STANDARDS AND INFORMATION DOCUMENTS

AES3-1-2009
(Rev. AES3-2003)

STANDARDS

AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly represented digital audio data Part 1: Audio Content

(Multi-part revision of AES3-2003, incorporating Amendments 5 & 6)

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AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly-represented digital audio data — Part 1: Audio Content

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Abstract

AES3 provides for the serial digital transmission of two channels of periodically sampled and uniformly quantized audio signals on various media.

This Part specifies the semantics of the audio data, including the "validity" flag. It also specifies the sampling frequency by reference to AES5, *AES recommended practice for professional digital audio — Preferred sampling frequencies for applications employing pulse-code modulation*.

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Foreword

This foreword is not part of the *AES3-1-2009, AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly represented digital audio data, Part 1: Audio Content*.

AES3 has been under constant review since the standard was first issued in 1985, and the present edition reflects the collective experience and opinions of many users, manufacturers, and organizations familiar with equipment or systems employing AES3.

This document was adapted by R. Caine from the 2003 edition as amended by Amendments 5 and 6, and its technical content is believed to be identical to the relevant parts of that version. Other members of the writing group that developed this document in draft included: C. Travis, C. Langen, H. Jahne, J. Grant, J. Woodgate, M. Natter, M. Poimboeuf, R. Cabot, S. Heinzmann, M. Werwein, and M. Yonge.

J Grant, chair
SC-02-02 Working Group on Digital Input-Output Interfacing
May 2009

Note on normative language

In AES standards documents, sentences containing the word “shall” are requirements for compliance with the document. Sentences containing the verb “should” are strong suggestions (recommendations). Sentences giving permission use the verb “may”. Sentences expressing a possibility use the verb “can”.

AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly-represented digital audio data — Part 1: Audio Content

1 Scope

These four documents specify an interface for the serial digital transmission of two channels of periodically sampled and linearly represented digital audio data from one transmitter to one receiver. This Part 1 defines the format for coding audio used for the audio content.

It is expected that the audio data will have been sampled at any of the sampling frequencies recognized by the AES5 *Recommended Practice for Professional Digital Audio Applications Employing Pulse-Code Modulation — Preferred Sampling Frequencies*. Note that conformance with this interface specification does not require equipment to utilise these rates. The capability of the interface to indicate other sample rates does not imply that it is recommended that equipment support these rates. To eliminate doubt, equipment specifications should define supported sampling frequencies.

2 Normative references

The following standards contain provisions which, through reference in this text, constitute provisions of this document. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this document are encouraged to investigate the possibility of applying the most recent editions of the indicated standards.

AES5-2008, *AES Recommended Practice for Professional Digital Audio Applications Employing Pulse Code Modulation—Preferred Sampling Frequencies*, Audio Engineering Society, New York, NY, USA.

ITU-R BS.450-3, *Transmission standards for FM sound broadcasting at VHF*, International Telecommunication Union, Geneva, Switzerland (was previously CCIR Rec 450-1).

ITU-T J.17, *Pre-emphasis used on sound-program circuits*, International Telecommunication Union, Geneva, Switzerland.

3 Definitions and abbreviations

3.1
sampling frequency
the frequency of the samples representing an audio signal. See 5.2

3.2
audio sample word
a series of binary digits representing the amplitude of an audio sample, also known as a PCM sample. See 4.1

3.3
auxiliary sample bits
the four least significant bits (LSBs) of those allocated to audio which can be assigned as auxiliary sample bits and used for auxiliary information when the number of audio sample bits is less than or equal to 20

3.4
validity bit
bit indicating whether the audio sample bits in the same subframe are suitable for direct conversion to an analog audio signal

3.5
MSB
in the context of this standard: the Most Significant Bit of an audio sample word, being the sign bit in the case of two's complement code

3.6
LSB
in the context of this standard: the Least Significant Bit of an audio sample word

3.7
Subframe
the smallest structural element in an AES3 transport, carrying one PCM sample and ancillary information; see Part 3

4 Audio Content

4.1 Audio content coding

The audio content shall be coded as linear PCM using 2's complement code.

4.2 PCM polarity

Positive analogue voltages at the ADC input shall be represented by positive binary numbers.

4.3 Coding precision options

The accuracy of the coding shall be between 16 and 24 bits, in two ranges for the purpose of indicating which length is in use in channel status data, 16 to 20 bits and 20 to 24 bits (see Part 2).

4.4 Intermediate coding precision

The interface permits maximum word lengths of either 20 or 24 bits. A source which provides fewer bits than this shall be justified to the MSB of the available word length and the unused LSBs shall be set to logic 0.

NOTE If a low-resolution signal were not so justified, then sign extension would be needed.

4.5 Non-audio content

The interface may alternatively carry data or audio which is compressed or in a different format in place of linear PCM audio, in either channel B or both channels. In such cases the validity bit shall be set independently in each channel and channel status encoded to indicate this. See Part 2.

NOTE Such use is not standardized here: provision is only made to protect standard equipment from such use.

4.6 DC content

The coded audio shall contain as little equivalent DC offset as possible, and in any case less than the analogue equivalent noise level.

5 Sampling Frequency

5.1 Channel interdependency

The sampling frequency shall be the same in both channels.

5.2 Choice of sampling frequency

The sampling frequency shall be in accordance with AES5-2003.

6 Validity bit

6.1 Channel validity usage

The validity bit shall be set to logic 0 if the associated audio sample word is suitable for direct conversion to an analog audio signal, and shall be set to logic 1 if it is not suitable. Where channel status indicates (in Byte 0 bit 1 - see Part 2) that the audio sample word is not in linear PCM form the validity bit shall be set to logic 1 in every subframe.

There is no default state for the validity bit.

6.2 Independent channel validity

Validity shall be set or reset for each and every sample independently in each channel.

7 Pre-emphasis

7.1 Pre-emphasis characteristic

The audio signal may be coded with a flat frequency response, or with 50 µs pre-emphasis as per ITU-R BS.450-3 or with J.17 pre-emphasis as per ITU-T J.17.

7.2 Pre-emphasis indication

The use of pre-emphasis shall be indicated in channel status as defined in Part 2. Where no pre-emphasis is used, this may be indicated.

NOTE Positive indication is strongly preferred. The default value will normally be taken to indicate no pre-emphasis, but this condition is undefined. See AES-2id-2006 for clarification.

Annex A (informative) Informative references

AES-2id-2006, *AES information document for digital audio engineering - Guidelines for the use of the AES3 interface*, Audio Engineering Society, New York, NY, USA.

IEC 60958-1, *Digital audio interface - Part 1: General*, International Electrotechnical Commission, Geneva, Switzerland.

IEC 60958-4, *Digital audio interface - Part 4: Professional applications*, International Electrotechnical Commission, Geneva, Switzerland.

ITU-R BS.647, *A digital audio interface for broadcasting studios*, International Telecommunication Union, Geneva, Switzerland.

EBU T3250 Ed.3, *Specification of the digital audio interface (the AES/ EBU interface), Third Edition*, European Broadcasting Union, Geneva Switzerland.

SMPTE 337-2008, *Format for Non-PCM Audio and Data in an AES3 Serial Digital Audio Interface*. Society of Motion Picture and Television Engineers, White Plains, NY 10601, US.

SMPTE 338-2008, *Format for Non-PCM Audio and Data in AES3 - Data Types*. Society of Motion Picture and Television Engineers, White Plains, NY 10601, US.

SMPTE 339-2008, *Format for Non-PCM Audio and Data in AES3 - Generic Data* Types. Society of Motion Picture and Television Engineers, White Plains, NY 10601, US.

SMPTE 340-2008, *Format for Non-PCM Audio and Data in AES3 - ATSC A/52B Digital Audio Compression Standard for AC-3 and Enhanced AC-3 Data Types*. Society of Motion Picture and Television Engineers, White Plains, NY 10601, US.