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ATSC Digital Television Standard: Part 6 – Enhanced AC-3 Audio System Characteristics

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The revision history of this document is given below.

A/53 Part 6 Revision History

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1. SCOPE

This Part describes the robust mode audio system characteristics and normative specifications of the Digital Television Standard. Audio encoded per this Part may be transmitted over a TS-E (see A/53-3 [5]).

2. REFERENCES

All standards are subject to revision and amendment, and parties to agreement based on this standard are encouraged to investigate the possibility of applying the most recent editions of the documents listed below.

2.1 Normative References

The following documents contain provisions which in whole or part, through reference in this text, constitute provisions of this standard. At the time of publication, the editions indicated were valid.

[1] ATSC: "Digital Audio Compression (AC-3, E-AC-3) Standard," Doc. A/52B, Advanced Television Systems Committee, Washington, D.C., 14 June 2005.

[2] AES: "AES Recommended Practice for digital audio engineering—Serial transmission format for two-channel linearly represented digital audio data," Doc. AES3-2003, Audio Engineering Society, New York, N.Y., 2003. (This document is a revision of AES3-1992.)

[3] ANSI: "Specification for Sound Level Meters," Doc. ANSI S1.4-1983 (R 2001) with Amd.S1.4A-1995, American National Standards Institute, Washington, D.C., 2001.

[4] ITU-R, Recommendation BS.1770, "Algorithms to measure audio programme loudness and true-peak audio level", International Telecommunications Union, Geneva.

2.2 Informative References

[5] ATSC: "ATSC Digital Television Standard, Part 3 – Service Multiplex and Transport Subsystem Characteristics," Doc. A/53 Part 3:2009, Advanced Television Systems Committee, Washington, D.C.,7 August 2009.

[6] ATSC: "ATSC Digital Television Standard, Part 5 – AC-3 Audio System Characteristics," Doc. A/53 Part 5:2010, Advanced Television Systems Committee, Washington, D.C., 6 July 2010.

[7] ATSC: "Digital Television Standard, Part 1 – Digital Television System," Doc. A/53 Part 1:2009, Advanced Television Systems Committee, Washington, D.C., 7 August 2009.

[8] ATSC: "Recommended Practice - Techniques for Establishing and Maintaining Audio Loudness for Digital Television," Doc. A/85, Advanced Television Systems Committee, Washington, D.C. , 4 November 2009.

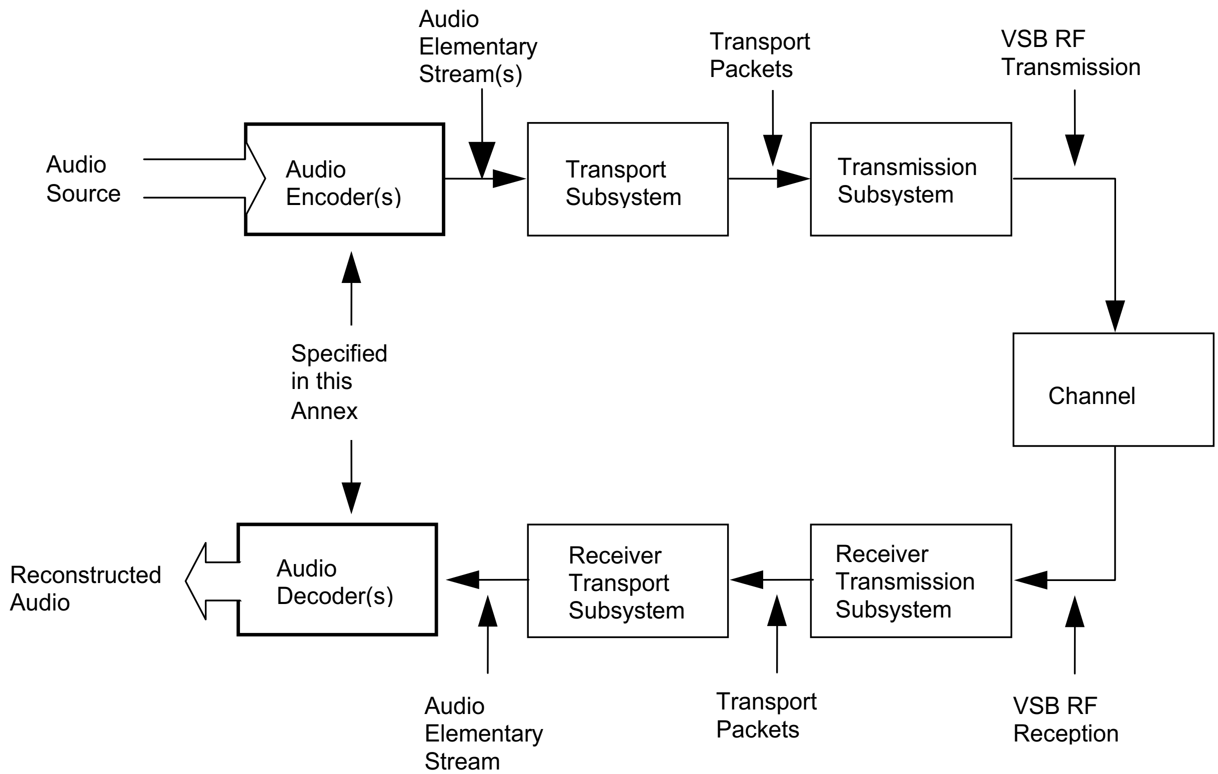

Figure 4.1 Audio subsystem in the digital television system.

3. COMPLIANCE NOTATION

As used in this document, “shall” or “will”, denotes a mandatory provision of the standard. “Should” denotes a provision that is recommended but not mandatory. “May” denotes a feature whose presence does not preclude compliance, and that may or may not be present at the option of the implementer.

3.1 Treatment of Syntactic Elements

This document contains symbolic references to syntactic elements used in the audio, video, and transport coding subsystems. These references are typographically distinguished by the use of a different font (e.g., restricted), may contain the underscore character (e.g., sequence_end_code) and may consist of character strings that are not English words (e.g., dynrng).

3.2 Symbols, Abbreviations, and Mathematical Operators

The symbols, abbreviations, and mathematical operators used herein are as found in Section 3.4 of ATSC A/53 Part 1 [7].

4. SYSTEM OVERVIEW

As illustrated in Figure 4.1, the audio subsystem comprises the audio encoding/decoding function and resides between the audio inputs/outputs and the transport subsystem. The audio encoder(s) is (are) responsible for generating the audio elementary stream(s) which are encoded representations of the baseband audio input signals. At the receiver, the audio subsystem is responsible for decoding the audio elementary stream(s) back into baseband audio.

5. SPECIFICATION

This Section forms the normative specification for the robust mode audio system that may be transmitted as part of TS-E (see A/53 Part 3 [5]). The robust mode audio compression system conforms to Annex E of the A/52 [1] Digital Audio Compression (AC-3) Standard, subject to the constraints outlined in this Section.

5.1 Constraints With Respect to ATSC Standard A/52 Annex E

The robust mode digital television audio coding system shall use the Enhanced AC-3 Digital Audio Compression Standard specified in Annex E of ATSC Doc. A/52 [1], and as constrained by this Part. Audio bit streams encoded per that specification may be included the TS-E that is delivered by E-VSB. Constraints on the robust mode audio system shall be as shown in Table 5.1, which shows permitted values of certain syntactical elements. These constraints are further described in Sections 5.2 – 5.4, and Section 6.

Table 5.1 Audio Constraints

AC-3 Syntactical Element	Comment	Allowed value
fscod	Indicates sampling rate	‘00’ (indicates 48 kHz)
frmsize	Indicates the size of the audio frame	≤ ’011 1000 0000’ (indicates a frame size ≤ 448 kb/s for a six block frame)
bstyp	Indicates an independent stream (no sub-streams)	‘00’
acmod	Indicates number of channels, prohibits 1+1 mode	≥ ‘001’
bsmod	Restricts audio service types to CM, VI, HI, C	0, 2, 3, or 5

5.2 Sampling Frequency

The system conveys digital audio sampled at a frequency of 48 kHz that shall be locked to the 27 MHz MPEG-2 system clock. The 48 kHz audio sampling clock is defined as:

$$48 \text{ kHz audio sample rate} = (2 \div 1125) \times (27 \text{ MHz MPEG-2 system clock})$$

If analog signal inputs are employed[1], the A/D converters shall sample at 48 kHz locked to the 27 MHz clock. If digital inputs are employed, the input sampling rate shall be 48 kHz locked to the system clock, or the audio encoder shall contain sampling rate converters which convert the sampling rate to 48 kHz locked to the system clock.

5.3 Frame Size

The audio frame size shall be less than or equal to 1792 bytes. This implies a bit-rate limitation of 448 kb/s for AC-3 frames of 1536 samples (32 msec at 48 kHz).

1. Either via AES3 [2] signals or embedded in the corresponding video.

5.4 Audio Coding Modes

Audio services shall be encoded using any of the audio coding modes specified in A/52, with the exception of the 1+1 mode. The value of acmod in the AC-3 bit stream shall have a value in the range of 1–7, with the value 0 prohibited.

5.5 Dialogue Level

The value of the dialnorm parameter in the AC-3 elementary bit stream shall indicate the loudness[2] of the encoded audio content (typically of the average spoken dialogue) using LKFS units. LKFS and its loudness measurement algorithm are specified in ITU-R Recommendation BS.1770, Annex 1 [4][3]. (Receivers use the value of dialnorm to adjust the reproduced audio level to normalize the loudness.) In order to enable clean switching (i.e., without level shifts) between main and fallback audio services (that might have a different number of audio channels), linked audio services shall have values of dialnorm that result in matched dialogue levels when decoded by compliant decoders.

5.6 Dynamic Range Compression - Artistic

Each encoded audio block may contain a dynamic range control word (dynrng) that is used by decoders (by default) to alter the level of the reproduced audio. The control words allow the decoded signal level to be increased or decreased by up to 24 dB. In general, elementary streams may have dynamic range control words inserted or modified without affecting the encoded audio. When it is necessary to alter the dynamic range of audio programs that are broadcast, the dynamic range control word should be used. In order to enable clean switching between main and fallback audio services (that might have a different number of audio channels), linked audio services shall have values of dynrng that result in matched audio levels when decoded by compliant decoders.

5.7 Dynamic Range Compression - Heavy

Each encoded audio frame may contain a dynamic range control word (compr) that may be optionally used by decoders to render the audio with a very narrow dynamic range. The control words allow the decoded signal level to be increased or decreased by up to 48 dB. In order to enable clean switching between main and fallback audio services (that might have a different number of audio channels), linked audio services shall have values of compr that result in matched audio levels when decoded by compliant decoders.

6. MAIN AND ASSOCIATED SERVICES

An AC-3 elementary stream contains the encoded representation of a single audio service. Multiple audio services are provided by multiple elementary streams. Each elementary stream is conveyed by the transport multiplex with a unique PID. There are a number of audio service types that may (individually) be coded into each elementary stream. Each AC-3 elementary stream is tagged as to its service type using the bsmod bit field. There is a *complete main service* and there are three types of *associated services*.

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2. Methods to measure loudness and different modes for dialnorm management are explained in the ATSC Recommended Practice A/85 "Techniques for Establishing and Maintaining Audio Loudness for Digital Television" [8]; see particularly Section 5.2.
3. A dialnorm value based on an "A" weighted integrated measurement (LAeq) (ANSI S1.4) [3] prior to the publication of this document need not be re-measured with BS.1770.

Associated services delivered in a TS-E shall contain complete program mixes containing all audio program elements (dialog, music, effects, etc.) that are intended to be presented to a listener. This is indicated by the full_svc bit in the AC-3_audio_stream_descriptor() being set to a value of '1' (see A/53-3 [5] and A/52, Annex A [1]).

This section specifies the meaning and use of each type of service.

6.1 Summary of Service Types

The audio service types shall be as listed in Table 6.1.

Table 6.1 Audio Service Types

bsmod	Type of Service
'000' (0)	Main audio service: complete main (CM)
'010' (2)	Associated service: visually impaired (VI)
'011' (3)	Associated service: hearing impaired (HI)
'101' (5)	Associated service: commentary (C)

6.2 Complete Main Audio Service (CM)

The CM type of main audio service shall contain a complete audio program (complete with dialogue, music, and effects). This is the type of audio service normally provided. The CM service may contain from 1 to 5.1 audio channels. Audio in multiple languages may be provided by supplying multiple CM services, each in a different language.

6.3 Visually Impaired (VI)

The VI associated service a complete program mix containing music, effects, dialogue, and additionally a narration that describes the picture content. The VI service may be coded using any number of channels (up to 5.1).

6.4 Hearing Impaired (HI)

The HI service is a complete program mix containing music, effects, and dialogue with enhanced intelligibility. The HI service may be coded using any number of channels (up to 5.1).

6.5 Commentary (C)

The commentary associated service is a complete program mix containing music, effects, dialogue, and additionally some special commentary. This service may be provided using any number of channels (up to 5.1).

7. AUDIO ENCODER INTERFACES

See A/53 Part 5 [6], Section 7.

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