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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.

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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:

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	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

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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 resillient 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 teleconferencing 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

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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

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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 scalablity, it can improve the quality of an

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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

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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

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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,

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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

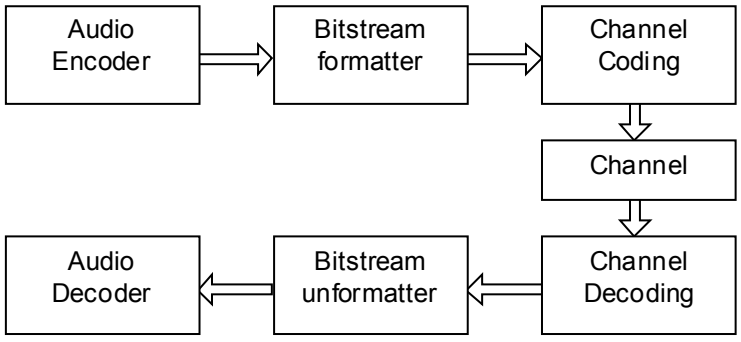
The EP tool provides unequal error protection. It receives several classes of bits from the audio coding tools, and then applies forward error correction codes (FEC) and/or cyclic redundancy codes (CRC) for each class, according to its error sensitivity.

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.

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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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Contents for Subpart 1

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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

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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.

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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. **Fs**: 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.

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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.

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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.

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* 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. Sign(x) = 1 when x > 0
Sign(x) = 0 when x == 0
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.

√ Square root.

$\log_{10}$ Logarithm to base ten.

$\log_e$ Logarithm to base e.

$\log_2$ 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.

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<= 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 twos 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*

π 3,14159265358...

e 2,71828182845...

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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 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.

while (condition) {
data_element
. . .
}
	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.

do { **data_element** . . . } while (condition)	The data element always occurs at least once. The data element is repeated until the condition is not true.

if (condition) {
data_element
. . .
}
	If the condition is true, then the first group of data elements occurs next in the data stream

else {
data_element
. . .
}
	If the condition is not true, then the second group of data elements occurs next in the data stream.

for (expr1; expr2; expr3)
{
data_element
. . .
}
	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.

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

for (i = 0; i < n; i++) {
data_element
. . .
}
	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.

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

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

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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 / Audio Object Type	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
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

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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 ot 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.

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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.

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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 scalablility 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.

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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

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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	

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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:

PCU = (1 + 0.15*ns) * 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

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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

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- **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 *2	Max RCU *2	EP-Tool: Max. redundancy by class FEC *1	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 *3	0 %	0
4	5.1	48	100	42 *3	0 %	0
5	2	22.05	5	8	20%	9
6	2	48	10	8	20%	9
7	5.1	48	25	12 *3	20%	22
8	5.1	48	100	42 *3	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 *2	Max RCU *2	EP-Tool: Max. redundancy by class FEC *1	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 *3	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 *3	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.

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- **Levels for the Natural Audio Profile**

Table 1.6 – Levels for the Natural Audio Profile

Level	Max. sampling rate [kHz]	Max PCU *2	EP-Tool: Max. redundancy by class FEC *1	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 *3	Max RCU *2 *3	Max. # audio objects	EP-Tool: Max. redundancy by class FEC *1	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 *4	-	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 *4	-	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 channnel 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.

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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 ()		
{		
audioObjectType;	**5**	**bslbf**
samplingFrequencyIndex;	**4**	**bslbf**
if (samplingFrequencyIndex==0xf)		
samplingFrequency;	**24**	**uimsbf**
channelConfiguration;	**4**	**bslbf**
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) {		
epConfig;	**2**	**bslbf**
if (epConfig == 2 || epConfig == 3) {		
ErrorProtectionSpecificConfig();		
}		

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```
            if ( epConfig == 3 ) {
                directMapping;                                  1           bslbf
                if ( ! directMapping ) {
                    /* tbd */
                }
            }
        }
}
```

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.

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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

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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.

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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 restictions 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()		
{		
upStreamType;	**4**	**uimsbf**
switch (upStreamType) {		
case 0: /* scalablity control */		
numOfLayer;	**6**	**uimsbf**
for (layer = 0; layer < numOfLayer; layer++) {		
avgBitrate[layer];	**24**	**uimsbf**
}		

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```
            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         uimsbf
                                                    [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 $\left(2^{numFrameExp}-1\right)$ 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.

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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.

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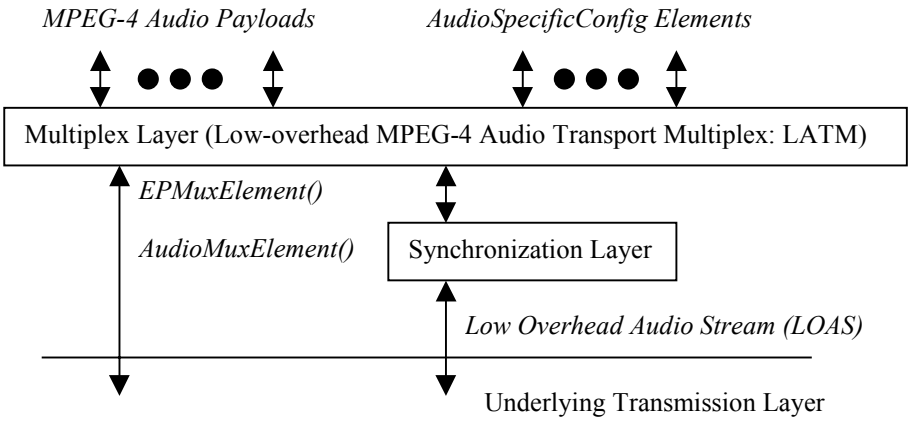

Figure 1.1 – Concept of MPEG-4 Audio Transport

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.

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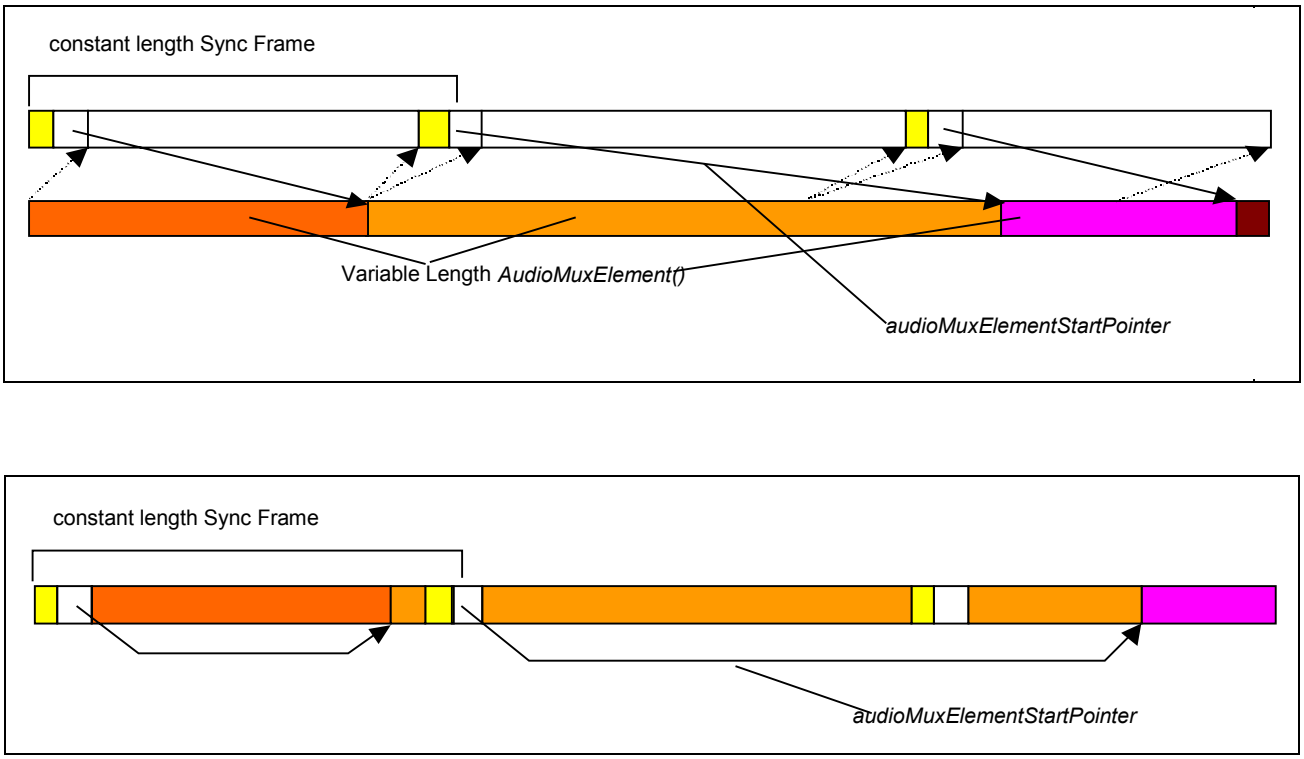

Figure 1.2 – Synchronization in AudioPointerStream()

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**	**uimsbf**
AudioMuxElement(1);		
ByteAlign();		
}		
}		

Table 1.17 – Syntax of EPAudioSyncStream()

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

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```
    }
}
```

Table 1.18 – Syntax of AudioPointerStream()

Syntax	No. of bits	Mnemonic
AudioPointerStream(length)		
{		
ByteAlign();		
audioMuxElementStartPointer;	ceil(ld(length))	**uimsbf**
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 = ceil(log2(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.*

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- *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
EPMuxElement(epDataPresent , muxConfigPresent)		
{		
if(epDataPresent) {		
epUsePreviousMuxConfig;	**1**	**bslbf**
epUsePreviousMuxConfigParity;	**2**	**bslbf**
if(!epUsePreviousMuxConfig) {		
epSpecificConfigLength;	**10**	**bslbf**
epSpecificConfigLengthParity;	**11**	**bslbf**
ErrorProtectionSpecificConfig();		
ErrorProtectionSpecificConfigParity();		
}		
ByteAlign();		
EPAudioMuxElement(muxConfigPresent);		
} else {		
AudioMuxElement(muxConfigPresent);		
ByteAlign();		
}		
}		

Table 1.20 – Syntax of AudioMuxElement()

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

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```
        }
        if( otherDataPresent ) {
            for( i=0; i<otherDataLenBits; I++ ) {
                otherDataBit;                                   1       bslbf
            }
        }
    }
    else {
        /* tbd */
    }
}
```

Table 1.21 – Syntax of StreamMuxConfig()

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

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

Table 1.22 – Syntax of PayloadLengthInfo()

Syntax	No. of bits	Mnemonic
PayloadLengthInfo()		
{		
if(allStreamsSameTimeFraming) {		
for (prog = 0; prog <= numProgram; prog++) {		
for (lay = 0; lay <= numLayer; lay++) {		
if(frameLengthType[streamID[prog][lay]] == 0) {		
do { /* always one complete access unit */		
tmp;	**8**	**uimsbf**
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) {		
MuxSlotLengthCoded[streamID[prog][lay]];	**2**	**uimsbf**
}		
}		
}		
}		
} else {		
numChunk;	**4**	**uimsbf**
for (chunkCnt=0; chunkCnt <= numChunk; chunkCnt++) {		
streamIndx;	**4**	**uimsbf**
prog = progCIndx[chunkCnt] = progSIndx[streamIndx]**;**		
lay = layCIndx[chunkCnt] = laySIndx [streamIndx];		
if(frameLengthType[streamID[prog][lay]] == 0) {		
do { /* not necessarily a complete access unit */		
tmp;	**8**	**uimsbf**
MuxSlotLengthBytes[streamID[prog][lay]] += tmp;		
} while (tmp == 255);		
AuEndFlag[streamID[prog][lay]];	**1**	**bslbf**
} else {		
if (frameLengthType[streamID[prog][lay]] == 5 ||		
frameLengthType[streamID[prog][lay]] == 7 ||		
frameLengthType[streamID[prog][lay]] == 3) {		
MuxSlotLengthCoded[streamID[prog][lay]];	**2**	**uimsbf**
}		
}		

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```
        }
    }
}
```

Table 1.23 – Syntax of PayloadMux()

```
Syntax                                                          No. of bits  Mnemonic
PayloadMux()
{
    if( allStreamsSameTimeFraming ) {
        for ( prog = 0; prog <= numProgram; prog++ ) {
            for ( lay = 0; lay <= numLayer; lay++ ) {
                payload [streamID[prog][ lay]];
            }
        }
    } else {
        for (chunkCnt=0; chunkCnt <= numChunk; chunkCnt++) {
            prog = progCIndx[chunkCnt];
            lay = layCIndx  [chunkCnt];

            payload [streamID[prog][ lay]];
        }
    }
}
```

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

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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 + StreamMuxConfig()	0
PayloadLengthInfo()	1
PayloadMux()	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).

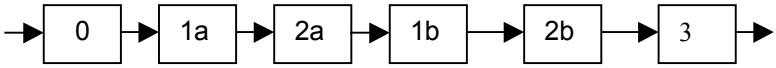

Figure 1.3 – Instance order in EPAudioMuxElement()

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.

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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 * (frameLength + 20).

CELPframeLengthTableIndex A field indicating one of two indexes for pointing out the frame length for a CELP or ER_CELP object. (Table 1.25and 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.

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

otherDataLenBits — A helper variable indicating the length in bits of the other data.

otherDataLenEsc — A field indicating whether there is more than one otherDataLenTmp data element following.

otherDataLenTmp — A field used to calculate otherDataLenBits.

crcCheckPresent — 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 — A field indicating the CRC check bits.

tmp — 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 — A field indicating one of two indexes for pointing out the payload length for CELP, HVXC, ER_CELP, and ER_HVXC objects.

numChunk — 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 — A field indicating the stream. Used if payloads are splitted into chunks.

chunkCnt — Helper variable to count number of chunks.

progSIndx,laySIndx — Helper variables to identify program and layer number from **streamIndx.**

progCIndx,layCIndx — Helper variables to identify program and layer number from **chunkCnt.**

AuEndFlag — 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]

		MuxSlotLengthCoded			
frameLengthType[]	HVXCframeLengthTableIndex[]	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]

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

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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

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

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

CELPframeLenghTableIndex	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.

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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.

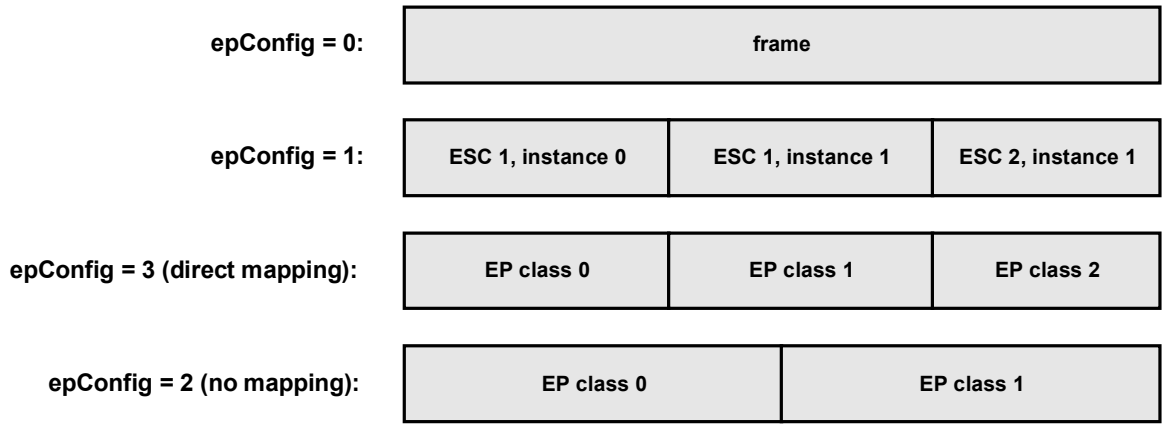

Figure 1.4 – EP classes for different epConfig values

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

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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.

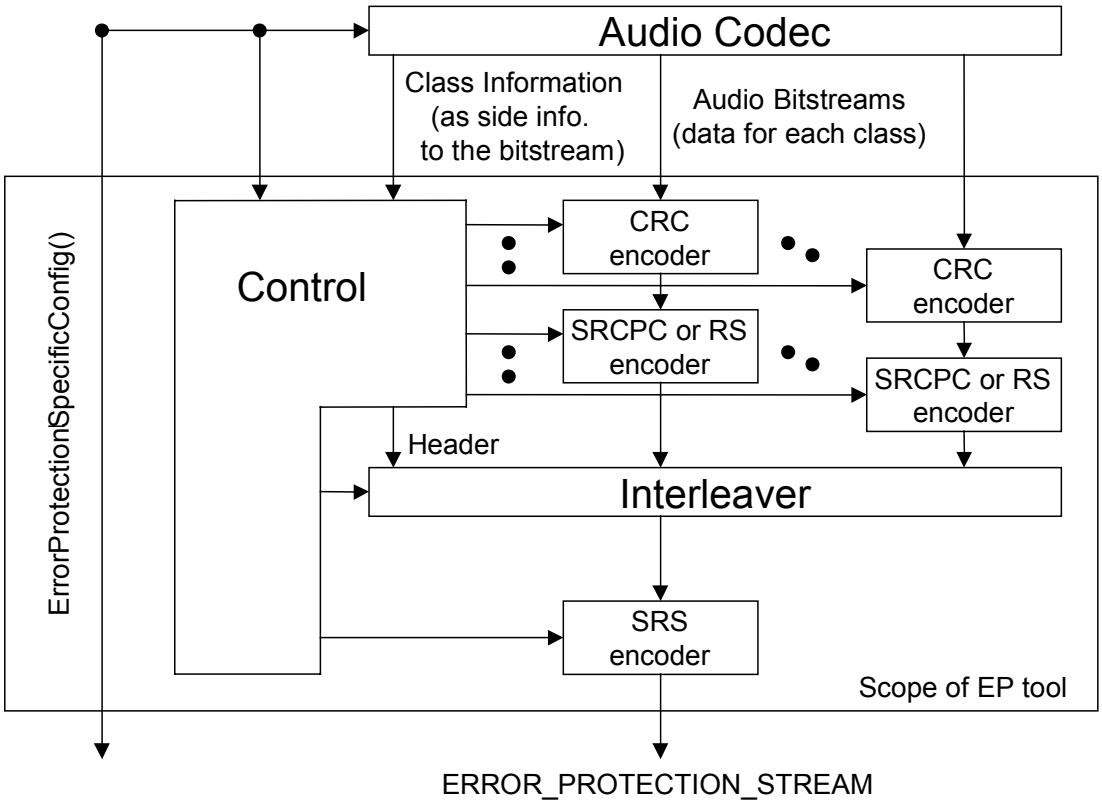

Figure 1.5 – Outline of EP encoder

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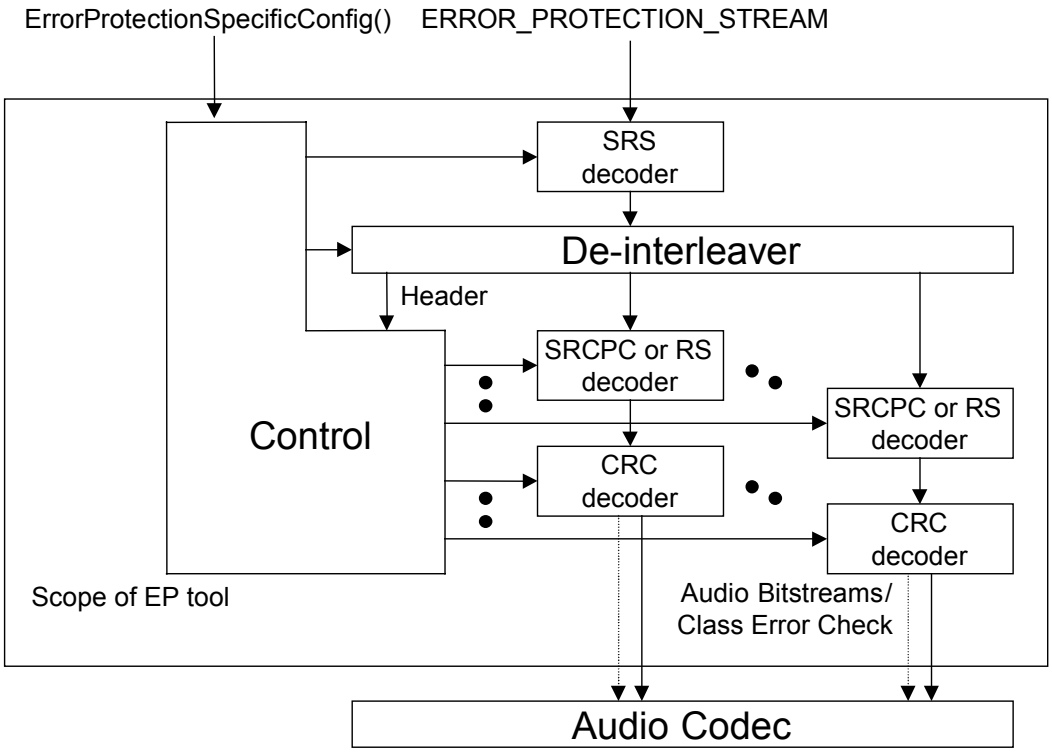

Figure 1.6 – Outline of EP decoder

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()		
{		
number_of_predefined_set;	**8**	**uimsbf**
interleave_type;	**2**	**uimsbf**
bit_stuffing;	**3**	**uimsbf**
number_of_concatenated_frame;	**3**	**uimsbf**
for (i = 0; i < number_of_predefined_set; i++) {		
number_of_class[i];	**6**	**uimsbf**
for (j = 0; j < number_of_class[i]; j++) {		
length_escape[i][j];	**1**	**uimsbf**
rate_escape[i][j];	**1**	**uimsbf**
crclen_escape[i][j];	**1**	**uimsbf**
if (number_of_concatenated_frame != 1) {		
concatenate_flag[i][j];	**1**	**uimsbf**
}		
fec_type[i][j];	**2**	**uimsbf**
if(fec_type[i][j] == 0) {		
termination_switch[i][j];	**1**	**uimsbf**
}		
if (interleave_type == 2) {		

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	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

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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){		
interleaved_frame_mode1;	**1 -**	**bslbf**
}		
if (interleave_type == 2){		
interleaved_frame_mode2;	**1 -**	**bslbf**
}		
stuffing_bits;	**Nstuff**	**bslbf**
}		

Table 1.30 – Syntax of ep_header ()

Syntax	No. of bits	Mnemonic
ep_header()		
{		
choice_of_pred;	**N_{pred}**	**uimsbf**
choice_of_pred_parity;	**N_{pred_parity}**	**bslbf**
class_attrib();		
class_attrib_parity;	**N_{attrib_parity}**	**bslbf**
}		

Npred: the smallest integer value greater than log2 (# of pre-defined set).
Npred_parity: See subclause 1.8.4.3
Nattrib_parity: 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){		
class_bit_count[j];	**Nbitcount**	**uimsbf**
}		
if (rate_escape[frame_pred][j] == 1){		
class_code_rate[j];	**3**	**uimsbf**
}		
if (crclen_escape[frame_pred][j] == 1){		
class_crc_count[j];	**3**	**uimsbf**
}		
}		
if (bit_stuffing == 1){		
num_stuffing_bits;	**3**	**uimsbf**
}		
}		

class_count: number of class for this frame
frame_pred: selected predefined set for this frame

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Table 1.32 – Syntax of ep_encoded_classes ()

Syntax	No. of bits	Mnemonic

ep_encoded_classes(class_count)
{
for(j=0; j<class_count; j++){
ep_encoded_class[j];
}
}
		bslbf

1.8.3 *General information*

1.8.3.1 Definitions

ErrorProtectionSpecificConfig (): Error protection specific configuration that is out-of-band information.

number_of_predefined_set The number of pre-defined set.

interleave_type 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.

bit_stuffing 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.

number_of_concatenated_frame 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

number_of_class[i] The number of classes for i-th pre-defined set.

length_escape[i][j] 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 **number_of_bits_for_length[i][j]** value should be 0.

rate_escape[i][j] 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.

crclen_escape[i][j] 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.

concatenate_flag[i][j] 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)

fec_type[i][j] 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.

termination_switch[i][j] 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.

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interleave_switch[i][j] 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 wihtin 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)

class_optional 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^N predefinition sets, where N equals the number of optional classes.(See subclause 1.8.4.2)

number_of_bits_for_length[i][j] This field exists only when the **length_escape[i][j]** 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).

class_length[i][j] This field exists only when the **length_escape[i][j]** 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.

class_rate[i][j] This field exists only when the **rate_escape[i][j]** is 0. In case **fec_type[i][j]** 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 **fec_type[i][j]** 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 **fec_type[i][j]** shall have the same value of **class_rate[i][j]**.

class_crclen[i][j] This field exists only when the **crclen_escape[i][j]** 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 subclause1.8.4.5)

class_reordered_output 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.

class_output_order[i][j] 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 (**class_output_order[i][j]**)-th class from ep decoder. See subclause 1.8.4.9.

header_protection 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.

header_rate, header_crclen These values have the same semantics with **class_rate[i][j]** and **class_crclen[i][j]** respectively, while these error protection is utilized for the protection of header part.

rs_fec_capability This field indicates the correction capability of SRS code to protect the whole frame (see subclause1.8.4.7). This capability is given as number of erroneous bytes that can be corrected. The value "0" indicates SRS is not used.

rs_ep_frame() Reed-Solomon error protected frame that is applied Reed-Solomon code.

rs_parity_bits The Reed-Solomon parity bits for ep_frame(). See subclause 1.8.4.7.

ep_frame() error protected frame.

ep_header() EP frame header information.

ep_encoded_classes() The EP encoded audio information.

interleaved_frame_mode1 The information bits after interleaving with interleaving mode 1. See subclause 1.8.4.1 and subclause 1.8.4.8.

interleaved_frame_mode2 The information bits after interleaving with interleaving mode 2. See subclause 1.8.4.1 and subclause 1.8.4.8.

stuffing_bits 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.

choice_of_pred The choice of pre-defined set. See subclause 1.8.4.2.

choice_of_pred_parity The parity bits for **choice_of_pred**. See subclause 1.8.4.2.

class_attrib_parity The parity bits for class_attrib(). See subclause 1.8.4.2.

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class_attrib() Attribution information for each class

class_bit_count[j] The number of information bits included in the class. This field only exists in case the length_escape in out-of-band information is 1 (escape). The number of bits of this parameter Nbitcount is also signaled in the out-of-band information.

class_code_rate[j] The coding rate for the audio data belonging to the class, as defined in the table below. This field only exists in case the rate_escape 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

class_crc_count[j] 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 crclen_escape 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

num_stuffing_bits the number of stuffing bits for the EP frame octet alignment. This field only exists in case the **bit_stuffing** in out-of-band information is 1.

ep_encoded_class[j] CRC/SRCPC encoded audio data of j-th class.Note that if **class_bit_count[j]** == 0, 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:

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- **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 all predefinition sets */
    for ( j = 0; j < pow ( 2, NCO[i] ); j++ ) {  /* unwraping */
        for ( k = 0; k < NCO[i]; k++ ) {      /* for all optional classes */
            if ( j & ( 0x01 << k ) ) {
                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		

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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 -	-

Npred_parity(or Nattrib_parity) = total number of bits – Number of bit to be protected

Note that Number of bit to be protected is Npred (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:

BCH(7,4): x^3+x+1

BCH(15,7): $x^8+x^7+x^6+x^4+1$

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

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

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

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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.

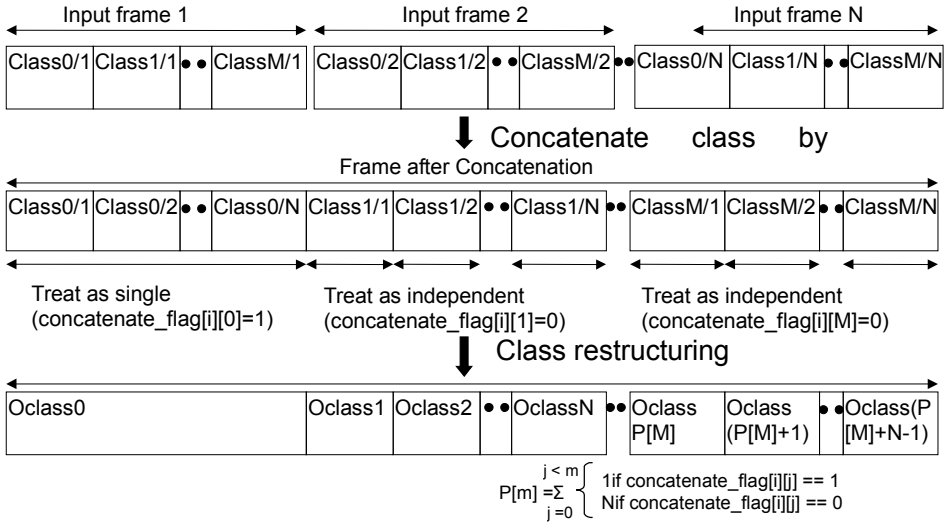

Figure 1.7 – Concatenation procedure

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***: $x^5+x^4+x^2+x+1$

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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.

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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.

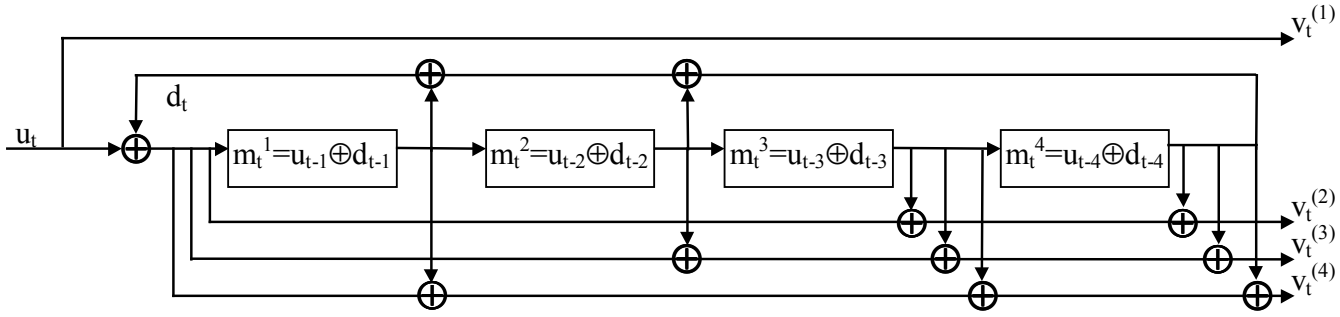

Figure 1.8 – Shift register realization for systematic recursive convolutional encoder

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

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

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

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

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

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

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

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

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

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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*, 255-2*k*-*l*) defined over GF(2^8) 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 (255-2*k*) octets. This division is made from the beginning of the frame so that the length of the sub-frame becomes (255-2*k*) 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.

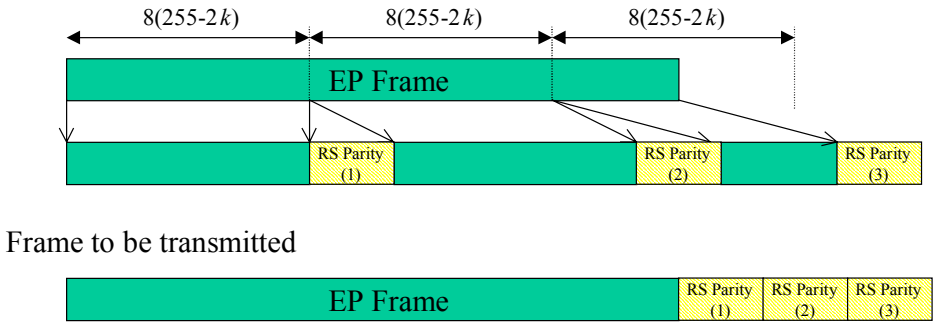

Figure 1.9 – RS encoding of EP frame

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 $0 \le 2k \le 254$. The SRS code defined in the Galois Field GS(2^8) is generated from a generator polynomial $g(x) = (x-\alpha)(x-\alpha^2)\cdots(x-\alpha^{2k})$, where α denotes a root of the primitive polynomial $m(x)=x^8+x^4+x^3+x^2+1$. The binary representative of α^j is shown in the Table 1.41 below, where the MSB of the octet is transmitted first.

Table 1.41 – Binary representation for α^i ($0 \le i \le 254$) over GF(2^8)

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

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

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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 = minimum integer small than (L / (255-2k))

l_i = 255-2k, for i<N

L mod (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:

Nrsparity = 8N

1.8.4.8 Recursive interleaving

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

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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.

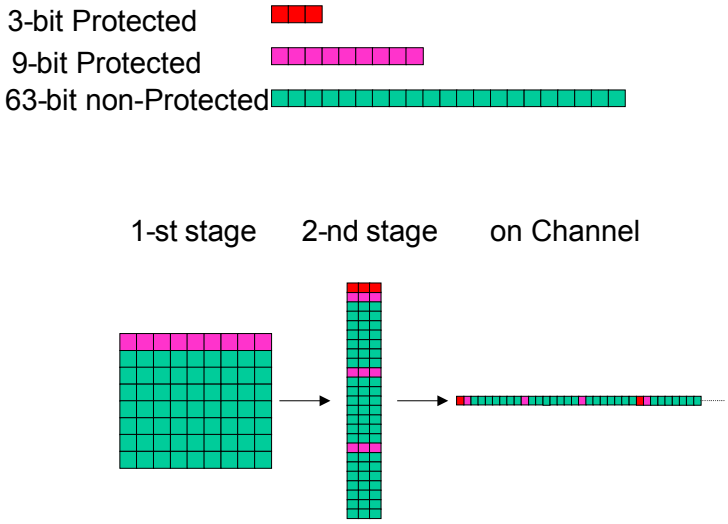

Figure 1.11 – Example of multi-stage interleaving

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.

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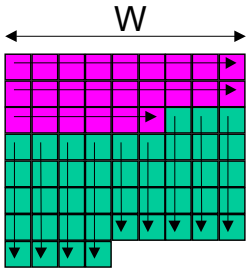

Figure 1.12 – Interleave matrix in non-rectangular case

1.8.4.8.1 Definition of recursive interleaver

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

X_i, $0<=i<I_x$

Y_j, $0<=j<I_y$,

where I_x and I_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_i 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,

$D = (I_x + I_y) / W$

$d = I_x + I_y - D * W$

Where '/' indicates division by truncation.

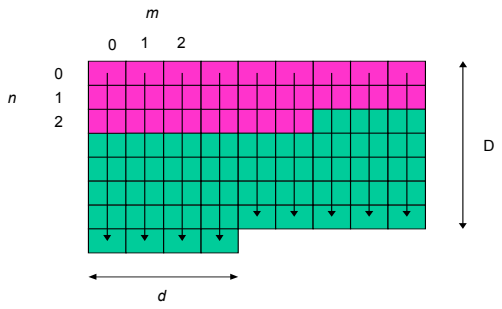

Figure 1.13 – The size of interleaving matrix

The output bitstream Z_k ($0 < k <= I_x+I_y$) 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

$m = i \bmod W$, $\quad n = i / W$,

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Thus Z_k which is set by the X_i becomes:

$Z_k = X_i$, 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:

$D' = I_x / W$

$d' = I_x - D' * W$

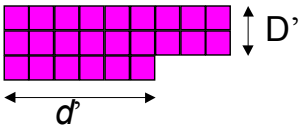

Figure 1.14 – The bits which are set with Xi in the interleaving matrix

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.

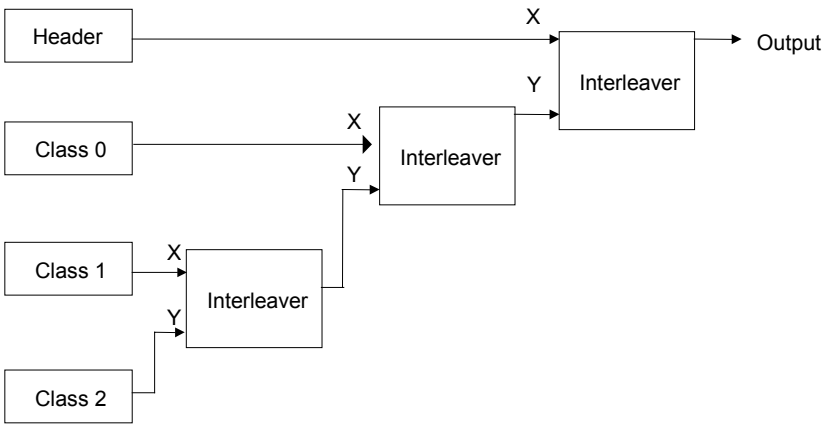

Figure 1.15 – Interleaving process of mode1 specification

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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.

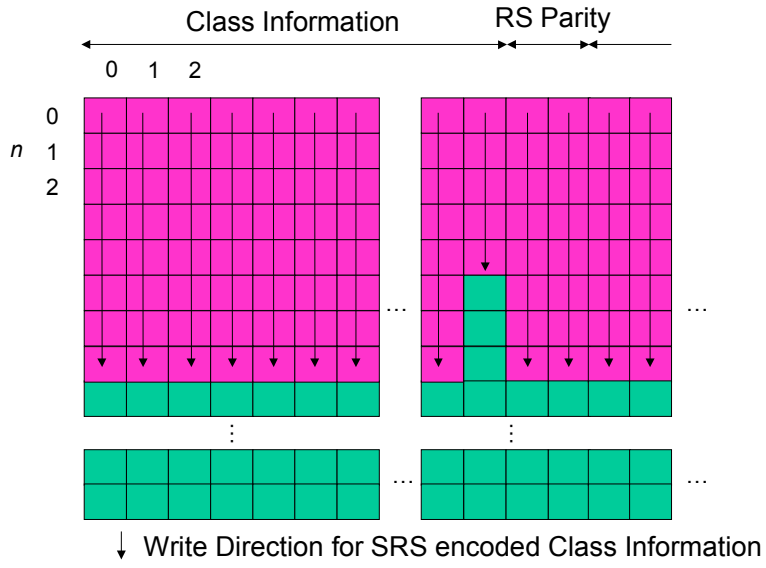

Figure 1.16 – Interleaving matrix in RS encoded class case

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_N0;
   }
}
set BUF_N0 into BUF_Y;
clear buffer BUF_N0
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;
   }
}
```


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```
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;
```

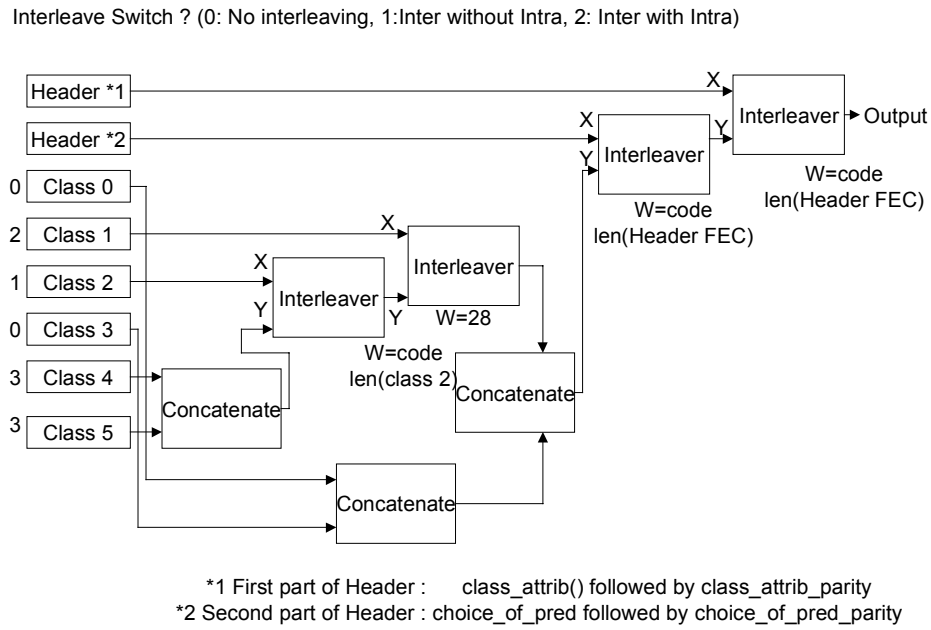

Figure 1.17 – Interleave process with class-wise control of interleaving

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.

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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

adif_sequence()
{
  adif_header()
  raw_data_stream()
}		

Table 1. A.2 – Syntax of adif_header()

Syntax	No. of bits	Mnemonic
adif_header() {		
adif_id	**32**	**bslbf**
copyright_id_present	**1**	**bslbf**
if(copyright_id_present)		
copyright_id	**72**	**bslbf**
original_copy	**1**	**bslbf**
home	**1**	**bslbf**
bitstream_type	**1**	**bslbf**
bitrate	**23**	**uimsbf**
num_program_config_elements	**4**	**bslbf**
for (i = 0; i < num_program_config_elements + 1; i++) {		
if(bitstream_type == ‘0’)		
adif_buffer_fullness	**20**	**uimsbf**

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```
            program_config_element()
        }
}
```

Table 1. A.3 – Syntax of raw_data_stream()

Syntax	No. of bits	Mnemonic

raw_data_stream()
{
  while (data_available()) {
  raw_data_block()
  byte_alignment()
  }
}		

1.A.2.2 Audio_Data_Transport_Stream frame, ADTS

Table 1. A.4 – Syntax of adts_sequence()

Syntax	No. of bits	Mnemonic

adts_sequence()
{
  while (nextbits()==syncword) {
  adts_frame()
  }
}		

Table 1. A.5 – Syntax of adts_frame()

Syntax	No. of bits	Mnemonic

adts_frame()
{
  byte_alignment()
  adts_fixed_header()
  adts_variable_header()
  adts_error_check()
  for(i=0; i<no_raw_data_blocks_in_frame+1; i++) {
  raw_data_block()
  }
}		

1.A.2.2.1 Fixed Header of ADTS

Table 1. A.6 – Syntax of adts_fixed_header()

Syntax	No. of bits	Mnemonic
adts_fixed_header() {		
Syncword	**12**	**bslbf**
ID	**1**	**bslbf**
Layer	**2**	**uimsbf**
protection_absent	**1**	**bslbf**
Profile_ObjectType	**2**	**uimsbf**

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sampling_frequency_index	**4**	**uimsbf**
private_bit	**1**	**bslbf**
channel_configuration	**3**	**uimsbf**
original/copy	**1**	**bslbf**
home	**1**	**bslbf**
Emphasis	**2**	**bslbf**
}		

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()		
{		
copyright_identification_bit	**1**	**bslbf**
copyright_identification_start	**1**	**bslbf**
aac_frame_length	**13**	**bslbf**
adts_buffer_fullness	**11**	**bslbf**
no_raw_data_blocks_in_frame	**2**	**uimsbf**
}		

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')		
crc_check	**16**	**Rpchof**
}		

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**.

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- **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

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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).

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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.

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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_lentgh[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 */
```


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

1.B.2.1 Example for AAC

based on the error sensitivity category assignement 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	SRCPC 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),SRCPC code rate 8/12, 8 bit CRC
  Class 2: 839 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  */
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 */
```

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- 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

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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

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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

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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

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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

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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 */
```

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```
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 sesitivity 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

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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

frame_size: the number of samples in a frame
g_ac: the adaptive codebook gain
g_ec: the MPE gain
lpc_order: the order of LP
signal_mode: the speech mode
signal_mode_pre: 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.

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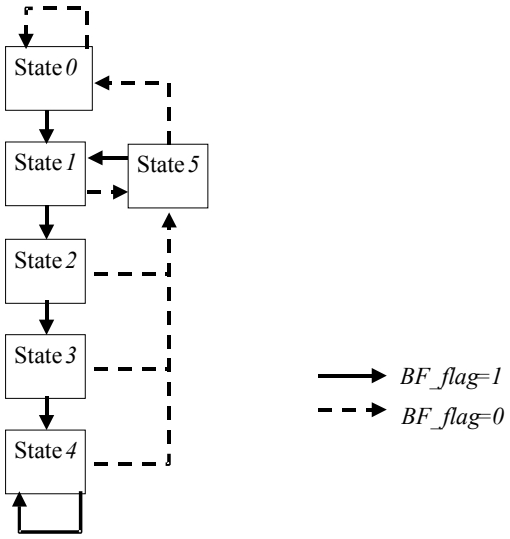

Figure 1.B.1 – State transition model for controlling the error concealment

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.

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BE mode:

The speech mode (*singal_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:

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

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

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 (*blsp*[*0*][]) is estimated from the LSP (*qlsp_pre*[]) in the previous frame, the predicted LSP (*vec_hat*[]) and the prediction coefficient (*cb*[*0*][]) in the current frame as follows:

```
for ( i = 0; i < lpc_order; i++ ) {
    blsp[0] [i] = (qlsp_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.

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

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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 occures, 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)

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--- 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**						
	class I bits	**class II bits**	**class III bits**	**class IV bits**	**class V bits**	**class VI bits**	**total**
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**
total	**44**	**4**	**4**	**4**	**4**	**20**	**80**

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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						
	class I bits	**class II bits**	**class III bits**	**class IV bits**	**Class V bits**	**class VI bits**	**total**
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**
total	**44**	**0**	**0**	**0**	**0**	**36**	**80**

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							
	class I bits	**class II bits**	**class III bits**	**class IV bits**	**class V bits**	**class VI bits**	**class VII bits**	**total**
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**

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

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**							
	class I bits	**class II bits**	**class III bits**	**class IV bits**	**class V bits**	**class VI bits**	**class VII bits**	**total**
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**
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**
VX_gain2[0]p/c	3/3	-	-	-	-	-	-	**6**
VX_gain2[1]p/c	3/3	-	-	-	-	-	-	**6**
VX_gain2[2]p/c	3/3	-	-	-	-	-	-	**6**
VX_gain2[3]p/c	2/2	-	-	-	-	-	1/1	**6**
VX_shape2[0]p/	-	-	-	-	-	-	5/5	**10**
VX_shape2[1]p/	-	-	-	-	-	-	5/5	**10**
VX_shape2[2]p/	-	-	-	-	-	-	5/5	**10**
VX_shape2[3]p/	-	-	-	-	-	-	5/5	**10**
total	**66**	**0**	**0**	**0**	**0**	**0**	**94**	**160**

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**
Class I Bit			**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	**Class V Bit**		

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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	**Class VI Bit**		
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	**Class II Bit**			**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	**Class III Bit**			**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	**Class IV Bit**			**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**
Class I Bit			**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	**Class V Bit**		
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	**Class VI Bit**		
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	**Class II Bit**			**68**	LSP3p	0
16	Pitchp	3	**44**	SE_shape1p	3	**69**	Pitchp	0

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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	**Class III Bit**			**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	**Class IV Bit**			**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**
Class I Bit			**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	**Class V Bit**		
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	**Class VI Bit**		
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	**Class II Bit**			**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	**Class III Bit**			**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	**Class IV Bit**			**78**	LSP3c	0
26	SE_gainc	3	**52**	SE_shape1c	3	**79**	Pitchc	0
27	SE_gainc	2	**53**	SE_shape1c	2			

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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**
Class I Bit			**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	**Class V Bit**		
2	LSP4p	0	**31**	VX_gain1[1]c	1	**56**	LSP3c	1
3	VX_gain1[0]p	3	**32**	VX_gain1[1]c	0	**57**	LSP3c	0
4	VX_gain1[0]p	2	**33**	LSP1c	4	**58**	VX_shape1[0]c	5
5	VX_gain1[0]p	1	**34**	LSP1c	3	**59**	VX_shape1[0]c	4
6	VX_gain1[0]p	0	**35**	LSP1c	2	**Class VI Bit**		
7	VX_gain1[1]p	3	**36**	LSP1c	1	**60**	VX_shape1[0]p	3
8	VX_gain1[1]p	2	**37**	LSP1c	0	**61**	VX_shape1[0]p	2
9	VX_gain1[1]p	1	**38**	LSP2c	6	**62**	VX_shape1[0]p	1
10	VX_gain1[1]p	0	**39**	LSP2c	5	**63**	VX_shape1[0]p	0
11	LSP1p	4	**40**	LSP2c	4	**64**	VX_shape1[1]p	5
12	LSP1p	3	**41**	LSP2c	3	**65**	VX_shape1[1]p	4
13	LSP1p	2	**42**	LSP3c	4	**66**	VX_shape1[1]p	3
14	LSP1p	1	**43**	LSP3c	3	**67**	VX_shape1[1]p	2
15	LSP1p	0	**Class II Bit**			**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**	VX_shape1[0]c	3
18	LSP2p	4	**46**	LSP2p	0	**71**	VX_shape1[0]c	2
19	LSP2p	3	**47**	LSP3p	2	**72**	VX_shape1[0]c	1
20	LSP3p	4	**Class III Bit**			**73**	VX_shape1[0]c	0
21	LSP3p	3	**48**	LSP3p	1	**74**	VX_shape1[1]c	5
22	VUVc	1	**49**	LSP3p	0	**75**	VX_shape1[1]c	4
23	VUVc	0	**50**	VX_shape1[0]p	5	**76**	VX_shape1[1]c	3
24	LSP4c	0	**51**	VX_shape1[0]p	4	**77**	VX_shape1[1]c	2
25	VX_gain1[0]c	3	**Class IV Bit**			**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.12 – Bit Order for 4kbit/s(voiced frame – voiced frame)

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

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10	LSP1p	2	**Class II Bit**			**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	**Class VI Bit**		
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	**Class VII Bit**		
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			

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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**
Class 1bit			**55**	VX_gain2[0]c	2	**Class III Bit**		
0	VUVp	1	**56**	VX_gain2[0]c	1	**110**	SE_shape1p	3
1	VUVp	0	**57**	VX_gain2[0]c	0	**111**	SE_shape1p	2
2	LSP4p	0	**58**	VX_gain2[1]c	2	**112**	SE_shape1p	1
3	SE_gainp	4	**59**	VX_gain2[1]c	1	**113**	SE_shape1p	0
4	SE_gainp	3	**60**	VX_gain2[1]c	0	**Class IV Bit**		
5	SE_gainp	2	**61**	VX_gain2[2]c	2	**114**	SE_shape2p	3
6	SE_gainp	1	**62**	VX_gain2[2]c	1	**115**	SE_shape2p	2
7	SE_gainp	0	**63**	VX_gain2[2]c	0	**116**	SE_shape2p	1
8	LSP1p	4	**64**	VX_gain2[3]c	2	**117**	SE_shape2p	0
9	LSP1p	3	**65**	VX_gain2[3]c	1	**Class V Bit**		
10	LSP1p	2	**Class II Bit**			**118**	VX_shape1[1]c	4
11	LSP1p	1	**66**	SE_shape4p	8	**119**	VX_shape1[1]c	3
12	LSP1p	0	**67**	SE_shape4p	7	**120**	VX_shape1[1]c	2
13	Pitchp	6	**68**	SE_shape4p	6	**121**	VX_shape1[1]c	1
14	Pitchp	5	**69**	SE_shape4p	5	**Class VI Bit**		
15	Pitchp	4	**70**	SE_shape4p	4	**122**	VX_shape1[1]c	0
16	Pitchp	3	**71**	SE_shape4p	3	**123**	VX_shape2[0]c	4
17	Pitchp	2	**72**	SE_shape4p	2	**124**	VX_shape2[0]c	3
18	Pitchp	1	**73**	SE_shape4p	1	**125**	VX_shape2[0]c	2
19	LSP2p	6	**74**	SE_shape4p	0	**Class VII Bit**		
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**	VX_gain2[3]c	0	**139**	LSP5p	0
34	VUVc	0	**89**	LSP2c	2	**140**	Pitchp	0
35	LSP4c	0	**90**	LSP2c	1	**141**	SE_shape3p	1
36	VX_gain1[0]c	3	**91**	LSP2c	0	**142**	SE_shape3p	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

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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**
Class 1bit			**55**	LSP2c	3	**Class III Bit**		
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	**Class IV Bit**		
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	**Class V Bit**		
10	VX_gain1[1]p	0	**Class II Bit**			**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	**Class VI Bit**		
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	**Class VII Bit**		
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

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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**
Class 1bit			**55**	VX_gain2[0]c	2	**Class III Bit**		
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	**Class IV Bit**		
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	**Class V Bit**		
10	VX_gain1[1]p	0	**Class II Bit**			**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	**Class VI Bit**		
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

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18	LSP2p	4	**73**	LSP3p	0	**125**	VX_shape2[0]c	2
19	LSP2p	3	**74**	LSP5p	6	**Class VII Bit**		
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.

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

	2kbit/s source coder	4kbit/s Source Coder
Class I		
Source coder bits	44	66
CRC parity	6	6
Code Rate	8/16	8/16
Class I total	100	144
Class II		
Source coder bits	4	44
CRC parity	1	6
Code Rate	8/8	8/8
Class II total	5	50
Class III		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class III total	5	5
Class IV		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class IV total	5	5
Class V		
Source coder bits	4	4
CRC parity	1	1
Code Rate	8/8	8/8
Class V total	5	5
Class VI		
Source coder bits	20	4
CRC parity	0	1
Code Rate	8/8	8/8
Class VI total	20	5
Class VII		
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.

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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.

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

$LSP_{base}(n)$ 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

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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 "$Am[0\ldots127]$" 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.

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

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

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

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).

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

$s[i]$ 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 $res[0\ldots159]$ is gain controlled as shown in Eq.(4). In the equation, $res_{(org)}[i]$ is computed from stochastic codebook parameters.

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

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1.B.4.4.1.5 Frame Masking State Transitions

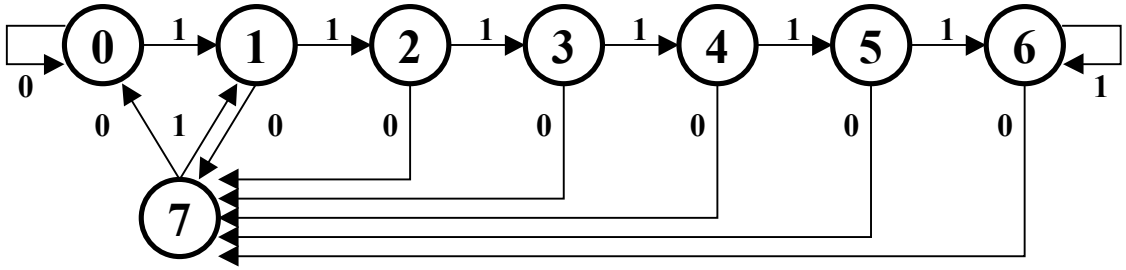

Figure 1.B.2 – Frame Masking State Transitions

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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.

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Notes:
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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

	Company	Part 1	Part 2	Part 3	Part 5	Part 6
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.	DirecTV	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	

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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			

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