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Information technology — Generic coding of moving pictures and associated audio information —

Part 7: Advanced Audio Coding (AAC)

Technologies de l'information — Codage générique des images animées et du son associé —

Partie 7: Codage du son avancé (AAC)

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Foreword

ISO (the International Organization for Standardization) is a worldwide federation of national standards bodies (ISO member bodies). The work of preparing International Standards is normally carried out through ISO technical committees. Each member body interested in a subject for which a technical committee has been established has the right to be represented on that committee. International organizations, governmental and non-governmental, in liaison with ISO, also take part in the work. ISO collaborates closely with the International Electrotechnical Commission (IEC) on all matters of electrotechnical standardization.

International Standards are drafted in accordance with the rules given in the ISO/IEC Directives, Part 2.

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

ISO/IEC 13818-7 was prepared by Joint Technical Committee ISO/IEC JTC 1, *Information technology*, Subcommittee SC 29, *Coding of audio, picture, multimedia and hypermedia information*.

This third edition cancels and replaces the second edition (ISO/IEC 13818-7:2003), which has been technically revised. It also incorporates the Amendment ISO/IEC 13818-7:2003/Amd.1:2004.

ISO/IEC 13818 consists of the following parts, under the general title *Information technology — Generic coding of moving pictures and associated audio information*:

— *Part 1: Systems*

— *Part 2: Video*

— *Part 3: Audio*

— *Part 4: Conformance testing*

— *Part 5: Software simulation*

— *Part 6: Extensions for DSM-CC*

— *Part 7: Advanced Audio Coding (AAC)*

— *Part 9: Extension for real time interface for systems decoders*

— *Part 10: Conformance extensions for Digital Storage Media Command and Control (DSM-CC)*

— *Part 11: IPMP on MPEG-2 systems*

Introduction

The standardization body ISO/IEC JTC 1/SC 29/WG 11, also known as the Moving Pictures Experts Group (MPEG), was established in 1988 to specify digital video and audio coding schemes at low data rates. MPEG completed its first phase of audio specifications (MPEG-1) in November 1992, ISO/IEC 11172-3. In its second phase of development, the MPEG Audio subgroup defined a multichannel extension to MPEG-1 audio that is backwards compatible with existing MPEG-1 systems (MPEG-2 BC) and defined an audio coding standard at lower sampling frequencies than MPEG-1, ISO/IEC 13818-3.

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Information technology — Generic coding of moving pictures and associated audio information —

Part 7:
Advanced Audio Coding (AAC)

1 Scope

This International Standard describes the MPEG-2 audio non-backwards compatible standard called MPEG-2 Advanced Audio Coding, AAC [1], a higher quality multichannel standard than achievable while requiring MPEG-1 backwards compatibility. This MPEG-2 AAC audio standard allows for ITU-R 'indistinguishable' quality according to [2] at data rates of 320 kbit/s for five full-bandwidth channel audio signals.

The AAC decoding process makes use of a number of required tools and a number of optional tools. Table 1 lists the tools and their status as required or optional. Required tools are mandatory in any possible profile. Optional tools may not be required in some profiles.

Table 1 – AAC decoder tools

Tool Name	Required / Optional
Bitstream Formatter	Required
Noiseless Decoding	Required
Inverse quantization	Required
Rescaling	Required
M/S	Optional
Prediction	Optional
Intensity	Optional
Dependently switched coupling	Optional
TNS	Optional
Filterbank / block switching	Required
Gain control	Optional
Independently switched coupling	Optional

1.1 MPEG-2 AAC Tools Overview

The basic structure of the MPEG-2 AAC system is shown in Figure 1 and Figure 2. As is shown in Table 1, there are both required and optional tools in the decoder. The data flow in this diagram is from left to right, top to bottom. The functions of the decoder are to find the description of the quantized audio spectra in the bitstream, decode the quantized values and other reconstruction information, reconstruct the quantized spectra, process the reconstructed spectra through whatever tools are active in the bitstream in order to arrive at the actual signal spectra as described by the input bitstream, and finally convert the frequency domain spectra to the time domain, with or without an optional gain control tool. Following the initial reconstruction and scaling of the spectrum reconstruction, there are many optional tools that modify one or more of the spectra in order to provide more efficient coding. For each of the optional tools that operate in the spectral domain, the option to "pass through" is retained, and in all cases where a spectral operation is omitted, the spectra at its input are passed directly through the tool without modification.

The input to the bitstream demultiplexer tool is the MPEG-2 AAC bitstream. The demultiplexer separates the parts of the MPEG-AAC data stream into the parts for each tool, and provides each of the tools with the bitstream information related to that tool.

The outputs from the bitstream demultiplexer tool are:

- The sectioning information for the noiselessly coded spectra
- The noiselessly coded spectra
- The M/S decision information (optional)
- The predictor state information (optional)
- The intensity stereo control information and coupling channel control information (both optional)
- The temporal noise shaping (TNS) information (optional)
- The filterbank control information
- The gain control information (optional)

The noiseless decoding tool takes information from the bitstream demultiplexer, parses that information, decodes the Huffman coded data, and reconstructs the quantized spectra and the Huffman and DPCM coded scalefactors.

The inputs to the noiseless decoding tool are:

- The sectioning information for the noiselessly coded spectra
- The noiselessly coded spectra

The outputs of the Noiseless Decoding tool are:

- The decoded integer representation of the scalefactors:
- The quantized values for the spectra

The inverse quantizer tool takes the quantized values for the spectra, and converts the integer values to the non-scaled, reconstructed spectra. This quantizer is a non-uniform quantizer.

The input to the Inverse Quantizer tool is:

- The quantized values for the spectra

The output of the inverse quantizer tool is:

- The un-scaled, inversely quantized spectra

The rescaling tool converts the integer representation of the scalefactors to the actual values, and multiplies the un-scaled inversely quantized spectra by the relevant scalefactors.

The inputs to the rescaling tool are:

- The decoded integer representation of the scalefactors
- The un-scaled, inversely quantized spectra

The output from the scalefactors tool is:

- The scaled, inversely quantized spectra

The M/S tool converts spectra pairs from Mid/Side to Left/Right under control of the M/S decision information in order to improve coding efficiency.

The inputs to the M/S tool are:

- The M/S decision information
- The scaled, inversely quantized spectra related to pairs of channels

The output from the M/S tool is:

- The scaled, inversely quantized spectra related to pairs of channels, after M/S decoding

Note: The scaled, inversely quantized spectra of individually coded channels are not processed by the M/S block, rather they are passed directly through the block without modification. If the M/S block is not active, all spectra are passed through this block unmodified.

The prediction tool reverses the prediction process carried out at the encoder. This prediction process re-inserts the redundancy that was extracted by the prediction tool at the encoder, under the control of the predictor state information. This tool is implemented as a second order backward adaptive predictor. The inputs to the prediction tool are:

- The predictor state information
- The scaled, inversely quantized spectra

The output from the prediction tool is:

- The scaled, inversely quantized spectra, after prediction is applied.

Note: If the prediction is disabled, the scaled, inversely quantized spectra are passed directly through the block without modification.

The intensity stereo tool implements intensity stereo decoding on pairs of spectra.

The inputs to the intensity stereo tool are:

- The inversely quantized spectra
- The intensity stereo control information

The output from the intensity stereo tool is:

- The inversely quantized spectra after intensity channel decoding.

Note: The scaled, inversely quantized spectra of individually coded channels are passed directly through this tool without modification, if intensity stereo is not indicated. The intensity stereo tool and M/S tool are arranged so that the operation of M/S and intensity stereo are mutually exclusive on any given scalefactor band and group of one pair of spectra.

The coupling tool for dependently switched coupling channels adds the relevant data from dependently switched coupling channels to the spectra, as directed by the coupling control information.

The inputs to the coupling tool are:

- The inversely quantized spectra
- The coupling control information

The output from the coupling tool is:

- The inversely quantized spectra coupled with the dependently switched coupling channels.

Note: The scaled, inversely quantized spectra are passed directly through this tool without modification, if coupling is not indicated. Depending on the coupling control information, dependently switched coupling channels might either be coupled before or after the TNS processing.

The coupling tool for independently switched coupling channels adds the relevant data from independently switched coupling channels to the time signal, as directed by the coupling control information.

The inputs to the coupling tool are:

- The time signal as output by the filterbank
- The coupling control information

The output from the coupling tool is:

- The time signal coupled with the independently switched coupling channels.

Note: The time signal is passed directly through this tool without modification, if coupling is not indicated.

The temporal noise shaping (TNS) tool implements a control of the fine time structure of the coding noise. In the encoder, the TNS process has flattened the temporal envelope of the signal to which it has been applied. In the decoder, the inverse process is used to restore the actual temporal envelope(s), under control of the TNS information. This is done by applying a filtering process to parts of the spectral data.

The inputs to the TNS tool are:

- The inversely quantized spectra
- The TNS information

The output from the TNS block is:

- The inversely quantized spectra

Note: If this block is disabled, the inversely quantized spectra are passed through without modification.

The filterbank / block switching tool applies the inverse of the frequency mapping that was carried out in the encoder. An inverse modified discrete cosine transform (IMDCT) is used for the filterbank tool. The IMDCT can be configured to support either one set of 128 or 1024, or four sets of 32 or 256 spectral coefficients.

The inputs to the filterbank tool are:

- The inversely quantized spectra
- The filterbank control information

The output(s) from the filterbank tool is (are):

- The time domain reconstructed audio signal(s).

When present, the gain control tool applies a separate time domain gain control to each of 4 frequency bands that have been created by the gain control PQF filterbank in the encoder. Then, it assembles the 4 frequency bands and reconstructs the time waveform through the gain control tool's filterbank.

The inputs to the gain control tool are:

- The time domain reconstructed audio signal(s)
- The gain control information

The output(s) from the gain control tool is (are):

- The time domain reconstructed audio signal(s)

If the gain control tool is not active, the time domain reconstructed audio signal(s) are passed directly from the filterbank tool to the output of the decoder. This tool is used for the scalable sampling rate (SSR) profile only.

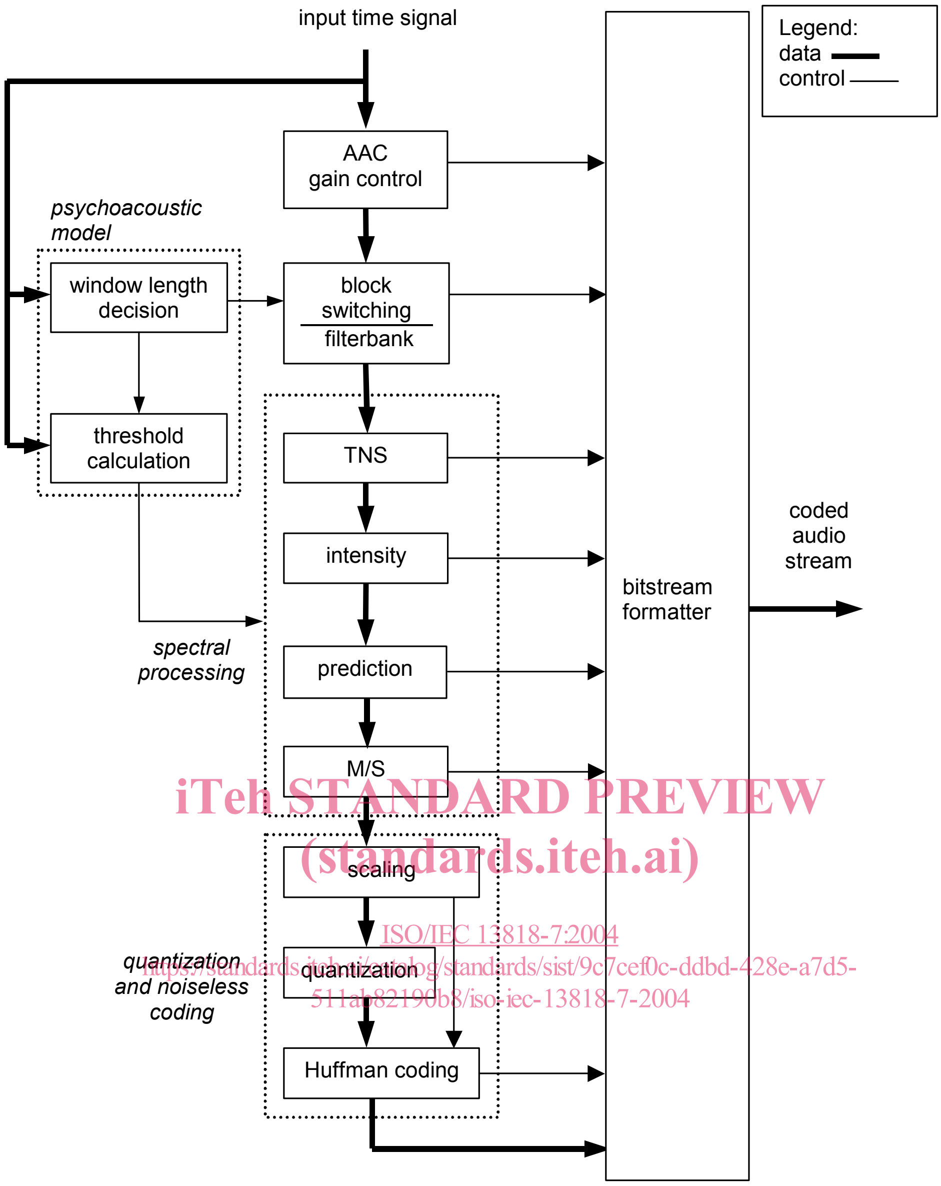

Figure 1 – MPEG-2 AAC Encoder Block Diagram

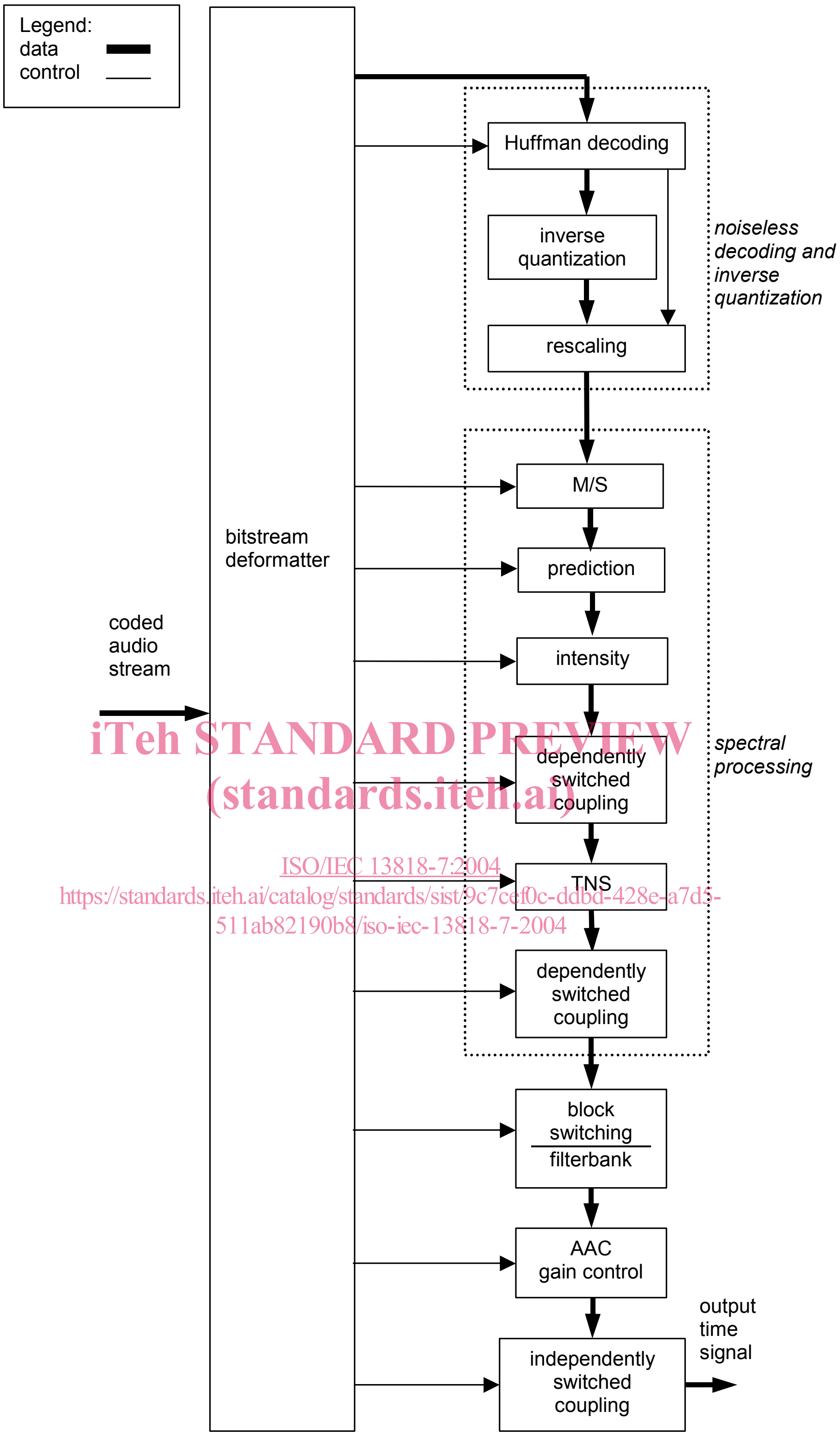

Figure 2 – MPEG-2 AAC Decoder Block Diagram

2 Normative References

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

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

ISO/IEC 13818-1, *Information technology — Generic coding of moving pictures and associated audio information — Part 1: Systems*

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

ISO/IEC 14496-3, *Information technology — Coding of audio-visual objects — Part 3: Audio*

3 Terms and Definitions

For the purposes of this part of ISO/IEC 13818, the following definitions apply:

3.1
access unit
in the case of compressed audio an access unit is an audio access unit.

3.2
alias
mirrored signal component resulting from sampling.

3.3
analysis filterbank
filterbank in the encoder that transforms a broadband PCM audio signal into a set of spectral coefficients.

3.4
ancillary data
part of the bitstream that might be used for transmission of ancillary data.

3.5
audio access unit
for AAC, an audio access unit is defined as the smallest part of the encoded bitstream which can be decoded by itself, where decoded means "fully reconstructed sound". Typically this is a segment of the encoded bitstream starting after the end of the byte containing the last bit of one ID_END id_syn_ele() through the end of the byte containing the last bit of the next ID_END id_syn_ele.

3.6
audio buffer
a buffer in the system target decoder (see ISO/IEC 13818-1) for storage of compressed audio data.

3.7
Bark
the Bark is the standard unit corresponding to one critical band width of human hearing.

3.8
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.

3.9
bitrate
the rate at which the compressed bitstream is delivered to the input of a decoder.

3.10
bitstream; stream
an ordered series of bits that forms the coded representation of the data.

3.11
bitstream verifier
a process by which it is possible to test and verify that all the requirements specified in this part of ISO/IEC 13818 are met by the bitstream.

3.12
block companding
normalising of the digital representation of an audio signal within a certain time period.

3.13
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 6.1) or the first bit in the syncword for the Audio Data Transport Stream Format (see subclause 6.2).

3.14
Byte
sequence of 8-bits.

3.15
centre channel
an audio presentation channel used to stabilise the central component of the frontal stereo image.

3.16
channel
a sequence of data representing an audio signal intended to be reproduced at one listening position.

3.17
coded audio bitstream
a coded representation of an audio signal.

3.18
coded representation
a data element as represented in its encoded form.

3.19
compression:
reduction in the number of bits used to represent an item of data.

3.20
constant bitrate
operation where the bitrate is constant from start to finish of the coded bitstream.