BS EN 61937-11:2010

BSI Standards Publication

Digital audio — Interface for non-linear PCM encoded audio bitstreams applying IEC 60958

Part 11: MPEG-4 AAC and its extensions in LATM/LOAS

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The UK participation in its preparation was entrusted to Technical Committee EPL/100, Audio, video and multimedia systems and equipment.

A list of organizations represented on this committee can be obtained on request to its secretary.

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English version

Digital audio - Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 - Part 11: MPEG-4 AAC and its extensions in LATM/LOAS

(IEC 61937-11:2010)

Audionumérique -
Interface pour les flux de bits audio
à codage MIC non linéaire
conformément à la CEI 60958 -
Partie 11 : MPEG-4 AAC
et ses extensions en LATM/LOAS
(CEI 61937-11:2010)

Digitalton – Schnittstelle für nichtlinear-
PCM-codierte Audio-Bitströme
unter Verwendung von IEC 60958 –
Teil 11: MPEG-4-AAC
und dessen Erweiterungen in LATM/LOAS
(IEC 61937-11:2010)

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Foreword

The text of document 100/1491/CDV, future edition 1 of IEC 661937-11, prepared by IEC TC 100, Audio, video and multimedia systems and equipment, was submitted to the IEC-CENELEC parallel vote and was approved by CENELEC as EN 61937-11 on 2010-06-01.

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– latest date by which the EN has to be implemented at national level by publication of an identical national standard or by endorsement	(dop)	2011-03-01
– latest date by which the national standards conflicting with the EN have to be withdrawn	(dow)	2013-06-01

Annex ZA has been added by CENELEC.

Endorsement notice

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IEC 61937 series	NOTE	Harmonized in EN 61937 series (not modified).
IEC 61937-6	NOTE	Harmonized as EN 61937-6.

Annex ZA
(normative)

Normative references to international publications with their corresponding European publications

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.

NOTE When an international publication has been modified by common modifications, indicated by (mod), the relevant EN/HD applies.

Publication	Year	Title	EN/HD	Year
IEC 60958	Series	Digital audio interface	EN 60958	Series
IEC 61937-1	-	Digital audio - Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 - Part 1: General	EN 61937-1	-
IEC 61937-2	-	Digital audio - Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 - Part 2: Burst-info	EN 61937-2	-
ISO/IEC 14496-3	2009	Information technology - Coding of audio-visual objects - Part 3: Audio	-	-

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INTRODUCTION

Modern digital video broadcasting standards such as DVB include support for the MPEG-4 HE AAC and/or HE AAC v2 audio codecs as specified in ISO/IEC 14496-3. An increasing number of countries are adopting these new codecs for their standard definition and high definition digital video broadcasting services and have started with implementations.

For MPEG-2 AAC audio (ISO/IEC 13818-7) the specified framing format for the audio bit stream is ADTS and its transport over an IEC 60958 interface is specified in IEC 61937-6.

However, the MPEG-4 (ISO/IEC 14496-3) audio codecs introduce new features and capabilities that require a framing format that supports more flexible signaling and delivery mechanisms. Therefore, MPEG-2 Systems (ISO/IEC 13818-1) specifies the MPEG-4 LATM/LOAS framing format for MPEG-4 audio codecs to overcome the limitations of ADTS.

In order to be able to pass the MPEG-4 audio bit stream from a Set Top Box to an A/V receiver connected via the IEC 60958 interface without needing to reframe the audio bit stream within ADTS, the MPEG-4 LATM/LOAS framing format needs to be supported by IEC 61937.

DIGITAL AUDIO – INTERFACE FOR NON-LINEAR PCM ENCODED AUDIO BITSTREAMS APPLYING IEC 60958 –

Part 11: MPEG-4 AAC and its extensions in LATM/LOAS

1 Scope

This part of IEC 61937 describes the method to convey non-linear PCM bitstreams encoded according to the MPEG-4 AAC format and its extensions spectral band replication, parametric stereo and MPEG surround, framed in MPEG-4 LATM/LOAS.

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.

IEC 60958 (all parts), *Digital audio interface*

IEC 61937-1, *Digital audio – Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 – Part 1: General*

IEC 61937-2, *Digital audio – Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 – Part 2: Burst-info*

ISO/IEC 14496-3:2009, *Information technology – Coding of audio-visual objects – Part 3: Audio*

3 Terms, definitions and abbreviations

For the purposes of this document the terms, definitions and abbreviations of IEC 61937-1, IEC 61937-2 and the following apply.

3.1 Terms and definitions

3.1.1
access unit
smallest entity to which timing information can be attributed; an access unit is the smallest individually decodable unit; a decoder consumes access units

3.1.2
AudioMuxElement(1)
LATM element that carries payload data for at least one audio elementary stream, related payload length information and multiplex configuration information

NOTE This element carries payload data in form of PayloadMux elements. The number in brackets indicates multiplexing configuration (StreamMuxConfig) is multiplexed into AudioMuxElements, that is in-band transmission.

3.1.3
AudioSpecificConfig
configuration structure used to convey parameters to initialize the MPEG-4 audio decoder

3.1.4
low overhead MPEG-4 audio transport multiplex
LATM
multiplexing layer defined by ISO/IEC 14496-3; used for multiplexing of audio elementary streams

3.1.5
low overhead audio stream
LOAS
synchronisation layer defined by ISO/IEC 14496-3; three different formats of LOAS are defined, each of which is designed to address the specific characteristics of the underlying transmission layer

3.1.6
MPEG-4 AAC profile
contains only the MPEG-4 AAC low complexity audio object type; MPEG-4 AAC low complexity object type is the counterpart to the MPEG-2 AAC low complexity profile; in addition to the MPEG-2 AAC LC profile the MPEG-4 AAC low complexity object type enables the usage of the PNS tool

NOTE The MPEG-4 AAC Low Complexity object type is used when there are restrictions on the usage of RAM and processing complexity.

3.1.7
MPEG-4 high efficiency AAC profile
contains the spectral band replication object type in conjunction with the MPEG-4 AAC low complexity object type

NOTE For further information please refer to ISO/IEC 14496-3. The MPEG-4 high efficiency AAC profile is a superset of the MPEG-4 AAC profile.

3.1.8
MPEG-4 high efficiency AAC profile version 2
contains the parametric stereo object type and the spectral band replication object type in conjunction with the AAC low complexity object type

NOTE The MPEG-4 high efficiency AAC profile version 2 is a superset of the MPEG-4 high efficiency AAC profile.

3.1.9
MPEG surround
technology used for coding of multichannel signals based on a downmixed signal of the original multichannel signal, and associated spatial parameters

NOTE MPEG surround is defined in ISO/IEC 23003-1.

3.1.10
PayloadMux
payload data chunk in an AudioMuxElement that contains potentially multiplexed payload data for multiple audio elementary streams; in general PayloadMux elements can be concatenated inside AudioMuxElements

3.1.11
SpatialSpecificConfig
configuration structure used to initialize the MPEG surround decoder

3.1.12
StreamMuxConfig
configuration structure that describes the structure of the LATM payload multiplex

3.1.13
Sub-data-type
reference to the type of payload of the data-bursts defined for the use with the specified data-type

3.1.14
modified discrete cosine transformation
MDCT
transformation schema used by AAC

3.1.15
transformation length (of the AAC codec or core codec)
AAC can operate in two modes using either a 960 lines or 1 024 lines MDCT transformation for long blocks; an MDCT line is a spectral component described by frequency, amplitude and phase

3.2 Abbreviations

AAC	Advanced Audio Coding
AAC LC	MPEG-4 AAC Low Complexity
HE AAC	MPEG-4 High Efficiency AAC and MPEG-4 High Efficiency AAC Version 2
ADTS	Audio Data Transport Stream
DVB	Digital Video Broadcasting
MDCT	Modified Discrete Cosine Transformation
MPEG	Moving Picture Experts Group
MPS	MPEG Surround
PNS	Perceptual Noise Substitution
PS	Parametric Stereo
SBR	Spectral Band Replication
TL	AAC Transformation Length

4 Mapping of the audio bit stream on to IEC 61937-1

4.1 General

The coding of the bit stream and data-burst is in accordance with IEC 61937-1 and IEC 61937-2.

4.2 Burst-info for MPEG-4 AAC and its extensions in LATM/LOAS

The 16-bit burst-info contains information about the data which will be found in the data-burst (see Table 1).

Table 1 – Values for data-type and sub-data-type

Data-type according to IEC 61937-2 Value of Pc bits 0–4	Sub-data-type Value of Pc bits 5–6	Contents	Reference point R	Repetition period of data-bursts in IEC 60958 frames
0–22	0–3	According to IEC 61937		
23	0	According to IEC 61937-10		Definition specific to IEC 61937-10
	1	AAC LC	Bit 0 of Pa	960 / 1 024
	2	HE AAC	Bit 0 of Pa	1 920 / 2 048
	3	Reserved for future definition of other applications	reserved	Reserved for future definition of other applications
24–31	0–3	According to IEC 61937		
Bits 0–4 of the burst-info (Pc) signal the data-type used for transmission. For MPEG-4 AAC-based audio in LATM/LOAS, the signaled data-type is 23. The Pc bits 5–6 indicate if the transmitted data stream contains audio encoded in AAC LC or HE AAC (including high efficiency AAC version 2). Only values 1 and 2 refer to the transmission of AAC LC or HE AAC based audio. The values 0 and 3 are used for indication of codec types which are described by other or future parts of IEC 61937.				

5 Format of data-burst for MPEG-4 AAC and its extensions in LATM/LOAS

5.1 General

This clause specifies the data-burst for MPEG-4 AAC audio and its extensions in LATM/LOAS. Specific properties such as reference points, repetition period, the method of filling stream gaps and decoding latency are specified.

The decoding latency (or delay), indicated for the sub-data-types, should be taken into account by the transmitter to schedule data-bursts as necessary to establish synchronisation between picture and decoded audio.

5.2 Pause data-bursts for MPEG-4 AAC and its extensions in LATM/LOAS

Pause data-bursts for MPEG-4 AAC and its extensions in LATM/LOAS are defined in Table 2.

Table 2 – Repetition period of pause data-bursts

Data-type of audio data-burst	Repetition period of pause data-burst	
	Mandatory	Recommended
Sub-data-type for MPEG-4 audio in LATM/LOAS based on MPEG-4 AAC core codec	—	64 IEC 60958 frames

If regular audio data-bursts are not being transmitted due to for example a PAUSE condition, it is recommended to use pause data-bursts to fill such stream gaps. The repetition period of the pause data-bursts should be selected according to Table 2. If other repetition periods are necessary to precisely fill the stream gap length, or to meet the requirement on audio data-bursts spacing (see IEC 61937) pause data-bursts may have other lengths which may not be an integer multiple of 64 IEC 60958 frames.

When a stream gap in an audio stream is filled by a sequence of pause data-bursts, the Pa of the first pause data-burst shall occur after exactly that amount of IEC 60958 frames as indicated by the AAC transformation length in conjunction with the codec type information from Table 3. It is recommended that the sequence(s) of pause data-bursts which fill the stream gap should continue from this point up to the Pa of the first audio data-burst which follows the stream gap, or as close as possible considering the specific IEC 60958 frame length of the pause data-burst with respect to the AAC core codec transformation length. The repetition-period-length parameter contained in the pause data-burst is intended to be interpreted by the receiver as an indication of the number of decoded PCM samples that are missing (due to the resulting audio gap).

5.3 Audio data-bursts

5.3.1 MPEG-4 AAC and its extensions in LATM/LOAS

The stream of data-bursts consists of sequences of MPEG-4 AAC and its extensions in LATM/LOAS frames. Each data-burst consists of a preamble followed by the payload and stuffing. The data-type of a data-burst according to this specification is 23.

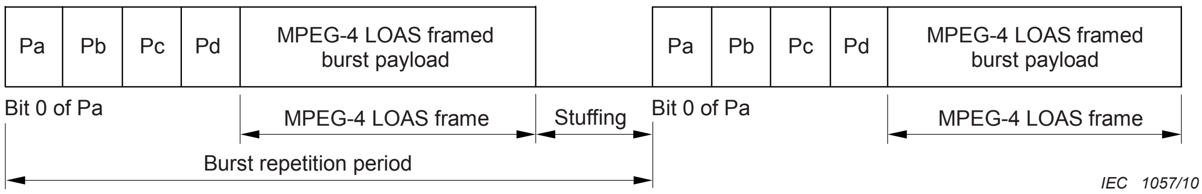


Figure 1 – Data-burst structure

The length of the audio payload data in the data-burst depends on the bit rate and other parameters of the encoded audio. The size of the data-burst payload is indicated by the Pd preamble word and is measured in bits.

P_{AD} = 4 × 16 bit (P_{AD} is the size of the preamble words P_A – P_D measured in bits)

B_S = 4 × 16 bit (B_S is the size of the burst spacing measured in bits.)

TL = 1 024 or 960 lines (*TL* is the used MDCT transformation length in MDCT lines)

The maximum data-burst payload size for AAC not utilizing SBR is calculated according to the following equation:

$$2 \times 16 \text{ bit} \times TL - (P_{AD} + B_S) = \text{maximum payload size in bits.}$$

If HE AAC is used the maximum data-burst payload size is calculated according to the following equation:

$$4 \times 16 \text{ bit} \times TL - (P_{AD} + B_S) = \text{maximum payload size in bits}$$

The data-type-dependent information for MPEG-4 AAC and its extensions in LATM/LOAS is given in Table 3. Bits 8–12 of Pc contain information about the audio codec used and about the LATM configuration.

Table 3 – Data-type-dependent information

Bits of Pc LSB…MSB	Value	Definition	Description
8	0	AAC Transformation Length	1 024 lines
	1		960 lines
9	0	PS	PS data not present
	1		PS data present
10–11	0	MPS	MPS data not present
	1		Embedded MPS data present / LATM single layer transport mode (implicit MPS signaling)
	2		Embedded MPS data present / explicit signaling of MPS in second LATM layer
	3		Do not use until further definition
12	0	Reserved	Set to “0” until further definition
	1		Do not use until further definition

The Pc bit 8 indicates the transformation length of the AAC core codec which is used to encode the transmitted audio stream. Information from Pc bit 8 does not define the repetition period of data-bursts on its own. This information is required in conjunction with the codec signaled by the sub-data-type to calculate the data-burst repetition period. Receivers shall read the sub-data-types as well as the data-type-dependent information in order to compute the repetition period of data bursts. Examples can be found in Annex A.

The Pc bit 9 indicates whether PS data is present in the encoded audio stream.

The two Pc bits 10–11 indicate the presence and transport configuration of MPS data in the encoded audio stream. The value 0 indicates that no MPS data is present. Values 1 and 2 indicate that MPS data is present in the audio bit stream. Pc bits 10-11 signaling 1 indicates that the MPS payload as well as the MPS SpatialSpecificConfig is embedded inside the payload of the first LATM layer which conveys the AAC LC or HE AAC data stream. Pc bits 10–11 signaling 2 indicates that the MPS payload is also embedded inside the payload of the first layer. But in this case the MPS SpatialSpecificConfig is signaled explicitly associated to the second layer inside the LATM StreamMuxConfig. The value 3 signaled by the Pc bits 10–11 is reserved for future use.

This specification does not allow the transmission of MPS payload which is not embedded inside the AAC LC or HE AAC payload but resides separated from the AAC LC or HE AAC payload inside another LATM layer.

The presence of the PS or MPS extensions does not influence the data-burst repetition rate or the calculation of the transmission and decoding latency as described in 5.3.3.

The Pc bit 12 is reserved for future use. This bit shall be set to 0.

One complete AAC access unit represents a time interval of 1 024 or 960 audio samples embedded into the data-burst payload. When transmitting MPEG-4 HE AAC encoded audio programs, SBR is used as an extension to AAC. In this case the sampling frequency of the MPEG-4 AAC core component is usually half the sampling frequency of the SBR tool and audio program. One complete HE AAC access unit represents a time interval of 2 048 or 1 920 audio samples embedded into the data-burst payload.

HE AAC bit streams with downsampled SBR shall be transmitted signaling AAC LC in the sub-data-type. Therefore one access unit corresponds to 1 024 or 960 AAC encoded audio samples.

If HE AAC is signaled by the sub-data-type the IEC 60958 frame rate shall be equal to the sampling frequency of the SBR tool. If AAC LC is signaled the IEC 60958 frame rate shall correspond to the sampling frequency of AAC.

The reference point of a data-burst is bit 0 of Pa and occurs exactly once every number of IEC 60958 sampling periods which is computed using the information from Table 1 and Table 3. The data-burst containing one LATM/LOAS audio frame shall occur at a constant rate. The intervals for data-bursts sharing the same bit-stream-number shall correspond exactly to the amount of IEC 60958 frames which is calculated using the information from Table 1 and Table 3.

It is not allowed to transmit audio data streams using IEC 60958 frame rates below 32 kHz.

5.3.2 LATM/LOAS framing

The LOAS frame as described in ISO/IEC 14496-3 shall be mapped directly to the payload section, right after the preamble words of the data-burst. The first bit of the LOAS frame shall always correspond to the first bit after the preamble section in the data-burst.

The payload in a data-burst consists of one complete LOAS frame containing one LATM AudioMuxElement. It is not allowed to convey one LATM/LOAS frame using multiple data-bursts. LOAS frames exceeding the payload capacity of a data-burst shall be dropped and the actual data-burst shall be replaced by a sequence of pause-bursts to match the duration of that data-burst.

The parameter numSubFrames from the LATM StreamMuxConfig shall be 0. The parameter numProgram from the LATM StreamMuxConfig shall be 0. The parameter numLayer from the LATM StreamMuxConfig shall be 0 except for audio streams signaling the availability of MPS with payload embedding and explicit signaling of MPS in the second LATM layer. In such cases the presence of a second layer in LATM frames is allowed and therefore numLayer shall be 1 indicating 2 layers. In this configuration there exists no payload associated to the second LATM layer and therefore the payload size indication for the second layer in LATM is set to zero.

Only the LOAS AudioSyncStream() scheme shall be used in the context of this specification.

The LATM StreamMuxConfig structure shall be conveyed inside the LATM multiplex. This is the main structure that is utilized by the decoder for configuration. The StreamMuxConfig may not be present in each LATM frame in order to save bandwidth. It may be sent in intervals to allow decoders to tune in to a running stream.

5.3.3 Latency

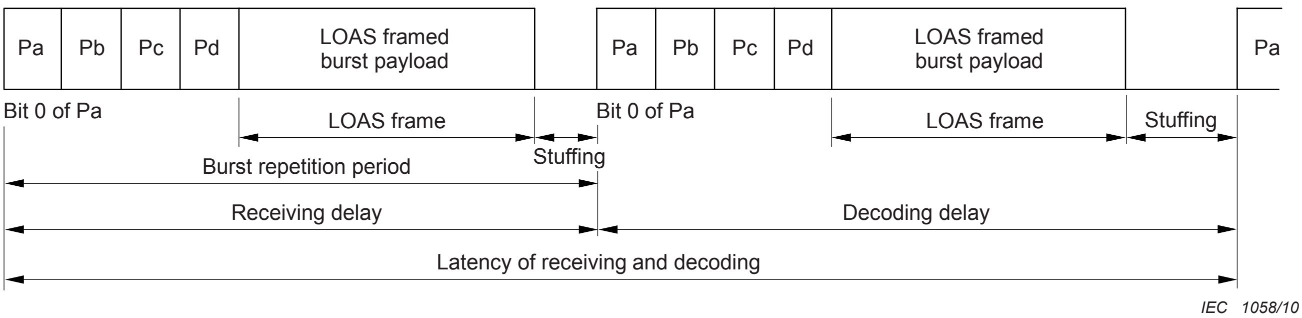
The latency of an external audio decoder to decode MPEG-4 AAC and its extensions in LATM/LOAS is defined as the sum of the receiving time of the audio payload in one data-burst and the time used for decoding of one access unit.

Each data-burst contains a minimum of 4 stuffing words (Pz of 16 bits). The repetition period of data-bursts in IEC 60958 frames is computed according to information from Table 1 and Table 3. The reception delay for one audio access unit is calculated as the time elapsed counting from the first bit of the data-burst until the last bit of the actual audio payload inside the data-burst received. Subsequent stuffing is not taken into account. After a complete frame is received immediate decoding and subsequent rendering of the audio frame is not recommended as the size of the next audio frame and therefore time required for receiving it completely cannot be determined accurately.

In order to simplify the timing mechanism for receiving and decoding of content of data-bursts, the receiving delay should be calculated as the time necessary to receive the complete data-burst including the stuffing. The maximum time available to decode (the decoding delay) should be selected to correspond to the length of one full data-burst. This results in an overall delay corresponding to two complete data-bursts for reception and decoding.

For synchronisation (for example with video), the recommended value for latency corresponds to the time necessary to receive two complete data-bursts. Picture 3 shows the simplified and recommended method for calculating the latency for reception and decoding.

A shorter latency may be acceptable if synchronisation is not required.



NOTE This diagram shows the recommended method.

Figure 2 – Latency diagram for burst reception and decoding

Annex A
(informative)

Calculation of delay and data-burst repetition rates, guidelines

A.1 Examples

Some examples for the calculation of data-burst-repetition rates and latencies can be found in Table A.1.

Table A.1 – Examples – Calculation of delay and data-burst repetition rates

Bits 5-6 of Pc codec indication	**Bit 8 of Pc TL** lines	**AAC sampling rate** kHz	**SBR sampling rate** kHz	**IEC 60958 frame rate** kHz	**Data-burst repetition rate** IEC 60958 frames	**Overall latency** ms
AAC LC	1 024	32	n/a	32	1 024	64
		44,1	n/a	44,1		46,44
		48	n/a	48		42,67
		96	n/a	96		21,33
	960	32	n/a	32	960	60
		44,1	n/a	44,1		43,54
		48	n/a	48		40
		96	n/a	96		20
HE AAC	1 024	16	32	32	2 048	128
		22,05	44,1	44,1		92,88
		24	48	48		85,33
		48	96	96		42,67
	960	16	32	32	1 920	120
		22,05	44,1	44,1		87,07
		24	48	48		80
		48	96	96		40

The presence of the PS or MPS extensions signaled by Pc bits 9–11 does not influence the data-burst repetition rate or the calculation of the transmission and decoding latency as described in 5.3.3.

A.2 Guidelines

The following guidelines should be taken into account.

a) Receivers which receive an indication in the data-burst-dependent information that signals the presence of MPS, but that are not capable of decoding MPS, should not refuse decoding of that stream. It is highly recommended that non-MPS capable decoders decode just the AAC LC / HE AAC channel configuration as indicated by the downmix codec configuration record and ignore the MPS extension in the bit stream.

b) The IEC 60958 frame rate may be calculated by making use of the audio sampling rate indication from the AudioSpecificConfig inside the LATM StreamMuxConfig. It is highly recommended that the correct codec indication as well as the matching sampling frequency or IEC 60958 frame rate indication is available before starting transmission of IEC 61937-11 data bursts. In case of signaled audio configuration changes upstream it is highly recommended that audio data-bursts referring to the new program are only transmitted after the relevant information (new codec and new sampling frequency) is available to the transmitter and signaled properly.

c) It is highly recommended that decoders do not attempt to decode an audio stream before they have received the corresponding decoder configuration records (e.g. AudioSpecificConfig).

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IEC 61937-10, *Digital audio – Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 – Part 10: Non-linear PCM bitstreams according to the MPEG-4 Audio Lossless Coding (ALS) format* (under consideration)

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

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

ISO/IEC 23003-1:2007, *Information technology – MPEG audio technologies – Part 1: MPEG Surround*[1]

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[1] NOTE Technical corrigendum 1 from 2008 has to be applied.

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