STANDARDS AND INFORMATION DOCUMENTS

AES3-3-2009
(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 3: Transport

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

Users of this standard are encouraged to determine if they are using the latest printing incorporating all current amendments and editorial corrections. Information on the latest status, edition, and printing of a standard can be found at: http://www.aes.org/standards

AUDIO ENGINEERING SOCIETY, INC.
60 East 42nd Street, New York, New York 10165, US.

The **AES Standards Committee** is the organization responsible for the standards programme of the Audio Engineering Society. It publishes a number of technical standards, information documents and technical reports.

Working groups and task groups with a fully international membership are engaged in writing standards covering fields that include topics of specific relevance to professional audio.

Membership of any AES standards working group is open to all individuals who are materially and directly affected by the documents that may be issued under the scope of that working group.

Complete information, including scopes of working groups and project status is available at http://www.aes.org/standards. Enquiries may be addressed to standards@aes.org

AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly-represented digital audio data — Part 3: Transport

Published by
Audio Engineering Society, Inc.

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 framing and channel coding for transmission on a unidirectional point-to-point physical link. The specified format minimizes the direct-current (DC) component on the transmission line, facilitates clock recovery from the data stream, and makes the interface insensitive to the polarity of connections.

2010-01-11 printing

Contents

Foreword

This foreword is not part of the *AES3-3-2009, AES standard for digital audio — Digital input-output interfacing — Serial transmission format for two-channel linearly represented digital audio data, Part 3: Transport*.

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 3: Transport

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 3 defines the format for transport of an AES3 digital audio interface.

Specific synchronization issues are covered in AES11 *AES recommended practice for digital audio engineering -- Synchronization of digital audio equipment in studio operations*. An engineering guideline document to accompany this interface specification has been published as AES-2id *AES information document for digital audio engineering - Guidelines for the use of the AES3 interface*.

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.

None.

3 Definitions and abbreviations

3.1
Biphase-mark
channel-coding (or line-coding) technique which minimizes DC content and maximizes clock-recovery energy relative to the original binary bitstream

3.2
even parity bit
a bit whose value is chosen such that the total number of ones in the field which includes it is even

3.3
preambles
specific unique patterns used for synchronization, compatible with but not part of the biphase mark code See 7

3.4
subframe
smallest structural element in an AES3 transport, used to carry the information described in 4

3.5
frame
sequence of two successive and associated subframes, see 5

3.6
block
group of 192 consecutive frames with a defined start point, see 8

> NOTE The start of a block is designated by a special subframe preamble. See 7 and 8.

3.7
channel coding/line coding
coding describing the method by which the binary digits are represented for transmission through the interface, see biphase mark above

3.8
unit interval
(UI)
shortest nominal time interval in the coding scheme

> NOTE There are 128 UI in a sample frame.

4 Subframe

4.1 Subframe time slots

Each subframe shall be divided into 32 time slots, numbered from 0 to 31. See figure 1. Time slot 0 is transmitted first. Each time slot shall consist of 2 UI

4.2 Preambles

Time slots 0 to 3, the preambles, shall comprise one of the three permitted preambles designated X, Y and Z. See 7 and 8 and figure 4.

4.3 Audio data content

Time slots 4 to 27 shall contain the audio sample word, or some other data such as compressed audio, or some combination of audio and other data (see Part 1, and Part 2 clause 6).

4.4 Sample word orientation

The sample is carried LSB first.

4.5 MSB position

The most significant bit (MSB), the sign bit, shall be carried by time slot 27. If the source provides fewer bits than the interface allows, either 20 or 24, the unused LSBs shall be set to logic 0 and the active bits shall be justified to the MSB end of the available word length.

When a 24-bit coding range is used, the LSB shall be in time slot 4.

When a 20-bit coding range is sufficient, the LSB shall be in time slot 8. Time slots 4 to 7 may be used for other applications. Under these circumstances, the bits in time slots 4 to 7 are designated auxiliary sample bits. (See Part 2).

4.6 Validity bit

Time slot 28 shall carry the validity bit associated with the audio sample word transmitted in the same subframe. (See Part 1).

4.7 User data bit

Time slot 29 shall carry 1 bit of the user data channel associated with the audio channel transmitted in the same subframe. (See Part 2).

4.8 Channel status bit

Time slot 30 the channel status bit, shall carry 1 bit of the channel status information associated with the audio channel transmitted in the same subframe. (See Part 2).

4.9 Parity bit

Time slot 31 shall carry an even parity bit such that time slots 4 to 31 inclusive will carry an even number of ones and an even number of zeros.

0 3	4		27	28			31
Preamble	LSB	24-bit audio sample word	MSB	V	U	C	P

(a)

V	Validity bit
U	User data bit
C	Channel status bit
P	Parity bit
AUX	Auxiliary sample bits

0 3	4 7	8		27	28			31
Preamble	AUX	LSB	20-bit audio sample word	MSB	V	U	C	P

(b)

Figure 1 — Subframe format

5 Frame

A frame shall comprise two subframes (see figure 2). Except where otherwise specified the rate of transmission of frames corresponds exactly to the source sampling frequency, and in the case of stereophonic signals the two subframes in a frame shall carry samples taken at the same instant.

Examples include:

Two-channel mode:	Channel 1 is in subframe 1, and channel 2 is in subframe 2.
Stereophonic mode	The interface is used to transmit stereophonic audio in which the two channels are presumed to have been simultaneously sampled. The left, or `A`, channel is in subframe 1, and the right, or `B`, channel is in subframe 2.
Single-channel mode (monophonic)	The transmitted bit rate remains at the normal two-channel rate and the audio sample word is placed in subframe 1. Time slots 4 to 31 of subframe 2 either carry the bits identical to subframe 1 or are set to logic 0. A receiver normally defaults to channel 1 unless manual override is provided.
Primary-secondary mode	In some applications requiring two channels where one of the channels is the main or primary channel while the other is a secondary channel, the primary channel is in subframe 1, and the secondary channel is in subframe 2.
single-channel double sampling-frequency mode	The frame rate is half the audio sampling frequency. Channel 2 in each frame carries the sample immediately following the sample in channel 1 of the same frame.

NOTE The modes of transmission are signaled by setting bits 0 to 3 of byte 1 of channel status. (See Part 2)

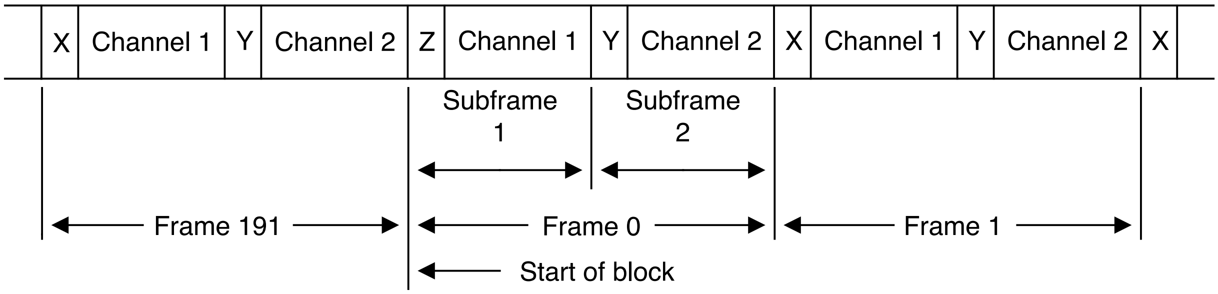

Figure 2 — Frame format

6 Channel coding (line coding)

Time slots 4 to 31 shall be encoded in biphase-mark form.

Each bit to be transmitted shall be represented by a symbol comprising two consecutive binary states. The first state of a symbol shall always be different from the second state of the previous symbol. The second state of the symbol shall be identical to the first if the bit to be transmitted is logic 0 and it shall be different if the bit is logic 1. See figure 3. Each state shall occupy one unit interval (UI).

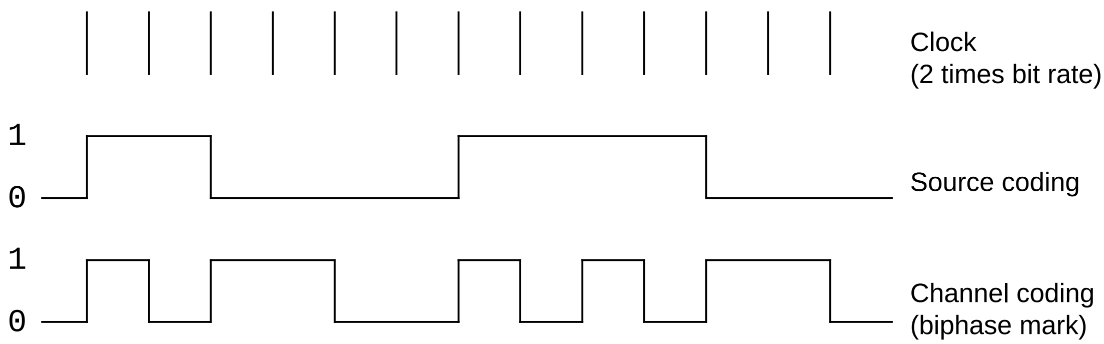

Figure 3 — Channel coding

NOTE Biphase-mark coding minimizes the direct-current (DC) component on the transmission line, facilitates clock recovery from the data stream, and makes the interface insensitive to the polarity of connections.

7. Preambles

7.1 Preamble time slots

Time slots 0 to 3 shall be encoded as preambles

7.2 First subframe preamble

The first subframe in every frame shall start with a preamble type `X`, except for that at the start of a 192-frame block, when it shall carry a preamble type `Z`. This defines the block structure used to organize the channel status information.

7.3 Second subframe preamble

The second subframe shall always start with a preamble type `Y`.

NOTE Preambles are specific patterns providing synchronization and identification of the subframes and blocks. To achieve synchronization within one sampling period and to make this process completely reliable, these patterns violate the biphase-mark code rules, thereby avoiding the possibility of data imitating the preambles. The preambles have even parity as an explicit property.

7.4 Preamble codes

The form of the three types of preamble shall be as shown in the table, represented by eight successive states, occupying four timeslots. Figure 4 represents preamble X.

Table x - Preamble codes

	Channel Coding		
Preceding state	0	1	
Preamble			
X	11100010	00011101	Subframe 1
Y	11100100	00011011	Subframe 2
Z	11101000	00010111	Subframe 1 and block start

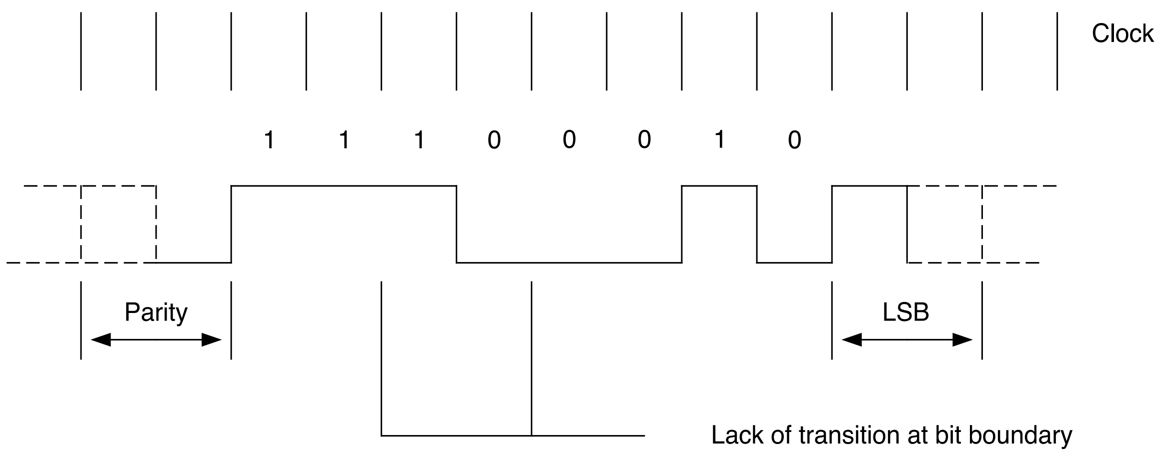

Figure 4 — Preamble X (11100010)

NOTE 1 The first state of the preamble is always different from the second state of the previous symbol, representing the parity bit.

NOTE 2 these preambles are DC-free and provide clock recovery as with biphase code. They differ in at least two states from any valid biphase sequence.

NOTE 3 The state is always inverted once per timeslot plus once per data "one" bit. There are an even number of timeslots in a subframe, and, owing to the even-parity bit in time slot 31 (see 4.9), an even number of "one" bits, so the total number of inversions in any subframe is even. Hence, all preambles will start with the same state. Thus only one of these sets of preambles will, in practice, be transmitted through the interface. However, it is necessary for either set to be decodable in order to maintain immunity to polarity change.

8. Block

A sequence of 192 frames shall be designated a Block. The first frame in this sequence shall contain a preamble type Z in place of the type X preamble. The subframes comprising this frame shall contain the first bit of the first byte of the channel status code described in Part 2.

Annex A (informative) Informative references

AES11, *AES recommended practice for digital audio engineering—Synchronization of digital audio equipment in studio operations*, Audio Engineering Society, New York, NY, USA.

AES-2id, *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.