 0; slayer--)
    layer_bit_offset[layer] = layer_bit_offset[layer+1] - layer_si_maxlen[layer];
overflow_size = (header_length + 7) * 8 - layer_bit_offset[0];
layer_bit_offset[0] = (header_length + 7) * 8;
if (overflow_size > 0) {
    for ( layer = (top_layer+slayer_size-1); layer >= slayer_size; layer--) {
        layer_bit_size = layer_bit_offset[layer+1] - layer_bit_offset[layer];
        layer_bit_size -= layer_si_maxlen[layer];
        if (layer_bit_size >= overflow_size) {
            layer_bit_size = overflow_size;
            overflow_size = 0;
        }
        else
            overflow_size = overflow_size - layer_bit_size;
        for (m=1; m<=layer; m++)
            layer_bit_offset[m] += layer_bit_size;
        if (overflow_size<=0)
            break;
    }
}
else {
    underflow_size = -overflow_size;
    for (m=1; m < slayer_size; m++) {
        layer_bit_offset[m] = layer_bit_offset[m-1] + layer_si_maxlen[m-1];
        layer_bit_offset[m] += underflow_size / slayer_size;
        if (layer <= (underflow_size%slayer_size);
            layer_bit_offset[m] += 1;
    }
}
for (layer=0; layer <(top_layer+slayer_size); layer++)
    available_len[layer] = layer_bit_offset[layer+1] - layer_bit_offset[layer];

```

Some help variables and arrays are needed to describe the bit-sliced decoding process of the spectral values in each BSAC fine grain layer. `cur_snf[ch][g][i]` is initialized as the MSB plane (`MSBplane[ch][g][cband]`) allocated to the coding band `cband`, where we can get `MSBplane[[]]` from `cband_si[ch][g][cband]` using Table 4.A.31. And, we start the decoding of the bit-sliced data in each layer from the maximum significance, `maxsnf`.

These help variables and arrays must be built up for each `bsac_spectral_data()`. The pseudo code shown below describes

- how to initialize `cur_snf[[]][[]]`, the current significance of the spectra to be added newly due to the spectral band extension in each enhancement scalability layer.

```

/* set current snf */
g = layer_group[layer];
for(ch = 0; ch < nch; ch++) {
    for (i=layer_start_index[layer]; i<layer_end_index[layer]; i++) {
        cband = i/32;
        cur_snf[ch][g][i] = MSBplane[ch][g][cband];
    }
}

```

- how to determine `maxsnf`, the maximum significance of all vectors to be decoded.
- ```

maxsnf = 0;
for (g = start_g; g < end_g; g++)

```

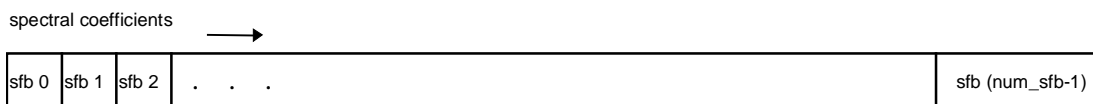
```
for(ch = 0; ch < nch; ch++) {
    for(i = start_index[g]; i < end_index[g]; i++)
        if (maxsnf < cur_snf[ch][g][i])
            maxsnf = cur_snf[ch][g][i];
}
```

- how to store *cur\_snf* for the secondary coding (*bsac\_lower\_spectra()* and *bsac\_higher\_spectra()*) after the sliced bits of the new spectra has been coded in each layer (*bsac\_layer\_spectra()*).

```
/* store current snf */
for(g=0; g<no_window_groups;g++)
for(ch = 0; ch < nch; ch++) {
    for (i=layer_start_index[layer]; i<layer_end_index[layer]; i++) {
        unc_snf[ch][g][i] = cur_snf[ch][g][i];
    }
}
```

**4.5.2.6.2.6 Order of spectral coefficients in spectral\_data**

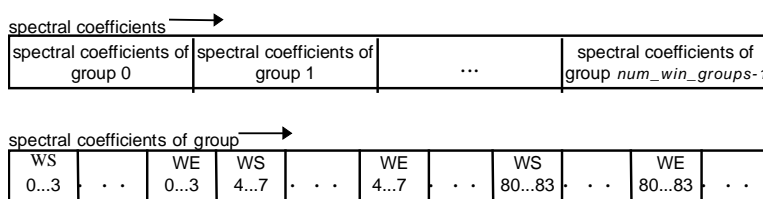
For ONLY\_LONG\_SEQUENCE windows (*num\_window\_groups* = 1, *window\_group\_length*[0] = 1) the spectral data is in ascending spectral order, as shown in Figure 4.18.



Order of scalefactor bands for ONLY\_LONG\_SEQUENCE

**Figure 4.18 – Order of scalefactor bands for ONLY\_LONG\_SEQUENCE**

For the EIGHT\_SHORT\_SEQUENCE window, each 4 spectral coefficients of blocks within each group are interleaved in ascending spectral order and the interleaved spectral coefficients are interleaved in ascending group number, as shown in Figure 4.19.



where, WS is the start window index and WE is the end window index of group *g*

Order of spectral data for EIGHT\_SHORT\_SEQUENCE

**Figure 4.19 – Order of spectral data for EIGHT\_SHORT\_SEQUENCE**

**4.5.2.6.2.7 Arithmetic coding procedure**

Arithmetic Coding consists of the following 2 steps:

- Initialization which is performed prior to the coding of the first symbol
- Coding of the symbol themselves.

#### 4.5.2.6.2.7.1 Registers, symbols and constants

Several registers, symbols and constants are defined to describe the arithmetic decoder.

- `half[]` : 32-bit fixed point array equal to  $\frac{1}{2}$
- `range`: 32-bit fixed point register. Contains the range of the interval.
- `value`: 32-bit fixed point register. Contains the value of the arithmetic code.
- `est_cw_len`: 16-bit fixed point register. Contains the estimated length of the arithmetic codeword to be decoded.
- `p0`: 16-bit fixed point register (Upper 6 MSBs are available, Other LSBs are 0). Probability of the “0” symbol.
- `p1`: 16-bit fixed point register (Upper 6 MSBs are available, Other LSBs are 0). Probability of the “1” symbol.
- `cum_freq` : 16-bit fixed point registers. Cumulative Probabilities of the symbols.

#### 4.5.2.6.2.7.2 Initialization

The bitstreams of each segment are read in the buffer of each segment. And 32-bit zero is concatenated to the buffer of each segment. If the segmented arithmetic coding is not, all the bitstreams of a frame is a segment and the zero stuffing is used. See subclause 4.6.4.6.3 for the detailed description of the segment.

The register *value* is set to 0, *range* to 1 and *est\_cw\_len* to 30. Using these initialized registers, the 30 bits are read in register *value* and registers are updated when the first symbol is decoded.

#### 4.5.2.6.2.7.3 Decoding a symbol

Arithmetic decoding procedure varies on the symbol to be decoded. If the symbol is the sliced bit of the spectral data, the binary arithmetic decoding is used. Otherwise, the general arithmetic decoding is used.

When a symbol is binary arithmetic-decoded, the probability *p0* of the “0” symbol is provided according to the context computed properly and using the probability table. *p0* uses a 6-bit fixed-point number representation. Since the decoder is binary, the probability of the “1” symbol is defined to be 1 minus the probability of the “0” symbol, i.e.  $p1 = 1 - p0$ .

When a symbol is arithmetic-decoded, the cumulative probability values of multiple symbols are provided. The probability values are regarded as the arithmetic model. The arithmetic model for decoding a symbol is given in the bitstream elements. For example, arithmetic models of scalefactor and *cband\_si* are given in the bitstream elements, **base\_scf\_model**, **enh\_scf\_model** and **cband\_si\_type**. Each value of the arithmetic model uses a 14-bit fixed-point representation.

#### 4.5.2.6.2.7.4 Software

```
unsigned long half[16] =
{
    0x20000000, 0x10000000, 0x08000000, 0x04000000,
    0x02000000, 0x01000000, 0x00800000, 0x00400000,
    0x00200000, 0x00100000, 0x00080000, 0x00040000,
    0x00020000, 0x00010000, 0x00008000, 0x00004000
};
/* Initialize the Parameters of the Arithmetic Decoder */
void initArDecode()
{
    value = 0;
    range = 1;
    est_cw_len = 30;
}
/* GENEARL ARITHMETIC DECODE */
int decode_symbol (buf_idx, cum_freq, symbol)
int buf_idx;      /* buffer index to save the arithmetic code word */
int cum_freq[];  /* Cumulative symbol frequencies */
int *symbol;      /* Symbol decoded */
{
    if (est_cw_len) {
```

```

    range = (range<<est_cw_len);
    value = (value<<est_cw_len) | readBits(buf_idx, est_cw_len);
    /* read bitstream from the buffer */
}

range >>= 14;
cum = value/range;      /* Find cum freq */

/* Find symbol */
for (sym=0; cum_freq[sym]>cum; sym++);
*symbol = sym;

/* Narrow the code region to that allotted to this symbol. */
value -= (range * cum_freq[sym]);

if (sym > 0) {
    range = range * (cum_freq[sym-1]-cum_freq[sym]);
}
else {
    range = range * (16384-cum_freq[sym]);
}

for (est_cw_len = 0; range < half[est_cw_len]; est_cw_len++);
return est_cw_len;
}
/* BINARY ARITHMETIC-DECODE THE NEXT SYMBOL. */
int decode_symbol2 (buf_idx, freq0, symbol)
int buf_idx; /* buffer index to save the arithmetic code word */
int p0; /* Normalized probability of symbol 0 */
int *symbol; /* Symbol decoded */
{
    if (est_cw_len) {
        range = (range<<est_cw_len);
        value = (value<<est_cw_len) | readBits(buf_idx, est_cw_len);
        /* read bitstream from the buffer */
    }

    range >>= 14;

    /* Find symbol */
    if ( (p0 * range) <= value ) {
        *symbol = 1;

        /* Narrow the code region to that allotted to this symbol. */
        value -= range * p0;
        p1 = 16384 - p0;
        range = range * p1;
    }
    else {
        *symbol = 0;

        /* Narrow the code region to that allotted to this symbol. */
        range = range * p0;
    }

    for(est_cw_len=0; range<half[est_cw_len]; est_cw_len++);
    return est_cw_len;
}

```

#### 4.5.2.6.3 Error sensitivity category assignment

BSAC has the layered structure where the syntax is arranged in order of importance in order to support the fine grain scalability and error resilience. Therefore the BSAC syntax can be channel coded effectively without the bitstream reordering for advanced channel coding techniques like unequal error protection (UEP) since the error resilient syntax are included in the subclause 4.4.2.6. However, error sensitivity categories (ESC) of the bitstream

elements should be defined for advanced channel coding. The bitstream element can be classified into the error sensitivity categories depending upon its error sensitivity as follows:

**Table 4.69 – BSAC error sensitivity category assignment**

| Category | bitstream elements                               | description                                    |
|----------|--------------------------------------------------|------------------------------------------------|
| 0        | frame_length, bsac_header() and general_header() | commonly used side information                 |
| 1        | bsac_layer_element(0)                            | BSAC base layer except common side             |
| 2        | bsac_layer_element(1) –                          | 1 <sup>st</sup> quarternary enhancement layers |
| 3        | bsac_layer_element(top_layer/4+1) –              | 2 <sup>nd</sup> quarternary enhancement layers |
| 4        | bsac_layer_element(top_layer/2+1) ~              | 3 <sup>rd</sup> quarternary enhancement layers |
| 5        | basc_layer_element(top_layer*3/4+1) ~            | 4 <sup>th</sup> quarternary enhancement layers |

The lower category indicates the class with the higher error sensitivity, whereas the higher category indicates the class with the lower sensitivity.

#### 4.5.2.7 Dynamic Range Control (DRC)

##### 4.5.2.7.1 Definitions

|                                      |                                                                                                                                                                                                                                                                                                                                                                                                                           |
|--------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>pce_tag_present</b>               | One bit indicating that program element tag is present;                                                                                                                                                                                                                                                                                                                                                                   |
| <b>pce_instance_tag</b>              | Tag field that indicates with which program the dynamic range information is associated                                                                                                                                                                                                                                                                                                                                   |
| <b>drc_tag_reserved_bits</b>         | Reserved                                                                                                                                                                                                                                                                                                                                                                                                                  |
| <b>excluded_chns_present</b>         | One bit indicating that excluded channels are present                                                                                                                                                                                                                                                                                                                                                                     |
| <b>drc_bands_present</b>             | One bit indicating that DRC multi-band information is present                                                                                                                                                                                                                                                                                                                                                             |
| <b>drc_band_incr</b>                 | Number of DRC bands greater than 1 having DRC information                                                                                                                                                                                                                                                                                                                                                                 |
| <b>drc_bands_reserved_bits</b>       | Reserved                                                                                                                                                                                                                                                                                                                                                                                                                  |
| <b>drc_band_top[i]</b>               | Indicates top of i-th DRC band in units of 4 spectral lines. If drc_band_top[i]=k, then the index (w.r.t zero) of the highest spectral coefficient (w.r.t zero) that is in the i-th DRC band is = $k/4-1+3$ . In case of an EIGHT_SHORT_SEQUENCE window_sequence the index is interpreted as pointing into the concatenated array of $8*128$ (unde-interleaved) frequency points corresponding to the 8 short transforms. |
| <b>prog_ref_level_present</b>        | One bit indicating that reference level is present                                                                                                                                                                                                                                                                                                                                                                        |
| <b>pce_tag_present</b>               | One bit indicating that program element tag is present                                                                                                                                                                                                                                                                                                                                                                    |
| <b>excluded_chns_present</b>         | One bit indicating that excluded channels are present                                                                                                                                                                                                                                                                                                                                                                     |
| <b>drc_bands_present</b>             | One bit indicating that DRC multi-band information is present                                                                                                                                                                                                                                                                                                                                                             |
| <b>prog_ref_level</b>                | Reference level. A measure of long-term program audio level for all channels combined.                                                                                                                                                                                                                                                                                                                                    |
| <b>prog_ref_level_reserved_bits</b>  | Reserved                                                                                                                                                                                                                                                                                                                                                                                                                  |
| <b>pce_instance_tag</b>              | Tag field that indicates with which program the dynamic range information is associated                                                                                                                                                                                                                                                                                                                                   |
| <b>drc_tag_reserved_bits</b>         | Reserved                                                                                                                                                                                                                                                                                                                                                                                                                  |
| <b>exclude_mask[ i ]</b>             | Boolean array indicating that a SCE, CPE or LFE is excluded from dynamic range control. Mask information remains in effect until a new mask is transmitted.                                                                                                                                                                                                                                                               |
| <b>additional_excluded_chns[ i ]</b> | One bit indicating that additional excluded channels are present                                                                                                                                                                                                                                                                                                                                                          |
| <b>dyn_rng_sgn</b>                   | One bit indicating the sign of dyn_rng_ctl (0 if positive, 1 if negative)                                                                                                                                                                                                                                                                                                                                                 |
| <b>drc_band_incr</b>                 | Number of DRC bands greater than 1 having DRC information                                                                                                                                                                                                                                                                                                                                                                 |
| <b>drc_bands_reserved</b>            | Reserved                                                                                                                                                                                                                                                                                                                                                                                                                  |
| <b>drc_band_top[i]</b>               | Indicates top of ith DRC band in units of 4 spectral lines. If drc_band_top[i]=k, the index of the highest spectral coefficient (w.r.t zero) that in i-th DRC band is = $k/4-1$ . In case of an EIGHT_SHORT_SEQUENCE window_sequence the index is interpreted as pointing into the concatenated array of $8*128$ (un-interleaved) frequency points corresponding to the 8 short transforms.                               |
| <b>dyn_rng_sgn[i]</b>                | Dynamic range control sign information. One bit indicating the sign of dyn_rng_ctl (0 if positive, 1 if negative)                                                                                                                                                                                                                                                                                                         |
| <b>dyn_rng_ctl[i]</b>                | Dynamic range control magnitude information                                                                                                                                                                                                                                                                                                                                                                               |

- exclude\_mask[ i ]** Boolean array indicating the audio channels of a program that are excluded from DRC processing using this DRC information.
- exclude\_mask[ i ]** Boolean array indicating that one channel of a SCE, CPE, or LFE or CCE is excluded from dynamic range control. Mask information remains in effect until a new mask is transmitted.
- additional\_excluded\_chns[ i ]** One bit indicating that additional excluded channels are present

#### 4.5.2.7.2 Decoding process:

The transport of the DRC information does not involve the MPEG-4 System layer but is handled completely within the GA bitstream elements instead. Furthermore, the dynamic range control processing is carried out within the GA decoder. No DRC related information is passed on to a subsequent audio compositor.

Advisory: Interactions between DRC and the audio compositor (see ISO/IEC 14496-1) are possible. If a mix between several audio objects in the audio compositor is intended, DRC has to be applied with care.

**prog\_ref\_level\_present** indicates that **prog\_ref\_level** is being transmitted. This permits **prog\_ref\_level** to be sent as infrequently as desired (e.g. once), although periodic transmission would permit break-in.

**prog\_ref\_level** is quantized in 0.25 dB steps using 7 bits, and therefore has a range of approximately 32 dB. It indicates program level relative to full scale (i.e. dB below full scale), and is reconstructed as:

$$level = 32767 \cdot 2^{\frac{-prog\_ref\_level}{24}}$$

and where „full scale level,, is 32767 (prog\_ref\_level equal to 0).

**pce\_tag\_present** indicates that **pce\_instance\_tag** is being transmitted. This permits **pce\_instance\_tag** to be sent as infrequently as desired (e.g. once), although periodic transmission would permit break-in.

**pce\_instance\_tag** indicates with which program the dynamic range information is associated. If this is not present then the default program is indicated. Since each AAC bitstream typically has just one program, this would be the most common mode. Each program in a multi-program bitstream would send its dynamic range information in a distinct `extension_payload()` of the `fill_element()`. In the multiple program case, the **pce\_instance\_tag** would always have to be signaled.

The **drc\_tag\_reserved\_bits** fill out the optional fields to an integral number of bytes in length.

The **excluded\_chns\_present** bit indicates that channels that are to be *excluded* from dynamic range processing will be signaled immediately following this bit. The excluded channel mask information must be transmitted in each frame where channels are excluded. The following ordering principles are used to assign the `exclude_mask` to symmetric channel outputs:

- If a PCE is present (explicit speaker mapping), the **exclude\_mask** bits correspond to the relevant audio channels in the SCE, CPE, CCE and LFE syntax elements (SCE, CPE, LFE) in the order of their appearance in the PCE. In the case of a CPE, the first transmitted mask bit corresponds to the first channel in the CPE, the second transmitted mask bit to the second channel. In the case of a CCE, a mask bit is transmitted only if the coupling channel is specified to be an independently switched coupling channel.
- For the case of an implicit speaker mapping (no PCE present), the **exclude\_mask** bits correspond to the audio channels in therelevant SCE, CPE and LFE syntax elements (SCE, CPE, LFE) in the order of their appearance in the bitstream, followed by the audio channels in the CCE syntax elements in the order of their appearance in the bitstream. In the case of a CPE, the first transmitted mask bit corresponds to the first channel in the CPE, the second transmitted mask bit to the second channel. In the case of CCE, a mask bit is transmitted only if the coupling channel is specified to be an independently switched coupling channel.

**drc\_band\_incr** is the number of bands greater than one if there is multi-band DRC information.

**dyn\_rng\_ctl** is quantized in 0.25 dB steps using a 7-bit unsigned integer, and therefore, in association with **dyn\_rng\_sgn**, has a range of +/-31.75 dB. It is interpreted as a gain value that shall be applied to the decoded audio output samples of the current frame.

The range supported by the dynamic range information is summarized in the following table:

**Table 4.70 – Dynamic range information**

| Field                                     | bits    | steps   | stepsize, dB | range, dB |
|-------------------------------------------|---------|---------|--------------|-----------|
| <b>prog_ref_level</b>                     | 7       | 128     | 0.25         | 31.75     |
| <b>dyn_rng_sgn</b> and <b>dyn_rng_ctl</b> | 1 and 7 | +/- 127 | 0.25         | +/- 31.75 |

The dynamic range control process is applied to the spectral data `spec[i]` of one frame immediately before the synthesis filterbank. In case of an `EIGHT_SHORT_SEQUENCE` window\_sequence the index `i` is interpreted as pointing into the concatenated array of  $8 \times 128$  (de-interleaved) frequency points corresponding to the 8 short transforms.

The following pseudo code is for illustrative purposes only, showing one method for applying one set of dynamic range control information to a frame of a target audio channel. The constants `ctrl1` and `ctrl2` are compression constants (typically between 0 and 1, zero meaning no compression) that may optionally be used to scale the dynamic range compression characteristics for positive and negative level adjustment levels greater than or less than the program reference level, respectively. The constant `target_level` describes the output level desired by the user, expressed in the same scaling as `prog_ref_level`.

The process is applied to the spectral data `spec[i]` of one frame immediately before the synthesis filterbank. In case of an `EIGHT_SHORT_SEQUENCE` window\_sequence the index `i` is interpreted as pointing into the concatenated array of  $8 \times 128$  (non-interleaved) frequency points corresponding to the 8 short transforms.

```

bottom = 0;
drc_num_bands = 1;
if (drc_bands_present)
    drc_num_bands += drc_band_incr;
if (drc_num_bands == 1)
    drc_band_top[0] = 1024/4 - 1;
for (bd=0; bd < drc_num_bands; bd++) {
    top = 4 * (drc_band_top[bd] + 1);

    /* Decode DRC gain factor */
    if (dyn_rng_sgn[bd])
        factor = 2^(-ctrl1*dyn_rng_ctl[bd]/24); /* compress */
    else
        factor = 2^(ctrl2*dyn_rng_ctl[bd]/24); /* boost */

    /* Level alignment between different programs (if desired) */
    /* If program reference normalization is done in the digital domain, modify
    * factor to perform normalization.
    * prog_ref_level can alternatively be passed to the system for modification
    * of the level in the analog domain. Analog level modification avoids problems
    * with reduced DAC SNR (if signal is attenuated) or clipping (if signal is boosted)
    */
    factor *= 0.5^((target_level-prog_ref_level)/24);

    /* Apply gain factor */
    for (i=bottom; i<top; i++)
        spec[i] *= factor;
    bottom = top;
}

```

Note the relation between dynamic range control and coupling channels:



- Dependently switched coupling channels are always coupled onto their target channels as spectral coefficients, i.e. prior to the DRC and synthesis filterbank processing of these channels. This means that the output of a dependently switched coupling channel's signal that couples onto to a specific target channel will undergo the DRC processing of this target channel.
- Since independently switched coupling channels couple to their target channels in the time domain, each independently switched coupling channel will include its own DRC and subsequent synthesis filterbank processing separate from its target channels. This permits the independently switched coupling channel to have distinct DRC processing if desired.

### 4.5.2.7.3 Persistence of DRC information:

At the beginning of a stream, all DRC information for all channels is assumed to be set to its default value: program reference level equal to the decoder's target reference level, one DRC band, with no DRC gain modification for that band. Unless this data is specifically overwritten, this remains in effect.

There are two cases for the persistence of DRC information that has been transmitted:

- The program reference level is per audio program, and persists until a new value is transmitted, at which point the new data overwrites the old and takes effect that frame. (It may be appropriate to send this value periodically to allow bitstream break-in.)
- Other DRC information persists on a per-channel basis. Note that if a channel is excluded via the appropriate **exclude\_mask[]** bit, then effectively no information is transmitted for that channel in that call to `dynamic_range_info()`. The excluded channel mask information must be transmitted in each frame where channels are excluded.

The rules for retaining per-channel DRC information are as follows:

- If there is no DRC information in a given frame for a given channel, use the information that was used in the previous frame. (This means that one adjustment can hold for a long time, although it may be appropriate to transmit the DRC information periodically to permit break-in.)

If any DRC information for this channel appears in the current frame, the following sequence occurs: first, overwrite all per-channel DRC information for that channel with the default values (one DRC band, with no DRC gain modification for that band), then overwrite any per-channel DRC information with the transmitted values.

### 4.5.2.7.4 Usage of DRC with the audio object type AAC scalable

If DRC is used with the AAC scalable audio object type, the following additional restrictions and information apply:

1. The field **pce\_tag\_present** must be '0' (no reference to a PCE).
2. The field **excluded\_chns\_present** must be '0' (common control of all audio channels).
3. DRC information may be transmitted in several layers of the scalable audio object. The DRC information to be used for the DRC processing is the one carried in the highest layer available to the decoder.

## 4.5.3 Buffer requirements

### 4.5.3.1 Minimum decoder input buffer

The following rules are used to calculate the maximum number of bits in the input buffer both for the bitstream as a whole, for any given program, or for any given SCE/CPE/CCE:

The input buffer size is 6144 bits per SCE or independently switched CCE, plus 12288 bits per CPE. Both, the total buffer size, and the individual buffer size are limited, so that the buffering limit can be calculated for either the entire bitstream, any entire program, or the individual audio elements permitting the decoder to break a multichannel bitstream into separate mono and stereo bitstreams which are decoded by separate mono and stereo decoders, respectively. Although the 6144 bit/CCE must be obeyed for dependent CCEs as well, any bits for dependent CCEs must be supplied from the total buffer requirements based on the independent CCEs, SCEs, and CPEs.

For the audio object type AAC scalable the same restrictions apply, however, here they apply for the combined size of the input buffers of all ASME and ASEE. This means, if a mono program is encoded a buffer size of

6144=1024\*6 bits is required, and for a stereo program a total buffer size of 12288 bits is available. In the case of the scalable configurations with both, mono- and stereo-layers, the maximum buffer size for all mono layers is 6144 bits. The total buffer size for all layers is 12288 bits.

#### 4.5.3.2 Bit reservoir

The bit reservoir is controlled at the encoder. The maximum bit reservoir in the encoder depends on the number of channels and the mean bitrate. The maximum bit reservoir size for constant rate channels can be calculated by subtracting the mean number of bits per block from the minimum decoder input buffer size. For example, at 96 kbit/s for a stereo signal at 48 kHz sampling frequency the maximum bit reservoir size is 12288 bit- (96000 bit/s / 48000 1/s \* frameLength) = 10240 bit in case of fixed bitrate operation. For variable bitrate channels the encoder must operate in a way that the input buffer requirements do not exceed the minimum decoder input buffer.

#### 4.5.3.3 Maximum bit rate

The maximum bit rate depends on the audio sampling rate. The maximum bit rate per channel can be calculated based on the minimum input buffer according to the formula:

$$\frac{6144 \cdot \frac{\text{bit}}{\text{block}}}{\text{frameLength} \cdot \frac{\text{samples}}{\text{block}}} \cdot \text{sampling\_frequency}$$

So, for example, this leads to the following maximum bit rates:

| sampling frequency | maximum bit rate / channel |
|--------------------|----------------------------|
| 48 kHz             | 288 kbit/sec               |
| 44.1 kHz           | 264.6 kbit/sec             |
| 32 kHz             | 192 kbit/sec               |

#### 4.5.4 Tables

Table 4.71 – Transform windows

| Window            | num_swb | #coeffs  | looks like |
|-------------------|---------|----------|------------|
| LONG_WINDOW       | 49      | 1024/960 |            |
| SHORT_WINDOW      | 14      | 128/120  |            |
| LONG_START_WINDOW | 49      | 1024/960 |            |
| LONG_STOP_WINDOW  | 49      | 1024/960 |            |

Table 4.72 – Window Sequences




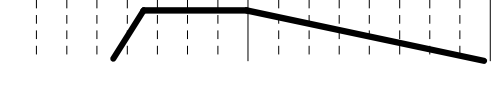
| Value | window_sequence                            | num_windows | looks like                                                                         |
|-------|--------------------------------------------|-------------|------------------------------------------------------------------------------------|
| 0     | ONLY_LONG_SEQUENCE<br>= LONG_WINDOW        | 1           |  |
| 1     | LONG_START_SEQUENCE<br>= LONG_START_WINDOW | 1           |  |
| 2     | EIGHT_SHORT_SEQUENCE<br>= 8 * SHORT_WINDOW | 8           |  |
| 3     | LONG_STOP_SEQUENCE<br>= LONG_STOP_WINDOW   | 1           |  |

Table 4.73 – scalefactor bands for a window length of 2048 and 1920 (values for 1920 in brackets) for LONG\_WINDOW, LONG\_START\_WINDOW, LONG\_STOP\_WINDOW at 44.1 and 48 kHz

| fs [kHz]            | 44.1,48                |
|---------------------|------------------------|
| num_swb_long_window | 49                     |
| swb                 | swb_offset_long_window |
| 0                   | 0                      |
| 1                   | 4                      |
| 2                   | 8                      |
| 3                   | 12                     |
| 4                   | 16                     |
| 5                   | 20                     |
| 6                   | 24                     |
| 7                   | 28                     |
| 8                   | 32                     |
| 9                   | 36                     |
| 10                  | 40                     |
| 11                  | 48                     |
| 12                  | 56                     |
| 13                  | 64                     |
| 14                  | 72                     |
| 15                  | 80                     |
| 16                  | 88                     |
| 17                  | 96                     |
| 18                  | 108                    |
| 19                  | 120                    |
| 20                  | 132                    |
| 21                  | 144                    |
| 22                  | 160                    |
| 23                  | 176                    |
| 24                  | 196                    |

| swb | swb_offset_long_window |
|-----|------------------------|
| 25  | 216                    |
| 26  | 240                    |
| 27  | 264                    |
| 28  | 292                    |
| 29  | 320                    |
| 30  | 352                    |
| 31  | 384                    |
| 32  | 416                    |
| 33  | 448                    |
| 34  | 480                    |
| 35  | 512                    |
| 36  | 544                    |
| 37  | 576                    |
| 38  | 608                    |
| 39  | 640                    |
| 40  | 672                    |
| 41  | 704                    |
| 42  | 736                    |
| 43  | 768                    |
| 44  | 800                    |
| 45  | 832                    |
| 46  | 864                    |
| 47  | 896                    |
| 48  | 928                    |
|     | 1024 (960)             |

**Table 4.74 – scalefactor bands for a window length of 256 and 240 (values for 240 in brackets) for SHORT\_WINDOW at 32, 44.1 and 48 kHz**

| fs [kHz]             | 32,44.1,48 | swb | swb_offset_short<br>window |
|----------------------|------------|-----|----------------------------|
| num_swb_short_window | 14         |     |                            |
|                      |            | 0   | 0                          |
|                      |            | 1   | 4                          |
|                      |            | 2   | 8                          |
|                      |            | 3   | 12                         |
|                      |            | 4   | 16                         |
|                      |            | 5   | 20                         |
|                      |            | 6   | 28                         |
|                      |            | 7   | 36                         |
|                      |            | 8   | 44                         |
|                      |            | 9   | 56                         |
|                      |            | 10  | 68                         |
|                      |            | 11  | 80                         |
|                      |            | 12  | 96                         |
|                      |            | 13  | 112                        |
|                      |            |     | 128 (120)                  |

**Table 4.75 – scalefactor bands for a window length of 2048 and 1920 (values for 1920 in brackets) for LONG\_WINDOW, LONG\_START\_WINDOW, LONG\_STOP\_WINDOW at 32 kHz**

| fs [kHz]            | 32 | swb | swb_offset_long<br>window |
|---------------------|----|-----|---------------------------|
| num_swb_long_window | 51 |     |                           |
|                     |    | 0   | 0                         |
|                     |    | 1   | 4                         |
|                     |    | 2   | 8                         |
|                     |    | 3   | 12                        |
|                     |    | 4   | 16                        |
|                     |    | 5   | 20                        |
|                     |    | 6   | 24                        |
|                     |    | 7   | 28                        |
|                     |    | 8   | 32                        |
|                     |    | 9   | 36                        |
|                     |    | 10  | 40                        |
|                     |    | 11  | 48                        |
|                     |    | 12  | 56                        |
|                     |    | 13  | 64                        |
|                     |    | 14  | 72                        |
|                     |    | 15  | 80                        |
|                     |    | 16  | 88                        |
|                     |    | 17  | 96                        |
|                     |    | 18  | 108                       |
|                     |    | 19  | 120                       |
|                     |    | 20  | 132                       |
|                     |    | 21  | 144                       |
|                     |    | 22  | 160                       |
|                     |    | 23  | 176                       |
|                     |    | 24  | 196                       |
|                     |    | 25  | 216                       |
|                     |    | 26  | 240                       |
|                     |    | 27  | 264                       |
|                     |    | 28  | 292                       |
|                     |    | 29  | 320                       |
|                     |    | 30  | 352                       |
|                     |    | 31  | 384                       |
|                     |    | 32  | 416                       |
|                     |    | 33  | 448                       |
|                     |    | 34  | 480                       |
|                     |    | 35  | 512                       |
|                     |    | 36  | 544                       |
|                     |    | 37  | 576                       |
|                     |    | 38  | 608                       |
|                     |    | 39  | 640                       |
|                     |    | 40  | 672                       |
|                     |    | 41  | 704                       |
|                     |    | 42  | 736                       |
|                     |    | 43  | 768                       |
|                     |    | 44  | 800                       |
|                     |    | 45  | 832                       |
|                     |    | 46  | 864                       |
|                     |    | 47  | 896                       |
|                     |    | 48  | 928                       |
|                     |    | 49  | 960                       |
|                     |    | 50  | 992 (960)                 |
|                     |    |     | 1024 (960)                |

**Table 4.76 – scalefactor bands for a window length of 2048 and 1920 (values for 1920 in brackets) for LONG\_WINDOW, LONG\_START\_WINDOW, LONG\_STOP\_WINDOW at 8 kHz**

| fs [kHz]            | 8                      |     |                        |
|---------------------|------------------------|-----|------------------------|
| num_swb_long_window | 40                     |     |                        |
| swb                 | swb_offset_long_window | swb | Swb_offset_long_window |
| 0                   | 0                      | 21  | 288                    |
| 1                   | 12                     | 22  | 308                    |
| 2                   | 24                     | 23  | 328                    |
| 3                   | 36                     | 24  | 348                    |
| 4                   | 48                     | 25  | 372                    |
| 5                   | 60                     | 26  | 396                    |
| 6                   | 72                     | 27  | 420                    |
| 7                   | 84                     | 28  | 448                    |
| 8                   | 96                     | 29  | 476                    |
| 9                   | 108                    | 30  | 508                    |
| 10                  | 120                    | 31  | 544                    |
| 11                  | 132                    | 32  | 580                    |
| 12                  | 144                    | 33  | 620                    |
| 13                  | 156                    | 34  | 664                    |
| 14                  | 172                    | 35  | 712                    |
| 15                  | 188                    | 36  | 764                    |
| 16                  | 204                    | 37  | 820                    |
| 17                  | 220                    | 38  | 880                    |
| 18                  | 236                    | 39  | 944                    |
| 19                  | 252                    |     | 1024 (960)             |
| 20                  | 268                    |     |                        |

**Table 4.77 – scalefactor bands for a window length of 256 and 240 (values for 240 in brackets) for SHORT\_WINDOW at 8 kHz**

| fs [kHz]             | 8                       |     |                         |
|----------------------|-------------------------|-----|-------------------------|
| num_swb_short_window | 15                      |     |                         |
| swb                  | swb_offset_short_window | swb | swb_offset_short_window |
| 0                    | 0                       | 8   | 36                      |
| 1                    | 4                       | 9   | 44                      |
| 2                    | 8                       | 10  | 52                      |
| 3                    | 12                      | 11  | 60                      |
| 4                    | 16                      | 12  | 72                      |
| 5                    | 20                      | 13  | 88                      |
| 6                    | 24                      | 14  | 108                     |
| 7                    | 28                      |     | 128 (120)               |

**Table 4.78 – scalefactor bands for a window length of 2048 and 1920 (values for 1920 in brackets) for LONG\_WINDOW, LONG\_START\_WINDOW, LONG\_STOP\_WINDOW at 11.025, 12 and 16 kHz**

| fs [kHz]            | 11.025, 12, 16         |     |                        |
|---------------------|------------------------|-----|------------------------|
| num_swb_long_window | 43                     |     |                        |
| swb                 | swb_offset_long_window | swb | swb_offset_long_window |
| 0                   | 0                      | 22  | 228                    |
| 1                   | 8                      | 23  | 244                    |
| 2                   | 16                     | 24  | 260                    |
| 3                   | 24                     | 25  | 280                    |
| 4                   | 32                     | 26  | 300                    |
| 5                   | 40                     | 27  | 320                    |

|    |     |    |            |
|----|-----|----|------------|
| 6  | 48  | 28 | 344        |
| 7  | 56  | 29 | 368        |
| 8  | 64  | 30 | 396        |
| 9  | 72  | 31 | 424        |
| 10 | 80  | 32 | 456        |
| 11 | 88  | 33 | 492        |
| 12 | 100 | 34 | 532        |
| 13 | 112 | 35 | 572        |
| 14 | 124 | 36 | 616        |
| 15 | 136 | 37 | 664        |
| 16 | 148 | 38 | 716        |
| 17 | 160 | 39 | 772        |
| 18 | 172 | 40 | 832        |
| 19 | 184 | 41 | 896        |
| 20 | 196 | 42 | 960        |
| 21 | 212 |    | 1024 (960) |

**Table 4.79 – scalefactor bands for a window length of 256 and 240 (values for 240 in brackets) for SHORT\_WINDOW at 11.025, 12 and 16 kHz**

| fs [kHz]             | 11.025, 12, 16 | swb | swb_offset_short_window |
|----------------------|----------------|-----|-------------------------|
| num_swb_short_window | 15             |     |                         |
| 0                    | 0              | 8   | 32                      |
| 1                    | 4              | 9   | 40                      |
| 2                    | 8              | 10  | 48                      |
| 3                    | 12             | 11  | 60                      |
| 4                    | 16             | 12  | 72                      |
| 5                    | 20             | 13  | 88                      |
| 6                    | 24             | 14  | 108                     |
| 7                    | 28             |     | 128 (120)               |

**Table 4.80 – scalefactor bands for a window length of 2048 and 1920 (values for 1920 in brackets) for LONG\_WINDOW, LONG\_START\_WINDOW, LONG\_STOP\_WINDOW at 22.05 and 24 kHz**

| fs [kHz]            | 22.05 and 24 | swb | swb_offset_long_window |
|---------------------|--------------|-----|------------------------|
| num_swb_long_window | 47           |     |                        |
| 0                   | 0            | 24  | 160                    |
| 1                   | 4            | 25  | 172                    |
| 2                   | 8            | 26  | 188                    |
| 3                   | 12           | 27  | 204                    |
| 4                   | 16           | 28  | 220                    |
| 5                   | 20           | 29  | 240                    |
| 6                   | 24           | 30  | 260                    |
| 7                   | 28           | 31  | 284                    |
| 8                   | 32           | 32  | 308                    |
| 9                   | 36           | 33  | 336                    |
| 10                  | 40           | 34  | 364                    |
| 11                  | 44           | 35  | 396                    |
| 12                  | 52           | 36  | 432                    |
| 13                  | 60           | 37  | 468                    |

|    |     |
|----|-----|
| 14 | 68  |
| 15 | 76  |
| 16 | 84  |
| 17 | 92  |
| 18 | 100 |
| 19 | 108 |
| 20 | 116 |
| 21 | 124 |
| 22 | 136 |
| 23 | 148 |

|    |            |
|----|------------|
| 38 | 508        |
| 39 | 552        |
| 40 | 600        |
| 41 | 652        |
| 42 | 704        |
| 43 | 768        |
| 44 | 832        |
| 45 | 896        |
| 46 | 960        |
|    | 1024 (960) |

**Table 4.81 – scalefactor bands for a window length of 256 and 240 (values for 240 in brackets) for SHORT\_WINDOW at 22.05 and 24 kHz**

| fs [kHz]             | 22.05 and 24            |
|----------------------|-------------------------|
| num_swb_short_window | 15                      |
| swb                  | swb_offset_short_window |
| 0                    | 0                       |
| 1                    | 4                       |
| 2                    | 8                       |
| 3                    | 12                      |
| 4                    | 16                      |
| 5                    | 20                      |
| 6                    | 24                      |
| 7                    | 28                      |

| swb | swb_offset_short_window |
|-----|-------------------------|
| 8   | 36                      |
| 9   | 44                      |
| 10  | 52                      |
| 11  | 64                      |
| 12  | 76                      |
| 13  | 92                      |
| 14  | 108                     |
|     | 128 (120)               |

**Table 4.82 – scalefactor bands for a window length of 2048 and 1920 (values for 1920 in brackets) for LONG\_WINDOW, LONG\_START\_WINDOW, LONG\_STOP\_WINDOW at 64 kHz**

| fs [kHz]            | 64                     |
|---------------------|------------------------|
| num_swb_long_window | 47                     |
| swb                 | swb_offset_long_window |
| 0                   | 0                      |
| 1                   | 4                      |
| 2                   | 8                      |
| 3                   | 12                     |
| 4                   | 16                     |
| 5                   | 20                     |
| 6                   | 24                     |
| 7                   | 28                     |
| 8                   | 32                     |
| 9                   | 36                     |
| 10                  | 40                     |
| 11                  | 44                     |
| 12                  | 48                     |
| 13                  | 52                     |
| 14                  | 56                     |
| 15                  | 64                     |
| 16                  | 72                     |
| 17                  | 80                     |
| 18                  | 88                     |
| 19                  | 100                    |

| swb | swb_offset_long_window |
|-----|------------------------|
| 24  | 172                    |
| 25  | 192                    |
| 26  | 216                    |
| 27  | 240                    |
| 28  | 268                    |
| 29  | 304                    |
| 30  | 344                    |
| 31  | 384                    |
| 32  | 424                    |
| 33  | 464                    |
| 34  | 504                    |
| 35  | 544                    |
| 36  | 584                    |
| 37  | 624                    |
| 38  | 664                    |
| 39  | 704                    |
| 40  | 744                    |
| 41  | 784                    |
| 42  | 824                    |
| 43  | 864                    |

|    |     |
|----|-----|
| 20 | 112 |
| 21 | 124 |
| 22 | 140 |
| 23 | 156 |

|    |            |
|----|------------|
| 44 | 904        |
| 45 | 944        |
| 46 | 984 (960)  |
|    | 1024 (960) |

**Table 4.83 – scalefactor bands for a window length of 256 and 240 (values for 240 in brackets) for SHORT\_WINDOW at 64 kHz**

| fs [kHz]             | 64                      |
|----------------------|-------------------------|
| num_swb_short_window | 12                      |
| swb                  | swb_offset_short_window |
| 0                    | 0                       |
| 1                    | 4                       |
| 2                    | 8                       |
| 3                    | 12                      |
| 4                    | 16                      |
| 5                    | 20                      |
| 6                    | 24                      |

| swb | swb_offset_short_window |
|-----|-------------------------|
| 7   | 32                      |
| 8   | 40                      |
| 9   | 48                      |
| 10  | 64                      |
| 11  | 92                      |
|     | 128 (120)               |
|     |                         |

**Table 4.84 – scalefactor bands for a window length of 2048 and 1920 (values for 1920 in brackets) for LONG\_WINDOW, LONG\_START\_WINDOW, LONG\_STOP\_WINDOW at 88.2 and 96 kHz**

| fs [kHz]            | 88.2 and 96            |
|---------------------|------------------------|
| num_swb_long_window | 41                     |
| swb                 | swb_offset_long_window |
| 0                   | 0                      |
| 1                   | 4                      |
| 2                   | 8                      |
| 3                   | 12                     |
| 4                   | 16                     |
| 5                   | 20                     |
| 6                   | 24                     |
| 7                   | 28                     |
| 8                   | 32                     |
| 9                   | 36                     |
| 10                  | 40                     |
| 11                  | 44                     |
| 12                  | 48                     |
| 13                  | 52                     |
| 14                  | 56                     |
| 15                  | 64                     |
| 16                  | 72                     |
| 17                  | 80                     |
| 18                  | 88                     |
| 19                  | 96                     |
| 20                  | 108                    |

| swb | swb_offset_long_window |
|-----|------------------------|
| 21  | 120                    |
| 22  | 132                    |
| 23  | 144                    |
| 24  | 156                    |
| 25  | 172                    |
| 26  | 188                    |
| 27  | 212                    |
| 28  | 240                    |
| 29  | 276                    |
| 30  | 320                    |
| 31  | 384                    |
| 32  | 448                    |
| 33  | 512                    |
| 34  | 576                    |
| 35  | 640                    |
| 36  | 704                    |
| 37  | 768                    |
| 38  | 832                    |
| 39  | 896                    |
| 40  | 960                    |
|     | 1024 (960)             |



**Table 4.85 – scalefactor bands for a window length of 256 and 240 (values for 240 in brackets) for SHORT\_WINDOW at 88.2 and 96 kHz**

| fs [kHz]             | 88.2 and 96             |
|----------------------|-------------------------|
| num_swb_short_window | 12                      |
| swb                  | swb_offset_short_window |
| 0                    | 0                       |
| 1                    | 4                       |
| 2                    | 8                       |
| 3                    | 12                      |
| 4                    | 16                      |
| 5                    | 20                      |
| 6                    | 24                      |

| swb | swb_offset_short_window |
|-----|-------------------------|
| 7   | 32                      |
| 8   | 40                      |
| 9   | 48                      |
| 10  | 64                      |
| 11  | 92                      |
|     | 128 (120)               |

**Table 4.86 – Scalefactor bands for 44.1 and 48 kHz, N=480**

| fs [kHz]            | 44.1, 48               |
|---------------------|------------------------|
| num_swb_long_window | 35                     |
| swb                 | swb_offset_long_window |
| 0                   | 0                      |
| 1                   | 4                      |
| 2                   | 8                      |
| 3                   | 12                     |
| 4                   | 16                     |
| 5                   | 20                     |
| 6                   | 24                     |
| 7                   | 28                     |
| 8                   | 32                     |
| 9                   | 36                     |
| 10                  | 40                     |
| 11                  | 44                     |
| 12                  | 48                     |
| 13                  | 52                     |
| 14                  | 56                     |
| 15                  | 64                     |
| 16                  | 72                     |
| 17                  | 80                     |

| swb | swb_offset_long_window |
|-----|------------------------|
| 18  | 88                     |
| 19  | 96                     |
| 20  | 108                    |
| 21  | 120                    |
| 22  | 132                    |
| 23  | 144                    |
| 24  | 156                    |
| 25  | 172                    |
| 26  | 188                    |
| 27  | 212                    |
| 28  | 240                    |
| 29  | 272                    |
| 30  | 304                    |
| 31  | 336                    |
| 32  | 368                    |
| 33  | 400                    |
| 34  | 432                    |
|     | 480                    |

**Table 4.87 – Scalefactor bands for 44.1 and 48 kHz, N=512**

| fs [kHz]            | 44.1, 48               |
|---------------------|------------------------|
| num_swb_long_window | 36                     |
| swb                 | swb_offset_long_window |
| 0                   | 0                      |
| 1                   | 4                      |
| 2                   | 8                      |
| 3                   | 12                     |
| 4                   | 16                     |
| 5                   | 20                     |
| 6                   | 24                     |
| 7                   | 28                     |
| 8                   | 32                     |

| swb | swb_offset_long_window |
|-----|------------------------|
| 19  | 92                     |
| 20  | 100                    |
| 21  | 112                    |
| 22  | 124                    |
| 23  | 136                    |
| 24  | 148                    |
| 25  | 164                    |
| 26  | 184                    |
| 27  | 208                    |

|    |    |
|----|----|
| 9  | 36 |
| 10 | 40 |
| 11 | 44 |
| 12 | 48 |
| 13 | 52 |
| 14 | 56 |
| 15 | 60 |
| 16 | 68 |
| 17 | 76 |
| 18 | 84 |

|    |     |
|----|-----|
| 28 | 236 |
| 29 | 268 |
| 30 | 300 |
| 31 | 332 |
| 32 | 364 |
| 33 | 396 |
| 34 | 428 |
| 35 | 460 |
|    | 512 |

Table 4.88 – Scalefactor bands for 32 kHz, N=480

|                         |                            |
|-------------------------|----------------------------|
| fs [kHz]                | 32                         |
| num_swb_long_<br>window | 37                         |
| Swb                     | swb_offset_long_<br>window |
| 0                       | 0                          |
| 1                       | 4                          |
| 2                       | 8                          |
| 3                       | 12                         |
| 4                       | 16                         |
| 5                       | 20                         |
| 6                       | 24                         |
| 7                       | 28                         |
| 8                       | 32                         |
| 9                       | 36                         |
| 10                      | 40                         |
| 11                      | 44                         |
| 12                      | 48                         |
| 13                      | 52                         |
| 14                      | 56                         |
| 15                      | 60                         |
| 16                      | 64                         |
| 17                      | 72                         |
| 18                      | 80                         |

|     |                            |
|-----|----------------------------|
| Swb | swb_offset_long_<br>window |
| 19  | 88                         |
| 20  | 96                         |
| 21  | 104                        |
| 22  | 112                        |
| 23  | 124                        |
| 24  | 136                        |
| 25  | 148                        |
| 26  | 164                        |
| 27  | 180                        |
| 28  | 200                        |
| 29  | 224                        |
| 30  | 256                        |
| 31  | 288                        |
| 32  | 320                        |
| 33  | 352                        |
| 34  | 384                        |
| 35  | 416                        |
| 36  | 448                        |
|     | 480                        |

Table 4.89 – Scalefactor bands for 32 kHz, N=512

|                         |                            |
|-------------------------|----------------------------|
| fs [kHz]                | 32                         |
| num_swb_long_<br>window | 37                         |
| Swb                     | swb_offset_long_<br>window |
| 0                       | 0                          |
| 1                       | 4                          |
| 2                       | 8                          |
| 3                       | 12                         |
| 4                       | 16                         |
| 5                       | 20                         |
| 6                       | 24                         |
| 7                       | 28                         |
| 8                       | 32                         |
| 9                       | 36                         |

|     |                            |
|-----|----------------------------|
| swb | swb_offset_long_<br>window |
| 19  | 96                         |
| 20  | 108                        |
| 21  | 120                        |
| 22  | 132                        |
| 23  | 144                        |
| 24  | 160                        |
| 25  | 176                        |
| 26  | 192                        |
| 27  | 212                        |
| 28  | 236                        |

|    |    |
|----|----|
| 10 | 40 |
| 11 | 44 |
| 12 | 48 |
| 13 | 52 |
| 14 | 56 |
| 15 | 64 |
| 16 | 72 |
| 17 | 80 |
| 18 | 88 |

|    |     |
|----|-----|
| 29 | 260 |
| 30 | 288 |
| 31 | 320 |
| 32 | 352 |
| 33 | 384 |
| 34 | 416 |
| 35 | 448 |
| 36 | 480 |
|    | 512 |

**Table 4.90 – Scalefactor bands for 22.05 and 24 kHz, N=480.**

| fs [kHz]            | 24, 22.05              |
|---------------------|------------------------|
| num_swb_long_window | 30                     |
| swb                 | swb_offset_long_window |
| 0                   | 0                      |
| 1                   | 4                      |
| 2                   | 8                      |
| 3                   | 12                     |
| 4                   | 16                     |
| 5                   | 20                     |
| 6                   | 24                     |
| 7                   | 28                     |
| 8                   | 32                     |
| 9                   | 36                     |
| 10                  | 40                     |
| 11                  | 44                     |
| 12                  | 52                     |
| 13                  | 60                     |
| 14                  | 68                     |
| 15                  | 80                     |

| swb | swb_offset_long_window |
|-----|------------------------|
| 16  | 92                     |
| 17  | 104                    |
| 18  | 120                    |
| 19  | 140                    |
| 20  | 164                    |
| 21  | 192                    |
| 22  | 224                    |
| 23  | 256                    |
| 24  | 288                    |
| 25  | 320                    |
| 26  | 352                    |
| 27  | 384                    |
| 28  | 416                    |
| 29  | 448                    |
|     | 480                    |

**Table 4.91 – Scalefactor bands for 22.05 and 24 kHz, N=512**

| fs [kHz]            | 24, 22.05              |
|---------------------|------------------------|
| num_swb_long_window | 31                     |
| swb                 | swb_offset_long_window |
| 0                   | 0                      |
| 1                   | 4                      |
| 2                   | 8                      |
| 3                   | 12                     |
| 4                   | 16                     |
| 5                   | 20                     |
| 6                   | 24                     |
| 7                   | 28                     |
| 8                   | 32                     |
| 9                   | 36                     |
| 10                  | 40                     |
| 11                  | 44                     |
| 12                  | 52                     |
| 13                  | 60                     |

| swb | swb_offset_long_window |
|-----|------------------------|
| 16  | 92                     |
| 17  | 104                    |
| 18  | 120                    |
| 19  | 140                    |
| 20  | 164                    |
| 21  | 192                    |
| 22  | 224                    |
| 23  | 256                    |
| 24  | 288                    |
| 25  | 320                    |
| 26  | 352                    |
| 27  | 384                    |
| 28  | 416                    |
| 29  | 448                    |

|    |    |    |     |
|----|----|----|-----|
| 14 | 68 | 30 | 480 |
| 15 | 80 |    | 512 |

Table 4.92 – AAC error sensitivity category assignment for main payload

| SCE / LFE / mono layer | CPE, common_window == 0 | CPE, common_window == 1 / stereo layer | data_element                      | function                        |
|------------------------|-------------------------|----------------------------------------|-----------------------------------|---------------------------------|
| 1                      | -                       | 0                                      | max_sfb                           | aac_scalable_extension_header() |
| -                      | -                       | 0                                      | ms_mask_present                   | aac_scalable_extension_header() |
| 1                      | -                       | 1                                      | tns_data_present                  | aac_scalable_extension_header() |
| 1                      | -                       | 0                                      | ics_reserved_bit                  | aac_scalable_main_header()      |
| 1                      | -                       | 1                                      | ltp_data_present                  | aac_scalable_main_header()      |
| 1                      | -                       | 0                                      | max_sfb                           | aac_scalable_main_header()      |
| -                      | -                       | 0                                      | ms_mask_present                   | aac_scalable_main_header()      |
| 1                      | -                       | 0                                      | scale_factor_grouping             | aac_scalable_main_header()      |
| 1                      | -                       | 0                                      | tns_channel_mono_layer            | aac_scalable_main_header()      |
| 1                      | -                       | 1                                      | tns_data_present                  | aac_scalable_main_header()      |
| 1                      | -                       | 0                                      | window_sequence                   | aac_scalable_main_header()      |
| 1                      | -                       | 0                                      | window_shape                      | aac_scalable_main_header()      |
| -                      | 0                       | 0                                      | common_window                     | channel_pair_element()          |
| -                      | 0                       | 0                                      | element_instance_tag              | channel_pair_element()          |
| -                      | 0                       | 0                                      | ms_mask_present                   | channel_pair_element()          |
| -                      | 0                       | 0                                      | ms_used                           | channel_pair_element()          |
| 1                      | 1                       | 1                                      | diff_control                      | diff_control_data()             |
| 1                      | 1                       | 1                                      | diff_control_lr                   | diff_control_data_lr()          |
| 1                      | 1                       | 0                                      | ics_reserved_bit                  | ics_info()                      |
| 1                      | 1                       | 1                                      | ltp_data_present                  | ics_info()                      |
| 1                      | 1                       | 0                                      | max_sfb                           | ics_info()                      |
| 1                      | 1                       | 0                                      | predictor_data_present            | ics_info()                      |
| 1                      | 1                       | 0                                      | scale_factor_grouping             | ics_info()                      |
| 1                      | 1                       | 0                                      | window_sequence                   | ics_info()                      |
| 1                      | 1                       | 0                                      | window_shape                      | ics_info()                      |
| 1                      | 1                       | 1                                      | gain_control_data_present         | individual_channel_stream()     |
| 1                      | 1                       | 1                                      | global_gain                       | individual_channel_stream()     |
| 1                      | 1                       | 1                                      | length_of_longest_codeword        | individual_channel_stream()     |
| 1                      | 1                       | 1                                      | length_of_reordered_spectral_data | individual_channel_stream()     |
| 1                      | 1                       | 1                                      | pulse_data_present                | individual_channel_stream()     |
| 1                      | 1                       | 1                                      | tns_data_present                  | individual_channel_stream()     |
| 1                      | -                       | -                                      | element_instance_tag              | lfe_channel_element()           |
| 1                      | 1                       | 1                                      | ltp_coef                          | ltp_data()                      |
| 1                      | 1                       | 1                                      | ltp_lag                           | ltp_data()                      |
| 1                      | 1                       | 1                                      | ltp_lag_update                    | ltp_data()                      |
| 1                      | 1                       | 1                                      | ltp_long_used                     | ltp_data()                      |
| 1                      | 1                       | 1                                      | ltp_short_lag                     | ltp_data()                      |
| 1                      | 1                       | 1                                      | ltp_short_lag_present             | ltp_data()                      |
| 1                      | 1                       | 1                                      | ltp_short_used                    | ltp_data()                      |

| SCE / LFE / mono layer | CPE, common_window == 0 | CPE, common_window == 1 / stereo layer | data_element             | function                  |
|------------------------|-------------------------|----------------------------------------|--------------------------|---------------------------|
| -                      | -                       | 0                                      | ms_used                  | ms_data()                 |
| 1                      | 1                       | 1                                      | number_pulse             | pulse_data()              |
| 1                      | 1                       | 1                                      | pulse_amp                | pulse_data()              |
| 1                      | 1                       | 1                                      | pulse_offset             | pulse_data()              |
| 1                      | 1                       | 1                                      | pulse_start_sfb          | pulse_data()              |
| 4                      | 4                       | 4                                      | reordered_spectral_data  | reordered_spectral_data() |
| 1                      | 1                       | 1                                      | dpcm_noise_last_position | scale_factor_data()       |
| 1                      | 1                       | 1                                      | dpcm_noise_nrg           | scale_factor_data()       |
| 1                      | 1                       | 1                                      | hcod_sf                  | scale_factor_data()       |
| 1                      | 1                       | 1                                      | length_of_rvlc_escapes   | scale_factor_data()       |
| 1                      | 1                       | 1                                      | length_of_rvlc_sf        | scale_factor_data()       |
| 1                      | 1                       | 1                                      | rev_global_gain          | scale_factor_data()       |
| 2                      | 2                       | 2                                      | rvlc_cod_sf              | scale_factor_data()       |
| 2                      | 2                       | 2                                      | rvlc_esc_sf              | scale_factor_data()       |
| 1                      | 1                       | 1                                      | sf_concealment           | scale_factor_data()       |
| 1                      | 1                       | 1                                      | sf_escapes_present       | scale_factor_data()       |
| 1                      | 1                       | 1                                      | sect_cb                  | section_data()            |
| 1                      | 1                       | 1                                      | sect_len_incr            | section_data()            |
| 1                      | -                       | -                                      | element_instance_tag     | single_channel_element()  |
| 4                      | 4                       | 4                                      | hcod                     | spectral_data()           |
| 4                      | 4                       | 4                                      | hcod_esc_y               | spectral_data()           |
| 4                      | 4                       | 4                                      | hcod_esc_z               | spectral_data()           |
| 4                      | 4                       | 4                                      | pair_sign_bits           | spectral_data()           |
| 4                      | 4                       | 4                                      | quad_sign_bits           | spectral_data()           |
| 3                      | 3                       | 3                                      | coef                     | tns_data()                |
| 3                      | 3                       | 3                                      | coef_compress            | tns_data()                |
| 3                      | 3                       | 3                                      | coef_res                 | tns_data()                |
| 3                      | 3                       | 3                                      | direction                | tns_data()                |
| 3                      | 3                       | 3                                      | length                   | tns_data()                |
| 3                      | 3                       | 3                                      | n_filt                   | tns_data()                |
| 3                      | 3                       | 3                                      | order                    | tns_data()                |

Table 4.93 – AAC error sensitivity category assignment for extended payload

| extension_payload | data_element                 | function             |
|-------------------|------------------------------|----------------------|
| 6                 | drc_band_top                 | dynamic_range_info() |
| 6                 | drc_bands_incr               | dynamic_range_info() |
| 6                 | drc_bands_present            | dynamic_range_info() |
| 6                 | drc_bands_reserved_bits      | dynamic_range_info() |
| 6                 | drc_tag_reserved_bits        | dynamic_range_info() |
| 6                 | dyn_rng_ct                   | dynamic_range_info() |
| 6                 | dyn_rng_sgn                  | dynamic_range_info() |
| 6                 | excluded_chns_present        | dynamic_range_info() |
| 6                 | pce_instance_tag             | dynamic_range_info() |
| 6                 | pce_tag_present              | dynamic_range_info() |
| 6                 | prog_ref_level               | dynamic_range_info() |
| 6                 | prog_ref_level_present       | dynamic_range_info() |
| 6                 | prog_ref_level_reserved_bits | dynamic_range_info() |
| 6                 | additional_excluded_chns     | excluded_channels()  |
| 6                 | exclude_mask                 | excluded_channels()  |
| 5                 | extension_type               | extension_payload()  |
| 7                 | fill_byte                    | extension_payload()  |
| 7                 | fill_nibble                  | extension_payload()  |
| 7                 | other_bits                   | extension_payload()  |

### 4.5.5 Figures

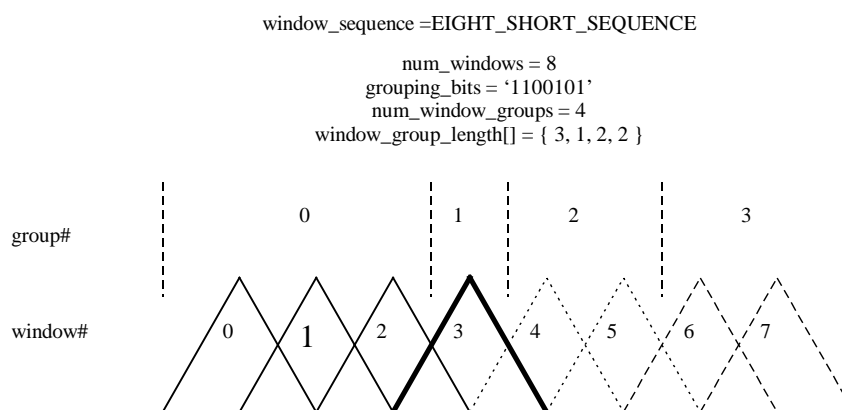
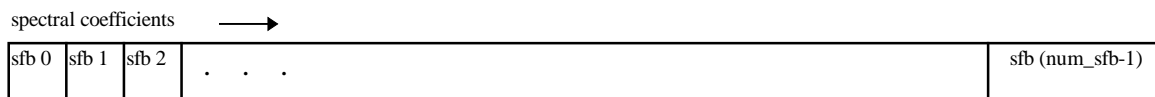
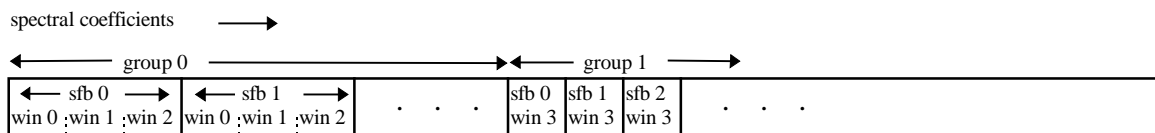


Figure 4.20 – Example for short window grouping



Order of scalefactor bands for ONLY\_LONG\_SEQUENCE

Figure 4.21 – Spectral order of scalefactor bands in case of ONLY\_LONG\_SEQUENCE



Order of scale factor bands for EIGHT\_SHORT\_SEQUENCE  
window\_group\_length[] = { 3, 1, ... }

Figure 4.22 – Spectral order of scalefactor bands in case of EIGHT\_SHORT\_SEQUENCE

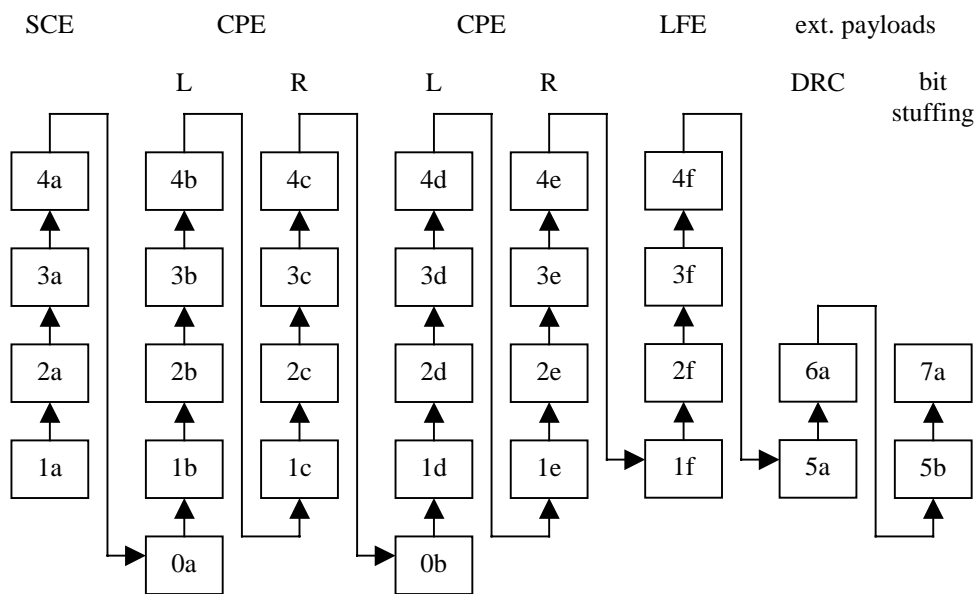


Figure 4.23 – Dependency structure in case of error resilient multichannel AAC syntax (channelConfiguration == 6)

## 4.6 GA-Tool Descriptions

### 4.6.1 Quantization

(subclause identical to ISO/IEC 13818-7)

#### 4.6.1.1 Tool description

For quantization of the spectral coefficients in the encoder a non uniform quantizer is used. Therefore the decoder must perform the inverse non uniform quantization after the Huffman decoding of the scalefactors (see subclauses 4.6.3 and 4.6.2) and spectral data (see subclause 4.6.3).

#### 4.6.1.2 Definitions

##### Help elements:

`x_quant[g][win][sfb][bin]` quantized spectral coefficient for group *g*, window *win*, scalefactor band *sfb*, coefficient *bin*.  
`x_invquant[g][win][sfb][bin]` spectral coefficient for group *g*, window *win*, scalefactor band *sfb*, coefficient *bin* after inverse quantization.

#### 4.6.1.3 Decoding process

The inverse quantization is described by the following formula:

$$x\_invquant = \text{Sign}(x\_quant) \cdot |x\_quant|^{\frac{4}{3}}$$

The maximum allowed absolute amplitude for `x_quant` is 8191. The inverse quantization is applied as follows:

```
for( g=0; g<num_window_groups; g++ ) {
  for( sfb=0; sfb < max_sfb; sfb++ ) {
    width = (swb_offset [sfb+1] - swb_offset [sfb] );
    for( win = 0; win < window_group_len[g]; win++ ) {
      for( bin=0; bin<width; bin++ ) {
        x_invquant[g][win][sfb][bin] = sign(x_quant[g][win][sfb][bin]) *
          abs(x_quant[g][win][sfb][bin]) ^ (4/3);
      }
    }
  }
}
```

### 4.6.2 Scalefactors

(subclause similar to ISO/IEC 13818-7)

#### 4.6.2.1 Tool description

The basic method to adjust the quantization noise in the frequency domain is the noise shaping using scalefactors. For this purpose the spectrum is divided in several groups of spectral coefficients called scalefactor bands which share one scalefactor (see subclause 4.5.2.3.4). A scalefactor represents a gain value which is used to change the amplitude of all spectral coefficients in that scalefactor band. This mechanism is used to change the allocation of the quantization noise in the spectral domain generated by the non uniform quantizer.

For window\_sequences which contain SHORT\_WINDOWS grouping can be applied, i.e. a specified number of consecutive SHORT\_WINDOWS may have only one set of scalefactors. Each scalefactor is then applied to a group of scalefactor bands corresponding in frequency (see subclause 4.5.2.3.4).

In this tool the scalefactors are applied to the inverse quantized coefficients to reconstruct the spectral values.

#### 4.6.2.2 Definitions

##### Bit stream elements:

**global\_gain** An 8-bit unsigned integer value representing the value of the first scalefactor. It is also the start value for the following differential coded scalefactors  
**scale\_factor\_data()** Part of bit stream which contains the differential coded scalefactors  
**hcod\_sf[]** Huffman codeword from the Huffman code table used for coding of scalefactors, see subclause 4.6.3.2

##### Help elements:



dpcm\_sf[g][sfb] Differential coded scalefactor of group g, scalefactor band sfb.  
 x\_rescal[] rescaled spectral coefficients  
 sf[g][sfb] Array for scalefactors of each group  
 get\_scale\_factor\_gain() Function that returns the gain value corresponding to a scalefactor

### 4.6.2.3 Decoding process

#### 4.6.2.3.1 Scalefactor bands

Scalefactors are used to shape the quantization noise in the spectral domain. For this purpose, the spectrum is divided into several scalefactor bands (see subclause 4.5.2.3.4). Each scalefactor band has a scalefactor, which represents a certain gain value which has to be applied to all spectral coefficients in this scalefactor band. In case of EIGHT\_SHORT\_SEQUENCE a scalefactor band may contain multiple scalefactor window bands of consecutive SHORT\_WINDOWS (see subclause 4.5.2.3.4 and 4.5.2.3.5).

#### 4.6.2.3.2 Decoding of scalefactors

For all scalefactors the difference to the preceeding value is coded using the Huffman code book given in Table 4.A.1. See subclause 4.6.3 for a detailed description of the Huffman decoding process. The start value is given explicitly as a 8 bit PCM in the bitstream element **global\_gain**. A scalefactor is not transmitted for scalefactor bands which are coded with the Huffman codebook ZERO\_HCB. If the Huffman codebook for a scalefactor band is coded with INTENSITY\_HCB or INTENSITY\_HCB2, the scalefactor is used for intensity stereo (see subclauses 4.6.3 and 4.6.8.2). In that case a normal scalefactor does not exist (but is initialized to zero to have an valid entry in the array).

The following pseudo code describes how to decode the scalefactors sf[g][sfb]:

```

last_sf = global_gain;
for( g=0; g < num_window_groups; g++ ) {
    for( sfb=0; sfb<max_sfb; sfb++ ) {
        if( sfb_cb[g][sfb] != ZERO_HCB && sfb_cb[g][sfb] != INTENSITY_HCB
            && sfb_cb[g][sfb] != INTENSITY_HCB2 ) {
            dpcm_sf = decode_huffman() - index_offset; /* see subclause 4.6.3 „Noiseless Coding“ */
            sf[g][sfb] = dpcm_sf + last_sf;
            last_sf = sf[g][sfb];
        }
        else {
            sf[g][sfb] = 0;
        }
    }
}

```

Note that scalefactors, sf[g][sfb], must be within the range of zero to 256, both inclusive.

In case of error resilient scalefactor coding, a RVLC has been used instead of a Huffman code. The decoding process of the RVLC words is the same as for the Huffman codewords; just another codebook has to be used. This codebook uses symmetric codewords. Due to this it is possible to detect errors, because asymmetric codewords are illegal. Furthermore, decoding can be started at both sides. To allow backward decoding, an additional value is available within the bitstream, which contains the last scalefactor value. In case of intensity an additional codeword is available, which allows backwards decoding. In case of PNS an additional DPCM value is available for the same reason.

A decoded value of ±7 is used as ESC\_FLAG. It signals that an escape value exists, that has to be added to +7 or subtracted from -7 in order to find the actual scalefactor value. This escape value is Huffman encoded.

#### 4.6.2.3.3 Applying scalefactors

The spectral coefficients of all scalefactor bands which correspond to a scalefactor have to be rescaled according to their scalefactor. In case of a window sequence that contains groups of short windows all coefficients in grouped scalefactor window bands have to be scaled using the same scalefactor.

In case of window\_sequences with only one window, the scalefactor bands and their corresponding coefficients are in spectral ascending order. In case of EIGHT\_SHORT\_SEQUENCE and grouping the spectral coefficients of grouped short windows are interleaved by scalefactor window bands. See subclause 4.5.2.3.5 for more detailed information.

The rescaling operation is done according to the following pseudo code:

```
for( g=0; g<num_window_groups; g++ ) {
  for( sfb=0; sfb < max_sfb; sfb++ ) {
    width = (swb_offset [sfb+1] - swb_offset [sfb] );
    for( win = 0; win < window_group_len[g]; win++ ) {
      gain = get_scale_factor_gain( sf[g][sfb] );
      for( k=0; k<width; k++ ) {
        x_rescal[g][window][sfb][k] =
          x_invquant[g][window][sfb][k] * gain;
      }
    }
  }
}
```

The function `get_scale_factor_gain(sf[g][sfb])` returns the gain factor that corresponds to a scalefactor. The return value follows the equation:

$$gain = 2^{0.25 \cdot (sf[g][sfb] - SF\_OFFSET)}$$

The constant `SF_OFFSET` must be set to 100.

The following pseudo code describes this operation:

```
get_scale_factor_gain( sf[g][sfb] ) {
  SF_OFFSET = 100;
  gain = 2^(0.25 * ( sf[g][sfb] - SF_OFFSET));
  return( gain );
}
```

### 4.6.3 Noiseless coding

(subclause similar to ISO/IEC 13818-7)

#### 4.6.3.1 Tool description

Noiseless coding is used to further reduce the redundancy of the scalefactors and the quantized spectrum of each audio channel.

The `global_gain` is coded as an 8 bit unsigned integer. The first scalefactor associated with the quantized spectrum is differentially coded relative to the `global_gain` value and then Huffman coded using the scalefactor codebook. The remaining scalefactors are differentially coded relative to the previous scalefactor and then Huffman coded using the scalefactor codebook.

Noiseless coding of the quantized spectrum relies on two divisions of the spectral coefficients. The first is a division into scalefactor bands that contain a multiple of 4 quantized spectral coefficients (see subclauses 4.5.2.3.4 and 4.5.2.3.5).

The second division, which is dependent on the quantized spectral data, is a division by scalefactor bands to form sections. The significance of a section is that the quantized spectrum within the section is represented using a single Huffman codebook chosen from a set of 11 possible codebooks. The length of a section and its associated Huffman codebook must be transmitted as side information in addition to the section's Huffman coded spectrum. Note that the length of a section is given in scalefactor bands rather than scalefactor window bands (see subclauses 4.5.2.3.4). In order to maximize the match of the statistics of the quantized spectrum to that of the Huffman codebooks the number of sections is permitted to be as large as the number of scalefactor bands. The maximum size of a section is `max_sfb` scalefactor bands.

As indicated in Table 4.95, spectrum Huffman codebooks can represent signed or unsigned n-tuples of coefficients. For unsigned codebooks, sign bits for every non-zero coefficient in the n-tuple immediately follow the associated codeword.

The noiseless coding has two ways to represent large quantized spectra. One way is to send the escape flag from the escape (ESC) Huffman codebook, which signals that the bits immediately following that codeword plus optional sign bits are an escape sequence that encodes values larger than those represented by the ESC Huffman codebook. A second way is the pulse escape method, in which relatively large-amplitude coefficients can be replaced by coefficients with smaller amplitudes in order to enable the use of Huffman code tables with higher coding efficiency. This replacement is corrected by sending the position of the spectral coefficient and the differences in amplitude as side information. The frequency information is represented by the combination of the scalefactor band number to indicate a base frequency and an offset into that scalefactor band.

#### 4.6.3.2 Definitions

|                                        |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
|----------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>sect_cb[g][i]</b>                   | spectrum Huffman codebook used for section i in group g.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
| <b>sect_len_incr</b>                   | used to compute the length of a section, measures number of scalefactor bands from start of section. The length of <b>sect_len_incr</b> is 3 bits if <code>window_sequence</code> is <code>EIGHT_SHORT_SEQUENCE</code> and 5 bits otherwise.                                                                                                                                                                                                                                                                                                                                                                                                                       |
| <b>global_gain</b>                     | global gain of the quantized spectrum, sent as unsigned integer value.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                             |
| <b>hcod_sf[]</b>                       | Huffman codeword from the Huffman code table used for coding of scalefactors.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                      |
| <b>hcod[sect_cb[g][i]][w][x][y][z]</b> | Huffman codeword from codebook <b>sect_cb[g][i]</b> that encodes the next 4-tuple (w, x, y, z) of spectral coefficients, where w, x, y, z are quantized spectral coefficients. Within an n-tuple, w, x, y, z are ordered as described in subclause 4.5.2.3.5. so that $x_{\text{quant}}[\text{group}][\text{win}][\text{sfb}][\text{bin}] = w$ , $x_{\text{quant}}[\text{group}][\text{win}][\text{sfb}][\text{bin}+1] = x$ , $x_{\text{quant}}[\text{group}][\text{win}][\text{sfb}][\text{bin}+2] = y$ and $x_{\text{quant}}[\text{group}][\text{win}][\text{sfb}][\text{bin}+3] = z$ . N-tuples progress from low to high frequency within the current section. |
| <b>hcod[sect_cb[g][i]][y][z]</b>       | Huffman codeword from codebook <b>sect_cb[g][i]</b> that encodes the next 2-tuple (y, z) of spectral coefficients, where y, z are quantized spectral coefficients. Within an n-tuple, y, z are ordered as described in subclause 4.5.2.3.5 so that $x_{\text{quant}}[\text{group}][\text{win}][\text{sfb}][\text{bin}] = y$ and $x_{\text{quant}}[\text{group}][\text{win}][\text{sfb}][\text{bin}+1] = z$ . N-tuples progress from low to high frequency within the current section.                                                                                                                                                                              |
| <b>quad_sign_bits</b>                  | sign bits for non-zero coefficients in the spectral 4-tuple. A '1' indicates a negative coefficient, a '0' a positive one. Bits associated with lower frequency coefficients are sent first.                                                                                                                                                                                                                                                                                                                                                                                                                                                                       |
| <b>pair_sign_bits</b>                  | sign bits for non-zero coefficients in the spectral 2-tuple. A '1' indicates a negative coefficient, a '0' a positive one. Bits associated with lower frequency coefficients are sent first.                                                                                                                                                                                                                                                                                                                                                                                                                                                                       |
| <b>hcod_esc_y</b>                      | escape sequence for quantized spectral coefficient y of 2-tuple (y,z) associated with the preceding Huffman codeword.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
| <b>hcod_esc_z</b>                      | escape sequence for quantized spectral coefficient z of 2-tuple (y,z) associated with the preceding Huffman codeword.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
| <b>pulse_data_present</b>              | 1 bit indicating whether the pulse escape is used (1) or not (0). Note that <code>pulse_data_present</code> must be 0, if <code>window_sequence == EIGHT_SHORT_SEQUENCE</code> ;                                                                                                                                                                                                                                                                                                                                                                                                                                                                                   |
| <b>number_pulse</b>                    | 2 bits indicating how many pulse escapes are used. The number of pulse escapes is from 1 to 4.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |
| <b>pulse_start_sfb</b>                 | 6 bits indicating the index of the lowest scalefactor band where the pulse escape is achieved.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |

|                           |                                                                                                                |
|---------------------------|----------------------------------------------------------------------------------------------------------------|
| <b>pulse_offset[i]</b>    | 5 bits indicating the offset.                                                                                  |
| <b>pulse_amp[i]</b>       | 4 bits indicating the unsigned magnitude of the pulse.                                                         |
| sect_start[g][i]          | offset to first scalefactor band in section i of group g.                                                      |
| sect_end[g][i]            | offset to one higher than last scalefactor band in section i of group g.                                       |
| num_sec[g]                | number of sections in group g.                                                                                 |
| escape_flag               | the value of 16 in the ESC Huffman codebook                                                                    |
| escape_prefix             | the bit sequence of N 1's                                                                                      |
| escape_separator          | one 0 bit                                                                                                      |
| escape_word               | an N+4 bit unsigned integer word, msb first                                                                    |
| escape_sequence           | the sequence of escape_prefix, escape_separator and escape_word                                                |
| escape_code               | $2^{(N+4)} + \text{escape\_word}$                                                                              |
| x_quant[g][win][sfb][bin] | <i>Huffman decoded value for group g, window win, scalefactor band sfb, coefficient bin</i>                    |
| spec[w][k]                | <i>de-interleaved spectrum. w ranges from 0 to num_windows-1 and k ranges from 0 to swb_offset[num_swb]-1.</i> |

The noiseless coding tool requires these constants (see Table 4.50).

|                |    |
|----------------|----|
| ZERO_HCB       | 0  |
| FIRST_PAIR_HCB | 5  |
| ESC_HCB        | 11 |
| QUAD_LEN       | 4  |
| PAIR_LEN       | 2  |
| NOISE_HCB      | 13 |
| INTENSITY_HCB2 | 14 |
| INTENSITY_HCB  | 15 |
| ESC_FLAG       | 16 |

#### 4.6.3.3 Decoding process

Four-tuples or 2-tuples of quantized spectral coefficients are Huffman coded and transmitted starting from the lowest-frequency coefficient and progressing to the highest-frequency coefficient. For the case of multiple windows per block (*EIGHT\_SHORT\_SEQUENCE*), the grouped and interleaved set of spectral coefficients is treated as a single set of coefficients that progress from low to high. The set of coefficients may need to be de-interleaved after they are decoded (see subclause 4.5.2.3.5). Coefficients are stored in the array `x_quant[g][win][sfb][bin]`, and the order of transmission of the Huffman codewords is such that when they are decoded in the order received and stored in the array, *bin* is the most rapidly incrementing index and *g* is the most slowly incrementing index. Within a codeword, for those associated with spectral four-tuples, the order of decoding is *w, x, y, z*; for codewords associated with spectral two-tuples, the order of decoding is *y, z*. The set of coefficients is divided into sections and the sectioning information is transmitted starting from the lowest frequency section and progressing to the highest frequency section. The spectral information for sections that are coded with the „zero“ codebook is not sent as this spectral information is zero. Similarly, spectral information for sections coded with the „intensity“ codebooks is not sent. The spectral information for all scalefactor bands at and above **max\_sfb**, for which there is no section data, is zero.

There is a single differential scalefactor codebook which represents a range of values as shown in Table 4.94. The differential scalefactor codebook is shown in Table 4.A.1. There are eleven Huffman codebooks for the spectral data, as shown in Table 4.95. The codebooks are shown in Table 4.A.2 through Table 4.A.12. There are four other „codebooks“ above and beyond the actual Huffman codebooks, specifically the „zero“ codebook, indicating that neither scalefactors nor quantized data will be transmitted, and the „intensity“ codebooks indicating that this individual channel is part of a channel pair, and that the data that would normally be scalefactors is instead steering data for intensity stereo. Similarly, the „noise substitution“ codebook indicates that the spectral coefficients are derived from random numbers rather than quantized spectral values, and that the data that would normally be scalefactors is instead noise energy data. In these cases, no quantized spectral data are transmitted. Codebook index 12 is reserved.

The spectrum Huffman codebooks encode 2- or 4-tuples of signed or unsigned quantized spectral coefficients, as shown in Table 4.95. This table also indicates the largest absolute value (LAV) able to be encoded by each codebook and defines a boolean helper variable array, `unsigned_cb[]`, that is 1 if the codebook is unsigned and 0 if signed.

The result of Huffman decoding each differential scalefactor codeword is the codeword index, listed in the first column of Table 4.A.1. This is translated to the desired differential scalefactor by adding `index_offset` to the index. `Index_offset` has a value of -60, as shown in Table 4.94. Likewise, the result of Huffman decoding each spectrum n-tuple is the codeword index, listed in the first column of Table 4.A.2 through Table 4.A.12. This index is translated to the n-tuple spectral values as specified in the following pseudo C-code:

`unsigned` = Boolean value `unsigned_cb[i]`, listed in second column of Table 4.95.  
`dim` = Dimension of codebook, listed in the third column of Table 4.95.  
`lav` = LAV, listed in the fourth column of Table 4.95.  
`idx` = codeword index

```

if (unsigned) {
    mod = lav + 1;
    off = 0;
}
else {
    mod = 2*lav + 1;
    off = lav;
}

if (dim == 4) {
    w = INT(idx/(mod*mod*mod)) - off;
    idx -= (w+off)*(mod*mod*mod)
    x = INT(idx/(mod*mod)) - off;
    idx -= (x+off)*(mod*mod)
    y = INT(idx/mod) - off;
    idx -= (y+off)*mod
    z = idx - off;
}
else {
    y = INT(idx/mod) - off;
    idx -= (y+off)*mod
    z = idx - off;
}

```

If the Huffman codebook represents signed values, the decoding of the quantized spectral n-tuple is complete after Huffman decoding and translation of codeword index to quantized spectral coefficients. If the codebook represents unsigned values then the sign bits associated with non-zero coefficients immediately follow the Huffman codeword, with a '1' indicating a negative coefficient and a '0' indicating a positive one. For example, if a Huffman codeword from codebook 7

**hcod[7][y][z]**

has been parsed, then immediately following this in the bitstream is

**pair\_sign\_bits**

which is a variable length field of 0 to 2 bits. It can be parsed directly from the bitstream as

```

if (y != 0)
    if (one_sign_bit == 1)
        y = -y
if (z != 0)
    if (one_sign_bit == 1)
        z = -z

```

where `one_sign_bit` is the next bit in the bitstream and `pair_sign_bits` is the concatenation of the `one_sign_bit` fields.

The ESC codebook is a special case. It represents values from 0 to 16 inclusive, but values from 0 to 15 encode actual data values, and the value 16 is an escape\_flag that signals the presence of **hcod\_esc\_y** or **hcod\_esc\_z**, either of which will be denoted as an escape\_sequence. This escape\_sequence permits quantized spectral elements of  $LAV > 15$  to be encoded. It consists of an escape\_prefix of N 1's, followed by an escape\_separator of one zero, followed by an escape\_word of N+4 bits representing an unsigned integer value. The escape\_sequence has a decoded value of  $2^{(N+4)} + \text{escape\_word}$ . The desired quantized spectral coefficient is then the sign indicated by the pair\_sign\_bits applied to the value of the escape\_sequence. In other words, an escape\_sequence of 00000 would decode as 16, an escape\_sequence of 01111 as 31, an escape\_sequence of 1000000 as 32, one of 1011111 as 63, and so on. Note that restrictions in subclause 4.6.1.3 dictate that the length of the escape\_sequence is always less than 22 bits. For escape Huffman codewords the ordering of bitstream elements is Huffman codeword followed by 0 to 2 sign bits followed by 0 to 2 escape sequences.

When **pulse\_data\_present** is 1 (the pulse escape is used), one or several quantized coefficients have been replaced by coefficients with smaller amplitudes in the encoder. The number of coefficients replaced is indicated by **number\_pulse**. In reconstructing the quantized spectral coefficients  $x_{\text{quant}}$  this replacement is compensated by adding **pulse\_amp** to or subtracting **pulse\_amp** from the previously decoded coefficients whose frequency indices are indicated by **pulse\_start\_sfb** and **pulse\_offset**. Note that the pulse escape method is illegal for a block whose **window\_sequence** is *EIGHT\_SHORT\_SEQUENCE*. The decoding process is specified in the following pseudo-C code:

```

if (pulse_data_present) {
    g = 0;
    win = 0;
    k = swb_offset[pulse_start_sfb];
    for (j = 0; j < number_pulse + 1; j++) {
        k += pulse_offset[j];

        /* translate_pulse_parameters(); */
        for (sfb = pulse_start_sfb; sfb < num_swb; sfb++) {
            if (k < swb_offset[sfb + 1]) {
                bin = k - swb_offset[sfb];
                break;
            }
        }

        /* restore coefficients */
        if (x_quant[g][win][sfb][bin] > 0)
            x_quant[g][win][sfb][bin] += pulse_amp[j];
        else
            x_quant[g][win][sfb][bin] -= pulse_amp[j];
    }
}

```

Several decoder tools (TNS, filterbank) access the spectral coefficients in a non-interleaved fashion, i.e. all spectral coefficients are ordered according to window number and frequency within a window. This is indicated by using the notation  $\text{spec}[w][k]$  rather than  $x_{\text{quant}}[g][w][\text{sfb}][\text{bin}]$ .

The following pseudo C-code indicates the correspondence between the four-dimensional, or interleaved, structure of array  $x_{\text{quant}}[ ][ ][ ][ ]$  and the two-dimensional, or de-interleaved, structure of array  $\text{spec}[ ][ ]$ . In the latter array the first index increments over the individual windows in the window sequence, and the second index increments over the spectral coefficients that correspond to each window, where the coefficients progress linearly from low to high frequency.

```

quant_to_spec() {
    k=0;
    for (g=0; g < num_window_groups; g++) {
        j=0;
        for (sfb=0; sfb < num_swb; sfb++) {
            width = swb_offset[sfb + 1] - swb_offset[sfb];
            for (win=0; win < window_group_length[g]; win++) {
                for (bin=0; bin < width; bin++) {
                    spec[win+k][bin+j] = x_quant[g][win][sfb][bin];
                }
            }
        }
    }
}

```

```

        }
        j+=width;
    }
    k+=window_group_length[g];
}
}

```

Decoding of reordered spectral data cannot be done straightforward. The following c-like description shows the decoding process:

```

/* helper functions */
void InitReordering(void);
/* Initializes variables used by the reordering functions like the segment
widths and the used offsets in segments and codewords */

void InitRemainingBitsInSegment(void);
/* Initializes remainingBitsInSegment[] array for each segment with the
total size of the segment */

int DecodeCodeword(codewordNr, segmentNr, direction);
/* Try to decode the codeword indexed by codewordNr using data already read
for this codeword and using data from the segment index by segmentNr.
The read direction in the segment is given by direction.
DecodeCodeword returns the number of bits read from the indexed segment. */

void MoveFromSegmentToCodeword(codewordNr, segmentNr, bitLen, direction);
/* Move bitLen bits from the segment indexed by segmentNr to the codeword
indexed by codewordNr using direction as read direction in the segment.
The bits are appended to existing bits for the codeword and the codeword
length is adjusted. */

void AdjustOffsetsInSegment(segmentNr, bitLen, direction);
/* Like MoveFromSegmentToCodeword(), but no bits are moved. Only the offsets
for the segment indexed by segmentNr are adjusted according bitLen and
direction. */

void MarkCodewordAsDecoded(codewordNr);
/* Marks the codeword indexed by codewordNr as decoded. */

bool CodewordIsNotDecoded(codewordNr);
/* Returns TRUE if the codeword indexed by codewordNr is not decoded. */

void ToggleReadDirection(void);
/* Toggles the read direction in the segments between forward and backward. */

/* (input) variables */
numberOfCodewords;
numberOfSegments;
numberOfSets;

DecodeReorderedSpectralData()
{
    InitReordering();
    InitRemainingBitsInSegment();

    /* first step: decode PCWs (set 0) */
    readDirection = forward;
    for (codeword = 0; codeword < numberOfSegments; codeword++) {
        cwLen = DecodeCodeword(codeword, codeword, readDirection);
        if (cwLen <= remainingBitsInSegment[codeword]) {
            AdjustOffsetsInSegment(codeword, cwLen, readDirection);
            MarkCodewordAsDecoded(codeword);
            remainingBitsInSegment[codeword] -= cwLen;
        }
        else {
            /* error !!! (PCWs do always fit into segments) */

```

```

}
}

/* second step: decode nonPCWs */
for (set = 1; set < numberOfSets; set++) {
  ToggleReadDirection();
  for (trial = 0; trial < numberOfSegments; trial++) {
    for (codewordBase = 0; codewordBase < numberOfSegments; codewordBase++) {
      segment = (trial + codewordBase) % numberOfSegments;
      codeword = codewordBase + set*numberOfSegments;

      if (CodewordIsNotDecoded(codeword) &&
          (remainingBitsInSegment[segment] > 0)) {
        cwLenInSegment = DecodeCodeword(codeword, segment, readDirection);
        if (cwLenInSegment <= remainingBitsInSegment[segment]) {
          AdjustOffsetsInSegment(segment, cwLenInSegment, readDirection);
          MarkCodewordAsDecoded(codeword);
          remainingBitsInSegment[segment] -= cwLenInSegment;
        }
        else { /* only part of codeword in segment */
          MoveFromSegmentToCodeword(codeword,
                                     segment,
                                     remainingBitsInSegment[segment],
                                     readDirection);
          remainingBitsInSegment[segment] = 0;
        }
      }
    }
  }
}
}
}
}

```

#### 4.6.3.4 Tables

**Table 4.94 – Scalefactor Huffman codebook parameters**

| Codebook Number | Dimension of Codebook | index_offset | Range of values | Codebook listed in Table |
|-----------------|-----------------------|--------------|-----------------|--------------------------|
| 0               | 1                     | -60          | -60 to +60      | Table 4.A.1              |

**Table 4.95 – Spectrum Huffman codebooks parameters**

| Codebook number, i | unsigned_cb[i] | Dimension of codebook | LAV for codebook              | Codebook listed in Table |
|--------------------|----------------|-----------------------|-------------------------------|--------------------------|
| 0                  | -              | -                     | 0                             | -                        |
| 1                  | 0              | 4                     | 1                             | Table 4.A.2              |
| 2                  | 0              | 4                     | 1                             | Table 4.A.3              |
| 3                  | 1              | 4                     | 2                             | Table 4.A.4              |
| 4                  | 1              | 4                     | 2                             | Table 4.A.5              |
| 5                  | 0              | 2                     | 4                             | Table 4.A.6              |
| 6                  | 0              | 2                     | 4                             | Table 4.A.7              |
| 7                  | 1              | 2                     | 7                             | Table 4.A.8              |
| 8                  | 1              | 2                     | 7                             | Table 4.A.9              |
| 9                  | 1              | 2                     | 12                            | Table 4.A.10             |
| 10                 | 1              | 2                     | 12                            | Table 4.A.11             |
| 11                 | 1              | 2                     | 16 (with ESC 8191)            | Table 4.A.12             |
| 12                 | -              | -                     | (reserved)                    | -                        |
| 13                 | -              | -                     | perceptual noise substitution | -                        |
| 14                 | -              | -                     | intensity out-of-phase        | -                        |
| 15                 | -              | -                     | intensity in-phase            | -                        |
| 16                 | 1              | 2                     | 16 (w/o ESC 15)               | Table 4.A.12             |



|    |   |   |                    |              |
|----|---|---|--------------------|--------------|
| 17 | 1 | 2 | 16 (with ESC 31)   | Table 4.A.12 |
| 18 | 1 | 2 | 16 (with ESC 47)   | Table 4.A.12 |
| 19 | 1 | 2 | 16 (with ESC 63)   | Table 4.A.12 |
| 20 | 1 | 2 | 16 (with ESC 95)   | Table 4.A.12 |
| 21 | 1 | 2 | 16 (with ESC 127)  | Table 4.A.12 |
| 22 | 1 | 2 | 16 (with ESC 159)  | Table 4.A.12 |
| 23 | 1 | 2 | 16 (with ESC 191)  | Table 4.A.12 |
| 24 | 1 | 2 | 16 (with ESC 223)  | Table 4.A.12 |
| 25 | 1 | 2 | 16 (with ESC 255)  | Table 4.A.12 |
| 26 | 1 | 2 | 16 (with ESC 319)  | Table 4.A.12 |
| 27 | 1 | 2 | 16 (with ESC 383)  | Table 4.A.12 |
| 28 | 1 | 2 | 16 (with ESC 511)  | Table 4.A.12 |
| 29 | 1 | 2 | 16 (with ESC 767)  | Table 4.A.12 |
| 30 | 1 | 2 | 16 (with ESC 1023) | Table 4.A.12 |
| 31 | 1 | 2 | 16 (with ESC 2047) | Table 4.A.12 |

#### 4.6.4 Noiseless coding for the Fine Grain Scalability

##### 4.6.4.1 Tool description

BSAC stands for bit sliced arithmetic coding and is the name of a noiseless coder and bitstream formatter that provides a fine grain scalability and error resilience in the MPEG-4 General Audio (GA) coder. The BSAC noiseless coding module is an alternative to the AAC coding module, with all other modules of the AAC-based coder remaining unchanged. The BSAC noiseless coding is used to make the bitstream scalable and error resilience and further reduce the redundancy of the scalefactors and the quantized spectrum. The BSAC noiseless decoding process is split into 4 subclauses. Subclause 4.6.4.2 to 4.6.4.6 describes the detailed decoding process of the spectral data, the stereo or pns related data, the scalefactors and the coding band side information.

##### 4.6.4.2 Decoding of bit sliced spectral data (`bsac_spectral_data`)

###### 4.6.4.2.1 Description

BSAC uses the bit-slicing scheme of the quantized spectral coefficients in order to provide the fine grain scalability. And it encode the bit-sliced data using binary arithmetic coding scheme in order to reduce the average bits transmitted while suffering no loss of fidelity.

In BSAC scalable coding scheme, a quantized sequence is divided into coding bands, as shown in subclause 4.5.2.6.2.5. And, a quantized sequence is mapped into a bit-sliced sequence within a coding band. The noiseless coding of the sliced bits relies on the probability table of the coding band, the significance and the other contexts.

The significance of the bit-sliced data is the position of the sliced bit to be coded.

The flags, `sign_is_coded[]` are updated with coding the vectors from MSB to LSB. They are initialized to 0. And they are set to 1 when the sign of the quantized spectrum is coded.

The probability table for encoding the bit-sliced data within each coding band is included in the bistream element `cband_si_type` and transmitted starting from the lowest coding band and progressing to the highest coding band allocated to each layer. For the detailed description of the coding band side information `cband_si_type`, see subclause 4.6.4.5. Table 4.A.31 lists 23 probability tables which are used for encoding/decoding the bit-sliced data. The BSAC probability table consists of several sub-tables. Sub-tables are classified and chosen according to the significance and the coded upper bits as shown Table 4.A.54 to Table 4.A.75. Every sliced bit is arithmetic encoded using the probability value chosen among several possible sub-tables of BSAC probability table.

#### 4.6.4.2.2 Definitions

##### Bitstream elements:

|                                  |                                                                                                                                                                                                                                                                                                                                                                                                         |
|----------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>acod_sliced_bit[ch][g][i]</b> | Arithmetic codeword necessary for arithmetic decoding of the sliced bit. Using this decoded bit, we can reconstruct each bit value of the quantized spectral value. The actually reconstructed bit-value is dependent on the significance of the sliced bit.                                                                                                                                            |
| <b>acod_sign[ch][g][i]</b>       | Arithmetic codeword from binary arithmetic coding sign_bit. The probability of the "0" symbol is defined to 0.5 which uses 8192 as a 14-bit fixed-point number. sign_bit indicates sign bit for non-zero coefficient. A "1" indicates a negative coefficient, a "0" a positive one. When the bit value of the quantized signal is assigned 1 for the first time, sign bit is arithmetic coded and sent. |

##### Help elements:

|                                 |                                                                                                                                                                                                                      |
|---------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <i>layer</i>                    | scalability layer index                                                                                                                                                                                              |
| <i>snf</i>                      | significance of vector to be decoded.                                                                                                                                                                                |
| <i>ch</i>                       | channel index                                                                                                                                                                                                        |
| <i>nch</i>                      | the number of channel                                                                                                                                                                                                |
| <i>cur_snf [i]</i>              | current significance of the <i>i</i> -th vector. <i>cur_snf</i> [] is initialized to <i>Abit</i> [cband]. See subclause 4.5.2.6.2.5.                                                                                 |
| <i>maxsnf</i>                   | maximum of current significance of the vectors to be decoded. See subclause 4.5.2.6.2.5.                                                                                                                             |
| <i>snf</i>                      | significance index                                                                                                                                                                                                   |
| <i>layer_data_available()</i>   | function that returns "1" as long as each layer's bitstream is available, otherwise "0". In other words, it indicates whether the remaining bitstream of each layer is available or not.                             |
| <i>layer_group[layer]</i>       | indicates the group index of the spectral data to be added newly in the scalability layer. See subclause 4.5.2.6.2.5                                                                                                 |
| <i>layer_start_index[layer]</i> | indicates the index of the lowest spectral component to be added newly in the scalability layer. See subclause 4.5.2.6.2.5                                                                                           |
| <i>layer_end_index[layer]</i>   | indicates the index of the highest spectral component to be added newly in the scalability layer. See subclause 4.5.2.6.2.5                                                                                          |
| <i>start_index[g]</i>           | indicates the index of the lowest spectral component to be coded in the group <i>g</i>                                                                                                                               |
| <i>end_index[g]</i>             | indicates the index of the highest spectral component to be coded in the group <i>g</i>                                                                                                                              |
| <i>sliced_bit</i>               | the decoded value of the sliced bits of the quantized spectrum.                                                                                                                                                      |
| <i>sample[ch][g][i]</i>         | quantized spectral coefficients reconstructed from the decoded bit-sliced data of spectral line <i>i</i> in channel <i>ch</i> and group index <i>g</i> . See subclause 4.5.2.6.2.2                                   |
| <i>sign_is_coded[ch][g][i]</i>  | flag that indicates whether the sign of the <i>i</i> <sup>th</sup> quantized spectrum is already coded (1) or not (0) in channel <i>ch</i> and group index <i>g</i> .                                                |
| <i>sign_bit[ch][g][i]</i>       | sign bit for non-zero coefficient. A "1" indicates a negative coefficient, a "0" a positive one. When the bit value of the quantized signal is assigned 1 for the first time, sign bit is arithmetic coded and sent. |

#### 4.6.4.2.3 Decoding process

In BSAC encoder, the absolute values of quantized spectral coefficients is mapped into a bit-sliced sequence. These sliced bits are the symbols of the arithmetic coding. Every sliced bits are binary arithmetic coded from the lowest-frequency coefficient to the highest-frequency coefficient of the scalability layer, starting the Most Significant Bit(MSB) plane and progressing to the Least Significant Bit(LSB) plane. The arithmetic coding of the sign bits associated with non-zero coefficient follows that of the sliced bit when the bit-slice of the spectral coefficient is 1 for the first time.

For the case of multiple windows per block, the concatenated and possibly grouped and interleaved set of spectral coefficients is treated as a single set of coefficients that progress from low to high as described in subclause 4.5.2.6.2.6. This set of spectral coefficients may need to be de-interleaved after they are decoded. The spectral information for all scalefactor bands equal to or greater than *max\_sfb* is set to zero.

After all MSB data are encoded from the lowest frequency line to the highest, the same encoding process is repeated until LSB data is encoded or the layer data is not available.

The length of the available bitstream (*available\_len*[]) is initialized at the beginning of each layer as described in subclause 4.5.2.6.2.5. The estimated length of the codeword (*est\_cw\_len*) to be decoded is calculated from the

arithmetic decoding process as described in subclause 4.5.2.6.2.7. After the arithmetic decoding of a symbol, the length of the available bitstream should be updated by subtracting the estimated codeword length from it. We can detect whether the remaining bitstream of each layer is available or not by checking the available\_len.

The bit-sliced data is decoded with the probability, which is selected among values listed in from Table 4.A.54 to Table 4.A.75.

The probability value should be defined in order to arithmetic-code the symbols (the sliced bits). Binary probability table is made up of probability values (p0) of the symbol '0'. First of all, probability table is selected using cband\_si as shown in Table 4.A.29. Next, the sub-table is selected in the probability table according to the context such as the current significance of the spectral coefficient and the higher bit-slices that have been decoded. All the vector of the higher bit-slices, *higher\_bit\_vector* are initialized to 0 before the coding of the bit-sliced data is started. Whenever the bit-slice is coded, the vector, *higher\_bit\_vector* is updated as follows:

```
higher_bit_vector[ch][g][i] = (higher_bit_vector[ch][g][i]<<1) + decoded_bitslice;
if (higher_bit_vector[ch][g][i] > 15)
    higher_bit_vector[ch][g][i] = 15;
```

And, the probability (p0) is selected among the several values in the sub-table. In order to select one of the several probability values in the sub-table, the index of the probability should be decided. If the higher bit-slice vector is non-zero, the index of the probability (p0) is (higher\_bit\_vector[ch][g][i] - 1). Otherwise, it relies upon the sliced bits of successive non-overlapping 4 spectral data as shown in Table 4.A.32.

However if the available codeword size is smaller than 14, there is a constraints on the selected probability value as follows:

```
if (available_len < 14) {
    if (p0 < min_p0[available_len])
        p0 = min_p0[available_len];
    else if (p0 > max_p0[available_len])
        p0 = max_p0[available_len];
}
```

The minimum probability min\_p0[] and the maximum probability max\_p0[] is listed in Table 4.A.33 and Table 4.A.34.

Detailed arithmetic decoding procedure is described in this subclause 4.5.2.6.2.7.

There are 23 probability tables which can be used for encoding/decoding the bit-sliced data. 23 probability table are provided to cover the different statistics of the bit-slices. In order to transmit the probability table used in encoding process, the probability table is included in the syntax element, **cband\_si**. After **cband\_si** is decoded, the probability table is mapped from **cband\_si** using Table 4.A.31 and the decoding of the bit-sliced data shall be started.

The current significance of the spectral coefficient represents the bit-plane of the bit-slice to be decoded. Table 4.A.31 shows the MSB plane of the decoded sample according to **cband\_si**. Current significance, *cur\_snf[]* of all spectral coefficient within a coding band are initialized to the MSB plane. For the detailed initialization process, see subclause 4.5.2.6.2.5

The arithmetic decoding of the sign bit associated with non-zero coefficient follows the arithmetic decoding of the sliced bit when the bit-value of the quantized spectral coefficient is 1 for the first time, with a 1 indicating a negative coefficient and a 0 indicating a positive one. The flag, *sign\_is\_coded[]* represents whether the sign bit of the quantized spectrum has been decoded or not. Before the decoding of the bit-sliced data is started, all the *sign\_is\_coded* flags are set to 0. The flag, *sign\_is\_coded* is set to 1 after the sign bit is decoded. The decoding process of the sign bit can be summarized as follows:

```
i = the spectral line index
if (sample[ch][g][i] && !sign_is_coded[ch][g][i]) {
    arithmetic decoding of the sign bit;
```

```

    sign_is_coded[ch][g][i] = 1;
}

```

Decoded symbol need to be reconstructed to the sample. For the detailed reconstruction of the bit-sliced data, see **Reconstruction of the decoded sample from bit-sliced data** part in subclause 4.5.2.6.2.2.

#### 4.6.4.3 Decoding of stereo\_info, ms\_used and noise\_flag

##### 4.6.4.3.1 Descriptions

The BSAC scalable coding scheme includes the noiseless coding which is different from MPEG-4 AAC coding and further reduce the redundancy of the stereo-related data.

Decoding of the stereo-related data and Perceptual Noise Substitution(pns) data is depended on pns\_data\_present and stereo\_info which indicates the stereo mask. Since the decoded data is the same value with MPEG-4 AAC, the MPEG-4 AAC stereo-related and pns processing follows the decoding of the stereo-related data and pns data.

##### 4.6.4.3.2 Definitions

###### Bitstream elements:

|                                   |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                |
|-----------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>acode_ms_used[g][sfb]</b>      | arithmetic codeword from the arithmetic coding of ms_used which is one-bit flag per scalefactor band indicating that M/S coding is being used in window group g and scalefactor band sfb, as follows:<br><b>0</b> Independent<br><b>1</b> ms_used                                                                                                                                                                                                                                                                                                              |
| <b>acode_stereo_info[g][sfb]</b>  | arithmetic codeword from the arithmetic coding of stereo_info which is two-bit flag per scalefactor band indicating that M/S coding or Intensity coding is being used in window group g and scalefactor band sfb, as follows:<br><b>00</b> Independent<br><b>01</b> ms_used<br><b>10</b> Intensity_in_phase<br><b>11</b> Intensity_out_of_phase or noise_flag_is_used<br>Note : If ms_mask_present is 3, noise_flag_l and noise_flag_r are 0 value, then stereo_info is interpreted as out-of-phase intensity stereo regardless the value of pns_data_present. |
| <b>acode_noise_flag[g][sfb]</b>   | arithmetic codeword from the arithmetic coding of noise_flag which is 1-bit flag per scalefactor band indicating whether the perceptual noise substitution is used(1) or not(0) in window group g and scalefactor band sfb.                                                                                                                                                                                                                                                                                                                                    |
| <b>acode_noise_flag_l[g][sfb]</b> | arithmetic codeword from the arithmetic coding of noise_flag_l which is 1-bit flag per scalefactor band indicating whether the perceptual noise substitution is used(1) or not(0) in the left channel, window group g and scalefactor band sfb .                                                                                                                                                                                                                                                                                                               |
| <b>acode_noise_flag_r[g][sfb]</b> | arithmetic codeword from the arithmetic coding of noise_flag_r which is 1-bit flag per scalefactor band indicating whether the perceptual noise substitution is used(1) or not(0) in the right channel, window group g and scalefactor band sfb.                                                                                                                                                                                                                                                                                                               |
| <b>acode_noise_mode[g][sfb]</b>   | arithmetic codeword from the arithmetic coding of noise_mode which is two-bit flag per scalefactor band indicating that which noise substitution is being used in window group g and scalefactor band sfb, as follows:<br><b>00</b> Noise Subst L+R (independent)<br><b>01</b> Noise Subst L+R (correlated)<br><b>10</b> Noise Subst L+R (correlated, out-of-phase)<br><b>11</b> reserved                                                                                                                                                                      |

###### Help elements:

|            |                                      |
|------------|--------------------------------------|
| <i>ch</i>  | channel index                        |
| <i>g</i>   | group index                          |
| <i>sfb</i> | scalefacotor band index within group |

|                               |                                                                                                                                                                                                                                                                                                                      |
|-------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <i>layer</i>                  | scalability layer index                                                                                                                                                                                                                                                                                              |
| <i>nch</i>                    | the number of channel                                                                                                                                                                                                                                                                                                |
| <i>ms_mask_present</i>        | this two bit field indicates that the stereo mask is<br><b>00</b> Independent<br><b>01</b> 1 bit mask of <i>ms_used</i> is located in the layer sfb side information part.<br><b>10</b> All <i>ms_used</i> are ones<br><b>11</b> 2 bit mask of <i>stereo_info</i> is located in the layer sfb side information part. |
| <i>layer_group[layer]</i>     | indicates the group index of the spectral data to be added newly in the scalability layer. See subclause 4.5.2.6.2.5                                                                                                                                                                                                 |
| <i>layer_start_sfb[layer]</i> | indicates the index of the lowest scalefactor band index to be added newly in the scalability layer. See subclause 4.5.2.6.2.5                                                                                                                                                                                       |
| <i>layer_end_sfb[layer]</i>   | indicates the highest scalefactor band index to be added newly in the scalability layer. See subclause 4.5.2.6.2.5                                                                                                                                                                                                   |

#### 4.6.4.3.3 Decoding process

Decoding process of *ms\_mask\_present*, *noise\_flag* or *ms\_used* is depended on *pns\_data\_present*, number of channel and *ms\_mask\_present*. *pns\_data\_present* flag is conveyed as a element in syntax of *general\_header()*. *pns\_data\_present* indicates whether pns tool is used or not at each frame. *stereo\_info* indicates the stereo mask as follows :

- 00 Independent
- 01 1 bit mask of *ms\_used* is located in the layer sfb side information part.
- 10 All *ms\_used* are ones
- 11 2 bit mask of *stereo\_info* is located in the layer sfb side information part.

Detailed arithmetic decoding procedure is described in this subclause 4.5.2.6.2.7.

Decoding process is classified as follows :

- 1 channel, no pns data  
If the number of channel is 1 and pns data is not present, there is no bitstream elements related to stereo or pns.
- 1 channel, pns data  
If the number of channel is 1 and pns data is present, noise flag of the scalefactor bands between **pns\_start\_sfb** to **max\_sfb** is arithmetic decoded using model shown in Table 4.A.52. Perceptual noise substitution is done according to the decoded noise flag.
- 2 channel, *ms\_mask\_present*=0 (Independent), No pns data  
If *ms\_mask\_present* is 0 and pns data is not present, arithmetic decoding of *stereo\_info* or *ms\_used* is not needed.
- 2 channel, *ms\_mask\_present*=0 (Independent), pns data  
If *ms\_mask\_present* is 0 and pns data is present, noise flag for pns is arithmetic decoded using model shown in Table 4.A.52. Perceptual noise substitution of independent mode is done according to the decoded noise flag.
- 2 channel, *ms\_mask\_present*=2 (all *ms\_used*), pns data or no pns data  
All *ms\_used* values are ones in this case. So, M/S stereo processing of AAC is done at all scalefactor band. And naturally there can be no pns processing regardless of *pns\_data\_present* flag.
- 2 channel, *ms\_mask\_present*=1 (optional *ms\_used*), pns data or no pns data  
1 bit mask of *max\_sfb* bands of *ms\_used* is conveyed in this case. So, *ms\_used* is arithmetic decoded using the *ms\_used* model given in Table 4.A.50. M/S stereo processing of AAC is done or not according to the decoded *ms\_used*. And there is no pns processing regardless of *pns\_data\_present* flag
- 2 channel, *ms\_mask\_present*=3 (optional *ms\_used/intensity/pns*), no pns data  
At first, *stereo\_info* is arithmetic decoded using the *stereo\_info* model given in Table 4.A.51. *stereo\_info* is is two-bit flag per scalefactor band indicating that M/S coding or Intensity coding is being used in window group *g* and scalefactor band *sfb* as follows :  
  - 00** Independent
  - 01** *ms\_used*
  - 10** *Intensity\_in\_phase*
  - 11** *Intensity\_out\_of\_phase*

If stereo\_info is not 0, M/S stereo or intensity stereo of AAC is done with these decoded data. Since pns data is not present, we don't have to process pns.

- 2 channel, ms\_mask\_present=3 (optional ms\_used/intensity/pns), pns data  
stereo\_info is arithmetic decoded using the stereo\_info model given in Table 4.A.51.  
If stereo\_info is 1 or 2, M/S stereo or intensity stereo processing of AAC is done with these decoded data and there is no pns processing.  
If stereo\_info is 3 and scalefactor band is larger than or equal to pns\_start\_sfb, noise flag for pns is arithmetic decoded using model given in Table 4.A.52. And then if the both noise flags of two channel are 1, noise substitution mode is arithmetic decoded using model given in Table 4.A.53. The perceptual noise is substituted or out\_of\_phase intensity stereo processing is done according to the substitution mode. Otherwise, the perceptual noise is substituted only if noise flag is 1.  
If stereo\_info is 3 and scalefactor band is smaller than pns\_start\_sfb, out\_of\_phase intensity stereo processing is done.

#### 4.6.4.4 Decoding of scalefactors, noise energy and intensity stereo position

##### 4.6.4.4.1 Description

The BSAC scalable coding scheme includes the noiseless coding which is different from AAC and further reduce the redundancy of the scalefactors.

The max\_scalefactor is coded as an 8 bit unsigned integer. The scalefactors are differentially coded relative to the max\_scalefactor value and then Arithmetic coded using the differential scalefactor model.

##### 4.6.4.4.2 Definitions

###### Bitstream element:

|                                                  |                                                                             |
|--------------------------------------------------|-----------------------------------------------------------------------------|
| <b>acode_scf[ch][g][sfb]</b>                     | Arithmetic codeword from the coding of the differential scalefactors.       |
| <b>acode_max_noise_energy [ch]</b>               | Arithmetic codeword from the coding of the maximum of the noise energies.   |
| <b>acode_dpcm_noise_energy_index[ch][g][sfb]</b> | Arithmetic codeword from the coding of the differential noise energy index. |
| <b>acode_is_position_index[g][sfb]</b>           | Arithmetic codeword from the coding of the intensity stereo position index. |

###### Help elements:

|                                            |                                                                                                                                |
|--------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------|
| <i>ch</i>                                  | channel index                                                                                                                  |
| <i>g</i>                                   | group index                                                                                                                    |
| <i>sfb</i>                                 | scalefactor band index within group                                                                                            |
| <i>layer</i>                               | scalability layer index                                                                                                        |
| <i>nch</i>                                 | the number of channel                                                                                                          |
| <i>layer_group[layer]</i>                  | indicates the group index of the spectral data to be added newly in the scalability layer. See subclause 4.5.2.6.2.5           |
| <i>layer_start_sfb[layer]</i>              | indicates the index of the lowest scalefactor band index to be added newly in the scalability layer. See subclause 4.5.2.6.2.5 |
| <i>layer_end_sfb[layer]</i>                | indicates the highest scalefactor band index to be added newly in the scalability layer. See subclause 4.5.2.6.2.5             |
| <i>scf[ch][g][sfb]</i>                     | indicates the scalefactors.                                                                                                    |
| <i>max_noise_energy[ch]</i>                | indicates the maximum of the noise energy.                                                                                     |
| <i>dpcm_noise_energy_index[ch][g][sfb]</i> | indicates the differential noise energy index.                                                                                 |
| <i>is_position_index[g][sfb]</i>           | indicates the intensity stereo position index.                                                                                 |

#### 4.6.4.4.3 Decoding process

The spectral coefficients are divided into scalefactor bands that contain a multiple of 4 quantized spectral coefficients. Each scalefactor band has a scalefactor.

The differential scalefactor index is arithmetic-decoded using the arithmetic model given in Table 4.A.30. The arithmetic model of the scalefactor for the base layer is given as a 3 bit unsigned integer bitstream element, **base\_scf\_model**. The arithmetic model of the scalefactor for the enhancement layers is given as a 3 bit unsigned integer bitstream element, **enh\_scf\_model**.

For all scalefactors the difference to the offset value is arithmetic-decoded. All scalefactors are calculated from the difference and the offset value. The offset value is given explicitly as a 8 bit PCM in the bitstream element **max\_scalefactor[ch]**. Detailed arithmetic decoding procedure is described in this subclause 4.5.2.6.2.7.

The following pseudo code describes how to decode the scalefactors  $scf[ch][g][sfb]$  in base layer and each enhancement layer:

```
for (ch =0; ch<nch; ch++) {
    g = layer_group[ch][g][sfb];
    for (sfb = layer_start_sfb[layer]; sfb < layer_end_sfb[layer]; sfb++) {
        diff_scf = arithmetic_decoding();
        scf[ch][g][sfb] = max_scalefactor[ch] - diff_scf;
    }
}
```

If noise substitution coding is active for a particular group and scalefactor band, a noise energy value is transmitted instead of the scalefactor of the respective channel.

Noise energies are arithmetic-coded of differential values. For all noise energies the difference to the offset value is arithmetic-decoded. All noise energies are calculated from the difference and the offset value. The offset value,  $max\_noise\_energy[ch]$  is arithmetic-decoded before the first differential noise energy is decoded.

Noise substitution decoding process is same as the PNS part of MPEG-4 Audio General Audio. The noise energy decoding in each layer is defined by the following pseudo code:

```
for (ch =0; ch<nch; ch++) {
    g = layer_group[ch][g][sfb];
    for (sfb = layer_start_sfb[layer]; sfb < layer_end_sfb[layer]; sfb++) {
        if (noise_flag[ch][g][sfb]) {
            dpcm_noise_energy_index[ch][g][sfb] = arithmetic_decoding();
            noise_nrg[ch][g][sfb] = max_noise_energy[ch] - dpcm_noise_energy[ch][g][sfb];
        }
    }
}
```

The direction information for the intensity stereo decoding is represented by an "intensity stereo position" value indicating the relation between left and right channel scaling. If intensity stereo is active for a particular group and scalefactor band, an intensity stereo position value is transmitted in stead of the scalefactor of the right channel.

When intensity positions are arithmetic-coded, the same arithmetic model is used. The intensity decoding process is same as the intensity stereo of MPEG-4 Audio General Audio. The intensity stereo position decoding in each layer is defined by the following pseudo code:

```
g = layer_group[1][g][sfb]
for (sfb = layer_start_sfb[layer]; sfb < layer_end_sfb[layer]; sfb++) {
    if (stereo_info[g][sfb] && ch==1) {
        is_position_index[g][sfb] = arithmetic_decoding();
        if (is_position_sign[g][sfb]%2)
            is_position[g][sfb] = -(int)((is_position_index[g][sfb]+1)/2);
        else
            is_position[g][sfb] = (int)(is_position_index[g][sfb]/2);
    }
}
```

```

}
}

```

#### 4.6.4.5 Decoding of coding band side information

##### 4.6.4.5.1 Description

In BSAC scalable coding scheme, the spectral coefficients are divided into coding bands which contain 32 quantized spectral coefficients for the noiseless coding. Coding bands are the basic units used for the noiseless coding. The set of bit-sliced sequence is divided into coding bands. The MSB plane and the probability table of each coding band are included in this layer coding band side information, **cband\_si** as shown in Table 4.A.31. The coding band side informations of each layer are transmitted starting from the lowest coding band (*layer\_start\_cband[layer]*) and progressing to the highest coding band (*layer\_end\_cband[layer]*). For all *cband\_si*, it is arithmetic-coded using the arithmetic model as given in Table 4.A.29.

##### 4.6.4.5.2 Definitions

###### Bitstream element:

**acode\_cband\_si[ch][g][cband]** Arithmetic codeword from the arithmetic coding of **cband\_si** for each coding-band.

###### Help elements:

|                                 |                                                                                                                      |
|---------------------------------|----------------------------------------------------------------------------------------------------------------------|
| <i>g</i>                        | group index                                                                                                          |
| <i>cband</i>                    | coding band index within group                                                                                       |
| <i>ch</i>                       | channel index                                                                                                        |
| <i>nch</i>                      | the number of channel                                                                                                |
| <i>layer_group[layer]</i>       | indicates the group index of the spectral data to be added newly in the scalability layer. See subclause 4.5.2.6.2.5 |
| <i>layer_start_cband[layer]</i> | indicates the lowest coding band index to be added newly in the scalability layer. See subclause 4.5.2.6.2.5         |
| <i>layer_end_cband[layer]</i>   | indicates the highest coding band index to be added newly in the scalability layer. See subclause 4.5.2.6.2.5        |

##### 4.6.4.5.3 Decoding process

*cband\_si* is arithmetic-coded using the arithmetic model as given in Table 4.A.29. The arithmetic model used for coding *cband\_si* is dependent on a 5-bit unsigned integer in the bitstream element, **cband\_si\_type** as shown in Table 4.A.29. And, the largest value of the decodable *cband\_si* is given in Table 4.A.29. If the decoded *cband\_si* larger than this value, it can be considered that there was a bit-error in the bitstream. Detailed arithmetic decoding procedure is described in this subclause 4.5.2.6.2.7.

The following pseudo code describes how to decode the *cband\_si* *cband\_si[ch][g][cband]* in base layer and each enhancement layer.

```

g = layer_group[layer];
for ( ch=0; ch<nch; ch++ ) {
  for ( cband=layer_start_cband[g][layer]; cband<layer_cband[g][layer+1]; cband++ ) {
    cband_si[ch][g][cband] = arithmetic_decoding();
    if ( cband_si[ch][g][cband] > largest_cband_si )
      bit_error_is_generated;
  }
}

```

where, *layer\_cband[g][layer]* is the start coding band and *layer\_cband[g][layer+1]* is the end coding band for decoding the arithmetic model index in each layer.



4.6.4.6 Segmented binary arithmetic coding (SBA)

4.6.4.6.1 Tool Description

Segmented Binary Arithmetic Coding (SBA) is based on the fact that the arithmetic codewords can be partitioned at known positions so that these codewords can be decoded independent of any error within other sections. Therefore, this tool avoids error propagation to those sections. The arithmetic coding should be initialized at the beginning of these segments and terminated at the end of these segments in order to localize the arithmetic codewords. This tool is activated if the syntax element, **sba\_mode** is 1. And this flag should be set to 1 if the BSAC is used in the error-prone environment.

4.6.4.6.2 Definitions

There is no definition because only the initialization and termination process are added at the beginning and the end of the segments in order to localize the arithmetic codewords.

4.6.4.6.3 Decoding process

The arithmetic coding is terminated at the end of the segments, and re-initialized at the beginning of the next segment. The segment is made up of the scalability layers. `terminal_layer[layer]` indicates whether each layer is the last layer of the segment, which is set as follows :

```

for (layer = 0; layer < (top_layer+slayer_size-1); layer++) {
    if (layer_start_cband[layer] != layer_start_cband[layer+1])
        terminal_layer[layer] = 1;
    else
        terminal_layer[layer] = 0;
}
    
```

where, `toplayer` is the top layer to be encoded, `layer_max_cband[]` are the maximum coding band limit to be encoded and `slayer_size` is the sub-layer size of the base layer. Figure 4.24 shows an example of the segmented bitstream to be made in the encoder.

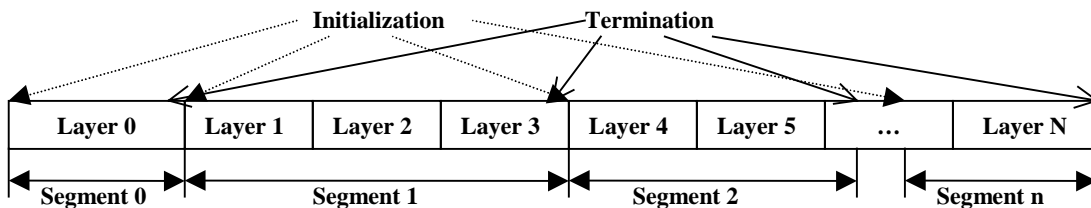


Figure 4.24 – The structure of SBA coded bitstream

In the decoder, the bitstream of each layer is split from the total bitstream. If the previous layer is the last of the segment, the split bitstream is stored in the independent buffer and arithmetic decoding process is re-initialized. Otherwise, the split bitstream is concatenated to that of the previous layer and used for arithmetic decoding sequentially.

In order to do the arithmetic decoding perfectly, 32-bit zero value should be concatenated to the split bitstream if the layer is the last of the segment. Figure 4.25 shows an example of the bitstream splitting and zero stuffing in decoder part.

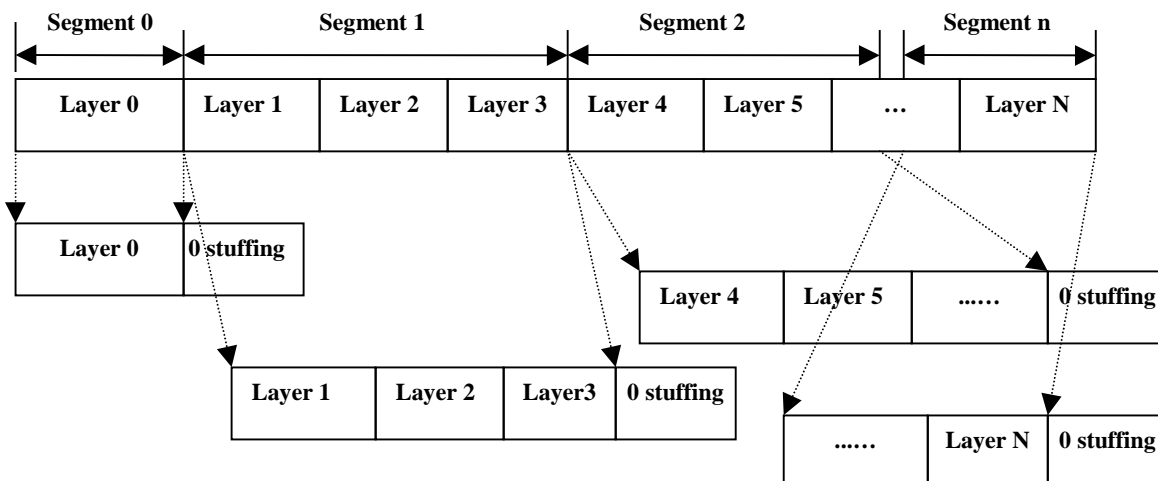


Figure 4.25 – Decoding of SBA bitstream

## 4.6.5 Interleaved vector quantization

### 4.6.5.1 Tool description

This process generates flattened MDCT spectrum using vector quantization. This quantization tool provides high coding gain, even at lower bitrates. Bitstream for this quantizer has a simple fixed-length structure, thus it is robust against transmission channel errors.

The decoding process consists of vector quantization part and reconstruction part. In the vector quantization part, subvectors are specified by codevector index. Then, subvectors are interleaved and combined into one output vector (see Figure 4.26).

### 4.6.5.2 Definitions

#### Inputs:

|                        |                                                                                        |
|------------------------|----------------------------------------------------------------------------------------|
| <b>fb_shift[][]</b>    | Syntax element indicating the active frequency band of the adaptive bandwidth control. |
| <b>index0[]</b>        | bitstream element indicating the codevector number of codebook 0                       |
| <b>index1[]</b>        | bitstream element indicating the codevector number of codebook 1                       |
| <b>window_sequence</b> | bitstream element indicating window sequence type                                      |
| <i>side_info_bits</i>  | number of bits for side information                                                    |
| <i>bitrate</i>         | system parameter indicating bitrate                                                    |
| <i>used_bits</i>       | number of bits used by variable bit-rate tool, such as long term prediction tool       |
| <i>lyr</i>             | indicates enhancement layer number. Number 0 is assigned for the base layer.           |

#### Outputs:

|                  |                                 |
|------------------|---------------------------------|
| <i>x_flgaf[]</i> | reconstructed MDCT coefficients |
|------------------|---------------------------------|

#### Parameters:

|                   |                                                      |
|-------------------|------------------------------------------------------|
| <i>FRAME_SIZE</i> | frame length                                         |
| <i>MAXBIT</i>     | maximum bits for shape codebook index representation |

|                     |                                                                                                                             |
|---------------------|-----------------------------------------------------------------------------------------------------------------------------|
| <i>N_CH</i>         | number of channels                                                                                                          |
| <i>N_DIV</i>        | number of subvectors                                                                                                        |
| <i>N_SF</i>         | number of subframes in a frame                                                                                              |
| <i>sp_cv0</i> [][]  | shape codebook of conjugate channel 0 (Elements are given in Table 4.A.19, Table 4.A.21, Table 4.A.23, Table 4.A.25)        |
| <i>sp_cv1</i> [][]  | shape codebook of conjugate channel 1 (Elements are given in tables Table 4.A.20, Table 4.A.22, Table 4.A.24, Table 4.A.26) |
| <i>SP_CB_SIZE</i>   | shape codebook size                                                                                                         |
| <i>shape_index0</i> | points the selected codevector of MDCT shape codebook 0                                                                     |
| <i>shape_index1</i> | points the selected codevector of MDCT shape codebook 1                                                                     |
| <i>pol0</i>         | negates the selected codevector of MDCT shape codebook 0                                                                    |
| <i>pol1</i>         | negates the selected codevector of MDCT shape codebook 1                                                                    |

#### 4.6.5.3 Parameter settings

The shape codebook vectors of the flattened MDCT coefficients, *sp\_cv0*[][] and *sp\_cv1*[][] are listed in Annex A.

Parameters are set initially as listed below:

```

MAXBIT_SHAPE = 5
MAXBIT       = MAXBIT_SHAPE + 1
SP_CB_SIZE   = (1<<MAXBIT_SHAPE)
    
```

#### 4.6.5.4 Decoding process

##### 4.6.5.4.1 Initializations

Based on the *bits\_available\_vq* defined in 4.5.2.5.3.2,

*N\_DIV* and the length of each subvector, *length*[], are calculated by

*N\_DIV* and the length of each subvector, *length*[] is calculated by

```

N_DIV = ((int)((bits_available_vq + MAXBIT*2-1)/(MAXBIT*2)))

for(idiv=0; idiv<ntt_N_DIV; idiv++){
    length[idiv] = ((int) (N_FR*qsample)*N_SF*N_CH+N_DIV-1-idiv) / N_DIV;
}
    
```

where *N\_FR* is the number of samples in a subframe and *qsample* is defined in subclause 4.6.5.4.4

Detailed calculation of *side\_info\_bits* is described in subclause 4.5.2.5.3 with *bits\_for\_side\_information*.

If the codevector length, *length*[], exceeds the number of codevector elements which are described in the subclause 4.A.4, the undefined elements of *sp\_cv0*[] and *sp\_cv1*[] (i.e. elements beyond the defined area) are set to zero.

##### 4.6.5.4.2 Index unpacking

The quantization index consists of the polarity and shape code information. So in the first stage of the inverse quantization, the input indices are unpacked, and the polarities and shapes are extracted.

The extracting of the polarities is described as follows:

```

for (idiv=0; idiv<N_DIV; idiv++){
    pol0[idiv] = 2 * (index0 [idiv] / SP_CB_SIZE) - 1;
    pol1[idiv] = 2 * (index1 [idiv] / SP_CB_SIZE) - 1;
}
    
```

where

pol0[]: polarity of conjugate channel 0  
 pol1[]: polarity of conjugate channel 1

The shape code extraction is described as follows:

```
for (idiv=0; idiv<N_DIV; idiv++){
  index_shape0[idiv] = index0 [idiv] % SP_CB_SIZE;
  index_shape1[idiv] = index1 [idiv] % SP_CB_SIZE;
}
```

#### 4.6.5.4.3 Reconstruction

Output coefficients are reconstructed as follows:

```
for (idiv=0; idiv<N_DIV; idiv++){
  for (icv=0; icv<length[idiv]; icv++){
    if ((icv<length[0]-1) &&
        ((N_DIV%(N_SF*N_CH)==0 && (N_SF*N_CH)>1) || ((N_SF*N_CH)&0x1)==0))
      itmp = ((idiv+icv)%N_DIV)+icv*N_DIV;
    else
      itmp = idiv + icv * N_DIV;
    ismp = itmp / (N_SF*N_CH) + ((itmp % (N_SF*N_CH)) * (FRAME_SIZE / (N_SF*N_CH)));
    x_flat_tmp[ismp] =
      (pol0[idiv]*sp_cv0[index_shape0[idiv]][icv]
       + pol1[idiv]*sp_cv1[index_shape1[idiv]][icv]) / 2;
  }
}
```

where

icv: indicates sample number in the shape code vector

idiv: indicates interleaved-division subvector

ismp: indicates sample number in the subframe

itmp: an integer

#### 4.6.5.4.4 Adaptive active band selection

This procedure selects the active band (see Figure 4.27) depending on the **fb\_shift** parameter.

If lyr=0, the active band is fixed as listed below:

```
bandUpper_i = 95 * BPS / ISAMPF;
bandUpper_i = min(100000, bandUpper_i);
bandUpper_i *= 16384;
bandUpper_i += 1562;
bandUpper_i /= 3125;
qsample = (double)(bandupper_i)/524288.;
AC_TOP[lyr][i_ch][0] = qsample;
AC_BTM[lyr][i_ch][0] = 0.0;
```

where ISAMP is an integer sampling frequency in [kHz] truncated from the standard frequency values listed in the right column of Table 4.55 and BPS is bitrate in [bits/s/ch] based on the byte-aligned bits for a frame and it equals  $(\text{int})(((\text{FRAME\_SIZE} * \text{bitrate} / \text{sampling\_frequency}) / 8 + 0.5) * 8) / \text{sampling\_frequency} / \text{FRAME\_SIZE} / \text{N\_CH}$ .

If lyr is greater or equal '1', the upper limit of the quantization bandwidth of a certain layer is defined as follows using the accumulated bitrate totalbps up to the layer.

```
totalbps=bpsbase;
for(iscl0=1; iscl0<=lyr; iscl0++){
  totalbps += BPS_SCL[iscl0];
}

upperlimit_i = (totalbps* 100) / ISAMPF;
```

```

upperlimit_i = min(100000, upperlimit_i);
upperlimit_i *= 16384;
upperlimit_i += 1562;
upperlimit_i /= 3125;
upperlimit = (double)(upperlimit_i)/524288.;

```

UPPER\_BOUNDARY and LOWER\_BOUNDARY of the quantization bandwidth is defined in the tables of AC\_TOP[lyr][i\_ch][fb\_shift] and AC\_BTM[lyr][i\_ch][fb\_shift] as follows, depending on the band selection code and the value of bitrate per channel BPS\_SCL[lyr] in [bits/s/ch] for each enhancement layer. Note that BPS\_SCL[lyr] should be based on the byte-aligned bits for a frame and it equals

$(\text{int})(((\text{FRAME\_SIZE} * \text{bitrate}/\text{sampling\_frequency})/8+0.5)*8)/\text{sampling\_frequency}/\text{FRAME\_SIZE}/\text{N\_CH}$ .

```

qsample_i = (BPS_SCL[lyr]* 130) / ISAMPF;
qsample_i = min(100000, qsample_i);
qsample_i *= 16384;
qsample_i += 1562;
qsample_i /= 3125;
bias_i = (upperlimit_i- qsample_i)/4;
if(qsample_i < bias_i){
    bias_i = upperlimit_i/4;
    qsample_i = bias_i;
}
bias = (double) bias_i;
qsample = (double)(qsample_i)/524288./ * 16384*32 */
if(bias <= 0.0) bias =0.0;
for(i_ch=0; i_ch<N_CH; i_ch++){
    AC_TOP[lyr][i_ch][0] = qsample;
    AC_BTM[lyr][i_ch][0] = 0.0;
    AC_TOP[lyr][i_ch][1] = qsample+bias;
    AC_BTM[lyr][i_ch][1] = bias;
    AC_TOP[lyr][i_ch][2] = qsample + bias*2;
    AC_BTM[lyr][i_ch][2] = bias*2;
    AC_TOP[lyr][i_ch][3] = qsample + bias*3;
    AC_BTM[lyr][i_ch][3] = bias*3;
}

```

where AC\_BTM and AC\_TOP is the bottom and top frequency of the active band, respectively. Values are ranged from 0 to 1.0 (i.e. 1.0 means the highest frequency).

The lower and upper boundaries in MDCT domain are calculated as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    LOWER_BOUNDARY[lyr][i_ch] = AC_BTM[lyr][i_ch][fb_shift] * N_FR;
    UPPER_BOUNDARY[lyr][i_ch] = AC_TOP[lyr][i_ch][fb_shift] * N_FR;
}

```

Then, the output x\_flat[] is copied from x\_flat\_tmp[] as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    for (isf=0; isf<N_SF; isf++){
        for (ismp=0; ismp<LOWER_BOUNDARY[lyr][i_ch]; ismp++){
            x_flat[ismp+(isf+i_ch*N_SF)*N_FR] = 0.;
        }
        for (ismp=LOWER_BOUNDARY[lyr][i_ch]; ismp<UPPER_BOUNDARY[lyr][i_ch]; ismp++){
            ismp2 = ismp - LOWER_BOUNDARY[lyr][i_ch];
            x_flat[ismp+(isf+i_ch*N_SF)*N_FR] = x_flat_tmp[ismp2+(isf+i_ch*N_SF)*N_FR];
        }
        for (ismp=UPPER_BOUNDARY[lyr][i_ch]; ismp<N_FR; ismp++){
            x_flat[ismp+(isf+i_ch*N_SF)*N_FR] = 0.;
        }
    }
}

```

4.6.5.5 Diagrams

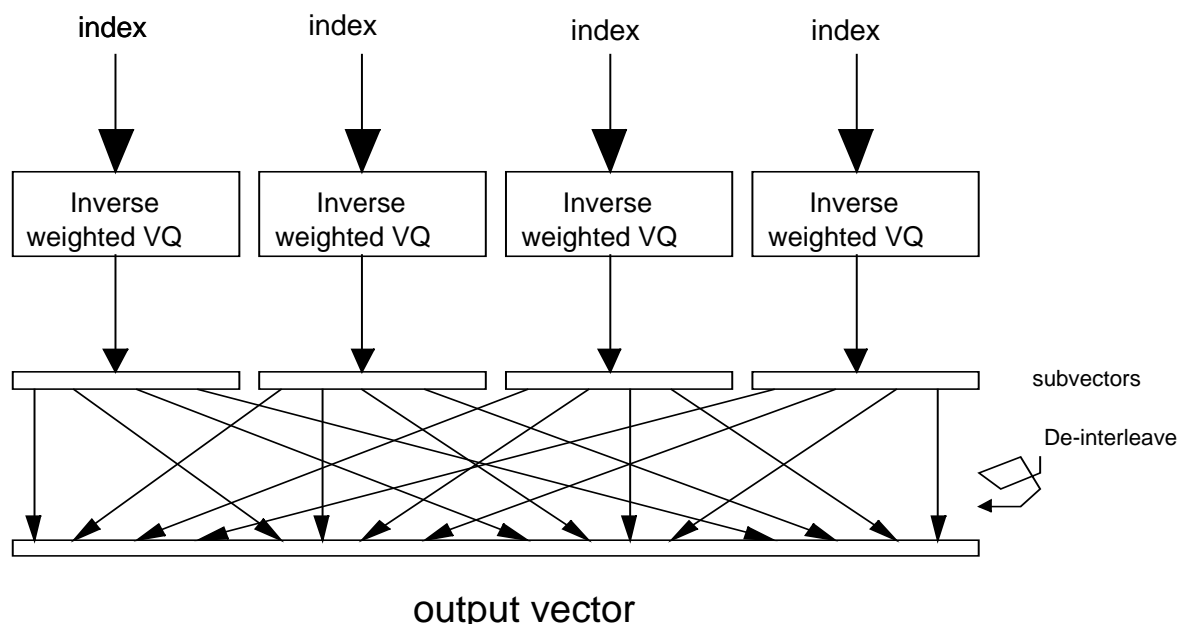


Figure 4.26 – Decoding process of interleaved vector quantization tool.

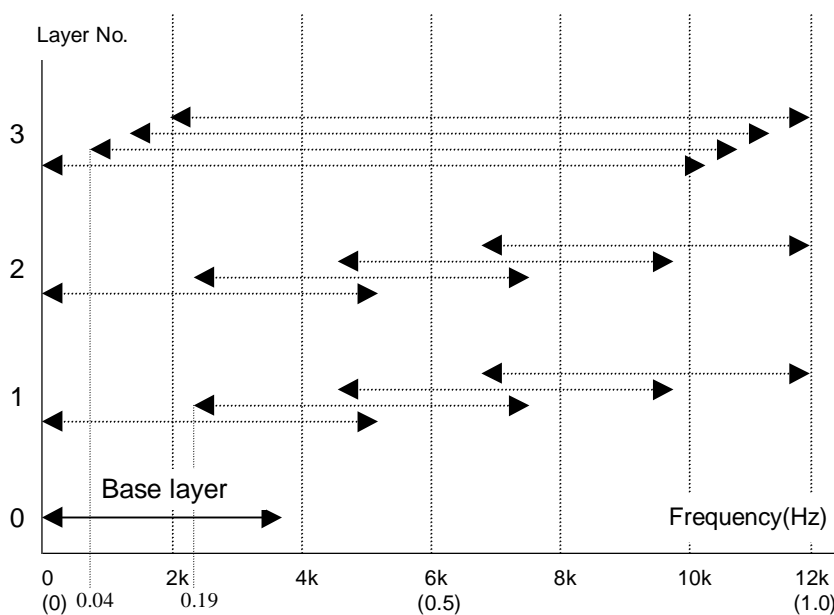


Figure 4.27 – Example of adaptive active band selection.  
 (Sampling\_Frequency=24kHz, Total bitrate=40kbit/s(8+8+8+16kbps)  
 In this case,  $q_{sample} = 0.43$ ,  $bias = 0.19$  for layer 1 and layer 2  
 $q_{sample} = 0.87$ ,  $bias = 0.04$  for layer 3)

4.6.6 Frequency domain prediction

(similar to ISO/IEC 13818-7)

4.6.6.1 Tool description

Prediction is used for an improved redundancy reduction and is especially effective in case of more or less stationary parts of a signal which belong to the most demanding parts in terms of required bitrate. Prediction can be applied to every channel using an intra channel (or mono) predictor which exploits the auto-correlation between the spectral components of consecutive frames. Because a window\_sequence of type EIGHT\_SHORT\_SEQUENCE indicates signal changes, i.e. non-stationary signal characteristics, prediction is only used if window\_sequence is of type ONLY\_LONG\_SEQUENCE, LONG\_START\_SEQUENCE or LONG\_STOP\_SEQUENCE. The use of the prediction tool is object type / profile dependent. See subpart 1 for detailed information on the MPEG-4 Audio object types and profiles.

For each channel prediction is applied to the spectral components resulting from the spectral decomposition of the filterbank. For each spectral component up to limit specified by PRED\_SFB\_MAX, there is one corresponding predictor resulting in a bank of predictors, where each predictor exploits the auto-correlation between the spectral component values of consecutive frames.

The overall coding structure using a filterbank with high spectral resolution implies the use of backward adaptive predictors to achieve high coding efficiency. In this case, the predictor coefficients are calculated from preceding quantized spectral components in the encoder as well as in the decoder and no additional side information is needed for the transmission of predictor coefficients - as would be required for forward adaptive predictors. A second order backward-adaptive lattice structure predictor is used for each spectral component, so that each predictor is working on the spectral component values of the two preceding frames. The predictor parameters are adapted to the current signal statistics on a frame by frame base, using an LMS based adaptation algorithm. If prediction is activated, the quantizer is fed with a prediction error instead of the original spectral component, resulting in a coding gain.

In order to keep storage requirements to a minimum, predictor state variables are quantized prior to storage.

Note: The frequency domain prediction tool can be used only for AudioObjectType AAC Main, but not for AAC LC, AAC SSR or AAC LTP.

4.6.6.2 Definitions

- predictor\_data\_present** 1 bit indicating whether prediction is used in current frame (1) or not (0) (always present for ONLY\_LONG\_SEQUENCE, LONG\_START\_SEQUENCE and LONG\_STOP\_SEQUENCE).
- predictor\_reset** 1 bit indicating whether predictor reset is applied in current frame (1) or not (0) (only present if **predictor\_data\_present** flag is set).
- predictor\_reset\_group\_number** 5 bit number specifying the reset group to be reset in current frame if predictor reset is enabled (only present if **predictor\_reset** flag is set).
- prediction\_used** 1 bit for each scalefactor band (sfb) where prediction can be used indicating whether prediction is switched on (1) / off (0) in that sfb. If **max\_sfb** is less than PRED\_SFB\_MAX then for i greater than or equal to max\_sfb, prediction\_used[i] is not transmitted and therefore is set to off (0) (only present if **predictor\_data\_present** flag is set).

The following table specifies the upper limit of scalefactor bands up to which prediction can be used:

Table 4.96 – Upper limit of scalefactor bands

| Sampling Frequency (Hz) | Pred_SFB_MAX | Number of Predictors | Maximum Frequency using Prediction (Hz) |
|-------------------------|--------------|----------------------|-----------------------------------------|
| 96000                   | 33           | 512                  | 24000.00                                |
| 88200                   | 33           | 512                  | 22050.00                                |
| 64000                   | 38           | 664                  | 20750.00                                |
| 48000                   | 40           | 672                  | 15750.00                                |
| 44100                   | 40           | 672                  | 14470.31                                |
| 32000                   | 40           | 672                  | 10500.00                                |
| 24000                   | 41           | 652                  | 7640.63                                 |

|       |    |     |         |
|-------|----|-----|---------|
| 22050 | 41 | 652 | 7019.82 |
| 16000 | 37 | 664 | 5187.50 |
| 12000 | 37 | 664 | 3890.63 |
| 11025 | 37 | 664 | 3574.51 |
| 8000  | 34 | 664 | 2593.75 |

This means that at 48 kHz sampling rate prediction can be used in scalefactor bands 0 through 39. According to Table 5.8 these 40 scalefactor bands include the MDCT lines 0 through 671, hence resulting in max. 672 predictors.

#### 4.6.6.3 Decoding process

For each spectral component up to the limit specified by PRED\_SFB\_MAX of each channel there is one predictor. Prediction is controlled on a `single_channel_element` or `channel_pair_element` basis by the transmitted side information in a two step approach, first for the whole frame at all and then conditionally for each scalefactor band individually, see subclause 4.6.6.3.1. The predictor coefficients for each predictor are calculated from preceding reconstructed values of the corresponding spectral component. The details of the required predictor processing are described in subclause 4.6.6.3.2. At the start of the decoding process, all predictors are initialized. The initialization and a predictor reset mechanism are described in subclause 4.6.6.3.3.

##### 4.6.6.3.1 Predictor side information

The following description is valid for either one `single_channel_element` or one `channel_pair_element` and has to be applied to each such element. For each frame the predictor side information has to be extracted from the bitstream to control the further predictor processing in the decoder. In case of a `single_channel_element` the control information is valid for the predictor bank of the channel associated with that element. In case of a `channel_pair_element` there are the following two possibilities: If **common\_window** == 1 then there is only one set of the control information which is valid for the two predictor banks of the two channels associated with that element. If **common\_window** == 0 then there are two sets of control information, one for each of the two predictor banks of the two channels associated with that element.

If `window_sequence` is of type `ONLY_LONG_SEQUENCE`, `LONG_START_SEQUENCE` or `LONG_STOP_SEQUENCE`, the **predictor\_data\_present** bit is read. If this bit is not set (0) then prediction is switched off at all for the current frame and there is no further predictor side information present. In this case the **prediction\_used** bit for each scalefactor band stored in the decoder has to be set to zero. If the **predictor\_data\_present** bit is set (1) then prediction is used for the current frame and the **predictor\_reset** bit is read which determines whether predictor reset is applied in the current frame (1) or not (0). If **predictor\_reset** is set then the next 5 bits are read giving a number specifying the group of predictors to be reset in the current frame, see also subclause 4.6.6.3.3 for the details. If the **predictor\_reset** is not set then there is no 5 bit number in the bitstream. Next, the **prediction\_used** bits are read from the bitstream, which control the use of prediction in each scalefactor band individually, i.e. if the bit is set for a particular scalefactor band, then prediction is enabled for all spectral components of this scalefactor band and the quantized prediction error of each spectral component is transmitted instead of the quantized value of the spectral component. Otherwise, prediction is disabled for this scalefactor band and the quantized values of the spectral components are transmitted.

##### 4.6.6.3.2 Predictor processing

###### 4.6.6.3.2.1 General

The following description is valid for one single predictor and has to be applied to each predictor. A second order backward adaptive lattice structure predictor is used. Figure 4.28 shows the corresponding predictor flow graph on the decoder side. In principle, an estimate  $x_{\text{est}}(n)$  of the current value of the spectral component  $x(n)$  is calculated from preceding reconstructed values  $x_{\text{rec}}(n-1)$  and  $x_{\text{rec}}(n-2)$ , stored in the register elements of the predictor structure, using the predictor coefficients  $k_1(n)$  and  $k_2(n)$ . This estimate is then added to the quantized prediction error  $e_q(n)$  reconstructed from the transmitted data resulting in the reconstructed value  $x_{\text{rec}}(n)$  of the current spectral component  $x(n)$ . Figure 4.29 shows the block diagram of this reconstruction process for one single predictor.



Due to the realization in a lattice structure, the predictor consists of two so-called basic elements which are cascaded. In each element, the part  $x_{est,m}(n)$ ,  $m=1, 2$  of the estimate is calculated according to

$$X_{est,m}(n) = b \cdot k_m(n) \cdot r_{q,m-1}(n-1),$$

where

$$r_{q,0}(n) = ax_{rec}(n),$$

$$r_{q,1}(n) = a(r_{q,0}(n-1) - b \cdot k_1(n) \cdot e_{q,0}(n))$$

and  $e_{q,m}(n) = e_{q,m-1}(n) - x_{est,m}(n)$ .

Hence, the overall estimate results to:  $x_{est}(n) = x_{est,1}(n) + x_{est,2}(n)$

The constants

$$a \text{ and } b, \quad 0 < a, b \leq 1$$

are attenuation factors which are included in each signal path contributing to the recursivity of the structure for the purpose of stabilization. By this means, possible oscillations due to transmission errors or drift between predictor coefficients on the encoder and decoder side due to numerical inaccuracy can be faded out or even prevented.

In the case of stationary signals and with  $a = b = 1$ , the predictor coefficient of element  $m$  is calculated by

$$k_m = \frac{E[e_{q,m-1}(n) \cdot r_{q,m-1}(n-1)]}{\frac{1}{2} \cdot (E[e_{q,m-1}^2(n)] + E[r_{q,m-1}^2(n-1)])}, \quad m = 1, 2 \quad \text{and} \quad e_{q,0}(n) = r_{q,0}(n) = x_{rec}(n)$$

In order to adapt the coefficients to the current signal properties, the expected values in the above equation are substituted by time average estimates measured over a limited past signal period. A compromise has to be chosen between a good convergence against the optimum predictor setting for signal periods with quasi-stationary characteristic and the ability of fast adaptation in case of signal transitions. In this context algorithms with iterative improvement of the estimates, i.e. from sample to sample, are of special interest. Here, a "least mean square" (LMS) approach is used and the predictor coefficients are calculated as follows

$$k_m(n+1) = \frac{COR_m(n)}{VAR_m(n)}$$

with

$$COR_m(n) = \alpha \cdot COR_m(n-1) + r_{q,m-1}(n-1) \cdot e_{q,m-1}(n)$$

$$VAR_m(n) = \alpha \cdot VAR_m(n-1) + 0.5 \cdot (r_{q,m-1}^2(n-1) + e_{q,m-1}^2(n))$$

where  $\alpha$  is an adaptation time constant which determines the influence of the current sample on the estimate of the expected values. The value of  $\alpha$  is chosen to

$$\alpha = 0.90625$$

The optimum values of the attenuation factors  $a$  and  $b$  have to be determined as a compromise between high prediction gain and small fade out time. The chosen values are

$$a = b = 0.953125.$$

Independent of whether prediction is disabled - either at all or only for a particular scalefactor band - or not, all the predictors are run all the time in order to always adapt the coefficients to the current signal statistics.

If `window_sequence` is of type `ONLY_LONG_SEQUENCE`, `LONG_START_SEQUENCE` and

LONG\_STOP\_SEQUENCE only the calculation of the reconstructed value of the quantized spectral components differs depending on the value of the **prediction\_used** bit:

- If the bit is set (1), then the quantized prediction error reconstructed from the transmitted data is added to the estimate  $x_{est}(n)$  calculated by the predictor resulting in the reconstructed value of the quantized spectral component, i.e.  $x_{rec}(n) = x_{est}(n) + e_q(n)$
- If the bit is not set (0), then the quantized value of the spectral component is reconstructed directly from the transmitted data.

In case of short blocks, i.e. window\_sequence is of type EIGHT\_SHORT\_SEQUENCE, prediction is always disabled and a reset is carried out for all predictors in all scalefactor bands, which is equivalent to a reinitialization, see subclause 4.6.6.3.3.

For a single\_channel\_element, the predictor processing for one frame is done according to the following pseudo code:

(It is assumed that the reconstructed value  $y_{rec}(c)$  - which is either the reconstructed quantized prediction error or the reconstructed quantized spectral coefficient - is available from previous processing.)

```

if (ONLY_LONG_SEQUENCE || LONG_START_SEQUENCE || LONG_STOP_SEQUENCE) {
  for (sfb=0; sfb<PRED_SFB_MAX; sfb++) {
    fc = swb_offset_long_window[fs_index][sfb];
    lc = swb_offset_long_window[fs_index][sfb+1];
    for (c=fc; c<lc; c++) {
      x_est[c] = predict();
      if (predictor_data_present && prediction_used[sfb])
        x_rec[c] = x_est[c] + y_rec[c];
      else
        x_rec[c] = y_rec[c];
    }
  }
}
else {
  reset_all_predictors();
}

```

In case of channel\_pair\_elements with **common\_window** = 1, the only difference is that the computation of  $x_{est}$  and  $x_{rec}$  in the inner for loop is done for both channels associated with the channel\_pair\_element. In case of channel\_pair\_elements with **common\_window** = 0, each channel has prediction applied using that channel's prediction side information.

#### 4.6.6.3.2.2 Quantization in predictor calculations

For a given predictor six state variables need to be saved:  $r_0$ ,  $r_1$ ,  $COR_1$ ,  $COR_2$ ,  $VAR_1$  and  $VAR_2$ . These variables will be saved as truncated IEEE floating-point numbers (i.e. the 16 msb of a float storage word).

The predicted value  $x_{est}$  will be rounded to a 16-bit floating point representation (i.e. round to a 7-bit mantissa) prior to being used in any calculation. The exact rounding algorithm to be used is shown in pseudo-C function `flt_round_inf()`. Note that for complexity considerations, *round to nearest, infinity* is used instead of *round to nearest, even*.

The expressions  $(b / VAR_1)$  and  $(b / VAR_2)$  will be rounded to a 16-bit floating point representation (i.e. round to a 7-bit mantissa), which permits the ratio to be computed via a pair of small look-up tables. C-code for generating such tables is shown in pseudo-C function `make_inv_tables()`.

All intermediate results in every floating point computation in the prediction algorithm will be represented in single precision floating point using rounding described below.

The IEEE Floating Point computational unit used in executing all arithmetic in the prediction tool will enable the following options:

- Round-to-Nearest, Even - Round to nearest representable value; round to the value with the least significant bit equal to zero (even) when the two nearest representable values are equally near.

- Overflow exception - Values whose magnitude is greater than the largest representable value will be set to the representation for infinity.
- Underflow exception - Gradual underflow (de-normalized numbers) will be supported; values whose magnitude is less than the smallest representable value will be set to zero.

#### 4.6.6.3.2.3 Fast algorithm for rounding

```
/* this does not conform to IEEE conventions of round to
 * nearest, even, but it is fast
 */
```

```
static void
flt_round_inf(float *pf)
{
    int flg;
    ulong tmp;
    float *pt = (float *)&tmp; /* note: this presumes 32 bit ulong */
    *pt = *pf;
    flg = tmp & (ulong)0x00008000;
    tmp &= (ulong)0xffff0000;
    *pf = *pt;
    /* round 1/2 lsb toward infinity */
    if (flg) {
        tmp &= (ulong)0xff800000; /* extract exponent and sign */
        tmp |= (ulong)0x00010000; /* insert 1 lsb */
        *pf += *pt; /* add 1 lsb and elided one */
        tmp &= (ulong)0xff800000; /* extract exponent and sign */
        *pf -= *pt; /* subtract elided one */
    }
}
```

#### 4.6.6.3.2.4 Generating rounded b / Var

```
static float mnt_table[128];
static float exp_table[256];
```

```
/* function flt_round_even() only works for arguments in the range
 * 1.0 < *pf < 2.0 - 2^-24
 */
```

```
static void
flt_round_even(float *pf)
{
    int exp;
    double mnt;
    float offset;
    mnt = frexp((double)*pf, &exp);
    offset = (float)ldexp(1.0, exp+15);
    *pf += offset; /* WARNING: This shifts out LSB's. Do not remove this pair of operations! */
    *pf -= offset;
}

static void
make_inv_tables(void)
{
    int i;
    ulong tmp1, tmp;
    float *pf = (float *)&tmp;
    float ftmp;

    *pf = 1.0;
    tmp1 = tmp; /* float 1.0 */
    /* mantissa table */
    for (i=0; i<128; i++) {
        tmp = tmp1 + (i<16); /* float 1.m, 7 msb only */
        ftmp = b / *pf; /* predictor constant b as in 8.3.2 */
        flt_round_even(&ftmp); /* round to 16 bits */
        mnt_table[i] = ftmp;
    }
}
```

```

/* exponent table */
for (i=0; i<256; i++) {
    tmp = i<<23;          /* float 1.0 * 2^exp */
    if (*pf > 1.0) {
        ftmp = 1.0 / *pf;
    }
    else {
        ftmp = 0;
    }
    exp_table[i] = ftmp;
}
}

```

### Quantization in Predictor Calculations

For a given predictor seven state variables need to be saved:

$$x_{rec}(n-4), x_{rec}(n-3), x_{rec}(n-2), x_{rec}(n-1), x_{rec}(n), a_1, a_2$$

Because we update the predictor every four samples, not all variables are necessary to be saved. Actually, for reconstructed spectral components, we only need to save ten variables every time. Including eight prediction coefficients, a total of eighteen variables needs to be saved. The ten reconstructed spectral components will be saved as truncated IEEE floating-point numbers (i.e. the 16 msb of a float storage word) and other eight prediction coefficients will be uniformly quantized into eight bits.

The dequantized error spectral components are rounded to a 16-bit floating point representation (i.e. round to a 7-bit mantissa). The exact rounding algorithm to be used is shown in pseudo-C function `flt_round_inf()`. Note that for complexity considerations, *round to nearest, infinity* is used instead of *round to nearest, even*. The reconstructed spectral components (the predicted value  $x_{est}$  plus the dequantized spectral component, error) will be truncated to a 16-bit floating point representation prior to being used in any calculation.

#### 4.6.6.3.3 Predictor reset

Initialization of a predictor means that the predictor's state variables are set as follows:  $r_0 = r_1 = 0$ ,  $COR_1 = COR_2 = 0$ ,  $VAR_1 = VAR_2 = 1$ . When the decoding process is started, all predictors are initialized.

A cyclic reset mechanism is applied by the encoder and signaled to the decoder, in which all predictors are initialized again in a certain time interval in an interleaved way. On one hand this increases predictor stability by re-synchronizing the predictors of the encoder and the decoder and on the other hand it allows defined entry points in the bitstream.

The whole set of predictors is subdivided into 30 so-called reset groups according to the following table:

**Table 4.97 – Reset Groups**

| Reset group number | Predictors of reset group |
|--------------------|---------------------------|
| 1                  | P0, P30, P60, P90,...     |
| 2                  | P1, P31, P61, P91,...     |
| 3                  | P2, P32, P62, P92,...     |
| ...                |                           |
| 30                 | P29, P59, P89, P119,...   |

where  $P_i$  is the predictor which corresponds to the spectral coefficient indexed by  $i$ .

Whether or not a reset has to be applied in the current frame is determined by the **predictor\_reset** bit. If this bit is set then the number of the predictor reset group to be reset in the current frame is specified in **predictor\_reset\_group\_number**. All predictors belonging to that reset group are then initialized as described above. This initialization has to be done after the normal predictor processing for the current frame has been carried out. Note that **predictor\_reset\_group\_number** cannot have the value 0 or 31.

A typical reset cycle starts with reset group number 1 and the reset group number is then incremented by 1 until it reaches 30, and then it starts with 1 again. Nevertheless, it may happen, e.g. due to switching between programs (bitstreams) or cutting and pasting, that there will be a discontinuity in the reset group numbering. If this is the case, these are the following three possibilities for decoder operation:

- Ignore the discontinuity and carry on the normal processing. This may result in a short audible distortion due to a mismatch (drift) between the predictors in the encoder and decoder. After one complete reset cycle (reset group  $n, n+1, \dots, 30, 1, 2, \dots, n-1$ ) the predictors are re-synchronized again. Furthermore, a possible distortion is faded out because of the attenuation factors  $a$  and  $b$ .
- Detect the discontinuity, carry on the normal processing but mute the output until one complete reset cycle is performed and the predictors are re-synchronized again.
- Reset all predictors.

An encoder is required to signal the reset of a group at least once every 8 frames. Groups do not have to be reset in ascending order, but every group must be reset within the maximum reset interval of  $8 \times 30 = 240$  frames. The bitstream syntax permits the encoder to signal the reset of a group at every frame, resulting in a minimum reset interval of  $1 \times 30 = 30$  frames.

In case of a `single_channel_element` or a `channel_pair_element` with `common_window = 0`, the reset has to be applied to the predictor bank(s) of the channel(s) associated with that element. In case of a `channel_pair_element` with `common_window = 1`, the reset has to be applied to the two predictor banks of the two channels associated with that element.

In the case of a short block (i.e. `window_sequence` of type `EIGHT_SHORT_SEQUENCE`) all predictors in all scalefactor bands must be reset.

4.6.6.4 Diagrams

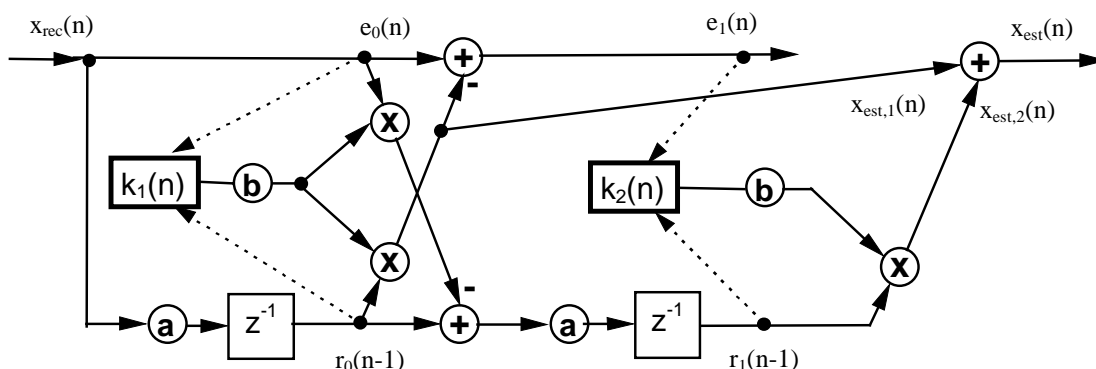


Figure 4.28 – Flow graph of AAC intra channel predictor for one spectral component in the decoder. The dotted lines indicate the signal flow for the adaptation of the predictor coefficients.

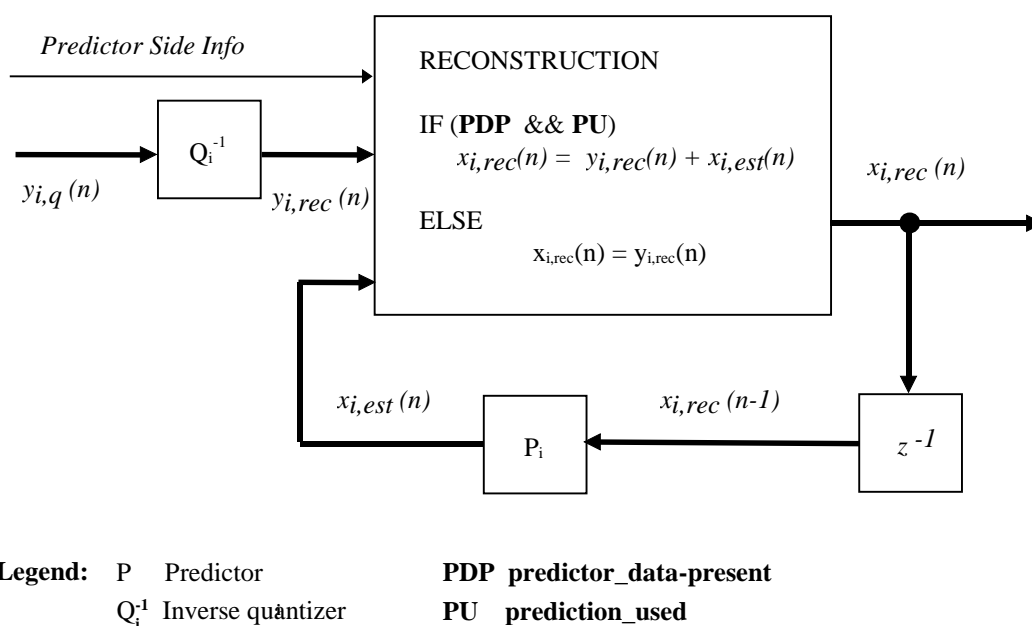


Figure 4.29 – Block diagram of decoder prediction unit for one single spectral component

## 4.6.7 Long Term Prediction (LTP)

### 4.6.7.1 Tool description

Long term prediction (LTP) is an efficient tool for reducing the redundancy of a signal between successive coding frames. This tool is especially effective for the parts of a signal which have clear pitch property. The implementation complexity of LTP is significantly lower than the complexity of Backward Adaptive Prediction. Because the Long Term Predictor is a forward adaptive predictor (prediction coefficients are sent as side information), it is inherently less sensitive to round-off numerical errors in the decoder or bit errors in the transmitted spectral coefficients.

With the MPEG-4 AAC scalable audio object type LTP can be used for any window type. For the AAC LTP audio object type LTP is restricted to long windows only, to achieve bitstream compatibility with MPEG-2 AAC such that if LTP is not used, an ISO/IEC 13818-7 (MPEG-2) AAC LC or Main profile decoder can parse the MPEG-4 AAC LTP bitstream.

### 4.6.7.2 Definitions

**ltp\_data\_present** 1 bit indicating whether prediction is used in current frame (1) or not (0) (always present)  
**ltp\_lag** 11-bit number specifying the optimal delay from 0 to 2047.  
**ltp\_coef** 3-bit index indicating the LTP coefficient in the table below. For all short windows in the current frame, the same coefficient is always used.

Table 4.98 – LTP coefficient

| Value of <b>ltp_coef</b> | value of LTP coefficient |
|--------------------------|--------------------------|
| 000                      | 0.570829                 |
| 001                      | 0.696616                 |
| 010                      | 0.813004                 |
| 011                      | 0.911304                 |
| 100                      | 0.984900                 |
| 101                      | 1.067894                 |
| 110                      | 1.194601                 |
| 111                      | 1.369533                 |

|                              |                                                                                                                                                                        |
|------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>ltp_short_used</b>        | 1 bit indicating whether LTP is used for each short window (1) or not (0)                                                                                              |
| <b>ltp_short_lag_present</b> | 1 bit indicating whether <b>ltp_short_lag</b> is actually transmitted (1), or omitted (0) from the bit-stream, which means that the value of <b>ltp_short_lag</b> is 0 |
| <b>ltp_short_lag</b>         | 4-bit number specifying the relative delay for each short window to <b>ltp_lag</b> from -8 to 7                                                                        |
| <b>ltp_long_used</b>         | 1 bit for each scalefactor band (sfb) where LTP can be used indicating whether LTP is switched on (1) or off (0) in that sfb.                                          |

For the audio object type ER AAC LD the following bitstream elements are available:

|                       |                                                                                                                                                                                                                                                                                                                                         |
|-----------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>ltp_lag_update</b> | 1 bit indicating whether a new value for the ltp lag is transmitted. Due to the high consistency of the LTP lag for many signals, one additional bit is introduced signaling that the lag of the previous frame is repeated ( $ltp\_lag\_update==0$ ). Otherwise, a new value for the ltp lag is transmitted ( $ltp\_lag\_update==1$ ). |
| <b>ltp_lag</b>        | 10-bit number specifying the optimal delay from 0 to 1023.                                                                                                                                                                                                                                                                              |

#### 4.6.7.3 Decoding process

The decoding process for LTP is carried out on each window of the current frame by applying 1-tap IIR filtering in the time domain to predict samples in the current frame by (quantized) samples in the previous frames. The process is controlled by the transmitted side information in a two step approach. The first control step defines whether LTP is used at all for the current frame. In the case of long window, the second control step defines, on which scalefactor bands LTP is used. In case of short windows the second control step defines which of the short windows in the coding block LTP is applied to. At the start of the decoding process, the reconstructed time domain samples are initialized by zeros.

For each frame, the LTP side information is extracted from the bitstream to control the further predictor processing at the decoder. In case of a `single_channel_element` the control information is valid for the channel with that element. In case of a `channel_pair_element` there are two sets of control data.

First, the **ltp\_data\_present** bit is read. If this bit is not set (0) then LTP is switched off for the current frame and there is no further predictor side information present. In this case the **ltp\_long\_used** flag for each scalefactor band stored in the decoder has to be set to zero. If the **ltp\_data\_present** bit is set (1) then LTP is used for the current frame and the LTP parameters are read. The decoding process is different for long and short windows.

For long windows, the LTP parameters are used to calculate the predicted time domain signals using the following formula:

$$x\_est(i) = ltp\_coef * x\_rec(i - M - ltp\_lag)$$

$$i = 0, \dots, N-1$$

where  $x\_est(i)$  are the predicted samples  
 $x\_rec(i)$  are reconstructed time domain samples  
 N is the length of the transform window  
 $M = N/2$  if aot == ER AAC LD, otherwise  $M = 0$

The different value for M used in combination with the low delay codec is to achieve a similar range of possible lag values in absolute time, despite of the shorter frame.

The reference point for index i and the content of the buffer  $x\_rec$  are arranged so that  $x\_rec(0 \dots N/2 - 1)$  contains the last aliased half window from the IMDCT, and  $x\_rec(N/2 \dots N-1)$  is always all zeros. The rest of  $x\_rec$  ( $i < 0$ ) contains the previous fully reconstructed time domain samples, i.e., output of the decoder.

Using the MDCT for long windows, the predicted spectral components are obtained for the current frame from the predicted time domain signal. Next, the **ltp\_long\_used** bits are read from the bitstream, which control the use of prediction in each scalefactor band individually, i.e. if the bit is set for a particular scalefactor band, all the predicted spectral components of this scalefactor band are used. Otherwise the predicted spectral components are set to zeros. That is, if the **ltp\_long\_used** bit is set, then the quantized prediction error reconstructed from the transmitted

data is added to the predicted spectral component. If the bit is not set (0), then the quantized value of spectral component is reconstructed directly from the transmitted data.

For each short window, the bit **ltp\_short\_used** is read from the bitstream. If the **ltp\_short\_used** is not set, the quantized value of spectral component is reconstructed directly from the transmitted data and the time domain signal can be reconstructed for this particular subframe. If the **ltp\_short\_used** is set, the **ltp\_short\_lag\_present** is read. If **ltp\_short\_lag\_present** is set then **ltp\_short\_lag** is read. If **ltp\_short\_lag\_present** is not set, the value of **ltp\_short\_lag** is set to 0. The value of **ltp\_short\_lag** is combined with **ltp\_lag** and **ltp\_coef** to calculate the predicted time domain signal for this particular subframe. Using the MDCT for short windows, the predicted spectral components are calculated and the spectral components in the first eight scalefactor bands are added to the quantized prediction error reconstructed from the transmitted data.

The signal reconstruction part of decoding process for one channel can be described as following pseudo code. Here  $x\_est$  is the predicted time domain signal,  $X\_est$  is the corresponding frequency domain vector,  $Y\_rec$  is the vector of decoded spectral coefficients and  $X\_rec$  is the vector of reconstructed spectral coefficients.

```

if (ONLY_LONG_SEQUENCE || LONG_START_SEQUENCE || LONG_STOP_SEQUENCE) {
    x_est = predict();
    X_est = MDCT(x_est)
    for (sfb=0; sfb<NUMBER_SCALEFACTOR_BAND; sfb++) {
        if (ltp_data_present && ltp_long_used[sfb] )
            X_rec = X_est + Y_rec;
        else
            X_rec = Y_rec;
    }
}
else {
    for (w=0; w<num_windows; w++) {
        if (ltp_data_present && ltp_short_used[w] {
            x_est = predict();
            X_est = MDCT(x_est)
            for ( sfb=0; sfb<8; sfb++)
                X_rec = X_est + Y_rec;
        }
        else
            X_rec = Y_rec;
    }
}

```

#### 4.6.7.4 Integration of LTP with other GA tools

##### 4.6.7.4.1 LTP with TNS

Because TNS needs to be applied to a reconstructed spectrum, the TNS filter must occur after LTP in the decoding chain. This makes an additional TNS analysis filter necessary in the LTP loop, as shown in Figure 4.30.

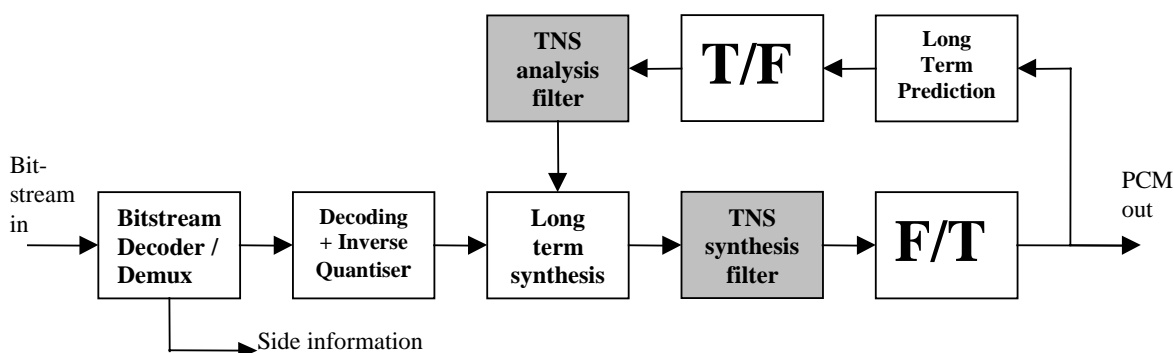


Figure 4.30 – Long Term Prediction with TNS.



**4.6.7.4.2 LTP with PNS**

Simultaneous use of LTP and PNS is not prevented in the syntax. If both LTP, and PNS are enabled on the same scalefactor band, PNS takes precedence, and no prediction is applied to this band.

**4.6.7.5 LTP in a scalable GA decoder**

In a scalable coder LTP is used only on the lowest GA coding layer and the predictor update is based on the time domain output of the first GA layer. Either LTP or a core coder can be used, but not both at the same time. The LTP decoding process is similar to the process used when the lowest layer is a core coder. The LTP part itself is decoded exactly in the same way as in a non-scalable GA decoder.

In a scalable configuration the simultaneous use of LTP and Intensity Stereo is not prevented in the syntax. However if both LTP and Intensity Stereo are enabled in the same scalefactor band in the first GA layer, Intensity Stereo takes precedence and no prediction is applied to this band.

**4.6.8 Joint Coding**

(similar to ISO/IEC 13818-7)

**4.6.8.1 M/S stereo**

**4.6.8.1.1 Tool description**

The M/S joint channel coding operates on channel pairs. Channels are most often paired such that they have symmetric presentation relative to the listener, such as left/right or left surround/right surround. The first channel in the pair is denoted „left“ and the second „right.“ On a per-spectral-coefficient basis, the vector formed by the left and right channel signals is reconstructed or de-matrixed by either the identity matrix

$$\begin{bmatrix} l \\ r \end{bmatrix} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix} \begin{bmatrix} l \\ r \end{bmatrix}$$

or the inverse M/S matrix

$$\begin{bmatrix} l \\ r \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} \begin{bmatrix} m \\ s \end{bmatrix}$$

The decision on which matrix to use is done on a scalefactor band by scalefactor band basis as indicated by the `ms_used` flags. For the audio object types AAC Main, LC, SSR, and LTP, M/S joint channel coding can only be used if **common\_window** is '1'. For the AAC scalable audio object type always common windows between the two audio channels are used, so that M/S coding is always possible.

**4.6.8.1.2 Definitions**

- ms\_mask\_present** this two bit field indicates that the MS mask is
  - 00 All zeros
  - 01 A mask of max\_sfb bands of ms\_used follows this field
  - 10 All ones
  - 11 Reserved
- ms\_used[g][sfb]** one-bit flag per scalefactor band indicating that M/S coding is being used in windowgroup g and scalefactor band sfb.
- l\_spec[]** Array containing the left channel spectrum of the respective channel pair.
- r\_spec[]** Array containing the right channel spectrum of the respective channel pair.
- is\_intensity(g,sfb)** function returning the intensity status, defined in subclause 4.6.8.2.3.
- is\_noise(g,sfb)** function returning the noise substitution status, defined in subclause 4.6.13.3.

### 4.6.8.1.3 Decoding process

Reconstruct the spectral coefficients of the first („left“) and second („right“) channel as specified by the **mask\_present** and the **ms\_used[][]** flags as follows:

```

if (mask_present >= 1) {
  for (g=0; g<num_window_groups; g++) {
    for (b=0; b<window_group_length[g]; b++) {
      for (sfb=0; sfb<max_sfb; sfb++) {
        if ((ms_used[g][sfb] || mask_present == 2) &&
            !is_intensity(g,sfb) && !is_noise(g,sfb)) {
          for (i=0; i<swb_offset[sfb+1]-swb_offset[sfb]; i++) {
            tmp = l_spec[g][b][sfb][i] - r_spec[g][b][sfb][i];
            l_spec[g][b][sfb][i] = l_spec[g][b][sfb][i] +
                                  r_spec[g][b][sfb][i];
            r_spec[g][b][sfb][i] = tmp;
          }
        }
      }
    }
  }
}

```

Please note that **ms\_used[][]** is also used in the context of intensity stereo coding and perceptual noise substitution. If intensity stereo coding or noise substitution is on for a particular scalefactor band, no M/S stereo decoding is carried out.

## 4.6.8.2 Intensity Stereo (IS)

### 4.6.8.2.1 Tool description

This tool is used to implement joint intensity stereo coding between both channels of a channel pair. Thus, both channel outputs are derived from a single set of spectral coefficients after the inverse quantization process. This is done selectively on a scalefactor band basis when intensity stereo is flagged as active.

### 4.6.8.2.2 Definitions

|                                |                                                                                                      |
|--------------------------------|------------------------------------------------------------------------------------------------------|
| <b>hcod_sf[]</b>               | Huffman codeword from the Huffman code table used for coding of scalefactors (see subclause 4.6.3.2) |
| <i>dpcm_is_position[][]</i>    | Differentially encoded intensity stereo position                                                     |
| <i>is_position[group][sfb]</i> | Intensity stereo position for each group and scalefactor band                                        |
| <i>l_spec[]</i>                | Array containing the left channel spectrum of the respective channel pair                            |
| <i>r_spec[]</i>                | Array containing the right channel spectrum of the respective channel pair                           |

### 4.6.8.2.3 Decoding process

The use of intensity stereo coding is signaled by the use of the pseudo codebooks INTENSITY\_HCB and INTENSITY\_HCB2 (15 and 14) in the right channel (use of these codebooks in a left channel of a channel pair element is illegal). INTENSITY\_HCB and INTENSITY\_HCB2 signal in-phase and out-of-phase intensity stereo coding, respectively.

In addition, in case of a non-scalable GA decoder the phase relationship of the intensity stereo coding can be reversed by means of the **ms\_used** field: Because M/S stereo coding and intensity stereo coding are mutually exclusive for a particular scalefactor band and group, the primary phase relationship indicated by the Huffman code tables is changed from in-phase to out-of-phase or vice versa if the corresponding **ms\_used** bit is set for the respective band.

The directional information for the intensity stereo decoding is represented by an "intensity stereo position" value indicating the relation between left and right channel scaling. If intensity stereo coding is active for a particular group and scalefactor band, an intensity stereo position value is transmitted instead of the scalefactor of the right

channel. Intensity positions are coded just like scalefactors, i.e. by Huffman coding of differential values with two differences:

- There is no first value that is sent as PCM. Instead, the differential decoding is started assuming the last intensity stereo position value to be zero.
- Differential decoding is done separately between scalefactors and intensity stereo positions. In other words, the scalefactor decoder ignores interposed intensity stereo position values and vice versa (see subclause 4.6.2.3.2)

The same codebook is used for coding intensity stereo positions as for scalefactors.

Two pseudo functions are defined for use in intensity stereo decoding:

```
function is_intensity(group,sfb) {
+1   for window groups / scalefactor bands with right channel
      codebook sfb_cb[group][sfb] == INTENSITY_HCB
-1   for window groups / scalefactor bands with right channel
      codebook sfb_cb[group][sfb] == INTENSITY_HCB2
  0   otherwise
}

function invert_intensity(group,sfb) {
  1-2*ms_used[group][sfb]   if (ms_mask_present == 1) && aot != AAC scalable
+1   otherwise
}
```

The intensity stereo decoding for one channel pair is defined by the following pseudo code:

```
p = 0;
for (g=0; g<num_window_groups; g++) {

  /* Decode intensity positions for this group */
  for (sfb=0; sfb<max_sfb; sfb++)
    if (is_intensity(g,sfb))
      is_position[g][sfb] = p += dpcm_is_position[g][sfb];

  /* Do intensity stereo decoding */
  for (b=0; b<window_group_length[g]; b++) {
    for (sfb=0; sfb<max_sfb; sfb++) {
      if (is_intensity(g,sfb)) {

        scale = is_intensity(g,sfb) * invert_intensity(g,sfb) *
                0.5^(0.25*is_position[g][sfb]);
        /* Scale from left to right channel,
           do not touch left channel */
        for (i=0; i<swb_offset[sfb+1]-swb_offset[sfb]; i++)
          r_spec[g][b][sfb][i] = scale * l_spec[g][b][sfb][i];

      }
    }
  }
}
```

In case of reversible variable length coding (RVLC) there is no last value that is sent as PCM. Instead, the backwards differential decoding is started assuming the last intensity stereo position value to be zero. The decoding of the intensity stereo positions is defined by the following pseudo code:

```
p=dpcm_is_last_position;

for (g=win-1;g>=0;g--) {
  for (sfb=sfbmax-1;sfb>=0;sfb--) {
    is_pos[g][sfb]=p;
```

```

    p=dpcm_is_pos[g][sfb];
  }
}

```

#### 4.6.8.2.4 Integration with the intra channel prediction tool

For scalefactor bands coded in intensity stereo the corresponding predictors in the right channel are switched to "off" thus effectively overriding the status specified by the prediction\_used mask. The update of these predictors is done by feeding the intensity stereo decoded spectral values of the right channel as the "last quantized value"  $x_{rec}(n-1)$ . These values result from the scaling process from left to right channel as described in the pseudo code.

In case of a non-scalable configuration the function of Long Term Prediction does not depend on Intensity Stereo.

#### 4.6.8.2.5 Integration of the intensity tool for the audio object type AAC scalable

If a particular scalefactor band and group is coded by intensity stereo, its contribution to the spectral components of the output signal is omitted if spectral coefficients are transmitted for this scalefactor band and group in any of the higher (enhancement) layers (that contributes to the output signal) by means of a non-intensity codebook number.

If a particular scalefactor band and group is coded by intensity stereo, and if the same scalefactor band in the next enhancement layer is also coded by intensity stereo, only the spectral components of the left channel are combined. The spectral components of the right channel are re-calculated for the enhancement layer using the intensity factor of this enhancement layer.

In a scalable configuration the simultaneous use of Intensity Stereo and LTP is not prevented in the syntax. However if both Intensity Stereo and LTP are enabled in the same scalefactor band in the first GA layer, Intensity Stereo takes precedence and no prediction is applied to this band.

In case of an AAC scalable configuration the ms\_used field is ignored in Intensity Stereo decoded scalefactor bands but may still signal the use of M/S stereo decoding in higher (enhancement) layers.

Additional information on decoding intensity tool in a scalable bitstreams is given in subclause 4.5.2.2.7.

### 4.6.8.3 Coupling channel

(identical to ISO/IEC 13818-7)

#### 4.6.8.3.1 Tool description

Coupling channel elements provide two functionalities: First, coupling channels may be used to implement generalized intensity stereo coding where channel spectra can be shared across channel boundaries. Second, coupling channels may be used to dynamically perform a downmix of one sound object into the stereo image.

Note that this tool includes certain object type dependent parameters (see 4.6.9).

#### 4.6.8.3.2 Definitions

|                             |                                                                                                                              |
|-----------------------------|------------------------------------------------------------------------------------------------------------------------------|
| <b>ind_sw_cce_flag</b>      | one bit indicating whether the coupled target syntax element is an independently switched (1) or a dependently switched (0). |
| <b>num_coupled_elements</b> | number of coupled target elements                                                                                            |
| <b>cc_target_is_cpe</b>     | one bit indicating if the coupled target syntax element is a CPE (1) or a SCE (0).                                           |
| <b>cc_target_tag_select</b> | four bit field specifying the element_instance_tag of the coupled target syntax element.                                     |
| <b>cc_l</b>                 | one bit indicating that a list of gain_element values is applied to the left channel of a channel pair.                      |
| <b>cc_r</b>                 | one bit indicating that a list of gain_element values is applied to the right channel of a channel pair).                    |
| <b>cc_domain</b>            | one bit indicating whether the coupling is performed before (0) or after (1) the TNS                                         |

|                                       |                                                                                                                                                                                                                               |
|---------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
|                                       | decoding of the coupled target channels                                                                                                                                                                                       |
| <b>gain_element_sign</b>              | one bit indicating if the transmitted gain_element values contain information about in-phase / out-of-phase coupling (1) or not (0)                                                                                           |
| <b>gain_element_scale</b>             | determines the amplitude resolution <i>cc_scale</i> of the scaling operation according to Table 4.100.                                                                                                                        |
| <b>common_gain_element_present[c]</b> | one bit indicating whether Huffman coded common_gain_element values are transmitted (1) or whether Huffman coded differential gain_elements are sent (0)                                                                      |
| <i>dpcm_gain_element[[]]</i>          | Differentially encoded gain element                                                                                                                                                                                           |
| <i>gain_element[group][sfb]</i>       | Gain element for each group and scalefactor band                                                                                                                                                                              |
| <i>common_gain_element[]</i>          | Gain element that is used for all window groups and scalefactor bands of one coupling target channel                                                                                                                          |
| <i>spectrum_m(idx, domain)</i>        | Pointer to the spectral data associated with the single_channel_element with index idx. Depending on the value of "domain", the spectral coefficients before (0) or after (1) TNS decoding are pointed to.                    |
| <i>spectrum_l(idx, domain)</i>        | Pointer to the spectral data associated with the left channel of the channel_pair_element with index idx. Depending on the value of "domain", the spectral coefficients before (0) or after (1) TNS decoding are pointed to.  |
| <i>spectrum_r(idx, domain)</i>        | Pointer to the spectral data associated with the right channel of the channel_pair_element with index idx. Depending on the value of "domain", the spectral coefficients before (0) or after (1) TNS decoding are pointed to. |

#### 4.6.8.3.3 Decoding process

The coupling channel is based on an embedded single\_channel\_element which is combined with some dedicated fields to accommodate its special purpose.

The coupled target syntax elements (SCEs or CPEs) are addressed using two syntax elements. First, the cc\_target\_is\_cpe field selects whether a SCE or CPE is addressed. Second, a cc\_target\_tag\_select field selects the instance\_tag of the SCE/CPE.

The scaling operation involved in channel coupling is defined by gain\_element values which describe the applicable gain factor and sign. In accordance with the coding procedures for scalefactors and intensity stereo positions, gain\_element values are differentially encoded using the Huffman table for scalefactors. Similarly, the decoded gain factors for coupling relate to window groups of spectral coefficients.

#### Independently switched CCEs vs. dependently switched CCEs

There are two kinds of CCEs. They are „independently switched“ and „dependently switched“ CCEs. An independently switched CCE is a CCE in which the window state (i.e. window\_sequence and window\_shape) of the CCE does not have to match that of any of the SCE or CPE channels that the CCE is coupled onto (target channels). This has several important implications:

- First, it is required that an independently switched CCE must only use the common\_gain element, not a list of gain\_elements.
- Second, the independently switched CCE must be decoded all the way to the time domain (i.e. including the synthesis filterbank) before it is scaled and added onto the various SCE and CPE channels that it is coupled to in the case that window state does not match.

A dependently switched CCE, on the other hand, must have a window state that matches all of the target SCE and CPE channels that it is coupled onto as determined by the list of cc\_l and cc\_r elements. In this case, the CCE only needs to be decoded as far as the frequency domain and then scaled as directed by the gain list before it is added to the target SCE or CPE channels.

The following pseudo code in function decode\_coupling\_channel() defines the decoding operation for a dependently switched coupling channel element. First the spectral coefficients of the embedded single\_channel\_element are decoded into an internal buffer. Since the gain elements for the first coupled target (list\_index == 0) are not transmitted, all gain\_element values associated with this target are assumed to be 0, i.e. the coupling channel is added to the coupled target channel in its natural scaling. Otherwise the spectral

coefficients are scaled and added to the coefficients of the coupled target channels using the appropriate list of gain\_element values.

An independently switched CCE is decoded like a dependently switched CCE having only common\_gain\_elements. However, the resulting scaled spectrum is transformed back into its time representation and then coupled in the time domain.

Please note that the gain\_element lists may be shared between the left and the right channel of a target channel pair element. This is signalled by both cc\_l and cc\_r being zero as indicated in the table below:

**Table 4.99 – Shared gain\_element lists**

| cc_l,<br>cc_r | shared gain list<br>present | left gain list present | right gain list present |
|---------------|-----------------------------|------------------------|-------------------------|
| 0, 0          | yes                         | no                     | no                      |
| 0, 1          | no                          | no                     | yes                     |
| 1, 0          | no                          | yes                    | no                      |
| 1, 1          | no                          | yes                    | yes                     |

```

decode_coupling_channel()
{
  - decode spectral coefficients of embedded single_channel_element
  into buffer "cc_spectrum[]".

  /* Couple spectral coefficients onto target channels */
  list_index = 0;
  for (c=0; c<num_coupled_elements+1; c++) {
    if (!cc_target_is_cpe[c]) {
      couple_channel( cc_spectrum,
                     spectrum_m( cc_target_tag_select[c], cc_domain ),
                     list_index++ );
    }
    if (cc_target_is_cpe[c]) {
      if (!cc_l[c] && !cc_r[c]) {
        couple_channel( cc_spectrum,
                       spectrum_l( cc_target_tag_select[c], cc_domain ),
                       list_index );
        couple_channel( cc_spectrum,
                       spectrum_r( cc_target_tag_select[c], cc_domain ),
                       list_index++ );
      }
      if (cc_l[c]) {
        couple_channel( cc_spectrum,
                       spectrum_l( cc_target_tag_select[c], cc_domain ),
                       list_index++ );
      }
      if (cc_r[c]) {
        couple_channel( cc_spectrum,
                       spectrum_r( cc_target_tag_select[c], cc_domain ),
                       list_index++ );
      }
    }
  }
}

```

```

couple_channel( source_spectrum[], dest_spectrum[], gain_list_index )
{
  idx = gain_list_index;
  a = 0;
  cc_scale = cc_scale_table[gain_element_scale];
  for (g=0; g<num_window_groups; g++) {

    /* Decode coupling gain elements for this group */
    if (common_gain_element_present[idx]) {

```

```

for (sfb=0; sfb<max_sfb; sfb++) {
    cc_sign[idx][g][sfb] = 1;
    gain_element[idx][g][sfb] = common_gain_element[idx];
}
} else {

for (sfb=0; sfb<max_sfb; sfb++) {
    if ( sfb_cb[g][sfb] == ZERO_HCB )
        continue;

    if (gain_element_sign) {
        cc_sign[idx][g][sfb] =
            1 - 2*(dpcm_gain_element[idx][g][sfb] & 0x1);
        gain_element[idx][g][sfb] =
            a += (dpcm_gain_element[idx][g][sfb] >> 1);
    }
    else {
        cc_sign[idx][g][sfb] = 1;
        gain_element[idx][g][sfb] =
            a += dpcm_gain_element[idx][g][sfb];
    }
}
}

/* Do coupling onto target channels */
for (b=0; b<window_group_length[b]; b++) {
    for (sfb=0; sfb<max_sfb; sfb++) {

        if ( sfb_cb[g][sfb] != ZERO_HCB ) {
            cc_gain[idx][g][sfb] =
                cc_sign[idx][g][sfb] * cc_scale^gain_element[idx][g][sfb];

            for (i=0; i<swb_offset[sfb+1]-swb_offset[sfb]; i++)
                dest_spectrum[g][b][sfb][i] +=
                    cc_gain[idx][g][sfb] * source_spectrum[g][b][sfb][i];
        }
    }
}
}
}

```

Note: The array sfb\_cb represents the codebook data respect to the CCE's embedded single\_channel\_element (not the coupled target channel).

4.6.8.3.4 Tables

**Table 4.100 – Scaling resolution for channel coupling (cc\_scale\_table)**

| Value of "gain_element_scale" | Amplitude Resolution "cc_scale" | Stepsize [dB] |
|-------------------------------|---------------------------------|---------------|
| 0                             | 2 <sup>^(1/8)</sup>             | 0.75          |
| 1                             | 2 <sup>^(1/4)</sup>             | 1.50          |
| 2                             | 2 <sup>^(1/2)</sup>             | 3.00          |
| 3                             | 2 <sup>^1</sup>                 | 6.00          |

Note: the Coupling Channel Tool can only be applied to AudioObjectTypes AAC Main, LC, SSR, and LTP.

## 4.6.9 Temporal Noise Shaping (TNS)

(similar to ISO/IEC 13818-7)

### 4.6.9.1 Tool description

Temporal Noise Shaping is used to control the temporal shape of the quantization noise within each window of the transform. This is done by applying a filtering process to parts of the spectral data of each channel.

### 4.6.9.2 Definitions

|                               |                                                                                                                                                                 |
|-------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>n_filt[w]</b>              | number of noise shaping filters used for window w                                                                                                               |
| <b>coef_res[w]</b>            | token indicating the resolution of the transmitted filter coefficients for window w, switching between a resolution of 3 bits (0) and 4 bits (1)                |
| <b>length[w][filt]</b>        | length of the region to which one filter is applied in window w (in units of scalefactor bands)                                                                 |
| <b>order[w][filt]</b>         | order of one noise shaping filter applied to window w                                                                                                           |
| <b>direction[w][filt]</b>     | 1 bit indicating whether the filter is applied in upward (0) or downward (1) direction                                                                          |
| <b>coef_compress[w][filt]</b> | 1 bit indicating whether the most significant bit of the coefficients of the noise shaping filter filt in window w are omitted from transmission (1) or not (0) |
| <b>coef[w][filt][i]</b>       | coefficients of one noise shaping filter applied to window w                                                                                                    |
| <b>spec[w][k]</b>             | Array containing the spectrum for the window w of the channel being processed                                                                                   |

Depending on the window\_sequence the size of the following bitstream fields is switched for each transform window according to its window size:

**Table 4.101 – Size of bitstream fields**

| Name     | Window with 128 spectral lines | Other window size |
|----------|--------------------------------|-------------------|
| 'n_filt' | 1                              | 2                 |
| 'length' | 4                              | 6                 |
| 'order'  | 3                              | 5                 |

### 4.6.9.3 Decoding process

The decoding process for Temporal Noise Shaping is carried out separately on each window of the current frame by applying all-pole filtering to selected regions of the spectral coefficients (see function `tns_decode_frame`). The number of noise shaping filters applied to each window is specified by "n\_filt". The target range of spectral coefficients is defined in units of scalefactor bands counting down "length" bands from the top band (or the bottom of the previous noise shaping band).

First the transmitted filter coefficients have to be decoded, i.e. conversion to signed numbers, inverse quantization, conversion to LPC coefficients as described in function `tns_decode_coef`(). Then the all-pole filters are applied to the target frequency regions of the channel's spectral coefficients (see function `tns_ar_filter`()). The token "direction" is used to determine the direction the filter is slid across the coefficients (0=upward, 1=downward). The constant `TNS_MAX_BANDS` defines the maximum number of scalefactor bands to which Temporal Noise Shaping is applied. The maximum possible filter order is defined by the constant `TNS_MAX_ORDER`. Both constants are object type dependent parameters.

The decoding process for one channel can be described as follows pseudo code:

```
/* TNS decoding for one channel and frame */
tns_decode_frame()
{
    for (w=0; w<num_windows; w++) {

        bottom = num_swb;
        for (f=0; f<n_filt[w]; f++) {
```



```

top = bottom;
bottom = max( top - length[w][f], 0 );
tns_order = min( order[w][f], TNS_MAX_ORDER );
if (!tns_order) continue;

tns_decode_coef( tns_order, coef_res[w]+3, coef_compress[w][f],
                coef[w][f], lpc[] );

start = swb_offset[ min(bottom, TNS_MAX_BANDS, max_sfb) ];
end = swb_offset[ min(top, TNS_MAX_BANDS, max_sfb) ];
if ((size = end - start) <= 0) continue;

if (direction[w][f]) {
    inc = -1; start = end - 1;
} else {
    inc = 1;
}

tns_ar_filter( &spec[w][start], size, inc, lpc[], tns_order );
}
}
}

/* Decoder transmitted coefficients for one TNS filter */
tns_decode_coef( order, coef_res_bits, coef_compress, coef[], a[] )
{
    /* Some internal tables */
    sgn_mask[] = { 0x2, 0x4, 0x8 };
    neg_mask[] = { ~0x3, ~0x7, ~0xf };

    /* size used for transmission */
    coef_res2 = coef_res_bits - coef_compress;
    s_mask = sgn_mask[ coef_res2 - 2 ]; /* mask for sign bit */
    n_mask = neg_mask[ coef_res2 - 2 ]; /* mask for padding neg. values */

    /* Conversion to signed integer */
    for (i=0; i<order; i++)
        tmp[i] = (coef[i] & s_mask) ? (coef[i] | n_mask) : coef[i];

    /* Inverse quantization */
    iqfac = ((1 << (coef_res_bits-1)) - 0.5) / (π/2.0);
    iqfac_m = ((1 << (coef_res_bits-1)) + 0.5) / (π/2.0);
    for (i=0; i<order; i++) {
        tmp2[i] = sin( tmp[i] / ((tmp[i] >= 0) ? iqfac : iqfac_m) );
    }
}

/* Conversion to LPC coefficients */
a[0] = 1;
for (m=1; m<=order; m++) {
    for (i=1; i<m; i++) { /* loop only while i<m */
        b[i] = a[i] + tmp2[m-1] * a[m-i];
    }
    for (i=1; i<m; i++) { /* loop only while i<m */
        a[i] = b[i];
    }
    a[m] = tmp2[m-1]; /* changed */
}

tns_ar_filter( spectrum[], size, inc, lpc[], order )
{
    - Simple all-pole filter of order "order" defined by
    y(n) = x(n) - lpc[1]*y(n-1) - ... - lpc[order]*y(n-order)

```

- The state variables of the filter are initialized to zero every time
  - The output data is written over the input data ("in-place operation")
  - An input vector of "size" samples is processed and the index increment to the next data sample is given by "inc"
- }

Please note that this pseudo code uses a „C“-style interpretation of arrays and vectors, i.e. if `coef[w][filt][i]` describes the coefficients for all windows and filters, `coef[w][filt]` is a pointer to the coefficients of one particular window and filter. Also, the identifier `coef` is used as a formal parameter in function `tns_decode_coef()`.

#### 4.6.9.4 Maximum TNS order and bandwidth

The value for the constant `MAX_TNS_ORDER` depends on audio object type, windowing and sampling rate. Table 4.102 defines `MAX_TNS_ORDER` depending on these parameters.

**Table 4.102 – Definition of `TNS_MAX_ORDER` depending on AOT, windowing and sampling rate**

| Windowing           | short windows | long windows | long windows |
|---------------------|---------------|--------------|--------------|
| sampling rate       | -             | > 32kHz      | <= 32kHz     |
| AOT 1 (AAC Main)    | 7             | 20           | 20           |
| AOT 2 (AAC LC)      | 7             | 12           | 12           |
| AOT 3 (AAC SSR)     | 7             | 12           | 12           |
| other AOT using TNS | 7             | 20           | 12           |

According to the sampling rate and audio object type in use, the value for the constant `TNS_MAX_BANDS` is set as follows:

**Table 4.103 – Definition of `TNS_MAX_BANDS` depending on AOT, windowing and sampling rate**

| Sampling Rate [Hz] | AudioObject types without PQF filterbank (long windows) | AudioObject types without PQF filterbank (short windows) | AudioObject types with PQF filterbank (long windows) | AudioObject types with PQF filterbank (short windows) |
|--------------------|---------------------------------------------------------|----------------------------------------------------------|------------------------------------------------------|-------------------------------------------------------|
| 96000              | 31                                                      | 9                                                        | 28                                                   | 7                                                     |
| 88200              | 31                                                      | 9                                                        | 28                                                   | 7                                                     |
| 64000              | 34                                                      | 10                                                       | 27                                                   | 7                                                     |
| 48000              | 40                                                      | 14                                                       | 26                                                   | 6                                                     |
| 44100              | 42                                                      | 14                                                       | 26                                                   | 6                                                     |
| 32000              | 51                                                      | 14                                                       | 26                                                   | 6                                                     |
| 24000              | 46                                                      | 14                                                       | 29                                                   | 7                                                     |
| 22050              | 46                                                      | 14                                                       | 29                                                   | 7                                                     |
| 16000              | 42                                                      | 14                                                       | 23                                                   | 8                                                     |
| 12000              | 42                                                      | 14                                                       | 23                                                   | 8                                                     |
| 11025              | 42                                                      | 14                                                       | 23                                                   | 8                                                     |
| 8000               | 39                                                      | 14                                                       | 19                                                   | 7                                                     |

#### 4.6.9.5 TNS in the scalable coder

For the first mono layer, and for both channels of the first stereo layer a `tns_data_present` bit is available which activates the usage of TNS for a certain channel. The TNS filter information, however, is not necessarily transmitted along with the enable bit. Table 4.104 lists the source channel from which the TNS filter information for a specific output channel must be taken.

In all TwinVQ-mono/AAC-mono and TwinVQ-stereo/AAC-stereo configurations either the first TwinVQ layer uses TNS (**tns\_data\_present**==1 in tvq\_main\_header(), no **tns\_data\_present** bit in ASME), or the ASME (tns\_data\_present==0 in tvq\_main\_header(), **tns\_data\_present** bit in ASME), but not both at the same time..

In all TwinVQ-mono/AAC-stereo configurations the first TwinVQ layer carries the **tns\_data\_present** bit and two tns\_data\_present bits occur in the first AAC stereo layer (one for each audio output channel).

In all TwinVQ-mono/AAC-mono/AAC-stereo configurations either the first TwinVQ layer uses TNS and therefore the tns\_data\_present bit and tns\_data() occurs in the tvq\_main\_header(), or the first AAC mono-layer and in both cases two tns\_data\_present bits in the first AAC stereo layer.

**Table 4.104 – TNS filter information source channel information depending on tns\_data\_present in the scalable mono / stereo coder**

| tns_data_present<br>M-Channel | tns_data_present<br>L-Channel | tns_data_present<br>R-Channel | TNS Info M<br>Source Chan. | TNS Info L<br>Source Chan. | TNS Info R<br>Source Chan. |
|-------------------------------|-------------------------------|-------------------------------|----------------------------|----------------------------|----------------------------|
| 0                             | 0                             | 0                             | -                          | -                          | -                          |
| 1                             | 0                             | 0                             | M                          | M                          | M                          |
| 0                             | 1                             | 1                             | -                          | L                          | R                          |
| 0                             | 1                             | 0                             | -                          | L                          | -                          |
| 0                             | 0                             | 1                             | -                          | -                          | R                          |
| 1                             | 0                             | 1                             | M                          | M                          | R/M                        |
| 1                             | 1                             | 0                             | M                          | L/M                        | M                          |
| 1                             | 1                             | 1                             | M                          | L/M                        | R/M                        |

The L/M and R/M TNS entries describe a serial layout of two TNS filters (See also the block diagrams in subclause 4.5.2.2 ). Additionally, the following rules apply in this case:

- The execution of the M-Filter must stop at the scale factor band, which is denoted by the max\_sfb parameter of the highest mono-layer.
- The M-filter is not calculated if the lower boundary of the L-, or R-Filter, respectively, is lower than max\_sfb of the highest mono layer. This allows to override the M-Filter for the lower frequency bands, if desired.

With this definition each spectral line is still filtered once at most. Therefore, the overall complexity of this combined filter is the same as the complexity of the usual single filter

If TNS is used in a scalable coder with a core coder, the TNS encoder filter of the first mono AAC layer have to be applied to the output of the MDCT which is employed to generate the spectrum of the core coder. These encoder filters use the LPC coefficients already decoded for the corresponding TNS decoder filters. The filters are slid across the specified target frequency range exactly the way described for the decoder filter. The difference between decoder and encoder filtering is that each all-pole (auto-regressive) decoder filter used for TNS decoding is replaced by its inverse (all-zero, moving average) filter.

If TNS is used in a scalable coder with a TwinVQ coder, and if the TwinVQ layer does not use TNS, the TNS encoder filter of the first mono AAC layer have to be applied to the output of the TwinVQ decoder. These encoder filters use the LPC coefficients already decoded for the corresponding TNS decoder filters. The filters are slid across the specified target frequency range exactly the way described for the decoder filter. The difference between decoder and encoder filtering is that each all-pole (auto-regressive) decoder filter used for TNS decoding is replaced by its inverse (all-zero, moving average) filter.

The filter equation is :

$$y[n] = x[n] + lpc[1] * x[n-1] + ... + lpc[order] * x[n-order]$$

The number of filters, filtering direction etc. is controlled exactly like in the decoding process.

#### 4.6.9.5.1 Mono base + Stereo AAC without any AAC mono layer

If TNS is used in a scalable coder with a core coder, the TNS encoder filter of either the left or the right channel (depending on the **tns\_channel\_mono\_layer** flag) of the first AAC stereo layer has to be applied to the output of the core coder.

If TNS is used in a scalable coder with a TwinVQ coder, and if the TwinVQ layer does not use TNS, the TNS encoder filter of either the left or the right channel (depending on the **tns\_channel\_mono\_layer** flag) of the first AAC stereo layer have to be applied to the output of the TwinVQ decoder.

### 4.6.10 Spectrum normalization

#### 4.6.10.1 Tool description

In the TwinVQ decoder, spectral de-normalization is used in combination with inverse vector quantization of the MDCT coefficients, whose reproduction has globally flat shape. Using this tool, the spectral envelope is regenerated by decoding gain, Bark-scale envelope and an envelope specified with LPC parameters. Bark-scale envelope is reconstructed using a vector quantization decoder. LPC coefficients are quantized in LSP domain by means of 2-stage split vector quantization with moving average interframe prediction. Decoded LSP coefficients are directly used for generating an amplitude spectrum (square root of the power spectral envelope).

In a long MDCT block size mode, periodic peak components are optionally added to the flattened MDCT coefficients for the low rate coder.

#### 4.6.10.2 Definitions

|                             |                                                                                          |
|-----------------------------|------------------------------------------------------------------------------------------|
| <i>a</i>                    | MA prediction coefficients used for LSP quantization                                     |
| <i>alfq</i> [][]            | Predictive coefficients for Bark-envelope                                                |
| <i>AMP_MAX</i>              | maximum value of mu-law quantizer for global gain                                        |
| <i>AMP_NM</i>               | normalization factor for global gain                                                     |
| <i>BASF_STEP</i>            | step size of base frequency quantizer for periodic peak components coding                |
| <i>bfreq</i> []             | base frequency of periodic peak components                                               |
| <i>blim_h</i> []            | bandwidth control factor (higher part)                                                   |
| <i>blim_l</i> []            | bandwidth control factor (lower part)                                                    |
| <i>BLIM_STEP_H</i>          | number of steps of bandwidth control quantization (higher part)                          |
| <i>BLIM_STEP_L</i>          | number of steps of bandwidth control quantization (lower part)                           |
| <i>CUT_M_H</i>              | minimum bandwidth ratio (higher part)                                                    |
| <i>CUT_M_L</i>              | maximum bandwidth ratio (lower part)                                                     |
| <i>cv_env</i> [][]          | code vectors of envelope codebook                                                        |
| <i>env</i> [][]             | Bark-scale envelope projected onto Bark-scale frequency axis                             |
| <b>fb_shift</b> [][]        | Syntax element indicating the active frequency band of the adaptive bandwidth control.   |
| <i>FW_ALF_STEP</i>          | MA prediction coefficient for quantization of Bark-scale envelope                        |
| <i>FW_CB_LEN</i>            | length of code vector of Bark-scale envelope codebook                                    |
| <i>FW_N_DIV</i>             | number of interleaved division of Bark-scale envelope vector quantization                |
| <i>gain_p</i> [][]          | gain factors of the periodic peak components                                             |
| <i>gain</i> [][]            | gain factors of MDCT coefficients                                                        |
| <i>global_gain</i> []       | global gain of MDCT coefficients normalized by AMP_NM                                    |
| <b>index_blim_h</b> []      | syntax element indicating higher part bandwidth control                                  |
| <b>index_blim_l</b> []      | syntax element indicating lower part of bandwidth control                                |
| <b>index_env</b> [][][]     | syntax elements indicating Bark-scale envelope elements                                  |
| <b>index_fw_alf</b> []      | syntax element indicating the MA prediction switch of Bark-scale envelope quantization   |
| <b>index_gain</b> [][]      | syntax elements indicating global gain of MDCT coefficients                              |
| <b>index_gain_sb</b> [][][] | syntax elements indicating subblock gain of MDCT coefficients                            |
| <b>index_lsp0</b> [][]      | syntax elements indicating MA prediction coefficients used for LSP quantization          |
| <b>index_lsp1</b> []        | syntax element indicating the first-stage LSP quantization                               |
| <b>index_lsp2</b> [][]      | syntax element indicating the second-stage LSP quantization                              |
| <b>index_pgain</b> []       | syntax elements indicating gain of periodic peak components                              |
| <b>index_pit</b> [][]       | syntax elements indicating base frequency of periodic peak components                    |
| <b>index_shape0_p</b> []    | syntax element indicating peak elements quantization index for shape vector of conjugate |

|                         |                                                                                                                                                   |
|-------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------|
|                         | channel 0                                                                                                                                         |
| <b>index_shape1_p[]</b> | periodic peak elements quantization index for shape vector of conjugate channel 1                                                                 |
| <i>isp[]</i>            | split point table for second-stage LSP quantization                                                                                               |
| <i>lengthp[]</i>        | lengths of code vectors for periodic peak components quantization                                                                                 |
| <i>lnenv[][]</i>        | Bark-scale envelope projected onto linear-scale frequency axis                                                                                    |
| LOWER_BOUNDARY[][]      | lower boundary of active frequency band used in scalable layers                                                                                   |
| <i>lpenv[][]</i>        | LPC spectral envelope                                                                                                                             |
| <i>lsp[][]</i>          | LPC coefficients which range is set from zero to $\pi$ .                                                                                          |
| <i>lyr</i>              | indicates enhancement layer number. Number 0 is assigned for the base layer.                                                                      |
| LSP_SPLIT               | number of splits of 2nd-stage vector quantization for LSP coding                                                                                  |
| MU                      | mu factor for mu-law quantization for gain                                                                                                        |
| N_CRB                   | number of subbands for Bark-scale envelope coding                                                                                                 |
| N_DIV_P                 | number of interleave division for periodic peak components coding                                                                                 |
| N_FR                    | number of samples in a subframe                                                                                                                   |
| N_FR_P                  | number of elements of periodic peak components                                                                                                    |
| N_SF                    | number of subframe in a frame. The value is set in subclause 4.5.2.5.2.                                                                           |
| <i>p_cv_env[][]</i>     | Bark-scale envelope vector reconstructed in the previous frame                                                                                    |
| <i>pit[]</i>            | periodic peak components                                                                                                                          |
| <i>pit_seq[][][]</i>    | periodic peak components projected into linear scale                                                                                              |
| PIT_CB_SIZE             | size of codebook for periodic peak components quantization                                                                                        |
| PGAIN_MAX               | maximum value of mu-law quantizer for gain of periodic peak components                                                                            |
| PGAIN_MU                | mu factor for mu-law quantization for periodic peak components                                                                                    |
| PGAIN_STEP              | step size of mu-law quantizer for gain of periodic peak components                                                                                |
| <i>pol0_p</i>           | polarity of conjugate channel 0 for periodic peak components quantization                                                                         |
| <i>pol1_p</i>           | polarity of conjugate channel 1 for periodic peak components quantization                                                                         |
| <i>pit_cv0[] r</i>      | econstructed shape of conjugate channel 0 for periodic peak components quantization<br>(Elements are given in the first 64 rows of Table 4.A.28.) |
| <i>pit_cv1[]</i>        | reconstructed shape of conjugate channel 1 for periodic peak components quantization<br>(Elements are given in the last 64 rows of Table 4.A.28)  |
| STEP                    | step size of mu-law quantizer for global gain                                                                                                     |
| SUB_AMP_MAX             | maximum amplitude of mu-law quantizer for subframe gain ratio                                                                                     |
| SUB_AMP_NM              | normalization factor for subframe gain ratio                                                                                                      |
| SUB_STEP                | step size of mu-law quantizer for subframe gain                                                                                                   |
| <i>subg_ratio[]</i>     | subframe gain ratio                                                                                                                               |
| UPPER_BOUNDARY[][]      | upper boundary of active frequency band used in scalable layers                                                                                   |
| <i>v[]</i>              | LSP coefficients                                                                                                                                  |
| <i>v1[]</i>             | reconstructed vector from 1st-stage VQ for LSP coding                                                                                             |
| <i>v2[]</i>             | reconstructed vector from 2nd-stage VQ for LSP coding                                                                                             |
| <i>x_flat[]</i>         | normalized MDCT coefficients (input)                                                                                                              |
| <i>spec[][][]</i>       | de-normalized MDCT coefficients (output)                                                                                                          |

#### 4.6.10.3 Decoding process

The decoding process consists of five parts: gain decoding, Bark-scale envelope decoding, periodic peak components decoding, LPC spectrum decoding, and inverse normalization (see Figure 4.31).

##### 4.6.10.3.1 Initializations

Before starting any process, prediction memories *p\_cv\_env[][]* and  $\alpha_i^{(j)}$  are cleared.

##### 4.6.10.3.2 Gain decoding

In the first step of gain decoding, the global gain is decoded using mu-law inverse quantizer described as follows:

```
for (i_ch=0; i_ch<N_CH; i_ch++) {
    g_temp = index_gain * STEP + STEP / 2;
    global_gain =
        (AMP_MAX * (exp10(g_temp * log10(1.+MU) / AMP_MAX) - 1) / MU) / AMP_NM;
}
```

Next, subband gain ratios are decoded using mu-law inverse quantizer described as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  if (N_SF > 1){
    for(isf=0; isf<N_SF; isf++){
      g_temp = index_gain_sb[i_ch][isf+1] * SUB_STEP +SUB_STEP/2.;
      subg_ratio[isf] =
        (SUB_AMP_MAX*(exp10(g_temp*log10(1.+MU)/SUB_AMP_MAX)-1)/MU)
        / SUB_AMP_NM;
    }
  }
  else{
    subg_ratio[i_ch][0] = 1;
  }
}

```

Finally, gain factors are reconstructed as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  for(isf=0; isf<N_SF; isf++){
    gain[i_ch][isf] = global_gain[i_ch] * subg_ratio[i_ch][isf] / SUB_AMP_NM;
  }
}

```

#### 4.6.10.3.3 Decoding of periodic peak components

Periodic peak components are optionally added to the input coefficients. The periodic peak components are coded using vector quantization. This process is active when the parameter **ppc\_present** is set to TRUE. Otherwise, all the elements of output array, `pit_seq[]` are set to zero and the process is skipped.

```

MAXBIT_P = 6
PIT_CB_SIZE=(1<<MAXBIT_P)

```

##### 4.6.10.3.3.1 Decoding of polarity

```

for (idiv=0; idiv<N_DIV_P; idiv++){
  pol0[idiv] = 2*(index_shape0_p[idiv] / PIT_CB_SIZE) - 1;
  pol1[idiv] = 2*(index_shape1_p[idiv] / PIT_CB_SIZE) - 1;
}

```

##### 4.6.10.3.3.2 Decoding of shape code

```

for (idiv=0; idiv<N_DIV_P; idiv++){
  index0[idiv] = index_shape0_p[idiv] % PIT_CB_SIZE;
  index1[idiv] = index_shape1_p[idiv] % PIT_CB_SIZE;
}

```

Decoding gain of periodic peak components

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  temp = index_pgain[i_ch] * PGAIN_STEP + PGAIN_STEP / 2;
  gain_p[i_ch] =
    (PGAIN_MAX*(exp10(temp*log10(1.+PGAIN_MU)/PGAIN_MAX)-1)/PGAIN_MU)/AMP_NM;
}

```

##### 4.6.10.3.3.3 Reconstruction of periodic peak components

There are two steps of procedures. First the lengths of code vectors for periodic peak components, `lengthp[]`, are calculated. Then, the periodic peak components `pit[]` are calculated.

```

for (idiv=0; idiv<N_DIV_P; idiv++){
  lengthp[idiv] = (N_FR_P*N_CH+N_DIV_P-1-idiv) / N_DIV_P;
}

```

```

for (idiv=0; idiv<N_DIV_P; idiv++){

```

```

if(N_CH==1) {
  for (icv=0; icv<lengthp[idiv]; icv++){
    ismp = idiv + icv * N_DIV_P;
    pit[ismp] = (pol0[idiv]*pit_cv0[index0[idiv][icv]] + pol1[idiv]*pit_cv1[index1[idiv][icv]]) / 2;
  }
}
else{
  for (icv=0; icv<lengthp[idiv]-1; icv++){
    ismp = ((icv+idiv)%N_DIV_P)+ icv * N_DIV_P;
    ismp = ismp/2+(ismp%2)*20;
    pit[ismp] = (pol0[idiv]*pit_cv0[index0[idiv][icv]] + pol1[idiv]*pit_cv1[index1[idiv][icv]]) / 2;
  }
  icv= lengthp[idiv]-1;
  ismp = idiv + icv* N_DIV_P;
  ismp = ismp/2+(ismp%2)*20;
  pit[ismp] = (pol0[idiv]*pit_cv0[index0[idiv][icv]] + pol1[idiv]*pit_cv1[index1[idiv][icv]]) / 2;
}
}

```

#### 4.6.10.3.3.4 Projecting periodic peak components into linear scale

First, parameters are calculated as following:

```

fcmín = log2((N_FR/(double)ISAMPF)*0.2);
fcmáx = log2((N_FR/(double)ISAMPF)*2.4);
if (ISAMPF == 8) bandwidth = 1.5;
  else if (ISAMPF >= 11) bandwidth = 2.0;
  else if (ISAMPF >= 22) bandwidth = 4.0;
if (bandwidth < 1./UPPER_BOUNDARY[0][0] ) bandwidth = 1./UPPER_BOUNDARY[0][0];

```

where  $f_{min}$  is the minimum frequency to be quantization expressed in log scale,  $f_{max}$  is the maximum and ISAMPF is an integer sampling frequency in [kHz] truncated from the standard frequency values listed in the right column of Table 4.55.

Next, the base frequency of the periodic peak components is decoded according to the following procedure:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  pow_i = (int)(pow( 1.009792f, (float)index_pit[i_sup] ) *4096.+0.5);
  bl_i = (int)((float)block_size_samples/(float)isampf * 0.2*1024+0.5);
  pitch_i = pow_i *bl_i /256.;
  bfreq[i_ch] = pitch_i/16384.;
}

```

Before projecting the periodic peak components into linear scale, all the elements of target array `pit_seq[][]` is set to zero:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  for (ismp=0; ismp<N_FR; ismp++){
    pit_seq[i_ch][ismp] = 0.;
  }
}

```

Then, reconstructed periodic peak components are projected to linear-scale as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  if (bandwidth * upperlimit_i < 16384 ) {
    tmpnp0_i = pitch_i*16384. / (upperlimit_i);
    tmpnp1_i = tmpnp0_i *N_FR_P;
    tmpnp0_j = tmpnp1_i / N_FR;
    npcount = tmpnp0_i/16384.;
  } else {
    tmpnp0_i = pitch_i * bandwidth*2;
  }
}

```

```

    tmpnp1_i = tmpnp0_i * N_FR_P;
    tmpnp0_i = tmpnp1_i / N_FR;
    npcount = tmpnp0_i / 32768;
}

iscount=0;
for (jj=0; jj<npcount/2; jj++){
    pit_seq[i_ch][jj] = pit[jj+i_ch*N_FR_P];
    iscount ++;
}
for ( ii=0; ii<(ntt_N_FR_P)&& (iscount<ntt_N_FR_P); ii++) {
    tmpnp0_i = pitch_i * (ii+1);
    tmpnp0_i += 8192;
    i_smp = tmpnp0_i / 16384;

    for (jj=-npcount/2; jj<(npcount-1)/2+1; jj++){
        pit_seq[i_ch][i_smp+jj] = pit[iscount+i_ch*N_FR_P];
        iscount ++;
        if(iscount >= N_FR_P) break;
    }
}
}
}

```

In case of skipping the periodic peak components decoding process, all the elements of pitch component array `pit_seq[][]` are set to zero.

#### 4.6.10.3.4 Decoding of Bark-scale envelope

The Bark-scale envelope is decoded in each subframe. There are two procedure stages: inverse quantizing of the envelope vectors `env[][]` and projecting the bark-scale envelopes `env[][]` onto the linear scale envelopes `lenv[][]`.

##### 4.6.10.3.4.1 Inverse quantization of envelope vector

The inverse quantization part is illustrated in Figure 4.26.

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    for (isf=0; isf<N_SF; isf++){
        alfq[i_ch][isf] = index_fw_alf[i_ch][isf] * FW_ALF_STEP;
        for (ifdiv=0; ifdiv<FW_N_DIV; ifdiv++){
            for (icv=0; icv<FW_CB_LEN; icv++){
                ienv = FW_N_DIV * icv + ifdiv;
                dtmp = cv_env[index_env[ich][isf][ifdiv]][icv];
                env[i_ch][isf][ienv] = dtmp + alfq[i_ch][isf] * p_cv_env[i_ch][icv] + 1;
                p_cv_env[i_ch][icv] = dtmp;
            }
        }
    }
}
}

```

The `cv_env[][]` is the Bark-scale envelope codebook listed in Table 4.A.27.

##### 4.6.10.3.4.2 Projecting the Bark-scale envelope onto a linear scale

The envelopes `env[][]` are expressed using the Bark scale on the frequency axis. The de-normalization procedure requires a linear-scale envelopes.

Before the projecting process, the boundary table of the bark-scale subband, `crb_tbl[]`, is determined. In case of the base layer (`lyr = 0`),

```

if(sampling_rate <= 16000)
    use the scalefactor band table of AAC for 16 kHz up to 41st value (Table 4.78)
else
    use the scalefactor band table of AAC for 24 kHz up to 41st value (Table 4.80)

```



Number of scalefactor bands is 42 for long frame size.

42nd value is the frame length of the long frame (1024 or 960).

Barkscale estimation is not used for short frames.

If  $lyr \geq 1$ , the values of the Bark-scale subband table are calculated as follows.

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  lower_band_i = (int)(AC_BTM[lyr][i_ch][fb_shift]*16384.);
  upper_band_i = (int)( AC_TOP[lyr][i_ch][fb_shift]*16384.);
  average_number_of_lines=
    (int)(frame_length*(upper_band_i-lower_band_i))/N_CRB;
    for (i=0; i<N_CRB-1; i++){
      crb_tbl[i_ch][i] = (int)((i+1)*(i+1)* average_number_of_lines/ N_CRB/2.0)
      crb_tbl[i_ch][i] += (i+1)* average_number_of_lines/2.0 +8192;
      crb_tbl[i_ch][i] /= 16384.
      crb_tbl[i_ch][i] += (int)(frame_length*lower_band_i)/16384.;
    }
    crb_tbl[i_ch][N_CRB-1] = (int)(frame_length*lower_band_i)/16384.
    +(int)(frame_length*(upper_band_i-lower_band_i))/16384.;
}

```

After the  $crb\_tbl[][]$  is determined, the projecting process is done as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
  for (isf=0; isf<N_SF; isf++){
    ismp=0
    for (ienv=0; ienv<N_CRB; ienv++){
      while (ismp<crb_tbl[i_ch][ienv]){
        lnenv[i_ch][isf][ismp] = env[i_ch][isf][ienv];
        ismp++;
      }
    }
  }
}

```

The values of the UPPER\_BOUNDARY[][] and LOWER\_BOUNDARY[][] are defined in subclause 4.6.5.4.4.

If **postprocess\_present** is active,  $lnenv[i\_ch][isf][ismp]$  is modified as follows:

If  $(lnenv[i\_ch][isf][ismp] < 1.0)$   $lnenv[i\_ch][isf][ismp] = lnenv[i\_ch][isf][ismp] * lnenv[i\_ch][isf][ismp];$

#### 4.6.10.3.5 Decoding of LPC spectrum

The LPC spectrum is represented by LSP coefficients. In the decoding process the LSP coefficients are reconstructed first; then they are transformed into the LPC spectrum, which represents the square root of the power spectrum.

##### 4.6.10.3.5.1 LSP coefficients decoding using the MA prediction

MA prediction coefficients are determined by referring to the coefficient table  $\alpha t_i^{(j)}$ . The rule is:

$$\alpha_i^{(j)}[i\_ch] = \alpha t_i^{(j)}[i\_ch](index\_lsp0[i\_ch]) \quad \text{for } i = 1 \text{ to } N\_PR, j = 1 \text{ to } MA\_NP, i\_ch = 0 \text{ to } N\_CH - 1,$$

where  $i$  is LPC order and  $j$  is MA prediction order. The coefficient table  $\alpha_i^{(j)}$  is listed in subclause Table 4.A.4, Table 4.A.17 and Table 4.A.18 (81th and 82th row).

#### 4.6.10.3.5.2 First-stage inverse quantization of LSP decoding

$$v1_i[i\_ch] = \text{lspcode1}_i(\text{index\_lsp1}[i\_ch]) \quad \text{for } i = 1 \text{ to } N\_PR, i\_ch = 0 \text{ to } N\_CH - 1,$$

where  $\text{lspcode1}$  is the first-stage LSP codebook listed in subclause Table 4.A.4, Table 4.A.17 and Table 4.A.18 (from 1st to 64th row).

#### 4.6.10.3.5.3 Second-stage inverse quantization of LSP decoding

$$v2_i[i\_ch] = \text{lspcode2}_i(\text{index\_lsp2}[i\_ch][k]) \\ \text{for } k = 0 \text{ to } LSP\_SPLIT - 1, i = \text{isp}(k) + 1 \text{ to } \text{isp}(k + 1) - 1, i\_ch = 0 \text{ to } N\_CH - 1,$$

where  $\text{lspcode2}$  is the second-stage LSP codebook listed in subclause 4.A.4 (from 65th to 80th row). Values of  $\text{isp}(k)$  are listed in Table 4.107.

#### 4.6.10.3.5.4 Reconstruction of LSP coefficients

LSP coefficients  $\text{lsp}[][]$  are calculated as follows:

$$\vec{v}[i\_ch] = v1[i\_ch] + v2[i\_ch], \quad \text{for } i\_ch = 0 \text{ to } N\_CH - 1 \\ \alpha_i^{(0)}[i\_ch] = 1 - \sum_{j=1}^{MA\_NP} \alpha_i^{(j)}[i\_ch] \quad \text{for } i = 1 \text{ to } N\_PR, i\_ch = 0 \text{ to } N\_CH - 1 \\ \text{lsp}[i\_ch][i] = \sum_{j=0}^{MA\_NP} \alpha_i^{(j)}[i\_ch] \cdot v_i^{(-j)}[i\_ch] \quad \text{for } i = 1 \text{ to } N\_PR, i\_ch = 0 \text{ to } N\_CH - 1 \\ \hat{v}^{(j-1)}[i\_ch] = \hat{v}^{(j)}[i\_ch] \quad \text{for } j = -MA\_NP - 1 \text{ to } 0, i\_ch = 0 \text{ to } N\_CH - 1$$

#### 4.6.10.3.5.5 Transforming LSP parameters into LPC spectrum

The LPC amplitude spectrum envelope corresponding to the  $ii$ -th MDCT coefficient,  $\text{lpenv}[[ii]$ , is defined as follows:

Note that  $\text{lpenv}[[ii]$  seems to represent the power spectrum envelope, but actually represents the amplitude envelope, since the original LPC spectral envelope is derived from the square root of the power spectrum at the encoder.

```
for (i_ch=0; i_ch<N_CH; i_ch++){
  for (ii=1; ii<=N_FR-1; ii++){
    for (i=2, P[i_ch]=1.0; i<=N_PR; i+=2)
      P[i_ch] *= (cos(PI*ii/N_FR)-cos(lsp[i]))^2;
    for (i=1, Q[i_ch]=1.0; i<=N_PR; i+=2)
      Q[i_ch] *= (cos(PI*ii/N_FR)-cos(lsp[i]))^2;
    lpenv[i_ch][ii] = 1/((1-cos(PI*ii/N_FR))*P[i_ch] + (1+cos(PI*ii/N_FR))*Q[i_ch]);
  }
}
```

In the case of long frames ( $N\_FR == 1024$ , or  $960$ ),  $\text{lpenv}[[ii]$  shall be calculated only at frequency points  $ii$  that satisfy the following conditions:

```
(
  ((ii = 0 <= ii < N_FR/2) && (ii%4==0))
)
```

```

    || ((ii= 0 <= ii <N_FR/2-4) && ((ii%2==0) && (lpenv[][ii/4*4]>lpenv[][ii/4*4+8] ) &&
        (lpenv[][ii/4*4+4]*1.95 > lpenv[][ii/4*4] + lpenv[][ii/4*4+8] )))
    || ((ii= N_FR/2<= ii <N_FR) && (ii%8==0))
    || ((ii= N_FR/2<= ii <N_FR-8) && ((ii%4==0) && (lpenv[][ii/8*8]>lpenv[][ii/8*8+16] ) &&
        (lpenv[][ii/8*8+8]*1.95 > lpenv[][ii/8*8] + lpenv[][ii/8*8+16] )))
).

```

For the remaining frequency points, the `lpenv[ii]` values are calculated by linear interpolation from the values already calculated at the nearest two frequency points. If frequency points are larger than `N_FR-8`, `lpenv[][ii]` shall be equal to `lpenv[][N_FR-8]`.

If this tool is used as an element of scalable coder, LPC spectrum is squeezed into the active frequency band:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    nfr_lu = UPPER_BOUNDARY[lyr][i_ch] - LOWER_BOUNDARY[lyr][i_ch];
    for (ismp=0; ismp<LOWER_BOUNDARY[lyr][i_ch]; ismp++){
        lpenv_tmp[i_ch][ismp] = 0;
    }
    upperband_i = (int)( AC_TOP[lyr][i_ch][fb_shift] * 16384.);
    lowerband_i = (int)( AC_BTM[lyr][i_ch][fb_shift] * 16384.);
    ftmp = (16384*16384)/( upperband_i - lowerband_i );
    for(ismp=0; ismp<nfr_lu; ismp++){
        ftmp = (int)(ismp*ftmp)/16384;
        lpenv_tmp[i_ch][ismp+LOWER_BOUNDARY[lyr][i_ch]] = lpenv[i_ch][ftmp];
    }

    for (ismp=UPPER_BOUNDARY[lyr][i_ch]; ismp<N_FR; ismp++){
        lpenv_tmp[i_ch][ismp] = 0;
    }
    for(ismp=0; ismp<N_FR; ismp++){
        lpenv[i_ch][ismp] = lpenv_tmp[i_ch][ismp];
    }
}

```

The values of `UPPER_BOUNDARY`, `LOWER_BOUNDARY`, `AC_TOP` and `AC_BTM` are defined in subclause 4.6.5.4.4.

#### 4.6.10.3.6 Inverse normalization

The input coefficients `x_flat[]` are applied to inverse normalization according to the following procedure, and output coefficients `spec[][][]` are created.

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    for (isf=0; isf<N_SF; isf++){
        for (ismp=0; ismp<N_FR; ismp++){
            spec[isf][ismp] =
                (x_flat[ismp+(isf+i_ch*N_SF)*N_FR]
                 *lnenv[i_ch][isf][ismp]* gain[i_ch][isf]+ p_gain[i_ch]*pit_seq[i_ch][ismp])
                 *lpenv[i_ch][ismp];
        }
    }
}

```

#### 4.6.10.3.7 Bandwidth control

This functionality is valid only if `bandlimit_present` flag is ON.

After the inverse normalization the upper and lower bands of the output coefficients `spec[][][]` are set to zero.

In the bandwidth decoding modules the higher signal bandwidth ratio `blim_h[]` is decoded as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    blim_h[i_ch] =
        (1. - (1.-CUT_M_H) * (double)index_blim_h[i_ch]/(double)BLIM_STEP_H)

```

```

    * UPPER_BOUNDARY[0][i_ch];
}

```

The `blim_l[]` is decoded as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    if (index_blim_l[i_ch] == 1) blim_l[i_ch] = LOWER_BOUNDARY[0][i_ch]+CUT_M_L;
    else
        blim_l[i_ch] = 0;
}

```

In the bandwidth limitation module, higher and lower parts of MDCT coefficients are set to zero as follows:

```

for (i_ch=0; i_ch<N_CH; i_ch++){
    NbaseH = blim_h[i_ch] * N_FR;
    NbaseL = blim_l[i_ch] * N_FR;
    for (isf=0; isf<N_SF; isf++){
        for (ismp=NbaseH; ismp<N_FR; ismp++){
            spec[i_ch][isf][ismp] = 0;
        }
        for (ismp=0; ismp<NbaseL; ismp++){
            spec[i_ch][isf][ismp] = 0;
        }
    }
}

```

#### 4.6.10.4 Diagrams

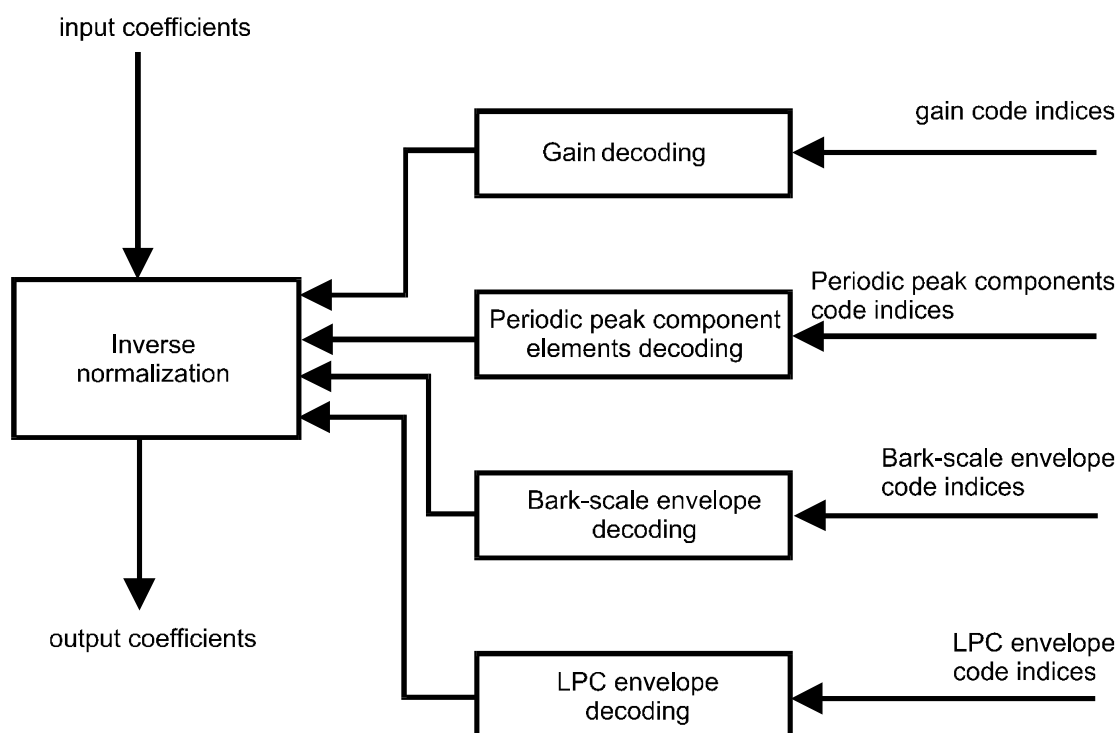


Figure 4.31 – Decoding process of the TwinVQ

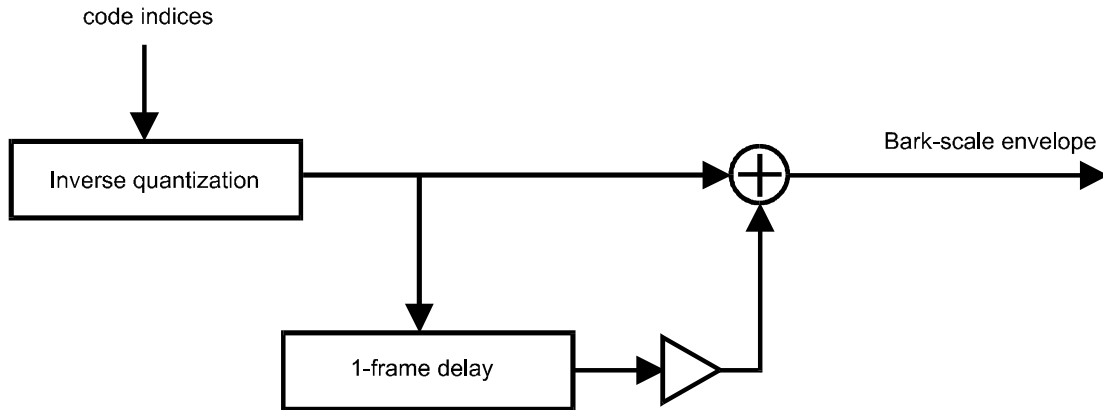


Figure 4.32 – Reconstruction process of the Bark-scale envelope

4.6.10.5 Tables

Parameters used in the inverse spectrum normalization process are set as follows. Other parameters related to the syntax are defined in subclause 4.5.2.5.3.

Table 4.105 – Parameter table for the core and core\_960 mode

| Parameter    | Value                                          | Meaning                                                  |
|--------------|------------------------------------------------|----------------------------------------------------------|
| Common:      |                                                |                                                          |
| AMP_MAX      | 16000                                          | Maximum amplitude in the mu-law coding of the frame gain |
| AMP_NM       | 1024                                           | Normalized amplitude of flattened coefficients           |
| BLIM_BITS_H  | 2                                              | Bits for higher bandwidth control                        |
| BLIM_BITS_L  | 1                                              | Bits for lower bandwidth control                         |
| BLIM_STEP_H  | 4                                              | Number of steps of band limitation quantization          |
| CUT_M_H      | 0.7                                            | Minimum bandwidth ratio (higher part)                    |
| CUT_M_L      | 0.0025                                         | Maximum bandwidth ratio (lower part)                     |
| FW_ALF_STEP  | 0.5                                            | MA prediction coefficient for the envelope coding        |
| MA_NP        | 1                                              | MA prediction order (LSP coding)                         |
| MU           | 100                                            | Parameter mu in the mu-law coding of the frame power     |
| N_PR         | 20                                             | LPC order                                                |
| NUM_STEP     | 512                                            | Number of gain-quantization steps                        |
| STEP         | $AMP\_MAX / (NUM\_STEP - 1) = 31.31$           | Step width of gain-quantization                          |
| SUB_AMP_MAX  | 4700                                           | Maximum of subframe-gain to frame-gain ratio             |
| SUB_AMP_NM   | 1024                                           | Normalized subframe amplitude of flattened coefficients  |
| SUB_GAIN_BIT | 4                                              | Number of bits for the subframe gain coding              |
| SUB_NUM_STEP | 16                                             | Number of sub-gain-quantization steps                    |
| SUB_STEP     | $SUB\_AMP\_MAX / (SUB\_NUM\_STEP - 1) = 313.3$ | Step width of sub-gain-quantization                      |
| PGAIN_MAX    | 20000                                          | maximum value of mu-law quantizer for gain of ppc        |
| PGAIN_MU     | 200                                            | mu factor for mu-law quantization for ppc                |
| PGAIN_STEP   | $PGAIN\_MAX / (1 \ll PGAIN\_BIT)$              | step size of mu-law quantizer for gain of ppc            |
| Long block:  |                                                |                                                          |
| FW_CB_LEN    | 6                                              | Envelope code vector length                              |
| FW_N_BIT     | 6                                              | Envelope code bits                                       |
| FW_N_DIV     | 7                                              | Number of divisions in the envelope coding               |
| N_CRB        | 42                                             | Number of bark-scale subbands                            |

|              |          |                 |
|--------------|----------|-----------------|
| N_FR         | 1024/960 | MDCT block size |
| Short block: |          |                 |
| N_FR         | 128/120  | MDCT block size |

Table 4.106 – Parameter table for the enhance and enhance\_960 mode

| Parameter    | Value                                          | Meaning                                                  |
|--------------|------------------------------------------------|----------------------------------------------------------|
| Common:      |                                                |                                                          |
| AMP_MAX      | 8000                                           | Maximum amplitude in the mu-law coding of the frame gain |
| AMP_NM       | 1024                                           | Normalized amplitude of flattened coefficients           |
| FW_ALF_STEP  | 0.5                                            | MA prediction coefficient for the envelope coding        |
| MA_NP        | 1                                              | MA prediction order (LSP coding)                         |
| MU           | 100                                            | Parameter mu in the mu-law coding of the frame power     |
| N_PR         | 20                                             | LPC order                                                |
| NUM_STEP     | 256                                            | Number of gain-quantization steps                        |
| STEP         | $AMP\_MAX / (NUM\_STEP - 1) = 31.37$           | Step width of gain-quantization                          |
| SUB_AMP_MAX  | 6000                                           | Maximum of subframe-gain to frame-gain ratio             |
| SUB_AMP_NM   | 1024                                           | Normalized subframe amplitude of flattened coefficients  |
| SUB_GAIN_BIT | 4                                              | Number of bits for the subframe gain coding              |
| SUB_NUM_STEP | 16                                             | Number of sub-gain-quantization steps                    |
| SUB_STEP     | $SUB\_AMP\_MAX / (SUB\_NUM\_STEP - 1) = 400.0$ | Step width of sub-gain-quantization                      |
| Long block:  |                                                |                                                          |
| FW_CB_LEN    | 7                                              | Envelope code vector length                              |
| FW_N_BIT     | 6                                              | Envelope code bits                                       |
| FW_N_DIV     | 7                                              | Number of division in the envelope coding                |
| N_CRB        | 42                                             | Number of bark-scale subbands                            |
| N_FR         | 1024/960                                       | MDCT block size                                          |
| Short block: |                                                |                                                          |
| N_FR         | 128/120                                        | MDCT block size                                          |

Table 4.107 – Values of isp[]

| split_num | isp[split_num] |
|-----------|----------------|
| 0         | 0              |
| 1         | 5              |
| 2         | 14             |
| 3         | 20             |

#### 4.6.11 Filterbank and block switching

(Similar to ISO/IEC 13818-7)

##### 4.6.11.1 Tool description

The time/frequency representation of the signal is mapped onto the time domain by feeding it into the filterbank module. This module consists of an inverse modified discrete cosine transform (IMDCT), and a window and an overlap-add function. In order to adapt the time/frequency resolution of the filterbank to the characteristics of the input signal, a block switching tool is also adopted.  $N$  represents the window length, where  $N$  is a function of the **window sequence** (see subclauses 4.5.2.3.3). For each channel, the  $N/2$  time-frequency values  $X_{i,k}$  are transformed into the  $N$  time domain values  $x_{i,n}$  via the IMDCT. After applying the window function, for each channel, the first half of the  $z_{i,n}$  sequence is added to the second half of the previous block windowed sequence  $z_{(i-1),n}$  to reconstruct the output samples for each channel  $out_{i,n}$ .

4.6.11.2 Definitions

**window\_sequence** 2 bit indicating which window sequence (i.e. block size) is used.  
**window\_shape** 1 bit indicating which window function is selected.

Table 4.72 shows the four **window\_sequences** (ONLY\_LONG\_SEQUENCE, LONG\_START\_SEQUENCE, EIGHT\_SHORT\_SEQUENCE, LONG\_STOP\_SEQUENCE).

4.6.11.3 Decoding process

4.6.11.3.1 IMDCT

The analytical expression of the IMDCT is:

$$x_{i,n} = \frac{2}{N} \sum_{k=0}^{\frac{N}{2}-1} spec[i][k] \cos\left(\frac{2\pi}{N} \left(n + n_0\right) \left(k + \frac{1}{2}\right)\right) \text{ for } 0 \leq n < N$$

where:

- n = sample index
- i = window index
- k = spectral coefficient index
- N = window length based on the window\_sequence value
- n<sub>0</sub> = (N / 2 + 1) / 2

The synthesis window length N for the inverse transform is a function of the syntax element **window\_sequence** and the algorithmic context. It is defined as follows:

Window length 2048:

$$N = \begin{cases} 2048, & \text{if ONLY\_LONG\_SEQUENCE (0x0)} \\ 2048, & \text{if LONG\_START\_SEQUENCE (0x1)} \\ 256, & \text{if EIGHT\_SHORT\_SEQUENCE (0x2), (8 times)} \\ 2048, & \text{if LONG\_STOP\_SEQUENCE (0x3)} \end{cases}$$

Window length 1920:

$$N = \begin{cases} 1920, & \text{if ONLY\_LONG\_SEQUENCE (0x0)} \\ 1920, & \text{if LONG\_START\_SEQUENCE (0x1)} \\ 240, & \text{if EIGHT\_SHORT\_SEQUENCE (0x2), (8 times)} \\ 1920, & \text{if LONG\_STOP\_SEQUENCE (0x3)} \end{cases}$$

The meaningful block transitions are as follows:

|                           |    |                                              |
|---------------------------|----|----------------------------------------------|
| from ONLY_LONG_SEQUENCE   | to | { ONLY_LONG_SEQUENCE<br>LONG_START_SEQUENCE  |
| from LONG_START_SEQUENCE  | to | { EIGHT_SHORT_SEQUENCE<br>LONG_STOP_SEQUENCE |
| from LONG_STOP_SEQUENCE   | to | { ONLY_LONG_SEQUENCE<br>LONG_START_SEQUENCE  |
| from EIGHT_SHORT_SEQUENCE | to | { EIGHT_SHORT_SEQUENCE<br>LONG_STOP_SEQUENCE |

In addition to the meaningful block transitions the following transitions are possible:

|                           |    |                                              |
|---------------------------|----|----------------------------------------------|
| from ONLY_LONG_SEQUENCE   | to | { EIGHT_SHORT_SEQUENCE<br>LONG_STOP_SEQUENCE |
| from LONG_START_SEQUENCE  | to | { ONLY_LONG_SEQUENCE<br>LONG_START_SEQUENCE  |
| from LONG_STOP_SEQUENCE   | to | { EIGHT_SHORT_SEQUENCE<br>LONG_STOP_SEQUENCE |
| from EIGHT_SHORT_SEQUENCE | to | { ONLY_LONG_SEQUENCE<br>LONG_START_SEQUENCE  |

This will still result in a reasonably smooth transition from one block to the next.

#### 4.6.11.3.2 Windowing and block switching

Depending on the **window\_sequence** and **window\_shape** element different transform windows are used. A combination of the window halves described as follows offers all possible window\_sequences.

For **window\_shape** == 1, the window coefficients are given by the Kaiser - Bessel derived (KBD) window as follows:

$$W_{KBD\_LEFT, N}(n) = \frac{\sum_{p=0}^n [W'(p, \alpha)]}{\sum_{p=0}^{N/2} [W'(p, \alpha)]} \quad \text{for } 0 \leq n < \frac{N}{2}$$



$$W_{KBD\_RIGHT,N}(n) = \sqrt{\frac{\sum_{p=0}^{N-n-1} [W'(p,\alpha)]}{\sum_{p=0}^{N/2} [W'(p,\alpha)]}} \quad \text{for } \frac{N}{2} \leq n < N$$

where:

$W'$ , Kaiser - Bessel kernel window function, see also [5], is defined as follows:

$$W'(n,\alpha) = \frac{I_0\left[\pi\alpha\sqrt{1.0 - \left(\frac{n - N/4}{N/4}\right)^2}\right]}{I_0[\pi\alpha]} \quad \text{for } 0 \leq n \leq \frac{N}{2}$$

$$I_0[x] = \sum_{k=0}^{\infty} \left[ \frac{\left(\frac{x}{2}\right)^k}{k!} \right]^2$$

$\alpha$  = kernel window alpha factor,  $\alpha = \begin{cases} 4 & \text{for } N = 2048 \text{ (1920)} \\ 6 & \text{for } N = 256 \text{ (240)} \end{cases}$

Otherwise, for **window\_shape** == 0, a sine window is employed as follows:

$$W_{SIN\_LEFT,N}(n) = \sin\left(\frac{\pi}{N}\left(n + \frac{1}{2}\right)\right) \quad \text{for } 0 \leq n < \frac{N}{2}$$

$$W_{SIN\_RIGHT,N}(n) = \sin\left(\frac{\pi}{N}\left(n + \frac{1}{2}\right)\right) \quad \text{for } \frac{N}{2} \leq n < N$$

The window length  $N$  can be 2048 (1920) or 256 (240) for the KBD and the sine window. How to obtain the possible window sequences is explained in the parts a)-d) of this subclause. All four window\_sequences described below have a total length of 2048 samples.

For all kinds of window\_sequences the window\_shape of the left half of the first transform window is determined by the window shape of the previous block. The following formula expresses this fact:

$$W_{LEFT,N}(n) = \begin{cases} W_{KBD\_LEFT,N}(n), & \text{if } \text{window\_shape\_previous\_block} == 1 \\ W_{SIN\_LEFT,N}(n), & \text{if } \text{window\_shape\_previous\_block} == 0 \end{cases}$$

where:

*window\_shape\_previous\_block*: **window\_shape** of the previous block (i-1).

For the first block of the bitstream to be decoded the **window\_shape** of the left and right half of the window are identical.

**a) ONLY\_LONG\_SEQUENCE:**

The **window\_sequence** == ONLY\_LONG\_SEQUENCE is equal to one LONG\_WINDOW with a total window length  $N_l$  of 2048 (1920).

For **window\_shape**==1 the window for ONLY\_LONG\_SEQUENCE is given as follows:

$$W(n) = \begin{cases} W_{LEFT,N_l}(n), & \text{for } 0 \leq n < N_l/2 \\ W_{KBD\_RIGHT,N_l}(n), & \text{for } N_l/2 \leq n < N_l \end{cases}$$

If **window\_shape**==0 the window for ONLY\_LONG\_SEQUENCE can be described as follows:

$$W(n) = \begin{cases} W_{LEFT,N_l}(n), & \text{for } 0 \leq n < N_l/2 \\ W_{SIN\_RIGHT,N_l}(n), & \text{for } N_l/2 \leq n < N_l \end{cases}$$

After windowing, the time domain values ( $z_{i,n}$ ) can be expressed as:

$$z_{i,n} = w(n) \cdot x_{i,n};$$

#### b) LONG\_START\_SEQUENCE:

The LONG\_START\_SEQUENCE is needed to obtain a correct overlap and add for a block transition from a ONLY\_LONG\_SEQUENCE to a EIGHT\_SHORT\_SEQUENCE.

Window length  $N_l$  and  $N_s$  is set to 2048 (1920) and 256 (240) respectively.

If **window\_shape**==1 the window for LONG\_START\_SEQUENCE is given as follows:

$$W(n) = \begin{cases} W_{LEFT,N_l}(n), & \text{for } 0 \leq n < N_l/2 \\ 1.0, & \text{for } N_l/2 \leq n < \frac{3N_l - N_s}{4} \\ W_{KBD\_RIGHT,N_s}(n + \frac{N_s}{2} - \frac{3N_l - N_s}{4}), & \text{for } \frac{3N_l - N_s}{4} \leq n < \frac{3N_l + N_s}{4} \\ 0.0, & \text{for } \frac{3N_l + N_s}{4} \leq n < N_l \end{cases}$$

If **window\_shape**==0 the window for LONG\_START\_SEQUENCE looks like:

$$W(n) = \begin{cases} W_{LEFT,N_l}(n), & \text{for } 0 \leq n < N_l/2 \\ 1.0, & \text{for } N_l/2 \leq n < \frac{3N_l - N_s}{4} \\ W_{SIN\_RIGHT,N_s}(n + \frac{N_s}{2} - \frac{3N_l - N_s}{4}), & \text{for } \frac{3N_l - N_s}{4} \leq n < \frac{3N_l + N_s}{4} \\ 0.0, & \text{for } \frac{3N_l + N_s}{4} \leq n < N_l \end{cases}$$

The windowed time-domain values can be calculated with the formula explained in a).

#### c) EIGHT\_SHORT

The **window\_sequence** == EIGHT\_SHORT comprises eight overlapped and added SHORT\_WINDOWS with a length  $N_s$  of 256 (240) each. The total length of the window\_sequence together with leading and following zeros is 2048 (1920). Each of the eight short blocks are windowed separately first. The short block number is indexed with the variable  $j = 0, \dots, M-1$  ( $M=N_l/N_s$ ).

The **window\_shape** of the previous block influences the first of the eight short blocks ( $W_0(n)$ ) only. If **window\_shape**==1 the window functions can be given as follows:

$$W_0(n) = \begin{cases} W_{LEFT,N_s}(n), & \text{for } 0 \leq n < N_s/2 \\ W_{KBD\_RIGHT,N_s}(n), & \text{for } N_s/2 \leq n < N_s \end{cases}$$

$$W_{1-(M-1)}(n) = \begin{cases} W_{KBD\_LEFT,N_s}(n) & \text{for } 0 \leq n < N_s/2 \\ W_{KBD\_RIGHT,N_s}(n), & \text{for } N_s/2 \leq n < N_s \end{cases}$$

Otherwise, if **window\_shape**==0, the window functions can be described as:

$$W_0(n) = \begin{cases} W_{LEFT,N_s}(n), & \text{for } 0 \leq n < N_s/2 \\ W_{SIN\_RIGHT,N_s}(n), & \text{for } N_s/2 \leq n < N_s \end{cases}$$

$$W_{1-(M-1)}(n) = \begin{cases} W_{SIN\_LEFT,N_s}(n), & \text{for } 0 \leq n < N_s/2 \\ W_{SIN\_RIGHT,N_s}(n), & \text{for } N_s/2 \leq n < N_s \end{cases}$$

The overlap and add between the EIGHT\_SHORT **window\_sequence** resulting in the windowed time domain values  $z_{i,n}$  is described as follows:

$$z_{i,n} = \begin{cases} 0, & \text{for } 0 \leq n < \frac{N_l - N_s}{4} \\ x_{0,n - \frac{N_l - N_s}{4}} \cdot W_0\left(n - \frac{N_l - N_s}{4}\right), & \text{for } \frac{N_l - N_s}{4} \leq n < \frac{N_l + N_s}{4} \\ x_{j-1,n - \frac{N_l + (2j-3)N_s}{4}} \cdot W_{j-1}\left(n - \frac{N_l + (2j-3)N_s}{4}\right) + x_{j,n - \frac{N_l + (2j-1)N_s}{4}} \cdot W_j\left(n - \frac{N_l + (2j-1)N_s}{4}\right), & \text{for } 1 \leq j < M, \frac{N_l + (2j-1)N_s}{4} \leq n < \frac{N_l + (2j+1)N_s}{4} \\ x_{M-1,n - \frac{N_l + (2M-3)N_s}{4}} \cdot W_{M-1}\left(n - \frac{N_l + (2M-3)N_s}{4}\right), & \text{for } \frac{N_l + (2M-1)N_s}{4} \leq n < \frac{N_l + (2M+1)N_s}{4} \\ 0, & \text{for } \frac{N_l + (2M+1)N_s}{4} \leq n < N_l \end{cases}$$

**d) LONG\_STOP\_SEQUENCE**

This **window\_sequence** is needed to switch from a EIGHT\_SHORT\_SEQUENCE back to a ONLY\_LONG\_SEQUENCE.

If **window\_shape** == 1 the window for LONG\_STOP\_SEQUENCE is given as follows:

$$W(n) = \begin{cases} 0.0, & \text{for } 0 \leq n < \frac{N_l - N_s}{4} \\ W_{LEFT,N_s}\left(n - \frac{N_l - N_s}{4}\right), & \text{for } \frac{N_l - N_s}{4} \leq n < \frac{N_l + N_s}{4} \\ 1.0, & \text{for } \frac{N_l + N_s}{4} \leq n < N_l/2 \\ W_{KBD\_RIGHT,N_l}(n), & \text{for } N_l/2 \leq n < N_l \end{cases}$$

If **window\_shape** == 0 the window for LONG\_START\_SEQUENCE is determined by:

$$W(n) = \begin{cases} 0.0, & \text{for } 0 \leq n < \frac{N_l - N_s}{4} \\ W_{LEFT, N_s}(n - \frac{N_l - N_s}{4}), & \text{for } \frac{N_l - N_s}{4} \leq n < \frac{N_l + N_s}{4} \\ 1.0, & \text{for } \frac{N_l + N_s}{4} \leq n < N_l / 2 \\ W_{SIN\_RIGHT, N_l}(n), & \text{for } N_l / 2 \leq n < N_l \end{cases}$$

The windowed time domain values can be calculated with the formula explained in a).

#### 4.6.11.3.3 Overlapping and adding with previous window sequence

Besides the overlap and add within the EIGHT\_SHORT **window\_sequence** the first (left) half of every **window\_sequence** is overlapped and added with the second (right) half of the previous **window\_sequence** resulting in the final time domain values  $out_{i,n}$ . The mathematic expression for this operation can be described as follows. It is valid for all four possible **window\_sequences**.

$$out_{i,n} = z_{i,n} + z_{i-1, n + \frac{N}{2}}; \quad \text{for } 0 \leq n < \frac{N}{2}, \quad N = 2048 \text{ (1920)}$$

### 4.6.12 Gain Control

#### 4.6.12.1 Tool description

The gain control tool is made up of several gain compensators and overlap/add processing stages, and an IPQF (Inverse Polyphase Quadrature Filter) stage. This tool receives non-overlapped signal sequences provided by the IMDCT stages, **window\_sequence** and **gain\_control\_data**, and then reproduces the output PCM data. The block diagram for the gain control tool is shown in Figure 4.33.

Due to the characteristics of the PQF filterbank, the order of the MDCT coefficients in each even PQF band must be reversed. This is done by reversing the spectral order of the MDCT coefficients, i.e. exchanging the higher frequency MDCT coefficients with the lower frequency MDCT coefficients.

If the gain control tool is used, the configuration of the filter bank tool is changed as follows. In the case of an EIGHT\_SHORT\_SEQUENCE **window\_sequence**, the number of coefficients for the IMDCT is 32 instead of 128 and eight IMDCTs are carried out. In the case of other **window\_sequence** values, the number of coefficients for the IMDCT is 256 instead of 1024 and one IMDCT is performed. In all cases, the filter bank tool outputs a total of 2048 non-overlapped values per frame. These values are supplied to the gain control tool as  $U_{W,B}(j)$  defined in subclause 4.6.12.3.3.

The IPQF combines four uniform frequency bands and produces a decoded time domain output signal. The aliasing components introduced by the PQF in the encoder are cancelled by the IPQF.

The gain values for each band can be controlled independently except for the lowest frequency band. The step size of gain control is  $2^n$  where  $n$  is an integer.

The gain control tool outputs a time signal sequence, which is  $AS(n)$  defined in subclause 4.6.12.3.4.

#### 4.6.12.2 Definitions

|                          |                                                                                                                   |
|--------------------------|-------------------------------------------------------------------------------------------------------------------|
| <b>gain control data</b> | side information indicating the gain values and the positions used for the gain change.                           |
| <b>IPQF band</b>         | each split band of IPQF.                                                                                          |
| <b>adjust_num</b>        | 3-bit field indicating the number of gain changes for each IPQF band. The maximum number of gain changes is seven |
| <b>max_band</b>          | 2-bit field indicating the number of IPQF bands in which their signal gain have been controlled.                  |

The meanings of this value are shown below:

0: no bands have activated gain control.

1: signal gain on 2nd IPQF band has been controlled.

2: signal gain on 2nd and 3rd IPQF bands have been controlled.

3: signal gain on 2nd, 3rd and 4th IPQF bands have been controlled.

**alevcode** 4-bit field indicating the gain value for one gain change.

**alocode** 2-, 4-, or 5-bit field indicating the position for one gain change. The length of this data varies depending on the window sequence.

#### 4.6.12.3 Decoding process

The following four processes are required for decoding.

- (1) Gain control data decoding
- (2) Gain control function setting
- (3) Gain control windowing and overlapping
- (4) Synthesis filter

##### 4.6.12.3.1 Gain control data decoding

Gain control data are reconstructed as follows.

(1)

$$NAD_{W,B} = \mathbf{adjust\_num}[B][W]$$

(2)

$$ALOC_{W,B}(m) = \mathit{AdjLoc}(\mathbf{alocode}[B][W][m-1]), \quad 1 \leq m \leq NAD_{W,B}$$

$$ALEV_{W,B}(m) = 2^{\mathit{AdjLev}(\mathbf{alevcode}[B][W][m-1])}, \quad 1 \leq m \leq NAD_{W,B}$$

(3)

$$ALOC_{W,B}(0) = 0$$

$$ALEV_{W,B}(0) = \begin{cases} 1, & \text{if } NAD_{W,B} = 0 \\ ALEV_{W,B}(1), & \text{otherwise} \end{cases}$$

(4)

$$ALOC_{W,B}(NAD_{W,B} + 1) = \begin{cases} \left. \begin{array}{l} 256, \quad W = 0 \\ 112, \quad W = 0 \\ 32, \quad W = 1 \end{array} \right\} & \text{if } \mathbf{ONLY\_LONG\_SEQUENCE} \\ \left. \begin{array}{l} 32, \quad 0 \leq W \leq 7 \\ 112, \quad W = 0 \\ 256, \quad W = 1 \end{array} \right\} & \text{if } \mathbf{EIGHT\_SHORT\_SEQUENCE} \\ \left. \begin{array}{l} 112, \quad W = 0 \\ 256, \quad W = 1 \end{array} \right\} & \text{if } \mathbf{LONG\_STOP\_SEQUENCE} \end{cases}$$

$$ALEV_{W,B}(NAD_{W,B} + 1) = 1$$

where

|                   |                                                   |
|-------------------|---------------------------------------------------|
| $NAD_{W,B}$ :     | Gain Control Information Number, an integer       |
| $ALOC_{W,B}(m)$ : | Gain Control Location, an integer                 |
| $ALEV_{W,B}(m)$ : | Gain Control Level, an integer-valued real number |
| $B$ :             | Band ID, an integer from 1 to 3                   |
| $W$ :             | Window ID, an integer from 0 to 7                 |
| $m$ :             | an integer                                        |

**alocode**[ $B$ ][ $W$ ][ $m$ ] must be set so that  $\{ALOC_{W,B}(m)\}$  satisfies the following conditions.

$$ALOC_{W,B}(m_1) < ALOC_{W,B}(m_2), \quad 1 \leq m_1 < m_2 \leq NAD_{W,B} + 1$$

In cases of LONG\_START\_SEQUENCE and LONG\_STOP\_SEQUENCE, the values 14 and 15 of **alocode**[ $B$ ][ $0$ ][ $m$ ] are invalid. *AdjLoc*() is defined in Table 4.108. *AdjLev*() is defined in Table 4.109.

#### 4.6.12.3.2 Gain control function setting

The Gain control function is obtained as follows.

(1)

$$M_{W,B,j} = \text{Max}\{m: ALOC_{W,B}(m) \leq j\},$$

$$0 \leq j \leq 255, \quad W = 0 \quad \text{if ONLY\_LONG\_SEQUENCE}$$

$$\left. \begin{array}{l} 0 \leq j \leq 111, \quad W = 0 \\ 0 \leq j \leq 31, \quad W = 1 \end{array} \right\} \quad \text{if LONG\_START\_SEQUENCE}$$

$$0 \leq j \leq 31, \quad 0 \leq W \leq 7 \quad \text{if EIGHT\_SHORT\_SEQUENCE}$$

$$\left. \begin{array}{l} 0 \leq j \leq 111, \quad W = 0 \\ 0 \leq j \leq 255, \quad W = 1 \end{array} \right\} \quad \text{if LONG\_STOP\_SEQUENCE}$$

(2)

$$FMD_{W,B}(j) = \begin{cases} \text{Inter} \left( \begin{array}{l} ALEV_{W,B}(M_{W,B,j}), \\ ALEV_{W,B}(M_{W,B,j} + 1), \\ j - ALOC_{W,B}(M_{W,B,j}) \end{array} \right), & \text{if } ALOC_{W,B}(M_{W,B,j}) \leq j \leq ALOC_{W,B}(M_{W,B,j}) + 7 \\ ALEV_{W,B}(M_{W,B,j} + 1), & \text{otherwise} \end{cases}$$

(3)

if ONLY\_LONG\_SEQUENCE

$$GMF_{0,B}(j) = \begin{cases} ALEV_{0,B}(0) \times PFMD_B(j), & 0 \leq j \leq 255 \\ FMD_{0,B}(j - 256), & 256 \leq j \leq 511 \end{cases}$$

$$PFMD_B(j) = FMD_{0,B}(j), \quad 0 \leq j \leq 255$$

if LONG\_START\_SEQUENCE

$$GMF_{0,B}(j) = \begin{cases} ALEV_{0,B}(0) \times ALEV_{1,B}(0) \times PFMD_B(j), & 0 \leq j \leq 255 \\ ALEV_{1,B}(0) \times FMD_{0,B}(j-256), & 256 \leq j \leq 367 \\ FMD_{1,B}(j-368), & 368 \leq j \leq 399 \\ 1, & 400 \leq j \leq 511 \end{cases}$$

$$PFMD_B(j) = FMD_{1,B}(j), \quad 0 \leq j \leq 31$$

if EIGHT\_SHORT\_SEQUENCE

$$GMF_{W,B}(j) = \begin{cases} ALEV_{W,B}(0) \times PFMD_B(j), & W = 0, \quad 0 \leq j \leq 31 \\ ALEV_{W,B}(0) \times FMD_{W-1,B}(j), & 1 \leq W \leq 7, \quad 0 \leq j \leq 31 \\ FMD_{W,B}(j-32), & 0 \leq W \leq 7, \quad 32 \leq j \leq 63 \end{cases}$$

$$PFMD_B(j) = FMD_{7,B}(j), \quad 0 \leq j \leq 31$$

if LONG\_STOP\_SEQUENCE

$$GMF_{0,B}(j) = \begin{cases} 1, & 0 \leq j \leq 111 \\ ALEV_{0,B}(0) \times ALEV_{1,B}(0) \times PFMD_B(j-112), & 112 \leq j \leq 143 \\ ALEV_{1,B}(0) \times FMD_{0,B}(j-144), & 144 \leq j \leq 255 \\ FMD_{1,B}(j-256), & 256 \leq j \leq 511 \end{cases}$$

$$PFMD_B(j) = FMD_{1,B}(j), \quad 0 \leq j \leq 255$$

(4)

$$AD_{W,B}(j) = \frac{1}{GMF_{W,B}(j)},$$

$$0 \leq j \leq 511, \quad W = 0 \quad \text{if ONLY\_LONG\_SEQUENCE}$$

$$0 \leq j \leq 511, \quad W = 0 \quad \text{if LONG\_START\_SEQUENCE}$$

$$0 \leq j \leq 63, \quad 0 \leq W \leq 7 \quad \text{if EIGHT\_SHORT\_SEQUENCE}$$

$$0 \leq j \leq 511, \quad W = 0 \quad \text{if LONG\_STOP\_SEQUENCE}$$

where

$FMD_{W,B}(j)$ : Fragment Modification Function, a real number

$PFMD_B(j)$ : Fragment Modification Function of previous frame, a real number

$GMF_{W,B}(j)$ : Gain Modification Function, a real number

$AD_{W,B}(j)$ : Gain Control Function, a real number

$ALOC_{W,B}(m)$ : Gain Control Location defined in subclause 4.6.12.3.1, an integer

$ALEV_{W,B}(m)$ : Gain Control Level defined in subclause 4.6.12.3.1, an integer-valued real number

$B$ : Band ID, an integer from 1 to 3

$W$ : Window ID, an integer from 0 to 7

$M_{W,B,j}$ : an integer  
 $m$ : an integer

and

$$\text{Inter}(a,b,j) = 2^{\frac{(8-j)\log_2(a)+j\log_2(b)}{8}}$$

Note that the initial value of  $PFMD_B(j)$  must be set to 1.0.

#### 4.6.12.3.3 Gain control windowing and overlapping

Band Sample Data are obtained through the processes (1) to (2) shown below.

##### (1) Gain Control Windowing

if  $B=0$

$$T_{W,B}(j) = U_{W,B}(j),$$

$0 \leq j \leq 511, W = 0$  if ONLY\_LONG\_SEQUENCE  
 $0 \leq j \leq 511, W = 0$  if LONG\_START\_SEQUENCE  
 $0 \leq j \leq 63, 0 \leq W \leq 7$  if EIGHT\_SHORT\_SEQUENCE  
 $0 \leq j \leq 511, W = 0$  if LONG\_STOP\_SEQUENCE

else

$$T_{W,B}(j) = AD_{W,B}(j) \times U_{W,B}(j),$$

$0 \leq j \leq 511, W = 0$  if ONLY\_LONG\_SEQUENCE  
 $0 \leq j \leq 511, W = 0$  if LONG\_START\_SEQUENCE  
 $0 \leq j \leq 63, 0 \leq W \leq 7$  if EIGHT\_SHORT\_SEQUENCE  
 $0 \leq j \leq 511, W = 0$  if LONG\_STOP\_SEQUENCE

##### (2) Overlapping

if ONLY\_LONG\_SEQUENCE

$$V_B(j) = PT_B(j) + T_{0,B}(j), \quad 0 \leq j \leq 255$$

$$PT_B(j) = T_{0,B}(j + 256), \quad 0 \leq j \leq 255$$

if LONG\_START\_SEQUENCE

$$V_B(j) = PT_B(j) + T_{0,B}(j), \quad 0 \leq j \leq 255$$

$$V_B(j + 256) = T_{0,B}(j + 256), \quad 0 \leq j \leq 111$$

$$PT_B(j) = T_{0,B}(j + 368), \quad 0 \leq j \leq 31$$

if EIGHT\_SHORT\_SEQUENCE



$$V_B(j) = PT_B(j) + T_{W,B}(j), \quad W = 0, \quad 0 \leq j \leq 31$$

$$V_B(32W + j) = T_{W-1,B}(j + 32) + T_{W,B}(j), \quad 1 \leq W \leq 7, \quad 0 \leq j \leq 31$$

$$PT_B(j) = T_{w,B}(j + 32), \quad W = 7, \quad 0 \leq j \leq 31$$

if LONG\_STOP\_SEQUENCE

$$V_B(j) = PT_B(j) + T_{0,B}(j + 112), \quad 0 \leq j \leq 31$$

$$V_B(j + 32) = T_{0,B}(j + 144), \quad 0 \leq j \leq 111$$

$$PT_B(j) = T_{0,B}(j + 256), \quad 0 \leq j \leq 255$$

where

- $U_{W,B}(j)$ : Band Spectrum Data, a real number
- $T_{W,B}(j)$ : Gain Controlled Block Sample Data, a real number
- $PT_B(j)$ : Gain Controlled Block Sample Data of previous frame, a real number
- $V_B(j)$ : Band Sample Data, a real number
- $AD_{W,B}(j)$ : Gain Control Function defined in subclause 4.6.12.3.2, a real number
- $B$ : Band ID, an integer from 0 to 3
- $W$ : Window ID, an integer from 0 to 7
- $j$ : an integer

Note that the initial value of  $PT_B(j)$  must be set to 0.0.

#### 4.6.12.3.4 Synthesis filter

Audio Sample Data are obtained from the following equations.

(1)

$$\tilde{V}_B(j) = \begin{cases} V_B(k), & \text{if } j = 4k, \\ 0, & \text{else} \end{cases} \quad 0 \leq B \leq 3$$

(2)

$$Q_B(j) = Q(j) \times \cos\left(\frac{(2B+1)(2j-3)\pi}{16}\right), \quad 0 \leq j \leq 95, \quad 0 \leq B \leq 3$$

(3)

$$AS(n) = \sum_{B=0}^3 \sum_{j=0}^{95} Q_B(j) \times \tilde{V}_B(n-j)$$

where

$AS(n)$ : Audio Sample Data

- $V_B(n)$ : Band Sample Data defined in subclause 4.6.12.3.3, a real number
- $\tilde{V}_B(j)$ : Interpolated Band Sample Data, a real number
- $Q_B(j)$ : Synthesis Filter Coefficients, a real number
- $Q(j)$ : Prototype Coefficients given below, a real number
- $B$ : Band ID, an integer from 0 to 3
- $W$ : Window ID, an integer from 0 to 7
- $n$ : an integer
- $j$ : an integer
- $k$ : an integer

The values of  $Q(0)$  to  $Q(47)$  are shown in Table 4.110. The values of  $Q(48)$  to  $Q(95)$  are obtained from the following equation.

$$Q(j) = Q(95 - j), \quad 48 \leq j \leq 95$$

#### 4.6.12.4 Diagrams

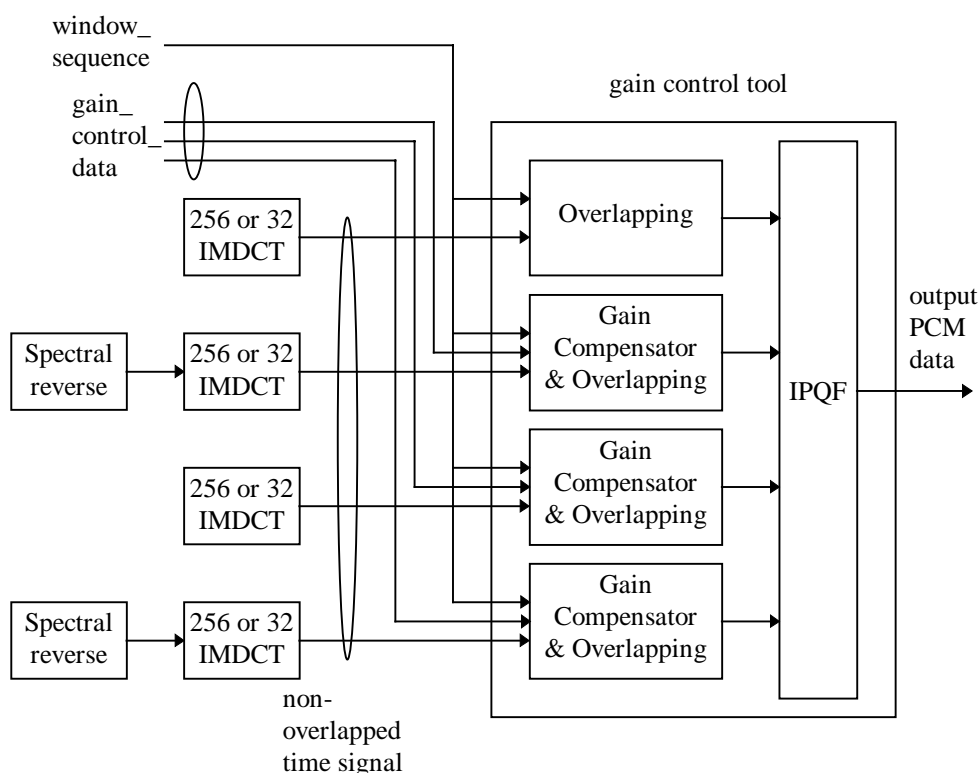


Figure 4.33 – Block diagram of gain control tool

#### 4.6.12.5 Tables

Table 4.108 – AdjLoc()

| AC | AdjLoc(AC) | AC | AdjLoc(AC) |
|----|------------|----|------------|
| 0  | 0          | 16 | 128        |
| 1  | 8          | 17 | 136        |
| 2  | 16         | 18 | 144        |

|    |     |    |     |
|----|-----|----|-----|
| 3  | 24  | 19 | 152 |
| 4  | 32  | 20 | 160 |
| 5  | 40  | 21 | 168 |
| 6  | 48  | 22 | 176 |
| 7  | 56  | 23 | 184 |
| 8  | 64  | 24 | 192 |
| 9  | 72  | 25 | 200 |
| 10 | 80  | 26 | 208 |
| 11 | 88  | 27 | 216 |
| 12 | 96  | 28 | 224 |
| 13 | 104 | 29 | 232 |
| 14 | 112 | 30 | 240 |
| 15 | 120 | 31 | 248 |

Table 4.109 – AdjLev()

| <i>AV</i> | <i>AdjLev(AV)</i> |
|-----------|-------------------|
| 0         | -4                |
| 1         | -3                |
| 2         | -2                |
| 3         | -1                |
| 4         | 0                 |
| 5         | 1                 |
| 6         | 2                 |
| 7         | 3                 |
| 8         | 4                 |
| 9         | 5                 |
| 10        | 6                 |
| 11        | 7                 |
| 12        | 8                 |
| 13        | 9                 |
| 14        | 10                |
| 15        | 11                |

Table 4.110 – Q()

| <i>j</i> | <i>Q(j)</i>             | <i>j</i> | <i>Q(j)</i>             |
|----------|-------------------------|----------|-------------------------|
| 0        | 9.7655291007575512E-05  | 24       | -2.2656858741499447E-02 |
| 1        | 1.3809589379038567E-04  | 25       | -6.8031113858963354E-03 |
| 2        | 9.8400749256623534E-05  | 26       | 1.5085400948280744E-02  |
| 3        | -8.6671544782335723E-05 | 27       | 3.9750993388272739E-02  |
| 4        | -4.6217998911921346E-04 | 28       | 6.2445363629436743E-02  |
| 5        | -1.0211814095158174E-03 | 29       | 7.7622327748721326E-02  |
| 6        | -1.6772149340010668E-03 | 30       | 7.9968338496132926E-02  |
| 7        | -2.2533338951411081E-03 | 31       | 6.5615493068475583E-02  |
| 8        | -2.4987888343213967E-03 | 32       | 3.3313658300882690E-02  |
| 9        | -2.1390815966761882E-03 | 33       | -1.4691563058190206E-02 |
| 10       | -9.5595397454597772E-04 | 34       | -7.2307890475334147E-02 |
| 11       | 1.1172111530118943E-03  | 35       | -1.2993222541703875E-01 |
| 12       | 3.9091309127348584E-03  | 36       | -1.7551641029040532E-01 |
| 13       | 6.9635703420118673E-03  | 37       | -1.9626543957670528E-01 |
| 14       | 9.5595442159478339E-03  | 38       | -1.8073330670215029E-01 |
| 15       | 1.0815766540021360E-02  | 39       | -1.2097653136035738E-01 |
| 16       | 9.8770514991715300E-03  | 40       | -1.4377370758549035E-02 |
| 17       | 6.1562567291327357E-03  | 41       | 1.3522730742860303E-01  |
| 18       | -4.1793946063629710E-04 | 42       | 3.1737852699301633E-01  |
| 19       | -9.2128743097707640E-03 | 43       | 5.1590021798482233E-01  |

|    |                         |    |                        |
|----|-------------------------|----|------------------------|
| 20 | -1.8830775873369020E-02 | 44 | 7.1080020379761377E-01 |
| 21 | -2.7226498457701823E-02 | 45 | 8.8090632488444798E-01 |
| 22 | -3.2022840857588906E-02 | 46 | 1.0068321641150089E+00 |
| 23 | -3.0996332527754609E-02 | 47 | 1.0737914947736096E+00 |

### 4.6.13 Perceptual Noise Substitution (PNS)

#### 4.6.13.1 Tool description

This tool is used to implement perceptual noise substitution coding within an ICS. Thus, certain sets of spectral coefficients are derived from random vectors rather than from Huffman coded symbols and an inverse quantization process. This is done selectively on a scalefactor band and group basis when perceptual noise substitution is flagged as active.

#### 4.6.13.2 Definitions

|                              |                                                                                                    |
|------------------------------|----------------------------------------------------------------------------------------------------|
| <b>hcod_sf[]</b>             | Huffman codeword from the Huffman code table used for coding of scalefactors (see subclause 4.6.2) |
| <i>dpcm_noise_nrg[][]</i>    | Differentially encoded noise energy                                                                |
| <i>noise_nrg[group][sfb]</i> | Noise energy for each group and scalefactor band                                                   |
| <i>spec[]</i>                | Array containing the channel spectrum of the respective channel                                    |

#### 4.6.13.3 Decoding process

The use of the perceptual noise substitution tool is signaled by the use of the pseudo codebook NOISE\_HCB (13).

Furthermore, if the same scalefactor band and group is coded by perceptual noise substitution in both channels of a channel pair, the correlation of the noise signal can be controlled by means of the *ms\_used* field: While the default noise generation process works independently for each channel (separate generation of random vectors), the same random vector is used for both channels if *ms\_used[]* is set for a particular scalefactor band and group. In this case, no M/S stereo coding is carried out (because M/S stereo coding and noise substitution coding are mutually exclusive). If the same scalefactor band and group is coded by perceptual noise substitution in only one channel of a channel pair the setting of *ms\_used[]* is not evaluated.

The energy information for perceptual noise substitution decoding is represented by a "noise energy" value indicating the overall power of the substituted spectral coefficients in steps of 1.5 dB. If noise substitution coding is active for a particular group and scalefactor band, a noise energy value is transmitted instead of the scalefactor of the respective channel.

Noise energies are coded just like scalefactors, i.e. by Huffman coding of differential values:

- the start value for the DPCM decoding is given by *global\_gain*.
- Differential decoding is done separately between scalefactors, intensity stereo positions and noise energies. In other words, the noise energy decoder ignores interposed scalefactors and intensity stereo position values and vice versa (see subclause 4.6.2.3.2)

The same codebook is used for coding of noise energies as for scalefactors.

One pseudo function is defined for use in perceptual noise substitution decoding:

```
function is_noise(group,sfb) {
    1    for window groups / scalefactor bands with
        codebook sfb_cb[group][sfb] == NOISE_HCB
    0    otherwise
}
```

The noise substitution decoding process for one channel is defined by the following pseudo code:

```

nrg = global_gain - NOISE_OFFSET - 256;
for (g=0; g<num_window_groups; g++) {

    /* Decode noise energies for this group */
    for (sfb=0; sfb<max_sfb; sfb++)
        if (is_noise(g,sfb))
            noise_nrg[g][sfb] = nrg += dpcm_noise_nrg[g][sfb];

    /* Do perceptual noise substitution decoding */
    for (b=0; b<window_group_length[g]; b++) {
        for (sfb=0; sfb<max_sfb; sfb++) {
            if (is_noise(g,sfb)) {

                offs = swb_offset[sfb];
                size = swb_offset[sfb+1] - offs;

                /* Generate random vector */
                gen_rand_vector( &spec[g][b][sfb][0], size );
                scale = 1/(size * sqrt(MEAN_NRG));
                scale *= 2.0^(0.25*noise_nrg [g][sfb]);
                /* Scale random vector to desired target energy */
                for (i=0; i<len; i++)
                    spec[g][b][sfb][i] *= scale;

            }
        }
    }
}

```

The constant NOISE\_OFFSET is used to adapt the range of average noise energy values to the usual range of scalefactors and has a value of 90.

The function gen\_rand\_vector( addr, size ) generates a vector of length <size> with signed random values of average energy MEAN\_NRG per random value. A suitable random number generator can be realized using one multiplication/accumulation per random value.

In case of reversible variable length coding (RVLC) the start value for the DPCM backwards decoding is given by reversible\_global\_gain. The decoding of the noise energies is defined by the following pseudo code:

```

nrg=rev_global_gain-NOISE_OFFSET-256+dpcm_noise_last_position;

for (g=win-1;g>=0;g--) {
    for (sfb=sfbmax-1;sfb>=0;sfb--) {
        noise_nrg[g][sfb]=nrg;
        nrg-=dpcm_noise_nrg[g][sfb];
    }
}

```

#### 4.6.13.4 Integration with the intra channel prediction tools

For scalefactor bands coded using PNS the corresponding predictors are switched to „off,, thus effectively overriding the status specified by the prediction\_used mask. In addition, for scalefactor bands coded by perceptual noise substitution the predictors belonging to the corresponding spectral coefficients are reset (see ISO/IEC13818-7 subclause 8.3.3). The update of these predictors is done by feeding a value of zero as the "last quantized value"  $x_{rec}(n-1)$ .

If both Long Term Prediction and PNS are active for a particular scalefactor band and group, PNS takes precedence, i.e. the spectral coefficients in this scalefactor band are produced by the PNS tool only.

#### 4.6.13.5 Integration with other AAC tools

The following interactions between the perceptual noise substitution tool and other AAC tools take place:

- Definition of a new pseudo Huffman codebook number NOISE\_HCB = 13
- During Huffman decoding of the quantized spectral coefficients, the Huffman codebook table NOISE\_HCB is treated exactly like the zero codebook ZERO\_HCB, i.e. no Huffman codewords are read for the corresponding scalefactor band and group.
- If the same scalefactor band and group is coded by perceptual noise substitution in both channels of a channel pair, no M/S stereo decoding is carried out for this scalefactor band and group .
- The pseudo noise components generated by the perceptual noise substitution tool are injected into the output spectrum prior to the temporal noise shaping (TNS) processing step.

#### 4.6.13.6 Integration into a scalable AAC-based coder (AudioObjectType AAC scalable)

The following rules apply for usage of the perceptual noise substitution tool in a scalable AAC-based coder:

- If a particular scalefactor band and group is coded by perceptual noise substitution, its contribution to the spectral components of the reconstructed output signal for the update of the intra channel predictor is omitted.
- If a particular scalefactor band is coded by perceptual noise substitution in layer N, it only contributes to the output spectrum of the combined layers N and N+1 if all spectral coefficients in this scalefactor band equal zero in the higher layer and no M/S or Intensity Stereo decoding takes place (see subclause 4.5.2.2.7). In case of a mono-stereo layer combination the condition of all spectral coefficients being equal to zero is expanded to both stereo channels and the FSS tool is used to control the addition.
- If a particular scalefactor band and group is coded by perceptual noise substitution in both channels of a channel pair, the higher (enhancement) layers may still use the M/S stereo flag ms\_used[][] to signal the use of M/S stereo decoding.

#### 4.6.14 Frequency Selective Switch (FSS) Module

The Frequency Selective Switching Unit (FSS) is used in various scalable coder configurations (see the diagrams in Figure 4.4 to Figure 4.15). It consists of a bank of switches which have the function to select one of two input signals, independently for each scale factor band (sfb). For each sfb a control bit is available which controls the selection. The form of the transmission of these control bits differs depending on the configuration in which the FSS module is used.

##### 4.6.14.1 FSS in combined TwinVQ /CELP- AAC systems:

###### 4.6.14.1.1 Definitions

|                          |                                                                                                                                                                                             |
|--------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| dc_group                 | Four consecutive scalefactor bands if the window type is not SHORT_WINDOW. One band of diff_short_lines, if the window type is SHORT_WINDOW.                                                |
| no_of_dc_groups          | If the window type is not SHORT_WINDOW, the number of groups depending on the sampling frequency is given in Table 4.112 below. If the window type is SHORT_WINDOW, no_of_dc_groups is '1'. |
| diff_short_lines         | Only used, if the window type is SHORT_WINDOW. The number of spectral lines in the single dc_group per window, depending on the sampling rate, is given in Table 4.112 below.               |
| diff_control_sfb[w][sfb] | Only applies, if the window type is not SHORT_WINDOW; These are the decoded <b>diff_control[w][dc_group]</b> values; One control bit for each switched scale factor band is available.      |

4.6.14.1.2 Decoding for the CELP – AAC combination

In the bitstream **diff\_control[w][dc\_group]** is used to transmit the huffman encoded values of **diff\_control\_sfb[w][sfb]** according to the following table:

**Table 4.111 – Huffman code table for diff\_control[w][dc\_group]**

|        |   |    |    |    |    |    |    |   |   |    |    |    |    |    |    |    |
|--------|---|----|----|----|----|----|----|---|---|----|----|----|----|----|----|----|
| Index  | 0 | 1  | 2  | 3  | 4  | 5  | 6  | 7 | 8 | 9  | 10 | 11 | 12 | 13 | 14 | 15 |
| Code   | 0 | 20 | 21 | 22 | 23 | 24 | 25 | 8 | 9 | 26 | 27 | 28 | 29 | 30 | 31 | 1  |
| Length | 2 | 5  | 5  | 5  | 5  | 5  | 5  | 4 | 4 | 5  | 5  | 5  | 5  | 5  | 5  | 2  |

The number of dc\_groups and the value of diff\_short lines, depending on the sampling rate, is given in Table 4.112:

**Table 4.112 – Number of dc\_groups and diff\_short\_lines for each sampling rate**

|                    |    |      |    |    |      |    |    |       |    |     |        |     |
|--------------------|----|------|----|----|------|----|----|-------|----|-----|--------|-----|
| Sampling Rate      | 96 | 88.2 | 64 | 48 | 44.1 | 32 | 24 | 22.05 | 16 | 12  | 11.025 | 8   |
| Core Sampling Rate | 8  | 7.35 | 8  | 8  | 7.35 | 8  | 8  | 7.35  | 8  | 12  | 11.025 | 8   |
| no_of_dc_groups    | 4  | 4    | 5  | 5  | 5    | 6  | 8  | 8     | 8  | 10  | 10     | 9   |
| Diff_short_lines   | 8  | 8    | 13 | 18 | 18   | 26 | 36 | 36    | 54 | 104 | 104    | 104 |

If sampling rates are used, which are in between the sampling rates specified in Table 4.112, the entries of the next higher sampling rate have to be used for this table (in contradiction to what is stated in subclause 4.5.1.1 about the parameter setting for non-standard sampling rates).

4.6.14.1.3 FSS for the TwinVQ – AAC combination:

The same huffman encoding (Table 4.111) of the values of **diff\_control\_sfb[w][sfb]** which is defined for the CELP – AAC combination, is also used for combined TwinVQ – AAC systems. However, the value of **no\_of\_dc\_groups** is calculated from the value of **max\_sfb** of the last TwinVQ layer as follows:

$$\text{no\_of\_dc\_groups} = \text{int}((\text{max\_sfb} + 3) / 4)$$

The value for **diff\_short\_lines** is taken from the scalefactor band table for **SHORT\_WINDOW** of the corresponding sampling rate:

$$\text{diff\_short\_lines} = \text{swb\_offset}[\text{max\_sfb}]$$

4.6.14.1.4 Common decoding process

Decoding if the window type is not SHORT\_WINDOW:

After huffman decoding **diff\_control[w][dc\_group]** from the bitstream, the array **diff\_control\_sfb[w][sfb]** is generated according to:

```

if ( ! SHORT_WINDOW ) {
    dc_group = 0;
    while( dc_group < no_of_dc_groups ) {
        for( i=0; i<4; i++ ) {
            diff_control_sfb[0][dc_group*4+i] = diff_control[0][dc_group] & 0x8;
            diff_control[0][dc_group] <<= 1;
        }
        dc_group++;
    }
}

```

For all scale factor bands which did not get a value assigned in **diff\_control\_sfb[w][sfb]** in the above procedure, **diff\_control\_sfb[w][sfb]** is set to '1';

Finally the switching for all scale factor bands is done according to:

```

if ( diff_control_sfb[w][sfb] == 0 ) {
    spectrum_out[w][sfb] = spectrum_AAC[w][sfb] + spectrum_Celp/TwinVQ[w][sfb];
} else {
    spectrum_out[w][sfb] = spectrum_AAC[w][sfb];
}

```

#### Decoding if the window type is SHORT\_WINDOW:

If the window type is SHORT\_WINDOW, there is only one band of diff\_short\_lines per window where the diff control mechanism is applied:

For the spectral lines from 0 to diff\_short\_lines-1:

```

if ( diff_control_sfb[w][0] == 0 ) {
    spectrum_out[w] = spectrum_AAC [w] + spectrum_Celp/TwinVQ[w];
} else {
    spectrum_out[w] = spectrum_AAC[w];
}

```

For the remaining lines the output of the switch is identical to the input:

```
spectrum_out[w] = spectrum_AAC [w];
```

### 4.6.14.2 FSS in combined mono / stereo scalable configurations

#### 4.6.14.2.1 Decoding process

In a combined mono-stereo coder, where a mono (M) signal which is derived from a stereo input is coded with one or more mono layers, and later coded with one or more stereo layers, the FSS tool is also used to control the addition of the output signal of the combined M-coding stages to the Left (L) or Right (R) channel signals of the stereo coding stages. In this case the number of processed bands in the current layer is equal to **max\_sfb** of the current layer. However, if last\_max\_sfb is larger than **max\_sfb** of the current layer, the unused **diff\_control\_lr[w][sfb]** bits are preserved for a subsequent layer. The control bits for these FSS modules are directly available from the **diff\_control\_lr[w][sfb]** syntax elements. Since the combined L+M, or R+M signals in one scale factor band (sfb) are only needed in subsequent coding stages of the current layer, if MS-coding is not selected for this specific sfb, **diff\_control\_lr[w][sfb]** is only transmitted for a sfb for which MS-coding is not selected.

#### Decoding if the window type is not SHORT\_WINDOW:

For all scale factor bands for which no value is transmitted **diff\_control\_lr[w][sfb]** is set to '1';

The switching for all scale factor bands is done according to:

```

if ( diff_control_lr[w][sfb] == 0 ) {
    spectrum_L/R_out[w][sfb] = spectrum_L/R[w][sfb] + 2 * spectrum_M[w][sfb];
} else {
    spectrum_L/R_out[w][sfb] = spectrum_L/R[w][sfb];
}

```

#### Decoding if the window type is SHORT\_WINDOW:

If the window type is SHORT\_WINDOW, the value **diff\_control\_lr[win][0]** is used for all scale factor bands from 0 to last\_max\_sfb-1. For all other bands a value of '1' is used.



## 4.6.15 Upsampling filter tool

### 4.6.15.1 Tool description

The upsampling filter tool is used to adapt the sampling rate of the (CELP) core coder to the sampling rate of the time/frequency coder. The upsampling filter uses the MDCT filterbank of the AAC encoder. This filterbank is very similar to the IMDCT filterbank, which is used in the decoder. Both use the same window functions.

The filterbank takes a block of time samples of the core coder output and inserts an appropriate number of zeroes between these samples to generate a signal at the desired higher sampling rate. These up-sampled values are then delayed by the number of samples given in „coreCoderDelay“ (see „GASpecificConfig()“) of the first enhancement layer, and then modulated with the same window function which is used for the IMDCT of this block. The window type and the window shape from the IMDCT are used (To save RAM in the decoder it is also possible to delay the core bitstream by an appropriate number of core frames instead of delaying the upsampled core signal). Each block of input samples is overlapped by 50% with the immediately preceding block. The transform input block length N is set to either 2048 (1920) or 256 (240) samples depending on the value of **frameLengthFlag**.

The output of the MDCT filterbank is connected to the FSS module, which only uses the output values in the FSS-bands. Since the upper FSS band doesn't exceed half of the lower sampling rate, aliasing effects are omitted.

### 4.6.15.2 Definitions

|                         |                                                                                        |
|-------------------------|----------------------------------------------------------------------------------------|
| up-sampling-factor      | ratio of T/F coder sampling rate and core coder sampling rate.                         |
| $X_{in\_mdct\_core}[i]$ | temporal data field, which is used to hold the up-sampled input to the MDCT filterbank |
| $X_{out\_core}[i]$      | Output samples of the core decoder                                                     |

### 4.6.15.3 Decoding process

The analysis window length N for the transform is a function of the syntax element **window\_sequence** and the algorithmic context. It is derived in an identical way to the procedure described for the Filterbank and Blockswitching tool.

#### 4.6.15.3.1 Upsampling by insertion of zeroes

The input to the filterbank is generated by :

$$\begin{aligned}
 X_{in\_mdct\_core}[k] &= 0 && \text{for } k = [0 : N/2-1] \\
 X_{in\_mdct\_core}[up\_sampling\_factor*i] &= X_{out\_core}[i] * up\_sampling\_factor && \text{for } i = [0 : N/2/up\_sampling\_factor-1]
 \end{aligned}$$

#### 4.6.15.3.2 Windowing and block switching

The adaptation of the time-frequency resolution of the filterbank is done by shifting between transforms whose input lengths are either 2048 (1920) or 256 (240) samples, synchronously to the decoder IMDCT filterbank. The selection between the 2048/256 or the 1920/240 pairs is done depending on the value of **frameLengthFlag**.

The windowed time domain values can be calculated by using exactly the same windows  $w(n)$  as defined for the IMDCT filterbank (see subclause 4.6.11).

The windowed coefficients are calculated by

$$z_{i,n} = w(n) \cdot x'_{in\_mdct\_core}(n);$$

#### 4.6.15.3.3 MDCT

The windowed coefficients are transformed into the frequency domain with an MDCT.

The MDCT spectral coefficient,  $X_{i,k}$ , are defined as follows:

$$X_{i,k} = 2 \cdot \sum_{n=0}^{N-1} z_{i,n} \cos\left(\frac{2\pi}{N}(n+n_0)\left(k+\frac{1}{2}\right)\right) \text{ for } 0 \leq k < N/2.$$

where:

$z_{i,n}$  = windowed input sequence

$n$  = sample index

$k$  = spectral coefficient index

$i$  = block index

$N$  = window length of the one transform window based on the window\_sequence value

$n_0 = (N/2 + 1)/2$

Only the output values from 0 to  $N/2/\text{up-sampling-factor}-1$  can be used without aliasing distortions. This is ensured by a subsequently following FSS module.

## 4.6.16 Tools for AAC Error resilience

### 4.6.16.1 Virtual codebooks for AAC section data

#### 4.6.16.1.1 Tool description

Virtual codebooks are used to limit the largest absolute value permitted within a certain scale factor band where escape values are allowed, i. e. where codebook 11 is used originally. This tool allows 17 different codebook indices (11, 16...31) for the escape codebook. All these codebook indices refer to codebook 11. They are therefore called virtual codebooks. The difference between these codebook indices is the allowed maximum of spectral values belonging to the appropriate section. Due to this, errors within spectral data resulting in too large spectral values can be located and the according spectral lines can be concealed.

#### 4.6.16.1.2 Decoding process

See subclause 4.5.2.3.2.

### 4.6.16.2 RVLC for AAC scalefactors

#### 4.6.16.2.1 Tool description

RVLC (reversible variable length coding) is used instead of Huffman coding to achieve entropy coding of the scalefactors, because of its better performance in terms of error resilience. It can be considered to be a plug-in of the noiseless coding tool defined in 4.6.3 which allows decoding error resilient encoded scalefactor data.

RVLC enables additional backward decoding. Some error detection is possible in addition because not all nodes of the coding tree are used as codewords. The error resilience performance of the RVLC is as better as smaller the number of codewords. Therefore the RVLC table contains only values from -7 to +7, whereas the original Huffman codebook contains values from -60 to +60. A decoded value of  $\pm 7$  is used as ESC\_FLAG. It signals that an escape value exists, that has to be added to +7 or subtracted from -7 in order to find the actual scalefactor value. This escape value is Huffman encoded.

It is necessary to transmit an additional value in order to have a starting point for backward decoding for the DPCM encoded scalefactors. This value is called reversible global gain. If intensity stereo coding or PNS is used, additional values are also necessary for them. The length of the RVLC bitstream part has to be transmitted to allow backward decoding. Furthermore the length of the bitstream part containing the escape codewords should be transmitted to keep synchronization in case of bitstream errors.

4.6.16.2.2 Definitions

The following bitstream elements are available within the bitstream, if the GASpecificConfig() enables the RVLC tool.

- sf\_concealment:** is a data field that indicates the similarity between scale factors of the last frame and those of the current one. It shall be set to '0', if the scale factors of the last frame are dissimilar to those of the current frame. It shall set to '1', if they are similar.  
Note: This data field is not required to decode error-free payload, but might be used to apply appropriate concealment strategies in case of corrupted scale factor data. No similarity criterion is specified since concealment is out of the scope of this standard.
- rev\_global\_gain** contains the last scalefactor value as a start value for the backward decoding. The length of this data field is 8 bits.
- length\_of\_rvlc\_sf** is a data field that contains the length of the current RVLC data part in bits, including the DPCM start value for PNS. The length of this data field depends on window\_sequence: If window\_sequence == EIGHT\_SHORT\_SEQUENCE, the field consists of 11 bits, otherwise it consists of 9 bits.
- rvlc\_cod\_sf** RVLC word from the RVLC table used for coding of scalefactors, intensity positions or noise energy.
- sf\_escapes\_present** is a data field that signals whether there are escapes coded in the bitstream or not. The length of this data is 1 bit.
- length\_of\_rvlc\_escapes** is a data field that contains the length of the current RVLC escape data part in bits. The length of this data is 8 bits.
- rvlc\_esc\_sf** Huffman codeword from the Huffman table for RVLC-ESC-values used for coding values larger than ±6.
- dpcm\_is\_last\_position** DPCM value allowing backward decoding of intensity stereo data part. It is the symmetric value to dpcm\_is\_position.
- dpcm\_noise\_last\_position** DPCM value allowing backward decoding of PNS data part. The length of this data is 9 bit. It is the symmetric value to dpcm\_noise\_nrg.

4.6.16.2.3 Decoding process

See subclause 4.6.2.3.2.

4.6.16.2.4 Tables

Table 4.113 – RVLC codebook

| index | length | codeword |
|-------|--------|----------|
| -7    | 7      | 65       |
| -6    | 9      | 257      |
| -5    | 8      | 129      |
| -4    | 6      | 33       |
| -3    | 5      | 17       |
| -2    | 4      | 9        |
| -1    | 3      | 5        |
| 0     | 1      | 0        |
| 1     | 3      | 7        |
| 2     | 5      | 27       |
| 3     | 6      | 51       |
| 4     | 7      | 107      |
| 5     | 8      | 195      |
| 6     | 9      | 427      |
| 7     | 7      | 99       |

Table 4.114 – Asymmetric (forbidden) codewords

| length | codeword |
|--------|----------|
| 6      | 50       |
| 7      | 96       |
| 9      | 256      |
| 8      | 194      |
| 7      | 98       |
| 6      | 52       |
| 9      | 426      |
| 8      | 212      |

Table 4.115 – Huffman codebook for RVLC escape values

| index | length | codeword | index | length | codeword |
|-------|--------|----------|-------|--------|----------|
| 0     | 2      | 2        | 27    | 20     | 473482   |
| 1     | 2      | 0        | 28    | 20     | 473483   |
| 2     | 3      | 6        | 29    | 20     | 473484   |
| 3     | 3      | 2        | 30    | 20     | 473485   |
| 4     | 4      | 14       | 31    | 20     | 473486   |
| 5     | 5      | 31       | 32    | 20     | 473487   |
| 6     | 5      | 15       | 33    | 20     | 473488   |
| 7     | 5      | 13       | 34    | 20     | 473489   |
| 8     | 6      | 61       | 35    | 20     | 473490   |
| 9     | 6      | 29       | 36    | 20     | 473491   |
| 10    | 6      | 25       | 37    | 20     | 473492   |
| 11    | 6      | 24       | 38    | 20     | 473493   |
| 12    | 7      | 120      | 39    | 20     | 473494   |
| 13    | 7      | 56       | 40    | 20     | 473495   |
| 14    | 8      | 242      | 41    | 20     | 473496   |
| 15    | 8      | 114      | 42    | 20     | 473497   |
| 16    | 9      | 486      | 43    | 20     | 473498   |
| 17    | 9      | 230      | 44    | 20     | 473499   |
| 18    | 10     | 974      | 45    | 20     | 473500   |
| 19    | 10     | 463      | 46    | 20     | 473501   |
| 20    | 11     | 1950     | 47    | 20     | 473502   |
| 21    | 11     | 1951     | 48    | 20     | 473503   |
| 22    | 11     | 925      | 49    | 19     | 236736   |
| 23    | 12     | 1848     | 50    | 19     | 236737   |
| 24    | 14     | 7399     | 51    | 19     | 236738   |
| 25    | 13     | 3698     | 52    | 19     | 236739   |
| 26    | 15     | 14797    | 53    | 19     | 236740   |

#### 4.6.16.3 Huffman Codeword Reordering for AAC spectral data

##### 4.6.16.3.1 Tool description

The Huffman codeword reordering (HCR) algorithm for AAC spectral data is based on the fact that some of the codewords can be placed at known positions so that these codewords can be decoded independent of any error within other codewords. Therefore, this algorithm avoids error propagation to those codewords, the so-called priority codewords (PCW). To achieve this, segments of known length are defined and those codewords are placed at the beginning of these segments.

The remaining codewords (non-priority codewords, non-PCW) are filled into the gaps left by the PCWs using a special algorithm that minimizes error propagation to the non-PCWs codewords.

This reordering algorithm does not increase the size of spectral data.

Before applying the reordering algorithm itself, a pre-sorting process is applied to the codewords. It sorts all codewords depending on their importance, i. e. it determines the PCWs.

### 4.6.16.3.2 Definitions

The following bitstream elements are available within the bitstream, if the GASpecificConfig() enables the HCR tool.

**length\_of\_reordered\_spectral\_data** is a 14-bit data field that contains the length of spectral data in bits. The maximum value is 6144 in case of a single\_channel\_element, a coupling\_channel\_element and a lfe\_channel\_element and 12288 in case of a channel\_pair\_element. Larger values are reserved for future use. If those values occur, current decoders have to replace them by the valid maximum value.

**length\_of\_longest\_codeword** is a 6-bit data field that contains the length of the longest codeword available within the current spectral data in bits. This field is used to decrease the distance between protected codewords. Valid values are between 0 and 49. Values between 50 and 63 are reserved for future use. If those values occur, current decoders have to replace them by 49.

### 4.6.16.3.3 Bitstream structure

#### 4.6.16.3.3.1 Pre-sorting

Subclause 4.5.2.3.5 is not valid if this tool is used. Instead, the procedure described in the following paragraphs has to be applied:

For explanation of the pre-sorting steps the term “unit” is introduced. A unit covers four spectral lines, i. e. two two-dimensional codewords or one four-dimensional codeword. In case of two two-dimensional codewords their natural order is kept, i. e. the higher frequency codeword follows the lower frequency codeword.

In case of one long window (1024 spectral lines per long block, one long block per frame), each window contains 256 units.

In case of eight short windows (128 spectral lines per short block, eight short blocks per frame), each window contains 32 units.

First pre-sorting step:

Units representing the same part of the spectrum are collected together in temporal order and denoted as unit group. In case of one long window, each unit group contains one unit. In case of eight short windows, each unit group contains eight units.

Unit groups are collected ascending in spectral direction. For one long window, that gives the original codeword order, but for eight short windows a unit based window interleaving has been applied.

Using this scheme, the codewords representing the lowest frequencies are the first codewords within spectral data for both, long and short blocks.

Table 4.116 shows an example output of the first pre-sorting step for short blocks, assuming two-dimensional codebooks for window 0, 1, 6, and 7 and four-dimensional codebooks for window 2, 3, 4, and 5.

Second pre-sorting step:

The more energy a spectral line contains, the more audible is its distortion. The energy within spectral lines is related to the used codebook. Codebooks with low numbers can represent only low values and allow only small errors, while codebooks with high numbers can represent high values and allow large errors.

Therefore, the codewords are pre-sorted depending on the used codebook. If the error resilient section data is used, the order is 11, 31, 30, 29, 28, 27, 26, 25, 24, 23, 22, 21, 20, 19, 18, 17, 16, 9/10, 7/8, 5/6, 3/4, 1/2. If the normal section data is used, the order is 11, 9/10, 7/8, 5/6, 3/4, 1/2. This order is based on the largest absolute value of the tables. This second pre-sorting step is done on the described unit by unit base used in the first pre-sorting step. The output of the first pre-sorting step is scanned in a consecutive way for each codebook.

Assigning numbers to units according the following metric can perform these two pre-sorting steps:

$$\text{codbookPriority}[32] = \{x, 21, 21, 20, 20, 19, 19, 18, 18, 17, 17, 0, x, x, x, x, 16, 15, 14, 13, 12, 11, 10, 9, 8, 7, 6, 5, 4, 3, 2, 1\}$$

$$\text{assignedUnitNr} = (\text{codbookPriority}[\text{cb}] * \text{maxNrOfLinesInWindow} + \text{nrOfFirstLineInUnit}) * \text{MaxNrOfWindows} + \text{window}$$

with:

|                      |                                                                                                                                                        |
|----------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------|
| codebookPriority[cb] | codebook priority according the second pre-sorting step.                                                                                               |
| maxNrOfLinesInWindow | constant number: f in case of one long window and 128 in case of eight short windows                                                                   |
| nrOfFirstLineInUnit  | a number between 0 and 1020 in case of one long window and between 0 and 124 in case of eight short windows (this number is always a multiple of four) |
| maxNrOfWindows       | constant number: 1 in case of one long window and 8 in case of eight short windows                                                                     |
| window               | always 0 in case of one long window, a number between 0 and 7 in case of eight short windows                                                           |

and sort the units in ascending order using these assigned unit numbers.

Encoder note: In order to reduce audible artifacts in case of errors within spectral data it is strongly recommended to use codebook 11 only if necessary!

#### 4.6.16.3.3.2 Segment width and segment instantiation

The segment widths depend on the Huffman codebook used. They are derived as the minimums of the (codebook dependent) maximum codeword length and the transmitted longest codeword length:

$$\text{segmentWidth} = \min(\text{maxCwLen}, \text{length\_of\_longest\_codeword})$$

Table 4.117 shows the values of maxCwLen depending on the Huffman codebook.

Segments are instantiated until the available buffer is exhausted, whereas the size of this buffer is given by the data element length\_of\_reordered\_spectral\_data. The remaining bits at the end of the buffer increase the size of the last segment.

#### 4.6.16.3.3.3 Order of Huffman codewords in spectral data

Figure 4.34 shows the general scheme of the segmentation and the arrangement of the PCWs. In this Figure 4.34, five segments can be provided to protect codewords from section 0 and section 1 against error propagation. Segment widths are different, because the length of the longest possible codeword depends on the current codebook.

The writing scheme for the non-PCWs is as follows (PCWs have been written already):

The proposed scheme introduces the term set. A set contains a certain number of codewords. Given N is the number of segments; all sets except the last one contain N codewords. Non-PCWs are written consecutively into these sets. Due to the pre-sorting algorithm set one contains the most important non-PCWs. The importance of the codewords stored within a set is the smaller the higher the set number.

Sets are written consecutively. Writing of a set might need several trials.

First trial: The first codeword of the current set is written into the remaining part of the first segment, the second codeword into the remaining part of the second segment and so on. The last codeword of the current set is written into the remaining part of the last segment.

Second trial: The remaining part of the first codeword (if any) is written into the remaining part of the second segment, the remaining part of the second codeword (if any) into the remaining part of the third segment and so on. The remaining part of the last codeword (if any) is written into the remaining part of the first segment (modulo shift).

If a codeword does not fit into the remaining part of a segment, it is only partly written and its remaining part is stored. At least after a maximum of N trials all codewords are completely written into segments.

If one set was written completely, writing of the next set starts. To improve the error propagation behavior between consecutive sets, the writing direction within segments changes from set to set. While PCWs are written from left to right, codewords of set one are written from right to left, codewords of set two are again written from left to right and so on.

#### **4.6.16.3.3.4 Encoding process**

The structure of the reordered spectral data cannot be described within the C like syntax commonly used. Therefore, Figure 4.35 shows an example and the following c-like description is provided:

```

/* helper functions */
void InitReordering(void);
/* Initializes variables used by the reordering functions like the segment
widths and the used offsets in segments and codewords. */

void InitRemainingBitsInCodeword(void);
/* Initializes remainingBitsInCodeword[] array for each codeword with
the total size of the codeword. */

int WriteCodewordToSegment(codewordNr, segmentNr, direction);
/* Writes a codeword or only a part of a codeword indexed by codewordNr
to the segment indexed by segmentNr with a given direction.
Write offsets for each segment are handled internally.
The function returns the number of bits written to the segment.
This number may be lower than the codeword length.
WriteCodewordToSegment handles already written parts of the codeword
internally. */

void ToggleWriteDirection(void);
/* Toggles the write direction in the segments between forward and backward. */

/* (input) variables */
numberOfCodewords; /* 15 in the example */
numberOfSegments; /* 6 in the example */
numberOfSets; /* 3 in the example */

ReorderSpectralData()
{
  InitReordering();
  InitRemainingBitsInCodeword();

  /* first step: write PCWs (set 0) */
  writeDirection = forward;
  for (codeword = 0; codeword < numberOfSegments; codeword++) {
    WriteCodewordToSegment(codeword, codeword, writeDirection);
  }

  /* second step: write nonPCWs */
  for (set = 1; set < numberOfSets; set++) {
    ToggleWriteDirection();
    for (trial = 0; trial < numberOfSegments; trial++) {
      for (codewordBase = 0; codewordBase < numberOfSegments; codewordBase++) {
        segment = (trial + codewordBase) % numberOfSegments;
      }
    }
  }
}

```

```
codeword = codewordBase + set*numberOfSegments;
```

```
if (remainingBitsInCodeword[codeword] > 0) {
    remainingBitsInCodeword[codeword] -= WriteCodewordToSegment(codeword,
        segment,
        writeDirection);
}
}
}
}
}
```

#### 4.6.16.3.4 Decoding process

See subclauses 4.5.2.3.2 and 4.6.3.3.

#### 4.6.16.3.5 Tables

**Table 4.116 – Example output of the first pre-sorting step for short blocks, assuming two-dimensional codebooks for window 0, 1, 6, and 7 and four-dimensional codebooks for window 2, 3, 4, and 5**

| index | codeword entry |              |
|-------|----------------|--------------|
|       | window         | window index |
| 0     | 0              | 0            |
| 1     | 0              | 1            |
| 2     | 1              | 0            |
| 3     | 1              | 1            |
| 4     | 2              | 0            |
| 5     | 3              | 0            |
| 6     | 4              | 0            |
| 7     | 5              | 0            |
| 8     | 6              | 0            |
| 9     | 6              | 1            |
| 10    | 7              | 0            |
| 11    | 7              | 1            |
| 12    | 0              | 2            |
| 13    | 0              | 3            |
| 14    | 1              | 2            |
| 15    | 1              | 3            |
| 16    | 2              | 1            |
| 17    | 3              | 1            |
| 18    | 4              | 1            |
| 19    | 5              | 1            |
| 20    | 6              | 2            |
| 21    | 6              | 3            |
| 22    | 7              | 2            |
| 23    | 7              | 3            |
| ...   | ...            | ...          |

**Table 4.117 – Values of maxCwLen depending on the Huffman codebook**

| codebook | maximum codeword length (maxCwLen) |
|----------|------------------------------------|
| 0        | 0                                  |



|    |    |
|----|----|
| 1  | 11 |
| 2  | 9  |
| 3  | 20 |
| 4  | 16 |
| 5  | 13 |
| 6  | 11 |
| 7  | 14 |
| 8  | 12 |
| 9  | 17 |
| 10 | 14 |
| 11 | 49 |
| 16 | 14 |
| 17 | 17 |
| 18 | 21 |
| 19 | 21 |
| 20 | 25 |
| 21 | 25 |
| 22 | 29 |
| 23 | 29 |
| 24 | 29 |
| 25 | 29 |
| 26 | 33 |
| 27 | 33 |
| 28 | 33 |
| 29 | 37 |
| 30 | 37 |
| 31 | 41 |

4.6.16.3.6 Figures

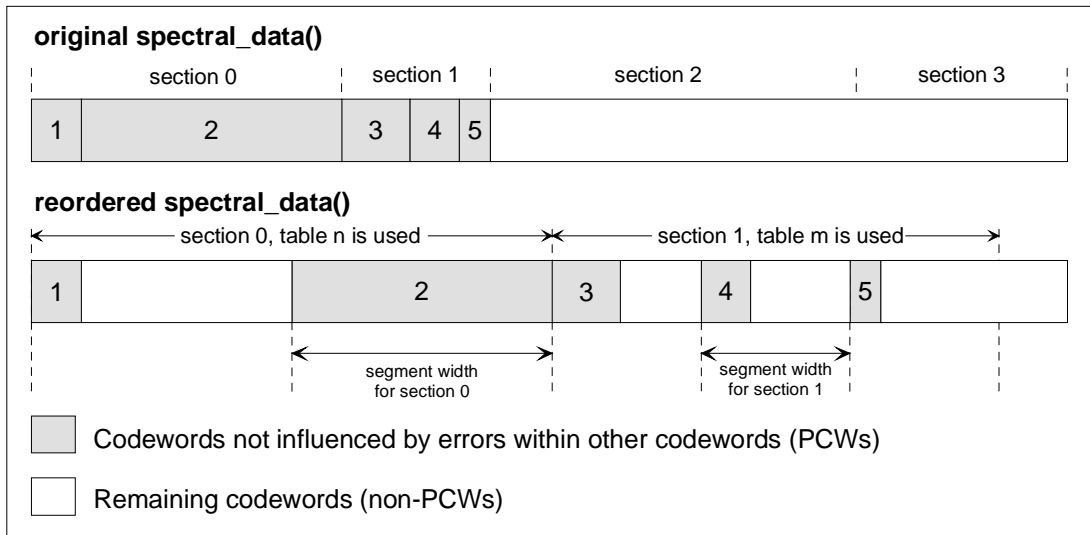


Figure 4.34 – General scheme of segmentation and arrangement of PCWs

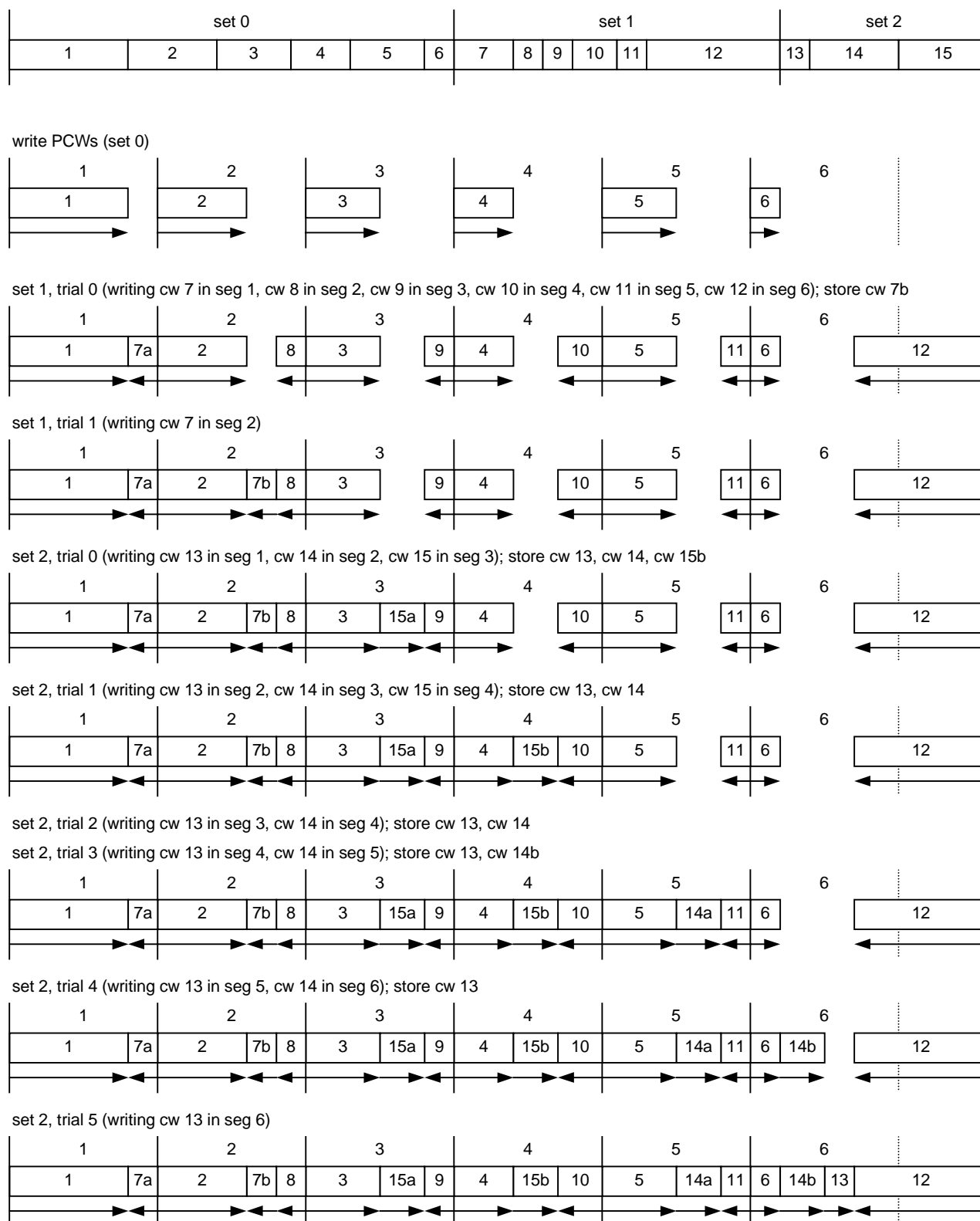


Figure 4.35 – Example for HCR encoding algorithm (only one segment width, pre-sorting has been done before)

### 4.6.17 Low delay codec

#### 4.6.17.1 Introduction

The low delay coding functionality provides the ability to extend the usage of generic low bitrate audio coding to applications requiring a very low delay of the encoding / decoding chain (e.g. full-duplex real-time communications).

This subclause specifies a low delay audio coder providing a mode with an algorithmic delay not exceeding 20 ms.

The overall algorithmic delay of a general audio coder is determined by the following factors:

- **Frame length**  
For block-based processing, a certain amount of time has to pass to collect the samples belonging to one block
- **Filterbank delay**  
Use of an analysis-synthesis filterbank pair causes a certain amount of delay.
- **Look-ahead for block switching decision**  
Due to the underlying principles of the block switching scheme, the detection of transients has to use a certain amount of “look-ahead” in order to ensure that all transient signal parts are covered properly by short windows.
- **Use of bit reservoir**  
While the bit reservoir facilitates the use of a locally varying bitrate, this implies an additional delay depending on the size of the bit reservoir relative to the average bitrate per block.

The overall algorithmic delay can be calculated as

$$t_{delay} = \frac{N_{Frame} + N_{FB} + N_{look\_ahead} + N_{bitres}}{F_s}$$

where  $F_s$  is the coder sampling rate,  $N_{Frame}$  is the frame size,  $N_{FB}$  is the delay due to the filterbank (s),  $N_{look\_ahead}$  corresponds to the look-ahead delay for block switching and  $N_{bitres}$  is the delay due to the use of the bit reservoir.

The low delay codec is derived from the AAC LTP audio object type, i.e. a coder consisting of the low complexity AAC codec plus the PNS (Perceptual Noise Substitution) and the LTP (Long Term Predictor) tools. Figure 4.36 shows the overall structure of the decoder.

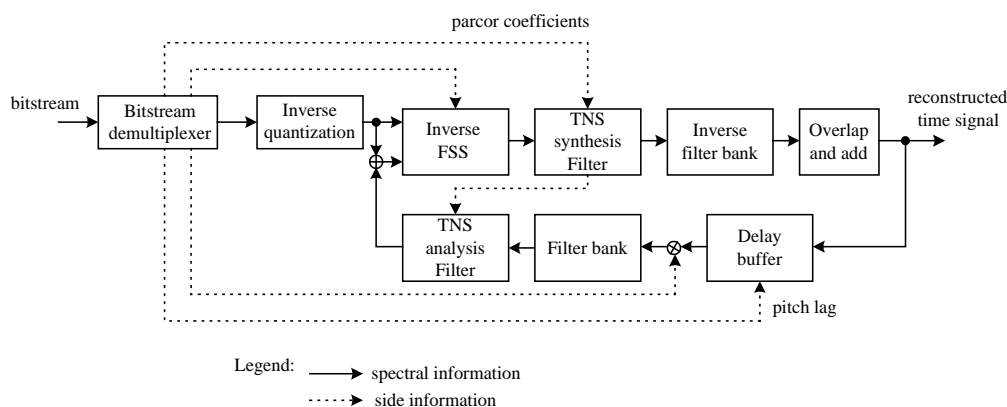


Figure 4.36 – Decoder structure

Figure 4.37 shows the overall structure of the encoder.

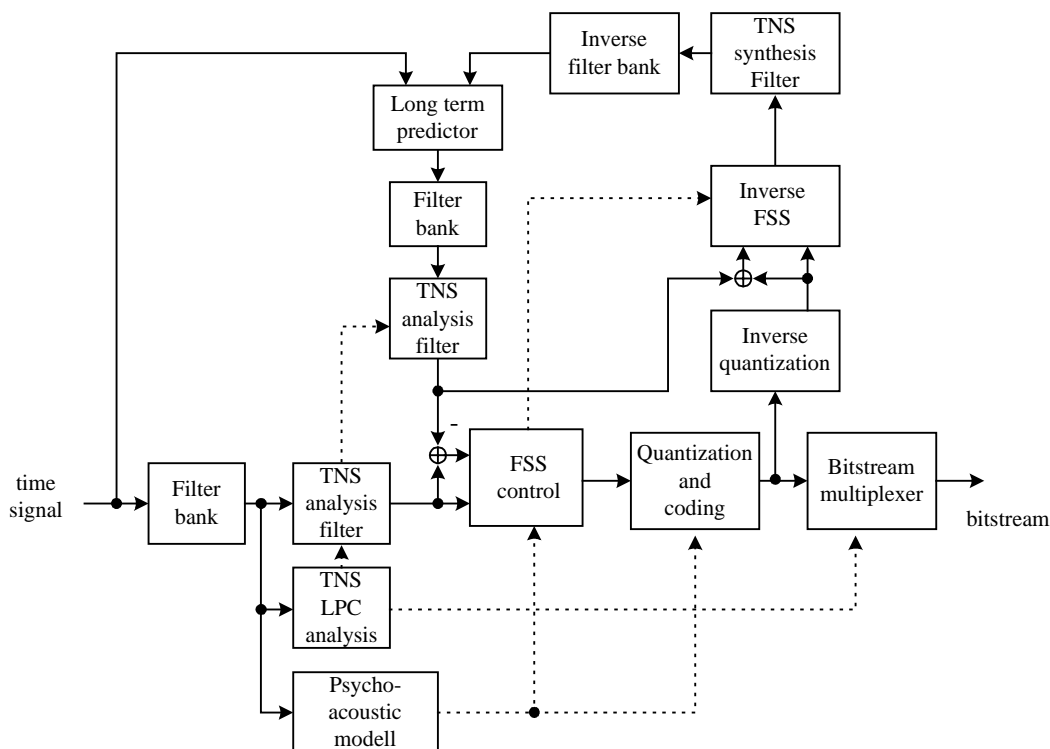


Figure 4.37 – Encoder structure

#### 4.6.17.2 Coder description

The low delay codec is defined by the following modifications with respect to the standard algorithm (i.e. AAC LTP audio object type) to achieve low delay operation:

##### 4.6.17.2.1 Frame size/window length

The length of the analysis window is reduced to 1024 or 960 time domain samples corresponding to 512 and 480 spectral values, respectively. The latter choice enables the coder to have a frame size that is commensurate with widely used speech codecs (20 ms). The corresponding scalefactor band tables are given in subclause 4.5.4.

##### 4.6.17.2.2 Block switching

Due to the contribution of the look-ahead time to the overall delay, no block switching is used.

##### 4.6.17.2.3 Window shape

As stated in the previous chapter, block switching is not used in the low delay coder to keep the delay as low as possible. As an alternative tool to improve coding of transient signals, the low delay coder uses the window shape switching feature with a slight modification compared to normal-delay AAC: The low delay coder still uses the sine window shape, but the Kaiser-Bessel derived window is replaced by a low-overlap window. As indicated by its name, this window has a rather low overlap with the following window, thus being optimized for the use of the TNS tool to prevent preecho artefacts in case of transient signals. For normal coding of non-transient signals the sine window is used because of its advantageous frequency response.

In line with normal-delay AAC, the `window_shape` indicates the shape of the trailing part (i.e. the second half) of the analysis window. The shape of the leading part (i.e. the first half) of the analysis window is identical to the `window_shape` of the last block.

**Table 4.118 – window depending on window\_shape**

| window_shape | window      |
|--------------|-------------|
| 0x0          | sine        |
| 0x1          | low-overlap |

The low-overlap window is defined by:

$$W(i) = \begin{cases} 0 & i = 0..3 \cdot N/16 - 1 \\ \sin \left[ \frac{\pi(i - 3 \cdot N/16 + 0.5)}{N/4} \right] & i = 3 \cdot N/16..5 \cdot N/16 - 1 \\ 1 & i = 5 \cdot N/16..11 \cdot N/16 - 1 \\ \sin \left[ \frac{\pi(i - 9 \cdot N/16 + 0.5)}{N/4} \right] & i = 11 \cdot N/16..13 \cdot N/16 - 1 \\ 0 & i = 13 \cdot N/16..N - 1 \end{cases}$$

with  $N = 1024$  or  $N = 960$ .

**4.6.17.2.4 Bit reservoir use**

Use of the bit reservoir is minimized in order to reach the desired target delay. As one extreme case, no bit reservoir is used at all.

**4.6.17.2.5 Tables for temporal noise shaping (TNS)**

The following tables specify the value of TNS\_MAX\_BANDS for the low delay coder:

**Table 4.119 – TNS\_MAX\_BANDS in case of 480 samples per frame**

| Sampling Rate | TNS_MAX_BANDS |
|---------------|---------------|
| 48000         | 31            |
| 44100         | 32            |
| 32000         | 37            |
| 24000         | 30            |
| 22050         | 30            |

Table 4.120 – TNS\_MAX\_BANDS in case of 512 samples per frame

| Sampling Rate | TNS_MAX_BANDS |
|---------------|---------------|
| 48000         | 31            |
| 44100         | 32            |
| 32000         | 37            |
| 24000         | 31            |
| 22050         | 31            |

#### 4.6.17.2.6 Long Term Prediction

The size of the LTP delay buffer size is scaled down proportionally with the frame size. Thus, the size is 2048 and 1920 samples for frame sizes of  $N=512$  and  $N=480$ , respectively (see subclause 4.6.7).

#### 4.6.17.2.7 Adaptation to systems using lower sampling rates

In certain applications it may be necessary to integrate the low delay decoder into an audio system running at lower sampling rates (e.g. 16 kHz) while the nominal sampling rate of the bitstream is much higher (e.g. 48 kHz, corresponding to an algorithmic codec delay of approx. 20 ms). In such cases, it is favorable to decode the output of the low delay codec directly at the target sampling rate rather than using an additional sampling rate conversion operation after decoding.

This can be approximated by appropriate downscaling of both, the frame size and the sampling rate, by some integer factor (e.g. 2, 3), resulting in the same time/frequency resolution of the codec. For example, the codec output can be generated at 16 kHz sampling rate instead of the nominal 48 kHz by retaining only the lowest third (i.e.  $480/3=160$ ) of the spectral coefficients prior to the synthesis filterbank and reducing the inverse transform size to one third (i.e. window size  $960/3=320$ ).

As a consequence, decoding for lower sampling rates reduces both memory and computational requirements, but may not produce exactly the same output as a full-bandwidth decoding, followed by band limiting and sample rate conversion.

Please note that decoding at a lower sampling rate, as described above, does not affect the interpretation of levels, which refers to the nominal sampling rate of a low delay coder bitstream.

## Annex A (normative) Normative Tables

### 4.A.1 Huffman codebook tables for AAC-type noiseless coding

Table 4.A.1 – Scalefactor Huffman Codebook

| Index | length | codeword<br>(hexadecimal) | Index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 18     | 3ffe8                     | 61    | 4      | a                         |
| 1     | 18     | 3ffe6                     | 62    | 4      | c                         |
| 2     | 18     | 3ffe7                     | 63    | 5      | 1b                        |
| 3     | 18     | 3ffe5                     | 64    | 6      | 39                        |
| 4     | 19     | 7fff5                     | 65    | 6      | 3b                        |
| 5     | 19     | 7fff1                     | 66    | 7      | 78                        |
| 6     | 19     | 7ffed                     | 67    | 7      | 7a                        |
| 7     | 19     | 7fff6                     | 68    | 8      | f7                        |
| 8     | 19     | 7fee                      | 69    | 8      | f9                        |
| 9     | 19     | 7ffef                     | 70    | 9      | 1f6                       |
| 10    | 19     | 7fff0                     | 71    | 9      | 1f9                       |
| 11    | 19     | 7fffc                     | 72    | 10     | 3f4                       |
| 12    | 19     | 7fffd                     | 73    | 10     | 3f6                       |
| 13    | 19     | 7fff                      | 74    | 10     | 3f8                       |
| 14    | 19     | 7ffe                      | 75    | 11     | 7f5                       |
| 15    | 19     | 7fff7                     | 76    | 11     | 7f4                       |
| 16    | 19     | 7fff8                     | 77    | 11     | 7f6                       |
| 17    | 19     | 7fffb                     | 78    | 11     | 7f7                       |
| 18    | 19     | 7fff9                     | 79    | 12     | ff5                       |
| 19    | 18     | 3ffe4                     | 80    | 12     | ff8                       |
| 20    | 19     | 7fffa                     | 81    | 13     | 1ff4                      |
| 21    | 18     | 3ffe3                     | 82    | 13     | 1ff6                      |
| 22    | 17     | 1ffef                     | 83    | 13     | 1ff8                      |
| 23    | 17     | 1fff0                     | 84    | 14     | 3ff8                      |
| 24    | 16     | fff5                      | 85    | 14     | 3ff4                      |
| 25    | 17     | 1fee                      | 86    | 16     | fff0                      |
| 26    | 16     | fff2                      | 87    | 15     | 7ff4                      |
| 27    | 16     | fff3                      | 88    | 16     | fff6                      |
| 28    | 16     | fff4                      | 89    | 15     | 7ff5                      |
| 29    | 16     | fff1                      | 90    | 18     | 3ffe2                     |
| 30    | 15     | 7ff6                      | 91    | 19     | 7ffd9                     |
| 31    | 15     | 7ff7                      | 92    | 19     | 7ffda                     |
| 32    | 14     | 3ff9                      | 93    | 19     | 7ffdb                     |
| 33    | 14     | 3ff5                      | 94    | 19     | 7ffdc                     |
| 34    | 14     | 3ff7                      | 95    | 19     | 7ffdd                     |
| 35    | 14     | 3ff3                      | 96    | 19     | 7ffde                     |
| 36    | 14     | 3ff6                      | 97    | 19     | 7ffd8                     |
| 37    | 14     | 3ff2                      | 98    | 19     | 7ffd2                     |
| 38    | 13     | 1ff7                      | 99    | 19     | 7ffd3                     |
| 39    | 13     | 1ff5                      | 100   | 19     | 7ffd4                     |
| 40    | 12     | ff9                       | 101   | 19     | 7ffd5                     |
| 41    | 12     | ff7                       | 102   | 19     | 7ffd6                     |

|    |    |     |     |    |       |
|----|----|-----|-----|----|-------|
| 42 | 12 | ff6 | 103 | 19 | 7fff2 |
| 43 | 11 | 7f9 | 104 | 19 | 7ffdf |
| 44 | 12 | ff4 | 105 | 19 | 7ffe7 |
| 45 | 11 | 7f8 | 106 | 19 | 7ffe8 |
| 46 | 10 | 3f9 | 107 | 19 | 7ffe9 |
| 47 | 10 | 3f7 | 108 | 19 | 7ffea |
| 48 | 10 | 3f5 | 109 | 19 | 7ffeb |
| 49 | 9  | 1f8 | 110 | 19 | 7ffe6 |
| 50 | 9  | 1f7 | 111 | 19 | 7ffe0 |
| 51 | 8  | fa  | 112 | 19 | 7ffe1 |
| 52 | 8  | f8  | 113 | 19 | 7ffe2 |
| 53 | 8  | f6  | 114 | 19 | 7ffe3 |
| 54 | 7  | 79  | 115 | 19 | 7ffe4 |
| 55 | 6  | 3a  | 116 | 19 | 7ffe5 |
| 56 | 6  | 38  | 117 | 19 | 7ffd7 |
| 57 | 5  | 1a  | 118 | 19 | 7ffec |
| 58 | 4  | b   | 119 | 19 | 7fff4 |
| 59 | 3  | 4   | 120 | 19 | 7fff3 |
| 60 | 1  | 0   |     |    |       |

Table 4.A.2 – Spectrum Huffman Codebook 1

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 11     | 7f8                       | 41    | 5      | 14                        |
| 1     | 9      | 1f1                       | 42    | 7      | 65                        |
| 2     | 11     | 7fd                       | 43    | 5      | 16                        |
| 3     | 10     | 3f5                       | 44    | 7      | 6d                        |
| 4     | 7      | 68                        | 45    | 9      | 1e9                       |
| 5     | 10     | 3f0                       | 46    | 7      | 63                        |
| 6     | 11     | 7f7                       | 47    | 9      | 1e4                       |
| 7     | 9      | 1ec                       | 48    | 7      | 6b                        |
| 8     | 11     | 7f5                       | 49    | 5      | 13                        |
| 9     | 10     | 3f1                       | 50    | 7      | 71                        |
| 10    | 7      | 72                        | 51    | 9      | 1e3                       |
| 11    | 10     | 3f4                       | 52    | 7      | 70                        |
| 12    | 7      | 74                        | 53    | 9      | 1f3                       |
| 13    | 5      | 11                        | 54    | 11     | 7fe                       |
| 14    | 7      | 76                        | 55    | 9      | 1e7                       |
| 15    | 9      | 1eb                       | 56    | 11     | 7f3                       |
| 16    | 7      | 6c                        | 57    | 9      | 1ef                       |
| 17    | 10     | 3f6                       | 58    | 7      | 60                        |
| 18    | 11     | 7fc                       | 59    | 9      | 1ee                       |
| 19    | 9      | 1e1                       | 60    | 11     | 7f0                       |
| 20    | 11     | 7f1                       | 61    | 9      | 1e2                       |
| 21    | 9      | 1f0                       | 62    | 11     | 7fa                       |
| 22    | 7      | 61                        | 63    | 10     | 3f3                       |
| 23    | 9      | 1f6                       | 64    | 7      | 6a                        |
| 24    | 11     | 7f2                       | 65    | 9      | 1e8                       |
| 25    | 9      | 1ea                       | 66    | 7      | 75                        |
| 26    | 11     | 7fb                       | 67    | 5      | 10                        |
| 27    | 9      | 1f2                       | 68    | 7      | 73                        |
| 28    | 7      | 69                        | 69    | 9      | 1f4                       |
| 29    | 9      | 1ed                       | 70    | 7      | 6e                        |
| 30    | 7      | 77                        | 71    | 10     | 3f7                       |
| 31    | 5      | 17                        | 72    | 11     | 7f6                       |



|    |   |     |    |    |     |
|----|---|-----|----|----|-----|
| 32 | 7 | 6f  | 73 | 9  | 1e0 |
| 33 | 9 | 1e6 | 74 | 11 | 7f9 |
| 34 | 7 | 64  | 75 | 10 | 3f2 |
| 35 | 9 | 1e5 | 76 | 7  | 66  |
| 36 | 7 | 67  | 77 | 9  | 1f5 |
| 37 | 5 | 15  | 78 | 11 | 7ff |
| 38 | 7 | 62  | 79 | 9  | 1f7 |
| 39 | 5 | 12  | 80 | 11 | 7f4 |
| 40 | 1 | 0   |    |    |     |

Table 4.A.3 – Spectrum Huffman Codebook 2

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 9      | 1f3                       | 41    | 5      | 7                         |
| 1     | 7      | 6f                        | 42    | 6      | 1d                        |
| 2     | 9      | 1fd                       | 43    | 5      | b                         |
| 3     | 8      | eb                        | 44    | 6      | 30                        |
| 4     | 6      | 23                        | 45    | 8      | ef                        |
| 5     | 8      | ea                        | 46    | 6      | 1c                        |
| 6     | 9      | 1f7                       | 47    | 7      | 64                        |
| 7     | 8      | e8                        | 48    | 6      | 1e                        |
| 8     | 9      | 1fa                       | 49    | 5      | c                         |
| 9     | 8      | f2                        | 50    | 6      | 29                        |
| 10    | 6      | 2d                        | 51    | 8      | f3                        |
| 11    | 7      | 70                        | 52    | 6      | 2f                        |
| 12    | 6      | 20                        | 53    | 8      | f0                        |
| 13    | 5      | 6                         | 54    | 9      | 1fc                       |
| 14    | 6      | 2b                        | 55    | 7      | 71                        |
| 15    | 7      | 6e                        | 56    | 9      | 1f2                       |
| 16    | 6      | 28                        | 57    | 8      | f4                        |
| 17    | 8      | e9                        | 58    | 6      | 21                        |
| 18    | 9      | 1f9                       | 59    | 8      | e6                        |
| 19    | 7      | 66                        | 60    | 8      | f7                        |
| 20    | 8      | f8                        | 61    | 7      | 68                        |
| 21    | 8      | e7                        | 62    | 9      | 1f8                       |
| 22    | 6      | 1b                        | 63    | 8      | ee                        |
| 23    | 8      | f1                        | 64    | 6      | 22                        |
| 24    | 9      | 1f4                       | 65    | 7      | 65                        |
| 25    | 7      | 6b                        | 66    | 6      | 31                        |
| 26    | 9      | 1f5                       | 67    | 4      | 2                         |
| 27    | 8      | ec                        | 68    | 6      | 26                        |
| 28    | 6      | 2a                        | 69    | 8      | ed                        |
| 29    | 7      | 6c                        | 70    | 6      | 25                        |
| 30    | 6      | 2c                        | 71    | 7      | 6a                        |
| 31    | 5      | a                         | 72    | 9      | 1fb                       |
| 32    | 6      | 27                        | 73    | 7      | 72                        |
| 33    | 7      | 67                        | 74    | 9      | 1fe                       |
| 34    | 6      | 1a                        | 75    | 7      | 69                        |
| 35    | 8      | f5                        | 76    | 6      | 2e                        |
| 36    | 6      | 24                        | 77    | 8      | f6                        |
| 37    | 5      | 8                         | 78    | 9      | 1ff                       |
| 38    | 6      | 1f                        | 79    | 7      | 6d                        |
| 39    | 5      | 9                         | 80    | 9      | 1f6                       |
| 40    | 3      | 0                         |       |        |                           |

Table 4.A.4 – Spectrum Huffman Codebook 3

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 1      | 0                         | 41    | 10     | 3ef                       |
| 1     | 4      | 9                         | 42    | 9      | 1f3                       |
| 2     | 8      | ef                        | 43    | 9      | 1f4                       |
| 3     | 4      | b                         | 44    | 11     | 7f6                       |
| 4     | 5      | 19                        | 45    | 9      | 1e8                       |
| 5     | 8      | f0                        | 46    | 10     | 3ea                       |
| 6     | 9      | 1eb                       | 47    | 13     | 1ffc                      |
| 7     | 9      | 1e6                       | 48    | 8      | f2                        |
| 8     | 10     | 3f2                       | 49    | 9      | 1f1                       |
| 9     | 4      | a                         | 50    | 12     | ffb                       |
| 10    | 6      | 35                        | 51    | 10     | 3f5                       |
| 11    | 9      | 1ef                       | 52    | 11     | 7f3                       |
| 12    | 6      | 34                        | 53    | 12     | ffc                       |
| 13    | 6      | 37                        | 54    | 8      | ee                        |
| 14    | 9      | 1e9                       | 55    | 10     | 3f7                       |
| 15    | 9      | 1ed                       | 56    | 15     | 7ffe                      |
| 16    | 9      | 1e7                       | 57    | 9      | 1f0                       |
| 17    | 10     | 3f3                       | 58    | 11     | 7f5                       |
| 18    | 9      | 1ee                       | 59    | 15     | 7ffd                      |
| 19    | 10     | 3ed                       | 60    | 13     | 1ffb                      |
| 20    | 13     | 1ffa                      | 61    | 14     | 3ffa                      |
| 21    | 9      | 1ec                       | 62    | 16     | fff                       |
| 22    | 9      | 1f2                       | 63    | 8      | f1                        |
| 23    | 11     | 7f9                       | 64    | 10     | 3f0                       |
| 24    | 11     | 7f8                       | 65    | 14     | 3ffc                      |
| 25    | 10     | 3f8                       | 66    | 9      | 1ea                       |
| 26    | 12     | ff8                       | 67    | 10     | 3ee                       |
| 27    | 4      | 8                         | 68    | 14     | 3ffb                      |
| 28    | 6      | 38                        | 69    | 12     | ff6                       |
| 29    | 10     | 3f6                       | 70    | 12     | ffa                       |
| 30    | 6      | 36                        | 71    | 15     | 7ffc                      |
| 31    | 7      | 75                        | 72    | 11     | 7f2                       |
| 32    | 10     | 3f1                       | 73    | 12     | ff5                       |
| 33    | 10     | 3eb                       | 74    | 16     | ffe                       |
| 34    | 10     | 3ec                       | 75    | 10     | 3f4                       |
| 35    | 12     | ff4                       | 76    | 11     | 7f7                       |
| 36    | 5      | 18                        | 77    | 15     | 7ffb                      |
| 37    | 7      | 76                        | 78    | 12     | ff7                       |
| 38    | 11     | 7f4                       | 79    | 12     | ff9                       |
| 39    | 6      | 39                        | 80    | 15     | 7fa                       |
| 40    | 7      | 74                        |       |        |                           |

Table 4.A.5 – Spectrum Huffman Codebook 4

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 4      | 7                         | 41    | 7      | 6b                        |
| 1     | 5      | 16                        | 42    | 8      | e3                        |
| 2     | 8      | f6                        | 43    | 7      | 69                        |
| 3     | 5      | 18                        | 44    | 9      | 1f3                       |
| 4     | 4      | 8                         | 45    | 8      | eb                        |
| 5     | 8      | ef                        | 46    | 8      | e6                        |

|    |    |     |    |    |     |
|----|----|-----|----|----|-----|
| 6  | 9  | 1ef | 47 | 10 | 3f6 |
| 7  | 8  | f3  | 48 | 7  | 6e  |
| 8  | 11 | 7f8 | 49 | 7  | 6a  |
| 9  | 5  | 19  | 50 | 9  | 1f4 |
| 10 | 5  | 17  | 51 | 10 | 3ec |
| 11 | 8  | ed  | 52 | 9  | 1f0 |
| 12 | 5  | 15  | 53 | 10 | 3f9 |
| 13 | 4  | 1   | 54 | 8  | f5  |
| 14 | 8  | e2  | 55 | 8  | ec  |
| 15 | 8  | f0  | 56 | 11 | 7fb |
| 16 | 7  | 70  | 57 | 8  | ea  |
| 17 | 10 | 3f0 | 58 | 7  | 6f  |
| 18 | 9  | 1ee | 59 | 10 | 3f7 |
| 19 | 8  | f1  | 60 | 11 | 7f9 |
| 20 | 11 | 7fa | 61 | 10 | 3f3 |
| 21 | 8  | ee  | 62 | 12 | fff |
| 22 | 8  | e4  | 63 | 8  | e9  |
| 23 | 10 | 3f2 | 64 | 7  | 6d  |
| 24 | 11 | 7f6 | 65 | 10 | 3f8 |
| 25 | 10 | 3ef | 66 | 7  | 6c  |
| 26 | 11 | 7fd | 67 | 7  | 68  |
| 27 | 4  | 5   | 68 | 9  | 1f5 |
| 28 | 5  | 14  | 69 | 10 | 3ee |
| 29 | 8  | f2  | 70 | 9  | 1f2 |
| 30 | 4  | 9   | 71 | 11 | 7f4 |
| 31 | 4  | 4   | 72 | 11 | 7f7 |
| 32 | 8  | e5  | 73 | 10 | 3f1 |
| 33 | 8  | f4  | 74 | 12 | ffe |
| 34 | 8  | e8  | 75 | 10 | 3ed |
| 35 | 10 | 3f4 | 76 | 9  | 1f1 |
| 36 | 4  | 6   | 77 | 11 | 7f5 |
| 37 | 4  | 2   | 78 | 11 | 7fe |
| 38 | 8  | e7  | 79 | 10 | 3f5 |
| 39 | 4  | 3   | 80 | 11 | 7fc |
| 40 | 4  | 0   |    |    |     |

Table 4.A.6 – Spectrum Huffman Codebook 5

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 13     | 1fff                      | 41    | 4      | a                         |
| 1     | 12     | ff7                       | 42    | 7      | 71                        |
| 2     | 11     | 7f4                       | 43    | 8      | f3                        |
| 3     | 11     | 7e8                       | 44    | 11     | 7e9                       |
| 4     | 10     | 3f1                       | 45    | 11     | 7ef                       |
| 5     | 11     | 7ee                       | 46    | 9      | 1ee                       |
| 6     | 11     | 7f9                       | 47    | 8      | ef                        |
| 7     | 12     | ff8                       | 48    | 5      | 18                        |
| 8     | 13     | 1ffd                      | 49    | 4      | 9                         |
| 9     | 12     | ffd                       | 50    | 5      | 1b                        |
| 10    | 11     | 7f1                       | 51    | 8      | eb                        |
| 11    | 10     | 3e8                       | 52    | 9      | 1e9                       |
| 12    | 9      | 1e8                       | 53    | 11     | 7ec                       |
| 13    | 8      | f0                        | 54    | 11     | 7f6                       |
| 14    | 9      | 1ec                       | 55    | 10     | 3eb                       |
| 15    | 10     | 3ee                       | 56    | 9      | 1f3                       |

|    |    |     |    |    |      |
|----|----|-----|----|----|------|
| 16 | 11 | 7f2 | 57 | 8  | ed   |
| 17 | 12 | ffa | 58 | 7  | 72   |
| 18 | 12 | ff4 | 59 | 8  | e9   |
| 19 | 10 | 3ef | 60 | 9  | 1f1  |
| 20 | 9  | 1f2 | 61 | 10 | 3ed  |
| 21 | 8  | e8  | 62 | 11 | 7f7  |
| 22 | 7  | 70  | 63 | 12 | ff6  |
| 23 | 8  | ec  | 64 | 11 | 7f0  |
| 24 | 9  | 1f0 | 65 | 10 | 3e9  |
| 25 | 10 | 3ea | 66 | 9  | 1ed  |
| 26 | 11 | 7f3 | 67 | 8  | f1   |
| 27 | 11 | 7eb | 68 | 9  | 1ea  |
| 28 | 9  | 1eb | 69 | 10 | 3ec  |
| 29 | 8  | ea  | 70 | 11 | 7f8  |
| 30 | 5  | 1a  | 71 | 12 | ff9  |
| 31 | 4  | 8   | 72 | 13 | 1ffc |
| 32 | 5  | 19  | 73 | 12 | ffc  |
| 33 | 8  | ee  | 74 | 12 | ff5  |
| 34 | 9  | 1ef | 75 | 11 | 7ea  |
| 35 | 11 | 7ed | 76 | 10 | 3f3  |
| 36 | 10 | 3f0 | 77 | 10 | 3f2  |
| 37 | 8  | f2  | 78 | 11 | 7f5  |
| 38 | 7  | 73  | 79 | 12 | ffb  |
| 39 | 4  | b   | 80 | 13 | 1ffe |
| 40 | 1  | 0   |    |    |      |

Table 4.A.7 – Spectrum Huffman Codebook 6

| index | Length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 11     | 7fe                       | 41    | 4      | 3                         |
| 1     | 10     | 3fd                       | 42    | 6      | 2f                        |
| 2     | 9      | 1f1                       | 43    | 7      | 73                        |
| 3     | 9      | 1eb                       | 44    | 9      | 1fa                       |
| 4     | 9      | 1f4                       | 45    | 9      | 1e7                       |
| 5     | 9      | 1ea                       | 46    | 7      | 6e                        |
| 6     | 9      | 1f0                       | 47    | 6      | 2b                        |
| 7     | 10     | 3fc                       | 48    | 4      | 7                         |
| 8     | 11     | 7fd                       | 49    | 4      | 1                         |
| 9     | 10     | 3f6                       | 50    | 4      | 5                         |
| 10    | 9      | 1e5                       | 51    | 6      | 2c                        |
| 11    | 8      | ea                        | 52    | 7      | 6d                        |
| 12    | 7      | 6c                        | 53    | 9      | 1ec                       |
| 13    | 7      | 71                        | 54    | 9      | 1f9                       |
| 14    | 7      | 68                        | 55    | 8      | ee                        |
| 15    | 8      | f0                        | 56    | 6      | 30                        |
| 16    | 9      | 1e6                       | 57    | 6      | 24                        |
| 17    | 10     | 3f7                       | 58    | 6      | 2a                        |
| 18    | 9      | 1f3                       | 59    | 6      | 25                        |
| 19    | 8      | ef                        | 60    | 6      | 33                        |
| 20    | 6      | 32                        | 61    | 8      | ec                        |
| 21    | 6      | 27                        | 62    | 9      | 1f2                       |
| 22    | 6      | 28                        | 63    | 10     | 3f8                       |
| 23    | 6      | 26                        | 64    | 9      | 1e4                       |
| 24    | 6      | 31                        | 65    | 8      | ed                        |
| 25    | 8      | eb                        | 66    | 7      | 6a                        |

|    |   |     |    |    |     |
|----|---|-----|----|----|-----|
| 26 | 9 | 1f7 | 67 | 7  | 70  |
| 27 | 9 | 1e8 | 68 | 7  | 69  |
| 28 | 7 | 6f  | 69 | 7  | 74  |
| 29 | 6 | 2e  | 70 | 8  | f1  |
| 30 | 4 | 8   | 71 | 10 | 3fa |
| 31 | 4 | 4   | 72 | 11 | 7ff |
| 32 | 4 | 6   | 73 | 10 | 3f9 |
| 33 | 6 | 29  | 74 | 9  | 1f6 |
| 34 | 7 | 6b  | 75 | 9  | 1ed |
| 35 | 9 | 1ee | 76 | 9  | 1f8 |
| 36 | 9 | 1ef | 77 | 9  | 1e9 |
| 37 | 7 | 72  | 78 | 9  | 1f5 |
| 38 | 6 | 2d  | 79 | 10 | 3fb |
| 39 | 4 | 2   | 80 | 11 | 7fc |
| 40 | 4 | 0   |    |    |     |

Table 4.A.8 – Spectrum Huffman Codebook 7

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 1      | 0                         | 32    | 8      | f3                        |
| 1     | 3      | 5                         | 33    | 8      | ed                        |
| 2     | 6      | 37                        | 34    | 9      | 1e8                       |
| 3     | 7      | 74                        | 35    | 9      | 1ef                       |
| 4     | 8      | f2                        | 36    | 10     | 3ef                       |
| 5     | 9      | 1eb                       | 37    | 10     | 3f1                       |
| 6     | 10     | 3ed                       | 38    | 10     | 3f9                       |
| 7     | 11     | 7f7                       | 39    | 11     | 7fb                       |
| 8     | 3      | 4                         | 40    | 9      | 1ed                       |
| 9     | 4      | c                         | 41    | 8      | ef                        |
| 10    | 6      | 35                        | 42    | 9      | 1ea                       |
| 11    | 7      | 71                        | 43    | 9      | 1f2                       |
| 12    | 8      | ec                        | 44    | 10     | 3f3                       |
| 13    | 8      | ee                        | 45    | 10     | 3f8                       |
| 14    | 9      | 1ee                       | 46    | 11     | 7f9                       |
| 15    | 9      | 1f5                       | 47    | 11     | 7fc                       |
| 16    | 6      | 36                        | 48    | 10     | 3ee                       |
| 17    | 6      | 34                        | 49    | 9      | 1ec                       |
| 18    | 7      | 72                        | 50    | 9      | 1f4                       |
| 19    | 8      | ea                        | 51    | 10     | 3f4                       |
| 20    | 8      | f1                        | 52    | 10     | 3f7                       |
| 21    | 9      | 1e9                       | 53    | 11     | 7f8                       |
| 22    | 9      | 1f3                       | 54    | 12     | ffd                       |
| 23    | 10     | 3f5                       | 55    | 12     | ffe                       |
| 24    | 7      | 73                        | 56    | 11     | 7f6                       |
| 25    | 7      | 70                        | 57    | 10     | 3f0                       |
| 26    | 8      | eb                        | 58    | 10     | 3f2                       |
| 27    | 8      | f0                        | 59    | 10     | 3f6                       |
| 28    | 9      | 1f1                       | 60    | 11     | 7fa                       |
| 29    | 9      | 1f0                       | 61    | 11     | 7fd                       |
| 30    | 10     | 3ec                       | 62    | 12     | ffc                       |
| 31    | 10     | 3fa                       | 63    | 12     | fff                       |

Table 4.A.9 – Spectrum Huffman Codebook 8

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 5      | e                         | 32    | 7      | 71                        |
| 1     | 4      | 5                         | 33    | 6      | 2b                        |
| 2     | 5      | 10                        | 34    | 6      | 2d                        |
| 3     | 6      | 30                        | 35    | 6      | 31                        |
| 4     | 7      | 6f                        | 36    | 7      | 6d                        |
| 5     | 8      | f1                        | 37    | 7      | 70                        |
| 6     | 9      | 1fa                       | 38    | 8      | f2                        |
| 7     | 10     | 3fe                       | 39    | 9      | 1f9                       |
| 8     | 4      | 3                         | 40    | 8      | ef                        |
| 9     | 3      | 0                         | 41    | 7      | 68                        |
| 10    | 4      | 4                         | 42    | 6      | 33                        |
| 11    | 5      | 12                        | 43    | 7      | 6b                        |
| 12    | 6      | 2c                        | 44    | 7      | 6e                        |
| 13    | 7      | 6a                        | 45    | 8      | ee                        |
| 14    | 7      | 75                        | 46    | 8      | f9                        |
| 15    | 8      | f8                        | 47    | 10     | 3fc                       |
| 16    | 5      | f                         | 48    | 9      | 1f8                       |
| 17    | 4      | 2                         | 49    | 7      | 74                        |
| 18    | 4      | 6                         | 50    | 7      | 73                        |
| 19    | 5      | 14                        | 51    | 8      | ed                        |
| 20    | 6      | 2e                        | 52    | 8      | f0                        |
| 21    | 7      | 69                        | 53    | 8      | f6                        |
| 22    | 7      | 72                        | 54    | 9      | 1f6                       |
| 23    | 8      | f5                        | 55    | 9      | 1fd                       |
| 24    | 6      | 2f                        | 56    | 10     | 3fd                       |
| 25    | 5      | 11                        | 57    | 8      | f3                        |
| 26    | 5      | 13                        | 58    | 8      | f4                        |
| 27    | 6      | 2a                        | 59    | 8      | f7                        |
| 28    | 6      | 32                        | 60    | 9      | 1f7                       |
| 29    | 7      | 6c                        | 61    | 9      | 1fb                       |
| 30    | 8      | ec                        | 62    | 9      | 1fc                       |
| 31    | 8      | fa                        | 63    | 10     | 3ff                       |

Table 4.A.10 – Spectrum Huffman Codebook 9

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 1      | 0                         | 85    | 12     | fa                        |
| 1     | 3      | 5                         | 86    | 12     | fe3                       |
| 2     | 6      | 37                        | 87    | 12     | fe9                       |
| 3     | 8      | e7                        | 88    | 13     | 1fe6                      |
| 4     | 9      | 1de                       | 89    | 13     | 1ff3                      |
| 5     | 10     | 3ce                       | 90    | 13     | 1ff7                      |
| 6     | 10     | 3d9                       | 91    | 11     | 7d3                       |
| 7     | 11     | 7c8                       | 92    | 10     | 3d8                       |
| 8     | 11     | 7cd                       | 93    | 10     | 3e1                       |
| 9     | 12     | fc8                       | 94    | 11     | 7d4                       |
| 10    | 12     | fdd                       | 95    | 11     | 7d9                       |
| 11    | 13     | 1fe4                      | 96    | 12     | fd3                       |
| 12    | 13     | 1fec                      | 97    | 12     | fde                       |
| 13    | 3      | 4                         | 98    | 13     | 1fdd                      |
| 14    | 4      | c                         | 99    | 13     | 1fd9                      |
| 15    | 6      | 35                        | 100   | 13     | 1fe2                      |

|    |    |      |     |    |      |
|----|----|------|-----|----|------|
| 16 | 7  | 72   | 101 | 13 | 1fea |
| 17 | 8  | ea   | 102 | 13 | 1ff1 |
| 18 | 8  | ed   | 103 | 13 | 1ff6 |
| 19 | 9  | 1e2  | 104 | 11 | 7d2  |
| 20 | 10 | 3d1  | 105 | 10 | 3d4  |
| 21 | 10 | 3d3  | 106 | 10 | 3da  |
| 22 | 10 | 3e0  | 107 | 11 | 7c7  |
| 23 | 11 | 7d8  | 108 | 11 | 7d7  |
| 24 | 12 | fcf  | 109 | 11 | 7e2  |
| 25 | 12 | fd5  | 110 | 12 | fce  |
| 26 | 6  | 36   | 111 | 12 | fdb  |
| 27 | 6  | 34   | 112 | 13 | 1fd8 |
| 28 | 7  | 71   | 113 | 13 | 1fee |
| 29 | 8  | e8   | 114 | 14 | 3ff0 |
| 30 | 8  | ec   | 115 | 13 | 1ff4 |
| 31 | 9  | 1e1  | 116 | 14 | 3ff2 |
| 32 | 10 | 3cf  | 117 | 11 | 7e1  |
| 33 | 10 | 3dd  | 118 | 10 | 3df  |
| 34 | 10 | 3db  | 119 | 11 | 7c9  |
| 35 | 11 | 7d0  | 120 | 11 | 7d6  |
| 36 | 12 | fc7  | 121 | 12 | fca  |
| 37 | 12 | fd4  | 122 | 12 | fd0  |
| 38 | 12 | fe4  | 123 | 12 | fe5  |
| 39 | 8  | e6   | 124 | 12 | fe6  |
| 40 | 7  | 70   | 125 | 13 | 1feb |
| 41 | 8  | e9   | 126 | 13 | 1fef |
| 42 | 9  | 1dd  | 127 | 14 | 3ff3 |
| 43 | 9  | 1e3  | 128 | 14 | 3ff4 |
| 44 | 10 | 3d2  | 129 | 14 | 3ff5 |
| 45 | 10 | 3dc  | 130 | 12 | fe0  |
| 46 | 11 | 7cc  | 131 | 11 | 7ce  |
| 47 | 11 | 7ca  | 132 | 11 | 7d5  |
| 48 | 11 | 7de  | 133 | 12 | fc6  |
| 49 | 12 | fd8  | 134 | 12 | fd1  |
| 50 | 12 | fea  | 135 | 12 | fe1  |
| 51 | 13 | 1fdb | 136 | 13 | 1fe0 |
| 52 | 9  | 1df  | 137 | 13 | 1fe8 |
| 53 | 8  | eb   | 138 | 13 | 1ff0 |
| 54 | 9  | 1dc  | 139 | 14 | 3ff1 |
| 55 | 9  | 1e6  | 140 | 14 | 3ff8 |
| 56 | 10 | 3d5  | 141 | 14 | 3ff6 |
| 57 | 10 | 3de  | 142 | 15 | 7ffc |
| 58 | 11 | 7cb  | 143 | 12 | fe8  |
| 59 | 11 | 7dd  | 144 | 11 | 7df  |
| 60 | 11 | 7dc  | 145 | 12 | fc9  |
| 61 | 12 | fcd  | 146 | 12 | fd7  |
| 62 | 12 | fe2  | 147 | 12 | fdc  |
| 63 | 12 | fe7  | 148 | 13 | 1fdc |
| 64 | 13 | 1fe1 | 149 | 13 | 1fdf |
| 65 | 10 | 3d0  | 150 | 13 | 1fed |
| 66 | 9  | 1e0  | 151 | 13 | 1ff5 |
| 67 | 9  | 1e4  | 152 | 14 | 3ff9 |
| 68 | 10 | 3d6  | 153 | 14 | 3ffb |
| 69 | 11 | 7c5  | 154 | 15 | 7ffd |
| 70 | 11 | 7d1  | 155 | 15 | 7ffe |
| 71 | 11 | 7db  | 156 | 13 | 1fe7 |

|    |    |      |     |    |      |
|----|----|------|-----|----|------|
| 72 | 12 | fd2  | 157 | 12 | fcc  |
| 73 | 11 | 7e0  | 158 | 12 | fd6  |
| 74 | 12 | fd9  | 159 | 12 | fdf  |
| 75 | 12 | feb  | 160 | 13 | 1fde |
| 76 | 13 | 1fe3 | 161 | 13 | 1fda |
| 77 | 13 | 1fe9 | 162 | 13 | 1fe5 |
| 78 | 11 | 7c4  | 163 | 13 | 1ff2 |
| 79 | 9  | 1e5  | 164 | 14 | 3ffa |
| 80 | 10 | 3d7  | 165 | 14 | 3ff7 |
| 81 | 11 | 7c6  | 166 | 14 | 3ffc |
| 82 | 11 | 7cf  | 167 | 14 | 3ffd |
| 83 | 11 | 7da  | 168 | 15 | 7fff |
| 84 | 12 | fcb  |     |    |      |

Table 4.A.11 – Spectrum Huffman Codebook 10

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 6      | 22                        | 85    | 9      | 1c7                       |
| 1     | 5      | 8                         | 86    | 9      | 1ca                       |
| 2     | 6      | 1d                        | 87    | 9      | 1e0                       |
| 3     | 6      | 26                        | 88    | 10     | 3db                       |
| 4     | 7      | 5f                        | 89    | 10     | 3e8                       |
| 5     | 8      | d3                        | 90    | 11     | 7ec                       |
| 6     | 9      | 1cf                       | 91    | 9      | 1e3                       |
| 7     | 10     | 3d0                       | 92    | 8      | d2                        |
| 8     | 10     | 3d7                       | 93    | 8      | cb                        |
| 9     | 10     | 3ed                       | 94    | 8      | d0                        |
| 10    | 11     | 7f0                       | 95    | 8      | d7                        |
| 11    | 11     | 7f6                       | 96    | 8      | db                        |
| 12    | 12     | ffd                       | 97    | 9      | 1c6                       |
| 13    | 5      | 7                         | 98    | 9      | 1d5                       |
| 14    | 4      | 0                         | 99    | 9      | 1d8                       |
| 15    | 4      | 1                         | 100   | 10     | 3ca                       |
| 16    | 5      | 9                         | 101   | 10     | 3da                       |
| 17    | 6      | 20                        | 102   | 11     | 7ea                       |
| 18    | 7      | 54                        | 103   | 11     | 7f1                       |
| 19    | 7      | 60                        | 104   | 9      | 1e1                       |
| 20    | 8      | d5                        | 105   | 8      | d4                        |
| 21    | 8      | dc                        | 106   | 8      | cf                        |
| 22    | 9      | 1d4                       | 107   | 8      | d6                        |
| 23    | 10     | 3cd                       | 108   | 8      | de                        |
| 24    | 10     | 3de                       | 109   | 8      | e1                        |
| 25    | 11     | 7e7                       | 110   | 9      | 1d0                       |
| 26    | 6      | 1c                        | 111   | 9      | 1d6                       |
| 27    | 4      | 2                         | 112   | 10     | 3d1                       |
| 28    | 5      | 6                         | 113   | 10     | 3d5                       |
| 29    | 5      | c                         | 114   | 10     | 3f2                       |
| 30    | 6      | 1e                        | 115   | 11     | 7ee                       |
| 31    | 6      | 28                        | 116   | 11     | 7fb                       |
| 32    | 7      | 5b                        | 117   | 10     | 3e9                       |
| 33    | 8      | cd                        | 118   | 9      | 1cd                       |
| 34    | 8      | d9                        | 119   | 9      | 1c8                       |
| 35    | 9      | 1ce                       | 120   | 9      | 1cb                       |
| 36    | 9      | 1dc                       | 121   | 9      | 1d1                       |
| 37    | 10     | 3d9                       | 122   | 9      | 1d7                       |



|    |    |     |     |    |     |
|----|----|-----|-----|----|-----|
| 38 | 10 | 3f1 | 123 | 9  | 1df |
| 39 | 6  | 25  | 124 | 10 | 3cf |
| 40 | 5  | b   | 125 | 10 | 3e0 |
| 41 | 5  | a   | 126 | 10 | 3ef |
| 42 | 5  | d   | 127 | 11 | 7e6 |
| 43 | 6  | 24  | 128 | 11 | 7f8 |
| 44 | 7  | 57  | 129 | 12 | ffa |
| 45 | 7  | 61  | 130 | 10 | 3eb |
| 46 | 8  | cc  | 131 | 9  | 1dd |
| 47 | 8  | dd  | 132 | 9  | 1d3 |
| 48 | 9  | 1cc | 133 | 9  | 1d9 |
| 49 | 9  | 1de | 134 | 9  | 1db |
| 50 | 10 | 3d3 | 135 | 10 | 3d2 |
| 51 | 10 | 3e7 | 136 | 10 | 3cc |
| 52 | 7  | 5d  | 137 | 10 | 3dc |
| 53 | 6  | 21  | 138 | 10 | 3ea |
| 54 | 6  | 1f  | 139 | 11 | 7ed |
| 55 | 6  | 23  | 140 | 11 | 7f3 |
| 56 | 6  | 27  | 141 | 11 | 7f9 |
| 57 | 7  | 59  | 142 | 12 | ff9 |
| 58 | 7  | 64  | 143 | 11 | 7f2 |
| 59 | 8  | d8  | 144 | 10 | 3ce |
| 60 | 8  | df  | 145 | 9  | 1e4 |
| 61 | 9  | 1d2 | 146 | 10 | 3cb |
| 62 | 9  | 1e2 | 147 | 10 | 3d8 |
| 63 | 10 | 3dd | 148 | 10 | 3d6 |
| 64 | 10 | 3ee | 149 | 10 | 3e2 |
| 65 | 8  | d1  | 150 | 10 | 3e5 |
| 66 | 7  | 55  | 151 | 11 | 7e8 |
| 67 | 6  | 29  | 152 | 11 | 7f4 |
| 68 | 7  | 56  | 153 | 11 | 7f5 |
| 69 | 7  | 58  | 154 | 11 | 7f7 |
| 70 | 7  | 62  | 155 | 12 | ffb |
| 71 | 8  | ce  | 156 | 11 | 7fa |
| 72 | 8  | e0  | 157 | 10 | 3ec |
| 73 | 8  | e2  | 158 | 10 | 3df |
| 74 | 9  | 1da | 159 | 10 | 3e1 |
| 75 | 10 | 3d4 | 160 | 10 | 3e4 |
| 76 | 10 | 3e3 | 161 | 10 | 3e6 |
| 77 | 11 | 7eb | 162 | 10 | 3f0 |
| 78 | 9  | 1c9 | 163 | 11 | 7e9 |
| 79 | 7  | 5e  | 164 | 11 | 7ef |
| 80 | 7  | 5a  | 165 | 12 | ff8 |
| 81 | 7  | 5c  | 166 | 12 | ffe |
| 82 | 7  | 63  | 167 | 12 | ffc |
| 83 | 8  | ca  | 168 | 12 | fff |
| 84 | 8  | da  |     |    |     |

Table 4.A.12 – Spectrum Huffman Codebook 11

| index | length | codeword<br>(hexadecimal) | index | length | codeword<br>(hexadecimal) |
|-------|--------|---------------------------|-------|--------|---------------------------|
| 0     | 4      | 0                         | 145   | 10     | 38d                       |
| 1     | 5      | 6                         | 146   | 10     | 398                       |
| 2     | 6      | 19                        | 147   | 10     | 3b7                       |
| 3     | 7      | 3d                        | 148   | 10     | 3d3                       |

|    |    |     |     |    |     |
|----|----|-----|-----|----|-----|
| 4  | 8  | 9c  | 149 | 10 | 3d1 |
| 5  | 8  | c6  | 150 | 10 | 3db |
| 6  | 9  | 1a7 | 151 | 11 | 7dd |
| 7  | 10 | 390 | 152 | 8  | b4  |
| 8  | 10 | 3c2 | 153 | 10 | 3de |
| 9  | 10 | 3df | 154 | 9  | 1a9 |
| 10 | 11 | 7e6 | 155 | 9  | 19b |
| 11 | 11 | 7f3 | 156 | 9  | 19c |
| 12 | 12 | ffb | 157 | 9  | 1a1 |
| 13 | 11 | 7ec | 158 | 9  | 1aa |
| 14 | 12 | ffa | 159 | 9  | 1ad |
| 15 | 12 | ffe | 160 | 9  | 1b3 |
| 16 | 10 | 38e | 161 | 10 | 38b |
| 17 | 5  | 5   | 162 | 10 | 3b2 |
| 18 | 4  | 1   | 163 | 10 | 3b8 |
| 19 | 5  | 8   | 164 | 10 | 3ce |
| 20 | 6  | 14  | 165 | 10 | 3e1 |
| 21 | 7  | 37  | 166 | 10 | 3e0 |
| 22 | 7  | 42  | 167 | 11 | 7d2 |
| 23 | 8  | 92  | 168 | 11 | 7e5 |
| 24 | 8  | af  | 169 | 8  | b7  |
| 25 | 9  | 191 | 170 | 11 | 7e3 |
| 26 | 9  | 1a5 | 171 | 9  | 1bb |
| 27 | 9  | 1b5 | 172 | 9  | 1a8 |
| 28 | 10 | 39e | 173 | 9  | 1a6 |
| 29 | 10 | 3c0 | 174 | 9  | 1b0 |
| 30 | 10 | 3a2 | 175 | 9  | 1b2 |
| 31 | 10 | 3cd | 176 | 9  | 1b7 |
| 32 | 11 | 7d6 | 177 | 10 | 39b |
| 33 | 8  | ae  | 178 | 10 | 39a |
| 34 | 6  | 17  | 179 | 10 | 3ba |
| 35 | 5  | 7   | 180 | 10 | 3b5 |
| 36 | 5  | 9   | 181 | 10 | 3d6 |
| 37 | 6  | 18  | 182 | 11 | 7d7 |
| 38 | 7  | 39  | 183 | 10 | 3e4 |
| 39 | 7  | 40  | 184 | 11 | 7d8 |
| 40 | 8  | 8e  | 185 | 11 | 7ea |
| 41 | 8  | a3  | 186 | 8  | ba  |
| 42 | 8  | b8  | 187 | 11 | 7e8 |
| 43 | 9  | 199 | 188 | 10 | 3a0 |
| 44 | 9  | 1ac | 189 | 9  | 1bd |
| 45 | 9  | 1c1 | 190 | 9  | 1b4 |
| 46 | 10 | 3b1 | 191 | 10 | 38a |
| 47 | 10 | 396 | 192 | 9  | 1c4 |
| 48 | 10 | 3be | 193 | 10 | 392 |
| 49 | 10 | 3ca | 194 | 10 | 3aa |
| 50 | 8  | 9d  | 195 | 10 | 3b0 |
| 51 | 7  | 3c  | 196 | 10 | 3bc |
| 52 | 6  | 15  | 197 | 10 | 3d7 |
| 53 | 6  | 16  | 198 | 11 | 7d4 |
| 54 | 6  | 1a  | 199 | 11 | 7dc |
| 55 | 7  | 3b  | 200 | 11 | 7db |
| 56 | 7  | 44  | 201 | 11 | 7d5 |
| 57 | 8  | 91  | 202 | 11 | 7f0 |
| 58 | 8  | a5  | 203 | 8  | c1  |
| 59 | 8  | be  | 204 | 11 | 7fb |

|     |    |     |     |    |     |
|-----|----|-----|-----|----|-----|
| 60  | 9  | 196 | 205 | 10 | 3c8 |
| 61  | 9  | 1ae | 206 | 10 | 3a3 |
| 62  | 9  | 1b9 | 207 | 10 | 395 |
| 63  | 10 | 3a1 | 208 | 10 | 39d |
| 64  | 10 | 391 | 209 | 10 | 3ac |
| 65  | 10 | 3a5 | 210 | 10 | 3ae |
| 66  | 10 | 3d5 | 211 | 10 | 3c5 |
| 67  | 8  | 94  | 212 | 10 | 3d8 |
| 68  | 8  | 9a  | 213 | 10 | 3e2 |
| 69  | 7  | 36  | 214 | 10 | 3e6 |
| 70  | 7  | 38  | 215 | 11 | 7e4 |
| 71  | 7  | 3a  | 216 | 11 | 7e7 |
| 72  | 7  | 41  | 217 | 11 | 7e0 |
| 73  | 8  | 8c  | 218 | 11 | 7e9 |
| 74  | 8  | 9b  | 219 | 11 | 7f7 |
| 75  | 8  | b0  | 220 | 9  | 190 |
| 76  | 8  | c3  | 221 | 11 | 7f2 |
| 77  | 9  | 19e | 222 | 10 | 393 |
| 78  | 9  | 1ab | 223 | 9  | 1be |
| 79  | 9  | 1bc | 224 | 9  | 1c0 |
| 80  | 10 | 39f | 225 | 10 | 394 |
| 81  | 10 | 38f | 226 | 10 | 397 |
| 82  | 10 | 3a9 | 227 | 10 | 3ad |
| 83  | 10 | 3cf | 228 | 10 | 3c3 |
| 84  | 8  | 93  | 229 | 10 | 3c1 |
| 85  | 8  | bf  | 230 | 10 | 3d2 |
| 86  | 7  | 3e  | 231 | 11 | 7da |
| 87  | 7  | 3f  | 232 | 11 | 7d9 |
| 88  | 7  | 43  | 233 | 11 | 7df |
| 89  | 7  | 45  | 234 | 11 | 7eb |
| 90  | 8  | 9e  | 235 | 11 | 7f4 |
| 91  | 8  | a7  | 236 | 11 | 7fa |
| 92  | 8  | b9  | 237 | 9  | 195 |
| 93  | 9  | 194 | 238 | 11 | 7f8 |
| 94  | 9  | 1a2 | 239 | 10 | 3bd |
| 95  | 9  | 1ba | 240 | 10 | 39c |
| 96  | 9  | 1c3 | 241 | 10 | 3ab |
| 97  | 10 | 3a6 | 242 | 10 | 3a8 |
| 98  | 10 | 3a7 | 243 | 10 | 3b3 |
| 99  | 10 | 3bb | 244 | 10 | 3b9 |
| 100 | 10 | 3d4 | 245 | 10 | 3d0 |
| 101 | 8  | 9f  | 246 | 10 | 3e3 |
| 102 | 9  | 1a0 | 247 | 10 | 3e5 |
| 103 | 8  | 8f  | 248 | 11 | 7e2 |
| 104 | 8  | 8d  | 249 | 11 | 7de |
| 105 | 8  | 90  | 250 | 11 | 7ed |
| 106 | 8  | 98  | 251 | 11 | 7f1 |
| 107 | 8  | a6  | 252 | 11 | 7f9 |
| 108 | 8  | b6  | 253 | 11 | 7fc |
| 109 | 8  | c4  | 254 | 9  | 193 |
| 110 | 9  | 19f | 255 | 12 | ffd |
| 111 | 9  | 1af | 256 | 10 | 3dc |
| 112 | 9  | 1bf | 257 | 10 | 3b6 |
| 113 | 10 | 399 | 258 | 10 | 3c7 |
| 114 | 10 | 3bf | 259 | 10 | 3cc |
| 115 | 10 | 3b4 | 260 | 10 | 3cb |

|     |    |     |     |    |     |
|-----|----|-----|-----|----|-----|
| 116 | 10 | 3c9 | 261 | 10 | 3d9 |
| 117 | 10 | 3e7 | 262 | 10 | 3da |
| 118 | 8  | a8  | 263 | 11 | 7d3 |
| 119 | 9  | 1b6 | 264 | 11 | 7e1 |
| 120 | 8  | ab  | 265 | 11 | 7ee |
| 121 | 8  | a4  | 266 | 11 | 7ef |
| 122 | 8  | aa  | 267 | 11 | 7f5 |
| 123 | 8  | b2  | 268 | 11 | 7f6 |
| 124 | 8  | c2  | 269 | 12 | ffc |
| 125 | 8  | c5  | 270 | 12 | fff |
| 126 | 9  | 198 | 271 | 9  | 19d |
| 127 | 9  | 1a4 | 272 | 9  | 1c2 |
| 128 | 9  | 1b8 | 273 | 8  | b5  |
| 129 | 10 | 38c | 274 | 8  | a1  |
| 130 | 10 | 3a4 | 275 | 8  | 96  |
| 131 | 10 | 3c4 | 276 | 8  | 97  |
| 132 | 10 | 3c6 | 277 | 8  | 95  |
| 133 | 10 | 3dd | 278 | 8  | 99  |
| 134 | 10 | 3e8 | 279 | 8  | a0  |
| 135 | 8  | ad  | 280 | 8  | a2  |
| 136 | 10 | 3af | 281 | 8  | ac  |
| 137 | 9  | 192 | 282 | 8  | a9  |
| 138 | 8  | bd  | 283 | 8  | b1  |
| 139 | 8  | bc  | 284 | 8  | b3  |
| 140 | 9  | 18e | 285 | 8  | bb  |
| 141 | 9  | 197 | 286 | 8  | c0  |
| 142 | 9  | 19a | 287 | 9  | 18f |
| 143 | 9  | 1a3 | 288 | 5  | 4   |
| 144 | 9  | 1b1 |     |    |     |

#### 4.A.2 Window tables

Table 4.A.13 – Kaiser-Bessel window for AAC SSR object type EIGHT\_SHORT\_SEQUENCE

| $i$ | $w(i)$             | $i$ | $w(l)$             |
|-----|--------------------|-----|--------------------|
| 0   | 0.0000875914060105 | 16  | 0.7446454751465113 |
| 1   | 0.0009321760265333 | 17  | 0.8121892962974020 |
| 2   | 0.0032114611466596 | 18  | 0.8683559394406505 |
| 3   | 0.0081009893216786 | 19  | 0.9125649996381605 |
| 4   | 0.0171240286619181 | 20  | 0.9453396205809574 |
| 5   | 0.0320720743527833 | 21  | 0.9680864942677585 |
| 6   | 0.0548307856028528 | 22  | 0.9827581789763112 |
| 7   | 0.0871361822564870 | 23  | 0.9914756203467121 |
| 8   | 0.1302923415174603 | 24  | 0.9961964092194694 |
| 9   | 0.1848955425508276 | 25  | 0.9984956609571091 |
| 10  | 0.2506163195331889 | 26  | 0.9994855586984285 |
| 11  | 0.3260874142923209 | 27  | 0.9998533730714648 |
| 12  | 0.4089316830907141 | 28  | 0.9999671864476404 |
| 13  | 0.4959414909423747 | 29  | 0.9999948432453556 |
| 14  | 0.5833939894958904 | 30  | 0.9999995655238333 |
| 15  | 0.6674601983218376 | 31  | 0.9999999961638728 |

Table 4.A.14 – Kaiser-Bessel window for SSR object type for other window sequences.

| $i$ | $w(i)$             | $i$ | $w(l)$             |
|-----|--------------------|-----|--------------------|
| 0   | 0.0005851230124487 | 128 | 0.7110428359000029 |

|    |                    |     |                    |
|----|--------------------|-----|--------------------|
| 1  | 0.0009642149851497 | 129 | 0.7188474364707993 |
| 2  | 0.0013558207534965 | 130 | 0.7265597347077880 |
| 3  | 0.0017771849644394 | 131 | 0.7341770687621900 |
| 4  | 0.0022352533849672 | 132 | 0.7416968783634273 |
| 5  | 0.0027342299070304 | 133 | 0.7491167073477523 |
| 6  | 0.0032773001022195 | 134 | 0.7564342060337386 |
| 7  | 0.0038671998069216 | 135 | 0.7636471334404891 |
| 8  | 0.0045064443384152 | 136 | 0.7707533593446514 |
| 9  | 0.0051974336885144 | 137 | 0.7777508661725849 |
| 10 | 0.0059425050016407 | 138 | 0.7846377507242818 |
| 11 | 0.0067439602523141 | 139 | 0.7914122257259034 |
| 12 | 0.0076040812644888 | 140 | 0.7980726212080798 |
| 13 | 0.0085251378135895 | 141 | 0.8046173857073919 |
| 14 | 0.0095093917383048 | 142 | 0.8110450872887550 |
| 15 | 0.0105590986429280 | 143 | 0.8173544143867162 |
| 16 | 0.0116765080854300 | 144 | 0.8235441764639875 |
| 17 | 0.0128638627792770 | 145 | 0.8296133044858474 |
| 18 | 0.0141233971318631 | 146 | 0.8355608512093652 |
| 19 | 0.0154573353235409 | 147 | 0.8413859912867303 |
| 20 | 0.0168678890600951 | 148 | 0.8470880211822968 |
| 21 | 0.0183572550877256 | 149 | 0.8526663589032990 |
| 22 | 0.0199276125319803 | 150 | 0.8581205435445334 |
| 23 | 0.0215811201042484 | 151 | 0.8634502346476508 |
| 24 | 0.0233199132076965 | 152 | 0.8686552113760616 |
| 25 | 0.0251461009666641 | 153 | 0.8737353715068081 |
| 26 | 0.0270617631981826 | 154 | 0.8786907302411250 |
| 27 | 0.0290689473405856 | 155 | 0.8835214188357692 |
| 28 | 0.0311696653515848 | 156 | 0.8882276830575707 |
| 29 | 0.0333658905863535 | 157 | 0.8928098814640207 |
| 30 | 0.0356595546648444 | 158 | 0.8972684835130879 |
| 31 | 0.0380525443366107 | 159 | 0.9016040675058185 |
| 32 | 0.0405466983507029 | 160 | 0.9058173183656508 |
| 33 | 0.0431438043376910 | 161 | 0.9099090252587376 |
| 34 | 0.0458455957104702 | 162 | 0.9138800790599416 |
| 35 | 0.0486537485902075 | 163 | 0.9177314696695282 |
| 36 | 0.0515698787635492 | 164 | 0.9214642831859411 |
| 37 | 0.0545955386770205 | 165 | 0.9250796989403991 |
| 38 | 0.0577322144743916 | 166 | 0.9285789863994010 |
| 39 | 0.0609813230826460 | 167 | 0.9319635019415643 |
| 40 | 0.0643442093520723 | 168 | 0.9352346855155568 |
| 41 | 0.0678221432558827 | 169 | 0.9383940571861993 |
| 42 | 0.0714163171546603 | 170 | 0.9414432135761304 |
| 43 | 0.0751278431308314 | 171 | 0.9443838242107182 |
| 44 | 0.0789577503982528 | 172 | 0.9472176277741918 |
| 45 | 0.0829069827918993 | 173 | 0.9499464282852282 |
| 46 | 0.0869763963425241 | 174 | 0.9525720912004834 |
| 47 | 0.0911667569410503 | 175 | 0.9550965394547873 |
| 48 | 0.0954787380973307 | 176 | 0.9575217494469370 |
| 49 | 0.0999129187977865 | 177 | 0.9598497469802043 |
| 50 | 0.1044697814663005 | 178 | 0.9620826031668507 |
| 51 | 0.1091497100326053 | 179 | 0.9642224303060783 |
| 52 | 0.1139529881122542 | 180 | 0.9662713777449607 |
| 53 | 0.1188797973021148 | 181 | 0.9682316277319895 |
| 54 | 0.1239302155951605 | 182 | 0.9701053912729269 |
| 55 | 0.1291042159181728 | 183 | 0.9718949039986892 |
| 56 | 0.1344016647957880 | 184 | 0.9736024220549734 |
| 57 | 0.1398223211441467 | 185 | 0.9752302180233160 |

|     |                    |     |                    |
|-----|--------------------|-----|--------------------|
| 58  | 0.1453658351972151 | 186 | 0.9767805768831932 |
| 59  | 0.1510317475686540 | 187 | 0.9782557920246753 |
| 60  | 0.1568194884519144 | 188 | 0.9796581613210076 |
| 61  | 0.1627283769610327 | 189 | 0.9809899832703159 |
| 62  | 0.1687576206143887 | 190 | 0.9822535532154261 |
| 63  | 0.1749063149634756 | 191 | 0.9834511596505429 |
| 64  | 0.1811734433685097 | 192 | 0.9845850806232530 |
| 65  | 0.1875578769224857 | 193 | 0.9856575802399989 |
| 66  | 0.1940583745250518 | 194 | 0.9866709052828243 |
| 67  | 0.2006735831073503 | 195 | 0.9876272819448033 |
| 68  | 0.2074020380087318 | 196 | 0.9885289126911557 |
| 69  | 0.2142421635060113 | 197 | 0.9893779732525968 |
| 70  | 0.2211922734956977 | 198 | 0.9901766097569984 |
| 71  | 0.2282505723293797 | 199 | 0.9909269360049311 |
| 72  | 0.2354151558022098 | 200 | 0.9916310308941294 |
| 73  | 0.2426840122941792 | 201 | 0.9922909359973702 |
| 74  | 0.2500550240636293 | 202 | 0.9929086532976777 |
| 75  | 0.2575259686921987 | 203 | 0.9934861430841844 |
| 76  | 0.2650945206801527 | 204 | 0.9940253220113651 |
| 77  | 0.2727582531907993 | 205 | 0.9945280613237534 |
| 78  | 0.2805146399424422 | 206 | 0.9949961852476154 |
| 79  | 0.2883610572460804 | 207 | 0.9954314695504363 |
| 80  | 0.2962947861868143 | 208 | 0.9958356402684387 |
| 81  | 0.3043130149466800 | 209 | 0.9962103726017252 |
| 82  | 0.3124128412663888 | 210 | 0.9965572899760172 |
| 83  | 0.3205912750432127 | 211 | 0.9968779632693499 |
| 84  | 0.3288452410620226 | 212 | 0.9971739102014799 |
| 85  | 0.3371715818562547 | 213 | 0.9974465948831872 |
| 86  | 0.3455670606953511 | 214 | 0.9976974275220812 |
| 87  | 0.3540283646950029 | 215 | 0.9979277642809907 |
| 88  | 0.3625521080463003 | 216 | 0.9981389072844972 |
| 89  | 0.3711348353596863 | 217 | 0.9983321047686901 |
| 90  | 0.3797730251194006 | 218 | 0.9985085513687731 |
| 91  | 0.3884630932439016 | 219 | 0.9986693885387259 |
| 92  | 0.3972013967475546 | 220 | 0.9988157050968516 |
| 93  | 0.4059842374986933 | 221 | 0.9989485378906924 |
| 94  | 0.4148078660689724 | 222 | 0.9990688725744943 |
| 95  | 0.4236684856687616 | 223 | 0.9991776444921379 |
| 96  | 0.4325622561631607 | 224 | 0.9992757396582338 |
| 97  | 0.4414852981630577 | 225 | 0.9993639958299003 |
| 98  | 0.4504336971855032 | 226 | 0.9994432036616085 |
| 99  | 0.4594035078775303 | 227 | 0.9995141079353859 |
| 100 | 0.4683907582974173 | 228 | 0.9995774088586188 |
| 101 | 0.4773914542472655 | 229 | 0.9996337634216871 |
| 102 | 0.4864015836506502 | 230 | 0.9996837868076957 |
| 103 | 0.4954171209689973 | 231 | 0.9997280538466377 |
| 104 | 0.5044340316502417 | 232 | 0.9997671005064359 |
| 105 | 0.5134482766032377 | 233 | 0.9998014254134544 |
| 106 | 0.5224558166913167 | 234 | 0.9998314913952471 |
| 107 | 0.5314526172383208 | 235 | 0.9998577270385304 |
| 108 | 0.5404346525403849 | 236 | 0.9998805282555989 |
| 109 | 0.5493979103766972 | 237 | 0.9999002598526793 |
| 110 | 0.5583383965124314 | 238 | 0.9999172570940037 |
| 111 | 0.5672521391870222 | 239 | 0.9999318272557038 |
| 112 | 0.5761351935809411 | 240 | 0.9999442511639580 |
| 113 | 0.5849836462541291 | 241 | 0.9999547847121726 |
| 114 | 0.5937936195492526 | 242 | 0.9999636603523446 |

|     |                    |     |                    |
|-----|--------------------|-----|--------------------|
| 115 | 0.6025612759529649 | 243 | 0.9999710885561258 |
| 116 | 0.6112828224083939 | 244 | 0.9999772592414866 |
| 117 | 0.6199545145721097 | 245 | 0.9999823431612708 |
| 118 | 0.6285726610088878 | 246 | 0.9999864932503106 |
| 119 | 0.6371336273176413 | 247 | 0.9999898459281599 |
| 120 | 0.6456338401819751 | 248 | 0.9999925223548691 |
| 121 | 0.6540697913388968 | 249 | 0.9999946296375997 |
| 122 | 0.6624380414593221 | 250 | 0.9999962619864214 |
| 123 | 0.6707352239341151 | 251 | 0.9999975018180320 |
| 124 | 0.6789580485595255 | 252 | 0.9999984208055542 |
| 125 | 0.6871033051160131 | 253 | 0.9999990808746198 |
| 126 | 0.6951678668345944 | 254 | 0.9999995351446231 |
| 127 | 0.7031486937449871 | 255 | 0.9999998288155155 |

4.A.3 Differential scalefactor to index tables

Table 4.A.15 – transition table 0 (differential scalefactor to index )

| DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX |
|------|-------|------|-------|------|-------|------|-------|------|-------|------|-------|------|-------|------|-------|
| 0    | 68    | 16   | 87    | 32   | 46    | 48   | 25    | 64   | 9     | 80   | 40    | 96   | 96    | 112  | 112   |
| 1    | 69    | 17   | 88    | 33   | 47    | 49   | 19    | 65   | 10    | 81   | 43    | 97   | 97    | 113  | 113   |
| 2    | 70    | 18   | 89    | 34   | 48    | 50   | 20    | 66   | 12    | 82   | 44    | 98   | 98    | 114  | 114   |
| 3    | 71    | 19   | 72    | 35   | 49    | 51   | 14    | 67   | 13    | 83   | 45    | 99   | 99    | 115  | 115   |
| 4    | 75    | 20   | 90    | 36   | 50    | 52   | 15    | 68   | 17    | 84   | 52    | 100  | 100   | 116  | 116   |
| 5    | 76    | 21   | 73    | 37   | 51    | 53   | 16    | 69   | 18    | 85   | 53    | 101  | 101   | 117  | 117   |
| 6    | 77    | 22   | 65    | 38   | 41    | 54   | 11    | 70   | 21    | 86   | 63    | 102  | 102   | 118  | 118   |
| 7    | 78    | 23   | 66    | 39   | 42    | 55   | 7     | 71   | 22    | 87   | 56    | 103  | 103   | 119  | 119   |
| 8    | 79    | 24   | 58    | 40   | 35    | 56   | 8     | 72   | 26    | 88   | 64    | 104  | 104   | 120  | 120   |
| 9    | 80    | 25   | 67    | 41   | 36    | 57   | 5     | 73   | 27    | 89   | 57    | 105  | 105   | 121  | 121   |
| 10   | 81    | 26   | 59    | 42   | 37    | 58   | 2     | 74   | 28    | 90   | 74    | 106  | 106   | 122  | 122   |
| 11   | 82    | 27   | 60    | 43   | 29    | 59   | 1     | 75   | 31    | 91   | 91    | 107  | 107   | 123  | 123   |
| 12   | 83    | 28   | 61    | 44   | 38    | 60   | 0     | 76   | 32    | 92   | 92    | 108  | 108   | 124  | 124   |
| 13   | 84    | 29   | 62    | 45   | 30    | 61   | 3     | 77   | 33    | 93   | 93    | 109  | 109   | 125  | 125   |
| 14   | 85    | 30   | 54    | 46   | 23    | 62   | 4     | 78   | 34    | 94   | 94    | 110  | 110   | 126  | 126   |
| 15   | 86    | 31   | 55    | 47   | 24    | 63   | 6     | 79   | 39    | 95   | 95    | 111  | 111   | 127  | 127   |

Table 4.A.16 – transition table 1 (index to differential scalefactor )

| INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF | INDEX | DIFF |
|-------|------|-------|------|-------|------|-------|------|-------|------|-------|------|-------|------|-------|------|
| 0     | 60   | 16    | 53   | 32    | 76   | 48    | 34   | 64    | 88   | 80    | 9    | 96    | 96   | 112   | 112  |
| 1     | 59   | 17    | 68   | 33    | 77   | 49    | 35   | 65    | 22   | 81    | 10   | 97    | 97   | 113   | 113  |
| 2     | 58   | 18    | 69   | 34    | 78   | 50    | 36   | 66    | 23   | 82    | 11   | 98    | 98   | 114   | 114  |
| 3     | 61   | 19    | 49   | 35    | 40   | 51    | 37   | 67    | 25   | 83    | 12   | 99    | 99   | 115   | 115  |
| 4     | 62   | 20    | 50   | 36    | 41   | 52    | 84   | 68    | 0    | 84    | 13   | 100   | 100  | 116   | 116  |
| 5     | 57   | 21    | 70   | 37    | 42   | 53    | 85   | 69    | 1    | 85    | 14   | 101   | 101  | 117   | 117  |
| 6     | 63   | 22    | 71   | 38    | 44   | 54    | 30   | 70    | 2    | 86    | 15   | 102   | 102  | 118   | 118  |
| 7     | 55   | 23    | 46   | 39    | 79   | 55    | 31   | 71    | 3    | 87    | 16   | 103   | 103  | 119   | 119  |
| 8     | 56   | 24    | 47   | 40    | 80   | 56    | 87   | 72    | 19   | 88    | 17   | 104   | 104  | 120   | 120  |
| 9     | 64   | 25    | 48   | 41    | 38   | 57    | 89   | 73    | 21   | 89    | 18   | 105   | 105  | 121   | 121  |
| 10    | 65   | 26    | 72   | 42    | 39   | 58    | 24   | 74    | 90   | 90    | 20   | 106   | 106  | 122   | 122  |
| 11    | 54   | 27    | 73   | 43    | 81   | 59    | 26   | 75    | 4    | 91    | 91   | 107   | 107  | 123   | 123  |
| 12    | 66   | 28    | 74   | 44    | 82   | 60    | 27   | 76    | 5    | 92    | 92   | 108   | 108  | 124   | 124  |
| 13    | 67   | 29    | 43   | 45    | 83   | 61    | 28   | 77    | 6    | 93    | 93   | 109   | 109  | 125   | 125  |
| 14    | 51   | 30    | 45   | 46    | 32   | 62    | 29   | 78    | 7    | 94    | 94   | 110   | 110  | 126   | 126  |
| 15    | 52   | 31    | 75   | 47    | 33   | 63    | 86   | 79    | 8    | 95    | 95   | 111   | 111  | 127   | 127  |

## 4.A.4 Tables for TwinVQ

Table 4.A.17 – LSP codebook for core coder

| filename   | contents | mode | number of elements | number of vectors |
|------------|----------|------|--------------------|-------------------|
| 20b19s48bs | LSP      | core | 20                 | 64+16+2           |

| VN | EN | 0      | 1      | 2      | 3      | 4      | 5      | 6      | 7      | 8      | 9      | 10     | 11     |
|----|----|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|
| 0  | 0  | 0.1174 | 0.2629 | 0.4047 | 0.5952 | 0.7249 | 0.8803 | 1.0465 | 1.1687 | 1.3174 | 1.4778 | 1.6125 | 1.779  |
|    | 12 | 1.8951 | 2.0787 | 2.2225 | 2.3546 | 2.5002 | 2.6572 | 2.8355 | 2.9783 |        |        |        |        |
| 1  | 0  | 0.1828 | 0.3386 | 0.4116 | 0.5621 | 0.721  | 0.8888 | 1.0088 | 1.1501 | 1.3288 | 1.4473 | 1.5727 | 1.7789 |
|    | 12 | 1.8824 | 2.0304 | 2.1998 | 2.3585 | 2.4955 | 2.6511 | 2.814  | 2.9605 |        |        |        |        |
| 2  | 0  | 0.1293 | 0.3249 | 0.4623 | 0.6582 | 0.7948 | 0.9477 | 1.1068 | 1.2104 | 1.3098 | 1.4293 | 1.569  | 1.757  |
|    | 12 | 1.8739 | 2.0543 | 2.2133 | 2.363  | 2.5169 | 2.6593 | 2.8147 | 2.9676 |        |        |        |        |
| 3  | 0  | 0.1359 | 0.3157 | 0.477  | 0.634  | 0.6863 | 0.797  | 1.0202 | 1.2062 | 1.3237 | 1.4243 | 1.571  | 1.788  |
|    | 12 | 1.9209 | 2.0588 | 2.1726 | 2.3476 | 2.5309 | 2.6718 | 2.8079 | 2.9564 |        |        |        |        |
| 4  | 0  | 0.1311 | 0.2912 | 0.4227 | 0.6131 | 0.7523 | 0.9053 | 1.0771 | 1.2221 | 1.3648 | 1.534  | 1.6759 | 1.8169 |
|    | 12 | 1.902  | 2.0612 | 2.2039 | 2.3483 | 2.4994 | 2.6446 | 2.803  | 2.9602 |        |        |        |        |
| 5  | 0  | 0.1264 | 0.2699 | 0.3763 | 0.581  | 0.7562 | 0.9114 | 1.0854 | 1.2163 | 1.373  | 1.5429 | 1.6488 | 1.7651 |
|    | 12 | 1.8514 | 2.0338 | 2.1947 | 2.3444 | 2.4975 | 2.6497 | 2.8091 | 2.9622 |        |        |        |        |
| 6  | 0  | 0.1437 | 0.2588 | 0.3554 | 0.5655 | 0.7454 | 0.9205 | 1.0799 | 1.171  | 1.3025 | 1.446  | 1.5693 | 1.7406 |
|    | 12 | 1.8593 | 2.0314 | 2.1916 | 2.3495 | 2.5033 | 2.648  | 2.8094 | 2.9634 |        |        |        |        |
| 7  | 0  | 0.1114 | 0.2943 | 0.4312 | 0.5984 | 0.7299 | 0.8952 | 1.0942 | 1.2548 | 1.3628 | 1.4482 | 1.5589 | 1.7495 |
|    | 12 | 1.8897 | 2.0738 | 2.2136 | 2.3605 | 2.5199 | 2.6703 | 2.8201 | 2.9603 |        |        |        |        |
| 8  | 0  | 0.0948 | 0.2009 | 0.3228 | 0.52   | 0.6759 | 0.8336 | 1.0034 | 1.169  | 1.3346 | 1.4869 | 1.6169 | 1.7833 |
|    | 12 | 1.8962 | 2.0733 | 2.2408 | 2.3867 | 2.5308 | 2.6702 | 2.822  | 2.9715 |        |        |        |        |
| 9  | 0  | 0.1354 | 0.2516 | 0.3765 | 0.5569 | 0.6437 | 0.7706 | 0.9587 | 1.1167 | 1.2847 | 1.448  | 1.5884 | 1.7576 |
|    | 12 | 1.8759 | 2.0557 | 2.2131 | 2.3606 | 2.5117 | 2.6561 | 2.8135 | 2.9681 |        |        |        |        |
| 10 | 0  | 0.1457 | 0.3447 | 0.4823 | 0.6299 | 0.7071 | 0.8266 | 1.0111 | 1.159  | 1.3168 | 1.4711 | 1.6095 | 1.7785 |
|    | 12 | 1.8965 | 2.0814 | 2.2277 | 2.3749 | 2.5471 | 2.6981 | 2.8343 | 2.9631 |        |        |        |        |
| 11 | 0  | 0.1308 | 0.2559 | 0.3849 | 0.5877 | 0.6716 | 0.7929 | 0.9911 | 1.1467 | 1.2875 | 1.4331 | 1.5804 | 1.7507 |
|    | 12 | 1.8768 | 2.0533 | 2.2073 | 2.3571 | 2.5075 | 2.6526 | 2.8114 | 2.9668 |        |        |        |        |
| 12 | 0  | 0.11   | 0.2342 | 0.3655 | 0.5678 | 0.7011 | 0.8539 | 1.0156 | 1.1344 | 1.2878 | 1.4475 | 1.5857 | 1.7553 |
|    | 12 | 1.8788 | 2.0568 | 2.209  | 2.3575 | 2.5091 | 2.6533 | 2.8127 | 2.9664 |        |        |        |        |
| 13 | 0  | 0.1276 | 0.2569 | 0.3779 | 0.5755 | 0.7111 | 0.8433 | 1.012  | 1.1497 | 1.2934 | 1.4561 | 1.5821 | 1.7393 |
|    | 12 | 1.8597 | 2.0338 | 2.1981 | 2.3514 | 2.5005 | 2.6466 | 2.8099 | 2.9643 |        |        |        |        |
| 14 | 0  | 0.1449 | 0.2648 | 0.3846 | 0.5961 | 0.7299 | 0.8612 | 1.0124 | 1.1361 | 1.276  | 1.4033 | 1.5325 | 1.7194 |
|    | 12 | 1.8567 | 2.038  | 2.1982 | 2.3487 | 2.5037 | 2.6505 | 2.8093 | 2.9663 |        |        |        |        |
| 15 | 0  | 0.1191 | 0.2724 | 0.3804 | 0.562  | 0.7344 | 0.8313 | 1.0047 | 1.1589 | 1.2462 | 1.4182 | 1.5826 | 1.7253 |
|    | 12 | 1.8671 | 2.0428 | 2.1839 | 2.3524 | 2.4969 | 2.6468 | 2.8092 | 2.9619 |        |        |        |        |
| 16 | 0  | 0.1605 | 0.3428 | 0.4476 | 0.5915 | 0.7067 | 0.8606 | 1.0336 | 1.1538 | 1.3018 | 1.4734 | 1.6169 | 1.7796 |
|    | 12 | 1.9074 | 2.1093 | 2.2537 | 2.3646 | 2.5005 | 2.6536 | 2.8113 | 2.9592 |        |        |        |        |
| 17 | 0  | 0.198  | 0.3043 | 0.3485 | 0.5137 | 0.6684 | 0.836  | 1.034  | 1.18   | 1.3348 | 1.4823 | 1.6125 | 1.7743 |
|    | 12 | 1.8685 | 2.0274 | 2.1877 | 2.3437 | 2.5005 | 2.6494 | 2.8088 | 2.966  |        |        |        |        |
| 18 | 0  | 0.1522 | 0.2816 | 0.3968 | 0.5617 | 0.7004 | 0.8729 | 1.0133 | 1.1174 | 1.2474 | 1.4405 | 1.6196 | 1.7822 |
|    | 12 | 1.8452 | 2.0062 | 2.1803 | 2.3318 | 2.484  | 2.6414 | 2.8079 | 2.9638 |        |        |        |        |
| 19 | 0  | 0.0838 | 0.2169 | 0.3715 | 0.6054 | 0.7642 | 0.8653 | 0.9821 | 1.1144 | 1.2896 | 1.4547 | 1.6024 | 1.7732 |
|    | 12 | 1.8876 | 2.0665 | 2.2032 | 2.3498 | 2.507  | 2.6526 | 2.8113 | 2.9673 |        |        |        |        |
| 20 | 0  | 0.175  | 0.3555 | 0.4484 | 0.5662 | 0.6469 | 0.7965 | 1.0043 | 1.1502 | 1.302  | 1.4691 | 1.6097 | 1.774  |
|    | 12 | 1.8883 | 2.0712 | 2.2111 | 2.3544 | 2.5066 | 2.6538 | 2.813  | 2.9669 |        |        |        |        |
| 21 | 0  | 0.1553 | 0.2518 | 0.3129 | 0.494  | 0.6681 | 0.84   | 1.0047 | 1.1351 | 1.2916 | 1.4545 | 1.6012 | 1.7689 |
|    | 12 | 1.8857 | 2.0635 | 2.2054 | 2.3529 | 2.5061 | 2.6522 | 2.8115 | 2.9686 |        |        |        |        |
| 22 | 0  | 0.151  | 0.2829 | 0.373  | 0.5356 | 0.672  | 0.8278 | 1.0133 | 1.1413 | 1.2788 | 1.4408 | 1.5875 | 1.7644 |
|    | 12 | 1.8764 | 2.0482 | 2.1994 | 2.3476 | 2.5006 | 2.6488 | 2.8092 | 2.9658 |        |        |        |        |
| 23 | 0  | 0.1242 | 0.2415 | 0.3427 | 0.5296 | 0.6996 | 0.8341 | 0.9758 | 1.1223 | 1.2826 | 1.4479 | 1.5923 | 1.7595 |
|    | 12 | 1.8802 | 2.0585 | 2.1987 | 2.3476 | 2.5022 | 2.6495 | 2.8098 | 2.9672 |        |        |        |        |



|    |    |        |        |        |        |        |        |        |        |        |        |        |        |
|----|----|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|
| 24 | 0  | 0.137  | 0.2369 | 0.3319 | 0.4979 | 0.6176 | 0.7802 | 0.9803 | 1.1446 | 1.3162 | 1.4684 | 1.6044 | 1.7707 |
|    | 12 | 1.8874 | 2.063  | 2.2188 | 2.3651 | 2.5155 | 2.6591 | 2.8162 | 2.9694 |        |        |        |        |
| 25 | 0  | 0.1082 | 0.2604 | 0.3933 | 0.5205 | 0.6302 | 0.8223 | 1.0015 | 1.1183 | 1.2792 | 1.4528 | 1.5918 | 1.7571 |
|    | 12 | 1.8807 | 2.0589 | 2.1995 | 2.3482 | 2.5029 | 2.6504 | 2.8099 | 2.967  |        |        |        |        |
| 26 | 0  | 0.1655 | 0.3502 | 0.4789 | 0.6442 | 0.7489 | 0.8833 | 1.0259 | 1.1458 | 1.276  | 1.4114 | 1.5562 | 1.7398 |
|    | 12 | 1.8638 | 2.0451 | 2.2086 | 2.3589 | 2.5122 | 2.6577 | 2.8144 | 2.9662 |        |        |        |        |
| 27 | 0  | 0.1398 | 0.2609 | 0.3662 | 0.539  | 0.6898 | 0.8598 | 1.0172 | 1.1053 | 1.2365 | 1.4228 | 1.5771 | 1.7457 |
|    | 12 | 1.8741 | 2.0544 | 2.2075 | 2.3562 | 2.5101 | 2.6536 | 2.8122 | 2.9669 |        |        |        |        |
| 28 | 0  | 0.0999 | 0.2563 | 0.3139 | 0.4686 | 0.6594 | 0.8453 | 1.0207 | 1.155  | 1.2962 | 1.4475 | 1.5626 | 1.7421 |
|    | 12 | 1.8998 | 2.0971 | 2.2265 | 2.3679 | 2.5087 | 2.6455 | 2.7782 | 2.9457 |        |        |        |        |
| 29 | 0  | 0.1138 | 0.2129 | 0.3229 | 0.5337 | 0.6762 | 0.8288 | 1.0247 | 1.1512 | 1.2682 | 1.4174 | 1.5696 | 1.7451 |
|    | 12 | 1.8698 | 2.0503 | 2.2105 | 2.3574 | 2.5101 | 2.6545 | 2.8123 | 2.9678 |        |        |        |        |
| 30 | 0  | 0.1464 | 0.2629 | 0.3682 | 0.5432 | 0.6492 | 0.8349 | 1.066  | 1.1853 | 1.2746 | 1.4003 | 1.5387 | 1.7244 |
|    | 12 | 1.8658 | 2.0481 | 2.2015 | 2.3521 | 2.5052 | 2.6515 | 2.8106 | 2.967  |        |        |        |        |
| 31 | 0  | 0.1281 | 0.2253 | 0.3181 | 0.5448 | 0.738  | 0.8919 | 1.0314 | 1.1421 | 1.2714 | 1.4252 | 1.5776 | 1.7548 |
|    | 12 | 1.8818 | 2.06   | 2.2127 | 2.3616 | 2.5133 | 2.6548 | 2.8141 | 2.9675 |        |        |        |        |
| 32 | 0  | 0.1671 | 0.2982 | 0.3779 | 0.5418 | 0.6953 | 0.8556 | 1.0118 | 1.1498 | 1.3094 | 1.4649 | 1.6072 | 1.7726 |
|    | 12 | 1.8847 | 2.0659 | 2.2223 | 2.3653 | 2.5163 | 2.6591 | 2.8135 | 2.9657 |        |        |        |        |
| 33 | 0  | 0.1584 | 0.313  | 0.4545 | 0.612  | 0.7032 | 0.8315 | 1.0019 | 1.0994 | 1.2239 | 1.4114 | 1.5721 | 1.7441 |
|    | 12 | 1.8778 | 2.0618 | 2.1931 | 2.3407 | 2.4986 | 2.6482 | 2.8099 | 2.9659 |        |        |        |        |
| 34 | 0  | 0.1343 | 0.3271 | 0.4606 | 0.6457 | 0.8085 | 0.9335 | 1.0208 | 1.1017 | 1.2672 | 1.4591 | 1.6018 | 1.7899 |
|    | 12 | 1.9118 | 2.0595 | 2.2023 | 2.3582 | 2.512  | 2.6632 | 2.8272 | 2.9739 |        |        |        |        |
| 35 | 0  | 0.1772 | 0.3363 | 0.4214 | 0.5756 | 0.727  | 0.8934 | 1.0098 | 1.0963 | 1.2554 | 1.4493 | 1.5827 | 1.7606 |
|    | 12 | 1.892  | 2.0618 | 2.1981 | 2.3514 | 2.5063 | 2.6505 | 2.8097 | 2.9658 |        |        |        |        |
| 36 | 0  | 0.1629 | 0.3196 | 0.422  | 0.5608 | 0.6796 | 0.8588 | 1.007  | 1.1124 | 1.3047 | 1.4725 | 1.5731 | 1.7454 |
|    | 12 | 1.8934 | 2.0498 | 2.1861 | 2.3488 | 2.5041 | 2.6429 | 2.8095 | 2.9683 |        |        |        |        |
| 37 | 0  | 0.196  | 0.3317 | 0.4323 | 0.6092 | 0.6915 | 0.8002 | 0.9693 | 1.1223 | 1.2892 | 1.4404 | 1.5735 | 1.7424 |
|    | 12 | 1.8701 | 2.0495 | 2.201  | 2.3498 | 2.5061 | 2.6522 | 2.8101 | 2.9654 |        |        |        |        |
| 38 | 0  | 0.1468 | 0.3363 | 0.4748 | 0.6645 | 0.7704 | 0.8472 | 0.9688 | 1.1161 | 1.2929 | 1.4648 | 1.6101 | 1.7753 |
|    | 12 | 1.8918 | 2.0788 | 2.2212 | 2.3697 | 2.5156 | 2.6535 | 2.8097 | 2.9669 |        |        |        |        |
| 39 | 0  | 0.2093 | 0.3679 | 0.4453 | 0.613  | 0.7225 | 0.8482 | 1.0341 | 1.1537 | 1.2889 | 1.4605 | 1.5785 | 1.7303 |
|    | 12 | 1.8466 | 2.0314 | 2.1931 | 2.3403 | 2.4949 | 2.6475 | 2.8109 | 2.9672 |        |        |        |        |
| 40 | 0  | 0.099  | 0.2279 | 0.3506 | 0.5357 | 0.6608 | 0.8117 | 0.993  | 1.1395 | 1.3025 | 1.4607 | 1.5989 | 1.7658 |
|    | 12 | 1.883  | 2.062  | 2.2171 | 2.3647 | 2.5148 | 2.6583 | 2.8151 | 2.9689 |        |        |        |        |
| 41 | 0  | 0.1526 | 0.2818 | 0.3926 | 0.5601 | 0.6776 | 0.7982 | 0.9554 | 1.0972 | 1.2681 | 1.4457 | 1.5934 | 1.7644 |
|    | 12 | 1.8867 | 2.0678 | 2.2277 | 2.3753 | 2.5217 | 2.661  | 2.8151 | 2.9671 |        |        |        |        |
| 42 | 0  | 0.1664 | 0.2865 | 0.3753 | 0.5542 | 0.7075 | 0.8239 | 0.9482 | 1.1018 | 1.2804 | 1.4409 | 1.5888 | 1.7569 |
|    | 12 | 1.876  | 2.0564 | 2.2125 | 2.3592 | 2.5109 | 2.6547 | 2.8128 | 2.9673 |        |        |        |        |
| 43 | 0  | 0.1497 | 0.2776 | 0.404  | 0.6093 | 0.6903 | 0.8241 | 0.9759 | 1.0756 | 1.2293 | 1.4062 | 1.564  | 1.7444 |
|    | 12 | 1.869  | 2.0471 | 2.2058 | 2.3548 | 2.5082 | 2.6541 | 2.8123 | 2.9663 |        |        |        |        |
| 44 | 0  | 0.1856 | 0.3246 | 0.416  | 0.5572 | 0.6681 | 0.8443 | 1.0393 | 1.1434 | 1.2551 | 1.4226 | 1.5953 | 1.7619 |
|    | 12 | 1.8762 | 2.0626 | 2.2266 | 2.3616 | 2.5056 | 2.6552 | 2.8174 | 2.9657 |        |        |        |        |
| 45 | 0  | 0.1963 | 0.3169 | 0.3824 | 0.5766 | 0.6988 | 0.8159 | 1.013  | 1.111  | 1.2497 | 1.4341 | 1.5718 | 1.7518 |
|    | 12 | 1.8737 | 2.051  | 2.2107 | 2.3543 | 2.5119 | 2.6529 | 2.812  | 2.9667 |        |        |        |        |
| 46 | 0  | 0.1856 | 0.3454 | 0.4117 | 0.5668 | 0.7447 | 0.8623 | 0.9852 | 1.169  | 1.2915 | 1.3994 | 1.5688 | 1.7447 |
|    | 12 | 1.8442 | 2.0565 | 2.2031 | 2.3417 | 2.5085 | 2.6514 | 2.8045 | 2.9678 |        |        |        |        |
| 47 | 0  | 0.166  | 0.2923 | 0.4013 | 0.5757 | 0.6739 | 0.8321 | 1.0109 | 1.1455 | 1.2546 | 1.3844 | 1.5564 | 1.7446 |
|    | 12 | 1.8702 | 2.0453 | 2.2067 | 2.3576 | 2.5086 | 2.6522 | 2.8103 | 2.9659 |        |        |        |        |
| 48 | 0  | 0.1399 | 0.3315 | 0.4603 | 0.6454 | 0.7743 | 0.9153 | 1.0794 | 1.2339 | 1.3972 | 1.5176 | 1.5984 | 1.7353 |
|    | 12 | 1.8565 | 2.0569 | 2.2083 | 2.3508 | 2.5053 | 2.6524 | 2.8188 | 2.9722 |        |        |        |        |
| 49 | 0  | 0.1288 | 0.3059 | 0.4501 | 0.6671 | 0.7666 | 0.8348 | 0.9684 | 1.1432 | 1.3353 | 1.5161 | 1.6186 | 1.7363 |
|    | 12 | 1.8551 | 2.0594 | 2.207  | 2.3614 | 2.5017 | 2.6432 | 2.8081 | 2.9672 |        |        |        |        |
| 50 | 0  | 0.1283 | 0.3108 | 0.4497 | 0.6447 | 0.7964 | 0.8913 | 0.9786 | 1.1139 | 1.3089 | 1.4954 | 1.6582 | 1.7989 |
|    | 12 | 1.8601 | 2.0313 | 2.2017 | 2.357  | 2.5284 | 2.6688 | 2.8041 | 2.953  |        |        |        |        |
| 51 | 0  | 0.1353 | 0.3106 | 0.4598 | 0.6512 | 0.7234 | 0.8098 | 0.9998 | 1.1857 | 1.3597 | 1.486  | 1.5771 | 1.7548 |
|    | 12 | 1.9111 | 2.1113 | 2.2261 | 2.3281 | 2.4854 | 2.6665 | 2.8452 | 2.9846 |        |        |        |        |
| 52 | 0  | 0.1604 | 0.3143 | 0.4318 | 0.6003 | 0.7132 | 0.8728 | 1.0448 | 1.1722 | 1.3083 | 1.4632 | 1.5988 | 1.7629 |

|    |    |         |         |         |         |         |         |         |         |         |         |         |         |
|----|----|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
| 53 | 12 | 1.8779  | 2.0539  | 2.2067  | 2.3572  | 2.5069  | 2.6503  | 2.8111  | 2.9627  | 1.2412  | 1.4131  | 1.5947  | 1.7693  |
|    | 0  | 0.1028  | 0.2826  | 0.45    | 0.6391  | 0.7665  | 0.9555  | 1.1078  | 1.1585  |         |         |         |         |
| 54 | 12 | 1.8957  | 2.0951  | 2.2557  | 2.3628  | 2.503   | 2.6596  | 2.8153  | 2.9674  | 1.2882  | 1.4453  | 1.6329  | 1.8118  |
|    | 0  | 0.1349  | 0.3392  | 0.4855  | 0.6061  | 0.6773  | 0.8411  | 1.0665  | 1.1875  |         |         |         |         |
| 55 | 12 | 1.8896  | 2.035   | 2.201   | 2.3721  | 2.5096  | 2.6424  | 2.8095  | 2.9693  | 1.3288  | 1.4701  | 1.5703  | 1.7315  |
|    | 0  | 0.2062  | 0.3369  | 0.3912  | 0.5671  | 0.7037  | 0.8188  | 1.0099  | 1.1641  |         |         |         |         |
| 56 | 12 | 1.8461  | 2.022   | 2.1849  | 2.3378  | 2.497   | 2.6473  | 2.8072  | 2.9652  | 1.3263  | 1.4822  | 1.6103  | 1.7697  |
|    | 0  | 0.1611  | 0.2965  | 0.3994  | 0.576   | 0.7066  | 0.8527  | 1.0117  | 1.1601  |         |         |         |         |
| 57 | 12 | 1.885   | 2.037   | 2.1847  | 2.3501  | 2.5087  | 2.6489  | 2.8095  | 2.963   | 1.3099  | 1.4713  | 1.611   | 1.7712  |
|    | 0  | 0.174   | 0.3077  | 0.3979  | 0.5763  | 0.6674  | 0.7658  | 0.9572  | 1.1408  |         |         |         |         |
| 58 | 12 | 1.8662  | 2.0278  | 2.1799  | 2.3338  | 2.4943  | 2.6458  | 2.8068  | 2.9654  | 1.2975  | 1.4555  | 1.5783  | 1.722   |
|    | 0  | 0.1644  | 0.2687  | 0.3549  | 0.558   | 0.7462  | 0.8957  | 0.9979  | 1.1157  |         |         |         |         |
| 59 | 12 | 1.8323  | 2.0177  | 2.1885  | 2.3429  | 2.5006  | 2.6499  | 2.8095  | 2.9659  | 1.346   | 1.5024  | 1.6307  | 1.8283  |
|    | 0  | 0.1396  | 0.3211  | 0.4387  | 0.591   | 0.7217  | 0.8955  | 1.059   | 1.1835  |         |         |         |         |
| 60 | 12 | 1.9458  | 2.0646  | 2.1776  | 2.3532  | 2.5179  | 2.6556  | 2.8191  | 2.9694  | 1.2978  | 1.4558  | 1.5979  | 1.7627  |
|    | 0  | 0.182   | 0.288   | 0.3636  | 0.5345  | 0.6327  | 0.7978  | 1.0031  | 1.1387  |         |         |         |         |
| 61 | 12 | 1.8847  | 2.061   | 2.1793  | 2.354   | 2.5294  | 2.6627  | 2.8095  | 2.9644  | 1.309   | 1.4242  | 1.5993  | 1.7636  |
|    | 0  | 0.1252  | 0.2473  | 0.3853  | 0.5624  | 0.7105  | 0.8498  | 0.9762  | 1.1477  |         |         |         |         |
| 62 | 12 | 1.8327  | 2.0352  | 2.2026  | 2.332   | 2.5015  | 2.6514  | 2.8022  | 2.9628  | 1.2857  | 1.4605  | 1.6015  | 1.7364  |
|    | 0  | 0.2073  | 0.3438  | 0.3918  | 0.5472  | 0.7194  | 0.8508  | 1.0187  | 1.176   |         |         |         |         |
| 63 | 12 | 1.8728  | 2.0492  | 2.1888  | 2.3526  | 2.5016  | 2.6466  | 2.8089  | 2.9626  | 1.3292  | 1.4451  | 1.5527  | 1.7176  |
|    | 0  | 0.1434  | 0.2665  | 0.369   | 0.5565  | 0.6735  | 0.8049  | 1.0074  | 1.1739  |         |         |         |         |
| 64 | 12 | 1.8464  | 2.0366  | 2.1979  | 2.3497  | 2.5038  | 2.6518  | 2.8105  | 2.9653  | 0.0483  | 0.0216  | -0.0111 | -0.0329 |
|    | 0  | 0.0561  | 0.0312  | 0.0155  | -0.0042 | 0.0325  | 0.0171  | 0.0047  | 0.0264  |         |         |         |         |
| 65 | 12 | 0.0229  | 0.0045  | -0.0032 | 0.0015  | 0.0081  | 0.0198  | 0.0181  | 0.0193  | 0.0252  | -0.0252 | -0.009  | -0.003  |
|    | 0  | 0.0423  | 0.0088  | -0.0058 | -0.0414 | -0.0039 | 0.0202  | 0.0252  | 0.0543  |         |         |         |         |
| 66 | 12 | 0.0404  | 0.0114  | 0.068   | 0.1005  | 0.0661  | 0.0072  | -0.0267 | -0.0069 | 0.0163  | 0.0187  | 0.0435  | 0.0378  |
|    | 0  | -0.0397 | -0.0683 | -0.0203 | -0.0272 | 0.0078  | 0.004   | -0.0156 | 0.0067  |         |         |         |         |
| 67 | 12 | 0.0811  | 0.0438  | -0.0306 | -0.0252 | -0.0009 | 0.0161  | 0.0134  | 0.0158  | 0.0359  | 0.024   | 0.0296  | 0.0116  |
|    | 0  | 0.02    | -0.0239 | -0.0162 | -0.01   | 0.0278  | 0.0185  | 0.0115  | 0.0347  |         |         |         |         |
| 68 | 12 | 0.0473  | 0.0197  | 0.0079  | 0.0177  | 0.0266  | 0.043   | 0.0641  | 0.0436  | 0.0485  | 0.0428  | 0.0564  | 0.0481  |
|    | 0  | -0.0128 | 0.0234  | 0.0498  | -0.0345 | -0.0268 | 0.0514  | 0.0364  | 0.0499  |         |         |         |         |
| 69 | 12 | 0.0919  | 0.0574  | 0.013   | 0.0282  | 0.0017  | -0.0071 | 0.0073  | 0.0156  | 0.0195  | 0.0434  | 0.0361  | -0.0285 |
|    | 0  | -0.0074 | 0.0132  | 0.0242  | -0.0196 | 0.0217  | 0.0452  | 0.0326  | 0.0172  |         |         |         |         |
| 70 | 12 | -0.0001 | 0.0035  | 0.0074  | -0.0221 | -0.0314 | -0.0046 | 0.0049  | 0.0143  | 0.0235  | 0.0337  | 0.0322  | -0.0196 |
|    | 0  | -0.0005 | -0.0231 | 0       | -0.0433 | -0.019  | -0.0067 | -0.0218 | 0.0094  |         |         |         |         |
| 71 | 12 | 0.0049  | -0.0037 | 0.0272  | -0.0061 | -0.0058 | 0.021   | 0.017   | 0.0165  | 0.013   | 0.0054  | 0.0201  | 0.0062  |
|    | 0  | 0.0121  | -0.0081 | 0.0233  | 0.0268  | 0.0422  | -0.0014 | -0.0065 | 0.0172  |         |         |         |         |
| 72 | 12 | 0.0431  | 0.016   | 0.0193  | 0.0402  | 0.0449  | 0.0152  | -0.0066 | 0.0082  | 0.0064  | -0.0207 | -0.0137 | -0.0211 |
|    | 0  | -0.0388 | -0.0291 | 0.0236  | -0.0205 | -0.0011 | -0.0001 | -0.0203 | 0.0054  |         |         |         |         |
| 73 | 12 | 0.0169  | -0.0032 | -0.0093 | -0.0004 | 0.0357  | 0.0354  | -0.0014 | -0.0013 | -0.0025 | -0.0027 | 0.0182  | 0.0029  |
|    | 0  | -0.0215 | -0.0158 | 0.0584  | 0.0477  | 0.0426  | 0.0422  | 0.0241  | 0.0117  |         |         |         |         |
| 74 | 12 | 0.0412  | 0.0151  | 0.0549  | 0.0632  | 0.0653  | 0.0757  | 0.0695  | 0.0421  | -0.0203 | -0.0141 | 0.0073  | -0.0058 |
|    | 0  | 0.0022  | 0.0325  | 0.0862  | 0.0287  | 0.0114  | 0.013   | 0.0068  | -0.0033 |         |         |         |         |
| 75 | 12 | 0.0333  | 0.0088  | 0.0676  | 0.0589  | 0.0051  | -0.0155 | -0.0006 | 0.0127  | 0.0253  | 0.0278  | 0.0644  | 0.0385  |
|    | 0  | -0.0292 | -0.0195 | 0.0575  | 0.0156  | 0.0006  | 0.0107  | -0.0123 | 0.0147  |         |         |         |         |
| 76 | 12 | 0.0303  | -0.0152 | -0.0222 | 0.0136  | 0.0206  | 0.0226  | 0.0215  | 0.0208  | 0.0075  | 0.0045  | 0.0157  | 0.002   |
|    | 0  | -0.0255 | -0.0425 | 0.0049  | 0.0131  | 0.0372  | 0.039   | -0.015  | -0.0178 |         |         |         |         |
| 77 | 12 | 0.0357  | 0.0057  | 0.0122  | 0.0176  | 0.0198  | 0.0277  | 0.0226  | 0.0208  | 0.0105  | 0.0036  | 0.0152  | 0.002   |
|    | 0  | 0.0567  | 0.052   | 0.0237  | -0.0588 | -0.0326 | -0.0352 | -0.0394 | 0.0085  |         |         |         |         |
| 78 | 12 | 0.0395  | 0.0105  | 0.028   | 0.0407  | 0.0652  | 0.0661  | 0.0186  | -0.008  | -0.0027 | -0.0019 | 0.017   | 0.0029  |
|    | 0  | 0.032   | 0.0601  | 0.0632  | -0.0162 | -0.0058 | 0       | -0.0441 | -0.0256 |         |         |         |         |
| 79 | 12 | 0.0321  | 0.0081  | 0.0478  | 0.026   | 0.0004  | 0.0197  | 0.0281  | 0.0243  | -0.0023 | -0.0106 | 0.0283  | 0.0012  |
|    | 0  | 0.0268  | 0.0004  | 0.0397  | 0.0076  | -0.0044 | 0.0231  | 0.0238  | 0.0318  |         |         |         |         |
| 80 | 12 | 0.0055  | -0.0158 | 0.0285  | 0.032   | 0.0307  | 0.0348  | 0.0273  | 0.0219  | 0.3672  | 0.3619  | 0.3276  | 0.3173  |
|    | 0  | 0.4987  | 0.4924  | 0.4713  | 0.4427  | 0.4144  | 0.4168  | 0.3813  | 0.3695  |         |         |         |         |
|    | 12 | 0.3115  | 0.278   | 0.3223  | 0.2951  | 0.2826  | 0.2568  | 0.2145  | 0.2527  |         |         |         |         |

|    |    |         |         |        |        |         |         |         |        |        |        |       |         |
|----|----|---------|---------|--------|--------|---------|---------|---------|--------|--------|--------|-------|---------|
| 81 | 0  | 0.3519  | 0.3181  | 0.2607 | 0.206  | 0.1473  | 0.1563  | 0.1073  | 0.0633 | 0.0178 | 0.0178 | -0.03 | -0.0538 |
|    | 12 | -0.0466 | -0.0529 | 0.0389 | -0.024 | -0.0405 | -0.1279 | -0.1221 | 0.0012 |        |        |       |         |

Table 4.A.18 – LSP codebook for scalable coder

| filename   | contents | mode    | number of elements | number of vectors |
|------------|----------|---------|--------------------|-------------------|
| 20b19s48sc | LSP      | enhance | 20                 | 64+16+2           |

| VN | EN | 0      | 1      | 2      | 3      | 4      | 5      | 6      | 7      | 8      | 9      | 10     | 11     |
|----|----|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|
| 0  | 0  | 0.1567 | 0.293  | 0.4518 | 0.6063 | 0.7601 | 0.9216 | 1.067  | 1.2147 | 1.3665 | 1.5191 | 1.6681 | 1.8001 |
|    | 12 | 1.9483 | 2.1034 | 2.2565 | 2.3958 | 2.5371 | 2.6784 | 2.8201 | 2.9648 |        |        |        |        |
| 1  | 0  | 0.1363 | 0.2451 | 0.3925 | 0.524  | 0.6463 | 0.8015 | 0.9244 | 1.0729 | 1.2474 | 1.4157 | 1.5815 | 1.7372 |
|    | 12 | 1.8908 | 2.0462 | 2.2112 | 2.3506 | 2.4974 | 2.653  | 2.7995 | 2.9567 |        |        |        |        |
| 2  | 0  | 0.1649 | 0.2806 | 0.4123 | 0.5608 | 0.7206 | 0.8699 | 0.9973 | 1.1528 | 1.3042 | 1.45   | 1.6105 | 1.7487 |
|    | 12 | 1.8897 | 2.0482 | 2.2069 | 2.3509 | 2.499  | 2.6508 | 2.7986 | 2.9581 |        |        |        |        |
| 3  | 0  | 0.1559 | 0.2632 | 0.4053 | 0.5361 | 0.7076 | 0.8445 | 0.9292 | 1.0251 | 1.1718 | 1.3625 | 1.5449 | 1.7128 |
|    | 12 | 1.879  | 2.0404 | 2.2004 | 2.3505 | 2.5068 | 2.6627 | 2.82   | 2.9759 |        |        |        |        |
| 4  | 0  | 0.1529 | 0.2209 | 0.3506 | 0.4867 | 0.6205 | 0.7836 | 0.8989 | 1.0645 | 1.2688 | 1.3758 | 1.4801 | 1.6433 |
|    | 12 | 1.8303 | 2      | 2.1712 | 2.3409 | 2.5026 | 2.6545 | 2.817  | 2.9744 |        |        |        |        |
| 5  | 0  | 0.1433 | 0.2288 | 0.3578 | 0.4937 | 0.6217 | 0.7847 | 0.9273 | 1.0626 | 1.2336 | 1.3957 | 1.5397 | 1.704  |
|    | 12 | 1.8573 | 2.0034 | 2.1756 | 2.3409 | 2.5    | 2.6518 | 2.807  | 2.9664 |        |        |        |        |
| 6  | 0  | 0.1664 | 0.2349 | 0.3465 | 0.4486 | 0.5634 | 0.7389 | 0.9071 | 1.0751 | 1.243  | 1.3872 | 1.5295 | 1.7175 |
|    | 12 | 1.8303 | 1.9659 | 2.1052 | 2.2759 | 2.4874 | 2.6327 | 2.7668 | 2.9633 |        |        |        |        |
| 7  | 0  | 0.1458 | 0.2419 | 0.3502 | 0.4939 | 0.6698 | 0.8162 | 0.9193 | 1.0387 | 1.2036 | 1.3958 | 1.5755 | 1.7351 |
|    | 12 | 1.8583 | 1.9894 | 2.163  | 2.3414 | 2.4937 | 2.6384 | 2.7951 | 2.9633 |        |        |        |        |
| 8  | 0  | 0.1491 | 0.2608 | 0.4017 | 0.5588 | 0.719  | 0.8763 | 1.0197 | 1.1674 | 1.3192 | 1.477  | 1.6349 | 1.7749 |
|    | 12 | 1.9286 | 2.0907 | 2.2376 | 2.3789 | 2.5253 | 2.6725 | 2.8259 | 2.975  |        |        |        |        |
| 9  | 0  | 0.1812 | 0.2982 | 0.4199 | 0.549  | 0.6944 | 0.7986 | 0.8602 | 0.9987 | 1.2019 | 1.4017 | 1.5757 | 1.7307 |
|    | 12 | 1.8879 | 2.05   | 2.1715 | 2.319  | 2.4852 | 2.6498 | 2.8074 | 2.9718 |        |        |        |        |
| 10 | 0  | 0.164  | 0.2604 | 0.3885 | 0.5396 | 0.6878 | 0.8403 | 0.9959 | 1.1211 | 1.2562 | 1.4133 | 1.5712 | 1.7299 |
|    | 12 | 1.8956 | 2.0557 | 2.2103 | 2.3572 | 2.5108 | 2.6634 | 2.8211 | 2.9763 |        |        |        |        |
| 11 | 0  | 0.1566 | 0.297  | 0.4632 | 0.5803 | 0.6433 | 0.7481 | 0.9195 | 1.1048 | 1.2769 | 1.4439 | 1.6033 | 1.7564 |
|    | 12 | 1.9081 | 2.0646 | 2.2041 | 2.349  | 2.4886 | 2.6269 | 2.7689 | 2.9472 |        |        |        |        |
| 12 | 0  | 0.1467 | 0.2153 | 0.361  | 0.5554 | 0.7325 | 0.8634 | 0.9628 | 1.1097 | 1.3069 | 1.4554 | 1.5611 | 1.6825 |
|    | 12 | 1.8505 | 2.0227 | 2.1789 | 2.3352 | 2.4982 | 2.6573 | 2.815  | 2.9725 |        |        |        |        |
| 13 | 0  | 0.1285 | 0.2004 | 0.354  | 0.5039 | 0.6163 | 0.7581 | 0.9083 | 1.0782 | 1.2559 | 1.4238 | 1.5749 | 1.724  |
|    | 12 | 1.8709 | 2.0279 | 2.1951 | 2.3786 | 2.541  | 2.6805 | 2.8318 | 2.9801 |        |        |        |        |
| 14 | 0  | 0.1463 | 0.2438 | 0.3903 | 0.5417 | 0.6666 | 0.7941 | 0.9412 | 1.1156 | 1.295  | 1.4665 | 1.6099 | 1.722  |
|    | 12 | 1.8562 | 2.0184 | 2.1976 | 2.3507 | 2.5054 | 2.6602 | 2.8182 | 2.9738 |        |        |        |        |
| 15 | 0  | 0.1344 | 0.1615 | 0.2467 | 0.4187 | 0.6347 | 0.81   | 0.9542 | 1.1096 | 1.2665 | 1.4258 | 1.5829 | 1.7285 |
|    | 12 | 1.8835 | 2.0436 | 2.1891 | 2.3415 | 2.4915 | 2.6198 | 2.7569 | 2.9449 |        |        |        |        |
| 16 | 0  | 0.187  | 0.3036 | 0.4315 | 0.535  | 0.6405 | 0.7804 | 0.9596 | 1.1519 | 1.3195 | 1.4341 | 1.5404 | 1.6821 |
|    | 12 | 1.8442 | 2.009  | 2.1585 | 2.3114 | 2.4752 | 2.6417 | 2.8005 | 2.9638 |        |        |        |        |
| 17 | 0  | 0.1751 | 0.2719 | 0.3828 | 0.5296 | 0.7036 | 0.9034 | 1.0637 | 1.1863 | 1.3125 | 1.4169 | 1.5111 | 1.6425 |
|    | 12 | 1.8131 | 1.9929 | 2.1476 | 2.3055 | 2.4739 | 2.6423 | 2.8053 | 2.9703 |        |        |        |        |
| 18 | 0  | 0.1547 | 0.3035 | 0.4669 | 0.5746 | 0.672  | 0.8425 | 1.033  | 1.1782 | 1.2909 | 1.4494 | 1.6306 | 1.7793 |
|    | 12 | 1.8982 | 2.038  | 2.2171 | 2.378  | 2.4998 | 2.6364 | 2.808  | 2.9789 |        |        |        |        |
| 19 | 0  | 0.16   | 0.2923 | 0.4369 | 0.5654 | 0.6968 | 0.8422 | 0.9896 | 1.1608 | 1.3081 | 1.4465 | 1.5856 | 1.7046 |
|    | 12 | 1.8411 | 1.9932 | 2.1526 | 2.3165 | 2.4797 | 2.6425 | 2.8061 | 2.968  |        |        |        |        |
| 20 | 0  | 0.1562 | 0.2721 | 0.4059 | 0.5534 | 0.7148 | 0.8458 | 0.9977 | 1.1582 | 1.3127 | 1.5004 | 1.6237 | 1.7315 |
|    | 12 | 1.912  | 2.0342 | 2.1649 | 2.3464 | 2.4693 | 2.6224 | 2.7995 | 2.9671 |        |        |        |        |
| 21 | 0  | 0.0886 | 0.1482 | 0.3145 | 0.5203 | 0.7025 | 0.8746 | 1.0101 | 1.1515 | 1.3107 | 1.4647 | 1.6207 | 1.7669 |
|    | 12 | 1.9143 | 2.0666 | 2.1859 | 2.3295 | 2.4918 | 2.6541 | 2.8129 | 2.9723 |        |        |        |        |
| 22 | 0  | 0.1526 | 0.2884 | 0.4275 | 0.5815 | 0.7543 | 0.9188 | 1.0653 | 1.2176 | 1.3491 | 1.4997 | 1.69   | 1.8325 |
|    | 12 | 1.9063 | 1.9897 | 2.1563 | 2.3462 | 2.4988 | 2.6451 | 2.7962 | 2.9644 |        |        |        |        |

|    |    |        |        |        |        |        |        |        |        |        |        |        |        |
|----|----|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|
| 23 | 0  | 0.1488 | 0.2703 | 0.4265 | 0.5861 | 0.744  | 0.8934 | 1.0286 | 1.1766 | 1.3575 | 1.5352 | 1.646  | 1.6996 |
|    | 12 | 1.8178 | 2.0112 | 2.2095 | 2.3733 | 2.4985 | 2.632  | 2.7964 | 2.9644 |        |        |        |        |
| 24 | 0  | 0.1572 | 0.2429 | 0.3593 | 0.5062 | 0.6562 | 0.8095 | 0.961  | 1.118  | 1.2825 | 1.4421 | 1.6008 | 1.7497 |
|    | 12 | 1.901  | 2.0565 | 2.208  | 2.3977 | 2.5566 | 2.6866 | 2.8352 | 2.9804 |        |        |        |        |
| 25 | 0  | 0.1483 | 0.2405 | 0.3764 | 0.5123 | 0.6309 | 0.7726 | 0.9575 | 1.1598 | 1.2968 | 1.3939 | 1.5335 | 1.6989 |
|    | 12 | 1.8671 | 2.0255 | 2.1687 | 2.3193 | 2.4833 | 2.6474 | 2.8075 | 2.9693 |        |        |        |        |
| 26 | 0  | 0.14   | 0.2833 | 0.4514 | 0.588  | 0.7017 | 0.838  | 1.0121 | 1.1987 | 1.3348 | 1.4524 | 1.6014 | 1.769  |
|    | 12 | 1.9371 | 2.0673 | 2.2007 | 2.3651 | 2.5405 | 2.6912 | 2.8146 | 2.9573 |        |        |        |        |
| 27 | 0  | 0.147  | 0.2935 | 0.4763 | 0.6216 | 0.7255 | 0.8457 | 0.9767 | 1.1172 | 1.2633 | 1.4177 | 1.5811 | 1.744  |
|    | 12 | 1.8958 | 2.0268 | 2.1556 | 2.3108 | 2.476  | 2.6424 | 2.8053 | 2.9671 |        |        |        |        |
| 28 | 0  | 0.1452 | 0.2348 | 0.3705 | 0.5244 | 0.6837 | 0.8371 | 0.9823 | 1.1497 | 1.3086 | 1.4343 | 1.5756 | 1.734  |
|    | 12 | 1.8896 | 2.043  | 2.1931 | 2.3458 | 2.5031 | 2.6589 | 2.818  | 2.9747 |        |        |        |        |
| 29 | 0  | 0.1752 | 0.2676 | 0.3921 | 0.5289 | 0.6385 | 0.8015 | 1.0014 | 1.1711 | 1.3076 | 1.4502 | 1.5964 | 1.7216 |
|    | 12 | 1.8779 | 2.0451 | 2.1952 | 2.3347 | 2.4832 | 2.6366 | 2.7888 | 2.9543 |        |        |        |        |
| 30 | 0  | 0.1758 | 0.2815 | 0.3937 | 0.5605 | 0.6842 | 0.7915 | 0.9867 | 1.1111 | 1.2982 | 1.4349 | 1.5369 | 1.7311 |
|    | 12 | 1.8721 | 2.0534 | 2.1813 | 2.31   | 2.4778 | 2.6298 | 2.7919 | 2.9531 |        |        |        |        |
| 31 | 0  | 0.1484 | 0.2614 | 0.406  | 0.5621 | 0.6949 | 0.838  | 1.0034 | 1.1408 | 1.3059 | 1.4644 | 1.5745 | 1.7082 |
|    | 12 | 1.8647 | 2.0263 | 2.1836 | 2.3269 | 2.4801 | 2.6403 | 2.7939 | 2.9558 |        |        |        |        |
| 32 | 0  | 0.1859 | 0.2648 | 0.405  | 0.5544 | 0.7034 | 0.853  | 0.9523 | 1.0875 | 1.2349 | 1.3468 | 1.4844 | 1.6538 |
|    | 12 | 1.8288 | 1.9979 | 2.1465 | 2.3027 | 2.4674 | 2.6434 | 2.8068 | 2.9675 |        |        |        |        |
| 33 | 0  | 0.1523 | 0.2676 | 0.4172 | 0.5676 | 0.6782 | 0.7917 | 0.938  | 1.0911 | 1.2527 | 1.4017 | 1.5428 | 1.7005 |
|    | 12 | 1.8617 | 2.0219 | 2.1783 | 2.3305 | 2.4898 | 2.6492 | 2.809  | 2.9696 |        |        |        |        |
| 34 | 0  | 0.1429 | 0.2452 | 0.3811 | 0.5172 | 0.6436 | 0.78   | 0.9157 | 1.0533 | 1.2193 | 1.3916 | 1.5431 | 1.6877 |
|    | 12 | 1.8483 | 2.0113 | 2.1709 | 2.3314 | 2.4933 | 2.6543 | 2.8159 | 2.9774 |        |        |        |        |
| 35 | 0  | 0.1429 | 0.2755 | 0.4623 | 0.6348 | 0.7367 | 0.8006 | 0.9024 | 1.082  | 1.2703 | 1.4362 | 1.597  | 1.7439 |
|    | 12 | 1.8992 | 2.0596 | 2.2391 | 2.3879 | 2.5199 | 2.6538 | 2.8044 | 2.9714 |        |        |        |        |
| 36 | 0  | 0.1755 | 0.2674 | 0.3797 | 0.5093 | 0.6128 | 0.768  | 0.9463 | 1.0888 | 1.2435 | 1.3842 | 1.5028 | 1.653  |
|    | 12 | 1.8138 | 1.9719 | 2.1481 | 2.3143 | 2.4863 | 2.643  | 2.8036 | 2.9727 |        |        |        |        |
| 37 | 0  | 0.1028 | 0.2019 | 0.4564 | 0.5772 | 0.6168 | 0.7817 | 1.0249 | 1.1519 | 1.2362 | 1.3784 | 1.5467 | 1.7077 |
|    | 12 | 1.8748 | 2.0347 | 2.1629 | 2.3126 | 2.4791 | 2.633  | 2.7958 | 2.9645 |        |        |        |        |
| 38 | 0  | 0.1552 | 0.2539 | 0.3842 | 0.4973 | 0.6185 | 0.7823 | 0.9783 | 1.2079 | 1.3287 | 1.4312 | 1.5682 | 1.6595 |
|    | 12 | 1.7912 | 1.9505 | 2.1305 | 2.3021 | 2.4812 | 2.6502 | 2.8077 | 2.973  |        |        |        |        |
| 39 | 0  | 0.144  | 0.2506 | 0.4179 | 0.5561 | 0.6573 | 0.7961 | 0.9659 | 1.1313 | 1.2651 | 1.3966 | 1.5404 | 1.691  |
|    | 12 | 1.85   | 2.0157 | 2.1777 | 2.334  | 2.4927 | 2.6505 | 2.8105 | 2.9723 |        |        |        |        |
| 40 | 0  | 0.1231 | 0.2362 | 0.392  | 0.5375 | 0.6822 | 0.839  | 0.9828 | 1.1363 | 1.2963 | 1.4559 | 1.6114 | 1.7592 |
|    | 12 | 1.9089 | 2.0622 | 2.2153 | 2.3634 | 2.5155 | 2.6669 | 2.8236 | 2.9771 |        |        |        |        |
| 41 | 0  | 0.1644 | 0.3065 | 0.4254 | 0.5421 | 0.7221 | 0.8851 | 0.9748 | 1.1407 | 1.3215 | 1.4231 | 1.5704 | 1.757  |
|    | 12 | 1.8728 | 2.0114 | 2.1897 | 2.3369 | 2.4832 | 2.6524 | 2.7991 | 2.9625 |        |        |        |        |
| 42 | 0  | 0.1541 | 0.2444 | 0.3696 | 0.4892 | 0.584  | 0.7198 | 0.9029 | 1.0817 | 1.2504 | 1.4125 | 1.5675 | 1.7227 |
|    | 12 | 1.8809 | 2.0445 | 2.2064 | 2.3566 | 2.5114 | 2.6657 | 2.8226 | 2.9783 |        |        |        |        |
| 43 | 0  | 0.1685 | 0.2822 | 0.4073 | 0.531  | 0.678  | 0.826  | 0.9442 | 1.0943 | 1.2424 | 1.3927 | 1.5633 | 1.721  |
|    | 12 | 1.8769 | 2.0336 | 2.1866 | 2.3397 | 2.4986 | 2.6565 | 2.8165 | 2.9738 |        |        |        |        |
| 44 | 0  | 0.1595 | 0.2721 | 0.4006 | 0.518  | 0.6605 | 0.8306 | 0.9808 | 1.1277 | 1.3003 | 1.4401 | 1.5503 | 1.6986 |
|    | 12 | 1.8766 | 2.0486 | 2.2281 | 2.3824 | 2.5148 | 2.6498 | 2.7995 | 2.9672 |        |        |        |        |
| 45 | 0  | 0.1576 | 0.2128 | 0.3353 | 0.4837 | 0.5821 | 0.6951 | 0.8633 | 1.013  | 1.2138 | 1.4012 | 1.5417 | 1.7026 |
|    | 12 | 1.8413 | 2.004  | 2.1891 | 2.342  | 2.5054 | 2.6629 | 2.8164 | 2.9719 |        |        |        |        |
| 46 | 0  | 0.1623 | 0.2444 | 0.3882 | 0.541  | 0.6619 | 0.8141 | 0.9687 | 1.1125 | 1.2987 | 1.4352 | 1.5293 | 1.6206 |
|    | 12 | 1.7733 | 1.9671 | 2.1671 | 2.3417 | 2.5032 | 2.6485 | 2.8044 | 2.975  |        |        |        |        |
| 47 | 0  | 0.1645 | 0.2494 | 0.4065 | 0.5624 | 0.7081 | 0.8553 | 0.9783 | 1.117  | 1.2605 | 1.3666 | 1.5037 | 1.6825 |
|    | 12 | 1.8595 | 2.0285 | 2.2063 | 2.3573 | 2.512  | 2.6642 | 2.8209 | 2.9768 |        |        |        |        |
| 48 | 0  | 0.1412 | 0.2599 | 0.4089 | 0.5763 | 0.7575 | 0.9404 | 1.0963 | 1.2039 | 1.2875 | 1.4104 | 1.5831 | 1.7469 |
|    | 12 | 1.9057 | 2.0655 | 2.2198 | 2.3592 | 2.5008 | 2.6519 | 2.8075 | 2.9645 |        |        |        |        |
| 49 | 0  | 0.1506 | 0.2249 | 0.3354 | 0.4702 | 0.6394 | 0.8209 | 0.9874 | 1.1137 | 1.2158 | 1.3713 | 1.5632 | 1.7234 |
|    | 12 | 1.8511 | 2.0006 | 2.1659 | 2.3298 | 2.4952 | 2.6518 | 2.8204 | 2.9774 |        |        |        |        |
| 50 | 0  | 0.1427 | 0.2519 | 0.4087 | 0.5698 | 0.73   | 0.8871 | 1.0136 | 1.1402 | 1.2784 | 1.4188 | 1.5715 | 1.7279 |
|    | 12 | 1.8821 | 2.0373 | 2.1915 | 2.3442 | 2.5019 | 2.6578 | 2.8168 | 2.9741 |        |        |        |        |
| 51 | 0  | 0.1546 | 0.2464 | 0.3871 | 0.5358 | 0.6887 | 0.8628 | 1.0297 | 1.1593 | 1.2707 | 1.3877 | 1.5348 | 1.7029 |

|    |    |         |         |         |         |         |         |         |         |         |         |         |         |
|----|----|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
|    | 12 | 1.8544  | 2.0102  | 2.1646  | 2.3194  | 2.485   | 2.647   | 2.8099  | 2.971   |         |         |         |         |
| 52 | 0  | 0.1563  | 0.2624  | 0.3977  | 0.5517  | 0.7151  | 0.8666  | 0.9931  | 1.1265  | 1.2623  | 1.419   | 1.5703  | 1.7052  |
|    | 12 | 1.8638  | 2.0244  | 2.1786  | 2.333   | 2.4926  | 2.6513  | 2.8121  | 2.9714  |         |         |         |         |
| 53 | 0  | 0.1756  | 0.3055  | 0.4336  | 0.5642  | 0.6999  | 0.8519  | 1.0053  | 1.1666  | 1.3314  | 1.4941  | 1.6493  | 1.7804  |
|    | 12 | 1.9026  | 2.041   | 2.1964  | 2.3556  | 2.5057  | 2.6603  | 2.8194  | 2.972   |         |         |         |         |
| 54 | 0  | 0.1584  | 0.3036  | 0.4517  | 0.5956  | 0.7302  | 0.8717  | 1.0275  | 1.1963  | 1.3597  | 1.5018  | 1.6108  | 1.7443  |
|    | 12 | 1.9113  | 2.0817  | 2.223   | 2.3474  | 2.489   | 2.6508  | 2.8187  | 2.9778  |         |         |         |         |
| 55 | 0  | 0.1863  | 0.2831  | 0.4032  | 0.5544  | 0.7006  | 0.8505  | 1.0197  | 1.1847  | 1.3216  | 1.427   | 1.5489  | 1.695   |
|    | 12 | 1.8516  | 2.0187  | 2.152   | 2.3037  | 2.4735  | 2.6409  | 2.804   | 2.9682  |         |         |         |         |
| 56 | 0  | 0.1365  | 0.2121  | 0.3269  | 0.4585  | 0.6069  | 0.7824  | 0.9466  | 1.1138  | 1.282   | 1.4431  | 1.6006  | 1.7496  |
|    | 12 | 1.9023  | 2.0596  | 2.2231  | 2.3686  | 2.5198  | 2.6706  | 2.8259  | 2.9791  |         |         |         |         |
| 57 | 0  | 0.1418  | 0.2012  | 0.3235  | 0.4318  | 0.5561  | 0.7305  | 0.9193  | 1.0594  | 1.2154  | 1.3873  | 1.5191  | 1.6691  |
|    | 12 | 1.8355  | 2.0049  | 2.1635  | 2.3106  | 2.4786  | 2.6498  | 2.7995  | 2.9615  |         |         |         |         |
| 58 | 0  | 0.1298  | 0.2535  | 0.4368  | 0.6313  | 0.8039  | 0.9111  | 0.9697  | 1.0732  | 1.2481  | 1.4316  | 1.5946  | 1.7482  |
|    | 12 | 1.8986  | 2.0577  | 2.219   | 2.3698  | 2.5216  | 2.6719  | 2.829   | 2.9839  |         |         |         |         |
| 59 | 0  | 0.164   | 0.2683  | 0.3859  | 0.5464  | 0.721   | 0.8791  | 0.9947  | 1.1248  | 1.2728  | 1.42    | 1.5777  | 1.6956  |
|    | 12 | 1.8111  | 1.9719  | 2.1449  | 2.334   | 2.5168  | 2.6671  | 2.83    | 2.982   |         |         |         |         |
| 60 | 0  | 0.16    | 0.2608  | 0.4067  | 0.5504  | 0.6802  | 0.817   | 0.9717  | 1.1255  | 1.2888  | 1.4648  | 1.6376  | 1.8126  |
|    | 12 | 1.9239  | 2.0146  | 2.1353  | 2.3044  | 2.4523  | 2.618   | 2.7863  | 2.9651  |         |         |         |         |
| 61 | 0  | 0.1642  | 0.2736  | 0.3797  | 0.5014  | 0.6611  | 0.8243  | 0.9655  | 1.1203  | 1.2834  | 1.4431  | 1.604   | 1.7534  |
|    | 12 | 1.9015  | 2.0469  | 2.1684  | 2.318   | 2.484   | 2.6484  | 2.8086  | 2.9696  |         |         |         |         |
| 62 | 0  | 0.15    | 0.273   | 0.4163  | 0.5685  | 0.7419  | 0.9107  | 1.0222  | 1.1298  | 1.2903  | 1.4624  | 1.6191  | 1.7762  |
|    | 12 | 1.9303  | 2.0511  | 2.1871  | 2.3528  | 2.5078  | 2.6608  | 2.8287  | 2.9768  |         |         |         |         |
| 63 | 0  | 0.1448  | 0.2603  | 0.3473  | 0.4684  | 0.6635  | 0.8312  | 0.9606  | 1.124   | 1.2812  | 1.3949  | 1.5717  | 1.7249  |
|    | 12 | 1.9085  | 2.0455  | 2.1565  | 2.29    | 2.4086  | 2.6323  | 2.8375  | 2.9776  |         |         |         |         |
| 64 | 0  | 0.0383  | 0.0567  | 0.0096  | 0.0058  | 0.0176  | 0.0025  | 0.0011  | -0.0004 | 0.0199  | 0.0127  | -0.0218 | -0.0168 |
|    | 12 | 0.0079  | 0.0142  | -0.0017 | 0.0067  | 0.0106  | 0.0125  | 0.01    | 0.0094  |         |         |         |         |
| 65 | 0  | 0.0081  | 0.0043  | -0.0335 | -0.0402 | -0.012  | 0.0213  | 0.042   | 0.0507  | 0.0238  | -0.0191 | -0.0249 | -0.0022 |
|    | 12 | 0.0109  | 0.0117  | 0.0497  | 0.0557  | 0.0464  | 0.035   | 0.0275  | 0.019   |         |         |         |         |
| 66 | 0  | -0.0261 | -0.048  | -0.046  | -0.014  | 0.0035  | 0.0071  | 0.0125  | 0.0034  | -0.0005 | 0.0004  | 0.0096  | 0.034   |
|    | 12 | 0.0524  | 0.0495  | -0.0303 | -0.0281 | -0.008  | 0.0009  | -0.0004 | 0.0039  |         |         |         |         |
| 67 | 0  | 0.0102  | -0.0001 | -0.0296 | -0.0006 | 0.0327  | 0.0346  | 0.0448  | 0.0414  | 0.0344  | 0.0273  | 0.0219  | 0.0264  |
|    | 12 | 0.0257  | 0.0208  | -0.012  | -0.0134 | 0.0141  | 0.0377  | 0.0392  | 0.0162  |         |         |         |         |
| 68 | 0  | 0.0104  | 0.0071  | 0.011   | 0.0152  | -0.0102 | 0.0115  | 0.0289  | 0.0372  | 0.0476  | 0.0562  | 0.0607  | 0.0667  |
|    | 12 | 0.0618  | 0.0449  | 0.0256  | -0.0128 | -0.0182 | 0.0032  | 0.005   | 0.0107  |         |         |         |         |
| 69 | 0  | -0.0177 | 0.0338  | 0.0173  | 0.0015  | 0.0224  | 0.0144  | 0.034   | 0.0447  | 0.0526  | 0.0358  | -0.0029 | -0.0166 |
|    | 12 | -0.0123 | -0.0052 | 0.0066  | -0.0246 | -0.0336 | -0.0143 | -0.0047 | 0.0054  |         |         |         |         |
| 70 | 0  | -0.0163 | 0.0226  | 0.0214  | -0.0236 | -0.0324 | -0.0239 | -0.0003 | 0.0098  | -0.0045 | -0.0015 | 0.0045  | 0.0126  |
|    | 12 | 0.0129  | 0.0104  | -0.0208 | -0.0085 | 0.0011  | 0.0066  | 0.0228  | 0.0289  |         |         |         |         |
| 71 | 0  | 0.0096  | 0.0184  | 0.0142  | 0.0422  | 0.0419  | -0.0037 | 0.015   | 0.0267  | 0.0238  | 0.0191  | 0.0134  | 0.0177  |
|    | 12 | 0.0218  | 0.0207  | 0.041   | 0.0594  | 0.0599  | 0.0113  | -0.0264 | -0.0092 |         |         |         |         |
| 72 | 0  | -0.0428 | -0.0164 | -0.002  | -0.0084 | -0.0008 | 0.0143  | 0.0203  | 0.0009  | -0.0134 | -0.0132 | -0.0215 | -0.0289 |
|    | 12 | -0.0202 | -0.0083 | 0.0009  | 0.0297  | 0.0171  | -0.015  | -0.0062 | 0.0079  |         |         |         |         |
| 73 | 0  | -0.0259 | 0.013   | 0.0487  | 0.0691  | 0.0466  | 0.035   | 0.0465  | 0.0052  | -0.0155 | 0.0023  | 0.0134  | 0.0146  |
|    | 12 | 0.0142  | 0.011   | 0.029   | 0.0326  | 0.0437  | 0.0669  | 0.0835  | 0.0498  |         |         |         |         |
| 74 | 0  | -0.0035 | 0.0321  | 0.054   | 0.0382  | 0.0055  | 0.0101  | 0.0146  | -0.0031 | -0.0324 | -0.0381 | -0.0189 | 0.0044  |
|    | 12 | 0.0177  | 0.0185  | 0.0471  | 0.0252  | -0.0066 | -0.0072 | -0.0129 | -0.0052 |         |         |         |         |
| 75 | 0  | -0.0373 | -0.0028 | 0.0375  | 0.0295  | -0.0016 | -0.0017 | -0.0022 | 0.0084  | 0.0219  | 0.0322  | 0.0421  | 0.0369  |
|    | 12 | 0.0094  | -0.0019 | 0.0096  | 0.0242  | 0.0324  | 0.031   | 0.0363  | 0.0306  |         |         |         |         |
| 76 | 0  | -0.0433 | -0.0333 | 0.001   | 0.0338  | 0.034   | 0.0181  | -0.0051 | -0.0337 | -0.017  | 0.0115  | 0.0188  | 0.0132  |
|    | 12 | 0.011   | 0.0114  | 0.0232  | 0.025   | 0.0232  | 0.0203  | 0.015   | 0.0128  |         |         |         |         |
| 77 | 0  | 0.0552  | 0.0672  | -0.0093 | -0.0535 | -0.022  | -0.0204 | -0.0284 | -0.0194 | -0.0177 | -0.0228 | -0.0217 | -0.0114 |
|    | 12 | -0.0043 | -0.0033 | -0.0012 | 0.0158  | 0.0533  | 0.0514  | 0.0011  | -0.0072 |         |         |         |         |
| 78 | 0  | 0.0209  | 0.0653  | 0.0635  | 0.0541  | 0.0327  | -0.0325 | -0.0418 | -0.0221 | -0.002  | 0.0078  | 0.0096  | 0.0202  |
|    | 12 | 0.0285  | 0.0278  | 0.0574  | 0.0428  | -0.0096 | -0.011  | 0.026   | 0.0358  |         |         |         |         |
| 79 | 0  | 0.0146  | 0.062   | 0.0578  | 0.0008  | -0.0285 | 0.0054  | 0.0183  | 0.0136  | 0.0081  | 0.001   | -0.0056 | 0.0145  |
|    | 12 | -0.0033 | -0.0204 | 0.038   | 0.0141  | 0.0198  | 0.0573  | 0.031   | -0.0055 |         |         |         |         |

|    |    |        |        |        |        |        |        |        |        |        |        |        |        |
|----|----|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|--------|
| 80 | 0  | 0.4989 | 0.4993 | 0.499  | 0.4995 | 0.4999 | 0.4985 | 0.4973 | 0.499  | 0.4977 | 0.4971 | 0.4977 | 0.4995 |
|    | 12 | 0.4969 | 0.4986 | 0.4996 | 0.4996 | 0.4999 | 0.4989 | 0.4937 | 0.4906 |        |        |        |        |
| 81 | 0  | 0.2664 | 0.3126 | 0.3112 | 0.3132 | 0.2987 | 0.2609 | 0.2724 | 0.2535 | 0.2363 | 0.2679 | 0.2746 | 0.2536 |
|    | 12 | 0.2708 | 0.269  | 0.2831 | 0.2939 | 0.262  | 0.2428 | 0.2456 | 0.2665 |        |        |        |        |

Table 4.A.19 – Interleave VQ codebook 0 for core coder (WINDOW TYPE : LONG)

| filename | Contents | mode      | number of elements | number of vectors |
|----------|----------|-----------|--------------------|-------------------|
| cdmdct_0 | MDCT     | core LONG | 20                 | 32                |

| VN | EN | 0       | 1       | 2       | 3       | 4        | 5      | 6       | 7        | 8        | 9      | 10     | 11     |
|----|----|---------|---------|---------|---------|----------|--------|---------|----------|----------|--------|--------|--------|
| 0  | 0  | -165.2  | 44.8    | 83.7    | -31.8   | -54.7    | 79.8   | 104.3   | -11322.7 | -46.7    | -5.7   | 139.8  | -38.4  |
|    | 12 | -59.7   | -101.2  | -5.6    | 134.4   | 34       | 113    | -738.8  | 80.1     |          |        |        |        |
| 1  | 0  | 17.2    | 4.2     | 148.2   | 54.7    | 54.2     | -11.9  | 74.7    | 35.2     | 73.9     | 90     | 9523.9 | 73.2   |
|    | 12 | 1.1     | -47.8   | -5.7    | -966.6  | 60       | 120.8  | -189.5  | -59.4    |          |        |        |        |
| 2  | 0  | 395.5   | 136     | 5816.1  | -64.1   | -89.2    | 17.3   | 371.2   | -9.6     | 11.2     | -10.6  | -31.9  | 27.1   |
|    | 12 | -41.5   | 31.5    | -496.7  | -138.5  | -2108.3  | -272.2 | -99.2   | 262.8    |          |        |        |        |
| 3  | 0  | -1136.4 | -142.9  | 73.5    | -92     | -544.9   | -28.6  | -79.1   | -85.1    | 136.3    | -5.4   | -122.4 | -81.2  |
|    | 12 | -6.2    | 5.8     | -123.2  | 11554.5 | 4.3      | -71.2  | -22.1   | -203.1   |          |        |        |        |
| 4  | 0  | -56     | 14.4    | 30      | -2.6    | 29.7     | -150   | 96.5    | 71.3     | 24.7     | -130.8 | -7.1   | 5682.9 |
|    | 12 | -7.6    | 43.1    | 15.6    | 35.9    | -219.7   | -84.9  | -1051.1 | 647.5    |          |        |        |        |
| 5  | 0  | 209.2   | -93.8   | -105.8  | -18.2   | 51.5     | 22.9   | -38.3   | 88       | 4.6      | -38.3  | -118.2 | 328.6  |
|    | 12 | 12156.2 | -111    | -47     | 109.5   | -3.2     | 116.4  | -156.3  | 51       |          |        |        |        |
| 6  | 0  | 69.1    | -74.3   | 15.4    | 184.4   | -165.6   | -97.8  | 23.8    | 12.2     | -12330.6 | -72.9  | 57.2   | 42     |
|    | 12 | -55.7   | 61.3    | -107.8  | 98.3    | 167.9    | 32.8   | 146     | 6.3      |          |        |        |        |
| 7  | 0  | 58.7    | 728     | -3068.5 | -263.4  | -353.1   | -398.7 | 2696.7  | 95.2     | -202.9   | -28.2  | -5.6   | 45.7   |
|    | 12 | 22.5    | 13.1    | -66.7   | 42      | -269.3   | -467.2 | -208.8  | 2168     |          |        |        |        |
| 8  | 0  | 17.8    | -3857.2 | -214.1  | 221.1   | -3488.2  | 83.3   | -162.9  | -260.7   | 3.5      | -46.1  | 32.8   | -73.8  |
|    | 12 | -1      | -53     | 397.9   | 227.4   | -961.9   | 286.9  | -229    | 364.3    |          |        |        |        |
| 9  | 0  | -7050.2 | 4.7     | 168.8   | -40.4   | 51.1     | -143.8 | -141.4  | -56.7    | -159.6   | 79.5   | -8.1   | 23.2   |
|    | 12 | 2.3     | -51.7   | 54.7    | 186.3   | 6.3      | 44.6   | -91.9   | 54.7     |          |        |        |        |
| 10 | 0  | -329.7  | -636.1  | 207.9   | 115     | -178.2   | -103.3 | 65      | -120.6   | -4683.1  | -975.1 | -77.1  | 85.8   |
|    | 12 | 53.5    | 42      | 184.9   | -51.3   | 355.2    | -112.8 | -321.5  | 79.8     |          |        |        |        |
| 11 | 0  | 14.4    | 32.5    | 1394.3  | -151.4  | 48.1     | -35.6  | 293.8   | 4278.1   | -118     | 173.6  | 21     | -36.7  |
|    | 12 | -7.3    | 37.1    | 975.2   | 253.8   | 213.2    | 476.8  | -1456   | -23.3    |          |        |        |        |
| 12 | 0  | 255.3   | -5356.6 | 89      | 351     | 429.9    | 175.9  | 381.5   | 39.8     | -315.5   | 43.2   | 57.3   | 196    |
|    | 12 | 9.8     | 33.2    | 1035    | 92.4    | 1186     | 31.7   | 145.9   | -27.5    |          |        |        |        |
| 13 | 0  | -2680.3 | 260.6   | 3250.2  | -8.1    | -326.9   | 12     | -191.2  | 237.9    | -30.4    | -41.6  | 28.7   | 10.7   |
|    | 12 | -20.2   | 11.9    | -456.6  | 146.7   | 50       | -546   | -1884.2 | 295.5    |          |        |        |        |
| 14 | 0  | 144.6   | -204.4  | 166.4   | 6474.3  | -373.7   | 49     | 117     | -49      | -2.2     | 9.8    | -7.6   | -70.1  |
|    | 12 | 88.3    | -27.5   | 201.1   | 102.2   | 58.5     | 34     | 95.8    | 36.3     |          |        |        |        |
| 15 | 0  | 953.3   | 23.7    | 296.6   | 91.9    | 50.3     | -338.5 | 6794.1  | 7        | 0.2      | 77.2   | 27.9   | 96.8   |
|    | 12 | 99.2    | 11.7    | -48.9   | 7.5     | 138.1    | 39.4   | 426.9   | 26       |          |        |        |        |
| 16 | 0  | -1.2    | -408.9  | -306.6  | -234.8  | 89       | -555.9 | 22      | 106.6    | 98       | 11.5   | -4.6   | -15.3  |
|    | 12 | -52.1   | 58.4    | 11645.8 | -104.3  | -52      | -66.8  | 162.1   | -38.7    |          |        |        |        |
| 17 | 0  | -2503   | -167.9  | -131.7  | 159.9   | 75.5     | -314.6 | 3624.7  | -25.9    | 229.1    | -144.9 | 30.6   | -191.3 |
|    | 12 | 51.7    | -35.8   | -69.2   | -178.6  | -196.6   | 202.2  | 165.9   | -165.6   |          |        |        |        |
| 18 | 0  | 14.1    | -8789.4 | -46     | 49.4    | -14.4    | 112.4  | -30.6   | 82.4     | -160.4   | 21.3   | 74     | 50.2   |
|    | 12 | 82.7    | 53.1    | 59.1    | -15.9   | -285.9   | 91.9   | 145.8   | 240.5    |          |        |        |        |
| 19 | 0  | -331.2  | -1859.9 | 19.9    | 343.1   | -63.4    | 3389.2 | 541.5   | 485.4    | 129.3    | 214.5  | 29.4   | 19     |
|    | 12 | 33.4    | -15.1   | -1094.3 | -528.6  | 266.3    | -185.5 | -478.7  | -185.8   |          |        |        |        |
| 20 | 0  | -123    | 3966.6  | -17.5   | 4279.8  | -15.7    | -47.5  | 239.8   | 76.4     | 9.8      | -3.7   | -2     | 29.3   |
|    | 12 | 49.4    | -109.1  | 182.2   | -47.5   | 158      | -80.9  | 542     | -328.9   |          |        |        |        |
| 21 | 0  | -213.5  | 492.4   | -408.5  | 2968.7  | 99.5     | -651.8 | -379.6  | 728.1    | -230.4   | 192.2  | 84.5   | -130   |
|    | 12 | -20.3   | -78.7   | -502.3  | -74     | -11514.1 | -172.9 | -882    | -26.7    |          |        |        |        |
| 22 | 0  | -401.2  | 120.3   | 23.1    | -233.7  | 3403     | -332.7 | 47.4    | 23.8     | 190.7    | -735.1 | 131.1  | -83    |

|    |    |         |         |        |        |         |         |         |        |       |          |        |        |
|----|----|---------|---------|--------|--------|---------|---------|---------|--------|-------|----------|--------|--------|
|    | 12 | -30.7   | -147.3  | -263.5 | 306.2  | -54.8   | 11629.9 | 308.2   | -528   |       |          |        |        |
| 23 | 0  | -308.5  | 1931.8  | 659.7  | 126    | -4247.4 | -208    | 347.9   | 207.5  | 76    | -238.6   | 184    | -58    |
|    | 12 | -52.3   | -23.7   | 194.3  | 165.8  | 46.6    | 400.7   | -1299.6 | -120.9 |       |          |        |        |
| 24 | 0  | -3467.8 | -4196.2 | -49    | -28.7  | -107.2  | -172.7  | -355.4  | 210.5  | -72   | -118.8   | 41.6   | -159.4 |
|    | 12 | 77.6    | -97.3   | -136.3 | 47.9   | -402.7  | 132.7   | -101.5  | -288.8 |       |          |        |        |
| 25 | 0  | -71     | 517.2   | -413.7 | -1414  | -374.7  | -86.5   | -285.7  | 322.6  | -473  | 351.9    | 436.9  | -71.8  |
|    | 12 | 62.9    | -4541.1 | -314.8 | 271.7  | 25.2    | 250.9   | -436.7  | 500.3  |       |          |        |        |
| 26 | 0  | 94.6    | -66.5   | 311.2  | 3960.3 | 3911.6  | 528.5   | 309.7   | -213.5 | 74.9  | 52.6     | 28.3   | -52.8  |
|    | 12 | -48.6   | -101.2  | -18.1  | -283   | 671.4   | -38.7   | 282.7   | 39.6   |       |          |        |        |
| 27 | 0  | 692.1   | -48.2   | -87.8  | 112.4  | 43.7    | 7489    | -30.8   | -63.6  | 87.2  | -13.4    | 59.8   | 25.1   |
|    | 12 | -82.8   | 4.7     | -162   | 26.4   | -94.8   | -86     | 190.6   | -353.1 |       |          |        |        |
| 28 | 0  | -4033.6 | 182.3   | -237.9 | -49    | -181.6  | 453.2   | -508.9  | -56.1  | -86.6 | -51.4    | 90.5   | -4.4   |
|    | 12 | -19.8   | 53.4    | -97.7  | -284.1 | 180.3   | 263.8   | 3747    | 0.8    |       |          |        |        |
| 29 | 0  | 1019.4  | -1791.3 | 15.8   | 252.2  | 83      | -3434.4 | 132.8   | -87.3  | 93.9  | 199.1    | -149.5 | -30.6  |
|    | 12 | 14.1    | 31.4    | -17.3  | 828.3  | 76.8    | -403    | 21.1    | 541.3  |       |          |        |        |
| 30 | 0  | 7.4     | -0.4    | 10.4   | 165.8  | -24.5   | 49.6    | 54      | -88.5  | 96.7  | -12001.5 | -61.2  | 2.8    |
|    | 12 | 43.7    | -39.3   | 292.9  | -87.4  | -132.9  | -36.3   | 538.9   | 45.2   |       |          |        |        |
| 31 | 0  | 83.6    | -75.6   | 37.8   | -16.5  | 10805.3 | 65.4    | -23.6   | 17     | 17.1  | -73.1    | 21.4   | 10.9   |
|    | 12 | 64      | 19.7    | 46     | 84.1   | -112.2  | -55.3   | 244.2   | 134.2  |       |          |        |        |

Table 4.A.20 – Interleave VQ codebook 1 for core coder (WINDOW TYPE : LONG)

| filename | Contents | mode      | number of elements | number of vectors |
|----------|----------|-----------|--------------------|-------------------|
| cdmdct_1 | MDCT     | core LONG | 20                 | 32                |

| VN | EN | 0       | 1       | 2      | 3       | 4       | 5        | 6      | 7      | 8      | 9      | 10     | 11     |
|----|----|---------|---------|--------|---------|---------|----------|--------|--------|--------|--------|--------|--------|
| 0  | 0  | 235.3   | 67.3    | 50.9   | -70.8   | 141     | -11826.5 | -180.2 | -63.5  | 101.1  | -67.7  | -52.8  | -26.3  |
|    | 12 | 3.1     | 9.5     | -213.3 | 91.1    | -19.6   | -208.5   | 185.1  | 155.8  |        |        |        |        |
| 1  | 0  | -2551.4 | -264.8  | 68.9   | -3066.5 | -340.7  | 139.4    | 130.4  | 26.9   | 60.2   | 35.7   | -4.1   | 74.7   |
|    | 12 | -35.7   | 10.4    | -215.4 | -28.2   | 182.3   | 258.3    | 272.9  | -157.2 |        |        |        |        |
| 2  | 0  | 625.1   | -205.8  | 260.9  | -388.8  | -46.3   | -37.3    | 40.9   | -162.9 | -59.8  | -66.2  | -57.3  | -204.2 |
|    | 12 | -46.2   | 54.2    | 121.4  | -154.5  | -12125  | 83.4     | 421.2  | 168.3  |        |        |        |        |
| 3  | 0  | -209.8  | 227     | -229.9 | -47     | 6981.7  | -61.2    | 43.9   | -60.2  | -68.3  | 50.4   | 34.3   | -0.2   |
|    | 12 | 16.8    | 10.1    | -144.3 | -9.7    | -127.6  | -128.7   | -320.2 | 195.7  |        |        |        |        |
| 4  | 0  | 3358.8  | 295.2   | -171.2 | -205.5  | -2008.6 | -196.7   | 25.4   | -112.9 | -50.6  | -304.3 | -65    | -49.7  |
|    | 12 | -54     | -35.7   | -2190  | 54.1    | -279    | 491.8    | 37.6   | -327.2 |        |        |        |        |
| 5  | 0  | -11.3   | -341.2  | 158.5  | 133.3   | -158.6  | 2.8      | -11.8  | -38.9  | 6211.1 | -519.8 | -58.8  | -26.8  |
|    | 12 | 17.3    | -57.9   | 341.8  | -294.8  | 20.2    | 38.5     | 378.2  | 101.3  |        |        |        |        |
| 6  | 0  | 66      | 85.5    | 1894.5 | 3096.2  | -1994.1 | 40       | -103.1 | -369.4 | -93.3  | -65.3  | 151.1  | 34     |
|    | 12 | 12.1    | -27.7   | -488.9 | -136.9  | 39.4    | 1058.2   | 385.7  | 370.2  |        |        |        |        |
| 7  | 0  | -8      | -8.6    | -44.9  | 18.7    | -73.3   | 50.5     | -4     | -61.6  | -14.4  | 1.7    | 5132.2 | -25.8  |
|    | 12 | 147.1   | 42.4    | 18.2   | 2672.9  | 182.8   | -251.4   | -329.7 | 378.2  |        |        |        |        |
| 8  | 0  | -4.6    | 165.2   | -55.1  | 9383.7  | 117.1   | -95.3    | -115.3 | 18.8   | -0.1   | 59.2   | 70.4   | -45.5  |
|    | 12 | 21.6    | -8.3    | 29.7   | -51.5   | 294.9   | -117.1   | -209.5 | -44.2  |        |        |        |        |
| 9  | 0  | -214.8  | -80.8   | 47.6   | -433.9  | 371.7   | 5001.4   | -330.3 | 27.5   | 37.4   | -44    | -255.9 | -153.2 |
|    | 12 | 63.1    | -12.2   | -152.5 | 552.9   | 216.9   | -145     | -903.1 | 512.9  |        |        |        |        |
| 10 | 0  | 1342.8  | -240.9  | -920.2 | 98.8    | -132.1  | -192.2   | -158.8 | 3771.8 | -21.2  | -158.4 | -48.4  | 107.4  |
|    | 12 | 60.7    | -10.7   | -13.8  | -227.2  | 240.3   | 478.1    | 319.2  | 496.1  |        |        |        |        |
| 11 | 0  | 4.8     | -5600.1 | 492    | 159.7   | -55.6   | -26.9    | -318.6 | 121    | 149.8  | 91.1   | 126.5  | -130.8 |
|    | 12 | -63.2   | 89.2    | -228.1 | -49     | 132.7   | -101.9   | 95.8   | -197.4 |        |        |        |        |
| 12 | 0  | 356.8   | 2654.8  | -358.9 | 114     | 103.3   | 85.6     | 22.1   | -37.8  | 106.3  | 1.2    | -15    | -49.8  |
|    | 12 | -72.9   | 170.8   | -66    | 36.2    | 2617.9  | -162.5   | 503.6  | 1981.6 |        |        |        |        |
| 13 | 0  | -9.4    | 127.7   | 23.1   | -66.9   | -122.5  | -198.8   | 101.7  | 56.8   | 41.2   | 89.5   | -60.9  | 30.3   |
|    | 12 | 5664.6  | 159.7   | 213.8  | -162.9  | -59.2   | 84.3     | -33.3  | -259.1 |        |        |        |        |
| 14 | 0  | -103.3  | 42.4    | -1     | -180.8  | 44.4    | 743.2    | 4688.1 | 28.2   | -81.9  | 51.3   | 69.5   | 61.2   |

|    |    |          |         |         |         |         |         |          |         |        |        |        |        |
|----|----|----------|---------|---------|---------|---------|---------|----------|---------|--------|--------|--------|--------|
|    | 12 | -106.5   | -11.6   | -37     | -126.5  | -27.4   | 164.2   | -136.7   | -88.4   |        |        |        |        |
| 15 | 0  | -1042.2  | -313.7  | -154.2  | -414.9  | 650.3   | -2755.9 | -161.8   | 364.9   | 103.8  | 27.5   | -94.9  | -172.6 |
|    | 12 | -75.1    | 108.1   | -7700   | -310.2  | 161.1   | -56.1   | -341.6   | -34.1   |        |        |        |        |
| 16 | 0  | -54.2    | -3008.6 | -3909.4 | -85.6   | -62.9   | -56.7   | 108.9    | 40      | 267.7  | -151.7 | 144.4  | -63.7  |
|    | 12 | -221.4   | -67.8   | 193.4   | 55.7    | 572.8   | -265.8  | -565.6   | -407.7  |        |        |        |        |
| 17 | 0  | -63.3    | 231.6   | -3415.5 | 601     | 575.7   | 1192.1  | -742.8   | 47      | -427.7 | 29.6   | 58.8   | 46.8   |
|    | 12 | -87.6    | 101.1   | -843.8  | 314.8   | -1669.2 | -178.3  | -315.2   | -230.1  |        |        |        |        |
| 18 | 0  | 558      | -233.4  | 130.9   | 5.3     | 42.9    | -227.1  | 63.8     | -7070.4 | -35.4  | 27.3   | -70.9  | -14.2  |
|    | 12 | 30.6     | 127.3   | 499.7   | 11.2    | 343.1   | 789.6   | 860.6    | 325.5   |        |        |        |        |
| 19 | 0  | -3560.6  | 3198.6  | -27.2   | 19      | 3.2     | -363.7  | 52.9     | 257.5   | -150.4 | 128.9  | -159.8 | -58.2  |
|    | 12 | 1.2      | 17.3    | -15.3   | 28.7    | -376.5  | -186.5  | 121.6    | -72.7   |        |        |        |        |
| 20 | 0  | -1971.3  | -21.1   | -60.6   | 3407    | -171.8  | 35.7    | -15.5    | 65.3    | 173.8  | -58.9  | 2.5    | 70.6   |
|    | 12 | -48.5    | -24.8   | 306.4   | 156.2   | 476.7   | -327.6  | -2828.6  | -905.5  |        |        |        |        |
| 21 | 0  | -580.4   | -214.9  | -1765.9 | 76.6    | -4029.4 | -5.1    | -172.3   | -113    | 51.4   | -152   | 15     | -37.4  |
|    | 12 | 27.1     | -56.2   | 266.6   | -408    | 589.2   | 298.9   | 941.4    | -136    |        |        |        |        |
| 22 | 0  | 297.5    | -16.5   | 62.7    | -49.7   | 3.1     | 97.1    | -11428.2 | 64.4    | 11.5   | 64     | 32.8   | -114.3 |
|    | 12 | 2.9      | 10      | -32.2   | 117.6   | -48.1   | 68.9    | 293.1    | 14.3    |        |        |        |        |
| 23 | 0  | -49.9    | 2192.6  | 578.3   | -4411.8 | -44.1   | 134.5   | -64.7    | 55.5    | -10.8  | -3.7   | -90.7  | 3.1    |
|    | 12 | -46.5    | 92.6    | 431     | 139.3   | -407.1  | 15      | -167.1   | -108.2  |        |        |        |        |
| 24 | 0  | 28.8     | -7.1    | 121     | -63.5   | 79.6    | -35.6   | -14      | 45.6    | -201.3 | 5649.4 | -13.3  | -12.9  |
|    | 12 | 43.6     | 26.2    | 155.9   | -306.9  | -134.7  | 22.8    | 556      | -21     |        |        |        |        |
| 25 | 0  | 6.7      | -62.4   | -23.1   | 790.1   | 199.2   | -93.1   | 72.3     | -203.9  | -26.1  | -374.6 | -437.6 | 219.8  |
|    | 12 | -139.7   | -9142.4 | 30.3    | -176.5  | 119.9   | -146.4  | -595     | -391.8  |        |        |        |        |
| 26 | 0  | 2816.7   | 122.3   | 3753.5  | -33.5   | 242.4   | 245.3   | 233.1    | -340    | 17.7   | -215.5 | 40.5   | -58.3  |
|    | 12 | 6.6      | 29.7    | -254.2  | -2.6    | 675.4   | -347.6  | 506.2    | 227.3   |        |        |        |        |
| 27 | 0  | -682.7   | -375.4  | 5715.8  | 196.7   | 236.8   | 214     | 21.2     | 229.6   | -232.8 | -41.2  | -54.1  | 2.8    |
|    | 12 | -74.8    | -148.1  | -103.5  | -55.2   | 298.6   | -263.6  | 393.1    | -462.4  |        |        |        |        |
| 28 | 0  | -10807.7 | 112.1   | -283.5  | 173.9   | 368.5   | 172.3   | 46.9     | 208     | 51.4   | 226.4  | 72.3   | 108.1  |
|    | 12 | -76.5    | 37      | 543.5   | -277.6  | -291    | -178.8  | -246     | 228.4   |        |        |        |        |
| 29 | 0  | -58.2    | -42.1   | -36.1   | -25.6   | -78.7   | 315.2   | 193.4    | 11.3    | 47.5   | 46     | 44.7   | 12181  |
|    | 12 | -7.5     | -10     | -31.3   | 106     | 255.7   | 78.9    | 571.9    | -518.5  |        |        |        |        |
| 30 | 0  | -102.9   | 20.3    | -9260.7 | 53.2    | 55.9    | 133.6   | 125.5    | 202.7   | -66.6  | -62.6  | -7.3   | 2.2    |
|    | 12 | -0.6     | 56.8    | -6.7    | -14.7   | -340.4  | -160.4  | -419.5   | -241.2  |        |        |        |        |
| 31 | 0  | 5816.2   | 60.8    | -128.6  | -76.2   | 350     | 99.9    | -81.6    | 133.1   | -103.3 | -19.4  | 56     | -81    |
|    | 12 | -87.3    | -14.7   | 276.2   | 194.7   | -148.2  | 129.3   | -495.4   | -270.1  |        |        |        |        |

Table 4.A.21 – Inteleave VQ codebook 0 for core coder (WINDOW TYPE : SHORT)

| filename | contents | mode       | number of elements | number of vectors |
|----------|----------|------------|--------------------|-------------------|
| cdmdct_2 | MDCT     | core SHORT | 17                 | 32                |

| VN | EN | 0      | 1       | 2       | 3       | 4       | 5      | 6       | 7       | 8       | 9      | 10     | 11     |
|----|----|--------|---------|---------|---------|---------|--------|---------|---------|---------|--------|--------|--------|
| 0  | 0  | 16384  | -534    | 296.2   | 242.6   | -1011.7 | -156.3 | -25.8   | -720    | 135.3   | -288.9 | -197.1 | 339.2  |
|    | 12 | 12.6   | 484.7   | 68.3    | 1615.2  | -524.1  |        |         |         |         |        |        |        |
| 1  | 0  | -7.6   | -264.3  | -165    | 453.1   | -8874   | -37.8  | 918.3   | -257.6  | 183.8   | -144.9 | 115.8  | 63.6   |
|    | 12 | 352.4  | 137.9   | 322.3   | -630.2  | -5343   |        |         |         |         |        |        |        |
| 2  | 0  | 15.7   | 337.7   | -1005.3 | 4231.3  | -2785.4 | 214    | -3.1    | 311.3   | -296.8  | 99.8   | -496.3 | 186.7  |
|    | 12 | -33.9  | -685.2  | 642.6   | 633.4   | 4888.4  |        |         |         |         |        |        |        |
| 3  | 0  | 149.2  | -252.5  | -238.4  | -191.4  | 45.6    | 154.2  | -498.2  | -4264.6 | -1516.9 | 985.7  | -939.2 | -419.5 |
|    | 12 | 1279.1 | 581.1   | -1809   | -951.5  | 1015.2  |        |         |         |         |        |        |        |
| 4  | 0  | 239    | -72     | -104.5  | 882.2   | -73.4   | 24.2   | -187.9  | 141.5   | -9608   | 337.9  | 253.2  | 433.7  |
|    | 12 | -533.1 | -1421.2 | -2313.7 | 877.7   | -1429.2 |        |         |         |         |        |        |        |
| 5  | 0  | 233.8  | -356.3  | -704.4  | 218.4   | -766.8  | -226.7 | -7255.4 | -139.7  | 229.3   | -439.2 | -523   | 230.3  |
|    | 12 | -173.3 | 1689    | -806.6  | 2720    | -1824.9 |        |         |         |         |        |        |        |
| 6  | 0  | -651.9 | 677.9   | -425    | -7670.7 | -637.7  | 168.6  | -751.9  | -37.3   | -471.9  | 193    | 137.8  | 293    |



|    |    |          |         |         |         |         |         |         |         |         |         |         |         |
|----|----|----------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
|    | 12 | 367.1    | -910.5  | -542    | -4284.2 | 5243.1  |         |         |         |         |         |         |         |
| 7  | 0  | -45.2    | 609     | -135.9  | -740.3  | 2550.1  | 3678.4  | 1197    | 774.5   | -1783.1 | 355.6   | -191.8  | -292.4  |
|    | 12 | -293.4   | -196.9  | 199.3   | -2391.9 | 429.2   |         |         |         |         |         |         |         |
| 8  | 0  | 819.6    | -838.5  | -374.1  | -704.7  | -329.2  | 310.2   | 404     | -204.9  | 94.1    | 1035.5  | -487.3  | 1144.9  |
|    | 12 | -0.1     | -8832   | 1367.5  | 120.2   | -1549.2 |         |         |         |         |         |         |         |
| 9  | 0  | -274.8   | -263.5  | 590.7   | 723.8   | 420.4   | -272.4  | 346.6   | -478.9  | 3294.9  | 1890.3  | -1243.6 | -200.1  |
|    | 12 | 96.6     | 232.9   | -3382.5 | 953.2   | -1167.4 |         |         |         |         |         |         |         |
| 10 | 0  | -113.3   | -594.2  | -9768.3 | 80.5    | -315.6  | 12.5    | 281.2   | -97.8   | -37.7   | 427.1   | 310.7   | -30.2   |
|    | 12 | -90.6    | 92.8    | 909.2   | -684.6  | -954.4  |         |         |         |         |         |         |         |
| 11 | 0  | -92.3    | 141     | 576.4   | -24.4   | 56.3    | -1025.6 | 115.2   | -8990.2 | 176.6   | 144.3   | 86.6    | -190.4  |
|    | 12 | -1808.1  | -673    | 452.2   | -29     | 3351.3  |         |         |         |         |         |         |         |
| 12 | 0  | 152.2    | 2705.2  | 2093.3  | -1395.5 | 27.3    | -76.7   | 835.9   | -54.8   | -363.6  | 690.4   | -316.8  | -151.5  |
|    | 12 | -539.3   | 492.3   | 2115    | 777.4   | -3297.1 |         |         |         |         |         |         |         |
| 13 | 0  | -266.9   | -272.8  | -464.7  | 262.5   | 0.4     | 11.9    | 441.1   | -267.7  | -33.5   | 361.3   | 4513.6  | -476.3  |
|    | 12 | 12.5     | -138.8  | -965.3  | 2941    | -760.6  |         |         |         |         |         |         |         |
| 14 | 0  | -5805.6  | 289.6   | 469.5   | 1712.9  | -169.3  | 150.8   | -1835.6 | 734.9   | 90      | 551.8   | -236.2  | -323.6  |
|    | 12 | -172.4   | -5589.5 | -2574.5 | -1050.4 | 1301.2  |         |         |         |         |         |         |         |
| 15 | 0  | -3543.8  | 329.1   | -598.6  | -274.4  | 1067    | 973.2   | 856.7   | 549.2   | -31.2   | -82.9   | -907.1  | 671.3   |
|    | 12 | -115.7   | 2434.7  | 2131.6  | -931.5  | -1369.2 |         |         |         |         |         |         |         |
| 16 | 0  | -355.6   | 242.4   | -267.9  | 182.5   | 177.8   | -134    | 85.2    | 806.1   | 117.7   | -121.4  | -738.8  | -6574   |
|    | 12 | -813.8   | -615.8  | -632.4  | 390.3   | 1416.5  |         |         |         |         |         |         |         |
| 17 | 0  | -18.8    | 247.7   | 2109.5  | -2071.1 | -3397   | 1029.1  | 474.4   | -52.4   | 620.7   | 63.1    | 518.2   | -206.5  |
|    | 12 | 288.5    | 1795.3  | -2903   | -2856.7 | 1327.6  |         |         |         |         |         |         |         |
| 18 | 0  | 260.8    | 496.6   | 695.4   | -1262.8 | 409     | -521.2  | -2073   | 1709    | 1929.4  | 69.5    | -1172.1 | -1298.6 |
|    | 12 | 3453.2   | -77.8   | -1448.3 | 1117.8  | 1309.2  |         |         |         |         |         |         |         |
| 19 | 0  | 3500.7   | -3781.1 | 337.1   | 350.8   | 620.7   | -166.5  | 507.2   | -305    | -15.5   | -51     | -343.1  | 310     |
|    | 12 | 21.2     | 1445.3  | 1738.2  | -576    | 1721.7  |         |         |         |         |         |         |         |
| 20 | 0  | -32.2    | -54.2   | -1286.9 | 76.5    | -1713.8 | -455.4  | 3055.5  | 1436.6  | 345.3   | -55     | 94.7    | -580.7  |
|    | 12 | -39.9    | 1640.6  | 2128    | 625.4   | -2599.5 |         |         |         |         |         |         |         |
| 21 | 0  | -178.5   | -715.7  | -315.2  | -671.3  | -1472.1 | 4815.1  | -753.8  | 83.8    | 317.6   | 136     | 559.3   | -192.3  |
|    | 12 | 690      | -2273.6 | -2537.7 | 1253    | -3324.6 |         |         |         |         |         |         |         |
| 22 | 0  | 591      | -267.2  | -299.1  | -485.2  | -423.5  | -38.2   | 674.1   | -2357.8 | 1102.6  | -3306.7 | -757.9  | -302.3  |
|    | 12 | 564.8    | 2143.9  | -754.7  | 2681    | 611.6   |         |         |         |         |         |         |         |
| 23 | 0  | -525.9   | -124.6  | -189.3  | 111.3   | 427.3   | -505.8  | -420.1  | 127.8   | -364.7  | -7157.5 | -47.4   | 226.7   |
|    | 12 | -380.5   | 195.8   | -1269   | -18.3   | -204.7  |         |         |         |         |         |         |         |
| 24 | 0  | -492.2   | -4261.3 | 1940.7  | 1797.6  | 341.4   | 601.3   | 611.3   | 27.5    | 252.7   | 151.2   | 390.8   | -54.1   |
|    | 12 | -1216.6  | 2894.2  | -3176   | -1731.3 | -1433.6 |         |         |         |         |         |         |         |
| 25 | 0  | -12834.3 | -93.1   | -374.5  | 287     | -1899.4 | 35.7    | -715.3  | -307.9  | 10.6    | 78.9    | 1096.8  | 34.3    |
|    | 12 | 103.6    | 2076.4  | 615.2   | 1465.3  | 758     |         |         |         |         |         |         |         |
| 26 | 0  | 1855.5   | 683.3   | -272.6  | -279.2  | 818.5   | -796.5  | 85.7    | 57.3    | 161.8   | -7.5    | 271.7   | -197.5  |
|    | 12 | -6743    | 2581.4  | -1422.1 | 175     | -1154.1 |         |         |         |         |         |         |         |
| 27 | 0  | -129.1   | -379.4  | -4225.6 | -2308.9 | 247.6   | 246.7   | -478.2  | 234.3   | 327.2   | -193.4  | -380.8  | 165.5   |
|    | 12 | -322.4   | -1003.3 | -706.7  | 135.4   | 2359.3  |         |         |         |         |         |         |         |
| 28 | 0  | -8970.6  | 625.4   | 316.8   | -108.1  | 625.2   | 1014.1  | 934.6   | -815.1  | -137.6  | -365.9  | -392.8  | -204.2  |
|    | 12 | -130.7   | -195.9  | 1439.7  | 3056.3  | 603.4   |         |         |         |         |         |         |         |
| 29 | 0  | -90      | 1303.7  | -1332.6 | 167.8   | -600.4  | 2072    | -1718.3 | 213.6   | 125     | -37.1   | 21.4    | 106.7   |
|    | 12 | -3498.4  | 2942.1  | -421.4  | 1483    | 5828.9  |         |         |         |         |         |         |         |
| 30 | 0  | 478.7    | 7218.4  | 129.5   | 141.4   | 732.8   | 36.2    | 296.8   | -233.7  | 72.8    | 121.4   | -214.2  | -431.6  |
|    | 12 | 2.4      | 741.3   | 1486.1  | 1177.2  | 1714.4  |         |         |         |         |         |         |         |
| 31 | 0  | -4432.8  | -852.9  | -2584.9 | -89.8   | -780.6  | -722    | -88.5   | -1466.4 | 32.9    | 161.5   | -65.6   | -642.8  |
|    | 12 | 207.1    | 735.6   | -5723.4 | -841.8  | -361.4  |         |         |         |         |         |         |         |

Table 4.A.22 – Inter leave VQ codebook 1 for core coder (WINDOW TYPE : SHORT)

| filename | contents | mode       | number of elements | number of vectors |
|----------|----------|------------|--------------------|-------------------|
| cdmdct_3 | MDCT     | core SHORT | 17                 | 32                |

| VN | EN | 0       | 1        | 2       | 3        | 4       | 5       | 6       | 7       | 8       | 9       | 10      | 11      |
|----|----|---------|----------|---------|----------|---------|---------|---------|---------|---------|---------|---------|---------|
| 0  | 0  | 210.8   | -13952.3 | -184.9  | -354     | 562.2   | 506.7   | -239    | -758.3  | -199.4  | 205.1   | 32.8    | -1105.2 |
|    | 12 | -674.6  | 577.5    | -224.6  | -325.9   | -1149.9 |         |         |         |         |         |         |         |
| 1  | 0  | -413.1  | 83.5     | -336.7  | -1688    | -1187.9 | -3898.7 | -171.5  | 897.3   | -485.8  | 695.3   | 643.6   | -319.2  |
|    | 12 | 221.8   | -3611.9  | 3212    | -1658.3  | -2700.4 |         |         |         |         |         |         |         |
| 2  | 0  | -331.7  | 418.3    | -46.9   | 126      | 865.1   | 463.8   | -6325.3 | 271.8   | 176.6   | -314.8  | -18.7   | -555.5  |
|    | 12 | 795.3   | -474.9   | 628.7   | -2206.3  | 55.7    |         |         |         |         |         |         |         |
| 3  | 0  | -4269   | -392.4   | 3219.7  | -539.8   | 14.4    | -774.7  | -262.6  | -302.9  | 46.9    | 70.7    | -63.6   | 120.8   |
|    | 12 | 59.4    | 811.7    | 4138.9  | -9.2     | 112.3   |         |         |         |         |         |         |         |
| 4  | 0  | 767.6   | -710.4   | -7343.7 | 108.3    | 660.1   | -256.2  | -137.4  | -77.8   | -77.4   | 276.4   | -25     | 325.9   |
|    | 12 | 46.3    | -578.7   | 2070.7  | 1799.3   | 1214.2  |         |         |         |         |         |         |         |
| 5  | 0  | -141.8  | 3251.9   | -1072.7 | 343.9    | -593.7  | -29     | 191.1   | 1221.3  | -369.9  | -22.8   | -530.7  | 1110.2  |
|    | 12 | -1079.3 | -47.4    | -3613.6 | -3544.2  | 1237.2  |         |         |         |         |         |         |         |
| 6  | 0  | -241    | 877.9    | -36.2   | 125.9    | 253.3   | 8498.6  | -447.2  | -525.9  | 685.6   | 502.4   | -196.4  | 1546.3  |
|    | 12 | 755.3   | 382      | 305.2   | -797.4   | -1413   |         |         |         |         |         |         |         |
| 7  | 0  | 849.4   | -216.3   | 375.9   | -12034.9 | -438    | -444.1  | 250.6   | 80.4    | 1147.9  | 193.8   | 413.3   | 66.6    |
|    | 12 | -56.1   | 1401     | -1495.1 | -445.3   | -475.5  |         |         |         |         |         |         |         |
| 8  | 0  | 3884.2  | 4137.2   | 298.4   | 3.6      | 326     | 225.8   | 367.8   | -384.3  | 147     | -7.4    | 358     | -281.7  |
|    | 12 | 447.3   | 423.2    | 1928.4  | -3456.7  | -2995.4 |         |         |         |         |         |         |         |
| 9  | 0  | 482     | 292.9    | -1031.8 | 47.7     | -3246.4 | 248.2   | 666.6   | -548.3  | -470.8  | 430.4   | -840.3  | -332    |
|    | 12 | 156.3   | -5991.7  | -387.9  | -641.7   | -316.6  |         |         |         |         |         |         |         |
| 10 | 0  | -231.9  | -198.8   | -307.8  | 452.8    | 12.6    | -418.5  | 88.7    | -41     | 358.8   | -388.4  | -428.7  | 4450.3  |
|    | 12 | 384.5   | 737.2    | -4.5    | 4933.9   | 1082.1  |         |         |         |         |         |         |         |
| 11 | 0  | 5421.3  | 346.1    | 127     | 3133.9   | -586.4  | 76.3    | 585.8   | -319    | 147     | -67     | -700.3  | 206.2   |
|    | 12 | 93.4    | -4386.2  | 443.2   | 682.7    | 2974    |         |         |         |         |         |         |         |
| 12 | 0  | -303    | -58.5    | -133.6  | 609.8    | -339.1  | -79.9   | 112.4   | 577.4   | 3997.3  | -635.8  | 275     | -193.2  |
|    | 12 | 573.9   | 1187.4   | 1662.3  | 1321.8   | 527.8   |         |         |         |         |         |         |         |
| 13 | 0  | 102.2   | -1280.8  | -539.2  | -4552.8  | -744.6  | 694.4   | -49     | 94.7    | -22     | -5.5    | -577.7  | -254.2  |
|    | 12 | 114.7   | -531.6   | -3841.1 | 900.7    | 3404.9  |         |         |         |         |         |         |         |
| 14 | 0  | -573.8  | 117.5    | -2      | 259.8    | -324.1  | -48.6   | -1858.6 | -3521.6 | 94.6    | -161.4  | 908.1   | -404    |
|    | 12 | -416.5  | 361.2    | -792    | -303.6   | -1569   |         |         |         |         |         |         |         |
| 15 | 0  | 128.5   | -2026.6  | 78.6    | -476.9   | -975.5  | -358.2  | -1236.6 | 3351.7  | -75     | 44.7    | 636.2   | 67.4    |
|    | 12 | 6.3     | 2449.2   | -374.2  | -126.5   | -3959.9 |         |         |         |         |         |         |         |
| 16 | 0  | -30.5   | -640.4   | 1339.6  | 1507.5   | -493.6  | -403.1  | 22.8    | -368.6  | -1413.1 | -2039.5 | -1884.3 | -1548.5 |
|    | 12 | -654.6  | 2228.7   | 545.6   | 1654.6   | -4512.5 |         |         |         |         |         |         |         |
| 17 | 0  | -9067.5 | -617.7   | -460.9  | 264.7    | 1549.6  | 54.8    | -82.9   | 204.5   | 22      | 90.4    | 120.8   | 421.4   |
|    | 12 | -36.4   | -445.4   | 524.9   | 811.8    | 1014.5  |         |         |         |         |         |         |         |
| 18 | 0  | 6359    | -1291.6  | -175.5  | -279.3   | 1966    | 183.8   | -1376.2 | -322.4  | 91.9    | 124.1   | 987.5   | -377.8  |
|    | 12 | 466     | 2217.7   | -1671.1 | -1155.7  | -88.9   |         |         |         |         |         |         |         |
| 19 | 0  | -108.4  | -170.5   | -2132.2 | 3481.2   | -207.9  | 1148.2  | -640.2  | 202.6   | 191     | 324     | 103.3   | -949.2  |
|    | 12 | 3242.5  | -123.1   | 129.5   | -2247.7  | 125.2   |         |         |         |         |         |         |         |
| 20 | 0  | 18.2    | -309.1   | -693.2  | -63.8    | 5058.6  | 1085.2  | 1284.9  | -674.8  | 395.6   | 253.9   | -412.8  | -290.1  |
|    | 12 | 37.7    | -810     | 1208.4  | -624.7   | -8533.2 |         |         |         |         |         |         |         |
| 21 | 0  | 66.8    | 103.7    | 136     | 296.5    | 275.6   | 86.2    | -373.8  | -81.1   | 5       | -404.3  | 6575.4  | -38.4   |
|    | 12 | 417.6   | 183.3    | 1674.7  | -342.1   | 1390.6  |         |         |         |         |         |         |         |
| 22 | 0  | 97.2    | -6.6     | 764.7   | 417.9    | 339.4   | -160.5  | -22     | -560.2  | 16.2    | 4354.3  | -714.5  | -186.2  |
|    | 12 | -792.8  | 5761.3   | -2868.7 | 426      | 264.2   |         |         |         |         |         |         |         |
| 23 | 0  | -267.9  | -69.1    | 165.6   | -718.6   | -826.1  | 2625.5  | 3479.2  | 511.1   | 528.3   | 172     | 259.7   | -380.7  |
|    | 12 | 507.4   | 278.7    | 803.8   | 373.8    | 870.7   |         |         |         |         |         |         |         |
| 24 | 0  | 598     | -1202.7  | 5005.8  | -1061.1  | 1246.4  | 27.8    | -6.3    | 358     | -147.8  | -604.8  | -50.5   | -175.6  |
|    | 12 | -131.6  | -2995.7  | -1675.1 | -1108.7  | 1871.5  |         |         |         |         |         |         |         |
| 25 | 0  | -2505.2 | -282.2   | 145.1   | 399.2    | 1227.1  | -1103.3 | 1568.9  | -203    | -65.6   | 104     | 427.1   | -180.8  |
|    | 12 | 102.3   | -109.7   | -2583.3 | 1662.1   | 1983.7  |         |         |         |         |         |         |         |
| 26 | 0  | 60      | 3970.3   | 4094.5  | 844.5    | -1026.5 | 114.6   | -830.2  | -38.5   | 568.2   | -555.8  | -49.7   | 242.7   |
|    | 12 | 853.8   | -1240.9  | 151.6   | 418.9    | 485.3   |         |         |         |         |         |         |         |

|    |    |        |        |         |        |        |        |        |         |         |        |        |        |
|----|----|--------|--------|---------|--------|--------|--------|--------|---------|---------|--------|--------|--------|
| 27 | 0  | 58.5   | 56.9   | 156     | 37.8   | -305.9 | -437.6 | 548.6  | 651.9   | -6621.1 | 284.5  | -295.6 | -333.5 |
|    | 12 | 1616.7 | 1988   | 1497.3  | -51.3  | 2570.1 |        |        |         |         |        |        |        |
| 28 | 0  | -275.8 | 8445.6 | -1183.1 | 849.5  | -337.1 | 48.2   | -161.5 | -270.5  | 431.1   | 24.4   | 507.4  | 17.2   |
|    | 12 | -571.8 | 580.6  | -2151.6 | 674.3  | -1505  |        |        |         |         |        |        |        |
| 29 | 0  | -43.4  | -348.7 | 2831.3  | 459.9  | -784.6 | 1878.2 | -188.4 | -91.6   | -819.7  | 1119.6 | 344.8  | 709.9  |
|    | 12 | 160.9  | 1134.7 | -777.8  | -431.3 | 63     |        |        |         |         |        |        |        |
| 30 | 0  | 120.6  | -343.5 | 15.8    | -248.1 | 143.5  | 528.9  | 761.2  | -6620.7 | 476.2   | -192.8 | 669.9  | 546.6  |
|    | 12 | 467.6  | -65.3  | 2492.2  | 916.1  | -781.6 |        |        |         |         |        |        |        |
| 31 | 0  | 1232.8 | -125.6 | -52.2   | -361.1 | 669.9  | -600.5 | 359.3  | 125     | -195.9  | -437.1 | -282.8 | 216.6  |
|    | 12 | 6506.1 | 1893.6 | -117.7  | -55.5  | 7375.8 |        |        |         |         |        |        |        |

Table 4.A.23 – Interleave VQ codebook 0 for scalable coder (WINDOW TYPE : LONG)

| Filename | contents | mode         | number of elements | number of vectors |
|----------|----------|--------------|--------------------|-------------------|
| scmdct_0 | MDCT     | enhance LONG | 28                 | 32                |

| VN | EN | 0       | 1       | 2       | 3       | 4       | 5       | 6      | 7       | 8       | 9       | 10      | 11     |
|----|----|---------|---------|---------|---------|---------|---------|--------|---------|---------|---------|---------|--------|
| 0  | 0  | 128.3   | -932.2  | -269.6  | -417.6  | -94.2   | 340.3   | -83.8  | -8130.1 | 144.4   | 249     | 156.5   | -315.9 |
|    | 12 | -95.4   | 404.7   | -662.3  | -283.8  | 94      | 208.8   | -97.2  | -56.2   | 97.9    | 396.1   | 182     | -135   |
|    | 24 | -38.2   | 306.2   | 91.3    | 106.5   |         |         |        |         |         |         |         |        |
| 1  | 0  | 201.2   | -4276   | 1196.9  | 392.1   | -5216.8 | 598.6   | 376.6  | 131.8   | 56.3    | -2237   | 894.1   | -832.2 |
|    | 12 | -256.5  | -506.3  | 481.2   | 321.3   | 393.9   | -91.1   | 103.8  | 452     | 170.5   | 235.3   | 269     | 459    |
|    | 24 | 166.1   | -664    | -17.3   | 339     |         |         |        |         |         |         |         |        |
| 2  | 0  | 2617    | 85.4    | -167.7  | -1804.1 | 4414.1  | 498.8   | 1432.6 | -1359.2 | 632.5   | -4937.2 | -415.2  | 174.3  |
|    | 12 | -1008.8 | 300     | 112.1   | 70      | 158     | -152.1  | 275.7  | -391.9  | 50.5    | 136     | -80     | -413.4 |
|    | 24 | 9       | 468.8   | 46.9    | 325.5   |         |         |        |         |         |         |         |        |
| 3  | 0  | 301.2   | 137.9   | -1784.8 | 45.9    | -823.1  | 4.9     | 144.8  | 154     | 322.9   | 156.4   | 51.7    | -162.2 |
|    | 12 | 275.2   | 319.9   | -146.4  | -2.5    | 221.7   | 35.6    | -16.2  | 688     | 148.6   | 348.9   | 8192    | -112.7 |
|    | 24 | -629.1  | 218.2   | 373.9   | 72.8    |         |         |        |         |         |         |         |        |
| 4  | 0  | -3885.1 | -446.4  | -519.5  | 566.8   | 662.8   | -1059.8 | 5754.8 | 251.3   | -416.1  | -798.9  | -728.2  | 697.9  |
|    | 12 | -679.3  | -473.4  | -192.3  | -159.6  | -119.9  | -81.8   | -66.8  | -37.8   | -143.7  | -33.9   | -20     | 40     |
|    | 24 | -93.6   | -47.5   | -192.2  | -117.1  |         |         |        |         |         |         |         |        |
| 5  | 0  | 754     | 275.7   | -345.7  | 129     | 259.6   | -803.1  | -85.8  | 374     | -198.4  | 323.5   | 470     | 1314.9 |
|    | 12 | 956.8   | -319.2  | -30.2   | 1585.1  | -12.5   | -123.9  | -8192  | -343.6  | -109.5  | -101.9  | 130     | -686.1 |
|    | 24 | 384     | -180.1  | -413.2  | -23.9   |         |         |        |         |         |         |         |        |
| 6  | 0  | 2396.1  | 355.6   | -207.3  | 1833.8  | -10.6   | -1321.6 | 7373   | -158    | 144.5   | -284.1  | 93.2    | -317   |
|    | 12 | -545.6  | -237.8  | -271.6  | -132.6  | 91.9    | -17.1   | -94    | 374     | 460.8   | -193.5  | -409.3  | 92.1   |
|    | 24 | -235.6  | 88.2    | 96.8    | -121.1  |         |         |        |         |         |         |         |        |
| 7  | 0  | -651.8  | -2127.6 | 655.4   | 7212.4  | -323.1  | -477.9  | 1955.8 | 303.2   | -420.9  | 1250.9  | -27.1   | -552.6 |
|    | 12 | 59.2    | -45.7   | 71.2    | -178.6  | 107.9   | -89.2   | 88.7   | -236.6  | -200.7  | 253.5   | -280.3  | 265.6  |
|    | 24 | -188    | -223.3  | -172.2  | 338.6   |         |         |        |         |         |         |         |        |
| 8  | 0  | -353.3  | -490.6  | 53.3    | 1929.3  | -386    | 7211.5  | 669.8  | -267    | -1117.5 | -390.9  | -264.1  | 583.9  |
|    | 12 | 1183.9  | 1311.3  | -79.4   | -333.2  | 24.9    | 159.3   | -210.4 | 116.9   | -194.3  | 52      | -82     | -105.1 |
|    | 24 | 239.1   | 134.9   | -69     | -116.3  |         |         |        |         |         |         |         |        |
| 9  | 0  | 14.3    | 348.3   | 868.3   | -63.9   | 321.9   | -2221.2 | 53.3   | 215.6   | -7890.4 | -3.5    | 233.1   | 163.6  |
|    | 12 | 499.4   | 131     | -0.4    | -131.6  | 13.7    | -230.4  | 221.1  | -61.7   | -189.2  | -252    | 117.3   | 426.7  |
|    | 24 | 89.1    | -323.3  | 209.5   | 16.4    |         |         |        |         |         |         |         |        |
| 10 | 0  | -221    | -173.4  | 59.4    | -119.5  | 116.1   | -19.6   | 45.3   | 96.6    | 152     | -19     | 195.5   | 141.4  |
|    | 12 | 2.7     | -32.6   | 125.8   | -104.3  | -8192   | 12.6    | -132.6 | 264.3   | -136.2  | -180.6  | 76.3    | -98.8  |
|    | 24 | 141.3   | -164.8  | -325.5  | 128.1   |         |         |        |         |         |         |         |        |
| 11 | 0  | -3222.5 | 377.5   | 4063.5  | 834.4   | 1458.5  | 2100.7  | 378.5  | 73.9    | -311.6  | -200.6  | -139.7  | -125.7 |
|    | 12 | 581.7   | 259.8   | 151.5   | 82      | -30.6   | -35.4   | -247.7 | 402     | 6035.4  | -166.7  | -100.8  | -75.2  |
|    | 24 | 284.6   | -53.3   | -318.5  | 232.8   |         |         |        |         |         |         |         |        |
| 12 | 0  | 1721.7  | -4502.2 | -852.4  | 1847.9  | 46.2    | -228.1  | 1950.1 | -221    | -671.4  | 1833.4  | -4362.2 | 1558.5 |
|    | 12 | -1244.3 | 20.6    | 34.9    | 294.9   | 144     | 202     | -90.3  | -813.7  | -142.5  | 117.9   | 260.1   | -37.7  |

|    |    |         |         |         |         |         |         |         |         |         |         |         |         |
|----|----|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
|    | 24 | 1205.4  | -476    | -1403.6 | 36.6    |         |         |         |         |         |         |         |         |
| 13 | 0  | -7301.4 | 1080.6  | -901.3  | 97.1    | -380.6  | 996.1   | -699.1  | 1045.8  | -514.7  | 920.1   | 283     | -1027.7 |
|    | 12 | -166.3  | 371.3   | 190.4   | 89.6    | -808.9  | -141.1  | 118.3   | 169.7   | -481.7  | -209.4  | 92.2    | 27.4    |
|    | 24 | 374.3   | -135.2  | 477     | 15.8    |         |         |         |         |         |         |         |         |
| 14 | 0  | -147.3  | 4031.9  | 1435.8  | -46.5   | -5117.2 | 81      | 885.1   | -1175.5 | -1107.2 | -887.5  | -763.1  | 504.5   |
|    | 12 | -304.1  | 419.4   | -131.7  | -258.4  | 44.3    | -169.2  | -284.9  | -656.5  | -168.8  | 249.9   | 92.2    | 434.8   |
|    | 24 | 399.3   | 2742.8  | -2269.8 | -422.2  |         |         |         |         |         |         |         |         |
| 15 | 0  | 55      | 2056.7  | 41.7    | 644.4   | -94.3   | -222.1  | 403.9   | 173.3   | 78.1    | 285.3   | 213     | -356.7  |
|    | 12 | -415.1  | -64.4   | 8192    | 27.7    | 129.3   | -171.8  | 124.2   | -475.3  | -119.8  | 553.9   | 168.1   | 405.4   |
|    | 24 | 231.8   | -115.4  | 357.8   | -11.8   |         |         |         |         |         |         |         |         |
| 16 | 0  | -4373.6 | -4707.6 | -234.2  | -154.3  | -32.1   | 544.7   | -566.6  | -793.3  | 514.2   | 542.4   | -440.2  | -49.7   |
|    | 12 | 595.3   | -94     | 102.5   | 178.9   | -85.8   | -285.5  | -218.5  | 143.9   | 31.4    | 98.4    | 180.4   | -214.9  |
|    | 24 | 3.5     | 262.8   | -484.5  | -88     |         |         |         |         |         |         |         |         |
| 17 | 0  | 290     | -8192   | -278.4  | 297.9   | 146.5   | -0.5    | -588.2  | -248.6  | -199.9  | -520.7  | 497.6   | 475.9   |
|    | 12 | 118.3   | -89     | 107.1   | -91.6   | -86.8   | 91.8    | -77.5   | 233.9   | -58.3   | 394.4   | -145.9  | -215.6  |
|    | 24 | 265.4   | -149.4  | -72.8   | 135.5   |         |         |         |         |         |         |         |         |
| 18 | 0  | -110.7  | -1113   | -290.1  | 4806.1  | 1204.9  | -730.4  | -5190   | 400.1   | -593    | -1403.1 | -340.4  | 677.9   |
|    | 12 | 29.2    | -122.1  | 20.3    | 320.7   | 79.6    | 181.3   | 373.8   | 20      | -197.5  | -354.6  | -208.2  | 129.6   |
|    | 24 | -98.4   | 103.4   | 252.8   | -91.8   |         |         |         |         |         |         |         |         |
| 19 | 0  | 378.7   | 137     | -1140   | -592.3  | -251.4  | -532.3  | 1373.9  | 238.4   | 1098.1  | -27.9   | 403.4   | 32.6    |
|    | 12 | 8140.6  | -375.4  | -137.7  | -83.2   | -147.7  | 168.8   | 57      | 360.3   | 179.8   | -195.7  | -220.1  | -226.8  |
|    | 24 | 170.5   | 230.1   | -163    | -56.9   |         |         |         |         |         |         |         |         |
| 20 | 0  | -987.2  | 30.1    | 866.7   | 526.6   | 8093.1  | 70.7    | -55.7   | -699.4  | -1164.6 | -316.1  | 1375.7  | 189     |
|    | 12 | 22.1    | -498.7  | 347.6   | -16.3   | 22.9    | -76.5   | 0.5     | 97.8    | -269.5  | 74      | -329.3  | 416.5   |
|    | 24 | 51.1    | -208.5  | -687.8  | -355.3  |         |         |         |         |         |         |         |         |
| 21 | 0  | 630.7   | -1598.8 | -450.4  | -489.3  | 168     | 1335.9  | -243.7  | 841     | -433.4  | -23.2   | 561.6   | 413.4   |
|    | 12 | 17.6    | -575.4  | 49.2    | -282.7  | -210    | -28.5   | 12.3    | 228.6   | 70.7    | 7983.5  | 43.6    | -92.4   |
|    | 24 | -191.5  | 124.7   | -265.7  | -638.3  |         |         |         |         |         |         |         |         |
| 22 | 0  | -890.3  | 548.5   | -274.7  | 1330.1  | -126    | 104.6   | 497.7   | 85.7    | 217.6   | 21.2    | 8192    | 370.7   |
|    | 12 | -722.8  | 0.7     | -477.6  | 20.6    | 1.1     | -385.2  | 46.2    | -386.8  | -204.8  | -104.7  | -154.2  | -134.7  |
|    | 24 | 131.3   | -264.4  | -509.6  | -312.1  |         |         |         |         |         |         |         |         |
| 23 | 0  | -3520.7 | 1720.9  | -920.1  | -958.3  | 633.2   | -725.3  | -1123.3 | -240.1  | 2405    | -1595.2 | -2771.2 | -464    |
|    | 12 | -995.5  | -2664.6 | -962    | -1557.6 | 529.4   | 176.4   | -432.5  | 26.9    | -311.8  | 68.5    | -222.6  | 162.7   |
|    | 24 | 4052    | -508.5  | -260.2  | -5318.8 |         |         |         |         |         |         |         |         |
| 24 | 0  | -1135.5 | -686.1  | -738.4  | 27      | 225.8   | -2268.9 | 219.7   | 608.7   | 276.7   | 586.2   | -316.9  | 43.1    |
|    | 12 | -28.4   | 8192    | 415.8   | -84.4   | 174.8   | -61.7   | -40.6   | -286.4  | 148.3   | 101.6   | -161    | -82.4   |
|    | 24 | 110.8   | 268.8   | -79.3   | 121.3   |         |         |         |         |         |         |         |         |
| 25 | 0  | -311.8  | -137.4  | -670.6  | -884.3  | 113.8   | -261.8  | -146.9  | -241.6  | -489.4  | 155.7   | -64     | 8192    |
|    | 12 | -116.1  | -97     | -146.1  | 357.8   | 7.3     | 254.2   | 220.7   | -1634.1 | 353.3   | -134.5  | -26.2   | -222.5  |
|    | 24 | -382.3  | 4.3     | 112.5   | -157.4  |         |         |         |         |         |         |         |         |
| 26 | 0  | 1862.8  | -495.5  | 6418.9  | -149.9  | 340.6   | -2238.5 | 196.6   | -164.2  | -1437.8 | 435.9   | 710.5   | -514.6  |
|    | 12 | 169.9   | -348.1  | -74     | 106     | 138.6   | -177.7  | 21.5    | 283.2   | -545.6  | 89.3    | -63.1   | 431     |
|    | 24 | 764.4   | -249.3  | -414.9  | -587.9  |         |         |         |         |         |         |         |         |
| 27 | 0  | -41.7   | 270.5   | -359.4  | -10.9   | -27.3   | -261    | 142.7   | 77.9    | 14.5    | 19.9    | 345.9   | 52      |
|    | 12 | -78.3   | 81      | 122.3   | 102.6   | 3.7     | 8192    | -30.7   | -356.9  | -97.6   | 73.5    | -216.4  | -302.7  |
|    | 24 | -153.8  | 42.4    | 2.2     | 98      |         |         |         |         |         |         |         |         |
| 28 | 0  | -583.8  | 365     | -4062.4 | 1.6     | 1181    | 833.6   | 650.5   | -1345.9 | -5367.4 | 231.7   | 93.2    | -2805.4 |
|    | 12 | 366.5   | -317.6  | -158.5  | 116.1   | -199.2  | -92.6   | 223.6   | 817     | 174.4   | -114.9  | 17.9    | -1200.2 |
|    | 24 | 148.8   | -180.4  | -170    | -80.1   |         |         |         |         |         |         |         |         |
| 29 | 0  | -1043   | -79.6   | -359.2  | -298.3  | -661.8  | -1090.7 | 128.3   | -837.1  | -1261.5 | -8192   | 216.3   | 141.4   |
|    | 12 | -51.2   | 582.6   | 205     | 182.4   | -69.7   | 335.1   | 110.7   | 118.4   | -108    | -75.7   | 274.5   | -77.5   |
|    | 24 | 18.7    | -89.6   | 179.5   | -99.4   |         |         |         |         |         |         |         |         |
| 30 | 0  | 709.8   | -1626.7 | -165.6  | 1293.3  | 19.2    | -1666.5 | 133.5   | 138.4   | -806.8  | -440.8  | 518.5   | 696.7   |
|    | 12 | 75      | 129.9   | -23.3   | -8192   | 38.3    | -374.9  | -127.1  | -946.8  | 129.2   | -201    | -42.1   | 251.4   |
|    | 24 | 293     | 430.6   | -511.5  | 637     |         |         |         |         |         |         |         |         |
| 31 | 0  | -710.9  | 4344.6  | -134.9  | 5527.1  | -120.1  | 713.5   | 394.8   | -680.5  | 404     | 66.3    | 336.9   | 478     |
|    | 12 | -302.1  | -30.3   | 27.7    | 318     | -498.2  | 23      | 85.9    | -151.9  | -74.7   | -77.4   | -225.1  | 242.8   |

|  |    |       |      |       |       |  |  |  |  |  |  |  |  |
|--|----|-------|------|-------|-------|--|--|--|--|--|--|--|--|
|  | 24 | 780.9 | -393 | 656.9 | 715.2 |  |  |  |  |  |  |  |  |
|--|----|-------|------|-------|-------|--|--|--|--|--|--|--|--|

**Table 4.A.24 – Interleave VQ codebook 1 for scalable coder (WINDOW TYPE : LONG)**

| filename | contents | mode         | number of elements | number of vectors |
|----------|----------|--------------|--------------------|-------------------|
| scmdct_1 | MDCT     | enhance LONG | 28                 | 32                |

| VN | EN | 0       | 1       | 2       | 3       | 4       | 5       | 6       | 7       | 8       | 9       | 10      | 11      |
|----|----|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
| 0  | 0  | -561.2  | -713.8  | -16.7   | 1735.2  | 215     | -8119.3 | -300    | -116.6  | -321.4  | 830.9   | 355.2   | -58.3   |
|    | 12 | 280.3   | 464.6   | 167.6   | 47.9    | -11.4   | 96.1    | 137.8   | -252.5  | 137.8   | -303.4  | -98.5   | 14.5    |
|    | 24 | -225.3  | 160.7   | -12     | -58.6   |         |         |         |         |         |         |         |         |
| 1  | 0  | -775    | 315.3   | -620.2  | -992.4  | -12.8   | -1335.7 | 143.6   | -1198.4 | -1288.1 | 8015.5  | 204.3   | 158.9   |
|    | 12 | -591.5  | 731.1   | 144.8   | 179.3   | 93.2    | 145.5   | 38.9    | 257.6   | 101.6   | -67.5   | 50.9    | 11.4    |
|    | 24 | 99.4    | -210.4  | -40     | -58.5   |         |         |         |         |         |         |         |         |
| 2  | 0  | -3915   | -782.6  | -1029.4 | 4961    | 534.9   | -181.7  | 606.6   | 369.6   | -1139.2 | -1963.3 | -426.4  | -941.3  |
|    | 12 | 259     | 276.8   | 121.6   | 84.6    | 202     | 449.1   | 144.5   | 205.4   | -366.1  | -335.5  | 20.2    | 82.8    |
|    | 24 | -357.4  | 119.7   | -107.2  | -358.5  |         |         |         |         |         |         |         |         |
| 3  | 0  | -1610.2 | -1283.8 | -1449.3 | 437.8   | -267.9  | 5562.3  | -2744.8 | -305.7  | 278     | 2618.1  | 326.5   | -169.6  |
|    | 12 | -2019.1 | -1045.5 | 321.3   | 281.7   | -564.1  | 159.5   | 358.6   | -793.5  | -231.2  | -47.2   | 172.5   | 195.9   |
|    | 24 | -473.5  | 609.3   | -268.5  | -18.1   |         |         |         |         |         |         |         |         |
| 4  | 0  | 203.1   | 1157.9  | -332    | 8192    | 246.5   | -60.8   | -900.1  | 285.3   | 878.8   | -932.6  | 72.4    | 223.6   |
|    | 12 | 206.3   | 117.7   | 195.7   | -205.6  | 50.3    | -217.2  | -262.4  | 197.6   | 33.2    | -89.9   | 313.1   | 270.3   |
|    | 24 | -78.6   | 208.8   | 135.8   | -92     |         |         |         |         |         |         |         |         |
| 5  | 0  | -182.8  | 3165.2  | 500.3   | -3345.9 | 631.5   | -220.1  | -219.9  | 4608.3  | -3178.4 | -399.9  | -312.2  | 378.8   |
|    | 12 | -559.3  | 196.1   | 354.2   | 24.5    | 397.2   | 24.5    | -122.3  | -579.3  | -29.6   | -351.2  | 29      | 306     |
|    | 24 | -518.6  | 329.2   | 259     | -361.8  |         |         |         |         |         |         |         |         |
| 6  | 0  | 3560.4  | -556.8  | 901.4   | -1009.7 | -3115.6 | -1781.1 | 20.4    | -448.8  | -2473.8 | -1018.8 | -3821.8 | -655    |
|    | 12 | -247.1  | 231.3   | -350.5  | -76.9   | -2524.5 | -11.8   | 601.6   | -42.9   | 503.4   | 11.8    | -98.8   | 563.1   |
|    | 24 | -855.1  | -613.3  | 877.7   | -836.3  |         |         |         |         |         |         |         |         |
| 7  | 0  | 99.3    | 803.3   | -350.6  | -103.9  | 556.6   | -2584.9 | 501.2   | 24.7    | 7101.9  | 42.6    | 83.8    | -947.8  |
|    | 12 | 630.4   | 491.2   | -69     | 3.7     | -182.6  | -165.1  | 372.6   | 64.4    | 120.3   | -103.6  | 51.5    | 3.1     |
|    | 24 | -483.5  | -39.7   | 254.2   | 315.2   |         |         |         |         |         |         |         |         |
| 8  | 0  | -684.1  | -186.1  | 1408    | 467.3   | 516.3   | 647.9   | -1823.9 | -209.9  | -1533.2 | 214.2   | -735.8  | -223.6  |
|    | 12 | 7325.9  | 349.4   | 210.1   | 66.3    | 160.8   | 63.6    | -153.7  | -434.8  | -205.1  | 214     | 479.9   | 138.6   |
|    | 24 | -90.4   | -36.2   | 260.9   | -82     |         |         |         |         |         |         |         |         |
| 9  | 0  | 326     | 510.6   | 664.5   | 933.7   | -171.2  | 614.6   | 264.1   | 528.9   | 97.5    | -301.1  | 157     | 6638.7  |
|    | 12 | -70.2   | 209.6   | 74.7    | -734.7  | 91.6    | -496.3  | -258.8  | 4242    | -162.9  | -179.7  | 101.5   | 346.6   |
|    | 24 | 250.4   | -181.2  | 165     | -184    |         |         |         |         |         |         |         |         |
| 10 | 0  | 536.5   | 1332.9  | 164     | 1325    | -231.2  | -62     | -8136.5 | 13.3    | -8.8    | -183.4  | -21.5   | -156.9  |
|    | 12 | -748.1  | 78.5    | -84     | -435.3  | 12.9    | -236    | -118.6  | 182.9   | -26.3   | 52.5    | -93.9   | 402.3   |
|    | 24 | -43.3   | -265.1  | 249.3   | 96.7    |         |         |         |         |         |         |         |         |
| 11 | 0  | 2532.6  | -1240.8 | 4875.4  | -395.9  | 744.2   | 4754.6  | -212.9  | 538.6   | 38.7    | -791.5  | 311.9   | -400    |
|    | 12 | -141.9  | -34.1   | 122.9   | 20.4    | 310.5   | 289     | -264    | 34.3    | -295.5  | 13.5    | -98.2   | 85.9    |
|    | 24 | 155.5   | -421.3  | -420.2  | -1113.5 |         |         |         |         |         |         |         |         |
| 12 | 0  | -854.1  | -4737.3 | -4964.8 | -42.6   | 36      | 271     | 402.8   | 216     | -361.4  | -1031.3 | 896.6   | 303.3   |
|    | 12 | -882.9  | 146.9   | 407.8   | -57.6   | 140.3   | -62.1   | -208.2  | -299    | 62.6    | -104    | -54     | -22     |
|    | 24 | 125     | 541     | 167.3   | 62.6    |         |         |         |         |         |         |         |         |
| 13 | 0  | -37.3   | -320.7  | -479.8  | -1002   | 6982.4  | 428.5   | 429.6   | 1185.6  | 270.6   | -38.5   | -2430   | -216.1  |
|    | 12 | -258.1  | 1022.9  | 7.5     | 31.9    | 145.5   | -95.5   | 57.1    | -248.5  | -362.5  | 386.9   | -647.1  | -296.8  |
|    | 24 | -215.8  | 657.7   | 158     | -170.8  |         |         |         |         |         |         |         |         |
| 14 | 0  | -936.6  | -6447.3 | 1963.8  | -43.3   | 621.1   | -589.6  | 790.4   | 34.8    | 192     | -681.5  | -75.1   | -1008.5 |
|    | 12 | -592.1  | 372.8   | -407.7  | -232.1  | 84.5    | 48      | 735.8   | -190.7  | -189    | -36.1   | -20.3   | 69      |
|    | 24 | -379.5  | 24.9    | -290.1  | -187.7  |         |         |         |         |         |         |         |         |
| 15 | 0  | -903    | -538.2  | -466.7  | 82.9    | 323.1   | -2047.7 | 296     | 732.5   | -316.9  | 625.5   | -87.2   | 19.2    |
|    | 12 | 117     | -8038.9 | 371.8   | 126.6   | -22.1   | -100.3  | -177.6  | -126.4  | 260.5   | 0.5     | 0.6     | 34.8    |

|    |      |         |         |         |         |         |         |         |         |        |         |         |         |
|----|------|---------|---------|---------|---------|---------|---------|---------|---------|--------|---------|---------|---------|
| 16 | 24   | -285.4  | 103.3   | -347.5  | 389.4   |         |         |         |         |        |         |         |         |
|    | 0    | 1747.9  | 2234.3  | -277.5  | -109.8  | -171.4  | -878.5  | -34.1   | -6668.5 | -277.8 | -71.8   | -929.3  | 77.1    |
|    | 12   | -55.3   | -834.8  | 1929.4  | 4.5     | 18.3    | 59.5    | -95.9   | 305.9   | 158.7  | 59.7    | 248.2   | -149.5  |
| 17 | 24   | 128     | -65.1   | -1.6    | 213.8   |         |         |         |         |        |         |         |         |
|    | 0    | -739.1  | -212.2  | 400.8   | 70.1    | -189.8  | 683.7   | 40.3    | -532    | 3      | -695    | -1249.8 | -1958.4 |
|    | 12   | -1494   | 142.9   | -97.3   | -2738.4 | -112.9  | 159     | -7679.3 | 260.8   | -267.6 | -203.5  | 147.6   | 1002.2  |
| 18 | 24   | -304.2  | 317.3   | 792.7   | 427.9   |         |         |         |         |        |         |         |         |
|    | 0    | 3208.6  | -221.6  | -1632.8 | 444.5   | -382.3  | -171.5  | 130.6   | 5996.9  | 1416.9 | 299.8   | -641.1  | -1334.1 |
|    | 12   | -135    | -295.7  | 999.2   | -588.2  | -109.4  | 0.8     | 45.1    | 241.5   | 716    | 75.3    | 181.6   | -63.4   |
| 19 | 24   | 608     | 302.1   | -2.8    | 186.6   |         |         |         |         |        |         |         |         |
|    | 0    | -4161.7 | -205.9  | -505    | -5407   | 132.5   | 913.8   | 441.8   | -43.6   | 281.5  | -884    | 369     | 306.5   |
|    | 12   | 236.6   | 16.9    | 321.3   | -136    | -285.3  | -57     | 117.7   | 172.7   | -341.1 | -286.4  | 518.2   | -36.8   |
| 20 | 24   | -165.2  | 38.2    | -128    | 2036.2  |         |         |         |         |        |         |         |         |
|    | 0    | 4768.2  | 651.6   | -1323.6 | -21.5   | 3555.3  | 2592.6  | 672.9   | -1588.3 | 776.8  | 1531.6  | 170.8   | -1284.5 |
|    | 12   | -351.6  | 29.5    | 637.3   | -571.3  | 401.2   | -84.3   | -197.4  | -192.6  | 304.2  | -91.6   | -144.7  | -112.9  |
| 21 | 24   | 135.2   | -150.4  | 49.5    | 492.3   |         |         |         |         |        |         |         |         |
|    | 0    | 566.2   | 193.6   | -8192   | -1008.5 | 311     | -703.7  | 143.8   | -322.4  | -345.2 | 493.5   | -122.1  | -583.3  |
|    | 12   | -72.1   | 189.9   | -399.4  | -105    | 209     | -118.6  | 97.1    | 220.5   | 479.3  | -14.2   | 198.2   | -104.4  |
| 22 | 24   | 202.3   | 83.1    | 12.5    | 226.3   |         |         |         |         |        |         |         |         |
|    | 0    | -529.2  | 317.2   | -3662.3 | -2057.5 | -4863.5 | 1738    | 495.1   | 783.9   | 9.1    | 105.2   | -514.3  | 84.7    |
|    | 12   | 258.6   | 538.8   | 134     | -14.6   | 359.9   | -259.7  | -39.4   | -164.9  | 525.2  | 19.7    | -3801.9 | 78.5    |
| 23 | 24   | -237.3  | -526.4  | 466.5   | 20.8    |         |         |         |         |        |         |         |         |
|    | 0    | 1203.4  | -1301.8 | 348.1   | -1694.5 | -83.5   | -667.8  | -69.4   | -248.8  | -806.8 | -32.7   | 7341    | -343.4  |
|    | 12   | 80      | -278.2  | 174.6   | -342.1  | -312.2  | 498.4   | -382.4  | -16.3   | 252.4  | 246.8   | 1.9     | 111.2   |
| 24 | 24   | 321.4   | 77.7    | 248.6   | -645.2  |         |         |         |         |        |         |         |         |
|    | 0    | -2744   | -226.8  | 5684.9  | -3196.3 | 201     | -1143.2 | -68.9   | 48      | 625.6  | -12.2   | -1007.2 | -911.8  |
|    | 12   | -619.4  | 413.5   | -113.2  | 45.1    | 334.9   | 15.2    | -7.5    | 363.2   | -1040  | -165.1  | 230.6   | -309.3  |
| 25 | 24   | 510.4   | -149.2  | -171.9  | 454.3   |         |         |         |         |        |         |         |         |
|    | 0    | 482.8   | 2739.8  | -178.6  | 648.8   | 19.1    | -422    | 129.6   | 155.7   | -32.2  | 218.2   | 255.1   | -766.4  |
|    | 12   | -511.5  | -441.8  | -7895.9 | -95.4   | 88.4    | -507.1  | -117.5  | -160.8  | 110.5  | 478.6   | -43.8   | 375.2   |
| 26 | 24   | -422.4  | -239.4  | 273     | 43.8    |         |         |         |         |        |         |         |         |
|    | 0    | 459.8   | 2520.3  | -3700.9 | -219    | 604.4   | -398.3  | -3011.9 | -165.8  | 611    | -3859.3 | 549.1   | -468.7  |
|    | 12   | -54.6   | 378.4   | 259.5   | -6.5    | 291     | 215.1   | 225     | -25.2   | 51.5   | 94.8    | -151.8  | -382.5  |
| 27 | 24   | -18.1   | -1051.7 | -4932.4 | 174.7   |         |         |         |         |        |         |         |         |
|    | 0    | -7985.6 | -471.4  | 536.2   | -173.5  | 565.7   | 67.8    | 802.8   | -1040.1 | 832.5  | -106.7  | -361.2  | 453.6   |
|    | 12   | 40.1    | -622.4  | -140.1  | -63.5   | 383     | -80.7   | -149.3  | -162.5  | -72.1  | 84.6    | -199.5  | 84.3    |
| 28 | 24   | -575.8  | -136    | 87.8    | -713.6  |         |         |         |         |        |         |         |         |
|    | 0    | 350.8   | -1145.1 | -210.1  | 1038.6  | -18.8   | -1098   | 210.8   | 210.4   | -802.5 | -696.9  | 212.9   | 381.2   |
|    | 12   | -524.5  | 73.2    | -33.8   | 8192    | 130.9   | -435.1  | -880.7  | -549.6  | 133    | -97.3   | 76.1    | 177.6   |
| 29 | 24   | -12     | 261     | -168.5  | 177.8   |         |         |         |         |        |         |         |         |
|    | 0    | -3820.4 | 4929    | 586.4   | 357.3   | 208.7   | -1327.9 | 525.2   | -426.1  | -290   | -84.7   | -283.1  | -992.5  |
|    | 12   | -777    | 727.9   | 522.5   | -130.6  | -31.9   | 58.6    | 655.4   | -233.7  | -2.1   | 1969.1  | -24.5   | 122.3   |
| 30 | 24   | -1434.3 | -529.6  | -127.7  | 399.8   |         |         |         |         |        |         |         |         |
|    | 0    | -253.6  | 580.2   | 3009.4  | 2563.5  | -2683.8 | 34.2    | -168.6  | 349     | 322.5  | -399.6  | 626.4   | -743.8  |
|    | 12   | -938.8  | 151.5   | 8.4     | -310.6  | 199.1   | 116.5   | 181.6   | -192.6  | -90    | 93.5    | 178.1   | -6795.7 |
| 31 | 24   | 77.2    | 689.1   | 269.7   | 216.1   |         |         |         |         |        |         |         |         |
|    | 0    | -74.9   | 2728.3  | 143     | 1142.2  | 181.9   | 3492.9  | 5593.3  | 631     | -296.2 | -136.5  | 464.6   | -234.4  |
|    | 12   | 639.1   | -401.2  | 206.2   | -0.4    | 244.2   | -132.8  | 323.2   | -611.7  | -270.7 | -438    | 756.8   | 194.9   |
| 24 | 30.6 | -370.5  | -55.2   | -210.9  |         |         |         |         |         |        |         |         |         |

Table 4.A.25 – Interleave VQ codebook 0 for scalable coder (WINDOW TYPE : SHORT)

| filename | contents | mode          | number of elements | number of vectors |
|----------|----------|---------------|--------------------|-------------------|
| scmdct_2 | MDCT     | enhance SHORT | 24                 | 32                |

ISO/IEC 14496-3:2001(E)

| VN | EN | 0       | 1       | 2       | 3       | 4       | 5       | 6       | 7       | 8       | 9       | 10      | 11      |
|----|----|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
| 0  | 0  | 613     | -1918.4 | -968.9  | 892.6   | 419     | 79.3    | -77.8   | -240.8  | 162.1   | 493.8   | -1223.7 | 720     |
|    | 12 | 181.3   | -1005.3 | -10.1   | 1160.7  | 464.2   | -1110.3 | 138.9   | -7755.7 | -603.2  | -621    | 779.9   | -144.7  |
| 1  | 0  | 199.2   | 449.8   | -147.5  | -650.1  | 474.8   | -258.1  | 635.7   | -975.8  | 768.7   | -1061.6 | 606.4   | -437.7  |
|    | 12 | -535.1  | 214.6   | -934.9  | -8192   | -104.9  | 84      | -393.5  | -261.4  | 26.9    | 1021.8  | -110.7  | -818    |
| 2  | 0  | -2191.5 | 516.5   | -512.3  | -7841.8 | -1863.1 | 930.8   | 296.5   | -854.1  | 321.5   | -104.4  | 745.6   | 1428.2  |
|    | 12 | -71.9   | -406.6  | 549.5   | 1065.5  | 1224.3  | -899.9  | -418.3  | 4       | -588.7  | 125.3   | -942.3  | -1317.2 |
| 3  | 0  | -5741.3 | 1175.1  | -3780.8 | -1979.1 | 3997.5  | 2363.4  | 1948.9  | -1075.8 | 270.2   | 1113.4  | -1514.8 | 36.5    |
|    | 12 | -215.7  | 1244.3  | 1454.6  | -629.9  | 526.6   | -31.9   | 960.9   | -292.7  | 1256.4  | 1770.9  | -1638.3 | -1141.1 |
| 4  | 0  | 1869.4  | 1185.6  | 515.5   | -381    | -1273   | -942.5  | -988.9  | -595.9  | -954    | -663.4  | -547.1  | 731.1   |
|    | 12 | -360.1  | -93.3   | 7036.1  | -1416   | 1070.4  | -2318.2 | 314.9   | 440.7   | 72.7    | -60.8   | -1731.8 | 1086.1  |
| 5  | 0  | -535.5  | -152.7  | -523.3  | 1189.4  | 100.9   | -560.4  | -269.2  | -1528.7 | -482    | -672.3  | 204.8   | -1138.3 |
|    | 12 | -85.9   | -914.3  | -266.2  | 1683.5  | -411.3  | -522.6  | 427.8   | 436.3   | 166.2   | 8119    | -1297.1 | 173.8   |
| 6  | 0  | 1996.2  | 1353.1  | 2318.5  | 4260    | -3061.5 | 2192.3  | 702     | -2385.7 | 1653.8  | 1494.3  | -1128.3 | 807.4   |
|    | 12 | -1586.9 | 1139.7  | -1618.1 | 217.8   | -808.5  | 827     | -756.3  | 1528    | 493.9   | -1344.4 | -4055.4 | -118.7  |
| 7  | 0  | 164     | 620.7   | -359.6  | 201.5   | -932.1  | 8192    | -101.1  | -1723.8 | -94.3   | -950.4  | 198     | 903.7   |
|    | 12 | -602.2  | -198.7  | -514.4  | 1396.4  | -756.6  | -429    | 370.1   | -249.4  | -59.5   | 221.6   | -1645.2 | 1282.2  |
| 8  | 0  | -959.2  | -2799.8 | 2031.5  | 1081.8  | 3374.3  | 3582.4  | 603.2   | 777.2   | -1314   | 2646.5  | 340     | -1727.3 |
|    | 12 | 1645.3  | 173.9   | -558.3  | -807.2  | -645.4  | -2908.5 | -525.6  | 2070    | 269.2   | -86.7   | 3020.5  | -6239.7 |
| 9  | 0  | -474    | -489.6  | 511.5   | -707.2  | 247.4   | 1316.2  | -45.1   | -823.5  | -8016.4 | -1546.4 | 236.2   | -143.7  |
|    | 12 | 143.7   | -385.4  | -375.3  | -1493.8 | 716.5   | 334.3   | -156.9  | 658.5   | -48.5   | -1005.8 | -2546.6 | 320.7   |
| 10 | 0  | 896.7   | -1018.5 | -8192   | -1013.5 | -380.1  | 657.3   | 94.6    | -263.9  | 421.1   | -783.9  | -149    | -427.7  |
|    | 12 | 555.6   | 264.4   | 393.1   | -322.4  | 255.5   | -175    | -628.9  | 673.4   | 1279.4  | -641    | -1016.1 | -1884.3 |
| 11 | 0  | -513.2  | -335.3  | -37.1   | -805.8  | 1171.8  | 262.9   | -1791.4 | -8099.1 | 815.1   | -783.7  | 296.2   | -601.5  |
|    | 12 | -192    | 86.9    | -3.9    | 974     | 2015.1  | 161.7   | -1157   | -490.5  | -1972.9 | -985.8  | 381.7   | 525.7   |
| 12 | 0  | -2340.4 | -5661.8 | 4040    | -2018   | 620.7   | -1192.3 | 82.4    | -2415   | -531.9  | -408.1  | 1410.8  | -124.5  |
|    | 12 | 1669.3  | 1686    | 351.5   | -150.9  | -46.5   | 775.2   | -550    | -189.9  | 1333.1  | 33.9    | -1745.5 | -417.2  |
| 13 | 0  | 65      | 22.3    | 770.3   | -610    | 1222.8  | 625.5   | 923.8   | 611.1   | 778.8   | -72.4   | -793.9  | -8138.9 |
|    | 12 | -652.8  | -443.5  | -326.3  | 372     | -835.5  | -932.2  | -438    | 636.3   | 552.6   | -645.1  | 1783.7  | -1578.1 |
| 14 | 0  | -1400   | -98.5   | 3057    | 3613.4  | 2282.3  | 2572.4  | 3484.4  | -3571.9 | 481.7   | -3392.1 | 1297    | 1167.4  |
|    | 12 | 161     | -1064.2 | -313.3  | 1599.9  | -6.9    | 470.8   | 653.7   | 136.3   | 2168.9  | 1038.2  | 856.7   | 363     |
| 15 | 0  | -785.2  | -384.5  | -166.7  | 305.1   | -465.8  | -565.9  | -1368.1 | -287.8  | 52.2    | 207.1   | 2756.1  | 487.7   |
|    | 12 | 422.8   | -1724.3 | -646.7  | 219.3   | 52.2    | -7309.5 | -398.8  | 871.1   | 614.9   | -965.9  | -94.9   | -1695.4 |
| 16 | 0  | 3931    | -7212.4 | -30.7   | -853.7  | -141.4  | -138.3  | -196    | 295.3   | -78.4   | 364     | 388.5   | -1395.2 |
|    | 12 | -1237   | 1038.9  | 703.5   | 87.8    | 546.8   | -9.3    | -696.1  | -777.8  | -78.2   | -475.6  | 224     | -268.8  |
| 17 | 0  | 2059    | 2976.1  | 2990.4  | -3869.7 | -1417   | 1535.6  | -2343.5 | -983.8  | -969.7  | 1746.4  | 149.8   | 2876.3  |
|    | 12 | -1391.5 | -262.8  | -1512.9 | 980.8   | -1948.6 | -752.1  | -564.2  | 1323.3  | 1.1     | -518.9  | 2387.7  | 617.5   |
| 18 | 0  | 263.7   | 78.7    | 663.8   | -310.6  | -2441.4 | -1218.3 | -481.4  | 1743.2  | 514.8   | -483.3  | 1094.8  | 655.3   |
|    | 12 | 269.5   | 146.2   | -913    | -744.6  | 7872.2  | 2233.3  | 401.5   | 948.2   | -1182.8 | 1871.3  | 459.9   | -672.7  |
| 19 | 0  | 23.6    | 461.8   | 754.2   | -411.9  | 451.3   | 245.3   | -8192   | -294.2  | -192.5  | -819.3  | -791.4  | -1601.9 |
|    | 12 | -430.8  | -131.7  | 573.7   | 1347.4  | -345.2  | 46.1    | 1877.7  | -798.3  | -42.5   | -84.7   | 96      | 1252.6  |
| 20 | 0  | 381.9   | 3174.6  | -608.4  | -2156.8 | -3019.3 | 781     | 4767.2  | -1575.8 | 920.3   | -660.6  | -779.4  | -2567.7 |
|    | 12 | 1139.5  | 647.3   | 98.2    | 2453.2  | -1062.3 | -608.9  | -1438.9 | 1269.2  | -1554.3 | -979.2  | -244    | 1879.8  |
| 21 | 0  | -502.3  | -843.6  | 631.9   | 843.7   | 708.5   | -992    | -471.3  | -1323.4 | 788.6   | -952.4  | 833.7   | 1104.1  |
|    | 12 | 555.9   | 27.4    | 1181.2  | -183.5  | 291     | 803.2   | -106.9  | -517    | -504.5  | 108.5   | 8192    | 2984.5  |
| 22 | 0  | 433.6   | 246     | -1589.2 | -817.6  | -708.4  | -1522.4 | -1661.6 | 1227.3  | -1094.2 | -491.1  | 237     | -887.9  |
|    | 12 | 1532.3  | 1569.8  | -299.8  | -219.7  | -517.8  | 407.6   | 414.2   | 122.3   | 7957.8  | -501.4  | -406    | 202     |
| 23 | 0  | -4188.3 | 247.2   | 437.9   | 1234.2  | -3749.7 | -5089.5 | -159.7  | -401    | 1524.7  | -1408   | -3081.6 | -60.8   |
|    | 12 | 1318.4  | -63     | -1692.2 | -71     | -575.4  | -1530.1 | -6.8    | 838.7   | -913.4  | -431.7  | 1032.4  | -2078.2 |
| 24 | 0  | 656.5   | 1578.5  | -2239.9 | 764.9   | 1307.3  | -4063   | 2133.6  | -4375.1 | -2720.8 | -82.3   | -4.4    | 1379.8  |
|    | 12 | 1452.4  | -794.8  | -1084.7 | -720.3  | -1685.5 | -105.5  | 1770.7  | 1205.6  | -635.2  | 799.4   | 1327.4  | -776.7  |
| 25 | 0  | 406     | 353.4   | 1207.9  | 100.3   | -554.4  | -363.5  | 812.3   | -219.5  | 368.2   | -818    | -1283.4 | 115.5   |
|    | 12 | 949.7   | -348.7  | 864.5   | -336.3  | 326.9   | -779.5  | 7874.7  | -142.3  | 1076.4  | -848    | -246.6  | -3475.1 |
| 26 | 0  | -8192   | -3.9    | -103.4  | -300.1  | -451.7  | -318.6  | 197.6   | 139.9   | -167.5  | -386.7  | 1758.9  | 755.7   |
|    | 12 | 75.4    | -465.2  | 329.6   | 557.2   | -219    | 526.8   | 896.1   | 435.7   | -509.5  | 65.4    | -407.4  | -1614.7 |
| 27 | 0  | 246.7   | -2.2    | -202    | 763.1   | -8192   | 985.9   | 776.4   | 51.7    | 74.6    | 180.7   | -360    | 463.2   |
|    | 12 | 226     | -83.2   | 1488.2  | -1249.9 | -1186.7 | 167.1   | -211.1  | -562.4  | 75.1    | 1007.6  | 400.6   | -2326.2 |

|    |    |         |         |        |        |         |         |         |        |         |         |        |        |
|----|----|---------|---------|--------|--------|---------|---------|---------|--------|---------|---------|--------|--------|
| 28 | 0  | 336     | 441.6   | 215.6  | 104.9  | -216.9  | 860.4   | -1256.2 | 505.1  | 596.4   | -7828   | 542.6  | -218.9 |
|    | 12 | -1139.2 | 175.8   | -1623  | 383.9  | 66.5    | -230.2  | -732.1  | 505.2  | 417.1   | 587     | 210.5  | -1437  |
| 29 | 0  | 572.1   | -128    | 823    | -312.2 | 59.7    | 100.7   | -720.4  | 695.3  | -790.7  | -1235.8 | 891.5  | 1663.5 |
|    | 12 | 347.7   | 7774.4  | -121.7 | -516.7 | -906.6  | 929.5   | -315.7  | -397.6 | -1038.2 | -1197.8 | -92.7  | -929   |
| 30 | 0  | -1050.2 | -1276.6 | -428.5 | -141.9 | 88.4    | -1400.9 | 34.3    | 763.6  | -203.7  | -854.5  | -713.6 | 13.7   |
|    | 12 | -7944.3 | 265.6   | -520.4 | -532.2 | 1207.4  | -1606.8 | 480.2   | -31.7  | -2769.1 | -386.9  | -52.8  | -45.7  |
| 31 | 0  | -4196.5 | -786.1  | -474.8 | 2859.1 | -3151.7 | 3652.5  | -2296   | 1179.6 | -1717.4 | 1482.6  | -2578  | -401.3 |
|    | 12 | 1222.3  | 522.7   | 124.4  | 2285.2 | -180    | -849.8  | -180.3  | 1514.7 | -1489.2 | -946    | 1162.9 | 1043.9 |

Table 4.A.26 – Interleave VQ codebook 1 for scalable coder (WINDOW TYPE : SHORT)

| filename | contents | mode          | number of elements | number of vectors |
|----------|----------|---------------|--------------------|-------------------|
| scmdct_3 | MDCT     | enhance SHORT | 28                 | 32                |

| VN | EN | 0       | 1       | 2       | 3       | 4       | 5       | 6       | 7       | 8       | 9       | 10      | 11      |
|----|----|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
| 0  | 0  | -1587.4 | -8151.2 | 466.2   | 476.5   | 1310.7  | -36.4   | -835.2  | -385.1  | -329.8  | -983.2  | -724.4  | -861.8  |
|    | 12 | -354.7  | -992.7  | 197.9   | 554.3   | -77.3   | 207.1   | -977.7  | -660    | -331.6  | 602.2   | 706.1   | -1584.4 |
| 1  | 0  | -495    | 61.2    | -263.6  | 1248.6  | -1573.1 | -713.3  | 2527.9  | -6633.3 | -696.5  | 247.3   | 1391.3  | -289.7  |
|    | 12 | -324.9  | 489     | -1072   | -459.6  | -211.1  | -736    | 385.2   | -1296   | -37.3   | 1785.9  | -258.4  | -459    |
| 2  | 0  | 433     | -524.1  | 264.9   | -41.6   | 7135.3  | 1978.2  | 105.5   | -52.2   | -396.5  | -306.8  | -1282.4 | 412.9   |
|    | 12 | -21.3   | 1616.7  | -46.3   | -844.6  | -1902.3 | 1368.5  | -133.6  | 211.3   | 1114.3  | 457.7   | 2710.2  | -1426.3 |
| 3  | 0  | 1621.1  | -382.5  | 941.6   | 1426.9  | -4604.8 | 2534.3  | -1221.5 | 267.8   | -4996.4 | -1924.1 | 763.4   | 1417.4  |
|    | 12 | 254.9   | -620.3  | 325.2   | 358.6   | 669.7   | -799.6  | 529.5   | -319.1  | -364    | -423.1  | -2416.2 | -1041   |
| 4  | 0  | -357.9  | 860.6   | 339.1   | 143.7   | -188.7  | 1506.9  | -903.6  | -898.6  | 6330.3  | -3820.3 | 829.1   | -11.5   |
|    | 12 | 220.8   | -477.8  | 1134.2  | 253     | 383     | -1115.3 | -1149.9 | 848.9   | -1113.5 | 107.8   | -868.4  | 295     |
| 5  | 0  | -8192   | -591.2  | -299.6  | 524.8   | 1471.7  | 73.1    | -869.7  | -211.1  | -162.2  | 139.5   | -1121.5 | -1009.6 |
|    | 12 | 91.9    | 160     | -270.5  | 67.1    | -275.7  | -861.7  | -722.4  | -488.8  | -623.8  | 698.3   | 697.8   | -264.1  |
| 6  | 0  | -818.4  | 786     | -608.1  | -61.4   | 1509.7  | 7783.9  | -37.6   | 2065.4  | -638.7  | 198.9   | 35.8    | -824.1  |
|    | 12 | 791.2   | 739     | 605.3   | -1152.7 | -372.9  | 424.6   | 1346.8  | -18.8   | -625.8  | 604.6   | -384.4  | 284.6   |
| 7  | 0  | 3029.7  | -1750.6 | 4506.6  | -870    | 513.2   | 2301.4  | 927.1   | 2530.7  | 228.1   | -2474.1 | 1344.5  | -78.5   |
|    | 12 | 153.2   | -1658.2 | -537.2  | -280.5  | 642.4   | 2369.1  | 500.4   | -584.8  | -170.5  | -303.1  | 1704    | 136.9   |
| 8  | 0  | -1569.8 | -1288.6 | 201.5   | -1157.1 | -466.5  | 797.8   | 371.5   | -1819.5 | 537.5   | 131.8   | -3898.4 | 5382.6  |
|    | 12 | 1775.5  | -542.9  | -1048.7 | -370.1  | -635.7  | -952.7  | -997.7  | -41.5   | 726     | 11.2    | -81.3   | 13.6    |
| 9  | 0  | -2450   | 1788.6  | -1971.3 | 614.4   | -787.8  | -345.9  | -1830.9 | 4736.7  | 154.9   | 76.6    | 1477.9  | 576.4   |
|    | 12 | 1549.4  | -873.6  | -1628.5 | -5.8    | -1817.4 | 1087.2  | -119.7  | -4009.5 | -362.6  | -2510   | -3032   | -1122.4 |
| 10 | 0  | 1489.3  | 5731.5  | 4239.4  | 396.9   | 1859.3  | -1909.5 | 302     | -790.5  | 1461.5  | -1084   | -510    | -1498.7 |
|    | 12 | 788.4   | -259    | 965     | -536.7  | 428.1   | 622.7   | -227.5  | -1693.9 | -98.6   | 217.7   | -852.7  | -1826.4 |
| 11 | 0  | 727.1   | -471.3  | -3938   | 4596.7  | 155.4   | 2508.6  | -1160.6 | -959.7  | 1337.4  | -1124.4 | 782     | 747.3   |
|    | 12 | 169.4   | -577.7  | 2942.3  | -638.7  | 1307.7  | 1236.2  | -449.5  | -649.7  | 294.4   | 1321.4  | -699.9  | 364.8   |
| 12 | 0  | -3489.7 | -557.6  | 1321    | -1750.9 | -324.9  | -704.9  | 5375.6  | 2709.4  | 343.7   | -3100.4 | -1881.5 | 172.9   |
|    | 12 | -1917   | 847.7   | 679.8   | -268.1  | 974.4   | 27.3    | -60.4   | -1128.4 | 12.1    | -2983.5 | -354.1  | 945     |
| 13 | 0  | 254.3   | 1387.7  | 1209.8  | 598.9   | 277.1   | -934.4  | -7093.7 | 1923.9  | 1015.7  | -357.3  | -390.3  | 1251.3  |
|    | 12 | -946.7  | 480     | 208.4   | 1134.5  | 750     | -716.6  | -726.8  | 794.7   | -336    | 75.3    | 807     | -2576.3 |
| 14 | 0  | 1188.4  | -248.5  | 718     | -1366.1 | -3130.4 | -1170.6 | -101.4  | 1117.5  | 1004.6  | 7.1     | -1020.9 | 63.7    |
|    | 12 | -483.1  | -1065   | 878.5   | -1709.5 | -6818.4 | 1088.2  | 363.8   | -686.1  | -1432.5 | -405.5  | 343.5   | -391.3  |
| 15 | 0  | -945.3  | 2896.2  | 362.4   | -2761.9 | 2342.6  | 805.7   | -619.8  | 490.8   | -2910   | -798.1  | -59.8   | 409.3   |
|    | 12 | -1168.1 | 149.5   | 550.7   | 4936.1  | -563.9  | -328.6  | -320    | 616     | 1443.7  | 381.4   | -1751.2 | -738.3  |
| 16 | 0  | 1408.8  | -324.5  | -250.1  | -42.1   | -718.3  | 654.3   | 608     | 324.9   | -863.8  | 596.2   | -2048.1 | -1469.7 |
|    | 12 | 1793.5  | -211.2  | -15.2   | 394.4   | 992.8   | -6976.7 | 1115.8  | -1024.9 | 823.3   | 216.9   | -1298.7 | 2454    |
| 17 | 0  | -2586.3 | -2557.9 | -1035.1 | -2009.8 | -1809.3 | -5664.7 | -661.2  | -987.2  | -261.8  | -731.4  | 806.3   | 972.9   |
|    | 12 | -2113.7 | 1892    | 1078.4  | 1751.2  | -443.4  | 1183.3  | 1097.3  | 570.2   | -57.2   | -956    | -1259.4 | 53      |
| 18 | 0  | 1515.5  | 941.4   | -381.6  | 1674.6  | -81.4   | 121.6   | 2131.6  | -556.5  | 1081.9  | 1872.4  | -816.3  | -1076.9 |
|    | 12 | 496.7   | 7226.8  | -416.5  | -382.9  | 508.1   | 1137.3  | -1334.6 | 790.6   | 1024.6  | 613.2   | -441.6  | -1065.9 |
| 19 | 0  | -406.3  | 380.5   | 681     | -1482.5 | -2079.6 | -1901.1 | -3583.9 | -3315.9 | -447.2  | -63.2   | -691    | -1701.2 |
|    | 12 | 578.3   | -2632.7 | 663.9   | 30.6    | 1611.2  | 3718.7  | 867.1   | 1683.5  | 50      | -1270.9 | -1571.8 | -1539.7 |



|    |    |         |         |         |         |         |         |         |         |         |         |         |         |
|----|----|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|---------|
| 20 | 0  | 321.7   | -618.9  | 1318.2  | -2858.6 | -95.2   | 109.2   | 1145.3  | 1812    | -278.3  | 2607.3  | -359.3  | 1530.9  |
|    | 12 | -1664.7 | 678     | 2133.4  | -1815.8 | 1561.3  | 56.1    | -134.9  | -12.2   | -342.6  | 6635.7  | 2642.9  | 2408.3  |
| 21 | 0  | 757.9   | -357.8  | 798.2   | 492.1   | -1311.4 | 742.9   | 4154.7  | 105.9   | 571.6   | 2169.7  | 903.3   | 334.5   |
|    | 12 | -3191.2 | -1928.7 | -1051.4 | 1526.9  | 245.2   | -1750.8 | -1104.9 | 1049.2  | 5111.9  | -1193.6 | -521.4  | -734    |
| 22 | 0  | -1838.4 | 1235.6  | 1095.1  | 2725.7  | 2497.5  | -1924.5 | 1151.5  | 603.4   | -1416.1 | 2696.4  | -268.5  | -1621.3 |
|    | 12 | 399     | -529.9  | 2481.5  | -123.4  | 2178.4  | 857.2   | -991.2  | 3837    | -385.4  | -212.5  | -3028.1 | 4468.3  |
| 23 | 0  | 1330.4  | 2003.6  | -5625.7 | -1541.8 | 709.1   | -203.4  | 722.3   | 468.9   | 1065.2  | 1010.1  | 1321.9  | 653.3   |
|    | 12 | -1503.7 | -890.1  | 126.5   | 595.1   | 1292    | -120.9  | 867.1   | 309     | -1041.5 | -1233.9 | 4458.3  | 1747.7  |
| 24 | 0  | 3786.4  | 166.8   | 659.1   | -7165.8 | 736.4   | 760.8   | -157.3  | -832.6  | -726.2  | -558.5  | -1335.9 | -1448.4 |
|    | 12 | 1227.1  | 310.8   | -1051.7 | -528    | -791.4  | 629.3   | 33.1    | -1012.9 | -632.4  | -129.1  | -499.5  | 1025.3  |
| 25 | 0  | -1451.7 | 246.9   | 244.9   | 149.8   | 5.9     | 156.5   | -51.5   | -109.4  | 478.2   | 306.5   | 7042.5  | 253.4   |
|    | 12 | 122.7   | 110.6   | -2.7    | -947.8  | -1074.1 | 318.6   | 81.2    | 44.4    | -1188.6 | 980.2   | -1001.7 | 1166.1  |
| 26 | 0  | -1751.7 | 1177.9  | -1175.2 | -2322.5 | -4825.7 | 1590.5  | 258     | 229.3   | -92     | 167.1   | 71.5    | -2142.2 |
|    | 12 | -2665.3 | 1556.1  | -2104.4 | 374.6   | 1627.4  | -847.9  | 1737.1  | 110.3   | 594.9   | 1048.1  | -177.5  | -207.9  |
| 27 | 0  | 121     | -145.1  | -561.2  | 420.7   | -152.6  | -871.5  | -1213.1 | -294.2  | -868.6  | 807.8   | -220.4  | 981.4   |
|    | 12 | 89.8    | -313.7  | -7218.6 | 424.1   | -315.4  | -975.3  | -199.4  | -893.3  | 283.3   | -108.3  | 775.2   | 785.3   |
| 28 | 0  | 2464.9  | -3039.7 | -673.7  | 188.1   | 1483.1  | 1089.5  | -446.4  | 283     | 648.4   | 2184.9  | -1143.1 | 576.4   |
|    | 12 | 1052.9  | 281     | -2367.4 | 1505.7  | -702.3  | 628.9   | 4176.2  | 1901.7  | -1451.3 | -460    | -4274.9 | 174.8   |
| 29 | 0  | 9.6     | 362.4   | 362.1   | 802.5   | 497.8   | -914.2  | -1697.2 | -45.5   | -1122.9 | 648.9   | 720.2   | 916.9   |
|    | 12 | -371.9  | 588.3   | 162.3   | -6838.7 | -849.4  | -560.4  | 316.3   | 180.4   | -229.1  | -2966.6 | -638.9  | 437.5   |
| 30 | 0  | 3819.9  | 685.6   | -662.9  | 1849.7  | 1157.1  | -2943.1 | 2673    | -153.4  | -1106   | -1384.2 | 869.4   | 1226.8  |
|    | 12 | -513.6  | -352    | 1064.3  | 4032.2  | 375.9   | -824.9  | -1518.7 | 1262    | -2905.1 | 160     | -315.9  | -3058   |
| 31 | 0  | -125.4  | -618.8  | 379.6   | -840.3  | -975.8  | -1430.1 | 1370.9  | 583     | -452.9  | -73     | -374.4  | -1129   |
|    | 12 | 7405    | 175.4   | 123.6   | -458.8  | 1692.4  | -318.4  | -1535.9 | 521.6   | -277.1  | -690.6  | 899.4   | 3196.9  |

Table 4.A.27 – Bark scale codebook

| Filename | contents      | mode         | number of elements | number of vectors |
|----------|---------------|--------------|--------------------|-------------------|
| Fcdl     | bark envelope | core/enhance | 6                  | 64                |

| VN | 0        | 1        | 2        | 3        | 4        | 5        |
|----|----------|----------|----------|----------|----------|----------|
| 0  | 2.59553  | 1.59708  | -0.21402 | -0.17514 | -0.09628 | -0.26982 |
| 1  | -0.31846 | -0.28922 | -0.32107 | 1.15714  | 1.03967  | -0.03709 |
| 2  | 2.80202  | -0.48079 | 2.15678  | -0.37126 | -0.36071 | -0.16559 |
| 3  | 0.19375  | -0.32843 | -0.30811 | 0.27793  | -0.2106  | -0.07445 |
| 4  | 2.2074   | -0.43946 | -0.3363  | 1.40869  | 0.0118   | 0.04888  |
| 5  | -0.50163 | -0.3072  | -0.34648 | -0.36553 | 0.08881  | 0.66861  |
| 6  | -0.5478  | 3.10285  | 2.923    | -0.42525 | -0.30431 | -0.12124 |
| 7  | 0.36265  | -0.59101 | -0.40694 | -0.37653 | -0.02823 | 8        |
| 8  | 0.6108   | -0.2944  | 0.84234  | 0.50441  | 0.15898  | -0.19708 |
| 9  | -0.26828 | -0.42775 | -0.55847 | 8        | 0.26906  | 0.09807  |
| 10 | 0.58163  | 0.69724  | -0.38205 | -0.32305 | -0.04204 | -0.18015 |
| 11 | -0.1531  | -0.52465 | -0.5112  | -0.46322 | 8        | 0.60858  |
| 12 | 8        | -0.37894 | -0.41379 | -0.57177 | -0.46141 | -0.54807 |
| 13 | -0.3409  | -0.46601 | -0.40139 | 2.77429  | -0.30628 | -0.29932 |
| 14 | -0.00875 | 1.00366  | 0.35027  | -0.14719 | -0.12343 | -0.01017 |
| 15 | -0.42826 | -0.25287 | 0.5889   | 0.00455  | -0.18978 | 0.72595  |
| 16 | -0.41718 | -0.42836 | 0.44293  | -0.26096 | 0.67149  | -0.15131 |
| 17 | -0.48094 | 0.40551  | 0.63479  | 0.10576  | 0.60523  | 0.00965  |
| 18 | 0.5875   | 0.17574  | 0.22913  | -0.07386 | -0.25141 | -0.28609 |
| 19 | 1.54175  | -0.38932 | -0.26886 | -0.37004 | 1.26474  | -0.04734 |
| 20 | -0.30997 | -0.45256 | 2.71793  | -0.29703 | -0.34868 | -0.27166 |
| 21 | 0.80891  | -0.39427 | -0.36544 | 1.05681  | -0.26354 | -0.07337 |
| 22 | 1.69802  | 0.38677  | -0.19213 | 0.04237  | -0.07601 | -0.16281 |
| 23 | 0.82683  | 0.70298  | 1.21216  | -0.23461 | -0.02006 | -0.24785 |

|    |          |          |          |          |          |          |
|----|----------|----------|----------|----------|----------|----------|
| 24 | -0.4438  | -0.46604 | -0.44381 | -0.40097 | 1.97582  | 0.05232  |
| 25 | 0.71883  | 0.61561  | -0.1844  | 0.68805  | 0.00106  | -0.01607 |
| 26 | -0.38766 | 8        | -0.06823 | -0.22391 | -0.52152 | -0.42881 |
| 27 | -0.4812  | -0.4593  | -0.4612  | 1.54307  | -0.31462 | 0.33229  |
| 28 | 0.72907  | -0.31141 | -0.38468 | -0.36775 | -0.3587  | 0.7      |
| 29 | 0.88654  | 2.10076  | -0.1964  | -0.23175 | -0.15172 | -0.15909 |
| 30 | -0.51999 | -0.53691 | -0.52863 | -0.4753  | 0.54786  | -0.32601 |
| 31 | -0.4771  | 2.06467  | 1.01919  | -0.28684 | -0.14439 | -0.05744 |
| 32 | -0.57224 | 0.42109  | 0.5242   | -0.2856  | -0.1884  | -0.15752 |
| 33 | -0.24585 | -0.44169 | 8        | -0.43055 | -0.40749 | -0.43133 |
| 34 | 0.20221  | -0.25994 | -0.31787 | -0.30064 | 0.6737   | 0.05431  |
| 35 | 0.32692  | -0.4254  | -0.45128 | -0.45458 | -0.16592 | -0.3783  |
| 36 | 1.42145  | -0.42477 | -0.43237 | -0.33452 | -0.32365 | -0.35176 |
| 37 | -0.52165 | -0.47377 | 0.32662  | -0.34749 | -0.24129 | -0.18172 |
| 38 | -0.46545 | 0.29674  | -0.31146 | 0.41441  | -0.12466 | -0.07952 |
| 39 | -0.26585 | 0.07241  | -0.26056 | -0.2953  | -0.24943 | -0.22879 |
| 40 | 0.34838  | -0.3777  | 0.48323  | -0.31334 | -0.0997  | -0.15367 |
| 41 | -0.36403 | -0.40947 | 1.43126  | -0.48284 | -0.33652 | 0.15962  |
| 42 | -0.50425 | -0.42948 | -0.31036 | 0.28637  | 0.32368  | 0.02942  |
| 43 | -0.44723 | 2.90793  | -0.35379 | -0.37063 | -0.16951 | -0.00506 |
| 44 | -0.41686 | -0.31531 | 0.3233   | 0.63998  | -0.13565 | -0.25397 |
| 45 | -0.57783 | -0.55613 | -0.52567 | -0.35549 | -0.35857 | 0.13331  |
| 46 | -0.5269  | 2.92107  | -0.39287 | 2.50676  | 0.83325  | -0.34838 |
| 47 | -0.35037 | 1.58353  | -0.14216 | -0.14513 | 1.25833  | 0.02395  |
| 48 | -0.54618 | -0.51853 | -0.5225  | 0.6915   | -0.34809 | -0.32408 |
| 49 | 0.91424  | -0.39755 | -0.23708 | 0.06298  | 0.32884  | -0.27221 |
| 50 | -0.26031 | -0.34193 | 1.87043  | -0.21861 | 1.45874  | -0.0149  |
| 51 | 2.71047  | -0.44695 | -0.35421 | -0.35093 | -0.21284 | 0.18185  |
| 52 | -0.54924 | -0.36134 | 2.97081  | 2.32648  | 0.01315  | 0.14583  |
| 53 | -0.51477 | -0.40314 | 1.23469  | 0.2071   | 0.00622  | -0.15786 |
| 54 | -0.37564 | 0.82963  | 0.77552  | 0.72145  | -0.21951 | -0.21676 |
| 55 | 1.59584  | -0.35791 | 0.89497  | -0.26614 | -0.12674 | -0.13373 |
| 56 | -0.42762 | 1.49492  | -0.31584 | 0.6556   | -0.06384 | -0.07354 |
| 57 | -0.38684 | 0.71379  | -0.43055 | -0.24995 | -0.29525 | 0.42418  |
| 58 | -0.55264 | -0.52329 | -0.57869 | -0.60007 | -0.61091 | -0.56922 |
| 59 | -0.46465 | 0.55867  | -0.28994 | -0.14188 | 0.53955  | -0.1124  |
| 60 | -0.40785 | -0.46448 | 1.26074  | 1.48856  | 0.10128  | -0.10592 |
| 61 | -0.427   | 0.70964  | -0.30237 | 1.50654  | 0.0617   | -0.27741 |
| 62 | -0.43966 | 1.39518  | -0.39417 | -0.44686 | -0.35716 | -0.39151 |
| 63 | -0.43383 | 0.9626   | 1.83882  | -0.37505 | -0.235   | -0.05237 |

Table 4.A.28 – Periodic Peak Component codebook

| name | contents | mode      | number of elements | number of vectors |
|------|----------|-----------|--------------------|-------------------|
| pcdl | PPC      | core LONG | 10                 | 64+64             |

| VN | 0        | 1        | 2        | 3        | 4        | 5        | 6        | 7        | 8        | 9        |
|----|----------|----------|----------|----------|----------|----------|----------|----------|----------|----------|
| 0  | 1.28888  | 0.26874  | 3.16758  | 1.37315  | 0.49535  | 0.94071  | 1.44487  | 0.37004  | -0.70932 | -0.05462 |
| 1  | -0.02234 | -0.20001 | 0.19604  | 0.26626  | 0.75466  | -0.81913 | 0.03708  | -0.29647 | -1.78632 | -0.18008 |
| 2  | -2.19075 | 0.13333  | 2.93822  | -0.22869 | 0.54111  | 0.51598  | 0.29629  | -1.60571 | 0.48978  | -0.40555 |
| 3  | -0.16435 | -1.43562 | 0.12102  | 0.1109   | 0.83013  | 0.16508  | -0.07873 | 2.78031  | 0.15479  | -0.28546 |
| 4  | 3.06554  | -1.41708 | -1.88582 | 0.14353  | -1.66046 | 0.26375  | 0.08646  | -0.41044 | -0.23005 | 0.47026  |
| 5  | -0.11899 | -0.14149 | 0.03797  | 0.01927  | 3.72378  | 0.16278  | 0.28207  | 0.15989  | -0.17563 | 0.48802  |
| 6  | -0.19166 | 0.05232  | 0.68887  | 0.24894  | -0.48014 | 0.01794  | -0.09456 | -0.03353 | -3.71716 | -0.37879 |
| 7  | -0.08722 | 0.69099  | 0.52234  | 0.74945  | -0.62316 | 0.58827  | 0.30028  | 1.0085   | 0.24395  | 0.06055  |

|    |          |          |          |          |          |          |          |          |          |          |
|----|----------|----------|----------|----------|----------|----------|----------|----------|----------|----------|
| 8  | 0.12958  | -0.22199 | 0.96387  | 3.13907  | -1.96687 | 0.53834  | -0.69258 | -0.04179 | -0.39869 | -0.03142 |
| 9  | -0.02278 | 0.45226  | -0.46875 | 0.03839  | 2.35515  | -0.02039 | -0.03662 | -0.33315 | -0.3716  | -3.22201 |
| 10 | 1.50588  | -0.78015 | 1.43802  | -0.42078 | -1.56427 | -0.32711 | -1.81516 | -0.1517  | -1.39942 | -0.70676 |
| 11 | 0.03549  | 0.97029  | 0.17778  | 2.40233  | -0.1207  | 1.79612  | -0.09612 | -3.32378 | -0.06238 | -0.47122 |
| 12 | 0.11414  | -0.06499 | 1.95345  | -0.03737 | -0.08911 | 1.86812  | 0.03298  | 0.3419   | -0.9445  | 0.26631  |
| 13 | -1.20479 | -0.70412 | 0.30736  | 1.66473  | 0.25032  | -1.62769 | 0.26336  | -2.39109 | 1.66663  | -1.01706 |
| 14 | 0.27057  | 0.28062  | 4.60481  | 0.14478  | -0.04332 | -0.26043 | -0.15716 | -0.52842 | 1.15402  | 0.02068  |
| 15 | -0.02468 | -2.63793 | -0.06035 | 2.17716  | 0.55198  | 1.33668  | 0.18469  | 0.73687  | -1.59535 | 0.47395  |
| 16 | 0.08763  | -1.03259 | 0.29432  | 0.23395  | -0.21428 | 0.13532  | -4.58495 | 0.5589   | 1.19783  | -1.8095  |
| 17 | -1.87625 | 1.93644  | 0.2355   | 0.60058  | 1.07984  | -0.13744 | -0.89547 | -0.8501  | 0.80741  | -0.42413 |
| 18 | 0.23718  | -1.82628 | 2.62266  | 0.48302  | 0.5773   | 0.00497  | -0.88499 | 0.11386  | 0.50195  | -1.61181 |
| 19 | 2.29405  | 0.76792  | 0.77194  | 0.45336  | -0.15374 | -1.68579 | 0.40538  | -0.73821 | -0.52721 | 0.27069  |
| 20 | 0.31311  | 0.67546  | 0.05131  | -0.53788 | 0.75649  | -1.92602 | 0.44063  | -3.20197 | 0.24121  | -1.52231 |
| 21 | 0.38234  | -0.34155 | -2.34677 | 0.44454  | -0.12667 | -0.01386 | 0.83684  | -3.42385 | 0.27861  | -0.62057 |
| 22 | -2.294   | -0.00411 | 0.10751  | -2.0974  | 0.55445  | -1.90704 | 0.46176  | -0.22817 | -1.05712 | -0.31803 |
| 23 | 0.08838  | 0.104    | 0.07782  | 0.28709  | 0.77533  | 0.5377   | -3.49315 | -0.15122 | -0.71822 | -0.55433 |
| 24 | -0.12964 | -0.15924 | 0.08426  | 4.31671  | 0.35474  | -0.3812  | 0.08329  | -0.5072  | 0.22773  | 0.10575  |
| 25 | 0.37323  | 0.11307  | -0.26726 | 1.19933  | -0.2052  | 3.97109  | 0.23754  | 0.42287  | -0.17526 | 1.6363   |
| 26 | -0.8932  | 0.98569  | -0.17186 | -0.66099 | 0.34743  | 0.31512  | 0.56153  | 0.03641  | -0.2279  | -0.32669 |
| 27 | 0.49296  | 1.50705  | 0.11747  | 3.64671  | -0.08886 | 0.95113  | -0.10827 | 0.24228  | -0.2188  | 0.09239  |
| 28 | -0.05516 | -0.36735 | 0.06542  | -0.10454 | -0.64008 | -0.07052 | -0.50515 | 2.55652  | -0.94717 | -1.30125 |
| 29 | 0.09405  | 0.74533  | -1.13331 | 0.37229  | -0.38024 | 0.17472  | 0.61134  | 0.20712  | -0.10988 | -0.49257 |
| 30 | -0.11187 | -0.17222 | -1.73824 | -0.13261 | -0.52195 | 0.62213  | -0.08559 | 0.17988  | 3.15089  | -0.24704 |
| 31 | 0.01548  | 3.74936  | 0.08098  | -0.20425 | -0.05489 | -0.31726 | 0.69418  | 0.32569  | -0.1634  | 0.50865  |
| 32 | -0.17257 | 0.28592  | 2.00649  | 0.53473  | 2.57991  | 0.33445  | -0.33589 | -0.71256 | 0.51466  | 2.46521  |
| 33 | -0.09483 | 1.08439  | 0.03998  | 0.11961  | 1.85047  | 0.00631  | -0.45111 | -0.71269 | 0.15621  | 1.16161  |
| 34 | -0.02632 | 2.00056  | -0.32356 | -0.56487 | 0.10923  | 1.26427  | -0.2674  | 2.41749  | -0.16922 | -0.8848  |
| 35 | -0.0544  | 0.16837  | 2.56225  | 0.1894   | -0.22447 | -0.39572 | 0.49943  | 0.11457  | 0.35493  | 0.89878  |
| 36 | -0.26616 | -0.43364 | 0.3558   | 0.05177  | 0.50149  | -0.24278 | 0.17047  | -0.12116 | 5.45574  | -0.15218 |
| 37 | 0.06203  | 0.23433  | -0.99429 | 0.12958  | -0.628   | -3.51385 | -0.24958 | 0.06293  | -0.89521 | 0.2426   |
| 38 | 3.72107  | 1.23981  | -0.88335 | 0.46989  | -0.3001  | -0.42813 | 0.41867  | 1.0639   | 0.31391  | -0.57627 |
| 39 | -1.12063 | 0.44526  | 1.91996  | 0.34752  | -0.524   | -0.6746  | -2.59639 | 0.11581  | -0.20243 | 0.28307  |
| 40 | -0.03832 | -0.01316 | -0.03846 | -0.007   | 0.04931  | -0.09357 | 0.06964  | -0.00049 | 0.21041  | -0.08892 |
| 41 | 0.10125  | -0.03012 | -0.03539 | -2.37631 | 0.28491  | 2.4899   | -0.34378 | -0.73052 | 0.52019  | -0.10068 |
| 42 | 1.15802  | 0.7305   | 0.86786  | -0.15069 | 2.3452   | -0.69111 | 0.57534  | 1.29397  | 0.15267  | -0.16541 |
| 43 | -0.12218 | -0.03125 | -0.01363 | -1.5446  | -0.27404 | -0.62664 | 0.46486  | -0.44052 | 0.17302  | 0.77978  |
| 44 | -0.03625 | -0.33463 | -1.59515 | -0.42669 | -0.3637  | 3.45674  | 0.35785  | 0.34057  | 0.26874  | -0.66518 |
| 45 | 0.12284  | 0.01668  | -0.36273 | 0.00005  | 0.05147  | -0.29728 | -0.00085 | 5.10734  | 0.41718  | -0.05075 |
| 46 | -0.20474 | 1.11663  | -1.4853  | -1.6009  | -0.02503 | -0.24016 | -0.09167 | 2.05302  | 0.41858  | -0.52989 |
| 47 | -1.22133 | -0.80464 | 3.2915   | 0.29576  | -1.01545 | 0.85362  | 0.25387  | 1.69078  | 0.05201  | -0.3414  |
| 48 | 0.10775  | -0.30053 | -0.19873 | 0.20553  | 0.38424  | 0.2795   | 0.33175  | 0.10667  | -0.43378 | 5.0211   |
| 49 | 0.06324  | -1.1164  | -0.34474 | -0.03852 | 2.47664  | 0.30992  | 0.34303  | -3.0838  | -0.26944 | -0.15003 |
| 50 | -3.87998 | 0.08673  | -0.92651 | -0.28277 | 0.7912   | 0.06152  | 0.05675  | -0.04378 | -0.15055 | -0.49744 |
| 51 | 0.06965  | -0.24747 | -0.30479 | 0.44796  | -0.38039 | 0.30899  | -2.49186 | 0.15352  | -0.07609 | 2.77728  |
| 52 | 0.03771  | -0.07037 | 1.44405  | -0.08627 | 0.01465  | 0.65797  | -0.20831 | -0.02516 | 3.82295  | 0.58679  |
| 53 | -0.28911 | 0.00076  | 2.25478  | -2.67418 | 1.08246  | 0.20249  | 0.62566  | -0.04001 | -0.20703 | -0.15767 |
| 54 | 1.76926  | 0.21839  | 0.4214   | -0.18353 | -0.23904 | 0.25714  | -0.23697 | 0.07767  | 0.53228  | 0.22643  |
| 55 | 0.19622  | -0.064   | 0.10897  | 0.79608  | -1.20257 | -1.10434 | 0.30441  | 0.49781  | 3.11769  | -0.29082 |
| 56 | -0.31973 | -1.36087 | 1.1628   | 0.16043  | -1.01046 | 2.64181  | -0.19644 | -0.27073 | -0.77847 | -0.42009 |
| 57 | 0.04065  | -2.4468  | -0.45823 | 0.51819  | 0.08004  | -1.72072 | -0.52    | 0.59359  | 0.37168  | 0.78352  |
| 58 | -2.06255 | 0.4418   | -1.32138 | 2.39634  | -0.89775 | 1.60514  | -1.52266 | -0.22682 | -0.10606 | -0.5798  |
| 59 | -3.31681 | 0.85384  | -0.27833 | 0.29965  | -1.55139 | -0.12387 | 0.08035  | -0.41261 | 0.26349  | -0.21486 |
| 60 | -0.91388 | -0.98165 | -1.13409 | -1.47777 | 0.23383  | 1.1217   | -0.27391 | -2.83883 | 0.37536  | -0.83449 |
| 61 | -0.30317 | 0.03241  | -2.27905 | 0.39328  | -0.10539 | 2.62031  | -0.21607 | -0.11414 | -3.2875  | 0.16852  |
| 62 | 0.05213  | 2.18937  | 0.84015  | 1.33077  | -0.05358 | 1.3409   | -0.09204 | 0.09985  | 0.93062  | 0.59825  |
| 63 | 0.12392  | 1.59708  | 0.9337   | -0.90451 | -0.12543 | -2.01707 | 0.37513  | 1.94555  | 0.23099  | -0.70055 |
| 64 | 2.13905  | 0.10191  | 1.24373  | 0.51358  | 1.63431  | -0.08351 | 0.30739  | -1.16675 | -0.35279 | -0.37096 |

|     |          |          |          |          |          |          |          |          |          |          |
|-----|----------|----------|----------|----------|----------|----------|----------|----------|----------|----------|
| 65  | 0.0812   | 1.07058  | 0.15907  | -0.09428 | -0.14151 | -0.4618  | -0.21325 | 0.45456  | 0.66027  | 3.37816  |
| 66  | -1.32206 | -0.4821  | 1.43573  | 1.74563  | 2.01948  | -1.56545 | 0.17182  | 0.45271  | 0.0628   | -0.91456 |
| 67  | -2.3898  | -2.35197 | -0.22554 | 0.33998  | -0.79031 | 0.06321  | -0.01377 | 1.07176  | 1.19673  | 0.2861   |
| 68  | 1.24864  | -0.63868 | -1.0763  | -0.28812 | 1.14537  | 1.80661  | -0.15262 | 2.16631  | 1.67133  | -0.29882 |
| 69  | -0.05193 | -1.29291 | -0.85556 | -0.03597 | 4.35258  | -0.32884 | 0.79375  | 0.41554  | 0.35156  | -0.57719 |
| 70  | -0.1198  | -0.05545 | 0.43083  | -0.10895 | -2.37948 | -0.49303 | 2.75568  | -0.04768 | -2.33369 | -0.07613 |
| 71  | 0.21686  | 1.41341  | -0.42804 | -2.70242 | 0.61944  | -0.06463 | -0.43017 | -0.1903  | -2.0755  | -0.07512 |
| 72  | -0.02184 | 0.4818   | 0.88869  | 4.23426  | 0.32367  | -0.16813 | -0.0971  | -0.93254 | 0.52898  | 0.05603  |
| 73  | -0.15532 | 0.52294  | -1.60824 | -0.18328 | 3.23327  | 0.1135   | -1.45172 | -0.0493  | -0.23632 | -0.25873 |
| 74  | -0.05867 | -2.59416 | 0.8312   | 0.31796  | -0.30826 | 0.2373   | 2.77116  | 0.02446  | 0.55313  | -0.22707 |
| 75  | -0.51455 | 1.30626  | -0.77779 | 2.71737  | 0.5061   | -1.37105 | 0.15878  | 0.48821  | 0.25587  | 1.37404  |
| 76  | -0.12972 | 0.16455  | 2.0636   | 0.25143  | 0.28128  | 2.95291  | 0.08582  | 0.09976  | -0.25123 | -0.30165 |
| 77  | -0.04632 | -1.3975  | -0.14369 | 0.86661  | 0.47791  | -3.43773 | -0.08283 | 0.40818  | -0.40507 | 1.10291  |
| 78  | -0.0182  | -0.15582 | 1.34311  | 0.06751  | -0.14121 | -0.53351 | 0.03831  | 0.5043   | 2.10788  | -0.58678 |
| 79  | 0.39477  | -3.04864 | -0.45931 | 2.01978  | 0.04376  | -0.52536 | -0.31599 | -0.12492 | -0.10121 | 0.11121  |
| 80  | -0.03452 | -0.03057 | -0.09894 | 0.03226  | 0.01192  | -0.48283 | -4.65939 | -0.11003 | -0.3613  | 0.079    |
| 81  | -1.31754 | 0.0574   | -0.2971  | 0.08453  | -0.30721 | 0.86434  | 2.45968  | 2.51238  | 1.23547  | -0.50231 |
| 82  | 1.50294  | -0.24216 | 2.28121  | -0.90775 | 1.04895  | 0.21859  | -2.06629 | 0.08329  | 0.34822  | 0.1581   |
| 83  | 0.36466  | 0.70677  | -0.918   | 2.47546  | 3.79165  | -0.24031 | -0.37853 | -0.32197 | -0.37636 | -0.02901 |
| 84  | -0.12627 | -0.04525 | -0.82218 | -2.26543 | -0.84347 | -1.03939 | 0.26653  | -2.47079 | 1.23837  | 0.75649  |
| 85  | 0.24064  | -0.62543 | -2.94637 | 0.47164  | 1.58428  | 0.23203  | -0.59003 | 0.3325   | -0.13853 | 0.14422  |
| 86  | -0.17463 | -0.12583 | 0.85598  | -0.17722 | 1.54264  | 0.05216  | -0.08885 | 0.88151  | 0.46255  | -0.48946 |
| 87  | -0.06676 | 0.36942  | 0.58086  | 0.11589  | -1.18536 | -0.12857 | -2.80603 | 0.00658  | 1.06034  | -0.58457 |
| 88  | -0.1561  | -0.18171 | -0.63499 | 2.6653   | 0.46641  | 0.59787  | 0.30223  | -0.53096 | 1.29943  | -0.98973 |
| 89  | -0.00976 | 1.04356  | 0.16697  | 1.99034  | 0.23572  | 2.65679  | -0.00325 | 1.03831  | -0.0701  | -0.77931 |
| 90  | 0.13381  | 0.49307  | 0.04619  | -1.68299 | 0.19455  | 0.52731  | 0.14125  | 0.99717  | 0.43132  | -0.97925 |
| 91  | -0.1563  | 1.8791   | 0.19313  | 0.23078  | 0.30029  | 1.5847   | 0.36806  | -0.91846 | -0.10561 | 2.76858  |
| 92  | -0.03141 | -0.27603 | -1.25022 | 0.76867  | 0.37696  | -1.86725 | 0.45545  | 2.35843  | 0.57733  | -0.51505 |
| 93  | 0.19081  | -0.02002 | 0.89507  | -0.17124 | 0.07392  | 0.07283  | -0.47206 | -0.03043 | -0.34827 | -4.29424 |
| 94  | -0.29    | 0.04272  | -1.72027 | -0.11905 | 0.27361  | -1.60983 | 0.1195   | -0.37867 | 3.50315  | 0.36605  |
| 95  | -0.1328  | 4.18273  | 0.04841  | 0.61437  | 0.60526  | -0.23885 | -0.05236 | -0.82974 | -0.29851 | 0.51121  |
| 96  | 0.03186  | -0.24892 | 0.25614  | -0.17325 | 3.12521  | -0.12148 | 0.07243  | -0.3332  | -0.04557 | -0.77374 |
| 97  | -0.02997 | 2.20189  | 0.2339   | 0.05403  | 3.47503  | -0.14415 | 0.23471  | 0.75477  | 0.00188  | 0.06228  |
| 98  | -1.11961 | 0.43678  | 0.61581  | -3.07064 | -0.02222 | -0.20063 | -0.19664 | -0.1743  | 1.29354  | 0.31722  |
| 99  | 0.80714  | -0.31437 | 3.21672  | 0.05728  | 0.42811  | -0.01624 | 0.4669   | -0.14758 | 0.79094  | -0.49991 |
| 100 | -0.04883 | 0.18854  | -0.0422  | -0.24325 | 0.33396  | 0.58096  | 0.07173  | 0.18625  | 4.80098  | -0.04774 |
| 101 | 0.12902  | 0.04455  | -0.58451 | 0.22529  | 0.18288  | -3.31484 | 0.20874  | 0.1439   | -2.33226 | -0.00004 |
| 102 | 1.4006   | 2.53879  | 2.23308  | -0.40397 | -0.91191 | 0.30424  | -0.59309 | -0.28013 | -0.53681 | -0.72096 |
| 103 | 0.14981  | -0.34273 | -0.1096  | -2.08103 | -0.55452 | 1.22105  | -2.32346 | 0.71767  | -0.04973 | 0.88489  |
| 104 | -0.06296 | 0.00779  | 0.37141  | -0.07311 | -0.43599 | -0.69232 | 0.37183  | 0.59265  | -0.20391 | 0.76627  |
| 105 | -0.04811 | 0.81816  | -0.06811 | -3.60488 | -0.30926 | 1.2619   | -0.25378 | -2.67757 | 0.01421  | 0.15661  |
| 106 | -0.18735 | -0.29123 | -0.1443  | -0.17479 | -0.2036  | 1.97977  | -0.22341 | -0.17369 | -2.54577 | -0.41017 |
| 107 | -0.15575 | 0.64521  | -0.20741 | -0.06303 | 2.55173  | 0.08849  | -1.3168  | -0.0852  | -3.59554 | -0.16002 |
| 108 | 0.2538   | 0.42032  | -2.50621 | 0.13451  | -0.06898 | 2.41402  | 0.09969  | -0.09772 | 1.45985  | 0.45849  |
| 109 | 0.19057  | -0.22542 | -0.16274 | -0.19255 | -0.085   | 0.1524   | -0.06344 | 3.54373  | -0.19634 | 0.42005  |
| 110 | -0.00046 | 1.3075   | 0.44001  | 0.29739  | 0.12153  | 0.45681  | -0.24122 | 3.97823  | 1.01631  | -0.5718  |
| 111 | -1.0474  | 0.76204  | 0.79817  | 0.02734  | -0.85256 | 0.27437  | -0.36512 | 0.45476  | -0.34809 | -0.26607 |
| 112 | 0.39011  | -0.04813 | 0.28325  | -0.40521 | -0.26705 | 0.11646  | -1.00205 | -0.09658 | 0.39184  | 6.38045  |
| 113 | 0.18221  | -3.1436  | 0.56642  | -0.41265 | 2.10605  | 0.00959  | 0.32737  | -0.30618 | -0.46143 | 0.62372  |
| 114 | -3.41695 | -0.29748 | 0.93423  | 0.2968   | -0.22186 | -0.04393 | 0.4199   | -0.02121 | -0.0349  | 0.24145  |
| 115 | 0.08292  | -0.34821 | -0.98374 | 0.76693  | -2.30082 | 0.11379  | -2.00866 | 0.10299  | -1.80156 | 0.49759  |
| 116 | 0.09801  | 0.02398  | 0.87932  | -0.29853 | -0.23085 | 0.60271  | 0.283    | 0.17743  | -0.0655  | 0.11407  |
| 117 | 0.0495   | -2.7049  | 0.93988  | -0.84697 | -0.47967 | -0.45333 | -0.34552 | -0.60481 | 0.77467  | 1.79495  |
| 118 | 2.31525  | -1.35283 | 1.51663  | 0.46045  | -0.94423 | 0.66598  | 0.14087  | -1.23472 | 0.07825  | 0.23723  |
| 119 | -0.01891 | 0.01146  | 0.05543  | -0.00939 | -0.25309 | -0.05408 | -0.08464 | -0.08331 | 0.28796  | -0.23134 |
| 120 | 0.14456  | -1.82728 | 0.54162  | -1.45394 | 0.65876  | 2.88044  | 0.04801  | 1.52839  | 0.4177   | 0.24366  |
| 121 | 0.00192  | -1.51934 | -0.28914 | 0.27953  | -0.43474 | 0.3359   | 0.34038  | 0.09476  | -0.16812 | 0.55648  |

|     |          |          |          |         |          |          |          |          |          |          |
|-----|----------|----------|----------|---------|----------|----------|----------|----------|----------|----------|
| 122 | 1.58963  | 0.62054  | -1.19708 | 0.95618 | -0.19293 | 0.13055  | -0.46467 | -1.08598 | 1.0294   | -0.00537 |
| 123 | -1.63945 | 1.27057  | 1.17362  | 0.77949 | -0.25979 | 1.31919  | 1.09939  | -0.59425 | -1.04889 | 0.33754  |
| 124 | -0.36973 | -1.0016  | 0.18249  | 0.95907 | 0.72926  | 2.32241  | 0.60848  | -2.12186 | -0.24255 | -0.65101 |
| 125 | -0.00988 | -0.04139 | -0.76083 | 0.14922 | 0.57784  | 3.97886  | -0.30668 | -0.34279 | -0.84522 | 0.11366  |
| 126 | -0.12785 | 1.334    | 0.11998  | 0.17243 | -1.82888 | 0.49492  | 0.09101  | -0.99747 | 0.39561  | -1.90151 |
| 127 | 0.1718   | 1.22115  | 0.41649  | 2.03242 | -0.20785 | -2.41295 | -0.75    | 1.09243  | 0.65831  | -2.08726 |

4.A.5 Tables for ER BSAC

Table 4.A.29 – cband\_si\_type parameters

| cband_si_type | max_cband_si_len | Largest cband_si      |             | Model listed in       |              |
|---------------|------------------|-----------------------|-------------|-----------------------|--------------|
|               |                  | 0 <sup>th</sup> cband | Other cband | 0 <sup>th</sup> cband | Other cband  |
| 0             | 6                | 6                     | 4           | Table 4.A.49          | Table 4.A.42 |
| 1             | 5                | 6                     | 6           | Table 4.A.49          | Table 4.A.43 |
| 2             | 6                | 8                     | 4           | Table 4.A.49          | Table 4.A.42 |
| 3             | 5                | 8                     | 6           | Table 4.A.49          | Table 4.A.43 |
| 4             | 6                | 8                     | 8           | Table 4.A.49          | Table 4.A.44 |
| 5             | 6                | 10                    | 4           | Table 4.A.49          | Table 4.A.42 |
| 6             | 5                | 10                    | 6           | Table 4.A.49          | Table 4.A.43 |
| 7             | 6                | 10                    | 8           | Table 4.A.49          | Table 4.A.44 |
| 8             | 5                | 10                    | 10          | Table 4.A.49          | Table 4.A.45 |
| 9             | 6                | 12                    | 4           | Table 4.A.49          | Table 4.A.42 |
| 10            | 5                | 12                    | 6           | Table 4.A.49          | Table 4.A.43 |
| 11            | 6                | 12                    | 8           | Table 4.A.49          | Table 4.A.44 |
| 12            | 8                | 12                    | 12          | Table 4.A.49          | Table 4.A.46 |
| 13            | 6                | 14                    | 4           | Table 4.A.49          | Table 4.A.42 |
| 14            | 5                | 14                    | 6           | Table 4.A.49          | Table 4.A.43 |
| 15            | 6                | 14                    | 8           | Table 4.A.49          | Table 4.A.44 |
| 16            | 8                | 14                    | 12          | Table 4.A.49          | Table 4.A.46 |
| 17            | 9                | 14                    | 14          | Table 4.A.49          | Table 4.A.47 |
| 18            | 6                | 15                    | 4           | Table 4.A.49          | Table 4.A.42 |
| 19            | 5                | 15                    | 6           | Table 4.A.49          | Table 4.A.43 |
| 20            | 6                | 15                    | 8           | Table 4.A.49          | Table 4.A.44 |
| 21            | 8                | 15                    | 12          | Table 4.A.49          | Table 4.A.46 |
| 22            | 10               | 15                    | 15          | Table 4.A.49          | Table 4.A.48 |
| 23            | 8                | 16                    | 12          | Table 4.A.49          | Table 4.A.46 |
| 24            | 10               | 16                    | 16          | Table 4.A.49          | Table 4.A.48 |
| 25            | 9                | 17                    | 14          | Table 4.A.49          | Table 4.A.47 |
| 26            | 10               | 17                    | 17          | Table 4.A.49          | Table 4.A.48 |
| 27            | 10               | 18                    | 18          | Table 4.A.49          | Table 4.A.48 |
| 28            | 12               | 19                    | 19          | Table 4.A.49          | Table 4.A.48 |
| 29            | 12               | 20                    | 20          | Table 4.A.49          | Table 4.A.48 |
| 30            | 12               | 21                    | 21          | Table 4.A.49          | Table 4.A.48 |
| 31            | 12               | 22                    | 22          | Table 4.A.49          | Table 4.A.48 |

**Table 4.A.30 – Scalefactor model parameters**

| scf_model | Largest Differential ArModel | Model listed in |
|-----------|------------------------------|-----------------|
| 0         | 0                            | not used        |
| 1         | 3                            | Table 4.A.35    |
| 2         | 7                            | Table 4.A.36    |
| 3         | 15                           | Table 4.A.37    |
| 4         | 15                           | Table 4.A.38    |
| 5         | 31                           | Table 4.A.39    |
| 6         | 31                           | Table 4.A.40    |
| 7         | 63                           | Table 4.A.41    |

**Table 4.A.31 – BSAC cband\_si parameters**

| cband_si | MSB plane | Table listed in | cband_si | MSB plane | Table listed in |
|----------|-----------|-----------------|----------|-----------|-----------------|
| 0        | 0         | -               | 12       | 6         | Table 4.A.65    |
| 1        | 1         | Table 4.A.54    | 13       | 7         | Table 4.A.66    |
| 2        | 1         | Table 4.A.55    | 14       | 7         | Table 4.A.67    |
| 3        | 2         | Table 4.A.56    | 15       | 8         | Table 4.A.68    |
| 4        | 2         | Table 4.A.57    | 16       | 9         | Table 4.A.69    |
| 5        | 3         | Table 4.A.58    | 17       | 10        | Table 4.A.70    |
| 6        | 3         | Table 4.A.59    | 18       | 11        | Table 4.A.71    |
| 7        | 4         | Table 4.A.60    | 19       | 12        | Table 4.A.72    |
| 8        | 4         | Table 4.A.61    | 20       | 13        | Table 4.A.73    |
| 9        | 5         | Table 4.A.62    | 21       | 14        | Table 4.A.74    |
| 10       | 5         | Table 4.A.63    | 22       | 15        | Table 4.A.75    |
| 11       | 6         | Table 4.A.64    |          |           |                 |

**Table 4.A.32 – Position of probability value in probability table**

|   |   |   |   | h | 0  | 0  | 0  | 0  | 0  | 0  | 0  | 0  | 1  | 1  | 1  | 1  | 1  | 1  | 1  |   |  |
|---|---|---|---|---|----|----|----|----|----|----|----|----|----|----|----|----|----|----|----|---|--|
|   |   |   |   | g | 0  | 0  | 0  | 0  | 1  | 1  | 1  | 1  | 0  | 0  | 0  | 0  | 1  | 1  | 1  | 1 |  |
|   |   |   |   | f | 0  | 0  | 1  | 1  | 0  | 0  | 1  | 1  | 0  | 0  | 1  | 1  | 0  | 0  | 1  | 1 |  |
|   |   |   |   | e | 0  | 1  | 0  | 1  | 0  | 1  | 0  | 1  | 0  | 1  | 0  | 1  | 0  | 1  | 0  | 1 |  |
| a | b | c | d |   |    |    |    |    |    |    |    |    |    |    |    |    |    |    |    |   |  |
| 0 | x | x | x |   | 0  | 15 | 22 | 29 | 32 | 39 | 42 | 45 |    |    |    |    |    |    |    |   |  |
| 1 | x | x | 0 |   | 1  | 16 | 23 | 30 |    |    |    |    | 46 | 53 | 56 | 59 |    |    |    |   |  |
|   | x | x | 1 |   | 2  | 17 | 24 | 31 |    |    |    |    | 46 | 53 | 56 | 59 |    |    |    |   |  |
| 2 | x | 0 | 0 |   | 3  | 18 |    |    | 33 | 40 |    |    | 47 | 54 |    |    | 60 | 63 |    |   |  |
|   | x | 0 | 1 |   | 4  | 19 |    |    | 33 | 40 |    |    | 48 | 55 |    |    | 60 | 63 |    |   |  |
|   | x | 1 | 0 |   | 5  | 20 |    |    | 34 | 41 |    |    | 47 | 54 |    |    | 60 | 63 |    |   |  |
|   | x | 1 | 1 |   | 6  | 21 |    |    | 34 | 41 |    |    | 48 | 55 |    |    | 60 | 63 |    |   |  |
| 3 | 0 | 0 | 0 |   | 7  |    | 25 |    | 35 |    | 43 |    | 49 |    | 57 |    | 61 |    | 64 |   |  |
|   | 0 | 0 | 1 |   | 8  |    | 25 |    | 36 |    | 43 |    | 50 |    | 57 |    | 62 |    | 64 |   |  |
|   | 0 | 1 | 0 |   | 9  |    | 26 |    | 35 |    | 43 |    | 51 |    | 58 |    | 61 |    | 64 |   |  |
|   | 0 | 1 | 1 |   | 10 |    | 26 |    | 36 |    | 43 |    | 52 |    | 58 |    | 62 |    | 64 |   |  |
|   | 1 | 0 | 0 |   | 11 |    | 27 |    | 37 |    | 44 |    | 49 |    | 57 |    | 61 |    | 64 |   |  |
|   | 1 | 0 | 1 |   | 12 |    | 27 |    | 38 |    | 44 |    | 50 |    | 57 |    | 62 |    | 64 |   |  |
|   | 1 | 1 | 0 |   | 13 |    | 28 |    | 37 |    | 44 |    | 51 |    | 58 |    | 61 |    | 64 |   |  |
|   | 1 | 1 | 1 |   | 14 |    | 28 |    | 38 |    | 44 |    | 52 |    | 58 |    | 62 |    | 64 |   |  |

where, i = spectral index

$$a = i \% 4$$

b = the sliced bit of (i-3)th spectral data whose significance is same with that of i-th spectral data

c = the sliced bit of (i-2)th spectral data whose significance is same with that of i-th spectral data

d = the sliced bit of (i-1)th spectral data whose significance is same with that of i-th spectral data

e = whether the higher bits of the (i-a+3)th spectral data whose significance is larger than that of i-th spectral data is nonzero (1) or zero(0)

f = whether the higher bits of the (i-a+2)th spectral data whose significance is larger than that of i-th spectral data is nonzero (1) or zero(0)

g = whether the higher bits of the (i-a+1)th spectral data whose significance is larger than that of i-th spectral data is nonzero (1) or zero(0)

h = whether the higher bits of the (i-a)th spectral data whose significance is larger than that of i-th spectral data is nonzero (1) or zero(0)

**Table 4.A.33 – The minimum probability(min\_p0) in proportion to the available length of the layer**

|                      |      |      |     |     |     |     |    |    |    |    |    |    |    |
|----------------------|------|------|-----|-----|-----|-----|----|----|----|----|----|----|----|
| Available length     | 1    | 2    | 3   | 4   | 5   | 6   | 7  | 8  | 9  | 10 | 11 | 12 | 13 |
| min_p0 (hexadecimal) | 2000 | 1000 | 800 | 400 | 200 | 100 | 80 | 40 | 20 | 10 | 8  | 4  | 2  |

**Table 4.A.34 – The maximum probability(max\_p0) in proportion to the available length of the layer**

|                      |   |   |   |    |    |    |    |     |     |     |     |      |      |
|----------------------|---|---|---|----|----|----|----|-----|-----|-----|-----|------|------|
| Available length     | 1 | 2 | 3 | 4  | 5  | 6  | 7  | 8   | 9   | 10  | 11  | 12   | 13   |
| max_p0 (hexadecimal) | 2 | 4 | 8 | 10 | 20 | 40 | 80 | 100 | 200 | 400 | 800 | 1000 | 2000 |

**Table 4.A.35 – Scalefactor arithmetic model 1**

|      |                                      |      |      |    |  |  |  |  |  |  |  |  |  |
|------|--------------------------------------|------|------|----|--|--|--|--|--|--|--|--|--|
| size | cumulative frequencies (hexadecimal) |      |      |    |  |  |  |  |  |  |  |  |  |
| 4    | 752,                                 | 3cd, | 14d, | 0, |  |  |  |  |  |  |  |  |  |

**Table 4.A.36 – Scalefactor arithmetic model 2**

|      |                                      |      |      |      |      |      |     |    |  |  |  |  |  |
|------|--------------------------------------|------|------|------|------|------|-----|----|--|--|--|--|--|
| size | cumulative frequencies (hexadecimal) |      |      |      |      |      |     |    |  |  |  |  |  |
| 8    | 112f,                                | de7, | a8b, | 7c1, | 47a, | 23a, | d4, | 0, |  |  |  |  |  |

**Table 4.A.37 – Scalefactor arithmetic model 3**

|      |                                      |       |       |       |       |      |      |      |      |      |     |     |     |     |    |    |
|------|--------------------------------------|-------|-------|-------|-------|------|------|------|------|------|-----|-----|-----|-----|----|----|
| size | cumulative frequencies (hexadecimal) |       |       |       |       |      |      |      |      |      |     |     |     |     |    |    |
| 16   | 1f67,                                | 1c5f, | 18d8, | 1555, | 1215, | eb4, | adc, | 742, | 408, | 1e6, | df, | 52, | 32, | 23, | c, | 0, |

**Table 4.A.38 – Scalefactor arithmetic model 4**

|      |                                      |       |       |       |       |       |       |       |      |      |      |      |      |     |     |    |
|------|--------------------------------------|-------|-------|-------|-------|-------|-------|-------|------|------|------|------|------|-----|-----|----|
| size | cumulative frequencies (hexadecimal) |       |       |       |       |       |       |       |      |      |      |      |      |     |     |    |
| 16   | 250f,                                | 22b8, | 2053, | 1deb, | 1b05, | 186d, | 15df, | 12d9, | f77, | c01, | 833, | 50d, | 245, | 8c, | 33, | 0, |

**Table 4.A.39 – Scalefactor arithmetic model 5**

| size | cumulative frequencies (hexadecimal) |      |      |      |      |      |      |      |
|------|--------------------------------------|------|------|------|------|------|------|------|
| 32   | 8a8,                                 | 74e, | 639, | 588, | 48c, | 3cf, | 32e, | 272, |
|      | 1bc,                                 | 13e, | e4,  | 97,  | 69,  | 43,  | 2f,  | 29,  |
|      | 20,                                  | 1b,  | 18,  | 15,  | 12,  | f,   | d,   | c,   |
|      | a,                                   | 9,   | 7,   | 6,   | 4,   | 3,   | 1,   | 0,   |

**Table 4.A.40 – Scalefactor arithmetic model 6**

| size | cumulative frequencies (hexadecimal) |      |      |      |      |      |      |      |
|------|--------------------------------------|------|------|------|------|------|------|------|
| 32   | c2a,                                 | 99f, | 809, | 6ec, | 603, | 53d, | 491, | 40e, |
|      | 394,                                 | 30a, | 2a5, | 259, | 202, | 1bc, | 170, | 133, |
|      | 102,                                 | c9,  | 97,  | 73,  | 4f,  | 37,  | 22,  | 16,  |
|      | f,                                   | b,   | 9,   | 7,   | 5,   | 3,   | 1,   | 0,   |

**Table 4.A.41 – Scalefactor arithmetic model 7**

| size | cumulative frequencies (hexadecimal) |       |       |       |       |       |       |       |
|------|--------------------------------------|-------|-------|-------|-------|-------|-------|-------|
| 64   | 3b5e,                                | 3a90, | 39d3, | 387c, | 3702, | 3566, | 33a7, | 321c, |
|      | 2f90,                                | 2cf2, | 29fe, | 26fa, | 23e4, | 20df, | 1e0d, | 1ac4, |
|      | 1804,                                | 159a, | 131e, | 10e7, | e5b,  | c9c,  | b78,  | a21,  |
|      | 8fd,                                 | 7b7,  | 6b5,  | 62c,  | 55d,  | 4f6,  | 4d4,  | 44b,  |
|      | 38e,                                 | 2e2,  | 29d,  | 236,  | 225,  | 1f2,  | 1cf,  | 1ad,  |
|      | 19c,                                 | 179,  | 168,  | 157,  | 146,  | 135,  | 123,  | 112,  |
|      | 101,                                 | f0,   | df,   | ce,   | bc,   | ab,   | 9a,   | 89,   |
|      | 78,                                  | 67,   | 55,   | 44,   | 33,   | 22,   | 11,   | 0,    |

**Table 4.A.42 – cband\_si arithmetic model 0**

| size | cumulative frequencies (hexadecimal) |       |       |       |    |
|------|--------------------------------------|-------|-------|-------|----|
| 5    | 3ef6,                                | 3b59, | 1b12, | 12a3, | 0, |

**Table 4.A.43 – cband\_si arithmetic model 1**

| size | cumulative frequencies (hexadecimal) |       |       |      |      |      |    |
|------|--------------------------------------|-------|-------|------|------|------|----|
| 7    | 3d51,                                | 33ae, | 1cff, | fb7, | 7e4, | 22b, | 0, |

**Table 4.A.44 – cband\_si arithmetic model 2**

| size | cumulative frequencies (hexadecimal) |       |       |       |      |      |      |      |
|------|--------------------------------------|-------|-------|-------|------|------|------|------|
| 9    | 3a47,                                | 2aec, | 1e05, | 1336, | e7d, | 860, | 5e0, | 44a, |
|      | 0,                                   |       |       |       |      |      |      |      |

**Table 4.A.45 – cband\_si arithmetic model 3**

| size | cumulative frequencies (hexadecimal) |       |       |       |       |      |      |      |
|------|--------------------------------------|-------|-------|-------|-------|------|------|------|
| 11   | 36be,                                | 27ae, | 20f4, | 1749, | 14d5, | d46, | ad3, | 888, |
|      | 519,                                 | 20b,  | 0,    |       |       |      |      |      |



**Table 4.A.46 – cband\_si arithmetic model 4**

| size | cumulative frequencies (hexadecimal) |               |               |              |             |       |       |      |
|------|--------------------------------------|---------------|---------------|--------------|-------------|-------|-------|------|
| 13   | 3983,<br>94c,                        | 2e77,<br>497, | 2b03,<br>445, | 1ee8,<br>40, | 1df9,<br>0, | 1307, | 11e4, | b4d, |

**Table 4.A.47 – cband\_si arithmetic model 5**

| size | cumulative frequencies (hexadecimal) |               |               |               |               |              |            |      |
|------|--------------------------------------|---------------|---------------|---------------|---------------|--------------|------------|------|
| 15   | 306f,<br>af2,                        | 249e,<br>7a8, | 1f56,<br>71a, | 1843,<br>454, | 161a,<br>413, | 102d,<br>16, | f6c,<br>0, | c81, |

**Table 4.A.48 – cband\_si arithmetic model 6**

| size | cumulative frequencies (hexadecimal) |                      |                      |                     |                    |                    |                    |              |
|------|--------------------------------------|----------------------|----------------------|---------------------|--------------------|--------------------|--------------------|--------------|
| 23   | 31af,<br>955,<br>198,                | 2001,<br>825,<br>77, | 162d,<br>7dd,<br>10, | 127e,<br>6a9,<br>c, | f05,<br>688,<br>8, | c34,<br>55b,<br>4, | b8f,<br>54b,<br>0, | a61,<br>2f7, |

**Table 4.A.49 – cband\_si arithmetic model for 0<sup>th</sup> coding band**

| size | cumulative frequencies (hexadecimal) |                       |                       |                       |                      |                     |                     |              |
|------|--------------------------------------|-----------------------|-----------------------|-----------------------|----------------------|---------------------|---------------------|--------------|
| 23   | 3ff8,<br>3074,<br>30,                | 3ff0,<br>2bcf,<br>28, | 3fe8,<br>231b,<br>20, | 3fe0,<br>13db,<br>18, | 3fd7,<br>d51,<br>10, | 3f31,<br>603,<br>8, | 3cd7,<br>44c,<br>0, | 3bc9,<br>80, |

**Table 4.A.50 – MS\_used model**

| size | Cumulative frequencies (hexadecimal) |    |
|------|--------------------------------------|----|
| 2    | 2000,                                | 0, |

**Table 4.A.51 – stereo\_info model**

| size | Cumulative frequencies(hexadecimal) |       |      |    |
|------|-------------------------------------|-------|------|----|
| 4    | 3666,                               | 1000, | 666, | 0, |

**Table 4.A.52 – noise\_flag arithmetic model**

| size | Cumulative frequencies(hexadecimal) |    |
|------|-------------------------------------|----|
| 2    | 2000,                               | 0, |

**Table 4.A.53 – noise\_mode arithmetic model**

| size | cumulative frequencies(hexadecimal) |       |       |    |
|------|-------------------------------------|-------|-------|----|
| 4    | 3000,                               | 2000, | 1000, | 0, |

**Table 4.A.54 – BSAC probability table 1**

(MSB plane = 1)

| Significance | Probability Value of symbol '0' (Hexadecimal) |                |                |                |                |                |                |                |
|--------------|-----------------------------------------------|----------------|----------------|----------------|----------------|----------------|----------------|----------------|
| 1            | 3900,<br>3000,                                | 3a00,<br>3600, | 2f00,<br>2d00, | 3b00,<br>3900, | 2f00,<br>2f00, | 3700,<br>3700, | 2c00,<br>2c00, | 3b00,<br>3b00, |

**Table 4.A.55 – BSAC probability table 2**

(MSB plane = 1)

| Significance | Probability Value of symbol '0' (Hexadecimal) |                |                |                |                |                |                |                |
|--------------|-----------------------------------------------|----------------|----------------|----------------|----------------|----------------|----------------|----------------|
| 1            | 2800,<br>2700,                                | 2800,<br>2800, | 2500,<br>2400, | 2900,<br>2800, | 2600,<br>2500, | 2700,<br>2600, | 2300,<br>2200, | 2a00,<br>2a00, |

**Table 4.A.56 – BSAC probability table 3**

(MSB plane = 2)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)                                 |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
|--------------|---------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|
| 2            | zero                | 3d00,<br>3200,                                                                | 3d00,<br>3b00,                                                       | 3300,<br>3100,                                                       | 3d00,<br>3e00,                                                       | 3300,<br>3700,                                                       | 3b00,<br>3c00,                                                       | 3300,<br>3300,                                                       | 3d00,<br>3d00,                                                       |
| 1            | zero                | 3700,<br>2500,<br>2c00,<br>1a00,<br>1e00,<br>1300,<br>2300,<br>1400,<br>2200, | 3a00,<br>2b00,<br>1d00,<br>1d00,<br>1f00,<br>1a00,<br>3600,<br>2100, | 2800,<br>2400,<br>2200,<br>1900,<br>1c00,<br>2000,<br>2800,<br>2200, | 3b00,<br>3100,<br>1a00,<br>1c00,<br>2b00,<br>1800,<br>3100,<br>1000, | 2600,<br>2300,<br>1c00,<br>1e00,<br>2400,<br>2300,<br>2500,<br>1e00, | 2c00,<br>2900,<br>1600,<br>2c00,<br>2900,<br>2500,<br>1400,<br>3000, | 2400,<br>2300,<br>2700,<br>2400,<br>2700,<br>1f00,<br>1200,<br>2600, | 3a00,<br>3000,<br>2200,<br>1900,<br>2400,<br>2c00,<br>1800,<br>1200, |
|              | non-zero            | 3100,                                                                         |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |

**Table 4.A.57 – BSAC probability table 4**

(MSB plane = 2)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)                                 |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
|--------------|---------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|
| 2            | zero                | 3900,<br>3000,                                                                | 3a00,<br>3500,                                                       | 2e00,<br>2c00,                                                       | 3a00,<br>3600,                                                       | 2f00,<br>2b00,                                                       | 3400,<br>3100,                                                       | 2a00,<br>2500,                                                       | 3a00,<br>3a00,                                                       |
| 1            | zero                | 1e00,<br>1e00,<br>1a00,<br>1700,<br>1700,<br>1c00,<br>1a00,<br>1400,<br>1400, | 1d00,<br>1e00,<br>1800,<br>1700,<br>1800,<br>1500,<br>1e00,<br>1600, | 1c00,<br>1a00,<br>1800,<br>1800,<br>1900,<br>1600,<br>1800,<br>1500, | 1d00,<br>1e00,<br>1800,<br>1800,<br>1c00,<br>1c00,<br>1c00,<br>1700, | 1c00,<br>1c00,<br>1700,<br>1600,<br>1700,<br>1800,<br>1b00,<br>1600, | 1d00,<br>1d00,<br>1700,<br>1700,<br>1900,<br>1700,<br>1400,<br>1500, | 1b00,<br>1b00,<br>1800,<br>1600,<br>1700,<br>1700,<br>1700,<br>1800, | 1d00,<br>1a00,<br>1a00,<br>1500,<br>1500,<br>1a00,<br>1500,<br>1400, |
|              | non-zero            | 3600,                                                                         |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |

**Table 4.A.58 – BSAC probability table 5**

(MSB plane = 3)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)                                 |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
|--------------|---------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|
| 3            | zero                | 3d00,<br>3500,                                                                | 3d00,<br>3c00,                                                       | 3200,<br>3500,                                                       | 3d00,<br>3f00,                                                       | 3300,<br>3b00,                                                       | 3d00,<br>3f00,                                                       | 3600,<br>3d00,                                                       | 3d00,                                                                |
| 2            | zero                | 3c00,<br>2b00,<br>3400,<br>1a00,<br>2600,<br>1800,<br>3000,<br>1900,<br>3100, | 3d00,<br>3400,<br>2400,<br>2a00,<br>2500,<br>1600,<br>3c00,<br>2900, | 2b00,<br>2b00,<br>2a00,<br>2200,<br>2700,<br>2900,<br>3300,<br>2a00, | 3d00,<br>3800,<br>1c00,<br>2b00,<br>3500,<br>2500,<br>3b00,<br>2400, | 2900,<br>2b00,<br>1f00,<br>2a00,<br>2d00,<br>3100,<br>3400,<br>2700, | 3500,<br>3700,<br>1600,<br>3500,<br>3800,<br>2c00,<br>1700,<br>3c00, | 2c00,<br>2a00,<br>3500,<br>2600,<br>3200,<br>2300,<br>1a00,<br>3600, | 3d00,<br>3900,<br>2500,<br>1a00,<br>2e00,<br>3600,<br>1c00,<br>1d00, |
|              | non-zero            | 3100,                                                                         |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
| 1            | zero                | 3400,<br>2500,<br>2300,<br>1b00,<br>1900,<br>1200,<br>1e00,<br>1300,<br>1a00, | 3800,<br>2d00,<br>1a00,<br>1c00,<br>1b00,<br>1400,<br>3000,<br>1e00, | 2700,<br>2000,<br>1a00,<br>1b00,<br>1a00,<br>1a00,<br>2900,<br>1f00, | 3900,<br>3300,<br>1b00,<br>1a00,<br>1d00,<br>1300,<br>2d00,<br>1100, | 2700,<br>2000,<br>1800,<br>1800,<br>1e00,<br>1c00,<br>2500,<br>1900, | 2f00,<br>2900,<br>1700,<br>1d00,<br>1f00,<br>1b00,<br>1300,<br>2100, | 2200,<br>1e00,<br>1e00,<br>1b00,<br>1b00,<br>1900,<br>1700,<br>1e00, | 3800,<br>2b00,<br>1c00,<br>1800,<br>1e00,<br>2000,<br>1400,<br>1500, |
|              | non-zero            | 2a00,                                                                         | 2b00,                                                                | 2800,                                                                |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |

**Table 4.A.59 – BSAC probability table 6**

(MSB plane = 3)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)                                 |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
|--------------|---------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|
| 3            | zero                | 3800,<br>2d00,                                                                | 3a00,<br>3600,                                                       | 2d00,<br>2b00,                                                       | 3a00,<br>3a00,                                                       | 2d00,<br>2800,                                                       | 3600,<br>3600,                                                       | 2d00,<br>2700,                                                       | 3a00,                                                                |
| 2            | zero                | 2b00,<br>2500,<br>2900,<br>1f00,<br>1d00,<br>1800,<br>2800,<br>1c00,<br>1a00, | 3000,<br>2b00,<br>2300,<br>1d00,<br>1f00,<br>1a00,<br>2f00,<br>1e00, | 2500,<br>2400,<br>2200,<br>2200,<br>1f00,<br>1d00,<br>2300,<br>2100, | 2f00,<br>2d00,<br>1e00,<br>1b00,<br>2900,<br>2000,<br>2f00,<br>1700, | 2600,<br>2500,<br>1b00,<br>1800,<br>2600,<br>1c00,<br>2600,<br>2200, | 2d00,<br>2800,<br>1900,<br>2100,<br>2a00,<br>1a00,<br>1d00,<br>2300, | 2400,<br>2500,<br>2600,<br>2100,<br>2100,<br>1e00,<br>1700,<br>2300, | 3000,<br>2a00,<br>2300,<br>1d00,<br>2300,<br>2900,<br>1d00,<br>1400, |
|              | non-zero            | 3000,                                                                         |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
| 1            | zero                | 1900,<br>1900,<br>1700,<br>1600,<br>1300,<br>1400,<br>1600,<br>1300,<br>1300, | 1900,<br>1600,<br>1500,<br>1600,<br>1600,<br>1400,<br>1f00,<br>1400, | 1900,<br>1800,<br>1500,<br>1200,<br>1600,<br>1500,<br>1a00,<br>1300, | 1b00,<br>1e00,<br>1500,<br>1300,<br>1c00,<br>1400,<br>1e00,<br>1100, | 1700,<br>1900,<br>1700,<br>1200,<br>1400,<br>1300,<br>1800,<br>1500, | 1b00,<br>1a00,<br>1400,<br>1600,<br>1700,<br>1300,<br>1700,<br>1600, | 1a00,<br>1700,<br>1900,<br>1500,<br>1600,<br>1500,<br>1600,<br>1500, | 1000,<br>1b00,<br>1700,<br>1500,<br>1400,<br>1800,<br>1600,<br>1200, |
|              | non-zero            | 2b00,                                                                         | 2800,                                                                | 2700,                                                                |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |

Table 4.A.60 – BSAC probability table 7

(MSB plane = 4)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)                                 |                                                                                        |                                                                               |                                                                               |                                                                      |                                                                      |                                                                      |                                                                      |
|--------------|---------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------------------------|-------------------------------------------------------------------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|
| MSB          | zero                | 3d00,<br>3200,                                                                | 3d00,<br>3f00,                                                                         | 3500,<br>3a00,                                                                | 3e00,<br>3f00,                                                                | 3500,<br>3d00,                                                       | 3f00,<br>3f00,                                                       | 3b00,<br>3b00,                                                       | 3e00,                                                                |
| MSB-1        | zero                | 3f00,<br>2d00,<br>3900,<br>1b00,<br>2800,<br>1a00,<br>3800,<br>1800,<br>3500, | 3f00,<br>3c00,<br>2600,<br>2600,<br>2f00,<br>3300,<br>3f00,<br>3b00,<br>3b00,<br>3a00, | 3200,<br>3000,<br>2f00,<br>2300,<br>2500,<br>2500,<br>2800,<br>3b00,<br>3a00, | 3f00,<br>3f00,<br>1e00,<br>3a00,<br>3e00,<br>2800,<br>3f00,<br>3f00,<br>1200, | 3500,<br>3700,<br>2400,<br>3900,<br>3700,<br>3c00,<br>3a00,<br>2f00, | 3e00,<br>3e00,<br>1500,<br>3e00,<br>3e00,<br>3800,<br>1e00,<br>3f00, | 3700,<br>3400,<br>3700,<br>2b00,<br>3d00,<br>2c00,<br>1b00,<br>3b00, | 3f00,<br>3f00,<br>3100,<br>2200,<br>3900,<br>3d00,<br>1800,<br>1b00, |
|              | non-zero            | 2100,                                                                         |                                                                                        |                                                                               |                                                                               |                                                                      |                                                                      |                                                                      |                                                                      |
| MSB-2        | zero                | 3c00,<br>2c00,<br>3100,<br>2800,<br>2100,<br>1800,<br>2b00,<br>1900,<br>2b00, | 3e00,<br>3900,<br>2100,<br>2400,<br>2b00,<br>1800,<br>3e00,<br>3400,                   | 3000,<br>2e00,<br>2c00,<br>2200,<br>2700,<br>1f00,<br>3d00,<br>3500,          | 3e00,<br>3c00,<br>2600,<br>2100,<br>3200,<br>1e00,<br>3d00,<br>1c00,          | 3100,<br>2d00,<br>2800,<br>2300,<br>2d00,<br>2e00,<br>3a00,<br>2600, | 3a00,<br>3c00,<br>1d00,<br>2d00,<br>3400,<br>2a00,<br>1e00,<br>3300, | 3100,<br>3100,<br>2b00,<br>2500,<br>2a00,<br>2400,<br>2b00,<br>2a00, | 3d00,<br>3d00,<br>2800,<br>1f00,<br>3500,<br>3000,<br>2600,<br>1c00, |
|              | non-zero            | 2800,<br>2b00,                                                                | 2900,<br>2400,                                                                         |                                                                               |                                                                               |                                                                      |                                                                      |                                                                      |                                                                      |
| others       | zero                | 3500,<br>2600,<br>2700,<br>1b00,<br>1e00,<br>1b00,<br>2200,<br>1900,<br>1b00, | 3b00,<br>2f00,<br>1c00,<br>1d00,<br>2400,<br>1500,<br>3700,<br>2500,                   | 2900,<br>2400,<br>2400,<br>2000,<br>2100,<br>1b00,<br>2f00,<br>2300,          | 3b00,<br>3400,<br>1c00,<br>1b00,<br>2b00,<br>1400,<br>3200,<br>1500,          | 2a00,<br>2300,<br>1c00,<br>1a00,<br>2100,<br>1a00,<br>2a00,<br>1900, | 3100,<br>2d00,<br>1900,<br>2300,<br>2800,<br>1a00,<br>1700,<br>2500, | 2700,<br>2000,<br>2700,<br>1d00,<br>2000,<br>2000,<br>1700,<br>2200, | 3b00,<br>3300,<br>2800,<br>1700,<br>2300,<br>2a00,<br>1600,<br>1400, |
|              | non-zero            | 2d00,<br>2400,                                                                | 2500,<br>2300,                                                                         | 2300,<br>2500,                                                                | 2500,<br>2500,                                                                | 2600,<br>2400,                                                       |                                                                      |                                                                      |                                                                      |

Table 4.A.61 – BSAC probability table 8

(MSB plane = 4)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)                                 |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
|--------------|---------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|
| MSB          | zero                | 3b00,<br>3200,                                                                | 3c00,<br>3a00,                                                       | 3400,<br>3100,                                                       | 3c00,<br>3c00,                                                       | 3400,<br>3000,                                                       | 3a00,<br>3900,                                                       | 3000,<br>2f00,                                                       | 3c00,                                                                |
| MSB-1        | zero                | 3500,<br>2e00,<br>3100,<br>2000,<br>2500,<br>1e00,<br>2c00,<br>1b00,<br>2400, | 3800,<br>3400,<br>2600,<br>2600,<br>2400,<br>1c00,<br>3700,<br>2900, | 2c00,<br>2d00,<br>2900,<br>2300,<br>2400,<br>2500,<br>2b00,<br>2a00, | 3900,<br>3600,<br>2000,<br>2500,<br>3000,<br>1d00,<br>3400,<br>1d00, | 2c00,<br>2a00,<br>2300,<br>2100,<br>2800,<br>2300,<br>2c00,<br>2600, | 3400,<br>3300,<br>1f00,<br>2c00,<br>3000,<br>2300,<br>1e00,<br>3200, | 2b00,<br>2800,<br>2d00,<br>2400,<br>2900,<br>2500,<br>1c00,<br>2a00, | 3800,<br>3100,<br>2600,<br>1d00,<br>2200,<br>3300,<br>2100,<br>2000, |
|              | non-zero            | 3200,                                                                         |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |

|        |          |                                                                               |                                                                      |                                                                               |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
|--------|----------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|-------------------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|----------------------------------------------------------------------|
| MSB-2  | zero     | 2900,<br>2500,<br>2800,<br>1c00,<br>1e00,<br>1a00,<br>2500,<br>1a00,<br>1c00, | 2e00,<br>2b00,<br>2100,<br>2100,<br>2100,<br>1a00,<br>2d00,<br>1b00, | 2600,<br>2600,<br>2400,<br>2200,<br>2100,<br>2100,<br>2700,<br>1d00,          | 2f00,<br>2f00,<br>2000,<br>1d00,<br>2900,<br>2100,<br>2a00,<br>1800, | 2600,<br>2300,<br>2000,<br>1c00,<br>2200,<br>1c00,<br>2300,<br>2000, | 2d00,<br>2a00,<br>1b00,<br>1f00,<br>2300,<br>1c00,<br>1c00,<br>2300, | 2600,<br>2300,<br>2400,<br>1c00,<br>2100,<br>1f00,<br>1d00,<br>1f00, | 2e00,<br>2800,<br>1f00,<br>1900,<br>1c00,<br>2700,<br>1a00,<br>1900, |
|        | non-zero | 2b00,                                                                         | 2900,                                                                | 2800,                                                                         |                                                                      |                                                                      |                                                                      |                                                                      |                                                                      |
| others | zero     | 1c00,<br>1f00,<br>1a00,<br>1900,<br>1600,<br>1500,<br>1a00,<br>1400,<br>1400, | 1e00,<br>1f00,<br>1900,<br>1700,<br>1800,<br>1500,<br>2300,<br>1800, | 1b00,<br>1900,<br>1800,<br>1800,<br>1a00,<br>1600,<br>1c00,<br>1500,<br>1500, | 1e00,<br>2000,<br>1900,<br>1700,<br>1c00,<br>1600,<br>1d00,<br>1300, | 1c00,<br>1a00,<br>1800,<br>1800,<br>1c00,<br>1500,<br>1a00,<br>1700, | 1e00,<br>1f00,<br>1600,<br>1600,<br>1c00,<br>1400,<br>1600,<br>1900, | 1900,<br>1700,<br>1900,<br>1700,<br>1700,<br>1700,<br>1600,<br>1600, | 1a00,<br>1b00,<br>1a00,<br>1400,<br>1700,<br>1b00,<br>1500,<br>1400, |
|        | non-zero | 2800,                                                                         | 2500,                                                                | 2500,                                                                         | 2700,                                                                | 2500,                                                                | 2600,                                                                | 2500,                                                                |                                                                      |

**Table 4.A.62 – BSAC probability table 9**

(MSB plane = 5)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)       |                |                |                |                |                |                |       |
|--------------|---------------------|-----------------------------------------------------|----------------|----------------|----------------|----------------|----------------|----------------|-------|
| MSB          | zero                | 3d00,<br>3400,                                      | 3e00,<br>3e00, | 3300,<br>3500, | 3e00,<br>3f00, | 3500,<br>3d00, | 3e00,<br>3f00, | 3700,<br>3c00, | 3e00, |
| MSB-1        | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 2e00,                                               |                |                |                |                |                |                |       |
| MSB-2        | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 2900,                                               | 2a00,          | 2700,          |                |                |                |                |       |
| MSB-3        | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 2d00,                                               | 2500,          | 2400,          | 2500,          | 2400,          | 2500,          | 2300,          |       |
| others       | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 2800,<br>2200,                                      | 2500,<br>2200, | 2300,<br>2200, | 2300,<br>2100, | 2200,<br>2000, | 2200,<br>2200, | 2200,<br>2100, | 2200, |

**Table 4.A.63 – BSAC probability table 10**

(MSB plane = 5)

| Significance | decoded higher bits | Probability Value of symbol '0' (Hexadecimal)       |                |                |                |                |                |                |       |
|--------------|---------------------|-----------------------------------------------------|----------------|----------------|----------------|----------------|----------------|----------------|-------|
| MSB          | zero                | 3b00,<br>3400,                                      | 3c00,<br>3900, | 3400,<br>2f00, | 3c00,<br>3c00, | 3200,<br>2d00, | 3900,<br>3700, | 2e00,<br>2d00, | 3d00, |
| MSB-1        | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 3100,                                               |                |                |                |                |                |                |       |
| MSB-2        | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 2b00,                                               | 2a00,          | 2900,          |                |                |                |                |       |
| MSB-3        | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 2700,                                               | 2600,          | 2500,          | 2500,          | 2500,          | 2200,          | 2200,          |       |
| others       | zero                | same as BSAC probability table 8 (see Table 4.A.61) |                |                |                |                |                |                |       |
|              | non-zero            | 2200,<br>2200,                                      | 2300,<br>2200, | 2300,<br>2200, | 2300,<br>2200, | 2200,<br>2200, | 2300,<br>2000, | 2200,<br>2100, | 2300, |

**Table 4.A.64 – BSAC probability table 11****Same as BSAC probability table 10, but MSB plane = 6****Table 4.A.65 – BSAC probability table 12****Same as BSAC probability table 11, but MSB plane = 6****Table 4.A.66 – BSAC probability table 13****Same as BSAC probability table 10, but MSB plane = 7****Table 4.A.67 – BSAC probability table 14****Same as BSAC probability table 11, but MSB plane = 7****Table 4.A.68 – BSAC probability table 15****Same as BSAC probability table 10, but MSB plane = 8****Table 4.A.69 – BSAC probability table 16****Same as BSAC probability table 10, but MSB plane = 9****Table 4.A.70 – BSAC probability table 17****Same as BSAC probability table 10, but MSB plane = 10****Table 4.A.71 – BSAC probability table 18****Same as BSAC probability table 10, but MSB plane = 11****Table 4.A.72 – BSAC probability table 19****Same as BSAC probability table 10, but MSB plane = 12****Table 4.A.73 – BSAC probability table 20****Same as BSAC probability table 10, but MSB plane = 13****Table 4.A.74 – BSAC probability table 21****Same as BSAC probability table 10, but MSB plane = 14****Table 4.A.75 – BSAC probability table 22****Same as BSAC probability table 10, but MSB plane = 15**

## Annex B (informative) Encoder tools

### 4.B.1 *Weighted interleave vector quantization*

In the weighted interleave VQ section, the flattened MDCT coefficients vector and the weight vector is divided into subvector at the first stage. Then, the subvectors are applied to weighted vector quantization.

#### 4.B.1.1 Initialization

Necessary parameters are initialized depending on the coding modes as described in subclause 4.6.5 of interleaved vector quantization tool.

#### 4.B.1.2 Vector quantization

##### 4.B.1.2.1 Basic structure

This vector quantizer has a two-channel conjugate structure, where two sets of codebooks are prepared. The search procedure consists of three steps; pre-selection, main selection and index packing.

##### 4.B.1.2.2 Pre-selection

At the first step of the vector quantization procedure, candidates for the nearest codevector are selected from each codebook. To select the candidates, weighted distortion measures are calculated.

$$dist[idiv][icb] = \sum_{ismp=0}^{vec\_len[idiv]-1} wt\_sub^2 (sp\_cv0[icb][ismp] - U\_sub[idiv][ismp])^2,$$

for  $idiv = 0$  to  $N\_DIV - 1$ ,  $icb = 0$  to  $2^{bits[idiv]} - 1$

If the  $bits0[idiv]$  is greater than  $MAXBIT\_SHAPE$ , the following distortion are also calculated.

$$dist_{negative}[idiv][icb] = \sum_{ismp=0}^{vec\_len[idiv]-1} wt\_sub^2 (sp\_cv0[icb][ismp] + U\_sub[idiv][ismp])^2,$$

for  $idiv = 0$  to  $N\_DIV - 1$ ,  $icb = 0$  to  $2^{bits[idiv]} - 1$

The  $N\_CAN$  candidates of  $dist$  and  $dist_{negative}$  are selected with minimum distortion measure

##### 4.B.1.2.3 Main selection

The distortion measures are calculated as below.

$$dist_{cross}[idiv][icb] = \sum_{ismp=0}^{vec\_len[idiv]-1} wt_{sub}[ismp] \left( \begin{array}{l} pol0[idiv] \\ \cdot sp\_cv0[can\_ind0[ican0]][ismp] \\ + pol1[idiv] \\ \cdot sp\_cv1[can\_ind1[ican1]][ismp] \\ \hline 2 - U_{sub}[idiv][ismp] \end{array} \right)$$

for  $idiv = 0$  to  $N\_DIV - 1$ ,  $ican0 = 0$  to  $N\_CAN - 1$ ,  $ican1 = 0$  to  $N\_CAN - 1$

where

$$pol0[idiv] = \begin{cases} 1 & \text{if } bits0[idiv] = MAXBIT\_SHAPE \\ \begin{cases} 1 \\ -1 \end{cases} & \text{if } bits0[idiv] > MAXBIT\_SHAPE \end{cases}$$

$$pol1[idiv] = \begin{cases} 1 & \text{if } bits1[idiv] = MAXBIT\_SHAPE \\ \begin{cases} 1 \\ -1 \end{cases} & \text{if } bits1[idiv] > MAXBIT\_SHAPE \end{cases}$$

Indexes  $can\_ind0[ican0]$  and  $can\_ind1[ican1]$  which give the least distortion measure is the quantization indexes  $index0[idiv]$  and  $index1[idiv]$ .

#### 4.B.1.2.4 Index packing

For each VQ index, the polarity information is added as follows.

If  $pol0[idiv] = -1$  then  $index0[idiv] = index0[idiv] + 2^{MAXBIT\_SHAPE}$ , for  $idiv = 0$  to  $N\_DIV$

If  $pol1[idiv] = -1$  then  $index1[idiv] = index1[idiv] + 2^{MAXBIT\_SHAPE}$ , for  $idiv = 0$  to  $N\_DIV$



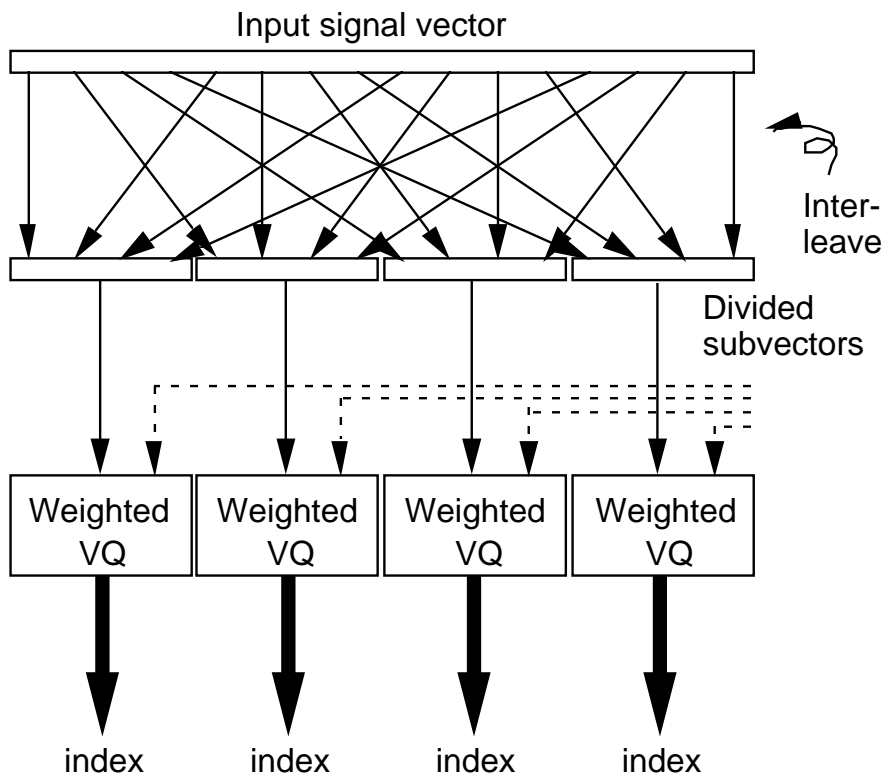


Figure 4.B.1 – Weighted interleave vector quantization.

## 4.B.2 Spectrum normalization

### 4.B.2.1 Encoding process

#### 4.B.2.1.1 Bandwidth control

This process limits the bandwidth of MDCT spectrum for the quantization process.

#### 4.B.2.1.2 LPC spectrum coding

At the first stage of the spectrum normalization process, input samples are divided into two paths. In one path they are transformed into frequency domain coefficients by MDCT. In the other path, LPC coefficients are calculated by LPC analysis; then, the LPC modeled spectrum envelope is calculated; and finally, each MDCT coefficient is divided by its corresponding spectrum envelope and first-stage flattened coefficients are produced. The main advantage of this flattening method is that fewer bits are required to normalize the frequency characteristics of input signals as a result of making use of an efficient quantization scheme of LSP (Line Spectral Pair) parameters. In order to avoid square root calculation in the de-flattening procedure at the decoder, LPC coefficients for the square root envelope are quantized; they are derived from the half-value of the LPC cepstrum of the normal LPC coefficients. The procedure for obtaining spectral envelope is listed as follows.

- Autocorrelation function is calculated from the windowed input signal.
- LPC coefficients are calculated by the Levinson-Durbin-Shur method.
- LPC cepstrum coefficients are generated from LPC coefficients.
- LPC cepstrum coefficients are multiplied by 0.5.
- LPC coefficients (corresponding to square root spectrum) are converted from the LPC cepstrum coefficients by solving a normal equation.
- Stability of the obtained coefficients is checked.
- LPC coefficients are converted to LSP parameters.

LSP parameters are quantized by 2-stage split vector quantizer with moving average inter-frame prediction. Inverse LPC spectral envelope values are calculated by the same method in subclause 4.6.10 of the spectrum normalization tool

#### 4.B.2.1.3 First-stage flattening

MDCT coefficients  $X[isf][j]$  are flattened using LPC spectrum as follows.

$$X_{flat1}[isf][ismp] = X[isf][ismp] / lpc\_spectrum[j],$$

$$\text{for } 0 \leq isf < N\_SF, 0 \leq ismp < N\_FR$$

#### 4.B.2.1.4 Periodic peak components coding

If there are peak components with a constant period in the MDCT coefficients due to the harmonic nature of speech or some musical instruments in the long frame of the core coding mode, the process of the following periodic components coding is activated.

##### 4.B.2.1.4.1 Fundamental frequency estimation

The best period value is searched that maximizes the energy of the periodic peak components. Number of extracted samples are fixed to 20 indifferent to the period. The definition of the period and the extracted samples are identical to those for the decoding process.

##### 4.B.2.1.4.2 Vector quantization of the periodic peak components

The extracted peak components are normalized by the averaged amplitude and quantized by the interleaved weighted vector quantization similar to those used for flattened MDCT coefficients.

The averaged amplitude is quantized in a mu-law scale.

##### 4.B.2.1.4.3 Subtraction of periodic peak components

The periodic peak components are locally decoded and subtracted from the original MDCT coefficients. The difference signals are used for the proceeding processes.

#### 4.B.2.1.5 Bark-scale envelope coding

##### 4.B.2.1.5.1 Calculation of the Bark-scale envelope

The inputs of the Bark-scale envelope calculation procedure are the MDCT coefficients  $xflat[]$  flattened by the LPC spectrum. First the square-rooted power of the input coefficients corresponding to each Bark-scale subband is calculated. The upper frequency boundary of each subband is described in subclause 4.6.10 of spectrum normalization tool.

$$env\_tmp[isf][ib] = \sqrt{\frac{\sum_{ismp=iblow}^{ibhigh-1} X_{flat1}[ismp]^2}{ibhigh - iblow}},$$

$$\text{for } 0 \leq isf < N\_SF, 0 \leq ib < N\_CRB$$

Next, the average of all square-rooted powers is calculated.

$$env\_avr[isf] = \frac{\sum_{ib=0}^{N\_CRB-1} env\_tmp[isf][ib]}{N\_CRB}, \quad \text{for } 0 \leq isf < N\_SF$$

Then, square-rooted powers are normalized by the average to create the MDCT envelope.

$$env[isf][ib] = env\_tmp[isf][ib] / env\_avr[isf],$$

for  $0 \leq isf < N\_SF, 0 \leq ib < N\_CRB$

#### 4.B.2.1.5.2 Weight calculation

The weight is used for the vector quantization of the MDCT envelope.

$$env\_fwt[ib] = \frac{\sum_{ismp=iblow}^{ibhigh-1} (lpc\_spectrum[ismp])}{ibhigh - iblow}$$

#### 4.B.2.1.5.3 Quantization

##### 4.B.2.1.5.3.1 Interframe prediction

Before quantizing the MDCT envelope, the prediction switch is set. The values of envelope elements are subtracted by their average (is 1).

$$env[is][ib] = env[isf][ib] - 1, \quad \text{for } 0 \leq isf < N\_SF, 0 \leq ib < N\_CRB$$

Correlation between current-frame  $env[isf][ib]$  and previous-frame  $env[isf][ib]$  is calculated.

$$correlation = \frac{\sum_{ib=0}^{N\_CRB-1} env\_current[isf][ib] \cdot env\_previous[isf][ib]}{\sum_{ib=0}^{N\_CRB-1} (env\_current[isf][ib])^2}, \quad \text{for } 0 \leq isf < N\_SF$$

if  $N\_SF$  is not equal to 1, the envelope of the previous frame is set by a following equation:

$$env\_previous[0][ib] = 0,$$

$$env\_previous[isf][ib] = env\_current[isf-1][ib], \quad \text{for } 0 \leq isf < N\_SF, 0 \leq ib < N\_CRB$$

If the correlation measure is greater than 0.5, prediction is on, otherwise, prediction is off.

$$\text{if } (correlation[isf] > 0.5) \text{ then } index\_fw\_alf[isf] = 1$$

else  $index\_fw\_alf[isf] = 0,$

for  $0 \leq isf < N\_SF$

#### 4.B.2.1.5.3.2 Vector quantization of the MDCT envelope

At the first step of quantization, the envelope and the weight vector are divided into  $FW\_N\_DIV$  subvectors.

$$denv[isf][ifdiv][icv] = env[isf][FW\_N\_DIV * icv + ifdiv],$$

for  $0 \leq isf < N\_SF$ ,  $0 \leq ifdiv < N\_CRB$ ,  $0 \leq icv < FW\_CB\_LEN$

$$dfwt[ifdiv][icv] = env\_fwt[FW\_N\_DIV * icv + ifdiv],$$

for  $ifdiv = 0$  to  $N\_CRB - 1$ ,  $icv = 0$  to  $FW\_CB\_LEN - 1$

Distortion measures  $fwdist[itmp]$  are calculated as follows.

$$alfq[isf] = index\_fw\_alf[isf] \cdot FW\_ALF\_STEP$$

$$fwdist[isf][itmp] =$$

$$\sum_{icv=0}^{FW\_CB\_LEN-1} dfwt[ifdiv][icv]^2 \cdot$$

$$(denv[isf][ifdiv][icv] - alfq \cdot p\_cv\_env[isf][icv] - cb\_env[itmp][icv])^2$$

for  $0 \leq isf < N\_SF$ ,  $0 \leq itmp < FW\_CB\_SIZE$ ,  $0 \leq ifdiv < FW\_N\_DIV$

The  $cv\_env[[]]$  are elements of the envelope codebook. The  $p\_cv\_env[[]]$  represents the MDCT envelope of the previous frame. It is calculated by the same procedure mentioned in subclause 4.6.10 of spectrum normalization tool.

Finally,  $itmp$  for the minimum distortion measure  $fwdist[itmp]$  is the quantization index  $index\_env[ifdiv]$ .

#### 4.B.2.1.5.3.3 Local decoding

After the vector quantization, the MDCT envelope is reproduced as follows.

```
for (isf=0; isf<N_SF; isf++){
  alfq[isf] = index_fw_alf[isf] * FW_ALF_STEP
  for (ifdiv=0; ifdiv<FW_N_DIV; ifdiv++){
    for (icv=0; icv<FW_CB_LEN; icv++){
      ienv = FW_N_DIV * icv + ifdiv
      dtmp = cv_env[index_env[isf][ifdiv]][icv]
      qenv_b[isf][ienv] = dtmp + alfq * p_cv_env[isf][icv] + 1
      if (N_SF==1) p_cv_env[isf][icv] = dtmp
      else      p_cv_env[isf-1][icv] = dtmp
    }
  }
}
```

The  $cv\_env[[]]$  are the envelope codebook.

Then, Bark-scale envelope  $qenv\_b[isf][ienv]$  are projected into linear-scale envelope  $qenv[isf][ismp]$ .

```
for (isf=0; isf<N_SF; ist++){
  for (ib=0; ib<N_CRB; ib++){
    for (ismp=iblow[ib]; ismp<ibhigh[ib]; ismp++){
      qenv[isf][ismp]=qenv_b[isf][ib];
    }
  }
}
```

**4.B.2.1.6 Second-stage flattening**

$X_{flat2}[][]$  are flattened as follows:

$$X_{flat2}[isf][j] = X_{flat1}[isf][j] / qenv[j] \quad \text{for } 0 \leq isf < N\_SF, 0 \leq j < N\_FR$$

**4.B.2.1.7 Gain quantization and amplitude normalization**

Before quantizing the flattened MDCT coefficients, their amplitudes are normalized by the gain factor.

The gain factor is calculated as follow:

$$gain[isf] = \frac{\sqrt{\frac{\sum_{ismp=0}^{N\_FR-1} X_{flat2}[isf]^2}{N\_FR}}}{AMP\_NM}, \quad \text{for } 0 \leq isf < N\_SF$$

If  $N\_SF == 1$ , the gain factor is quantized using the following equation:

$$index\_gain[0] = (\text{int}) \frac{\log_{10}(1 + MU \cdot gain[0] / AMP\_MAX)}{\log_{10}(1 + MU)}$$

Then, gain factor is locally decoded.

$$g\_temp = index\_gain * STEP + STEP / 2$$

$$qgain = AMP\_MAX * (\exp_{10}(g\_temp * \log_{10}(1+MU) / AMP\_MAX) - 1) / MU$$

$$qgain /= AMP\_NM$$

If  $N\_SF > 1$ , the global gain and subframe gains are quantized:

$$index\_gain[0] = (\text{int}) \frac{\log_{10}(1 + MU \cdot gain[0] / AMP\_MAX)}{\log_{10}(1 + MU)}$$

Finally, the flattened MDCT coefficients are normalized by the decoded gain factor  $qgain$ .

$$U[ismp] = X_{flat2}[ismp] / qgain \quad \text{for } ismp = 0 \text{ to } N\_FR - 1$$

### 4.B.3 Psychoacoustic model

#### 4.B.3.1 General

This Annex presents the general Psychoacoustic Model for the AAC encoder. The psychoacoustic model calculates the maximum distortion energy which is masked by the signal energy. This energy is called *threshold*. The threshold generation process has three inputs. They are:

1. The shift length for the threshold calculation process is called *iblen*. This *iblen* must remain constant over any particular application of the threshold calculation process. Since it is necessary to calculate thresholds for two different shift lengths, two processes, each running with a fixed shift length, are necessary. For long FFT  $iblen = 1024$ , for short FFT  $iblen = 128$ .
2. For each FFT type the newest *iblen* samples of the signal, with the samples delayed (either in the filterbank or psychoacoustic calculation) such that the window of the psychoacoustic calculation is centered in the time-window of the codec time/frequency transform.
3. The sampling rate. There are sets of tables provided for the standard sampling rates. Sampling rate, just as *iblen*, must necessarily remain constant over one implementation of the threshold calculation process.

The output from the psychoacoustic model is :

1. a set of Signal-to-Mask Ratios and thresholds, which are adapted to the encoder as described below,
2. the delayed time domain data (PCM samples), which are used by the MDCT,
3. the block type for the MDCT ( long, start, stop or short type )
4. an estimation of how many bits should be used for encoding in addition to the average available bits.

The delay of the PCM samples is necessary, because if the switch decision algorithm detects an attack, so that *short blocks* have to be used for the actual frame, the *long block* before the *short blocks* has to be 'patched' to a *start block type* in this case.

Before running the model initially, the array used to hold the preceding FFT source data window and the arrays used to hold  $r(w)$  and  $f(w)$  should be zeroed to provide a known starting point.

#### 4.B.3.2 Comments on notation

Throughout this threshold calculation process, three indices for data values are used. These are:

- $w$ - indicates that the calculation is indexed by frequency in the FFT spectral line domain. An index of 0 corresponds to the DC term and an index of 1023 corresponds to the spectral line at the Nyquist frequency.
- $b$  - indicates that the calculation is indexed in the threshold calculation partition domain. In the case where the calculation includes a convolution or sum in the threshold calculation partition domain,  $bb$  will be used as the summation variable. Partition numbering starts at 0.
- $n$  - indicates that the calculation is indexed in the coder scalefactor band domain. An index of 0 corresponds to the lowest scalefactor band.

#### 4.B.3.3 The "spreading function"

Several points in the following description refer to the "spreading function". It is calculated by the following method:

```

if  $j >= i$ 
     $tmpx = 3,0 (j-i)$ 
else
     $tmpx = 1.5(j-i)$ 

```

Where  $i$  is the Bark value of the signal being spread,  $j$  is the Bark value of the band being spread into, and  $tmpx$  is a temporary variable.

$$tmpz = 8 * \text{minimum} ((tmpx-0,5)^2 - 2(tmpx-0,5), 0)$$

Where  $tmpz$  is a temporary variable, and  $\text{minimum}(a, b)$  is a function returning the more negative of  $a$  or  $b$ .

$$tmpy = 15.811389 + 7.5(tmpx + 0.474) - 17,5(1,0 + (tmpx + 0.474)^2)^{0,5}$$

where  $tmpy$  is another temporary variable.

$$\text{if } (tmpy < -100) \text{ then } \{sprdngf(i, j) = 0\} \text{ else } \{sprdngf(i, j) = 10^{((tmpz + tmpy)/10)}\}$$

#### 4.B.3.4 Steps in threshold calculation

The following are the necessary steps for the calculation of  $SMR(n)$  and  $xmin(n)$  used in the coder for long and short FFT.

1. Reconstruct  $2 * iblen$  samples of the input signal.

$iblen$  new samples are made available at every call to the threshold generator. The threshold generator must store  $2 * iblen - iblen$  samples, and concatenate those samples to accurately reconstruct  $2 * iblen$  consecutive samples of the input signal,  $s(i)$ , where  $i$  represents the index,  $0 \leq i < 2 * iblen$ , of the current input stream.

2. Calculate the complex spectrum of the input signal.

First,  $s(i)$  is windowed by a Hann window, i.e.

$$sw(i) = s(i) * (0,5 - 0,5 * \cos((\pi * (i+0,5)) / iblen)).$$

Second, a standard forward FFT of  $sw(i)$  is calculated.

Third, the polar representation of the transform is calculated.  $r(w)$  and  $f(w)$  represent the magnitude and phase components of the transformed  $sw(i)$ , respectively.

3. Calculate a predicted  $r(w)$  and  $f(w)$ .

A predicted magnitude,  $r\_pred(w)$  and phase,  $f\_pred(w)$  are calculated from the preceding two threshold calculation blocks  $r(w)$  and  $f(w)$ :

$$r\_pred(w) = 2.0 * r(t-1) - r(t-2)$$

$$f\_pred(w) = 2.0 * f(t-1) - f(t-2)$$

where  $t$  represents the current block number,  $t-1$  indexes the previous block's data, and  $t-2$  indexes the data from the threshold calculation block before that.

4. Calculate the unpredictability measure  $c(w)$ :

$$c(w) = (((r(w) * \cos(f(w)) - r\_pred(w) * \cos(f\_pred(w)))^2 + (r(w) * \sin(f(w)) - r\_pred(w) * \sin(f\_pred(w)))^2)^{0,5}) / (r(w) + \text{abs}(r\_pred(w)))$$

This formula is used for each of the short blocks with the short FFT, for long blocks for the first 6 lines the unpredictability measure is calculated from the long FFT, for the remaining lines the minimum of the unpredictability of all short FFT's is used. If calculation power should be saved, the unpredictability of the upper part of the spectrum can be set to 0.4.

Calculate the energy and unpredictability in the threshold calculation partitions.

The energy in each partition,  $e(b)$ , is:

do for each partition  $b$ :

$$e(b) = 0$$

do from lower index to upper index  $w$  of partition  $b$

```

    e(b) = e(b) + r(w)^2
end do

```

and the weighted unpredictability,  $c(b)$ , is:

```

do for each partition b:
    c(b) = 0
    do from lower index to upper index w of partition b
        c(b) = c(b) + r(w)^2 * c(w)
    end do
end do

```

The threshold calculation partitions provide a resolution of approximately either one FFT line or 1/3 critical band, whichever is wider. At low frequencies, a single line of the FFT will constitute a calculation partition. At high frequencies, many lines will be combined into one calculation partition. A set of partition values is provided for each of the three sampling rates in Tables 4.B.1 to 4.B.24. These table elements will be used in the threshold calculation process. There are several elements in each table entry:

1. The index of the calculation partition,  $b$ .
2. The lowest frequency line in the partition,  $w\_low(b)$ .
3. The highest frequency line in the partition,  $w\_high(b)$
4. The median bark value of the partition,  $bval(b)$
5. The threshold in quiet  $qsthr(b)$

A largest value of  $b$ ,  $bmax$ , equal to the largest index, exists for each sampling rate.

5. Convolve the partitioned energy and unpredictability with the spreading function.

```

for each partition b:
    ecb(b) = 0
    do for each partition bb:
        ecb(b) = ecb(b) + e(bb) * sprdngf(bval(bb), bval(b))
    end do
end do

do for each partition b:
    ct(b) = 0
    do for each partition bb:
        ct(b) = ct(b) + c(bb) * sprdngf(bval(bb), bval(b))
    end do
end do

```

Because  $ct(b)$  is weighted by the signal energy, it must be renormalized to  $cb(b)$ .

$$cb(b) = ct(b) / ecb(b)$$

Just as this, due to the non-normalized nature of the spreading function,  $ecbb$  should be renormalized and the normalized energy  $enb$ , calculated.

$$en(b) = ecb(b) * norm(b)$$

The normalization coefficient,  $norm(b)$ , is:

```

do for each partition b
    tmp(b) = 0
    do for each partition bb
        tmp(b) = tmp(b) + sprdngf(bval(bb), bval(b))
    end do
    norm(b) = 1 / tmp(b)
end do

```



6. Convert  $cb(b)$  to  $tb(b)$ , the tonality index.

$$tb(b) = -0,299 - 0,43 \log_e (cb(b))$$

Each  $tb(b)$  is limited to the range of  $0 < tb(b) < 1$ .

7. Calculate the required SNR in each partition.

$NMT(b) = 6$  dB for all  $b$ .  $NMT(b)$  is the value for noise masking tone (in dB) for the partition.  $TMN(b) = 18$  dB for all  $b$ .  $TMN(b)$  is the value for tone masking noise (in dB). The required signal to noise ratio,  $SNR(b)$ , is:

$$SNR(b) = tb(b) * TMN(b) + (1 - tb(b)) * NMT(b)$$

8. Calculate the power ratio.

The power ratio,  $bc(b)$ , is:

$$bc(b) = 10^{(-SNR(b) / 10)}$$

9. Calculation of actual energy threshold,  $nb(b)$ .

$$nb(b) = en(b) * bc(b)$$

$nb(b)$  is also used in the M/S-module (see subclause 4.B.9. "Joint coding"):  $nb(b)$  is equal to  $X_{thr}$  with 'X'=[R,L,M,S]

10. Pre-echo control and threshold in quiet.

To avoid pre-echoes the pre-echo control is calculated for short and long FFT, the threshold in quiet is also considered here:

$nb\_l(b)$  is the threshold of partition  $b$  for the last block,  $qsthr(b)$  is the threshold in quiet. The dB values of  $qsthr(b)$  shown in Tables 4.B.1 to 4.B.24 are relative to the level that a sine wave of + or - 1/2 lsb has in the FFT used for threshold calculation. The dB values must be converted into the energy domain after considering the FFT normalization actually used.

$$nb(b) = \max(qsthr(b), \min(nb(b), nb\_l(b) * rpelev))$$

$rpelev$  is set to '1' for short blocks and '2' for long blocks

11. The PE is calculated for each block type from the ratio  $e(b) / nb(b)$ , where  $nb(b)$  is the threshold and  $e(b)$  is the energy for each threshold partition:

```
PE = 0
do for threshold partition b
    PE = PE - (w_high(b) - w_low(b)) * log10 ( nb(b) / ( e(b) + 1 ) )
end do
```

12. The decision, whether long or short block type is used for encoding is made according to this pseudo code:

```
if PE for long block is greater than switch_pe then
    coding_block_type = short_block_type
else
    coding_block_type = long_block_type
end if
if ( coding_block_type == short_block_type ) and ( last_coding_block_type == long_type ) then
    last_coding_block_type = start_type
else
    last_coding_block_type = short_type
```

The last four lines are necessary since there is no combined stop/start block type in AAC. *switch\_pe* is a implementation dependend constant

13. Calculate the signal-to-mask ratios,  $SMR(n)$  and the codec threshold  $xmin(n)$ .

Tables 4.72 to 4.90 (normative part) show:

1. The index, *swb*, of the coder partition called scalefactor band.
2. The offset of mdct line for the scalefactor band *swb\_offset\_long/short\_window*.

we define the following variable :

```
n = swb
w_low(n) = swb_offset_long/short_window(n)
w_high(n) = swb_offset_long/short_window(n+1) - 1
```

The FFT energy in the scalefactor band,  $epart(n)$ , is:

```
do for each scalefactor band n
  epart(n) = 0
  do for w = lower index w_low(n) to n = upper index w_high(n)
    epart(n) = epart ( n) + r(w)^2
  end do
end do
```

the threshold for one line of the spectrum is calculated according to:

```
do for each threshold partition b
  thr(all line_indices in this partition b) =
    thr (w_low(b),...,w_high(b)) = nb(b) / (w_high(b)+1-w_low(b))
end do
```

the noise level in the scalefactor band on FFT level,  $npart(n)$  is calculated as:

```
do for each scalefactor band n
  npart(n) = minimum( thr(w_low(n)),..., thr(w_high(n)) ) * (w_high(n)+1-w_low(n))
end do
```

Where, in this case, minimum (a,...,z) is a function returning the smallest positive argument of the arguments a...z.

The ratios to be sent to the quantization module,  $SMR(n)$ , are calculated as:

$$SMR(n) = epart(n) / npart(n)$$

For the calculation of coder thresholds  $xmin(n)$  the MDCT energy for each scalefactor band is calculated:

```
do for all scalefactor bands n
  codec_e(n) = 0
  do for lower index i to higher index i of this scalefactor band
    codec_e(n) = codec_e(n) +( mdct_line(i))^2
  end do
end do
```

Then  $xmin(n)$ , the maximum allowed error energy on MDCT level, can be calculated according to this formula :

$$xmin (n) = npart(n) * codec_e(n) / epart (n)$$

Calculate the bit allocation out of the psychoacoustic entropy (PE).

$$\text{bit\_allocation} = \text{pew1} * \text{PE} + \text{pew2} * \text{sqrt}(\text{PE});$$

for long blocks the constants are defined as:

$$\text{pew1} = 0.3, \text{pew2} = 6.0$$

for short blocks the PE of the eight short blocks is summed up and the constants are :

$$\text{pew1} = 0.6, \text{pew2} = 24$$

then  $\text{bit\_allocation}$  is limited to  $0 < \text{bit\_allocation} < 3000$  and

$\text{more\_bits}$  is calculated :

$$\text{more\_bits} = \text{bit\_allocation} - (\text{average\_bits} - \text{side\_info\_bits})$$

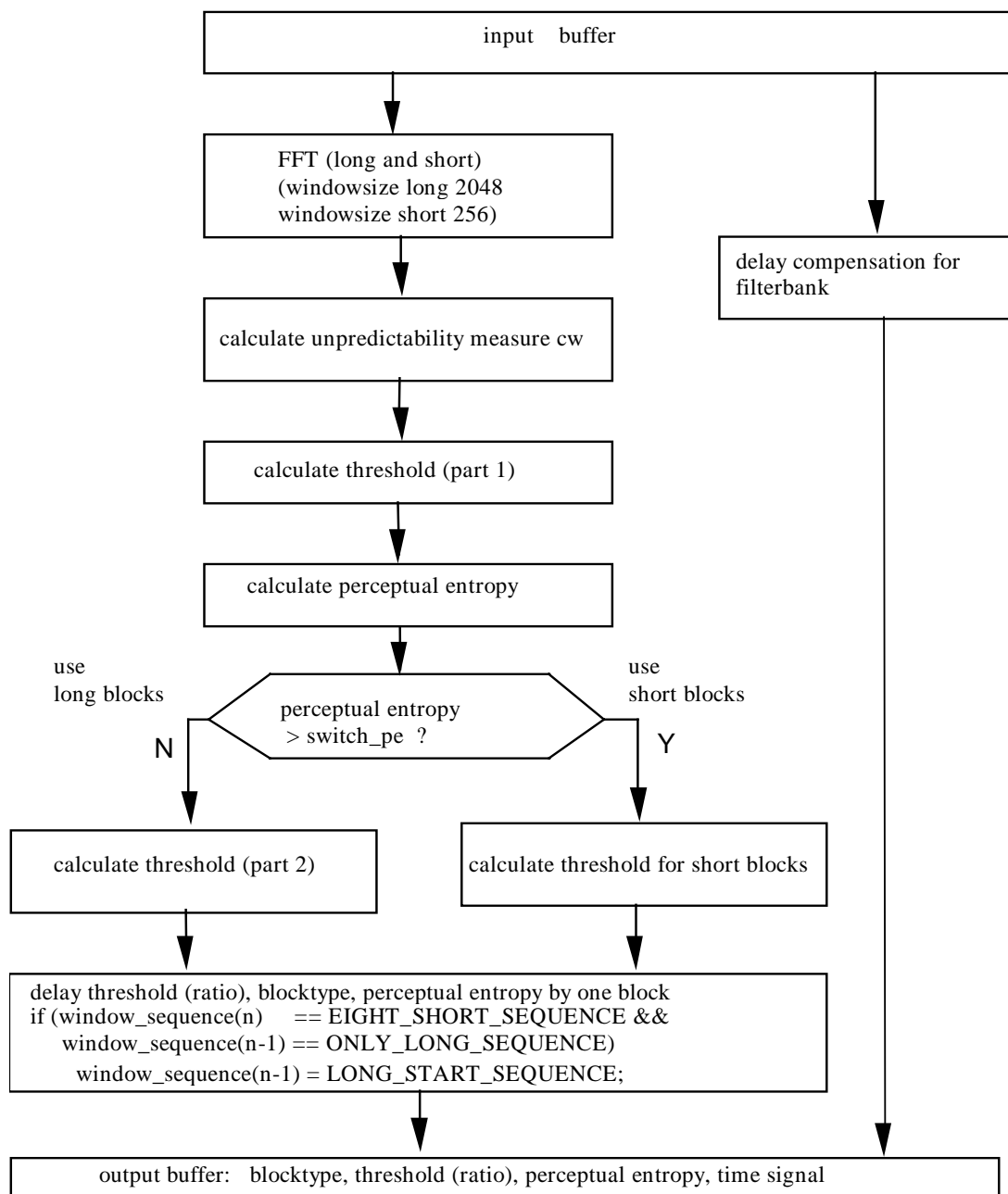


Figure 4.B.2 – block diagram psychoacoustic model

Table 4.B.1 – Psychoacoustic parameters for 8 kHz long FFT

| index | w low | w high | width | bval | gstr  |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 8      | 9     | 0,18 | 46,82 |
| 1     | 9     | 17     | 9     | 0,53 | 46,82 |
| 2     | 18    | 26     | 9     | 0,89 | 46,82 |
| 3     | 27    | 35     | 9     | 1,24 | 41,82 |
| 4     | 36    | 44     | 9     | 1,59 | 41,82 |
| 5     | 45    | 53     | 9     | 1,94 | 41,82 |
| 6     | 54    | 62     | 9     | 2,29 | 38,82 |
| 7     | 63    | 71     | 9     | 2,63 | 38,82 |

|    |      |      |    |       |       |
|----|------|------|----|-------|-------|
| 8  | 72   | 80   | 9  | 2,98  | 38,82 |
| 9  | 81   | 89   | 9  | 3,31  | 33,82 |
| 10 | 90   | 98   | 9  | 3,65  | 33,82 |
| 11 | 99   | 108  | 10 | 3,99  | 34,28 |
| 12 | 109  | 118  | 10 | 4,35  | 32,28 |
| 13 | 119  | 128  | 10 | 4,71  | 32,28 |
| 14 | 129  | 138  | 10 | 5,05  | 32,28 |
| 15 | 139  | 148  | 10 | 5,39  | 32,28 |
| 16 | 149  | 159  | 11 | 5,74  | 32,69 |
| 17 | 160  | 170  | 11 | 6,10  | 32,69 |
| 18 | 171  | 181  | 11 | 6,45  | 32,69 |
| 19 | 182  | 192  | 11 | 6,79  | 32,69 |
| 20 | 193  | 204  | 12 | 7,13  | 33,07 |
| 21 | 205  | 216  | 12 | 7,48  | 33,07 |
| 22 | 217  | 228  | 12 | 7,82  | 33,07 |
| 23 | 229  | 241  | 13 | 8,17  | 33,42 |
| 24 | 242  | 254  | 13 | 8,51  | 33,42 |
| 25 | 255  | 268  | 14 | 8,85  | 33,74 |
| 26 | 269  | 282  | 14 | 9,20  | 33,74 |
| 27 | 283  | 297  | 15 | 9,54  | 34,04 |
| 28 | 298  | 312  | 15 | 9,88  | 34,04 |
| 29 | 313  | 328  | 16 | 10,22 | 34,32 |
| 30 | 329  | 345  | 17 | 10,56 | 34,58 |
| 31 | 346  | 363  | 18 | 10,91 | 34,83 |
| 32 | 364  | 381  | 18 | 11,25 | 34,83 |
| 33 | 382  | 400  | 19 | 11,58 | 35,06 |
| 34 | 401  | 420  | 20 | 11,91 | 35,29 |
| 35 | 421  | 441  | 21 | 12,24 | 35,50 |
| 36 | 442  | 464  | 23 | 12,58 | 35,89 |
| 37 | 465  | 488  | 24 | 12,92 | 36,08 |
| 38 | 489  | 514  | 26 | 13,26 | 36,43 |
| 39 | 515  | 541  | 27 | 13,59 | 36,59 |
| 40 | 542  | 570  | 29 | 13,93 | 36,90 |
| 41 | 571  | 601  | 31 | 14,26 | 37,19 |
| 42 | 602  | 634  | 33 | 14,60 | 37,46 |
| 43 | 635  | 670  | 36 | 14,93 | 37,84 |
| 44 | 671  | 708  | 38 | 15,27 | 38,07 |
| 45 | 709  | 749  | 41 | 15,60 | 38,40 |
| 46 | 750  | 793  | 44 | 15,93 | 38,71 |
| 47 | 794  | 841  | 48 | 16,26 | 39,09 |
| 48 | 842  | 893  | 52 | 16,60 | 39,44 |
| 49 | 894  | 949  | 56 | 16,93 | 39,76 |
| 50 | 950  | 1009 | 60 | 17,26 | 40,06 |
| 51 | 1010 | 1023 | 14 | 17,47 | 33,74 |

Table 4.B.2 – Psychoacoustic parameters for 8 kHz short FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 1      | 2     | 0,32 | 30,29 |
| 1     | 2     | 3      | 2     | 0,95 | 30,29 |
| 2     | 4     | 5      | 2     | 1,57 | 25,29 |
| 3     | 6     | 7      | 2     | 2,19 | 22,29 |
| 4     | 8     | 9      | 2     | 2,80 | 22,29 |
| 5     | 10    | 11     | 2     | 3,40 | 17,29 |
| 6     | 12    | 13     | 2     | 3,99 | 17,29 |
| 7     | 14    | 15     | 2     | 4,56 | 15,29 |

|    |     |     |   |       |       |
|----|-----|-----|---|-------|-------|
| 8  | 16  | 17  | 2 | 5,12  | 15,29 |
| 9  | 18  | 19  | 2 | 5,66  | 15,29 |
| 10 | 20  | 21  | 2 | 6,18  | 15,29 |
| 11 | 22  | 23  | 2 | 6,68  | 15,29 |
| 12 | 24  | 25  | 2 | 7,16  | 15,29 |
| 13 | 26  | 27  | 2 | 7,63  | 15,29 |
| 14 | 28  | 29  | 2 | 8,07  | 15,29 |
| 15 | 30  | 31  | 2 | 8,50  | 15,29 |
| 16 | 32  | 33  | 2 | 8,90  | 15,29 |
| 17 | 34  | 35  | 2 | 9,29  | 15,29 |
| 18 | 36  | 37  | 2 | 9,67  | 15,29 |
| 19 | 38  | 39  | 2 | 10,03 | 15,29 |
| 20 | 40  | 41  | 2 | 10,37 | 15,29 |
| 21 | 42  | 44  | 3 | 10,77 | 17,05 |
| 22 | 45  | 47  | 3 | 11,23 | 17,05 |
| 23 | 48  | 50  | 3 | 11,66 | 17,05 |
| 24 | 51  | 53  | 3 | 12,06 | 17,05 |
| 25 | 54  | 56  | 3 | 12,44 | 17,05 |
| 26 | 57  | 59  | 3 | 12,79 | 17,05 |
| 27 | 60  | 63  | 4 | 13,18 | 18,30 |
| 28 | 64  | 67  | 4 | 13,59 | 18,30 |
| 29 | 68  | 71  | 4 | 13,97 | 18,30 |
| 30 | 72  | 75  | 4 | 14,32 | 18,30 |
| 31 | 76  | 80  | 5 | 14,69 | 19,27 |
| 32 | 81  | 85  | 5 | 15,07 | 19,27 |
| 33 | 86  | 90  | 5 | 15,42 | 19,27 |
| 34 | 91  | 96  | 6 | 15,77 | 20,06 |
| 35 | 97  | 102 | 6 | 16,13 | 20,06 |
| 36 | 103 | 109 | 7 | 16,49 | 20,73 |
| 37 | 110 | 116 | 7 | 16,85 | 20,73 |
| 38 | 117 | 124 | 8 | 17,20 | 21,31 |
| 39 | 125 | 127 | 3 | 17,44 | 17,05 |

Table 4.B.3 – Psychoacoustic parameters for 11.025 kHz long FFT

| index | w low | w high | width | bval | qstr  |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 6      | 7     | 0,19 | 45,73 |
| 1     | 7     | 13     | 7     | 0,57 | 45,73 |
| 2     | 14    | 20     | 7     | 0,95 | 45,73 |
| 3     | 21    | 27     | 7     | 1,33 | 40,73 |
| 4     | 28    | 34     | 7     | 1,71 | 40,73 |
| 5     | 35    | 41     | 7     | 2,08 | 37,73 |
| 6     | 42    | 48     | 7     | 2,45 | 37,73 |
| 7     | 49    | 55     | 7     | 2,82 | 37,73 |
| 8     | 56    | 62     | 7     | 3,18 | 32,73 |
| 9     | 63    | 69     | 7     | 3,54 | 32,73 |
| 10    | 70    | 76     | 7     | 3,89 | 32,73 |
| 11    | 77    | 83     | 7     | 4,24 | 30,73 |
| 12    | 84    | 90     | 7     | 4,59 | 30,73 |
| 13    | 91    | 97     | 7     | 4,92 | 30,73 |
| 14    | 98    | 105    | 8     | 5,28 | 31,31 |
| 15    | 106   | 113    | 8     | 5,65 | 31,31 |
| 16    | 114   | 121    | 8     | 6,01 | 31,31 |
| 17    | 122   | 129    | 8     | 6,36 | 31,31 |
| 18    | 130   | 137    | 8     | 6,70 | 31,31 |
| 19    | 138   | 146    | 9     | 7,06 | 31,82 |

|    |      |      |    |       |       |
|----|------|------|----|-------|-------|
| 20 | 147  | 155  | 9  | 7,42  | 31,82 |
| 21 | 156  | 164  | 9  | 7,77  | 31,82 |
| 22 | 165  | 173  | 9  | 8,11  | 31,82 |
| 23 | 174  | 183  | 10 | 8,46  | 32,28 |
| 24 | 184  | 193  | 10 | 8,82  | 32,28 |
| 25 | 194  | 203  | 10 | 9,16  | 32,28 |
| 26 | 204  | 214  | 11 | 9,50  | 32,69 |
| 27 | 215  | 225  | 11 | 9,85  | 32,69 |
| 28 | 226  | 237  | 12 | 10,19 | 33,07 |
| 29 | 238  | 249  | 12 | 10,54 | 33,07 |
| 30 | 250  | 262  | 13 | 10,88 | 33,42 |
| 31 | 263  | 275  | 13 | 11,22 | 33,42 |
| 32 | 276  | 289  | 14 | 11,56 | 33,74 |
| 33 | 290  | 304  | 15 | 11,90 | 34,04 |
| 34 | 305  | 320  | 16 | 12,24 | 34,32 |
| 35 | 321  | 337  | 17 | 12,59 | 34,58 |
| 36 | 338  | 355  | 18 | 12,94 | 34,83 |
| 37 | 356  | 374  | 19 | 13,28 | 35,06 |
| 38 | 375  | 394  | 20 | 13,62 | 35,29 |
| 39 | 395  | 415  | 21 | 13,96 | 35,50 |
| 40 | 416  | 438  | 23 | 14,29 | 35,89 |
| 41 | 439  | 462  | 24 | 14,63 | 36,08 |
| 42 | 463  | 488  | 26 | 14,96 | 36,43 |
| 43 | 489  | 516  | 28 | 15,29 | 36,75 |
| 44 | 517  | 546  | 30 | 15,63 | 37,05 |
| 45 | 547  | 579  | 33 | 15,96 | 37,46 |
| 46 | 580  | 614  | 35 | 16,30 | 37,72 |
| 47 | 615  | 652  | 38 | 16,63 | 38,07 |
| 48 | 653  | 693  | 41 | 16,97 | 38,40 |
| 49 | 694  | 737  | 44 | 17,30 | 38,71 |
| 50 | 738  | 785  | 48 | 17,64 | 39,09 |
| 51 | 786  | 836  | 51 | 17,97 | 39,35 |
| 52 | 837  | 891  | 55 | 18,30 | 39,68 |
| 53 | 892  | 950  | 59 | 18,64 | 39,98 |
| 54 | 951  | 1014 | 64 | 18,97 | 40,34 |
| 55 | 1015 | 1023 | 9  | 19,16 | 31,82 |

Table 4.B.4 – Psychoacoustic parameters for 11.025 kHz short FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 0      | 1     | 0,00 | 27,28 |
| 1     | 1     | 1      | 1     | 0,44 | 27,28 |
| 2     | 2     | 2      | 1     | 0,87 | 27,28 |
| 3     | 3     | 3      | 1     | 1,30 | 22,28 |
| 4     | 4     | 4      | 1     | 1,73 | 22,28 |
| 5     | 5     | 5      | 1     | 2,16 | 19,28 |
| 6     | 6     | 6      | 1     | 2,58 | 19,28 |
| 7     | 7     | 7      | 1     | 3,00 | 14,28 |
| 8     | 8     | 8      | 1     | 3,41 | 14,28 |
| 9     | 9     | 9      | 1     | 3,82 | 14,28 |
| 10    | 10    | 10     | 1     | 4,22 | 12,28 |
| 11    | 11    | 11     | 1     | 4,61 | 12,28 |
| 12    | 12    | 12     | 1     | 4,99 | 12,28 |
| 13    | 13    | 13     | 1     | 5,37 | 12,28 |
| 14    | 14    | 14     | 1     | 5,74 | 12,28 |
| 15    | 15    | 15     | 1     | 6,10 | 12,28 |

|    |     |     |   |       |       |
|----|-----|-----|---|-------|-------|
| 16 | 16  | 16  | 1 | 6,45  | 12,28 |
| 17 | 17  | 17  | 1 | 6,79  | 12,28 |
| 18 | 18  | 19  | 2 | 7,44  | 15,29 |
| 19 | 20  | 21  | 2 | 8,05  | 15,29 |
| 20 | 22  | 23  | 2 | 8,64  | 15,29 |
| 21 | 24  | 25  | 2 | 9,19  | 15,29 |
| 22 | 26  | 27  | 2 | 9,70  | 15,29 |
| 23 | 28  | 29  | 2 | 10,19 | 15,29 |
| 24 | 30  | 31  | 2 | 10,65 | 15,29 |
| 25 | 32  | 33  | 2 | 11,08 | 15,29 |
| 26 | 34  | 35  | 2 | 11,48 | 15,29 |
| 27 | 36  | 37  | 2 | 11,86 | 15,29 |
| 28 | 38  | 39  | 2 | 12,22 | 15,29 |
| 29 | 40  | 42  | 3 | 12,64 | 17,05 |
| 30 | 43  | 45  | 3 | 13,10 | 17,05 |
| 31 | 46  | 48  | 3 | 13,53 | 17,05 |
| 32 | 49  | 51  | 3 | 13,93 | 17,05 |
| 33 | 52  | 54  | 3 | 14,30 | 17,05 |
| 34 | 55  | 58  | 4 | 14,69 | 18,30 |
| 35 | 59  | 62  | 4 | 15,11 | 18,30 |
| 36 | 63  | 66  | 4 | 15,49 | 18,30 |
| 37 | 67  | 70  | 4 | 15,84 | 18,30 |
| 38 | 71  | 75  | 5 | 16,21 | 19,27 |
| 39 | 76  | 80  | 5 | 16,58 | 19,27 |
| 40 | 81  | 85  | 5 | 16,92 | 19,27 |
| 41 | 86  | 91  | 6 | 17,27 | 20,06 |
| 42 | 92  | 97  | 6 | 17,62 | 20,06 |
| 43 | 98  | 104 | 7 | 17,97 | 20,73 |
| 44 | 105 | 111 | 7 | 18,32 | 20,73 |
| 45 | 112 | 119 | 8 | 18,67 | 21,31 |
| 46 | 120 | 127 | 8 | 19,02 | 21,31 |

Table 4.B.5 – Psychoacoustic parameters for 12 kHz long FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 5      | 6     | 0,18 | 45,06 |
| 1     | 6     | 11     | 6     | 0,53 | 45,06 |
| 2     | 12    | 17     | 6     | 0,89 | 45,06 |
| 3     | 18    | 23     | 6     | 1,24 | 40,06 |
| 4     | 24    | 29     | 6     | 1,59 | 40,06 |
| 5     | 30    | 35     | 6     | 1,94 | 40,06 |
| 6     | 36    | 41     | 6     | 2,29 | 37,06 |
| 7     | 42    | 47     | 6     | 2,63 | 37,06 |
| 8     | 48    | 53     | 6     | 2,98 | 37,06 |
| 9     | 54    | 59     | 6     | 3,31 | 32,06 |
| 10    | 60    | 65     | 6     | 3,65 | 32,06 |
| 11    | 66    | 72     | 7     | 4,00 | 30,73 |
| 12    | 73    | 79     | 7     | 4,38 | 30,73 |
| 13    | 80    | 86     | 7     | 4,75 | 30,73 |
| 14    | 87    | 93     | 7     | 5,11 | 30,73 |
| 15    | 94    | 100    | 7     | 5,47 | 30,73 |
| 16    | 101   | 107    | 7     | 5,82 | 30,73 |
| 17    | 108   | 114    | 7     | 6,15 | 30,73 |
| 18    | 115   | 122    | 8     | 6,51 | 31,31 |
| 19    | 123   | 130    | 8     | 6,88 | 31,31 |
| 20    | 131   | 138    | 8     | 7,24 | 31,31 |



|    |     |      |    |       |       |
|----|-----|------|----|-------|-------|
| 21 | 139 | 146  | 8  | 7,58  | 31,31 |
| 22 | 147 | 154  | 8  | 7,92  | 31,31 |
| 23 | 155 | 163  | 9  | 8,27  | 31,82 |
| 24 | 164 | 172  | 9  | 8,62  | 31,82 |
| 25 | 173 | 181  | 9  | 8,96  | 31,82 |
| 26 | 182 | 191  | 10 | 9,31  | 32,28 |
| 27 | 192 | 201  | 10 | 9,66  | 32,28 |
| 28 | 202 | 212  | 11 | 10,01 | 32,69 |
| 29 | 213 | 223  | 11 | 10,36 | 32,69 |
| 30 | 224 | 235  | 12 | 10,71 | 33,07 |
| 31 | 236 | 247  | 12 | 11,06 | 33,07 |
| 32 | 248 | 260  | 13 | 11,41 | 33,42 |
| 33 | 261 | 273  | 13 | 11,75 | 33,42 |
| 34 | 274 | 287  | 14 | 12,09 | 33,74 |
| 35 | 288 | 302  | 15 | 12,43 | 34,04 |
| 36 | 303 | 318  | 16 | 12,77 | 34,32 |
| 37 | 319 | 335  | 17 | 13,11 | 34,58 |
| 38 | 336 | 353  | 18 | 13,46 | 34,83 |
| 39 | 354 | 372  | 19 | 13,80 | 35,06 |
| 40 | 373 | 392  | 20 | 14,13 | 35,29 |
| 41 | 393 | 414  | 22 | 14,47 | 35,70 |
| 42 | 415 | 437  | 23 | 14,81 | 35,89 |
| 43 | 438 | 462  | 25 | 15,14 | 36,26 |
| 44 | 463 | 489  | 27 | 15,48 | 36,59 |
| 45 | 490 | 518  | 29 | 15,81 | 36,90 |
| 46 | 519 | 549  | 31 | 16,15 | 37,19 |
| 47 | 550 | 583  | 34 | 16,48 | 37,59 |
| 48 | 584 | 619  | 36 | 16,82 | 37,84 |
| 49 | 620 | 658  | 39 | 17,15 | 38,19 |
| 50 | 659 | 700  | 42 | 17,48 | 38,51 |
| 51 | 701 | 745  | 45 | 17,81 | 38,81 |
| 52 | 746 | 794  | 49 | 18,14 | 39,18 |
| 53 | 795 | 847  | 53 | 18,48 | 39,52 |
| 54 | 848 | 904  | 57 | 18,81 | 39,83 |
| 55 | 905 | 965  | 61 | 19,15 | 40,13 |
| 56 | 966 | 1023 | 58 | 19,47 | 39,91 |

Table 4.B.6 – Psychoacoustic parameters for 12 kHz short FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 0      | 1     | 0,00 | 27,28 |
| 1     | 1     | 1      | 1     | 0,47 | 27,28 |
| 2     | 2     | 2      | 1     | 0,95 | 27,28 |
| 3     | 3     | 3      | 1     | 1,42 | 22,28 |
| 4     | 4     | 4      | 1     | 1,88 | 22,28 |
| 5     | 5     | 5      | 1     | 2,35 | 19,28 |
| 6     | 6     | 6      | 1     | 2,81 | 19,28 |
| 7     | 7     | 7      | 1     | 3,26 | 14,28 |
| 8     | 8     | 8      | 1     | 3,70 | 14,28 |
| 9     | 9     | 9      | 1     | 4,14 | 12,28 |
| 10    | 10    | 10     | 1     | 4,57 | 12,28 |
| 11    | 11    | 11     | 1     | 4,98 | 12,28 |
| 12    | 12    | 12     | 1     | 5,39 | 12,28 |
| 13    | 13    | 13     | 1     | 5,79 | 12,28 |
| 14    | 14    | 14     | 1     | 6,18 | 12,28 |
| 15    | 15    | 15     | 1     | 6,56 | 12,28 |

|    |     |     |   |       |       |
|----|-----|-----|---|-------|-------|
| 16 | 16  | 16  | 1 | 6,93  | 12,28 |
| 17 | 17  | 17  | 1 | 7,28  | 12,28 |
| 18 | 18  | 18  | 1 | 7,63  | 12,28 |
| 19 | 19  | 20  | 2 | 8,28  | 15,29 |
| 20 | 21  | 22  | 2 | 8,90  | 15,29 |
| 21 | 23  | 24  | 2 | 9,48  | 15,29 |
| 22 | 25  | 26  | 2 | 10,02 | 15,29 |
| 23 | 27  | 28  | 2 | 10,53 | 15,29 |
| 24 | 29  | 30  | 2 | 11,00 | 15,29 |
| 25 | 31  | 32  | 2 | 11,45 | 15,29 |
| 26 | 33  | 34  | 2 | 11,86 | 15,29 |
| 27 | 35  | 36  | 2 | 12,25 | 15,29 |
| 28 | 37  | 38  | 2 | 12,62 | 15,29 |
| 29 | 39  | 40  | 2 | 12,96 | 15,29 |
| 30 | 41  | 43  | 3 | 13,36 | 17,05 |
| 31 | 44  | 46  | 3 | 13,80 | 17,05 |
| 32 | 47  | 49  | 3 | 14,21 | 17,05 |
| 33 | 50  | 52  | 3 | 14,59 | 17,05 |
| 34 | 53  | 55  | 3 | 14,94 | 17,05 |
| 35 | 56  | 59  | 4 | 15,32 | 18,30 |
| 36 | 60  | 63  | 4 | 15,71 | 18,30 |
| 37 | 64  | 67  | 4 | 16,08 | 18,30 |
| 38 | 68  | 72  | 5 | 16,45 | 19,27 |
| 39 | 73  | 77  | 5 | 16,83 | 19,27 |
| 40 | 78  | 82  | 5 | 17,19 | 19,27 |
| 41 | 83  | 88  | 6 | 17,54 | 20,06 |
| 42 | 89  | 94  | 6 | 17,90 | 20,06 |
| 43 | 95  | 101 | 7 | 18,26 | 20,73 |
| 44 | 102 | 108 | 7 | 18,62 | 20,73 |
| 45 | 109 | 116 | 8 | 18,97 | 21,31 |
| 46 | 117 | 124 | 8 | 19,32 | 21,31 |
| 47 | 125 | 127 | 3 | 19,55 | 17,05 |

Table 4.B.7 – Psychoacoustic parameters for 16 kHz long FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 4      | 5     | 0,20 | 43,30 |
| 1     | 5     | 9      | 5     | 0,59 | 43,10 |
| 2     | 10    | 14     | 5     | 0,99 | 38,30 |
| 3     | 15    | 19     | 5     | 1,38 | 38,10 |
| 4     | 20    | 24     | 5     | 1,77 | 38,00 |
| 5     | 25    | 29     | 5     | 2,16 | 35,10 |
| 6     | 30    | 34     | 5     | 2,54 | 35,30 |
| 7     | 35    | 39     | 5     | 2,92 | 30,00 |
| 8     | 40    | 44     | 5     | 3,29 | 30,00 |
| 9     | 45    | 49     | 5     | 3,66 | 28,30 |
| 10    | 50    | 54     | 5     | 4,03 | 28,30 |
| 11    | 55    | 59     | 5     | 4,39 | 28,30 |
| 12    | 60    | 64     | 5     | 4,74 | 28,30 |
| 13    | 65    | 69     | 5     | 5,09 | 28,30 |
| 14    | 70    | 74     | 5     | 5,43 | 28,30 |
| 15    | 75    | 80     | 6     | 5,79 | 28,30 |
| 16    | 81    | 86     | 6     | 6,18 | 28,30 |
| 17    | 87    | 92     | 6     | 6,56 | 28,00 |
| 18    | 93    | 98     | 6     | 6,92 | 29,27 |
| 19    | 99    | 104    | 6     | 7,28 | 29,27 |

|    |     |      |    |       |       |
|----|-----|------|----|-------|-------|
| 20 | 105 | 110  | 6  | 7,63  | 29,27 |
| 21 | 111 | 116  | 6  | 7,96  | 29,27 |
| 22 | 117 | 123  | 7  | 8,31  | 29,27 |
| 23 | 124 | 130  | 7  | 8,68  | 29,06 |
| 24 | 131 | 137  | 7  | 9,03  | 30,06 |
| 25 | 138 | 144  | 7  | 9,37  | 30,06 |
| 26 | 145 | 152  | 8  | 9,71  | 30,06 |
| 27 | 153 | 160  | 8  | 10,07 | 30,73 |
| 28 | 161 | 168  | 8  | 10,41 | 30,73 |
| 29 | 169 | 177  | 9  | 10,75 | 30,73 |
| 30 | 178 | 186  | 9  | 11,10 | 31,31 |
| 31 | 187 | 196  | 10 | 11,45 | 31,31 |
| 32 | 197 | 206  | 10 | 11,80 | 31,82 |
| 33 | 207 | 217  | 11 | 12,14 | 31,82 |
| 34 | 218 | 228  | 11 | 12,48 | 32,28 |
| 35 | 229 | 240  | 12 | 12,82 | 32,28 |
| 36 | 241 | 253  | 13 | 13,16 | 32,69 |
| 37 | 254 | 267  | 14 | 13,51 | 32,69 |
| 38 | 268 | 282  | 15 | 13,86 | 33,07 |
| 39 | 283 | 298  | 16 | 14,21 | 33,46 |
| 40 | 299 | 315  | 17 | 14,56 | 33,82 |
| 41 | 316 | 333  | 18 | 14,90 | 34,12 |
| 42 | 334 | 352  | 19 | 15,24 | 34,42 |
| 43 | 353 | 373  | 21 | 15,58 | 34,68 |
| 44 | 374 | 395  | 22 | 15,91 | 35,15 |
| 45 | 396 | 419  | 24 | 16,25 | 35,32 |
| 46 | 420 | 445  | 26 | 16,58 | 35,73 |
| 47 | 446 | 473  | 28 | 16,92 | 35,91 |
| 48 | 474 | 503  | 30 | 17,25 | 36,42 |
| 49 | 504 | 536  | 33 | 17,59 | 36,75 |
| 50 | 537 | 571  | 35 | 17,93 | 37,11 |
| 51 | 572 | 609  | 38 | 18,26 | 37,34 |
| 52 | 610 | 650  | 41 | 18,60 | 37,63 |
| 53 | 651 | 694  | 44 | 18,94 | 38,12 |
| 54 | 695 | 741  | 47 | 19,27 | 38,17 |
| 55 | 742 | 791  | 50 | 19,60 | 41,52 |
| 56 | 792 | 845  | 54 | 19,94 | 41,84 |
| 57 | 846 | 903  | 58 | 20,27 | 42,13 |
| 58 | 904 | 965  | 62 | 20,61 | 44,41 |
| 59 | 966 | 1023 | 58 | 20,92 | 44,87 |

Table 4.B.8 – Psychoacoustic parameters for 16 kHz short FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 0      | 1     | 0,00 | 27,28 |
| 1     | 1     | 1      | 1     | 0,63 | 27,28 |
| 2     | 2     | 2      | 1     | 1,26 | 22,28 |
| 3     | 3     | 3      | 1     | 1,88 | 22,28 |
| 4     | 4     | 4      | 1     | 2,50 | 19,28 |
| 5     | 5     | 5      | 1     | 3,11 | 14,28 |
| 6     | 6     | 6      | 1     | 3,70 | 14,28 |
| 7     | 7     | 7      | 1     | 4,28 | 12,28 |
| 8     | 8     | 8      | 1     | 4,85 | 12,28 |
| 9     | 9     | 9      | 1     | 5,39 | 12,28 |

|    |     |     |   |       |       |
|----|-----|-----|---|-------|-------|
| 10 | 10  | 10  | 1 | 5,92  | 12,28 |
| 11 | 11  | 11  | 1 | 6,43  | 12,28 |
| 12 | 12  | 12  | 1 | 6,93  | 12,28 |
| 13 | 13  | 13  | 1 | 7,40  | 12,28 |
| 14 | 14  | 14  | 1 | 7,85  | 12,28 |
| 15 | 15  | 15  | 1 | 8,29  | 12,28 |
| 16 | 16  | 16  | 1 | 8,70  | 12,28 |
| 17 | 17  | 17  | 1 | 9,10  | 12,28 |
| 18 | 18  | 18  | 1 | 9,49  | 12,28 |
| 19 | 19  | 19  | 1 | 9,85  | 12,28 |
| 20 | 20  | 20  | 1 | 10,20 | 12,28 |
| 21 | 21  | 22  | 2 | 10,85 | 15,29 |
| 22 | 23  | 24  | 2 | 11,44 | 15,29 |
| 23 | 25  | 26  | 2 | 11,99 | 15,29 |
| 24 | 27  | 28  | 2 | 12,50 | 15,29 |
| 25 | 29  | 30  | 2 | 12,96 | 15,29 |
| 26 | 31  | 32  | 2 | 13,39 | 15,29 |
| 27 | 33  | 34  | 2 | 13,78 | 15,29 |
| 28 | 35  | 36  | 2 | 14,15 | 15,29 |
| 29 | 37  | 39  | 3 | 14,57 | 17,05 |
| 30 | 40  | 42  | 3 | 15,03 | 17,05 |
| 31 | 43  | 45  | 3 | 15,45 | 17,05 |
| 32 | 46  | 48  | 3 | 15,84 | 17,05 |
| 33 | 49  | 51  | 3 | 16,19 | 17,05 |
| 34 | 52  | 55  | 4 | 16,57 | 18,30 |
| 35 | 56  | 59  | 4 | 16,97 | 18,30 |
| 36 | 60  | 63  | 4 | 17,33 | 18,30 |
| 37 | 64  | 68  | 5 | 17,71 | 19,27 |
| 38 | 69  | 73  | 5 | 18,09 | 19,27 |
| 39 | 74  | 78  | 5 | 18,44 | 19,27 |
| 40 | 79  | 84  | 6 | 18,80 | 20,06 |
| 41 | 85  | 90  | 6 | 19,17 | 20,06 |
| 42 | 91  | 97  | 7 | 19,53 | 20,73 |
| 43 | 98  | 104 | 7 | 19,89 | 20,73 |
| 44 | 105 | 112 | 8 | 20,25 | 24,31 |
| 45 | 113 | 120 | 8 | 20,61 | 24,31 |
| 46 | 121 | 127 | 7 | 20,92 | 23,73 |

Table 4.B.9 – Psychoacoustic parameters for 22.05 kHz long FFT

| index | w low | w high | width | bval | qstr  |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 3      | 4     | 0,22 | 43,30 |
| 1     | 4     | 7      | 4     | 0,65 | 43,30 |
| 2     | 8     | 11     | 4     | 1,09 | 38,30 |
| 3     | 12    | 15     | 4     | 1,52 | 38,30 |
| 4     | 16    | 19     | 4     | 1,95 | 38,30 |
| 5     | 20    | 23     | 4     | 2,37 | 35,30 |
| 6     | 24    | 27     | 4     | 2,79 | 35,30 |
| 7     | 28    | 31     | 4     | 3,21 | 30,30 |
| 8     | 32    | 35     | 4     | 3,62 | 30,30 |
| 9     | 36    | 39     | 4     | 4,02 | 28,30 |
| 10    | 40    | 43     | 4     | 4,41 | 28,30 |
| 11    | 44    | 47     | 4     | 4,80 | 28,30 |
| 12    | 48    | 51     | 4     | 5,18 | 28,30 |
| 13    | 52    | 55     | 4     | 5,55 | 28,30 |
| 14    | 56    | 59     | 4     | 5,92 | 28,30 |

|    |     |      |    |       |       |
|----|-----|------|----|-------|-------|
| 15 | 60  | 63   | 4  | 6,27  | 28,30 |
| 16 | 64  | 67   | 4  | 6,62  | 28,30 |
| 17 | 68  | 71   | 4  | 6,95  | 28,30 |
| 18 | 72  | 76   | 5  | 7,32  | 29,27 |
| 19 | 77  | 81   | 5  | 7,71  | 29,27 |
| 20 | 82  | 86   | 5  | 8,10  | 29,27 |
| 21 | 87  | 91   | 5  | 8,46  | 29,27 |
| 22 | 92  | 96   | 5  | 8,82  | 29,27 |
| 23 | 97  | 101  | 5  | 9,16  | 29,27 |
| 24 | 102 | 107  | 6  | 9,52  | 30,06 |
| 25 | 108 | 113  | 6  | 9,89  | 30,06 |
| 26 | 114 | 119  | 6  | 10,25 | 30,06 |
| 27 | 120 | 125  | 6  | 10,59 | 30,06 |
| 28 | 126 | 132  | 7  | 10,95 | 30,73 |
| 29 | 133 | 139  | 7  | 11,31 | 30,73 |
| 30 | 140 | 146  | 7  | 11,65 | 30,73 |
| 31 | 147 | 154  | 8  | 12,00 | 31,31 |
| 32 | 155 | 162  | 8  | 12,35 | 31,31 |
| 33 | 163 | 171  | 9  | 12,70 | 31,82 |
| 34 | 172 | 180  | 9  | 13,05 | 31,82 |
| 35 | 181 | 190  | 10 | 13,40 | 32,28 |
| 36 | 191 | 200  | 10 | 13,74 | 32,28 |
| 37 | 201 | 211  | 11 | 14,07 | 32,69 |
| 38 | 212 | 223  | 12 | 14,41 | 33,07 |
| 39 | 224 | 236  | 13 | 14,76 | 33,42 |
| 40 | 237 | 250  | 14 | 15,11 | 33,74 |
| 41 | 251 | 265  | 15 | 15,46 | 34,04 |
| 42 | 266 | 281  | 16 | 15,80 | 34,32 |
| 43 | 282 | 298  | 17 | 16,14 | 34,58 |
| 44 | 299 | 317  | 19 | 16,48 | 35,06 |
| 45 | 318 | 337  | 20 | 16,82 | 35,29 |
| 46 | 338 | 359  | 22 | 17,16 | 35,70 |
| 47 | 360 | 382  | 23 | 17,50 | 35,89 |
| 48 | 383 | 407  | 25 | 17,84 | 36,26 |
| 49 | 408 | 434  | 27 | 18,17 | 36,59 |
| 50 | 435 | 463  | 29 | 18,51 | 36,90 |
| 51 | 464 | 494  | 31 | 18,84 | 37,19 |
| 52 | 495 | 527  | 33 | 19,17 | 37,46 |
| 53 | 528 | 563  | 36 | 19,51 | 37,84 |
| 54 | 564 | 601  | 38 | 19,84 | 38,07 |
| 55 | 602 | 642  | 41 | 20,17 | 41,40 |
| 56 | 643 | 686  | 44 | 20,50 | 41,71 |
| 57 | 687 | 733  | 47 | 20,84 | 42,00 |
| 58 | 734 | 784  | 51 | 21,17 | 44,35 |
| 59 | 785 | 839  | 55 | 21,50 | 44,68 |
| 60 | 840 | 898  | 59 | 21,84 | 44,98 |
| 61 | 899 | 962  | 64 | 22,17 | 50,34 |
| 62 | 963 | 1023 | 61 | 22,48 | 50,13 |

Table 4.B.10 – Psychoacoustic parameters for 22.05 kHz short FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 0      | 1     | 0,00 | 27,28 |
| 1     | 1     | 1      | 1     | 0,87 | 27,28 |
| 2     | 2     | 2      | 1     | 1,73 | 22,28 |
| 3     | 3     | 3      | 1     | 2,58 | 19,28 |

|    |     |     |   |       |       |
|----|-----|-----|---|-------|-------|
| 4  | 4   | 4   | 1 | 3,41  | 14,28 |
| 5  | 5   | 5   | 1 | 4,22  | 12,28 |
| 6  | 6   | 6   | 1 | 4,99  | 12,28 |
| 7  | 7   | 7   | 1 | 5,74  | 12,28 |
| 8  | 8   | 8   | 1 | 6,45  | 12,28 |
| 9  | 9   | 9   | 1 | 7,12  | 12,28 |
| 10 | 10  | 10  | 1 | 7,75  | 12,28 |
| 11 | 11  | 11  | 1 | 8,36  | 12,28 |
| 12 | 12  | 12  | 1 | 8,92  | 12,28 |
| 13 | 13  | 13  | 1 | 9,45  | 12,28 |
| 14 | 14  | 14  | 1 | 9,96  | 12,28 |
| 15 | 15  | 15  | 1 | 10,43 | 12,28 |
| 16 | 16  | 16  | 1 | 10,87 | 12,28 |
| 17 | 17  | 17  | 1 | 11,29 | 12,28 |
| 18 | 18  | 18  | 1 | 11,68 | 12,28 |
| 19 | 19  | 19  | 1 | 12,05 | 12,28 |
| 20 | 20  | 21  | 2 | 12,71 | 15,29 |
| 21 | 22  | 23  | 2 | 13,32 | 15,29 |
| 22 | 24  | 25  | 2 | 13,86 | 15,29 |
| 23 | 26  | 27  | 2 | 14,35 | 15,29 |
| 24 | 28  | 29  | 2 | 14,80 | 15,29 |
| 25 | 30  | 31  | 2 | 15,21 | 15,29 |
| 26 | 32  | 33  | 2 | 15,58 | 15,29 |
| 27 | 34  | 35  | 2 | 15,93 | 15,29 |
| 28 | 36  | 38  | 3 | 16,32 | 17,05 |
| 29 | 39  | 41  | 3 | 16,75 | 17,05 |
| 30 | 42  | 44  | 3 | 17,15 | 17,05 |
| 31 | 45  | 47  | 3 | 17,51 | 17,05 |
| 32 | 48  | 51  | 4 | 17,89 | 18,30 |
| 33 | 52  | 55  | 4 | 18,30 | 18,30 |
| 34 | 56  | 59  | 4 | 18,67 | 18,30 |
| 35 | 60  | 63  | 4 | 19,02 | 18,30 |
| 36 | 64  | 68  | 5 | 19,37 | 19,27 |
| 37 | 69  | 73  | 5 | 19,74 | 19,27 |
| 38 | 74  | 78  | 5 | 20,09 | 22,27 |
| 39 | 79  | 84  | 6 | 20,44 | 23,06 |
| 40 | 85  | 90  | 6 | 20,79 | 23,06 |
| 41 | 91  | 97  | 7 | 21,15 | 25,73 |
| 42 | 98  | 104 | 7 | 21,50 | 25,73 |
| 43 | 105 | 112 | 8 | 21,85 | 26,31 |
| 44 | 113 | 120 | 8 | 22,20 | 31,31 |
| 45 | 121 | 127 | 7 | 22,49 | 30,73 |

Table 4.B.11 – Psychoacoustic parameters for 24 kHz long FFT

| index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 2      | 3     | 0,18 | 42,05 |
| 1     | 3     | 5      | 3     | 0,53 | 42,05 |
| 2     | 6     | 8      | 3     | 0,89 | 42,05 |
| 3     | 9     | 11     | 3     | 1,24 | 37,05 |
| 4     | 12    | 14     | 3     | 1,59 | 37,05 |
| 5     | 15    | 17     | 3     | 1,94 | 37,05 |
| 6     | 18    | 20     | 3     | 2,29 | 34,05 |
| 7     | 21    | 23     | 3     | 2,63 | 34,05 |
| 8     | 24    | 26     | 3     | 2,98 | 34,05 |
| 9     | 27    | 29     | 3     | 3,31 | 29,05 |

|    |     |      |    |       |       |
|----|-----|------|----|-------|-------|
| 10 | 30  | 32   | 3  | 3,65  | 29,05 |
| 11 | 33  | 36   | 4  | 4,03  | 28,30 |
| 12 | 37  | 40   | 4  | 4,46  | 28,30 |
| 13 | 41  | 44   | 4  | 4,88  | 28,30 |
| 14 | 45  | 48   | 4  | 5,29  | 28,30 |
| 15 | 49  | 52   | 4  | 5,69  | 28,30 |
| 16 | 53  | 56   | 4  | 6,08  | 28,30 |
| 17 | 57  | 60   | 4  | 6,46  | 28,30 |
| 18 | 61  | 64   | 4  | 6,83  | 28,30 |
| 19 | 65  | 68   | 4  | 7,19  | 28,30 |
| 20 | 69  | 72   | 4  | 7,54  | 28,30 |
| 21 | 73  | 76   | 4  | 7,88  | 28,30 |
| 22 | 77  | 81   | 5  | 8,25  | 29,27 |
| 23 | 82  | 86   | 5  | 8,64  | 29,27 |
| 24 | 87  | 91   | 5  | 9,02  | 29,27 |
| 25 | 92  | 96   | 5  | 9,38  | 29,27 |
| 26 | 97  | 101  | 5  | 9,73  | 29,27 |
| 27 | 102 | 107  | 6  | 10,09 | 30,06 |
| 28 | 108 | 113  | 6  | 10,47 | 30,06 |
| 29 | 114 | 119  | 6  | 10,83 | 30,06 |
| 30 | 120 | 125  | 6  | 11,18 | 30,06 |
| 31 | 126 | 132  | 7  | 11,53 | 30,73 |
| 32 | 133 | 139  | 7  | 11,89 | 30,73 |
| 33 | 140 | 146  | 7  | 12,23 | 30,73 |
| 34 | 147 | 154  | 8  | 12,57 | 31,31 |
| 35 | 155 | 162  | 8  | 12,92 | 31,31 |
| 36 | 163 | 171  | 9  | 13,26 | 31,82 |
| 37 | 172 | 180  | 9  | 13,61 | 31,82 |
| 38 | 181 | 190  | 10 | 13,95 | 32,28 |
| 39 | 191 | 201  | 11 | 14,29 | 32,69 |
| 40 | 202 | 213  | 12 | 14,65 | 33,07 |
| 41 | 214 | 225  | 12 | 15,00 | 33,07 |
| 42 | 226 | 238  | 13 | 15,33 | 33,42 |
| 43 | 239 | 252  | 14 | 15,66 | 33,74 |
| 44 | 253 | 267  | 15 | 16,00 | 34,04 |
| 45 | 268 | 284  | 17 | 16,34 | 34,58 |
| 46 | 285 | 302  | 18 | 16,69 | 34,83 |
| 47 | 303 | 321  | 19 | 17,02 | 35,06 |
| 48 | 322 | 342  | 21 | 17,36 | 35,50 |
| 49 | 343 | 364  | 22 | 17,70 | 35,70 |
| 50 | 365 | 388  | 24 | 18,03 | 36,08 |
| 51 | 389 | 414  | 26 | 18,37 | 36,43 |
| 52 | 415 | 442  | 28 | 18,70 | 36,75 |
| 53 | 443 | 472  | 30 | 19,04 | 37,05 |
| 54 | 473 | 504  | 32 | 19,38 | 37,33 |
| 55 | 505 | 538  | 34 | 19,71 | 37,59 |
| 56 | 539 | 575  | 37 | 20,04 | 40,96 |
| 57 | 576 | 614  | 39 | 20,38 | 41,19 |
| 58 | 615 | 656  | 42 | 20,71 | 41,51 |
| 59 | 657 | 701  | 45 | 21,04 | 43,81 |
| 60 | 702 | 750  | 49 | 21,37 | 44,18 |
| 61 | 751 | 803  | 53 | 21,70 | 44,52 |
| 62 | 804 | 860  | 57 | 22,04 | 49,83 |
| 63 | 861 | 922  | 62 | 22,37 | 50,20 |
| 64 | 923 | 989  | 67 | 22,70 | 50,54 |
| 65 | 990 | 1023 | 34 | 22,95 | 47,59 |

Table 4.B.12 – Psychoacoustic parameters for 24 kHz short FFT

| index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0,00  | 27,28 |
| 1     | 1     | 1      | 1     | 0,95  | 27,28 |
| 2     | 2     | 2      | 1     | 1,88  | 22,28 |
| 3     | 3     | 3      | 1     | 2,81  | 19,28 |
| 4     | 4     | 4      | 1     | 3,70  | 14,28 |
| 5     | 5     | 5      | 1     | 4,57  | 12,28 |
| 6     | 6     | 6      | 1     | 5,39  | 12,28 |
| 7     | 7     | 7      | 1     | 6,18  | 12,28 |
| 8     | 8     | 8      | 1     | 6,93  | 12,28 |
| 9     | 9     | 9      | 1     | 7,63  | 12,28 |
| 10    | 10    | 10     | 1     | 8,29  | 12,28 |
| 11    | 11    | 11     | 1     | 8,91  | 12,28 |
| 12    | 12    | 12     | 1     | 9,49  | 12,28 |
| 13    | 13    | 13     | 1     | 10,03 | 12,28 |
| 14    | 14    | 14     | 1     | 10,53 | 12,28 |
| 15    | 15    | 15     | 1     | 11,01 | 12,28 |
| 16    | 16    | 16     | 1     | 11,45 | 12,28 |
| 17    | 17    | 17     | 1     | 11,87 | 12,28 |
| 18    | 18    | 18     | 1     | 12,26 | 12,28 |
| 19    | 19    | 19     | 1     | 12,62 | 12,28 |
| 20    | 20    | 21     | 2     | 13,28 | 15,29 |
| 21    | 22    | 23     | 2     | 13,87 | 15,29 |
| 22    | 24    | 25     | 2     | 14,40 | 15,29 |
| 23    | 26    | 27     | 2     | 14,88 | 15,29 |
| 24    | 28    | 29     | 2     | 15,32 | 15,29 |
| 25    | 30    | 31     | 2     | 15,71 | 15,29 |
| 26    | 32    | 33     | 2     | 16,08 | 15,29 |
| 27    | 34    | 36     | 3     | 16,49 | 17,05 |
| 28    | 37    | 39     | 3     | 16,94 | 17,05 |
| 29    | 40    | 42     | 3     | 17,35 | 17,05 |
| 30    | 43    | 45     | 3     | 17,73 | 17,05 |
| 31    | 46    | 48     | 3     | 18,07 | 17,05 |
| 32    | 49    | 52     | 4     | 18,44 | 18,30 |
| 33    | 53    | 56     | 4     | 18,83 | 18,30 |
| 34    | 57    | 60     | 4     | 19,20 | 18,30 |
| 35    | 61    | 65     | 5     | 19,57 | 19,27 |
| 36    | 66    | 70     | 5     | 19,96 | 19,27 |
| 37    | 71    | 75     | 5     | 20,31 | 22,27 |
| 38    | 76    | 81     | 6     | 20,67 | 23,06 |
| 39    | 82    | 87     | 6     | 21,04 | 25,06 |
| 40    | 88    | 94     | 7     | 21,41 | 25,73 |
| 41    | 95    | 101    | 7     | 21,77 | 25,73 |
| 42    | 102   | 109    | 8     | 22,13 | 31,31 |
| 43    | 110   | 117    | 8     | 22,48 | 31,31 |
| 44    | 118   | 126    | 9     | 22,82 | 31,82 |
| 45    | 127   | 127    | 1     | 23,01 | 32,28 |

Table 4.B.13 – Psychoacoustic parameters for 32 kHz long FFT

| Index | w low | w high | width | bval | qsthr |
|-------|-------|--------|-------|------|-------|
| 0     | 0     | 2      | 3     | 0.24 | 42.05 |
| 1     | 3     | 5      | 3     | 0.71 | 42.05 |
| 2     | 6     | 8      | 3     | 1.18 | 37.05 |
| 3     | 9     | 11     | 3     | 1.65 | 37.05 |



|    |     |     |    |       |       |
|----|-----|-----|----|-------|-------|
| 4  | 12  | 14  | 3  | 2.12  | 34.05 |
| 5  | 15  | 17  | 3  | 2.58  | 34.05 |
| 6  | 18  | 20  | 3  | 3.03  | 29.05 |
| 7  | 21  | 23  | 3  | 3.48  | 29.05 |
| 8  | 24  | 26  | 3  | 3.92  | 29.05 |
| 9  | 27  | 29  | 3  | 4.35  | 27.05 |
| 10 | 30  | 32  | 3  | 4.77  | 27.05 |
| 11 | 33  | 35  | 3  | 5.19  | 27.05 |
| 12 | 36  | 38  | 3  | 5.59  | 27.05 |
| 13 | 39  | 41  | 3  | 5.99  | 27.05 |
| 14 | 42  | 44  | 3  | 6.37  | 27.05 |
| 15 | 45  | 47  | 3  | 6.74  | 27.05 |
| 16 | 48  | 50  | 3  | 7.10  | 27.05 |
| 17 | 51  | 53  | 3  | 7.45  | 27.05 |
| 18 | 54  | 56  | 3  | 7.80  | 27.05 |
| 19 | 57  | 60  | 4  | 8.18  | 28.30 |
| 20 | 61  | 64  | 4  | 8.60  | 28.30 |
| 21 | 65  | 68  | 4  | 9.00  | 28.30 |
| 22 | 69  | 72  | 4  | 9.39  | 28.30 |
| 23 | 73  | 76  | 4  | 9.76  | 28.30 |
| 24 | 77  | 80  | 4  | 10.11 | 28.30 |
| 25 | 81  | 84  | 4  | 10.45 | 28.30 |
| 26 | 85  | 89  | 5  | 10.81 | 29.27 |
| 27 | 90  | 94  | 5  | 11.19 | 29.27 |
| 28 | 95  | 99  | 5  | 11.55 | 29.27 |
| 29 | 100 | 104 | 5  | 11.90 | 29.27 |
| 30 | 105 | 110 | 6  | 12.25 | 30.06 |
| 31 | 111 | 116 | 6  | 12.62 | 30.06 |
| 32 | 117 | 122 | 6  | 12.96 | 30.06 |
| 33 | 123 | 129 | 7  | 13.31 | 30.73 |
| 34 | 130 | 136 | 7  | 13.66 | 30.73 |
| 35 | 137 | 144 | 8  | 14.01 | 31.31 |
| 36 | 145 | 152 | 8  | 14.36 | 31.31 |
| 37 | 153 | 161 | 9  | 14.71 | 31.82 |
| 38 | 162 | 171 | 10 | 15.07 | 32.28 |
| 39 | 172 | 181 | 10 | 15.42 | 32.28 |
| 40 | 182 | 192 | 11 | 15.76 | 32.69 |
| 41 | 193 | 204 | 12 | 16.10 | 33.07 |
| 42 | 205 | 217 | 13 | 16.45 | 33.42 |
| 43 | 218 | 231 | 14 | 16.80 | 33.74 |
| 44 | 232 | 246 | 15 | 17.14 | 34.04 |
| 45 | 247 | 262 | 16 | 17.48 | 34.32 |
| 46 | 263 | 279 | 17 | 17.82 | 34.58 |
| 47 | 280 | 298 | 19 | 18.15 | 35.06 |
| 48 | 299 | 318 | 20 | 18.49 | 35.29 |
| 49 | 319 | 340 | 22 | 18.84 | 35.70 |
| 50 | 341 | 363 | 23 | 19.17 | 35.89 |
| 51 | 364 | 388 | 25 | 19.51 | 36.26 |
| 52 | 389 | 415 | 27 | 19.85 | 36.59 |
| 53 | 416 | 444 | 29 | 20.19 | 39.90 |
| 54 | 445 | 475 | 31 | 20.53 | 40.19 |
| 55 | 476 | 508 | 33 | 20.87 | 40.46 |
| 56 | 509 | 543 | 35 | 21.20 | 42.72 |
| 57 | 544 | 581 | 38 | 21.53 | 43.07 |
| 58 | 582 | 622 | 41 | 21.86 | 43.40 |
| 59 | 623 | 667 | 45 | 22.20 | 48.81 |
| 60 | 668 | 715 | 48 | 22.53 | 49.09 |

|    |     |      |    |       |       |
|----|-----|------|----|-------|-------|
| 61 | 716 | 768  | 53 | 22.86 | 49.52 |
| 62 | 769 | 826  | 58 | 23.20 | 59.91 |
| 63 | 827 | 890  | 64 | 23.53 | 60.34 |
| 64 | 891 | 961  | 71 | 23.86 | 60.79 |
| 65 | 962 | 1023 | 62 | 24.00 | 65.89 |

Table 4.B.14 – Psychoacoustic parameters for 32 kHz short FFT

| Index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0.00  | 27.28 |
| 1     | 1     | 1      | 1     | 1.26  | 22.28 |
| 2     | 2     | 2      | 1     | 2.50  | 19.28 |
| 3     | 3     | 3      | 1     | 3.70  | 14.28 |
| 4     | 4     | 4      | 1     | 4.85  | 12.28 |
| 5     | 5     | 5      | 1     | 5.92  | 12.28 |
| 6     | 6     | 6      | 1     | 6.93  | 12.28 |
| 7     | 7     | 7      | 1     | 7.85  | 12.28 |
| 8     | 8     | 8      | 1     | 8.70  | 12.28 |
| 9     | 9     | 9      | 1     | 9.49  | 12.28 |
| 10    | 10    | 10     | 1     | 10.20 | 12.28 |
| 11    | 11    | 11     | 1     | 10.85 | 12.28 |
| 12    | 12    | 12     | 1     | 11.45 | 12.28 |
| 13    | 13    | 13     | 1     | 12.00 | 12.28 |
| 14    | 14    | 14     | 1     | 12.50 | 12.28 |
| 15    | 15    | 15     | 1     | 12.96 | 12.28 |
| 16    | 16    | 16     | 1     | 13.39 | 12.28 |
| 17    | 17    | 17     | 1     | 13.78 | 12.28 |
| 18    | 18    | 18     | 1     | 14.15 | 12.28 |
| 19    | 19    | 20     | 2     | 14.80 | 15.29 |
| 20    | 21    | 22     | 2     | 15.38 | 15.29 |
| 21    | 23    | 24     | 2     | 15.89 | 15.29 |
| 22    | 25    | 26     | 2     | 16.36 | 15.29 |
| 23    | 27    | 28     | 2     | 16.77 | 15.29 |
| 24    | 29    | 30     | 2     | 17.15 | 15.29 |
| 25    | 31    | 32     | 2     | 17.50 | 15.29 |
| 26    | 33    | 35     | 3     | 17.90 | 17.05 |
| 27    | 36    | 38     | 3     | 18.34 | 17.05 |
| 28    | 39    | 41     | 3     | 18.74 | 17.05 |
| 29    | 42    | 44     | 3     | 19.11 | 17.05 |
| 30    | 45    | 48     | 4     | 19.50 | 18.30 |
| 31    | 49    | 52     | 4     | 19.92 | 18.30 |
| 32    | 53    | 56     | 4     | 20.30 | 21.30 |
| 33    | 57    | 60     | 4     | 20.65 | 21.30 |
| 34    | 61    | 65     | 5     | 21.02 | 24.27 |
| 35    | 66    | 70     | 5     | 21.40 | 24.27 |
| 36    | 71    | 75     | 5     | 21.75 | 24.27 |
| 37    | 76    | 81     | 6     | 22.10 | 30.06 |
| 38    | 82    | 87     | 6     | 22.45 | 30.06 |
| 39    | 88    | 94     | 7     | 22.80 | 30.73 |
| 40    | 95    | 102    | 8     | 23.16 | 41.31 |
| 41    | 103   | 110    | 8     | 23.51 | 41.31 |
| 42    | 111   | 119    | 9     | 23.85 | 41.82 |
| 43    | 120   | 127    | 8     | 24.00 | 60.47 |

Table 4.B.15 – Psychoacoustic parameters for 44.1 kHz long FFT

| Index | w low | w high | width | bval  | gstr  |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 1      | 2     | 0.22  | 40.29 |
| 1     | 2     | 3      | 2     | 0.65  | 40.29 |
| 2     | 4     | 5      | 2     | 1.09  | 35.29 |
| 3     | 6     | 7      | 2     | 1.52  | 35.29 |
| 4     | 8     | 9      | 2     | 1.95  | 35.29 |
| 5     | 10    | 11     | 2     | 2.37  | 32.29 |
| 6     | 12    | 13     | 2     | 2.79  | 32.29 |
| 7     | 14    | 15     | 2     | 3.21  | 27.29 |
| 8     | 16    | 17     | 2     | 3.62  | 27.29 |
| 9     | 18    | 19     | 2     | 4.02  | 25.29 |
| 10    | 20    | 21     | 2     | 4.41  | 25.29 |
| 11    | 22    | 23     | 2     | 4.80  | 25.29 |
| 12    | 24    | 25     | 2     | 5.18  | 25.29 |
| 13    | 26    | 27     | 2     | 5.55  | 25.29 |
| 14    | 28    | 29     | 2     | 5.92  | 25.29 |
| 15    | 30    | 31     | 2     | 6.27  | 25.29 |
| 16    | 32    | 33     | 2     | 6.62  | 25.29 |
| 17    | 34    | 35     | 2     | 6.95  | 25.29 |
| 18    | 36    | 38     | 3     | 7.36  | 27.05 |
| 19    | 39    | 41     | 3     | 7.83  | 27.05 |
| 20    | 42    | 44     | 3     | 8.28  | 27.05 |
| 21    | 45    | 47     | 3     | 8.71  | 27.05 |
| 22    | 48    | 50     | 3     | 9.12  | 27.05 |
| 23    | 51    | 53     | 3     | 9.52  | 27.05 |
| 24    | 54    | 56     | 3     | 9.89  | 27.05 |
| 25    | 57    | 59     | 3     | 10.25 | 27.05 |
| 26    | 60    | 62     | 3     | 10.59 | 27.05 |
| 27    | 63    | 66     | 4     | 10.97 | 28.30 |
| 28    | 67    | 70     | 4     | 11.38 | 28.30 |
| 29    | 71    | 74     | 4     | 11.77 | 28.30 |
| 30    | 75    | 78     | 4     | 12.13 | 28.30 |
| 31    | 79    | 82     | 4     | 12.48 | 28.30 |
| 32    | 83    | 87     | 5     | 12.84 | 29.27 |
| 33    | 88    | 92     | 5     | 13.22 | 29.27 |
| 34    | 93    | 97     | 5     | 13.57 | 29.27 |
| 35    | 98    | 103    | 6     | 13.93 | 30.06 |
| 36    | 104   | 109    | 6     | 14.30 | 30.06 |
| 37    | 110   | 116    | 7     | 14.67 | 30.73 |
| 38    | 117   | 123    | 7     | 15.03 | 30.73 |
| 39    | 124   | 131    | 8     | 15.40 | 31.31 |
| 40    | 132   | 139    | 8     | 15.76 | 31.31 |
| 41    | 140   | 148    | 9     | 16.11 | 31.82 |
| 42    | 149   | 157    | 9     | 16.45 | 31.82 |
| 43    | 158   | 167    | 10    | 16.79 | 32.28 |
| 44    | 168   | 178    | 11    | 17.13 | 32.69 |
| 45    | 179   | 190    | 12    | 17.48 | 33.07 |
| 46    | 191   | 203    | 13    | 17.83 | 33.42 |
| 47    | 204   | 217    | 14    | 18.18 | 33.74 |
| 48    | 218   | 232    | 15    | 18.52 | 34.04 |
| 49    | 233   | 248    | 16    | 18.87 | 34.32 |
| 50    | 249   | 265    | 17    | 19.21 | 34.58 |
| 51    | 266   | 283    | 18    | 19.54 | 34.83 |
| 52    | 284   | 303    | 20    | 19.88 | 35.29 |
| 53    | 304   | 324    | 21    | 20.22 | 38.50 |
| 54    | 325   | 347    | 23    | 20.56 | 38.89 |

|    |     |      |    |       |       |
|----|-----|------|----|-------|-------|
| 55 | 348 | 371  | 24 | 20.90 | 39.08 |
| 56 | 372 | 397  | 26 | 21.24 | 41.43 |
| 57 | 398 | 425  | 28 | 21.57 | 41.75 |
| 58 | 426 | 455  | 30 | 21.91 | 42.05 |
| 59 | 456 | 488  | 33 | 22.24 | 47.46 |
| 60 | 489 | 524  | 36 | 22.58 | 47.84 |
| 61 | 525 | 563  | 39 | 22.91 | 48.19 |
| 62 | 564 | 606  | 43 | 23.25 | 58.61 |
| 63 | 607 | 653  | 47 | 23.58 | 59.00 |
| 64 | 654 | 706  | 53 | 23.91 | 59.52 |
| 65 | 707 | 765  | 59 | 24.00 | 69.98 |
| 66 | 766 | 832  | 67 | 24.00 | 70.54 |
| 67 | 833 | 908  | 76 | 24.00 | 71.08 |
| 68 | 909 | 996  | 88 | 24.00 | 71.72 |
| 69 | 997 | 1023 | 27 | 24.00 | 72.09 |

Table 4.B.16 – Psychoacoustic parameters for 44.1 kHz short FFT

| Index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0.00  | 27.28 |
| 1     | 1     | 1      | 1     | 1.73  | 22.28 |
| 2     | 2     | 2      | 1     | 3.41  | 14.28 |
| 3     | 3     | 3      | 1     | 4.99  | 12.28 |
| 4     | 4     | 4      | 1     | 6.45  | 12.28 |
| 5     | 5     | 5      | 1     | 7.75  | 12.28 |
| 6     | 6     | 6      | 1     | 8.92  | 12.28 |
| 7     | 7     | 7      | 1     | 9.96  | 12.28 |
| 8     | 8     | 8      | 1     | 10.87 | 12.28 |
| 9     | 9     | 9      | 1     | 11.68 | 12.28 |
| 10    | 10    | 10     | 1     | 12.39 | 12.28 |
| 11    | 11    | 11     | 1     | 13.03 | 12.28 |
| 12    | 12    | 12     | 1     | 13.61 | 12.28 |
| 13    | 13    | 13     | 1     | 14.12 | 12.28 |
| 14    | 14    | 14     | 1     | 14.59 | 12.28 |
| 15    | 15    | 15     | 1     | 15.01 | 12.28 |
| 16    | 16    | 16     | 1     | 15.40 | 12.28 |
| 17    | 17    | 17     | 1     | 15.76 | 12.28 |
| 18    | 18    | 19     | 2     | 16.39 | 15.29 |
| 19    | 20    | 21     | 2     | 16.95 | 15.29 |
| 20    | 22    | 23     | 2     | 17.45 | 15.29 |
| 21    | 24    | 25     | 2     | 17.89 | 15.29 |
| 22    | 26    | 27     | 2     | 18.30 | 15.29 |
| 23    | 28    | 29     | 2     | 18.67 | 15.29 |
| 24    | 30    | 31     | 2     | 19.02 | 15.29 |
| 25    | 32    | 34     | 3     | 19.41 | 17.05 |
| 26    | 35    | 37     | 3     | 19.85 | 17.05 |
| 27    | 38    | 40     | 3     | 20.25 | 20.05 |
| 28    | 41    | 43     | 3     | 20.62 | 20.05 |
| 29    | 44    | 47     | 4     | 21.01 | 23.30 |
| 30    | 48    | 51     | 4     | 21.43 | 23.30 |
| 31    | 52    | 55     | 4     | 21.81 | 23.30 |
| 32    | 56    | 59     | 4     | 22.15 | 28.30 |
| 33    | 60    | 64     | 5     | 22.51 | 29.27 |
| 34    | 65    | 69     | 5     | 22.87 | 29.27 |
| 35    | 70    | 75     | 6     | 23.23 | 40.06 |
| 36    | 76    | 81     | 6     | 23.59 | 40.06 |

|    |     |     |    |       |       |
|----|-----|-----|----|-------|-------|
| 37 | 82  | 88  | 7  | 23.93 | 40.73 |
| 38 | 89  | 96  | 8  | 24.00 | 51.31 |
| 39 | 97  | 105 | 9  | 24.00 | 51.82 |
| 40 | 106 | 115 | 10 | 24.00 | 52.28 |
| 41 | 116 | 127 | 12 | 24.00 | 53.07 |

Table 4.B.17 – Psychoacoustic parameters for 48 kHz long FFT

| Index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 1      | 2     | 0.24  | 40.29 |
| 1     | 2     | 3      | 2     | 0.71  | 40.29 |
| 2     | 4     | 5      | 2     | 1.18  | 35.29 |
| 3     | 6     | 7      | 2     | 1.65  | 35.29 |
| 4     | 8     | 9      | 2     | 2.12  | 32.29 |
| 5     | 10    | 11     | 2     | 2.58  | 32.29 |
| 6     | 12    | 13     | 2     | 3.03  | 27.29 |
| 7     | 14    | 15     | 2     | 3.48  | 27.29 |
| 8     | 16    | 17     | 2     | 3.92  | 27.29 |
| 9     | 18    | 19     | 2     | 4.35  | 25.29 |
| 10    | 20    | 21     | 2     | 4.77  | 25.29 |
| 11    | 22    | 23     | 2     | 5.19  | 25.29 |
| 12    | 24    | 25     | 2     | 5.59  | 25.29 |
| 13    | 26    | 27     | 2     | 5.99  | 25.29 |
| 14    | 28    | 29     | 2     | 6.37  | 25.29 |
| 15    | 30    | 31     | 2     | 6.74  | 25.29 |
| 16    | 32    | 33     | 2     | 7.10  | 25.29 |
| 17    | 34    | 35     | 2     | 7.45  | 25.29 |
| 18    | 36    | 37     | 2     | 7.80  | 25.29 |
| 19    | 38    | 40     | 3     | 8.20  | 27.05 |
| 20    | 41    | 43     | 3     | 8.68  | 27.05 |
| 21    | 44    | 46     | 3     | 9.13  | 27.05 |
| 22    | 47    | 49     | 3     | 9.55  | 27.05 |
| 23    | 50    | 52     | 3     | 9.96  | 27.05 |
| 24    | 53    | 55     | 3     | 10.35 | 27.05 |
| 25    | 56    | 58     | 3     | 10.71 | 27.05 |
| 26    | 59    | 61     | 3     | 11.06 | 27.05 |
| 27    | 62    | 65     | 4     | 11.45 | 28.30 |
| 28    | 66    | 69     | 4     | 11.86 | 28.30 |
| 29    | 70    | 73     | 4     | 12.25 | 28.30 |
| 30    | 74    | 77     | 4     | 12.62 | 28.30 |
| 31    | 78    | 81     | 4     | 12.96 | 28.30 |
| 32    | 82    | 86     | 5     | 13.32 | 29.27 |
| 33    | 87    | 91     | 5     | 13.70 | 29.27 |
| 34    | 92    | 96     | 5     | 14.05 | 29.27 |
| 35    | 97    | 102    | 6     | 14.41 | 30.06 |
| 36    | 103   | 108    | 6     | 14.77 | 30.06 |
| 37    | 109   | 115    | 7     | 15.13 | 30.73 |
| 38    | 116   | 122    | 7     | 15.49 | 30.73 |
| 39    | 123   | 130    | 8     | 15.85 | 31.31 |
| 40    | 131   | 138    | 8     | 16.20 | 31.31 |
| 41    | 139   | 147    | 9     | 16.55 | 31.82 |
| 42    | 148   | 157    | 10    | 16.91 | 32.28 |
| 43    | 158   | 167    | 10    | 17.25 | 32.28 |
| 44    | 168   | 178    | 11    | 17.59 | 32.69 |
| 45    | 179   | 190    | 12    | 17.93 | 33.07 |
| 46    | 191   | 203    | 13    | 18.28 | 33.42 |

|    |     |      |    |       |       |
|----|-----|------|----|-------|-------|
| 47 | 204 | 217  | 14 | 18.62 | 33.74 |
| 48 | 218 | 232  | 15 | 18.96 | 34.04 |
| 49 | 233 | 248  | 16 | 19.30 | 34.32 |
| 50 | 249 | 265  | 17 | 19.64 | 34.58 |
| 51 | 266 | 283  | 18 | 19.97 | 34.83 |
| 52 | 284 | 303  | 20 | 20.31 | 38.29 |
| 53 | 304 | 324  | 21 | 20.65 | 38.50 |
| 54 | 325 | 347  | 23 | 20.99 | 38.89 |
| 55 | 348 | 371  | 24 | 21.33 | 41.08 |
| 56 | 372 | 397  | 26 | 21.66 | 41.43 |
| 57 | 398 | 425  | 28 | 21.99 | 41.75 |
| 58 | 426 | 456  | 31 | 22.32 | 47.19 |
| 59 | 457 | 490  | 34 | 22.66 | 47.59 |
| 60 | 491 | 527  | 37 | 23.00 | 47.96 |
| 61 | 528 | 567  | 40 | 23.33 | 58.30 |
| 62 | 568 | 612  | 45 | 23.67 | 58.81 |
| 63 | 613 | 662  | 50 | 24.00 | 69.27 |
| 64 | 663 | 718  | 56 | 24.00 | 69.76 |
| 65 | 719 | 781  | 63 | 24.00 | 70.27 |
| 66 | 782 | 853  | 72 | 24.00 | 70.85 |
| 67 | 854 | 937  | 84 | 24.00 | 71.52 |
| 68 | 938 | 1023 | 86 | 24.00 | 70.20 |

Table 4.B.18 – Psychoacoustic parameters for 48 kHz short FFT

| Index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0.00  | 27.28 |
| 1     | 1     | 1      | 1     | 1.88  | 22.28 |
| 2     | 2     | 2      | 1     | 3.70  | 14.28 |
| 3     | 3     | 3      | 1     | 5.39  | 12.28 |
| 4     | 4     | 4      | 1     | 6.93  | 12.28 |
| 5     | 5     | 5      | 1     | 8.29  | 12.28 |
| 6     | 6     | 6      | 1     | 9.49  | 12.28 |
| 7     | 7     | 7      | 1     | 10.53 | 12.28 |
| 8     | 8     | 8      | 1     | 11.45 | 12.28 |
| 9     | 9     | 9      | 1     | 12.26 | 12.28 |
| 10    | 10    | 10     | 1     | 12.96 | 12.28 |
| 11    | 11    | 11     | 1     | 13.59 | 12.28 |
| 12    | 12    | 12     | 1     | 14.15 | 12.28 |
| 13    | 13    | 13     | 1     | 14.65 | 12.28 |
| 14    | 14    | 14     | 1     | 15.11 | 12.28 |
| 15    | 15    | 15     | 1     | 15.52 | 12.28 |
| 16    | 16    | 16     | 1     | 15.90 | 12.28 |
| 17    | 17    | 18     | 2     | 16.56 | 15.29 |
| 18    | 19    | 20     | 2     | 17.15 | 15.29 |
| 19    | 21    | 22     | 2     | 17.66 | 15.29 |
| 20    | 23    | 24     | 2     | 18.13 | 15.29 |
| 21    | 25    | 26     | 2     | 18.54 | 15.29 |
| 22    | 27    | 28     | 2     | 18.93 | 15.29 |
| 23    | 29    | 30     | 2     | 19.28 | 15.29 |
| 24    | 31    | 33     | 3     | 19.69 | 17.05 |
| 25    | 34    | 36     | 3     | 20.14 | 20.05 |
| 26    | 37    | 39     | 3     | 20.54 | 20.05 |
| 27    | 40    | 42     | 3     | 20.92 | 20.05 |
| 28    | 43    | 45     | 3     | 21.27 | 22.05 |
| 29    | 46    | 49     | 4     | 21.64 | 23.30 |

|    |     |     |    |       |       |
|----|-----|-----|----|-------|-------|
| 30 | 50  | 53  | 4  | 22.03 | 28.30 |
| 31 | 54  | 57  | 4  | 22.39 | 28.30 |
| 32 | 58  | 62  | 5  | 22.76 | 29.27 |
| 33 | 63  | 67  | 5  | 23.13 | 39.27 |
| 34 | 68  | 73  | 6  | 23.49 | 40.06 |
| 35 | 74  | 79  | 6  | 23.85 | 40.06 |
| 36 | 80  | 86  | 7  | 24.00 | 50.73 |
| 37 | 87  | 94  | 8  | 24.00 | 51.31 |
| 38 | 95  | 103 | 9  | 24.00 | 51.82 |
| 39 | 104 | 113 | 10 | 24.00 | 52.28 |
| 40 | 114 | 125 | 12 | 24.00 | 53.07 |
| 41 | 126 | 127 | 1  | 24.00 | 53.07 |

Table 4.B.19 – Psychoacoustic parameters for 64 kHz long FFT

| index | w low | w high | width | bval  | qstr  |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 1      | 2     | 0,32  | 40,29 |
| 1     | 2     | 3      | 2     | 0,95  | 40,29 |
| 2     | 4     | 5      | 2     | 1,57  | 35,29 |
| 3     | 6     | 7      | 2     | 2,19  | 32,29 |
| 4     | 8     | 9      | 2     | 2,80  | 32,29 |
| 5     | 10    | 11     | 2     | 3,40  | 27,29 |
| 6     | 12    | 13     | 2     | 3,99  | 27,29 |
| 7     | 14    | 15     | 2     | 4,56  | 25,29 |
| 8     | 16    | 17     | 2     | 5,12  | 25,29 |
| 9     | 18    | 19     | 2     | 5,66  | 25,29 |
| 10    | 20    | 21     | 2     | 6,18  | 25,29 |
| 11    | 22    | 23     | 2     | 6,68  | 25,29 |
| 12    | 24    | 25     | 2     | 7,16  | 25,29 |
| 13    | 26    | 27     | 2     | 7,63  | 25,29 |
| 14    | 28    | 29     | 2     | 8,07  | 25,29 |
| 15    | 30    | 31     | 2     | 8,50  | 25,29 |
| 16    | 32    | 33     | 2     | 8,90  | 25,29 |
| 17    | 34    | 35     | 2     | 9,29  | 25,29 |
| 18    | 36    | 37     | 2     | 9,67  | 25,29 |
| 19    | 38    | 39     | 2     | 10,03 | 25,29 |
| 20    | 40    | 41     | 2     | 10,37 | 25,29 |
| 21    | 42    | 44     | 3     | 10,77 | 27,05 |
| 22    | 45    | 47     | 3     | 11,23 | 27,05 |
| 23    | 48    | 50     | 3     | 11,66 | 27,05 |
| 24    | 51    | 53     | 3     | 12,06 | 27,05 |
| 25    | 54    | 56     | 3     | 12,44 | 27,05 |
| 26    | 57    | 59     | 3     | 12,79 | 27,05 |
| 27    | 60    | 63     | 4     | 13,18 | 28,30 |
| 28    | 64    | 67     | 4     | 13,59 | 28,30 |
| 29    | 68    | 71     | 4     | 13,97 | 28,30 |
| 30    | 72    | 75     | 4     | 14,32 | 28,30 |
| 31    | 76    | 80     | 5     | 14,69 | 29,27 |
| 32    | 81    | 85     | 5     | 15,07 | 29,27 |
| 33    | 86    | 90     | 5     | 15,42 | 29,27 |
| 34    | 91    | 96     | 6     | 15,77 | 30,06 |
| 35    | 97    | 102    | 6     | 16,13 | 30,06 |
| 36    | 103   | 109    | 7     | 16,49 | 30,73 |
| 37    | 110   | 116    | 7     | 16,85 | 30,73 |
| 38    | 117   | 124    | 8     | 17,20 | 31,31 |
| 39    | 125   | 132    | 8     | 17,54 | 31,31 |

|    |     |      |     |       |       |
|----|-----|------|-----|-------|-------|
| 40 | 133 | 141  | 9   | 17,88 | 31,82 |
| 41 | 142 | 151  | 10  | 18,23 | 32,28 |
| 42 | 152 | 161  | 10  | 18,58 | 32,28 |
| 43 | 162 | 172  | 11  | 18,91 | 32,69 |
| 44 | 173 | 184  | 12  | 19,25 | 33,07 |
| 45 | 185 | 197  | 13  | 19,60 | 33,42 |
| 46 | 198 | 211  | 14  | 19,94 | 33,74 |
| 47 | 212 | 226  | 15  | 20,29 | 37,04 |
| 48 | 227 | 242  | 16  | 20,63 | 37,32 |
| 49 | 243 | 259  | 17  | 20,97 | 37,58 |
| 50 | 260 | 277  | 18  | 21,31 | 39,83 |
| 51 | 278 | 297  | 20  | 21,64 | 40,29 |
| 52 | 298 | 318  | 21  | 21,98 | 40,50 |
| 53 | 319 | 341  | 23  | 22,31 | 45,89 |
| 54 | 342 | 366  | 25  | 22,65 | 46,26 |
| 55 | 367 | 394  | 28  | 22,98 | 46,75 |
| 56 | 395 | 424  | 30  | 23,32 | 57,05 |
| 57 | 425 | 458  | 34  | 23,66 | 57,59 |
| 58 | 459 | 495  | 37  | 23,99 | 57,96 |
| 59 | 496 | 537  | 42  | 24,00 | 68,51 |
| 60 | 538 | 584  | 47  | 24,00 | 69,00 |
| 61 | 585 | 638  | 54  | 24,00 | 69,60 |
| 62 | 639 | 701  | 63  | 24,00 | 70,27 |
| 63 | 702 | 774  | 73  | 24,00 | 70,91 |
| 64 | 775 | 861  | 87  | 24,00 | 71,67 |
| 65 | 862 | 966  | 105 | 24,00 | 72,49 |
| 66 | 967 | 1023 | 57  | 24,00 | 69,83 |

Table 4.B.20 – Psychoacoustic parameters for 64 kHz short FFT

| index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0,00  | 27,28 |
| 1     | 1     | 1      | 1     | 2,50  | 19,28 |
| 2     | 2     | 2      | 1     | 4,85  | 12,28 |
| 3     | 3     | 3      | 1     | 6,93  | 12,28 |
| 4     | 4     | 4      | 1     | 8,70  | 12,28 |
| 5     | 5     | 5      | 1     | 10,20 | 12,28 |
| 6     | 6     | 6      | 1     | 11,45 | 12,28 |
| 7     | 7     | 7      | 1     | 12,50 | 12,28 |
| 8     | 8     | 8      | 1     | 13,39 | 12,28 |
| 9     | 9     | 9      | 1     | 14,15 | 12,28 |
| 10    | 10    | 10     | 1     | 14,81 | 12,28 |
| 11    | 11    | 11     | 1     | 15,39 | 12,28 |
| 12    | 12    | 12     | 1     | 15,90 | 12,28 |
| 13    | 13    | 13     | 1     | 16,36 | 12,28 |
| 14    | 14    | 14     | 1     | 16,78 | 12,28 |
| 15    | 15    | 15     | 1     | 17,16 | 12,28 |
| 16    | 16    | 17     | 2     | 17,82 | 15,29 |
| 17    | 18    | 19     | 2     | 18,40 | 15,29 |
| 18    | 20    | 21     | 2     | 18,92 | 15,29 |
| 19    | 22    | 23     | 2     | 19,39 | 15,29 |
| 20    | 24    | 25     | 2     | 19,82 | 15,29 |
| 21    | 26    | 27     | 2     | 20,21 | 18,29 |
| 22    | 28    | 29     | 2     | 20,57 | 18,29 |
| 23    | 30    | 32     | 3     | 20,98 | 20,05 |
| 24    | 33    | 35     | 3     | 21,43 | 22,05 |



|    |     |     |    |       |       |
|----|-----|-----|----|-------|-------|
| 25 | 36  | 38  | 3  | 21,84 | 22,05 |
| 26 | 39  | 41  | 3  | 22,22 | 27,05 |
| 27 | 42  | 45  | 4  | 22,61 | 28,30 |
| 28 | 46  | 49  | 4  | 23,02 | 38,30 |
| 29 | 50  | 53  | 4  | 23,39 | 38,30 |
| 30 | 54  | 58  | 5  | 23,75 | 39,27 |
| 31 | 59  | 63  | 5  | 24,00 | 49,27 |
| 32 | 64  | 69  | 6  | 24,00 | 50,06 |
| 33 | 70  | 76  | 7  | 24,00 | 50,73 |
| 34 | 77  | 84  | 8  | 24,00 | 51,31 |
| 35 | 85  | 93  | 9  | 24,00 | 51,82 |
| 36 | 94  | 104 | 11 | 24,00 | 52,69 |
| 37 | 105 | 117 | 13 | 24,00 | 53,42 |
| 38 | 118 | 127 | 10 | 24,00 | 52,28 |

Table 4.B.21 – Psychoacoustic parameters for 88.2 kHz long FFT

| index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0,00  | 37,28 |
| 1     | 1     | 1      | 1     | 0,44  | 37,28 |
| 2     | 2     | 2      | 1     | 0,87  | 37,28 |
| 3     | 3     | 3      | 1     | 1,30  | 32,28 |
| 4     | 4     | 4      | 1     | 1,73  | 32,28 |
| 5     | 5     | 5      | 1     | 2,16  | 29,28 |
| 6     | 6     | 6      | 1     | 2,58  | 29,28 |
| 7     | 7     | 7      | 1     | 3,00  | 24,28 |
| 8     | 8     | 8      | 1     | 3,41  | 24,28 |
| 9     | 9     | 9      | 1     | 3,82  | 24,28 |
| 10    | 10    | 10     | 1     | 4,22  | 22,28 |
| 11    | 11    | 11     | 1     | 4,61  | 22,28 |
| 12    | 12    | 12     | 1     | 4,99  | 22,28 |
| 13    | 13    | 13     | 1     | 5,37  | 22,28 |
| 14    | 14    | 14     | 1     | 5,74  | 22,28 |
| 15    | 15    | 15     | 1     | 6,10  | 22,28 |
| 16    | 16    | 16     | 1     | 6,45  | 22,28 |
| 17    | 17    | 17     | 1     | 6,79  | 22,28 |
| 18    | 18    | 19     | 2     | 7,44  | 25,29 |
| 19    | 20    | 21     | 2     | 8,05  | 25,29 |
| 20    | 22    | 23     | 2     | 8,64  | 25,29 |
| 21    | 24    | 25     | 2     | 9,19  | 25,29 |
| 22    | 26    | 27     | 2     | 9,70  | 25,29 |
| 23    | 28    | 29     | 2     | 10,19 | 25,29 |
| 24    | 30    | 31     | 2     | 10,65 | 25,29 |
| 25    | 32    | 33     | 2     | 11,08 | 25,29 |
| 26    | 34    | 35     | 2     | 11,48 | 25,29 |
| 27    | 36    | 37     | 2     | 11,86 | 25,29 |
| 28    | 38    | 39     | 2     | 12,22 | 25,29 |
| 29    | 40    | 42     | 3     | 12,64 | 27,05 |
| 30    | 43    | 45     | 3     | 13,10 | 27,05 |
| 31    | 46    | 48     | 3     | 13,53 | 27,05 |
| 32    | 49    | 51     | 3     | 13,93 | 27,05 |
| 33    | 52    | 54     | 3     | 14,30 | 27,05 |
| 34    | 55    | 58     | 4     | 14,69 | 28,30 |
| 35    | 59    | 62     | 4     | 15,11 | 28,30 |
| 36    | 63    | 66     | 4     | 15,49 | 28,30 |
| 37    | 67    | 70     | 4     | 15,84 | 28,30 |

|    |     |      |     |       |       |
|----|-----|------|-----|-------|-------|
| 38 | 71  | 75   | 5   | 16,21 | 29,27 |
| 39 | 76  | 80   | 5   | 16,58 | 29,27 |
| 40 | 81  | 85   | 5   | 16,92 | 29,27 |
| 41 | 86  | 91   | 6   | 17,27 | 30,06 |
| 42 | 92  | 97   | 6   | 17,62 | 30,06 |
| 43 | 98  | 104  | 7   | 17,97 | 30,73 |
| 44 | 105 | 111  | 7   | 18,32 | 30,73 |
| 45 | 112 | 119  | 8   | 18,67 | 31,31 |
| 46 | 120 | 127  | 8   | 19,02 | 31,31 |
| 47 | 128 | 136  | 9   | 19,35 | 31,82 |
| 48 | 137 | 146  | 10  | 19,71 | 32,28 |
| 49 | 147 | 156  | 10  | 20,05 | 35,28 |
| 50 | 157 | 167  | 11  | 20,39 | 35,69 |
| 51 | 168 | 179  | 12  | 20,73 | 36,07 |
| 52 | 180 | 192  | 13  | 21,08 | 38,42 |
| 53 | 193 | 206  | 14  | 21,43 | 38,74 |
| 54 | 207 | 221  | 15  | 21,77 | 39,04 |
| 55 | 222 | 237  | 16  | 22,11 | 44,32 |
| 56 | 238 | 255  | 18  | 22,45 | 44,83 |
| 57 | 256 | 274  | 19  | 22,80 | 45,06 |
| 58 | 275 | 295  | 21  | 23,13 | 55,50 |
| 59 | 296 | 318  | 23  | 23,47 | 55,89 |
| 60 | 319 | 344  | 26  | 23,81 | 56,43 |
| 61 | 345 | 373  | 29  | 24,00 | 66,90 |
| 62 | 374 | 405  | 32  | 24,00 | 67,33 |
| 63 | 406 | 442  | 37  | 24,00 | 67,96 |
| 64 | 443 | 484  | 42  | 24,00 | 68,51 |
| 65 | 485 | 533  | 49  | 24,00 | 69,18 |
| 66 | 534 | 591  | 58  | 24,00 | 69,91 |
| 67 | 592 | 660  | 69  | 24,00 | 70,66 |
| 68 | 661 | 745  | 85  | 24,00 | 71,57 |
| 69 | 746 | 851  | 106 | 24,00 | 72,53 |
| 70 | 852 | 988  | 137 | 24,00 | 73,64 |
| 71 | 989 | 1023 | 35  | 24,00 | 67,72 |

Table 4.B.22 – Psychoacoustic parameters for 88.2 kHz short FFT

| index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0,00  | 27,28 |
| 1     | 1     | 1      | 1     | 3,41  | 14,28 |
| 2     | 2     | 2      | 1     | 6,45  | 12,28 |
| 3     | 3     | 3      | 1     | 8,92  | 12,28 |
| 4     | 4     | 4      | 1     | 10,87 | 12,28 |
| 5     | 5     | 5      | 1     | 12,39 | 12,28 |
| 6     | 6     | 6      | 1     | 13,61 | 12,28 |
| 7     | 7     | 7      | 1     | 14,59 | 12,28 |
| 8     | 8     | 8      | 1     | 15,40 | 12,28 |
| 9     | 9     | 9      | 1     | 16,09 | 12,28 |
| 10    | 10    | 10     | 1     | 16,69 | 12,28 |
| 11    | 11    | 11     | 1     | 17,21 | 12,28 |
| 12    | 12    | 12     | 1     | 17,68 | 12,28 |
| 13    | 13    | 13     | 1     | 18,11 | 12,28 |
| 14    | 14    | 14     | 1     | 18,49 | 12,28 |
| 15    | 15    | 15     | 1     | 18,85 | 12,28 |
| 16    | 16    | 17     | 2     | 19,48 | 15,29 |
| 17    | 18    | 19     | 2     | 20,05 | 18,29 |

|    |     |     |    |       |       |
|----|-----|-----|----|-------|-------|
| 18 | 20  | 21  | 2  | 20,55 | 18,29 |
| 19 | 22  | 23  | 2  | 21,01 | 20,29 |
| 20 | 24  | 25  | 2  | 21,43 | 20,29 |
| 21 | 26  | 27  | 2  | 21,81 | 20,29 |
| 22 | 28  | 29  | 2  | 22,15 | 25,29 |
| 23 | 30  | 32  | 3  | 22,55 | 27,05 |
| 24 | 33  | 35  | 3  | 22,98 | 27,05 |
| 25 | 36  | 38  | 3  | 23,36 | 37,05 |
| 26 | 39  | 42  | 4  | 23,75 | 38,30 |
| 27 | 43  | 46  | 4  | 24,00 | 48,30 |
| 28 | 47  | 51  | 5  | 24,00 | 49,27 |
| 29 | 52  | 56  | 5  | 24,00 | 49,27 |
| 30 | 57  | 62  | 6  | 24,00 | 50,06 |
| 31 | 63  | 69  | 7  | 24,00 | 50,73 |
| 32 | 70  | 77  | 8  | 24,00 | 51,31 |
| 33 | 78  | 87  | 10 | 24,00 | 52,28 |
| 34 | 88  | 99  | 12 | 24,00 | 53,07 |
| 35 | 100 | 115 | 16 | 24,00 | 54,32 |
| 36 | 116 | 127 | 12 | 24,00 | 53,07 |

Table 4.B.23 – Psychoacoustic parameters for 96 kHz long FFT

| index | w low | w high | width | bval  | qstr  |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0,00  | 37,28 |
| 1     | 1     | 1      | 1     | 0,47  | 37,28 |
| 2     | 2     | 2      | 1     | 0,95  | 37,28 |
| 3     | 3     | 3      | 1     | 1,42  | 32,28 |
| 4     | 4     | 4      | 1     | 1,88  | 32,28 |
| 5     | 5     | 5      | 1     | 2,35  | 29,28 |
| 6     | 6     | 6      | 1     | 2,81  | 29,28 |
| 7     | 7     | 7      | 1     | 3,26  | 24,28 |
| 8     | 8     | 8      | 1     | 3,70  | 24,28 |
| 9     | 9     | 9      | 1     | 4,14  | 22,28 |
| 10    | 10    | 10     | 1     | 4,57  | 22,28 |
| 11    | 11    | 11     | 1     | 4,98  | 22,28 |
| 12    | 12    | 12     | 1     | 5,39  | 22,28 |
| 13    | 13    | 13     | 1     | 5,79  | 22,28 |
| 14    | 14    | 14     | 1     | 6,18  | 22,28 |
| 15    | 15    | 15     | 1     | 6,56  | 22,28 |
| 16    | 16    | 16     | 1     | 6,93  | 22,28 |
| 17    | 17    | 17     | 1     | 7,28  | 22,28 |
| 18    | 18    | 18     | 1     | 7,63  | 22,28 |
| 19    | 19    | 20     | 2     | 8,28  | 25,29 |
| 20    | 21    | 22     | 2     | 8,90  | 25,29 |
| 21    | 23    | 24     | 2     | 9,48  | 25,29 |
| 22    | 25    | 26     | 2     | 10,02 | 25,29 |
| 23    | 27    | 28     | 2     | 10,53 | 25,29 |
| 24    | 29    | 30     | 2     | 11,00 | 25,29 |
| 25    | 31    | 32     | 2     | 11,45 | 25,29 |
| 26    | 33    | 34     | 2     | 11,86 | 25,29 |
| 27    | 35    | 36     | 2     | 12,25 | 25,29 |
| 28    | 37    | 38     | 2     | 12,62 | 25,29 |
| 29    | 39    | 40     | 2     | 12,96 | 25,29 |
| 30    | 41    | 43     | 3     | 13,36 | 27,05 |
| 31    | 44    | 46     | 3     | 13,80 | 27,05 |
| 32    | 47    | 49     | 3     | 14,21 | 27,05 |

|    |     |      |     |       |       |
|----|-----|------|-----|-------|-------|
| 33 | 50  | 52   | 3   | 14,59 | 27,05 |
| 34 | 53  | 55   | 3   | 14,94 | 27,05 |
| 35 | 56  | 59   | 4   | 15,32 | 28,30 |
| 36 | 60  | 63   | 4   | 15,71 | 28,30 |
| 37 | 64  | 67   | 4   | 16,08 | 28,30 |
| 38 | 68  | 72   | 5   | 16,45 | 29,27 |
| 39 | 73  | 77   | 5   | 16,83 | 29,27 |
| 40 | 78  | 82   | 5   | 17,19 | 29,27 |
| 41 | 83  | 88   | 6   | 17,54 | 30,06 |
| 42 | 89  | 94   | 6   | 17,90 | 30,06 |
| 43 | 95  | 101  | 7   | 18,26 | 30,73 |
| 44 | 102 | 108  | 7   | 18,62 | 30,73 |
| 45 | 109 | 116  | 8   | 18,97 | 31,31 |
| 46 | 117 | 124  | 8   | 19,32 | 31,31 |
| 47 | 125 | 133  | 9   | 19,67 | 31,82 |
| 48 | 134 | 143  | 10  | 20,03 | 35,28 |
| 49 | 144 | 153  | 10  | 20,38 | 35,28 |
| 50 | 154 | 164  | 11  | 20,72 | 35,69 |
| 51 | 165 | 176  | 12  | 21,07 | 38,07 |
| 52 | 177 | 189  | 13  | 21,42 | 38,42 |
| 53 | 190 | 203  | 14  | 21,77 | 38,74 |
| 54 | 204 | 218  | 15  | 22,12 | 44,04 |
| 55 | 219 | 234  | 16  | 22,46 | 44,32 |
| 56 | 235 | 252  | 18  | 22,80 | 44,83 |
| 57 | 253 | 271  | 19  | 23,14 | 55,06 |
| 58 | 272 | 292  | 21  | 23,47 | 55,50 |
| 59 | 293 | 316  | 24  | 23,81 | 56,08 |
| 60 | 317 | 342  | 26  | 24,00 | 66,43 |
| 61 | 343 | 372  | 30  | 24,00 | 67,05 |
| 62 | 373 | 406  | 34  | 24,00 | 67,59 |
| 63 | 407 | 445  | 39  | 24,00 | 68,19 |
| 64 | 446 | 490  | 45  | 24,00 | 68,81 |
| 65 | 491 | 543  | 53  | 24,00 | 69,52 |
| 66 | 544 | 607  | 64  | 24,00 | 70,34 |
| 67 | 608 | 685  | 78  | 24,00 | 71,20 |
| 68 | 686 | 783  | 98  | 24,00 | 72,19 |
| 69 | 784 | 910  | 127 | 24,00 | 73,31 |
| 70 | 911 | 1023 | 113 | 24,00 | 72,81 |

Table 4.B.24 – Psychoacoustic parameters for 96 kHz short FFT

| index | w low | w high | width | bval  | qsthr |
|-------|-------|--------|-------|-------|-------|
| 0     | 0     | 0      | 1     | 0,00  | 27,28 |
| 1     | 1     | 1      | 1     | 3,70  | 14,28 |
| 2     | 2     | 2      | 1     | 6,93  | 12,28 |
| 3     | 3     | 3      | 1     | 9,49  | 12,28 |
| 4     | 4     | 4      | 1     | 11,45 | 12,28 |
| 5     | 5     | 5      | 1     | 12,96 | 12,28 |
| 6     | 6     | 6      | 1     | 14,15 | 12,28 |
| 7     | 7     | 7      | 1     | 15,11 | 12,28 |
| 8     | 8     | 8      | 1     | 15,90 | 12,28 |
| 9     | 9     | 9      | 1     | 16,57 | 12,28 |
| 10    | 10    | 10     | 1     | 17,16 | 12,28 |
| 11    | 11    | 11     | 1     | 17,67 | 12,28 |
| 12    | 12    | 12     | 1     | 18,13 | 12,28 |
| 13    | 13    | 13     | 1     | 18,55 | 12,28 |

|    |     |     |    |       |       |
|----|-----|-----|----|-------|-------|
| 14 | 14  | 14  | 1  | 18,93 | 12,28 |
| 15 | 15  | 16  | 2  | 19,60 | 15,29 |
| 16 | 17  | 18  | 2  | 20,20 | 18,29 |
| 17 | 19  | 20  | 2  | 20,73 | 18,29 |
| 18 | 21  | 22  | 2  | 21,21 | 20,29 |
| 19 | 23  | 24  | 2  | 21,64 | 20,29 |
| 20 | 25  | 26  | 2  | 22,03 | 25,29 |
| 21 | 27  | 28  | 2  | 22,39 | 25,29 |
| 22 | 29  | 31  | 3  | 22,79 | 27,05 |
| 23 | 32  | 34  | 3  | 23,23 | 37,05 |
| 24 | 35  | 37  | 3  | 23,62 | 37,05 |
| 25 | 38  | 41  | 4  | 24,00 | 48,30 |
| 26 | 42  | 45  | 4  | 24,00 | 48,30 |
| 27 | 46  | 50  | 5  | 24,00 | 49,27 |
| 28 | 51  | 55  | 5  | 24,00 | 49,27 |
| 29 | 56  | 61  | 6  | 24,00 | 50,06 |
| 30 | 62  | 68  | 7  | 24,00 | 50,73 |
| 31 | 69  | 77  | 9  | 24,00 | 51,82 |
| 32 | 78  | 88  | 11 | 24,00 | 52,69 |
| 33 | 89  | 102 | 14 | 24,00 | 53,74 |
| 34 | 103 | 120 | 18 | 24,00 | 54,83 |
| 35 | 121 | 127 | 7  | 24,00 | 50,73 |

**4.B.4 Gain control**

**4.B.4.1 Encoding process**

The gain control tool consists of a PQF (Polyphase Quadrature Filter), gain detectors and gain modifiers. This tool receives the input time-domain signals and **window\_sequence**, and then outputs **gain\_control\_data** and a gain controlled signal whose length is equal to the length of the MDCT window. The block diagram for the gain control tool is shown in Figure 4.B.3.

Due to the characteristics of the PQF filterbank, the order of the MDCT coefficients in each even PQF band needs to be reversed. This is done by reversing the spectral order of the MDCT coefficients, i.e. exchanging the higher frequency MDCT coefficients with the lower frequency MDCT coefficients.

If the gain control tool is used, the configuration of the filterbank tool is changed as follows. In the case of an **EIGHT\_SHORT\_SEQUENCE** **window\_sequence**, the number of coefficients for the MDCT is 32 instead of 128 and eight MDCTs are carried out. In the case of other **window\_sequence** values, the number of coefficients for the MDCT is 256 instead of 1024 and one MDCT is carried out. In all cases, the filter bank tool receives a total of 2048 gain controlled signal values per frame, because the input samples have been overlapped.

**4.B.4.1.1 PQF**

The input signal is divided by a PQF into four equal width frequency bands. The coefficients of each band PQF are given as follows.

$$h_i(n) = \frac{1}{4} \cos\left(\frac{(2i+1)(2n+5)\pi}{16}\right) Q(n), \quad 0 \leq n \leq 95, 0 \leq i \leq 3$$

where

$$Q(n) = Q(95-n), \quad 48 \leq n \leq 95$$

and the values of  $Q(n)$  are the same values as those of the decoder.

#### 4.B.4.1.2 Gain detector

The gain detectors produce gain control data which satisfies the bitstream syntax. This information consists of the number of gain changes, the index of gain change positions and the index of gain change level. Note that the output gain control data applies to the previous input time signal. This means that the gain detector has a one frame delay.

The detection of the gain change point is done in the second half of the MDCT window region and in the non-overlapped region (of LONG\_START\_SEQUENCE and LONG\_STOP\_SEQUENCE). Thus the number of regions are one for ONLY\_LONG\_SEQUENCE, two for LONG\_START\_SEQUENCE and LONG\_STOP\_SEQUENCE, and eight for EIGHT\_SHORT\_SEQUENCE.

The samples in each region are divided into subregions, each having eight-tuple samples. Then one value (e.g. peak value of samples) is selected in these subregions. The ratios between the values of subregions and the value of the last subregion are calculated. If the ratio is greater or less than the value of  $2^n$  where  $n$  is an integer between -4 to 11, those subregions can be detected as the gain change points of the signals. The subregion number which is detected as the gain change point is set to be the position data. The exponent of the ratio is set to be the gain data. The time resolution of the gain control is approximately 0.7 ms at 48 kHz sampling rate.

#### 4.B.4.1.3 Gain modifier

The gain modifier for each PQF band controls the gain of each signal band. The complementary gain control process in the decoder decreases the pre-echo and reconstructs the original signal. A window function for gain control, the Gain Modification Function (GMF), which is defined in the decoding process, is derived from the gains and the gain-changed positions. The gain controlled signals are derived by applying the GMF to the corresponding band signals.

#### 4.B.4.2 Diagrams

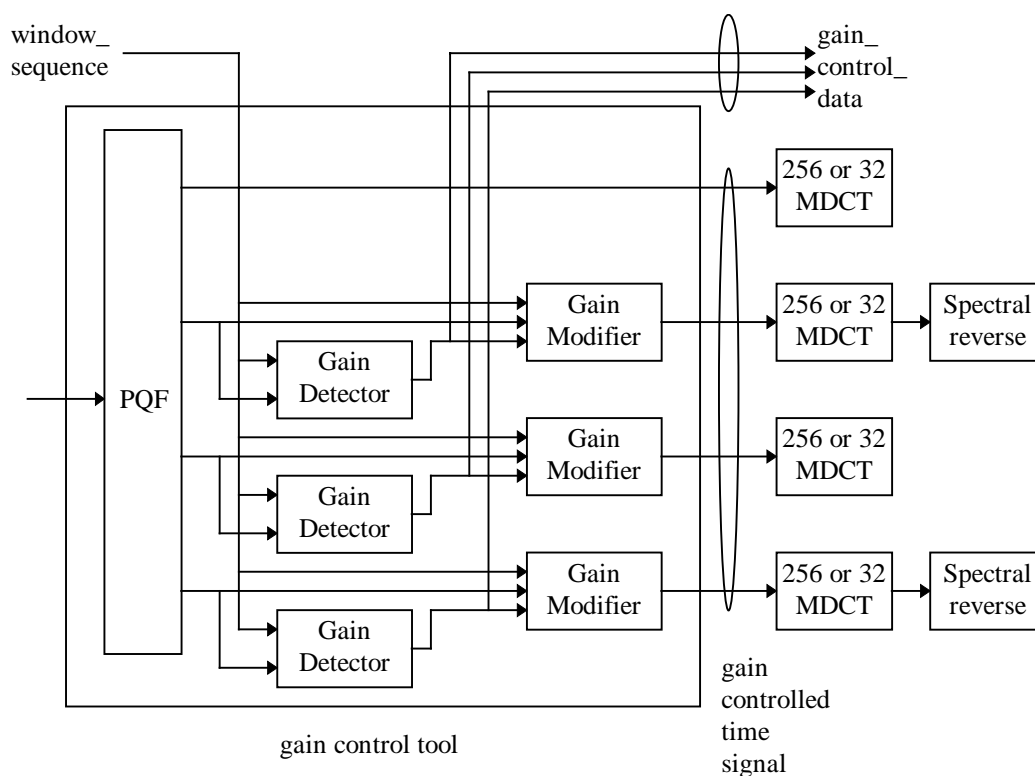


Figure 4.B.3 – Block diagram of gain control tool for encoder

### 4.B.5 Filterbank and block switching

A fundamental component in the audio coding process is the conversion of the time domain signals into a time-frequency representation. This conversion is done by a forward modified discrete cosine transform (MDCT).

#### 4.B.5.1 Encoding process

In the encoder the filterbank takes the appropriate block of time samples, modulates them by an appropriate window function, and performs the MDCT. Each block of input samples is overlapped by 50% with the immediately preceding block and the following block. The transform input block length N can be set to either 2048 or 256 samples. Since the window function has a significant effect on the filterbank frequency response, the filterbank has been designed to allow a change in window shape to best adapt to input signal conditions. The shape of the window is varied simultaneously in the encoder and decoder to allow the filterbank to efficiently separate spectral components of the input for a wider variety of input signals.

##### 4.B.5.1.1 Windowing and block switching

The adaptation of the time-frequency resolution of the filterbank to the characteristics of the input signal is done by shifting between transforms whose input lengths are either 2048 or 256 samples. By enabling the block switching tool, the following transitions are meaningful:

|                           |    |                                              |
|---------------------------|----|----------------------------------------------|
| from ONLY_LONG_SEQUENCE   | to | { ONLY_LONG_SEQUENCE<br>LONG_START_SEQUENCE  |
| from LONG_START_SEQUENCE  | to | { EIGHT_SHORT_SEQUENCE<br>LONG_STOP_SEQUENCE |
| from LONG_STOP_SEQUENCE   | to | { ONLY_LONG_SEQUENCE<br>LONG_START_SEQUENCE  |
| from EIGHT_SHORT_SEQUENCE | to | { EIGHT_SHORT_SEQUENCE<br>LONG_STOP_SEQUENCE |

Window shape decisions are made by the encoder on a frame-by-frame-basis. The window selected is applicable to the second half of the window function only, since the first half is constrained to use the appropriate window shape from the preceding frame. Figure 4.B.4 shows the sequence of blocks for the transition (D-E-F) to and from a frame employing the sine function window. The window shape selector generally produces window shape run-lengths greater than that shown in the figure.

The 2048 time-domain values  $x'_{i,n}$  to be windowed are the last 1024 values of the previous window\_sequence concatenated with 1024 values of the current block. The formula below shows this fact:

$$x'_{i,n} = \begin{cases} x_{(i-1),(n+1024)}, & \text{for } 0 \leq n < 1024 \\ x_{i,n}, & \text{for } 1024 \leq n < 2048 \end{cases}$$

Where i is the block index and n is the sample index within a block. Once the window shape is selected, the **window\_shape** syntax element is initialized. Together with the chosen **window\_sequence** all information needed for windowing exist.

With the window halves described below all **window\_sequences** can be assembled.

For **window\_shape** == 1, the window coefficients are given by the Kaiser - Bessel derived (KBD) window as follows:

$$W_{KBD\_LEFT,N}(n) = \sqrt{\frac{\sum_{p=0}^n [W'(n,\alpha)]}{\sum_{p=0}^{N/2} [W'(p,\alpha)]}} \quad \text{for } 0 \leq n < \frac{N}{2}$$

$$W_{KBD\_RIGHT,N}(n) = \sqrt{\frac{\sum_{p=0}^{N-n} [W'(n,\alpha)]}{\sum_{p=0}^{N/2} [W'(p,\alpha)]}} \quad \text{for } \frac{N}{2} \leq n < N$$

where:

$W'$ , Kaiser - Bessel kernel window function, see also [5], is defined as follows:

$$W'(n) = \frac{I_0 \left[ \pi \alpha \sqrt{1.0 - \left( \frac{n - N/4}{N/4} \right)^2} \right]}{I_0[\pi \alpha]} \quad \text{for } 0 \leq n \leq \frac{N}{2}$$

$$I_0[x] = \sum_{k=0}^{\infty} \left[ \frac{\left( \frac{x}{2} \right)^k}{k!} \right]^2$$

$$\alpha = \text{kernel window alpha factor, } \alpha = \begin{cases} 4 & \text{for } N = 2048 \\ 6 & \text{for } N = 256 \end{cases}$$

Otherwise, for **window\_shape** == 0, a sine window is employed as follows:

$$W_{SIN\_LEFT,N}(n) = \sin\left(\frac{\pi}{N}\left(n + \frac{1}{2}\right)\right) \quad \text{for } 0 \leq n < \frac{N}{2}$$

$$W_{SIN\_RIGHT,N}(n) = \sin\left(\frac{\pi}{N}\left(n + \frac{1}{2}\right)\right) \quad \text{for } \frac{N}{2} \leq n < N$$

The window length  $N$  can be 2048 or 256 for the KBD and the sine window. How to obtain the possible window sequences is explained in the parts a)-d) of this subclause. All four window\_sequences explained below have a total length of 2048 samples.

For all kinds of window\_sequences the window\_shape of the left half of the first transform window is determined by the window shape of the previous block. The following formula expresses this fact:

$$W_{LEFT,N}(n) = \begin{cases} W_{KBD\_LEFT,N}(n), & \text{if } \text{window\_shape\_previous\_block} == 1 \\ W_{SIN\_LEFT,N}(n), & \text{if } \text{window\_shape\_previous\_block} == 0 \end{cases}$$



where:

*window\_shape\_previous\_block*: **window\_shape** of the block (i-1).

For the first block to be encoded the **window\_shape** of the left and right half of the window are identical.

**a) ONLY\_LONG\_SEQUENCE:**

The **window\_sequence** == ONLY\_LONG\_SEQUENCE is equal to one LONG\_WINDOW with a total window length of 2048.

For **window\_shape**==1 the ONLY\_LONG\_SEQUENCE is given as follows:

$$W(n) = \begin{cases} W_{LEFT,2048}(n), & \text{for } 0 \leq n < 1024 \\ W_{KBD\_RIGHT,2048}(n), & \text{for } 1024 \leq n < 2048 \end{cases}$$

If **window\_shape**==0 the ONLY\_LONG\_SEQUENCE can be described as follows:

$$W(n) = \begin{cases} W_{LEFT,2048}(n), & \text{for } 0 \leq n < 1024 \\ W_{SIN\_RIGHT,2048}(n), & \text{for } 1024 \leq n < 2048 \end{cases}$$

The time domain values after windowing ( $z_{i,n}$ ) can be expressed as:

$$z_{i,n} = w(n) \cdot x'_{i,n};$$

**b) LONG\_START\_SEQUENCE:**

The LONG\_START\_SEQUENCE is needed to obtain a correct overlap and add for a block transition from a ONLY\_LONG\_SEQUENCE to a EIGHT\_SHORT\_SEQUENCE.

If **window\_shape**==1 the LONG\_START\_SEQUENCE is given as follows:

$$W(n) = \begin{cases} W_{LEFT,2048}(n), & \text{for } 0 \leq n < 1024 \\ 1.0, & \text{for } 1024 \leq n < 1472 \\ W_{KBD\_RIGHT,256}(n + 128 - 1472), & \text{for } 1472 \leq n < 1600 \\ 0.0, & \text{for } 1600 \leq n < 2048 \end{cases}$$

If **window\_shape**==0 the LONG\_START\_SEQUENCE looks like:

$$W(n) = \begin{cases} W_{LEFT,2048}(n), & \text{for } 0 \leq n < 1024 \\ 1.0, & \text{for } 1024 \leq n < 1472 \\ W_{SIN\_RIGHT,256}(n + 128 - 1472), & \text{for } 1472 \leq n < 1600 \\ 0.0, & \text{for } 1600 \leq n < 2048 \end{cases}$$

The windowed domain values can be calculated with the formula explained in a).

**c) EIGHT\_SHORT**

The **window\_sequence** == EIGHT\_SHORT comprises eight overlapped SHORT\_WINDOWS with a window length of 256 samples each (2048 in total). Each of the eight short windows are windowed separately first. The short window number is indexed with the variable  $j = 0, \dots, 7$ .

The **window\_shape** of the previous block influences the first of the eight short windows ( $W_0(n)$ ) only.

If **window\_shape**==1 the window functions can be given as follows:

$$W_0(n) = \begin{cases} W_{LEFT,256}(n), & \text{for } 0 \leq n < 128 \\ W_{KBD\_RIGHT,256}(n), & \text{for } 128 \leq n < 256 \end{cases}$$

$$W_{1-7}(n) = \begin{cases} W_{KBD\_LEFT,256}(n), & \text{for } 0 \leq n < 128 \\ W_{KBD\_RIGHT,256}(n), & \text{for } 128 \leq n < 256 \end{cases}$$

Otherwise, if **window\_shape**==0, the window functions can be described as:

$$W_0(n) = \begin{cases} W_{LEFT,256}(n), & \text{for } 0 \leq n < 128 \\ W_{SIN\_RIGHT,256}(n), & \text{for } 128 \leq n < 256 \end{cases}$$

$$W_{1-7}(n) = \begin{cases} W_{SIN\_LEFT,256}(n), & \text{for } 0 \leq n < 128 \\ W_{SIN\_RIGHT,256}(n), & \text{for } 128 \leq n < 256 \end{cases}$$

An EIGHT\_SHORT sequence  $x'_{i,n}$  is subdivided into eight overlapping segments first and then multiplied with the appropriate window function:

$$z_{i,n} = \begin{cases} x'_{i,n+448} \cdot W_0(n), & \text{for } 0 \leq n < 256 \\ x'_{i,n+576} \cdot W_1(n-256), & \text{for } 256 \leq n < 512 \\ x'_{i,n+704} \cdot W_2(n-512), & \text{for } 512 \leq n < 768 \\ x'_{i,n+832} \cdot W_3(n-768), & \text{for } 768 \leq n < 1024 \\ x'_{i,n+960} \cdot W_4(n-1024), & \text{for } 1024 \leq n < 1280 \\ x'_{i,n+1088} \cdot W_5(n-1280), & \text{for } 1280 \leq n < 1536 \\ x'_{i,n+1216} \cdot W_6(n-1536), & \text{for } 1536 \leq n < 1792 \\ x'_{i,n+1344} \cdot W_7(n-1792), & \text{for } 1792 \leq n < 2048 \end{cases}$$

#### d) LONG\_STOP\_SEQUENCE

This window\_sequence is needed to switch from a EIGHT\_SHORT\_SEQUENCE back to a ONLY\_LONG\_SEQUENCE.

If **window\_shape**==1 the LONG\_STOP\_SEQUENCE is given as follows:

$$W(n) = \begin{cases} 0.0, & \text{for } 0 \leq n < 448 \\ W_{LEFT,256}(n-448), & \text{for } 448 \leq n < 576 \\ 1.0, & \text{for } 576 \leq n < 1024 \\ W_{KBD\_RIGHT,2048}(n), & \text{for } 1024 \leq n < 2048 \end{cases}$$

If **window\_shape**==0 the LONG\_START\_SEQUENCE is determined by:

$$W(n) = \begin{cases} 0.0, & \text{for } 0 \leq n < 448 \\ W_{LEFT,256}(n - 448), & \text{for } 448 \leq n < 576 \\ 1.0, & \text{for } 576 \leq n < 1024 \\ W_{SIN\_RIGHT,2048}(n), & \text{for } 1024 \leq n < 2048 \end{cases}$$

The windowed time domain values can be calculated with the formula explained in a).

#### 4.B.5.1.2 MDCT

The spectral coefficient,  $X_{i,k}$ , are defined as follows:

$$X_{i,k} = 2 \cdot \sum_{n=0}^{N-1} z_{i,n} \cos\left(\frac{2\pi}{N}(n + n_0)\left(k + \frac{1}{2}\right)\right) \text{ for } 0 \leq k < N / 2 .$$

where:

$z_{i,n}$  = windowed input sequence

$n$  = sample index

$k$  = spectral coefficient index

$i$  = block index

$N$  = window length of the one transform window based on the window\_sequence value

$n_0$  =  $(N / 2 + 1) / 2$

The analysis window length  $N$  of one transform window of the mdct is a function of the syntax element **window\_sequence** and is defined as follows:

$$N = \begin{cases} 2048, & \text{if ONLY\_LONG\_SEQUENCE (0x0)} \\ 2048, & \text{if LONG\_START\_SEQUENCE (0x1)} \\ 256, & \text{if EIGHT\_SHORT\_SEQUENCE (0x2) (8 times)} \\ 2048, & \text{if LONG\_STOP\_SEQUENCE (0x3)} \end{cases}$$

4.B.5.2 Diagrams

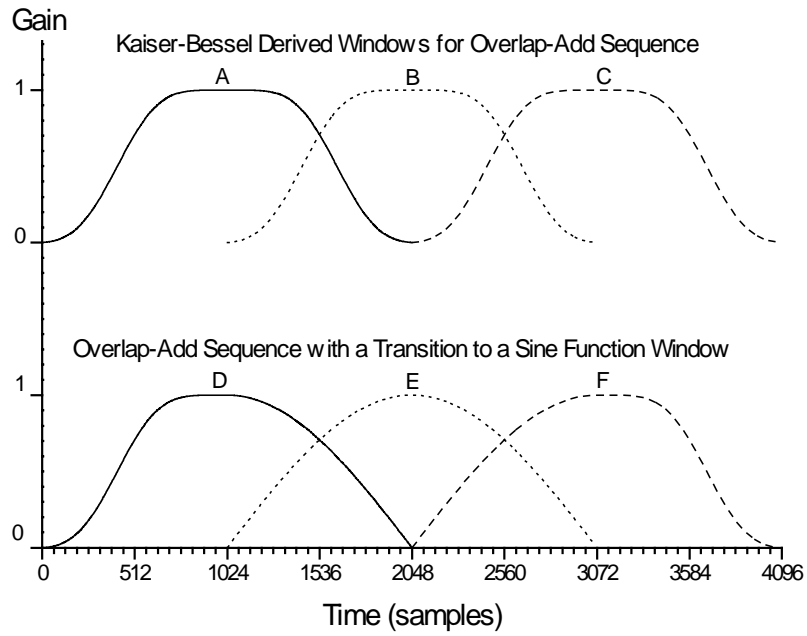


Figure 4.B.4 – An example of the Window Shape Adaptation Process.

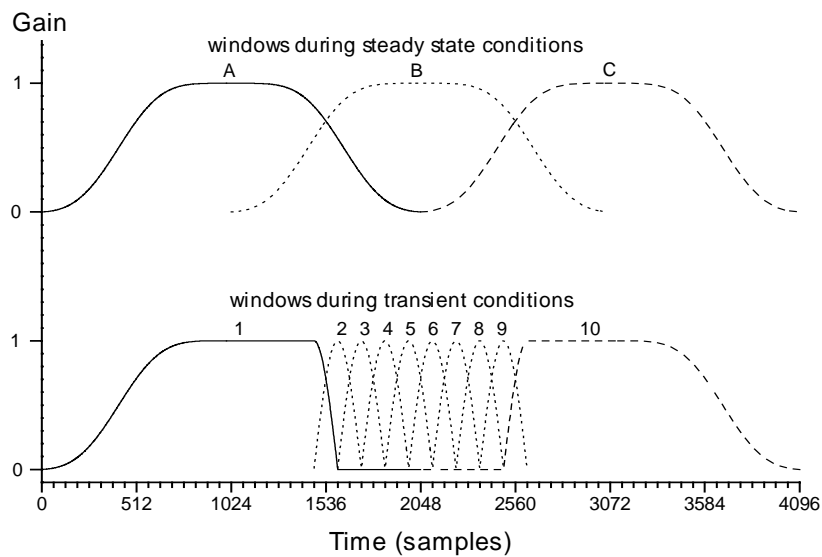


Figure 4.B.5 – Example of Block Switching During Transient Signal Conditions

## 4.B.6 Frequency domain prediction

### 4.B.6.1 Tool description

Since each predictor itself is identical on both, the encoder and decoder side, all descriptions and definitions as specified for the decoder in subclause 4.6.6 of the normative part are also valid here.

Prediction is used for an improved redundancy reduction and is especially effective in case of more or less stationary parts of a signal which belong to the most demanding parts in terms of required bitrate. Prediction can be applied to every channel using an intra channel (or mono) predictor which exploits the auto-correlation between the spectral components of consecutive frames. Because a `window_sequence` of type `EIGHT_SHORT_SEQUENCE` indicates signal changes, i.e. non-stationary signal characteristic, prediction is only used if `window_sequence` is of type `ONLY_LONG_SEQUENCE`, `LONG_START_SEQUENCE` or `LONG_STOP_SEQUENCE`.

For each channel prediction is applied to the spectral components resulting from the spectral decomposition of the filterbank. For each spectral component up to limit specified by `PRED_SFB_MAX`, there is one corresponding predictor resulting in a bank of predictors, where each predictor exploits the auto-correlation between the spectral component values of consecutive frames.

The overall coding structure using a filterbank with high spectral resolution implies the use of backward adaptive predictors to achieve high coding efficiency. In this case, the predictor coefficients are calculated from preceding quantized spectral components in the encoder as well as in the decoder and no additional side information is needed for the transmission of predictor coefficients - as would be required for forward adaptive predictors. A second order backward-adaptive lattice structure predictor is used for each spectral component, so that each predictor is working on the spectral component values of the two preceding frames. The predictor parameters are adapted to the current signal statistics on a frame by frame base, using an LMS based adaptation algorithm. If prediction is activated, the quantizer is fed with a prediction error instead of the original spectral component, resulting in a coding gain.

### 4.B.6.2 Encoding process

For each spectral component up to the limit specified by `PRED_SFB_MAX` of each channel there is one predictor. The following description is valid for one single predictor and has to be applied to each predictor. As said above, each predictor is identical on both, the encoder and decoder side. Therefore, the predictor structure is the same as shown in Figure 4.28 and the calculations of the estimate  $x_{est}(n)$  of the current spectral component  $x(n)$  as well as the calculation and adaptation of the predictor coefficients are exactly the same as those described for the decoder in subclause 4.6.6.3.2.

The only difference on the encoder side is that the prediction error has to be calculated according to

$$e(n) = x(n) - x_{est}(n)$$

to be fed to the quantizer. In this case the quantized prediction error is transmitted instead of the quantized spectral component.

#### 4.B.6.2.1 Predictor control

In order to guarantee that prediction is only used if this results in a coding gain, an appropriate predictor control is required and a small amount of predictor control information has to be transmitted to the decoder. For the predictor control, the predictors are grouped into scalefactor bands.

The following description is valid for either one `single_channel_element` or one `channel_pair_element` and has to be applied to each such element. Since prediction is only used if `window_sequence` is of type `ONLY_LONG_SEQUENCE`, `LONG_START_SEQUENCE` or `LONG_STOP_SEQUENCE` for the channel associated with the `single_channel_element` or for both channels associated with the `channel_pair_element`, the following applies only in these cases.

The predictor control information for each frame, which has to be transmitted as side information, is determined in two steps. First, it is determined for each scalefactor band whether or not prediction leads to a coding gain and if yes, the **prediction\_used** bit for that scalefactor band is set to one. After this has been done for all scalefactor bands up to PRED\_SFB\_MAX, it is determined whether the overall coding gain by prediction in this frame compensates at least the additional bit need for the predictor side information. If yes, the **predictor\_data\_present** bit is set to 1, the complete side information including that needed for predictor reset (see below) has to be transmitted and the prediction error value is fed to the quantizer. Otherwise, the **predictor\_data\_present** bit is set to 0, the **prediction\_used** bits are all reset to zero and are not transmitted. In this case, the spectral component value is fed to the quantizer. Figure 4.B.6 shows a block diagram of the prediction unit for one scalefactor band. As described above, the predictor control first operates on all predictors of one scalefactor band and is then followed by a second step over all scalefactor bands.

In case of a `single_channel_element` or a `channel_pair_element` with **common\_window** = 0 the control information is calculated and valid for the predictor bank(s) of the channel(s) associated with that element. In case of a `channel_pair_element` with **common\_window** = 1 the control information is calculated considering both channels associated with that element together. In this case the control information is valid for both predictor banks of the two channels in common.

#### 4.B.6.2.2 Predictor reset

When the encoding process is started, all predictors are initialized. This means that the predictor state variables for each predictor are set as follows:  $r_0 = r_1 = 0$ ,  $COR_1 = COR_2 = 0$ ,  $VAR_1 = VAR_2 = 1$ .

A cyclic reset mechanism is applied by the encoder, where all predictors are initialized again in a certain time interval in an interleaved way. On one hand this increases the stability by re-synchronizing the predictors of the encoder and the decoder and on the other hand it allows defined entry points in the bitstream.

The whole set of predictors is subdivided into 30 so-called reset groups according to the following table:

**Table 4.B.25 – Reset group**

| <i>Reset group number</i> | <i>Predictors of reset group</i>         |
|---------------------------|------------------------------------------|
| 1                         | $P_0, P_{30}, P_{60}, P_{90}, \dots$     |
| 2                         | $P_1, P_{31}, P_{61}, P_{91}, \dots$     |
| 3                         | $P_2, P_{32}, P_{62}, P_{92}, \dots$     |
| ...                       |                                          |
| 30                        | $P_{29}, P_{59}, P_{89}, P_{119}, \dots$ |

where  $P_i$  is the predictor which corresponds to the spectral coefficient indexed by  $i$ .

An encoder is required to reset at least one group every 8 frames and to reset every group within the maximum reset interval of  $8 \times 30 = 240$  frames. It is recommended that groups be reset in ascending order, although that is not mandatory. Shorter reset intervals are supported by the bitstream syntax, for example if one group is reset every fourth frame, then all predictors can be reset within an interval of  $4 \times 30 = 120$  frames.

A typical reset cycle starts with reset group number 1 and the reset group number is then incremented by 1 until it reaches 30, and then it starts with 1 again. Nevertheless, it may happen, e.g. due to switching between programs (bitstreams) or cutting and pasting, that there will be a discontinuity in the reset group numbering. If this is the case, there are the following three possibilities for the decoder to operate:

- Ignore the discontinuity and carry on the normal processing. This may result in a short audible distortion due to a mismatch (drift) between the predictors in the encoder and decoder. After one complete reset cycle (reset group  $n, n+1, \dots, 30, 1, 2, \dots, n-1$ ) the predictors are re-synchronized again. Furthermore, a possible distortion is faded out because of the attenuation factors  $a$  and  $b$ .
- Detect the discontinuity, carry on the normal processing but mute the output until one complete reset cycle is performed and the predictors are re-synchronized again.

- Reset all predictors.

The reset mechanism is controlled by the **pred\_reset** bit, which always has to be transmitted as soon as the **predictor\_data\_present** bit is set to 1, and the **pred\_reset\_group\_number**, which specifies the group of predictors to be reset and has only to be transmitted if the **pred\_reset** bit is set to 1. If a group reset is applied in the current frame, the **pred\_reset** bit has to be set to 1 and the number of the predictor group to be reset has to be coded and transmitted as 5 bit unsigned binary number in **pred\_reset\_group\_number**. After the quantized value has been reconstructed (see below), the reset of the specified group is then carried out by initializing all predictors belonging to that group as described above.

If no group reset is applied in the current frame, the **pred\_reset** bit has to be set to 0.

In case of a **single\_channel\_element** or a **channel\_pair\_element** with **common\_window** = 0 the reset has to be applied to the predictor bank(s) of the channel(s) associated with that element and the control information relates to each individual channel. In case of a **channel\_pair\_element** with **common\_window** = 1 the reset has to be applied to the two predictor banks of the two channels associated with that element and the control information relates to both channels together.

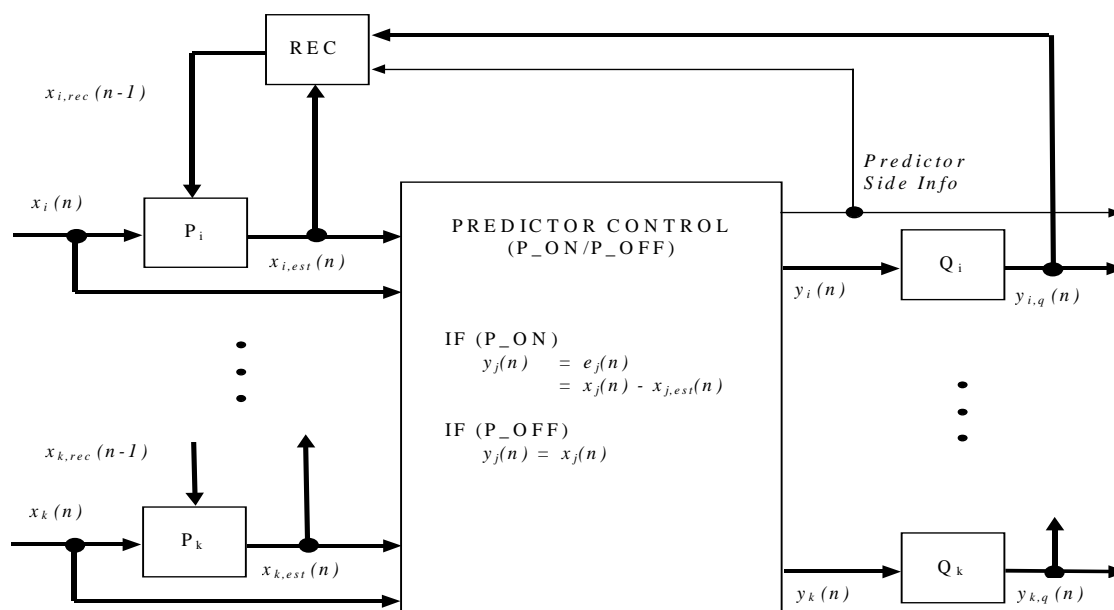
In case of short blocks, i.e. **window\_sequence** is of type **EIGHT\_SHORT\_SEQUENCE**, prediction is always disabled and a reset is carried out for all predictors in all scalefactor bands, which is equivalent to a complete reinitialization, see above.

#### 4.B.6.2.3 Reconstruction of the quantized spectral component

Since the reconstructed value of the quantized spectral component is needed as predictor input signal, it has to be calculated in the encoder, see also Figure 4.B.6. Depending on the value of the **prediction\_used** bit, the reconstructed value is either the quantized spectral component or the quantized prediction error. Therefore, the following steps are necessary:

- If the bit is set (1), then the quantized prediction error, reconstructed from data to be transmitted, is added to the estimate  $x_{est}(n)$ , calculated by the predictor, resulting in the reconstructed value of the quantized spectral component, i.e.  $x_{rec}(n) = x_{est}(n) + e_q(n)$
- If the bit is not set (0), then the quantized value of the spectral component is identical to the value reconstructed directly from the data to be transmitted.

## 4.B.6.3 Diagrams



**Figure 4.B.6 – Block diagram of prediction unit for one scalefactor band. The complete processing is only shown for predictor  $P_i$  (Q - quantizer, REC - reconstruction of last quantized value). Note that the predictor control operates on all predictors  $P_i \dots P_j \dots P_k$  of a scalefactor band and is followed by a second control over all scalefactor band.**

## 4.B.7 Long Term Prediction

The general structure of the perceptual audio coder with LTP is shown in Figure 4.B.7. First, the optimal LTP is estimated. Then, the predicted signals based on quantized data are obtained and corresponding error signals are calculated. Finally, the error signals are quantized and transmitted with the side information, such as predictor control and predictor parameters.



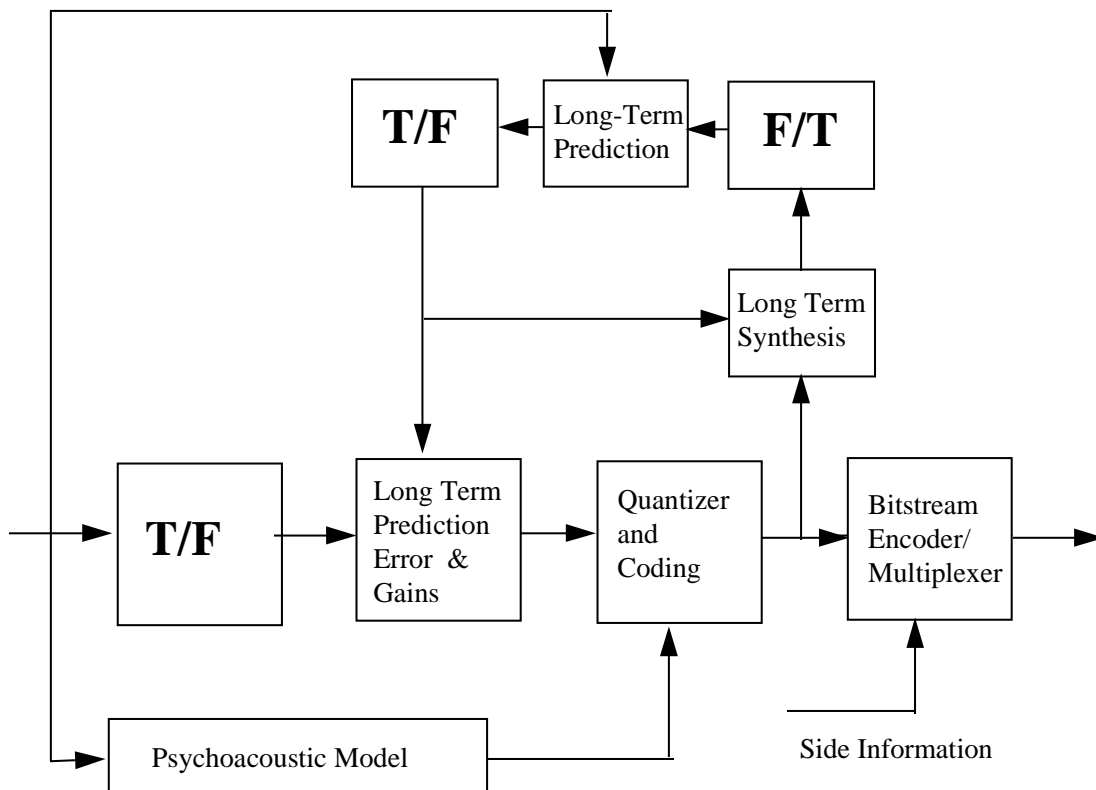


Figure 4.B.7 – Block diagram of the encoding process of Long Term Prediction

As the perceptual audio coder processes the signal frame by frame, it is obvious that when we want to quantize  $\mathbf{X}_{m+1}$ , the quantized data  $\tilde{\mathbf{X}}_k, k = m, m-1, m-2, \dots$ , are available both in encoder and decoder. In time domain, it means that the quantised time domain samples  $\tilde{x}(i), i = mN, mN-1, \dots$ , are available.

To remove the redundancy of the signal in present frame  $m+1$  based on the previously quantized data, an LTP is used

$$P(z) = \sum_{k=-m_1}^{m_2} b_k z^{-(\alpha+k)}$$

where  $\alpha$  represents a long delay in the range 1 to 2048. For  $m_1 = m_2 = 0$ , we have a one-tap predictor.

The parameters  $\alpha$  and  $b_k$  are determined by minimizing the mean squared error over the whole window. The delay  $\alpha$  is quantized with 11 bits with 2048 possible values in the range 1 to 2048. For prediction coefficients  $b_k$  are non-uniformly quantized to 3 bits. Due to their nonuniform distribution, nonuniform quantization has to be used. The table for coding the coefficient values is presented in the decoding process description.

After the LTP is determined, the predicted signal can be obtained for the  $(m+1)$ th frame

$$\hat{x}(i) = \sum_{k=-m_1}^{m_2} b_k \tilde{x}(i-1-k-\alpha),$$

$$i = mN + 1, mN + 2, \dots, (m+1)N$$

Using the forward MDCT, we have the predicted spectral coefficients  $\tilde{X}_{m+1}$  for the  $(m+1)$ th frame. In order to guarantee that prediction is only used if it results in a coding gain, an appropriate predictor control is required and a small amount of predictor control information has to be transmitted to the decoder. The predictor control scheme is the same as the backward predictor control scheme which has been used in MPEG-2 Advanced Audio Coding (AAC), i.e., one bit per scalefactor band is sent as side information to flag whether prediction is used for that band.

In the encoding process the prediction gain resulting from LTP is estimated for each scalefactor band by comparing the predicted energy to the amount of side information needed. LTP is switched on only for those scalefactor bands where there is positive net coding gain. The corresponding bits of side information are set accordingly. Finally, the net coding gain for the whole frame (all scalefactor bands) is checked, and the switch for using LTP at all for the frame is set accordingly.

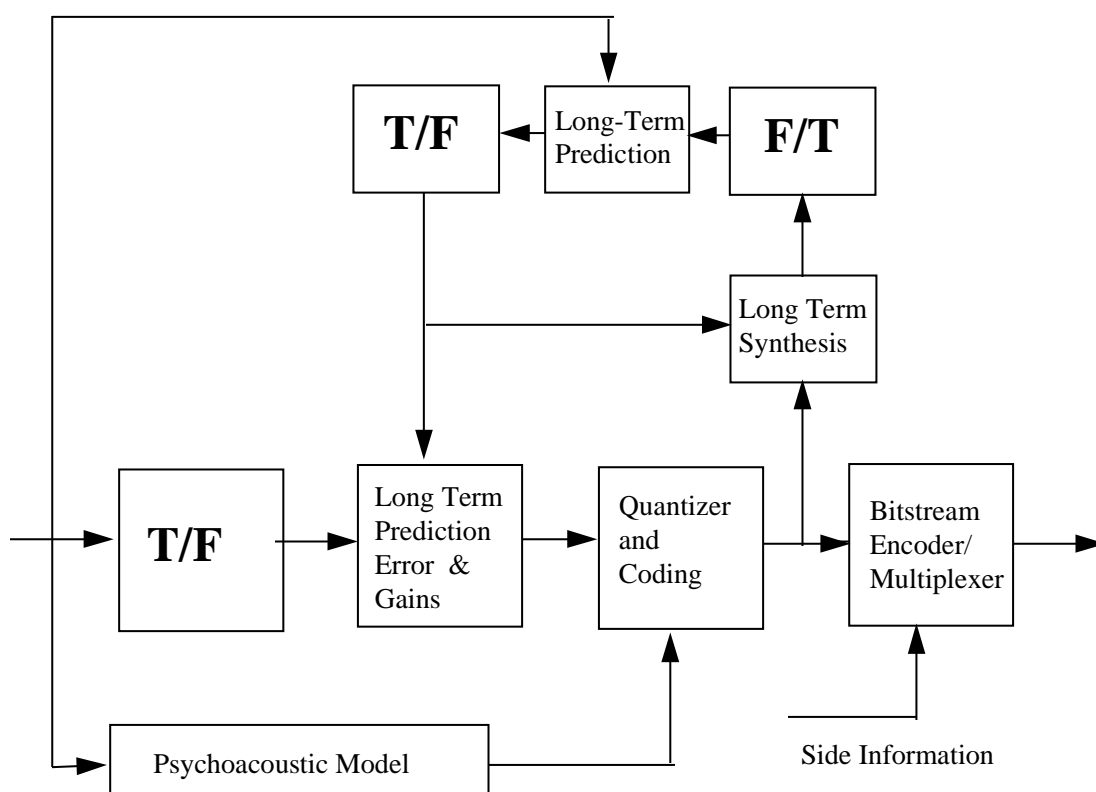


Figure 4.B.8 – Block diagram of the encoding process of Long Term Prediction

#### 4.B.8 Temporal Noise Shaping (TNS)

Temporal Noise Shaping is used to control the temporal shape of the quantization noise within each window of the transform. This is done by applying a filtering process to parts of the spectral data of each channel.

Encoding is done on a window basis. The following steps are carried out to apply the Temporal Noise Shaping tool to one window of spectral data:

- A target frequency range for the TNS tool is chosen. A suitable choice is to cover a frequency range from 1.5 kHz to the uppermost possible scalefactor band with one filter. Please note that this parameter (TNS\_MAX\_BANDS) depends on profile and sampling rate as indicated in the normative part.
- Next, a linear predictive coding (LPC) calculation is carried out on the spectral MDCT coefficients corresponding to the chosen target frequency range. For better stability, coefficients corresponding to frequencies below 2.5 kHz may be excluded from this process. Standard LPC procedures as known from speech processing can be used for the LPC calculation, e.g. the well-known Levinson-Durbin algorithm. The

calculation is carried out for the maximum permitted order of the noise shaping filter (TNS\_MAX\_ORDER). Please note that this value depends on the profile as indicated in the normative part.

- As a result of the LPC calculation, the expected prediction gain  $g_p$  is known as well as the TNS\_MAX\_ORDER reflection coefficients  $r[]$  (so-called PARCOR coefficients).
- If the prediction gain  $g_p$  does not exceed a certain threshold  $t$ , no temporal noise shaping is used. In this case, the `tns_data_present` bit is set to zero and TNS processing is finished. A suitable threshold value is  $t = 1.4$ .
- If the prediction gain  $g_p$  exceeds the threshold  $t$ , temporal noise shaping is used.
- In a next step the reflection coefficients are quantized using `coef_res` bits. An appropriate choice for `coef_res` is 4 bits. The following pseudo code describes the conversion of the reflection coefficients  $r[]$  to index values `index[]` and back to quantized reflection coefficients `rq[]`.

```

iqfac = ((1 << (coef_res-1)) - 0.5) / (π/2.0);
iqfac_m = ((1 << (coef_res-1)) + 0.5) / (π/2.0);

/* Reflection coefficient quantization */
for (i=0; i<TNS_MAX_ORDER; i++) {
    index[i] = NINT( arcsin( r[i] ) * ((r[i] >= 0) ? iqfac : iqfac_m) );
}
/* Inverse quantization */
for (i=0; i<TNS_MAX_ORDER; i++) {
    rq[i] = sin( index[i] / ((index[i] >= 0) ? iqfac : iqfac_m) );
}

```

where `arcsin()` denotes the inverse `sin()` function.

- The order of the used noise shaping filter is determined by subsequently removing all reflection coefficients with an absolute value smaller than a threshold  $p$  from the "tail" of the reflection coefficient array. The number of the remaining reflection coefficients is the order of the noise shaping filter. A suitable threshold for truncation is  $p = 0.1$ .
- The remaining reflection coefficients `rq[]` are converted into order+1 linear prediction coefficients `a[]` (known as "step-up procedure"). A description of this procedure is provided in the normative part as a part of the tool description (see "/\* Conversion to LPC coefficients \*/").
- The computed LPC coefficients `a[]` are used as the encoder noise shaping filter coefficients. This FIR filter is slid across the specified target frequency range exactly the way it is described in the normative part for the decoding process (tool description). The difference between the decoding and encoding filtering is that the all-pole (auto-regressive) filter used for decoding is replaced by its inverse all-zero (moving-average) filter, i.e. replacing the decoder filter equation

$$y[n] = x[n] - a[1]*y[n-1] - \dots - a[order]*y[n-order]$$

by the inverse (encoder) filter equation

$$y[n] = x[n] + a[1]*x[n-1] + \dots + a[order]*x[n-order]$$

By default, an upward direction of the filtering is appropriate.

- Finally, the side information for Temporal Noise Shaping is transmitted:

| Bitstream Element          | Algorithmic Variable or Value         |
|----------------------------|---------------------------------------|
| <code>n_filt</code>        | 1                                     |
| <code>coef_res</code>      | <code>coef_res-3</code>               |
| <code>coef_compress</code> | 0                                     |
| <code>length</code>        | Number of processed scalefactor bands |
| <code>direction</code>     | 0 (upwards)                           |

|        |                               |
|--------|-------------------------------|
| order  | Order of noise shaping filter |
| coef[] | index[]                       |

- Optionally, the use of the `coef_compress` field allows saving 1 bit per transmitted reflection coefficient if none of the reflection coefficients use more than half of their full range. Specifically, if the two most significant bits of each quantized reflection coefficient are either '00' or '11', `coef_compress` may be set to a value of one and the size of the transmitted quantized reflection coefficients decreased by one.

## 4.B.9 Joint coding

### 4.B.9.1 M/S stereo

The decision to code left and right coefficients as either left + right (L/R) or mid/side (M/S) is made on a noiseless coding band by noiseless coding band basis for all spectral coefficients in the current block. For each noiseless coding band the following decision process is used:

- For each noiseless coding band, not only L and R raw thresholds, but also  $M=(L+R)/2$  and  $S=(L-R)/2$  raw thresholds are calculated. For the raw M and S thresholds, rather than using the tonality for the M or S threshold, one uses the more tonal value from the L or R calculation in each threshold calculation band, and proceed with the psychoacoustic model for M and S from the M and S energies and the minimum of the L or R values for  $C(\omega)$  in each threshold calculation band. The values that are provided to the imaging control process are identified in the psychoacoustic model information section as  $en(b)$  (the spread normalized energy) and  $nb(b)$ , the raw threshold.
- The raw thresholds for M, S, L and R, and the spread energy for M, S, L and R, are all brought into an "imaging control process". The resulting adjusted thresholds are inserted as the values for  $cb(b)$  into step 11 of the psychoacoustic model for further processing.
- The final, protected and adapted to coder-band thresholds for all of M,S,L and R are directly applied to the appropriate spectrum by quantizing the actual L, R, M and S spectral values with the appropriate calculated and quantized threshold.
- The number of bits actually required to code M/S, and the number of bits required to code L/R are calculated.
- The method that uses the least bits is used in each given noiseless coding band, and the stereo mask is set accordingly.

With these definitions

Mthr, Sthr, Rthr, Lthr      raw thresholds. (the  $nb(b)$  from step 10 of the psychoacoustic model)  
Mengy, Sengy, Rengy, Sengy      spread energy. ( $en(b)$  from step 6 of the psychoacoustic model)  
Mfthr, Sfthr, Rfthr, Lfthr      final (output) thresholds. (returned as  $nb(b)$  in step 11 of the psychoacoustic model)  
**bmax(b)**      BMLD protection ratio, as can be calculated from

$$bmax(b) = 10^{-3} \left[ 0.5 + 0.5 \cos \left( \pi \cdot \frac{\min(bval(b), 15.5)}{15.5} \right) \right]$$

the imaging control process for each noiseless coding band is as follows:

```
t=Mthr/Sthr
if (t>1)
  t=1/t
Rfthr= max(Rthr*t, min (Rthr, bmax*Rengy))
```

$Lfthr = \max(Lthr \cdot t, \min(Lthr, bmax) \cdot Lengy)$   
 $t = \min(Lthr, Rthr)$   
 $Mfthr = \min(t, \max(Mthr, \min(Sengy \cdot bmax, Sthr)))$   
 $Sfthr = \min(t, \max(Sthr, \min(Mengy \cdot bmax, Mthr)))$

#### 4.B.9.2 Intensity stereo coding

Intensity stereo coding is used to exploit irrelevance in the between both channels of a channel pair in the high frequency region. The following procedure describes one possible implementation while several different implementations are possible within the framework of the defined bitstream syntax.

Encoding is done on a window group basis. The following steps are carried out to apply the intensity stereo coding tool to one window group of spectral data:

- A suitable approach is to code a consecutive region of scalefactor bands in intensity stereo technique starting above a lower border frequency  $f_0$ . An average value of  $f_0 = 6$  kHz is appropriate for most types of signals.
- For each scalefactor band, the energy of the left, right and the sum channel is calculated by summing the squared spectral coefficients, resulting in values  $E_l[sfb]$ ,  $E_r[sfb]$ ,  $E_s[sfb]$ . If the window group comprises several windows, the energies of the included windows are added.
- For each scalefactor band, the corresponding intensity position value is computed as

$$is\_position[sfb] = NINT \left( 2 \cdot \log_2 \left( \frac{E_l[sfb]}{E_r[sfb]} \right) \right)$$

- Next, the intensity signal spectral coefficients  $spec_i[i]$  are calculated for each scalefactor bands by adding spectral samples from the left and right channel ( $spec_l[i]$  and  $spec_r[i]$ ) and rescaling the resulting values like

$$spec_i[i] = (spec_l[i] + spec_r[i]) \cdot \sqrt{\frac{E_l[sfb]}{E_s[sfb]}}$$

- The intensity signal spectral components are used to replace the corresponding left channel spectral coefficients. The corresponding spectral coefficients of the right channel are set to zero.

Then, the standard process for quantization and encoding is performed on the spectral data of both channels. However, the prediction status of the right channel predictors is forced to "off" for the scalefactor bands coded in intensity stereo. These predictors are updated by using an intensity decoded version of the quantized spectral coefficients. The procedure for this is described in the tool description for the intensity stereo decoding process in the normative part.

Finally, before transmission the Huffman codebook INTENSITY\_HCB (15) is set in the sectioning information for all scalefactor bands that are coded in intensity stereo.

#### 4.B.10 Quantization

##### 4.B.10.1 Introduction

The description of the AAC quantization module is subdivided into three levels. The top level is called "loops frame program". The loops frame program calls a subroutine named "outer iteration loop" which calls the subroutine "inner iteration loop". For each level a corresponding flow diagram is shown.

The loops module quantizes an input vector of spectral data in an iterative process according to several demands. The inner loop quantizes the input vector and increases the quantizer step size until the output vector can be coded with the available number of bits. After completion of the inner loop an outer loop checks the distortion of each scalefactor band and, if the allowed distortion is exceeded, amplifies the scalefactor band and calls the inner loop again.

AAC loops module input:

1. vector of the magnitudes of the spectral values `mdct_line(0..1023)`.
2. `xmin(sb)` (see subclause 4.B.3.4. „Steps in threshold calculation,, Step 12)
3. `mean_bits` (average number of bits available for encoding the bitstream).
4. `more_bits`, the number of bits in addition to the average number of bits, calculated by the psychoacoustic module out of the perceptual entropy (PE).
5. the number and width of the scalefactor bands (see Table 4.73 to Table 4.91 normative part)
6. for short block grouping the spectral values have to be interleaved so that spectral lines that belong to the same scalefactor band but to different block types which shall be quantized with the same scalefactors are put together in one (bigger) scalefactor band (for a full description of grouping see subclause 4.5.2.3.4.)

AAC loops module output:

1. vector of quantized values `x_quant(0..1023)`.
2. a scalefactor for each scalefactor band (`sb`)
3. `common_scalefac` (quantizer step size information for all scalefactor bands)
4. number of unused bits available for later use.

## 4.B.10.2 Preparatory steps

### 4.B.10.2.1 Reset of all iteration variables

1. The start value of `global_gain` for the quantizer is calculated so that all quantized MDCT values can be encoded in the bitstream :

$$start\_common\_scalefac = ceiling(16/3 * (log_2( (max\_mdct\_line ^ (3/4) ) / MAX\_QUANT)))$$

`max_mdct_line` is the maximum MDCT coefficient. `MAX_QUANT` is the maximum quantized value which can be encoded in the bitstream, it is defined to 8191. During the iteration process, the `common_scalefac` must not become less than `start_common_scalefac`.

2. `Scalefactor[sb]` is set to zero for all values of `sb`.

### 4.B.10.3 Bit reservoir control

Bits are saved to the reservoir when fewer than the `mean_bits` are used to code one frame.

$$mean\_bits = bit\_rate * 1024 / sampling\_rate.$$

The number of bits which can be saved in the bit reservoir at maximum is called 'maximum\_bitreservoir\_size' which is calculated using the procedure outlined in subclause 4.5.3.2. If the reservoir is full, unused bits have to be encoded in the bitstream as fillbits.

The maximum amount of bits available for a frame is the sum of `mean_bits` and bits saved in the bit reservoir.

The number of bits that should be used for encoding a frame depends on the `more_bits` value which is calculated by the psychoacoustic model and the maximum available bits. The simplest way to control bit reservoir is :

*if more\_bits > 0 :*

$$available\_bits = average\_bits + min ( more\_bits, bitres\_bits)$$

*if more\_bits < 0 :*

$$available\_bits = average\_bits + max ( more\_bits, bitres\_bits - maximum\_bitreservoir\_size)$$

#### 4.B.10.4 Quantization of MDCT coefficients

The formula for the quantization in the encoder is the inverse of the decoder dequantization formula (see also the decoder description) :

$$x_{quant} = \text{int} \left( \left( \text{abs}(mdct\_line) \right) * \left( 2^{(-1/4 * (sf\_decoder - SF\_OFFSET))} \right) \right)^{3/4} + MAGIC\_NUMBER$$

MAGIC\_NUMBER is defined to 0.4054, SF\_OFFSET is defined as 100 and mdct\_line is one of spectral values, which is calculated from the MDCT. These values are also called 'coefficients'. The scalefactor sf\_decoder is the same as "sf[g][sfb]" used in the normative part.

For use in the iteration loops, the scalefactor 'sf\_decoder' is split in two variables:

$$sf\_decoder = scalefactor - common\_scalefac + SF\_OFFSET$$

It follows from this, that the formula used in the distortion control loop is:

$$x_{quant} = \text{int} \left( \left( \text{abs}(mdct\_line) \right) * \left( 2^{(-1/4 * (scalefactor - common\_scalefac))} \right) \right)^{3/4} + MAGIC\_NUMBER$$

The sign of the mdct\_line is saved separately and added again only for counting the bits and encoding the bitstream

##### 4.B.10.4.1 Outer iteration loop (distortion control loop)

The outer iteration loop controls the quantization noise which is produced by the quantization of the frequency domain lines within the inner iteration loop. The coloring of the noise is done by multiplication of the lines within scalefactor bands with the actual scalefactors before doing the quantization. The following pseudo-code illustrates the multiplication.

```
do for each scalefactor band sb:
  do from lower index to upper index i of scalefactor band
    mdct_scaled(i) = abs(mdct_line(i))3/4 * 2(3/16 * (scalefactor(sb)))
  end do
end do
```

##### 4.B.10.4.2 Call of inner iteration loop

For each outer iteration loop (distortion control loop) the inner iteration loop (rate control loop) is called. The parameters are the frequency domain values with the scalefactors applied to the values within the scalefactor bands ( mdct\_scaled(0..1023) ), a start value for common\_scalefac, and the number of bits which are available to the rate control loop. The result is the number of bits actually used and the quantized frequency lines x\_quant(*i*), and a new common\_scalefac.

The formula to calculate the quantized MDCT coefficients is:

$$x_{quant}(i) = \text{int} \left( \left( mdct\_scaled(i) \right) * \left( 2^{(-3/16 * common\_scalefac)} \right) \right) + MAGIC\_NUMBER$$

The bits, that would be needed to encode the quantizes values and the side information (scalefactors etc.) are counted according to the bitstream syntax, described in subclause 4.4.

##### 4.B.10.4.3 Amplification of scalefactor bands which violate the masking threshold

The calculation of the distortion (error\_energy(*sb*)) of the scalefactor band is done as follows:

```
do for each scalefactor band sb:
  error_energy(sb)=0
  do from lower index to upper index i of scalefactor band
    error_energy(sb) = error_energy(sb) + (abs( mdct_line(i))
```

```

- (x_quant(i)^(4/3) * 2^(1/4 * (scalefactor(sb) - common_scalefac))))^2
end do
end do

```

All spectral values of the scalefactor bands which have a distortion that exceeds the allowed distortion ( $xmin(sb)$ ) are amplified according to formula in Figure 4.B.10 (“Outer Iteration Loop”), the new scalefactors can be calculated according to this pseudocode:

```

do for each scalefactor band sb
  if ( error_energy(sb) > xmin(sb) ) then
    scalefactor(sb) = scalefactor(sb) + 1
  end if
end do

```

#### 4.B.10.4.4 Conditions for the termination of the loops processing

Normally the loops processing terminates, if there is no scalefactor band with more than the allowed distortion. However this is not always possible to obtain. In this case there are other conditions to terminate the outer loop. If

- All scalefactor bands are already amplified, or
- The difference between two consecutive scalefactors is greater than 64

The loop processing stops, and by restoring the saved scalefactors( $sb$ ) a useful output is available. For real-time implementation, there might be a third condition added which terminates the loops in case of a lack of computing time.

#### 4.B.10.4.5 Inner iteration loop (rate control loop)

The inner iteration loop calculates the actual quantization of the frequency domain data ( $mdct\_scaled$ ) with the following function, which uses the formula from clause 4.B.10.4.2 “call of Inner iteration loop”:

*quantize\_spectrum(x\_quant[], mdct\_scaled[], common\_scalefac):*

```

do for all MDCT coefficients i :
  x_quant(i) = int (( mdct_scaled (i) * 2^(-3/16 * common_scalefac) ) + MAGIC_NUMBER)
end do

```

and then calls a function *bit\_count()*. This function counts the number of bits that would be necessary to encode a bitstream according to clause 4.4.

The inner iteration loop can be implemented using successive approximation.:

```

inner_loop():
  if (outer_loop_count == 0)
    common_scalefac = start_common_scalefac
    quantizer_change = 64
  else
    quantizer_change = 2
  end if
  do
    quantize_spectrum()
    counted_bits = bit_count()
    if (counted_bits > available_bits) then
      common_scalefac = common_scalefac + quantizer_change
    else
      common_scalefac = common_scalefac - quantizer_change
    end if
    quantizer_change = int ( quantizer_change / 2)
    if (quantizer_change == 0) && (counted_bits > available_bits)

```



```
    quantizer_change = 1  
  end if  
  while ( quantizer_change != 0 )
```

Due to the choice of *start\_common\_scalefac*, after the first run through the inner loop the number of needed bits is always greater than the available bits , and therefore *common\_scalefac* is always increased by the *quantizer\_change*.

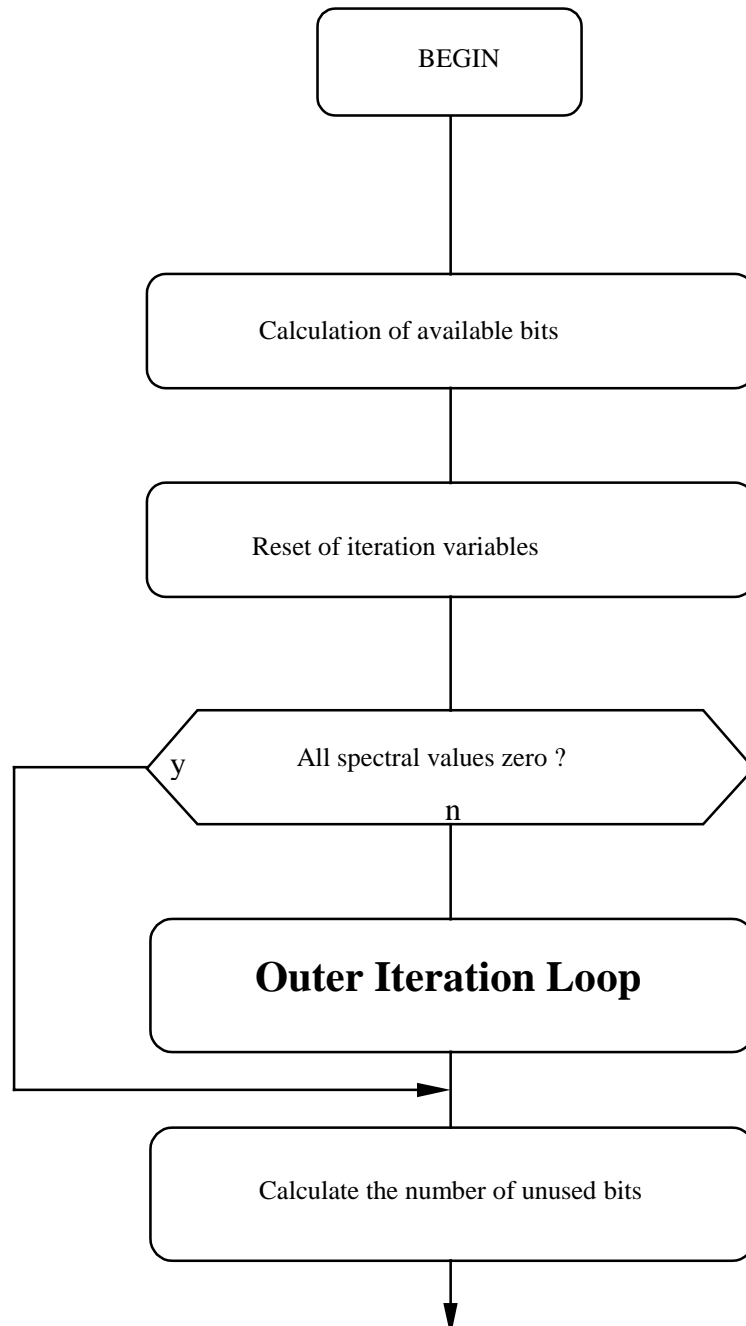


Figure 4.B.9 – AAC iteration loop

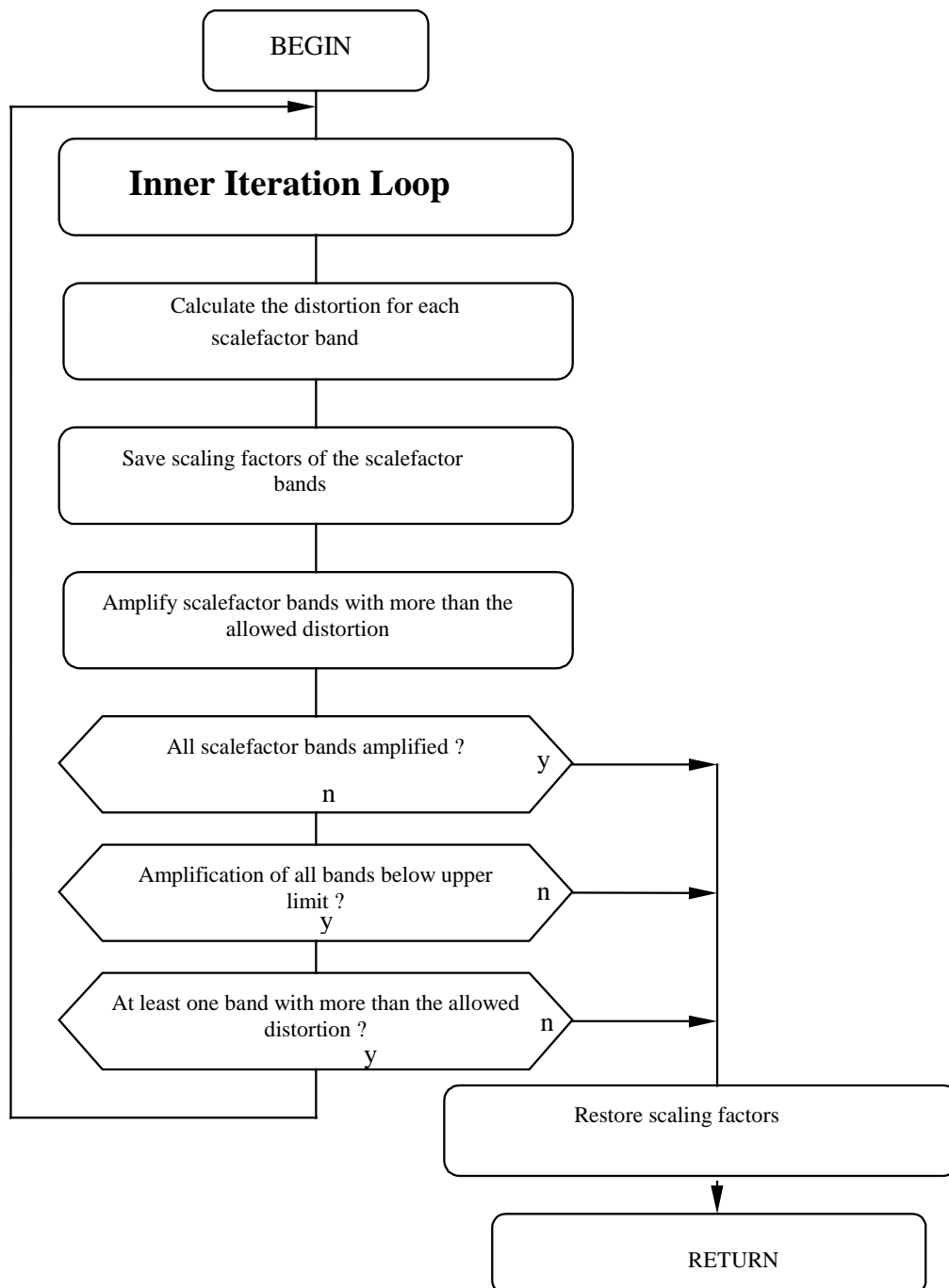


Figure 4.B.10 – AAC outer iteration loop

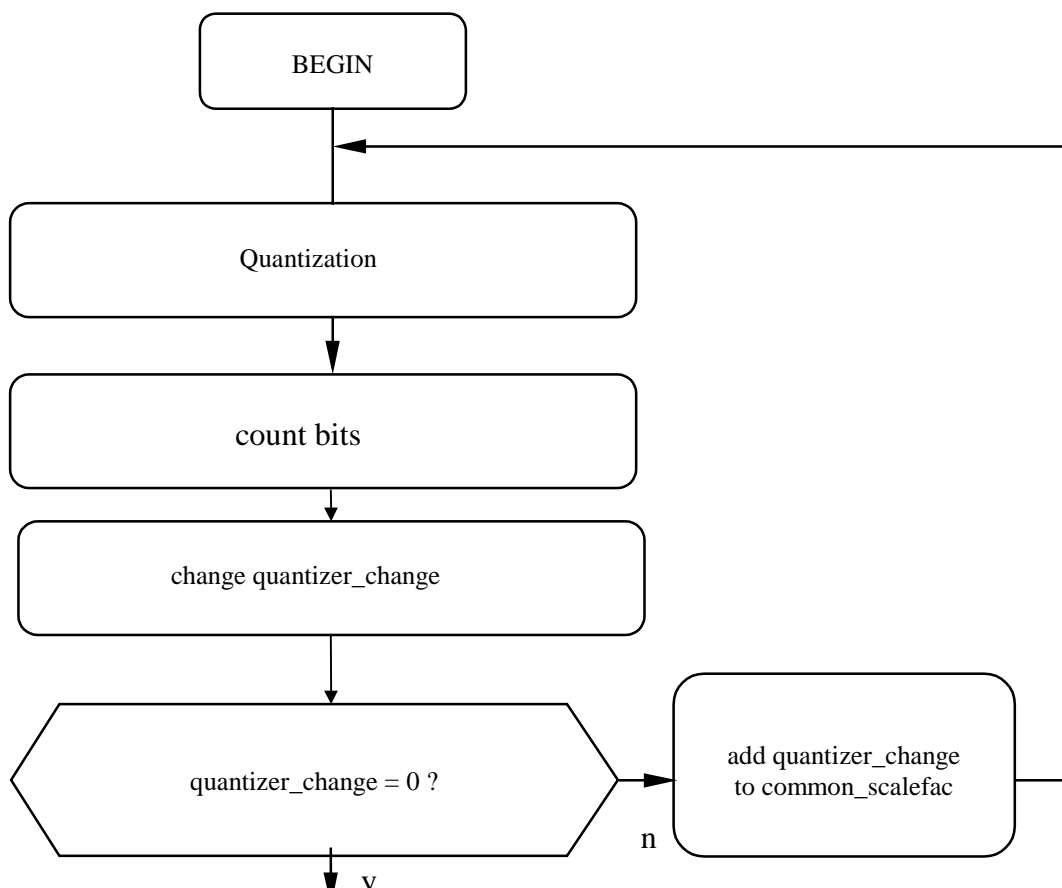


Figure 4.B.11 – AAC inner iteration loop

## 4.B.11 Noiseless coding

### 4.B.11.1 Introduction

In the AAC encoder the input to the noiseless coding module is the set of 1024 quantized spectral coefficients. Since the noiseless coding is done inside the quantizer inner loop, it is part of an iterative process that converges when the total bit count (of which the noiseless coding is the vast majority) is within some interval surrounding the allocated bit count. This subclause will describe the encoding process for a single call to the noiseless coding module.

Noiseless coding is done via the following steps:

- Spectrum clipping
- Preliminary Huffman coding using maximum number of sections
- Section merging to achieve lowest bit count

### 4.B.11.2 Spectrum clipping

As a first step a method of noiseless dynamic range limiting may be applied to the spectrum. Up to four coefficients can be coded separately as magnitudes in excess of one, with a value of +-1 left in the quantized coefficient array to carry the sign. The index of the scalefactor band containing the lowest-frequency “clipped” coefficients is sent in the bitstream. Each of the “clipped” coefficients is coded as a magnitude (in excess of 1) and an offset from the base of the previously indicated scalefactor band. For this the long block scalefactor bands and coefficient ordering

within those bands are used regardless of the window sequence. One strategy for applying spectrum clipping is to clip high-frequency coefficients whose absolute amplitudes are larger than one. Since the side information for carrying the clipped coefficients costs some bits, this noiseless compression is applied only if it results in a net savings of bits.

#### 4.B.11.3 Sectioning

The noiseless coding segments the set of 1024 quantized spectral coefficients into *sections*, such that a single Huffman codebook is used to code each section (the method of Huffman coding is explained in a later subclause). For reasons of coding efficiency, section boundaries can only be at scalefactor band boundaries so that for each section of the spectrum one must transmit the length of the section, in scalefactor bands, and the Huffman codebook number used for the section.

Sectioning is dynamic and typically varies from block to block, such that the number of bits needed to represent the full set of quantized spectral coefficients is minimized. This is done using a greedy merge algorithm starting with the maximum possible number of sections each of which uses the Huffman codebook with the smallest possible index. Sections are merged if the resulting merged section results in a lower total bit count, with merges that yield the greatest bit count reduction done first. If the sections to be merged do not use the same Huffman codebook then the codebook with the higher index must be used.

Sections often contain only coefficients whose value is zero. For example, if the audio input is band limited to 20 kHz or lower, then the highest coefficients are zero. Such sections are coded with Huffman codebook zero, which is an escape mechanism that indicates that all coefficients are zero and it does not require that any Huffman codewords be sent for that section.

#### 4.B.11.4 Grouping and interleaving

If the window sequence is eight short windows then the set of 1024 coefficients is actually a matrix of 8 by 128 frequency coefficients representing the time-frequency evolution of the signal over the duration of the eight short windows. Although the sectioning mechanism is flexible enough to efficiently represent the 8 zero sections, *grouping* and *interleaving* provide for greater coding efficiency. As explained earlier, the coefficients associated with contiguous short windows can be grouped such that they share scalefactors amongst all scalefactor bands within the group. In addition, the coefficients within a group are interleaved by interchanging the order of scalefactor bands and windows. To be specific, assume that before interleaving the set of 1024 coefficients  $c$  are indexed as

$$c[g][w][b][k]$$

where

$g$  is the index on groups

$w$  is the index on windows within a group

$b$  is the index on scalefactor bands within a window

$k$  is the index on coefficients within a scalefactor band

and the right-most index varies most rapidly.

After interleaving the coefficients are indexed as

$$c[g][b][w][k]$$

This has the advantage of combining all zero sections due to band-limiting within each group.

#### 4.B.11.5 Scalefactors

The coded spectrum uses one quantizer per scalefactor band. The step sizes of each of these quantizers is specified as a set of scalefactors and a global gain which normalizes these scalefactors. In order to increase compression, scalefactors associated with scalefactor bands that have only zero-valued coefficients are ignored in the coding process and therefore do not have to be transmitted. Both the global gain and scalefactors are

quantized in 1.5 dB steps. The global gain is coded as an 8-bit unsigned integer and the scalefactors are differentially encoded relative to the previous scalefactor (or global gain for the first scalefactor) and then Huffman coded. The dynamic range of the global gain is sufficient to represent full-scale values from a 24-bit PCM audio source.

**4.B.11.6 Huffman coding**

Huffman coding is used to represent n-tuples of quantized coefficients, with the Huffman code drawn from one of 11 codebooks. The spectral coefficients within n-tuples are ordered (low to high) and the n-tuple size is two or four coefficients. The maximum absolute value of the quantized coefficients that can be represented by each Huffman codebook and the number of coefficients in each n-tuple for each codebook is shown in Table B.26. There are two codebooks for each maximum absolute value, with each representing a distinct probability distribution function. The best fit is always chosen. In order to save on codebook storage (an important consideration in a mass-produced decoder), most codebooks represent unsigned values. For these codebooks the magnitude of the coefficients is Huffman coded and the sign bit of each non-zero coefficient is appended to the codeword.

**Table 4.B.26 – Huffman Codebooks**

| Codebook index | n-Tuple size | Maximum absolute value | Signed values |
|----------------|--------------|------------------------|---------------|
| 0              |              | 0                      |               |
| 1              | 4            | 1                      | yes           |
| 2              | 4            | 1                      | yes           |
| 3              | 4            | 2                      | no            |
| 4              | 4            | 2                      | no            |
| 5              | 2            | 4                      | yes           |
| 6              | 2            | 4                      | yes           |
| 7              | 2            | 7                      | no            |
| 8              | 2            | 7                      | no            |
| 9              | 2            | 12                     | no            |
| 10             | 2            | 12                     | no            |
| 11             | 2            | 16 (ESC)               | no            |

Two codebooks require special note: codebook 0 and codebook 11. As mentioned previously, codebook 0 indicates that all coefficients within a section are zero. Codebook 11 can represent quantized coefficients that have an absolute value greater than or equal to 16. If the magnitude of one or both coefficients is greater than or equal to 16, a special *escape coding* mechanism is used to represent those values. The magnitude of the coefficients is limited to no greater than 16 and the corresponding 2-tuple is Huffman coded. The sign bits, as needed, are appended to the codeword. For each coefficient magnitude greater or equal to 16, an *escape sequence* is also appended, as follows:

escape sequence = <escape\_prefix><escape\_separator><escape\_word>

where

<escape\_prefix> is a sequence of N binary “1’s”

<escape\_separator> is a binary “0”

<escape\_word> is an N+4 bit unsigned integer, msb first

and N is a count that is just large enough so that the magnitude of the quantized coefficient is equal to

$2^{(N+4)} + \text{<escape\_word>}$

#### 4.B.12 Perceptual Noise Substitution (PNS)

The encoding procedure for noise substitution is similar to the encoding procedure for intensity stereo and is performed as follows:

- For each scalefactor band containing spectral coefficients above a lower border frequency (e.g. 4 kHz) a noise detection is carried out. The scalefactor band is classified as noise-like if the corresponding signal is neither tonal nor contains strong changes in energy over time. The tonality of the signal can be estimated by using the tonality values calculated in the psychoacoustic model. Similarly, changes in signal energy can be evaluated using the FFT energies calculated in the psychoacoustic model.
- From the detection procedure, a map, `noise_flag[sfb]`, is constructed such that noise-like scalefactor bands are flagged with a non-zero value.
- For each flagged scalefactor band the energy (sum of squares) of the corresponding spectral coefficients is calculated and mapped to a logarithmic representation with a resolution of 1.5 dB. An offset (`NOISE_OFFSET=90`) is added to the logarithmic noise energy values.
- For each flagged scalefactor band, the corresponding spectral coefficients are set to zero before quantization of the coefficients is carried out as usual.
- During the noiseless coding procedure, the pseudo-codebook `NOISE_HCB` is set for all flagged scalefactor bands. Apart from this, the regular section / noiseless coding procedure is carried out on the quantized coefficient data.
- The logarithmic noise energy values are coded analogous to the regular scalefactors, i.e. with a differential encoding scheme starting with the „global\_gain value“. They are transmitted in place of the scalefactors belonging to the flagged scalefactor bands.

#### 4.B.13 Random access points for GA coded bit streams (Audio Object Types 0x1 to 0x7)

If neither the predictor nor the Long Term Predictor (LTP) is used in a bit stream the encoder should mark every frame as random accessible frame (on more details about random access see ISO/IEC 14496-1).

If the MPEG2 style AAC predictor is used in a bit stream, none of the bit stream frames is randomly accessible. But nevertheless a decoder is allowed to tune into a bit stream. For the MPEG2 style AAC predictor it will take at maximum 240 frames until the full quality of the audio quality can be guaranteed (until each predictor reset group gets at least one reset).

If the MPEG2 style AAC predictor is used in a bit stream, a frame is random accessible if the in non of the Predictor Reset Groups, the predictors are not used in the following frames until the corresponding Predictor Reset Group gets at least one reset.

Starting the decoder at a frame where LTP is on (and the previous frames are not available) can in some cases cause slight distortion. If evenly distributed entry points for random access without any possibility for distortion are desired, the LTP can just be switched off in the encoder for (at least) two frames every  $N^{\text{th}}$  frame. The Random Access Point (RAP) is then the first frame where LTP is off. This reset has no effect on the prediction gain in the next frames (no adaptation time). Therefore the number  $N$  can be small without significant effect on the average prediction gain. The same reset procedure can be used to stop propagation of frame loss errors when the LTP gain is very high. With very small values ( $<10$ ) of  $N$  (number of frames between resets) a trade-off between average prediction gain and frame loss resilience can be made.

More complicated reset (switch-off) processes can be run in the encoder to optimise the bitstream for specific applications. For example the LTP flags for individual scalefactor bands can be used instead of the global LTP flag. In any case this is an encoder optimisation issue (and possibly error concealment issue in the decoder), and does not require any additional side information to be sent in the bitstream.

It should be noted that no predictor reset is needed with LTP for stability reasons.

Regarding error resilience, it should be noted that LTP and the signal buffer it requires can be used as an efficient tool for error concealment. Actual concealment methods are outside the scope of this standard.

### 4.B.14 Scalable AAC with core coder

#### Mono or Stereo Encoder

The description provided below describes one possible way of achieving bit rate scalability. Bit-rate functionality can also be achieved using individual coding schemes as well. Here, a particular configuration is described where the GA modules and a core coder is used. Since it is based on the calculation of a difference signal, the core coder must encode the waveform of the input signal. The MPEG-4 CELP coder is the coder to be used with this tool in MPEG-4 Audio. Furthermore, the tool has been successfully tested with the ITU-T G.729 and G.723.1 codec and with the FS1016 CELP coder. However none of these coders is available in a MPEG-4 system.

In addition to the core coder there is at least one enhancement layer based on the MPEG-4 Audio GA coding modules. In order to allow for integer frame lengths, depending on the core coder, alternative frame lengths for the AAC module may be used. The GA modules provide, in addition to the standard 1024 length, also a 960 samples per frame implementation, which at 48 kHz sampling rate leads to 20 ms and at 32 kHz to a frame length of 30 ms. This allows for an easy integration of the MPEG-4 CELP core, and to construct bitstream frames which integrate standard speech coders, like G.729, which have a frame length of a multiple of 10 ms.

The bitrate of the additional layers can be any bit rate defined for the GA based enhancement stages. The ratio of the sampling rate of the core coder and the sampling rate of the enhancement coder must be an integer.

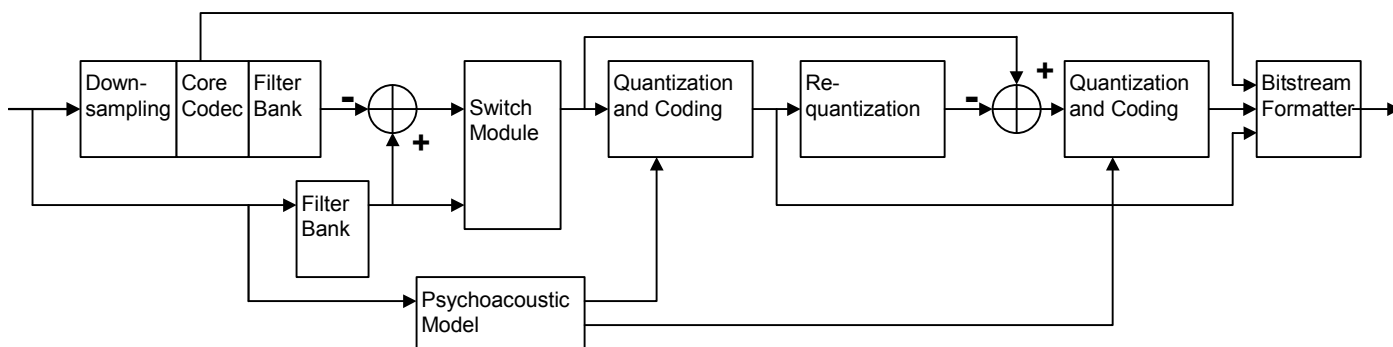


Figure 4.B.12 – Encoder process

#### Mixed Mono/Stereo Coder

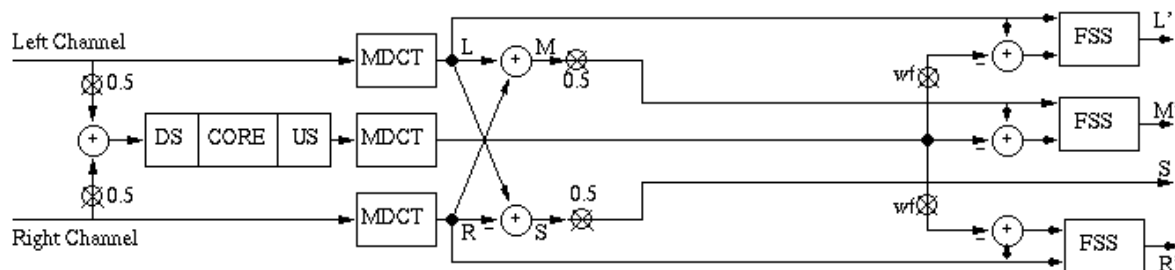


Figure 4.B.13 – Core + stereo GA

From the stereo input signal a mono signal is derived, which is then coded/decoded with a core coder, which - in general - operates at a lower sampling rate, up-sampled and transformed into the frequency-domain. This part is

identical to the corresponding part of the mono scalable coder. Next the standard M/S-Stereo matrix, as defined for the AAC coder is calculated on the spectra of the Left (L) and Right (R) signal, giving the Mid (M) and Side (S) signals. Three Frequency-Selective Switch modules (FSS) - identical to the switch modules of the mono scalable coder - then are used to generate the signals L', R' and M' from the L, R, and M signals. These signals and the original S signal then are used as the input signal to the standard AAC joint stereo coding modules. Both, M/S- and Intensity stereo coding is possible. The standard AAC `ms_mask_present` and `ms_used[]` flags are used to indicate M/S or L/R coding. However, since either M/S or L/R coding is used, it is not necessary to transmit all three FSS-related side-information. Instead, if M/S coding is selected for a band, only the FSS information for the M-switch needs to be transmitted in `diff_control[0][win][sfb]`. `diff_control[1][win][sfb]` is not transmitted in this case. If L/R coding is selected the FSS-side-information for the L and R channel needs to be transmitted in `diff_control[0][win][sfb]` for the left channel and in `diff_control[1][win][sfb]` for the right channel.

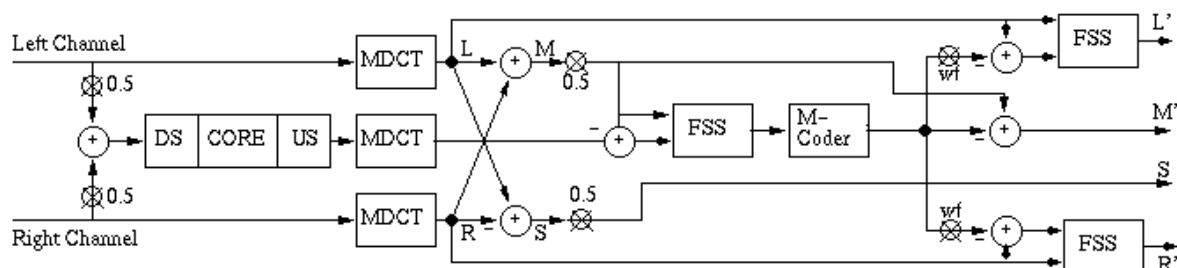


Figure 4.B.14 – Core + mono GA + stereo GA

This is an extended version of the Core + stereo GA coder. An additional GA - Coder is used to encode the output signal of the core-coder in the frequency domain. This makes the core plus M - GA coder combination identical to the scalable mono core plus GA coder. Two more FSS modules allow to select L'/R' to be identical to L, or R, respectively, or  $L - M \cdot wf$  or  $R - M \cdot wf$ . Again some of the switching conditions need not to be transmitted: If M/S coding is selected, only the FSS information for the M-switch needs to be transmitted in `diff_control[0][win][sfb]`. `Diff_control[1][win][sfb]` is not transmitted. If L/R coding is selected the FSS-side-information for the L and R and the M channel needs to be transmitted in `diff_control[0][win][sfb]` for the left channel and in `diff_control[1][win][sfb]` for the right channel, and `diff_control_m[win][0]` for the M channel.

#### 4.B.15 Scaleable controller

This tool controls the spectrum normalization and interleaved vector quantization tools to construct a scalable coder. In each scalable layer, spectrum normalization and interleaved vector quantization tool is cascaded. Each scalable layer has active frequency band which is shifted according to signal characteristics.

#### 4.B.16 Features of AAC dynamic range control

In order to handle source material with variable peak levels, mean levels and dynamic range in a manner that minimizes the variability for the consumer, it is necessary to control the reproduced level such that, for instance, dialogue level or mean music level is set to a consumer controlled level at reproduction, regardless of how the programme was originated. Additionally, not all consumers will be able to audition the programmes in a good (i.e. low noise) environment, with no constraint on how loud they make the sound. The car environment, for instance, has a high ambient noise level and it can therefore be expected that the listener will want to reduce the range of levels that would otherwise be reproduced.

For both of these reasons, dynamic range control has to be available within the specification of AAC. To achieve this, it is necessary to accompany the bit-rate reduced audio with data used to set and control the dynamic range of



the programme items. This control has to be specified relative to a reference level and in relationship to the important programme elements, e.g. the dialogue.

The features of the dynamic range control are as follows:

1. Dynamic Range Control is entirely optional. Therefore, with correct syntax, there is no change in complexity for those not wishing to invoke DRC.
2. The bit-rate reduced audio data is transmitted with the full dynamic range of the source material, with supporting data to assist in dynamic range control.
3. The dynamic range control data can be sent every frame to reduce to a minimum the latency in setting replay gains.
4. The dynamic range control data is sent using the 'fill\_element' feature of AAC.
5. The *Reference Level* is defined as Full-scale.
6. The *Programme Reference Level* is transmitted to permit level parity between the replay levels of different sources and to provide a reference about which the dynamic range control may be applied. It is that feature of the source signal that is most relevant to the subjective impression of the loudness of a programme, such as the level of the dialogue content of a programme or the average level of a music programme.
7. The *Programme Reference Level* represents that level of programme that may be reproduced at a set level relative to the *Reference Level* in the consumer hardware to achieve replay level parity. Relative to this, the quieter portions of the programme may be increased in level and the louder portions of the programme may be reduced in level.
8. *Programme Reference Level* is specified within the range 0 to -31.75 dB relative to *Reference Level*.
9. *Programme Reference Level* uses a 7 bit field with 0.25 dB steps.
10. The dynamic range control is specified within the range  $\pm 31.75$  dB.
11. The dynamic range control uses an 8 bit field (1 sign, 7 magnitude) with 0.25 dB steps.
12. The dynamic range control can be applied to all of an audio channel's spectral coefficients/frequency bands as a single entity or the coefficients can be split into with different scalefactor bands, each being controlled separately by separate sets of dynamic range control data.
13. The dynamic range control can be applied to all channels (of a stereo or multichannel bitstream) as a single entity or can be split, with sets of channels/channels being controlled separately by separate sets of dynamic range control data.
14. If an expected set of dynamic range control data is missing, the last received valid values should be used.
15. Not all elements of the dynamic range control data are sent every time. For instance, *Programme Reference Level* may only be sent on average once every 200 ms.
16. Where necessary, error detection/protection is provided by the Transport Layer.
17. The user shall be given the means to alter the amount of dynamic range control, present in the bitstream, that is applied to the level of the signal.

## 4.B.17 Fine grain scalability: BSAC (Bit-Sliced Arithmetic Coding)

### 4.B.17.1 Introduction

In the BSAC encoder the inputs to the noiseless coding module are the set of 1024 quantized spectral coefficients and the scalefactor of the scalefactor band. Since the noiseless coding is done inside the quantizer inner loop, it is part of an iterative process that converges when the total bit count (of which the noiseless coding is the vast majority) is within some interval surrounding the allocated bit count. This subclause will describe the encoding process for a single call to the noiseless coding module.

Noiseless coding is done via the following steps:

- Bit Slicing of the quantized spectral coefficient
- Preliminary Arithmetic coding of the set of quantized spectrum within a coding band using the probability table
- Preliminary Arithmetic coding of the scalefactors, stereo information, arithmetic model information.
- Probability table determination to achieve lowest bit count

#### 4.B.17.2 Bit Slicing of the quantized spectral coefficients

As a first step of BSAC encoding process, a sequence of the absolute values of quantized spectral coefficients is mapped into a bit-sliced sequence as shown in the following Figure 4.B.15.

| MSB       |             |     | LSB       |  |                                    |
|-----------|-------------|-----|-----------|--|------------------------------------|
| $B_{0,m}$ | $B_{0,m-1}$ | ... | $B_{0,0}$ |  | $0^{\text{th}}$ quantized spectral |
| $B_{1,m}$ | $B_{1,m-1}$ | ... | $B_{1,0}$ |  |                                    |
| $B_{2,m}$ | $B_{2,m-1}$ | ... | $B_{2,0}$ |  |                                    |
| ...       | ...         | ... | ...       |  |                                    |
| $B_{k,m}$ | $B_{k,m-1}$ | ... | $B_{k,0}$ |  | $k^{\text{th}}$ quantized spectral |

where, MSB plane is  $(m+1)$  bit and  $B_{k,m}$  indicates the binary value of the  $m^{\text{th}}$  bit-slice of  $k^{\text{th}}$  quantized spectral coefficients

Figure 4.B.15 – Bit slicing of the quantized spectral coefficients

For example, consider a sequence of the absolute values,  $x[n]$  as follows :

$$x[0] = 9, x[1]=0, x[2]=7 \text{ and } x[3]=11 \dots$$

If MSB plane is 5, bit-slices are formed from a quantized sequence as shown in Figure 4.B.16 :

|             | MSB |   |   |   | LSB |  |
|-------------|-----|---|---|---|-----|--|
| $x[0] : 09$ | 0   | 1 | 0 | 0 | 1   |  |
| $x[1] : 00$ | 0   | 0 | 0 | 0 | 0   |  |
| $x[2] : 07$ | 0   | 0 | 1 | 1 | 1   |  |
| $x[3] : 11$ | 0   | 1 | 0 | 1 | 1   |  |
|             | ↓   | ↓ | ↓ | ↓ | ↓   |  |

Figure 4.B.16 – Example of Bit-Slicing

#### 4.B.17.3 Probability Table Determinations

Several probability tables are provided to cover the different statistics of the bit-slices. One probability table is used for encoding the bit-sliced data of each coding band. The noiseless coding segments the set of 1024 quantized spectral coefficients into *coding bands*, such that a single probability table is used to code each coding band (the method of Arithmetic coding is explained in a later subclause). For reasons of coding efficiency, the quantized spectral coefficients are divided into *coding bands* which contain 32 quantized spectral coefficients for the noiseless coding. *Coding bands* are the basic units used for the noiseless coding for BSAC.

The bit-sliced data is decoded with the probability value which is selected among the values of the BSAC probability table.

The probability value should be defined in order to arithmetic-code the symbols (the sliced bits). Binary probability table is made up of probability values ( $p_0$ ) of the symbol '0'. First of all, sub-table is selected according to the significance and the coded higher bits of the quantized spectra. The offset for the probability ( $p_0$ ) can be decided using the sliced bits of successive non-overlapping 4 spectral data in order to select one of the several probability

values in the sub-table. However if the available codeword size is smaller than 14, there is a constraints on the selected probability value.

Probability table index is determined among the possible tables, such that the number of bits needed to represent the full set of the bit-sliced data of quantized spectral coefficients within each coding band is minimized. The possible arithmetic models have the number of the allocated bit larger than or equal to that of the bit needed to represent the PCM data of quantized spectral coefficients within a coding band.

Coding bands often contain only coefficients whose value is zero. For example, if the audio input is band limited to 20 kHz or lower, then the highest coefficients are zero. Such coding bands are coded with probability table 0, where the allocated bit is 0 and all coefficients are zero.

In order to transmit the probability table information used in encoding process, it is included the coding band side information (*cband\_si*) and coded in the syntax of *layer\_cband\_si()*. The probability table index for encoding the bit-sliced data within each coding band is transmitted starting from the lowest frequency coding band and progressing to the highest frequency coding band. For all arithmetic model indexes the value is arithmetic-coded. After the model index is encoded, the encoding of the bit-sliced data shall be started.

#### 4.B.17.4 Grouping and interleaving

If the window sequence is eight short windows then the set of 1024 coefficients is actually a matrix of 8 by 128 frequency coefficients representing the time-frequency evolution of the signal over the duration of the eight short windows. Although the sectioning mechanism is flexible enough to efficiently represent the 8 zero sections, *grouping* and *interleaving* provide for greater coding efficiency. As explained earlier, the coefficients associated with contiguous short windows can be grouped such that they share scalefactors amongst all scalefactor bands within the group. In addition, the coefficients within a group are interleaved by interchanging the order of scalefactor bands and windows. To be specific, assume that before interleaving the set of 1024 coefficients *c* are indexed as

$$c[g][w][k/4][k\%4]$$

where

*g* is the index on groups,

*w* is the index on windows within a group,

*k* is the index on coefficients within a window,

and the right-most index varies most rapidly.

After interleaving the coefficients are indexed as

$$c[g][k/4][w][k\%4]$$

This has the advantage of combining all zero sections due to band-limiting within each group.

#### 4.B.17.5 Scalefactors

The coded spectrum uses one quantizer per scalefactor band. The step size of each of these quantizers is specified as a set of scalefactors and a maximum scalefactor that normalizes these scalefactors. In order to increase compression, scalefactors associated with scalefactor bands that have only zero-valued coefficients are ignored in the coding process and therefore do not have to be transmitted. Both the maximum scalefactor and scalefactors are quantized in 1.5 dB steps.

The BSAC scalable coding scheme includes the noiseless coding to reduce the redundancy of the scalefactors. The maximum scalefactor is coded as an 8 bit unsigned integer. The first scalefactor associated with the quantized spectrum is differentially coded relative to the maximum scalefactor and the arithmetic coded using the differential scalefactor arithmetic model. The remaining scalefactors are differentially coded relative to the previously encoded scalefactor and then Arithmetic coded using the differential scalefactor model. The dynamic range of the maximum scalefactor is sufficient to represent full-scale values from a 24-bit PCM audio source.

#### 4.B.17.6 Arithmetic coding

BSAC uses the bit-slicing scheme of the quantized spectral coefficients in order to provide the fine grain scalability. And it encode the bit-sliced data using binary arithmetic coding scheme in order to reduce the average bits transmitted while suffering no loss of fidelity.

In BSAC scalable coding scheme, a quantized sequence is divided into coding bands. And, a quantized sequence is mapped into a bit-sliced sequence within a coding band. The noiseless coding of the sliced bits relies on probability table of the coding band, the significance and the other contexts.

The significance of the bit-sliced data is the position of the sliced bit to be coded.

The flags, *sign\_is\_coded[]* are updated with coding the vectors from MSB to LSB. They are initialized to 0. And they are set to 1 when the sign of the quantized spectrum is coded.

The probability table for encoding the bit-sliced data within each coding band is included in the bistream element **cband\_si\_type** and transmitted starting from the lowest coding band and progressing to the highest coding band allocated to each layer.

The length of the available bitstream (*available\_len[]*) is initialized at the beginning of each layer. The estimated length of the codeword (*est\_cw\_len*) to be decoded is calculated from the arithmetic decoding process. After the arithmetic encoding of a symbol, the length of the available bitstream should be updated by subtracting the estimated codeword length from it. We can detect whether the remaining bitstream of each layer is available or not by checking the *available\_len*.

The sign bits associated with non-zero coefficients follow the arithmetic codeword when the bit-value of the quantized spectral coefficient is 1 for the first time, with a 1 indicating a negative coefficient and a 0 indicating a positive one. The flag, *sign\_is\_coded[]* represents whether the sign bit of the quantized spectrum has been decoded or not. When the flag, *sign\_is\_coded* is 0 and the bit value of quantized spectral coefficient is nonzero, the sign bit of a sample is binary arithmetic encoded. The flag, *sign\_is\_coded* is set to 1 after the sign bit is encoded.

##### 4.B.17.6.1 Arithmetic Coding Procedure

Arithmetic Coding consists of the following 3 steps :

- Initialization which is performed prior to the coding of the first symbol
- Coding of the symbol themselves.
- Termination which is performed after the decoding of the last symbol

##### 4.B.17.6.1.1 Registers, symbols and constants

Several registers, symbols and constants are defined to describe the arithmetic encoder.

- *coding\_mode* : 1-bit fixed point register which represents the 1-bit scale-up is used or not.
- *half*: 32-bit fixed point constant equal to 1/2(0x20000000)
- *qtr*: 32-bit fixed point constant equal to 1/4(0x10000000)
- *qtr3*: 32-bit fixed point constant equal to 3/4(0x30000000)
- *low*: 32-bit fixed point register. Contains the lower bound of the interval
- *range*: 32-bit fixed point register. Contains the range of the interval.
- *p0*: 16-bit fixed point register. Probability of the "0" symbol.
- *p1*: 16-bit fixed point register. Probability of the "1" symbol.
- *cum\_freq* : 16-bit fixed point registers. Cummulative Probabilities of the symbols.

##### 4.B.17.6.1.2 Initialization

The lower bound *low* is set to 0, the range *range* to  $half * 2$  (0x40000000).

#### 4.B.17.6.1.3 Encoding a symbol

Arithmetic encoding procedure varies on the symbol to be encode. If the symbol is the sliced bit of the spectral data, the binary arithmetic encoding is used. Otherwise, the general arithmetic encoding is used.

When a symbol is binary arithmetic-encoded, the probability  $p_0$  of the "0" symbol is provided according to the context computed properly and using the probability table.  $p_0$  uses a 6-bit fixed-point number representation. Since the decoder is binary, the probability of the "1" symbol is defined to be 1 minus the probability of the "0" symbol, i.e.  $p_1 = 1-p_0$ .

When a symbol is encoded using arithmetic encoding, the cumulative probability values of multiple symbols are provided. The probability values are regarded as the arithmetic model. The arithmetic model for encoding a symbol is transmitted in the bitstream elements. For example, arithmetic models of scalefactor and `cband_si` are transmitted in the bitstream elements, **base\_scf\_model**, **enh\_scf\_model** and **cband\_si\_type**. Each value of the arithmetic model uses a 14-bit fixed-point representation.

#### 4.B.17.6.1.4 Termination

After the last symbol has been coded in the arithmetic encoder, additional bits need to be "introduced" to guarantee decodability.

#### 4.B.17.7 Stereo-related data and PNS data

The BSAC scalable coding scheme includes the noiseless coding which is different from MPEG-4 AAC coding and further reduce the redundancy of the stereo-related data. Encoding of the stereo-related data and Perceptual Noise Substitution(pns) data is depended on `pns_data_present` and **stereo\_info** which indicates the stereo mask. Since the decoded data is the same value with MPEG-4 AAC, the MPEG-4 AAC stereo-related data and pns processing follows the decoding of the stereo-related data and pns data. The detailed encoding process of stereo-related data and pns data is classified as follows:

- 1 channel, no pns data: If the number of channel is 1 and pns data is not present, we do not need bit-stream elements related to stereo or pns.
- 1 channel, pns data: If the number of channel is 1 and pns data is present, noise flag of the scalefactor bands between **pns\_start\_sfb** to **max\_sfb** is arithmetic encoded. Perceptual noise substitution is done according to the noise flag.
- 2 channel, `ms_mask_present=0` (Independent), No pns data: If `ms_mask_present` is 0 and pns data is not present, arithmetic decoding of `stereo_info` or `ms_used` is not needed.
- 2 channel, `ms_mask_present=0` (Independent), pns data: If `ms_mask_present` is 0 and pns data is present, noise flag for pns is arithmetic encoded. Perceptual noise substitution of independent mode is done according to the noise flag.
- 2 channel, `ms_mask_present=2` (all `ms_used`), pns data or no pns data: All `ms_used` values are ones in this case. So, M/S stereo processing of AAC is done at all scalefactor band. And naturally there can be no pns processing regardless of `pns_data_present` flag.
- 2 channel, `ms_mask_present=1` (optional `ms_used`), pns data or no pns data: One bit mask of `ms_used` per band for `max_sfb` bands is conveyed in this case. `ms_used` is arithmetic encoded using the `ms_used` model. M/S stereo processing of AAC is done or not according to the decoded `ms_used`. And there is no pns processing regardless of `pns_data_present` flag
- 2 channel, `ms_mask_present=3` (optional `ms_used/intensity/pns`), no pns data: At first, `stereo_info` is arithmetic encoded using the `stereo_info` model. `stereo_info` is is two-bit flag per scalefactor band indicating that M/S coding or Intensity coding is being used in window group `g` and scalefactor band `sfb` as follows :
  - 00 independent
  - 01 `ms_used`
  - 10 `intensity_in_phase`
  - 11 `intensity_out_of_phase`
- If `ms_mask_present` is not 0, M/S stereo or intensity stereo of AAC is done with these data. Since pns data is not present, we do not have to process pns.

2 channel, `ms_mask_present=3` (optional `ms_used/intensity/pns`), pns data: `stereo_info` is arithmetic encoded using the `stereo_info` model. If `stereo_info` is 1 or 2, M/S stereo or intensity stereo processing of AAC is done with these data and there is no pns processing. If `stereo_info` is 3 and scalefactor band is larger than or equal to

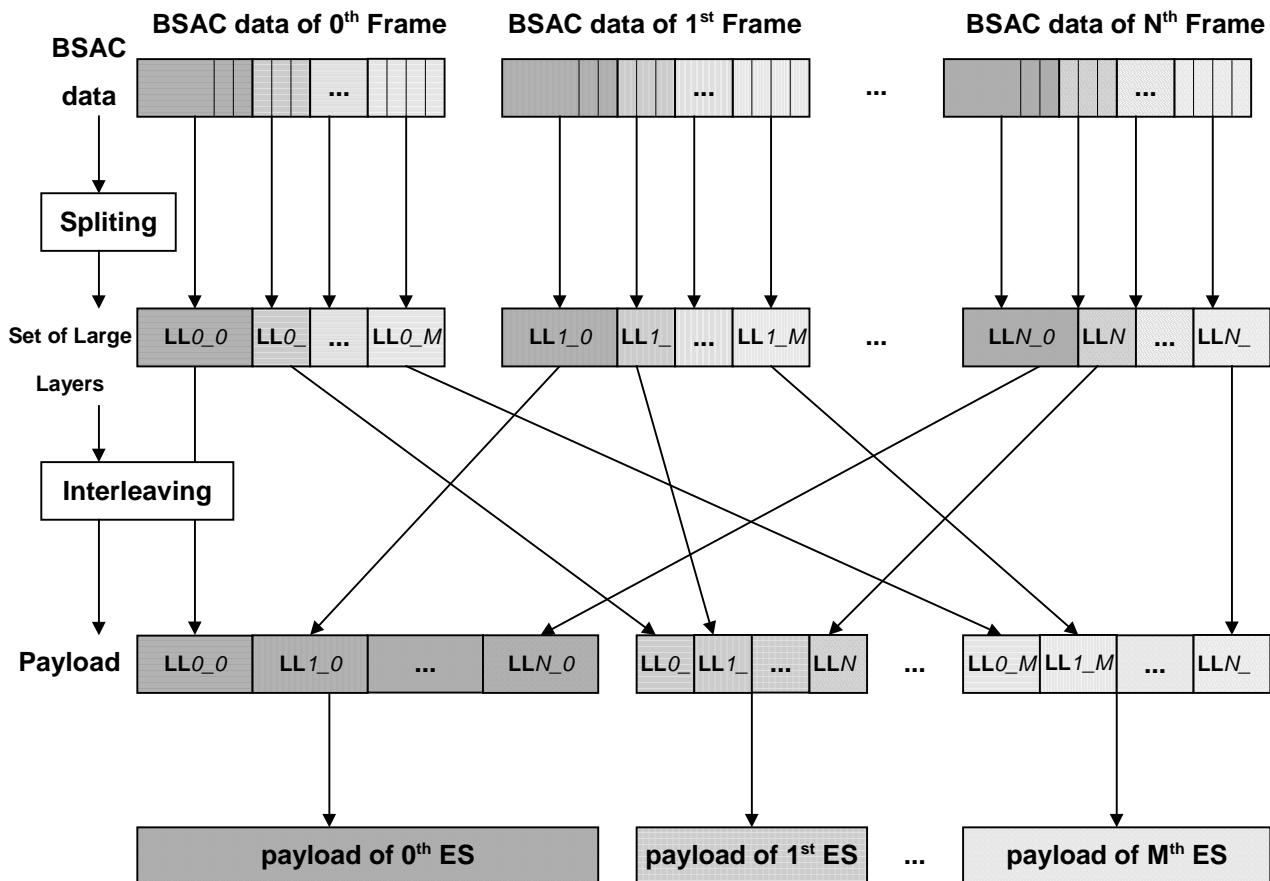
pns\_start\_sfb, noise flag for pns is arithmetic encoded. And then if the both noise flags of two channel are 1, noise substitution mode is arithmetic encoded. Otherwise, the perceptual noise is substituted only if noise flag is 1.

#### 4.B.17.8 Payload transmitted over Elementary Stream

Fine grain scalability would create large overhead if one would try to transmit fine step scalability over multiple elementary streams. So, the server can organize the BSAC data into the payloads in order to reduce overhead and implement the fine step scalability efficiently in MPEG-4 system. The configuration of the payload can be changed according to the environment such as the user interaction or the network traffic. The organization scheme is as follows:

- BSAC data of a frame is split into the large-step layers for transmission. The number of the layer and the length of each layer depend on the application that the content creator is willing to provide to the users. The length of the large-step layer, *layer\_length[]* can be calculated according to the bitrate of the layer to be transmitted, the sampling frequency and the frame length in sample. The length of the large-step layer is transmitted to the receiver through the syntax element **layer\_length** in GA Specific Configuration. In the course of the transmission, this value can be changed at a request of `bsac_backchan_stream()`. And if the user request the configuration of the payload through the back-channel, these values can be changed according to the syntax elements **numOfLayer** and **Avg\_bitrate** of `bsac_backchan_stream()`.
- BSAC data of several frames are grouped and transmitted at a time. The layers of the sub-frames are interleaved into the payload. The number of the grouped frame depends on the available delay of the application that will be transmitted to the users. The content provider should initialize this number to the proper value. And if the user request the configuration of the payload through the backchannel, this value can be changed. This number is transmitted to the receiver in the value of the syntax element **noOfSubFrame** in GA Specific Configuration. In the course of the transmission, this value can be changed at a request of `bsac_backchan_stream()`.

In the server, the payload is formed as shown in the following Figure 4.B.17.



where,  $LL_{i_k}$  is the k-th large-step layer of the i-th sub-frame  
 (M+1) is the number of the large-step layer to be transmitted (numOfLayer)  
 (N+1) is the number of the sub-frame to be grouped in an AU (numOfSubFrame)

Figure 4.B.17 – The manufacture process of the BSAC payload

1. The BSAC data of i-th sub-frame is split into several large layers for transmission over ES as in .

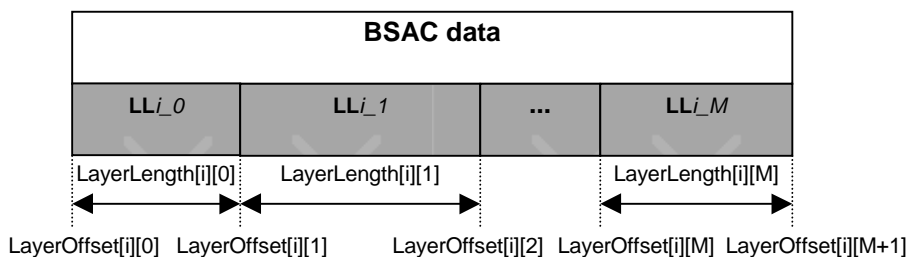


Figure 4.B.18 – The structure of the BSAC data

2. Large layers of M-subsequent frames are interleaved and concatenated to the payloads.

Some help variables and arrays are needed to describe the generation process of the payload transmitted over ES. These help variables depend on *layer*, *numOfLayer*, *numOfSubFrame*, **layer\_length** and **frame\_length** and must be built up for mapping *bsac\_raw\_data\_block()* of each sub-frame into the payloads. The pseudo code shown below describes

- how to calculate *LayerLength[i][k]*, the length of the large-step layer which is located on the fine granule audio data, *bsac\_raw\_data\_block()* of *i*<sup>th</sup> sub-frame.
- how to calculate *LayerOffset[i][k]* which indicates the start position of the large-step layer of *i*<sup>th</sup> frame which is located on the payload of the *k*<sup>th</sup> ES ( *bsac\_payload()* )
- how to calculate *LayerStartByte[i][k]* which indicates the start position of the large-step layer which is located on the fine granule audio data, *bsac\_raw\_data\_block()* of *i*<sup>th</sup> sub-frame

```

for (k = 0; k < numOfLayer; k++) {
  LayerStartByte[0][k] = 0;
  for (i = 0; i < numOfSubFrame; i++) {
    if (k == (numOfLayer-1) ) {
      LayerEndByte[i][k] = frame_length[i];
    } else {
      LayerEndByte[i][k]=LayerStartByte[i][k] + layer_length[k];
      if (frame_length[i] < LayerEndByte[i][k])
        LayerEndByte [i][k] = frame_length[i];
    }
    LayerStartByte[i+1][k] = LayerEndByte[i][k];
    LayerLength[i][k] = LayerEndByte[i][k] - LayerStartByte[i][k];
  }
}
for (k = 0; k < numOfLayer; k++) {
  LayerOffset[0][k] = 0;
  for (i = 0; i < numOfSubFrame; i++) {
    LayerOffset[i+1][k] = LayerOffset[i][k] + LayerLength[i][k]
  }
}

```

Where, *frame\_length[i]* is the length of *i*<sup>th</sup> frame bitstream which is obtained from the syntax element **frame\_length** and *layer\_length[i]* is the average length of the large-step layers in the payload of *i*<sup>th</sup> layer ES.



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## Subpart 5: Structured Audio (SA)

### 5.1 Scope

#### 5.1.1 Overview of subpart

##### 5.1.1.1 Purpose

The Structured Audio toolset enables the transmission and decoding of synthetic sound effects and music by standardising several different components. Using Structured Audio, high-quality sound can be created at extremely low bandwidth. Typical synthetic music may be coded in this format at bitrates ranging from 0 kbps (no continuous cost) to 2 or 3 kbps for extremely subtle coding of expressive performance using multiple instruments.

MPEG-4 does not standardise a particular set of synthesis methods, but a method for describing synthesis methods. Any current or future sound-synthesis method may be described in the MPEG-4 Structured Audio format.

##### 5.1.1.2 Introduction to major elements

There are five major elements to the Structured Audio toolset:

1. The Structured Audio Orchestra Language, or SAOL. SAOL is a digital-signal processing language which allows for the description of arbitrary synthesis and control algorithms as part of the content bitstream. The syntax and semantics of SAOL are standardised here in a normative fashion.
2. The Structured Audio Score Language, or SASL. SASL is a simple score and control language which is used in certain object types (see clause 5.6) to describe the manner in which sound-generation algorithms described in SAOL are used to produce sound.
3. The Structured Audio Sample Bank Format, or SASBF. The Sample Bank format allows for the transmission of banks of audio samples to be used in wavetable synthesis and the description of simple processing algorithms to use with them.
4. A normative scheduler description. The scheduler is the supervisory run-time element of the Structured Audio decoding process. It maps structural sound control, specified in SASL or MIDI, to real-time events dispatched using the normative sound-generation algorithms.
5. Normative reference to the MIDI standards, standardised externally by the MIDI Manufacturers Association. MIDI is an alternate means of structural control which can be used in conjunction with or instead of SASL. Although less powerful and flexible than SASL, MIDI support in this standard provides important backward-compatibility with existing content and authoring tools. MIDI support in this standard consists of a list of recognised MIDI messages and normative semantics for each.

### 5.2 Normative references

[DLS] MIDI Manufacturers Association, *The MIDI Downloadable Sounds Specification, v. 97.1*

[DLS2] MIDI Manufacturers Association, *The MIDI Downloadable Sounds Specification, v. 98.2*

[MIDI] MIDI Manufacturers Association, *The Complete MIDI 1.0 Detailed Specification v. 96.2*

### 5.3 Definitions

**5.3.1 Absolute time:** The time at which sound corresponding to a particular event is really created; time in the real-world. Contrast score time.

**5.3.2 Actual parameter:** The expression which, upon evaluation, is passed to an opcode as a parameter value.

**5.3.3 A-cycle:** See **audio cycle**.

**5.3.4 A-rate:** See **audio rate**.

**5.3.5 asig:** The lexical tag indicating an a-rate variable.

**5.3.6 Audio cycle:** The sequence of processing which computes new values for all a-rate expressions in a particular code block.

**5.3.7 Audio rate:** The rate type associated with a variable, expression or statement which may generate new values as often as the sampling rate.

**5.3.8 Audio sample:** A short snippet or clip of digitally represented sound. Typically used in wavetable synthesis.

**5.3.9 Authoring:** In Structured Audio, the combined processes of creatively composing music and sound control scripts, creating instruments which generate and alter sound, and encoding the instruments, control scripts, and audio samples in MPEG-4 Structured Audio format.

**5.3.10 Backus-Naur Format: (BNF)** A format for describing the syntax of programming languages, used here to specify the SAOL and SASL syntax. See subclause 5.4.2.2.

**5.3.11 Bank:** A set of samples used together to define a particular sound or class of sounds with wavetable synthesis.

**5.3.12 Beat:** The unit in which score time is measured.

**5.3.13 BNF:** See Backus-Naur Format.

**5.3.14 Bus:** An area in memory which is used to pass the output of one instrument into the input of another.

**5.3.15 Context:** See state space.

**5.3.16 Control:** An instruction used to describe how to use a particular synthesis method to produce sound.

#### EXAMPLES

“Using the piano instrument, play middle C at medium volume for 2 seconds.”

“Glissando the violin instrument up to middle C.”

“Turn off the reverberation for 8 seconds.”

**5.3.17 Control cycle:** The sequence of processing which computes new values for all control-rate expressions in a particular code block.

**5.3.18 Control period:** The length of time (typically measured in audio samples) corresponding to the control rate.

**5.3.19 Control rate:** (1) The rate at which instantiation and termination of instruments, parametric control of running instrument instances, sharing of global variables, and other non-sample-by-sample computation occurs in a particular orchestra. (2) The rate type of variables, expressions, and statements that can generate new values as often as the control rate.

**5.3.20 Decoding:** The process of turning an MPEG-4 Structured Audio bitstream into sound.

**5.3.21 Duration:** The amount of time between instantiation and termination of an instrument instance.

**5.3.22 Encoding:** The process of creating a legal MPEG-4 bitstream, whether automatically, by hand, or using special authoring tools.

- 5.3.23 Envelope:** A loudness-shaping function applied to a sound, or more generally, any function controlling a parametric aspect of a sound
- 5.3.24 Event:** One control instruction.
- 5.3.25 Expression:** A mathematical or functional combination of variable values, symbolic constants, and opcode calls.
- 5.3.26 Formal parameter:** The syntactic element that gives a name to one of the parameters of an opcode.
- 5.3.27 Future wavetable:** A wavetable that is declared but not defined in the SAOL orchestra; its definition must arrive in the bitstream before it is used.
- 5.3.28 Global block:** The section of the orchestra that describes global variables, route and send statements, sequence rules, and global parameters.
- 5.3.29 Global context:** The state space used to hold values of global variables and wavetables.
- 5.3.30 Global parameters:** The sampling rate, control rate, and number of input and output channels of audio associated with a particular orchestra.
- 5.3.31 Global variable:** A variable that can be accessed and/or changed by several different instruments.
- 5.3.32 Grammar:** A set of rules that describes the set of allowable sequences of lexical elements comprising a particular language.
- 5.3.33 Guard expression:** The expression standing at the front of an if, while, or else statement that determines whether or how many times a particular block of code is executed.
- 5.3.34 I-cycle:** See initialisation cycle.
- 5.3.35 Identifier:** A sequence of characters in a textual SAOL program that denotes a symbol.
- 5.3.36 Informative:** Aspects of a standards document that are provided to assist implementers, but are not required to be implemented in order for a particular system to be compliant to the standard.
- 5.3.37 I-pass:** See initialisation pass.
- 5.3.38 I-rate:** See initialisation rate.
- 5.3.39 Initialisation cycle:** See initialisation pass.
- 5.3.40 Initialisation rate:** The rate type of variables, expressions, and statements that are set once at instrument instantiation and then do not change.
- 5.3.41 Initialisation pass:** The sequence of processing that computes new values for each i-rate expression in a particular code block.
- 5.3.42 Instance:** See instrument instantiation.
- 5.3.43 Instantiation:** The process of creating a new instrument instantiation based on an event in the score or statement in the orchestra.
- 5.3.44 Instrument:** An algorithm for parametric sound synthesis, described using SAOL. An instrument encapsulates all of the algorithms needed for one sound-generation element to be controlled with a score.

NOTE - An MPEG-4 Structured Audio instrument does not necessarily correspond to a real-world instrument. A single instrument might be used to represent an entire violin section, or an ambient sound such as the wind. On the

other hand, a single real-world instrument that produces many different timbres over its performance range might be represented using several SAOL instruments.

**5.3.45 Instrument instantiation:** The state space created as the result of executing a note-creation event with respect to a SAOL orchestra.

**5.3.46 ivar:** The lexical tag indicating an i-rate variable.

**5.3.47 K-cycle:** See control cycle.

**5.3.48 K-rate:** See control rate.

**5.3.49 ksig:** The lexical tag indicating a k-rate variable.

**5.3.50 Lexical element:** See token.

**5.3.51 Looping:** A typical method of wavetable synthesis. Loop points in an audio sample are located and the sound between those endpoints is played repeatedly while being simultaneously modified by envelopes, modulators, etc.

**5.3.52 MIDI:** The Musical Instrument Digital Interface standards, see **[MIDI]** in subclause 5.2. MIDI is one method for specifying control of synthesis in MPEG-4 Structured Audio.

**5.3.53 Natural Sound:** A sound created through recording from a real acoustic space. Contrasted with synthetic sound.

**5.3.54 Normative:** Those aspects of a standard that must be implemented in order for a particular system to be compliant to the standard.

**5.3.55 Opcode:** A parametric signal-processing function that encapsulates a certain functionality so that it may be used by several instruments.

**5.3.56 Orchestra:** The set of sound-generation and sound-processing algorithms included in an MPEG-4 bitstream. Includes instruments, opcodes, routing, and global parameters.

**5.3.57 Orchestra cycle:** A complete pass through the orchestra, during which new instrument instantiations are created, expired ones are terminated, each instance receives one k-cycle and one control period worth of a-cycles, and output is produced.

**5.3.58 Parameter fields:** The names given to the parameters to an instrument.

**5.3.59 P-fields:** See parameter fields.

**5.3.60 Production rule:** In Backus-Naur Form grammars, a rule that describes how one syntactic element may be expressed in terms of other lexical and syntactic elements.

**5.3.61 Rate-mismatch error:** The condition that results when the rate semantics rules are violated in a particular SAOL construction. A type of syntax error.

**5.3.62 Rate semantics:** The set of rules describing how rate types are assigned to variables, expressions, statements, and opcodes, and the normative restrictions that apply to a bitstream regarding combining these elements based on their rate types.

**5.3.63 Rate type:** The “speed of execution” associated with a particular variable, expression, statement, or opcode.

**5.3.64 Route statement:** A statement in the global block that describes how to place the output of a certain set of instruments onto a bus.

**5.3.65 Run-time error:** The condition that results from improper calculations or memory accesses during execution of a SAOL orchestra.

**5.3.66 SASBF:** See Sample Bank Format

**5.3.67 SAOL:** The Structured Audio Orchestra Language, pronounced like the English word “sail.” SAOL is a digital-signal processing language that allows for the description of arbitrary synthesis and control algorithms as part of the content bitstream.

**5.3.68 SAOL orchestra:** See orchestra.

**5.3.69 SASL:** The Structured Audio Score Language. SASL is a simple format that allows for powerful and flexible control of music and sound synthesis.

**5.3.70 Sample:** See Audio sample.

**5.3.71 Sample Bank Format:** A component format of MPEG-4 Structured Audio that allows the description of a set of samples for use in wavetable synthesis and processing methods to apply to them.

**5.3.72 Scheduler:** The component of MPEG-4 Structured Audio that describes the mapping from control instructions to sound synthesis using the specified synthesis techniques. The scheduler description provides normative bounds on event-dispatch times and responses.

**5.3.73 Scope :** The code within which access to a particular variable name is allowed.

**5.3.74 Score:** A description in some format of the sequence of control parameters needed to generate a desired music composition or sound scene. In MPEG-4 Structured Audio, scores are described in SASL and/or MIDI.

**5.3.75 Score time:** The time at which an event happens in the score, measured in beats. Score time is mapped to absolute time by the current tempo.

**5.3.76 Send statement:** A statement in the global block that describes how to pass a bus on to an effect instrument for post-processing.

**5.3.77 Semantics:** The rules describing what a particular instruction or bitstream element should do. Most aspects of bitstream and SAOL semantics are normative in MPEG-4.

**5.3.78 Sequence rules:** The set of rules, both default and explicit, given in the global block that define in what order to execute instrument instantiations during an orchestra cycle.

**5.3.79 Signal variable:** A unit of memory, labelled with a name, that holds intermediate processing results. Each signal variable in MPEG-4 Structured Audio is instantaneously representable by a 32-bit floating point value.

**5.3.80 Spatialisation:** The process of creating special sounds that a listener perceives as emanating from a particular direction.

**5.3.81 State space:** A set of variable-value associations that define the current computational state of an instrument instantiation or opcode call. All the “current values” of the variables in an instrument or opcode call.

**5.3.82 Statement:** “One line” of a SAOL orchestra.

**5.3.83 Structured audio:** Sound-description methods that make use of high-level models of sound generation and control. Typically involving synthesis description, structured audio techniques allow for ultra-low bitrate description of complex, high-quality sounds. See [SAUD].

**5.3.84 Symbol:** A sequence of characters in a SAOL program, or a symbol token in a MPEG-4 Structured Audio bitstream, that represents a variable name, instrument name, opcode name, table name, bus name, etc.



**5.3.85 Symbol table:** In an MPEG-4 Structured Audio bitstream, a sequence of data that allows the tokenised representation of SAOL and SASL code to be converted back to a readable textual representation. The symbol table is an optional component.

**5.3.86 Symbolic constant:** A floating-point value explicitly represented as a sequence of characters in a textual SAOL orchestra, or as a token in a bitstream.

**5.3.87 Syntax:** The rules describing what a particular instruction or bitstream element should look like. All aspects of bitstream and SAOL syntax are normative in MPEG-4.

**5.3.88 Syntax error:** The condition that results when a bitstream element does not comply with its governing rules of syntax.

**5.3.89 Synthesis:** The process of creating sound based on algorithmic descriptions.

**5.3.90 Synthetic Sound:** Sound created through synthesis.

**5.3.91 Tempo:** The scaling parameter that specifies the relationship between score time and absolute time. A tempo of 60 beats per minute means that the score time measured in beats is equivalent to the absolute time measured in seconds; higher numbers correspond to faster tempi, so that 120 beats per minute is twice as fast.

**5.3.92 Terminal:** The “client side” of an MPEG transaction; whatever hardware and software are necessary in a particular implementation to allow the capabilities described in this document.

**5.3.93 Termination:** The process of destroying an instrument instantiation when it is no longer needed.

**5.3.94 Timbre:** The combined features of a sound that allow a listener to recognise such aspects as the type of instrument, manner of performance, manner of sound generation, etc. Those aspects of sound that distinguish sounds equivalent in pitch and loudness.

**5.3.95 Token:** A lexical element of a SAOL orchestra: a keyword, punctuation mark, symbol name, or symbolic constant.

**5.3.96 Tokenisation:** The process of converting a orchestra in textual SAOL format into a bitstream representation consisting of a stream of tokens.

**5.3.97 Variable:** See signal variable.

**5.3.98 Wavetable synthesis:** A synthesis method in which sound is created by simple manipulation of audio samples, such as looping, pitch-shifting, enveloping, etc.

**5.3.99 Width:** The number of channels of data that an expression represents.

## 5.4 Symbols and abbreviations

### 5.4.1 Mathematical operations

The mathematical operators used to describe this part of ISO/IEC 14496 are similar to those used in the C programming language.

|        |                               |
|--------|-------------------------------|
| +      | addition                      |
| -      | subtraction                   |
| x or * | multiplication                |
| /      | division                      |
| exp    | exponential function (base e) |
| log    | natural logarithm             |
| log10  | base-10 logarithm             |

abs absolute value  
floor(x) greatest integer less than or equal to x  
ceil(x) least integer greater than or equal to x  
> greater than  
< less than  
>= greater than or equal to  
<= less than or equal to  
<> or != not equal to

## 5.4.2 Description methods

### 5.4.2.1 Bitstream syntax

The Structured Audio bitstream syntax is described using SDL, the MPEG-4 Syntactic Description Language. See ISO/IEC 14496-1 clause 12.

### 5.4.2.2 SAOL syntax

The textual SAOL syntax (in subclause 5.8) is described using extended Backus-Naur format (BNF) notation [see **DRAG**]. BNF is a description for context-free grammars of programming languages.

BNF grammars are composed of terminals, also called tokens, and production rules. Terminals represent syntactic elements of the language, such as keywords and punctuation; production rules describe the composition of these elements into structural groups.

Terminals will be represented in **boldface**; production rules will be represented in <angle brackets>.

The rewrite rules, which map productions into sequences of other productions and terminals, are represented with the -> symbol.

#### EXAMPLE

```
<letter>      -> a  
<letter>      -> b  
<sequence>   -> <letter>  
<sequence>   -> <letter> <sequence>
```

This grammar (starting from the *sequence* token) describes, using a recursive rewrite rule and a two-symbol alphabet, all strings containing at least one letter that are made up of 'a' and 'b' characters.

In addition, rewrite rules using optional elements will be described using the [ ] symbols. Using this notation does not increase the power of the syntax description (in terms of the languages it can represent), but makes certain constructs simpler.

#### EXAMPLE

```
<head>        -> c  
<seqhead>     -> [<head>] <sequence>
```

This grammar (starting from the *seqhead* token) describes, in addition to the set above, all strings beginning with a 'c' character and followed by a sequence of 'a's and 'b's.

The **NULL** token may be used to indicate that a sequence of no characters (the empty string) is a permissible rewrite for a particular production.

Other symbols such as the ellipsis (...) will be used occasionally when their meaning is clear from the context.

Normative aspects of the relationship between the BNF grammar, other grammar representation methods, the bitstream syntax, and the textual description format are described in subclause 5.8.1.

### 5.4.2.3 SASL Syntax

The SASL syntax is specified using extended BNF grammars, as described in subclause 5.4.2.2.

## 5.5 Bitstream syntax and semantics

### 5.5.1 Introduction to bitstream syntax

This subclause describes the bitstream format defining an MPEG-4 Structured Audio bitstream.

Each group of classes is notated with normative semantics, which define the meaning of the data represented by those classes.

### 5.5.2 Bitstream syntax

```

/*****
    symbol table definitions
*****/

class symbol {
    unsigned int(16) sym;           // no more than 65535 symbols/orch + score
}

class sym_name {
    unsigned int(4) length;        // one name in a symbol table
    unsigned int(8) name[length]; // names up to 15 chars long
}

class symtable {
    unsigned int(16) length;       // a whole symbol table
    sym_name name[length];        // no more than 65535 symbols/orch+score
}

```

A bitstream may contain a symbol table, but this is not required. The symbol table allows textual SAOL and SASL code to be recovered from the tokenised bitstream representation. The inclusion or exclusion of a symbol table does not affect the decoding process.

If a symbol table is included, then all or some of the symbols in the orchestra and score shall be associated with a textual name in the following way: each symbol (a symbol is just an integer) shall be associated with the character string paired with that symbol in a `sym_name` object. There shall be no more than one name associated with a given symbol, otherwise the bitstream is invalid. It is permissible for the symbol table to be incomplete and contain names associated with some, but not all, symbols used in the orchestra and score. The presence of a zero-length string in a symbol table entry indicates that a name for this symbol is not included in the symbol table.

SAOL and SASL implementations that require textual input, rather than tokenised input, are permissible in a compliant decoder, in which case the decoder would detokenise the bitstream before it can be processed. In such a case, any symbols without associated names are suggested to be associated with a default name of the form `_sym_x`, where `x` is the symbol value. Names of this form are reserved in SAOL for this purpose, and so following this suggestion guarantees that names will not clash with symbol-table-defined symbol names.

```

/*****
    orchestra file definitions
*****/

class orch_token {
    int done; // a token in an orchestra

    unsigned int(8) token; // see standard token table, Annex 5.A
    switch (token) {

```

```

case 0xF0 : // a symbol
  symbol sym; // the symbol name
  break;
case 0xF1 : // a constant value
  float(32) val; // the floating-point value
  break;
case 0xF2 : // a constant int value
  unsigned int(32) val; // the integer value
  break;
case 0xF3 : // a string constant
  int(8) length;
  unsigned int(8) str[length]; // strings no more than 255 chars
  break;
case 0xF4 : // a one-byte constant
  int(8) val;
  break;
case 0xFF : // end of orch
  done = 1;
  break;
}
}

class orc_file { // a whole orch file
  unsigned int(16) length;
  orch_token data[length];
}

```

An orchestra file is a string of tokens. These tokens represent syntactic elements such as reserved words, core opcode names, and punctuation marks as given in the table in Annex 5.A; in addition, there are five special tokens. Token 0xF0 is the symbol token; when it is encountered, the next 16 bits in the bitstream shall be a symbol number. Token 0xF1 is the value token; when it is encountered, the next 32 bits in the bitstream shall be a floating-point value. This token shall be used for all symbolic constants within the SAOL program except for those encountered in special integer contexts, as described in subclause 5.12. Token 0xF2 is the integer token; when it is encountered, the next 32 bits in the bitstream shall be an unsigned integer value. Token 0xF3 is the string token; when it is encountered, the next several bits in the bitstream shall represent a character string (this token is currently unused). Token 0xF4 is the byte token; when it is encountered, the next 8 bits in the bitstream shall be an unsigned integer value. Token 0xFF is the end-of-orchestra token; this token has no syntactic function in the SAOL orchestra, but signifies the end of the orchestra file section of the bitstream.

Not every sequence of tokens is permitted to occur as an orchestra file. Subclause 5.8 contains extensive syntactic rules restricting the possible sequence of tokens, described according to the textual SAOL format. Normative rules for mapping back and forth between the tokenised format and the textual format are given in subclause 5.12. The overall sequence of orchestra tokens shall correspond to an <orchestra> production as given in subclause 5.8.4.

```

/*****
  score file definitions
*****/

class instr_event { // a note-on event
  bit(1) has_label;
  if (has_label)
    symbol label;
  symbol iname_sym; // the instrument name
  float(32) dur; // note duration
  unsigned int(8) num_pf;
  float(32) pf[num_pf]; // all the pfields (no more than 255)
}

class control_event { // a control event
  bit(1) has_label;
  if (has_label)
    symbol label;
  symbol varsym; // the controller name
  float(32) value; // the new value
}

class table_event {
  symbol tname; // the name of the table
}

```

```

bit(1) destroy;      // a table destructor
if (!destroy) {
    token tgen;      // a core wavetable generator
    bit(1) refers_to_sample;
    if (refers_to_sample)
        symbol table_sym;      // the name of the sample
    unsigned int(16) num_pf;    // the number of pfields
    if (tgen==0x7D) {          // concat
        float(32) size;
        symbol ft[num_pf - 1];
    } else {
        float (32) pf[num_pf];
    }
    // when coding sample generator, leave a blank array slot
    // for "which" parameter, to maintain alignment for "skip" parameter
}
}

class end_event {
    // fixed at nothing
}

class tempo_event { // a tempo event
    float(32) tempo;
}

class score_line {
    bit(1) has_time;
    if (has_time) {
        bit (1) use_if_late;
        float(32) time;          // the event time
    }
    bit (1) high_priority;
    bit(3) type;
    switch (type) {
        case 0b000 : instr_event inst; break;
        case 0b001 : control_event control; break;
        case 0b010 : table_event table; break;
        case 0b100 : end_event end; break;
        case 0b101 : tempo_event tempo; break;
    }
}

class score_file {
    unsigned int(20) num_lines; // a whole score file
    score_line lines[num_lines];
}

```

A score file is a set of lines of score information provided in the stream information header. Thus, events that are known before the real-time bitstream transmission begins may be included in the header, so that they are available to the decoder immediately, which may aid efficient computation in certain implementations. Each line shall be one of five events. Each type of event has different implications in the decoding and scheduling process, see subclause 5.7.3. An instrument event specifies the start time, instrument name symbol, duration, and any other parameters of a note played on a SAOL instrument. A control event specifies a control parameter that is passed to a instrument or instruments already generating sound. A table event dynamically creates or destroys a global wavetable in the orchestra. An end event signifies the end of orchestra processing. A tempo event dynamically changes the tempo of orchestra playback.

A score file need not be presented in increasing order of event times; the events shall be “sorted” by the scheduler as they are processed. In the score file, every score line shall have a time stamp (**has\_time** shall be 1).

The **high\_priority** bit indicates that the score event is a high-priority event as described in subclause 5.7.3.3.7. The **use\_if\_late** bit indicates, if the **has\_time** bit is set, that the score event shall be used whether or not it arrives on time (see subclause 5.7.3.3.8).

```

/*****
MIDI definitions

```

```

*****/

class midi_event {
    unsigned int(24) length
    unsigned int(8) data[length];
}

class midi_file {
    unsigned int(32) length;
    unsigned int(8) data[length];
}

```

The MIDI chunks allow the inclusion of MIDI score information in the bitstream header and bitstream. The MIDI event class contains a single MIDI instruction as specified in [MIDI]; the MIDI file class contains an array of bytes corresponding to a Standard Format 0 or Format 1 MIDIFile as specified in [MIDI]. Note that not every sequence of data may occur in either case; the legal syntaxes of MIDI events and MIDIFiles as specified in [MIDI] place normative bounds on syntactically valid MPEG-4 Structured Audio bitstreams. Only chunks of data up to  $2^{24}-1$  and  $2^{32}-1$  bytes long, respectively, may be included; longer messages shall be broken into several bitstream elements. The semantics of MIDI data are given in subclause 5.13 (for Object type 1 and 2 implementations) and subclause 5.14 (for Object type 3 and 4 implementations).

```

/*****
    sample data
*****/

class sample {
    /* note that 'sample' can be used for any big chunk of data
       that needs to get into a wavetable */
    symbol sample_name_sym;
    unsigned int(24) length; // length in samples
    bit(1) has_srate;
    if (has_srate)
        unsigned int(17) srate; // sampling rate (needs to go to 96 KHz)
    bit(1) has_loop;
    if (has_loop) {
        unsigned int(24) loopstart; // loop points in samples
        unsigned int(24) loopend;
    }
    bit(1) has_base;
    if (has_base)
        float(32) basecps; // base freq in Hz
    bit(1) float_sample;
    if (float_sample) {
        float(32) float_sample_data[length]; // data as floats ...
    }
    else {
        int(16) sample_data[length]; // ... or as ints
    }
}

```

A sample chunk includes a block of data that will be included in a wavetable in a SAOL orchestra. Each sample consists of a name, a length, a block of data, and four optional parameters: the sampling rate, the loop start and loop end points, and the base frequency. Access to the data in the sample is provided through the **sample** core wavetable generator, see subclause 5.10.2.

The sample data may be represented either as 32-bit floating point values, in which case it shall be scaled between -1 and 1, or may be represented as 16-bit integer values, in which case it shall be scaled between -32768 and 32767. In the case that the sample data is represented as integer values, upon inclusion in a wavetable, it shall be rescaled to floating-point as described in subclause 5.10.2.

Each sample is named with a symbol. If two samples in the decoder configuration header or in a single access unit have the same name, the result is unspecified. If a sample in an access unit has the same name as a sample in a previous access unit or one in the decoder configuration header, the new sample shall replace the old sample for accesses to that name through the **sample** core wavetable generator for any table generator executed at the same time or later than the decoding time of the access unit containing the new sample. Tables that have already been generated are not affected.

```

/*****
    sample bank data
*****/

```

The sample bank chunk describes a bank of wavetable data and associated processing parameters for use with the sample bank synthesis procedure in subclause 5.13.

```

class sbf {
    int(32)          length;
    int(8)          data[length];
}

```

The data chunk is opaque with regard to transport by MPEG-4 Systems. It shall conform to the format specification given in [DLS] (see subclause 5.2) – that is, it shall be a RIFF data chunk beginning “RIFF...”.

```

/*****
    bitstream formats
*****/

```

```

class StructuredAudioSpecificConfig { // the bitstream header
    bit more_data = 1;

    while (more_data) { // shall have at least one chunk
        bit(3) chunk_type;
        switch (chunk_type) {
            case 0b000 : orc_file orc; break;
            case 0b001 : score_file score; break;
            case 0b010 : midi_file SMF; break;
            case 0b011 : sample samp; break;
            case 0b100 : sbf sample_bank; break;
            case 0b101 : symtable sym; break;
        }
        bit(1) more_data;
    }
}

```

The bitstream decoder configuration contains all the information required to configure and start up a structured audio decoder. It contains a sequence of one or more chunks, where each chunk is of one of the following types: orchestra file, score file, midi file, sample data, sample bank, or symbol table. Multiple chunks of each of these types may occur in the bitstream (except for **midi\_file**), with the following semantics:

1. **orc\_file**: The multiple orchestra files shall be merged. It is a syntax error if more than one **global** block appears in the merged orchestra (see subclause 5.8.5).
2. **score\_file**: The multiple score files shall be sorted together by event times and merged.
3. **midi\_file**: Only one **midi\_file** element may occur in a single Structured Audio bitstream.
4. **sample**: The samples may be accessed by the orchestra as described in subclause 5.10.2.
5. **sbf**: The multiple sample banks are all accessible to the synthesis process. The behaviour is undefined if a particular combination of MIDI present and MIDI bank number is used more than once, whether in a single sample bank or in multiple sample banks.
6. **symtable**: The multiple symbol tables each give names to symbols in the orchestra. The  $N_0$  names in the first symbol table in the bitstream apply to symbols  $0..N_0-1$ ; the  $N_1$  names in the second symbol table to symbols  $N_0..N_0+N_1-1$ ; and so on.

```

class SA_access_unit { // the streaming data
    bit(1) more_data = 1;

    while (more_data) {
        bit(2) event_type;
    }
}

```

```
switch (event_type) {
  case 0b00 : score_line score_ev; break;
  case 0b01 : midi_event midi_ev; break;
  case 0b10 : sample samp; break;
}
bit(1) more_data;
}
```

The Structured Audio access unit contains real-time streaming control information to be provided to a running Structured Audio decoding process. It may contain as many control instructions as desired and as permitted by the available bandwidth. It shall not contain new instrument definitions; the orchestra configuration is fixed at decoder start-up. It may contain score lines, MIDI events, and new sample data. When provided as part of an access unit, the score line is not required to contain a timestamp. When **has\_time** is cleared in the **score\_line** class, the event is dispatched immediately according to the rules in subclause 5.7.3.3.6. Score lines without timestamps are not responsive to orchestra tempo changes.

Annex 5.E discusses when the random access point flag, conveyed in the Access Unit packaging in the Systems specification, may be set.

## 5.6 Object types

There are four object types standardised for Structured Audio. Each of these object types corresponds to a particular set of application requirements. The default object type is Object type 4; when reference is made to MPEG-4 Structured Audio format without reference to a object type, it shall be understood that the reference is to Object type 4.

Terminals implementing MPEG-4 Systems profiles containing the **AudioFX** node (see ISO/IEC 14496-1, subclause 9.4.2.7) shall also provide support for Structured Audio Object type 3 or 4.

1. **MIDI only.** In this object type, only the **midi\_file** chunk shall occur in the stream information header, and only the **midi\_event** event shall occur in the bitstream data. In this object type, the General MIDI patch mappings are used, and the decoding process is described in subclause 5.14.2. This object type is used to enable backward-compatibility with existing MIDI content and rendering devices. Normative and implementation-independent sound quality cannot be produced in this object type.
2. **Wavetable synthesis.** In this object type, only the **midi\_file** and **sbf** chunks shall occur in the stream information header, and only the **midi\_event** event shall occur in the bitstream data. This object type is used to describe music and sound-effects content in situations in which the full flexibility and functionality of SAOL, including 3-D audio, is not required. In this case, the decoding process is described in subclause 5.13.3.1.
3. **Algorithmic synthesis and AudioFX.** In this object type, the **sbf** and **midi\_file** chunks shall not occur in the **stream** information header. This object type is used to describe algorithmic synthesis and to provide audio effects processing in the AudioFX node when the use of the SASBF sample bank format (subclause 5.13) is not needed.
4. **Main synthetic.** All bitstream elements and stream information elements may occur.

The decoding process for Object types 3 and 4 is described in subclause 5.7.

## 5.7 Decoding process

### 5.7.1 Introduction

This subclause describes the algorithmic structured audio decoding process, in which a bitstream conforming to Object type 3 or 4 is converted into sound. The decoding process for Object type 1 bitstreams is described in subclause 5.14.2, and the decoding process for Object type 2 bitstreams in subclause 5.13.3.1.



## 5.7.2 Decoder configuration header

At the creation of a Structured Audio Elementary Stream, a Structured Audio decoder is instantiated and a bitstream object of class **SA\_decoder\_config** provided to that decoder as configuration information. At this time, the decoder shall initialise a run-time scheduler, and then parse the decoder configuration header into its component parts and use them as follows:

- Orchestra file: The orchestra file shall be checked for syntactic conformance with the SAOL grammar and rate semantics as specified in subclause 5.8. Whatever pre-processing (i.e., compilation, allocation of static storage, etc.) needs to be done to prepare for orchestra run-time execution shall be performed.
- Score file: Each event in the score file shall be registered with the scheduler. To “register” means to inform the scheduler of the presence of a particular parameterised event at a particular future time, and the scheduler’s associated actions.
- MIDI file: Each event in the MIDI file shall be converted into an appropriate event as described in subclause 5.13, and those events registered with the scheduler.
- Sample bank: The data in the bank shall be stored, and whatever pre-processing necessary to prepare for using the bank for synthesis shall be performed.
- Sample data: The data in the sample shall be stored, and whatever pre-processing necessary to prepare the data for reference from a SAOL wavetable generator shall be performed. If the sample data is represented as 16-bit integers in the bitstream, it shall be converted to floating-point format at this time.
- Symbol table: No normative decoder behaviour is associated with the symbol table.

If there is more than one orchestra file in the stream information header, the various files are combined together via concatenation and processed as one large orchestra file. That is, each orchestra file within the bitstream refers to the same global namespace, instrument namespace, and opcode namespace.

## 5.7.3 Bitstream data and sound creation

### 5.7.3.1 Relationship with systems layer

At each time step within the systems operation, the systems layer may present the Structured Audio decoder with an Access Unit containing data conforming to the **SA\_access\_unit** class. The run-time responsibility of the Structured Audio decoder is to receive these AU data elements, to parse and understand them as the various Structured Audio bitstream data elements, to execute the on-going SAOL orchestra to produce one Composition Unit of output, and to present the systems layer with that Composition Unit.

### 5.7.3.2 Bitstream data elements

As Access Units are received from the systems demultiplexer, they are parsed and used by the Structured Audio decoder in various ways, as follows:

- Sample data shall be stored, and whatever pre-processing is necessary for reference by forthcoming score lines containing references to that sample shall be performed. If the sample data is represented as 16-bit integers in the bitstream, it shall be converted to floating-point format at this time. Any samples in an access unit shall be processed before score lines, in case the score lines reference the samples.
- Score line events shall be registered with the scheduler if they have time stamps, or executed in the next k-cycle, if not.

- MIDI events shall be converted into appropriate SAOL events (see subclause 5.14) and then registered with the scheduler, if they have time stamps, or executed in the next k-cycle, if not.

### 5.7.3.3 Scheduler semantics

#### 5.7.3.3.1 Purpose of scheduler

The scheduler is the central control mechanism of a Structured Audio decoding system. It is responsible for handling events by instantiating and terminating instruments, keeping track of what instrument instantiations are active, instructing the various instrument instantiations to perform synthesis, routing the output of instruments onto busses, and sending busses to effects instruments. Although there are many ways to perform these tasks, the exact nature of what must be done can be clearly specified. This subclause provides normative bounds on the activities of the scheduler.

#### 5.7.3.3.2 Instrument instantiation

To instantiate an instrument is to create data space for its variables and the data space required for any opcodes called by that instrument. When an instrument is instantiated, the following tasks shall be performed. First, space for any parameter fields shall be allocated and their values set according to the p-fields of the instantiating expression or event. Then, space for any locally declared variables shall be allocated and these variable values set to 0. Then, the current values of any imported i-rate variables shall be copied into the local storage space. Then, locally declared wavetables shall be created and filled with data according to their declaration and the appropriate rules in subclause 5.10.

#### 5.7.3.3.3 Instrument termination

To terminate an instrument instantiation is to destroy the data space for that instance.

#### 5.7.3.3.4 Instrument execution

To execute an instrument instantiation at a particular rate is to calculate the results of the instructions given in that instrument definition. When an instrument instance is executed at a particular rate, the following steps shall be performed. First, the values of any global variables and wavetables imported by that instrument at that rate shall be copied into the storage space of the instrument. In addition, when executing at the a-rate an instrument instance that is the target of a **send** statement, the current value of the **input** standard name in the instance shall be set to the current value of the bus or busses referenced in the **send** statement. Then, the code block for that instrument shall be executed at the particular rate with regard to the data space of the instrument instantiation, as given by the rules in subclause 5.8.6.6. Then, the values of any global variables and wavetables exported by that instrument at that rate shall be copied into the global storage space. Finally, when executing an instrument instantiation at the a-rate, the value of the instance output shall be added to the bus onto which the instrument is routed according to the rules in subclause 5.8.5.4, unless the instance is the target of a **send** expression referencing the special bus **output\_bus**, in which case the output of the instrument instance is the output of the orchestra and may be turned into sound.

#### 5.7.3.3.5 Orchestra start-up and configuration

##### 5.7.3.3.5.1 Introduction

This subclause describes the steps required to begin the decoding process. These tasks (determination of instrument and bus width, global variable allocation, **startup** execution, global wavetable creation, bus and **send** instrument initialisation) shall be performed in the order indicated.

##### 5.7.3.3.5.2 Determination of instrument output width and bus width

The output width of each instrument is determined in the order specified by the global sequencing rules (subclause 5.8.5.6); the width of each bus is determined by the sum of the output widths (subclause 5.8.6.6.8) of the instruments routed to that bus in a single **route** statement (subclause 5.8.5.4). Only for the purposes of calculating

bus widths, any instrument that does not receive any bus data according to the sequence rules shall have an **inchannels** width of 1 (this specification is needed since output widths may depend on the value of **inchannels** or the width of **input**).

#### EXAMPLE 1

Consider the following orchestra.

```
global {
  route(bus1, i1);
  route(bus2, i2, i3);
  send(i2; ; bus1);
  send(i4; ; bus2);
}

instr i1(...) {
  asig a[2];
  ...
  output(a);
}

instr i2(...) {
  asig b;
  ...
  output(input + b);
}

instr i3(...) {
  asig c[inchannels];
  ...
  output(input,c);
}

instr i4(...) {
  asig d[inchannels];
  ...
  output(d + input);
}
```

In this orchestra, the global sequencing rules (subclause 5.8.5.6) specify that instrument **i1** precedes instrument **i2**, and instruments **i2** and **i3** precede instrument **i4**. Instrument **i1** has two channels, so bus **bus1** has two channels. Instrument **i2** has two channels, since it gets input from **bus1**. Instrument **i3** has two channels, since it gets no input, so **inchannels** is 1 and **c** and **input** have one channel each. The bus **bus2** has four channels, two each from **i1** and **i2**. The instrument **i4** has four channels, since it gets input from **bus2**.

#### EXAMPLE 2

Consider the following orchestra.

```
global {
  route(bus1, a);
  send(b; ;bus1);
  sequence(b,a);
  route(bus2, a, a);
  route(bus2, b);
  send(c; ; bus2);
}

instr a(...) {
  asig x,y;
  output(x,y,x);
}

instr b(...) {
  output(input,input);
}

instr c(...) { ... }
```

This orchestra contains a syntax error. The global sequence rules prescribe that **b** shall precede **a** (since the **sequence** directive overrides the implicit **send** sequencing), and that **a** and **b** precede **c**. The output width of **b** is two channels, since it does not receive any data according to the sequence rules, and so the width of its **input** is one channel. The output width of **a** is three channels. Thus, the width of **bus1** is three channels (although only the first is used by **b** since the width of its **input** is one channel). Thus, the two route statements onto **bus2** are incompatible, since the first uses six channels, but the second only two.

If the **sequence** directive were removed from the global block, then the syntax error would be resolved. In this case, instrument **a** precedes **b**, so **b** has three channels of input and six of output, and **bus2** can be correctly allocated with six channels.

#### 5.7.3.3.5.3 Variable allocation, startup execution and global wavetable creation

Space for any global signal variables (see subclause 5.8.5.3) shall be allocated and their values set to zero. If there is an instrument called **startup** in the orchestra, that instrument shall be instantiated and executed at the i-rate. After this execution is complete, then all global wavetables are created and filled with data according to their definitions in the global block of the orchestra and the appropriate rules in subclause 5.10.

#### 5.7.3.3.5.4 Initialisation of busses and send instruments

After the global wavetable creation, the orchestra busses are created and initialised. Each channel of each bus is set to 0 values. After the busses are created, all instruments that are the targets of **send** statements as described in subclause 5.8.5.5 shall be instantiated and executed at the i-rate in the order specified by the global sequencing rules described in the global block according to subclause 5.8.5.6. Finally, the global absolute orchestra time shall be set to 0.

NOTE - A time is called *absolute* if it is specified in seconds. When a tempo instruction is first decoded and the value of tempo changes from its default value, the score time and the absolute time are not identical anymore; all the times in the score, subsequent to a tempo line execution, are scaled according to the new tempo and enqueued in absolute dispatch and duration times as specified in subclause 5.7.3.3.6, list item 7.

#### 5.7.3.3.6 Decoder execution while streaming

In each orchestra cycle, one Composition Unit of samples is produced by the real-time synthesis process. This synthesis is performed according to the rules below and the resulting orchestra output, as described in list item 11, is presented to the Systems layer as a Composition Unit. To execute one orchestra cycle, the following tasks shall be performed in the order denoted:

1. If there is an end event whose dispatch time is earlier than or equal to the current absolute orchestra time, or an end event has been received without a timestamp since the last execution of this rule, no further output is produced, and all future requests from the systems layer to produce Composition Units are responded to with buffers of all 0s.
2. If there are any instrument events whose dispatch time is earlier than or equal to the current absolute orchestra time, or any instrument events have been received without timestamps since the last execution of this rule, an instrument instantiation is created for each such instrument event (see subclause 5.7.3.3.2), and those instantiations are each executed at the i-rate (see subclause 5.7.3.3.4) in the order prescribed by the global sequencing rules. If the instrument event specifies a duration for that instrument instantiation, the instrument instantiation shall be scheduled for termination at the time given by the sum of the current absolute orchestra time and the specified duration (scaled to absolute time units according to the actual tempo, if any).

NOTE - If the current orchestra time differs from the instrument dispatch time, the former shall be used to schedule instance termination.

3. If there are any active instrument instantiations whose termination time is earlier than or equal to the current absolute orchestra time, then the **released** standard name (see subclause 5.8.6.8.16) shall be

set to 1 within each such instrument instance, and the instance is marked for termination in step 12, below.

4. If there are any control events whose dispatch time is earlier than or equal to the current absolute orchestra time, or any control events have been received without timestamps since the last execution of this rule, the global variables or instrument variables within instrument instantiations labelled by each such control event shall have their values updated accordingly (see subclause 5.11.4). Note that this implies that no more than one control change per variable per control cycle may be received by the orchestra. If multiple control changes that reference the same variable are received in a single control cycle, the resulting value of the instrument or global variable is unspecified.
5. If there are any table events whose dispatch time is earlier than or equal to the current absolute orchestra time, or any table events have been received without timestamps since the last execution of this rule, global wavetables shall be created or destroyed as specified by the table event (see subclause 5.11.5).
6. If there are any MIDI events whose timestamp is earlier than or equal to the current orchestra time, or that have been received without timestamps since the last execution of this rule, they are dispatched according to their semantics in subclause 5.14.3.
7. If there are any tempo events whose dispatch time is earlier than or equal to the current absolute orchestra time, or any tempo events have been received since the last execution of this rule, then the global tempo variable shall be set to the specified value, and the dispatch times of all events pending for execution shall be scaled to new times according to the new tempo value. The already scheduled times for terminations are also scaled in their remaining part, according to the ratio between the old and new tempo. Existing **extend** times are not affected, since they are specified in absolute time and are thus "outside" the score. The value of the **dur** standard name (subclause 5.8.6.8.7) shall be changed in each active instrument instance to reflect the new duration of the instance.

If multiple tempo events are processed according to the preceding paragraph in the same control cycle, then the global tempo variable shall only be changed once, to the tempo indicated in the tempo event received last or with the latest timestamp, and the other tempo events are discarded. If there are multiple tempo events with the same timestamp, or both an un-timestamped event and a timestamped event shall be dispatched in the same control cycle, then the resulting value of the global tempo variable is unspecified.

NOTE - If the current orchestra time differs from the tempo dispatch time, the former shall be used to calculate the new durations and future dispatch times of events.

8. If the **speed** field of the **AudioSource** scene node responsible for instantiating this decoder (see subclause 5.15) has been changed in the last k-cycle, the tempo standard variable shall be set to 60 times the value specified in subclause 9.4.2.9 (the **AudioSource** node) of ISO/IEC 14496-1, and events in the orchestra shall be rescaled as specified in list item 7, above.
9. The value of each channel of each bus shall be set to 0.
10. Each active instrument instance shall be executed once at the k-rate and  $n$  times at the a-rate, where  $n$  is the number of samples in the control period (see subclause 5.8.5.2.2). Each execution at the k-rate shall be in the order given by the global sequencing rules, and each corresponding execution at the a-rate (that is, the first a-rate execution in a k-cycle of each, the second a-rate execution in a k-cycle of each, etc.) shall be in the order given by the global sequencing rules.

NOTE 1 - If instrument **a** is sequenced before instrument **b** according to the rules in subclause 5.8.5.6, then the k-rate execution of **a** shall be strictly before the k-rate execution of **b**, and the k-rate execution of **a** shall be strictly before the first a-rate execution of **a**, and the first a-rate execution of **a** shall be strictly before the first a-rate execution of **b**. However, there is no normative sequencing between the second a-rate execution of **a** and the first a-rate execution of **b**, or between the first a-rate execution of **a** and the k-rate execution of **b**, within a particular orchestra cycle.

NOTE 2 - In accordance with the conformance rules in subclause 5.7.4, the execution ordering described

in this subclause may be rearranged or ignored when it can be determined from examination of the orchestra that to do so will have no effect on the output of the decoding process. “Has no effect” shall be taken to mean that the output of the decoding process in rearranged order is sample-by-sample identical to the output of the decoding process performed strictly according to the rules in this subclause.

NOTE 3 - The k-cycle execution of each instrument instance shall be executed as an atomic operation; that is, the k-cycle execution of one instance shall be completed before the next begins. It is not permissible to execute k-cycles in parallel. This is not true of a-cycles; if two instruments have no sequencing relationship according to the global sequencing rules, their a-cycles may be executed in any order or in parallel.

11. If the special bus **output\_bus** is sent to an instrument, the output of that instrument at each a-cycle is the orchestra output at that a-cycle. Otherwise, the value of the special bus **output\_bus** after each instrument has been executed for an a-cycle is the orchestra output at that a-cycle. If the value of the current orchestra output is greater than 1 or less than -1, it shall be set to 1 or -1 respectively (hard clipping).
12. For each instance that was marked for termination in step 3, above: if that instrument instance called **extend** with a parameter greater than the amount of time in a control-cycle, the instrument is not terminated. All other instrument instances marked for termination in step 3 are terminated (see subclause 5.7.3.3.3). As discussed in subclause 5.14.3.2.11, in the case of an “All Notes Off” MIDI message, instances may not extend themselves, and are destroyed at this time.
13. The current global absolute orchestra time is advanced by one control period.

#### 5.7.3.3.7 Event priority

Certain events may be specified as “high priority” events, by setting the corresponding field in the bitstream element or using the \* token in the textual score. In the case that overloading of the capability of the decoder occurs, this flag allows the content author to have a minimum of control on the performance degradation.

If the **high\_priority** flag is set, then the event shall always be executed without degradation unless pathological conditions occur and no instantiations created at low priority levels are active. If the **high\_priority** flag is cleared, then the event shall be executed without degradation if no critical conditions occur. Instrument events with the **high\_priority** flag cleared may be prematurely terminated if resources are not available to dispatch an event with the **high\_priority** flag set.

NOTE - Degradation is not intended as an allowed, normative, technique to lower the computational complexity. Conforming decoders shall be able to decode, in normal conditions, bitstreams of the specified Profile@Level with no degradation. Instead, the priority level is intended as a help to implementers in critical situations due to resource overload, for instance in the case of a terminal attempting to decode a more complex bitstream than indicated by the level of the terminal, resource sharing with other applications, or high and unexpected degree of user interaction with the host terminal.

#### 5.7.3.3.8 Late-arriving events

In the case of transmission error or encoder error, certain events may arrive with timestamps that have already passed. The **use\_if\_late** field in the bitstream element indicates the proper behaviour in this case. If this field is cleared, then the event is ignored, and processing shall continue as if the event had never arrived. If this field is set, then the event is immediately dispatched according to the rules in subclause 5.7.3.3.6 as though it had been received with no timestamp.

### 5.7.4 Conformance

With regard to all normative language in this subpart of ISO/IEC 14496-3, conformance to the normative language is measured at the time of orchestra output. Any optimisation of SAOL code or rearrangement of processing sequence may be performed as long as to do so has no effect on the output of the orchestra. “Has no effect” in this

sense means that the output of the rearranged or optimised orchestra is sample-by-sample identical to the output of the original orchestra according to the decoding rules given in this subpart.

See also ISO/IEC 14496-4.

## 5.8 SAOL syntax and semantics

### 5.8.1 Relationship with bitstream syntax

The bitstream syntax description as given in subclause 5.5 specifies the representation of SAOL instruments and algorithms that shall be presented to the decoder in the bitstream. However, the tokenised description as presented there is not adequate to describe the SAOL language syntax and semantics. In addition, for purposes of enabling bitstream creation and exchange in a robust manner, it is useful to have a standard human-readable textual representation of SAOL code in addition to the tokenised binary format.

The Backus-Naur Format (BNF) grammar presented in this subclause denotes a *language*, or an infinite set of *programs*; the legal programs that may be transmitted in the bitstream are restricted to this set. Any program that cannot be parsed by this grammar is not a legal SAOL program – it has a *syntax error* – and a bitstream containing it is an invalid bitstream. Although the bitstream is made up of tokens, the grammar will be described in terms of lexical elements – a *textual representation* – for clarity of presentation. The syntactic rules expressed by the grammar that restrict the set of textual programs also normatively restrict the syntax of the bitstream, through the relationship of the bitstream and the textual format in the normative tokenisation process.

This subclause thus describes a textual representation of SAOL that is standardised, but stands outside of the bitstream-decoder relationship. Subclause 5.12 describes the mapping between this textual representation and the bitstream representation. The exact normative *semantics* of SAOL will be described in reference to the textual representation, but also apply to the tokenised bitstream representation as created via the normative tokenisation mapping.

Annex 5.C contains a grammar for the SAOL textual language, represented in the ‘lex’ and ‘yacc’ formats. Using these versions of the grammar, parsers can be automatically created using the ‘lex’ and ‘yacc’ tools. However, these versions are for informative purposes only; there is no requirement to use these tools in building a decoder.

Normative language regarding syntax in this subclause provides bounds on syntactically legal SAOL programs, and by extension, the syntactically legal bitstream sequences that can appear in an **orchestra** bitstream class. That is, there are constructions that appear to be permissible upon reading only the BNF grammar, but are disallowed in the normative text accompanying the grammar. The status of such constructions is exactly that of those which are outside of the language defined by the grammar alone. In addition, normative language describing static rate semantics further bounds the set of syntactically legal SAOL programs, and by extension, the set of syntactically legal bitstream sequences.

The decoding process for bitstreams containing syntactically illegal SAOL programs (i.e., SAOL programs that do not conform to the BNF grammar, or contain syntax errors or rate mismatch errors) is unspecified.

Normative language regarding semantics in this subclause describes the semantic bounds on the behaviour of the Structured Audio decoder. Certain constructions describe “run-time error” situations; the behaviour of the decoder in such circumstances is not normative, but implementations are encouraged to recover gracefully from such situations and continue decoding if possible.

### 5.8.2 Lexical elements

#### 5.8.2.1 Concepts

The textual SAOL orchestra contains punctuation marks, which syntactically disambiguate the orchestra; identifiers, which denote symbols of the orchestra; numbers, which denote constant values; string constants, which are not currently used; comments, which allow internal documentation of the orchestra; and whitespace, which lexically separates the various textual elements. These elements do not occur in the bitstream – since each is represented

there by a token – but we define them here to ground the subsequent discussion of SAOL. Within the rest of subclause 5.8, when we discuss the semantics of “an identifier”, this shall be taken to normatively refer to the semantics of the symbol denoted by that identifier; the language used is for clarity of presentation.

A lexical grammar for parsing SAOL, written in the ‘lex’ language, is provided for informative purposes in Annex 5.C.

#### 5.8.2.2 Identifiers

An identifier is a series of one or more letters, digits and the underscore that begins with a letter or underscore; it denotes a symbol of the orchestra. Every identifier that consists of the same characters in the first 16 characters (is equivalent under string comparison to the first 16 characters) denotes the same symbol. Identifiers are case-sensitive, meaning that identifiers that differ only in the case of one or more characters denote different symbols.

A string of characters equivalent to one of the reserved words listed in subclause 5.8.9, to one of the standard names listed in subclause 5.8.6.8, to the name of one of the core opcodes listed in subclause 5.9.3, or to the name of one of the core wavetable generators listed in clause 5.10 does not denote a symbol, but rather denotes that reserved word, standard name, core opcode, or core wavetable generator.

An identifier is denoted in the BNF grammar below by the terminal symbol **<ident>**.

#### 5.8.2.3 Numbers

There are two kinds of symbolic constants that hold numeric values in SAOL: integer constants and floating-point constants.

The integer constant is required to occur in certain contexts, such as array definitions. An integer token is a series of one or more digits. Since the contexts in which integers are required to occur in SAOL do not allow negative values, there is no provision for negative integers. A string of characters that appears to be a negative integer shall be lexically analysed as a floating-point constant. No integer constant greater than  $2^{32}$  (4294967296) shall occur in the orchestra.

There is no difference in SAOL between numbers coded with the bitstream token for integers and those coded with the bitstream token for bytes. The latter is only an aid to compression of the bitstream.

An integer constant is denoted in the BNF grammar below by the terminal symbol **<int>**.

The floating-point constant occurs in SAOL expressions, and denotes a constant numeric value. A floating-point token consists of a base, optionally followed by an exponent. A base is either a series of one or more digits, optionally followed by a decimal point and a series of zero or more digits, or a decimal point followed by a series of one or more digits. An exponent is the letter **e**, optionally followed by either a **+** or **-** character, followed by a series of one or more digits. Since the floating-point constant appears in a SAOL expression, where the unary negation operator is always available, floating-point constants need not be lexically negative. Every floating-point constant in the orchestra shall be representable by a 32-bit floating-point number.

A floating-point constant is denoted in the BNF grammar below by the terminal symbol **<number>**.

#### 5.8.2.4 String constants

String constants are not used in the normative SAOL specification, but a description is provided here so that they may be treated consistently by implementers who choose to add functionality over and above normative requirements to their implementations.

A string constant denotes a constant string value, that is, a character sequence. A string constant is a series of characters enclosed in double quotation marks (“”). The double quotation character may be included in the string constant by preceding it with a backslash (\) character. Any other character, including the line-break (newline) character, may be explicitly enclosed in the quotation marks.

The interpretation and use of string constants is left open to implementers.



### 5.8.2.5 Comments

Comments may be used in the textual SAOL representation to internally document an orchestra. However, they are not included in the bitstream, and so are lost on a tokenisation/detokenisation sequence.

A comment is any series of characters beginning with two slashes (*//*), and terminating with a new line. During lexical analysis, whenever the *//* element is found on a line, the rest of the line is ignored.

### 5.8.2.6 Whitespace

Whitespace serves to lexically separate the various elements of a textual SAOL orchestra. It has no syntactic function in SAOL, and is not represented in the bitstream, so the exact whitespacing of a textual orchestra is lost on a tokenisation/detokenisation sequence. Whitespace is not required in SAOL except so far as to disambiguate tokens and reserved words that appear next to each other (to separate “*asig*” from the variable name declared, for example).

A whitespace is any series of one or more space, tab, and/or newline characters.

## 5.8.3 Variables and values

Each signal variable within the SAOL orchestra holds a value, or an ordered set of values for array variables, as an intermediate calculation by the orchestra. At any point in time, the value of a variable, sample in a wavetable, or single element of an array variable, shall be represented by a 32-bit floating-point value.

Conformance to this subclause is in accordance with subclause 5.7.4; that is, implementations are free to use any internal representation for variable values, so long as the results calculated are identical to the results of the calculations using 32-bit floating-point values.

NOTE - For certain sensitive digital-filtering operations, the results of using greater precision in a calculation may be equivalently detrimental to orchestra output as the results of using less precision, as the stability of the filter may be critically dependent on the quantisation error that is provided with 32-bit values. It is strongly discouraged for bitstreams to contain code that generates very different results when calculated with 32-bit and 64-bit arithmetic.

At orchestra output, the values calculated by the orchestra should reside between a minimum value of  $-1$  and a maximum value of  $1$ . These values at orchestra output represent the maximum negatively- and positively-valued audio samples that can be produced by the terminal. If the values calculated by the orchestra fall outside that range, they are clipped to  $[-1, 1]$  as described in subclause 5.7.3.3 list item 11. This sound is presented, control cycle by control cycle, to the MPEG-4 system for use in AudioBIFS composition.

## 5.8.4 Orchestra

```
<orchestra>    -> <orchestra element> <orchestra>
<orchestra>    -> <orchestra element>
```

The orchestra is the collection of signal processing routines and declarations that make up a Structured Audio processing description. It shall consist of a list of one or more orchestra elements.

```
<orchestra element> -> <global block>
<orchestra element> -> <instrument declaration>
<orchestra element> -> <opcode declaration>
<orchestra element> -> <template declaration>
<orchestra element> -> NULL
```

There are four kinds of orchestra elements:

1. The global block contains instructions for global orchestra parameters, bus routings, global variable declarations, and instrument sequencing. It is not permissible to have more than one global block in an orchestra.

2. Instrument declarations describe sequences of processing instructions that can be parametrically controlled using SASL or MIDI score files.
3. Opcode declarations describe sequences of processing instruments that provide encapsulated functionality used by zero or more instruments in the orchestra.
4. Template declarations describe multiple instruments that differ only slightly using a concise parametric form.

Orchestra elements may appear in any order within the orchestra; in particular, opcode definitions may occur either syntactically before or after they are used in instruments or other opcodes.

## 5.8.5 Global block

### 5.8.5.1 Syntactic form

<global block> -> **global** { <global list> }  
<global list> -> <global statement> <global list>  
<global list> -> NULL

A global block shall contain a global list, which shall consist of a sequence of zero or more global statements.

<global statement> -> <global parameter>  
<global statement> -> <global variable declaration>  
<global statement> -> <route statement>  
<global statement> -> <send statement>  
<global statement> -> <sequence definition>  
<global statement> -> <interpolation level>

There are several kinds of global statement.

1. Global parameters set orchestra parameters such as sampling rate, control rate, and number of input and output channels of sound
2. Global variable declarations define global variables that can be shared by multiple instruments.
3. Route statements describe the routing of instrument outputs onto busses.
4. Send statements describe the sending of busses to effects instruments.
5. Sequence definitions describe the sequencing of instruments by the run-time scheduler.
6. The interpolation level specifies the quality of interpolation performed in the synthesis process.

### 5.8.5.2 Global parameter

#### 5.8.5.2.1 *srate* parameter

<global parameter> -> **srate** <int>;

The **srate** global parameter specifies the audio sampling rate of the orchestra. The decoding process shall create audio internally at this sampling rate. It is not permissible to simplify orchestra complexity or account for terminal capability by generating audio internally at other sampling rates, for to do so may have seriously detrimental effects on certain processing elements of the orchestra.

The **srate** parameter shall be an integer value between 4000 and 96000 inclusive, specifying the audio sampling rate in Hz. If the **srate** parameter is not provided in an orchestra, the default shall be the fastest of the audio signals

provided as input (see clause 5.15). If the sampling rate is not provided, and there are no input audio signals, the default sampling rate shall be 32000 Hz. It is a syntax error if more than one **srate** parameter instruction occurs in an orchestra.

In a Object type 3 terminal, or when the SAOL orchestra is used in an **AudioFX** AudioBIFS node (see subclause 5.15.3), the **srate** parameter shall only be one of the following values: (4000, 8000, 11025, 12000, 16000, 22050, 24000, 32000, 44100, 48000, 88200, 96000).

#### 5.8.5.2.2 **krate** parameter

<global parameter> -> **krate** <int>;

The **krate** global parameter specifies the control rate of the orchestra. The decoding process shall execute k-rate processing internally at this rate. It is not permissible to simplify orchestra complexity or account for terminal capability by executing k-rate processing at other rates, unless it can be determined that to do so will have no effect on orchestra output. In this case, “no effect” means that the resulting output of the orchestra is sample-by-sample identical to the output created if the control rate is not altered.

The **krate** parameter shall be an integer value between 1 and the sampling rate inclusive, specifying the control rate in Hz. If the **krate** parameter is not provided in an orchestra, the default control rate shall be 100 Hz. It is a syntax error if more than one **krate** parameter instruction occurs in an orchestra.

If the control rate as determined by the previous paragraph is not an even divisor of the sampling rate, then the control rate is the next larger integer that does evenly divide the sampling rate. The *control period* of the orchestra is the number of samples, or amount of time represented by these samples, in one control cycle.

#### 5.8.5.2.3 **inchannels** parameter

<global parameter> -> **inchannels** <int>;

The **inchannels** global parameter specifies the number of input channels to process. If there are fewer than this many audio channels provided as input sources, the additional channels shall be set to continuous zero-valued signals. If there are more than this many audio channels provided as input sources, the extra channels are ignored.

If the **inchannels** parameter is not provided in an orchestra, the default shall be the sum of the numbers of channels provided by the input sources (see clause 5.15). If there are no input sources provided, the value shall be 0. It is a syntax error if more than one **inchannels** parameter instruction occurs in an orchestra.

The only normative way in which audio input is processed by a SAOL orchestra is when the orchestra is embedded in an AudioBIFS **AudioFX** node, see ISO/IEC 14496-1 subclause 9.4.2.7. Nonnormative methods for audio processing include soundfile processing or microphone processing, see Annex 5.F.

#### 5.8.5.2.4 **outchannels** parameter

<global parameter> -> **outchannels** <int>;

The **outchannels** global parameter specifies the number of output channels of sound to produce. The run-time decoding process shall produce and render this number of channels internally. It is not permissible to simplify orchestra complexity or account for terminal capability by producing fewer channels.

If the **outchannels** parameter is not provided in an orchestra, the default shall be one channel. It is a syntax error if more than one **outchannels** parameter instruction occurs in an orchestra.

#### 5.8.5.2.5 **interp** parameter

<global parameter> -> **interp** <int>;

The **interp** global parameter specifies the quality of interpolation performed in the synthesis process. Various operations require access to wavetables at non-integer points; to access a wavetable at such a point requires interpolation among the available points in the wavetable.

If the **interp** parameter is 0, then “low level” interpolation is performed. Every interpolation shall be performed using a linear interpolation. That is, let *i* and *j* be two consecutive indices of a wavetable, and let *x* and *y* be the values at points *i* and *j* respectively. Assume that the value at point *k* is required, where  $i < k < j$ . Then let *q* be the value  $k - \text{floor}(k)$ . Then the interpolated value is  $x + q * (y - x)$ .

If the value required is that “between” the last point and first point of the table as it wraps around, consider *x* to be the value of the last point, *y* the value of the first point, and assert  $n < k < n+1$  where index *n* is the last point in the wavetable. Then calculate the interpolated value as  $x + q * (y - x)$  as above.

If the **interp** parameter is 1, then “high level” interpolation is performed. The method of high-level interpolation is non-normative, but it shall be a higher-quality method than linear interpolation.

It is a syntax error if the parameter is not 0 or 1. It is a syntax error if more than one **interp** parameter instruction occurs in an orchestra.

If the **interp** parameter is not specified in an orchestra, then the interpolation quality is “low” by default. If authors wish to have normative, high-quality interpolation for wavetables, they can rewrite their own versions of **tableread**, **oscil**, and other instructions to perform this.

### 5.8.5.3 Global variable declaration

#### 5.8.5.3.1 Syntactic form

```
<global variable declaration> -> ivar <namelist> ;
<global variable declaration> -> ksig <namelist> ;
<global variable declaration> -> <table declaration> ;
```

Global variable declarations declare variables that may be shared and accessed by all instruments and by a SASL score. Only **ivar** and **ksig** type variables, as well as wavetables, may be declared globally. A global variable declaration is either a table definition, or an allowed type name followed by a list of name declarations.

A global name declaration specifies that a name token shall be created and space equal to one signal value allocated for variable storage in the global context. A global array declaration specifies that a name token shall be created and space equal to the specified number of signal values allocated in the global context.

#### 5.8.5.3.2 Signal variables

```
<namelist>      -> <name>, <namelist>
<namelist>      -> <name>
```

A namelist is a sequence of one or more name declarations.

```
<name>          -> <ident>
<name>          -> <ident>[<array length>]
```

```
<array length> -> <int>
<array length> -> inchannels
<array length> -> outchannels
```

A name declaration is an identifier (see subclause 5.8.2.2), or an array declaration. For an array declaration, the parameter shall be either an integer strictly greater than 0, or one of the tokens **inchannels** or **outchannels**. If the latter, the array length shall be the same as the number of input channels or output channels to the instrument,

respectively, as described in subclause 5.7.3.3.5.2. It is illegal to use the token **inchannels** if the number of input channels to the instrument is 0.

Not every identifier may be used as a variable name; in particular, the reserved words listed in subclause 5.8.8, the standard names listed in subclause 5.8.6.8, the names of the core opcodes listed in clause 5.9, and the names of the core wavetable generators listed in clause 5.10 shall not be declared as variable names.

### 5.8.5.3.3 Wavetable declarations

<table declaration> -> **table** <ident> ( <ident> , <expr> [ , <expr list> ] ) ;

<expr> as defined in subclause 5.8.6.7.

<expr list> as defined in subclause 5.8.6.6.1.

Wavetables are structures of memory allocated for the typical purpose of allowing rapid oscillation, looping, and playback. The wavetable declaration associates a name (the first identifier) with a wavetable created by a core wavetable generator referenced by the second identifier. It is a syntax error if the second identifier is not one of the core wavetable generators named in clause 5.10. The first expression in the comma-delimited parameter sequence is termed the *size expression*; the remaining zero or more expressions comprise the *wavetable parameter list*.

The semantics of the size expression and wavetable parameter list are determined by the particular core wavetable generator, see clause 5.10. Any expression that is i-rate (see subclause 5.8.6.7.2) is legal as part of the table parameter list; in particular, reference to i-rate global variables is allowed (their values may be set by the special instrument **startup**). Each expression shall be single-valued, except in the case of the **concat** generator (subclause 5.10.16), in which case the expressions shall be table references. The order of creation of wavetables is not deterministic, with the exception of the table arguments of a **concat** generator, which are always generated before the **concat** generator that uses them. In this case, the tables used as arguments to the **concat** generator must be appear before the table which uses the **concat** generator, to prevent dependency loops.

It is not recommended for calls to the **tableread()** opcode to occur in the table parameter expressions, and to do so gives unspecified results.

A global wavetable may be referenced by a wavetable placeholder in any instrument or opcode. See subclause 5.8.6.5.4. Global wavetables shall be created and initialised with data at orchestra initialisation time, immediately after the execution of the special instrument **startup**. They shall not be destroyed unless they are explicitly destroyed or replaced by a **table** line in a SASL score.

To create a wavetable, first, the expression fields are evaluated in the order they appear in the syntax according to the rules in subclause 5.8.6.7. Then, the particular wavetable generator named in the second identifier is executed; the normative semantics of each wavetable generator detail exactly how large a wavetable shall be created, and which values placed in the wavetable, for each generator.

### 5.8.5.4 Route statement

<route statement> -> **route** ( <ident> , <identlist> ) ;

<identlist> -> <ident> , <identlist>

<identlist> -> <ident>

<identlist> -> <NULL>

A **route** statement consists of a single identifier, which specifies a bus, and a sequence of one or more instrument names, which specify instruments. The route statement specifies that the instruments listed do not produce sound output directly, but instead their results are placed on the given bus. The output channels from the instruments listed each are placed on a separate channel of the bus. Multiple **route** statements onto the same bus indicate that the given instrument outputs shall be summed on the bus. Multiple **route** statements with differing numbers of channels referencing the same bus are illegal, unless each statement has either *n* channels or 1 channel. In this case, each of the one-channel **route** statements places the same signal on each channel of the bus, which is *n* channels wide.

There shall be at least one instrument name in the instrument list (the NULL subclause in the grammar is provided so that constructions appearing later may use the same production).

EXAMPLES

Assume that instruments **a**, **b**, and **c** produce one, two, and three channels of output, respectively.

1. The sequence

```
route(bus1, a, b);
route(bus1, c);
```

is legal and specifies a three-channel bus. The first bus channel contains the sum of the output of **a** and the first channel of **c**; the second contains the sum of the first output channel of **b** and the second of **c**; and the third contains the sum of the second channel of **b** and the third channel of **c**.

2. The sequence

```
route(bus1,b);
route(bus1,c);
```

is illegal since the statements refer different numbers of channels to the same bus.

3. The sequence

```
route(bus1,a,c);
route(bus1,a);
route(bus1,b,b);
```

is legal and specifies a four-channel bus. The first and third **route** statements each refer to four channels of audio, and the second refers to one channel, which will be mapped to each of the four channels.

The resulting channel values are as follows, using array notation to indicate the channel outputs from each instrument:

**Table 5.1 - Example of calculating bus routing values**

Channel	Value
1	$a + a + b[1]$
2	$c[1] + a + b[2]$
3	$c[2] + a + b[1]$
4	$c[3] + a + b[2]$

It is illegal for a **route** statement to reference a bus that is not the special bus **output\_bus** and that does not occur in a **send** statement. See subclause 5.8.5.5.

It is illegal for a **route** statement to refer to the special bus **input\_bus** (see subclause 5.15.2).

All instruments that are not referred to in **route** statements place their output on the special bus **output\_bus**, except for an effect instrument to which **output\_bus** was sent (see subclause 5.8.5.5). The same rules for allowable channel combinations to the special bus **output\_bus** apply as if the route statements were explicit; these rules are implicit in the rules for the **output** statement, see subclause 5.8.6.6.8.

**5.8.5.5 Send statement**

```
<send statement> -> send ( <ident> ; <expr list> ; <identlist> );
<identlist>      as defined in subclause 5.8.5.4
<expr list>     as defined in subclause 5.8.6.6.1
```

The **send** statement creates an instrument instantiation, defines busses, and specifies that the referenced instrument is used as an effects processor for those busses.

All busses in the orchestra are defined by using **send** statements. It is illegal for a statement referencing a bus to refer to a bus that is not defined in a **send** statement. The exception is the special bus **output\_bus**, which is always defined.

The identifier in the **send** statement references an instrument that will be used as a bus-processing instrument, also called *effect instrument*. There is no syntactic distinction between effect instruments and other instruments. The identifier list references one or more busses that shall be made available to the effect instrument through its **input** standard name, as follows:

The first  $n_0$  channels of **input**, channels 0 through  $n_0-1$  are the  $n_0$  channels of the first referenced bus;  
Channels  $n_0$  through  $n_0+n_1-1$  of **input** are the  $n_1$  channels of the second bus,  
and so forth, with a total of  $n_0 + n_1 + \dots + n_k$  channels.

In addition, the grouping of busses in the **input** array shall be made available to the effect instrument through its **inGroup** standard name, as follows:

The first  $n_0$  values of **inGroup** have the value 1;  
Channels  $n_0$  through  $n_0+n_1-1$  of **inGroup** have the value 2,  
and so forth, through  $n_0 + n_1 + \dots + n_k$ , with the last  $n_k$  having the value  $k$ .

The expression list is a list of zero or more i-rate expressions that are provided to the effect instrument as its parameter fields. Any expression that is i-rate (see subclause 5.8.6.7.2) is legal as part of this list; in particular, reference to i-rate global variables is allowed. The number of expressions provided shall match the number of parameter fields defined in the instrument declaration; otherwise, it is a syntax error.

The effect instrument referred to in a **send** statement shall be instantiated at orchestra start-up; see subclause 5.7.3.3.5.4. These instrument instantiations shall remain in effect until the orchestra synthesis process terminates. One instrument instantiation shall be created for each **send** statement in the orchestra. If such an instrument instantiation utilises the **turnoff** statement, the instantiation is destroyed (and sound is no longer routed to it). No other changes are made in the orchestra.

Any bus may be routed to more than one effect instrument, except for the special bus **output\_bus**. The special bus **output\_bus** represents the second-to-finalmost processing of a sound stream; it may only be sent to at most one effect instrument, and it is a syntax error if that instrument is itself routed or makes use of the **outbus** statement. If **output\_bus** is not sent to an instrument, it is turned into sound at the end of an orchestra cycle (see subclause 5.7.3.3); if **output\_bus** is sent to an instrument, the output of that instrument is turned into sound at the end of an orchestra pass. This instrument is not permitted to use the **turnoff** statement.

In the case that the number of input channels received by an instrument instance differs from the width of the bus(es) providing that input (see subclause 5.7.3.3.5.2), the width of **input** and **inGroup** and the value of **inchan** respect the former rather than the latter. In the case that **inchan** is smaller than the number of channels on the buses providing input, only the first **inchan** channels are used; in the case that **inchan** is smaller, the “extra” channels are all 0’s in both **input** and **inGroup**.

At least one bus name shall be provided in the **send** instruction.

#### 5.8.5.6 Sequence specification

<sequence specification> -> **sequence** ( <identlist> );  
<identlist> as defined in subclause 5.8.5.4.

The **sequence** statement allows the specification of the ordering of execution of instrument instantiations by the run-time scheduler. The **identlist** references a list of instruments that describes a partial ordering on the set of instruments. If instrument **a** and instrument **b** are referenced in the same **sequence** statement with **a** preceding **b**, then instantiations of instrument **a** shall be executed strictly before instantiations of instrument **b**.

There are several default sequence rules:

1. The special instrument **startup** is instantiated and the instantiation executed at the i-rate at the very beginning of the orchestra.
2. Any instrument instances corresponding to the **startup** instrument are executed first in a particular orchestra cycle.
3. If **output\_bus** is sent to an instrument, the instrument instantiation corresponding to that **send** statement is the last instantiation executed in the orchestra cycle.
4. For each instrument routed to a bus that is sent to an effect instrument, instantiations of the routed instrument are executed before instantiations of the effect instrument. If loops are created using **route** and **send** statements, the ordering is resolved syntactically: whichever **send** statement occurs latest, that instrument instantiation is executed latest.

Default rules 2, 3, and 4 may be overridden by use of the **sequence** statement. Rule 1 cannot be overridden.

It is a syntax error if explicit **sequence** statements create loops in ordering. Any **send** statements that are the “backward” part of an implicit **send** loop have no effect.

If the sequence of two instruments is not defined by the default or explicit sequence rules, their instantiations may be executed in any order or in parallel.

It is not possible to specify the ordering of multiple instantiations of the same instrument; these instantiations can be run in any order or in parallel.

### EXAMPLES

An orchestra consists of five instruments, **a**, **b**, **c**, **d**, and **e**.

1. The following code fragment

```
route(bus1, a, b);
send(c; ; bus1);
```

is legal and specifies (using the default sequencing rules) that instantiations of instruments **a** and **b** shall be executed strictly before instantiations of instrument **c**. This ordering applies to all instantiations of instrument **c**, not only to the one corresponding to the **send** statement. No ordering is specified between instruments **a** and **b**.

2. The following code fragment

```
route(bus1, a, b);
send(c; ; bus1);
sequence(c,a);
send(d; ; bus1);
```

is legal and specifies that instantiations of instrument **b** shall be executed first, followed by instantiations of instrument **c**, followed by instantiations of instrument **a**, followed by instances of instrument **d**. (See Figure 5.8.1) The ordering of **b** and **c**, and **a** and **b** with **d**, follows from default rule 3; the placement of instrument **c** follows from the explicit **sequence** statement, which overrides default rule 3. Due to this ordering, the output samples of instrument **a** are not provided to instrument **c** (they get put on the bus “too late”), and however many channels of output this represents are set to 0 in instrument **c**. The output samples of instrument **a** are provided to instrument **d**.



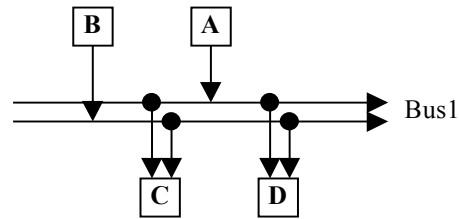


Figure 5.1 - Example of ordering instruments with 'sequence'

3. The following code fragment

```
sequence(a,b);
sequence(b,c,d);
sequence(c,e);
sequence(e,a);
```

is illegal, as it contains an explicit loop in sequencing.

4. The following code fragment

```
route(bus1, a);
send(b; ; bus1);
route(bus2, b);
send(a; ; bus2);
```

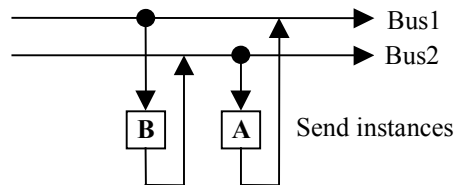


Figure 5.2 - Example of ordering instruments with 'sequence'

is legal, and specifies that instantiations of instrument **b** are executed first, followed by instantiations of instrument **a**. There is an implicit loop here that is resolved syntactically as described in default rule 3. Due to this ordering, the output values of instrument **a** are not provided to instrument **b**. Note that for deciding sequencing, only the order of **send** statements matters, not the order of **route** statements.

## 5.8.6 Instrument definition

### 5.8.6.1 Syntactic form

```
<instrument definition> -> instr <ident> ( <identlist> ) [ preset <int> [ <int> ... ] ] {
    <instr variable declarations>
    <block> }
```

An instrument definition has several elements. In order, they are

1. An identifier that defines the name of the instrument,
2. A list of zero or more identifiers that define names for the parameter fields, also called pfields, of the instrument,
3. An optional list of preset values for specifying MIDI preset mappings,

4. A list of zero or more instrument variable declarations, and
5. A block of statements defining the executable functionality of the instrument.

### 5.8.6.2 Instrument name

Any identifier may serve as the instrument name except that the instrument name shall not be a reserved word (see subclause 5.8.9), the name of a core opcode (see clause 5.9), or the name of a core wavetable generator (see clause 5.10). An instrument name may be the same as a variable in local or global scope; there is no ambiguity so created, since the contexts in which instrument names may occur are very restricted.

No two instruments or opcodes in an orchestra shall have the same name.

### 5.8.6.3 Parameter fields

<identlist>       -> as given in subclause 5.8.5.4

The parameter fields, also called pfields, of the instrument, are the interface through which the instrument is instantiated. In the instrument code, the pfields have the rate semantics of i-rate local variables. Their values shall be set on instrument instantiation, before the creation of local variables, with the appropriate values as given in the score line, score event, MIDI event, **send** statement, or **instr** statement corresponding to the instrument instantiation.

### 5.8.6.4 Preset tag

The **preset** tag specifies the preset number(s) of the instrument. When MIDI program change events arrive in a MIDI stream or MIDI file controlling the orchestra, the program change numbers refer to the **preset** tags given to the various instruments. No more than one instrument may have the same preset number; if multiple instruments in an orchestra specify the same **preset** tag, the one occurring syntactically last is assigned that preset number. If a **preset** tag is not associated with a particular instrument, then that instrument has no preset number and cannot be referenced with a program change. If more than one tag is given, the instrument responds to all of the listed preset values.

Preset tags in SAOL correspond to both the preset and bank value of a program in MIDI control; the program on preset **x**, bank **y** in MIDI syntax shall be indicated as preset  $(y - 1) * 128 + (x - 1)$  in SAOL (since presets and banks are numbered starting with 1 in MIDI).

See clause 5.14 for more normative semantics governing MIDI control of orchestras.

### 5.8.6.5 Instrument variable declarations

#### 5.8.6.5.1 Syntactic form

<instr variable declarations>       -> <instr variable declarations> <instr variable declaration>  
 <instr variable declarations>       -> <NULL>

<instr variable declaration>       -> [ <sharing tag> ] **ivar** <namelist> ;  
 <instr variable declaration>       -> [ <sharing tag> ] **ksig** <namelist> ;  
 <instr variable declaration>       -> **asig** <namelist> ;  
 <instr variable declaration>       -> <table declaration> ;  
 <instr variable declaration>       -> <sharing tag> **table** <identlist> ;  
 <instr variable declaration>       -> **oparray** <ident> [<array length> ] ;  
 <instr variable declaration>       -> <tablemap declaration> ;

<sharing tag>                       -> **imports**  
 <sharing tag>                       -> **exports**  
 <sharing tag>                       -> **imports exports**

<tablemap declaration>           -> **tablemap** <ident> ( <identlist> ) ;

<array length> and <namelist> as defined in subclause 5.8.5.3.2

<table declaration> as defined in subclause 5.8.5.3.3

<identlist> as defined in subclause 5.8.5.4

Instrument variable declarations declare variables that may be used within the scope of an instrument. Any rate type variable, as well as wavetables, tablemaps, and wavetable placeholders, may be declared in an instrument. An instrument variable declaration is either a wavetable declaration, or an type name, possibly preceded by a sharing tag, followed by a list of name declarations, or a sharing tag followed by the token **table** followed by a list of identifiers referencing global or future wavetables, or an opcode-array declaration, or a table-map definition.

#### 5.8.6.5.2   Wavetable declaration

The syntax and semantics of subclause 5.8.5.3.3 hold for instrument local wavetables, with the following exceptions and additions:

An instrument local wavetable is available only within the local scope of a single instrument instantiation. As such, it shall be created and initialised with data at the instrument instantiation time, immediately after the pfield values are assigned from the calling parameters. It may be deleted and freed when that instrument instantiation terminates.

Not every expression that is i-rate is legal as part of the table parameter list. Reference to constants, pfields, imported i-rate variables, and i-rate standard names is allowed. However, the instrument wavetable initialisation shall occur before the initialisation pass through the instrument code, and so reference to local i-rate variables is prohibited.

#### 5.8.6.5.3   Signal variables

The syntax and semantics of subclause 5.8.5.3.2 hold for instrument local signal variables, with the following exceptions and additions:

A local name declaration specifies that a name token shall be created and space equal to one signal value allocated for variable storage in each instrument instantiation associated with the instrument definition. A local array declaration specifies that a name token shall be created and space equal to the specified number of signal values allocated in each instrument instantiation associated with the instrument definition.

The sharing tags **imports** and/or **exports** may be used with local i-rate or k-rate signal variable declaration. They shall not be used with a-rate variables. If the **imports** tag is used, then the variable value shall be replaced with the value of the global variable of the same name at instrument initialisation time (for i-rate signal variables) or at the beginning of each control pass (for k-rate signal variables). The **imports** tag may be used for a local k-rate signal variable even if there is no global variable of the same name, in which case it is an indication that the k-rate variable so tagged may be modified with **control** lines in a SASL score. The **imports** tag shall not be used for local i-rate signal variables when there is no global variable of the same name.

If the **exports** tag is used, then the value of the global variable of the same name shall be replaced with the value of the local signal variable after instrument initialisation (for i-rate signal variables) or at the end of each control pass (for k-rate signal variables). The **exports** tag shall not be used if there is no global variable of the same name.

If, for a particular signal variable, the **imports** and/or **exports** tags are used, and there is a global variable with the same name, then the array width of the local and global variables shall be the same.

If, for a particular local variable, the **imports** tag is not used, then its value is set to 0 before instrument initialisation.

If, for a particular local variable declaration, the **imports** and **exports** tags are not used, even if there is a global variable of the same name, there is no semantic relationship between the two variables. The construction is syntactically legal.

#### 5.8.6.5.4 Wavetable placeholder

The sharing tags **imports** and **exports** may be used to reference global and future wavetables. In this case, the local declaration of the table reference is termed a wavetable placeholder. The wavetable placeholder definition does not contain a full wavetable definition, but only a reference to a global or future wavetable name.

If only the **imports** tag is used, and there is a global wavetable with the same name, then at instrument instantiation time, the current contents of the global wavetable are copied into a local wavetable with that name. If the contents of the global wavetable are modified after a particular instrument instantiation referencing that global wavetable is created, the new contents of the global wavetable shall not be copied into the instrument instantiation. Also, if the contents of the local wavetable are modified, these changes shall not be reflected in the global wavetable.

If the **imports** and **exports** tags are both used, and there is a global wavetable with the same name, then at instrument instantiation time and at the beginning of each control pass, the current contents of the global wavetable are made available to a local wavetable with that name. "Made available" in the preceding sentence means that access may be either in the form of copying data from one wavetable to another or by pointer reference to the same memory space, or by any equivalent implementation. Also, at the end of instrument instantiation and at the end of each control pass, the current contents of the local wavetable are similarly made available to the global wavetable with the same name. In this case, if a wavetable is modified at the a-rate in one instrument instance, it is unspecified exactly when these changes are visible to other instrument instances, but it shall be no later than the next orchestra cycle (it is permitted to be in the same orchestra cycle).

It is not permissible to use the **exports** tag alone for a wavetable placeholder.

If the **imports** tag is used, and there is no global wavetable with the same name, then the reference is to a *future wavetable* that will be provided in the bitstream. When the instrument is instantiated, the contents of the most recent wavetable provided in the bitstream with the same name shall be copied into the local wavetable. If no wavetable has been provided in the bitstream with the same name as the wavetable placeholder at the time of instrument instantiation, then the bitstream is invalid. If the wavetable with this name is changed by providing a new wavetable with the same name by using a **table** line in the bitstream (subclause 5.11.6), the reference immediately changes to the new wavetable when the **table** line is dispatched.

It is not permissible to use the **exports** tag if there is no global wavetable with the same name.

#### 5.8.6.5.5 Opcode array declaration

An opcode array, or "oparray" declaration, declares several opcode states for a particular opcode that may be used by the current instrument or opcode. By declaring the states in this manner, access to them is available through the oparray expression, see subclause 5.8.6.7.7. The identifier in the declaration shall be the name of a core opcode or a user-defined opcode declared elsewhere in the orchestra. The array length declares how many states are available for access to this oparray in the local code block; it shall be an integer value or the special tag **inchannels** or **outchannels**.

It is a syntax error if more than one oparray declaration references the same opcode name in a single instrument or opcode.

#### 5.8.6.5.6 Table map definition

<table map definition> -> **tablemap** <ident> ( <identlist> )

<identlist> as defined in subclause 5.8.5.4.

A table map is a data structure allowing indirect reference of wavetables via array notation. The identifier names the table map; it shall not be the same as the name of any other signal variable or other restricted word in the local scope. The identifier list gives a number of wavetable names for use with the table map. Each of these names shall correspond to a wavetable definition or wavetable placeholder within the current scope. The **tablemap** declaration may come before, after, or in the midst of wavetable declarations and wavetable placeholders in the instrument. All wavetables in the scope of the instrument may be referenced in a **tablemap**, regardless of the syntactic placement of the **tablemap**.

When the tablemap name is used in an array-reference expression (see subclause 5.8.6.7.5), the index of the expression determines to which of the wavetables in the list the expression refers. The first wavetable in the list is number 0, the second number 1, and so on.

#### EXAMPLE

For the following declarations

```
table t1(harm,2048,1);
imports table t2;
table t3(random,32,1);

tablemap tmap(t1,t2,t3,t2);
ivar i,x,y,z;
```

the following two code blocks are identical in semantics:

#### BLOCK 1

```
i = 3;
x = tableread(tmap[0],4);
y = tableread(tmap[i],3);
z = tableread(tmap[i > 4 ? 1 : 2],5);
```

#### BLOCK 2

```
x = tableread(t1,4);
y = tableread(t2,3);
z = tableread(t3,5);
```

Note that, like table references, array expressions using tablemaps may only occur in the context of an opcode or oparray call to an opcode accepting a wavetable reference.

### 5.8.6.6 Block of code statements

#### 5.8.6.6.1 Syntactic form

```
<block>      -> <statement> [ <block> ]
<block>      -> <NULL>

<statement> -> <lvalue> = <expr> ;
<statement> -> <expr> ;
<statement> -> if ( <expr> ) { <block> }
<statement> -> if ( <expr> ) { <block> } else { <block> }
<statement> -> while ( <expr> ) { <block> }
<statement> -> instr <ident> ( <expr list> ) ;
<statement> -> output ( <expr list> ) ;
<statement> -> spatialize ( <expr list> ) ;
<statement> -> outbus ( <ident> , <expr list> ) ;
<statement> -> extend ( <expr> ) ;
<statement> -> turnoff ;

<expr list> -> <expr> [, <expr list>]
<expr list> -> <NULL>
```

<lvalue> as given in subclause 5.8.6.6.2.

<expr> as given in subclause 5.8.6.7.

A block is a sequence of zero or more statements. A statement shall take one of several forms, which are enumerated and described in the subsequent subclauses. Each statement has rate-semantics rules governing the rate of the statement, the rate contexts in which it is allowable, and the times at which various subcomponents shall be executed.

To execute a block of statements at a particular rate, the statements within the block shall be executed, each at that rate, in such order as to produce equivalent results to executing the statements sequentially in linear order, according to the semantics below governing each type of statement.

#### 5.8.6.6.2 Assignment

<statement> -> <lvalue> = <expr> ;

<lvalue> -> <ident>

<lvalue> -> <ident> [ <expr> ]

<expr> as given in subclause 5.8.6.7.

An assignment statement calculates the value of an expression and changes the value of a signal variable or variables to match that value.

The lvalue, or left-hand-side value, denotes the signal variable or variables whose values are to be changed. An lvalue may be a local variable name, in which case the denotation is to the storage space associated with that name. An lvalue may also be a local array name, in which case the denotation is to the entire array storage space. An lvalue may also be a single element of a local array denoted by indexing a local array name with an expression. An lvalue shall not be a table reference or tablemap expression. An lvalue shall not be a standard name other than **MIDIctrl** or **params** (see subclause 5.8.6.8.9 and 5.8.6.8.26).

If the lvalue denotes an entire array, the right-hand-side expression of the assignment shall denote an array-valued expression with the same array length, or a single value, otherwise the construction is syntactically illegal.

If the lvalue denotes a single value, the right-hand-side expression of the assignment shall denote a single value, otherwise the construction is syntactically illegal.

The rate of the lvalue is the rate of the signal variable, if there is no indexing expression, or the faster of the rate of the signal array denoted by the signal variable and the rate of the indexing expression, if there is an indexing expression.

The rate of the right-hand side is the rate of the right-hand-side expression.

The rate of the statement is the rate of the lvalue, however, the statement is illegal if the rate of the right-hand side is faster than the rate of the lvalue.

The assignment shall be performed as follows:

At every pass through the statement occurring at equal rate to the rate of the assignment, the right-hand side expression shall be evaluated. Then, the storage space denoted by the lvalue shall be updated to be equal to the value of the right-hand expression. If the lvalue denotes an entire array, and the right-hand-side expression a single value, then each of the values of each of the elements of the array shall be changed to the single right-hand-side value.

#### 5.8.6.6.3 Null assignment

<statement> -> <expr> ;

A null assignment contains only an expression; it is provided so that opcodes that do not have useful return values need not be used in the context of an assignment to a dummy variable.

The rate of the statement is the rate of the expression. The expression may be single-valued or array-valued; it shall not be a table reference.

The null assignment shall be performed as follows:

At every pass through the statement occurring at equal rate to the rate of the statement, the expression shall be evaluated.

#### 5.8.6.6.4 If

<statement> -> **if** ( <expr> ) { <block> }

An **if** statement allows conditional evaluation of a block of code. The expression that is tested in the **if** statement is termed the guard expression.

The rate of the statement is the rate of the guard expression, or the rate of the fastest statement in the guarded code block, whichever is faster.

It is not permissible for the block of code governed by the **if** statement to contain statements slower than the guard expression. It is further not permissible for any of the statements in the governed block of code to contain calls to opcodes that would be executed slower than the guard expression. The guard expression shall be a single-valued expression.

#### EXAMPLE

The following code fragment contains a rate-mismatch error:

```
asig a; ksig k;

a = 0; if (a < 20) {
  k = kline(...);
}
```

The example is illegal because the **kline** assignment statement is slower than the guard **a < 20**. Even if the assignment were to an a-rate variable (“a2 = kline(...)”), thus making the assignment statement an a-rate statement, the example would be illegal, because the **kline** opcode itself is slower than the guard expression.

The **if** statement shall be executed as follows:

At every pass through the statement occurring at equal rate to the rate of the statement, the guard expression shall be evaluated. If the guard statement evaluates to any non-zero value in a particular pass, then the block of code shall be evaluated at the rate corresponding to that pass.

If a block of code executing at the **a-rate** or **k-rate** has **i-rate** statements, these statements should only be executed the first time the block executes, with regards to a particular state.

If a block of code executing at the **a-rate** has **k-rate** statements, these statements should only be executed the first time the block executes in a kcycle.

#### 5.8.6.6.5 Else

<statement> -> **if** ( <expr> ) { <block> } **else** { <block> }

An **else** statement allows disjunctive evaluation of two blocks of code. The expression that is tested in the **else** statement is termed the guard expression.

The rate of the statement is the rate of the guard expression, or the rate of the fastest statement in the first guarded block of code, or the rate of the fastest statement in the second guarded block of code, whichever is fastest.

It is not permissible for the blocks of code governed by the **else** statement to contain statements slower than the guard expression. It is further not permissible for any of the statements in the governed blocks of code to contain calls to opcodes that would be executed slower than the guard expression. The guard expression shall be a single-valued expression.

The **else** statement shall be executed as follows:

At every pass through the statement occurring at equal rate to the rate of the statement, the guard expression shall be evaluated. If the guard expression evaluates to any non-zero value in a particular pass, then the first guarded block of code shall be at the rate corresponding to that pass. If the guard statement evaluates to zero in a particular pass, then each statement in the second guarded block of code shall be so evaluated.

If a block of code executing at the **a-rate** or **k-rate** has **i-rate** statements, these statements should only be executed the first time the block executes, with regards to a particular state.

If a block of code executing at the **a-rate** has **k-rate** statements, these statements should only be executed the first time the block executes in a kcycle.

### 5.8.6.6.6 While

<statement> -> **while** ( <expr> ) { <block> }

The **while** statement allows a block of code to be conditionally evaluated several times in a single rate pass. The expression that is tested in the **while** statement is termed the guard expression.

The rate of the **while** statement is the rate of the guard expression.

It is not permissible for the block of code governed by the **while** statement to contain statements that run at a rate other than the rate of the guard expression. It is further not permissible for any of the statements in the governed block of code to contain calls to opcodes that would be executed at a rate other than the rate of the guard expression. The guard expression shall be a single-valued expression. It is not permissible for the guard expression to contain calls to core opcodes with type **specialop**, see subclause 5.9.2.

The **while** statement shall be executed as follows:

At every pass through the statement occurring at equal rate to the rate of the statement, the guard expression shall be evaluated. If the guard expression evaluates to any non-zero value in a particular pass, then each statement in the guarded block of code shall be evaluated according to the particular rules for that statement, and then the guard expression re-evaluated, iterating until the guard expression evaluates to zero.

### 5.8.6.6.7 Instr

<statement> -> **instr** <ident> ( <expr list> ) ;

The **instr** statement allows an instrument instantiation to dynamically create other instrument instantiations, for layering or synthetic-performance techniques. It shall consist of an identifier referring to an instrument defined in the current orchestra, a time delay, a duration, and a list of expressions defining parameters to pass to the referenced instrument.

It is a syntax error if the number of expressions in the expression list is not two greater than the number of pfields accepted by the referenced instrument (the first expression is the time delay and the second is the duration). Each expression in the expression list shall be a single-valued expression.

The rate of the **instr** statement is the rate of the fastest expression in the expression list, or the rate of the guarding expression containing the statement, or the rate of the opcode containing the statement, whichever is fastest.

It is not permissible for the rate of the **instr** statement to be a-rate.



The **instr** statement shall be executed as follows:

At every pass through the statement occurring at equal rate to the rate of the statement, each of the expressions in the expression list is evaluated. Then, a new instrument event is registered with the scheduler as described in subclause 5.11.3. The dispatch time of the new instrument event is the sum of the current orchestra time and the value of the first expression in the expression list, the latter scaled by the current global tempo; the duration of the new instrument event is the value of the second expression in the expression list; and the values of the p-fields for the new instrument event are the values of the remaining expressions in the expression list. To clarify, note that the values of the first and second expressions are considered to have units of score beats, not absolute time, and they are consequently scaled according to the actual **tempo** of the orchestra.

An exception to the above occurs when the time-delay (the first expression in the expression list) is less than the length of the orchestra control period. In this case, an instrument event is not created, but a new instrument instantiation is immediately created, where the duration of the new instantiation is the value of the second expression in the expression list, and the values of the instrument p-fields in the new instantiation are set to the values of the remaining expressions.

In this case, the i-rate pass through the new instrument instantiation shall be executed immediately upon its creation, before any more statements from the block of code containing the **instr** statement are executed. However, any changes to global i-rate variables made in the new instance during its i-rate pass are not respected in this instrument (the “caller”) (i-rate variables imported from the global context are set only during the initialisation pass of each instance, and never change afterward). The first k-rate and a-rate passes through the new instrument instantiation shall be executed as appropriate to the sequencing relation between the instantiating and instantiated instruments; that is, if the new instrument is sequenced later than the instantiating instrument, the new instantiation shall be executed at some later time in the same orchestra pass, but if the new instrument is sequenced earlier than the instantiating instrument, then the new instantiation shall not be executed in k-time or a-time until the subsequent orchestra pass.

A dynamically created instrument has access to the same **MIDCtrl** (subclause 5.8.6.8.9), **MIDtouch** (subclause 5.8.6.8.10), **MIDbend** (subclause 5.8.6.8.11), **channel** (subclause 5.8.6.8.12) and **preset** (subclause 5.8.6.8.13) standard name state as its parent. However, the dynamically created instrument is not scheduled for termination when the parent is terminated under MIDI control.

#### 5.8.6.6.8 Output

<statement> -> **output** ( <expr list> );

The **output** statement creates audio output from the instrument. This output does not get turned directly into sound, but rather gets buffered either on one or more busses based on instructions given in **route** statements (subclause 5.8.5.4) or on the special bus **output\_bus** by default. However, if the current instrument instantiation is the one created with a **send** statement referencing the special bus **output\_bus**, then the output of the current instantiation, created by summing its calls to **output**, may be turned directly into sound.

The expression list shall contain at least one expression.

The rate of the **output** statement is a-rate.

All statements within an orchestra that reference the same bus, whether through explicit sends, calls to **outbus**, or by default routing to the special bus **output\_bus**, shall have compatible numbers of expression parameters representing output channels. “Compatible” means that if any calls to **output** for a particular bus reference more than one expression parameter, then all other calls to **output** referencing this bus shall have either the same number of expression parameters, or else only a single expression parameter. In addition, the number of channels of the special bus **output\_bus** shall be the same as the global **outchannels** parameter and uses of **output** by instrument instances that are implicitly or explicitly routed to **output\_bus** shall be compatible with this number of channels.

The **output** statement is executed as follows:

At each k-rate pass through the instrument, an output buffer, with number of channels determined by the rules in subclause 5.7.3.3.5.2, shall be cleared to zero values. At every a-rate pass through the statement, the expression parameters shall each be evaluated. Then, the expression parameter values shall be placed in the output buffer: if the **output** statement has more than one parameter expression, then the value of each parameter shall be added to the current value of the output buffer in the corresponding channel. If the **output** statement has only one parameter expression, then the value of that expression shall be added to the current value of the output buffer in each channel.

The expression parameters to the **output** statement may be array-valued, in which the mapping described in the preceding paragraph is not from expressions to buffer channels, but from array value channels to buffer channels.

EXAMPLE

The following code fragment

```
asig a[2], b;  
  
. . .  
  
output(a,b);  
output(a[1],b,b);  
output(b);
```

is legal and describes an instrument that outputs three channels of sound. The first channel of output contains the value  $a[0] + a[1] + b$ , the second  $a[1] + b + b$ , and the third  $b + b + b$ .

After each a-rate pass through the instrument instantiation during a particular orchestra pass, the values in the output buffer shall be added channel-by-channel to the current values of the bus or busses referenced by the **route** expression or expressions that also reference this instrument. If there are no such **route** statements, the values in the output buffer shall be added channel-by-channel to the current values of the special bus **output\_bus**. If this is the instrument instantiation created by referencing the special bus **output\_bus** in a **send** statement, then the preceding two sentences do not hold, and instead the values in the output buffer are the output of the orchestra.

5.8.6.6.9 Spatialize

<statement> -> **spatialize** ( <expr list > );

The **spatialize** statement allows instruments to produce spatialised sound, using non-normative methods that are implementation-dependent.

The expression list shall contain four expressions. The second, third, and fourth shall not be a-rate expressions. The first expression represents the audio signal to be spatialised; the second, the azimuth (angle) from which the source sound shall apparently come, measuring in radians clockwise from 0 azimuth directly in front of the listener; the third, the elevation angle from which the sound source shall apparently come, measuring in radians upward from 0 elevation on the listener's horizontal plane; and the fourth, the distance from which the sound source shall apparently come, measuring in metres from the listener's position. Each of the four expressions shall be single-valued.

The rate of the **spatialize** statement is a-rate.

The **spatialize** statement shall be executed as follows:

At each a-rate pass through the instrument, the expressions in the expression list shall be evaluated. Then, the sound signal in the first expression shall be presented to the listener as though it has arrived from the azimuth, elevation, and distance given in the second, third, and fourth expressions. No normative requirements are placed on this spatialisation capability, although terminal implementers are encouraged to provide the maximum sophistication possible.

The sound produced via the **spatialize** statement is turned directly into orchestra output; it shall not be affected by bus routings or further manipulation within the orchestra. If multiple calls to **spatialize** occur within an orchestra, the various sounds so produced shall be mixed via simple summation after spatialisation. Similarly, if both spatialised and non-spatialised sound is produced within an orchestra, the final orchestra output of all non-spatialised sound shall be mixed via simple summation with the various spatialised sounds for presentation. The sound produced via each **spatialize** statement shall have as many channels as the global orchestra number of output channels (see subclause 5.8.5.2.4) in order to enable this mixing.

#### 5.8.6.6.10 Outbus

<statement> -> **outbus** ( <ident> , <expr list> ) ;

The **outbus** statement allows instruments to place dynamically-calculated signals on busses. The identifier parameter shall refer to the name of a bus defined with a **send** statement in the global block. The remaining expressions represent signals to place on the bus.

It is a syntax error if there are no expressions in the expression list, or if the identifier does not refer to a bus defined in the global block with a **send** statement. The number of expressions in the expression list shall be compatible with other statements making reference to the same bus, as defined in subclause 5.8.6.6.8.

The rate of the **outbus** statement is a-rate.

The **outbus** statement shall be executed as follows:

At each a-rate pass through the statement, the expression list shall be evaluated. Then, the expression values shall be added to the current values of the referenced bus. If there is more than one expression in the expression list, then each expression value shall be added to the corresponding channel of the referenced bus. If there is only one expression in the expression list, then the value of that expression shall be added to each channel of the referenced bus.

The expressions in the expression list may be array-valued, in which case the semantics are analogous to those in subclause 5.8.6.6.8.

The **outbus** statement shall not be used in an instrument that is the target of a **send** statement referencing the special bus **output\_bus**. In addition, an **outbus** statement may not write to the special bus **input\_bus**.

#### 5.8.6.6.11 Extend

<statement> -> **extend** ( <expr> ) ;

The **extend** statement allows an instrument instantiation to dynamically lengthen its duration.

The expression parameter shall not be a-rate. The expression shall be single-valued.

The rate of the **extend** statement is the rate of the expression parameter, or the rate of the guarding expression containing the statement, or the rate of the opcode containing the statement, whichever is fastest.

The **extend** statement shall be executed as follows:

At each pass through the statement at equal rate to the rate of the statement, the expression shall be evaluated. Then, the duration of the instrument instantiation shall be extended by the amount of time, in seconds, given by the value of the expression. That is, if the instrument instance had been previously scheduled to be terminated at time  $t$ , then after a call to **extend** with an expression evaluating to  $s$ , the instrument instance shall be scheduled to terminate at time  $t+s$ . If the instrument instance had no scheduled termination time (its duration was  $-1$  on instantiation), **extend** with an expression evaluating to  $s$  shall schedule termination of the instrument at time  $T + s$ , where  $T$  is the current orchestra time.

NOTE - The parameter of **extend** is specified in seconds, not in beats. If it is desirable to have time-extension dependant on tempo in a particular composition, the content author may enable this by rescaling the parameter by the current value of **gettempo()** (subclause 5.9.15.1).

**extend** may be called with a negative argument to shorten the duration of a note; if  $t+s < T$  (that is, if the negatively extended duration has already been exceeded in the instantiation), then the statement acts as the **turnoff** statement, see subclause 5.8.6.6.12.

When the **extend** statement is called, the standard name **dur** shall be updated to reflect the new duration; that is, **dur := dur + x** where **x** is the expression value of the parameter.

#### 5.8.6.6.12 Turnoff

<statement> -> **turnoff** ;

The **turnoff** statement allows an instrument instantiation to dynamically decide to terminate itself.

The rate of the **turnoff** statement is k-rate.

The **turnoff** statement shall be executed as follows:

When the **turnoff** statement is reached at k-rate, the instrument instance shall be scheduled to terminate after the following k-cycle; that is, if the current orchestra time is  $T$  and the k-pass duration  $k$ , the instrument instantiation shall be scheduled to terminate at time  $T+k$ .

The **turnoff** statement shall not update the **dur** standard name.

The **turnoff** statement shall not be executed in an instrument instance that is created as the result of a **send** statement referencing the special bus **output\_bus**.

NOTE - **turnoff** does not destroy the instantiation immediately; the instantiation is executed for one more orchestra pass, to allow the instrument time to examine the **released** variable. Instruments may call **turnoff** and then "save" themselves on the subsequent k-cycle by calling **extend**.

### 5.8.6.7 Expressions

#### 5.8.6.7.1 Syntactic form

<expr> -> <ident>  
 <expr> -> <number>  
 <expr> -> <int>  
 <expr> -> <ident> [ <expr> ]  
 <expr> -> <ident> ( <expr list> )  
 <expr> -> <ident> [ <expr> ] ( <expr list> )  
 <expr> -> <expr> ? <expr> : <expr>  
 <expr> -> <expr> <binop> <expr>  
 <expr> -> ! <expr>  
 <expr> -> - <expr>  
 <expr> -> ( <expr> )  
 <expr> -> **sasbf** ( <expr list> ) ;

<binop>-> +  
 <binop>-> -  
 <binop>-> \*  
 <binop>-> /  
 <binop>-> ==  
 <binop>-> >=

```

<binop>-> <=
<binop>-> !=
<binop>-> >
<binop>-> <
<binop>-> &&
<binop>-> ||

```

An expression can take one of several forms, the semantics of which are enumerated in the subclasses below. Each form has both rate semantics, which describe the rate of the expression in terms of the rates of the subexpressions, and value semantics, which describe the value of the expression in terms of the values of the subexpressions. The syntax above is ambiguous for many expressions; disambiguating precedence rules are given in subclass 5.8.6.7.14.

#### 5.8.6.7.2 Properties of expressions

Each expression is conceptually labelled with two properties: its rate and its width. The rate of an expression determines how fast the value of that expression might change; the width of an expression determines how many channels of sound or other data are represented by the expression. In each expression type, the rate and width of the expression are determined from the type of the expression, and perhaps from the rate and width of the component subexpressions.

NOTE - Any name declared as an array is an array-valued variable regardless of its length. That is, a variable declared as **asig name[1]** is not a single-valued variable.

#### 5.8.6.7.3 Identifier

```

<expr>      -> <ident>

```

An identifier expression denotes a storage location or locations that contain values stored in memory. It is illegal to reference an identifier that is not declared in the local instrument or opcode scope, and that is not a standard name (see subclass 5.8.6.7.14).

The rate of an identifier expression is the rate type at which the identifier was declared, or is implicitly declared in the case of standard names. The rate of a **table** identifier is i-rate.

If the identifier denotes a single-valued name (i.e., one that is not an array type), then the value of the identifier expression is the value stored in memory associated with that identifier in the current scope, and the width of the expression is 1.

If the identifier denotes an array-valued name, then the value of the identifier expression is the ordered sequence of values stored in memory and associated with that identifier in the current scope, and the width of the expression is the width of the array so denoted.

If the identifier denotes a table, then the value of the identifier expression is a reference to the table with the given name. Table references may only appear in calls to opcodes. A table reference has width 1.

#### 5.8.6.7.4 Constant value

```

<expr>      -> <number>
<expr>      -> <int>

```

A constant value expression denotes a single number.

The rate of a constant value expression is i-rate.

The width of a constant value expression is 1.

The value of a constant expression is the value of the number denoted by the constant. The value of a constant expression is always a floating-point value, whether the token or lexical expression denoting the value was an integer or floating-point token or expression.

#### 5.8.6.7.5 Array reference

<expr>           -> <ident> [ <expr> ]

An array reference expression allows the selection of one value from an array of several. The identifier in the array-reference syntax is termed the array name, and the expression the index expression. It is illegal to use an identifier in an array reference that is neither declared in the local instrument or opcode scope as an array, nor implicitly defined as an array-valued standard name or table map.

The index expression shall have width 1.

The rate of an array reference expression is the rate of the array name (which is the rate at which the array name was declared explicitly or implicitly), or the rate of the index expression, whichever is faster.

The width of an array reference expression is 1.

If the referenced array is an array-valued signal variable, then the value of the array reference expression is the value of that element of the sequence of values in the array storage corresponding to the value of the indexing expression, where element 0 corresponds to the first value in the sequence. It is a run-time error if the value of the indexing expression is less than 0, or equal to or greater than the declared size of the array. If the indexing expression is not an integer, it is rounded to the nearest integer.

If the referenced array is a table map, then the value of the array reference expression is a reference to that element of the sequence of tables corresponding to the value of the index expression, where element 0 corresponds to the first table in the sequence. It is a run-time error if the value of the indexing expression is less than 0, or equal to or greater than the declared size of the table map. If the indexing expression is not an integer, it is rounded to the nearest integer. Table references may only appear in calls to opcodes. See also the example in subclause 5.8.6.5.6.

NOTE - The syntax  $t[i]$ , where  $t$  is a table rather than a table map, is illegal. The **tableread** core opcode is used to directly access elements of a wavetable. See subclause 5.9.6.

#### 5.8.6.7.6 Opcode call

<expr>           -> <ident> ( <expr list> )

An opcode call expression allows the use of processing functionality encapsulated within an opcode.

The identifier is termed the opcode name, and the expression list the actual parameters of the opcode call expression. It is illegal to use an identifier that is not the name of a core opcode and is also not the name of a user-defined opcode declared elsewhere in the orchestra. For user-defined opcodes, the number of actual parameters shall be the same as the number of formal parameters in the opcode definition. For core opcodes without variable argument lists, the number of actual parameters required varies from opcode to opcode; see subclause 5.8.9. If a particular formal parameter in an opcode definition is an array, then the corresponding actual parameter shall be an array-typed expression of equal width. If a particular formal parameter in an opcode definition is a table, then the corresponding actual parameter shall be a table reference.

If a particular formal parameter in an opcode definition is at a particular rate, then the corresponding actual parameter expression shall not be at a faster rate.

The rate of the opcode call expression is determined according to the rules in subclause 5.8.7.7.

The width of the opcode call expression is the number of channels provided in the **return** statements in the opcode's code block.

For calls to core opcodes (see clause 5.9), in the absence of normative language specifying otherwise for a particular opcode, it is a syntax error if any of the following statements apply:

- there are fewer actual parameters in the opcode call than required formal parameters
- there are more actual parameters in the opcode call than required and optional formal parameters, and the opcode definition does not include a varargs “...” clause.
- a particular actual parameter expression is of faster rate than the corresponding formal parameter, or than the varargs formal parameter if that is the correspondence
- a particular actual parameter expression is not single-valued, or is not table-valued when the corresponding formal parameter specifies a table.

The context of the opcode call is restricted more than other expressions. When occurring within a block subsidiary to a guarding statement (**if**, **else**, or **while**), opcode calls shall not have a rate slower than the rate of the guarding expression (see subclauses 5.8.6.6.4 and 5.8.6.6.5 and 5.8.6.6.6). A call to an opcode with a particular name shall not occur within the code block of definition of that opcode, nor within the code blocks of any of the opcodes called by that opcode, or any of the opcodes called by them, etc. That is, recursive and mutually-recursive opcodes are prohibited.

If a **kopcode** opcode call occurs in a expression that runs at the **a-rate**, the first time this expression is executed in a k-cycle with regards to a particular opcode state, the **kopcode** is called, following the semantics described in this subclause. For all subsequent evaluations of the expression in the same k-cycle, the **kopcode** is not executed; instead, the return value from the first execution is used in the expression evaluation.

If an **iopcode** opcode call occurs in a expression that runs at the **a-rate** or the **k-rate**, the first time this expression is executed with regards to a particular opcode state, the **iopcode** is called, following the semantics described in this subclause. For all subsequent evaluations of the expression, the **iopcode** is not executed; instead, the return value from the first execution is used in the expression evaluation.

If a **specialop** core opcode call occurs in a expression that runs at the **a-rate**, the **k-rate** semantics of the **specialop** opcode call follows the rules for **kopcode** calls described above, while the **a-rate** semantics of the **specialop** opcode call happen at every a-cycle as described in subclause 5.9.2.

To calculate the value of an opcode call expression referencing a user-defined opcode at a particular rate, the values of the actual parameter expression shall be calculated in the order they appear in the expression list. The values of the formal parameters within the opcode scope shall be set to the values of the corresponding actual parameter expressions. If this is the first opcode call expression referencing this opcode scope, opcode storage space shall be created to store local signal variables and wavetables, the local signal variables set to 0, and the local wavetables created as discussed in subclause 5.8.6.5.2. Any global variables imported by the opcode at that rate shall be copied into the opcode storage space. The statement block of the opcode shall be executed at the current rate. The value of the opcode call expression is the value of the first **return** statement encountered when executing the opcode. The value of the opcode call expression may be array-valued (if the expression in the **return** statement is). After the end of opcode execution, any global variables exported by the opcode shall be copied into the global storage space.

NOTE - If an opcode changes and exports the value of a global variable that is imported by the calling instrument or opcode, the change in the global variable is not reflected in the caller until the next orchestra pass.

If a particular actual parameter expression in an opcode call expression is an identifier or an array-reference expression, then that parameter is a reference parameter in that call to that opcode. When the opcode statement block is executed, the final value of the formal parameter associated with that actual parameter shall be copied into the variable value denoted by the identifier or array-reference, unless the parameter is a standard name which may not be used as an lvalue (see subclause 5.8.6.6.2), in which case no copy is performed. This modification shall happen immediately after (but not until) the termination of the statement block, before any other calculation is done. Both single-value and array-value expressions may be reference parameters, but if an array-valued expression is used, the associated formal parameter shall be an array of the same length.

To calculate the value of an opcode call expression referencing a core opcode at a particular rate, the values of the actual parameter expressions shall be calculated in the order they appear in the expression list. Then, the return value of the core opcode shall be calculated according to the rules for the particular opcode given in subclause 5.8.9. If the rate of a formal parameter is slower than the rate of the opcode, the following rules apply:

- in an opcode call at the **k-rate**, actual variables associated to an **ivar** formal parameters are updated only the first time this opcode is executed with regards to a particular opcode state;
- in an opcode call at the **a-rate**, actual variables associated to an **ivar** formal parameters are updated only the first time this opcode is executed with regards to a particular opcode state; actual variables associated to a **ksig** formal parameters are updated only the first time this opcode is executed in a k-cycle with regards to a particular opcode state.

NOTE - The variables declared within the scope of a user-defined opcode are static-valued; that is, they preserve their values from call to call. The values of variables within the scope of a user-defined opcode are set to 0 before the opcode is called the first time. Each syntactically distinct call to an opcode creates one and only one opcode scope (see example in next subclause).

#### 5.8.6.7.7 Oparray call

<expr> -> <ident> [ <expr> ] ( <expr list> )

An oparray call expression allows the dynamic selection of an opcode state from a set of several, and the calculation of encapsulated functionality with respect to that opcode state.

The identifier is termed the opcode name, the expression in brackets is termed the index expression, and the expressions in the parameter list are termed the actual parameters. It is illegal to use an identifier that is not the name of a core opcode and is also not the name of a user-defined opcode declared elsewhere in the orchestra. It is also illegal to use an identifier for which oparray storage is not allocated in the local scope as described in subclause 5.8.6.5.5. For user-defined opcodes, the number of actual parameters shall be the same as the number of formal parameters in the opcode definition. For core, the number of actual parameters required varies from opcode to opcode; see subclause 5.8.9.

The index expression shall be a single-valued expression.

The rate of the oparray call expression is the rate of the opcode referenced, as determined by the rules in subclause 5.8.7. The rate of the index expression shall not be faster than the rate of the opcode referenced.

The width of the oparray call expression is the number of channels returned by **return** statements within the opcode code block.

The context of the oparray call expression is restricted in the same way as described for the opcode call expression in subclause 5.8.6.7.6. The rate semantics for oparray call execution follows the same rules described for the opcode call expression in subclause 5.8.6.7.6.

The value of the oparray call expression is determined in the same way as described for the opcode call in subclause 5.8.6.7.6, with the following exceptions and additions:

Before the values of the actual parameter expressions are calculated, the value of the index expression is calculated. It is a run-time error if the value of the index expression is not in the range [0..n-1], where n is the allocation size in the oparray definition for this oparray. If the index expression is not an integer, it is rounded to the nearest integer. The scope storage associated with the opcode name and the value of the index expression is selected from the set of oparray scopes in the local scope. The evaluation of the statement block in the referenced opcode is with regard to the selected scope. Within each oparray scope, local variables retain their values from call to call.

#### EXAMPLES

Some examples are provided to clarify the distinction between opcode calls and oparray calls.



The following user defined opcode

```

opcode inc() {
    ksig ct;

    ct = ct + 1;
    return(ct);
}

```

counts the number of times it is called.

1. After the first execution of the following code fragment

```

a = inc();
b = inc();

```

the value of **a** is 1, and the value of **b** is 1, since each call to **inc()** refers to a different scope.

2. After the first execution of the following code fragment

```

i = 0; while (i < 2) { a = inc(); i = i + 1; }

```

the value of **a** is 2, since there is only one scope for **inc()**.

3. After the first execution of the following code fragment

```

oparray inc[2];

a = inc[0]();
b = inc[0]();

```

the value of **a** is 1, and the value of **b** is 2, since each call to **inc()** refers to the same scope (since the value of the indexing expression is the same in both calls).

4. After the first execution of the following code fragment

```

oparray inc[2];

i = 0; while (i < 2) { a = inc[i](); i = i + 1; }

```

the value of **a** is 1, since each iteration refers to a different scope in the call to **inc()** (since the value of the indexing expression is 0 on the first iteration, and 1 on the second).

NOTE - Opcode calls and oparray calls referencing the same opcode may be used in the same scope. In this case, the scopes referenced by each of the opcode calls are different from any of the scopes defined in the oparray definition.

#### 5.8.6.7.8 Combination of vector and scalar elements in mathematical expressions

The subsequent subclauses (subclauses 5.8.6.7.9 through 5.8.6.7.13) describe mathematical expressions in SAOL. For each, the width of the expression is the maximum width of any of its subexpressions. For each expression type, each subexpression within an expression shall have the same width, or else width of 1. If subexpressions with width 1 and width different than 1 are combined in an expression, before the expression is computed, the subexpression(s) with width 1 shall be promoted to have the same width as the expression. That is, a width 1 expression with value **x** that is a subexpression of a width *n* expression shall be promoted to a width *n* expression where the value of each element is **x**.

For each expression type below, the semantics will be given for array-valued expressions. In each case, the semantics for the single-valued expression are the same as for an array-valued expression with width 1, except for the special cases of switch, logical AND, and logical OR, which will be described separately in those subclauses.

**5.8.6.7.9 Switch**

<expr> -> <expr> ? <expr> : <expr>

The switch expression combines values from two subexpressions based on the value of a third.

The rate of the switch expression is the rate of the fastest of the three subexpressions.

The value of the switch expression is calculated as follows: the three subexpressions are evaluated. Then, for each value of the first subexpression, if this value is non-zero, the corresponding value of the switch expression is the corresponding value of the second subexpression. If this value is zero, the corresponding value of the switch expression is the corresponding value of the third subexpression.

In the special case where all subexpressions have width 1, then the switch expression “short-circuits”: the first subexpression is evaluated, and if its value is non-zero, then the second subexpression is evaluated, and its value is the value of the switch expression. If the value of the first subexpression is zero, then the third subexpression is evaluated, and its value is the value of the switch expression. If the width of the switch expression is 1, then in no case are both the second and third subexpressions evaluated.

**5.8.6.7.10 Not**

<expr> -> ! <expr>

The not expression performs logical negation on a subexpression.

The rate of the not expression is the rate of the subexpression.

The value of the not expression is calculated as follows: the subexpression is evaluated. For each nonzero value in the subexpression, the corresponding value of the not expression is zero; for each zero value in the subexpression, the corresponding value of the not expression is 1.

**5.8.6.7.11 Negation**

<expr> -> - <expr>

The negation expression performs arithmetic negation on a subexpression.

The rate of the negation expression is the rate of the subexpression.

The value of the negation expression shall be calculated as follows: the subexpression is evaluated. For each value in the subexpression, the corresponding value of the negation expression is the arithmetic negative of the value.

**5.8.6.7.12 Binary operators**

<expr> -> <expr> <binop> <expr>

There are 12 binary operators. Each of them calculates a different function on binary subexpressions.

The value of the expression shall be calculated as follows. The two subexpressions shall be evaluated, and for each pair of values of the subexpressions, and the corresponding value of the binary expression shall be calculated according to the following table, where  $x_1$  and  $x_2$  are the values of the first and second subexpressions:

**Table 5.2 - Binary operators**

Operator	Value of expression
+	$x_1 + x_2$
-	$x_1 - x_2$
*	$x_1 x_2$

/	$x_1 / x_2$
==	if $x_1 = x_2$ , then 1, otherwise 0
>	if $x_1 > x_2$ , then 1, otherwise 0
<	if $x_1 < x_2$ , then 1, otherwise 0
<=	if $x_1 \leq x_2$ , then 1, otherwise 0
>=	if $x_1 \geq x_2$ , then 1, otherwise 0
!=	if $x_1 \neq x_2$ , then 1, otherwise 0

In each of these cases, if the particular operation would result in a NaN or Inf result (for example, division by 0), a run-time error shall result.

For the “logical and” operator **&&** in the special case where both subexpressions have width 1, the expression is calculated in a “short-circuit” fashion. The first subexpression shall be evaluated. If its value is 0, then the value of the expression is 0; if its value is nonzero, then the second subexpression shall be evaluated, and if its value is 0, then the value of the expression is 0, otherwise the value of the expression is 1.

For the “logical or” operator **||** in the special case where both subexpressions have width 1, the expression is calculated in a “short-circuit” fashion. The first subexpression shall be evaluated. If its value is nonzero, then the value of the expression is 1; if its value is 0, then the second subexpression shall be evaluated, and if its value is nonzero, then the value of the expression is 1, otherwise the value of the expression is 0.

#### 5.8.6.7.13 Parenthesis

<expr>           -> ( <expr> )

The parenthesis operator performs no new calculation, but allows the specification of arithmetic grouping.

The rate of the parenthesis expression is the rate of the subexpression.

The width of the parenthesis expression is the width of the subexpression.

The value of the parenthesis expression is the value of the subexpression.

#### 5.8.6.7.14 Order of operations

Expressions bind in the order prescribed in the following table. That is, operations listed higher in the table are performed before operations lower in the table whenever the ordering is syntactically ambiguous. Operations listed on the first and last row associate right-to-left, that is, the rightmost expression is performed first. Operations listed on the remaining rows associate left-to-right, that is, the leftmost expression is performed first.

**Table 5.3 - Order of operations**

Operator	Function
!, -	not, unary negation
*, /	multiply, divide
+, -	add, subtract
<, >, <=, >=	relational
==, !=	equality
&&	logical and
	logical or
?:	switch

#### 5.8.6.7.15 SASBF synthesis

<expr> -> **sasbf** ( <expr list> ) ;

The **sasbf** expression allows the use of the DLS-compatible bank synthesis procedure (see subclause 5.13) within a SAOL instrument. It shall not be used in a Object type 3 bitstream and capability for executing it does not have to be provided by a Object type 3 decoder. The parameter list shall have two, three, or four expressions. All shall be single-valued i-rate expressions.

1. The first expression shall correspond to the MIDI pitch desired for synthesis. If this value is not an integer, it shall be rounded to the nearest integer. It is a run-time error if this value is less than 1 or greater than 128.
2. The second expression shall correspond to the MIDI velocity desired for synthesis. If this value is not an integer, it shall be rounded to the nearest integer. It is a run-time error if this value is less than 0 or greater than 128.
3. The third expression, if given, corresponds to the MIDI preset number. If there are less than three expressions, the MIDI preset number is the default value given by  $p \bmod 128$ , where  $p$  is the preset number of the instrument to which the MIDI noteon that created the note instance was directed, or the lowest-numbered instrument preset number for the instrument containing this statement (subclause 5.8.6.4), if the note was not triggered with a MIDI event.
4. The fourth expression, if given, corresponds to the MIDI bank number. If there are less than four expressions in the list, the MIDI bank number is the default value given by  $\text{floor}(p/128) + 1$ , where  $p$  is the preset number to which the MIDI noteon that created the note instance was directed, or the lowest-numbered instrument preset number for the instrument (subclause 5.8.6.4), if the note was not triggered by a MIDI event. It is a syntax error if there are less than four expressions and no instrument preset number is provided.

The **sasbf** expression is an a-rate expression. The **sasbf** expression has two channels defined by the bank-synthesis procedure as described in subclause 5.13.

The value of the **sasbf** expression is calculated as follows:

On the first execution of the expression, each expression in the expression list shall be evaluated. Using these values, one note of synthesis shall be dispatched to the wavetable bank synthesis procedure described in clause 5.13.

On the first and each subsequent a-rate pass through the **sasbf** expression, the value of the expression shall be the next pair of audio samples from the stereo wavetable bank synthesis process.

During the wavetable bank synthesis process, the values of MIDI controllers and other continuous values on the current channel and note shall be respected. These values are not passed into the **sasbf** expression list, but are made available to the SASBF synthesiser in an implementation-dependent way. If the values of the global MIDIctrl[] standard name are changed by any instrument (see subclause 5.8.6.8.9), the new values shall be respected by all SASBF synthesis processes.

NOTE - Certain MIDI instructions (such as the Registered Parameter Number mechanism, see [MIDI]), cannot be properly understood on an event-by-event basis; rather, their semantics are understood to take effect when the entire RPN change is completed. To ensure this, all control events, whether or not they have SAOL semantics (see subclause 5.14.3.2), shall be passed through the SAOL scheduler to the **sasbf** synthesis processes in their original order.

If the instrument instance containing a particular **sasbf** expression was not instantiated in response to a MIDI event, then that instance is on no channel, and so the SASBF synthesis for that expression cannot be controlled by MIDI-based continuous controllers.

Each syntactically different instance of the **sasbf** expression results in one instance of a SASBF note synthesis procedure. There is no mechanism for interleaving samples from a single **sasbf** expression in multiple lines, or for instantiating multiple bank-synthesis procedures with one syntactic expression. **sasbf** is not an opcode and is not permitted to be used as an oparray construction (see subclause 5.8.6.7.7).

The value of the **released** standard name in the instrument instance (see subclause 5.8.6.8.16) containing the call to **sasbf** shall be made available to each SASBF process in an implementation-dependent way. The SASBF process shall use this flag to determine when to begin synthesis of the release portion of the given note. If a particular SASBF instance needs to extend the duration past the release time, it shall extend the note by one k-cycle in the manner of the **extend** statement (subclause 5.8.6.6.11). If, on the next k-cycle, the SASBF instance is still not finished, it may extend the note by a further k-cycle, and so on.

If multiple SASBF instances in an instrument each require extended duration, together they shall extend the duration by one k-cycle; the duration shall not be extended by one k-cycle per SASBF instance.

The SASBF synthesis process for each note terminates when the instrument instance containing this **sasbf** expression is destroyed. There is no mechanism for ending the SASBF synthesis earlier than this.

#### EXAMPLE

The following instrument layers two notes using the wavetable synthesiser and filters the result.

```
instr layer(mp, vel) preset 12 {
  asig a1[2], a2[2], i;
  oparray bandpass[2];

  a1 = sasbf(mp, vel);
  a2 = a1 + sasbf(mp, vel, 47, 2) / 2; // second note quieter
  i = 0;
  while (i < 2) {
    a2[i] = bandpass[i](a2[i], 400, 100);
    i = i + 1;
  }
  output(a2);
}
```

The two instances of **sasbf** operate simultaneously and in parallel. The first synthesises sound from preset 13, bank 1 (since this is the preset number to which the instrument responds); the second, from preset 48, bank 2. They return audio signals that are summed together and manipulated with the **bandpass** core opcode.

### 5.8.6.8 Standard names

#### 5.8.6.8.1 Definition

Not all identifiers to be referenced in an instrument or opcode are required to be declared as variables. Several identifiers, listed in this subclause, are termed standard names, shall not be used as variables, and have fixed semantics that shall be implemented in a compliant SAOL decoder. Standard names may otherwise be used as variables, embedded in expressions, etc. in any SAOL instrument or opcode. However, the semantics of using a standard name as an lvalue are undefined.

The implicit definition of each standard name, showing the rate semantics and width of that standard name, is listed, and the semantics of the value of the standard name specified in the subsequent subclauses.

#### 5.8.6.8.2 k\_rate

```
ivar k_rate
```

The standard name **k\_rate** shall contain the control rate of the orchestra, in Hz.

#### 5.8.6.8.3 s\_rate

```
ivar s_rate
```

The standard name **s\_rate** shall contain the sampling rate of the orchestra, in Hz.

#### 5.8.6.8.4 inchan

ivar inchan

The standard name **inchan**, in each scope, shall contain the number of channels of input being provided to the instrument instantiation with which that scope is associated. “Associated” shall be taken to mean, for instrument code, the instrument instantiation for which the scope memory was created; for opcode code, the instrument instantiation that called the opcode, or called the opcode’s caller, etc.

Different instances of the same instrument may have different numbers of input channels if, for example, they are the targets of different send statements. Instructions for calculating the value of this standard name are provided in subclause 5.7.3.3.5.2

#### 5.8.6.8.5 outchan

ivar outchan

The standard name **outchan** shall contain the number of channels of output being produced by the orchestra (not by the instrument instance).

#### 5.8.6.8.6 time

ivar time

The standard name **time**, in each scope, shall contain the time at which the instrument instantiation associated with that scope was created.

NOTE - If the “event time” of an instrument (for example, a score event more precisely timed than one control period) and the actual instantiation time differ, the name time shall contain the latter time, not the former.

#### 5.8.6.8.7 dur

ivar dur

The standard name **dur**, in each scope, shall contain the duration of the instrument instantiation as originally created, and as potentially revised by use of the **extend** statement (subclause 5.8.6.6.11), or  $-1$  if the duration was not known at instantiation.

Although **dur** is an i-rate variable, it may be changed during the duration of an instrument instance through the **extend** statement or through tempo changes. In this case, the value of expressions at the **ivar** do not change; expressions are only evaluated according to the rules in subclause 5.8.6.6.

#### 5.8.6.8.8 itime

ksig itime

The standard name **itime**, in each scope, shall contain the elapsed time of the instrument instance. That is, on the first k-cycle through an instrument, **itime** shall be 0, and thereafter shall be incremented by  $1/\mathbf{KR}$ , where **KR** is the orchestra sampling rate, at the beginning of each k-rate pass.

#### 5.8.6.8.9 MIDICTrl

ksig MIDICTrl[128]

The **MIDICTrl** standard variable shall contain, for each scope, the current values of the MIDI controllers on the channel corresponding to the channel to which the instrument instantiation associated with that scope is assigned. See clause 5.13 for more details on MIDI control of orchestras.

Instruments may use **MIDICtrl** as an lvalue, that is, to assign new values to it using the = statement (subclause 5.8.6.6.2). In this case, when an instrument assigns to **MIDICtrl**, the value for the indicated controller shall be changed on the channel to which the instrument instance associated with that scope is assigned. The value of **MIDICtrl** is changed in all other instrument instances associated with that channel to the new value, and this change shall take effect the next time each of these instrument instances is executed at the k-rate (see subclause 5.7.3.3.6, list item 10).

**MIDICtrl[64]** is a special controller. It is normatively defined as the sustain controller, and has a special relationship with the MIDI **noteoff** instruction, see subclause 5.14.3.2.4.

#### 5.8.6.8.10 MIDItouch

ksig MIDItouch

The **MIDItouch** standard variable shall contain, for each scope, the current value of the MIDI aftertouch on the note that caused the associated instrument instantiation to be created. See clause 5.13 for more details on MIDI control of orchestras.

#### 5.8.6.8.11 MIDIBend

ksig MIDIBend

The **MIDIBend** standard variable shall contain, for each scope, the current value of the MIDI pitchbend on the channel corresponding to the channel to which the instrument instantiation associated with that scope is assigned.

#### 5.8.6.8.12 channel

ivar channel

The **channel** standard name contains the extended MIDI channel of the note responsible for creating the current instrument instance. See subclause 5.14.3.2.2.

#### 5.8.6.8.13 preset

ivar preset

The **preset** standard name contains the SAOL preset number (which is one less than the MIDI preset number) of the note responsible for creating the current instrument instance. The **preset** standard name does not contain "all" of the preset numbers for the current instrument, only the one that led to the instantiation of the current instance.

#### 5.8.6.8.14 input

asig input[inchannels]

The **input** standard variable shall contain, for each scope, the input signal or signals being provided to the instrument instantiation through the **send** instruction. See subclause 5.8.5.5.

#### 5.8.6.8.15 inGroup

ivar inGroup[inchannels]

The **inGroup** standard variable shall contain, for each scope, the grouping of the input signals being provided to the instrument instantiation. See subclause 5.8.5.5.

#### 5.8.6.8.16 released

ksig released

The **released** standard name shall contain, for each scope, 1 if and only if the instrument instantiation associated with the scope is scheduled to be destroyed at the end of the current orchestra pass. Otherwise, **released** shall contain 0. See subclause 5.7.3.3.6, list item 3.

### 5.8.6.8.17 **cpuload**

```
ksig cpuload
```

The **cpuload** standard name shall contain, for each scope, a measure of the recent CPU load on the CPU most strongly associated with the instrument instantiation associated with the scope. If the instrument instantiation is running entirely on one CPU, then that CPU shall be measured; if the instrument instantiation is running on multiple CPUs, then the exact measurement procedure is nonnormative.

The measure of CPU load shall be as a percentage of real-time capability: if the CPU is entirely loaded and cannot perform any more calculations without slipping out of real-time performance, the value of **cpuload** shall be 1 on that CPU at that k-cycle. If the CPU is entirely unloaded and is not performing any calculations, the value of **cpuload** shall be 0 on that CPU at that k-cycle. If the CPU is half-loaded, and could perform twice as many calculations in real-time as it is currently performing, the value of **cpuload** shall be 0.5 on that CPU at that k-cycle.

The exact calculation method, time window, recency, etc. of the CPU load is left to implementers.

### 5.8.6.8.18 **position**

```
imports ksig position[3]
```

The **position** name contains the absolute position of the node responsible for creating the current orchestra in the BIFS scene graph (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82). The position is given by the current value of the **position** field of the **Sound** node that is the ancestor of this node in the scene graph, as transformed by its ancestors (that is, the final position in world co-ordinates of the **Sound** node). The value is global and shared by all instruments; it may not be changed by the orchestra.

### 5.8.6.8.19 **direction**

```
ksig direction[3]
```

The **direction** name contains the orientation of the node responsible for creating the current orchestra in the BIFS scene graph (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82). The direction is given by the current value of the **direction** field of the **Sound** node that is the ancestor of this node in the scene graph, as transformed by its ancestors (that is, the final direction in world co-ordinates of the **Sound** node). The value is global and shared by all instruments; it may not be changed by the orchestra.

### 5.8.6.8.20 **listenerPosition**

```
ksig listenerPosition[3]
```

The **listenerPosition** name contains the absolute position of the listener in the BIFS scene graph (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82). The position is given by the current value of the **position** field of the active **ListeningPoint** node in the scene graph, as transformed by its ancestors (that is, the final position in world co-ordinates of the **ListeningPoint** node).

### 5.8.6.8.21 **listenerDirection**

```
ksig listenerDirection[3]
```

The **listenerDirection** name contains the orientation of the listener in the BIFS scene graph (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82). The direction is given by the current value of the **direction** field of the active **ListeningPoint** node in the scene graph, as transformed by its ancestors (that is, the final direction in world co-ordinates of the **ListeningPoint** node).



**5.8.6.8.22 minFront**

ksig minFront

The **minFront** standard name gives one parameter of the sound radiation pattern of the sound that the current node is a part of. This parameter, and its semantics, are defined by the **minFront** field of the **Sound** node of which this node is an ancestor (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82).

**5.8.6.8.23 maxFront**

ksig maxFront

The **maxFront** standard name gives one parameter of the sound radiation pattern of the sound that the current node is a part of. This parameter, and its semantics, are defined by the **maxFront** field of the **Sound** node of which this node is an ancestor (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82).

**5.8.6.8.24 minBack**

ksig minBack

The **minBack** standard name gives one parameter of the sound radiation pattern of the sound that the current node is a part of. This parameter, and its semantics, are defined by the **minBack** field of the **Sound** node of which this node is an ancestor (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82).

**5.8.6.8.25 maxBack**

ksig maxBack

The **maxBack** standard name gives one parameter of the sound radiation pattern of the sound that the current node is a part of. This parameter, and its semantics, are defined by the **maxBack** field of the **Sound** node of which this node is an ancestor (see ISO/IEC 14496-1 clause 9, particularly subclause 9.4.2.82).

**5.8.6.8.26 params**

imports exports ksig params[128]

The **params** standard name is shared globally by all instruments. At each k-cycle of the orchestra, it shall contain the current values of the **params** field of the BIFS **AudioFX** node responsible for instantiating the current orchestra. If the orchestra is created by an **AudioSource** node rather than an **AudioFX** node, the value of **params** shall be 0 on every channel. See subclause 5.15.3 for more details.

Instruments may use **params** as an lvalue, that is, to assign new values to it using the = statement (subclause 5.8.6.6.2). In this case, when an instrument assigns to **params**, the value for the indicated controller shall be changed on the channel to which the instrument instance associated with that scope is assigned. The value of **params** is changed in all other instrument instances associated with that channel to the new value, and this change shall take effect the next time each of these instrument instances is executed at the k-rate (see subclause 5.7.3.3.6, list item 10).

**5.8.7 Opcode definition****5.8.7.1 Syntactic Form**

This subclause describes the definition of new opcodes. Bitstream authors may create their own opcodes according to these rules in order to encapsulate functionality and simplify instruments and the content authoring process.

```
<opcode definition>    -> <opcode rate> <ident> ( <formal param list> ) {
                        <opcode var declarations>
```

```

        <opcode statement block>
    }

```

```

<opcode rate>      -> aopcode
<opcode rate>      -> kopcode
<opcode rate>      -> iopcode
<opcode rate>      -> opcode

```

An opcode definition has several elements. In order, they are

1. A rate tag that defines the rate at which the opcode executes, or indicates that the opcode is rate-polymorphic,
2. An identifier that defines the name of the opcode,
3. A list of zero or more formal parameters of the opcode,
4. A list of zero or more opcode variable declarations,
5. A block of statements defining the executable functionality of the opcode.

#### 5.8.7.2 Rate tag

The rate tag describes the rate at which the opcode is to run, or else indicates that the opcode is rate-polymorphic. The four rate tags are

1. **iopcode**, indicating that the opcode runs at i-rate,
2. **kopcode**, indicating that the opcode runs at k-rate,
3. **aopcode**, indicating that the opcode runs at i-rate,
4. **opcode**, indicating that the opcode is rate-polymorphic.

See subclause 5.8.7.7 for instructions on determining the rate of a rate-polymorphic opcode.

#### 5.8.7.3 Opcode name

Any identifier may serve as the opcode name except that the opcode name shall not be a reserved word (see subclause 5.8.8), the name of one of the core opcodes listed in clause 5.9, or the name of one of the core wavetable generators listed in clause 5.10. An opcode name may be the same as the name of a variable in local or global scope; there is no ambiguity so created, since the contexts in which opcode names may occur are very restricted.

No two instruments or opcodes in an orchestra shall have the same name.

#### 5.8.7.4 Formal parameter list

##### 5.8.7.4.1 Syntactic form

```

<formal param list>      -> <formal param> [ , <formal param list> ]
<formal param list>      -> <NULL>

<formal param>           -> <opcode variable rate> <name>
<formal param>           -> table <ident>

```

<opcode variable rate> -> **asig**  
 <opcode variable rate> -> **ksig**  
 <opcode variable rate> -> **ivar**  
 <opcode variable rate> -> **xsig**

<name> as defined in subclause 5.8.5.3.2.

The formal parameter list defines the calling interface to the opcode. Each formal parameter in the list has a name, a rate type, and may have an array width. If the array width is the special token **inchannels**, then the array width shall be the same as the number of input channels to the associated instrument instantiation (in the sense of subclause 5.15.2); if the array width is the special token **outchannels**, then the array width shall be the same as the number of orchestra output channels as defined in the global block.

There is no way to create user-defined opcodes with variable number of arguments in SAOL, although certain of the core opcodes have this property.

Within the opcode statement block, formal parameters may be used like any other variable. The rate tag of each formal parameter defines the rate of the variable. If an opcode is declared to be at a particular rate, then no formal parameter shall be declared faster than that rate.

There is a special rate tag **xsig** that allows formal parameters to be rate-polymorphic, see subclause 5.8.7.7.2. **xsig** shall not be the rate tag of any formal parameter unless the opcode is of type **opcode**.

## 5.8.7.5 Opcode variable declarations

### 5.8.7.5.1 Syntactic form

<opcode var declarations> -> <opcode var declaration> [ <opcode var declarations> ]  
 <opcode var declarations> -> **<NULL>**

<opcode var declaration> -> <instr variable declaration>  
 <opcode var declaration> -> **xsig** <namelist> ;

<instr variable declaration> as defined in subclause 5.8.6.5.1.  
 <namelist> as defined in subclause 5.8.5.3.2.

The syntax and semantics of opcode variable declarations are the same as those of instrument variable declarations as given in subclause 5.8.6.5, with the following exceptions and additions:

The opcode variable names are available only within the scope of the opcode containing them. The instrument variable declarations for the instrument instantiation associated with the opcode call are not within the scope of an opcode, and references to these names shall not be made unless the names are also explicitly declared within the opcode, in which case the variable denoted is a different one. However, standard names (subclause 5.8.6.8) are within the scope of every opcode, may be referenced within opcodes, and shall have the semantics given in subclause 5.8.6.8 as applied to the instrument instantiation associated with the opcode call.

The values of opcode variables are static, and are preserved from call to call referencing a particular opcode state. The values of opcode variables shall be set to 0 in an opcode state before the first call referencing that state is executed.

The **tablemap** declaration may reference any tables declared in the local scope, as well as any formal parameters that are tables.

The values of opcode variables in different states of the same opcode (due to different syntactic uses of opcode expressions, or different indexing expressions in oparray expressions) are separate and have no relationship to one another.

There is a special rate tag called **xsig** that may be used to declare opcode variables in rate-polymorphic opcodes, see subclause 5.8.7.7.2. **xsig** shall not be the rate tag of any variable unless the opcode type is **opcode**.

### 5.8.7.6 Opcode statement block

#### 5.8.7.6.1 Syntactic form

<opcode statement block> -> <opcode statement> [ <opcode statement block> ]  
<opcode statement block> -> <NULL>

<opcode statement> -> <statement>  
<opcode statement> -> **return** ( <expr list> ) ;

<statement> as defined in subclause 5.8.6.6.1.

<expr list> as defined in subclause 5.8.6.6.

The syntax and semantics of statements in opcodes are the same as the syntax and semantics of statements in instruments, with the following exceptions and additions:

No statement in an opcode shall be faster than the rate of the opcode, as defined in subclause 5.8.7.7. A statement that is slower than the rate of the opcode shall be executed as described in subclause 5.8.7.7.3.

The assignment statement and the values of all variables refer to the opcode state associated with this particular call to this opcode, or associated with a particular indexing expression in an oparray call.

There is a special statement called **return** that is used in opcodes. This statement allows opcodes to return values back to their callers.

#### 5.8.7.6.2 Return statement

The **return** statement allows opcodes to return values back to their callers.

The expression parameter list may contain both single-valued and array-valued expressions.

The rate of the **return** statement is the rate of the opcode containing it. No expression in the expression parameter list shall be faster than the rate of the opcode.

The **return** statement shall be evaluated as follows. Each expression in the expression parameter list is evaluated, in the order they occur in the list. The return value of the opcode is the array-value formed by sequencing the values of the expression parameters. In the case that there is only one expression parameter that is a single-valued expression, then the return value of the opcode is the single value of that expression. The return value denoted by every **return** statement within an opcode shall have the same width (although it is permissible for them to differ in the number of expressions, so long as the sum of the widths of the expressions is equal).

If an opcode has a return statements with an <expr list> of length 0, it is considered to have width 1 for the purposes of this subclause. If the opcode executes this **return** statement, the return value is undefined.

If an opcode has no **return** statements, it has a width of 1, and its return value is undefined. If no **return** statement is encountered during an opcode execution, control is returned to the calling instrument or opcode at the end of the statement block.

After a **return** statement is encountered, no further statements in the opcode are evaluated, and control returns immediately to the calling instrument or opcode.

### 5.8.7.7 Opcode rate

#### 5.8.7.7.1 Introduction

This subclause describes the rules for determining the rate of a call to an opcode, and the semantics of the special tags **opcode** and **xsig**.

The rate of an opcode call depends on the type of the opcode, as follows:

1. If the opcode type is **aopcode**, calls to the opcode are a-rate.
2. If the opcode type is **kopcode**, calls to the opcode are k-rate.
3. If the opcode type is **iopcode**, calls to the opcode are i-rate.
4. If the opcode type is **opcode**, the opcode is rate-polymorphic, and the rate is as described in the next subclause.

#### 5.8.7.7.2 Rate-polymorphic opcodes

Opcodes that are rate-polymorphic take their rates from the context in which they are called. This allows the same opcode statement block to apply to multiple calling rate contexts. Without such a construct, three versions of each opcode of this sort would have to be created and used, depending on the context.

The rate of an **opcode** opcode for a particular call is the rate of the fastest actual parameter expression (not formal parameter expression) in that call, or the rate of the fastest formal parameter in the opcode definition, or the rate of the fastest guarding **if**, **while**, or **else** expression surrounding the opcode call, or the rate of the opcode enclosing the opcode call, whichever is fastest. If an **opcode** opcode has no non-**table** parameters, and is not enclosed in a guarded block or a opcode call, then it is a **kopcode** by default.

Rate-polymorphic opcodes may contain variable declarations and formal parameter declarations using the special rate tag **xsig**. A formal parameter of type **xsig**, for a particular call to that opcode, has the same rate as the actual parameter expression in the calling expression to which it corresponds. A variable of type **xsig**, for a particular call to that opcode, has the same rate as the opcode.

Rate-polymorphic opcodes shall not contain variable declarations and statements faster than the fastest formal parameter in the opcode declaration. In particular an opcode with all **xsig** formal parameters shall not contain variable declarations other than **xsig** and **ivar** and it shall not contain statements at a defined rate faster than the initialization rate.

#### EXAMPLES

Given the following opcode definition:

```
opcode xop(ksig p1, xsig p2) {
    xsig v1;
    . . .
}
```

1. For the following code fragment

```
ksig k;

k = xop(1,2);
```

the rate of the opcode call is k-rate, since the formal parameter **p1** is faster than either of the actual parameters. The rate of **p2** within the call to **xop()** is i-rate, matching the actual parameter. The rate of **v1** within **xop()** is k-rate.

2. For the following code fragment

```
asig a1, a2;  
  
a1 = xop(1,a2);
```

the rate of the opcode call is a-rate, since the actual parameter **a2** is faster than either of the formal parameters. The rates of **p2** and **v1** within the call to **xop()** are k-rate and a-rate respectively.

3. For the following code fragment

```
ksig k;  
asig a;  
  
k = xop(1,a)
```

there is a rate mismatch error, since the opcode call is a-rate, and thus shall not be assigned to a k-rate lvalue.

4. For the following code fragment

```
ksig k;  
asig a1,a2;  
oparray xop[10];  
  
a1 = 0; while (a1 < 10) {  
    a2 = a2 + xop[a1](1,k);  
    a1 = a1 + 1;  
}
```

the rate of the oparray call is a-rate, since the rate of the guarding expression is faster than any of the formal parameters or actual parameters. The rates of **p2** and **v1** within **xop()** are a-rate as well.

### 5.8.7.7.3 Shared variables and statements slower than the rate of the opcode

This subclause describes the rules for updating shared variables and for executing statements inside an opcode that are slower than the rate of the opcode.

In a **kopcode**, statements at the initialization rate are executed at the first call to the opcode with regard to a particular opcode state. At this call the **imports**, **exports** and **imports exports** ivars and tables are updated, as described in 5.8.6.5.3 and 5.8.6.5.4, and any wavetable generations are executed, as described in 5.8.6.5.2.

In an **aopcode**, statements at the initialization rate are executed at the first call to the opcode with regard to a particular opcode state. At this call the **imports**, **exports** and **imports exports** ivars and tables are updated, as described in 5.8.6.5.3 and 5.8.6.5.4, and any wavetable generations are executed, as described in 5.8.6.5.2. Statements at the control rate are executed at the first call of every k-cycle to the opcode with regard to a particular opcode state. At this call the **imports**, **exports** and **imports exports** ksig and tables are updated, as described in 5.8.6.5.3 and 5.8.6.5.4.

In an **opcode**, once the rate is determined as specified in subclause 5.8.7.7.2, the same rules apply for the cases of call at **k-rate** or **a-rate** respectively.

## 5.8.8 Template declaration

### 5.8.8.1 Syntactic form

```
<template declaration> -> template < <identlist> > [ preset <maplist> ] ( <identlist> )  
                           map { <identlist> } with { <maplist> }  
                           { <instr variable declarations> <block> }
```

```
<maplist> -> < <expr list> > , <maplist>
```

```
<maplist> -> < <expr list> >
```

<identlist> as given in subclause 5.8.5.4.  
 <namelist> as given in subclause 5.8.5.3.2.  
 <instr variable declarations> as given in subclause 5.8.6.5.1.  
 <expr list> as given in subclause 5.8.6.6.1.  
 <block> as given in subclause 5.8.6.6.1.

A template declaration allows the concise declaration of multiple instruments that are similar in processing structure and syntax, but differ in only a few key expressions or wavetable names.

### 5.8.8.2 Semantics

The first identifier list contains the names for the instruments declared with the template. There shall be at least one identifier in this list. The first optional maplist contains a list of the preset number lists to be associated with each instrument in the template. This maplist may be omitted, in which case there are no preset numbers associated with the template instruments. If the maplist is present, it shall contain as many lists as there are instrument names in the first identified list. The second identifier list contains the pfields for the template declaration. Each instrument declared with the template has the same list of pfields. The third identifier list contains a list of template variables that are to be replaced in the subsequent code block with expressions from the second (first required) map list. There may be no identifiers in this list, in which case each instrument declared by the template is exactly the same.

The map list takes the form of a list of lists. This list shall have as many elements as template variables declared in the third identifier list. Each sublist is a list of expressions, and shall have as many elements as instrument names in the first identifier list. The first (optional) maplist shall not contain identifiers, only numeric values.

### 5.8.8.3 Template instrument definitions

As many instruments are defined by the template definition as there are names in the first identifier list. To describe each of the instruments, the identifiers described in the third (template variable) list are replaced in turn by each of the expressions from the map list.

That is, to construct the code for the first instrument, the code block given is processed by replacing the first template variable with the first expression from the first map list sublist, the second template variable with the first expression from the second map list sublist, the third template variable with the first expression from the third map list sublist, and so on. To construct the code for the second instrument, the code block given is processed by replacing the first template variable with the *second* expression from the first map list sublist, the second template variable with the *second* expression from the second map list sublist, the third template variable with the second expression from the third map list sublist, and so on.

This code-block processing occurs before any other syntax-checking or rate-checking of the elements of the instruments so defined. That is, the template variables are not true signal variables, and do not need to be declared in the variable declaration block. Once the code-block processing and template expansion is complete, the resulting instruments are treated as any other instruments in the orchestra.

#### EXAMPLE

The following template declaration:

```
template <oneharm, threeharm> preset <3,4>,<24> (p)
  map {pitch,t,bar} with
  { <440, harm1, mysig>, <p, harm2, mysig * mysig + 2> } {

  table harm1(harm,4096,1);
  table harm3(harm,4096,3,2,1);
  asig mysig;

  mysig = oscil(t,pitch,-1);

  mysig = bar *3;
  output(mysig);
}
```

declares exactly the same two instruments as the following two instrument declarations:

```
instr oneharm(p) preset 3 4 {
  table harm1(harm,4096,1);
  table harm3(harm,4096,3,2,1);
  asig mysig;

  mysig = oscil(harm1,440,-1);

  mysig = mysig * 3;
  output(mysig);
}

instr threeharm(p) preset 24 {
  table harm1(harm,4096,1);
  table harm3(harm,4096,3,2,1);
  asig mysig;

  mysig = oscil(harm3,p,-1);

  mysig = (mysig * mysig + 2) * 3; // notice embedding of template expression
  output(mysig);
}
```

### 5.8.9 Reserved words

The following words are reserved, and shall not be used as identifiers in a SAOL orchestra or score.

**aopcode asig else exports extend global if imports inchannels instr interp iopcode ivar kopcode krate  
ksig map oparray opcode outbus outchannels output preset return route sasbf send sequence  
spatialize srate table tablemap template turnoff while with xsig**

Also, variable names starting with `_sym_` are reserved for implementation-specific use (for example, bitstream detokenisation) and shall not be the name of any instrument, signal variable, wavetable, or user-defined opcode in the orchestra.

## 5.9 SAOL core opcode definitions and semantics

### 5.9.1 Introduction

This clause describes the definitions and normative semantics for each of the core opcodes in SAOL. All core opcodes shall be implemented in every terminal that can decode Object Type 3 or 4.

For each core opcode, the following is described:

- The prototype, showing the rate of the opcode, the parameters which are required in a call to this opcode, and the rates of these parameters.
- The normative semantics of the return value. These semantics describe how to calculate the return value for each call to that opcode.
- The normative semantics of any side effects of the core opcode.

### 5.9.2 Specialop type

There is a special rate type for certain core opcodes called **specialop**. This rate type tag is not an actual lexical element of the SAOL language, and shall not appear in a SAOL orchestra, but is used in subsequent subclasses as a shorthand for core opcodes with these particular semantics.



Core opcodes with rate type **specialop** describe functions that map from one or more a-rate signals into a k-rate signal. That is, they have one or more parameters that vary at the a-rate, and they have normative semantics described at the a-rate, but they only return values and/or have side effects at the k-rate. When using these opcodes in expressions, they are treated as **kopcode** opcodes for the purposes of determining rate-mismatch errors (except that it is not a rate-mismatch error to pass them an a-rate signal), and as both **kopcode** and **aopcode** opcodes for the purposes of determining when to execute them.

An expression that is prohibited by the rules in subclauses 5.8.6.6 and 5.8.6.7 from being k-rate or a-rate shall not be of type **specialop**. Otherwise, an expression with one or more **specialop** components is also of type **specialop**. A **specialop** expression shall only occur in a guarded context (the code block of an **if**, or **else** statement) if the guard expression is also a **specialop** expression. A **specialop** expression shall not occur in the guarding expression or code block of the **while** statement.

The core opcodes with this type are: **fft**, **rms**, **sblock**, **downsamp**, and **decimate**.

#### EXAMPLE

Given the following code block:

```

asig x;
ksig y;

x = ...;
y = rms(x);           // #1
y = rms(x * 12);     // #2

if (rms(x)) {        // #3
  y = kline(...);   // #4
}

if (x) {             // #5
  y = rms(x);        // #6
}

if (rms(x) > y) {    // #7
  y = rms(x);        // #8
}

```

the following things are true. Statement #1 is a **ksig** rate statement; however, the right-side expression is a **specialop** expression and is executed at both the k-rate and at the a-rate (the assignment only occurs at the k-rate). Statement #2 is just like statement #1; the **x \* 12** expression is a **specialop** expression. Statement #3 is legal, and is a **ksig** rate statement. The guard expression is a **specialop** expression and is evaluated at both the a-rate and the k-rate, and on each k-cycle on which it is nonzero, statement #4 is executed at the k-rate. Statement #5 contains a rate-mismatch error. The guard rate of the **if** statement is an a-rate expression, but statement #6 is only at the k-rate for the purposes of determining rate-mismatch errors. Statement #7 is legal. The guard expression is a **specialop** expression.

### 5.9.3 List of core opcodes

The several core opcodes are described in the subsequent subclauses. They are divided by category into major subclauses, but there is no normative significance in this division; it is only for clarity of presentation.

**Math functions** `int`, `frac`, `dbamp`, `ampdb`, `abs`, `sgn`, `exp`, `log`, `sqrt`, `sin`, `cos`, `atan`, `pow`, `log10`, `asin`, `acos`, `floor`, `ceil`, `min`, `max`

**Pitch converters** `gettune`, `settune`, `octpch`, `pchoct`, `cspch`, `pchcps`, `cpsoct`, `octcps`, `midipch`, `pchmidi`, `midioct`, `octmidi`, `midicps`, `cpsmidi`

**Table operations** `ftlen`, `ftloop`, `ftloopend`, `ftsr`, `ftbasecps`, `ftsetloop`, `ftsetend`, `ftsetbase`, `ftsetsr`, `tableread`, `tablewrite`, `oscil`, `loscil`, `doscil`, `koscil`

<b>Signal generators</b>	kline, aline, kexpon, aexpon, kphasor, aphasor, pluck, buzz, grain
<b>Noise generators</b>	irand, krand, arand, ilinrand, klinrand, alinrand, iexprand, kexprand, aexprand, kpoissonrand, apoissonrand, igaussrand, kgaussrand, agaussrand
<b>Filters</b>	port, hipass, lopass, bandpass, bandstop, biquad, allpass, comb, fir, iir, firt, iirt
<b>Spectral analysis</b>	fft, ifft
<b>Gain control</b>	rms, gain, balance, compressor
<b>Sample conversion</b>	decimate, upsamp, downsamp, samphold, sblock
<b>Delays</b>	delay, delay1, fracedelay
<b>Effects</b>	reverb, chorus, flange, speedt, fx_speedc
<b>Tempo changes</b>	gettempo, settempo

For each core opcode, an opcode prototype is given. This shows the rate of the opcode, the number of required and optional formal parameters and the rate of each of the formal parameters. Certain parameters to certain core opcodes are presented in brackets, in which case that formal parameter is optional. Certain opcodes use the “...” notation, which means that the opcode can process an arbitrary number of parameters. The “...” is tagged with a rate for such opcodes, which is then the rate type of all of the parameters matching the varargs parameter. If there is not normative language for a particular opcode that specifies otherwise, it is a syntax error if any of the following statements apply:

- there are fewer actual parameters in the opcode call than required formal parameters
- there are more actual parameters in the opcode call than required and optional formal parameters, and the opcode definition does not include a varargs “...” section
- a particular actual parameter expression is of faster rate than the corresponding formal parameter, or than the varargs formal parameter if that is the correspondence
- a particular actual parameter expression is not single-valued, or is not table-valued when the corresponding formal parameter specifies a table.

The names associated with the formal parameters in the core opcode prototypes have no normative significance, but are used for clarity of exposition to refer to the values passed as the corresponding actual parameters when describing how to calculate the return value of the core opcode.

### 5.9.4 Math functions

#### 5.9.4.1 Introduction

Each of the opcodes in this subclause computes a mathematical function. Whenever the result of calculating the function on the argument or arguments provided results in a NaN or Inf value, a run-time error shall result..

#### 5.9.4.2 int

opcode `int(xsig x)`

The **int** core opcode calculates the integer part of its parameter.

The return value shall be the integer part of **x**.

**5.9.4.3 frac**opcode `frac(xsig x)`

The **frac** core opcode calculates the fractional part of its parameter.

The return value shall be the fractional part of  $x$ , i.e.,  $x - \text{int}(x)$ . If  $x$  is negative, then **frac(x)** is also negative.

**5.9.4.4 dbamp**opcode `dbamp(xsig x)`

The **dbamp** core opcode calculates the decibel equivalent of an amplitude parameter, where the amplitude of 1 corresponds to a decibel level of 90 dB. It is a run-time error if  $x$  is not strictly positive.

The return value shall be  $90 + 20 \log_{10} x$ .

**5.9.4.5 ampdb**opcode `ampdb(xsig x)`

The **ampdb** core opcode calculates the amplitude equivalent of a decibel-valued parameter, where the amplitude of 1 corresponds to a decibel level of 90 dB.

The return value shall be  $10^{(x - 90) / 20}$ .

**5.9.4.6 abs**opcode `abs(xsig x)`

The **abs** core opcode calculates the absolute value of a parameter.

The return value shall be  $-x$  if  $x < 0$ , or  $x$  otherwise.

**5.9.4.7 sgn**opcode `sgn(xsig x)`

The **sgn** core opcode calculates the signum (sign function) of a parameter.

The return value shall be  $-1$  if  $x < 0$ ,  $0$  if  $x = 0$ , or  $1$  if  $x > 0$ .

**5.9.4.8 exp**opcode `exp(xsig x)`

The **exp** core opcode calculates the exponential function.

The return value shall be  $e^x$ .

**5.9.4.9 log**opcode `log(xsig x)`

The **log** core opcode calculates the natural logarithm of a parameter.

It is a run-time error if  $x$  is not strictly positive.

The return value shall be  $\log x$ .

#### 5.9.4.10 **sqrt**

opcode `sqrt(xsig x)`

The **sqrt** core opcode calculates the square root of a parameter.

It is a run-time error if **x** is negative.

The return value shall be  $\sqrt{x}$ .

#### 5.9.4.11 **sin**

opcode `sin(xsig x)`

The **sin** core opcode calculates the sine of a parameter given in radians.

The return value shall be  $\sin x$ .

#### 5.9.4.12 **cos**

opcode `cos(xsig x)`

The **cos** core opcode calculates the cosine of a parameter given in radians.

The return value shall be  $\cos x$ .

#### 5.9.4.13 **atan**

opcode `atan(xsig x)`

The **atan** core opcode calculates the arctangent of a parameter, in radians.

The return value shall be  $\tan^{-1} x$ , in the range  $[-\pi/2, \pi/2)$ .

#### 5.9.4.14 **pow**

opcode `pow(xsig x, xsig y)`

The **pow** core opcode calculates the to-the-power-of operation.

It shall be a run-time error if **x** is negative and **y** is not an integer.

The return value shall be  $x^y$ .

#### 5.9.4.15 **log10**

opcode `log10(xsig x)`

The **log10** core opcode calculates the base-10 logarithm of a parameter.

It is a run-time error if **x** is not strictly positive.

The return value shall be  $\log_{10} x$ .

**5.9.4.16 asin**opcode `asin(xsig x)`

The **asin** core opcode calculates the arcsine of a parameter, in radians.

It is a run-time error if **x** is not in the range [-1, 1].

The return value shall be  $\sin^{-1} x$ , in the range  $[-\pi/2, \pi/2]$ .

**5.9.4.17 acos**opcode `acos(xsig x)`

The **acos** core opcode calculates the arccosine of a parameter, in radians.

It is a run-time error if **x** is not in the range [-1, 1].

The return value shall be  $\cos^{-1} x$ , in the range  $[0, \pi]$ .

**5.9.4.18 ceil**opcode `ceil(xsig x)`

The **ceil** core opcode calculates the ceiling of a parameter.

The return value shall be the smallest integer  $y$  such that  $x \leq y$ .

**5.9.4.19 floor**opcode `floor(xsig x)`

The **floor** core opcode calculates the floor of a parameter.

The return value shall be the greatest integer  $y$  such that  $y \leq x$ .

**5.9.4.20 min**opcode `min(xsig x1[, xsig ...])`

The **min** core opcode finds the minimum of a number of parameters.

The return value shall be the minimum value out of the parameter values.

**5.9.4.21 max**opcode `max(xsig x1[, xsig ...])`

The **max** core opcode finds the maximum out of the parameter values.

The return value shall be the maximum value out of the parameter values.

## 5.9.5 Pitch converters

### 5.9.5.1 Introduction to pitch representations

There are four representations for pitch in a SAOL orchestra; the following twelve functions (after **gettune** and **settune**) convert them from one to another. The four representations are as follows:

- pitch-class, or **pch** representation. A pitch is represented as an integer part, which represents the octave number, where 8 shall be the octave containing middle C (C4); plus a fractional part, which represents the pitch-class, where .00 shall be C, .01 shall be C#, .02 shall be D, and so forth. Fractional parts larger than .11 (B) have no meaning in this representation; fractional parts between the pitch-class steps are rounded to the nearest pitch-class.

For example, 7.09 is the A below middle C.

- octave-fraction, or **oct** representation. A pitch is represented as an integer part, which represents the octave number, where 8 shall be the octave containing middle C (C4); plus a fractional part, which represents a fraction of an octave, where each step of 1/12 represents a semitone.

For example, 7.75 is the A below middle C, in equal-tempered tuning.

- MIDI pitch number representation. A pitch is represented as an integer number of semitones above or below middle C, represented as 60.

For example, 57 is the A below middle C.

- Frequency, or **cps** representation. A pitch is represented as some number of cycles per second.

For example, 220 Hz is the A below middle C.

Each of the pitch converters represents the conversion that is done by its name, with the new representation first and the original (parameter) representation second. Thus, **cpsmidi** is the converter that returns the frequency corresponding to a particular MIDI pitch.

Changes to the **pitch** field of an AudioBIFS **AudioSource** node (see clause 5.15) controlling the decoding process scale the global tuning around 440 Hz just as if **settune()** was called. The value of the **pitch** field is a multiplier to be applied to 440; that is if the **pitch** field is changed to 1.1, the global tuning becomes 484 Hz. Through this mechanism, changes in the **pitch** field apply to the results of all pitch converters and the **gettune()** opcode, but to no other instructions in the orchestra.

### 5.9.5.2 **gettune**

```
opcode gettune([xsig dummy])
```

The **gettune** core opcode returns the value in Hz of the current orchestra global tuning, which is the frequency of A above middle C. The global tuning shall be set by default to 440, but can be changed using the **settune** core opcode, subclause 5.9.5.3.

The **dummy** parameter is used to specify the rate of the opcode call if desired; see subclause 5.8.7.7.2.

### 5.9.5.3 **settune**

```
kopcode settune(ksig x)
```

The **settune** core opcode sets and returns the value of the current orchestra global tuning. The global tuning is used by several pitch converters when converting between symbolic pitch representations and cycles-per-second representation.

It is a run-time error if  $x$  is not strictly positive. (Allowing a wide range for tuning parameters allows unusual “pitch” representations to be used).

This core opcode has side-effects, as follows: The global tuning variable shall be set to the value  $x$ .

The return value shall be  $x$ .

#### 5.9.5.4 **octpch**

opcode `octpch(xsig x)`

The **octpch** core opcode converts pitch-class representation to octave representation, with regard to equal scale tempering.

It is a run-time error if  $x$  is not strictly positive.

Let the integer part of  $x$  be  $y$  and the fractional part of  $x$  be  $z$ . Then, the return value shall be calculated as follows:

$z$  shall be “rounded” to the nearest value such that  $100z$  is an integer. If  $z < 0$  or  $z > 0.11$ , then  $z$  shall be set to 0 instead.

Then, the return value shall be  $y + 100z / 12$ .

#### 5.9.5.5 **pchoct**

opcode `pchoct(xsig x)`

The **pchoct** core opcode converts octave representation to pitch-class representation.

It is a run-time error if  $x$  is not strictly positive.

Let the integer part of  $x$  be  $y$  and the fractional part of  $x$  be  $z$ . Then, the return value shall be calculated as follows:

$z$  shall be rounded to the nearest value such that  $12z$  is an integer. Then, the return value shall be  $y + 12z / 100$ .

#### 5.9.5.6 **cpspch**

opcode `cpspch(xsig x)`

The **cpspch** core opcode converts pitch-class representation to cycles-per-second representation, with regard to equal scale tempering and the global tuning.

It is a run-time error if  $x$  is not strictly positive.

Let the integer part of  $x$  be  $y$  and the fractional part of  $x$  be  $z$ . Then, the return value shall be calculated as follows:

$z$  shall be “rounded” to the nearest value such that  $100z$  is an integer. If  $z < 0$  or  $z > 11$ , then  $z$  shall be set to 0 instead.

Further let  $t$  be the global tuning parameter. Then, the return value shall be  $t \times 2^{(y + 100z/12 - 8.75)}$ .

#### 5.9.5.7 **pchcps**

opcode `pchcps(xsig x)`

The **pchcps** core opcode converts cycles-per-second representation to pitch-class representation, with regard to the global tuning.

It is a run-time error if  $x$  is not strictly positive.

The return value shall be calculated as follows.

Let  $t$  be the global tuning parameter. Then, let  $k$  be  $\log_2(x/t) + 8.75$ . Then, let the integer part of  $k$  be  $y$  and the fractional part of  $k$  be  $z$ , "rounded" to the nearest value such that  $12z$  is an integer. The return value shall be  $y + 12z / 100$ .

#### 5.9.5.8 cpsoct

opcode cpsoct(xsig x)

The **cpsoct** core opcode converts octave representation to cycles-per-second representation, with regard to the global tuning.

It is a run-time error if  $x$  is not strictly positive.

Let  $t$  be the global tuning value; then, the return value shall be  $t \times 2^{(x-8.75)}$ .

#### 5.9.5.9 octcps

opcode octcps(xsig x)

The **octcps** core opcode converts cycles-per-second representation to octave representation, with regard to the global tuning.

It is a run-time error if  $x$  is not strictly positive.

Let  $t$  be the global tuning value; then, the return value shall be  $\log_2(x/t) + 8.75$ .

#### 5.9.5.10 midipch

opcode midipch(xsig x)

The **midipch** core opcode converts pitch-class representation to MIDI representation.

It is a run-time error if  $x \leq 3$ .

Let the integer part of  $x$  be  $y$  and the fractional part of  $x$  be  $z$ . Then, the return value shall be calculated as follows:

$z$  shall be "rounded" to the nearest value such that  $100z$  is an integer. If  $z < 0$  or  $z > 0.11$ , then  $z$  shall be set to 0 instead.

The return value shall be  $100z + 12(y - 3)$ .

#### 5.9.5.11 pchmidi

opcode pchmidi(xsig x)

The **midipch** core opcode converts MIDI representation to pitch-class representation.

It is a run-time error if  $x$  is not strictly positive.

The return value shall be calculated as follows:  $x$  shall be rounded to the nearest integer, then let  $k$  be  $(x + 36) / 12$ , and let  $y$  be the integer part of  $k$ , and let  $z$  be the fractional part of  $k$ . Then, the return value shall be  $y + 12z / 100$ .



**5.9.5.12 midiact**opcode `midiact(xsig x)`

The **midiact** core opcode converts octave representation to MIDI representation.

It is a run-time error if  $x \leq 3$ .

The return value shall be calculated as follows. Let  $k$  be  $12(x - 3)$ . Then, the value of  $k$  rounded to the nearest integer shall be the return value.

**5.9.5.13 octmidi**opcode `octmidi(xsig x)`

The **octmidi** core opcode converts MIDI representation to octave representation.

It is a run-time error if  $x$  is not strictly positive.

The return value shall be  $(x + 36) / 12$ .

**5.9.5.14 midicps**opcode `midicps(xsig x)`

The **midicps** core opcode converts cycles-per-second representation to MIDI representation, with regard to the global tuning.

It is a run-time error if  $x$  is not strictly positive.

Let  $t$  be the global tuning parameter, and let  $k$  be  $12 \log_2(x / t) + 69$ . Then, the return value shall be  $k$  rounded to the nearest nonnegative integer.

**5.9.5.15 cpsmidi**opcode `cpsmidi(xsig x)`

The **cpsmidi** core opcode converts MIDI representation to cycles-per-second representation, with regard to the global tuning and equal scale temperament.

It is a run-time error if  $x$  is not strictly positive.

Let  $t$  be the global tuning parameter. Then, the return value shall be  $t \times 2^{(x - 69) / 12}$ .

**5.9.6 Table operations****5.9.6.1 ftlen**opcode `ftlen(table t)`

The **ftlen** core opcode returns the length of a table. The length of a table is the value calculated based on the **size** parameter in the particular core wavetable generator as described in clause 5.10.

The return value shall be the length of the table referenced by **t**.

**5.9.6.2 ftloop**opcode `ftloop(table t)`

The **ftloop** core opcode returns the loop start point of a wavetable. The loop point is set either in a sound sample data block in the bitstream, or by the **ftsetloop** core opcode (see subclause 5.9.6.6), or else it is 0.

The return value shall be the loop start point (in samples) of the wavetable referenced by **t**.

### 5.9.6.3 ftloopend

opcode `ftloopend(table t)`

The **ftloopend** core opcode returns the loop end point of a wavetable. The loop point is set either in a sound sample data block in the bitstream, or by the **ftsetend** core opcode (see subclause 5.9.6.7), or else it is 0.

The return value shall be the loop end point (in samples) of the wavetable referenced by **t**.

### 5.9.6.4 ftsr

opcode `ftsr(table t)`

The **ftsr** core opcode returns the sampling rate of a wavetable. The sampling rate is set in a sound sample data block in the bitstream, or else it is 0.

The return value shall be the sampling rate, in Hz, of the wavetable referenced by **t**.

### 5.9.6.5 ftbasecps

opcode `ftbasecps(table t)`

The **ftbasecps** core opcode returns the base frequency of a wavetable, in cycles per second (Hz). The base frequency is set either in a sound sample data block in the bitstream, or in the core wavetable generator **sample** (subclause 5.10.2), or by the core opcode **ftsetbase** (subclause 5.9.6.8), or else it is 0.

The return value shall be the base frequency, in Hz, of the wavetable referenced by **t**.

### 5.9.6.6 ftsetloop

kopcode `ftsetloop(table t, ksig x)`

The **ftbasecps** core opcode sets the loop start point of a wavetable to a new value, and returns the new value.

It is a run-time error if **x** < 0, or if **x** is larger than the size of the wavetable referenced by **t**.

This core opcode has side effects, as follows: the loop start point of the wavetable **t** shall be set to sample number **x**.

The return value shall be **x**.

### 5.9.6.7 ftsetend

kopcode `ftsetend(table t, ksig x)`

The **ftsetend** core opcode sets the loop end point of a wavetable to a new value, and returns the new value. It is a run-time error if **x** < 0, or if **x** is larger than the size of the wavetable referenced by **t**.

This core opcode has side effects, as follows: the loop end point of the wavetable **t** shall be set to sample number **x**.

The return value shall be **x**.

**5.9.6.8 ftsetbase**

opcode `ftsetbase(table t, ksig x)`

The **ftsetbase** core opcode sets the base frequency of a wavetable to a new value, and returns the new value. It is a run-time error if **x** is not strictly positive.

This core opcode has side effects, as follows: the base frequency of the wavetable **t** shall be set to **x**, where **x** is a value in Hz.

The return value shall be **x**.

**5.9.6.9 ftsetsr**

opcode `ftsetsr(table t, ksig x)`

The **ftsetsr** core opcode sets the sampling rate parameter of a wavetable to a new value **x**, and returns the new value. It is a run-time error if **x** is not strictly positive.

This core opcode has side effects, as follows: the sampling rate of the wavetable **t** shall be set to **x**, where **x** is a value in Hz.

The return value shall be **x**.

**5.9.6.10 tableread**

opcode `tableread(table t, xsig index)`

The **tableread** core opcode returns a single value from a wavetable. It is a run-time error if **index** < 0, or if **index** is greater than or equal to (tsize-0.5), where tsize is the size of the wavetable referenced by **t**.

The return value shall be the value of the wavetable **t** at sample number **index**, where sample number 0 is the first sample in the wavetable. If **index** is not an integer, then the return value shall be interpolated from nearby points of the wavetable, as described by the global interpolation-quality parameter (subclause 5.8.5.2.5).

**5.9.6.11 tablewrite**

opcode `tablewrite(table t, xsig index, xsig val)`

The **tablewrite** core opcode sets a single value in a wavetable, and returns that value. It is a run-time error if **index** < 0, or if **index** is larger than the size of the wavetable referenced by **t**.

This core opcode has side effects, as follows: **index** shall be rounded to the nearest integer, and the value of sample number **index** in the wavetable **t** shall be set to the new value **val**, where sample number 0 is the first sample in the wavetable.

The return value shall be **val**.

If global tables are written to at the a-rate by one instrument instance, it is unspecified when the new values become available to other instrument instances. In particular, it is unspecified whether the changes are available in the same orchestra cycle. The changes shall be available to all instrument instances in the next orchestra cycle, and are permitted to be available to instances of instruments sequenced later in the same orchestra cycle.

**5.9.6.12 oscil**

aopcode `oscil(table t, asig freq[, ivar loops])`

The **oscil** core opcode loops several times around the wavetable **t** at a rate of **freq** loops per second, returning values at the audio-rate. **loops** shall be rounded to the nearest integer when the opcode is evaluated. If **loops** is not provided, its value shall be  $-1$ .

It is a run-time error if **loops** is not strictly positive and is also not  $-1$ .

The return value is calculated according to the following procedure.

On the first a-rate call to **oscil** relative to a particular state, the *internal phase* shall be set to 0, and the *internal number of loops* set to **loops**. On subsequent calls, the internal phase shall be incremented by  $\text{freq}/\text{SR}$ , where **SR** is the orchestra sampling rate. If, after the incrementation, the internal phase is not in the interval  $[0,1]$  and the internal loop count is strictly positive, the phase shall be set to the fractional portion of its value ( $p := p - \text{floor}(p)$ ) and the loop count decremented.

If the internal loop count is zero, the return value shall be 0. Otherwise the return value shall be the value of sample number  $x$  in the wavetable, where  $x = p * l$ , where  $p$  is the current internal phase, and  $l$  is the length of table **t**. If  $x$  is not an integer, then the value shall be interpolated from the nearby table values, as described by the global interpolation-quality parameter (subclause 5.8.5.2.5).

Note that a negative value for the frequency argument to this opcode is legal, and the phase pointer update algorithm works correctly for negative values.

NOTE - The **oscil** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **oscil** is referenced twice in the same a-cycle, then the effective loop frequency is twice as high as given by **freq**.

### 5.9.6.13 loscil

```
aopcode loscil(table t, asig freq[, ivar basefreq, ivar loopstart, ivar loopend])
```

This opcode loops around the wavetable **t**, returning values at the audio-rate. The looping continues as long as the opcode is active, and is performed at a special rate that depends on the base frequency **basefreq** and the sampling rate of the table. In this way, samples that were recorded at a particular known pitch may be interpolated to any other pitch.

If **basefreq** is not provided, it shall be set to the base frequency of the table **t** by default. If the table **t** has base frequency 0 and **basefreq** is not provided, it is a run-time error. If **basefreq** is not strictly positive, it is a runtime error. The **basefreq** parameter shall be specified in Hz.

If **loopstart** and **loopend** are not provided, they shall be set to the loop start point and loop end point of the table **t**, respectively. If **loopend** is not provided and the loop end point of **t** is 0, then it shall be set to the end of the table ( $l - 1$ ), where  $l$  is the length of the table in sample points). If **loopstart** is not strictly less than **loopend**, or either is negative, it is a runtime error.

The return value is calculated according to the following procedure.

Let  $l$  be the length of the table,  $m$  be the value  $\text{loopstart} / l$ , and  $n$  be the value  $\text{loopend} / l$ . On the first a-rate call to **loscil** relative to a particular state, the *internal phase* shall be set to 0. On subsequent calls, the internal phase shall be incremented by  $\text{freq} * \text{TSR} / (\text{basefreq} * \text{SR})$ , where **TSR** is the sampling rate of the table and **SR** is the orchestra sampling rate. If the incrementation has caused the internal phase to leave the interval  $[m,n]$  or to become less than 0, the phase shall be set to  $m + p - kn$ , where  $p$  is the internal phase and  $k$  is the value  $\text{floor}(p/n)$ . However, if the phase pointer has not yet passed the **loopstart** value, and if an incrementation has caused the phase pointer to become less than zero, the phase pointer takes on the value zero.

The return value shall be the value of sample number  $x$  in the wavetable, where  $x = p * l$ , where  $p$  is the current internal phase, and  $l$  is the length of table **t**. If  $x$  is not an integer, then the value shall be interpolated from the nearby table values, as described by the global interpolation-quality parameter (subclause 5.8.5.2.5).

Note that a negative value for the frequency argument to this opcode is legal, and the phase pointer update algorithm works correctly for negative values.

NOTE - The **loscil** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **loscil** is referenced twice in the same a-cycle, then the effective loop frequency is twice as high as given by **freq**.

#### 5.9.6.14 **doscil**

```
aopcode doscil(table t)
```

The **doscil** core opcode plays back a sample once, with no frequency control or looping. It is useful for sample-rate matching sampled drum sounds to an orchestra rate.

The return value is calculated according to the following procedure.

On the first a-rate call to **doscil** relative to a particular state, the *internal phase* shall be set to 0. On subsequent calls, the internal phase shall be incremented by **TSR/SR**, where **TSR** is the sampling rate of the table **t** and **SR** is the orchestra sampling rate. If, after the incrementation, the internal phase is greater than 1, then the opcode is done.

If the opcode is done, the return value shall be 0. Otherwise the return value shall be the value of sample number  $x$  in the wavetable, where  $x = p * l$ , where  $p$  is the current internal phase, and  $l$  is the length of table **t**. If  $x$  is not an integer, then the value shall be interpolated from the nearby table values, as described by the global interpolation-quality parameter (subclause 5.8.5.2.5).

NOTE - The **doscil** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **doscil** is referenced twice in the same a-cycle, then the sample is played back at twice its original frequency.

#### 5.9.6.15 **koscil**

```
kopcode koscil(table t, ksig freq[, ivar loops])
```

This opcode loops several times around the wavetable **t** at a rate of **freq** loops per second, returning values at the control-rate. **loops** shall be rounded to the nearest integer when the opcode is evaluated. If **loops** is not provided, its value shall be set to  $-1$ .

It is a run-time error if **loops** is not strictly positive and is also not  $-1$ .

The return value is calculated according to the following procedure.

On the first k-rate call to **koscil** relative to a particular state, the *internal phase* shall be set to 0, and the *internal number of loops* set to **loops**. On subsequent calls, the internal phase shall be incremented by **freq/KR**, where **KR** is the orchestra control rate. If, after the incrementation, the phase is not in the interval  $[0, 1]$  and the internal loop count is strictly positive, the phase shall be set to the fractional portion of its value ( $p := p - \text{floor}(p)$ ) and the loop count decremented.

If the internal loop count is zero, the return value shall be 0. Otherwise the return value shall be the value of sample number  $x$  in the wavetable, where  $x = p * l$ , where  $p$  is the current internal phase, and  $l$  is the length of table **t**. If  $x$  is not an integer, then the value shall be interpolated from the nearby table values, as described by the global interpolation-quality parameter (subclause 5.8.5.2.5).

Note that a negative value for the frequency argument to this opcode is legal, and the phase pointer update algorithm works correctly for negative values.

NOTE - The **koscil** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **koscil** is referenced twice in the same k-cycle, then the effective loop frequency is twice as high as given by **freq**.

## 5.9.7 Signal generators

### 5.9.7.1 **kline**

```
opcode kline(ivar x1, ivar dur1, ivar x2[, ivar dur2, ivar x3, ...]);
```

The **kline** core opcode produces a line-segmented or “ramp” function, with values changing at the k-rate. This function takes **dur1** seconds to go from **x1** to **x2**, **dur2** seconds to go from **x2** to **x3**, and so on.

It is a run-time error if any of the following conditions apply:

- there are an even number of parameters
- any of the **dur** values are negative

The return value shall be calculated as follows:

On the first call to **kline** with regard to a particular state, the *internal time* shall be set to 0, the *current left point* to **x1**, the *current right point* to **x2**, and the *current duration* to **dur1**. On subsequent calls, the internal time shall be incremented by  $1/KR$ , where **KR** is the orchestra control rate. So long as the internal time is thereby greater than the current duration and there is another duration parameter, the internal time shall be decremented by the current duration, the current duration shall be set to the next duration parameter, the current left point to the current right point, and the current right point to the next control point (x-value) (these steps repeat if necessary so long as the internal time is greater than the current duration). If there is no additional duration parameter, the generator is done.

If the generator is done, then the return value is 0. Otherwise, the return value is  $l + (r - l)t/d$ , where *l* is the current left point, *r* is the current right point, *t* is the internal time, and *d* is the current duration.

NOTE - The **kline** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **kline** is referenced twice in the same k-cycle, then the effective segment duration is half as long as given by the corresponding duration value.

### 5.9.7.2 **aline**

```
opcode aline(ivar x1, ivar dur1, ivar x2[, ivar dur2, ivar x3, ...]);
```

The **aline** core opcode produces a line-segmented or “ramp” function, with values changing at the a-rate. This function takes **dur1** seconds to go from **x1** to **x2**, **dur2** seconds to go from **x2** to **x3**, and so on.

It is a run-time error if any of the following conditions apply:

- there are an even number of parameters
- any of the **dur** values are negative

The return value shall be calculated as follows:

On the first call to **aline** with regard to a particular state, the *internal time* shall be set to 0, the *current left point* to **x1**, the *current right point* to **x2**, and the *current duration* to **dur1**. On subsequent calls, the internal time shall be incremented by  $1/SR$ , where **SR** is the orchestra sampling rate. So long as the internal time is thereby greater than the current duration and there is another duration parameter, the internal time shall be decremented by the current duration, the current duration shall be set to the next duration parameter, the current left point to the current right point, and the current right point to the next control point (x-value) (these steps repeat if necessary so long as the internal time is greater than the current duration). If there is no additional duration parameter, the generator is done.

If the generator is done, then the return value is 0. Otherwise, the return value is  $l + (r - l) t / d$ , where *l* is the current left point, *r* is the current right point, *t* is the internal time, and *d* is the current duration.

NOTE - The **aline** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **aline** is referenced twice in the same a-cycle, then the effective segment duration is half as long as given by the corresponding duration value.

### 5.9.7.3 kexpon

```
kopcode kexpon(ivar x1, ivar dur1, ivar x2[, ivar dur2, ivar x3, ...]);
```

The **kexpon** core opcode produces a segmented function made out of exponential curves, with values changing at the k-rate. This function takes **dur1** seconds to go from **x1** to **x2**, **dur2** seconds to go from **x2** to **x3**, and so on.

It is a run-time error if any of the following conditions apply:

- there are an even number of parameters
- any of the **dur** values are negative
- the **x** values are not all the same sign
- any **x** value is 0

The return value shall be calculated as follows:

On the first call to **kexpon** with regard to a particular state, the *internal time* shall be set to 0, the *current left point* to **x1**, the *current right point* to **x2**, and the *current duration* to **dur1**. On subsequent calls, the internal time shall be incremented by  $1/KR$ , where **KR** is the orchestra control rate. So long as the internal time is thereby greater than the current duration and there is another duration parameter, the internal time shall be decremented by the current duration, the current duration shall be set to the next duration parameter, the current left point to the current right point, and the current right point to the next control point (x-value) (these steps repeat if necessary so long as the internal time is greater than the current duration). If there is no additional duration parameter, the generator is done.

If the generator is done, then the return value is 0. Otherwise, the return value is  $l(r/l)^{t/d}$ , where *l* is the current left point, *r* is the current right point, *t* is the internal time, and *d* is the current duration.

NOTE - The **kexpon** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **kexpon** is referenced twice in the same k-cycle, then the effective segment duration is half as long as given by the corresponding duration value.

### 5.9.7.4 aexpon

```
aopcode aexpon(ivar x1, ivar dur1, ivar x2[, ivar dur2, ivar x3, ...]);
```

The **aexpon** core opcode produces a segmented function made out of exponential curves, with values changing at the a-rate. This function takes **dur1** seconds to go from **x1** to **x2**, **dur2** seconds to go from **x2** to **x3**, and so on.

It is a run-time error if any of the following conditions apply:

- there are an even number of parameters
- any of the **dur** values are negative
- the **x** values are not all the same sign
- any **x** value is 0

The return value shall be calculated as follows:

On the first call to **aexpon** with regard to a particular state, the *internal time* shall be set to 0, the *current left point* to **x1**, the *current right point* to **x2**, and the *current duration* to **dur1**. On subsequent calls, the internal time shall be incremented by  $1/\mathbf{SR}$ , where **SR** is the orchestra sampling rate. . So long as the internal time is thereby greater than the current duration and there is another duration parameter, the internal time shall be decremented by the current duration, the current duration shall be set to the next duration parameter, the current left point to the current right point, and the current right point to the next control point (x-value) (these steps repeat if necessary so long as the internal time is greater than the current duration). If there is no additional duration parameter, the generator is done.

If the generator is done, then the return value is 0. Otherwise, the return value is  $l(r/l)^{t/d}$ , where *l* is the current left point, *r* is the current right point, *t* is the internal time, and *d* is the current duration.

NOTE - The **aexpon** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **aexpon** is referenced twice in the same a-cycle, then the effective segment duration is half as long as given by the corresponding duration value.

### 5.9.7.5 kphasor

kopcode kphasor(ksig cps)

The **kphasor** core opcode produces a moving phase value, looping from 0 to 1 repeatedly, **cps** times per second.

The return value shall be calculated as follows:

On the first call to **kphasor** with regard to a particular state, the *internal phase* shall be set to 0. On subsequent calls, the internal phase shall be incremented by  $\mathbf{cps}/\mathbf{KR}$ , where **R** is the orchestra control rate. If the internal phase is thereby not in the interval [0,1], the internal phase shall be set to the fractional part of its value ( $p = \text{frac}(p)$ ). The return value is the internal phase.

Note that a negative value for the frequency argument to this opcode is legal, and the phase pointer update algorithm works correctly for negative values.

NOTE - The **kphasor** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **kphasor** is referenced twice in the same k-cycle, then the effective frequency is twice as fast as given by **cps**.

### 5.9.7.6 aphasor

aopcode aphasor(asig cps)

The **aphasor** opcode produces a moving phase value, looping from 0 to 1 repeatedly, **cps** times per second.

The return value shall be calculated as follows:

On the first call to **aphasor** with regard to a particular state, the *internal phase* shall be set to 0. On subsequent calls, the internal phase shall be incremented by  $\mathbf{cps}/\mathbf{SR}$ , where **SR** is the orchestra sampling rate. If the internal phase is thereby not in the interval [0,1], the internal phase shall be set to the fractional part of its value ( $p = \text{frac}(p)$ ). The return value is the internal phase.

Note that a negative value for the frequency argument to this opcode is legal, and the phase pointer update algorithm works correctly for negative values.

NOTE - The **aphasor** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **aphasor** is referenced twice in the same a-cycle, then the effective frequency is twice as fast as given by **cps**.

### 5.9.7.7 pluck

aopcode pluck(asig cps, ivar buflen, table init, ksig atten, ksig smoothrate)



This opcode uses a simple form of the Karplus-Strong algorithm to generate plucked-string sounds by repeated sampling and smoothing of a buffer.

It is a run-time error if **buflen** is not strictly positive.

The return value is calculated as follows:

On the first call to **pluck** with regard to a particular opcode state, a *buffer* of length **buflen** shall be created and filled with the values from the table **init**, as follows. Let  $x$  be the length of the table **init**. If  $x$  is less than **buflen**, then the values of the buffer shall be set to the first **buflen** sample values of the table **init**. If  $x$  is greater than or equal to **buflen**, then the first **buflen** values of the buffer shall be set to the sample values in the table **init**, and the remainder of the buffer filled as described in this paragraph for the whole table. That is, as many full and partial cycles of the table are used as necessary to fill the buffer.

Also on the first call to **pluck** with regard to a particular state, the *internal phase* shall be set to 0, and the *smooth count* shall be set to 0.

On subsequent calls to **pluck** with regard to a state, the smooth count is incremented. If the smooth count is equal to **smoothrate**, then **smoothrate** is set to 0, and the buffer shall be smoothed, as follows. A new buffer of length **buflen** shall be created, and its values set by averaging over the current buffer. Each sample value in the new buffer shall be set to the value of the attenuated mean of the five surrounding samples of the current buffer. That is, for each sample  $x$  of the new buffer, its value shall be set to  $\text{atten} * (b[x-2] + b[x-1] + b[x] + b[x+1] + b[x+2])/5$ , where the  $b[.]$  notation refers to values of the current buffer, and the indices are calculated modulo **buflen** (that is, they “wrap around”). Then, the values of the current buffer shall be set to the values of the new buffer.

Whether or not the buffer has just been smoothed, the internal phase shall be incremented by  $\text{cps}/\text{SR}$ , where **SR** is the orchestra sampling rate, and if the resulting value is not in the interval  $[0,1]$ , then the internal phase shall be set to the fractional part of the internal phase ( $p = p - \text{floor}(p)$ ).

The return value shall be the value of the buffer at the point  $p * \text{buflen}$ , where  $p$  is the internal phase. If this index is not an integer, the value shall be interpolated from nearby buffer values, as described by the global interpolation-quality parameter (subclause 5.8.5.2.5).

Note that a negative value for the frequency argument to this opcode is legal, and the phase pointer update algorithm works correctly for negative values.

NOTE - The **pluck** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **pluck** is referenced twice in the same a-cycle, then the effective frequency is twice as fast as given by **cps**.

### 5.9.7.8 **buzz**

```
aopcode buzz(asig cps, ksig nharm, ksig lowharm, ksig rolloff)
```

The **buzz** opcode produces a band-limited pulse train formed by adding together cosine overtones of a fundamental frequency **cps** given in Hz. These noisy sounds are useful as complex sound sources for subtractive synthesis.

**lowharm** gives the lowest harmonic used, where 0 is the fundamental, at frequency **cps**. It is a runtime error if **lowharm** is negative.

**nharm** gives the number of harmonics used starting from **lowharm**. If **nharm** is not strictly positive, then every overtone up to the orchestra Nyquist frequency is used (**nharm** shall be set to  $\text{SR} / 2 / \text{cps} - \text{lowharm}$ ).

**rolloff** gives the multiplicative rolloff that defines the spectral shape. If **rolloff** is negative, then the partials alternate in phase; if  $|\text{rolloff}| > 1$ , then the partials increase in amplitude rather than attenuating.

The return value is calculated as follows. On the first call to **buzz** with regard to a particular scope, the *internal phase* shall be set to 0. On subsequent calls, the internal phase shall be incremented by  $\text{cps} / \text{SR}$ , where **SR** is the

orchestra sampling rate. If, after this incrementation, the internal phase is greater than 1, the internal phase shall be set to the fractional part of its value ( $p := \text{frac}(p)$ ).

The return value shall be

$$\text{scale} * \sum_{f=\text{lowharm}}^{\text{lowharm}+\text{nharm}} \text{rolloff}^{(f-\text{lowharm})} \cos 2\pi(f+1)p$$

where  $p$  is the internal phase.

If  $|\text{rolloff}| = 1$ ,  $\text{scale} = (1/(\text{nharm}+1))$

else  $\text{scale} = (1-\text{abs}(\text{rolloff})) / (1-\text{abs}(\text{rolloff}^{(\text{nharm}+1)}))$ .

Note that a negative value for the frequency argument to this opcode is legal, and the phase pointer update algorithm works correctly for negative values.

### 5.9.7.9 grain

```
opcode grain(table wave, table env, ksig density, ksig freq, ksig amp,
            ksig dur, ksig time, ksig phase)
```

The **grain** opcode uses granular synthesis [**GRAN**] to synthesize periodic, quasi-periodic, noisy, and textured sounds. A sound in granular synthesis is represented as the sum of a number of short sound samples or “grains” distributed throughout the time-frequency space.

**wave** is the waveshape for the grain. **env** is the envelope to apply to the grain. **density** is the time-spacing of grain trigger points in Hz. **freq** is the frequency in Hz at which to place each new grain. **amp** is the amplitude of each new grain, as a scaling factor. **dur** is the duration of each grain in seconds. **time** is the offset in seconds from the trigger at which grains start (jitter). **phase** is the starting phase of each grain, in the range [0,1].

It is a run-time error if any of the following conditions apply: **density** is not positive, **dur** is negative, **time** is negative, or **phase** is not in the range [0,1].

On the first call to **grain** with regard to a particular opcode state, the *number of grains* is set to 0 and the *density clock* and *trigger clock* are both set to 0.

On the first and each subsequent a-rate call to **grain**, the following steps are executed:

The density clock is incremented by  $1/\text{SR}$ , where SR is the orchestra sampling rate. If the density clock is thereby greater than or equal to  $1/\text{density}$ , then the density clock is set to zero, and then if  $\text{time} < 1/\text{density}$ , the trigger clock is set to **time**. If  $\text{time} \geq 1/\text{density}$ , the trigger clock remains at zero and no grain is created.

If the trigger clock is positive, then it is decremented by  $1/\text{SR}$ . If the trigger clock is thereby less than or equal to zero, a new grain is dispatched. To dispatch a new grain, the number of active grains is incremented and henceforth denoted as  $i$ . Space shall be allocated to hold the current phase **phase[i]**, frequency **freq[i]**, amplitude **amp[i]**, duration **dur[i]**, and time **gtime[i]** of the new grain. These values shall be set to the current value of **phase**, **freq**, **amp**, **dur**, and 0 respectively.

For each active grain  $i$ , the current value of the grain **x[i]** is calculated as follows. There are three conditions, depending on the format of the wavetable **wave**:

1. If **wave** has both its sampling rate and base frequency parameters set, it's assumed to be a pitched sample, and is pitch-matched to **freq** in the manner of **loscil()**. Let **loopstart** be set to the loop start parameter of the table **wave**; let **loopend** be set to the loop end parameter of the table, or  $I-1$  where  $I$  is the length of the table **wave**, if the loop end parameter is 0. Let  $m$  be the value  $\text{loopstart} / I$ , and let  $n$  be the value  $\text{loopend} / I$ . Each time after the first that the grain value is calculated, the phase **phase[i]** shall be incremented by  $\text{freq}[i] * \text{TSR} / (\text{basefreq} * \text{SR})$ , where **TSR** is the table sampling rate and **basefreq** the base frequency of table **wave**. If, after this incrementation,

the phase is not in the range  $[0,1]$ , the phase shall be set to  $\text{phase}[i] - \text{floor}(\text{phase}[i])$ . The current value of the grain  $\mathbf{x}[i]$  is the value of sample number  $\mathbf{q}$  of the wavetable  $\mathbf{wave}$ , where  $\mathbf{q} = \text{phase}[i] * (\mathbf{m}-\mathbf{n}) + \mathbf{m}$ . If the value of  $\mathbf{q}$  is not an integer, then the value of  $\mathbf{x}[i]$  shall be interpolated from the nearby table values, as described in subclause 5.8.5.2.5.

2. If  $\mathbf{wave}$  has its sampling rate parameter set, but not its base frequency parameter, then it's assumed to be an unpitched sample, and is sample-rate-matched to the orchestra in the manner of  $\text{doscil}()$  and  $\text{freq}[i]$  is ignored. Each time after the first that the grain value is calculated, the phase  $\text{phase}[i]$  shall be incremented by  $\text{TSR}/\text{SR}$ , where  $\text{TSR}$  is the table sampling rate and  $\text{SR}$  is the orchestra sampling rate. If, after this incrementation, the phase is greater than 1, then the current and all future values of this grain are 0. Otherwise, the current value of the grain  $\mathbf{x}[i]$  is the value of sample number  $\mathbf{q}$  of the wavetable  $\mathbf{wave}$ , where  $\mathbf{q} = \text{phase}[i] * \mathbf{l}$  and  $\mathbf{l}$  is the length of the wavetable  $\mathbf{wave}$ . If the value of  $\mathbf{q}$  is not an integer, then the value of  $\mathbf{x}[i]$  shall be interpolated from the nearby table values, as described in subclause 5.8.5.2.5.

3. If  $\mathbf{wave}$  has neither its sampling rate nor base frequency parameters set, then it's assumed to be a waveshaped, and is oscillated in the manner of  $\text{oscil}()$ . Each time after the first that the grain value is calculated, the phase  $\text{phase}[i]$  shall be incremented by  $\text{freq}[i]/\text{SR}$ . If, after this incrementation, the phase value is outside the range  $[0,1]$ , then the phase shall be set to the fractional part of its value  $\text{phase}[i] - \text{floor}(\text{phase}[i])$ . The current value of the grain  $\mathbf{x}[i]$  shall be the value of sample number  $\mathbf{q}$  of the wavetable  $\mathbf{wave}$ , where  $\mathbf{q} = \text{phase}[i] * \mathbf{l}$  and  $\mathbf{l}$  is the length of the wavetable  $\mathbf{wave}$ . If the value of  $\mathbf{q}$  is not an integer, then the value of  $\mathbf{x}[i]$  shall be interpolated from the nearby table values, as described in subclause 5.8.5.2.5.

After the output value of the grain  $\mathbf{x}[i]$  is calculated in one of these three manners, it is modulated by the envelope wavetable according to the grain duration. The time  $\mathbf{gtime}[i]$  is incremented by  $1/\text{SR}$ , where  $\text{SR}$  is the orchestra sampling rate. If  $\mathbf{gtime}[i] > \text{dur}$ , then the grain is over; it shall be noted as non-active, and its space may be deallocated. Otherwise, the modulator value  $\mathbf{m}[i]$  is the value of sample number  $\mathbf{q}$  of the wavetable  $\mathbf{env}$ , where  $\mathbf{q} = \text{floor}(\mathbf{gtime}[i] / \text{dur}[i] * \mathbf{l})$  and  $\mathbf{l}$  is the length of the wavetable  $\mathbf{env}$ . The final output value of the grain  $\mathbf{x}[i]$  is rescaled by this modulator value as in  $\mathbf{x}[i] = \text{amp} * \mathbf{x}[i] * \mathbf{m}[i]$ .

The output value from the opcode is the sum of the output values for all active grains.

NOTE (1) - If the input values  $\text{freq}$ ,  $\text{dur}$ , and so forth change during the use of the instrument (as they would, for example, to stochastically vary grain distribution), the values of  $\text{freq}[i]$ ,  $\text{dur}[i]$  and so forth do not change for currently-active grains. Only the parameters of new grains are affected. The parameter values of active grains do not change during the lifetime of the grain.

NOTE (2) - The **grain** opcode shall not have a "proper" representation of time, but shall infer it from the number of calls. If the same state of **grain** is referenced twice in the same a-cycle, than the effective frequency, duration, density, jitter, and so forth are scaled appropriately.

## 5.9.8 Noise generators

### 5.9.8.1 Note on noise generators and pseudo-random sequences

The following core opcodes generate noise, that is, pseudo-random sequences of various statistical properties. In order to provide maximum decorrelation among multiple noise generators, it is important that all references to pseudo-random generation share a single feedback state. That is, all random values required by the various states of various noise generators shall make use of sequential values from a single "master" pseudo-random sequence.

It is strictly prohibited for an implementation to maintain multiple pseudo-random sequences to draw from (using the same algorithm) for various states of noise generation opcodes, because to do so may result in strong correlations between multiple noise generators.

This point does not apply to implementations that do not use "linear congruential", "modulo feedback", or similar mathematical structures to generate pseudo-random numbers.

The standard mathematical description of probability density functions is used in this subclause. This means that if the pdf of a random variable  $x$  is  $f(x)$ , then the probability of it taking a value in the range  $[y,z]$  is  $\int_y^z f(x)dx$ .

**5.9.8.2 irand**

iopcode irand(ivar p)

The **irand** core opcode generates a random number from a linear distribution.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} 1/2p : x \in [-p, p] \\ 0 : otherwise \end{cases}$$

**5.9.8.3 krand**

kopcode krand(ksig p)

The **krand** core opcode generates random numbers from a linear distribution.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} 1/2p : x \in [-p, p] \\ 0 : otherwise \end{cases}$$

**5.9.8.4 arand**

aopcode arand(asig p)

The **arand** core opcode generates random noise according to a linear distribution.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} 1/2p : x \in [-p, p] \\ 0 : otherwise \end{cases}$$

**5.9.8.5 ilinrand**

iopcode ilinrand(ivar p1, ivar p2)

The **ilinrand** core opcode generates a random number from a linearly-ramped distribution.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} \text{abs}(2 / (p2 - p1) * [(x - p1) / (p2 - p1)]) & \text{if } x \in [p1, p2] \\ 0 & \text{otherwise} \end{cases}$$

**5.9.8.6 klinrand**

kopcode klinrand(ksig p1, ksig p2)

The **klinrand** core opcode generates random numbers from a linearly-ramped distribution.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} \text{abs}(2 / (\mathbf{p2} - \mathbf{p1}) * [(x - \mathbf{p1}) / (\mathbf{p2} - \mathbf{p1})]) & \text{if } x \in [\mathbf{p1}, \mathbf{p2}] \\ 0 & \text{otherwise} \end{cases}$$

#### 5.9.8.7 alinrand

aopcode alinrand(asig p1, asig p2)

The **alinrand** core opcode generates random noise from a linearly-ramped distribution.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} \text{abs}(2 / (\mathbf{p2} - \mathbf{p1}) * [(x - \mathbf{p1}) / (\mathbf{p2} - \mathbf{p1})]) & \text{if } x \in [\mathbf{p1}, \mathbf{p2}] \\ 0 & \text{otherwise} \end{cases}$$

#### 5.9.8.8 iexprand

iopcode iexprand(ivar p1)

The **iexprand** core opcode generates a random number from an exponential distribution with mean  $\mathbf{p1}$ . It is a run-time error if  $\mathbf{p1}$  is not strictly positive.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} 0 & \text{if } x \leq 0, \text{ or} \\ k \exp(-kx), \text{ where } k = 1 / \mathbf{p1}, & \text{otherwise.} \end{cases}$$

#### 5.9.8.9 kexprand

kopcode kexprand(ksig p1)

The **kexprand** core opcode generates random numbers from an exponential distribution with mean  $\mathbf{p1}$ . It is a run-time error if  $\mathbf{p1}$  is not strictly positive.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} 0 & \text{if } x \leq 0, \text{ or} \\ k \exp(-kx), \text{ where } k = 1 / \mathbf{p1}, & \text{otherwise.} \end{cases}$$

#### 5.9.8.10 aexprand

aopcode aexprand(asig p1)

The **aexprand** core opcode generates random noise according to an exponential distribution with mean  $\mathbf{p1}$ . It is a run-time error if  $\mathbf{p1}$  is not strictly positive.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \begin{cases} 0 & \text{if } x \leq 0, \text{ or} \\ k \exp(-kx), \text{ where } k = 1 / \mathbf{p1}, & \text{otherwise.} \end{cases}$$

#### 5.9.8.11 kpoissonrand

kopcode kpoissonrand(ksig p1)

The **kpoissonrand** core opcode generates a random binary (0/1) sequence of numbers such that the mean time between 1's is  $\mathbf{p1}$  seconds. It is a run-time error if  $\mathbf{p1}$  is not strictly positive.

On the first call to **kpoissonrand** with regard to a particular opcode state, a random number  $x$  shall be chosen according to the pdf

$$p(x) = \begin{cases} 0 & \text{If } x \leq 0, \text{ or} \\ k \exp(-kx), \text{ where } k = 1 / (\mathbf{p1} * \mathbf{KR}), & \text{otherwise.} \end{cases}$$

where **KR** is the orchestra control rate.

The return value shall be 0 and the floor of this random value shall be stored.

On subsequent calls, the stored value shall be decremented by 1. If the decremented value is -1, the return value shall be 1 and a new random value shall be generated and stored as described above. Otherwise, the return value shall be 0.

NOTE - The **kpoissonrand** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **kpoissonrand** is referenced twice in the same  $k$ -cycle, then the effective mean time between 1 values is half as long as given by  $t$ .

### 5.9.8.12 a poissonrand

aopcode a poissonrand(asig p1)

The **apoissonrand** core opcode generates random binary (0/1) noise such that the mean time between 1's is **p1** seconds. It is a run-time error if **p1** is not strictly positive.

On the first call to **apoissonrand** with regard to a particular opcode state, a random number  $x$  shall be chosen according to the pdf

$$p(x) = \begin{cases} 0 & \text{If } x \leq 0, \text{ or} \\ k \exp(-kx), \text{ where } k = 1 / (\mathbf{p1} * \mathbf{SR}), & \text{otherwise.} \end{cases}$$

where **SR** is the orchestra sampling rate.

The return value shall be 0 and the floor of this random value shall be stored.

On subsequent calls, the stored value shall be decremented by 1. If the decremented value is -1, the return value shall be 1 and a new random value shall be generated and stored as described above. Otherwise, the return value shall be 0.

NOTE - The **apoissonrand** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **apoissonrand** is referenced twice in the same  $a$ -cycle, then the effective mean time between 1 values is half as long as given by  $t$ .

### 5.9.8.13 igaussrand

iopcode igaussrand(ivar mean, ivar var)

The **igaussrand** core opcode generates a random number drawn from a Gaussian (normal) distribution with mean **mean** and variance **var**.

It is a run-time error if **var** is not strictly positive.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \frac{e^{-(\mathbf{mean}-x)^2/(2\mathbf{var})}}{\sqrt{2\pi \times \mathbf{var}}}$$

that is,  $p(x) \sim N(\mathbf{mean}, \mathbf{var})$  where **mean** is the mean and **var** the variance of a normal distribution.

**5.9.8.14 kgaussrand**

opcode `kgaussrand(ksig mean, ksig var)`

The **kgaussrand** core opcode generates random numbers drawn from a Gaussian (normal) distribution with mean **mean** and variance **var**.

It is a run-time error if **var** is not strictly positive.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \frac{e^{-(\text{mean}-x)^2/(2\text{var})}}{\sqrt{2\pi \times \text{var}}},$$

that is,  $p(x) \sim N(\text{mean}, \text{var})$  where **mean** is the mean and **var** the variance of a normal distribution.

**5.9.8.15 agaussrand**

opcode `agaussrand(asig mean, asig var)`

The **agaussrand** core opcode generates random noise drawn from a Gaussian (normal) distribution with mean **mean** and variance **var**.

It is a run-time error if **var** is not strictly positive.

The return value shall be a random number  $x$  chosen according to the pdf

$$p(x) = \frac{e^{-(\text{mean}-x)^2/(2\text{var})}}{\sqrt{2\pi \times \text{var}}},$$

that is,  $p(x) \sim N(\text{mean}, \text{var})$  where **mean** is the mean and **var** the variance of a normal distribution.

**5.9.9 Filters****5.9.9.1 port**

opcode `port(ksig ctrl, ksig htime)`

The **port** core opcode converts a step-valued control signal into a portamento signal. **ctrl** is an incoming control signal, and **htime** is the half-transition time in seconds to slide from one value to the next.

The return value is calculated as follows. On the first call to **port** with regard to a particular state, the *current value* and *old value* are both set to **ctrl**. On subsequent calls, if **ctrl** is not equal to the new value, then the old value is set to the current value, the *new value* is set to **ctrl** and the *current time* is set to 0.

If **htime** is 0, the current value is set to the new value.

The return value is calculated as follows. If the current value and new value are equal, then the return value is the new value. Otherwise, the current time is incremented by  $1/\mathbf{KR}$ , where **KR** is the orchestra control rate. Then, the current value shall be set to  $o + (n - o)(1 - 2^{-t/\text{htime}})$ , where  $t$  is the current time,  $n$  is the new value, and  $o$  is the old value.

NOTE - The **port** opcode does not have a "proper" representation of time, but infers it from the number of calls. If the same state of **port** is referenced twice in the same k-cycle, then the effective half-transition time is half as long as given by **htime**.

### 5.9.9.2 hipass

```
aopcode hipass(asig input, ksig cut)
```

The **hipass** core opcode high-pass filters its input signal. **cut** is the –6 dB cut-off point of the filter, and is specified in Hz. It is a run-time error if **cut** is not strictly positive.

The particular method of high-pass filtering is not normative. Any filter with the specified characteristic may be used.

The return value shall be the result of filtering **input** with a high-pass filter at **cut**.

NOTE - The **hipass** opcode is not required to have a “proper” representation of time, but is permitted to infer it from the number of calls. If the same state of **hipass** is referenced twice in the same a-cycle, the result is unspecified.

### 5.9.9.3 lopass

```
aopcode lopass(asig input, ksig cut)
```

The **lopass** core opcode low-pass filters its input signal. **cut** is the –6 dB cut-off point of the filter, and is specified in Hz. It is a runtime error if **cut** is not strictly positive.

The particular method of low-pass filtering is not normative. Any filter with the specified characteristic may be used.

The return value shall be the result of filtering **input** with a low-pass filter at **cut**.

NOTE - The **lopass** opcode is not required to have a “proper” representation of time, but is permitted to infer it from the number of calls. If the same state of **lopass** is referenced twice in the same a-cycle, the result is unspecified.

### 5.9.9.4 bandpass

```
aopcode bandpass(asig input, ksig cf, ksig bw)
```

The **bandpass** core opcode band-pass filters its input signal. **cf** is the centre frequency of the passband, and is specified in Hz. **bw** is the bandwidth of the filter, measuring from the –6 dB cut-off point below the centre frequency to the –6 dB point above, and is specified in Hz. It is a runtime error if **cf** and **bw** are not both strictly positive.

The particular method of bandpass filtering is not normative. Any filter with the specified characteristic may be used.

The return value shall be the result of filtering **input** with a bandpass filter with centre frequency **cf** and bandwidth **bw**.

NOTE - The **bandpass** opcode is not required to have a “proper” representation of time, but is permitted to infer it from the number of calls. If the same state of **bandpass** is referenced twice in the same a-cycle, the result is unspecified.

### 5.9.9.5 bandstop

```
aopcode bandstop(asig input, ksig cf, ksig bw)
```

The **bandstop** core opcode band-stop (notch) filters its input signal. **cf** is the centre frequency of the stopband, and is specified in Hz. **bw** is the bandwidth of the filter, measuring from the –6 dB cut-off point below the centre frequency to the –6 dB point above, and is specified in Hz. It is a runtime error if **cf** and **bw** are not both strictly positive.

The particular method of notch filtering is not normative. Any filter with the specified characteristic may be used.



The return value shall be the result of filtering **input** with a bandstop filter with centre frequency **cf** and bandwidth **bw**.

NOTE - The **bandstop** opcode is not required to have a “proper” representation of time, but is permitted to infer it from the number of calls. If the same state of **bandstop** is referenced twice in the same a-cycle, the result is unspecified.

#### 5.9.9.6 **biquad**

```
aopcode biquad(asig input, ivar b0, ivar b1, ivar b2, ivar a1, ivar a2)
```

The **biquad** core opcode performs exactly normative filtering using the canonical second-order filter in a “Transposed Direct Form II” structure. Using cascades of **biquad** opcodes allows the construction of arbitrary filters with exactly normative results.

The return value is calculated as follows. On the first call to **biquad** with regard to a particular state, the intermediate variables **ti**, **to**, **w1**, and **w2** are set to 0. Then, on the first call and each subsequent call, the following pseudo-code defines the functionality:

```
ti = input;
to = w2 + b0*ti;
w2 = w1 - a1*to + b1*ti;
w1 = -a2*to + b2*ti;
```

and the return value is **to**.

A runtime error is produced if this process produces out-of-bounds values (i.e., the filter is unstable).

The **biquad** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **biquad** is referenced twice in the same a-cycle, the effective sampling rate of the filter is twice as high as the orchestra sampling rate.

#### 5.9.9.7 **allpass**

```
aopcode allpass(asig input, ivar time, ivar gain)
```

The **allpass** core opcode performs allpass filtering on an input signal. The length of the feedback delay is **time** and is specified in seconds. It is a run-time error if **time** is not strictly positive.

Let **t** be the value  $\text{floor}(\text{time} * \text{SR})$ , where **SR** is the orchestra sampling rate. On the first call to **comb** with regard to a particular state, a delay line of length **t** is initialised and set to all zeros.

On the first and each subsequent call, let **x** be the value that was inserted into the delay line **t** calls ago, or 0 if there have not been **t** calls to this state. Let the value **output** be  $\text{x} - \text{input} * \text{gain}$ . Insert the value **output \* gain + input** into the beginning of the delay line. The return value shall be **output**.

NOTE - The **allpass** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **allpass** is referenced twice in the same a-cycle, then the effective allpass length is half as long as **len**.

#### 5.9.9.8 **comb**

```
aopcode comb(asig input, ivar time, ivar gain)
```

The **comb** core opcode performs comb filtering on an input signal. The length of the feedback delay is **time** and is specified in seconds. It is a run-time error if **time** is not strictly positive.

Let **t** be the value  $\text{floor}(\text{time} * \text{SR})$ , where **SR** is the orchestra sampling rate. On the first call to **comb** with regard to a particular state, a delay line of length **t** is initialised and set to all zeros.

On the first and each subsequent call, let **x** be the value that was inserted into the delay line **t** calls ago, or 0 if there have not been **t** calls to this state. Insert the value **x \* gain + input** into the beginning of the delay line. The return value shall be **x**.

NOTE - The **comb** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. That is, if the same state of **comb** is referenced twice in the same a-cycle, the effective length is half of **t**.

#### 5.9.9.9 fir

`aopcode fir(asig input, ksig b0[, ksig b1, ksig b2, ksig ...])`

The **fir** core opcode applies a specified FIR filter of arbitrary order to an input signal. The particular method of implementing FIR filters is not specified and left open to implementers.

The parameters **b0, b1, b2, ...** specify a FIR filter

$$H(z) = \mathbf{b0} + \mathbf{b1} z^{-1} + \mathbf{b2} z^{-2} + \dots$$

The return value shall be the successive values given by the application of this filter to the signal given by the value of **input** in successive calls to **fir**.

NOTE - The **fir** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **fir** is referenced twice in the same a-cycle, the effective filter sampling rate is double that of the orchestra sampling rate.

#### 5.9.9.10 iir

`aopcode iir(asig input, ksig b0[, ksig a1, ksig b1, ksig a2, ksig b2, ksig ...])`

The **iir** core opcode applies a specified IIR filter of arbitrary order to an input signal. The particular method of implementing IIR filters is not specified and left open to implementers.

The parameters **b0, b1, b2, ...** and **a1, a2, ...** specify an IIR filter

$$H(z) = (\mathbf{b0} + \mathbf{b1}z^{-1} + \mathbf{b2}z^{-2} + \dots) / (1 + \mathbf{a1}z^{-1} + \mathbf{a2}z^{-2} + \dots).$$

The return value shall be the successive values of the signal given by the application of this filter to the signal given by **input** in successive calls to **iir**. It is a run-time error if this application produces out-of-range values (that is, if the filter is unstable).

NOTE - The **iir** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **iir** is referenced twice in the same a-cycle, the effective filter sampling rate is double that of the orchestra sampling rate.

#### 5.9.9.11 firt

`aopcode firt(asig input, table t[, ksig order])`

The **firt** core opcode applies a specified FIR filter of arbitrary order, given in a table, to an input signal. The particular method of implementing FIR filters is not specified and left open to implementers.

The values stored in samples 0, 1, 2, ... **order** of table **t** specify a FIR filter

$$H(z) = \mathbf{t[0]} + \mathbf{t[1]} z^{-1} + \mathbf{t[2]} z^{-2} + \dots \mathbf{t[order-1]} z^{-\mathbf{order}+1}$$

where array notation is used to indicate wavetable samples. If **order** is not given or is greater than the size of the wavetable **t**, then **order** shall be set to the size of the wavetable. It is a run-time error if **order** is zero or negative.

The return values shall be the successive values of the signal given by the application of this filter to the signal given by the value of **input** in successive calls to **firt**.

NOTE - The **firt** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **firt** is referenced twice in the same a-cycle, the effective filter sampling rate is double that of the orchestra sampling rate.

### 5.9.9.12 iirt

```
aopcode iirt(asig input, table a, table b[, ksig order])
```

The **iirt** core opcode applies a specified IIR filter of arbitrary order, given in two tables, to an input signal. The particular method of implementing IIR filters is not specified and left open to implementers.

The values stored in samples 1, 2, ... **order** of table **a** and samples 0, 1, 2, ..., **order** of wavetable **b** specify a IIR filter

$$H(z) = (\mathbf{b}[0] + \mathbf{b}[1]z^{-1} + \mathbf{b}[2]z^{-2} + \dots) / (1 + \mathbf{a}[1]z^{-1} + \mathbf{a}[2]z^{-2} + \dots)$$

where array notation is used to indicate wavetable samples. (Note that sample 0 of wavetable **a** is not used). If **order** is not given or is greater than the size of the larger of the two wavetables, then **order** shall be set to the size of the greater of the two wavetables. If one wavetable is smaller than given by **order**, then the “extra” values shall be taken as zero coefficients. It is a run-time error if **order** is zero or negative.

The return values shall be the successive values of the signal given by the application of this filter to the signal given by the value of **input** in successive calls to **iirt**. It is a run-time error if this application produces out-of-range values (that is, if the filter is unstable).

NOTE - The **iirt** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **iirt** is referenced twice in the same a-cycle, the effective filter sampling rate is double that of the orchestra sampling rate.

## 5.9.10 Spectral analysis

### 5.9.10.1 fft

```
specialop fft(asig input, table re, table im[, ivar len, ivar shift, ivar size, table win])
```

The **fft** core opcode calculates windowed and overlapped DFT frames and places the complex-valued result in two tables. It is a “special opcode”; that is, it accepts values at the audio rate, but only returns then at the control rate.

There are several optional parameters. **len** specifies the length of the sample frame (the number of input samples to use). If **len** is zero or not provided, it is set to the next power of two greater than **SR/KR**, where **SR** is the orchestra sampling rate and **KR** is the orchestra control rate. **shift** specifies the number of samples by which to shift the analysis window. If **shift** is zero or not provided, it is set to **len**. **size** is the length of the DFT calculated by the opcode. If **size** is zero or not provided, it is set to **len**. **win** is the analysis window to apply to the analysis. If **win** is not provided, a boxcar window of length **len** is used.

It is a runtime error if any of the following apply: **len** is negative, **shift** is negative, **size** is negative, **size** is not a power of two, **size** is greater than 8192, **win** has fewer than **len** samples, **re** has fewer than **size** samples, or **im** has fewer than **size** samples.

The calculation of this opcode is as follows. On the first call to the **fft** opcode with respect to a particular state, a holding buffer of length **len** is created, and the first (**len - shift**) buffer samples are initialized with zeros. On each a-

rate call to the opcode, the **input** sample is inserted into the next uninitialized buffer sample position. When all **len** samples in the buffer are initialized, the following steps are performed:

1. A new buffer is created of length **size**, for which each value **new[i]** is set to the value **buf[i] \* win[i]**, where **new[i]** is the *i*th value of the new buffer, **buf[i]** is the value of the holding buffer, and **win[i]** is the value of the *i*th sample in the analysis-window wavetable. (The new buffer contains the pointwise product of the holding buffer and the analysis window). If **size > len**, then the values of **new[i]** for **i > len** are set to zero. If **size < len**, then only the first **size** values of the holding buffer are used.
2. The first **shift** samples are removed from the holding buffer and the remaining **len-shift** samples shifted to the front of the holding buffer. The **shift** samples at the end of the buffer after this shift are set to zero. If **shift > len**, the holding buffer is cleared.
3. The real DFT of the new buffer is calculated, resulting in a length-**size** complex vector of frequency-domain values. The real components of the DFT are placed in the first **size** samples of table **re**; the imaginary components of the DFT are placed in the first **size** samples of table **im**. The DFT is arranged such that the lowest frequencies, starting with DC, are at the zero point of the output tables, going up to the Nyquist frequency at **size/2**; the reflection of the spectrum from the Nyquist to the sampling frequency is placed in the second half of the tables.

The DFT is defined as

$$d[i] = \frac{\sum_{k=0}^{size-1} e^{-ijk2\pi/size} new[k]}{\sqrt{size}}$$

where **d[i]** are the resulting complex components of the DFT,  $0 < i < size$ ;  
**new[k]** are the input samples,  $0 < k < size$ ;  
and **j** is the square root of  $-1$ .

The return value on a particular *k*-cycle is 1 if a DFT was calculated since the last *k*-cycle, or 0 if one was not.

The **fft** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **fft** is referenced twice in the same *a*-cycle, then the effective FFT period is half as long as given by **len** and **shift**.

### 5.9.10.2 **ifft**

```
aopcode ifft(table re, table im[, ivar len, ivar shift, ivar size, table win])
```

The **ifft** core opcode calculates windowed and overlapped IDFTs and streams the result out as audio. **re** and **im** are wavetables that contain the real and imaginary parts of a DFT, respectively. There are several optional arguments that control the synthesis procedure. **len** is the number of output samples to use as audio; if **len** is zero or not given, it is taken as the next power of two greater than **SR/KR**, where **SR** is the orchestra sampling rate and **KR** is the orchestra control rate. **shift** is the number of samples by which the analysis window is shifted between frames; if **shift** is not given or is zero, it is taken as **len**. **size** is the size of the IDFT; if **size** is not given or is zero, it is taken as **len**. **win** is the synthesis window; if it is not given, a boxcar window of length **len** is assumed.

It is a run-time error if any of the following apply: **re** or **im** are shorter than length **size**, **win**, if given, is shorter than length **len**, **size** is not a power of two, **size** is greater than 8192, or **len**, **shift**, or **size** are negative.

The calculation for this opcode is as follows. On the first call to **ifft** with respect to a particular state, the size **size** IDFT of the tables **re** and **im** is calculated. If **re** and/or **im** are longer than **size** samples, only the first **size** samples of these tables shall be used. The result of this IDFT is a sequence of **size** values, potentially complex-valued. The real components of the first **len** elements of this sequence are multiplied point-by-point by the corresponding samples of the window **win** and placed in an output buffer of length **len**. (**out[i] = seq[i] \* win[i]** for  $0 < i < len$ ).

The IDFT is calculated with the assumption that the lowest-numbered elements of the tables **re** and **im** are the lowest frequencies of the audio signal, beginning with DC in sample 0, proceeding up to the Nyquist frequency in sample **size/2**, and then the reflected spectrum in samples **size/2** up to **size-1**.

The IDFT is defined as

$$seq[i] = \frac{\sum_{k=0}^{size-1} e^{ijk2\pi / size} d[k]}{\sqrt{size}}$$

where **d[i]** are the complex frequency components of the DFT,  $0 < i < \mathbf{size}$  (**d[i] = re[i] + j im[i]**)

**seq[k]** are the output samples,  $0 < k < \mathbf{size}$ ;

and **j** is the square root of  $-1$ .

Also on the first call to **ifft** with respect to a particular state, the output point of the synthesis is set to 0.

At each call to **ifft**, the following calculation is performed. The output value of the opcode is the value of the output buffer at the output point. Then, the output point is incremented. If the output point is thereby equal to **shift**, then the following steps shall be performed:

1. The first **shift** samples of the output buffer are discarded, the remaining **len-shift** samples of the output buffer are shifted into the beginning of the buffer, and the last **shift** samples are set to 0.
2. The IDFT of the current values of the **re** and **im** wavetables is calculated as described above. The first **len** values of the real part of the resulting audio sequence are multiplied point-by-point by the synthesis window **win**, and the result is added point-by-point to the output buffer (**out[i] = out[i] + seq[i] \* win[i] \* shift / len** for  $0 < i < \mathbf{len}$ ).
3. The output point is set to 0.

NOTE - The **ifft** opcode shall not have a "proper" representation of table, but shall infer it from the number of calls. If the same state of **ifft** is referenced twice in the same a-cycle, the result is undefined.

#### EXAMPLE

The **ifft** and **fft** opcodes can be used together to write instruments that use FFT-based spectral modification techniques, with logical syntax at the instrument level, in which the FFT frame rate is asynchronous with the control rate. The structure of such an instrument is:

```
instr spec_mod() {
  asig out;
  ksig new_fft;
  ivar length;
  table re_original(empty, 1024);
  table im_original(empty, 1024);
  table re_modified(empty, 1024);
  table im_modified(empty, 1024);

  length = 256;
  // no windowing; place FFT in "original"
  new_fft = fft(input, re_original, im_original, 1024, length);
  if (new_fft) { // read original tables, generate modified values,
                // and place these values into modified tables.
    .
    .
    .
  }

  // and output IFFT
  out = ifft(re_modified, im_modified, 1024, length);
  output(out);
}
```

}

Thus, every 256 samples (assuming 256 is greater than the number of samples in the control period), we compute the 1024-point IFFT. On those k-cycles during which we compute the FFT, we modify the table values in some interesting way. The IFFT operator produces continuous output, where every 256 samples the new table data is inspected and the IFFT calculated

There is nothing preventing us from manipulating the table data every control period, but only those values present in the table at the IFFT times will actually be turned into sound.

FFTs and IFFTs do not need to be implemented in pairs; other methods (such as table calculations) can be used to generate spectra to be turned into sound with IFFT, or rudimentary audio-pattern-recognition tools can be constructed that compute functions of the FFT and return or export the results.

### 5.9.11 Gain control

#### 5.9.11.1 rms

`specialop rms(asig x[, ivar length])`

The **rms** core opcode calculates the power in a signal. It is a “special opcode”; that is, it accepts values at the audio rate, but only returns them at the k-rate.

If **length** is not provided, it shall be set to the length of the control period. It is a run-time error if **length** is provided and is negative. The **length** parameter is specified in seconds.

The return value is calculated as follows. Let  $l$  be the value  $\text{floor}(\text{length} * \text{SR})$ , where **SR** is the orchestra sampling rate. A buffer  $b[]$ , of length  $l$ , is maintained of the most recent values provided as the **x** parameter. Each control period, the RMS of these values is calculated as

$$p = \sqrt{\frac{\sum_{i=0}^{l-1} b[i]^2}{l}}$$

and the return value is  $p$ .

NOTE - The **rms** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **rms** is referenced twice in the same a-cycle, then the effective length is half as long as given by **length**.

#### 5.9.11.2 gain

`aoopcode gain(asig x, ksig gain[, ivar length])`

The **gain** core opcode attenuates or increases the amplitude of a signal to make its power equal to a specified power level.

If **length** is not provided, it shall be set to the length of the control period. It is a run-time error if **length** is provided and is not strictly positive. The **length** parameter is specified in seconds.

The return value is calculated as follows. Let  $l$  be the value  $\text{floor}(\text{length} * \text{SR})$ , where **SR** is the orchestra sampling rate.

At the first call to the opcode, the attenuation level is set to 1. At each subsequent call, the input value **x** shall be stored in a buffer  $b[]$  of length  $l$ . When the buffer is full, the attenuation level is recalculated as

$$\text{gain} / \sqrt{\frac{\sum_{i=0}^{l-1} b[i]^2}{l}}$$

and the buffer is cleared.

The return value at each call is  $\mathbf{x} * \mathbf{A}$ , where  $\mathbf{A}$  is the current attenuation level.

NOTE - The **gain** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **gain** is referenced twice in the same a-cycle, then the effective buffer length is half as long as given by **length**.

### 5.9.11.3 balance

```
aopcode balance(asig x, asig ref[, ivar length])
```

The **balance** core opcode attenuates or increases the amplitude of a signal to make its power equal to the power in a reference signal.

If **length** is not provided, it shall be set to the length of the control period. It is a run-time error if **length** is provided and is not strictly positive. The **length** parameter is specified in seconds.

The return value is calculated as follows. Let  $l$  be the value  $\text{floor}(\text{length} * \mathbf{SR})$ , where  $\mathbf{SR}$ , is the orchestra sampling rate.

At the first call to the opcode, the attenuation level is set to 1. At each subsequent call, the input value  $\mathbf{x}$  shall be stored in a buffer  $\mathbf{b}[]$  of length  $l$ , and the input value **ref** stored in a buffer  $\mathbf{r}[]$  of length  $l$ . When the buffers are full, the attenuation level is recalculated as

$$\sqrt{\frac{\sum_{i=0}^{l-1} r[i]^2}{l}} / \sqrt{\frac{\sum_{i=0}^{l-1} b[i]^2}{l}}$$

and the buffers are cleared.

The return value at each call is  $\mathbf{x} * \mathbf{A}$ , where  $\mathbf{A}$  is the current attenuation level.

NOTE - The **balance** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **balance** is referenced twice in the same a-cycle, then the effective buffer length is half as long as given by **length**.

### 5.9.11.4 compressor

```
aopcode compressor (asig x, asig comp, ksig nfloor, ksig thresh, ksig loknee, ksig hiknee,
                    ksig ratio, ksig att, ksig rel, ivar look)
```

The **compressor** core opcode functions as an audio compressor, limiter, expander, or noise gate, using either soft-knee or hard-knee mapping, and with dynamically variable performance characteristics. It takes two audio-rate input signals,  $\mathbf{x}$  and **comp**, the first of which is modified by a running analysis of the second. Both signals may be the same, or the first can be modified by a different controlling signal.

It is a run-time error if any of the following conditions apply: **thresh** < **nfloor**, **loknee** < **thresh**, **hiknee** < **loknee**, **ratio** is equal to or less than 0, **look** is negative, **att** is negative, or **rel** is negative.

**compressor** first examines the controlling signal **comp** by performing envelope detection. This is directed by two control values **att**, **rel**, and an initialisation value **look**, defining the attack, release, and look-ahead times (in seconds) of the envelope detector.

**look** is the look-ahead time (in seconds) of the envelope detector. This determines how far ahead the detector looks for a new peak in a decaying signal. If a new peak is found, the release envelope is adjusted to interpolate between the current and future peaks.

**att** and **rel** are the attack and release times of the envelope detector (in seconds). They define the time it takes the envelope to reach a detected peak value (for **att**) and to reach zero (for **rel**).

**nfloor** will set the absolute decibel floor for the system. It defines the minimum value to which the compressor is potentially reactive. For instance, if **nfloor** is 0, according to the decibel scale of SAOL, the minimum recognized absolute value for input will be  $10^{-90/20}$ , if **nfloor** is -10 the minimum recognized absolute value will be  $10^{-(90+10)/20}$ , and so on.

The envelope estimate is converted to decibels, then passed through a mapping function (see Figure 5.3) to determine what compressor action (if any) shall be taken. The mapping function is defined by four regions: the zero region, the 1:1 (no change) region, the knee, and the compression/expansion region.

The locations of these regions are defined by the decibel control values, **thresh**, **loknee**, **hiknee**, and **ratio**.

**thresh** is the minimum decibel level that will be allowed through. For a noise gate, this value will be greater than **nfloor**.

**loknee** and **hiknee** are the decibel values that define where compression or expansion will begin. These set the boundaries of a soft-knee curve joining the low-amplitude 1:1 line and the higher-amplitude compression/expansion line. If the two breakpoints are equal, a hard-knee mapping will result.

**ratio**, given in dB, defines compression above the knee. It defines the change in input power required for a 1dB change in output power. **ratio** values above 1 dB result in compression, with higher values yielding greater compression. Numbers between 0 and 1 result in expansion. For example, a **ratio** value of 3dB implies a compression (a 3dB change in input power produces a 1dB change in output power) while a **ratio** value of 0.5dB implies an expansion (a 0.5dB change in input power produces a 1dB change in output power).

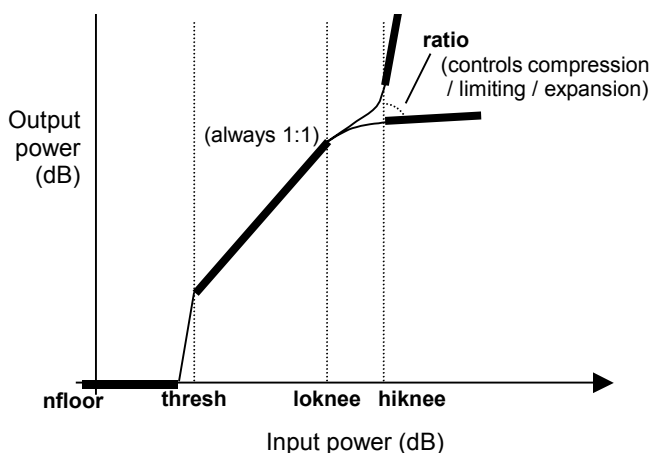


Figure 5.3 - Compressor characteristic function



The compression opcode parameters **nfloor**, **thresh**, **loknee**, and **hiknee** are defined on the decibel scale defined in subclause 5.9.4.4 for the **dbamp** core opcode. In this space, a waveform value of 1.0 maps to a decibel value of 90dB, a waveform value of 0.1 maps to 70dB, etc. Note that negative decibel values are permitted.

The actions of **compressor()** will depend on the parameter settings given. For instance, a hard-knee compressor is obtained from equal-value **loknee** and **hiknee** break-points. A noise-gate plus expander is obtained from some positive **thresh**, and a fractional **ratio** above the knee. A voice-activated music compressor (ducker) will result from feeding the music into **x** and the speech into **comp**. A voice de-esser will result from feeding the voice into both, with the **comp** version being preceded by a band-pass filter that emphasises the sibilants.

At initialisation, space shall be allocated for two buffers **xdly** and **compdly**. The length in samples of these buffers will be  $SR * look$ , where  $SR$  is the orchestra sampling rate. The initial values of both buffers will be set to zero.

Space is allocated for the following variables:

**gain**, the amplitude multiplier to be applied, initialised to 1.

**change**, the estimated change in envelope from sample to sample, initialised to 0.

**comp1**, the value of the current value of **comp** in dB, initialised to 0.

**comp2**, the delayed value of **comp** (from **compdelay**) in dB, initialised to zero.

**env**, the current envelope estimate, initialised to 0.

**projEnv**, the projected envelope value at the look-ahead point, initialised to 0.

At each a-rate call to **compressor()**, the following happens:

1. The sample **x** is placed into the beginning of the buffer **xdly**, pushing all values down and the oldest value off the end. The value pushed off the end is saved as **oldval**.
2. The next envelope value is calculated as follows:

$abs(comp)$  is converted to decibels. This value is called **comp1** with

$$comp1 = 90 + 20 * \log_{10}(abs(comp)).$$

It is put into the end of the buffer **compdly**. The value **comp2** is taken from the beginning of the buffer **compdly**.

if (**comp2** > **env**)

$$change = (comp2 - env) / (SR * att)$$

else {

$$projEnv = \max(change * SR * look, nfloor)$$

if ((**comp1** > **projEnv**) && (**comp2** > **comp1**))

$$change = (comp1 - comp2) / (SR * rel)$$

else

$$change = 0$$

}

$env = \max(env + change, nfloor)$

3. The amplitude multiplier **gain** is calculated as follows:

if (**env** < **thresh**)

**gain** shall first decrease monotonically from 1 to 0 and then stay at 0, moving from **thresh** towards **nfloor**, to create a noise gate. The exact input-output curve in this interval is not specified; it is left to the implementation. Implementations may also include temporal behavior in this regime to reduce onset and offset signal distortion.

else {

if (**env** <= **loknee**)

**gain** = 1

else if (**env** >= **hiknee**)

**gain** shall be calculated so that, above **hiknee**, an increase of **ratio** dB in input power must result in an increase of 1dB in output power. The decibel input/output curve shall be continuous at **hiknee** with either the soft or the hard knee curve.

else

**gain** shall be interpolated smoothly between the **loknee** and **hiknee** points to create a soft-knee input-output (decibel) curve. This curve shall be monotonically increasing, and have continuous derivative equal to 1 at **loknee** and  $1/ratio$  at **hiknee**. The exact input-output curve in this interval is not specified; it is left to the implementation.

}

4. The output value of the opcode shall be **oldval** multiplied by **gain**.

## 5.9.12 Sample conversion

### 5.9.12.1 decimate

specialop decimate(asig input)

The **decimate** core opcode decimates a signal from the audio rate to the control rate. It is a “special opcode”; that is, it accepts values at the audio rate, but only returns them at the k-rate.

The return value is calculated as follows. Each k-cycle, one of the values given as **input** in the preceding k-period of a-samples shall be returned.

NOTE - The **decimate** opcode is not required to have a “proper” representation of time, but is allowed to infer it from the number of calls. If the same state of **decimate** is referenced twice in the same a-cycle, then the return value for each call at the subsequent k-cycle may be taken from any of the values provided to the state during the preceding k-period.

#### EXAMPLE

```
oparray decimate[2];
ksig a,b,c;
```

```
a = decimate[0](1);
b = decimate[0](0);
c = decimate[1](2);
```

The value of **a** and **b** at each k-cycle shall be either 0 or 1, in an implementation-dependant manner. The value of **c** shall be 2.

### 5.9.12.2 upsamp

```
asig upsamp(ksig input[, table win])
```

The **upsamp** core opcode upsamples a control signal to an audio signal. **win** is an optional interpolation window. If **win** is not provided, it is taken to be a boxcar window (all values equal 1) of length **SR / KR**, where **SR** is the orchestra sampling rate and **KR** is the orchestra control rate. If **win** is provided and is shorter than **SR / KR** samples, it is zero-padded at the end to length **SR/KR** for use in this opcode (the table itself is not changed).

On the first call to **upsamp** with regard to a particular state, an output buffer of length **win** is created and set to zero. Also, the *output point* is set to 0.

On the first call to **upsamp** in a particular k-cycle with regard to a particular cycle, the output buffer is shifted by **SR / KR** samples: the first **SR / KR** samples are discarded, the remaining samples are shifted to the front of the output buffer, and the last **SR / KR** samples are set to 0. Then, the window function is scaled by **input** and added into the output buffer (**buf[i] = buf[i] + input \* win[i]**,  $0 < i < \text{length}(\text{win})$ ). Then, the output point is set to 0.

On the first call and each subsequent call to **upsamp**, the return value is the value of the output buffer at the current output point. Then, the output point shall be incremented.

It is a run-time error if the same state of **upsamp** is referenced more times than the length of **win** in a single k-cycle.

### 5.9.12.3 downsamp

```
specialop downsamp(asig input[, table win])
```

The **downsamp** core opcode downsamples an audio signal to a control signal. It is a “special opcode”; that is, it accepts samples at the audio rate but only returns values at the control rate. **win** is an optional analysis window.

It is a run-time error if **win** is shorter than **SR / KR** samples, where **SR** is the orchestra sampling rate and **KR** is the orchestra control rate.

The return value is calculated as follows: at each k-cycle, the values of each sample of **input** provided in the previous a-cycle are placed in a buffer. If **win** is not provided, then the return value is the mean of the samples in the buffer. If **win** is provided, then the return value is calculated by multiplying the analysis window point-by-point with the input signal ( **rtn =  $\sum \text{input}[i] * \text{win}[i]$**  for  $0 < i < \text{SR} * \text{KR}$ , where **SR** is the orchestra sampling rate and **KR** is the orchestra control rate).

NOTE - The **decimate** opcode does not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **decimate** is referenced twice in the same a-cycle, then the return value is calculated from the input values in the second half of the k-cycle.

### 5.9.12.4 samphold

```
opcode samphold(xsig input, ksig gate)
```

The **samphold** core opcode gates a signal with a control signal.

The return value is calculated as follows. On the first call to **samphold** with regard to a particular state, the last passed value is set to 0. If the value of **gate** is non-zero, then the last passed value is set to **input**. The last passed value is returned.

### 5.9.12.5 **sblock**

specialop sblock(asig x, table t)

The **sblock** core opcode creates control-rate blocks of samples and places them in a wavetable. It is a “special opcode”; that is, it accepts values at the audio rate, but only returns them at the k-rate.

It is a run-time error if the table **t** is not allocated with as much space as there are samples in the control period of the orchestra.

The return value of this opcode is always 0.

This opcode has side effects, as follows. Let *k* be the number of samples in a control period. At each k-cycle, the most recent *k* values of **x** are placed in table **t** such that the oldest value is placed in sample 0.

NOTE - The **sblock** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **sblock** is referenced twice in the same a-cycle, then the samples placed in the table shall be the interleaved values given in the two calls during the second half of the k-period.

## 5.9.13 Delays

### 5.9.13.1 **delay**

aopcode delay(asig x, ivar t)

The **delay** opcode implements a fixed-length end-to-end (i.e., untapped) delay line. **t** gives the length of the delay line in seconds. It is a run-time error if **t** < 0, unless the terminal is running in a negative-time universe.

Let *y* be floor(**t** \* **SR**) samples, where **SR** is the orchestra sampling rate. At each call to **delay** with respect to a particular opcode state, the value of **x** is inserted into a FIFO buffer of length *y*. The return value is the value that was inserted into the delay line *y* calls ago to **delay** with regard to the same state.

NOTE - The **delay** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **delay** is referenced twice in the same a-cycle, then the effective delay line is half as long as given by **t**.

### 5.9.13.2 **delay1**

aopcode delay1(asig x)

The **delay1** opcode implements a single-sample delay.

At each call to **delay1** with regard to a particular state, the value of **x** is stored, and the return value is the value stored on the previous call.

NOTE - The **delay1** opcode shall not have a “proper” representation of time, but shall infer it from the number of calls. If the same state of **delay1** is referenced twice in the same a-cycle, then the return value for the second call is the parameter value of the first.

### 5.9.13.3 **fracdelay**

aopcode fracdelay(ksig method[, xsig p1, xsig p2])

The **fracdelay** core opcode implements fractional, variable-length, and/or multitap delay lines. Several methods for manipulating the delay line are provided; in this way, **fracdelay** is like an object-oriented delay-line “class”.

The semantics of **p1** and **p2** and the calculation of the return value differ depending on the value of **method**. It is a run-time error if **method** is less than 1 or greater than 5.

If **method** is 1, the “initialise” method is specified. In this case, **p1** is the length of the delay line in seconds. It is a run-time error if **p1** is not provided, or is less than 0. Any currently existing delay line in this opcode state shall be destroyed, a new delay line with the specified length ( $\text{floor}(\mathbf{p1} * \mathbf{SR})$ , where **SR** is the orchestra sampling rate) shall be created, and all values on this delay line shall be initialised to 0. The return value is 0. **p2** is not used, and is ignored if provided.

If **method** is 2, the “tap” method is specified. In this case, **p1** is the position of the tap in seconds. It is a run-time error if method 1 has not yet been called for this opcode state, or if **p1** is not provided, or if **p1** is less than 0, or if **p1** is greater than the most recent initialisation length. The return value is the current value of the delay line at position  $\mathbf{p1} * \mathbf{SR}$ , where **SR** is the orchestra sampling rate. If  $\mathbf{p1} * \mathbf{SR}$  is not an integer, the return value shall be interpolated from the nearby values, as described by the global interpolation-quality parameter (subclause 5.8.5.2.5). **p2** is not used, and is ignored if provided.

If **method** is 3, the “set” method is specified. In this case, **p1** is the position of the insertion in seconds, and **p2** is the value to insert. It is a run-time error if method 1 has not yet been called for this opcode state, or if **p1** is not provided, or if **p1** is less than 0, or if **p1** is greater than the most recent initialisation length, or if **p2** is not provided. The value of the delay line at position  $\text{floor}(\mathbf{p1} * \mathbf{SR})$ , where **SR** is the orchestra sampling rate, is updated to **p2**. The return value is 0.

If **method** is 4, the “add into” method is specified. In this case, **p1** is the position of the insertion in seconds, and **p2** is the value to add in. It is a run-time error if method 1 has not yet been called for this opcode state, or if **p2** is not provided. Let *x* be the current value of the delay line at position  $\text{floor}(\mathbf{p1} * \mathbf{SR})$ , where **SR** is the orchestra sampling rate; then, the value of the delay line at this position is updated to  $x + \mathbf{p2}$ . The return value is  $x + \mathbf{p2}$ .

If **method** is 5, the “shift” method is specified. It is a run-time error if method 1 has not yet been called for this opcode state. All values of the delay line are shifted forward by one sample; that is, for each sample *x* where  $0 < x \leq L$ , where *L* is the length of the delay line, the new value of sample *x* of the delay line is the current value of sample *x* – 1. Sample 0 is set to value 0. The return value is the value shifted “off the end” of the delay line, that is the current value of sample *L*. **p1** and **p2** are not used, and are ignored if provided.

#### EXAMPLE

The following user-defined opcode implements the block diagram in Figure 5.4. We assume that the orchestra sampling rate is 10 Hz for clarity.

```

opcode example(asig a) {
  asig t1, t2, t3, x, first;
  oparray fracdelay[1];

  if (!itime) {
    fracdelay[0](1,1);      // initialise to 1 sec long
  }

  // flow network
  fracdelay[0](3,0,a);     // insert a at beginning
  t1 = fracdelay[0](2,0);  // tap at 0
  t2 = fracdelay[0](2,0.3); // tap at 0.3
  t3 = fracdelay[0](2,0.5); // tap at 0.5
  fracdelay[0](4,0.1,t3);  // feedback
  fracdelay[0](4,0.8,t1+t2); // feedforward
  x = fracdelay[0](5);     // shift and get output
  return(x);
}

```

Notice the use of the oparray construction (subclause 5.8.6.7.7) to implement this network. If an oparray is not used, then each call to **fracdelay** refers to a different delay line, and the algorithm makes no sense. Also note that **fracdelay**, unlike **delay**, does not shift automatically. For “typical” operations, method 5 should be called once per a-cycle.

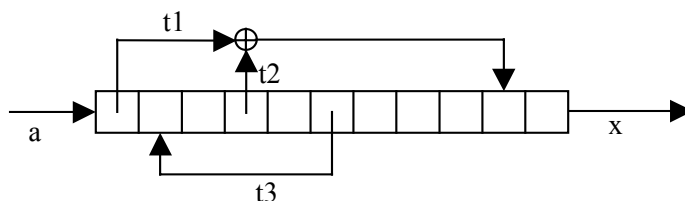


Figure 5.4 - Block diagram for 'fracdelay' example

## 5.9.14 Effects

### 5.9.14.1 reverb

```
opcode reverb(asig x, ivar f0[, ivar r0, ivar f1, ivar r1, ivar ...])
```

The **reverb** core opcode produces a reverberation effect according to the given parameters.

It is a run-time error if any **f** or **r** value is negative, or if there are an even number of parameters greater than 2.

If only one value **f0** is given as an argument, it is taken as a full-range reverberation time, that is, the amount of time delay until the sound amplitude is attenuated 60 dB compared to the source sound (RT60).

If more values are given, the **f** – **r** pairs represent responses at different frequencies. At each frequency **f** given as a parameter, the reverberation time (RT60) at that frequency is given by the corresponding **r** value.

The exact method of calculating the reverberation according to the specified parameters is not normative. If content authors wish to have exactly normative reverberations, they can easily be authored using the **comb**, **allpass**, **biquad**, **delay**, **fracdelay**, and other strictly normative core opcodes (q.v.).

The output shall be the reverberated sound signal.

### 5.9.14.2 chorus

```
asig chorus(asig x, ksig rate, ksig depth)
```

The **chorus** core opcode creates a sound with a chorusing effect, with rate **rate** and depth **depth**, from the input sound **x**. **rate** is specified in cycles per second; **depth** is specified as percent excursion with  $0 \leq \text{depth} \leq 100$ . It is a run-time error if **depth** < 0 or **depth** > 100.

The exact method of chorusing is non-normative and left open to implementers.

### 5.9.14.3 flange

```
asig flange(asig x, ksig rate, ksig depth)
```

The **flange** core opcode creates a sound with a flanged effect, with rate **rate** and depth **depth**, from the input sound **x**. **rate** is specified in cycles per second; **depth** is specified as percent excursion with  $0 \leq \text{depth} \leq 100$ . It is a run-time error if **depth** < 0 or **depth** > 100.

The exact method of flanging is non-normative and left open to implementers.

### 5.9.14.4 fx\_speedc

```
opcode fx_speedc(ivar speed_control_factor)
```

The **fx\_speedc** core opcode creates a sound with a speed change effect. This core opcode is only available in effects-processing orchestras (see subclause 5.15.3.5); it may not be used in the SAOL block of a Structured Audio bitstream.

The **fx\_speedc** core opcode directly accesses the special bus **input\_bus** (at ksmps per 1/k-rate) and performs a speed change at k-rate. Then the processed samples are once stored in a special buffer defined only for the speed control and are output as the signal at a rate of  $k/\text{speed\_control\_factor}$ , where  $k = \text{SR}/\text{KR}$ , SR is the orchestra sampling rate and KR is the orchestra control rate.

This indicates that the number of the input samples and that of the output samples are different when the sampling frequency is unchanged and the **fx\_speedc** core opcode shall be operated virtually at a rate of  $\text{speed\_control\_factor} * \text{KR}$  in order to keep the proper output sample-rate of  $k$  per control cycle. Therefore the input samples shall be also provided to the decoder at a rate of  $k$  per  $1/\text{speed\_control\_factor} * \text{KR}$  according to the **speed\_control\_factor** desired.

NOTE - This functionality is not supported for streaming transfer in ISO/IEC 14496-1 (MPEG-4 Systems). It is not therefore possible to apply the speed change to streaming media and the use is limited to applications such as storage media in which streaming data transfer is not required.

The exact method of speed change is non-normative and left open to implementers. Implementers are encouraged to provide the highest-quality speed change possible. As an example algorithm, the PICOLA speed change tool is described in Annex 5.D.

#### 5.9.14.5 speedt

```
iopcode speedt(table in, table out, ivar factor)
```

The **speedt** core opcode modifies a sound sample via time-scaling.

The **speedt** core opcode fills the sound sample in the wavetable **out** with a sound derived from the wavetable **in** by time-stretching without modifying the pitch. If **factor** < 1, the sound is compressed (accelerated) by the indicated factor; if **factor** > 1, the sound is expanded (slowed down) by the indicated factor. It is a run-time error if **factor** is not strictly positive, or if the wavetable **out** is not at least as long as **factor** multiplied by the length of the wavetable **in**.

The exact method of speed change is non-normative and left open to implementers. Implementers are encouraged to provide the highest-quality speed change possible. As an example algorithm, the PICOLA speed change tool is described in Annex 5.D.

### 5.9.15 Tempo functions

#### 5.9.15.1 gettempo

```
opcode gettempo([xsig dummy])
```

The **gettempo** core opcode returns the value in beats-per-minute of the current orchestra global tempo. The tempo by default is 60 beats per minute, but can be changed through the use of the **tempo** score line (subclause 5.11.5) or the **settempo** core opcode (subclause 5.9.15.2).

The **dummy** parameter is used to specify the rate of the opcode call if desired; see subclause 5.8.7.7.2.

#### 5.9.15.2 settempo

```
kopcode settempo(ksig x)
```

The **settempo** core opcode changes the value of the global orchestra tempo. The parameter **x** specifies the new tempo in beats-per-minute. It is a run-time error if **x** is not strictly positive. The return value is **x**.

This opcode has side-effects, as follows. All pending events are rescheduled as described in subclause 5.7.3.3.6, list item 7. The global orchestra tempo is set to **x**.

## 5.10 SAOL core wavetable generators

### 5.10.1 Introduction

This clause describes each of the core wavetable generators in SAOL. All core wavetable generators shall be implemented in every terminal that can decode Object type 3 or 4.

For each core wavetable generator, the following is described:

- A usage description, showing the parameters that are required to be provided in a table definition utilising this core wavetable generator.
- The normative semantics of the generator. These semantics describe how to calculate values and place them in the wavetable for each table definition using this generator.

For each core wavetable generator, the first field in the table definition is the name of the generator, and the value of the expression in the second field is the size of the wavetable. Many wavetable generators also allow the value  $-1$  in this field to signify dynamic calculation of the wavetable size. If the size is not  $-1$ , and is also not strictly greater than zero, then the syntax of the generator call is illegal. In each case, the **size** parameter shall be rounded to the nearest integer before evaluating the semantics as described below.

The subsequent expressions are the required and optional parameters to the generator. For ease of exposition, each of these parameter fields will be given a name in the description of the generators, but there is no normative significance to these names. Parameter fields enclosed in brackets are optional and may or may not occur in a table definition using that generator.

Each wavetable, as well as a block of data, has four parameters associated with it: the sampling rate loop start, loop end, and base frequency. For all wavetable generators except **sample**, these parameters shall be set to zero initially.

### 5.10.2 Sample

```
t1 table(sample, size, which[, skip])
```

The **sample** core wavetable generator allows the inclusion of audio samples (or other blocks of data) in the bitstream and subsequent access in the orchestra.

If **size** is  $-1$ , then the size of the table shall be the length of the audio sample. If **size** is given, and larger than the length of the audio sample, then the audio sample shall be zero-padded at the end to length **size**. If **size** is given, and smaller than the length of the audio sample, only the first **size** samples shall be used.

The **which** field identifies a sample. It is either a symbol, in which case the generator refers to a sample in the bitstream, by symbol number; or a number, in which case the generator refers to a sample stored as an **AudioBuffer** in the BIFS scene graph (ISO/IEC 14496-1 subclause 9.4.2.4).

In the case where the generator refers to a sample in the bitstream, for compliant bitstream implementations, the sample data is simply a stream of raw floating-point values. The most recent sample block of data with the given name (see subclause 5.5.2) shall be placed in the wavetable. If the bitstream sample data block contains sampling rate, loop start, loop end, and/or base frequency values, these parameters of the wavetable shall be set accordingly. If the sampling rate is not provided, it shall be set to the orchestra sampling rate by default. Any other parameters not so provided shall be set to 0.

In the case where the generator refers to a sample stored as an **AudioBuffer**, any audio coder described in ISO/IEC 14496-3 may be used to compress samples. The **children** fields of the **AudioSource** node responsible for instantiation of this orchestra refer to **AudioBuffer** nodes in this case. Each **AudioBuffer** contains, after buffering as described in ISO/IEC 14496-1 subclause 9.2.2.13, several channels of audio data. If the first child has  $n_0$  channels, the second  $n_1$  channels, and so forth up to child  $k-1$ , then this **AudioSource** node has  $K = n_0 + n_1 + \dots + n_{k-1}$  channels in all, and **which** shall be a value between 0 and  $K-1$ . Channel **which** (where **which** is rounded to



the nearest integer if necessary), numbering in order across children and their channels, shall be placed in the wavetable. The sampling rate of the wavetable shall be set to the sampling rate of the **AudioBuffer** node from which channel **which** is taken. The loop start, loop end, and base frequency values shall be set to 0.

If the selected **AudioBuffer** node is not finished capturing data when the generator is executed (that is, the generator is executed less than **length** seconds after the **length** field of the **AudioBuffer** is set or changed), then the bitstream is in error. That is, this form of this generator shall only be used in cases where there is time allotted in the bitstream for the other decoders to produce samples (in real-time) before the generator executes. This is likely done by including the table generator in a score line scheduled to execute after the Composition Time (see ISO/IEC 14496-1 subclause 7.3.5) of the last audio Access Unit needed in the **AudioBuffer** node.

For standalone systems such as authoring tools, implementers are encouraged to provide access to other audio file formats and disk file access using this field (for example, to allow a filename as a string constant here). However, the only normative behaviours are those described in this subclause.

If **skip** is provided and is a positive value, it is rounded to the nearest integer, and the data placed in the wavetable begins with sample **skip**+1 of the bitstream or **AudioBuffer** sample data.

### 5.10.3 Data

```
t1 table(data, size, p1, p2, p3, ...)
```

The **data** core wavetable generator allows the orchestra to place data values directly into a wavetable.

If **size** is –1, then the size of the table shall be the number of data values specified. If **size** is given, and larger than the number of data values, then the wavetable shall be zero-padded at the end to length **size**. If **size** is given, and smaller than the number of data values, then only the first **size** values shall be used.

The **p1**, **p2**, **p3** ... fields are floating-point values that shall be placed in the wavetable

### 5.10.4 Random

```
t1 table(random, size, dist, p1[, p2])
```

The **random** core wavetable generator fills a wavetable with pseudo-random numbers according to a given distribution. For all pseudo-random number generation algorithms, they shall be reseeded upon orchestra start-up such that each execution of an orchestra containing these instructions generates different numbers.

If **size** is –1, the generator is illegal and a run-time error generated. If the **size** field is a positive value, then this shall be the length of the table, and this many independent random numbers shall be computed to place in the table.

The **dist** field specifies which random distribution to use, and the meanings of the **p1** and **p2** fields vary accordingly.

If **dist** is 1, then a uniform distribution is used. Pseudo-random numbers are computed such that all floating-point values between **p1** and **p2** inclusive have equal probability of being chosen for any sample.

If **dist** is 2, then a linearly ramped distribution is used. Pseudo-random numbers are computed such that the probability distribution function of choosing  $x$  for any sample is given by

$$p(x) = \begin{cases} 0 & \text{if } x \leq \mathbf{p1} \text{ or } x > \mathbf{p2}, \text{ or} \\ \text{abs}(2 / (\mathbf{p2} - \mathbf{p1}) \times [(x - \mathbf{p1}) / (\mathbf{p2} - \mathbf{p1})]) & \text{otherwise.} \end{cases}$$

A run-time error is generated if **dist** is 2 and **p1** = **p2**.

If **dist** is 3, then an exponential distribution is used. Pseudo-random numbers are computed such that the probability distribution function of choosing  $x$  for any sample is

$$p(x) = \begin{cases} 0 & \text{if } x \leq 0, \text{ or} \\ k \exp(-kx), \text{ where } k = 1 / \mathbf{p1}, & \text{otherwise.} \end{cases}$$

If **dist** is 3, then **p2** is not used and is ignored if it is provided.

If **dist** is 4, then a Gaussian distribution is used. Pseudo-random numbers are computed such that the probability distribution function of choosing  $x$  for any sample is

$$p(x) = \frac{e^{-(\text{mean}-x)^2/(2\text{var})}}{\sqrt{2\pi \times \text{var}}},$$

that is,  $p(x) \sim N(\mathbf{p1}, \mathbf{p2})$  where **p1** is the mean and **p2** the variance of a normal distribution.

If **dist** is 4, then **p2** shall be strictly greater than 0, otherwise a run-time error is generated.

If **dist** is 5, then a Poisson process is modelled, where the mean number of samples between 1's is given by an exponential distribution with mean **p1**. A pseudo-random value is computed according to  $p(x)$  as given for **dist** = 3 (the exponential distribution), above. This value is rounded to the nearest integer  $y$ . The first  $y$  values of the table (elements 0 through  $y-1$ ) are set to 0, and the next value (element  $y$ ) to 1. Another pseudo-random value is computed as if **dist** =3, and rounded to the nearest integer  $z$ . The next  $z$  values (elements  $y + 1$  through  $y + z$  in the table) are set to 0, and the next value (element  $y + z + 1$ ) to 1. This process is repeated until the table is full through element **size**. The resulting table has length **size** regardless of the values generated in the pseudo-random process; the last element may be a zero or 1.

If **dist** is 5, then **p2** is not used and is ignored if provided.

If **dist** is less than 0 or greater than 5, a run-time error is generated.

### 5.10.5 Step

```
t1 table(step, size, x1, y1, x2, y2, ...)
```

The **step** core wavetable generator allows arbitrary step functions to be placed in a wavetable. The step function is computed from pairs of (**x**, **y**) values.

If **size** is  $-1$ , then the size of the wavetable shall be the size of the largest x-value parameter. If **size** is larger than the largest x-value parameter, then the wavetable shall be padded with 0 values at the end to size **size**. If **size** is smaller than the largest x-value provided, then only the first **size** values shall be computed and used.

It is a run-time error if:

- **x1** is not 0,
- the x-values are not a non-decreasing sequence, or
- there are an even number of parameters, not counting the **size** parameter.

For the **step** generator, sample values 0 through **x2-1** shall be set to **y1**, **x2** through **x3-1** shall be set to **y2**, **x3** through **x4-1** shall be set to **y3**, and so forth.

### 5.10.6 Lineseg

```
t1 table(lineseg, size, x1, y1, x2, y2, ...)
```

The **lineseg** core wavetable generator allows arbitrary line-segment functions to be placed in a wavetable. The line segment function is computed from pairs of (**x**, **y**) values.

If **size** is  $-1$ , then the size of the wavetable shall be the size of the largest x-value parameter. If **size** is larger than the largest x-value parameter, then the wavetable shall be padded with 0 values at the end to size **size**. If **size** is smaller than the largest x-value provided, then only the first **size** values shall be computed and used.

It is a run-time error if:

- **x1** is not 0,
- the x-values are not a non-decreasing sequence, or
- there are an odd number of parameters, not counting the **size** parameter.

For the **step** generator, sample values for samples

x in the range **x1** through **x2** shall be set to  $y1 + (y2-y1)(x - x1) / (x2 - x1)$ ,

x in the range **x2** through **x3** shall be set to  $y2 + (y3-y2)(x - x2) / (x3 - x2)$ ,

and so forth.

If any two successive x-values are equal, a discontinuous function is generated, and no values shall be calculated for the “range” corresponding to those values.

### 5.10.7 Expseg

```
t1 table(expseg, size, x1, y1, x2, y2, ...)
```

The **expseg** core wavetable generator allows arbitrary exponential-segment functions to be placed in a wavetable. The function is computed from pairs of (**x**, **y**) values.

If **size** is  $-1$ , then the size of the wavetable shall be the size of the largest x-value parameter. If **size** is larger than the largest x-value parameter, then the wavetable shall be padded with 0 values at the end to size **size**. If **size** is smaller than the largest x-value provided, then only the first **size** values shall be computed and used.

It is a run-time error if:

- **x1** is not 0,
- the x-values are not a non-decreasing sequence,
- the y-values are not all of the same sign,
- any y-value is equal to 0, or
- there are an odd number of parameters, not counting the **size** parameter.

For the **expseg** generator, sample values for samples

x in the range **x1** through **x2** shall be set to  $y1(y2/y1)^{(x-x1)/(x2-x1)}$ ,

x in the range **x2** through **x3** shall be set to  $y2(y3/y2)^{(x-x2)/(x3-x2)}$ ,

and so forth.

If any two successive x-values are equal, a discontinuous function is generated, and no values shall be calculated for the “range” corresponding to those values.

### 5.10.8 Cubicseg

```
t1 table(cubicseg, size, inf11, y1, x1, y2, inf12, y3, x2, y4, inf13, y5, ...)
```

The **cubicseg** core wavetable generator creates a function made up of segments of cubic polynomials. Each segment is specified in terms of endpoints and an inflection point. If, for successive segments, the y-values at the inflection points are between the y-values at the endpoints, then the function is smooth; otherwise, the function is pointy or “comb-like”.

If **size** is  $-1$ , then the size of the wavetable shall be the size of the largest x-value parameter. If **size** is larger than the largest x-value parameter, then the wavetable shall be padded with 0 values at the end to size **size**. If **size** is smaller than the largest x-value provided, then only the first **size** values shall be computed and used.

It is a run-time error if:

- **infl1** is not 0,
- the x-values are not a non-decreasing sequence,
- any infl-value is not strictly between the two surrounding x-values,
- there are less than two x-values, or
- the sequence of control values does not end with an y-value

For the **cubicseg** generator, sample values for samples numbered:

x in the range **infl1** to **infl2** shall be set to  $ax^3 + bx^2 + cx + d$ , where **a**, **b**, **c**, and **d** are the coefficients of a cubic polynomial that passes through (**infl1**,**y1**), (**x1**,**y2**), and (**infl2**,**y3**) and that has 0 derivative at **x1**;

x in the range **infl2** to **infl3** shall be set to  $ax^3 + bx^2 + cx + d$ , where **a**, **b**, **c**, and **d** are the coefficients of a cubic polynomial that passes through (**infl2**,**y3**), (**x2**,**y4**), and (**infl3**,**y5**) and that has 0 derivative at **x2**;

and so on.

If, for any segment, such a cubic polynomial does not exist or does not have real values through the segment range, it is a run-time error.

### 5.10.9 Spline

```
t1 table(spline, size, x1, y1, k2, x2, y2, k3, ...)
```

The **spline** core wavetable generator creates a smoothly varying “spline” function for a set of control points.

If **size** is  $-1$ , then the size of the wavetable shall be the size of the largest x-value parameter. If **size** is larger than the largest x-value parameter, then the wavetable shall be padded with 0 values at the end to size **size**. If **size** is smaller than the largest x-value provided, then only the first **size** values shall be computed and used.

It is a run-time error if:

- **x1** is not 0,
- the x-values are not a non-decreasing sequence,
- there are less than two x-values,
- there are less than 4 parameters, not including **size**, or
- the last parameter is not a *k* value

For the **spline** generator, sample values for samples numbered:

x in the range **x1** to **x2** shall be set to  $ax^3 + bx^2 + cx + d$ , where **a**, **b**, **c**, and **d** are the coefficients of a cubic polynomial that passes through (**x1**,**y1**), and (**x2**,**y2**) and that has derivative 0 at **x1** and derivative **k2** at **x2**;

x in the range **x2** to **x3** shall be set to  $ax^3 + bx^2 + cx + d$ , where **a**, **b**, **c**, and **d** are the coefficients of a cubic polynomial that passes through (**x2**,**y2**), and (**x3**,**y3**) and that has derivative **k2** at **x2** and derivative **k3** at **x3**;

x in the range **x3** to **x4** shall be set to  $ax^3 + bx^2 + cx + d$ , where **a**, **b**, **c**, and **d** are the coefficients of a cubic polynomial that passes through (**x3**,**y3**), and (**x4**,**y4**) and that has derivative **k3** at **x3** and derivative **k4** at **x4**; and so on.

The derivative of the last cubic section shall be zero at **xn**, the last x-point of the sequence.

If, for any segment, such a cubic polynomial does not exist or is not real-valued over the segment range, it is a run-time error.

### 5.10.10 Polynomial

```
t1 table(polynomial, size, xmin, xmax, a0, a1, a2, ...)
```

The **polynomial** core wavetable generator allows an arbitrary section of an arbitrary polynomial function to be placed in a wavetable. The polynomial function used is  $p(x) = a_0 + a_1x + a_2x^2 + \dots$ ; it is evaluated over the range [ **xmin**, **xmax** ].

It is a run-time error if **size** is not strictly positive, or if there are not at least 3 parameters, not counting the **size** parameter, or if **xmin** = **xmax**.

For the **polynomial** generator, the sample value for sample  $x$  in the range [0, **size-1**] inclusive shall be set to

$$a_0 + a_1y + a_2y^2 + \dots, \text{ where } y = \text{xmin} + (\text{size} - x) / \text{size} \times (\text{xmax} - \text{xmin}).$$

### 5.10.11 Window

```
t1 table(window, size, type[, p])
```

The **window** core wavetable generator allows a windowing function to be placed in a table.

It is a run-time error if the **size** parameter is not strictly positive, or if **type** = 5 and the **p** parameter is not included.

The window type is specified by the **type** parameter. This parameter shall be rounded to the nearest integer, and then interpreted as follows:

If **type**=1, a Hamming window shall be used. For sample number  $x$  in the range [ 0, **size** - 1], the value placed in the table shall be

$$0.54 - 0.46 \cos (2\pi x / (\text{size} - 1)).$$

If **type**=2, a Hanning (raised cosine) window shall be used. For sample number  $x$  in the range [ 0, **size** - 1], the value placed in the table shall be

$$0.50 ( 1 - \cos (2\pi x / (\text{size} - 1))).$$

If **type**=3, a Bartlett (triangular) window shall be used. For sample number  $x$  in the range [0, **size** - 1], the value placed in the table shall be

$$1 - 2 | x - (\text{size} - 1) / 2 | / (\text{size} - 1).$$

If **type**=4, a Gaussian window shall be used. For sample number  $x$  in the range [0, **size-1**], the value placed in the table shall be

$$e^{-c_1(c_2-x)(c_2+x)}, \text{ where } c_2 = \text{size}/2 \text{ and } c_1 = 18/(\text{size})^2.$$

If **type**=5, a Kaiser window shall be used, with parameter **p**. For sample number  $x$  in the range [ 0, **size** - 1], the value placed in the table shall be

$$\frac{I_0 \left[ p \sqrt{\left( \frac{\text{size}-1}{2} \right)^2 - \left( x - \frac{\text{size}-1}{2} \right)^2} \right]}{I_0 \left[ p \left( \frac{\text{size}-1}{2} \right) \right]},$$

where  $I_0(x)$  is the zero-order modified Bessel function of the first kind.

If **type**=6, a boxcar window shall be used. Each sample in the range [0, **size** – 1] shall be given the value 1.

### 5.10.12 Harm

`t1 table(harm, size, f1, f2, f3...)`

The **harm** generator creates one cycle of a composite waveform made up of a weighted sum of zero-phase sinusoids.

It is a run-time error if **size** is not strictly positive.

For each sample  $x$  in the range [0, **size** – 1], the sample shall be assigned the value

$$f1 \sin (2 \pi x / \text{size}) + f2 \sin (4 \pi x / \text{size}) + f3 \sin (6 \pi x / \text{size}) + \dots$$

### 5.10.13 Harm\_phase

`t1 table(harm_phase, size, f1, ph1, f2, ph2, ...)`

The **harm\_phase** core wavetable generator creates one cycle of a composite waveform made up of a weighted sum of zero-DC sinusoids, each with specified initial phase in radians.

It is a run-time error if **size** is not strictly positive, or if there are an odd number of parameters, not counting the **size** parameter.

For each sample  $x$  in the range [0, **size** – 1], the sample shall be assigned the value

$$f1 \sin (2 \pi x / \text{size} + \text{ph1}) + f2 \sin (4 \pi x / \text{size} + \text{ph2}) + f3 \sin (6 \pi x / \text{size} + \text{ph3}) + \dots$$

### 5.10.14 Periodic

`t1 table(periodic, size, p1, f1, ph1, p2, f2, ph2, ...)`

The **periodic** core wavetable generator creates one cycle of an arbitrary periodic waveform, parameterised as the sum of several sinusoids with arbitrary frequency, magnitude and phase. The phase values (**ph1**, **ph2**, ...) are specified in radians.

It is a run-time error if **size** is not strictly positive, or if the number of parameters, not counting the **size** parameter, is not evenly divisible by three.

For each sample  $x$  in the range [0, **size** – 1], the sample shall be assigned the value

$$f1 \sin (2 p1 \pi x / \text{size} + \text{ph1}) + f2 \sin (2 p2 \pi x / \text{size} + \text{ph2}) + f3 \sin (2 p3 \pi x / \text{size} + \text{ph3}) + \dots$$

Any of the **p1**, **p2**, **p3**, etc. values may be zero, in which case the corresponding term of the calculation is a DC term; or non-integral, in which case there is a discontinuity at the table wrap point, or negative, which means the corresponding term evolves as a negative phase term. In all cases, the above value expression holds as specified.

### 5.10.15 Buzz

`t1 table(buzz, size, nharm, lowharm, rolloff)`

The **buzz** core wavetable generator creates one cycle of the sum of a series of spectrally-sloped cosine partials (band-limited pulse train). This waveform is a good source for subtractive synthesis.

It is a run-time error if **size** is not strictly positive, and **nharm** is also not strictly positive.

**lowharm** and **nharm** shall be rounded to the nearest integer before further processing.

If **size** is not strictly positive, then the size of the table is given by the highest harmonic included, such that **size** = 2 (**lowharm** + **nharm**).

If **nharm** is not strictly positive, then the number of harmonics shall be given by the size of the table, such that **nharm** is the greatest integer smaller than **size/2** – **nharm**.

For each sample  $x$  in the range  $[0, \mathbf{size} - 1]$ , the sample shall be assigned the value

$$\mathbf{scale} * \sum_{f=\mathbf{lowharm}}^{\mathbf{lowharm}+\mathbf{nharm}} \mathbf{rolloff}^{(f-\mathbf{lowharm})} \cos 2\pi fp$$

where  $p$  is the value  $x / \mathbf{size}$  and **scale** is the value  $(1-\mathbf{abs}(\mathbf{rolloff})) / (1-\mathbf{abs}(\mathbf{rolloff}^{\mathbf{nharm}}))$ .

If **rolloff** is negative, then alternating partials alternate phase direction; if  $|\mathbf{r}| < 1$ , then partials attenuate as they get higher in frequency; otherwise, they stay the same or grow in magnitude; in all cases, the above value expression holds as specified.

### 5.10.16 Concat

```
table t1(concat, size, ft1, ft2, ...)
```

The **concat** generator allows several tables to be concatenated together into a new table.

It is a runtime error if no tables are provided as arguments.

If **size** is not strictly positive, the size of the wavetable shall be the sum of the sizes of the parameter wavetables. If **size** is strictly positive, but smaller than the sum of the sizes of the parameter wavetables, then only the first **size** points of the parameter wavetables shall be used. If **size** is larger than the sum of the sizes of the parameter wavetables, then the generated wavetables shall be zero-padded at the end to size **size**.

The values of the wavetable shall be calculated as follows: for each sample  $x$  in the range  $[0, s_1 - 1]$ , where  $s_1$  is the size of the wavetable referenced by **p1**, the sample shall be assigned the same value as sample  $x$  of **p1**; for each sample  $x$  in the range  $[s_1, s_1 + s_2 - 1]$ , where  $s_2$  is the size of the wavetable referenced by **p2**, the sample shall be assigned the same value as sample  $x - s_1$  of **p2**; and so on, up to sample **size**.

### 5.10.17 Empty

```
t1 table(empty, size)
```

The **empty** generator allocates space and fills it with zeros.

It is a run-time error if **size** is not strictly positive.

For each sample in the range  $[0, \mathbf{size} - 1]$ , the sample is assigned value 0.

This generator is useful in conjunction with user-defined opcodes that fill up a table with data.

## 5.11 SASL syntax and semantics

### 5.11.1 Introduction

This clause describes the syntax and semantics of the score language SASL. SASL allows the simple parametric description of events that use an orchestra to generate sound, including notes, controllers, and dynamic wavetable

generation. SASL is simpler than many previously existing score languages; this is intentional, as it enables easier cross-coding of score data from other formats into SASL. Since in many cases, SASL code is automatically generated by authoring tools, it is not a great disadvantage to have relatively simple syntax and few “defaults”.

As with the SAOL description in clause 5.8, this clause describes a textual representation of SASL that is standardised, but stands outside of the bitstream-decoder relationship. It also describes the mapping between the textual representation and the bitstream representation. The exact normative semantics of SASL will be described in reference to the textual representation, but also apply to the tokenised bitstream representation as created via the normative tokenisation mapping.

All times in the score file (start times and durations) are specified in *score time*, which is measured in *beats*. By default, the score time is equivalent to the absolute time, and thus events with duration of one beat last one second, and an event dispatched two beats of score time after another is dispatched two seconds later by the scheduler. However, this mapping can be changed with the **tempo** command, see below.

Each score line may be prefaced by an optional \* tag. This tag indicates that the event is a high-priority event as described in subclause 5.7.3.3.7.

NOTE - In streaming performance of Structured Audio bitstreams, some events have no timestamps. This is possible because the streaming mechanism contains intrinsic time, i.e. “right now”. For the textual score format, there is no such intrinsic time, and thus every score line in textual format is required to have a time field. A Structured Audio encoder (see Annex B) has the option of retaining or removing time fields when tokenizing the orchestra and packing tokenized score events into Access Units, depending on the requirements of the application. For the same reason, the “use if late” bitstream flag is not used in the textual score format.

### 5.11.2 Syntactic form

<score file> -> <score line> [ <score file> ]  
 <score file> -> <score line>

<score line> -> (\*) <instr line> <newline>  
 <score line> -> (\*) <control line> <newline>  
 <score line> -> (\*) <tempo line> <newline>  
 <score line> -> (\*) <table line> <newline>  
 <score line> -> <end line> <newline>

<instr line> -> [ <ident> :] <number> <ident> <number> <pflist>

<control line> -> <number> [ <ident> ] control <ident> <number>

<tempo line> -> <number> tempo <number>

<table line> -> <number> table <ident> <ident> <pflist>

<end line> -> <number> end

<pflist> -> <number> [ <pflist> ]  
 <pflist> -> <NULL>

<number> as given in subclause 5.8.2.3.

<ident> as given in subclause 5.8.2.2.

### 5.11.3 Instr line

The **instr** line specifies the construction of an instrument instantiation at a given time.



The first identifier, if given, is a label that is used to identify the instantiation for use with further control events.

The first number is the score time of the event. As much precision as desired may be used to specify times; however, instruments are only dispatched as fast as the orchestra control rate, as described in subclause 5.7.3.3. Event times do not have to be received, or present in the score file, in temporal order.

The second identifier (the first required identifier) is the name of the instrument, used to select one instrument from the orchestra described in the SAOL bitstream element. It is a syntax error if there is not an instrument with this name in the orchestra when the orchestra is started.

The second number is the score duration of the instrument instance. When the instrument instantiation is created, a termination event shall be scheduled (see subclause 5.7.3.3) at the time given by the sum of the instantiation time and the note duration. If this field is  $-1$ , then the instrument shall have no scheduled duration.

The pflist is the list of parameter fields to be passed to the instrument instance when it is created. If there are more pfields specified in the instrument declaration than elements of this list, the remaining pfields shall be set to 0 upon instantiation. If there are fewer pfields than elements, the extra elements shall be ignored.

#### 5.11.4 Control line

The **control** line specifies a control instruction to be passed to the orchestra, or to a set of running instruments.

The first number is the score time of the control event. When this time arrives in the orchestra, the control event is dispatched according to its particular semantics.

The first identifier, if provided, is a label specifying which instrument instances are to receive the event. If this label is provided, when the control event is dispatched, any active instrument instances that were created by **instr** events with the same label receive the control event. If the label is provided, and there are no such active instrument instances, the control event shall be ignored. If the label is not provided, then the control event references a global variable of the orchestra.

The second identifier (the first required identifier) is the name of a variable that will receive the event. For labelled control lines, the name references a variable in instruments that were created based upon **instr** events with the same label. If there is no such name in a particular instrument instance, then the control event shall be ignored for that instance. For unlabelled lines, the name references a global variable of the orchestra with the same name. If there is no such global variable, then the control event shall be ignored.

The second number is the new value for the control variable. When the control event is dispatched, variables in the orchestra as identified in the preceding paragraph shall have their values set to this value.

#### 5.11.5 Tempo line

The **tempo** line in the score specifies the new tempo for the decoding process. The tempo is specified in beats-per-minute; the default tempo shall be sixty beats per minute, and thus by default the score time is measured in seconds.

The first number in the tempo line is the score time at which the tempo changes. When this time arrives, the tempo event shall be dispatched as described in subclause 5.7.3.3, list item 7.

The second number is the new tempo, specified in beats per minute. Consequently, one beat lasts  $60/\text{tempo}$  seconds, so that a tempo of 120 beats per minute is twice as fast as the default. When a tempo line is decoded, the time numbers in the score continue progressively, with the increments now in accordance with the new time unit.

#### 5.11.6 Table line

The **table** line in the score specifies the creation or destruction of a wavetable.

The first number in the score line is the score time at which the wavetable is created or destroyed. For creation events, the wavetable shall be created at this time. For destruction events, the wavetable shall not be destroyed before this time.

The first identifier is the name of the wavetable. This name references a wavetable in the global orchestra scope.

The second identifier is either the name of the table generator, or the special name **destroy**. It is a syntax error if this identifier is not the name of one of the core wavetable generators listed in clause 5.10, or the special name **destroy**.

The pfield list is the list of parameters to the particular core wavetable generator. Not every sequence of parameters is legal for every table generator; see the definitions in clause 5.10.

The **sample** core wavetable generator refers to a sound sample (see subclause 5.10.2). Implementations providing textual interfaces are suggested to provide access to commonly-used “soundfile” formats in the first pfield as a string constant. However, this is non-normative; the only normative aspect is as follows. In a bitstream **table** score line object, the **refers\_to\_sample** bit may be set. If this is the case, then the **sample** token of that score line object shall refer to another bitstream object containing the sample data, and it is this sample data that shall be placed in the wavetable.

When the dispatch time of the table event is received, if the table line references the **destroy** name, then any global wavetable with that name may be destroyed and its memory freed. If the table line specifies creation of a wavetable, and there is already a global wavetable with the same name, the new wavetable replaces the existing wavetable. That is, the global wavetable with that name may be destroyed and its memory freed.

When a new table is to be created, memory space is allocated for the table and filled with data according to the particular wavetable generator. Any reference to a wavetable with this name (including indirect references through import into a instrument instance) in existing or new instrument instances shall be taken as direction to the new wavetable.

NOTE - According to this paragraph, the wavetables referenced by running instrument instances shall be replaced upon dispatch of a **table** score line using the same name. That is, in the midst of the sound generation process, when the **table** score line is dispatched, any table-reference opcodes in an instrument referencing that name shift reference to the new wavetable.

### 5.11.7 End line

The **end** line in the score specifies the end of the sound-generation process. The number given is the end time, in score time, for the orchestra. When this time is reached, the orchestra ceases, and all future Composition Buffers based on this Structured Audio decoding process contain only 0 values.

## 5.12 SAOL/SASL tokenisation

### 5.12.1 Introduction

This clause describes the normative process of mapping between the SAOL textual format used to describe syntax and semantics in clause 5.8, and the tokenised bitstream representation used in the bitstream definition in clause 5.5. The textual representation stands outside of the bitstream-decoder relationship, and as such is not required to be implemented or used. The only aspect of SAOL decoding that is strictly normative is the process of turning a tokenised bitstream representation into sound as described in clause 5.7. However, it is highly recommended that implementations that allow access to bitstream contents use the textual representation described in clause 5.8 rather than the tokenised representation. It is nearly impossible for a human reader to understand a SAOL program presented in tokenised format.

### 5.12.2 SAOL tokenisation

To tokenise a textual SAOL orchestra, the following steps shall be performed. First, the orchestra shall be divided into lexical elements, where a lexical element is one of the following:

1. A punctuation mark,
2. A reserved word (see subclause 5.8.9),
3. A standard name (see subclause 5.8.6.8),
4. A core opcode name (see subclause 5.9.3),
5. A core wavetable generator name (see clause 5.10),
6. A symbolic constant (a string, integer, or floating-point constant; see subclause 5.8.2.3), or
7. An identifier (see subclause 5.8.2.2).

Whitespace (see subclause 5.8.2.6) may be used to separate lexical elements as desired; in some cases, it is required in order to lexically disambiguate the orchestra. In neither case shall whitespace be treated as a lexical element of the orchestra. Comments (see subclause 5.8.2.5) may be used in the textual SAOL orchestra but are removed upon lexical analysis; comments are not preserved through a tokenisation/detokenisation sequence.

After lexical analysis, all identifiers in the orchestra shall be numbered with symbol values, so that a single symbol is associated with a particular textual identifier. All identifiers that are textually equivalent (equal under string comparison) shall be associated with the same symbol regardless of their syntactic scope. This association of symbols to identifiers is called the *symbol table*.

Using the lexical analysis and the symbol table, a tokenised representation of the orchestra may be produced. The lexical analysis is scanned in the order it was presented in the textual representation, and for each lexical element:

- If the element is of type (1) – (5) from above, the token value associated in the table in Annex 5.A with that element shall be produced.
- If the element is of type (6) from above, one of the special tokens 0xF1, 0xF2, 0xF3, 0xF4 shall be produced, depending on the type of the symbolic constant, and the succeeding bitstream element shall be the bitstream representation of the value. For integer constants in the range [0,255], either token 0xF1 or token 0xF4 may be produced.
- If the element is of type 7, the special token 0xF0 shall be produced, and the succeeding bitstream element shall be the symbol associated with the identifier in the symbol table.

After the sequence of lexical elements presented in the textual orchestra is tokenised, the special token 0xFF, representing end-of-orchestra, shall be produced.

### 5.12.3 SASL tokenisation

A SASL score shall be tokenised with respect to a particular SAOL orchestra, since the symbol values must correspond in order for the semantics to be according to the author's intent.

To tokenise a SASL file, the following steps are taken. First, the SASL file is divided into lexical elements, where each element is either an identifier, a reserved word, the name of a core wavetable generator, or a number. After lexical analysis, each identifier shall be associated with the appropriate symbol number from the SAOL orchestra reference. That is, for the associated SAOL orchestra, if there is an identifier in the orchestra equivalent to the identifier in the score, the identifier in the score shall receive the same symbol number that it received in the orchestra. If there is no such identifier in the orchestra, any unused symbol number may be assigned to the identifier in the score.

Using the lexical analysis and the symbol table, a tokenised representation of the orchestra may be produced. Each score line is taken in turn, in the order presented in the textual representation, and used to produce a **score\_line** bitstream element, according to the semantics in clause 5.11 and the bitstream syntax for the various score elements, as given in subclause 5.5.2.

## 5.13 Sample Bank syntax and semantics

### 5.13.1 Introduction

This subclause describes the operation of the Sample Bank synthesis method for Object types 2 and 4. In Object type 2, only Sample Bank and MIDI class types shall appear in the bitstream, and this subclause describes the normative process of generating sound from a Sample Bank bitstream data element and a sequence of MIDI instructions. In Object type 4, Sample Banks are used in the context of a SAOL instrument as described in subclause 5.8.6.7.15, and this subclause describes the normative process of generating sound and returning it to the SAOL decoding process, depending on the Sample Bank bitstream data element and the particular call to **sasbf**.

The Structured Audio Sample Bank Format is derived from the MIDI Manufacturers Association (MMA) Downloadable Sounds format, which has been adopted as a standard for sample-data exchange by electronic musical instrument and PC audio manufacturers. The MMA DLS-2 standard [**DLS2**] contains provisions which, through reference, constitute provisions of this part of this standard. Subsequent amendments or revision to this publication do not apply, but parties are encouraged to investigate the possibility of applying the most recent editions of the referenced document. In particular, though it is assumed that typically an MPEG encoder will strip all unnecessary chunks from valid DLS files, DLS “chunks” that are not defined in the DLS Level 2 text shall be syntactically valid within an MPEG Sample Bank stream, though it is acceptable to ignore them and not use them for synthesis if not specifically defined in the DLS-2 standard.

### 5.13.2 Elements of bitstream

The **SASBF** bitstream element is a block of data defined by the MIDI DLS file structure [**DLS**]. This block of data is opaque to the MPEG-4 bitstream parser; among other reasons for this opaqueness, it contains values that are byte-swapped (big-endian) compared to the rest of the MPEG-4 bitstream.

### 5.13.3 Decoding process

#### 5.13.3.1 Object type 2

##### 5.13.3.1.1 Overview

In Object type 2, all synthesis is performed through wavetable-bank synthesis as defined in the MIDI DLS Level 2 specification [**DLS2**]. Control is through the use of the Standard MIDIFile bitstream element and the MIDI command bitstream element.

##### 5.13.3.1.2 Channels, sample format, and sampling rate

For the purposes of attaching a Object type 2 Structured Audio (i.e., SASBF) decoder to an AudioBIFS **AudioSource** node, the resulting audio stream shall have two channels, 32-bit floating point samples, and a 22050 Hz sampling rate. Calculation is not required to occur in stereo 32-bit samples; if the internal representation is otherwise, the result shall be converted to this format after decoding is complete.

##### 5.13.3.1.3 Decoder configuration

In the stream header (decoder configuration element), one or more **sbf** chunks may appear; a standard MIDIFile **midi\_file** chunks may also appear. The **sbf** chunks are passed to the SASBF synthesiser, which uses the data

there to prepare for synthesis as described in [DLS2]. The **midi\_file** chunk, if any, is unrolled and organised in time according to its semantics as given in [MIDI].

#### 5.13.3.1.4 Runtime decoding

Two types of events may control runtime synthesis in Object type 2: cached MIDI events that were transmitted as a MIDI file in the stream header, and real-time MIDI events that are transmitted over the streaming connection.

A *decoding clock* is maintained to control dispatch of events, but the exact properties of this clock are nonnormative. At each step, the MIDI scheduler shall dispatch any MIDI events that have arrived in the bitstream with Decoding Time Stamp less than the current value of the decoding clock, as well as any MIDI events sequenced in **midi\_file** chunks in the stream header whose unrolled time-stamps are less than the current value of the decoding clock.

Interactive manipulations to the **speed** field of the **AudioSource** scene graph node pointing to this decoding process affect the playback speed of cached Standard MIDIFile events, but have no effect on the dispatch of streaming MIDI events. Thus, it might be the case that events that are synchronised between a MIDIFile and a streaming control end up no longer synchronised if the **speed** field is manipulated.

The sound that results shall be the sound described by the synthesis process in [DLS2] according to the sample banks in the stream header and the sequence of MIDI bytes dispatched by the scheduler. These sound samples are provided to the **AudioSource** node in the scene graph that references this bitstream as the output of the Object type 2 Structured Audio decoder.

NOTE - For compliant operation, the sound output from the SASBF synthesis process is not immediately turned into audio and played to the listener. Instead, the sound is made available for further processing by the AudioBIFS scene graph. Implementations that make use of DLS-2 hardware synthesisers that produce analogue output shall "recapture" this output and convert back to a digital signal for use in the scene graph. This is necessary because interactive scene-graph manipulations may alter, attenuate, or eliminate the sound produced in this synthesis process before it is finally played to the listener.

### 5.13.3.2 Object type 4

#### 5.13.3.2.1 Overview

In Object type 4, the SASBF synthesiser is not controlled directly by MIDI data, but dispatched note-by-note in response to commands in SAOL. The **sasbf** statement (subclause 5.8.6.7.15) performs this dispatching function.

#### 5.13.3.2.2 Decoder configuration

In Object type 4 operation, **sbf** data chunks in the bitstream configuration header are passed to the SASBF synthesiser, where they are used to prepare it for real-time synthesis.

#### 5.13.3.2.3 Runtime decoding

In Object type 4 operation, each note of synthesis is performed separately. The note-on command is executed when the i-rate pass of an instrument containing the **sasbf** expression is executed. This instruction contains note, velocity, preset, and bank select values; the synthesis of one note indicated by this preset number and bank is performed for this note number and velocity according to the SASBF instrument. The resulting stereo sound is returned by the **sasbf** expression (see subclause 5.8.6.7.15).

The SASBF decoder shall make use of the MIDI controller and other continuously changing MIDI information for the channel specified. This data is not passed directly into the SASBF synthesiser in the **sasbf** command, but is made available in an implementation-specific manner. See subclause 5.8.6.7.15.

## 5.14 MIDI semantics

### 5.14.1 Introduction

This clause describes the normative decoding process for Object type 1 implementations, and the normative mapping from MIDI events in the stream information header and bitstream data into SAOL semantics for Object type 3 and 4 implementations.

The MIDI standards referenced are standardised externally by the MIDI Manufacturers Association. In particular, we reference the Standard MIDI File format, the MIDI protocol, and the General MIDI patch mapping, all standardised in [MIDI]. The MIDI terminology used in this clause is defined in that document.

### 5.14.2 Object type 1 decoding process

Little normative needs be said about the Object type 1 decoding process. The rules given in [MIDI] apply as standardised in those documents. As described in clause 5.6, only **midi** and **midi\_file** bitstream elements shall occur in a Object type 1 bitstream.

There are no normative aspects to producing sound in Object type 1.

### 5.14.3 Mapping MIDI events into orchestra control

#### 5.14.3.1 Introduction

For Object types 3 and 4, events coded as MIDI data shall be converted, when they are received in the terminal as part of a Standard MIDI File or MIDI event, into the appropriate scheduler semantics. This subclause lists the various MIDI events and their corresponding semantics in MPEG-4. These semantics apply only to Object types 3 and 4, not to Object types 1 and 2. For the latter, the semantics of MIDI events are exactly as given in [MIDI].

#### 5.14.3.2 MIDI events

##### 5.14.3.2.1 Introduction

This subclause describes the semantics of the various types of events that may arrive in a continuous bitstream as a **MIDI\_event** object. The syntax of these objects is standardised externally in [MIDI].

##### 5.14.3.2.2 Extended channel values

An actual MIDI Channel event in [MIDI] has a channel number in the range 0...15. There is no direct way in [MIDI] to specify a channel number outside this range. Each MIDI input port, output port or track chunk is associated with a distinct stream or collection of MIDI events and a corresponding distinct set of 16 channels (some of which may be unused). MIDI applications commonly use port names, track names or other labels to identify different channel sets.

Extended channel numbers are used in MPEG-4 to avoid the need for such channel set labels. In MPEG-4, the **channel** value of a **MIDI\_event** is not limited to the range 0...15. Instead, an extended channel value is generated based on both the original MIDI channel number and a number associated with the port or stream that is the source of the event. Subclause 5.14.3.3.4 describes the mapping used with Standard MIDI Files. Annex 5.F.2 describes a recommended mapping that may be used with events from a live MIDI device.

##### 5.14.3.2.3 NoteOn

```
noteon channel note velocity
```

When a **noteon** event is received with nonzero velocity, the instrument in the orchestra (if any) currently assigned to channel **channel** shall be instantiated with duration  $-1$  and the first two p-fields set to **note** and **velocity**. Each

value of **MIDictrl[]** within the instrument instance is set to the most recent value of a controller change for that controller on channel **channel** or to the default value (see subclause 5.14.3.3.2) if there have been no controller changes for that controller on that channel. The value of **MIDibend** is set to the most recent value of the MIDI pitch bend. The value of **MIDitouch** is set to the most recent aftertouch value on the channel.

If there is no instrument currently assigned to channel **channel**, there is no action associated with this event.

An instrument instance created in response to a **noteon** message on a particular channel is referred to as being “on” that channel.

**noteon** messages with velocity 0 shall be treated as **noteoff** messages, see subclause 5.14.3.2.4.

#### 5.14.3.2.4 NoteOff

`noteoff channel note velocity`

When a **noteoff** event is received, each instrument instance on channel **channel** that was instantiated with note number **note** is scheduled for termination at the end of the k-cycle; that is, its **released** flag is set, and if the instrument does not call **extend**, it shall be de-instantiated after the current k-cycle of computation.

If **MIDictrl[64]** on the indicated channel is non-zero, then the execution of the **noteoff** event shall be delayed until **MIDictrl[64]** on the indicated channel becomes zero. This behaviour maintains whether the value of **MIDictrl[64]** is set in the bitstream or by assignment to the **MIDictrl** standard name (see subclause 5.8.6.8.9).

#### 5.14.3.2.5 Control change

`cc channel controller value`

When a **cc** or control change event is received, the new value of the specified controller is set to **value**. This value shall be cached so that future instrument instances on the given channel have access to it; also, all currently active instrument instances on the channel **channel** shall have the standard name **MIDictrl[controller]** updated to **value**.

#### 5.14.3.2.6 Aftertouch

`touch channel note velocity`

When a **touch** event is received, the value of the **MIDitouch** variable of each instrument instance on channel **channel** that was instantiated with note number **note** is set to **velocity**.

#### 5.14.3.2.7 Channel aftertouch

`ctouch channel velocity`

When a **ctouch** event is received, the value of the **MIDitouch** variable of each instrument instance on channel **channel** is set to **velocity**.

#### 5.14.3.2.8 Program change

`pchange channel program`

When a **pchange** event is received, the current instrument receiving events on channel **channel** shall be changed to the instrument with preset number **program** (see subclause 5.8.6.4). Only the instrument with preset number **program** is assigned to the channel. If there is no instrument with this preset number, then future note-on events on the channel, until another program change is received, shall be ignored.

#### 5.14.3.2.9 Bank select

`bankselect channel bank`

When a **bankselect** event is received, the next time a **pchange** event is received, the current instrument receiving events on channel **channel** shall be changed to the instrument with preset number **bank** \* 128 + **program**. The **bankselect** event has no direct effect by itself; it only changes the meaning of future **pchange** events on the channel.

### 5.14.3.2.10 Pitch wheel change

`pwheel channel value`

When a **pwheel** event is received, the **MIDIbend** value for each instrument instance on channel **channel** shall be set to **value**.

### 5.14.3.2.11 All notes off

`notesoff`

When a **notesoff** event is received, all instrument instances in the orchestra created by a MIDI NoteOn events (subclause 5.14.3.2.3) are scheduled for termination at the end of the current k-cycle; that is, the released flag is set, and if the instrument does not call extend, it shall be de-instantiated after the current k-cycle of computation.

If the **MIDIctrl**[64] value for an instance is non-zero, then the execution of the termination shall be delayed until the **MIDIctrl**[64] value becomes zero.

These semantics correspond to the behavioral intentions of the MIDI All Notes Off command: the release cycle of notes are permitted to decay naturally, and sustain pedal semantics are obeyed. The **MIDIctrl**[123] value, normally 0, shall be set to 1 for all associated **MIDIctrl** accesses during the orchestra cycle the All Notes Off Command is processed, so that dynamic instruments spawned from the MIDI instrument may detect the All Notes Off command.

### 5.14.3.2.12 Tempo change

`tempochange value`

When a **tempochange** event is received, the global orchestra tempo is changed as described in subclause 5.7.3.3.6, list item 7. **value** here indicates a beats/minute value as in the SASL **tempo** score event (subclause 5.11.5); it shall be converted from the native MIDI tempo format (see [MIDI]) to this format.

### 5.14.3.2.13 All sound off

`soundoff`

When a **soundoff** event is received, all instrument instances in the orchestra created by MIDI NoteOn events (subclause 5.14.3.2.3) are terminated at the end of the current k-cycle. Instruments may not save themselves from termination using the extend statement in this case, and the status of the **MIDIctrl**[64] is irrelevant. The **MIDIctrl**[120] value, normally 0, shall be set to 1 for all associated **MIDIctrl** accesses during the orchestra cycle the All Sound Off Command is processed, so that dynamic instruments spawned from the MIDI instrument may detect the All Sound Off command.

These semantics correspond to the behavioral intentions of the MIDI All Sound Off command: an immediate ending of all sound making.

### 5.14.3.2.14 MIDI messages not respected

The following MIDI messages have no meaning in MPEG-4 Object types 3 and 4:

Local Control  
Omni Mode On/Off  
Mono Mode On/Off  
Poly Mode On/Off  
System Exclusive



Tune Request  
 Timing Clock  
 Song Select/Continue/Stop  
 Song Position  
 Active Sensing  
 Reset

#### 5.14.3.2.15 The MIDI Master Channel

All instances created by SASL instr statements and SAOL send statements, and any dynamic instruments instantiated from instances created by SASL instr statements and SAOL send statements, see the status of the MIDI Master Channel in the values of the **MIDIctrl**[], **MIDibend**, **MIDIwheel**, and **channel**, and **preset** standard names. For these instances, the value of the **channel** standard name reflects the extended channel number of the MIDI master channel, and the value of the **preset** standard name reflects the value of the last pchange event on the MIDI Master Channel. The values of the **MIDIctrl**[], **MIDibend**, **MIDIwheel** are identical to the values seen in an instance instantiated via a MIDI NoteOn event from the MIDI Master Channel.

The identity of the MIDI Master Channel is determined as follows:

If a MIDI source is directly connected to the orchestra, as described in Annex 5.F, then the MIDI Master Channel is MIDI channel 0 of this source.

If no MIDI source is directly connected to the orchestra, if an SA\_access\_unit MIDI source is present, the MIDI Master Channel is channel 0 of this source.

If no MIDI source is directly connected to the orchestra, and if no SA\_access\_unit MIDI source is present, the MIDI Master Channel is channel 0 of the first MIDI File track that contains a NoteOff, NoteOn, CChange, PChange, Pwheel, Touch, or CTouch command on channel 0.

Decoder implementations may provide a way to specify the MIDI Master Channel directly, superseding these rules, if the decoder permits the direct connection of MIDI sources to the orchestra.

### 5.14.3.3 Standard MIDI Files

#### 5.14.3.3.1 Introduction

MIDI files have data with the same semantics as the MIDI messages described above; however, the timing semantics are more complicated due to the use of multiple tracks and delta-time timestamps.

#### 5.14.3.3.2 Overview of MIDI file processing

To process a **midi\_file** stream information element, the following steps shall be taken. First, the entire stream element is parsed and cached. Then, using the sequence instructions and the sequencer model described in [MIDI], the delta-times of the various **midi\_file** events are converted into score event times (in beats). During this step, the actual channel numbers of these events are also mapped to extended **channel** values, as described in subclause 5.14.3.3.4. Converting delta-times to score event times requires converting each **midi\_file** track chunk into a timelist containing **midi\_event** objects and then interleaving the various track timelists.

#### 5.14.3.3.3 Converting MIDI file track chunks

To convert the track chunk into a timelist, first parse the track chunk to generate a series of MIDI events. This requires converting MIDI file delta-times to MIDI event score times relative to the beginning of each track chunk. It is also necessary to map **midi\_file** event channel numbers to extended **midi\_event channel** values, as described in subclause 5.14.3.3.4. When a Set Tempo **midi\_file** meta-event is processed to generate a corresponding tempo change **midi\_event**, the tempo value shall be converted from the microseconds-per-quarter-note units used in [MIDI] to the beats-per-minute units used in MPEG-4. If the MIDI file does not specify a starting **tempo**, the default value of 120 beats per minute shall be used.

MIDI time-stamps may only appear in the “MIDI Time Code” syntax, not the “SMPTE” syntax, as described in [MIDI]. The SMPTE Offset meta-event and the SMPTE format delta-time object are not supported in Structured Audio Object 3 and Object 4 bitstreams.

As events are converted from MIDI to SAOL semantics, each event shall be registered with the scheduler according to its event time and semantics.

NOTE - MIDI tempo events are not used to calculate the event time of MIDI events in a MIDI file. Rather, they are scheduled as regular events and dispatched according to the tempo semantics in 5.7.3.3.6 list item 7.

#### 5.14.3.3.4 Converting MIDI channels to scheduler channels

Mapping the channels of events in MIDI files to **midi\_event channel** numbers is accomplished as follows. Successive track chunks within a **midi\_file** are assigned track numbers in ascending monotonic sequence, with an initial track number of 0. The **channel** value for a particular **midi\_file** event is given by:

$$\text{channel} = \text{midi\_file event channel number} + (\text{track number} * 16).$$

If there are multiple **midi\_file** chunks within the bitstream header, then subsequent **midi\_file** elements have tracks numbered sequentially from the end of the first chunk. That is, if the  $i^{\text{th}}$  **midi\_file** element has  $k_i$  tracks,  $0 < i < n$ , then the tracks from the first **midi\_file** are numbered  $0..k_0-1$ , those from the second are numbered  $k_0..k_1-1$ , and so forth.

#### EXAMPLE

A **midi\_file** containing ten track chunks would be mapped onto a set of 160 **midi\_event channel** values. Events in track chunk 0 would map to **channel** values in the range 0...15. Events in track chunk 1 would map to **channel** values in the range 16...31. Events in track chunk 9 would map to **channel** values in the range 144...159.

If some channel numbers within a given track chunk are unused, the corresponding **channel** values are also unused.

#### 5.14.3.4 Default controller values

The following table gives the default values for certain continuous controllers. If a particular controller is not listed here, then its default value shall be zero.

There is no normative significance to these “function names” excepting controller 64; however, content authors who wish to use General MIDI score files with SAOL orchestras are advised to consult [MIDI] for the normative meaning of the controllers and controller values within General MIDI bitstreams and MIDIfiles.

**Table 5.4 - Default MIDI Controller Values**

Controller	Function	Default
1	Mod Wheel	0
5	Portamento Speed	0
7	Volume	100
10	Pan	64
11	Expression	127
64	Sustain Pedal	0
65	Portamento On/Off	0
66	Sostenuto	0
67	Soft Pedal	0
84	Portamento Control	0
Pitch Bend	Pitch Bend	8192

## 5.15 Input sounds and relationship with AudioBIFS

### 5.15.1 Introduction

This clause describes the use of SAOL orchestras as the effects-processing functionality in the AudioBIFS (Binary Format for Scene Description) system, described in ISO/IEC 14496-1 subclause 9.2.2.13.3. In ISO/IEC 14496, SAOL is used not only as a sound-synthesis description method, but also as a description method for sound-effects and other post-production algorithms. The BIFS **AudioFX** node (ISO/IEC 14496-1, subclause 9.4.2.7) allows the inclusion of signal-processing algorithms described in SAOL that are applied to the outputs of the sound nodes subsidiary to that node in the scene graph. This functionality fits well into the bus-send methodology in Structured Audio, but requires some additional normative text to exactly describe the process.

### 5.15.2 Input sources and phaseGroup

Each node in a BIFS scene graph that contains SAOL code is either an **AudioSource** node or an **AudioFX** node. If the former, there are no input sources to the SAOL orchestra, and so the default orchestra global **inchannels** value is 0 (see subclause 5.8.5.2.3). In this case, the special bus **input\_bus** may not be sent to an instrument or otherwise used in the orchestra.

If the latter, the child nodes of the **AudioFX** node provide several channels of input sound to the orchestra. These channels of input sound, calculated as described in ISO/IEC 14496-1 subclause 9.4.2.7, are placed on the special bus **input\_bus**. From this bus, they may be sent to any instrument(s) desired and the audio data thereby provided shall be treated normally. The number of orchestra input channels---the default value of orchestra global **inchannels**---is the sum of the numbers of channels of sound provided by each of the children.

In any instrument that receives a **send** from the special bus **input\_bus**, the value of the **inGroup** standard name (see subclause 5.8.6.8.15) shall be constructed using the **phaseGroup** values of the child nodes in the scene graph, as follows. The **inGroup**[] values, when non-zero, shall have the property that **inGroup**[i] = **inGroup**[j] when  $i \neq j$  exactly when **input** channel **i** is output channel **n** of child **c1**, **input** channel **j** is output channel **m** of child **c2**, **c1** = **c2**, and **phaseGroup**[n] = **phaseGroup**[m] within **c1**. (That is, when the two channels come from the same child and are phase-grouped in that child's output).

This rule applies in addition to the usual **inGroup** rules as given in subclause 5.8.6.8.15.

#### NOTE

The **phaseGroup** values are made available to the SA decoder, but they are not visible inside the scope of the several orchestra elements, and therefore **phaseGroup** does not constitute a standard name as **inGroup**. **phaseGroup** is only used outside the orchestra by the SA decoder to construct the **inGroup** values.

#### EXAMPLE

Assume that the two child nodes of an **AudioFX** node produce two and three channels of output respectively, and their **phaseGroup** fields are [1,1] and [1,0,1] respectively. That is, in the first child, the two channels form a stereo pair; and in the second, the first and third channels form a stereo pair that has no phase relationships with the second channel.

For the following global orchestra definitions:

```
send(input_bus ; ; a);
route(a, bus2);
send(bus2, input_bus, bus2 ; ; b);
```

Assume that instrument **a** produces two channels of output. Then, a legal value for the **inGroup** name within **a** is [1,1,2,0,2], and a legal value for the **inGroup** name within **b** is [1,1,2,2,3,0,3,4,4]. The value for the **inGroup** name within **a** shall not be [1,1,1,0,1], and the value for the **inGroup** name within **b** shall not be [1,1,2,2,2,2,2,3,3] (among other illegal possibilities).

### 5.15.3 The AudioFX node

#### 5.15.3.1 Introduction

The **AudioFX** node in AudioBIFS is described in the MPEG-4 Systems document, ISO/IEC 14496-1, subclause 9.4.2.7. It is used therein to download audio effects-processing algorithms within the AudioBIFS toolset. SAOL is the language for description of audio effects-processing algorithms in AudioBIFS. This clause describes the execution of a Structured Audio orchestra for the purpose of processing sounds in the AudioBIFS scene.

The aspects described in this clause are normative, but the behaviour is not “decoding” behaviour. Rather, this subclause expands the normative processing semantics of the **AudioFX** node description in ISO/IEC 14496-1.

#### 5.15.3.2 AudioFX orchestra parameters

When a SAOL orchestra is instantiated due to an **AudioFX** BIFS node, only an orchestra file (the **orch** field in the node) and, optionally, a SASL score file (the **score** field) are provided. These files correspond to tokenised sequences of orchestra and score data forming legal **orchestra** and **score\_file** bitstream elements as described in subclause 5.5.2. Further, a score may not contain new instrument events, but only control parameters for the **send** instruments defined in the orchestra. The set of allowable sampling rates is restricted, see subclause 5.8.5.2.1.

#### 5.15.3.3 AudioFX orchestra instantiation

To instantiate the orchestra for the AudioFX node requires the following steps:

1. Decoding of the **orch** and **score** (if any) elements in the node
2. Parsing and syntax-checking of these elements
3. Instantiation of **send** instruments in the orchestra (as described in subclause 5.7.2).

Each of these **send** instances shall be maintained until it is turned off by the **turnoff** statement, or the node containing the orchestra is deleted from the scene graph. If the **turnoff** statement is used in one of these instruments, it shall be taken as producing zero values for all future time.

#### 5.15.3.4 AudioFX orchestra execution

The run-time synthesis process proceeds according to the rules cited in subclause 5.7.3.3 for a standard SA decoding process, with the following exceptions and additions:

As no access units will be received by an **AudioFX** process, no communication with the systems layer need be maintained for this purpose. The only events used are those that are in the **score** field of the node itself. At each time step, the **AudioFX** orchestra shall request from the systems layer the input audio buffers that correspond to the child nodes. These audio buffers shall be placed on the special bus **input\_bus** and then sent to whatever instruments are specified in the global orchestra header.

Also, at each control-rate step, the **params[]** fields of the **AudioFX** node shall be copied into the global **params[]** array of the orchestra. These fields are exposed in the scene graph so that interactive aspects of other parts of the scene graph may be used to control the orchestra. At the end of each control cycle, the **params[]** array values shall be copied back into the corresponding fields of the **AudioFX** node and then routed to other nodes as specified within the scene graph. (It is not possible to give a more semantically meaningful field name than **params** since the purpose of the field may vary greatly from application to application, depending on the needs of the content).

At every point in time, the output of the orchestra becomes the output of the **AudioFX** node.

#### 5.15.3.5 Speed change functionality in the AudioFX node

Speed change functionality for sounds provided from the input sources is supported in the AudioFX node. The SAOL core opcode **fx\_speedc** is provided for this purpose; see subclause 5.9.14.4.

#### 5.15.4 Interactive 3-D spatial audio scenes

When an **AudioSource** or **AudioFX** node is the child of a **Sound** node, the spatial location, direction, and propagation pattern of the sound subtree represented at the position of the **Sound** node, and the spatial location and direction of the listener, are provided to the SAOL code in the node. In this way, subtle spatial effects such as source directivity modelling may be written in SAOL.

The standard names **position**, **direction**, **listenerPosition**, **listenerDirection**, **minFront**, **maxFront**, **minBack**, and **maxBack** (see subclauses 5.8.6.8.18-5.8.6.8.25) are used for this purpose.

It is not recommended that content providing 3-D spatial audio in the context of audio-visual virtual reality applications in BIFS use the **spatialize** statement within SAOL to provide this functionality. In most terminals, the scene-composition 3-D audio functionality will be able to use more information about the interaction process to provide the best-quality audio rendering. In particular, spatial positioning and source directivity are implemented at the end terminal with a sophistication suitable for the terminal itself (see ISO/IEC 14496-1, **Sound** node specification, subclause 9.4.2.82). Content authors can use SAOL and the **AudioFX** node to create enhanced spatial effects that include reverberation, environmental attributes and complex attenuation functions, and then let the terminal-level spatial audio presentation be used to interface with the available rendering method for the terminal. The **spatialize** statement in SAOL is provided for the creation of non-interactive spatial audio effects in musical compositions, so that composers may tightly integrate the spatial presentation with other aspects of the musical material.

**Annex 5.A**  
(normative)

**Coding tables**

**5.A.1 Introduction**

This Annex contains the bitstream token table as referenced in clause 5.5 and clause 5.12. Certain tokens are indicated as **(reserved)**, which means they are not currently used in the bitstream, but may be used in future versions of the standard. Tokens 0xF5 through 0xFF may be used by implementers for implementation-dependent purposes.

**5.A.2 Bitstream token table**

Token	Text		
0x00	<b>(reserved)</b>	0x41	minFront
0x01	aopcode	0x42	minBack
0x02	asig	0x43	maxFront
0x03	else	0x44	maxBack
0x04	exports	0x45	params
0x05	extend	0x46	itime
0x06	global	0x47	<b>(reserved)</b>
0x07	if	0x48	channel
0x08	imports	0x49	input_bus
0x09	inchannels	0x4A	output_bus
0x0A	instr	0x4B	startup
0x0B	iopcode	0x4C-0x4F	<b>(reserved)</b>
0x0C	ivar	0x50	&&
0x0D	kopcode	0x51	
0x0E	krate	0x52	>=
0x0F	ksig	0x53	<=
0x10	map	0x54	!=
0x11	oparray	0x55	==
0x12	opcode	0x56	-
0x13	outbus	0x57	*
0x14	outchannels	0x58	/
0x15	output	0x59	+
0x16	return	0x5A	>
0x17	route	0x5B	<
0x18	send	0x5C	?
0x19	sequence	0x5D	:
0x1A	sasbf	0x5E	(
0x1B	spatialize	0x5F	)
0x1C	srate	0x60	{
0x1D	table	0x61	}
0x1E	tablemap	0x62	[
0x1F	template	0x63	]
0x20	turnoff	0x64	;
0x21	while	0x65	,
0x22	with	0x66	=
0x23	xsig	0x67	!
0x24	interp	0x68-0x6E	<b>(reserved)</b>
0x25	preset	0x6F	sample
0x26-0x2F	<b>(reserved)</b>	0x70	data
0x30	k_rate	0x71	random
0x31	s_rate	0x72	step
0x32	inchan	0x73	lineseg
0x33	outchan	0x74	expseg
0x34	time	0x75	cubicseg
0x35	dur	0x76	polynomial
0x36	MIDIctrl	0x77	spline
0x37	MIDItouch	0x78	window
0x38	MIDIbend	0x79	harm
0x39	input	0x7A	harm_phase
0x3A	inGroup	0x7B	periodic
0x3B	released	0x7C	buzz
0x3C	cpuload	0x7D	concat
0x3D	position	0x7E	empty
0x3E	direction	0x7F	<b>(reserved)</b>
0x3F	listenerPosition	0x80	int
0x40	listenerDirection	0x81	frac
		0x82	dbamp

0x83	ampdb	0xBC	alinrand
0x84	abs	0xBD	iexprand
0x85	exp	0xBE	kexprand
0x86	log	0xBF	aexprand
0x87	sqrt	0xC0	kpoissonrand
0x88	sin	0xC1	apoissonrand
0x89	cos	0xC2	igaussrand
0x8A	atan	0xC3	kgaussrand
0x8B	pow	0xC4	agaussrand
0x8C	log10	0xC5	port
0x8D	asin	0xC6	hipass
0x8E	acos	0xC7	lopass
0x8F	floor	0xC8	bandpass
0x90	ceil	0xC9	bandstop
0x91	min	0xCA	fir
0x92	max	0xCB	iir
0x93	pchoct	0xCC	firt
0x94	octpch	0xCD	iirt
0x95	cpspch	0xCE	biquad
0x96	pchcps	0xCF	fft
0x97	cpsoct	0xD0	ifft
0x98	octcps	0xD1	rms
0x99	pchmidi	0xD2	gain
0x9A	midipch	0xD3	balance
0x9B	octmidi	0xD4	decimate
0x9C	midioct	0xD5	upsamp
0x9D	cpsmidi	0xD6	downsamp
0x9E	midicps	0xD7	samphold
0x9F	sgn	0xD8	delay
0xA0	ftlen	0xD9	delay1
0xA1	ftloop	0xDA	fracdelay
0xA2	ftloopend	0xDB	comb
0xA3	ftsetloop	0xDC	allpass
0xA4	ftsetend	0xDD	chorus
0xA5	ftbasecps	0xDE	flange
0xA6	ftsetbase	0xDF	reverb
0xA7	tableread	0xE0	compressor
0xA8	tablewrite	0xE1	gettune
0xA9	oscil	0xE2	settune
0xAA	loscil	0xE3	ftsr
0xAB	doscil	0xE4	ftsetsr
0xAC	koscil	0xE5	gettempo
0xAD	kline	0xE6	settempo
0xAE	aline	0xE7	fx_speedc
0xAF	sblock	0xE8	speedt
0xB0	kexpon	0xE9-0xEF	<b>(reserved)</b>
0xB1	aexpon	0xF0	<symbol>
0xB2	kphasor	0xF1	<number>
0xB3	aphasor	0xF2	<integer>
0xB4	pluck	0xF3	<string>
0xB5	buzz	0xF4	<byte>
0xB6	grain	0xF5-0xFF	<b>(free)</b>
0xB7	irand	0xFF	<E00>
0xB8	krand		
0xB9	arand		
0xBA	ilinrand		
0xBB	klinrand		



## Annex 5.B (informative)

### Encoding

#### 5.B.1. Introduction

This Annex, provided for informative purposes only, provides guidelines for building a typical Structured Audio encoder. Unlike for the natural audio coders described in Subparts 2, 3 and 4, at the time of the completion of ISO/IEC 14496, there is no known technique for generally and automatically encoding Structured Audio bitstreams from acoustic data. The computational methods that would be required to accomplish this—known variously as “computational auditory scene analysis”, “automatic polyphonic transcription”, and “musical acoustic source separation”—are still in a research stage and not likely to be generally available for several years.

Thus, the creation of bitstreams complying to this clause is a process that requires human intervention and assistance. This Annex describes the functions of possible tools for creating such bitstreams; the techniques described here are for informative purposes only, and are not required for compliance to ISO/IEC 14496-3.

#### 5.B.2. Basic encoding

##### 5.B.2.1. Introduction

This clause describes the operation of a *basic encoder* of the sort provided with ISO/IEC 14496-5. A basic Structured Audio encoder takes as input the component units of a bitstream conforming to the description in clause 5.5 (such as orchestra files and sound samples), and converts them into in a legal bitstream representation. It is outside the scope of this Annex to discuss the origin of the component units themselves; perhaps they have been created by hand or with the use of other general-purpose computer tools.

For the purposes of this discussion, it is assumed that the component units are in the following formats: SAOL and SASL programs are in their respective textual formats as described in clause 5.8 and clause 5.11 respectively; sound samples are individually stored in computer sound-file format such as AIFF or WAVE; MIDI data is stored as a Standard MIDI File; and SASBF banks are stored as binary data files.

The steps required in bitstream creation are as follows: tokenisation of the SAOL and SASL programs, disassembly of the sound samples, assembly of the decoder configuration information, and (optionally) reorganisation of the score and MIDI events into streaming data. Each of these steps is described in the following subclauses.

##### 5.B.2.2. Tokenisation of SAOL data

The tokenisation of SAOL data is conducted as described in subclause 5.12.2. This process converts the SAOL program given in the textual format into a binary block of data. During this process, a *symbol table* may be constructed if desired by enumerating the names of the instruments, user-defined opcodes, wavetables, and signal variables in the orchestra, and associating each with a numeric value. This table may be incorporated in the decoder configuration header of the bitstream as described in subclause 5.5.2; it has no normative significance in decoding, but allows human-readable SAOL and SASL programs in the textual format to be recovered from the bitstream.

##### 5.B.2.3. Tokenisation of SASL data

The tokenisation of SASL data is conducted as described in subclause 5.12.3. This process converts the SASL program given in the textual format into a binary block of data. Any symbols used in the SASL score may be incorporated into the symbol table if one was constructed in the process described in subclause 5.B.2.2; however, at most one symbol table may be used in the orchestra.

#### 5.B.2.4. Disassembly of sound samples

The sound samples stored as computer sound files are disassembled into blocks of sample values. It is not permissible to include sound samples with formatted data (such as an AIFF or WAVE file) directly in the Structured Audio bitstream. The length (in samples), sampling rate (in Hz) if available, base frequency (in Hz) if needed, and loop start and end points (in sample number) if needed are accounted from the formatted information in the computer sound file. The sound samples are converted from whatever format they were stored in the computer sound file into either 16-bit signed integer values (that is, the values are scaled into the range [-32768, 32767]) or into 32-bit signed floating point values. Either format may be used for a sample in the Structured Audio bitstream; the first is more efficient and the second is more precise.

NOTE - If very long sound samples are to be used, and one or more natural sound decoders (that is, the functionality described in clauses 2, 3, and 4) are present in the terminal, the natural sound decoders may be used to compress sound samples as described in subclause 5.10.2 of this document and subclause 9.4.2.4 of ISO/IEC 14496-1. In this case, the sound sample(s) is (are) not contained in the Structured Audio bitstream, but in one or more natural audio bitstreams that are associated with the Structured Audio bitstream through use of the **AudioBuffer** AudioBIFS node as described in subclause 9.4.2.4 of ISO/IEC 14496-1. This process can result in more efficient transmission for Structured Audio bitstreams containing long samples.

#### 5.B.2.5 Assembly of decoder configuration information

The decoder configuration header is constructed according to the format given in clause 5.5 from a tokenised SAOL orchestra, and possibly one or more tokenised SASL scores, sound samples, MIDI files, and SASBF banks. The blocks may be in any desired order, and are indexed by the **more\_data** and **chunk\_type** bit fields. The **high\_priority** bit of each score event in the header may be set for each score line in the score that contained the \* tag, or for any set of important events desired.

#### 5.B.2.6 Assembly of streaming bitstream

In the Structured Audio bitstream format, streaming data in the form of access units is not strictly required; all of the information required for decoding may be present in the decoder configuration header. Including streaming data in the form of access units may make it easier for general-purpose MPEG-4 tools to reorganise the bitstream data for the purposes of editing or resynchronisation, or to execute random-access control of the bitstream (see Annex 5.C).

Sound samples, score events, and MIDI commands may all be included in the streaming-data part of the Structured Audio bitstream. A sound sample is included simply by packaging the sound data, after it has been disassembled from the computer sound file format, into an access unit as specified in subclause 5.5.2. A score event may be included with or without a timestamp as discussed in subclause 5.5.2. If it is included with a timestamp, it is subject to internal orchestra tempo control as discussed in subclause 5.7.3.3.6, but is difficult to reschedule at the Access Unit level; if it is included without timestamps, it is easier to reschedule at the Access Unit level, and is not subject to score-based control of the tempo. If there is no explicit timestamp, the event timing is controlled by the synchronisation information in the Access Unit, see subclause 7.2.3 of ISO/IEC 14496-1. In this case, the **use\_if\_late** tag may be used to indicate whether the event shall be used if it arrives late; see subclause 5.7.3.3.8.

For each score event, the **high\_priority** bit may be set if the corresponding line in the score contained the \* tag, or for any set of important events desired.

MIDI commands are first converted from the Standard MIDI File format to MIDI data representations. To accomplish this, the absolute time of each event in the Standard MIDI File is computed according to the syntax and semantics in [MIDI]. Then, the events are not included with delta-times, but are placed directly in the Access Unit, so that the synchronisation information in the Access Unit controls the event timing for the MIDI events. The MIDI data in each Access Unit in the bitstream is the same as that that is conveyed in the MIDI protocol in real-time MIDI-based performance; that is, it consists of un-timestamped note on, note off, and controller information.

MIDI events are wrapped in the length indicator as indicated in clause 5.5; the format of streaming MIDI data in MPEG-4 is not bit-for-bit identical (and thus, not as compact) as MIDI data in the strict MIDI protocol as specified in [MIDI], clause 5.2.

NOTE - The streaming data constructed by the process is noted to be “bursty” and highly variable-rate. That is, when a large access unit is conveyed, for example, a sound sample, the effective bitrate is suddenly much higher

than when only small access units such as note events are conveyed. The Access-Unit repackaging techniques described in clauses 10 and 11 of ISO/IEC 14496-1 may be used to smooth out the bitrate of the Structured Audio bitstream.

## Annex 5.C (informative)

### lex/yacc grammars for SAOL

#### 5.C.1 Introduction

This Annex provides grammars using the widely-available tools 'lex' and 'yacc' that conform to the SAOL specification in this document. They are provided for informative purposes only; implementers are free to use whichever tools they desire, or no tools, in building an implementation.

The reference software for Structured Audio in ISO/IEC 14496-5 builds the lexer and parser for SAOL out of these grammars by augmenting them with more processing and data-structures.

#### 5.C.2 Lexical grammar for SAOL in lex

```
STRCONST    \"(\\.|[^\\""])*\"
IDENT       [a-zA-Z_][a-zA-Z0-9_]*
INTGR       [0-9]+
NUMBER      [0-9]+(\\. [0-9]*)?(e[-+]?[0-9]+)?|-?\\. [0-9]*(e[-+]?[0-9]+)?
```

```
%{
void comment(void);
}%
```

```
%%
```

```
"//"      { comment(); }
"aopcode" { return(AOPCODE) ; }
"asig"    { return(ASIG) ; }
"else"    { return(ELSE) ; }
"exports" { return(EXPORTS) ; }
"extend"  { return(EXTEND) ; }
"global"  { return(GLOBAL) ; }
"if"      { return(IF) ; }
"imports" { return(IMPORTS) ; }
"inchannels" { return(INCHANNELS) ; }
"instr"   { return(INSTR) ; }
"interp"  { return(INTERP) ; }
"iopcode" { return(IOPCODE) ; }
"ivar"    { return(IVAR) ; }
"kopcode" { return(KOPCODE) ; }
"krate"   { return(KRATE) ; }
"ksig"    { return(KSIG) ; }
"map"     { return(MAP) ; }
"oparray" { return(OPARRAY) ; }
"opcode"  { return(OPCODE) ; }
"outbus"  { return(OUTBUS) ; }
"outchannels" { return(OUTCHANNELS) ; }
"output"  { return(OUTPUT) ; }
"preset"  { return(PRESET) ; }
"return"  { return(RETURN) ; }
"route"   { return(ROUTE) ; }
"send"    { return(SEND) ; }
"sequence" { return(SEQUENCE) ; }
"sasbf"   { return(SASBF) ; }
"spatialize" { return(SPATIALIZE) ; }
"srate"   { return(SRATE) ; }
"table"   { return(TABLE) ; }
"tablemap" { return(TABLEMAP) ; }
"template" { return(TEMPLATE) ; }
"turnoff" { return(TURNOFF) ; }
"while"   { return(WHILE) ; }
"with"    { return(WITH) ; }
```

```

"xsig"      { return(XSIG) ; }
"&&"       { return(AND); }
"||"        { return(OR); }
">="        { return(GEQ); }
"<="        { return(LEQ); }
"!="        { return(NEQ); }
"=="        { return(EQEQ); }
"-"         { return(MINUS); }
"*"         { return(STAR); }
"/"         { return(SLASH); }
"+"         { return(PLUS); }
">"         { return(GT); }
"<"         { return(LT); }
"?"         { return(Q); }
":"         { return(COL); }
"("         { return(LP); }
")"         { return(RP); }
"{"         { return(LC); }
"}"         { return(RC); }
"["         { return(LB); }
"]"         { return(RB); }
";"         { return(SEM); }
","         { return(COM); }
"="         { return(EQ); }
"!"         { return(NOT); }

{STRCONST}  { yytext[strlen(yytext)-1] = 0; /* strip quotes */
              yyval = strdup(&yytext[1]);
              return(STRCONST); }

{IDENT}     { yyval = strdup(yytext);
              return(IDENT); }

{INTGR}     { yyval = strdup(yytext);
              return(INTGR); }

{NUMBER}    { yyval = strdup(yytext);
              return(NUMBER); }

[ \t\n\r]   { /* whitespace */ }

.           { printf("Line %d: Unknown character: '%s'\n",
                    yyline,yytext); } /* parse error */

%%

void comment() {
    char c;

    while ((c = input()) != '\n'); /* skip */
    yyline++;
    thisline[0] = 0;
    yycol = 0;
}

```

## 5.C.3 Syntactic grammar for SAOL in yacc

```

/*
    This is a grammar for SAOL, written in 'yacc'.

    It has one shift/reduce conflict that arises when
    looking at table definitions.  Within the grammar,
    'table t1;' is allowed as a definition even though
    it is a prohibited construction (it is flagged in the syntax-
    check).  So there's an ambiguity between

        table t1(...)
        table t1;

    and you don't know with one lookahead where you are when you
    get 'table' if 't1' is the next token.

    This grammar is somewhat less strict than it could be about
    creating parse errors for all syntax errors.  As it is used
    in ISO/IEC 14496-5, many illegal constructions are allowed here,
    but then prohibited in the syntax check.

    The code in ISO/IEC 14496-5 for Structured Audio uses this grammar
    as a basis, and augments it with error productions and construction
    of a parse tree.

    Note that this grammar does not fully support the template construct
    as defined in subclause 5.8.8.2.  This subclause permits expressions in
    the preset and map lists, whereas this grammar only supports terminals
    in these lists.  To support the full template syntax, a lexical analyzer
    that detects expressions in preset and map lists and takes special action
    is necessary.
*/

%token IDENT INTGR NUMBER STRCONST AOPCODE ELSE EXPORTS EXTEND GLOBAL
%token IF IMPORTS INCHANNELS INTERP
%token INSTR IOPCODE IVAR TABLE KOPCODE KRATE KSIG ASIG MAP
%token OPARRAY OPCODE OUTBUS OUTCHANNELS OUTPUT ROUTE SEND SEQUENCE
%token SRATE TEMPLATE TURNOFF WHILE WITH XSIG AND OR GEQ LEQ
%token NEQ EQEQ MINUS STAR SPATIALIZE SASBF TABLEMAP
%token SLASH PLUS GT LT Q COL LP RP LC RC LB RB SEM COM EQ RETURN NOT
%token ARRAYREF OPCALL IMPEXP VARDECL NOTAG SPECIALOP PRESET

%

%start orcfile
%right Q
%left OR
%left AND
%left EQEQ NEQ
%left LT GT LEQ NEQ
%left PLUS MINUS
%left STAR SLASH
%right UNOT UMINUS
%token HIGHEST

%%

orcfile      : proclist
              ;

```

```

proclist      : proclist instrdecl
               | proclist opcodedecl
               | proclist globaldecl
               | proclist templatedecl
               | /* null */
               | error
               ;

instrdecl     : INSTR IDENT LP identlist RP miditag LC vardecls block RC
               ;

miditag       : PRESET int_list
               | /* null */
               ;

int_list      : int_list INTGR
               | INTGR
               ;

opcodedecl    : optype IDENT LP paramlist RP LC opvardecls block RC
               ;

globaldecl    : GLOBAL LC globalblock RC
               ;

templatedecl  : TEMPLATE LT identlist GT /* with preset */
               | PRESET mapblock
               | LP identlist RP
               | MAP LC identlist RC
               | WITH LC mapblock RC LC
               | vardecls block RC
               | TEMPLATE LT identlist GT /* no preset */
               | LP identlist RP
               | MAP LC identlist RC
               | WITH LC mapblock RC LC
               | vardecls block RC
               ;

mapblock      : mapblock COM LT terminal_list GT
               | LT terminal_list GT
               ;

terminal_list : terminal_list COM terminal
               | terminal
               ;

terminal      : IDENT
               | const
               | STRCONST
               ;

globalblock   : globalblock globaldef
               | /* null */
               ;

globaldef     : rtparam
               | vardecl
               | routedef
               | senddef
               | seqdef
               ;

rtparam       : SRATE INTGR SEM
               | KRATE INTGR SEM

```

```

| INCHANNELS INTGR SEM
| OUTCHANNELS INTGR SEM
| INTERP INTGR SEM
|
routedef      : ROUTE LP IDENT COM identlist RP SEM
|
senddef       : SEND LP IDENT SEM exprlist SEM identlist RP SEM
|
seqdef        : SEQUENCE LP identlist RP SEM
|
block         : block statement
| /* null */
|
statement     : lvalue EQ expr SEM
| expr SEM
| IF LP expr RP LC block RC
| IF LP expr RP LC block RC ELSE LC block RC
| WHILE LP expr RP LC block RC
| INSTR IDENT LP exprlist RP SEM
| OUTPUT LP exprlist RP SEM
| SPATIALIZE LP exprlist RP SEM
| OUTBUS LP IDENT COM exprlist RP SEM
| EXTEND LP expr RP SEM
| TURNOFF SEM
| RETURN LP exprlist RP SEM
|
lvalue        : IDENT
| IDENT LB expr RB
|
identlist     : identlist COM IDENT
| IDENT
| /* null */
|
paramlist     : paramlist COM paramdecl
| paramdecl
| /* null */
|
vardecls     : vardecls vardecl
| /* null */
|
vardecl       : taglist stype namelist SEM
| stype namelist SEM
| tabledecl SEM
| TABLEMAP IDENT LP identlist RP SEM
|
opvardecls   : opvardecls opvardecl
| /* null */
|
opvardecl     : taglist otype namelist SEM
| otype namelist SEM
| tabledecl SEM
| TABLEMAP IDENT LP identlist RP SEM
|

```



```

paramdecl      : otype name
                ;

namelist       : namelist COM name
                | name
                ;

name           : IDENT
                | IDENT LB INTGR RB
                | IDENT LB INCHANNELS RB
                | IDENT LB OUTCHANNELS RB
                ;

stype         : IVAR
                | KSIG
                | ASIG
                | TABLE
                | OPARRAY
                ;

otype         : XSIG
                | stype
                ;

tabledecl     : TABLE IDENT LP IDENT COM exprstrlist RP
                ;

taglist       : IMPORTS
                | EXPORTS
                | IMPORTS EXPORTS
                | EXPORTS IMPORTS
                ;

optype        : AOPCODE
                | KOPCODE
                | IOPCODE
                | OPCODE
                ;

expr          : IDENT
                | const
                | IDENT LB expr RB
                | SASBF LP exprlist RP
                | IDENT LP exprlist RP
                | IDENT LB expr RB LP exprlist RP
                | expr Q expr COL expr %prec Q
                | expr LEQ expr
                | expr GEQ expr
                | expr NEQ expr
                | expr EQEQ expr
                | expr GT expr
                | expr LT expr
                | expr AND expr
                | expr OR expr
                | expr PLUS expr
                | expr MINUS expr
                | expr STAR expr
                | expr SLASH expr
                | NOT expr %prec UNOT
                | MINUS expr
                | LP expr RP
                ;

exprlist      : exprlist COM expr

```

```
    | expr
    | /* null */
    ;

/* this is used for table declarations; string constants provide
   above-the-standard file handling for "sample" */

exprstrlist  : exprstrlist COM expr
              | exprstrlist COM STRCONST
              | STRCONST
              | expr
              ;

const        : INTGR
              | NUMBER
              ;
```

## Annex 5.D (informative)

### PICOLA Speed change algorithm

#### 5.D.1 Tool description

PICOLA (Pointer Interval Controlled OverLap Add) speed control tool supports speed change functionality for mono-channel signals. The speed control is achieved by replacing a part of the input signal with an overlap-added waveform or by inserting the overlap-added waveform into the input signal.

#### 5.D.2 Speed control process

The block diagram of the speed controller is shown in Figure 5.D.1. The input signal InputSignal, which is an output from the audio source with a given frame length NumInputSample, is stored in the buffer memory. Adjacent waveforms with the same length are extracted in pairs from the memory buffer and the pair with the minimum distortion between the two waveforms is selected. The selected waveforms are overlap-added. The speed control is achieved by replacing a part of the input signal with the overlap-added waveform or by inserting it into the input signal. The speed controller outputs the speed changed signal with a certain fixed length frame calculated as NumInputSample /SpeedControlFactor. Details of the processing are described below.

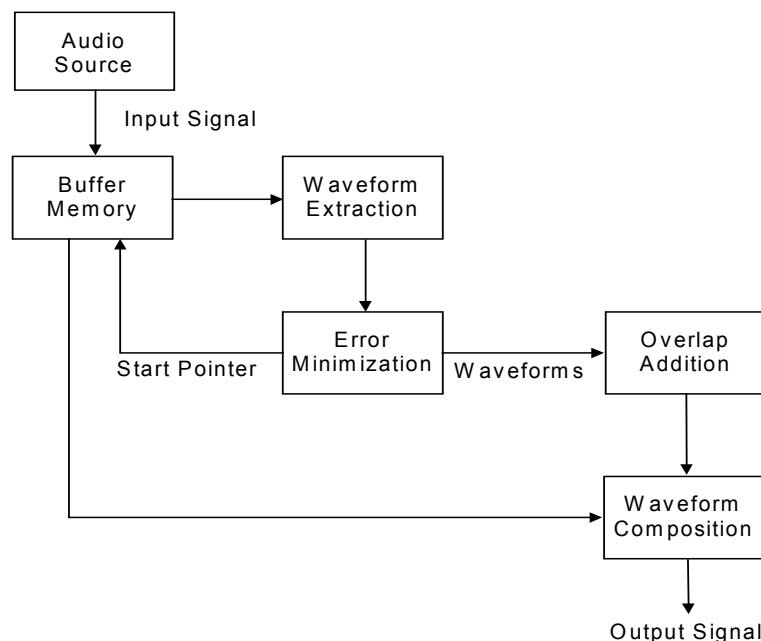


Figure 5.D.1 - Block Diagram of the Speed Controller

#### 5.D.3 Time scale compression (High speed replay)

The compression principle is shown in Figure 5.D.2. P0 is the pointer that indicates the starting sample of the current processing frame in the memory buffer. The processing frame has a length of LF samples and comprises adjacent waveforms each length of LW samples. The average distortion per sample between the first half of the processing frame (waveform A) and the second half (waveform B) is calculated as shown below.

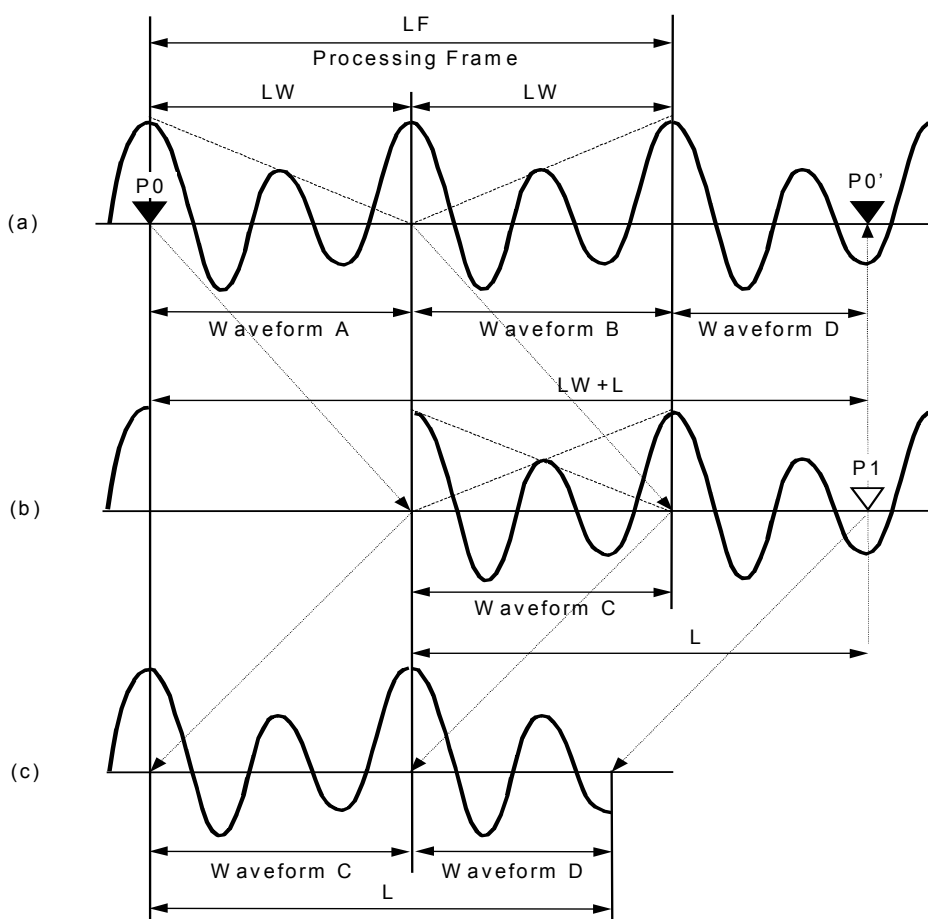
$$D(LW) = \frac{1}{LW} \sum_{n=0}^{LW-1} \{x(n) - y(n)\}^2 \quad (P_{MIN} \leq LW \leq P_{MAX})$$

where  $D(LW)$  is the average distortion between the two waveforms when the waveform length is  $LW$ ,  $x(n)$  is the waveform A,  $y(n)$  is the waveform B,  $PMIN$  is the minimum length of the candidate waveform and  $PMAX$  is the maximum length of the candidate waveform. Typically  $PMIN=32$  and  $PMAX=160$  for 8kHz sampling rate,  $PMIN=80$  and  $PMAX=320$  for 16kHz sampling.

The length  $LW$  that minimizes the distortion  $D(LW)$  is selected, and corresponding waveforms A and B are determined. If the cross-correlation between the selected waveforms A and B is negative, the length  $LW$  is set to  $PMIN-1$ . After the waveform length  $LW$  is determined, the waveform A is windowed by a half triangular window with a descending slope, and the waveform B is windowed by a half triangular window with an ascending slope. The overlap-added waveform C is obtained by linearly adding the windowed waveform A and waveform B. Then, the pointer  $P0$  moves to the point  $P1$ . The distance  $L$  from the beginning of waveform C to the pointer  $P1$  is given by;

$$L = LW \cdot \frac{1}{\text{SpeedControlFactor} - 1} \quad (\text{SpeedControlFactor} > 1)$$

$L$  samples from the beginning of waveform C are output as the compressed signal. If  $L$  is greater than  $LW$ , the original waveform D that follows the waveform B is also output. Therefore the length of the signal is shortened from  $LW+L$  samples to  $L$  samples. The updated pointer  $P1$  indicates the starting sample  $P0'$  of the next processing frame.



(a) Original signal; (b) Overlap-added waveform; (c) Compressed signal

Figure 5.D.2 - Principle of Time Scale Compression

#### 5.D.4 Time scale expansion (Low speed replay)

The expansion principle is shown in Figure 5.D.3. P0 is the pointer that indicates the starting sample of the current processing frame in the memory buffer. The processing frame has a length of LF samples and includes adjacent waveforms each length of LW samples. After the waveform length LW is determined using the same method as described in the time scale compression, the first half of the processing frame (waveform A) is outputted without any modification. Next, the first half (waveform A) is windowed by a half triangular window with an ascending slope, and the second half (waveform B) is windowed by a half triangular window with a descending slope. The overlap-added waveform C is obtained by linearly adding the windowed waveform A and waveform B. Then, the pointer P0 moves to the point P1. The distance L from the beginning of waveform C to the pointer P1 is given by;

$$L = LW \cdot \frac{\text{SpeedControlFactor}}{1 - \text{SpeedControlFactor}} \quad (0.5 \leq \text{SpeedControlFactor} < 1)$$

L samples from the beginning of waveform C are output as the expanded signal. If L is greater than LW, the original waveform B is repeated as the output. The length of the signal is therefore expanded from L samples to LW+L samples. The updated pointer P1 indicates the starting sample P0' of the next processing frame.

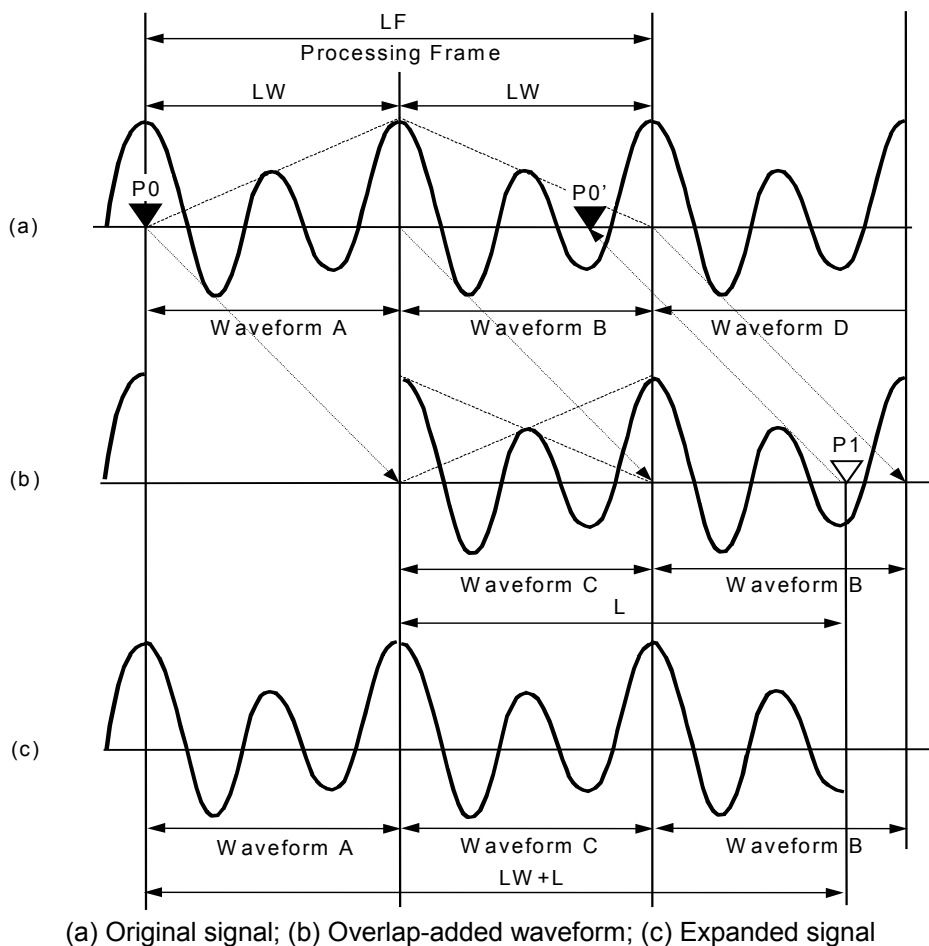


Figure 5.D.3 - Principle of Time Scale Expansion

## Annex 5.E (informative)

### Random access to Structured audio bitstreams

#### 5.E.1 Introduction

This Annex describes practices for constructing Structured Audio bitstreams that allow easy random access to sound content. As discussed in annex 5.E.2, not every sound described with a bitstream conforming to the Structured Audio standard can be easily accessed at random points. Thus, a presentation of guidelines for content authors regarding random access can encourage the development of randomly-accessible bitstreams when this functionality is needed for a particular application. The techniques here are presented for informative purposes only, and are not required for compliance to this part of ISO/IEC 14496.

#### 5.E.2 Difficulties in general-purpose random access

The major problem with enabling random access to every bitstream is that of hidden state. Consider the following orchestra:

```
global {
  srate 24000;
  krate 1000;
  ivar count;
}

instr alternate() {
  imports exports ivar count;
  count = count + 1;
  if ((floor(count/2) * 2 == count && ltime == 0) { // divisible by 2?
    instr tone(...);
  }
  // notice no regular output
}

instr tone(...) { ... } // make some sound
```

This orchestra counts up the number of times the `alternate()` instrument is instantiated. Every even-numbered instance, it passes on control to the `tone()` instrument (which presumably makes some noise); on the odd-numbered instances, nothing happens. Thus, with a score containing the following subclause:

```
.
.
.
25.0 alternate 1.0
26.0 alternate 1.0
27.0 alternate 1.0
.
.
.
```

it is impossible to determine from this segment alone the parity of the first instrument line referring to `alternate()`; we cannot randomly access this point in the sound except by scanning backward through the score to discover how many previous times the instrument has been instantiated.

The general problem here is with the *hidden state variable* `count`. The instrument `alternate()` depends on **count** for its behavior, and this variable has two properties: (1) it has a long “memory” of its past behavior (it maintains state), and (2) it is not controlled by the score (it is hidden). Any orchestra that depends on such hidden state variables is not easily randomly-accessible.

The example here generalizes, not only to instrument dispatch, but to controllers, tempo changes, and even to the development of very long notes containing ambient sound. Suppose, for instance, that there was a long note begun

in the above score at time 2.0 that is supposed to dominate the sound presentation at time 25.0 – in order to discover this, we have to similarly scan backward through the score.

### 5.E.3 Making Structured Audio bitstreams randomly-accessible

#### 5.E.3.1 Introduction

This Annex describes recommended practices for providing random access points to Structured Audio bitstreams. It covers two topics. First, a set of constructs to avoid is presented; then, a discussion is conducted regarding methods for converting non-randomly-accessible to randomly-accessible bitstreams.

#### 5.E.3.2 Constructs to avoid

In order to make bitstreams randomly-accessible, the following constructs should only be used carefully. Note that there is nothing “wrong” with this constructs in general, and they are useful in applications where random access to sound data is not required. Note also that examples can be constructed for each of these in which the described technique is used and random-accessibility is still maintained.

1. Very long notes. A note that lasts for a long time tends to prohibit random access while it is sustaining, particularly if the sound it makes changes a great deal over the duration.
2. Global variables. These are often used to preserve state between instrument instances or to share data between instruments. When used in this way, they tend to add hidden state to the orchestra.
3. Score-based creation of wavetables. If random access skips over the important point at which a wavetable is created, then it will not be available if some instrument instance needs it.
4. Modification of global wavetables. This is a special case of point (2).
5. Complex score-based control. If score-based controllers are used to modify greatly the behaviour of the orchestra, then random access will produce the wrong result if the control instructions are skipped. This applies to MIDI controllers as well as SASL controllers.

If these constructs are avoided, then in general a bitstream allows random access at the point of each instrument line in the score.

#### 5.E.3.3 Altering bitstreams to make them randomly accessible

##### 5.E.3.3.1 Introduction

For each of the points listed above, it is possible to modify a bitstream that uses the technique to create a bitstream that produces the same sound, but is higher-bandwidth and randomly accessible. This annex provides general guidelines for accomplishing this. Note that it is not possible (due to computational incompleteness) to develop an automated tool that determines whether a particular bitstream is randomly accessible, but it is possible to develop a tool that applies transforms of the following sort automatically.

##### 5.E.3.3.2 Break up very long notes

Very long notes can be broken into several shorter notes. To take a simple example:

```
global {
  krate 100;
}

instr tone(freq) {
  asig a;
  table sine(harm,2048,1);
  a = oscil(sine,freq);
  output(a);
}
```

```
}  
tone 10.0 440
```

This orchestra and score play a 440-Hz sine tone for 10 seconds. The same sound is produced by the score

```
0.0 tone 0.99 440  
1.0 tone 0.99 440  
2.0 tone 0.99 440  
3.0 tone 0.99 440  
4.0 tone 0.99 440  
5.0 tone 0.99 440  
6.0 tone 0.99 440  
7.0 tone 0.99 440  
8.0 tone 0.99 440  
9.0 tone 0.99 440
```

which is randomly-accessible once per second rather than only at the beginning. Note that the sound is the same here because the tone is only broken into a new note at the zero-phase point so that the phase remains continuous. (The duration is 0.99 because note instances execute for one extra control period, as specified in subclause 5.7.3.3.6).

For a note that changes an internal state during playback, the internal state must be exposed as a p-field (or control parameter). For example:

```
instr tone(freq) {  
  ksig env;  
  asig a;  
  table sine(harm,2048,1);  
  env = kline(0,dur/2,1,dur/2,0);  
  a = oscil(sine,freq) * env;  
  output(a);  
}  
0.0 tone 10.0 440
```

In this case, the **kline()** core opcode encapsulates hidden state (the “current value” of the line segment) that must be exposed to break the note into sections. Thus:

```
instr tone(freq, startenv, endenv) {  
  ksig env;  
  asig a;  
  table sine(harm,2048,1);  
  env = kline(startenv,dur,endenv);  
  a = oscil(sine,freq) * env;  
  output(a);  
}  
0.0 tone 0.99 440 0.0 0.2  
1.0 tone 0.99 440 0.2 0.4  
2.0 tone 0.99 440 0.4 0.6  
3.0 tone 0.99 440 0.6 0.8  
4.0 tone 0.99 440 0.8 1.0  
5.0 tone 0.99 440 1.0 0.8  
6.0 tone 0.99 440 0.8 0.6  
7.0 tone 0.99 440 0.6 0.4  
8.0 tone 0.99 440 0.4 0.2  
9.0 tone 0.99 440 0.2 0.0
```

This orchestra/score specifies the same sound, but is randomly accessible once per second. Application of this technique clearly becomes difficult for very complex instruments.

### 5.E.3.3.3 Replace global variables with controllers

Consider the initial example from subclause 5.E.2:

```
global {
```



```

srate 24000;
krate 1000;
ivar count;
}

instr alternate() {
  imports exports ivar count;
  count = count + 1;
  if ((floor(count/2) * 2 == count && ltime == 0) { // divisible by 2?
    instr tone(...);
  }
  // notice no regular output
}

instr tone(...) { ... } // make some sound

.
.
.
25.0 alternate 1.0
26.0 alternate 1.0
27.0 alternate 1.0
.
.
.

```

We can make this bitstream randomly accessible by moving the manipulation of the **count** global variable out of the **alternate** instrument and into the score.

```

global {
  srate 24000;
  krate 1000;
  imports ivar count;
}

instr alternate() {
  imports ivar count;

  if ((floor(count/2) * 2 == count && ltime == 0) { // divisible by 2?
    instr tone(...);
  }
  // notice no regular output
}

instr tone(...) { ... } // make some sound

.
.
25.0 control count 14.0
25.0 alternate 1.0
26.0 control count 15.0
26.0 alternate 1.0
27.0 control count 16.0
27.0 alternate 1.0
.
.
.

```

With this change, the bitstream is now randomly accessible (and requires twice the bandwidth).

#### 5.E.3.3.4 Replicate score-based wavetable generators

Consider the following example:

```

global {
  srate 24000;
  krate 1000;
}

```

```
instr tone(freq) {
  imports table shape;

  output(oscil(shape,freq));
}

0.0 table shape harm 2048 1
.
.
.
25.0 tone 1.0 440 // *
26.0 tone 1.0 485
```

At the random-access point (marked \*), it is not possible to determine the desired waveshape from local information. To fix this, we simply replicate the table generator throughout the score:

```
0.0 table shape harm 2048 1
.
.
.
25.0 table shape harm 2048 1
25.0 tone 1.0 440 // *
26.0 table shape harm 2048 1
26.0 tone 1.0 485
```

This bitstream is again more expensive, both in bandwidth and in computation, but it allows random access.

## Annex 5.F (informative)

### Directly-connected MIDI and microphone control of the orchestra

#### 5.F.1 Introduction

Although it is outside the normative scope of this part of ISO/IEC 14496, some discussion of connecting live MIDI devices and other controllers, and live microphones, directly to a sound generator making use of the Structured Audio tools (especially a SAOL orchestra) is presented and recommended practices developed. This discussion is presented for informative purposes only; the techniques presented here are not required for compliance to this part of ISO/IEC 14496.

The connection of live MIDI control of an orchestra allows a Structured Audio decoder device to be used as a real-time musical instrument in performance or recording situations. The sound quality and flexibility of the Structured Audio tools is an improvement over fixed hardware synthesizers for use in these situations. The connection of live microphones allows a Structured Audio decoder to act as an effects processor, a live or interactive karaoke device, or to enable other kinds of interactive electroacoustic performances.

This Annex presents recommendations for the connection of live MIDI devices and other controllers, and live microphones, to a Structured Audio decoder.

#### 5.F.2 MIDI controller recommended practices

As the MIDI-event bitstream format is very similar to the MIDI control data generated by a live MIDI device (it exactly encapsulates the data bytes generated by such a device in the MPEG-4 Access Unit), no direct modification needs to be made to a Structured Audio decoder to enable control of an orchestra by such a devices. All that need be accomplished is a connection between the MIDI input of the terminal and the scheduler input, so that non-timestamped events from the MIDI input are passed directly to the scheduler. The following practices are recommended in this method:

The MIDI input should not be converted into a legal Structured Audio bitstream, but should generate events directly in the scheduler.

Any note-on events generated with a live MIDI device should not be executed in the order prescribed by the global sequencing rules (see subclause 5.8.5.6). Rather, the note instantiation and first k-cycle of the instrument instance should be executed in the current orchestra pass as soon as possible after they are received by the orchestra. Upon the second k-cycle pass through the instrument instance, the instance begins to be processed according to the global sequencing rules. This practice may cause unpredictable results if it is used in conjunction with instruments containing certain global-variable or bus-routing constructions, and thus such constructions must be used advisedly. If possible, the latency between the time the event is triggered by the live performer and the time at which the first k-cycle of the instrument sound is audible should be no greater than 5 ms.

The live MIDI events should not be subject to orchestra tempo control.

The live MIDI events should otherwise be treated as any other orchestra MIDI event. Streaming performance and live performance should be possible at the same time.

If multiple MIDI devices are connected to the same terminal, the terminal should allow that the channel numbering be managed logically, so that "MIDI channel 1" from Device A is a different channel than "MIDI channel 1" from Device B. For this purpose, it is recommended that each such MIDI device be assigned device numbers in ascending monotonic sequence. The **channel** value for a particular live MIDI event is then given by:

$$\text{channel} = \text{live MIDI event channel} + (\text{device number} * 16).$$

The initial device number should be chosen so that **channel** values assigned to live MIDI events do not conflict with **channel** values assigned to **midi\_file** events as described in subclause 5.14.3.3.

Although at the time of writing ISO/IEC 14496, the vast majority of live musical control devices generate MIDI data, it is also possible to construct musical control devices that generate SASL events directly. In this case, the recommended practices are the same as above, except the incoming data is represented in terms of non-timestamped SASL events. The method of constructing devices that generate legal SASL events in real-time is outside the scope of this Annex.

### 5.F.3 Live microphone recommended practices

As the Structured Audio orchestra already handles acoustic data very well, it is easy to allow the connection of a live microphone. The special bus **input\_bus** is defined as in subclause 5.15.2, and the audio data captured by the microphone is placed on this bus, from which it can be sent to other instruments for processing. The following practices are recommended in this method:

A special bus called **input\_bus** is defined. This bus has the number of channels specified by the **inchannels** global parameter (see subclause 5.8.5.2.3). If the **inchannels** global parameter is not specified, this bus (and thus the microphone) cannot be used.

At the beginning of each control cycle (i.e., between step 9 and step 10 of subclause 5.7.3.3.6), the microphone input is sampled at the orchestra sampling rate and placed on the special bus **input\_bus**. If there are more channels of microphone input than on **input\_bus**, only the first channels are used and the rest are discarded; if there are fewer, then the “extra” channels of **input\_bus** are set to all 0 values. If there is no microphone connected, then all channels of **input\_bus** are set to all 0 values.

The **input\_bus** is treated as any other bus in the orchestra.

There is no recommended practice for using a microphone in conjunction with a SAOL orchestra that is used to process effects for an AudioFX node.

## Bibliography

- [BIFS]** Scheirer, Eric D., and Riitta Väänänen and Jyri Huopaniemi, "AudioBIFS: The MPEG-4 standard for effects processing." *Proc. 1<sup>st</sup> Cost-G6 Workshop on Digital Audio Effects Processing (DAFX-98)*, Barcelona, 1998.
- [BOOK]** Scheirer, Eric D., and Youngjik Lee and Jae-Woo Yang, "Synthetic audio and SNHC audio in MPEG-4." In Puri, Atul and Tsuhan Chen (eds.), *Advances in Multimedia: Systems, Standards, and Networks*. New York: Marcel Dekker, in press.
- [DRAG]** Aho, Alfred V., and Ravi Sethi and Jeffrey Ullman, *Compilers: Principles, Techniques, and Tools*. Reading, Mass: Addison-Wesley, 1984.
- [GRAN]** Cavaliere, Sergio, and Aldo Piccialli, "Granular synthesis of musical signals", in Curtis Roads, Stephen Travis Pope, Aldo Piccialli, and Giovanni de Poli (eds.), *Musical Signal Processing*. London: Swets & Zeitlinger, 1998, pp. 155-186.
- [ICASSP]** Scheirer, Eric D., "The MPEG-4 Structured Audio standard", *Proc 1998 IEEE ICASSP*, Seattle, 1998, pp. 3801-3804.
- [NETSOUND]** Casey, Michael A., and Paris G. Smaragdis, "Netsound", *Proc. 1996 ICMC*, Hong Kong, 1996, p. 143
- [SAFX]** Scheirer, Eric D., "Structured audio and effects processing in the MPEG-4 multimedia standard", *Multimedia Systems* 7:1 (Jan. 1999), pp. 11-22.
- [SAOL]** Scheirer, Eric D., and Barry L. Vercoe, "SAOL: The MPEG-4 Structured Audio Orchestra Language", *Computer Music Journal*, in press.
- [SAUD]** Vercoe, Barry, and William G. Gardner and Eric D. Scheirer, "Structured Audio: Creation, Transmission, and Rendering of Parametric Sound Descriptions". *Proc. IEEE* 85:5 (May 1998), pp. 922-940.
- [WAVE]** Scheirer, Eric D., and Lee Ray, "Algorithmic and wavetable synthesis in the MPEG-4 multimedia standard". Presented at the 105<sup>th</sup> Convention of the Audio Engineering Society, San Francisco, 1998 (AES reprint #4811).

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## Subpart 6: Text to Speech Interface (TTSI)

### 6.1 Scope

This subpart of ISO/IEC 14496-3 specifies the coded representation of MPEG-4 Audio Text-to-Speech (M-TTS) and its decoder for high quality synthesized speech and for enabling various applications. The exact synthesis method is not a standardization issue partly because there are already various speech synthesis techniques.

This subpart of ISO/IEC 14496-3 is intended for application to M-TTS functionalities such as those for facial animation (FA) and moving picture (MP) interoperability with a coded bitstream. The M-TTS functionalities include a capability of utilizing prosodic information extracted from natural speech. They also include the applications to the speaking device for FA tools and a dubbing device for moving pictures by utilizing lip shape and input text information.

The text-to-speech (TTS) synthesis technology is recently becoming a rather common interface tool and begins to play an important role in various multimedia application areas. For instance, by using TTS synthesis functionality, multimedia contents with narration can be easily composed without recording natural speech sound. Moreover, TTS synthesis with facial animation (FA) / moving picture (MP) functionalities would possibly make the contents much richer. In other words, TTS technology can be used as a speech output device for FA tools and can also be used for MP dubbing with lip shape information. In MPEG-4, common interfaces only for the TTS synthesizer and for FA/MP interoperability are defined. The M-TTS functionalities can be considered as a superset of the conventional TTS framework. This TTS synthesizer can also utilize prosodic information of natural speech in addition to input text and can generate much higher quality synthetic speech. The interface bitstream format is strongly user-friendly: if some parameters of the prosodic information are not available, the missed parameters are generated by utilizing preestablished rules. The functionalities of the M-TTS thus range from conventional TTS synthesis function to natural speech coding and its application areas, i.e., from a simple TTS synthesis function to those for FA and MP.

### 6.2 Definitions

**6.2.1 International Phonetic Alphabet; IPA** : The worldwide agreed symbol set to represent various phonemes appearing in human speech.

**6.2.2 lip shape pattern** : A number that specifies a particular pattern of the preclassified lip shape.

**6.2.3 lip synchronization** : A functionality that synchronizes speech with corresponding lip shapes.

**6.2.4 MPEG-4 Audio Text-to-Speech Decoder** : A device that produces synthesized speech by utilizing the M-TTS bitstream while supporting all the M-TTS functionalities such as speech synthesis for FA and MP dubbing.

**6.2.5 moving picture dubbing** : A functionality that assigns synthetic speech to the corresponding moving picture while utilizing lip shape pattern information for synchronization.

**6.2.6 M-TTS sentence** : This defines the information such as prosody, gender, and age for only the corresponding sentence to be synthesized.

**6.2.7 M-TTS sequence** : This defines the control information which affects all M-TTS sentences that follow this M-TTS sequence.

**6.2.8 phoneme/bookmark-to-FAP converter** : A device that converts phoneme and bookmark information to FAPs.

**6.2.9 text-to-speech synthesizer** : A device producing synthesized speech according to the input sentence character strings.

**6.2.10 trick mode** : A set of functions that enables stop, play, forward, and backward operations for users.

### 6.3 Symbols and abbreviations

**F0** fundamental frequency (pitch frequency)

<b>DEMUX</b>	demultiplexer
<b>FA</b>	facial animation
<b>FAP</b>	facial animation parameter
<b>ID</b>	identifier
<b>IPA</b>	International Phonetic Alphabet
<b>MP</b>	moving picture
<b>M-TTS</b>	MPEG-4 Audio TTS
<b>STOD</b>	story teller on demand
<b>TTS</b>	text-to-speech

## 6.4 MPEG-4 audio text-to-speech bitstream syntax

### 6.4.1 MPEG-4 audio TTSSpecificConfig

```
TTSSpecificConfig() {
    TTS_Sequence()
}
```

Table 6.1 - Syntax of TTS\_Sequence()

Syntax	No. of bits	Mnemonic
TTS_Sequence()		
{		
<b>TTS_Sequence_ID</b>	<b>5</b>	<b>uimsbf</b>
<b>Language_Code</b>	<b>18</b>	<b>uimsbf</b>
<b>Gender_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>Age_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>Speech_Rate_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>Prosody_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>Video_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>Lip_Shape_Enable</b>	<b>1</b>	<b>Bslbf</b>
<b>Trick_Mode_Enable</b>	<b>1</b>	<b>Bslbf</b>
}		

### 6.4.2 MPEG-4 audio text-to-speech payload

```
AIPduPayload {
    TTS_Sentence()
}
```

Table 6.2 - Syntax of TTS\_Sentence()

Syntax	No. of bits	Mnemonic
TTS_Sentence() {		
<b>TTS_Sentence_ID</b>	<b>10</b>	<b>uimsbf</b>
<b>Silence</b>	<b>1</b>	<b>bslbf</b>
if (Silence) {		
<b>Silence_Duration</b>	<b>12</b>	<b>uimsbf</b>
}		

else {		
if (Gender_Enable) {		
<b>Gender</b>	<b>1</b>	<b>bslbf</b>
}		
if (Age_Enable) {		
<b>Age</b>	<b>3</b>	<b>uimsbf</b>
}		
if (!Video_Enable && Speech_Rate_Enable) {		
<b>Speech_Rate</b>	<b>4</b>	<b>uimsbf</b>
}		
<b>Length_of_Text</b>	<b>12</b>	<b>uimsbf</b>
for (j=0; j<Length_of_Text; j++) {		
<b>TTS_Text</b>	<b>8</b>	<b>bslbf</b>
}		
if (Prosody_Enable) {		
<b>Dur_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>F0_Contour_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>Energy_Contour_Enable</b>	<b>1</b>	<b>bslbf</b>
<b>Number_of_Phonemes</b>	<b>10</b>	<b>uimsbf</b>
<b>Phoneme_Symbols_Length</b>	<b>13</b>	<b>uimsbf</b>
for (j=0 ; j<Phoneme_Symbols_Length ; j++) {		
<b>Phoneme_Symbols</b>	<b>8</b>	<b>bslbf</b>
}		
for (j=0 ; j<Number_of_Phonemes ; j++) {		
if(Dur_Enable) {		
<b>Dur_each_Phoneme</b>	<b>12</b>	<b>uimsbf</b>
}		
if (F0_Contour_Enable) {		
<b>Num_F0</b>	<b>5</b>	<b>uimsbf</b>
for (k=0; k<Num_F0;k++) {		
<b>F0_Contour_each_Phoneme</b>	<b>8</b>	<b>uimsbf</b>
<b>F0_Contour_each_Phoneme_Time</b>	<b>12</b>	<b>uimsbf</b>
}		
}		
if (Energy_Contour_Enable) {		
<b>Energy_Contour_each_Phoneme</b>	<b>8*3=24</b>	<b>uimsbf</b>
}		
}		
}		
if (Video_Enable) {		
<b>Sentence_Duration</b>	<b>16</b>	<b>uimsbf</b>
<b>Position_in_Sentence</b>	<b>16</b>	<b>uimsbf</b>
<b>Offset</b>	<b>10</b>	<b>uimsbf</b>

## 6.5 MPEG-4 audio text-to-speech bitstream semantics

### 6.5.1 MPEG-4 audio TTSSpecificConfig

**TTS\_Sequence\_ID** This is a five-bit ID to uniquely identify each TTS object appearing in one scene. Each speaker in a scene will have distinct TTS\_Sequence\_ID.

**Language\_Code** When this is "00" (00110000 00110000 in binary), the IPA is to be sent. In all other languages, this is the ISO 639 Language Code. In addition to this 16 bits, two bits that represent dialects of each language is added at the end (user defined).

**Gender\_Enable** This is a one-bit flag which is set to '1' when the gender information exists.

**Age\_Enable** This is a one-bit flag which is set to '1' when the age information exists.

**Speech\_Rate\_Enable** This is a one-bit flag which is set to '1' when the speech rate information exists.

**Prosody\_Enable** This is a one-bit flag which is set to '1' when the prosody information exists.

**Video\_Enable** This is a one-bit flag which is set to '1' when the M-TTS decoder works with MP. In this case, M-TTS should synchronize synthetic speech to MP and accommodate the functionality of ttsForward and ttsBackward. When VideoEnable flag is set, M-TTS decoder uses system clock to select adequate TTS\_Sentence frame and fetches Sentence\_Duration, Position\_in\_Sentence, Offset data. TTS synthesizer assigns appropriate duration for each phoneme to meet Sentence\_Duration. The starting point of speech in a sentence is decided by Position\_in\_Sentence. If Position\_in\_Sentence equals 0 (the starting point is the initial of sentence), TTS uses Offset as a delay time to synchronize synthetic speech to MP.

**Lip\_Shape\_Enable** This is a one-bit flag which is set to '1' when the coded input bitstream has lip shape information. With lip shape information, M-TTS request FA tool to change lip shape according to timing information (Lip\_Shape\_in\_Sentence) and predefined lip shape pattern.

**Trick\_Mode\_Enable** This is a one-bit flag which is set to '1' when the coded input bitstream permits trick mode functions such as stop, play, forward, and backward.

### 6.5.2 MPEG-4 audio text-to-speech payload

**TTS\_Sentence\_ID** This is a ten-bit ID to uniquely identify a sentence in the M-TTS text data sequence for indexing purpose. The first five bits equal to the TTS\_Sequence\_ID of the speaker defined in subclause 6.5.1, and the rest five bits are the sequential sentence number of each TTS object.

**Silence** This is a one-bit flag which is set to '1' when the current position is silence.

**Silence\_Duration** This defines the time duration of the current silence segment in milliseconds. It has a value from 1 to 4095. The value '0' is prohibited.

**Gender** This is a one-bit which is set to '1' if the gender of the synthetic speech producer is male and '0', if female.

**Age** This represents the age of the speaker for synthetic speech. The meaning of age is defined in Table 6.3.

**Table 6.3 - Age mapping table**

Age	Age of the speaker
000	below 6
001	6 – 12
010	13 – 18
011	19 – 25
100	26 – 34
101	35 – 45
110	45 – 60
111	over 60

**Speech\_Rate** This defines the synthetic speech rate in 16 levels. The level 8 corresponds the normal speed of the speaker defined in the current speech synthesizer, the level 0 corresponds to the slowest speed of the speech synthesizer, and the level 15 corresponds to the fastest speed of the speech synthesizer.

**Length\_of\_Text** This identifies the length of the TTS\_Text data in bytes.

**TTS\_Text** This is a character string containing the input text. The text bracketed by < and > contains bookmarks. If the text bracketed by < and > starts with FAP, the bookmark is handed to the face animation through the TtsFAPInterface as a string of characters. Otherwise, the text of the bookmark is ignored. The syntax of the bookmarks is defined in ISO/IEC 14496-2.

**Dur\_Enable** This is a one-bit flag which is set to '1' when the duration information for each phoneme exists.

**F0\_Contour\_Enable** This is a one-bit flag which is set to '1' when the pitch contour information for each phoneme exists.

**Energy\_Contour\_Enable** This is a one-bit flag which is set to '1' when the energy contour information for each phoneme exists.

**Number\_of\_Phonemes** This defines the number of phonemes needed for speech synthesis of the input text.

**Phonemes\_Symbols\_Length** This identifies the length of Phonemes\_Symbols (IPA code) data in bytes since the IPA code has optional modifiers and dialect codes.

**Phoneme\_Symbols** This defines the indexing number for the current phoneme by using the Unicode 2.0 numbering system. Each phoneme symbol is represented as a number for the corresponding IPA. Three two-byte numbers is used for each IPA representation including a two-byte integer for the character, and an optional two-byte integer for the spacing modifier, and another optional two-byte integer for the diacritical mark.

**Dur\_each\_Phoneme** This defines the duration of each phoneme in msec.

**Num\_F0** This defines the number of F0 values specified for the current phoneme.

**F0\_Contour\_each\_Phoneme** This defines half of the F0 value in Hz at time instant F0\_Contour\_each\_Phoneme\_Time.

**F0\_Contour\_each\_Phoneme\_Time** This defines the integer number of the time in ms for the position of the F0\_Contour\_each\_Phoneme.



**Energy\_Contour\_each\_Phoneme** These 3 8-bit data correspond to the energy values at the start, the middle, and the end positions of the phoneme. The energy value  $X$  is calculated as

$$X = \text{int}(50 \log_{10} A_{p-p}),$$

where  $A_{p-p}$  is the peak-to-peak value of the speech waveform at the defined position.

**Sentence\_Duration** This defines the duration of the sentence in msec.

**Position\_in\_Sentence** This defines the position of the current stop in a sentence as an elapsed time in msec.

**Offset** This defines the duration of a very short pause before the start of synthesized speech output in msec.

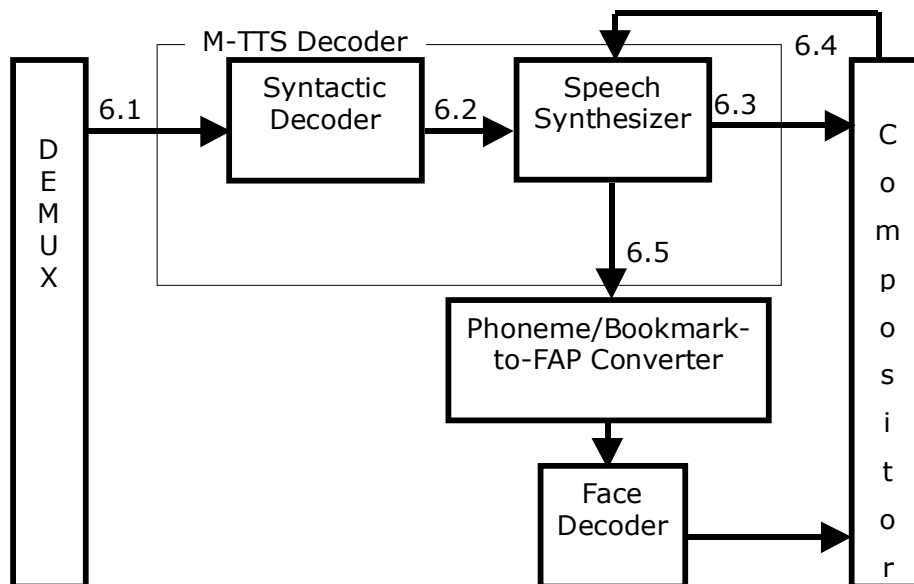
**Number\_of\_Lip\_Shape** This defines the number of lip-shape patterns to be processed.

**Lip\_Shape\_in\_Sentence** This defines the position of each lip shape from the beginning of the sentence in msec.

**Lip\_Shape** This defines the indexing number for the current lip-shape pattern to be processed that is defined in ISO/IEC 14496-2.

## 6.6 MPEG-4 audio text-to-speech decoding process

The architecture of the M-TTS decoder is described below and only the interfaces relevant to the M-TTS decoder are the subjects of standardization. The number above each arrow indicates the section describing each interface.



**Figure 6.1 - MPEG-4 Audio TTS decoder architecture.**

In this architecture the following types of interfaces is distinguished:

- Interface between DEMUX and the syntactic decoder
- Interface between the syntactic decoder and the speech synthesizer
- Interface from the speech synthesizer to the compositor
- Interface from the compositor to the speech synthesizer

Interface between the speech synthesizer and the phoneme/bookmark-to-FAP converter

**6.6.1 Interface between DEMUX and syntactic decoder**

Receiving a bitstream, DEMUX passes coded M-TTS bitstreams to the syntactic decoder.

**6.6.2 Interface between syntactic decoder and speech synthesizer**

Receiving a coded M-TTS bitstream, the syntactic decoder passes some of the following bitstreams to the speech synthesizer.

Input type of the M-TTS data: specifies synchronized operation with FA or MP

Control commands stream: Control command sequence

Input text: character string(s) for the text to be synthesized

Auxiliary information: Prosodic parameters including phoneme symbols

Lip shape patterns

Information for trick mode operation

The pseudo-C code representation of this interface is defined in subclause 6.4.

**6.6.3 Interface from speech synthesizer to compositor**

This interface is identical to the interface for digitized natural speech to the compositor. The dynamic range is from - 32767 to + 32768.

**6.6.4 Interface from compositor to speech synthesizer**

This interface is defined to allow the local control of the synthesized speech by users. This user interface supports trick mode of the synthesized speech in synchronization with MP and changes some prosodic properties of the synthesized speech by using the ttsControl defined as follows:

**Table 6.4 - Syntax of ttsControl()**

Syntax	No. of bits	Mnemonic
ttsControl() { ttsPlay() ttsForward() ttsBackward() ttsStopSyllable() ttsStopWord() ttsStopPhrase() TtsChangeSpeedRate() TtsChangePitchDynamicRange() TtsChangePitchHeight() TtsChangeGender() ttsChangeAge() }		

The member function ttsPlay allows a user to start speech synthesis in the forward direction while ttsForward and ttsBackward enable the user to change the starting play position in forward and backward direction, respectively. The ttsStopSyllable, ttsStopWord, and ttsStopPhrase functions define the interface for users to stop speech synthesis at the specified boundary such as syllable, word, and phrase. The member function ttsChangeSpeechRate is an interface to change the synthesized speech rate. The argument speed has the numbers from 1 to 16. The member function ttsChangePitchDynamicRange is an interface to change the dynamic

range of the pitch of synthesized speech. By using the argument of this function, level, a user can change the dynamic range from 1 to 16. Also a user can change the pitch height from 1 to 16 by using the argument height in the member function `ttsChangePitchHeight`. The member functions `ttsChangeGender` and `ttsChangeAge` allow a user to change the gender and the age of the synthetic speech producer by assigning numbers, as defined in subclause 6.5.2, to their arguments, gender and age, respectively.

### 6.6.5 Interface between speech synthesizer and phoneme/bookmark-to-FAP converter

In the MPEG-4 framework, the speech synthesizer and the face animation are driven synchronously. The speech synthesizer generates synthetic speech. At the same time, TTS gives `phonemeSymbol` and `phonemeDuration` as well as bookmarks to the Phoneme/Bookmark-to-FAP converter. The Phoneme/Bookmark to FAP converter generates relevant facial animation according to the `phonemeSymbol`, the `phonemeDuration` and bookmarks. Further description of the Phoneme/Bookmark to FAP converter is provided in ISO/IEC 14496-2.

The synthesized speech and facial animation have relative synchronization except the absolute composition time. The synchronization of the absolute composition time comes from the same composition time stamp of the TTS bitstream. If the `Lip_Shape_Enable` is set, the `Lip_Shape_in_Sentence` is used to generate the `phonemeDuration`. Otherwise, the TTS provides phoneme durations. The speech synthesizer generates stress and/or `wordBegin` bits when the corresponding phoneme has stress and/or start of a word, respectively.

Within the `MTTS_Text`, the beginning of a bookmark for using facial animation parameters is identified by '<FAP'. The bookmark lasts until the closing bracket '>'

A bookmark is handed to the `TtsFAPInterface` with the phoneme of the next word of the current sentence following the bookmark. If there is no word after the bookmark, the bookmark is handed to the `TtsFAPInterface` with the last phoneme of the previous word in the current sentence. In order to allow animation of complex expressions and motion, a sequence of up to 40 bookmarks is allowed without words between them. The `starttime` defines the time in msec relative to the beginning of the M-TTS sequence when the phoneme will start playing.

The class `ttsFAPInterface` defines the data structure for the interface between the speech synthesizer and the phoneme-to-FAP converter.

**Table 6.5 - Syntax of `TtsFAPInterface()`**

Syntax	No. of bits	Mnemonic
<code>TtsFAPInterface()</code>		
{		
<code>PhonemeSymbol</code>	<b>8</b>	<b>uimsbf</b>
<code>PhonemeDuration</code>	<b>12</b>	<b>uimsbf</b>
<code>f0Average</code>	<b>8</b>	<b>uimsbf</b>
<code>Stress</code>	<b>1</b>	<b>bslbf</b>
<code>WordBegin</code>	<b>1</b>	<b>bslbf</b>
<code>Bookmark</code>		<b>char *</b>
<code>Starttime</code>		<b>long int</b>
}		

## **Annex 6.A (informative)**

### **Applications of MPEG-4 audio text-to-speech decoder**

#### **6.A.1 General**

This annex part describes application scenarios for the M-TTS decoder.

#### **6.A.2 Application scenario: MPEG-4 Story Teller on Demand (STOD)**

In the STOD application, users can select a story from a huge database of story libraries which are stored in hard disks or compact disks. The STOD system reads aloud the story via the M-TTS decoder with the MPEG-4 facial animation tool or with appropriately selected images. The user can stop and resume speaking at any moment he wants through user interfaces of the local machine (for example, mouse or keyboard). The user can also select the gender, age, and the speech rate of the electronic story teller.

The synchronization between the M-TTS decoder with the MPEG-4 facial animation tool is realized by using the same composition time of the M-TTS decoder for the MPEG-4 facial animation tool.

#### **6.A.3 Application scenario: MPEG-4 audio text-to-speech with moving picture**

In this application, synchronized playback of the M-TTS decoder and encoded moving picture is the most important issue. The architecture of the M-TTS decoder can provide several granularities of synchronization. Aligning the composition time of each TTS\_Sentence, coarse granularity of synchronization and trick mode functionality can be easily achieved. To get finer granularity of synchronization, the information about the Lip\_Shape would be utilized. The finest granularity of synchronization can be achieved by using the prosody information and the video-related information such as Sentence\_Duration, Position\_in\_Sentence, and Offset.

With this synchronization capability, the M-TTS decoder can be used for moving picture dubbing by utilizing the Lip\_Shape and Lip\_Shape\_in\_Sentence.

#### **6.A.4 MPEG-4 audio TTS and face animation using bookmarks appropriate for trick mode**

Bookmarks allow to animate a face using facial animation parameters (FAP) in addition to the animation of the mouth derived from phonemes. The FAP of the bookmark is applied to the face until another bookmark resets the FAP. Designing contents that replay each sentence independent of trick mode requires that bookmarks of the text to be spoken are repeated at the beginning of each sentence to initialize the face to the state that is defined by the previous sentence. In this case, some mismatch of synchronization can occur in the beginning of a sentence. However, the system recovers when the new bookmark is processed.

#### **6.A.5 Random access unit**

Every TTS\_Sentence is a random access unit.

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## Subpart 7: Parametric Audio Coding - HILN

### 7.1 Scope

The Parametric Audio Coding Subpart provides the HILN tools which complement the other natural audio coding tools in the area of very low bit rates. Their focus is the representation of monophonic music signals with low and intermediate content complexity in the range of 4 to 16 kbit/s. HILN enables a high grade of interactivity by implicit support of speed and pitch change during playback and by the capability of bit rate scalability. Furthermore the possible combination with the parametric speech coding tools HVXC allows very efficient schemes for coding speech and music signals.

#### 7.1.1 Technical Overview

MPEG-4 parametric audio coding uses the HILN technique (Harmonic and Individual Lines plus Noise) to code audio signals like music at bitrates of 4 kbit/s and higher using a scalable parametric representation of the audio signal. HILN allows independent change of speed and pitch during decoding. Furthermore HILN can be combined with MPEG-4 parametric speech coding (HVXC, see Subpart 2) to form an integrated parametric coder covering a wider range of signals and bitrates.

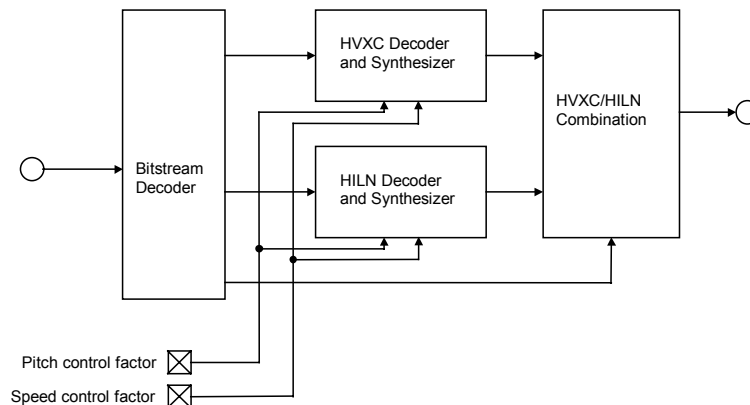


Figure 7.1 - Block diagram of the integrated parametric decoder

The integrated parametric coder can operate in the following modes:

Table 7.1 - Parametric coder operation modes

PARAMode	Description
0	HVXC only
1	HILN only
2	switched HVXC / HILN
3	mixed HVXC / HILN

PARAModes 0 and 1 represent the fixed HVXC and HILN modes. PARAMode 2 permits automatic switching between HVXC and HILN depending on the current input signal type. In PARAMode 3 the HVXC and HILN decoders can be used simultaneously and their output signals are added (mixed) in the parametric decoder.

In “switched HVXC / HILN” and “mixed HVXC / HILN” modes both HVXC and HILN decoder tools are operated alternatively or simultaneously according to the PARAswitchMode or PARAMixMode of the current frame. To obtain proper time alignment of both HVXC and HILN decoder output signals before they are added, a FIFO buffer compensates for the time difference between HVXC and HILN decoder delay.

To avoid hard transitions at frame boundaries when the HVXC or HILN decoders are switched on or off, the respective decoder output signals fade in and out smoothly. For the HVXC decoder a 20 ms linear fade is applied when it is switched on or off. The HILN decoder requires no additional fading because of the smooth synthesis windows utilized in the HILN synthesizer. It is only necessary to reset the HILN decoder (`numLine = 0`) if the current bitstream frame contains no `HILNframe()`.

## 7.2 Definitions

- 7.2.1 HVXC:** Harmonic Vector Excitation Coding (parametric speech coding).
- 7.2.2 HILN:** Harmonic and Individual Lines plus Noise (parametric audio coding).
- 7.2.3 individual line:** A spectral component described by frequency, amplitude and phase.
- 7.2.4 harmonic lines:** A set of spectral components having a common fundamental frequency.
- 7.2.5 noise component:** A signal component modeled as noise.
- 7.2.6 pi:** The constant  $\pi = 3.14159\dots$

A general glossary and list of symbols and abbreviations is located in Subpart 1.

## 7.3 Bitstream syntax

An MPEG-4 Natural Audio Object using Parametric Coding is transmitted in one or more Elementary Streams: The base layer stream, an optional enhancement layer stream, and one or more optional extension layer streams.

The bitstream syntax is described in pseudo-C code.

The mnemonics LARH1, LARH2, LARH3, LARN1, LARN2, DIA, DIF, DHF, DFS indicate that a "vlclbf" codeword is used. The corresponding codebooks are given in Subclause 7.3.2.3.

The mnemonic SDC indicates that a "vlclbf" codeword is used which is decoded by the HILN `SubDivisionCode` described in Subclause 7.3.2.4 using the parameters for `SDCdecode()` as given in the bitstream syntax description.

### 7.3.1 Decoder configuration (ParametricSpecificConfig)

The decoder configuration information for parametric coding is transmitted in the `ParametricSpecificConfig()` of the base layer and the enhancement or extension layer Elementary Stream (see Subpart 1).

#### Parametric Base Layer -- Configuration

The parametric coder in unscalable mode or the base layer in HILN scalable mode uses a `ParametricSpecificConfig()` with `isBaseLayer==1`.

#### Parametric HILN Enhancement / Extension Layer -- Configuration

To use HILN as core in a "T/F scalable with core" mode, in addition to the HILN base layer an HILN enhancement layer is required. In HILN bitrate scalable operation, in addition to the HILN base layer one or more HILN extension layers are permitted. Both the enhancement and extension layer use a `ParametricSpecificConfig()` with `isBaseLayer==0`.

**Table 7.2 - Syntax of ParametricSpecificConfig()**

Syntax	No. of bits	Mnemonic
ParametricSpecificConfig() { <b>isBaseLayer;</b> if (isBaseLayer) { PARAconfig(); } else { HILNenexConfig(); } }	<b>1</b>	<b>uimsbf</b>

**7.3.1.1 Parametric Audio decoder configuration**

**Table 7.3 - Syntax of PARAconfig()**

Syntax	No. of bits	Mnemonic
PARAconfig() { <b>PARAMode;</b> if (PARAMode != 1) { ErHVXCconfig(); } if (PARAMode != 0) { HILNconfig(); } <b>PARAextensionFlag;</b> if (PARAextensionFlag) { /* to be defined in MPEG-4 Phase 3 */ } }	<b>2</b>	<b>uimsbf</b>
	<b>1</b>	<b>uimsbf</b>

**Table 7.4 - PARAMode**

PARAMode	frameLength	Description
0	20 ms (N = 160 samples)	HVXC only
1	see Subclause 7.3.1.2 and Subclause 7.5.1.4.3.3	HILN only
2	40 ms (N = 320 samples)	HVXC/HILN switching
3	40 ms (N = 320 samples)	HVXC/HILN mixing

**7.3.1.2 HILN decoder configuration**

**Table 7.5 - Syntax of HILNconfig()**

Syntax	No. of bits	Mnemonic
HILNconfig() { <b>HILNquantMode;</b> <b>HILNmaxNumLine;</b> <b>HILNsampleRateCode;</b> <b>HILNframeLength;</b> <b>HILNcontMode;</b> }	<b>1</b> <b>8</b> <b>4</b> <b>12</b> <b>2</b>	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>



Table 7.6 - Syntax of HILNenexConfig()

Syntax	No. of bits	Mnemonic
HILNenexConfig() { <b>HILNenhaLayer;</b> if (HILNenhaLayer) { <b>HILNenhaQuantMode;</b> } }	<b>1</b>	<b>uimsbf</b>
	<b>2</b>	<b>uimsbf</b>

Table 7.7 - HILNsampleRateCode

HILNsampleRateCode	sampleRate	maxFIndex
0	96000	890
1	88200	876
2	64000	825
3	48000	779
4	44100	765
5	32000	714
6	24000	668
7	22050	654
8	16000	603
9	12000	557
10	11025	544
11	8000	492
12	7350	479
13	reserved	reserved
14	reserved	reserved
15	reserved	reserved

Table 7.8 - linebits

HILNmaxNumLine	0	1	2..3	4..7	8..15	16..31	32..63	64..127	128..255
linebits	0	1	2	3	4	5	6	7	8

Table 7.9 - HILNcontMode

HILNcontMode	additional decoder line continuation (see Subclause 7.5.1.4.3.1)
0	harmonic lines <-> individual lines and harmonic lines <-> harmonic lines
1	mode 0 plus individual lines <-> individual lines
2	no additional decoder line continuation
3	(reserved)

The number of frequency enhancement bits (fEnhbits[i]) in HILNenhaFrame() (see Subclause 7.3.2.2) is calculated as follows:

- individual line, see INDlenhaPara():

$$fEnhbits[i] = \max(0, fEnhbitsBase[ILFreqIndex[i]] + fEnhbitsMode[HILNenhaQuantMode])$$

- harmonic line, see HARMenhaPara():

$$fEnhbits[i] = \max(0, fEnhbitsBase[harmFreqIndex] + fEnhbitsMode[HILNenhaQuantMode] + fEnhbitsHarm[i])$$

**Table 7.10 - fEnhbitsBase**

ILFreqIndex	harmFreqIndex	fEnhbitsBase
0..159	0..1243	0
160..269	1244..1511	1
270..380	1512..1779	2
381..491	1780..2047	3
492..602		4
603..713		5
714..890		6

**Table 7.11 - fEnhbitsMode**

HILNenhaQuantMode	0	1	2	3
fEnhbitsMode	-3	-2	-1	0

**Table 7.12 - fEnhbitsHarm**

i	0	1	2..3	4..7	8..9
fEnhbitsHarm[i]	0	1	2	3	4

**Table 7.13 - HILN constants**

tmbits	4
atkbits	4
decbits	4
tmEnhbits	3
atkEnhbits	2
decEnhbits	2
phasebits	5

### 7.3.2 Bitstream frame (slPacketPayload)

The dynamic data for parametric coding is transmitted as SL packet payload in the base layer and the optional enhancement or extension layer Elementary Stream.

#### Parametric Base Layer -- Access Unit payload

For the parametric coder in unscalable mode or for the base layer in HILN scalable mode the following bitstream frame payload is defined:

```
slPacketPayload {
    PARAFrame();
}
```

#### Parametric HILN Enhancement / Extension Layer -- Access Unit payload

To parse and decode the HILN enhancement layer, information decoded from the HILN base layer is required.

To parse and decode an HILN extension layer, information decoded from the HILN base layer and possible lower HILN extension layers is required. The bitstream syntax of the HILN extension layers is described in a way which requires the HILN base and extension bitstream frames being parsed in proper order:

1. HILNbasicFrame()      base bitstream frame
2. HILNNextFrame(1)    1<sup>st</sup> extension bitstream frame (if base bitstream frame available)
3. HILNNextFrame(2)    2<sup>nd</sup> extension bitstream frame (if base and 1<sup>st</sup> extension bitstream frame available)
4. etc.

For the enhancement layer and the extension layer in HILN scalable mode the following bitstream frame payload is defined:

```
slPacketPayload {
    HILNenexFrame();
}
```

### 7.3.2.1 Parametric Audio bitstream frame

**Table 7.14 - Syntax of PARAFrame()**

Syntax	No. of bits	Mnemonic
<pre>PARAFrame() {     if (PARAMode == 0) {         ErHVXCframe(HVXCrate);     }     else if (PARAMode == 1) {         HILNframe();     }     else if (PARAMode == 2) {         switchFrame();     }     else if (PARAMode == 3) {         mixFrame();     } }</pre>		

**Table 7.15 - Syntax of switchFrame()**

Syntax	No. of bits	Mnemonic
<pre>switchFrame() {     <b>PARAswitchMode;</b>     if (PARAswitchMode == 0) {         ErHVXCdoubleframe(HVXCrate);     }     else {         HILNframe();     } }</pre>	<b>1</b>	<b>uimsbf</b>

One of the following PARAswitchModes is selected in each frame:

**Table 7.16 - PARAswitchMode**

PARAswitchMode	Description
0	HVXC only
1	HILN only

**Table 7.17 - Syntax of mixFrame()**

Syntax	No. of bits	Mnemonic
<pre> mixFrame() {   <b>PARAMixMode;</b>   if (PARAMixMode == 0) {     ErHVXCdoubleframe(HVXCrate);   }   else if (PARAMixMode == 1) {     HILNframe();     ErHVXCdoubleframe(2000);   }   else if (PARAMixMode == 2) {     HILNframe();     ErHVXCdoubleframe(4000);   }   else if (PARAMixMode == 3) {     HILNframe();   } } </pre>	<b>2</b>	<b>uimsbf</b>

One of the following PARAMixModes is selected in each frame:

**Table 7.18 - PARAMixMode**

PARAMixMode	Description
0	HVXC only
1	HVXC 2 kbit/s & HILN
2	HVXC 4 kbit/s & HILN
3	HILN only

**Table 7.19 - Syntax of HVXCdoubleframe()**

Syntax	No. of bits	Mnemonic
<pre> ErHVXCdoubleframe(rate) {   if (rate &gt;= 3000) {     ErHVXCfixframe(4000);     ErHVXCfixframe(rate);   }   else {     ErHVXCfixframe(2000);     ErHVXCfixframe(rate);   } } </pre>		





Table 7.26 - Syntax of HARMbasicParaESC0()

Syntax	No. of bits	Mnemonic
HARMbasicParaESC0() { prevHarmAmplIndex = harmAmplIndex; prevHarmFreqIndex = harmFreqIndex; <b>harmContFlag</b> ; <b>harmEnvFlag</b> ; if (!harmContFlag) { <b>harmAmplRel</b> ; harmAmplIndex = maxAmplIndex + harmAmplRel; <b>harmFreqIndex</b> ; } }	1 1 6 11	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 7.27 - Syntax of HARMbasicParaESC1()

Syntax	No. of bits	Mnemonic
HARMbasicParaESC1() { <b>numHarmParaIndex</b> ; numHarmPara = numHarmParaTable[numHarmParaIndex]; <b>numHarmLineIndex</b> ; numHarmLine = numHarmLineTable[numHarmLineIndex]; if (harmContFlag) { <b>contHarmAmpl</b> harmAmplIndex = prevHarmAmplIndex + contHarmAmpl; <b>contHarmFreq</b> harmFreqIndex = prevHarmFreqIndex + contHarmFreq; } for (i=0; i<2; i++) { <b>harmLAR[i]</b> ; } }	4 5 3..8 2..9 4..19	<b>uimsbf</b> <b>uimsbf</b> <b>DIA</b> <b>DHF</b> <b>LARH1</b>

Table 7.28 - Syntax of HARMbasicParaESC3()

Syntax	No. of bits	Mnemonic
HARMbasicParaESC3() { for (i=2; i<min(7,numHarmPara); i++) { <b>harmLAR[i]</b> ; } for (i=7; i<numHarmPara; i++) { <b>harmLAR[i]</b> ; } }	3..18 2..17	<b>LARH2</b> <b>LARH3</b>

Table 7.29 - Syntax of HARMbasicParaESC4()

Syntax	No. of bits	Mnemonic
HARMbasicParaESC4() { if (phaseFlag && ! harmContFlag) { <b>numHarmPhase;</b> } else { numHarmPhase = 0; } for (i=0; i<numHarmPhase; i++) { <b>harmPhase[i];</b> harmPhaseAvail[i] = 1; } for (i=numHarmPhase; i<numHarmLine; i++) { harmPhaseAvail[i] = 0; } }	6	uimsbf
	phasebits	uimsbf

Table 7.30 - numHarmParaTable

i	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
numHarmParaTable[i]	2	3	4	5	6	7	8	9	11	13	15	17	19	21	23	25

Table 7.31 - numHarmLineTable

i	0	1	2	3	4	5	6	7
numHarmLineTable[i]	3	4	5	6	7	8	9	10
i	8	9	10	11	12	13	14	15
numHarmLineTable[i]	12	14	16	19	22	25	29	33
i	16	17	18	19	20	21	22	23
numHarmLineTable[i]	38	43	49	56	64	73	83	94
i	24	25	26	27	28	29	30	31
numHarmLineTable[i]	107	121	137	155	175	197	222	250

Table 7.32 - Syntax of NOISEbasicParaESC0()

Syntax	No. of bits	Mnemonic
NOISEbasicParaESC0() { prevNoiseAmplIndex = noiseAmplIndex; <b>noiseContFlag;</b> <b>noiseEnvFlag;</b> if (!noiseContFlag) { <b>noiseAmplRel;</b> noiseAmplIndex = maxAmplIndex + noiseAmplRel; } }	1	uimsbf
	1	uimsbf
	6	uimsbf



Table 7.33 - Syntax of NOISEbasicParaESC1()

Syntax	No. of bits	Mnemonic
NOISEbasicParaESC1() { if (noiseContFlag) { <b>contNoiseAmpl;</b> noiseAmplIndex = prevNoiseAmplIndex + contNoiseAmpl; } <b>numNoiseParaIndex;</b> numNoisePara = numNoiseParaTable[numNoiseIndex]; for (i=0; i<min(2,numNoisePara); i++) { <b>noiseLAR[i];</b> } }	3..8  4  2..17	<b>DIA</b>  <b>uimsbf</b>  <b>LARN1</b>

Table 7.34 - Syntax of NOISEbasicParaESC3()

Syntax	No. of bits	Mnemonic
NOISEbasicParaESC3() { for (i=2; i<numNoisePara; i++) { <b>noiseLAR[i];</b> } }	1..18	<b>LARN2</b>

Table 7.35 - Syntax of NOISEbasicParaESC4()

Syntax	No. of bits	Mnemonic
NOISEbasicParaESC4() { if (noiseEnvFlag) { <b>noiseEnvTmax;</b> <b>noiseEnvRatk;</b> <b>noiseEnvRdec;</b> } }	<b>tmbits</b> <b>atkbits</b> <b>decbits</b>	<b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b>

Table 7.36 - numNoiseParaTable

i	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
numNoiseParaTable[i]	1	2	3	4	5	6	7	9	11	13	15	17	19	21	23	25

Table 7.37 - Syntax of INDbasicParaESC1()

Syntax	No. of bits	Mnemonic
INDbasicParaESC1() { for (k=0; k<prevNumLine; k++) { <b>prevLineContFlag[k];</b> } i = 0; for (k=0; k<prevNumLine; k++) { if (prevLineContFlag[k]) {	1	<b>uimsbf</b>

```

        linePred[i] = k;
        lineContFlag[i++] = 1;
    }
}
while (i<numLine) {
    lineContFlag[i++] = 0;
}
}

```

**Table 7.38 - Syntax of INDbasicParaESC2()**

Syntax	No. of bits	Mnemonic
<pre> INDbasicParaESC2() {     lastNLFreq = 0;     for (i=0; i&lt;prevNumLine; i++) {         prevILFreqIndex[i] = ILFreqIndex[i];         prevILAmplIndex[i] = ILAmplIndex[i];     }     for (i=0; i&lt;numLine; i++) {         if (envFlag) {             <b>lineEnvFlag[i];</b>         }     }     for (i=0; i&lt;numLine; i++) {         if (!lineContFlag[i]) {             if (numLine-1-i &lt; 7) {                 <b>ILFreqInc[i];</b>                 /* SDCdecode (maxFindex-lastNLFreq, */                 /*     sdcILFTable[numLine-1-i]) */             }             else {                 <b>ILFreqInc[i];</b>                 /* SDCdecode (maxFindex-lastNLFreq, */                 /*     sdcILFTable[7]) */             }             ILFreqIndex[i] = lastNLFreq + ILFreqInc[i];             lastNLFreq = ILFreqIndex[i];             if (HILNquantMode) {                 <b>ILAmplRel[i];</b>                 /* SDCdecode (50, sdcILATable) */                 ILAmplIndex[i] = maxAmplIndex + ILAmplRel[i];             }             else {                 <b>ILAmplRel[i];</b>                 /* SDCdecode (25, sdcILATable) */                 ILAmplIndex[i] = maxAmplIndex +                     2*ILAmplRel[i];             }         }     } } </pre>	<p><b>1</b></p> <p><b>0..14</b></p> <p><b>0..14</b></p> <p><b>4..10</b></p> <p><b>3..9</b></p>	<p><b>uimsbf</b></p> <p><b>SDC</b></p> <p><b>SDC</b></p> <p><b>SDC</b></p> <p><b>SDC</b></p>

Table 7.39 - Syntax of INDbasicParaESC3()

Syntax	No. of bits	Mnemonic
<pre> INDbasicParaESC3() {   for (i=0; i&lt;numLine; i++) {     if (lineContFlag[i]) {       <b>DILFreq[i];</b>       ILFreqIndex[i] = prevILFreqIndex[linePred[i]] +         DILFreq[i];       <b>DILAmpl[i];</b>       ILAmplIndex[i] = prevILAmplIndex[linePred[i]] +         DILAmpl[i];     }   } } </pre>	<p>2..10</p> <p>3..8</p>	<p>DIF</p> <p>DIA</p>

Table 7.40 - Syntax of INDbasicParaESC4()

Syntax	No. of bits	Mnemonic
<pre> INDbasicParaESC4() {   if (phaseFlag) {     <b>numLinePhase;</b>   }   else {     numLinePhase = 0;   }   j = 0;   for (i=0; i&lt;numLine; i++) {     if (! linePred[i] &amp;&amp; j&lt;numLinePhase) {       <b>linePhase[i];</b>       linePhaseAvail[i] = 1;       j++;     }     else {       linePhaseAvail[i] = 0;     }   } } </pre>	<p>linebits</p> <p>phasebits</p>	<p>uimsbf</p> <p>uimsbf</p>

Table 7.41 - Syntax of HILNenexFrame()

Syntax	No. of bits	Mnemonic
<pre> HILNenexFrame() {   /* HILNenexLayer value in ParametricSpecificConfig() of */   /* this Elementary Stream must be used here! */   if (HILNenexLayer) {     HILNenexFrame();   }   else {     numLayer++;     HILNextFrame(numLayer);   } } </pre>		

Table 7.42 - Syntax of HILNenhaFrame()

Syntax	No. of bits	Mnemonic
<pre>HILNenhaFrame() {   if (envFlag) {     envTmaxEnha;     envRatkEnha;     envRdecEnha;   }   if (harmFlag) {     HARMenhaPara();   }   INDlenhaPara(); }</pre>	<p><b>tmEnhbits</b> <b>atkEnhbits</b> <b>decEnhbits</b></p>	<p><b>uimsbf</b> <b>uimsbf</b> <b>uimsbf</b></p>

Table 7.43 - Syntax of HARMenhaPara()

Syntax	No. of bits	Mnemonic
<pre>HARMenhaPara() {   for (i=0; i&lt;min(numHarmLine,10); i++) {     harmFreqEnha[i];     harmPhase[i];   } }</pre>	<p><b>fEnhbits[i]</b> <b>phasebits</b></p>	<p><b>uimsbf</b> <b>uimsbf</b></p>

Table 7.44 - Syntax of INDlenhaPara()

Syntax	No. of bits	Mnemonic
<pre>INDlenhaPara() {   for (i=0; i&lt;numLine; i++) {     lineFreqEnha[i];     linePhase[i];   } }</pre>	<p><b>fEnhbits[i]</b> <b>phasebits</b></p>	<p><b>uimsbf</b> <b>uimsbf</b></p>

Table 7.45 - Syntax of HILNextFrame()

Syntax	No. of bits	Mnemonic
<pre>HILNextFrame(numLayer) {   layPrevNumLine[numLayer] = layNumLine[numLayer];   /* layPrevNumLine[numLayer] = 0 for the */   /* first bitstream frame */   addNumLine[numLayer];   if (phaseFlag) {     layNumLinePhase[numLayer];   }   layNumLine[numLayer] = layNumLine[numLayer-1] +     addNumLine[numLayer];   for (k=0; k&lt;layPrevNumLine[numLayer-1]; k++) {     if (layPrevLineContFlag[numLayer-1][k]) {       layPrevLineContFlag[numLayer][k] = 1;     }   } }</pre>	<p><b>linebits</b> <b>linebits</b></p>	<p><b>uimsbf</b> <b>uimsbf</b></p>

<pre> else {     <b>layPrevLineContFlag[numLayer][k];</b> } } for (k=layPrevNumLine[numLayer-1];     k&lt;layPrevNumLine[numLayer]; k++) {     <b>layPrevLineContFlag[numLayer][k];</b> } i = layNumLine[numLayer-1]; for (k=0; k&lt;layPrevNumLine[numLayer-1]; k++) {     if (!layPrevLineContFlag[numLayer-1][k] &amp;&amp;         layPrevLineContFlag[numLayer][k]) {         linePred[i] = k;         lineContFlag[i++] = 1     } } for (k=layPrevNumLine[numLayer-1];     k&lt;layPrevNumLine[numLayer]; k++) {     if (layPrevLineContFlag[numLayer][k]) {         linePred[i] = k;         lineContFlag[i++] = 1;     } } while (i&lt;layNumLine[numLayer]) {     lineContFlag[i++] = 0; } INDIextPara(numLayer); if (phaseFlag) {     INDIextPhasePara(numLayer); } } </pre>	<p><b>1</b></p> <p><b>1</b></p>	<p><b>uimsbf</b></p> <p><b>uimsbf</b></p>
----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------	---------------------------------	-------------------------------------------

Table 7.46 - Syntax of INDIextPara()

Syntax	No. of bits	Mnemonic
<pre> INDIextPara(numLayer) {     lastNLFreq = 0;     for (i=layPrevNumLine[numLayer-1];         i&lt;layPrevNumLine[numLayer]; i++) {         prevILFreqIndex[i] = ILFreqIndex[i];         prevILAmplIndex[i] = ILAmplIndex[i];     }     for (i=layNumLine[numLayer-1];         i&lt;layNumLine[numLayer]; i++) {         if (envFlag) {             <b>lineEnvFlag[i];</b>         }         if (lineContFlag[i]) {             <b>DILFreq[i];</b>             ILFreqIndex[i] = prevILFreqIndex[linePred[i]] +                 DILFreq[i];             <b>DILAmpl[i];</b>             ILAmplIndex[i] = prevILAmplIndex[linePred[i]] +                 DILAmpl[i];         }         else {             if (layNumLine[numLayer]-1-i &lt; 7) { </pre>	<p><b>1</b></p> <p><b>2..10</b></p> <p><b>3..8</b></p>	<p><b>uimsbf</b></p> <p><b>DIF</b></p> <p><b>DIA</b></p>



100000000000000000110	-5.750	00110	0.650
100000000000000000111	-5.650	00111	0.750
100000000000000000100	-5.550	000100	0.850
100000000000000000101	-5.450	000101	0.950
100000000000000000110	-5.350	000110	1.050
100000000000000000111	-5.250	000111	1.150
100000000000000000100	-5.150	0000100	1.250
100000000000000000101	-5.050	0000101	1.350
100000000000000000110	-4.950	0000110	1.450
100000000000000000111	-4.850	0000111	1.550
100000000000000000100	-4.750	00000100	1.650
100000000000000000101	-4.650	00000101	1.750
100000000000000000110	-4.550	00000110	1.850
100000000000000000111	-4.450	00000111	1.950
100000000000000000100	-4.350	000000100	2.050
100000000000000000101	-4.250	000000101	2.150
100000000000000000110	-4.150	000000110	2.250
100000000000000000111	-4.050	000000111	2.350
100000000000000000100	-3.950	0000000100	2.450
100000000000000000101	-3.850	0000000101	2.550
100000000000000000110	-3.750	0000000110	2.650
100000000000000000111	-3.650	0000000111	2.750
100000000000000000100	-3.550	00000000100	2.850
100000000000000000101	-3.450	00000000101	2.950
100000000000000000110	-3.350	00000000110	3.050
100000000000000000111	-3.250	00000000111	3.150
100000000000000000100	-3.150	000000000100	3.250
100000000000000000101	-3.050	000000000101	3.350
100000000000000000110	-2.950	000000000110	3.450
100000000000000000111	-2.850	000000000111	3.550
100000000000000000100	-2.750	0000000000100	3.650
100000000000000000101	-2.650	0000000000101	3.750
100000000000000000110	-2.550	0000000000110	3.850
100000000000000000111	-2.450	0000000000111	3.950
100000000000000000100	-2.350	00000000000100	4.050
100000000000000000101	-2.250	00000000000101	4.150
100000000000000000110	-2.150	00000000000110	4.250
100000000000000000111	-2.050	00000000000111	4.350
100000100	-1.950	000000000000100	4.450
100000101	-1.850	000000000000101	4.550
100000110	-1.750	000000000000110	4.650
100000111	-1.650	000000000000111	4.750
10000100	-1.550	0000000000000100	4.850
10000101	-1.450	0000000000000101	4.950
10000110	-1.350	0000000000000110	5.050
10000111	-1.250	0000000000000111	5.150
1000100	-1.150	00000000000000100	5.250
1000101	-1.050	00000000000000101	5.350
1000110	-0.950	00000000000000110	5.450
1000111	-0.850	00000000000000111	5.550
10100	-0.750	000000000000000100	5.650
10101	-0.650	000000000000000101	5.750
10110	-0.550	000000000000000110	5.850
10111	-0.450	000000000000000111	5.950
1100	-0.350	0000000000000000100	6.050
1101	-0.250	0000000000000000101	6.150

1110	-0.150	00000000000000000110	6.250
1111	-0.050	00000000000000000111	6.350

**Table 7.49 - LARH2 code ( harmLAR[2..6] )**

codeword	harmLAR[.]	codeword	harmLAR[.]
1000000000000000010	-4.725	010	0.075
1000000000000000011	-4.575	011	0.225
1000000000000000010	-4.425	0010	0.375
1000000000000000011	-4.275	0011	0.525
1000000000000000010	-4.125	00010	0.675
1000000000000000011	-3.975	00011	0.825
1000000000000000010	-3.825	000010	0.975
1000000000000000011	-3.675	000011	1.125
1000000000000000010	-3.525	0000010	1.275
1000000000000000011	-3.375	0000011	1.425
1000000000000000010	-3.225	00000010	1.575
1000000000000000011	-3.075	00000011	1.725
1000000000000000010	-2.925	000000010	1.875
1000000000000000011	-2.775	000000011	2.025
1000000000000000010	-2.625	0000000010	2.175
1000000000000000011	-2.475	0000000011	2.325
1000000000000000010	-2.325	00000000010	2.475
1000000000000000011	-2.175	00000000011	2.625
1000000000000000010	-2.025	000000000010	2.775
1000000000000000011	-1.875	000000000011	2.925
1000000000000000010	-1.725	0000000000010	3.075
1000000000000000011	-1.575	0000000000011	3.225
1000000000000000010	-1.425	00000000000010	3.375
1000000000000000011	-1.275	00000000000011	3.525
1000000000000000010	-1.125	000000000000010	3.675
1000000000000000011	-0.975	000000000000011	3.825
1000000000000000010	-0.825	0000000000000010	3.975
1000000000000000011	-0.675	0000000000000011	4.125
1000000000000000010	-0.525	00000000000000010	4.275
1000000000000000011	-0.375	00000000000000011	4.425
1000000000000000010	-0.225	000000000000000010	4.575
1000000000000000011	-0.075	000000000000000011	4.725

**Table 7.50 - LARH3 code ( harmLAR[7..24] )**

codeword	harmLAR[.]	codeword	harmLAR[.]
1000000000000000001	-2.325	01	0.075
1000000000000000001	-2.175	001	0.225
1000000000000000001	-2.025	0001	0.375
1000000000000000001	-1.875	00001	0.525
1000000000000000001	-1.725	000001	0.675
1000000000000000001	-1.575	0000001	0.825
1000000000000000001	-1.425	00000001	0.975
1000000000000000001	-1.275	000000001	1.125
1000000000000000001	-1.125	0000000001	1.275
1000000000000000001	-0.975	00000000001	1.425
1000000000000000001	-0.825	000000000001	1.575
1000000000000000001	-0.675	0000000000001	1.725



10001	-0.525	0000000000000001	1.875
1001	-0.375	0000000000000001	2.025
101	-0.225	0000000000000001	2.175
11	-0.075	0000000000000001	2.325

Table 7.51 - LARN1 code ( noiseLAR[0,1] )

codeword	noiseLAR[.]	codeword	noiseLAR[.]
100000000000000001	-4.65	01	0.15
100000000000000001	-4.35	001	0.45
100000000000000001	-4.05	0001	0.75
100000000000000001	-3.75	00001	1.05
100000000000000001	-3.45	000001	1.35
100000000000000001	-3.15	0000001	1.65
100000000001	-2.85	00000001	1.95
1000000001	-2.55	000000001	2.25
100000001	-2.25	0000000001	2.55
10000001	-1.95	00000000001	2.85
1000001	-1.65	000000000001	3.15
100001	-1.35	0000000000001	3.45
10001	-1.05	00000000000001	3.75
1001	-0.75	0000000000000001	4.05
101	-0.45	00000000000000001	4.35
11	-0.15	000000000000000001	4.65

Table 7.52 - LARN2 code ( noiseLAR[2..24] )

codeword	noiseLAR[.]	codeword	noiseLAR[.]
11000000000000000001	-6.35	101	0.35
11000000000000000001	-5.95	1001	0.75
11000000000000000001	-5.55	10001	1.15
11000000000000000001	-5.15	100001	1.55
11000000000000000001	-4.75	1000001	1.95
11000000000000000001	-4.35	10000001	2.35
11000000000000000001	-3.95	100000001	2.75
110000000001	-3.55	1000000001	3.15
1100000001	-3.15	10000000001	3.55
110000001	-2.75	100000000001	3.95
11000001	-2.35	1000000000001	4.35
1100001	-1.95	10000000000001	4.75
110001	-1.55	1000000000000001	5.15
11001	-1.15	10000000000000001	5.55
1101	-0.75	100000000000000001	5.95
111	-0.35	1000000000000000001	6.35
0	0.00		

**Table 7.53 - DIA code**

codeword	value	codeword	value
111 1 1111	-25	001	1
111 1 1110	-24	011 0	2
111 1 1101	-23	100 0	3
111 1 xxxx	-y	101 0 0	4
111 1 0001	-11	101 0 1	5
111 1 0000	-10	110 0 00	6
110 1 11	-9	110 0 01	7
110 1 10	-8	110 0 10	8
110 1 01	-7	110 0 11	9
110 1 00	-6	111 0 0000	10
101 1 1	-5	111 0 0001	11
101 1 0	-4	111 0 xxxx	y
100 1	-3	111 0 1101	23
011 1	-2	111 0 1110	24
010	-1	111 0 1111	25
000	0		

**Table 7.54 - DIF code**

codeword	value	codeword	value
11 11 1 11111	-42	01 0	1
11 11 1 11110	-41	10 0 0	2
11 11 1 11101	-40	10 0 1	3
11 11 1 xxxxx	-y	11 00 0	4
11 11 1 00001	-12	11 01 0 0	5
11 11 1 00000	-11	11 01 0 1	6
11 10 1 11	-10	11 10 0 00	7
11 10 1 10	-9	11 10 0 01	8
11 10 1 01	-8	11 10 0 10	9
11 10 1 00	-7	11 10 0 11	10
11 01 1 1	-6	11 11 0 00000	11
11 01 1 0	-5	11 11 0 00001	12
11 00 1	-4	11 11 0 xxxxx	y
10 1 1	-3	11 11 0 11101	40
10 1 0	-2	11 11 0 11110	41
01 1	-1	11 11 0 11111	42
00	0		

**Table 7.55 - DHF code**

codeword	value	codeword	value
11 1 111111	-69	01 0	1
11 1 111110	-68	10 0 00	2
11 1 111101	-67	10 0 01	3
11 1 xxxxxx	-y	10 0 10	4
11 1 000001	-7	10 0 11	5
11 1 000000	-6	11 0 000000	6
10 1 11	-5	11 0 000001	7
10 1 10	-4	11 0 xxxxxx	y
10 1 01	-3	11 0 111101	67
10 1 00	-2	11 0 111110	68

01 1	-1	11 0 111111	69
00	0		

Table 7.56 - HFS code

codeword	value	codeword	value
1 1 1 1111	-17	1 0 0	1
1 1 1 1110	-16	1 0 1 0000	2
1 1 1 1101	-15	1 0 1 0001	3
1 1 1 xxxx	-y	1 0 1 xxxx	y
1 1 1 0001	-3	1 0 1 1101	15
1 1 1 0000	-2	1 0 1 1110	16
1 1 0	-1	1 0 1 1111	17
0	0		

Notes on Table 7.53 to Table 7.56: The grouping of bits within a codeword (e.g. “1 1 1 1111”) is provided for easier readability only. Codewords not explicitly listed in the codebooks (e.g. “1 1 1 xxxx”) are defined by incrementing the implicit part of the codeword “xxxx” (uimbsf) and the magnitude “y” of the corresponding value. In all cases, the codewords and values for the two smallest and the three largest magnitudes are listed explicitly.

#### 7.3.2.4 HILN SubDivisionCode (SDC)

The SubDivisionCode (SDC) is an algorithmically generated variable length code, based on a given table and a given number of different codewords. The decoding process is defined below.

The idea behind this coding scheme is the subdivision of the probability density function into two parts which represent an equal probability. One bit is transmitted that determines the part the value to be coded is located. This subdivision is repeated until the width of the part is one and then its position is equal to the value being coded. The positions of the boundaries are taken out of a table of 32 quantized, fixed point values. Besides this table (parameter “tab”) the number of different codewords (parameter “k”) is needed too.

The following C function SDCDecode(k, tab) together with the 9 tables sdcILATable[32] and sdcILFTable[8][32] describe the decoding. The function GetBit() returns the next bit in the stream.

```
int sdcILATable[32] = {
    0, 13, 27, 41, 54, 68, 82, 96, 110, 124, 138, 152, 166, 180, 195, 210,
    225, 240, 255, 271, 288, 305, 323, 342, 361, 383, 406, 431, 460, 494, 538, 602
};

int sdcILFTable[8][32] = {
{ 0, 53, 87, 118, 150, 181, 212, 243, 275, 306, 337, 368, 399, 431, 462, 493,
  524, 555, 587, 618, 649, 680, 711, 743, 774, 805, 836, 867, 899, 930, 961, 992 },
{ 0, 34, 53, 71, 89, 106, 123, 141, 159, 177, 195, 214, 234, 254, 274, 296,
  317, 340, 363, 387, 412, 438, 465, 494, 524, 556, 591, 629, 670, 718, 774, 847 },
{ 0, 26, 41, 54, 66, 78, 91, 103, 116, 128, 142, 155, 169, 184, 199, 214,
  231, 247, 265, 284, 303, 324, 346, 369, 394, 422, 452, 485, 524, 570, 627, 709 },
{ 0, 23, 35, 45, 55, 65, 75, 85, 96, 106, 117, 128, 139, 151, 164, 177,
  190, 204, 219, 235, 252, 270, 290, 311, 334, 360, 389, 422, 461, 508, 571, 665 },
{ 0, 20, 30, 39, 48, 56, 64, 73, 81, 90, 99, 108, 118, 127, 138, 149,
  160, 172, 185, 198, 213, 228, 245, 263, 284, 306, 332, 362, 398, 444, 507, 608 },
{ 0, 18, 27, 35, 43, 50, 57, 65, 72, 79, 87, 95, 104, 112, 121, 131,
  141, 151, 162, 174, 187, 201, 216, 233, 251, 272, 296, 324, 357, 401, 460, 558 },
{ 0, 16, 24, 31, 38, 45, 51, 57, 64, 70, 77, 84, 91, 99, 107, 115,
  123, 132, 142, 152, 163, 175, 188, 203, 219, 237, 257, 282, 311, 349, 403, 493 },
{ 0, 12, 19, 25, 30, 35, 41, 46, 51, 56, 62, 67, 73, 79, 85, 92,
  99, 106, 114, 122, 132, 142, 153, 165, 179, 195, 213, 236, 264, 301, 355, 452 }
};
```

```
int SDCDecode (int k, int *tab)
{
    int *pp;
    int g, dp, min, max;

    min=0;
    max=k-1;
    pp=tab+16;
    dp=16;

    while ( min!=max )
    {
        if ( dp ) g=(k*(*pp))>>10; else g=(max+min)>>1;
        dp>>=1;
        if ( GetBit()==0 ) { pp-=dp; max=g; } else { pp+=dp; min=g+1; }
    }
    return max;
}
```

## 7.4 Bitstream semantics

### 7.4.1 Decoder configuration (ParametricSpecificConfig)

Bitstream elements:

**isBaseLayer** A one-bit identifier representing whether the corresponding layer is the base layer (1) or an enhancement or extension layer (0).

#### 7.4.1.1 Parametric Audio decoder configuration

Bitstream elements:

**PARAMode** A 2 bit field indicating parametric coder operation mode.

**PARAextensionFlag** A flag indicating the presence of MPEG-4 version 3 data (for future use).

#### 7.4.1.2 HILN decoder configuration

Bitstream elements:

**HILNquantMode** A 1 bit field indicating the individual line quantizer mode.

**HILNmaxNumLine** An 8 bit field indicating the maximum number of individual lines in a bitstream frame. It also determines linebits field size (see Table 7.8).

**HILNsampleRateCode** A 4 bit field indicating the sampling rate used for HILN parameter decoding (see Table 7.7).

**HILNframeLength** A 12 bit field indicating the HILN frame length in samples at the sampling rate indicated by HILNsampleRateCode.

**HILNcontMode** A 2 bit field indicating the additional decoder line continuation mode (see Table 7.9).

**HILNenhaLayer** A flag indicating whether this Elementary Stream contains an enhancement layer or an extension layer.

**HILNenhaQuantMode** A 2 bit field indicating frequency enhancement quantizer mode (see Table 7.11).

## 7.4.2 Bitstream frame (sIPacketPayload)

### 7.4.2.1 Parametric Audio bitstream frame

Bitstream elements:

<b>PARAswitchMode</b>	A flag indicating which coding tool is used in the current frame of a HVXC/HILN switching bitstream (see Table 7.16).
<b>PARAmixMode</b>	A 2 bit field indicating which coding tools are used in the current frame of a HVXC/HILN mixing bitstream (see Table 7.18).

### 7.4.2.2 HILN bitstream frame

Bitstream elements:

<b>numLine</b>	A field indicating the number of individual lines in the current frame.
<b>harmFlag</b>	A flag indicating the presence of harmonic line data in the current frame.
<b>noiseFlag</b>	A flag indicating the presence of noise component data in the current frame.
<b>phaseFlag</b>	A flag indicating the presence of line start phase data in the current frame.
<b>numLinePhase</b>	A field indicating the number of individual lines with start phase in the current frame.
<b>maxAmplIndexCoded</b>	A field indicating the maximum amplitude of a new signal component in the current frame.
<b>envFlag</b>	A flag indicating the presence of envelope data in the current frame.
<b>envTmax</b>	Coded envelope parameter: time of maximum.
<b>envRatk</b>	Coded envelope parameter: attack rate.
<b>envRdec</b>	Coded envelope parameter: decay rate.
<b>prevLineContFlag[k]</b>	A flag indicating that the k-th individual line of the previous frame is continued in the current frame.
<b>numHarmParaIndex</b>	A field indicating the number of harmonic line LPC parameters in the current frame (see Table 7.30).
<b>numHarmLineIndex</b>	A field indicating the number of harmonic lines in the current frame (see Table 7.31).
<b>harmContFlag</b>	A flag indicating that the harmonic lines are continued from the previous frame.
<b>numHarmPhase</b>	A field indicating the number of harmonic lines with start phase in the current frame.
<b>harmEnvFlag</b>	A flag indicating that the amplitude envelope is applied to the harmonic lines.
<b>contHarmAmpl</b>	Coded amplitude change of the harmonic lines (see Table 7.53).
<b>contHarmFreq</b>	Coded fundamental frequency change of the harmonic lines (see Table 7.55).
<b>harmAmplRel</b>	Coded relative amplitude of the harmonic lines.

<b>harmFreqIndex</b>	Coded fundamental frequency of the harmonic lines.
<b>harmFreqStretch</b>	Coded frequency stretching parameter of the harmonic lines (see Table 7.56).
<b>harmLAR[i]</b>	Coded LAR LPC parameters of the harmonic lines (see Table 7.48, Table 7.49, Table 7.50).
<b>harmPhase[i]</b>	Coded phase of i-th harmonic line.
<b>numNoiseParaIndex</b>	A field indicating the number of noise LPC parameters in the current frame (see Table 7.36).
<b>noiseContFlag</b>	A flag indicating that the noise is continued from the previous frame.
<b>noiseEnvFlag</b>	A flag indicating that noise envelope data is present in the current frame.
<b>contNoiseAmpl</b>	Coded amplitude change of the noise (see Table 7.53).
<b>noiseAmplRel</b>	Coded relative amplitude of the noise.
<b>noiseEnvTmax</b>	Coded noise envelope parameter: time of maximum
<b>noiseEnvRatk</b>	Coded noise envelope parameter: attack rate.
<b>noiseEnvRdec</b>	Coded noise envelope parameter: decay rate.
<b>noiseLAR[i]</b>	Coded LAR LPC parameters of the noise (see Table 7.51, Table 7.52).
<b>lineEnvFlag[i]</b>	A flag indicating that the amplitude envelope is applied to the i-th individual line.
<b>DILFreq[i]</b>	Coded frequency change of i-th individual line (see Table 7.54).
<b>DILAmpl[i]</b>	Coded amplitude change of i-th individual line (see Table 7.53).
<b>ILFreqInc[i]</b>	Coded frequency increment of i-th individual line (see Subclause 7.3.2.4).
<b>ILAmplRel[i]</b>	Coded relative amplitude of i-th individual line (see Subclause 7.3.2.4).
<b>linePhase[i]</b>	Coded phase of i-th individual line.
<b>envTmaxEnha</b>	Coded envelope enhancement parameter: time of maximum.
<b>envRatkEnha</b>	Coded envelope enhancement parameter: attack rate.
<b>envRdecEnha</b>	Coded envelope enhancement parameter: decay rate.
<b>harmFreqEnha[i]</b>	Coded frequency enhancement of i-th harmonic line.
<b>lineFreqEnha[i]</b>	Coded frequency enhancement of i-th individual line.
<b>addNumLine[i]</b>	A field indicating the number of individual lines in extension layer i of the current frame.
<b>layNumLinePhase[i]</b>	A field indicating the number of individual lines with start phase in extension layer i of the current frame.
<b>layPrevLineContFlag[i][k]</b>	A flag indicating that the k-th individual line of the previous frame is continued in extension layer i of the current frame.

Help elements:

<b>numLayer</b>	The number of extension layers available (0 if only base layer available).
<b>layNumLine[i]</b>	The total number of individual lines in the current frame as conveyed in the base layer and the first i extension layers.
<b>prevNumLine</b>	The number of individual lines in the previous frame.
<b>layPrevNumLine[i]</b>	The total number of individual lines in the previous frame as conveyed in the base layer and the first i extension layers.
<b>maxAmplIndex</b>	The maximum amplitude of a new signal component in the current frame.
<b>linePred[i]</b>	Index of the predecessor in the previous frame of the i-th individual line in the current frame.
<b>lineContFlag[i]</b>	A flag indicating that line i in the current frame is continued from the previous frame.
<b>numHarmPara</b>	The number of harmonic line LPC parameters in the current frame.
<b>numHarmLine</b>	The number of harmonic lines in the current frame.
<b>harmAmplIndex</b>	Amplitude index of the harmonic lines in the current frame.
<b>harmFreqIndex</b>	Fundamental frequency index of the harmonic lines in the current frame.
<b>prevHarmAmplIndex</b>	Amplitude index of the harmonic lines in the previous frame.
<b>prevHarmFreqIndex</b>	Fundamental frequency index of the harmonic lines in the previous frame.
<b>harmPhaseAvail[i]</b>	A flag indicating that start phase information is available for the i-th harmonic line.
<b>numNoisePara</b>	The number of noise LPC parameters in the current frame.
<b>noiseAmplIndex</b>	Amplitude index of the noise in the current frame.
<b>prevNoiseAmplIndex</b>	Amplitude index of the noise in the previous frame.
<b>lastNLFreq</b>	Individual line frequency increment accumulator.
<b>ILFreqIndex[i]</b>	Frequency index of the i-th individual line in the current frame.
<b>ILAmplIndex[i]</b>	Amplitude index of the i-th individual line in the current frame.
<b>prevILFreqIndex[i]</b>	Frequency index of the i-th individual line in the previous frame.
<b>prevILAmplIndex[i]</b>	Amplitude index of the i-th individual line in the previous frame.
<b>linePhaseAvail[i]</b>	A flag indicating that start phase information is available for the i-th individual line.
<b>linebits</b>	Number of bits for numLine.
<b>tmbits</b>	Number of bits for envTmax.
<b>atkbits</b>	Number of bits for encRatk.
<b>decbits</b>	Number of bits for envRdec.
<b>tmEnhbits</b>	Number of bits for envTmaxEnha.

- atkEnhbits**                      Number of bits for encRatkEnha.
- decEnhbits**                      Number of bits for envRdecEnha.
- fEnhbits[i]**                      Number of bits for lineFreqEnha[i] and harmFreqEnha[i] (see Subclause 7.3.1.2).
- phasebits**                      Number of bits for linePhase and harmPhase.

## 7.5 Parametric decoder tools

### 7.5.1 HILN decoder tools

The Harmonic and Individual Lines plus Noise (HILN) decoder utilizes a set of parameters which are encoded in the bitstream to describe the audio signal. Three different signal models are supported:

**Table 7.57 - HILN signal models**

signal model	description	essential parameters
harmonic lines	group of sinusoidal signals with common fundamental frequency	fundamental frequency and amplitudes of the spectral lines
individual lines	sinusoidal signals	frequency and amplitude of the individual spectral lines
noise	spectrally shaped noise signal	spectral shape and power of the noise

The HILN decoder first reconstructs these parameters from the bitstream with a set of decoding tools and then synthesizes the audio signal based on these parameters using a set of synthesizer tools:

- harmonic line decoder
- individual line decoder
- noise decoder
- harmonic and individual line synthesizer
- noise synthesizer

The HILN decoder tools reconstruct the parameters of the harmonic and individual lines (frequency, amplitude) and the noise (spectral shape) as well as possible envelope parameters from the bitstream.

The HILN synthesizer tools reconstruct one frame of the audio signal based on the parameters decoded by the HILN decoder tools for the current bitstream frame.

The samples of the decoded audio signal have a full scale range of [-32768, 32767] and eventual outliers must be limited (“clipped”) to these values.

The HILN decoder supports a wide range of frame lengths and sampling frequencies. By scaling the synthesizer frame length with an arbitrary factor, speed change functionality is available at the decoder. By scaling the line frequencies and resampling the noise signal with an arbitrary factor, pitch change functionality is available at the decoder.

The HILN decoder can operate in two different modes, as basic decoder and as enhanced decoder. The basic decoder which is used for normal operation only evaluates the information available in the bitstream elements HILNbasicFrame() to reconstruct the audio signal. To allow large step scalability in combination with other coder tools (e.g. GA scalable) the additional bitstream elements HILNenhaFrame() need to be transmitted and the HILN decoder must operate in the enhanced mode which exploits the information of both HILNbasicFrame() and



HILNenhaFrame(). This mode reconstructs an audio signal with well defined phase relationships which can be combined with a residual signal coded at higher bitrates using an enhancement coder (e.g. GA scalable). If the HILN decoder is used in this way as a core for a scalable coder no noise signal must be synthesized for the signal which is given to the enhancement decoder.

Due to the parametric signal representation utilized by the HILN parametric coder, it is well suited for applications requiring bitrate scalable coding. HILN bitrate scalable coding is accomplished by supplementing the data encoded in an HILNbasicFrame() of the basic bitstream by data encoded in one or more HILNextFrame() of one or more extension bitstreams transmitted as additional Elementary Streams. It should be noted that the coding efficiency of a combined bitstream consisting of a basic and one or more extension bitstreams is slightly lower than the coding efficiency of a non-scalable basic bitstream having the same total bitrate.

### 7.5.1.1 Harmonic line decoder

#### 7.5.1.1.1 Tool description

This tool decodes the parameters of the harmonic lines transmitted in the bitstream.

#### 7.5.1.1.2 Definitions

<b>prevNumHarmPara</b>	The number of harmonic line LPC parameters in the previous frame.
<b>harmLPCPara[i]</b>	Harmonic line LPC parameter $i$ in the current frame (LARs for harmonic tone spectrum).
<b>prevHarmLPCPara[i]</b>	Harmonic line LPC parameter $i$ in the previous frame (LARs for harmonic tone spectrum).
<b>hFreq</b>	Fundamental frequency of the harmonic lines.
<b>hStretch</b>	Frequency stretching of the harmonic lines.
<b>harmAmpl</b>	Harmonic tone amplitude.
<b>harmPwr</b>	Harmonic tone power.
<b>hLineAmpl[i]</b>	Amplitude of $i$ -th harmonic line.
<b>hLineFreq[i]</b>	Frequency of $i$ -th harmonic line (in Hz).
<b>hLineAmplEnh[i]</b>	Enhanced amplitude of $i$ -th harmonic line.
<b>hLineFreqEnh[i]</b>	Enhanced frequency of $i$ -th harmonic line (in Hz).
<b>hLinePhaseEnh[i]</b>	Phase of $i$ -th harmonic line (in rad).
<b>ha[i]</b>	Unscaled amplitude of $i$ -th harmonic line.
<b>r[i]</b>	LPC reflection coefficients.
<b>h[i]</b>	LPC impulse response.
<b>H(z)</b>	LPC system function.

#### 7.5.1.1.3 Decoding process

If the harmFlag is set and thus HARMbasicPara() data and in enhancement mode HARMenhaPara() data is available in the current frame, the parameters of the harmonic lines are decoded and dequantized as follows:

7.5.1.1.3.1 Basic decoder

A harmonic tone is represented by its fundamental frequency, its power and a set of LPC-Parameters.

First the harmNumPara LAR parameters are reconstructed. Prediction from the previous frame is used when harmContFlag is set.

```
float harmLPCMean[25] = { 5.0, -1.5, 0.0, 0.0, 0.0, ... , 0.0 };
float harmPredCoeff[25] = { 0.75, 0.75, 0.5, 0.5, 0.5, ... , 0.5 };

for (i=0; i<numHarmPara; i++) {
    if ( i<prevNumHarmPara && harmContFlag )
        pred = harmLPCMean[i] +
            (prevHarmLPCPara[i]-harmLPCMean[i])*harmPredCoeff[i];
    else
        pred = harmLPCMean[i];
    harmLPCPara[i] = pred + harmLAR[i];
}
```

Parameters needed in the following frame are stored in the frame-to-frame memory:

```
prevNumHarmPara = numHarmPara;
for (i=0; i<numHarmPara; i++)
    prevHarmLPCPara[i] = harmLPCPara[i];
```

The fundamental frequency and stretching of the harmonic lines are dequantized:

```
hFreq = 20 * exp( log(4000./20.) * (harmFreqIndex+0.5) / 2048.0 );
hStretch = harmFreqStretch / 16000.0;
```

The amplitude and power of the harmonic tone is dequantized as follows:

```
harmAmpl = 32768 * pow(10, -1.5*(harmAmplIndex+0.5)/20 );
harmPwr = harmAmpl*harmAmpl;
```

The harmEnvFlag and harmContFlag flags require no further dequantization; they are directly passed on to the synthesizer tool.

The LPC-Parameters are transmitted in the form of “Logarithmic Area Ratios” (LAR) as described above. After decoding the parameters the frequencies and amplitudes of the harmNumLine partials of the harmonic tone are calculated as follows:

The frequencies of the harmonic lines are calculated:

```
for (i=0; i<numHarmLine; i++)
    hLineFreq[i] = hFreq * (i+1) * (1 + hStretch*(i+1));
```

The LPC-Parameters represent an IIR-Filter. The amplitudes of the sinusoids are obtained by calculating the absolute value of this filter’s system function H(z) at the corresponding frequencies.

```
for (i=0; i<numHarmLine; i++)
    ha[i] = abs( H( exp( j * pi * (i+1)/(numHarmLine+1) ) ) );
```

with  $j = \sqrt{-1}$  and

$$H(z) = 1 / ( 1 - h[0]*pow(z,-1) - h[1]*pow(z,-2) - ... - h[numHarmPara-1]*pow(z,-numHarmPara) )$$

The impulse response h[i] is calculated from the LARs by the following algorithm:

In a first step the LARs are converted to reflection coefficients:

```
for (i=0; i<numHarmPara; i++)
    r[i] = ( exp(harmLPCPara[i]) - 1 ) / ( exp(harmLPCPara[i]) + 1 );
```

After this the reflection coefficients are converted to the time response. The C function given below does this conversion in place (call with  $x[i]=r[i]$  and  $N=numHarmPara$ ; returns with  $x[i]=h[i]$ ):

```
void Convert_k_to_h (float *x, int N)
{
    int i,j;
    float a,b,c;

    for (i=1; i<N; i++)
    {
        c = x[i];

        for (j=0; j<i-j-1; j++)
        {
            a = x[ j ];
            b = x[i-j-1];
            x[j] = a-c*b;
            x[i-j-1] = b-c*a;
        }
        if ( j==i-j-1 )
            x[j] -= c*x[j];
    }
}
```

After calculating the amplitudes  $ha[i]$  they must be normalized and multiplied with  $harmAmpl$  to find the harmonic line amplitudes meeting the condition:

$$\text{sum} ( hLineAmpl[i]*hLineAmpl[i] ) = \text{power of harmonic tone}$$

This is realized as follows:

```
p = 0.0;
for (i=0; i<numHarmLine; i++)
    p += ha[i]*ha[i];
s = sqrt( harmPwr / p );
for (i=0; i<numHarmLine; i++)
    hLineAmpl[i] = ha[i] * s;
```

The optional phase information is decoded as follows:

```
for (i=0; i<numHarmLine; i++) {
    if (harmPhaseAvail[i]) {
        hStartPhase[i] = 2*pi*(harmPhase[i]+0.5)/(1<<phasebits)-pi;
        hStartPhaseAvail[i] = 1;
    }
    else
        hStartPhaseAvail[i] = 0;
}
```

#### 7.5.1.1.3.2 Enhanced decoder

In this mode, the harmonic line parameters decoded by the basic decoder are refined and also line phases are decoded using the information contained in  $HARMenphaPara()$  as follows:

For the first maximum 10 harmonic lines  $i$

$$i = 0 \dots \min(\text{numHarmLine}, 10) - 1$$

the enhanced harmonic line parameters are calculated using the basic harmonic line parameters and the data in the enhancement bitstream:

```
hLineAmplEnh[i] = hLineAmpl[i];
hLineFreqEnh[i] = hLineFreq[i] *
    (1+((harmFreqEnh[i]+0.5)/(1<<fEnhbits[i])-0.5)*(hFreqRelStep-1));
```

where  $hFreqRelStep$  is the ratio of two neighboring fundamental frequency quantizer steps:

$$hFreqRelStep = \exp(\log(4000/20)/2048);$$

For both line types the phase is decoded from the enhancement bitstream:

$$hLinePhaseEnh[i] = 2\pi * (\text{harmPhase}[i] + 0.5) / (1 \ll \text{phasebits}) - \pi$$

### 7.5.1.2 Individual line decoder

#### 7.5.1.2.1 Tool description

The individual line basic bitstream decoder reconstructs the line parameters frequency, amplitude, and envelope from the bitstream. The enhanced bitstream decoder reconstructs the line parameters frequency, amplitude, and envelope with finer quantization and additionally reconstructs the line parameters phase.

#### 7.5.1.2.2 Definitions

<b>t_max</b>	Envelope parameter: time of maximum.
<b>r_atk</b>	Envelope parameter: attack rate.
<b>r_dec</b>	Envelope parameter: decay rate.
<b>ampl[i]</b>	Amplitude of i-th individual line.
<b>freq[i]</b>	Frequency of i-th individual line (in Hz).
<b>startPhase[i]</b>	Start phase of i-th individual line.
<b>startPhaseAvail[i]</b>	A flag indicating that start phase information is available for the i-th individual line.
<b>t_maxEnh</b>	Enhanced envelope parameter: time of maximum.
<b>r_atkEnh</b>	Enhanced envelope parameter: attack rate.
<b>r_decEnh</b>	Enhanced envelope parameter: decay rate.
<b>amplEnh[i]</b>	Enhanced amplitude of i-th individual line.
<b>freqEnh[i]</b>	Enhanced frequency of i-th individual line (in Hz).
<b>phaseEnh[i]</b>	Phase of i-th individual line (in rad).

#### 7.5.1.2.3 Decoding process

##### 7.5.1.2.3.1 Basic decoder

The basic decoder reconstructs the line parameters from the data contained in `HILNbasicFrame()` and `INDIbasicPara()` in the following way:

For each frame, first the number of individual lines encoded in this frame is read from `HILNbasicFrame()`:

```
numLine
```

Then the frame envelope flag is read from `HILNbasicFrame()`:

```
envFlag
```

If `envFlag = 1` then the 3 envelope parameters `t_max`, `r_atk`, and `r_dec` are decoded from `HILNbasicFrame()`:

```

t_max = (envTmax+0.5)/(1<<tmbits);
r_atk = tan(pi/2*max(0,envRatk-0.5)/((1<<atkbits)-1))/0.2;
r_dec = tan(pi/2*max(0,envRdec-0.5)/((1<<decbits)-1))/0.2;

```

These envelope parameters are valid for the harmonic lines as well as for the individual lines. Thus the envelope parameters envTmax, envRatk, envRdec must be dequantized if present, even if numLine == 0.

For each line k of the previous frame

```
k = 0 .. prevNumLine-1
```

the previous line continuation flag is read from HILNbasicFrame():

```
prevLineContFlag[k]
```

If prevLineContFlag[k] == 1 then line k of the previous frame is continued in the current frame. If prevLineContFlag[k] == 0 then line k of the previous frame is not continued.

In the current frame, first the parameters of all continued lines are encoded followed by the parameters of the new lines. Therefore, the line continuation flag and the line predecessor are determined before decoding the line parameters:

```

i = 0;
for (k=0;k<prevNumLine;k++)
  if (prevLineContFlag[k]) {
    linePred[i] = k;
    lineContFlag[i++] = 1;
  }
while (i<numLine)
  lineContFlag[i++] = 0;

```

The parameters of new lines are encoded with increasing frequency index, using a differential encoding scheme. Therefore the following initialization is required once for each frame:

```
lastNLFreq = 0;
```

For each line i of the current frame

```
i = 0 .. numLine-1
```

the line parameters are decoded from INDlbasicPara() now.

If envFlag == 1 then the line envelope flag is read from INDlbasicPara():

```
lineEnvFlag[i]
```

If lineContFlag[i] == 1 then the parameters of a continued line are decoded from INDlbasicPara() based on the amplitude and frequency index of its predecessor in the previous frame:

```

ILFreqIndex[i] = prevILFreqIndex[linePred[i]] + DILFreq[i];
ILAmplIndex[i] = prevILAmplIndex[linePred[i]] + DILAmpl[i];

```

If lineContFlag[i] == 0 then the parameters of a new line are decoded from INDlbasicPara():

```

if (numLine-1-i < 7)
  ILFreqInc[i] = SDCdecode (maxFindex-lastNLFreq, sdcILFTable[numLine-1-i]);
else
  ILFreqInc[i] = SDCdecode (maxFindex-lastNLFreq, dcILFTable[7]);
ILFreqIndex[i] = lastNLFreq + ILFreqInc[i];
lastNLFreq = ILFreqIndex[i];
if (HILNquantMode) {
  ILAmplRel[i] = SDCdecode (50, sdcILATable);
  ILAmplIndex[i] = maxAmplIndex + ILAmplRel[i];
}
else {

```

```

    ILAmpRel[i] = SDCdecode (25, sdcILATable);
    ILAmpIndex[i] = maxAmpIndex + 2*ILAmpRel[i];
}

```

The line parameter indices are stored for decoding of the line parameters of the next frame:

```

prevNumLine = numLine;
for (i=0; i<prevNumLine; i++) {
    prevILFreqIndex[i] = ILFreqIndex[i];
    prevILAmpIndex[i] = ILAmpIndex[i];
}

```

The basic decoder also handles combinations of a basic layer and one or more extension layers. If data from a total of numLayer extension layers is available to the basic decoder, the values of layNumLine[numLayer] and layPrevNumLine[numLayer] are to be used instead of numLine and prevNumLine respectively. The values of ILAmpIndex[i], ILFreqIndex[i], lineContFlag[i], and linePred[i] as determined by the bitstream syntax description are to be used.

The amplitudes and frequencies of the individual lines are now dequantized from the indices:

```

for (i=0; i<numLine; i++) {
    ampl[i] = 32768 * pow(10, -1.5*(ILAmpIndex+0.5)/20 );
    if ( ILFreqIndex<160 )
        freq[i] = (ILFreqIndex+0.5) * 3.125;
    else
        freq[i] = 500 * exp( 0.00625 * (ILFreqIndex+0.5-160) );
}

```

The optional start phase information is decoded as follows:

```

for (i=0; i<numLine; i++) {
    if (linePhaseAvail[i]) {
        startPhase[i] = 2*pi*(linePhase[i]+0.5)/(1<<phasebits)-pi;
        startPhaseAvail[i] = 1;
    }
    else
        startPhaseAvail[i] = 0;
}

```

If the decoding process starts with an arbitrary frame of a bitstream, all individual lines which are marked in the bitstream as to be continued from previous frames which have not been decoded are to be muted.

#### 7.5.1.2.3.2 Enhanced decoder

The enhanced decoder refines the line parameters obtained from the basic decoder and also decodes the line phases. The additional information is contained in bitstream element INDlenhaPara() and evaluated in the following way:

First, all operations of the basic decoder have to be carried out in order to allow correct decoding of parameters for continued lines.

If envFlag == 1 then the enhanced parameters t\_maxEnh, r\_atkEnh, and r\_decEnh are decoded using the envelope data conveyed in HILNbasicFrame() and HILNenhaFrame():

```

t_maxEnh = (envTmax+(envTmaxEnh+0.5)/(1<<tmEnhbits))/(1<<tmbits);
if (envRatk==0)
    r_atkEnh = 0;
else
    r_atkEnh = tan(pi/2*(envRatk-1+(envRatkEnh+0.5)/(1<<atkEnhbits)))/
                ((1<<atkbits)-1))/0.2;
if (envRdec==0)
    r_decEnh = 0;
else
    r_decEnh = tan(pi/2*(envRdec-1+(envRdecEnh+0.5)/(1<<decEnhbits)))/
                ((1<<decbits)-1))/0.2;

```

For each line  $i$  of the current frame

```
i = 0 .. numLine-1
```

the enhanced line parameters are obtained by refining the parameters from the basic decoder with the data in `INDIenhaPara()`:

```

amplEnh[i] = ampl[i];
if (fEnhbits[i]!=0) {
    if ( ILFreqIndex<160 )
        freqEnh[i] = (ILFreqIndex+0.5 + ((lineFreqEnh[i]+0.5)/(1<<fEnhbits[i])-0.5)) *
3.125;
    else
        freqEnh[i] = 500 * exp( 0.00625 * (ILFreqIndex+0.5-160 +
            ((lineFreqEnh[i]+0.5)/(1<<fEnhbits[i])-0.5)) );
}
else
    freqEnh[i] = freq[i];

```

For both line types the phase is decoded from the enhancement bitstream:

```
phaseEnh[i] = 2*pi*(linePhase[i]+0.5)/(1<<phasebits)-pi;
```

### 7.5.1.3 Noise decoder

#### 7.5.1.3.1 Tool description

This tool decodes the noise parameters transmitted in the bitstream.

#### 7.5.1.3.2 Definitions

<b>prevNumNoisePara</b>	The number of noise LPC parameters in the previous frame.
<b>noiseLPCPara[i]</b>	Noise LPC parameter $i$ in the current frame (LARs for noise spectrum).
<b>prevNoiseLPCPara[i]</b>	Noise LPC parameter $i$ in the previous frame (LARs for noise spectrum).
<b>noiseAmpl</b>	Noise amplitude.
<b>noisePwr</b>	Noise power.
<b>noiseT_max</b>	Noise envelope parameter: time of maximum.
<b>noiseR_atk</b>	Noise envelope parameter: attack rate.
<b>noiseR_dec</b>	Noise envelope parameter: decay rate.

#### 7.5.1.3.3 Decoding process

##### 7.5.1.3.3.1 Basic decoder

If the `noiseFlag` is set and thus `NOISEbasicPara()` data is available in the current frame, the parameters of the noise signal component are decoded and dequantized as follows:

The noise is represented by its power and a set of LPC-Parameters.

First the `noiseNumPara` LAR parameters are reconstructed. Prediction from the previous frame is used when `noiseContFlag` is set.

```
float noiseLPCMean[25] = { 2.0, -0.75, 0.0, 0.0, 0.0, ... , 0.0 };
```

```

for (i=0; i<numNoisePara; i++) {
    if ( i<prevNumNoisePara && noiseContFlag )
        pred = noiseLPCMean[i] + (prevNoiseLPCPara[i]-noiseLPCMean[i])*0.75;
    else
        pred = noiseLPCMean[i];
    noiseLPCPara[i] = pred + noiseLAR[i];
}

```

Parameters needed in the following frame are stored in the frame-to-frame memory:

```

prevNumNoisePara = numNoisePara;
for (i=0; i<numNoisePara; i++) {
    prevNoiseLPCPara[i] = noiseLPCPara[i];
}

```

The amplitude and power of the noise is dequantized as follows:

```

noiseAmpl = 32768 * pow(10, -1.5*(noiseAmplIndex+0.5)/20 );
noisePwr = noiseAmpl*noiseAmpl;

```

If noiseEnvFlag == 1 then the noise envelope parameters noiseEnvTmax, noiseEnvRatk, and noiseEnvRdec are dequantized into noiseT\_max, noiseR\_atk, and noiseR\_dec in the same way as described for the individual line decoder (see Subclause 7.5.1.2.3.1).

#### 7.5.1.3.3.2 Enhanced decoder

Since there is no enhancement data for noise components, there is no specific enhanced decoding mode for noise parameters. If noise is to be synthesized with enhancement data present for the other components, the basic noise parameter decoder can be used. However it has to be noted that if the HILN decoder is used as a core in a scalable coder no noise signal must be synthesized for the signal which is given to the enhancement decoder.

### 7.5.1.4 Harmonic and individual line synthesizer

#### 7.5.1.4.1 Tool description

This tool synthesizes the audio signal according to the harmonic and individual line parameters decoded by the corresponding decoder tools. It includes the combination of the harmonic and individual lines, the basic synthesizer and the enhanced synthesizer. To obtain the complete decoded audio signal, the output signal of this tool is added to the output signal of the noise synthesizer as described in Subclause 7.5.1.5.

#### 7.5.1.4.2 Definitions

<b>totalNumLine</b>	Total number of lines in the current frame to be synthesized (individual plus harmonic).
<b>sampleRate</b>	Sampling rate in Hz as indicated by HILNsampleRateCode (see Table 7.7).
<b>synthSampleRate</b>	Sampling rate in Hz of synthesized output signal $x[n]$ .
<b>speedFactor</b>	Synthesis speed change factor (>1 for faster than original playback speed).
<b>pitchFactor</b>	Synthesis pitch change factor (>1 for higher than original playback pitch).
<b>T</b>	Synthesis frame length in seconds.
<b>N</b>	Synthesis frame length in samples.
<b>env(t)</b>	Amplitude envelope function in the current frame.
<b>a(t)</b>	Instantaneous amplitude of the line being synthesized.



<b>p(t)</b>	Instantaneous phase of the line being synthesized.
<b>x(t)</b>	Synthesized output signal.
<b>x[n]</b>	Sampled synthesized output signal.
<b>startPhi[i]</b>	Start phase of the i-th line in the current frame (in rad).
<b>phi[i]</b>	End phase of the i-th line in the current frame (in rad).
<b>previousEnvFlag</b>	Envelope flag in the previous frame.
<b>previousT_max</b>	Envelope parameter t_max in the previous frame.
<b>previousR_atk</b>	Envelope parameter r_atk in the previous frame.
<b>previousR_dec</b>	Envelope parameter r_dec in the previous frame.
<b>previousEnv(t)</b>	Amplitude envelope function in the previous frame.
<b>previousTotalNumLine</b>	Total number of lines in the previous frame.
<b>previousAmpl[k]</b>	Amplitude of the k-th line in the previous frame.
<b>previousFreq[k]</b>	Frequency of the k-th line in the previous frame (in Hz).
<b>previousPhi[k]</b>	End phase of the k-th line in the previous frame (in rad).
<b>previousLineEnvFlag[k]</b>	A flag indicating that the previous amplitude envelope is applied to the k-th line in the previous frame.
<b>previousT_maxEnh</b>	Enhanced envelope parameter t_max in the previous frame.
<b>previousR_atkEnh</b>	Enhanced envelope parameter r_atk in the previous frame.
<b>previousR_decEnh</b>	Enhanced envelope parameter r_dec in the previous frame.
<b>previousAmplEnh[k]</b>	Enhanced amplitude of the k-th line in the previous frame.
<b>previousFreqEnh[k]</b>	Enhanced frequency of the k-th line in the previous frame (in Hz).
<b>previousPhaseEnh[k]</b>	Phase of the k-th line in the previous frame (in rad).

### 7.5.1.4.3 Synthesis process

#### 7.5.1.4.3.1 Combination of harmonic and individual lines

For the synthesis of the harmonic lines the same synthesis technique as for the individual lines is used.

If no harmonic component is decoded for the following steps numHarmLine has to be set to zero.

Otherwise the parameters of the harmonic lines are appended to the list of individual line parameters as decoded by the individual line decoder:

```
for (i=0; i<numHarmLine; i++) {
    freq[numLine+i] = hLineFreq[i];
    ampl[numLine+i] = hLineAmpl[i];
    lineContFlag[numLine+i] = harmContFlag;
    linePred[numLine+i] = prevNumLine+i;
    lineEnvFlag[numLine+i] = harmEnvFlag;
    startPhase[numLine+i] = hStartPhase[i];
}
```

```

    startPhaseAvail[numLine+i] = hStartPhaseAvail[i];
}

```

Thus the total number of line parameters passed to the harmonic and individual line synthesizer is:

```
totalNumLine = numLine + numHarmLine;
```

Depending on the value of HILNcontMode it is possible to connect lines in adjacent frames in order to avoid phase discontinuities in the case of transitions to and from harmonic lines (HILNcontMode == 0) or additionally from individual lines to individual lines for which the continue bit lineContFlag in the bitstream was not set by the encoder (HILNcontMode == 1). This additional line continuation as described below can also be completely disabled (HILNcontMode == 2).

For each line  $i = 0 \dots \text{totalNumLine}-1$  of the current frame that has no predecessor (i.e.  $\text{lineContFlag}[i] == 0$ ), the best-fitting line  $j$  of the previous frame having no successor and with the combination meeting the requirements specified by HILNcontMode as described above is determined by maximizing the following measure  $q$ :

```

df = freq[i] / previousFreq[j];
df = max(df, 1/df);
da = ampl[i] / previousAmpl[j];
da = max(da, 1/da);
q = (1 - (df-1)/(dfCont-1)) * (1 - (da-1)/(daCont-1));

```

where  $\text{dfCont} = 1.05$  and  $\text{daCont} = 4$  are the maximum relative frequency and amplitude changes permitted. For additional line continuations determined in this way, the line predecessor information is updated:

```

lineContFlag[i] = 1;
linePred[i] = j;

```

If there is not at least one possible predecessor with  $\text{df} < \text{dfCont}$  and  $\text{da} < \text{daCont}$ ,  $\text{lineContFlag}[i]$  and  $\text{linePred}[i]$  remain unchanged.

For the enhanced synthesizer, the enhanced harmonic (up to maximum of 10) and individual line parameters are combined as follows:

```

for (i=0; i<min(10,numHarmLine); i++) {
    freqEnh[numLine+i] = hLineFreqEnh[i];
    amplEnh[numLine+i] = hLineAmplEnh[i];
    lineContFlag[numLine+i] = harmContFlag;
    linePred[numLine+i] = prevNumLine+i;
    lineEnvFlag[numLine+i] = harmEnvFlag;
    phaseEnh[numLine+i] = hLinePhaseEnh[i];
}

```

Thus the total number of line parameters passed to the enhanced harmonic and individual line synthesizer, if the HILN decoder is used as a core in a scalable coder, is:

```
totalNumLine = numLine + min(10,numHarmLine);
```

Since phase information is available for all of these lines, no line continuation is introduced for the enhanced synthesizer.

#### 7.5.1.4.3.2 Speed and pitch change

Due to the parametric signal representation utilized by the HILN coder and the phase continuation provided by the basic line synthesizer, the playback speed and pitch can be easily modified during the signal synthesis in the decoder. If playback at the original speed and pitch is desired, the corresponding control factors are set to their default values:

```

speedFactor = 1.0;
pitchFactor = 1.0;

```

If the speed is controlled by the time scaling factor in the **speed** field of the AudioSource BIFS node, the speed change factor is:

```
speedFactor = 1 / speed;
```

If the pitch is controlled by the **pitch** field of the AudioSource BIFS node, the pitch change factor is:

```
pitchFactor = pitch;
```

When the enhanced synthesizer is used instead of the basic synthesiser, speedFactor and pitchFactor should always be set to their default value 1.0.

Speed change is realized by modifying the synthesis frame length according to the desired speedFactor as described in Subclause 7.5.1.4.3.3.

Pitch change is realized by modifying the frequency parameters of the harmonic and individual lines as follows:

```
for (i=0; i<totalNumLine; i++) {
    freq[i] *= pitchFactor;
}
```

The noise synthesizer described in Subclause 7.5.1.4.3.3 also supports speed and pitch change as detailed there.

#### 7.5.1.4.3.3 Synthesis framing

The harmonic and individual line synthesizer reconstructs one frame of the audio signal. Since the line parameters encoded in a bitstream frame are valid for the center of the corresponding frame of the audio signal, the synthesizer generates the one-frame long section of the audio signal  $x(t)$  that starts at the center of the previous frame ( $t = 0$ ) and that ends at the center of the current frame ( $t = T$ ).

As default, the HILN synthesizer operates at the sampling frequency synthSampleRate as indicated by the samplingFrequency conveyed in the AudioSpecificInfo() (see Subpart 1):

```
synthSampleRate = samplingFrequency
```

The synthesis frame contains  $N$  samples:

```
N = (int)(HILNframeLength * synthSampleRate / sampleRate / speedFactor + 0.5);
```

Thus the duration  $T$  of the synthesis frame is:

```
T = N / synthSampleRate;
```

In the following, the calculation of the synthesized output signal

```
x(t)
```

for  $0 \leq t < T$  is described. The time discrete version (i.e. the actual frame of output samples) is defined as

```
x[n] = x(t) with t = (n+0.5)*(T/N)
```

for  $0 \leq n < N$ .

The noise synthesizer described in Subclause 7.5.1.5 uses the same synthesis framing as the harmonic and individual line synthesizer described here.

7.5.1.4.3.4 Basic synthesizer

Some parameters of the previous frame (names starting with “previous”) are taken out of the frame-to-frame memory which has to be reset before decoding the first frame of a bitstream.

First the envelope functions  $previousEnv(t)$  and  $env(t)$  of the previous and current frame are calculated according to the following rules:

If  $envFlag == 1$  then the envelope function  $env(t)$  is derived from the envelope parameters  $t\_max$ ,  $r\_atk$ , and  $r\_dec$ . With  $T$  being the frame length,  $env(t)$  is calculated for  $-T/2 \leq t < 3/2 \cdot T$ :

```
if (-1/2 <= t/T && t/T < t_max)
    env(t) = max(0, 1-(t_max-t/T)*r_atk);
if (t_max <= t/T && t/T < 3/2)
    env(t) = max(0, 1-(t/T-t_max)*r_dec);
```

If  $envFlag == 0$  then a constant envelope function  $env(t)$  is used:

```
env(t) = 1;
```

Accordingly  $previousEnv(t)$  is calculated from the parameters  $previousT\_max$ ,  $previousR\_atk$ ,  $previousR\_dec$  and  $previousEnvFlag$ .

The envelope parameters transmitted in case of  $envFlag == 1$  are valid for the harmonic lines as well as for the individual lines. Thus the envelope functions always must be generated, even if all  $lineEnvFlag[i] == 0$ .

Before the synthesis is performed, the accumulator  $x(t)$  for the synthesized audio signal is cleared for  $0 \leq t < T$ :

```
x(t) = 0;
```

The lines  $i$  continuing from the previous frame to the current frame

```
all i=0 .. totalNumLine-1 that have lineContFlag[i] == 1
```

are synthesized as follows for  $0 \leq t < T$ :

```
k = linePred[i];
ap(t) = previousAmpl[k];
if (previousLineEnvFlag[k] == 1)
    ap(t) *= previousEnv(t+T/2);
ac(t) = ampl[k];
if (lineEnvFlag[i] == 1)
    ac(t) *= env(t-T/2);
short_x_fade = (previousLineEnvFlag[k] && !(previousR_atk < 5 &&
    (previousT_max > 0.5 || previousR_dec < 5))) ||
    (lineEnvFlag[i] && !(r_dec < 5 && (t_max < 0.5 || r_atk < 5)));
if (short_x_fade == 1) {
    if (0 <=t && t < 7/16*T)
        a(t) = ap(t);
    if (7/16*T <=t && t < 9/16*T)
        a(t) = ap(t) + (ac(t)-ap(t))*(t/T-7/16)*8;
    if (9/16*T <= t&& t < T)
        a(t) = ac(t);
}
else
    a(t) = ap(t) + (ac(t)-ap(t))*t/T;
p(t) = previousPhi[k]+2*pi*previousFreq[k]*t+
    2*pi*(freq[i]-previousFreq[k])/(2*T)*t*t;
x(t) += a(t)*sin(p(t));
phi[i] = p(T)
```

The lines  $i$  starting in the current frame

```
all i=0 .. totalNumLine-1 that have lineContFlag[i] == 0
```

are synthesized as follows for  $0 \leq t < T$ :

```

if (lineEnvFlag[i] && !(r_dec < 5 && (t_max < 0.5 || r_atk < 5))) {
  if (0 <= t && t < 7/16*T)
    fade_in(t) = 0;
  if (7/16*T <= t && t < 9/16*T)
    fade_in(t) = 0.5 - 0.5*cos((8*t/T-7/2)*pi);
  if (9/16*T <= t && t < T)
    fade_in(t) = 1;
}
else
  fade_in(t) = 0.5-0.5*cos(t/T*pi);
a(t) = fade_in(t)*ampl[i];
if (lineEnvFlag[i] == 1)
  a(t) *= env(t-T/2);
if (startPhaseAvail[i])
  startPhi[i] = startPhase[i];
else
  startPhi[i] = random(2*pi);
p(t) = startPhi[i] + 2*pi*freq[i]*(t-T);
x(t) += a(t)*sin(p(t));
phi[i] = p(T)

```

random(x) is a function returning a random number with uniform distribution in the interval

$$0 \leq \text{random}(x) < x$$

The lines k ending in the previous frame

all  $k=0 \dots \text{previousTotalNumLine}-1$  that have  $\text{prevLineContFlag}[k] == 0$

are synthesized as follows for  $0 \leq t < T$ :

```

if (previousLineEnvFlag[k] && !(previousR_atk < 5 &&
  (previousT_max > 0.5 || previousR_dec < 5))) {
  if (0 <= t && t < 7/16*T)
    fade_out(t) = 1;
  if (7/16*T <= t && t < 9/16*T)
    fade_out(t) = 0.5 + 0.5*cos((8*t/T-7/2)*pi);
  if (9/16*T <= t && t < T)
    fade_out(t) = 0;
}
else
  fade_out(t) = 0.5+0.5*cos(t/T*pi);
a(t) = fade_out(t)*previousAmpl[k];
if (previousLineEnvFlag[k] == 1)
  a(t) *= previousEnv(t+T/2);
p(t) = previousPhi[k]+2*pi*previousFreq[k]*t;
x(t) += a(t)*sin(p(t));

```

In order to avoid aliasing distortion, synthesized lines are muted (i.e.  $a(t) = 0$ ) while their instantaneous frequency is above or equal half the sampling frequency, i.e.

$$d \text{ phi}(t) / dt \geq \pi * N / T$$

Parameters needed in the following frame are stored in the frame-to-frame memory:

```

previousEnvFlag = envFlag;
previousT_max = t_max;
previousR_atk = r_atk;
previousR_dec = r_dec;
previousTotalNumLine = totalNumLine;
for (i=0; i<totalNumLine; i++) {
  previousFreq[i] = freq[i];
  previousAmpl[i] = ampl[i];
  previousPhi[i] = fmod(p[i], 2*pi);
  previousLineEnvFlag[i] = lineEnvFlag[i];
}

```

}

fmod(x,2\*pi) is a function returning the 2\*pi modulus of x.

#### 7.5.1.4.3.5 Enhanced synthesizer

The enhanced synthesizer is based on the basic synthesizer but evaluates also the line phases for reconstructing one frame of the audio signal. Since the line parameters encoded in a bitstream frame and the corresponding enhancement frame are valid for the middle of the corresponding frame of the audio signal, the harmonic and individual line synthesizer generates the one frame long section of the audio signal that starts in the middle of the previous frame and ends in the middle of the current frame.

Some parameters of the previous frame (names starting with "previous") are taken out of the frame-to-frame memory which has to be reset before decoding the first frame of a bitstream.

First the envelope functions previousEnv(t) and env(t) of the previous and current frame are calculated according to the following rules:

If envFlag == 1 then the envelope function env(t) is derived from the envelope parameters t\_maxEnh, r\_atkEnh, and r\_decEnh. With T being the frame length, env(t) is calculated for  $-T/2 \leq t < 3/2 * T$ :

```
if (-1/2 <= t/T && t/T < t_maxEnh)
    env(t) = max(0, 1-(t_maxEnh-t/T)*r_atkEnh);
if (t_maxEnh <= t/T && t/T < 3/2)
    env(t) = max(0, 1-(t/T-t_maxEnh)*r_decEnh);
```

If envFlag == 0 then a constant envelope function env(t) is used:

```
env(t) = 1;
```

Accordingly previousEnv(t) is calculated from the parameters previousT\_maxEnh, previousR\_atkEnh, previousR\_decEnh and previousEnvFlag.

The envelope parameters transmitted in case of envFlag == 1 are valid for the harmonic lines as well as for the individual lines. Thus the envelope functions always must be generated, even if all lineEnvFlag[i] == 0.

Before the synthesis is performed, the accumulator x(t) for the synthesized audio signal is cleared for  $0 \leq t < T$ :

```
x(t) = 0
```

All lines i in the in the current frame

```
all i =0 .. totalNumLine-1
```

are synthesized as follows for  $0 \leq t < T$ :

```
if (envFlag && !(r_decEnh < 5 && (t_maxEnh < 0.5 || r_atkEnh < 5))) {
    if (0 <= t&& t < 7/16*T)
        fade_in(t) = 0;
    if (7/16*T <= t&& t < 9/16*T)
        fade_in(t) = 0.5 - 0.5*cos((8*t/T-7/2)*pi);
    if (9/16*T <= t&& t < T)
        fade_in(t) = 1;
}
else
    fade_in(t) = 0.5-0.5*cos(t/T*pi);
a(t) = fade_in(t)*amplEnh[i];
if (envFlag[i] == 1)
    a(t) *= env(t-T/2);
phi(t) = 2*pi*freqEnh[i]*(t-T)+phaseEnh[i];
x(t) += a(t)*sin(phi(t));
```

The lines  $k$  in the previous frame

```
all k=0 .. previousTotalNumLine-1
```

are synthesized as follows for  $0 \leq t < T$ :

```
if (previousEnvFlag && !(previousR_atkEnh < 5 &&
    (previousT_maxEnh > 0.5 || previousR_decEnh < 5))) {
    if (0 <= t && t < 7/16*T)
        fade_out(t) = 1;
    if (7/16*T <= t && t < 9/16*T)
        fade_out(t) = 0.5 + 0.5*cos((8*t/T-7/2)*pi);
    if (9/16*T <= t && t < T)
        fade_out(t) = 0;
}
else
    fade_out(t) = 0.5+0.5*cos(t/T*pi);
a(t) = fade_out(t)*previousAmplEnh[k];
if (previousEnvFlag[k] == 1)
    a(t) *= previousEnv(t+T/2);
phi(t) = 2*pi*previousFreqEnh[k]*t+previousPhaseEnh[i];
x(t) += a(t)*sin(phi(t));
```

If the instantaneous frequency of a line is above or equal half the sampling frequency, i.e.

$$d \phi(t) / dt \geq \pi * N / T$$

it is not synthesized to avoid aliasing distortion.

Parameters needed in the following frame are stored in the frame-to-frame memory:

```
previousEnvFlag = envFlag;
previousT_maxEnh = t_maxEnh;
previousR_atkEnh = r_atkEnh;
previousR_decEnh = r_decEnh;
previousTotalNumLine = totalNumLine;
for (i=0; i<totalNumLine; i++) {
    previousFreqEnh[i] = freqEnh[i];
    previousAmplEnh [i] = amplEnh[i];
    previousPhaseEnh [i] = phaseEnh [i];
}
```

### 7.5.1.5 Noise synthesizer

#### 7.5.1.5.1 Tool description

This tool synthesizes the noise part of the output signal based on the noise parameters decoded by the noise decoder. Finally the noise signal is added to the output signal of the harmonic and individual line synthesizer (Subclause 7.5.1.4) to obtain the complete decoded audio signal

#### 7.5.1.5.2 Definitions

<b>noiseWin[n]</b>	Window for noise overlap-add.
<b>noiseEnv[n]</b>	Envelope for noise component in the current frame.
<b>M</b>	Frame length in samples before resampling.
<b>w[m]</b>	White noise with power $p_w$ .
<b>xf[m]</b>	Filtered noise signal in the current frame.
<b>xn[n]</b>	Synthesized noise signal in the current frame.

<b>previousXn[n]</b>	Synthesized noise signal in the previous frame.
<b>previousNoiseWin[n]</b>	Window and envelope for noise component in the previous frame.
<b>r[i]</b>	LPC reflection coefficients.
<b>h[i]</b>	LPC impulse response.
<b>hlp[i]</b>	Low-pass resampling filter impulse response.

### 7.5.1.5.3 Synthesis process

#### 7.5.1.5.3.1 Basic synthesizer

If noise parameters are transmitted for the current frame, a noise signal with a spectral shape as described by the noise parameters decoded from the bitstream is synthesized and added to the audio signal generated by the harmonic and individual line synthesizer. The synthesis framing is as described in Subclause 7.5.1.4.3.3.

The noise is represented by its power and a set of LPC-Parameters. As described in the harmonic tone decoder (Subclause 7.5.1.1.3.1), the noise LPC parameters are converted to the reflection coefficients  $r[i]$  and to the time response  $h[i]$ :

```
for ( i = 0; i < numNoisePara; i++ )
    r[i] = ( exp(noiseLPCPara[i]) - 1 ) / ( exp(noiseLPCPara[i]) + 1 );
```

After this the reflection coefficients  $r[i]$  are converted to the time response  $h[i]$  using the C function

```
void Convert_k_to_h (float *x, int N)
```

given in Subclause 7.5.1.1.3.1 (call with  $x[i]=r[i]$  and  $N=numNoisePara$ ; returns with  $x[i]=h[i]$ ).

Now the filtered noise signal  $xf[m]$  is generated by applying the LPC synthesis IIR filter to a white noise represented by random numbers  $w[m]$ . The power of this zero-mean white noise is denoted  $pw$ . For a noise with uniform distribution in  $[-1,1]$  the power is

```
pw = 1/3
```

To achieve the required noise signal power, the following scaling factor  $s$  is required:

```
ss = 1.0;
for (i=0; i<numNoisePara; i++)
    ss *= 1-r[i]*r[i];
s = noiseAmpl * sqrt( ss/pw );
```

Then the white noise  $w[m]$  is IIR-filtered to obtain the synthesized noise signal  $xf[m]$

```
for (m=startup; m<2*M; m++) {
    xf[m] = s * w[m];
    for (i=0; i<min(m-startup,numNoisePara); i++)
        xf[m] += h[i]*xf[m-i-1];
}
```

To ensure that the IIR filter can reach a sufficiently steady state, a startup phase is used:

```
startup = -numNoisePara
```

If decoded with a pitch change (i.e.  $pitchFactor \neq 1.0$ , see Subclause 7.5.1.4.3.2) or with a different sample rate than the encoder (i.e.  $synthSampleRate \neq sampleRate$ ), a resample operation must be applied to the signal  $xf[m]$  using the resampling factor

```
resampleFactor = ( sampleRate * pitchFactor ) / synthSampleRate;
```



where e.g. pitchFactor of 2 indicates that this signal is synthesized at twice its original pitch. Otherwise the resampleFactor is set to 1.0. Based on the resampleFactor, the frame length M before resampling is defined:

```
M = N * resampleFactor;
```

The resampling can be realized by applying two lowpass FIR filter operations to the signal  $x_f[m]$  and linearly interpolating between these two values to obtain the final noise signal  $x_n[n]$ .

```
if ( resampleFactor<1 )
    fc = 1;
else
    fc = 1/resampleFactor;
```

The following function calculates the time response  $hlp[0..31]$  of an appropriate lowpass FIR filter with 16 taps and an oversampling factor of 4. The cutoff frequency is  $fc$ .

```
void GenLPFilter (float *hlp, double fc)
{
    double x,f;
    int i;

    hlp[0] = (float) fc;
    for (i=1; i<32; i++)
    {
        x = i*pi/4.0;
        hlp[i] = (float) ((0.54+0.46*cos(0.125*x))*sin(fc*x)/x);
    }
}
```

To perform the FIR filter operation the following C function can be used. The parameters are the signal, the time response (as returned by the function above) and the position of the sampling point. The position is given as the difference between the nearest sample position prior to the desired sample position ( $x[7]$ ) and the desired sample position. Therefore  $0 \leq \text{pos} < 1$ . The interpolation is done between  $x[7]$  and  $x[8]$ , the returned value represents a sample position of  $7+\text{pos}$ .

```
float LPInterpolate (float *x, float *hlp, double pos)
{
    long j;
    double s,t;

    pos *= 4.0;
    j = (long) pos;
    pos -= (double) j;

    s = t = 0.0;
    j = 32-j;

    if ( j==32 )
    {
        t = h[31]*(*x);
        x++;
        j -= 4;
    }
    while (j>0)
    {
        s += h[j ]*(*x);
        t += h[j-1]*(*x);
        x++;
        j -= 4;
    }
    j = -j;
    while (j<32-1)
    {
        s += h[j ]*(*x);
        t += h[j+1]*(*x);
        x++;
        j += 4;
    }
}
```

```

    }
    if (j<32)
        s += h[j]*(*x);

    return (float) (s+pos*(t-s));
}

```

Using the functions GenLPFilter() and LPInterpolate(), the resampling is done as described below.  $x_f[m]$  is set to 0.0 for  $m < \text{startup}$  and  $m \geq 2*M$ .

```

GenLPFilter(hlp, fc);
for (n=0; n<2*N; n++)
    xn[n] = LPInterpolate(xf+((int)n*resampleFactor)-7, hlp, frac(n*resampleFactor));

```

If  $\text{resampleFactor} == 1.0$ ,  $x_f[m]$  is simply copied to  $x_n[n]$  without resampling:

```

for (n=0; n<2*N; n++)
    xn[n] = xf[n];

```

For smooth cross-fade of the noise signal at the boundary between two adjacent frames, the following window is used for this overlap-add operation:

```

for (n=0; n<N; n++) {
    if (n < N*3/8)
        noiseWin[n] = 0;
    if (N*3/8 <= n && n < N*5/8)
        noiseWin[n] = sin(pi/2 * (n-N*3/8+0.5)/(N*2/8));
    if (N*5/8 <= n)
        noiseWin[n] = 1;
    noiseWin[2*N-1-n] = noiseWin[n];
}

```

Now the envelope function  $\text{noiseEnv}[n]$  is calculated. If  $\text{noiseEnvFlag} == 1$  then the envelope function

```

noiseEnv[n] = noiseEnv(t) with  $t = (n+0.5)*(T/N)-0.5$ 

```

is derived from the envelope parameters  $\text{noiseT\_max}$ ,  $\text{noiseR\_atk}$ , and  $\text{noiseR\_dec}$  for  $-T/2 \leq t < 3/2*T$  (i.e.  $0 \leq n < 2*N$ ):

```

if (-1/2 <= t/T && t/T < noiseT_max)
    noiseEnv(t) = max(0,1-(noiseT_max-t/T)*noiseR_atk);
if (noiseT_max <= t/T && t/T < 3/2)
    noiseEnv(t) = max(0,1-(t/T-noiseT_max)*noiseR_dec);

```

If  $\text{noiseEnvFlag} == 0$  then a constant envelope function  $\text{noiseEnv}(t)$  is used:

```

noiseEnv[n] = 1;

```

The noise signal  $x_n[n]$  is windowed for overlap-add and multiplied with the envelope  $\text{noiseEnv}[n]$ . Then this signal and the noise from the previous frame  $\text{previousXn}[n]$  are added to the signal  $x[n]$  from the harmonic and individual line synthesizer to construct the complete synthesized signal  $\text{signalx}[n]$ :

```

for (n=0; n<N; n++)
    x[n] += xn[n]*noiseWin[n]*noiseEnv[n] + previousXn[n];

```

The second half of the generated noise signal  $x_n[n]$  is stored in the frame-to-frame memory  $\text{previousXn}[n]$  for overlap-add:

```

for (n=0; n<N; n++)
    previousXn[n] = xn[N+n]*noiseWin[N+n]*noiseEnv[N+n];

```

The  $\text{previousXn}[n]$  memory has to be reset to 0.0 before decoding of the first frame.

### 7.5.1.5.3.2 Enhanced synthesizer

Since there is no enhancement data for noise components, there is no specific enhanced synthesizer mode for noise components. If noise is to be synthesized with enhancement data present for the other components, the basic noise synthesizer decoder can be used. However it has to be noted that if the HILN decoder is used as a core in a scalable coder no noise signal must be synthesized for the signal which is given to the enhancement decoder.

## 7.5.2 Integrated parametric coder

The integrated parametric coder can operate in four different modes as shown in Table 7.1. PARAModes 0 and 1 represent the fixed HVXC and HILN modes. PARAMode 2 permits automatic switching between HVXC and HILN depending on the current input signal type. In PARAMode 3 the HVXC and HILN coders can be used simultaneously and their output signals are added (mixed) in the decoder.

The integrated parametric coder uses a frame length of 40 ms and a sampling rate of 8 kHz and can operate at 2025 bit/s or any higher bitrate. Operation at 4 kbit/s or higher is suggested.

### 7.5.2.1 Integrated parametric decoder

For the “HVXC only” and “HILN only” modes the parametric decoder is not modified.

In “switched HVXC / HILN” and “mixed HVXC / HILN” modes both HVXC and HILN decoder tools are operated alternatively or simultaneously according to the PARAswitchMode or PARAMixMode of the current frame. To obtain proper time alignment of both HVXC and HILN decoder output signals before they are added, the difference between HVXC and HILN decoder delay has to be compensated with a FIFO buffer:

- If HVXC is used in the low delay decoder mode, its output must be delayed for 100 samples (i.e. 12.5 ms).
- If HVXC is used in the normal delay decoder mode, its output must be delayed for 80 samples (i.e. 10 ms).

To avoid hard transitions at frame boundaries when the HVXC or HILN decoders are switched on or off, the respective decoder output signals are faded in and out smoothly. For the HVXC decoder a 20 ms linear fade is applied when it is switched on or off. The HILN decoder requires no additional fading because of the smooth synthesis windows utilized in the HILN synthesizer. It is only necessary to operate the HILN decoder with no new components for the current frame (i.e. force numLine = 0, harmFlag = 0, noiseFlag = 0) if the current bitstream frame contains no “HILNframe()”.

## 7.6 Error resilient bitstream payloads

### 7.6.1 Overview of the tools

Error resilient bitstream payloads allow the effective usage of advanced channel coding techniques like unequal error protection (UEP), which can be perfectly adapted to the needs of the different coding tools. The basic idea is to rearrange the conventional bitstream payload depending on its error sensitivity in one or more instances belonging to different error sensitivity categories (ESC). This re-arrangement works either data element-wise or even bit-wise. The error resilient bitstream payload is built by concatenating these instances.

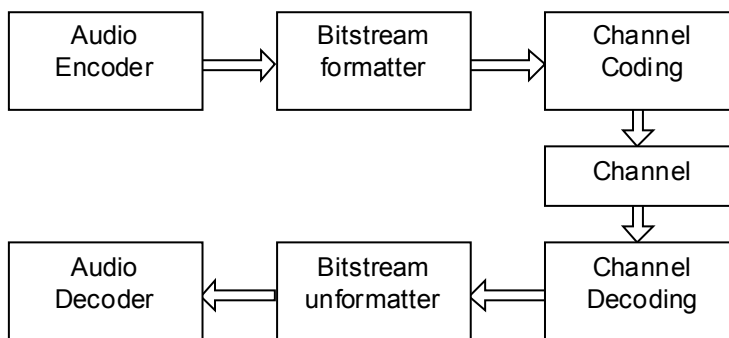


Figure 7.2 - Basic principle of error resilient bitstream reordering

The basic principle is depicted in Figure 7.2. A bitstream is reordered according to the error sensitivity of single bitstream elements or even single bits. This new arranged bitstream is channel coded, transmitted and channel decoded. Prior decoding, the bitstream is rearranged to its original order. Instead of performing the reordering in the described way, the reordered syntax, that is the bitstream order prior the bitstream formatter at the decoder site, is defined in this amendment.

In the subsequent subparts, a detailed description of error resilient bitstream reordering for these tools can be found.

### 7.6.2 ER HILN

Error sensitivity categories (ESC) are defined in the parametric bitstream as described in Subclause 7.3. Below, the ordering of the different ESCx is described for the four different modes PARAMode == 0, 1, 2, 3.

- **PARAMode == 0 (HVXC only)**  
HVXC: ESC0 ESC1 ESC2 ESC3 ESC4
- **PARAMode == 1 (HILN only)**  
PARA/HILN: ESC0 ESC1 ESC2 ESC3 ESC4
- **PARAMode == 2 (switched HVXC / HILN)**  
PARA/HILN: ESC0 [ ESC1 ESC2 ESC3 ESC4 ]  
HVXC 1/double: [ ESC0 ESC1 ESC2 ESC3 ESC4 ]  
HVXC 2/double: [ ESC0 ESC1 ESC2 ESC3 ESC4 ]
- **PARAMode == 3 (mixed HVXC / HILN)**  
PARA/HILN: ESC0 [ ESC1 ESC2 ESC3 ESC4 ]  
HVXC 1/double: [ ESC0 ESC1 ESC2 ESC3 ESC4 ]  
HVXC 2/double: [ ESC0 ESC1 ESC2 ESC3 ESC4 ]

In PARAMode 3 for example, the ordering is PARA/HILN ESC0, ESC1, ..., ESC4, followed by HVXC 1/double ESC0, ESC1, ..., ESC4 and HVXC 2/double ESC0, ESC1, ..., ESC4.

The ESC0 of "PARA/HILN" consists of the PARASwitchMode or PARAMixMode bitstream element in PARAFrame() followed by the bitstream elements in HILNbasicFrameESC0(). The actual presence of these bitstream elements can depend on the current values of PARAMode, PARASwitchMode, and PARAMixMode. "HVXC 1/double" and "HVXC 2/double" denote the first and second ErHVXCfixframe() within an ErHVXCdoubleframe() respectively. The presence of ESCs in squared brackets depends on the value of PARASwitchMode or PARAMixMode in the current frame.

For the HILN enhancement and extension layer Elementary Streams, no error sensitivity categories are defined and all bitstream elements of HILNenexFrame() are handled as "ESC0".

## Annex 7.A (informative)

### Parametric audio encoder

#### 7.A.1 Overview of the encoder tools

The following Figure 7.A.1 shows a general block diagram of a parametric encoder. First the input signal is separated into the two parts which are coded by HVXC and by HILN tools. This can be done manually or automatically. Automatic switching between speech and music signals is supported (see Subclause 7.A.3.1), allowing the use of HVXC for speech and HILN for music. For both HVXC and HILN parameter estimation and parameter encoding can be performed. A common bitstream formatter allows operation either in HVXC only, HILN only, or also in combined modes, i.e. switched or mixed mode.

The following description of an HILN parametric encoder is informative and also alternative techniques for signal separation and parameter estimation can be used in an encoder.

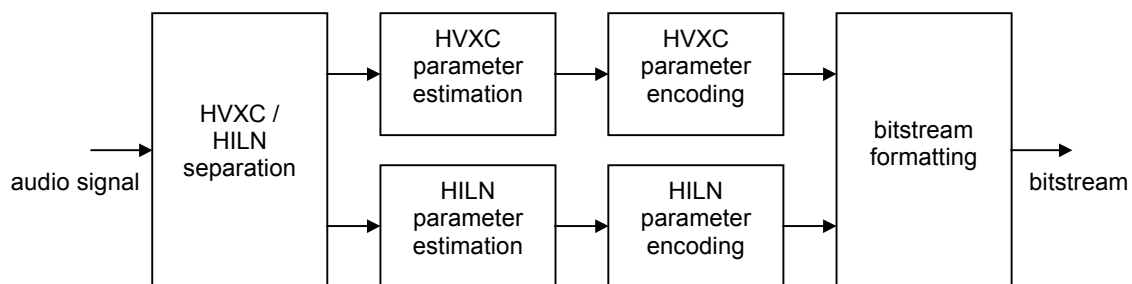


Figure 7.A.1 - General block diagram of the integrated parametric encoder

#### 7.A.2 HILN encoder tools

The basic principle of the “Harmonic and Individual Lines plus Noise” (HILN) encoder is to analyze the input signal in order to extract parameters describing the signal. These parameters are coded and transmitted as a bitstream. In the decoder the output signal is synthesized based on the parameters extracted and transmitted by the encoder.

The encoder consists of two main parts: “Parameter Extraction” and “Parameter Coding”. In the encoder, the input signal is divided into consecutive frames and for each frame a set of parameters describing the signal in this frame is extracted and coded. Due to this parametric description, a wide range of bitrates, sampling rates and frame lengths are possible. Typically a frame length of 32 ms is used. For input signals with 8 to 16 kHz sampling rate typically a bitrate of 6 to 16 kbit/s is used.

The “Parameter Extraction” and “Parameter Coding” is described in detail in the following Subclauses.

##### 7.A.2.1 HILN parameter extraction

Since different parameter sets and different synthesis techniques can be applied, the input signal of the encoder has to be split up in an appropriate way. This is performed by the *Separation* unit. Depending on the most appropriate synthesis technique, a parameter set is derived for each part of the input signal in the *Model Based Parameter Estimation* unit. The two units *Separation* and *Model Based Parameter Estimation* can be regarded as the analysis stage which produces a parametric description of the input signal. The separation of the input signal is

enhanced by feeding back the signals which are generated in the *Synthesis* unit from all the parameters of previously separated parts. The *Separation* and the *Model Based Parameter Estimation* additionally receive data from a synthesis model independent *Pre-Analysis*. Prior to transmission, the parameters are fed through the *Quantization and Coding* unit, which is controlled by a *Psychoacoustic Model*. This *Psychoacoustic Model* processes the input signal in order to derive information about the relevancy of synthesis parameters. In addition, the synthesized signal is fed into the *Psychoacoustic Model*, which thus is allowed to assist the *Model Based Parameter Estimation*.

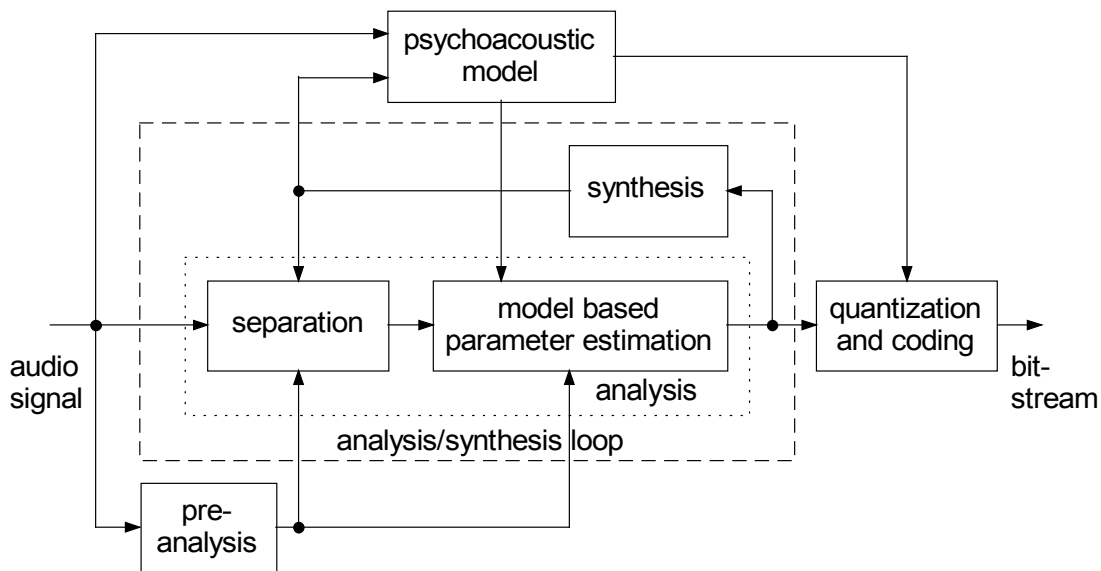


Figure 7.A.2 - Block diagram of the HILN encoder

In the parameter extraction, the input signal is separated into three different parts: “harmonic lines”, “individual lines” and “noise”.

For each of these parts parameters describing the signal are extracted. These are basically:

- harmonic lines: fundamental frequency and amplitudes of the harmonic components
- individual lines: frequency and amplitude of each individual line
- noise: spectral shape of the noise

Additionally parameters for amplitude envelopes and for continuation of spectral lines from one frame to the next can be determined.

The signal separation and parameter estimation is implemented in three steps: First the fundamental frequency of the harmonic part of the signal is estimated. Then the parameters of the relevant spectral lines are estimated and these lines are classified as “individual lines” or “harmonic lines” depending on the frequency with respect to the fundamental frequency. After all relevant spectral lines are extracted, the remaining residual signal is assumed to be noise-like and its spectral shape is described by a set of parameters.

The harmonic line extraction of the HILN tools can also be utilized in an integrated parametric coder utilizing both the HVXC speech coding tools as well as the HILN coding tools simultaneously. If the input signal is e.g. a speech signal mixed with background music, the HILN encoder can be used to extract only those individual spectral lines that do not belong to the harmonic part of the signal. These individual lines are encoded by the HILN tools and the remaining signal - consisting of the harmonic signal part and noise - then is encoded by the HVXC parametric speech codec tools. In the decoder, the audio signal is reconstructed by adding the output of the “individual line” synthesizer and the HVXC decoder.

### 7.A.2.1.1 Fundamental frequency estimation

A “cepstrum”-based fundamental frequency estimation technique is employed by the HILN tools. First the input signal is windowed with a Hanning window of twice the frame length centered around the current frame. For the windowed signal, the magnitude spectrum is calculated and the logarithm is applied to the magnitude spectrum. Then the log spectrum is multiplied with the window

$$w(f) = (1 + \cos(2\pi f/fs))/2 \quad 0 \leq f \leq fs/2$$

and zero-padding is used to virtually double the sampling frequency before the cepstrum is calculated. Finally the local maxima in the cepstrum are determined and the largest maximum within the permitted “pitch lag” search range is identified. The fundamental frequency is calculated from the “pitch lag” (period of the fundamental frequency) of the largest maximum.

The fundamental frequency determined by this cepstrum-based technique is used as an initial (coarse) estimate in the following line parameter estimation.

### 7.A.2.1.2 Harmonic and Individual Line Parameter Estimation

The estimation of harmonic and individual line parameters is based on an “Analysis/Synthesis Loop” described in the following Subclauses.

In a first step the parameters of all harmonic lines are estimated. This is done by performing the regression-based high accuracy frequency estimation described below for all integer multiples of the coarse fundamental frequency as initial estimates. Based on the accurate frequencies of the harmonic lines, a fine estimate of the fundamental frequency  $hFreq$  and the so-called “stretching”  $hStretch$  is calculated which minimizes the total error between the real harmonic line frequencies and those calculated according to

$$hLinefreq[i] = hFreq * (i+1) * (1 + hStretch*(i+1)) \quad i = 0 \dots harmNumLine-1$$

where the total number of harmonic lines is determined by the bandwidth  $w$  of the signal and the current fundamental frequency  $hFreq$ :

$$harmNumLine = \text{floor}(w/hFreq)$$

The harmonic envelope flag is set if using the current amplitude envelope for all harmonic lines results in a lower residual error than if no envelope is used. If the relative change of the fundamental frequency between the previous and the current frame is less than 15%, the harmonic continuation flag is set.

In the second step the relevant spectral lines are extracted from the input signal by means of the “Analysis/Synthesis Loop”. This loop utilizes a psychoacoustic model to extract the spectral lines in order of their subjective relevance. If the frequency of an extracted spectral line is close to the frequency of a harmonic line as calculated from  $hFreq$  and  $hStretch$ , this extracted line is classified as harmonic line. Otherwise it is classified as individual line. The “Analysis/Synthesis Loop” is terminated if the requested number of individual lines was extracted or if the remaining signal components cannot be properly modeled by spectral lines. The ratio between the number of harmonic lines extracted and the total lines extracted is passed on to the encoder as measure of “relevance” of the harmonic lines.

If less than three extracted lines were classified as “harmonic line”, these lines are added to the list of “individual lines” and  $numHarmLine$  is set to 0. Finally all harmonic lines that were not extracted by the “Analysis/Synthesis Loop” are also removed from the residual signal. This residual signal is then passed to the noise parameter estimation.

#### 7.A.2.1.2.1 Pre-Analysis

The pre-analysis module determines the signal amplitude envelope which is used in the analysis/synthesis loop.

7.A.2.1.2.2 Analysis/Synthesis Based on Single Spectral Lines

The Individual Line encoder is based on the model of single spectral lines, which can be generated with the help of sine wave generators. The according *Model Based Parameter Estimation* for the  $i$ -th line in the loop consists of the following steps:

- calculation of the deviation between FFT spectra of input and synthesized signals
- selection of the most relevant FFT line with center frequency  $f_{i,m}$
- high resolution frequency estimation in the surrounding of  $f_{i,m}$
- amplitude and phase estimation, selection of envelope information
- synthesis with the determined parameters
- calculation of the residual error signal by subtraction of the synthesized signal from the input signal

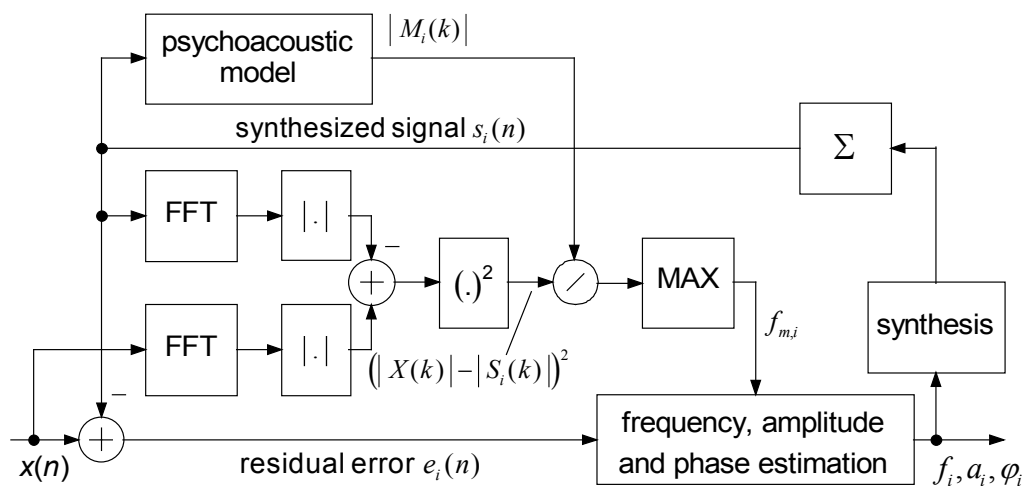


Figure 7.A.3 - Analysis/Synthesis Loop based on the synthesis method “single spectral lines”

The FFT line to be processed is determined by calculating the deviation between input spectrum and synthesized spectrum and searching the maximum ratio of the square of this deviation and the masking threshold derived from the signal synthesized from the previously determined spectral lines.

Based on the center frequency  $f_{i,m}$  of the selected FFT line a frequency estimation is performed in order to obtain a frequency parameter of higher accuracy than the FFT resolution.



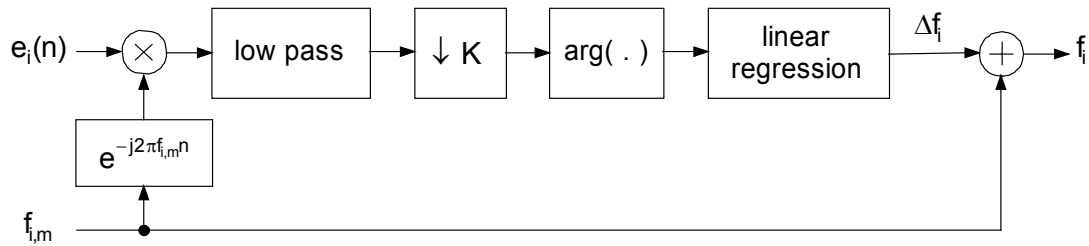


Figure 7.A.4 - High accuracy frequency estimation

For this frequency estimation, first the spectrum of the residual error signal is shifted in a way that the center frequency  $f_{i,m}$  of the selected FFT line becomes zero. The complex output values of this operation are fed into lowpass filter and sampling rate reduction, which are realized by applying time-shifted versions of the window function. The slope of the regression line for the phase values of the obtained complex samples gives a frequency offset, which is added to  $f_{i,m}$  in order to give a high resolution frequency parameter  $f_i$ . The time shift of the window function ranges from -0.32 to 0.32 times the frame length. The time shift step size is 0.08 times the frame length and thus 9 data points are used for the linear regression.

Based on  $f_i$  the amplitude and phase of the spectral line are calculated. This is realized by calculating a complex correlation coefficient of the residual error signal and a complex harmonic signal of frequency  $f_i$ . The absolute value of the correlation coefficient gives the amplitude parameter  $a_i$  and the argument gives the phase parameter  $\varphi_i$ . If the *Pre-Analysis* has generated envelope parameters, a second set of parameters  $a_{i,e}$  and  $\varphi_{i,e}$  is generated by correlation of the residual error signal and a complex harmonic signal of frequency  $f_i$  multiplied with the according envelope.

In the *Separation* unit, a new residual error signal is generated by subtracting the signal synthesized from the parameters  $f_i$ ,  $a_i$  and  $\varphi_i$ . If the parameters  $a_{i,e}$  and  $\varphi_{i,e}$  are also available, a second signal is synthesized accordingly and the parameter set leading to the lowest residual error variance is selected.

#### 7.A.2.1.2.3 Psychoacoustic model

The psychoacoustic model calculates the masked threshold for the synthesized signal components in the analysis/synthesis loop.

#### 7.A.2.1.3 Noise Parameter Estimation

The noise parameters are used to model the spectral shape of the residual signal. First the power spectrum of the Hanning-windowed residual signal is calculated. Then this spectrum is transformed back into the autocorrelation function. Based on this autocorrelation function, LPC parameters are calculated using Durbin's algorithm. These LPC parameters are then transformed into reflection coefficients which are represented as Log Area Ratios (LAR). Besides the LAR parameters also the power of the noise signal is calculated.

Additionally a new set of envelope parameters is calculated for the residual signal. Thus also the temporal shape of the residual signal can be modeled. The ratio of residual signal power to input signal power is calculated and passed to encoder as a measure of "relevance" of the noise-like signal component.

#### 7.A.2.2 HILN parameter encoder

The extracted parameters of the harmonic, individual line and noise parts of the signal are quantized and encoded to generate the bitstream output of the HILN encoder.

The allocation of the bits available in a frame to the parameters for the three parts of the signal is determined by the harmonic and noise component "relevance" measures calculated during the parameter estimation.

### 7.A.2.2.1 Harmonic Line Parameter Quantization

The number of bits available for the harmonic line parameters depends on the “relevance” measure of the harmonic signal component. If this measure is low, the number of harmonic lines encoded can be less than the number of lines extracted. This corresponds to a bandwidth limitation of the harmonic signal.

The fundamental frequency is quantized with 2048 steps on a logarithmic scale ranging from 20 Hz to 4 kHz. The “stretching” parameter is quantized with 35 steps on a uniform scale ranging from  $-17/16000$  to  $+17/16000$ .

To describe the spectrum of the harmonic tone, the autocorrelation function of the harmonic signal is calculated. From this LAR LPC coefficients are derived that approximately model the spectral envelope of the harmonic signal. This process is similar to the LPC spectral modeling used for the noise signal. Besides the LAR parameters also the power of the harmonic tone is calculated.

For a new harmonic tone, the indices of the quantised fundamental frequency and amplitude are directly written to the bitstream, while for a continued harmonic tone the index differences to the previous frame are coded with an entropy code.

### 7.A.2.2.2 Individual Line Parameter Quantization

In the *Quantization and Coding* unit, the parameters are processed in the order as they are obtained from the *Analysis/Synthesis Loop*, since this order corresponds to the relevancy with respect to the reproduction of the sound. This unit is able to generate two bitstreams, one *basic bitstream* which allows the generation of the basic quality audio signal, and an *enhancement bitstream* which can be used in applications where a difference signal between input and decoder output is needed, e.g. for scalability. The *basic bitstream* mainly contains the frequency and amplitude parameters, while the *enhancement bitstream* contains phase parameters and information for finer quantization of frequency and envelope parameters.

For each frame of the input signal, a number of bits according to the desired bitrate is transmitted. In each frame an *envelope bit* is transmitted, which indicates whether envelope parameters are used or not. If this bit is set, the 3 envelope parameters are transmitted, and for each transmitted line an additional *line envelope bit* is transmitted, which indicates, whether a constant amplitude or the envelope is to be used for the synthesis of the corresponding line.

Since the human auditory system is not very sensitive to phase changes, only the frequency and amplitude information of the spectral lines are coded and transmitted in the *basic bitstream* to obtain a signal with the basic audio quality. But in this case, it is necessary to provide information for the decoder which enables it to generate a signal free of phase discontinuities at frame boundaries. Therefore the first processing stage detects lines which continue from one frame to another. If a line is to be continued from the previous frame, only the frequency and amplitude changes are quantized and transmitted instead of the absolute frequency and amplitude values. For this purpose the frequency and amplitude parameters of the  $i$ -th line in the current frame  $m$  are compared with those of the  $k$ -th line in the previous frame  $m-1$  for all possible combinations of  $i$  and  $k$ . The line continuation is used, if the relative frequency change

$$q_f(i, k) = \frac{|f_i(m) - f_k(m-1)|}{f_i(m)}$$

does not exceed a given threshold  $q_{f,max}$  and if the ratio of amplitudes

$$q_a(i, k) = \begin{cases} a_i(m)/a_k(m-1) & \text{if } a_i(m) \geq a_k(m-1) \\ a_k(m-1)/a_i(m) & \text{if } a_i(m) < a_k(m-1) \end{cases}$$

lies within the interval  $[1...q_{a,max}]$ . If there is more than one possibility to continue a line from the previous frame, that line in the previous frame is selected, for which the following similarity criterion reaches its maximum:

$$Q = \frac{q_{f,max} - q_f(i,k)}{q_{f,max}} \cdot \frac{q_{a,max} - q_a(i,k)}{(q_{a,max} - 1) q_a(i,k)}$$

The frequencies and amplitudes of the individual lines are quantized according to a Bark-like frequency scale and a logarithmic amplitude scale. For each line of the previous frame, a *continuation bit* is transmitted in the bitstream, which indicates whether the line is continued in the current frame or not. For new lines, the indices for the quantised frequency and amplitude are encoded using a SubDivisionCode (SDC) as described below. For all lines continued from the previous frame, the frequency and amplitude index difference is encoded with an entropy code.

Since the transmission of absolute frequency and amplitude values requires more bits per line than for the relative values, the number of lines transmitted per frame is varied in if a constant bitrate for the *basic bitstream* is desired.

Since the *basic bitstream* does not contain phase information, it is not useful to calculate a residual error signal by subtracting the corresponding decoder output signal from the input signal. In order to enable scalability modes, in which the residual signal is transmitted in a bitstream of higher bitrate, an additional *enhancement bitstream* is generated. It is constructed in the following way:

- if the envelope parameters are transmitted in the *basic bitstream*, additional bits for finer quantization of the 3 envelope parameters are transmitted
- if a line is starting, i.e. not continued from the previous frame, and its frequency exceeds a given threshold, additional bits for finer quantization of the absolute frequency are transmitted
- for each line the phase parameter is transmitted after uniform quantization

The number of bits per frame in the *enhancement bitstream* can vary, this has to be taken into account in the calculation of available bits for the coding of the residual error.

Since the position of a continued line in the current frame depends on the position of its predecessor in the previous frame, a bit allocation algorithm is used which ensures that the *N* lines transmitted in the current frame are always the *N* most relevant lines found by the *Analysis/Synthesis Loop*.

The system delay of the encoder is 1.5 times the frame length. This delay results from the length of the frame itself plus an additional delay of (0.5) times the frame length caused by the shifted overlapping window used for the frequency estimation.

### SDC-Encoding:

k : number of codewords (0...k-1)

i : value to encode

tab: table containing domain limits

```
void SDCEncode(int k,int i,int *tab)
{
    int *pp;
    int g,dp,min,max,cwl;
    long cw;

    cw=cwl=0;
    min=0;
    max=k-1;
    pp=tab+16;
    dp=16;

    while ( min!=max )
    {
        if ( dp ) g=(k*( *pp )>>10; else g=(max+min)>>1;
        dp>>=1;
        cw<<=1;
        cwl++;
        if ( i<=g ) { pp-=dp; max=g; } else { cw|=1; pp+=dp; min=g+1; }
```

```
    }  
    PutBits(cw,cwl);  
}
```

PutBits() writes the codeword to the bitstream, where the LSBs in cw are the “vclcbf” codeword and cwl determines number of bits to be transmitted.

### 7.A.2.2.3 Noise parameter quantization

The number of noise parameters that are quantized and encoded depends on the “relevance” measure of the noise signal component. If it is very low, no noise parameters are transmitted. For higher values of this measure an adequate number of LAR parameters are quantised and coded. Due to the properties of the reflection coefficients, the number of LAR parameters transmitted can be decided during bit-allocation in the encoder and no re-calculation of these parameters is required.

If the noiseEnvFlag is set then also the additional set of noise envelope parameters is quantized and coded.

### 7.A.2.3 HILN bitrate scalability

Due to the parametric signal representation utilized by the HILN parametric coder, it is well suited for applications requiring bitrate scalable coding. In such an application, the bitrate received by a decoder can be adapted dynamically to the properties of the transmission link or chosen according to some other rules. In case of a reduced-rate bitstream, only the parameters of the perceptually most relevant signal components (individual lines, harmonic tone, noise) are transmitted. In case of a full-rate bitstream also the parameters of additional signal components (e.g. individual lines) which are perceptually less relevant than those in the reduced-rate bitstream as well as additional parameters which refine the description of signal components already present in the reduced-rate bitstream are transmitted.

This bitrate scalability for HILN bitstreams can be implemented using the base and extension layer bitstreams as described in the normative part of the standard or by a dynamically controlled parameter encoding as described below.

#### 7.A.2.3.1 HILN Bitrate Scalability by Dynamically Controlled Parameter Encoding

To implement bitrate scalability by means of dynamically controlled parameter encoding, it is utilized that both the HILN Parameter Extraction and the HILN Parameter Encoder can be operated independently. The parameters generated by the Parameter Extraction tool can be fed to multiple Parameter Encoder tools, each of which generates a bitstream with a different bitrate. This is computationally very advantageous since HILN encoder complexity is mainly determined by the Parameter Extraction tool. It is also possible to store the unquantized parameters generated by the Parameter Extraction tool in a file. Then the Parameter Encoder tool can be used to generate a bitstream with the currently requested bitrate from the parameters stored in this file.

### 7.A.3 Music/Speech Mixed Encoder tool

In MPEG-4 Audio the parametric coder is utilized for coding natural audio signals at very low bitrates ranging from 2 kbit/s to about 16 kbit/s. The parametric coder provides two sets of tools suited for coding speech and non-speech audio signals respectively:

- The Harmonic Vector Excitation (HVXC) tools are suited for coding speech signals at 2 kbit/s and 4 kbit/s.
- The Harmonic and Individual Lines plus Noise (HILN) tools are suited for coding non-speech audio signals at bitrates of about 4 kbit/s and above.

In the HVXC only or HILN only mode, the encoding mode is selected manually during encoding and the selected mode is used for all of the audio signal being encoded.

In this Subclause, an integrated parametric coder that utilizes the HVXC and HILN tools alternatively or simultaneously is described. This integrated coder automatically selects the coding tools that are suited best for the actual input signal characteristics. In case of a speech signal the HVXC tools are used and for music the HILN tools are used. This selection is done based on the decision of an automatic speech/music classification tool. For signals that are a mixture of speech and music it is also possible to use the HVXC and HILN tools simultaneously.

### 7.A.3.1 Music/Speech classification tool

This is a tool for the parametric speech coder, which enables automatic music/speech identification for the parametric speech/audio coder (HVXC and HILN). The tool makes decisions using internal parameters of the HVXC.

The features of input sequences used for the identification are: behavior of “pitch strength” and “frame energy”. In general, speech has higher “pitch strength” and frequent and higher energy change than music.

This music/speech classification tool can be applied in two ways:

- The first 5 seconds of the signal to be encoded are analyzed by the classification tool and then either HVXC or HILN is selected to encode the signal according to the speech/music decision.
- The classification tool is operated continuously and its current speech/music decision is used to select HVXC or HILN for the current frame. In this application the decision delay of 5 sec has to be taken into account.

#### 7.A.3.1.1 Frame energy

Frame Energy  $P$  is computed as:

$$P = \sum_{n=0}^{159} s(n)^2$$

where  $s(n)$  is the input signal.

In this case, frames with energy levels higher than a pre-determined minimum level are used (ex.  $> -78$  dB). A short-term average frame energy is defined as

$$P_{av} = \sum_{t=0}^3 P\{t\} / 4$$

which is computed from the last four Frame energies.

A difference between frame energy and short term average frame energy is computed as:

$$Pd[frm] = |P - P_{av}| / P_{av}$$

$Pd[frm]$  is kept for around 250 frames (5 seconds).

#### 7.A.3.1.2 Pitch Strength

In HVXC, maximum autocorrelation of LPC residual ( $r0r$ ) is computed during pitch detection process.  $r0r$  are kept for around 250 frames.

#### 7.A.3.1.3 Music/Speech decision

Mean and variance of frame energies and  $r0r$ s are computed respectively as:

$$Pd(av) = \sum_{frm=0}^{249} Pd[frm] / 250$$

$$Pd(va) = \sqrt{\sum_{frm=0}^{249} (Pd[frm] - Pd(av))^2 / 250}$$

$$r0r(av) = \sum_{frm=0}^{249} r0r[frm] / 250$$

$$r0r(va) = \sqrt{\sum_{frm=0}^{249} (r0r[frm] - r0r(av))^2 / 250}$$

Speech data has higher variances than music data in the same range of mean value of r0r. The matrix is classified to three areas.

(1) speech  $r0r(va) \geq 0.153 r0r(av) + 0.113$

(2) unknown  $0.07 r0r(av) + 0.137 < r0r(va) < 0.153 r0r(av) + 0.113$

(3) music  $0.07 r0r(av) + 0.137 \geq r0r(va)$

If mean and variance are included in the area (1), the data is classified as speech. If they are in the area (3), the data is classified as music.

If the mean and variance exist in the area (2), the mean and variance of (differential) frame energy Pd are used additionally. Speech data has larger means and variances of Pd than music data. Speech and music data is separated into the following two areas.

(1) speech  $Pd(va) \geq -0.5Pd(av) + 0.8$

(2) music  $Pd(va) < -0.5Pd(av) + 0.8$

Using the above two criteria, speech and music are separated.

### 7.A.3.2 Integrated parametric coder

The integrated parametric coder can operate in the following modes:

PARAMode	Description
0	HVXC only
1	HILN only
2	switched HVXC / HILN
3	mixed HVXC / HILN

PARAModes 0 and 1 represent the fixed HVXC and HILN modes. PARAMode 2 permits automatic switching between HVXC and HILN depending on the current input signal type. In PARAMode 3 the HVXC and HILN coders can be used simultaneously and their output signals are added (mixed) in the decoder.

The integrated parametric coder uses a frame length of 40 ms and a sampling rate of 8 kHz and can operate at 2025 bit/s or any higher bitrate. Operation at 4 kbit/s or higher is suggested.

### 7.A.3.2.1 Integrated Parametric encoder

For the “HVXC only” and “HILN only” modes the parametric encoder is not modified. The “switched HVXC / HILN” and “mixed HVXC / HILN” modes are described below.

### 7.A.3.2.2 Switched HVXC/HILN mode

Because the speech/music classification tool is based on the HVXC encoder, the HVXC encoder is operated continuously for every frame. The bitstream frame generated by the HVXC encoder and the input audio signal are stored in two FIFO buffers to compensate for the 5 sec delay of the speech/music decision. If a frame is classified as “speech” then PARASwitchMode is set to 0 and the HVXC bitstream frame available at the bitstream FIFO output is transmitted. In case of a “music” decision, then PARASwitchMode is set to 1 and the output of the signal FIFO buffer is encoded by the HILN encoder and this HILN bitstream frame is transmitted. If HVXC is used for a frame, the HILN encoder is reset (prevNumLine = 0).

### 7.A.3.2.3 Mixed HVXC/HILN mode

To operate the parametric codec in “mixed HVXC / HILN” mode, speech and music components of the input signal have to be separated. If both components are already available separately (e.g. speech and background music) encoding is straightforward.

