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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)

Reference number
ISO/IEC 13818-7:2006(E)

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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 2.

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.

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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 fourth edition cancels and replaces the third edition (ISO 13818-7:2004), which has been technically revised. It also incorporates the Technical Corrigendum ISO/IEC 13818-7:2004/Cor.1:2005.

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* [Technical Report]

— *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

1.1 General

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

1.2 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, and
- 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, and
- The noiselessly coded spectra.

The outputs of the Noiseless Decoding tool are:

- The decoded integer representation of the scalefactors, and
- 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, and
- 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, and
- 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, and
- 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, and
- 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, and
- 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, and
- 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, and
- 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, and
- 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 four frequency bands that have been created by the gain control PQF filterbank in the encoder. Then, it assembles four 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), and
- 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.

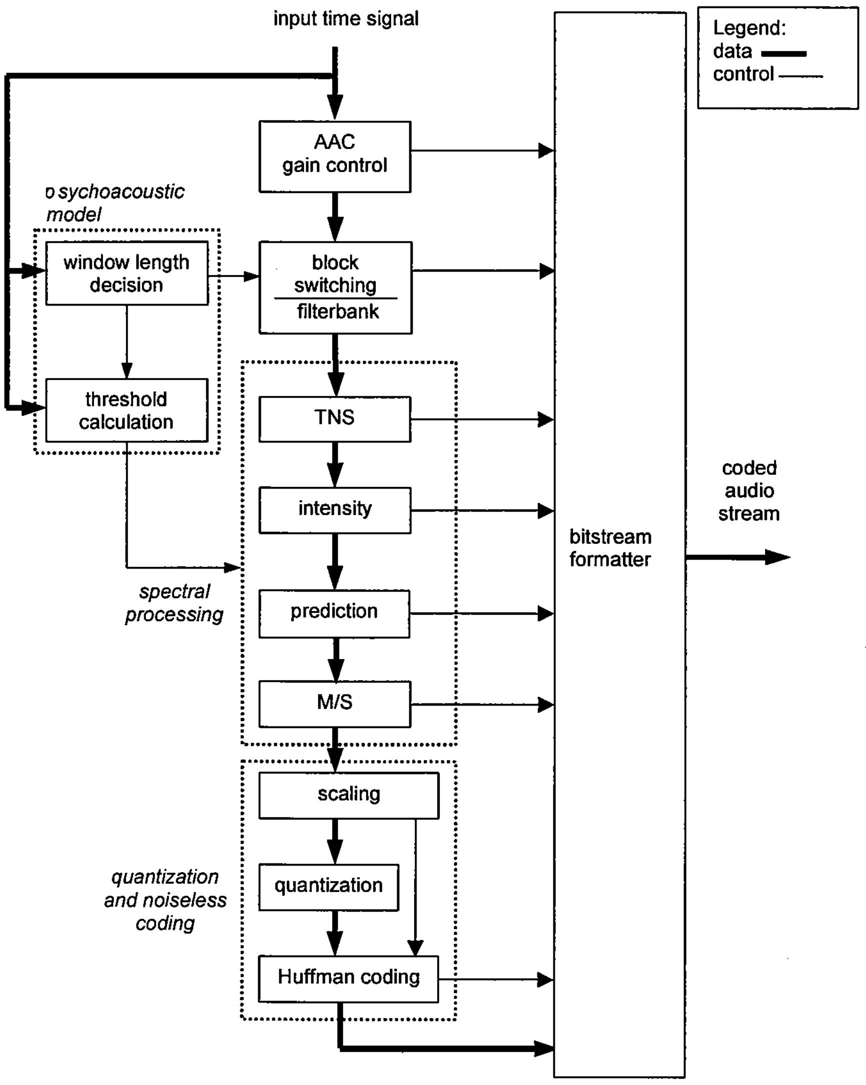


Figure 1 — MPEG-2 AAC Encoder Block Diagram

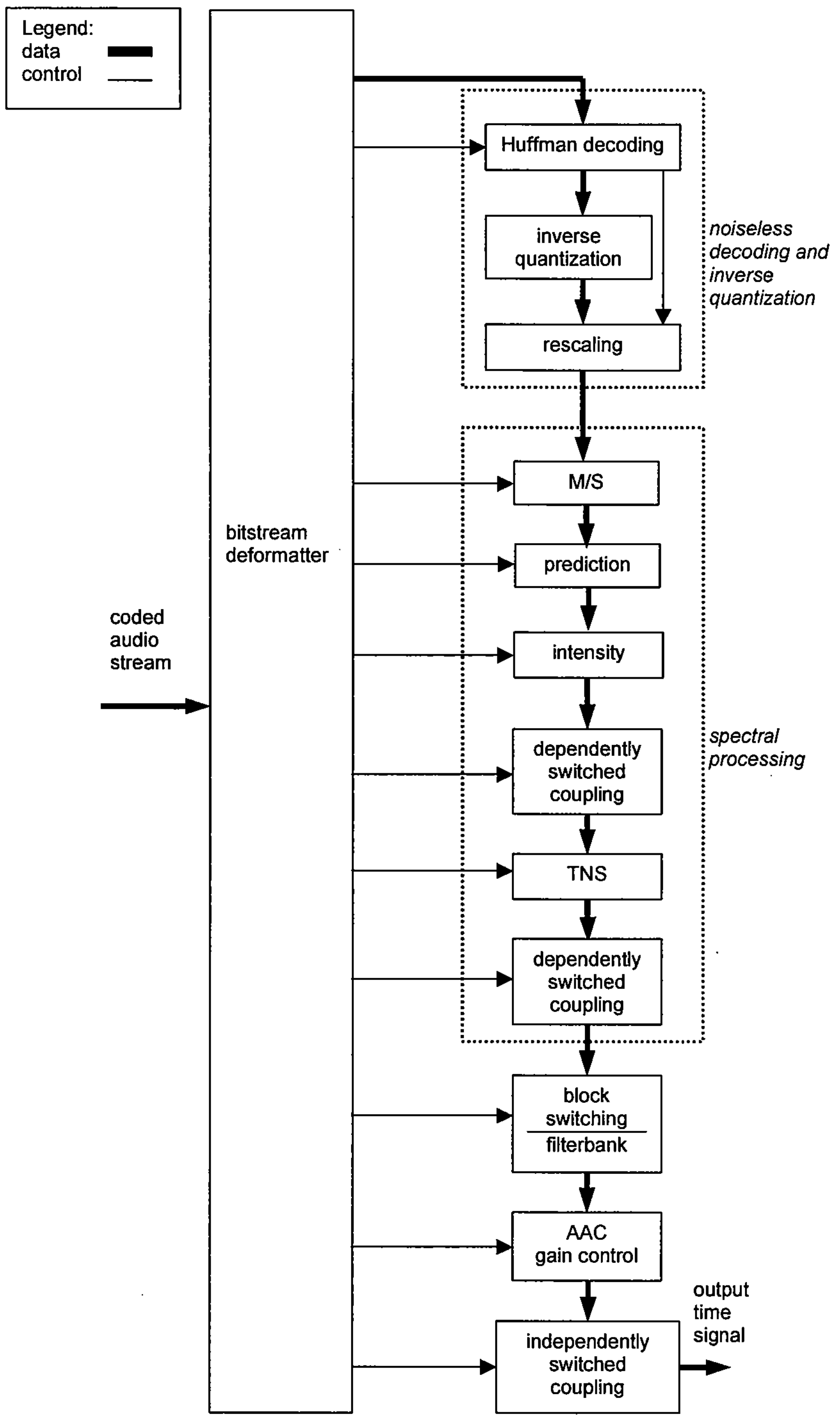


Figure 2 — MPEG-2 AAC Decoder Block Diagram

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 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, the smallest part of the encoded bitstream which can be decoded by itself, where decoded means "fully reconstructed sound"

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

3.7
bark
standard unit corresponding to one critical band width of human hearing

3.8
backward compatibility
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
rate at which the compressed bitstream is delivered to the input of a decoder

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

3.11
bitstream verifier
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
normalizing of the digital representation of an audio signal within a certain time period

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

3.14
byte
sequence of 8 bits

3.15
centre channel
audio presentation channel used to stabilize the central component of the frontal stereo image

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

3.17
coded audio bitstream
coded representation of an audio signal

3.18
coded representation
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 in which the bitrate is constant from start to finish of the coded bitstream

3.21
CRC
Cyclic Redundancy Check to verify the correctness of data

3.22
critical band
unit of bandwidth which represents the standard unit of bandwidth expressed in human auditory terms, corresponding to a fixed length on the human cochlea, approximately equal to 100 Hz at low frequencies and 1/3 octave at higher frequencies, above approximately 700 Hz

3.23
data element
item of data as represented before encoding and after decoding

3.24
decoded stream
decoded reconstruction of a compressed bitstream

3.25
decoder
embodiment of a decoding process

3.26
decoding (process)
process defined in this part of ISO/IEC 13818 that reads an input coded bitstream and outputs decoded audio samples

3.27
digital storage media
DSM
digital storage or transmission device or system

3.28
discrete cosine transform
DCT
either the forward discrete cosine transform or the inverse discrete cosine transform, an invertible, discrete orthogonal transformation

3.29
downmix
matrixing of n channels to obtain less than n channels

3.30
editing
process by which one or more coded bitstreams are manipulated to produce a new coded bitstream

NOTE Conforming edited bitstreams are defined in this part of ISO/IEC 13818.

3.31
encoder
embodiment of an encoding process

3.32
encoding (process)
process, not specified in ISO/IEC 13818, that reads a stream of input audio samples and produces a valid coded bitstream as defined in this part of ISO/IEC 13818

3.33
entropy coding
variable length lossless coding of the digital representation of a signal to reduce statistical redundancy

3.34
Fast Fourier Transformation
FFT
fast algorithm for performing a discrete Fourier transform (an orthogonal transform)

3.35
filterbank
set of band-pass filters covering the entire audio frequency range

3.36
flag
variable which can take one of only the two values defined in this part of ISO/IEC 13818

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

3.38
frame
part of the audio signal that corresponds to audio PCM samples from an audio access unit

3.39
Fs
sampling frequency

3.40
Hann window
time function applied sample-by-sample to a block of audio samples before Fourier transformation

3.41
Huffman coding
specific method for entropy coding

3.42
hybrid filterbank
serial combination of subband filterbank and MDCT

3.43
IDCT
Inverse Discrete Cosine Transform

3.44
IMDCT
Inverse Modified Discrete Cosine Transform

3.45
intensity stereo
method of exploiting stereo irrelevance or redundancy in stereophonic audio programmes based on retaining at high frequencies only the energy envelope of the right and left channels

3.46
joint stereo coding
any method that exploits stereophonic irrelevance or stereophonic redundancy

3.47
joint stereo mode
mode of the audio coding algorithm using joint stereo coding

3.48
low frequency enhancement (LFE) channel
limited bandwidth channel for low frequency audio effects in a multichannel system

3.49
main audio channels
all channels represented by either single_channel_element()'s (see 8.2.1) or channel_pair_element()´s (see 8.2.1)

3.50
mapping
conversion of an audio signal from time to frequency domain by subband filtering and/or by MDCT

3.51
masking
property of the human auditory system by which an audio signal cannot be perceived in the presence of another audio signal

3.52
masking threshold
function in frequency and time below which an audio signal cannot be perceived by the human auditory system

3.53
modified discrete cosine transform
MDCT
transform which has the property of time domain aliasing cancellation

NOTE An analytical espression for the MDCT can be found in C.3.1.2.

3.54
M/S stereo
method of removing imaging artefacts as well as exploiting stereo irrelevance or redundancy in stereophonic audio programmes based on coding the sum and difference signal instead of the left and right channels

3.55
multichannel
combination of audio channels used to create a spatial sound field

3.56
multilingual
presentation of dialogue in more than one language

3.57
non-tonal component
noise-like component of an audio signal

3.58
Number of Considered Channels
NCC
number of channels represented by the elements SCE, independently switched CCE and CPE, i.e. once the number of SCEs plus once the number of independently switched CCEs plus twice the number of CPEs, with respect to the naming conventions of the MPEG-AAC decoders and bitstreams, NCC=A+I

NOTE This number is used to derive the required decoder input buffer size (see 8.2.3).

3.59
Nyquist sampling
sampling at or above twice the maximum bandwidth of a signal

3.60
padding
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

3.61
parameter
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

3.62
parser
functional stage of a decoder which extracts from a coded bitstream a series of bits representing coded elements

3.63
polyphase filterbank
set of equal bandwidth filters with special phase interrelationships, allowing for an efficient implementation of the filterbank

3.64
prediction error
difference between the actual value of a sample or data element and its predictor

3.65
prediction
use of a predictor to provide an estimate of the sample value or data element currently being decoded

3.66
predictor
linear combination of previously decoded sample values or data elements

3.67
presentation channel
audio channel at the output of the decoder

3.68
presentation unit
in the case of compressed audio, a decoded audio access unit

3.69
program
set of main audio channels, coupling_channel_element()'s (see 8.2.1), lfe_channel_element()'s (see 8.2.1), and associated data streams intended to be decoded and played back simultaneously

NOTE A program may be defined by default (see 8.5.3.1 and 8.5.3.3) or specifically by a program_config_element() (see 8.5.3.2). A given single_channel_element() (see 8.2.1), channel_pair_element() (see 8.2.1), coupling_channel_element(), lfe_channel_element() or data channel may accompany one or more programs in any given bitstream.

3.70
psychoacoustic model
mathematical model of the masking behaviour of the human auditory system

3.71
random access
process of beginning to read and decode the coded bitstream at an arbitrary point

3.72
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

3.73
sampling frequency
Fs
rate in Hertz which is used to digitize an audio signal during the sampling process

3.74
scalefactor
factor by which a set of values is scaled before quantization

3.75
scalefactor band
set of spectral coefficients which are scaled by one scalefactor

3.76
scalefactor index
numerical code for a scalefactor

3.77
side information
information in the bitstream necessary for controlling the decoder

3.78
spectral coefficients
discrete frequency domain data output from the analysis filterbank

3.79
spreading function
function that describes the frequency spread of masking effects

3.80
stereo-irrelevant
portion of a stereophonic audio signal which does not contribute to spatial perception

3.81
stuffing (bits)
stuffing (bytes)
code words that may be inserted at particular locations in the coded bitstream that are discarded in the decoding process whose purpose is to increase the bitrate of the stream which would otherwise be lower than the desired bitrate

3.82
surround channel
audio presentation channel added to the front channels (L and R or L, R, and C) to enhance the spatial perception

3.83
Syncword
a 12-bit code embedded in the audio bitstream that identifies the start of a adts_frame() (see 6.2, Table 5)

3.84
synthesis filterbank
filterbank in the decoder that reconstructs a PCM audio signal from subband samples

3.85
tonal component
sinusoid-like component of an audio signal

3.86
variable bitrate
operation in which the bitrate varies with time during the decoding of a coded bitstream

3.87
variable length coding
reversible procedure for coding that assigns shorter code words to frequent symbols and longer code words to less frequent symbols

3.88
variable length code
VLC
code word assigned by variable length encoder (see variable length coding)

3.89
variable length decoder
procedure to obtain the symbols encoded with a variable length coding technique

3.90
variable length encoder
procedure to assign variable length codewords to symbols

4 Symbols and Abbreviations

The mathematical operators used to describe this International Standard 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 twos-complement representation of integers. Numbering and counting loops generally begin from zero.

4.1 Arithmetic Operators

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

| |	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.	

4.2 Logical Operators

||	Logical OR.
&&	Logical AND.
!	Logical NOT

4.3 Relational Operators

>	Greater than.
>=	Greater than or equal to.
<	Less than.
<=	Less than or equal to.
==	Equal to.
!=	Not equal to.
max [,...,]	the maximum value in the argument list.
min [,...,]	the minimum value in the argument list.

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.

4.5 Assignment

=	Assignment operator.

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 13818. 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.

4.7 Constants

π	3.14159265358...
e	2.71828182845...

5 Method of Describing Bitstream Syntax

The bitstream retrieved by the decoder is described in clause 6. Each data item in the bitstream is in bold type. It is described by

- its name;
- its length in bits, where "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;
- a mnemonic for its type and order of transmission.

The action caused by a decoded data element in a bitstream depends on the value of that data element and on data elements previously decoded. The decoding of the data elements and the definition of the state variables used in their decoding are described in the clauses following the syntax clause. 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.

```
switch (expression) {
    case const-expr:
        data_element;
         break;
    case const-expr:
        data_element;
}
```

If the condition formed by the comparison of expression and const-expr. is true, then the data stream continues with the subsequent data elements. An optionally break statement can be used to immediately leave the switch, data elements beyond a break do not occur 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.
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 clause 6 implements a satisfactory decoding procedure. In particular, it defines a correct and error-free input bitstream. Actual decoders must include a means to look for start codes in order to begin decoding correctly.

Definition of nextbits function

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

6 Syntax

6.1 Audio Data Interchange Format, ADIF

Table 2 — Syntax of adif_sequence()

Syntax	No. of bits	Mnemonic
adif_sequence()		
{		
adif_header();		
byte_alignment();		
raw_data_stream();		
}		

Table 3 — 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**
if (bitstream_type == '0') {		
adif_buffer_fullness;	**20**	**uimsbf**
}		
for (i = 0; i < num_program_config_elements + 1; i++) {		
program_config_element();		
}		
}		

6.2 Audio Data Transport Stream, ADTS

Table 4 — Syntax of adts_sequence()

Syntax	No. of bits	Mnemonic
adts_sequence()		
{		
while (nextbits() == syncword) {		
adts_frame();		
}		
}		

Table 5 — Syntax of adts_frame()

Syntax	No. of bits	Mnemonic
adts_frame()		
{		
adts_fixed_header();		
adts_variable_header();		
if (number_of_raw_data_blocks_in_frame == 0) {		
adts_error_check();		
raw_data_block();		
}		
else {		
adts_header_error_check();		
for (i = 0; i <= number_of_raw_data_blocks_in_frame;		
i++) {		
raw_data_block();		
adts_raw_data_block_error_check();		
}		
}		
}		

Table 6 — Syntax of adts_header_error_check()

Syntax	No. of bits	Mnemonic
adts_header_error_check ()		
{		
if (protection_absent == '0') {		
for (i = 1; i <= number_of_raw_data_blocks_in_frame; i++) {		
raw_data_block_position[i];	**16**	**uimsfb**
}		
crc_check;	**16**	**rpchof**
}		
}		

Table 7 — Syntax of adts_raw_data_block_error_check()

Syntax	No. of bits	Mnemonic
adts_raw_data_block_error_check()		
{		
if (protection_absent == '0')		
crc_check;	**16**	**rpchof**
}		

6.2.1 Fixed Header of ADTS

Table 8 — 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;	**2**	**uimsbf**
sampling_frequency_index;	**4**	**uimsbf**
private_bit;	**1**	**bslbf**
channel_configuration;	**3**	**uimsbf**
original_copy;	**1**	**bslbf**
home;	**1**	**bslbf**
}		

6.2.2 Variable Header of ADTS

Table 9 — 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**
number_of_raw_data_blocks_in_frame;	**2**	**uimsfb**
}		

6.2.3 Error Detection

Table 10 — Syntax of adts_error_check()

Syntax	No. of bits	Mnemonic
adts_error_check()		
{		
if (protection_absent == '0')		
crc_check;	**16**	**rpchof**
}		

6.3 Raw Data

Table 11 — Syntax of raw_data_stream()

Syntax	No. of bits	Mnemonic
raw_data_stream()		
{		
while (data_available()) {		
raw_data_block();		
}		
}		

Table 12 — Syntax of raw_data_block()

Syntax	No. of bits	Mnemonic
raw_data_block()		
{		
while ((id = **id_syn_ele**) != ID_END) {	**3**	**uimsbf**
switch (id) {		
case ID_SCE: single_channel_element();		
break;		
case ID_CPE: channel_pair_element();		
break;		
case ID_CCE: coupling_channel_element();		
break;		
case ID_LFE: lfe_channel_element();		
break;		
case ID_DSE: data_stream_element();		
break;		
case ID_PCE: program_config_element();		
break;		
case ID_FIL: fill_element();		
}		
}		
byte_alignment();		
}		

Table 13 — Syntax of single_channel_element()

Syntax	No. of bits	Mnemonic
single_channel_element()		
{		
element_instance_tag;	**4**	**uimsbf**
individual_channel_stream(0);		
}		

Table 14 — Syntax of channel_pair_element()

Syntax	No. of bits	Mnemonic
channel_pair_element()		
{		
element_instance_tag;	**4**	**uimsbf**
common_window;	**1**	**uimsbf**
if (common_window) {		
ics_info();		
ms_mask_present;	**2**	**uimsbf**
if (ms_mask_present == 1) {		
for (g = 0; g < num_window_groups; g++) {		
for (sfb = 0; sfb < max_sfb; sfb++) {		
ms_used[g][sfb];	**1**	**uimsbf**
}		
}		
}		
}		
individual_channel_stream(common_window);		
individual_channel_stream(common_window);		
}		

Table 15 — Syntax of ics_info()

Syntax	No. of bits	Mnemonic
ics_info()		
{		
ics_reserved_bit;	**1**	**bslbf**
window_sequence;	**2**	**uimsbf**
window_shape;	**1**	**uimsbf**
if (window_sequence == EIGHT_SHORT_SEQUENCE) {		
max_sfb;	**4**	**uimsbf**
scale_factor_grouping;	**7**	**uimsbf**
}		
else {		
max_sfb;	**6**	**uimsbf**
predictor_data_present;	**1**	**uimsbf**
if (predictor_data_present) {		
predictor_reset;	**1**	**uimsbf**
if (predictor_reset) {		
predictor_reset_group_number;	**5**	**uimsbf**
}		
for (sfb = 0; sfb < min(max_sfb,		
PRED_SFB_MAX); sfb++) {		
prediction_used[sfb];	**1**	**uimsbf**
}		
}		
}		
}		

Table 16 — Syntax of individual_channel_stream()

Syntax	No. of bits	Mnemonic
individual_channel_stream(common_window)		
{		
global_gain;	**8**	**uimsbf**
if (!common_window)		
ics_info();		
section_data();		
scale_factor_data();		
pulse_data_present;	**1**	**uismbf**
if (pulse_data_present) {		
pulse_data();		
}		
tns_data_present;	**1**	**uimsbf**
if (tns_data_present) {		
tns_data();		
}		
gain_control_data_present;	**1**	**uimsbf**
if (gain_control_data_present) {		
gain_control_data();		
}		
spectral_data();		
}		

Table 17 — Syntax of section_data()

Syntax	No. of bits	Mnemonic
section_data()		
{		
if (window_sequence == EIGHT_SHORT_SEQUENCE)		
sect_esc_val = (1<<3) - 1;		
else		
sect_esc_val = (1<<5) - 1;		
for (g = 0; g < num_window_groups; g++) {		
k = 0;		
i = 0;		
while (k < max_sfb) {		
sect_cb[g][i];	**4**	**uimsbf**
sect_len = 0;		
while (**sect_len_incr** == sect_esc_val) {	**{3;5}**	**uimsbf**
sect_len += sect_esc_val;		
}		
sect_len += sect_len_incr;		
sect_start[g][i] = k;		
sect_end[g][i] = k+sect_len;		
for (sfb = k; sfb < k+sect_len; sfb++)		
sfb_cb[g][sfb] = sect_cb[g][i];		
k += sect_len;		
i++;		
}		
num_sec[g] = i;		
}		
}		

Table 18 — Syntax of scale_factor_data()

Syntax	No. of bits	Mnemonic
scale_factor_data()		
{		
for (g = 0; g < num_window_groups; g++) {		
for (sfb = 0; sfb < max_sfb; sfb++) {		
if (sfb_cb[g][sfb] != ZERO_HCB) {		
if (is_intensity(g,sfb))		
hcod_sf[dpcm_is_position[g][sfb]];	**1..19**	**vlclbf**
else		
hcod_sf[dpcm_sf[g][sfb]];	**1..19**	**vlclbf**
}		
}		
}		
}		

Table 19 — Syntax of tns_data()

Syntax	No. of bits	Mnemonic
tns_data()		
{		
for (w = 0; w < num_windows; w++) {		
n_filt[w];	**1..2**	**uimsbf**
if (n_filt[w])		
coef_res[w];	**1**	**uimsbf**
for (filt = 0; filt < n_filt[w]; filt++) {		
length[w][filt];	**{4;6}**	**uimsbf**
order[w][filt];	**{3;5}**	**uimsbf**
if (order[w][filt]) {		
direction[w][filt];	**1**	**uimsbf**
coef_compress[w][filt];	**1**	**uimsbf**
for (i = 0; i < order[w][filt]; i++)		
coef[w][filt][i];	**2..4**	**uimsbf**
}		
}		
}		
}		

Table 20 — Syntax of spectral_data()

Syntax	No. of bits	Mnemonic
spectral_data()		
{		
for (g = 0; g < num_window_groups; g++) {		
for (i = 0; i < num_sec[g]; i++) {		
if (sect_cb[g][i] != ZERO_HCB &&		
sect_cb[g][i] <= ESC_HCB) {		
for (k = sect_sfb_offset[g][sect_start[g][i]];		
k < sect_sfb_offset[g][sect_end[g][i]];) {		
if (sect_cb[g][i]<FIRST_PAIR_HCB) {		
hcod[sect_cb[g][i]][w][x][y][z];	**1..16**	**vlclbf**
if (unsigned_cb[sect_cb[g][i]])		
quad_sign_bits;	**0..4**	**bslbf**
k += QUAD_LEN;		
}		
else {		
hcod[sect_cb[g][i]][y][z];	**1..15**	**vlclbf**
if (unsigned_cb[sect_cb[g][i]])		
pair_sign_bits;	**0..2**	**bslbf**
k += PAIR_LEN;		
if (sect_cb[g][i] == ESC_HCB) {		
if (y == ESC_FLAG)		
hcod_esc_y;	**5..21**	**vlclbf**
if (z == ESC_FLAG)		
hcod_esc_z;	**5..21**	**vlclbf**
}		
}		
}		
}		
}		
}		
}		

Table 21 — Syntax of pulse_data()

Syntax	No. of bits	Mnemonic
pulse_data() {		
number_pulse;	**2**	**uimsbf**
pulse_start_sfb;	**6**	**uimsbf**
for (i = 0; i < number_pulse+1; i++) {		
pulse_offset[i];	**5**	**uimsbf**
pulse_amp[i];	**4**	**uimsbf**
}		
}		

Table 22 — Syntax of coupling_channel_element()

Syntax	No. of bits	Mnemonic
coupling_channel_element()		
{		
element_instance_tag;	**4**	**uimsbf**
ind_sw_cce_flag;	**1**	**uimsbf**
num_coupled_elements;	**3**	**uimsbf**
num_gain_element_lists = 0;		
for (c = 0; c < num_coupled_elements+1; c++) {		
num_gain_element_lists++;		
cc_target_is_cpe[c];	**1**	**uimsbf**
cc_target_tag_select[c];	**4**	**uimsbf**
if (cc_target_is_cpe[c]) {		
cc_l[c];	**1**	**uimsbf**
cc_r[c];	**1**	**uimsbf**
if (cc_l[c] && cc_r[c])		
num_gain_element_lists++;		
}		
}		
cc_domain;	**1**	**uimsbf**
gain_element_sign;	**1**	**uimsbf**
gain_element_scale;	**2**	**uimsbf**
individual_channel_stream(0);		
for (c = 1; c < num_gain_element_lists; c++) {		
if (ind_sw_cce_flag) {		
cge = 1;		
} else {		
common_gain_element_present[c];	**1**	**uimsbf**
cge = common_gain_element_present[c];		
}		
if (cge)		
hcod_sf[common_gain_element[c]];	**1..19**	**vlclbf**
else {		
for (g = 0; g < num_window_groups; g++) {		
for (sfb = 0; sfb < max_sfb; sfb++) {		
if (sfb_cb[g][sfb] != ZERO_HCB);		
hcod_sf[dpcm_gain_element[c][g][sfb]];	**1..19**	**vlclbf**
}		
}		
}		
}		
}		

Table 23 — Syntax of lfe_channel_element()

Syntax	No. of bits	Mnemonic
lfe_channel_element()		
{		
element_instance_tag;	**4**	**uimsbf**
individual_channel_stream(0);		
}		

Table 24 — Syntax of data_stream_element()

Syntax	No. of bits	Mnemonic
data_stream_element()		
{		
element_instance_tag;	**4**	**uimsbf**
data_byte_align_flag;	**1**	**uimsbf**
cnt = **count;**	**8**	**uimsbf**
if (cnt == 255) {		
cnt += **esc_count**;	**8**	**uimsbf**
}		
if (data_byte_align_flag) {		
byte_alignment();		
}		
for (i = 0; i < cnt; i++) {		
data_stream_byte[element_instance_tag][i];	**8**	**uimsbf**
}		
}		

Table 25 — Syntax of program_config_element()

Syntax	No. of bits	Mnemonic
program_config_element()		
{		
element_instance_tag;	**4**	**uimsbf**
profile;	**2**	**uimsbf**
sampling_frequency_index;	**4**	**uimsbf**
num_front_channel_elements;	**4**	**uimsbf**
num_side_channel_elements;	**4**	**uimsbf**
num_back_channel_elements;	**4**	**uimsbf**
num_lfe_channel_elements;	**2**	**uimsbf**
num_assoc_data_elements;	**3**	**uimsbf**
num_valid_cc_elements;	**4**	**uimsbf**
mono_mixdown_present;	**1**	**uimsbf**
if (mono_mixdown_present == 1)		
mono_mixdown_element_number;	**4**	**uimsbf**
stereo_mixdown_present;	**1**	**uimsbf**
if (stereo_mixdown_present == 1)		
stereo_mixdown_element_number;	**4**	**uimsbf**
matrix_mixdown_idx_present;	**1**	**uimsbf**
if (matrix_mixdown_idx_present == 1) {		
matrix_mixdown_idx ;	**2**	**uimsbf**
pseudo_surround_enable;	**1**	**uimsbf**
}		
for (i = 0; i < num_front_channel_elements; i++) {		
front_element_is_cpe[i];	**1**	**bslbf**
front_element_tag_select[i];	**4**	**uimsbf**
}		
for (i = 0; i < num_side_channel_elements; i++) {		
side_element_is_cpe[i];	**1**	**bslbf**
side_element_tag_select[i];	**4**	**uimsbf**
}		
for (i = 0; i < num_back_channel_elements; i++) {		
back_element_is_cpe[i];	**1**	**bslbf**
back_element_tag_select[i];	**4**	**uimsbf**
}		
for (i = 0; i < num_lfe_channel_elements; i++)		
lfe_element_tag_select[i];	**4**	**uimsbf**
for (i = 0; i < num_assoc_data_elements; i++)		
assoc_data_element_tag_select[i];	**4**	**uimsbf**
for (i = 0; i < num_valid_cc_elements; i++) {		
cc_element_is_ind_sw[i];	**1**	**uimsbf**
valid_cc_element_tag_select[i];	**4**	**uimsbf**
}		
byte_alignment();		
comment_field_bytes;	**8**	**uimsbf**
for (i = 0; i < comment_field_bytes; i++)		
comment_field_data[i];	**8**	**uimsbf**
}		

Table 26 — Syntax of fill_element()

Syntax	No. of bits	Mnemonic
fill_element()		
{		
cnt = **count;**	**4**	**uimsbf**
if (cnt == 15)		
cnt += **esc_count** - 1;	**8**	**uimsbf**
while (cnt > 0) {		
cnt -= extension_payload(cnt);		
}		
}		

Table 27 — Syntax of gain_control_data()

Syntax	No. of bits	Mnemonic
gain_control_data()		
{		
max_band;	**2**	**uimsbf**
if (window_sequence == ONLY_LONG_SEQUENCE) {		
for (bd = 1; bd <= max_band; bd++) {		
for (wd = 0; wd < 1; wd++) {		
adjust_num[bd][wd];	**3**	**uimsbf**
for (ad = 0; ad < adjust_num[bd][wd]; ad++) {		
alevcode[bd][wd][ad];	**4**	**uimsbf**
aloccode[bd][wd][ad];	**5**	**uimsbf**
}		
}		
}		
}		
else if (window_sequence == LONG_START_SEQUENCE)		
{		
for (bd = 1; bd <= max_band; bd++) {		
for (wd = 0; wd < 2; wd++) {		
adjust_num[bd][wd];	**3**	**uimsbf**
for (ad = 0; ad < adjust_num[bd][wd]; ad++) {		
alevcode[bd][wd][ad];	**4**	**uimsbf**
if (wd == 0)		
aloccode[bd][wd][ad];	**4**	**uimsbf**
else		
aloccode[bd][wd][ad];	**2**	**uimsbf**
}		
}		
}		
}		
else if (window_sequence ==		
EIGHT_SHORT_SEQUENCE) {		
for (bd = 1; bd <= max_band; bd++) {		
for (wd = 0; wd < 8; wd++) {		
adjust_num[bd][wd];	**3**	**uimsbf**
for (ad = 0; ad < adjust_num[bd][wd]; ad++) {		
alevcode[bd][wd][ad];	**4**	**uimsbf**
aloccode[bd][wd][ad];	**2**	**uimsbf**
}		
}		
}		
}		
else if (window_sequence == LONG_STOP_SEQUENCE) {		
for (bd = 1; bd <= max_band; bd++) {		
for (wd = 0; wd < 2; wd++) {		
adjust_num[bd][wd];	**3**	**uimsbf**
for (ad = 0; ad < adjust_num[bd][wd]; ad++) {		
alevcode[bd][wd][ad];	**4**	**uimsbf**
if (wd == 0)		
aloccode[bd][wd][ad];	**4**	**uimsbf**
else		
aloccode[bd][wd][ad];	**5**	**uimsbf**
}		
}		
}		
}		
}		

Table 28 — Syntax of extension_payload()

```
extension_payload(cnt)
{
    extension_type;                                              4       uimsbf
    switch (extension_type) {
        case EXT_DYNAMIC_RANGE:
                n = dynamic_range_info();
                return n;
        case EXT_SBR_DATA:
                return sbr_extension_data(id_aac, 0);                    Note 1
        case EXT_SBR_DATA_CRC:
                return sbr_extension_data(id_aac, 1);                    Note 1
        case EXT_FILL_DATA:
                fill_nibble;        /* must be '0000' */         4       uimsbf
                for (i = 0; i < cnt-1; i++)
                    fill_byte[i];   /* must be '10100101' */     8       uimsbf
                return cnt;
        case default:
                for (i = 0; i < 8*(cnt-1)+4; i++)
                    other_bits[i];                               1       uimsbf
                return cnt;
    }
}
```

Note 1: id_aac is the id_syn_ele of the corresponding AAC element (ID_SCE or ID_CPE) or ID_SCE in case of CCE.

Table 29 — Syntax of dynamic_range_info()

Syntax	No. of bits	Mnemonic
dynamic_range_info()		
{		
n = 1;		
drc_num_bands = 1;		
pce_tag_present;	**1**	**uimsbf**
if (pce_tag_present == 1) {		
pce_ instance_tag;	**4**	**uimsbf**
drc_tag_reserved_bits;	**4**	
n++;		
}		
excluded_chns_present;	**1**	**uimsbf**
if (excluded_chns_present == 1) {		
n += excluded_channels();		
}		
drc_bands_present ;	**1**	**uimsbf**
if (drc_bands_present == 1) {		
drc_band_incr;	**4**	**uimsbf**
drc_bands_reserved_bits;	**4**	**uimsbf**
n++;		
drc_num_bands = drc_num_bands + drc_band_incr;		
for (i = 0; i < drc_num_bands; i++) {		
drc_band_top[i];	**8**	**uimsbf**
n++;		
}		
}		
prog_ref_level_present;	**1**	**uimsbf**
if (prog_ref_level_present == 1) {		
prog_ref_level;	**7**	**uimsbf**
prog_ref_level_reserved_bits;	**1**	**uimsbf**
n++;		
}		
for (i = 0; i < drc_num_bands; i++) {		
dyn_rng_sgn[i];	**1**	**uimsbf**
dyn_rng_ctl[i];	**7**	**uimsbf**
n++;		
}		
return n;		
}		

Table 30 — Syntax of excluded_channels()

Syntax	No. Of bits	Mnemonic
excluded_channels( )		
{		
n = 0;		
num_excl_chan = 70;		
for (i = 0; i < 7; i++)		
exclude_mask[i];	**1**	**uimsbf**
n++;		
while (**additional_excluded_chns[n-1]** == 1) {	**1**	**uimsbf**
for (i = num_excl_chan; i < num_excl_chan+7; i++)		
exclude_mask[i];	**1**	**uimsbf**
n++;		
num_excl_chan += 7;		
}		
return n;		
}		

7 Profiles and Profile Interoperability

7.1 Profiles

There are three profiles identified in the MPEG-2 AAC standard:

Main Profile

Low Complexity Profile

Scalable Sampling Rate Profile

In the program_config_element() and adts_fixed_header(), a two bit field indicates the profile in use:

Table 31 — Profiles

index	profile
0	Main profile
1	Low Complexity profile (LC)
2	Scalable Sampling Rate profile (SSR)
3	(reserved)

7.1.1 Main

The Main profile is used when memory cost is not significant, and when there is substantial processing power available. With the exception of the gain control tool, all parts of the tools may be used in order to provide the best data compression possible. There shall be only one program (in the sense of what is specified in a program_config_element()) in a Main profile bitstream. The program in a Main profile bitstream shall not contain any mono or stereo mixdown elements.

7.1.2 Low Complexity

The Low Complexity profile is used when RAM usage, processing power, and compression requirements are all present. In the low complexity profile, prediction, and gain control tool are not permitted and TNS order is limited. There shall be only one program (in the sense of what is specified in a program_config_element()) in a Low Complexity profile bitstream. The program in a Low Complexity profile bitstream shall not contain any mono or stereo mixdown elements.

7.1.3 Scalable Sampling Rate

In the Scalable Sampling Rate profile, the gain-control tool is required. Prediction and coupling channels are not permitted, and TNS order and bandwidth are limited. Gain control is not used in the lowest of the 4 PQF subbands. In the case of a reduced audio bandwidth, the SSR profile will scale accordingly in complexity. There shall be only one program (in the sense of what is specified in a program_config_element()) in a Scalable Sampling Rate profile bitstream. The program in a Scalable Sampling Rate profile bitstream shall not contain any mono or stereo mixdown elements.

7.1.4 Naming Convention for MPEG-2 AAC Decoders and Bitstreams

A decoder or bitstream may be specified as an A.L.I.D Channel <Profile Name> Profile MPEG-2 AAC decoder or bitstream, where A is replaced by the number of main audio channels, L by the number of LFE channels, I by the number of independently switched coupling channels, D by the number of dependently switched coupling channels, and Profile Name by the actual profile name. An example would be a 5.1.1.1 Channel Main Profile MPEG-2 AAC Decoder, indicating a decoder capable of decoding 5 main audio channels, one LFE channel, and one each independently and dependently switched CCE, with each of the channels using the profile specified. This can be abbreviated as M.5.1.1.1 where the "M" indicates a main profile decoder. Similarly, a Low Complexity decoder can be specified by a leading "L", and an SSR profile by an "S".

7.1.4.1 Naming Convention for MPEG-2 AAC + MPEG-4 SBR Decoders and Bitstreams

A decoder or bitstream conforming additionally to the MPEG-4 AOT SBR at a certain level may be referenced in a similar manner by appending "+ SBR / X [HQ/LP]" to the name, where X is replaced with the level of the HE-AAC profile decoder/bitstream with the same characteristics as specified by ISO/IEC 14496-3. An example would be a 5.1.1.1 Channel Main Profile MPEG-2 AAC + SBR / 5 HQ Decoder.

7.1.5 Minimum Decoder Capability for Specified Number of Main Audio Channels and Profile

To insure a certain level of interoperability the following minimum decoder capabilities for decoders of a given profile and number of main audio channels are specified.

Table 32 — Profile dependent minimum decoder capabilities in terms of channel configuration

Number of Main Audio Channels	Main Profile Capability	Low Complexity Profile Capability	SSR Profile Capability
1	1.0.0.0	1.0.0.0	1.0.0.0
2	2.0.0.0	2.0.0.0	2.0.0.0
3	3.0.1.0	3.0.0.1	3.0.0.0
4	4.0.1.0	4.0.0.1	4.0.0.0
5	5.1.1.1	5.1.0.1	5.1.0.0
7	7.1.1.2	7.1.0.2	7.1.0.0

7.1.6 Profile Dependent Tool Parameters

Maximum TNS order and bandwidth:

According to the profile in use, the value for the constant TNS_MAX_ORDER is set as follows for long windows: For the main profile the constant TNS_MAX_ORDER is 20, for the low complexity profile and the scalable sampling rate profile the constant TNS_MAX_ORDER is 12. For short windows, the constant TNS_MAX_ORDER is 7 for all profiles.

According to the sampling rate and profile in use, the value for the constant TNS_MAX_BANDS is set as follows:

Table 33 — Profile and sampling rate dependent definition of TNS_MAX_BANDS

Sampling Rate [Hz]	Low Complexity / Main Profile (long windows)	Low Complexity / Main Profile (short windows)	Scalable Sampling Rate Profile (long windows)	Scalable Sampling Rate Profile (short windows)
96000	31	9	28	7
88200	31	9	28	7
64000	34	10	27	7
48000	40	14	26	6
44100	42	14	26	6
32000	51	14	26	6
24000	46	14	29	7
22050	46	14	29	7
16000	42	14	23	8
12000	42	14	23	8
11025	42	14	23	8
8000	39	14	19	7

7.2 Profile Interoperability

7.2.1 Interoperability of Bitstreams and Decoders

Any bitstream of a given profile (see Table 34) whose number of main audio channels, LFE channels, independent coupling channels, and dependent coupling channels is less than or equal to the corresponding number of channels supported by a decoder of the same profile can be decoded by that decoder.

Table 34 describes the interoperability of the three profiles.

Table 34 — Profile Interoperability

	Encoder Profile		
Decoder Profile	Main Profile	LC Profile	SSR Profile
Main Profile	yes	yes	no *
LC Profile	no	yes	no *
SSR Profile	no	no **	yes

*In Table 34, these entries can be decoded if the main or LC profile decoder is able to parse, but not decode, the gain control information, but the reconstructed audio will have a limited bandwidth.

**In Table 33, this entry can be decoded, but the bandwidth of the decoded signal will be limited to approximately 5 kHz, corresponding to the nonaliased portion of the first PQMF filter band.

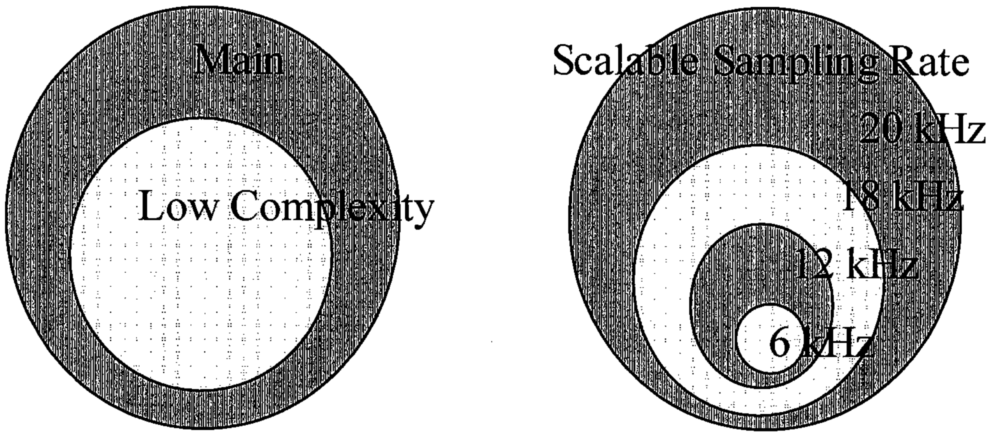


Figure 3 — Profile Interoperability

8 Overall Data Structure

8.1 AAC Interchange Formats

8.1.1 Overview

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.

For specific applications some or all of the syntax elements like those specified in the header of the ADIF, e.g. sampling_rate, may be known to the decoder by other means and hence do not appear in the bitstream.

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.

8.1.2 Audio Data Interchange Format (ADIF)

8.1.2.1 Overview

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 program_config_element()'s.

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.

8.1.2.2 Definitions

8.1.2.2.1 Data Functions

adif_sequence()	a sequence according to the Audio Data Interchange Format (Table 2).
adif_header()	header of the Audio Data Interchange Format located at the beginning of an adif_sequence (Table 3).
byte_alignment()	Align with respect to the first bit of the header.
raw_data_stream()	see subclause 8.2.1 and Table 11.
program_config_element()	contains information about the configuration for one program (Table 3). See subclause 8.5.

8.1.2.2.2 Data Elements

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 3).
copyright_id_present	indicates whether **copyright_id** is present or not (Table 3).
copyright_id	The field consists of an 8-bit copyright_identifier, followed by a 64-bit copyright_number (Table 3). 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, definition of data element **copyright_identification_bit**.
original_copy	see ISO/IEC 11172-3, definition of data element **copyright.**
home	see ISO/IEC 11172-3, definition of data element **original/copy.**
bitstream_type	a flag indicating the type of a bitstream (Table 3): ‘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 channels
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 3).
num_program_config_element	number of program_config_element()´s specified for this adif_sequence() is equal to num_program_config_element+1 (Table 3). The minimum value is 0 indicating 1 program_config_element().
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 (Table 3).

8.1.2.2.3 Help Elements

data_available()	Function that returns '1' as long as data is available, otherwise '0'.

8.1.3 Audio Data Transport Stream (ADTS)

8.1.3.1 Overview

The Audio Data Transport Stream (ADTS) is similar to syntax used in ISO/IEC 11172-3 and ISO/IEC 13818-3. This will be recognized by ISO/IEC 11172-3 and ISO/IEC 13818-3 decoders as a "Layer 4" bitstream.

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.

8.1.3.2 Definitions

8.1.3.2.1 Data Functions

adts_sequence()	a sequence according to Audio Data Transport Stream ADTS (Table 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_block()'s (Table 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 8).
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 9).
adts_error_check()	The following bits are protected and fed into the CRC algorithm in order of their appearance:

- all bits of adts_fixed_header()
- all bits of adts_variable_header()
- first 192 bits of any
  - single_channel_element()
  - channel_pair_element()
  - coupling_channel_element()
  - lfe_channel_element()
- First 128 bits of the second individual_channel_stream() in the channel_pair_element() must be protected.
- All information in any program_config_element() or data_stream_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.

	The id_syn_ele bits shall be excluded from CRC protection. 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, each time in order of their appearance. For example, if the second ICS starts at the 190th bit of CPE, the first 3 bits of the second ICS are protected twice. Finally, if the length of the second ICS is shorter than 128 bits, zero data are appended to achieve the length of 128 bits.
adts_header_error_check()	The following bits are protected and fed into the CRC algorithm in order of their appearance: • all bits of adts_fixed_header() • all bits of adts_variable_header() • all bits of every raw_data_block_position[i].
adts_raw_data_block_error_check()	With regard to the i-th adts_raw_data_block_error_check(), the bits of the i-th raw_data_block() are protected and fed into the CRC algorithm in order of their appearance according to what is specified with regard to the adts_error_check() with the exception that no header bits are considered.
raw_data_block()	see subclause 8.2.1 and Table 12.

8.1.3.2.2 Data Elements

raw_data_block_position[i]	Start position of the i-th raw_data_block() in the adts_frame(), measured as an offset in bytes from the start position of the first raw_data_block() in the adts_frame().
crc_check	CRC error detection data generated as described in ISO/IEC 11172-3, subclause 2.4.3.1 (Table 6, Table 7 and Table 10).
syncword	The bit string '1111 1111 1111'. See ISO/IEC 11172-3, subclause 2.4.2.3 (Table 8).
ID	MPEG identifier, set to '1'. See ISO/IEC 11172-3, subclause 2.4.2.3 (Table 8).
layer	Indicates which layer is used. Set to '00'. See ISO/IEC 11172-3, subclause 2.4.2.3 (Table 8).
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 (Table 8).
profile	profile used. See clause 2 (Table 8).
sampling_frequency_index	indicates the sampling frequency used according to the following table (Table 8):

Table 35 — Sampling frequency dependent on sampling_frequency_index

sampling_frequency_index	sampling frequeny [Hz]
0x0	96000
0x1	88200
0x2	64000
0x3	48000
0x4	44100
0x5	32000
0x6	24000
0x7	22050
0x8	16000
0x9	12000
0xa	11025
0xb	8000
0xc	reserved
0xd	reserved
0xe	reserved
0xf	reserved

private_bit	see ISO/IEC 11172-3, subclause 2.4.2.3 (Table 8).
channel_configuration	indicates the channel configuration used. If channel_configuration is greater than 0, the channel configuration is given in Table 42, see subclause 8.5.3.1. 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 syntactic element in the first raw_data_block() after the header (see subclause 8.5.3.2), or by the implicit configuration (see subclause 8.5.3.3) or must be known in the application (Table 8).
original_copy	see definition in 8.1.2.2.2.
home	see definition in 8.1.2.2.2.
copyright_identification_bit	One bit 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 (Table 9).
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 (Table 9).
aac_frame_length	Length of the frame including headers and error_check in bytes (Table 9).
adts_buffer_fullness	state of the bit reservoir in the course of encoding the ADTS frame, up to and including the first raw_data_block() and the

optionally following adts_raw_data_block_error_check(). It is transmitted as the number of available bits in the bit reservoir divided by NCC divided by 32 and truncated to an integer value (Table 9). A value of hexadecimal 7FF signals that the bitstream is a variable rate bitstream. In this case, buffer fullness is not applicable.

number_of_raw_data_blocks_in_frame	Number of raw_data_block()'s that are multiplexed in the adts_frame() is equal to number_of_raw_data_blocks_in_frame + 1. The minimum value is 0 indicating 1 raw_data_block() (Table 9).

8.2 Raw Data

8.2.1 Definitions

8.2.1.1 Data Functions

raw_data_stream()	sequence of raw_data_block()'s.
raw_data_block()	block of raw data that contains audio data for a time period of 1024 samples, related information and other data. There are seven syntactic elements, identified by the data element id_syn_ele. The audio_channel_element()'s in one raw_data_stream() and one raw_data_block() must have one and only one sampling rate. In the raw_data_block(), several instances of the same syntactic element may occur, but must have a different 4 bit element_instance_tag, except for data_stream_element()'s and fill_element()'s. Therefore, in one raw_data_block(), there can be from 0 to at most 16 instances of any syntactic element, except for data_stream_element()'s and fill_element()'s, where this limitation does not apply. If multiple data_stream_element()'s occur which have the same element_instance_tag then they are part of the same data stream. The fill_element() has no element_instance_tag (since the content does not require subsequent reference) and can occur any number of times. The end of a raw_data_block() is indicated with a special id_syn_ele (TERM), which may occur only once in a raw_data_block(). (Table 12).
single_channel_element()	abbreviaton SCE. Syntactic element of the bitstream containing coded data for a single audio channel. A single_channel_element() basically consists of an individual_channel_stream(). There may be up to 16 such elements per raw data block, each one must have a unique element_instance_tag (Table 13).
channel_pair_element()	abbreviation CPE. Syntactic element of the bitstream containing data for a pair of channels. A channel_pair_element() consists of two individual_channel_stream()'s and additional joint channel coding information. The two channels may share common side information. The channel_pair_element() has the same restrictions as the single channel element as far as element_instance_tag, and number of occurrances (Table 14).
coupling_channel_element()	Abbreviation CCE. Syntactic element that contains audio data for a coupling channel. A coupling channel represents the information for multi-channel intensity for one block, or alternately for dialogue for multilingual programming. The rules for number

	of coupling_channel_element()'s and instance tags are as for single_channel_element()'s (Table 22). See subclause 12.3.
lfe_channel_element()	Abbreviation LFE. Syntactic element that contains a low sampling frequency enhancement channel. The rules for the number of lfe_channel_element()'s and instance tags are as for single_channel_element()'s (Table 23). See subclause 8.4.
audio_channel_element**()**	generic term for single_channel_element(), channel_pair_element(), coupling_channel_element() and lfe_channel_element().
program_config_element()	Abbreviation PCE. Syntactic element that contains program configuration data. The rules for the number of program_config_element()'s and element_instance_tag's are the same as for single_channel_element()'s (Table 25). PCE's must come before all other syntactic elements in a raw_data_block(). See subclause 8.5.
fill_element()	Abbreviation FIL. Syntactic element that contains fill data. There may be any number of fill elements, that can come in any order in the raw data block (Table 26). See subclause 8.7.
data_stream_element()	Abbreviation DSE. Syntactic element that contains data. Again, there are 16 element_instance_tags. There is, however, no restriction on the number of data_stream_element()'s with any one instance tag, as a single data stream may continue across multiple data_stream_element()'s with the same instance tag (Table 24). See subclause 8.5.3.
byte_alignment()	Align with respect to the first bit of the raw_data_block().

8.2.2 Data Elements

id_syn_ele	a data element that identifies either a syntactic element or the end of a raw_data_block() (Table 12):

Table 36 — Syntaxtic element identification

ID name	encoding	Abbreviation	Syntactic Element
ID_SCE	0x0	SCE	single_channel_element()
ID_CPE	0x1	CPE	channel_pair_element()
ID_CCE	0x2	CCE	coupling_channel_element()
ID_LFE	0x3	LFE	lfe_channel_element()
ID_DSE	0x4	DSE	data_stream_element()
ID_PCE	0x5	PCE	program_config_element()
ID_FIL	0x6	FIL	fill_element()
ID_END	0x7	TERM	

element_instance_tag Unique instance tag for syntactic elements other than fill_element(). All syntactic elements containing instance tags may occur more than once, but, except for data_stream_element()'s, must have a unique element_instance_tag in each raw_data_block(). This tag is also used to reference audio syntactic elements in single_channel_element()'s, channel_pair_element()'s, lfe_channel_element()'s, data_channel_element()'s, and coupling_channel_element()'s inside a program_config_element(), and provides the possibility of up to 16 independent program_config_element()'s (Table 13, Table 14, Table 22, Table 23, Table 24, Table 25, Table 26).

8.2.3 Buffer Requirements

8.2.3.1 Minimum Decoder Input Buffer

The following rules are used to calculate the maximum number of bits in the input buffer both for the bitstream as a whole, for any given program, or for any given SCE/CPE/CCE:

The input buffer size is 6144 bits per SCE or independently switched CCE, plus 12288 bits per CPE (6144*NCC). Both the total buffer and the individual buffer sizes are limited, so that the buffering limit can be calculated for either the entire bitstream, any entire program, or the individual audio_channel_element()'s permitting the decoder to break a multichannel bitstream into separate mono and stereo bitstreams which are decoded by separate mono and stereo decoders, respectively. All bits for LFE's or dependent CCE's must be supplied from the total buffer requirements based on the independent CCE's, SCE's, and CPE's. Furthermore, all bits required for any DSE's, PCE's, FIL's, or fixed headers, variable headers, byte_alignment, and CRC must also be supplied from the same total buffer requirements.

8.2.3.2 Bit Reservoir

The bit reservoir is controlled at the encoder. The maximum bit reservoir in the encoder depends on the NCC and the mean bitrate. The maximum bit reservoir size for constant rate channels can be calculated by subtracting the mean number of bits per block from the minimum decoder input buffer size. For example, at 96 kbit/s for a stereo signal at 44.1 kHz sampling frequency the mean number of bits per block (mean_framelength) is (96000 bit/s / 44100 1/s * 1024) = 2229.1156... . This leads to a maximum bit reservoir size (max_bit_reservoir) of INT (12288 bit - 2229.1156...) = 10058. For variable bitrate channels the encoder must operate in a way that the input buffer requirements do not exceed the minimum decoder input buffer.

The state of the bit reservoir (bit_reservoir_state) is transmitted in the buffer_fullness field, either as the state of the bit reservoir truncated to an integer value (adif_buffer_fullness) or as the state of the bit reservoir divided by NCC divided by 32 and truncated to an integer value (adts_buffer_fullness).

The bit_reservoir_state of subsequent frames can be derived as follows:

$$bit\_reservoir\_state[frame] = bit\_reservoir\_state[frame-1] + mean\_framelength - framelength[frame]$$

Framelengths have to be adjusted such that the following restriction is met

$$0 \le bit\_reservoir\_state[frame] \le max\_bit\_reservoir$$

8.2.3.3 Maximum Bitrate

<u>Maximum bitrate:</u>

The maximum bitrate depends on the audio sampling rate. It can be calculated based on the minimum input buffer size according to the formula:

$$\frac{6144\,\frac{bit}{block}}{1024\,\frac{samples}{block}} \cdot sampling\_frequency \cdot NCC$$

Table 37 gives some examples of the maximum bitrates per channel depending on the used sampling frequency.

Table 37 — Maximum bitrate depending on the sampling frequency

sampling_frequency	maximum bitrate / NCC
48 kHz	288 kbit/s
44.1 kHz	264.6 kbit/s
32 kHz	192 kbit/s

8.2.4 Decoding Process

Assuming that the start of a raw_data_block() is known, it can be decoded without any additional „transport-level" information and produces 1024 audio samples per output channel. The sampling rate of the audio signal, as specified by the **sampling_frequency_index,** may be specified in a program_config_element() or it may be implied in the specific application domain. In the latter case, the **sampling_frequency_index** must be deduced in order for the bitstream to be parsed.

Since a given sampling frequency is associated with only one sampling frequency table, and since maximum flexibility is desired in the range of possible sampling frequencies, the following Table shall be used to associate an implied sampling frequency with the desired sampling frequency dependent tables.

Table 38 — Sampling frequency mapping

Frequency range (in Hz)	Use tables for sampling frequency (in Hz)
f >= 92017	96000
92017 > f >= 75132	88200
75132 > f >= 55426	64000
55426 > f >= 46009	48000
46009 > f >= 37566	44100
37566 > f >= 27713	32000
27713 > f >= 23004	24000
23004 > f >= 18783	22050
18783 > f >= 13856	16000
13856 > f >= 11502	12000
11502 > f >= 9391	11025
9391 > f	8000

The raw_data_stream supports encoding for both constant rate and variable rate channels. In each case the structure of the bitstream and the operation of the decoder are identical except for some minor qualifications. For constant rate channels, the encoder may have to insert a FIL element to adjust the rate upwards to exactly the desired rate. A decoder reading from a constant rate channel must accumulate a minimum number of bits in its input buffer prior to the start of decoding so that output buffer underrun does not occur. In the case of variable rate, demand read channels, each raw_data_block() can have the minimum length (rate) such that the desired audio quality is achieved, and in the decoder there is no minimum input data requirement prior to the start of decoding.

Examples of the simplest possible bitstreams are:

bitstream segment	**output signal**
<SCE><TERM><SCE><TERM>...	mono signal
<CPE><TERM><CPE><TERM>...	stereo signal
<SCE><CPE><CPE><LFE><TERM><SCE><CPE><CPE><LFE><TERM>...	5.1 channel signal

where angle brackets (< >) are used to delimit syntactic elements. For the mono signal each SCE must have the same value in its **element_instance_tag**, and similarly, for the stereo signal each CPE must have the same value in its **element_instance_tag**. For the 5.1 channel signal each SCE must have the same value in its **element_instance_tag**, each CPE associated with the front channel pair must have the same value in its **element_instance_tag**, and each CPE associated with the back channel pair must have the same value in its **element_instance_tag**.

If these bitstreams are to be transmitted over a constant rate channel then they might include a fill_element() to adjust the instantaneous bitrate. In this case an example of a coded stereo signal is

<CPE><FIL><TERM><CPE><FIL><TERM>...

If the bitstreams are to carry ancillary data and run over a constant rate channel then an example of a coded stereo signal is

<CPE><DSE><FIL><TERM><CPE><DSE><FIL><TERM>...

All data_stream_element()'s have the same element_instance_tag if they are part of the same data stream.

8.3 Single Channel Element (SCE), Channel Pair Element (CPE) and Individual Channel Stream (ICS)

8.3.1 Definitions

8.3.1.1 Data Elements

common_window	a flag indicating whether the two individual_channel_stream()'s share a common ics_info() or not. In case of sharing, the ics_info() is part of the channel_pair_element() and must be used for both channels. Otherwise, the ics_info() is part of each individual_channel_stream() (Table 14).
ics_reserved_bit	flag reserved for future use. Shall be '0'.
window_sequence	indicates the sequence of windows as defined in Table 44 (Table 15).
window_shape	A 1 bit field that determines what window is used for the trailing part of this analysis window (Table 15).
max_sfb	number of scalefactor bands transmitted per group (Table 15).
scale_factor_grouping	A bit field that contains information about grouping of short spectral data (Table 15).

8.3.1.2 Data Functions

individual_channel_stream()	contains data necessary to decode one channel (Table 16).
ics_info()	contains side information necessary to decode an individual_channel_stream(). The individual_channel_stream()'s of a channel_pair_element() may share one common ics_info() (Table 15).

8.3.1.3 Help Elements

scalefactor window band	term for scalefactor bands within a window, given in Table 45 to Table 57.
scalefactor band	term for scalefactor band within a group. In the case of EIGHT_SHORT_SEQUENCE and grouping a scalefactor band

	may contain several scalefactor window bands of corresponding frequency. For all other window_sequences scalefactor bands and scalefactor window bands are identical.
g	group index.
win	window index within group.
sfb	scalefactor band index within group.
swb	scalefactor window band index within window.
bin	coefficient index.
num_window_groups	number of groups of windows which share one set of scalefactors.
window_group_length[g]	number of windows in each group.
bit_set(bit_field,bit_num)	function that returns the value of bit number bit_num of a bit_field (most right bit is bit 0).
num_windows	number of windows of the actual window sequence.
num_swb_long_window	number of scalefactor bands for long windows. This number has to be selected depending on the sampling frequency. See subclause 8.9.
num_swb_short_window	number of scalefactor window bands for short windows. This number has to be selected depending on the sampling frequency. See subclause 8.9.
num_swb	number of scalefactor window bands for shortwindows in case of EIGHT_SHORT_SEQUENCE, number of scalefactor window bands for long windows otherwise.
swb_offset_long_window[swb]	Table containing the index of the lowest spectral coefficient of scalefactor band sfb for long windows. This Table has to be selected depending on the sampling frequency. See subclause 8.9.
swb_offset_short_window[swb]	Table containing the index of the lowest spectral coefficient of scalefactor band sfb for short windows. This Table has to be selected depending on the sampling frequency. See subclause 8.9.
swb_offset[swb]	Table containing the index of the lowest spectral coefficient of scalefactor band sfb for short windows in case of EIGHT_SHORT_SEQUENCE, otherwise for long windows.
sect_sfb_offset[g][section]	Table that gives the number of the start coefficient for the section_data() within a group. This offset depends on the window_sequence and scale_factor_grouping.
sampling_frequency_index	see subclause 8.1.2.1.

8.3.2 Decoding Process

8.3.2.1 Decoding a single_channel_element() and channel_pair_element()

A single_channel_element() is composed of an element_instance_tag and an individual_channel_stream. In this case ics_info() is always located in the individual_channel_stream.

A channel_pair_element() begins with an element_instance_tag and common_window flag. If the common_window equals '1', then ics_info() is shared amongst the two individual_channel_stream elements and the MS information is transmitted. If common_window equals '0', then there is an ics_info() within each individual_channel_stream and there is no MS information.

8.3.2.2 Decoding an individual_channel_stream()

In the individual_channel_stream, the order of decoding is:

get global_gain

get ics_info() (parse bitstream if common information is not present)

get section_data()

get scalefactor_data(), if present

get pulse_data(), if present

get tns_data(), if present

get gain_control_data(), if present

get spectral_data(), if present.

The process of recovering pulse_data is described in clause 9, tns_data in clause 14, and gain_control data in clause 16. An overview of how to decode ics_info() (subclause 8.3), section data (clause 9), scalefactor data (clause 9 and 11), and spectral data (clause 9) will be given here.

8.3.2.3 Recovering ics_info()

For single_channel_element()'s ics_info() is always located immediately after the global_gain in the inidividual_channel_stream(). For a channel_pair_element() there are two possible locations for the ics_info(). If each individual channel in the pair window switch together then the ics_info() is located immediately after common_window in the channel_pair_element() and common_window is set to 1. Otherwise there is an ics_info() immediately after global_gain in each of the two individual_channel_stream() in the channel_pair_element() and common_window is set to 0.

ics_info() carries window information associated with an ICS and thus permits channels in a channel_pair to switch separately if desired. In addition it carries the max_sfb which places an upper limit on the number of ms_used[] and predictor_used[] bits that must be transmitted. If the window_sequence is EIGHT_SHORT_SEQUENCE then scale_factor_grouping is transmitted. If a set of short windows form a group then they share scalefactors as well as intensity stereo positions and have their spectral coefficients interleaved. The first short window is always a new group so no grouping bit is transmitted. Subsequent short windows are in the same group if the associated grouping bit is 1. A new group is started if the associated grouping bit is 0. It is assumed that grouped short windows have similar signal statistics. Hence their spectra are interleaved so as to place correlated coefficients next to each other. The manner of interleaving is indicated in Figure 6. ics_info() also carries the prediction data for the individual channel or channel pair (see clause 13).

8.3.2.4 Recovering Sectioning Data

In the ICS, the information about one long window, or eight short windows, is recovered. The sectioning data is the first field to be decoded, and describes the Huffman codes that apply to the scalefactor bands in the ICS (see clause 9 and 11). The form of the section data is:

sect_cb The codebook for the section

and

sect_len The length of the section.

This length is recovered by reading the bitstream sequentially for a section length, adding the escape value to the total length of the section until a non-escape value is found, which is added to establish the total length of the section. This process is clearly explained in the C-like syntax description. Note that within each group the sections must delineate the scalefactor bands from zero to **max_sfb** so that the first section within each group starts at bands zero and the last section within each group ends at **max_sfb**.

The sectioning data describes the codebook, and then the length of the section using that codebook, starting from the first scalefactor band and continuing until the total number of scalefactor bands is reached.

After this description is provided, all scalefactors and spectral data corresponding to codebook zero are zeroed, and no values corresponding to these scalefactors or spectral data will be transmitted. When scanning for scale-factor data it is important to note that scalefactors for any scalefactor bands whose Huffman codebook is zero will be omitted. Similarly, all spectral data associated with Huffman codebook zero are omitted (see clause 9 and 11).

In addition spectral data associated with the scalefactor bands that have an intensity codebook will not be transmitted, but intensity steering coefficients will be transmitted in place of the scalefactors, as described in subclause 12.2.

8.3.2.5 Scalefactor Data Parsing and Decoding

For each scalefactor band that is not in a section coded with the zero codebook (ZERO_HCB), a scalefactor is transmitted. These will be denoted as 'active' scalefactor bands and the associated scalefactors as active scalefactors. Global gain, the first data element in an ICS, is typically the value of the first active scalefactor. All scalefactors (and steering coefficients) are transmitted using Huffman coded DPCM relative to the previous active scalefactor (see clause 9 and 11). The first active scalefactor is differentially coded relative to the global gain. Note that it is not illegal, merely inefficient, to provide a global_gain that is different from the first active scalefactor and then a non-zero DPCM value for the first scalefactor DPCM value. If any intensity steering coefficients are received interspersed with the DPCM scalefactor elements, they are sent to the intensity stereo module, and are not involved in the DPCM coding of scalefactor values (see subclause 12.2). The value of the first active scalefactor is usually transmitted as the global_gain with the first DPCM scalefactor having a zero value. Once the scalefactors are decoded to their integer values, the actual values are found via a power function (see clause 11).

8.3.2.6 Spectral Data Parsing and Decoding

The spectral data is recovered as the last part of the parsing of an ICS. It consists of all the non-zeroed coefficients remaining in the spectrum or spectra, ordered as described in the ICS_info. For each non-zero, non-intensity codebook, the data are recovered via Huffman decoding in quads or pairs, as indicated in the noiseless coding tool (see clause 9). If the spectral data is associated with an unsigned Huffman codebook, the necessary sign bits follow the Huffman codeword (see subclause 9.3). In the case of the ESCAPE codebook, if any escape value is received, a corresponding escape sequence will appear after that Huffman code. There may be zero, one or two escape sequences for each codeword in the ESCAPE codebook, as indicated by the presence of escape values in that decoded codeword. For each section the Huffman decoding continues until all the spectral values in that section have been decoded. Once all sections have been decoded, the data is multiplied by the decoded scalefactors and deinterleaved if necessary.

8.3.3 Windows and Window Sequences

Quantization and coding is done in the frequency domain. For this purpose, the time signal is mapped into the frequency domain in the encoder. The decoder performs the inverse mapping as described in clause 15. Depending on the signal, the coder may change the time/frequency resolution by using two different windows: LONG_WINDOW and SHORT_WINDOW. To switch between windows, the transition windows LONG_START_WINDOW and LONG_STOP_WINDOW are used. Table 43 lists the windows, specifies the corresponding transform length and shows the shape of the windows schematically. Two transform lengths are used: 1024 (referred to as long transform) and 128 coefficients (refered to as short transform).

Window sequences are composed of windows in a way that a raw_data_block() always contains data representing 1024 output samples. The data element **window_sequence** indicates the window sequence that is actually used. Table 44 lists how the window sequences are composed of individual windows. Refer to clause 15 for more detailed information about the transform and the windows.

8.3.4 Scalefactor Bands and Grouping

Many tools of the decoder perform operations on groups of consecutive spectral values called scalefactor bands (abbreviation 'sfb'). The width of the scalefactor bands is built in imitation of the critical bands of the human auditory system. For that reason the number of scalefactor bands in a spectrum and their width depend on the transform length and the sampling frequency. Table 45 to Table 57 list the offset to the beginning of each scalefactor band for the transform lengths 1024 and 128 and the different sampling frequencies, respectively.

To reduce the amount of side information in case of sequences which contain SHORT_WINDOWS, consecutive SHORT_WINDOWs may be grouped (see Figure 4). The information about the grouping is contained in the **scale_factor_grouping** data element. Grouping means that only one set of scalefactors is transmitted for all grouped windows as if there was only one window. The scalefactors are then applied to the corresponding spectral data in all grouped windows. To increase the efficiency of the noiseless coding (see clause 9), the spectral data of a group is transmitted in an interleaved order given in subclause 8.3.5. The interleaving is done on a scalefactor band by scalefactor band basis, so that the spectral data can be grouped to form a virtual scalefactor band to which the common scalefactor can be applied. Within this document the expression 'scalefactor band' (abbreviation 'sfb') denotes these virtual scalefactor bands. If the scalefactor bands of the single windows are referred to, the expression 'scalefactor window band' (abbreviation 'swb') is used. Due to its influence on the scalefactor bands, grouping affects the meaning of section_data (see clause 9), the order of spectral data (see subclause 8.3.5), and the total number of scalefactor bands. For a LONG_WINDOW scalefactor bands and scalefactor window bands are identical since there is only one group with only one window.

To reduce the amount of information needed for the transmission of side information specific to each scalefactor band, the data element **max_sfb** is transmitted. Its value is one greater than the highest active scalefactor band in all groups. **max_sfb** has influence on the interpretation of section data (see clause 9), the transmission of scalefactors (see clause 9 and 11), the transmission of predictor data (see clause 13) and the transmission of the ms_mask (see subclause 12.1).

Since scalefactor bands are a basic element of the coding algorithm, some help variables and arrays are needed to describe the decoding process in all tools using scalefactor bands. These help variables depend on sampling_frequency, **window_sequence, scalefactor_grouping** and **max_sfb** and must be built up for each raw_data_block(). The pseudo code shown below describes

- how to determine the number of windows in a window_sequence *num_windows*
- how to determine the number of window_groups *num_window_groups*
- how to determine the number of windows in each group *window_group_length[g]*
- how to determine the total number of scalefactor window bands *num_swb* for the actual window type
- how to determine *swb_offset[swb]*, the offset of the first coefficient in scalefactor window band *swb* of the window actually used

- how to determine *sect_sfb_offset[g][section]*, the offset of the first coefficient in section *section*. This offset depends on **window_sequence** and **scale_factor_grouping** and is needed to decode the spectral_data().

A long transform window is always described as a window_group containing a single window. Since the number of scalefactor bands and their width depend on the sampling frequency, the affected variables are indexed with sampling_frequency_index to select the appropriate table.

```
fs_index = sampling_frequency_index;
switch (window_sequence) {
   case ONLY_LONG_SEQUENCE:
   case LONG_START_SEQUENCE:
   case LONG_STOP_SEQUENCE:
      num_windows = 1;
      num_window_groups = 1;
      window_group_length[num_window_groups-1] = 1;
      num_swb = num_swb_long_window[fs_index];
      /* preparation of sect_sfb_offset for long blocks */
      /* also copy the last value! */
      for (i = 0; i < max_sfb + 1; i++) {
         sect_sfb_offset[0][i] = swb_offset_long_window[fs_index][i];
         swb_offset[i] = swb_offset_long_window[fs_index][i];
      }
      break;
   case EIGHT_SHORT_SEQUENCE:
      num_windows = 8;
      num_window_groups = 1;
      window_group_length[num_window_groups-1] = 1;
      num_swb = num_swb_short_window[fs_index];
      for (i = 0; i < num_swb_short_window[fs_index] + 1; i++)
         swb_offset[i] = swb_offset_short_window[fs_index][i];
      for (i = 0; i < num_windows-1; i++) {
         if(bit_set(scale_factor_grouping,6-i)) == 0) {
            num_window_groups += 1;
            window_group_length[num_window_groups-1] = 1;
         }
         else {
            window_group_length[num_window_groups-1] += 1;
         }
      }
      /* preparation of sect_sfb_offset for short blocks */
      for (g = 0; g < num_window_groups; g++) {
         sect_sfb = 0;
         offset = 0;
         for (i = 0; i < max_sfb; i++) {
            width = swb_offset_short_window[fs_index][i+1] -
                    swb_offset_short_window[fs_index][i];
            width *= window_group_length[g];
            sect_sfb_offset[g][sect_sfb++] = offset;
            offset += width;
         }
         sect_sfb_offset[g][sect_sfb] = offset;
      }
      break;
   default:
      break;
}
```

8.3.5 Order of Spectral Coefficients in spectral_data()

For ONLY_LONG_SEQUENCE windows (num_window_groups = 1, window_group_length[0] = 1) the spectral data is in ascending spectral order, as shown in Figure 5.

For the EIGHT_SHORT_SEQUENCE window, the spectral order depends on the grouping in the following manner:

- Groups are ordered sequentially
- Within a group, a scalefactor band consists of the spectral data of all grouped SHORT_WINDOWs for the associated scalefactor window band. To clarify via example, the length of a group is in the range of one to eight SHORT_WINDOWs.
    - If there are eight groups each with length one (num_window_groups = 8, window_group_length[0] = 1), the result is a sequence of eight spectrums, each in ascending spectral order.
    - If there is only one group with length eight (num_window_group = 1, window_group_length[0] = 8), the results is that spectral data of all eight SHORT_WINDOWs is interleaved by scalefactor window bands.
    - Figure 6 shows the spectral ordering for an EIGHT_SHORT_SEQUENCE with grouping of SHORT_WINDOWs according to Figure 4 (num_window_groups = 4).
- Within a scalefactor window band, the coefficients are in ascending spectral order.

8.3.6 Output Word Length

The global gain for each audio channel is scaled such that the integer part of the output of the IMDCT can be used directly as a 16-bit PCM audio output to a digital-to-analog (D/A) converter. This is the default mode of operation and will result in correct audio levels. If the decoder has a D/A converter that has greater than 16-bit resolution then the output of the IMDCT can be scaled up such that the appropriate number of fractional bits are included to form the desired D/A word size. In this case the level of the converter output would be matched to that of a 16-bit D/A, but would have the advantage of greater signal dynamic range and lower converter noise floor. Similarly, shorter D/A word lengths can be accommodated.

8.4 Low Frequency Enhancement Channel (LFE)

8.4.1 General

In order to maintain a regular structure of the decoder, the lfe_channel_element() is defined as a standard individual_channel_stream(0) element, i.e. equal to a single_channel_element(). Thus, decoding can be done using the standard procedure for decoding a single_channel_element().

In order to accomodate a more bitrate and hardware efficient implementation of the LFE decoder, however, several restrictions apply to the options used for the encoding of this element:

- The window_shape field is always set to 0, i.e. sine window (see subclause 6.3, Table 15).
- The window_sequence field is always set to 0 (ONLY_LONG_SEQUENCE) (see subclause 6.3, Table 15).
- Only the lowest 12 spectral coefficients of any LFE may be non-zero.
- No Temporal Noise Shaping is used, i.e. tns_data_present is set to 0 (see subclause 6.3, Table 16).
- No prediction is used, i.e. predictor_data_present is set to 0 (see subclause 6.3, Table 15).

The presence of LFE channels depends on the profile used. Refer to clause 7 for detailed information.

8.5 Program Config Element (PCE)

A program_config_element() may occur outside the AAC payload e. g. in the adif_header(), but also inside the AAC payload as syntactic element in a raw_data_block().

8.5.1 Data Functions

byte_alignment()	For PCEs within a raw_data_block(), align with respect to the first bit of the raw_data_block(). For PCEs within the adif_header(), align with respect to the first bit of the header.

8.5.2 Data Elements

profile	The two-bit profile index from Table 31 (Table 25).
sampling_frequency_index	Indicates the sampling rate of the program (and all other programs in this bitstream). See definition in subclause 8.1.2.1 (Table 25).
num_front_channel_elements	The number of audio syntactic elements in the front channels, front center to back center, symmetrically by left and right, or alternating by left and right in the case of single channel elements (Table 25).
num_side_channel_elements	Number of elements to the side as above (Table 25).
num_back_channel_elements	As number of side and front channel elements, for back channels (Table 25).
num_lfe_channel_elements	Number of LFE channel elements associated with this program (Table 25).
num_assoc_data_elements	The number of associated data elements for this program (Table 25).
num_valid_cc_elements	The number of CCE's that can add to the audio data for this program (Table 25).
mono_mixdown_present	One bit, indicating the presence of the mono mixdown element (Table 25).
mono_mixdown_element_number	The number of a specified SCE that is the mono mixdown (Table 25).
stereo_mixdown_present	One bit, indicating that there is a stereo mixdown present (Table 25).
stereo_mixdown_element_number	The number of a specified CPE that is the stereo mixdown element (Table 25).
matrix_mixdown_idx_present	One bit indicating the presence of matrix mixdown information by means of a stereo matrix coefficient index (see Table 39). For all configurations other than the 3/2 format this bit must be zero (Table 25).
matrix_mixdown_idx	Two bit field, specifying the index of the mixdown coefficient to be used in the 5-channel to 2-channel matrix-mixdown. Possible matrix coefficients are listed in Table 39 (Table 25).
pseudo_surround_enable	One bit, indicating the possibility of mixdown for pseudo surround reproduction (Table 25).
front_element_is_cpe	indicates whether a SCE or a CPE is addressed as a front element (Table 25).'0' selects an SCE.'1' selects an CPE. The

	instance of the SCE or CPE addressed is given by front_element_tag_select.
front_element_tag_select	The instance_tag of the SCE/CPE addressed as a front element (Table 25).
side_element_is_cpe	see front_element_is_cpe, but for side elements (Table 25).
side_element_tag_select	see front_element_tag_select, but for side elements (Table 25).
back_element_is_cpe	see front_element_is_cpe, but for back elements (Table 25).
back_element_tag_select	see front_element_tag_select, but for back elements (Table 25).
lfe_element_tag_select	instance_tag of the LFE addressed (Table 25).
assoc_data_element_tag_select	instance_tag of the DSE addressed (Table 25).
valid_cc_element_tag_select	instance_tag of the CCE addressed (Table 25).
cc_element_is_ind_sw	One bit, indicating that the corresponding CCE is an independently switched coupling channel (Table 25).
comment_field_bytes	The length, in bytes, of the following comment field (Table 25).
comment_field_data	The data in the comment field (Table 25).

SCE or CPE elements within the PCE are addressed with two syntax elements. First, an is_cpe syntax element selects whether a SCE or CPE is addressed. Second, a tag_select syntax element selects the instance_tag of a SCE/CPE. LFE, CCE and DSE elements are directly addressed with their instance_tag.

8.5.3 Channel configuration

The AAC audio syntax provides three ways to convey the mapping of channels within a set of syntactic elements to physical locations of speakers.

8.5.3.1 Explicit channel mapping using default channel settings

Default channel mappings are defined in Table 42 (values greater than 0).

8.5.3.2 Explicit channel mapping using a program_config_element()

Any possible channel configuration can be specified using a program_config_element().There are 16 available PCE's, and each one can specify a distinct program that is present in the raw data stream. All available PCE's within a raw_data_block() must come before all other syntactic elements. Programs may or may not share audio syntactic elements, for example, programs could share a channel_pair_element() and use distinct coupling channels for voice over in different languages. A given program_config_element() contains information pertaining to only one program out of many that may be included in the raw_data_stream(). Included in the PCE are "list of front channels", using the rule center outwards, left before right. In this list, a center channel SCE, if any, must come first, and any other SCE's must appear in pairs, constituting an LR pair. If only two SCE's are specified, this signifies one LR stereophonic pair.

After the list of front channels, there is a list of "side channels" consisting of CPE's, or of pairs of SCE's. These are listed in the order of front to back. Again, in the case of a pair of SCE's, the first is a left channel, the second a right channel.

After the list of side channels, a list of back channels is available, listed from outside in. Any SCE's except the last SCE must be paired, and the presence of exactly two SCE's (alone or preceeded by a CPE) indicates that the two SCE's are Left and Right Rear center, respectively.

The configuration indicated by the PCE takes effect at the raw_data_block() containing the PCE. The number of front, side and back channels as specified in the PCE must be present in that block and all subsequent raw_data_block()'s until a raw_data_block() containing a new PCE is transmitted.

Other elements are also specified. A list of one or more LFE's is specified for application to this program. A list of one or more CCE's (profile-dependent) is also provided, in order to allow for dialog management as well as different intensity coupling streams for different channels using the same main channels. A list of data streams associated with the program can also associate one or more data streams with a program. The program configuration element also allows for the specification of one monophonic and one stereophonic simulcast mixdown channel for a program. Note that the MPEG-2 Systems standard ISO/IEC 13818-1 supports alternate methods of simulcast.

A PCE element is not intended to allow for rapid program changes. At any time when a given PCE, as selected by its element_instance_tag, defines a new (as opposed to repeated) program, the decoder is not obliged to provide audio signal continuity.

8.5.3.3 Implicit channel mapping

If no explicit channel mapping is given, the following methods describe the implicit channel mapping:

1) Any number of SCE's may appear (as long as permitted by other constraints, for example profile). If this number of SCE's is odd, then the first SCE represents the front center channel, and the other SCE's represent L/R pairs of channels, proceeding from center front outwards and back to center rear.

If the number of SCE's is even, then the SCE's are assigned as pairs as center-front L/R, in pairs proceeding out and back from center front toward center back.

2) Any number of CPE's or pairs of SCE's may appear. Each CPE or pair of SCE's represents one L/R pair, proceeding from where the first sets of SCE's left off, pairwise until reaching either center back pair.

3) Any number of SCEs may appear. If this number is even, allocating pairs of SCEs Left/Right, from 2), back to center back. If this number is odd, allocated as L/R pairs, except for the final SCE, which is assigned to center back..

4) Any number of LFEs may appear. No speaker mapping is defined in case of multiple LFEs.

In the case of such implicit channel mapping the number and order of SCEs, CPEs and LFEs and the resulting configuration may not change within the bitstream without sending a program_config_element(), i.e. an implicit reconfiguration is not allowed.

Other audio syntactic elements that do not imply additional output speakers, such as coupling channel_element, may follow the listed set of syntactic elements. Obviously non-audio syntactic elements may be received in addition and in any order relative to the listed syntactic elements.

8.5.4 Matrix-mixdown Method

8.5.4.1 Description

The matrix-mixdown method applies only for mixing a 3-front/2-back speaker configuration, 5-channel program, down to a stereo or a mono program. It is not applicable to any program with other than the 3/2 configuration.

8.5.4.2 Matrix-mixdown Process

A derived stereo signal can be generated within a matrix-mixdown decoder by use of one of the two following sets of equations.

Set 1:

$$L' = \frac{1}{1+1/\sqrt{2}+A} \cdot [L + C/\sqrt{2} + A \cdot L_S]$$

$$R' = \frac{1}{1+1/\sqrt{2}+A} \cdot [R + C/\sqrt{2} + A \cdot R_S]$$

Set 2:

$$L' = \frac{1}{1+1/\sqrt{2}+2 \cdot A} \cdot [L + C/\sqrt{2} - A \cdot (L_S + R_S)]$$

$$R' = \frac{1}{1+1/\sqrt{2}+2 \cdot A} \cdot [R + C/\sqrt{2} + A \cdot (L_S + R_S)]$$

Where L, C, R, LS and RS are the source signals, L' and R' are the derived stereo signals and A is the matrix coefficient indicated by matrix_mixdown_idx. LFE channels are omitted from the mixdown.

If pseudo_surround_enable is not set, then only set 1 should be used. If pseudo_surround_enable is set, then either set 1 or set 2 equations can be used, depending on whether the receiver has facilities to invoke some form of surround synthesis.

As further information it should be noted that one can derive a mono signal using the following equation:

$$M = \frac{1}{3+2 \cdot A} \cdot [L + C + R + A \cdot (L_S + R_S)]$$

8.5.4.3 Advisory

The matrix-mixdown provision enables a mode of operation which may be beneficial to some operators in some circumstances. However, it is advised that this method should not be used. The psychoacoustic principles on which the audio coding are based are violated by this form of post-processing, and a perceptually faithful reconstruction of the signal cannot be guaranteed. The preferred method is to use the stereo or mono mixdown channels in the AAC syntax to provide stereo or mono programming which is specifically created by conventional studio mixing prior to bitrate reduction.

The stereo and mono mixdown channels additionally enable the content provider to separately optimize the stereo and multichannel program mixes - this is not possible by using the matrix-mixdown method.

It is additionally relevant to note that, due to the algorithms used for the multichannel and stereo mixdown coding, a better combination of quality and bitrate is usually provided by use of the stereo mixdown channels than can be provided by the matrix-mixdown process.

8.5.4.4 Tables

Table 39 — Matrix-mixdown coefficients

matrix_mixdown_idx	A
0	$1/\sqrt{2}$
1	$1/2$
2	$1/(2\sqrt{2})$
3	0

8.6 Data Stream Element (DSE)

8.6.1 Data Functions

byte_alignment()	align with respect to the first bit of the raw_data_block().

8.6.2 Data Elements

data_byte_align_flag	One bit indicating that a byte aligment is performed within the data stream element (Table 24)
count	Initial value for length of data stream (Table 24)
esc_count	Incremental value of length of data or padding element (Table 24)
data_stream_byte	A data stream byte extracted from bitstream (Table 24)

A data element contains any additional data, e.g. auxiliary information, that is not part of the audio information itself. Any number of data elements with the same element_instance_tag or up to 16 data elements with different element_instance_tags are possible. The decoding process of the data element is described in this clause.

8.6.3 Decoding Process

The first syntactic element to be read is the 1 bit **data_byte_align_flag**. Next is the 8 bit value **count**. It contains the initial byte-length of the data stream. If **count** equals 255, its value is incremented by a second 8 bit value, **esc_count**, this final value represents the number of bytes in the data stream element. If **data_byte_align_flag** is set, a byte alignment is performed. The bytes of the data stream follow.

8.7 Fill Element (FIL)

8.7.1 Data Elements

count	Initial value for length of extension_payload() (Table 26).
esc_count	Incremental value for length of extension_payload() (Table 26).

8.7.2 Decoding Process

fill_element()'s might be added to allow for several kinds of extension payloads. Any number of fill_element()'s is allowed.

The syntactic element **count** gives the initial value of the length of the fill data. In the same way as for the data element this value is incremented with the value of **esc_count** if **count** equals 15. The resulting number gives the number of **fill_bytes** to be read.

8.8 Extension Payload

8.8.1 General

8.8.1.1 Data Elements

extension_type	Four bit field indicating the type of fill element content (Table 26).

8.8.1.2 Decoding Process

Any number of extension_payload()'s are allowed.

The following symbolic abbreviations for values of the extension_type field are defined:

Table 40 — Values of the extension_type data element

Symbol	Value of extension_type	Purpose
EXT_FILL	'0000'	Bitstream filler
EXT_FILL_DATA	'0001'	Bitstream data as filler
EXT_DYNAMIC_RANGE	'1011'	Dynamic range control
EXT_SBR_DATA	'1101'	SBR enhancement
EXT_SBR_DATA_CRC	'1110'	SBR enhancement with CRC
-	all other values	reserved

The 'reserved' values might be used for further extension of the syntax in a compatible way.

8.8.2 Fill data and other bits

8.8.2.1 Data Elements

fill_nibble	Four bit field for fill (Table 28).
fill_byte	Byte to be discarded by the decoder (Table 28).
other_bits	Bits to be discarded by the decoder (Table 28).

8.8.2.2 Decoding Process

Fill data shall be added if the total bits for all audio data together with all additional data is lower than the minimum allowed number of bits in this frame necessary to reach the target bitrate. Under normal conditions fill bits are avoided and free bits are used to fill up the bit reservoir. Fill bits are written only if the bit reservoir is full.

Note that fill_nibble is normatively defined to be '0000' and fill_byte is normatively defined to be '10100101' (to ensure that self-clocked data streams, such as radio modems, can perform reliable clock recovery).

8.8.3 Dynamic Range Control (DRC)

8.8.3.1 Data Elements

pce_tag_present	One bit indicating that program element tag is present (Table 29).
pce_instance_tag	Tag field that indicates with which program the dynamic range information is associated (Table 29)

drc_tag_reserved_bits	Reserved (Table 29)
excluded_chns_present	One bit indicating that excluded channels are present (Table 29)
drc_bands_present	One bit indicating that DRC multi-band information is present (Table 29)
drc_band_incr	Number of DRC bands greater than 1 having DRC information (Table 29)
drc_bands_reserved_bits	Reserved (Table 29)
drc_band_top[i]	Indicates top of i-th DRC band in units of 4 spectral lines (Table 29).If drc_band_top[i] = k, then the index (w.r.t zero) of the highest spectral coefficient that is in the i-th DRC band is = k*4+3. In case of an EIGHT_SHORT_SEQUENCE window_sequence the index is interpreted as pointing into the concatenated array of 8*128 (de-interleaved) frequency points corresponding to the 8 short transforms.
prog_ref_level_present	One bit indicating that reference level is present (Table 29).
prog_ref_level	Reference level. A measure of long-term program audio level for all channels combined (Table 29).
prog_ref_level_reserved_bits	Reserved (Table 29)
dyn_rng_sgn[i]	Dynamic range control sign information. One bit indicating the sign of dyn_rng_ctl (0 if positive, 1 if negative, (Table 29)
dyn_rng_ctl[i]	Dynamic range control magnitude information (Table 29)
exclude_mask[i]	Boolean array indicating the audio channels of a program that are excluded from DRC processing using this DRC information.
additional_excluded_chns[i]	One bit indicating that additional excluded channels are present (Table 30)

8.8.3.2 Decoding Process

The evaluation of potentially available dynamic range control information in the decoder is optional.

prog_ref_level_present indicates that **prog_ref_level** is being transmitted. This permits **prog_ref_level** to be sent as infrequently as desired (e.g. once), although periodic transmission would permit break-in.

prog_ref_level is quantized in 0.25 dB steps using 7 bits, and therefore has a range of approximately 32 dB. It indicates program level relative to full scale (i.e. dB below full scale), and is reconstructed as:

$$level = 32767 \cdot 2^{-prog\_ref\_level/24}$$

where „full scale level„ is 32767 (prog_ref_level equal to 0).

pce_tag_present indicates that **pce_instance_tag** is being transmitted. This permits **pce_instance_tag** to be sent as infrequently as desired (e.g. once), although periodic transmission would permit break-in.

pce_instance_tag indicates with which program the dynamic range information is associated. If this is not present then the default program is indicated. Since each AAC bitstream typically has just one program, this would be the most common mode. Each program in a multi-program bitstream would send its dynamic range information in a distinct extension_payload() of the fill_element(). In the multiple program case, the **pce_instance_tag** would always have to be signaled.

The **drc_tag_reserved_bits** fill out the optional fields to an integral number of bytes in length.

The **excluded_chns_present** bit indicates that channels that are to be *excluded* from dynamic range processing will be signaled immediately following this bit. The excluded channel mask information must be transmitted in each frame where channels are excluded. The following ordering principles are used to assign the exclude_mask to channel outputs:

- If a PCE is present, the **exclude_mask** bits correspond to the audio channels in the SCE, CPE, CCE and LFE syntax elements in the order of their appearance in the PCE. In the case of a CPE, the first transmitted mask bit corresponds to the first channel in the CPE, the second transmitted mask bit to the second channel. In the case of a CCE, a mask bit is transmitted only if the coupling channel is specified to be an independently switched coupling channel.
- If no PCE is present, the **exclude_mask** bits correspond to the audio channels in the SCE, CPE and LFE syntax elements in the order of their appearance in the bitstream, followed by the audio channels in the CCE syntax elements in the order of their appearance in the bitstream. In the case of a CPE, the first transmitted mask bit corresponds to the first channel in the CPE, the second transmitted mask bit to the second channel. In the case of a CCE, a mask bit is transmitted only if the coupling channel is specified to be an independently switched coupling channel.

drc_band_incr is the number of bands greater than one if there is multi-band DRC information.

dyn_rng_ctl is quantized in 0.25 dB steps using a 7-bit unsigned integer, and therefore, in association with **dyn_rng_sgn,** has a range of +/-31.75 dB. It is interpreted as a gain value that shall be applied to the decoded audio output samples of the current frame.

The range supported by the dynamic range information is summarized in the following table:

Table 41 — Range supported by the DRC information

Field	bits	steps	stepsize, dB	range, dB
prog_ref_level	7	128	0.25	31.75
dyn_rng_sgn and **dyn_rng_ctl**	1 and 7	+/- 127	0.25	+/- 31.75

The dynamic range control process is applied to the spectral data `spec[i]` of one frame immediately before the synthesis filterbank. In case of an EIGHT_SHORT_SEQUENCE window_sequence the index i is interpreted as pointing into the concatenated array of 8*128 (de-interleaved) frequency points corresponding to the 8 short transforms.

This following pseudo code is for illustrative purposes only, showing one method for applying one set of dynamic control information to a frame of a target audio channel. The constants `ctrl1` and `ctrl2` are compression constants (typically between 0 and 1, zero meaning no compression) that may optionally be used to scale the dynamic range compression characteristics for levels greater than or less than the program reference level, respectively. The constant `target_level` describes the output level desired by the user, expressed in the same scaling as `prog_ref_level`.

```
bottom = 0;
drc_num_bands = 1;
if (drc_bands_present)
   drc_num_bands += drc_band_incr;
else
   drc_band_top[0] = 1024/4 - 1;
for (bd = 0; bd < drc_num_bands; bd++) {
   top = 4 * (drc_band_top[bd] + 1);
   /* Decode DRC gain factor */
   if (dyn_rng_sgn[bd])
      factor = 2^(-ctrl1*dyn_rng_ctl[bd]/24); /* compress */
   else
      factor = 2^(ctrl2*dyn_rng_ctl[bd]/24);  /* boost */

   /* If program reference normalization is done in the digital domain, modify
    * factor to perform normalization.
    * prog_ref_level can alternatively be passed to the system for modification
    * of the level in the analog domain. Analog level modification avoids
problems
    * with reduced DAC SNR (if signal is attenuated) or clipping
    * (if signal is boosted)
    */
   factor *= 0.5^((target_level-prog_ref_level)/24);

   /* Apply gain factor */
   for (i = bottom; i < top; i++)
      spec[i] *= factor;
   bottom = top;
}
```

Note the relation between dynamic range control and coupling channels:

- Dependently switched coupling channels are always coupled onto their target channels as spectral coefficients prior to the DRC processing and synthesis filtering of these channels. Therefore a dependently switched coupling channel's signal that couples onto a specific target channel will undergo the DRC processing of that target channel.
- Since independently switched coupling channels couple to their target channels in the time domain, each independently switched coupling channel will undergo DRC processing and subsequent synthesis filtering separate from its target channels. This permits the independently switched coupling channel to have distinct DRC processing if desired.

8.8.3.3 Persistence of DRC Information

At the beginning of a stream, all DRC information for all channels is assumed to be set to its default value: program reference level equal to the decoder's target reference level, one DRC band, with no DRC gain modification for that band. Unless this data is specifically overwritten, this remains in effect.

There are two cases for the persistence of DRC information that has been transmitted:

- The program reference level is per audio program, and persists until a new value is transmitted, at which point the new data overwrites the old and takes effect that frame. (It may be appropriate to send this value periodically to allow bitstream break-in.)
- Other DRC information persists on a per-channel basis. Note that if a channel is excluded via the appropriate **exclude_mask[]** bit, then effectively no information is transmitted for that channel in that call to dynamic_range_info(). The excluded channel mask information must be transmitted in each frame where channels are excluded.

The rules for retaining per-channel DRC information are as follows:

- If there is no DRC information in a given frame for a given channel, use the information that was used in the previous frame. (This means that one adjustment can hold for a long time, although it may be appropriate to transmit the DRC information periodically to permit break-in.)
- If any DRC information for this channel appears in the current frame, the following sequence occurs: first, overwrite all per-channel DRC information for that channel with the default values (one DRC band, with no DRC gain modification for that band), then overwrite any per-channel DRC information with the transmitted values.

8.8.4 Bandwidth Extension (SBR)

Fill elements containing an extension_payload with an extension_type of EXT_SBR_DATA or EXT_SBR_DATA_CRC are reserved for SBR enhancement data. In this case, the fill_element count field must be set equal to the total length in bytes, including the SBR enhancement data plus the extension_type field.

sbr_extension_data() and the decoding process are defined in ISO/IEC 14496-3.

The SBR fill elements shall be handled according to ISO/IEC 14496-3, subclause 4.5.2.8.2.2 "SBR Extension Payload for the Audio Object Types AAC main, AAC SSR, AAC LC and AAC LTP". The signaling of SBR shall be done implicitly as outlined in ISO/IEC 14496-3, subclause 1.6.5 "Signaling of SBR".

8.9 Tables

Table 42 — Channel Configuration

value	number of speakers	audio syntactic elements, listed in order received	element to speaker mapping
0	-	-	**defined in program_config_element() (see subclause 8.5.3.2) or implicitly given (see subclause 8.5.3.3)**
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
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

Table 43 — Transform windows (for 48 kHz)

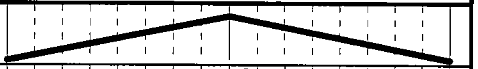
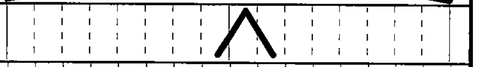
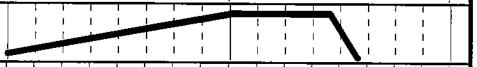
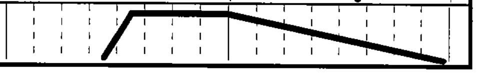
window	num_swb	#coeffs	looks like
LONG_WINDOW	49	1024	
SHORT_WINDOW	14	128	
LONG_START_WINDOW	49	1024	
LONG_STOP_WINDOW	49	1024	

Table 44 — Window Sequences

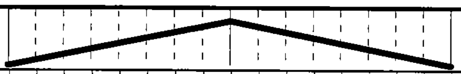
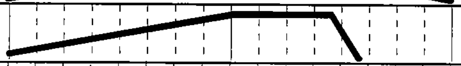
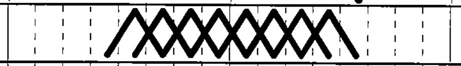
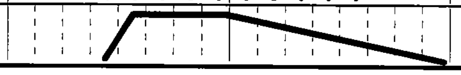
value	window_sequence	num_ windows	looks like
0	ONLY_LONG_SEQUENCE = LONG_WINDOW	1	
1	LONG_START_SEQUENCE = LONG_START_WINDOW	1	
2	EIGHT_SHORT_SEQUENCE = 8 * SHORT_WINDOW	8	
3	LONG_STOP_SEQUENCE = LONG_STOP_WINDOW	1	

Table 45 — Scalefactor bands for LONG_WINDOW, LONG_START_WINDOW, LONG_STOP_WINDOW at 44.1 kHz and 48 kHz

fs [kHz]	44.1, 48
num_swb_long_window	49
swb	swb_offset_long_window
0	0
1	4
2	8
3	12
4	16
5	20
6	24
7	28
8	32
9	36
10	40
11	48
12	56
13	64
14	72
15	80
16	88
17	96
18	108
19	120
20	132
21	144
22	160
23	176
24	196
25	216
26	240
27	264
28	292
29	320
30	352
31	384
32	416
33	448
34	480
35	512
36	544
37	576
38	608
39	640
40	672
41	704
42	736
43	768
44	800
45	832
46	864
47	896
48	928
	1024

Table 46 — Scalefactor bands for SHORT_WINDOW at 32 kHz, 44.1 kHz and 48 kHz

fs [kHz]	32, 44.1, 48
num_swb_short_window	14
swb	swb_offset_short_window
0	0
1	4
2	8
3	12
4	16
5	20
6	28
7	36
8	44
9	56
10	68
11	80
12	96
13	112
	128

Table 47 — Scalefactor bands for LONG_WINDOW, LONG_START_WINDOW, LONG_STOP_WINDOW at 32 kHz

fs [kHz]	**32**
num_swb_long_window	**51**
swb	**swb_offset_long_window**
0	0
1	4
2	8
3	12
4	16
5	20
6	24
7	28
8	32
9	36
10	40
11	48
12	56
13	64
14	72
15	80
16	88
17	96
18	108
19	120
20	132
21	144
22	160
23	176
24	196
25	216
26	240
27	264
28	292
29	320
30	352
31	384
32	416
33	448
34	480
35	512
36	544
37	576
38	608
39	640
40	672
41	704
42	736
43	768
44	800
45	832
46	864
47	896
48	928
49	960
50	992
	1024