AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly-represented digital audio data — Part 2: Metadata and Subcode

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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 information transmitted with the audio data: principally the "channel status" but also user data and the use of the auxiliary bits to carry a co-ordination signal.

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Contents

1 Scope	4
2 Normative references	4
3 Definitions and abbreviations	5
4 User data format	5
5 Channel status format	
5.1 Channel status bit	5
5.2 Channel status block	
5.3 Implementation	
5.3.1 Implementation levels	
5.3.1.2 Standard Level	
5.3.1.3 Enhanced Level	6
5.3.2 Transmitter requirement	e
5.3.3 Receiver requirement	
5.4 Documentation	
5.5 Channel status content	
5.5.0 Byte 0: Basic audio parameters	
5.5.1 Byte 1: Channel modes, user bits management	
5.5.2 Byte 2: Auxiliary bits, word length and alignment level	
5.5.3 Byte 3: Multichannel modes	
5.5.4 Byte 4: DARS, hidden information, multiple-rate sampling frequencies	
5.5.5 Byte 5: Reserved	
5.5.6 Bytes 6 to 9: Alphanumeric channel origin	
5.5.7 Bytes 10 to 13: Alphanumeric channel destination	
5.5.8 Bytes 14 to 17: Local sample address code	
5.5.9 Bytes 18 to 21: Time-of-day sample address code	
5.5.10 Byte 22: Reserved	
5.5.11 Byte 23: Channel status data CRCC	
5.6 Channel Status when non-PCM audio is flagged	
6. Auxiliary bits	14
6.1 Availability of auxiliary bits	
6.2 Use of auxiliary bits	
Annex A (informative) Informative references	15
Annex B (informative) Provision of additional, voice-quality channels	16
Annex C (informative) Generation of CRCC (byte 23) for channel status	17

Foreword

This foreword is not part of the AES3-2-2009, AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly represented digital audio data, Part 2: Metadata and Subcode

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".

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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 2 defines the format for coding metadata, or subcode, relating to the audio content and carried with it.

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.

AES18, AES recommended practice for digital audio engineering—Format for the user data channel of the AES digital audio interface, Audio Engineering Society, New York, NY, USA.

AES52-2006: AES standard for digital audio engineering — Insertion of unique identifiers into the AES3 transport stream, Audio Engineering Society, New York, NY, USA.

IEC 60958-3 *Digital audio interface - Part 3: Consumer applications*, International Electrotechnical Commission, Geneva, Switzerland.

ISO 646, *Information processing—ISO 7-bit coded character set for information interchange*, International Organization for Standardization, Geneva, Switzerland.

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

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

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