Digital terrestrial television broadcasting – Audio coding

Televisão digital terrestre – Codificação de vídeo, áudio e multiplexação – Parte 2: Codificação de áudio

Televisión digital terrestre — Codificación de video, audio y multiplexación – Parte 2: Codificación de audio

Digital terrestrial television – Video coding, audio coding and multiplexing – Part 2: Audio coding

デジタル放送における映像符号化、音声符号化及び多重化方式　第２部　音声信号と符号化方式

Video coding, audio coding, and multiplexing specifications for digital broadcasting – Part 2: Audio signal and coding system

Foreword

This document is the result of the joint efforts of the ABNT, ARIB and SBTVD Forum under the standardization and technical cooperation activities of the Brazil-Japan Digital Television Joint Working Group.

The Brazilian Association for Standardization (ABNT) is the organism responsible for technical standardization in Brazil, providing essential support for Brazilian technical development. It is private, non-profit organization, recognized as the only National Standardization Body. It provides Brazilian society with systematic knowledge, through normative documents, enabling the production, commercialization and use of goods and services, in a competitive and sustainable manner, in the internal and external markets, contributing to scientific and technological development, environmental protection and consumer's protection.

The Association of Radio Industries and Businesses (ARIB) was designated as "the Center for Promotion of Efficient Use of the Radio Spectrum" and "the Designated Frequency Change Support Agency" by the Minister of Internal Affairs and Communications (MIC) of Japan under the provisions of the Radio Law. Under this designation, ARIB conducts studies and R&D, establishes standards, provides consultation services for radio spectrum coordination, cooperates with other overseas organizations and provides frequency change support services for the smooth introduction of digital terrestrial television broadcasting. These activities are carried out in cooperation with and/or participation by telecommunication operators, broadcasters, radio equipment manufacturers and related organizations as well as under the support by MIC.

The Brazilian Digital Terrestrial Television Forum (SBTVD Forum) is a non-profit entity, created with the objective of aiding and stimulating the development and implementation of best practices aiming at the success of systems reality for digital broadcasting of images and sounds in Brazil. Since the creation of the SBTVD Forum in February, 2007, its members have endeavored to establish standards of technical quality which permit deployment of digital television in Brazil. The Technical Module has contributed to the preparation of standards, with active participation by universities, research centers, related industry organizations and broadcasters.

This document does not describe the industrial property rights mandatory to these standards.

This document has no standardization value. Its purpose is to serve as a reference for characterizing the specificities of Brazilian and Japanese digital terrestrial television standards within the scope of the Brazil-Japan Digital Television Joint Working Group.

This document is drafted in accordance with the rules established in the ISO/IEC Directives, Part 2.

In the Brazilian and Japanese harmonized documents, commonalities are described in Clause 5 where Table 1 includes all references to ABNT and ARIB related documents. Differences are described in Clause 6. In each subclause, a reference to the corresponding Brazilian and Japanese related session is included in separate boxes in *italic text*.

No reference is made to the domestic policies of the countries.

1 Scope

This document addresses the standard for the transmission system for digital terrestrial television broadcasting in Brazil and Japan.

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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.

ABNT NBR 15602-1:2007, *Digital terrestrial television – Video coding, audio coding and multiplexing – Part 1: Video coding*

ABNT NBR 15602-2:2007, *Digital terrestrial television – Video coding, audio coding and multiplexing – Part 2: Audio coding*

ABNT NBR 15602-3:2007, *Digital terrestrial television – Video coding, audio coding and multiplexing – Part 3: Signal multiplexing systems*

ABNT NBR 15603-2:2017, *Digital terrestrial television – Multiplexing and service information (SI) – Part 2: Data structure and definitions of basic information of SI*

ARIB STD-B21:V5.9:2016, *Receiver for digital broadcasting*

ARIB STD-B32:V3.9:2016, *Video coding, audio coding, and multiplexing specifications for digital broadcasting*

ARIB TR-B14:V6.2:2016*, Operational guidelines for digital terrestrial television broadcasting*

ISO/IEC 13818-1:2007, *Information technology – Generic coding of moving pictures and associated audio information: Systems*

ISO/IEC 13818-7:2003, *Information technology – Generic coding of moving pictures and associated audio information: Advanced audio coding*

ISO/IEC 14496-2:2004/Amd 4:2008, *Information technology – Coding of audio-visual objects – Part 2: Visual*

ISO/IEC 14496-3:2005, *Information technology – Coding of audio-visual objects – Part 3: Audio*

3 Terms and definitions

For the purposes of this document, the terms and definitions given in ABNT NBR 15602-2, and ARIB STD-B32, part 2, apply.

4 Abbreviated terms

For the purposes of this document, the abbreviated terms given in ABNT NBR 15602-1, and ARIB STD-B32, part 2, apply.

5 Commonalities of the audio coding system

The common parts of ABNT NBR 15602-2 and ARIB STD-B32, part 2, and how they correspond are described in Table 1.

Table 1 — Correspondence between ABNT NBR 15602-2 and ARIB STD-B32, part 2

Description	ABNT NBR 15602-2 reference clause	ARIB STD-B32, part 2 reference clause
Audio input format [a]	5.1	2
Audio coding system [b]	7	3.1
Audio compression and transmission	8	4.1

procedures [b]		
Audio coding parameter restrictions for full-seg service – audio coding modes [c]	9.1.1, Table 3	5.1
Audio coding parameter restrictions for full-seg service – Channel configurations	9.1.1, Table 4	5.2.3 (2)
Audio coding parameter restrictions for full-seg service – Operational restrictions with respect to stereo receiver compatibility [d]	9.1.3	5.2.4

[a] Some items have different specifications as shown in Subsection 6.1 of this document.

[b] ABNT NBR 15602-2 refers to MPEG-4 AAC, while ARIB STD-B32 part 2, refers to MPEG-2 AAC. It should be noted that the basic coding algorithm of MPEG-4 AAC and MPEG-2 AAC is almost the same, but there are differences between the standards. The differences are shown in Subsection 6.3 of this document.

[C] ABNT NBR 15602-2 adopts "2/0 + LFE" as one of the audio modes, which ARIB STD-B32 part 2 does not. ABNT NBR 15602-2 recommends two modes, 2/0 and 3/2 + LFE, among several modes specified, while other modes are also recommended in ARIB STD-B32 part 2.

[d] The basic ideas are the same but the transport formats are different.

6 Differences in the audio coding system

6.1 Audio input format

6.1.1 General conditions

The quantization resolution of audio signals and the number of audio channels are specified differently.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Subclause 5.1:

5.1 General conditions

The general conditions for the audio input format shall be the following:

a) sample frequency of audio signal: 32 kHz, 44,1 kHz or 48 kHz;

b) configuration of stereophonic and multichannel signals (in other words, signals consisting of two or more audio signals to obtain an evolving reproduction or spatial sound): sample rate for all the signals shall be the same;

c) quantization of input signals shall use 16 bits or 20 bits;

d) one audio program shall have at least one audio channel. The maximum number of channels in the program shall be limited to maximum number of channels allowed for ISO/IEC 14496-3;

e) it is recommended that multichannel programs be prepared in accordance with ITU Recommendation BS.775-1;

f) for audio programs in multichannel mode compatible with the modes foreseen in the standard ITU Recommendation BS.775-1, this shall be in one of the permitted configurations presented on Table 3;

g) where a multichannel program is transmitted without a stereo program, the multichannel program.

In the Japanese digital terrestrial television system, according to ARIB STD-B32, part 2, chapter 2.
ARIB STD-B32 prescribes technical systems used for not only digital terrestrial broadcasting but also other broadcast systems in Japan. Only digital terrestrial broadcasting related sentences are cited below:

Chapter 2: Audio Input Signal

(1) The sampling frequency for audio signals shall be 32 kHz, 44.1 kHz, or 48 kHz.

(2) To configure stereophonic signals (consisting of two or more audio signals to achieve a three-dimensional reproduction of sound), the sampling timing for all signals shall be the same.

(3) The number of quantization bits for the input signal shall be 16 or more.

(4) The maximum number of audio input channels shall be five, in addition to the channel used to enhance low frequencies.

6.1.2 Main parameters

ABNT NBR 15602-2 specifies formats, interfaces, audio signal levels, and metadata for audio input format, while ARIB STD-B32, part 2, does not specify these items.

6.2 Audio services and auxiliary channels

No such topic related to audio services and auxiliary channels could be found in ARIB STD-B32, part 2.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Clause 6:

6 Audio services and auxiliary channels

Audio services include transmission of additional audio programs to the main program and shall be considered optional services, excluding the audio description channel whose transmission is required by current legislation.

The transmission of these services takes place through the allocation of additional auxiliary channels in distinct audio programs (PID) or in the same bitstream of a single PID, with due regard for the maximum number of channels permitted in the same bitstream by the coding profile/level used.

Additional channels to the main program may be used to transmit audio in other languages (for instance, SAP), to transmit additional programs to the main program, audio description services (AD), and secondary audio from other sound takes (additional content, such as effects).

All additional channels referring to auxiliary audio services shall be appropriately signaled using valid component_type identification in the respective audio_component_descriptor of the program.

The auxiliary channels shall be transmitted in distinct programs (distinct PID) with proper signaling and channel identification to be selected, decoded and played with, or in substitution of, the main audio program channels.

The audio description service usually consists of a single-voice monaural channel and provides a scene description as a subcomponent associated with the television service. It must support comprehension of the main entertainment (but not exclusively) for viewers with visual impairment.

The AD transmission shall be implemented using at least one of the mechanisms below:

a) as an auxiliary channel (monaural or stereo) containing the AD previously mixed with the main audio program;

b) as a auxiliary channel containing separate AD for later processing with the main audio program;

In both cases, the service should be signaled on the component_type parameter described in the "Audio component descriptor", as per ABNT NBR 15603-2, Table 28.

The ability to mix one or more supplementary description channels with the main audio program may have other applications, including multilingual commentary, interactivity and educational purposes.

6.3 Audio coding system

ABNT NBR 15602-2 adopts MPEG-4 AAC defined in ISO/IEC 14496-3, while ARIB STD-B32, part 2, subclause 3.1, adopts MPEG-2 AAC defined in ISO/IEC 13818-7. Although the basic coding algorithms of MPEG-4 AAC and MPEG-2 AAC are almost the same, there are differences between the standards as listed below. Therefore, no direct comparison could be established on most of the items under coding parameter restrictions.

— MPEG-4 AAC defines a mandatory tool called PNS (pseudo noise subtraction) for the LC profile, which is not defined in MPEG-2 AAC;

— MPEG-4 AAC adopts the LATM/LOAS for the transport of coded streams, while MPEG-2 AAC adopts ADTS;

— MPEG -4 AAC specifies the extended coding algorithms such as HE-AAC, which are not covered by MPEG-2 AAC.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Clause 7:

7 Audio coding system

Audio signals shall be coded by a combination of time-frequency transform coding. The frequency transform shall decompose the input signal into its frequency components by means of a modified discrete cosine transform (DCT), which reduces the amount of information by reducing the decrease in the frequency deviation of each component.

An additional compression tool is the psycho-acoustic weighted bit assignment in which codes shall be weighted to minimize signal degradation in the frequency bandwidth perceived by human hearing.

Audio compression and transmission procedures shall comply with ISO/IEC 14496-3.

Decoders shall be made under the assumption that any legal structure as permitted by ISO/IEC 13818-1 may occur in the broadcast stream, even if presently reserved or unused. The audio decoder shall be able to skip over "reserved" structures and data structures which correspond to functions not implemented by receivers.

6.4 Audio compression and transmission procedures

Basically ABNT NBR 15602-2 and ARIB STD-B32, part 2, differ in the ISO/IEC 14496-2 regarding the coding system. ABNT NBR 15602-2 takes into account MPEG-4 AAC while ARIB STD-B32, part 2, subclause 4.1, takes into account MPEG-2 AAC.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Subclauses 8.2 and 8.3:

8.2 Profiles and levels

The audio coding shall be compatible with ISO/IEC 14496-3. The following profiles and levels of MPEG-4 AAC standard shall be permitted:

a) LC (low complexity), basic profile of AAC standard; L2 and L4 levels;

b) HE (high efficiency), advanced profile of high efficiency, combining the LC profile with the use of the SBR (spectral band replication) tool for version 1 of this profile, L2 and L4 levels;

c) HE combined with PS (parametric stereo) tool for version 2 of this profile, L2 level.

The profile and level of the MPEG-4 AAC coder shall be adequately signalized according to ABNT NBR 15602-3 and ABNT NBR 15603-2.

8.3 Transport and multiplex layer

Intermediate audio coding and framing shall be compatible with LATM/LOAS according to ISO/IEC 14496-3. The elementary stream shall firstly be encapsulated in the LATM transport format and shall use the AudioMuxElement() multiplex element.

The audio transport synchronization layer (LOAS) shall use the AudioSyncStream() transmission format, according to ISO/IEC 14496-3.

The MPEG-4 audio transported by the MPEG-2 transport stream using the LATM/LOAS transport syntax shall be identified by the stream_type 0x11, according to the stream_type_assignments in ISO/IEC 13818-1.

To decode audio, the receiver shall identify the type, profile and level transmitted and shall be capable of extracting the audio object payloads. It is mandatory to use explicit SBR signaling without PES alignment to transmit MPEG-4 *audio over MPEG-2 transport streams.*

Receivers shall be capable of processing the SBR tool. The SBR presence signaling shall be explicit using the non-backward compatible explicit signalization mechanism, according to ISO/IEC 14496-3.

Table 2 describes the LATM/LOAS transport syntax fields within StreamMuxConfig(), which shall be formatted to identify and recover audio payloads, according to ISO/IEC 14496-3.

6.5 Audio coding modes

ABNT NBR 15602-2 refers directly to multichannel modes while ARIB STD-B32, part 2, Subclause 5.1, refers to input audio formats as multichannel modes. ABNT NBR 15602-2 recommends only two audio modes for full-seg operation. ARIB STD-B32, part 2, refers to “emphasis”, ABNT NBR 15602-2 does not. ABNT NBR 15602-2, Table 3, inserts a “downmix” clarification, which does not exist in ARIB STD-B32, Part 2.

In ABNT NBR 15602-2, Table 4, names “coding modes” using both terminologies “5.1” and “3/2+LFE”. ABNT NBR 15602-2 does not refer to 2/1 and 2/2 audio coding modes, and explains dual mono coding mode after table 4.

ADTS comments are not present in ABNT NBR 15602-2.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Subclause 9.1.1:

9.1 Audio coding parameter restrictions for full-seg services

9.1.1 Audio coding modes

The coding mode determines the number of available channels on the audio service. The audio coding modes for digital transmission shall be in accordance with the restrictions of Table 3.

Table 1 - Restrictions on audio coding modes

Parameter	***Restriction***
Permitted audio modes	*Monaural (1/0), stereo (2/0 and 2/0 + LFE)[a], multichannel stereo (3/0, 2/1, 3/1, 2/2, 3/2, 3/2 + LFE)[a], two independent audio signals (dual monaural), multi-audio (three or more audio signals) and a any combination of the above modes*
Recommended audio modes	*Stereo (2/0), multichannel (3/2 + LFE)*
Downmix	*The signaling presented in Table 1 shall be used for 5.0 and 5.1 configurations. In any other multichannel configuration, the receiver can use other downmix schemes, since audio intelligibility is assured. The stereo-to-mono downmix scheme is not covered by this Standard, but clipping shall be avoided*
[a] *Number of channels for front/surround loudspeakers.*	

EXAMPLE 3/1 = 3 front + 1 surround; 3/2 = 5.0 = 3 front channels and 2 surround.

The decoder shall be capable of processing any of the recommended audio modes.

The second channel configuration according to its operation mode, and its transmission order within the payload, shall be in accordance with Table 4.

Table 2 - Channel configurations and recommended modes for MPEG-4 AAC

Mode	***Channel configuration***	***SE order of transmission [a]***	***Standard element for loudspeaker mapping [b]***
Monaural (1/0)	*1*	*<SCE1><TERM>*	*SCE1 = C*
Stereo (2/0)	*2*	*<CPE1><TERM>*	*CPE1 = L and R*
3/0	*3*	*<SCE1><CPE1><TERM>*	*SCE1 = C, CPE1 = L and R*
3/1	*4*	*<SCE1><CPE1><SCE2>*	*SCE1 = C, CPE1 = L and R,*

		<TERM>	SCE2 = MS
Multichannel 5.0 (3/2)	5	*<SCE1><CPE1><CPE2> <TERM>*	*SCE1 = C, CPE1 = L and R, CPE2 = LS and RS*
Multichannel 5.1 (3/2 + LFE)	6	*<SCE1><CPE1><CPE2> <LFE><TERM>*	*SCE1 = C, CPE1 = L and R, CPE2 = LS and RS, LFE = LFE*

[a] Abbreviations related to the syntactic element (SE): SCE – single channel element, CPE – channel pair element, LFE – LFE channel element, TERM – terminator.
[b] Abbreviations related to loudspeaker arrangement: L – front-left loudspeaker / R – front-right loudspeaker / C – front-central loudspeaker / LFE – low frequency enhancement / LS – left-surround loudspeaker / RS – right-surround loudspeaker / MS – monaural surround loudspeaker.

Where two independent audio signals are transmitted (monaural dual or 1/0 + 1/0) the recommended SE order of transmission is: <SCE1><SCE2><TERM>, SCE1 being the first (main) channel and SCE2 the second program channel.

If the configuration used is not present on Table 4, it shall be reproduced using a configuration with the same number of channels and with the respective signaling.

In the Japanese digital terrestrial television system, according to ARIB STD-B32, part 2, Subclause 5.1:

5.1 Input audio format based on MPEG-2 AAC System

The input audio format for digital broadcasting is subject to the following restrictions:

Parameter		Restriction
Audio mode	*Possible audio mode*	*mono, stereo, multichannel stereo (3/0, 2/1, 3/1, 2/2, 3/2, 3/2+LFE (3/2.1))* (Note)*, 2-audio signals (dual mono), multi-audio (3 or more audio signals) and combinations of the above*
	Recommended audio mode	*mono, stereo, multichannel stereo (3/1, 3/2, 3/2+LFE (3/2.1))* (Note)*, 2-audio signals (dual mono)*
Emphasis		*None*

(Note) *Notation for audio mode of multichannel stereo: Audio mode of multichannel stereo is denoted as "front/rear.LFE".*
There is a case to denote "+ LFE" when the assigned channel for LFE (low frequency enhance effect channel) is one.
There is a related record about notation for audio mode in Description 2.

6.6 Main parameters

The transport scheme is ADTS in ARIB STD-B32, part 2, and LATM/LOAS in ABNT NBR 15602-2; four profiles/levels are permitted in ABNT NBR 15602-2, while only LC (Low Complexity) is permitted in ARIB STD-B32, part 2, chapter 5; maximum number of channels in ABNT NBR 15602-2 refers to ISO/IEC 14496-3, while ARIB STD-B32, part 2 sets it at 5.1 per ADTS; ABNT NBR 15602-2 includes information on samples/frame and recommendations of profiles/levels for high fidelity; ABNT NBR 15602-2 states that DRC controls of MPEG-4 may be used.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Subclause 9.1.2:

9.1.2 Main parameters

The audio coding system main parameters shall be as presented in Table 5.

Table 3 - Main audio coding parameters – Full-seg services

Parameter	*Restriction*
Allowed transport	*LATM/LOAS (according to ISO/IEC 14496-3)*

mechanisms	
Recommended channel number	*Mono (1.0), 2 channels (stereo or 2.0) or multichannel (5.1)*
Allowed profiles and levels	*Low complexity AAC: level 2 (LC-AAC@L2) for two channels* *Low complexity AAC: level 4 (LC-AAC@L4) for multichannel* *High-Efficiency (HE): level 2 (HE-AAC v1@L2) for two channels* *High-Efficiency (HE): level 4 (HE-AAC v1@L4) for multichannel*
Maximum allowed bit rate	*In accordance with ISO/IEC 14496-3*
Samples per frame	*frameLengthFlag in GASpecificConfig() shall be set to 0, indicating that the frame length shall be 1024 samples for AAC and 2048 when using SBR. 960 samples for AAC (or 1920 when using SBR) are not allowed*

For high-fidelity transmission it is recommended using the AAC@L4 profile/level in the multichannel mode and AAC@L2 profile/level for stereo mode. In stereo audio transmission, level 4 (L4) shall not be used.

Signals may be encoded on any bit rate supported by the selected profile and level. At the same time, the multichannel signal may use any of the profile sample rates.

MPEG-4 AAC dynamic range control tools may be used.

In the Japanese digital terrestrial television system, according to ARIB STD-B32, part 2, Subclause 5.2.1:

5.2.1 Main parameters

Parameter	*Restriction*
Bitstream format	*AAC Audio Data Transport Stream (ADTS)*
Profile	*Low Complexity (LC) profile*
Max. number of coded channels	*5.1 channels[(Note)] per ADTS*
Max. bitrate	*Compliant with ISO/IEC 13818-7*

(Note) 5 channels + LFE channel

6.7 Operational restrictions with respect to stereo receiver compatibility

ABNT NBR 15602-2, Subclause 9.1.3, and ARIB STD-B32, part 2, Subclause 5.2.4, approach the same issues, but with differences: ABNT NBR 15602-2 refers to the ability of receptors to decode PCE downmix coefficients transmitted; ARIB STD-B32, part 2, Subclause 5.2.4 (3) refers to ADTS and is not covered in ABNT NBR 15602-2, Subclause 9.1.3.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Subclause 9.1.3:

9.1.3 Operational restrictions with respect to stereo receiver compatibility

When the multichannel service is available:

a) transmission shall occur with a minimum of one program in two channels (2/0 or stereo) or one multichannel program (3/2);

b) simultaneous transmission of two channels is not mandatory when the multichannel service is available. Basically, two-channel receivers (stereo) shall be capable of processing the signal through downmixing;

c) the receiver shall be capable of interpreting the downmix coefficient using PCE according to the AAC standard (see Table 1) when the 5 (3/2) and 5.1 (3/2 + LFE) channel services are available.

In the Japanese digital terrestrial television system, according to ARIB STD-B32, part 2, Subclause 5.2.4:

5.2.4 Operational provisions regarding downmixing when multichannel stereo service is provided

This section defines the conditions and lists considerations in relation to compatibility with 2-channel stereo-capable receiver when multichannel stereo service of 5.1-channel stereo or less is provided.

(1) Two-channel stereo simulcasting is not obligatory when multichannel stereo service of 5.1-channel stereo (3/2+LFE (3/2.1)) or less is provided. Basically, 2-channel stereo-capable receiver shall handle the service by downmixing.

(2) It shall be possible to transmit the downmix coefficient using PCE according to the AAC Standard when 5-channel stereo (3/2) and 5.1-channel stereo (3/2+LFE (3/2.1)) services are provided. For the detailed provisions regarding transmission of PCE, refer to the section 5.2.3 (3).

(3) It shall be possible to provide 2-channel stereo simulcasting service at the request of broadcasting stations. In this case, two streams shall be treated as different ADTSs, multiplexed, and stream-controlled by the systems layer.

(4) For more information on downmixing operations of a 2-channel stereo-capable receiver other than the above mentioned cases (2) and (3), refer to the ARIB STD-B21 section 6.2.1(7), "Downmixing function from multi-channel to 2-channel stereo".

6.8 Audio coding parameter restrictions for one-seg services

ABNT NBR 15602-2, Subclauses 9.2.1 and 9.2.2, refer to restrictions for the "one-seg services". There are no specific clauses in ARIB STD-B32, part 2, but rather in ARIB TR-B14, for distinguishing one-seg from full-seg coding parameter restrictions.

In the Brazilian digital terrestrial television system, according to ABNT NBR 15602-2, Subclauses 9.2.1 and 9.2.2:

9.2.1 Audio coding modes

The coding mode determines the number of available channels on the audio service. The audio coding modes for digital transmission shall be in accordance with the restrictions described on Table 6.

Table 4 - Restrictions on audio coding modes – one-seg service

Parameter	***Restriction***
Permitted audio modes	*Monaural (1/0), stereo (2/0)*

The audio decoder shall be capable of processing any of the recommended audio modes.

The channel configuration, according to the operation mode, and its transmission order within the payload shall be in accordance with Table 7.

Table 5 - Channel configuration and standard modes for MPEG-4 AAC

Mode	***Channel configuration***	***SE order of transmission[a]***	***Standard element for loudspeaker mapping[b]***
Monaural (1/0)	*1*	*<SCE1><TERM>*	*SCE1 = C*
Stereo (2/0)	*2*	*<CPE1><TERM>*	*CPE1 = L and R*
[a] Abbreviations related to the syntactic element (SE): SCE – single channel element, CPE – channel pair element, LFE – LFE channel element, TERM – terminator. [b] Abbreviations related to loudspeaker arrangement: L – front-left loudspeaker / R – front-right loudspeaker / C – front-central loudspeaker.			

9.2.1 Main parameters

The main audio coding parameters for portable devices shall be as described in Table 8.

Table 6 - Main audio coding parameters for one-seg services

Parameter	***Restriction***
Permitted transport mechanisms	*LATM/LOAS, according to ISO/IEC 14496-3*
Permitted profiles and levels	*High-efficiency (HE): level 2 (HE-AAC v2@L2)*
Maximum number of encoded channels	*2 channels per bitstream (stereo or 2 monaural channels).*
Maximum bit rate	*According to ISO/IEC 14496-3*

The MPEG-4 HE-AAC version 2 shall be used in the transmission for portable devices and is mandatory for fixed and mobile devices if they receive the one-seg service.

The signals may be decoded with any rate and any sample rate supported by the profile and level of Table 8.

When using the PS extension, the audio decoder shall be capable of processing the syntactic element sbr_extension(), whose bs_extension_id is equal to EXTENSION_ID_PS, according to ISO/IEC 14496-3 (PS implicit signaling).