STANDARDS AND INFORMATION DOCUMENTS

AES3-2-2009
(Rev. AES3-2003)

AUDIO
ES
STANDARDS

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

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

AUDIO ENGINEERING SOCIETY, INC.
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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 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

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".

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

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..

3 Definitions and abbreviations

3.1
channel status
bits carrying, in a fixed format derived from the block (see Part 3) information associated with each audio channel which is decodable by any interface user

3.2
user data
channel provided to carry any other information

3.3
metadata
Information relating to the audio content in the same channel

3.4
subframe
smallest structural element in an AES3 transport, carrying one PCM sample and ancillary information; see Part 3

3.5
frame
sequence of two successive and associated subframes

4 User data format

One bit of User data may be carried in each subframe. Different user data may be carried in each channel and may be related to the associated audio or not. Its capacity in kbit/s is therefore equal to the sampling frequency in use, in kilosamples/s, for each channel.

User data bits may be used in any way desired by the user.

Known possible formats for the user data channel are indicated by the channel status byte 1, bits 4 to 7.
Other possible formats may be used and may or may not be standardized in future.

The default value of the user data bit is logic 0.

5 Channel status format

5.1 Channel status bit

One bit of Channel Status data shall be carried in each sub-frame. Different channel status data may be carried in each channel. Its capacity in kbit/s is therefore equal to the sampling frequency in use, in kilosamples/s.

> NOTE The channel status for each audio signal carries information associated with that audio signal, and thus it is possible for different channel status data to be carried in the two subframes of the digital audio signal. Examples of information to be carried in the channel status are: length of audio sample words, number of audio channels, sampling frequency, sample address code, alphanumeric source and destination codes, and emphasis.

5.2 Channel status block

Channel status information shall be organized in 192-bit blocks, subdivided into 8-bit bytes numbered from 0 to 23. The transmission format shall mark every 192nd frame to show that it carries the first bit of a block. Within each byte, the bits are numbered from 0 to 7, 0 being the first bit transmitted, so bit 0 of byte 0 is the first bit in the block. Where a byte holds a numerical value, bit 0 is the least significant bit.

> NOTE In Part 3, the frame that begins with preamble Z contains the first bit of a block in both channels. In other transports (eg AES10 and AES47) a 'block start' flag is used to mark the first subframe in a block, and may be applied to each channel independently.

5.3 Implementation

5.3.1 Implementation levels

5.3.1.1 General

The following two implementations are defined: standard and enhanced. These terms are used to communicate in a simple manner the level of implementation of the interface transmitter involving the many features of channel status. Irrespective of the level of implementation, all reserved states of bits defined in 5.5 shall remain unchanged.

5.3.1.2 Standard Level

The standard implementation provides a fundamental level of implementation which should prove sufficient for general applications in professional audio or broadcasting. In the standard implementation, transmitters shall correctly encode and transmit all channel status bits in byte 0, byte 1, byte 2, and byte 23 (CRCC) in the manner specified in this document.

5.3.1.3 Enhanced Level

In addition to conforming to the requirements described in 5.3.1.2 for the standard implementation, the enhanced implementation shall provide further capabilities.

5.3.2 Transmitter requirement

Transmitters shall encode channel status to follow all the formatting and channel coding rules to one of the two specified implementation levels. All transmitters shall correctly encode and transmit channel status with the correct juxtaposition with respect to the Z preamble or block start (see Part 3)

5.3.3 Receiver requirement

Receivers shall decode channel status as required by their application. Receivers shall interpret CRCC errors as needing to reject the channel status block with the error. Receivers shall not interpret any errors in a channel status block such as CRCC or block length errors as a reason to mute or alter the audio content.

> NOTE The purpose of the CRCC in byte 23 is to indicate corruption of the channel status block due to switching or editing effects (for example). Due consideration should be given to the implications of any action on downstream equipment and the associated system in general.

5.4 Documentation

Documentation shall be provided describing the channel status features supported by interface transmitters and receivers.

> NOTE To promote compatible operation between items of equipment built to this specification it is necessary to establish which information bits and operational bits need to be encoded and sent by a transmitter and decoded by an interface receiver.

5.5 Channel status content

The specific organization follows. Multiple-bit quantities are shown in the tables with the most significant bit to the left; note that the order in which the bits are transmitted is therefore from right to left.

Byte	Bit 7	6	5	4	3	2	1	0
0	e		d	c			b	a
1	g				f			
2	j		i			h		
3	n=0	k						
3	n=1	m			l			
4	r	q				p	o	
5	s							
6–9	Alphanumeric channel origin data							
10–13	Alphanumeric channel destination data							
14–17	Local sample address code (32-bit binary)							
18–21	Time-of-day sample address code (32-bit binary)							
22	Reliability flags							
23	Cyclic redundancy check character							

Key:

a	use of channel status block	j	indication of alignment level
b	linear PCM identification	k	channel number
c	audio signal pre-emphasis	l	channel number
d	lock indication	m	multichannel mode number
e	sampling frequency	n	multichanel mode
f	channel mode	o	digital audio reference signal
g	user bits management	p	reserved but undefined
h	use of auxiliary sample bits	q	sampling frequency
i	source word length	r	sampling frequency scaling flag
		s	reserved but undefined

Figure 1 - Channel status data format

5.5.0 Byte 0: Basic audio parameters

Bit	0	Use of channel status block
state	0	Consumer use of channel status block (see note 1).
	1	Professional use of channel status block.

Bit	1	Linear PCM identification
state	0	Audio sample word represents linear PCM samples.
	1	Audio sample word used for purposes other than linear PCM samples.

Bits	4 3 2	Audio signal emphasis
states	0 0 0	Emphasis not indicated. Receiver defaults to no emphasis with manual override enabled.
	0 0 1	No emphasis. Receiver manual override is disabled.
	0 1 1	50 µs + 15 µs emphasis, see ITU-R BS.450. Receiver manual override is disabled.
	1 1 1	ITU-T J.17 emphasis (with 6,5-dB insertion loss at 800 Hz). Receiver manual override is disabled
	All other states of bits 2 to 4 are reserved and are not to be used until further defined.	

Bit	5	Lock indication
state	0	Default. Lock condition not indicated.
	1	Source sampling frequency unlocked.

Bits	7 6	Sampling frequency
states	0 0	Sampling frequency not indicated. Receiver default to interface frame rate and manual override or auto set is enabled.
	1 0	48-kHz sampling frequency. Manual override or auto set is disabled.
	0 1	44,1-kHz sampling frequency. Manual override or auto set is disabled.
	1 1	32-kHz sampling frequency. Manual override or auto set is disabled.

NOTE 1 The significance of byte 0, bit 0 is such that a transmission from an interface conforming to IEC 60958-3 consumer use can be identified, and a receiver conforming only to IEC 60958-3 consumer use will correctly identify a transmission from a professional-use interface as defined in this standard. Connection of a professional-use transmitter with a consumer-use receiver or vice versa might result in unpredictable operation. Thus the following byte definitions only apply when bit 0 = logic 1 (professional use of the channel status block).

NOTE 2 The indication that the audio sample words are not in linear PCM form requires that the validity bit be set for that channel. See 5.6 and Part 1.

NOTE 3 The indication of sampling frequency, or the use of one of the sampling frequencies that can be indicated in this byte, is not a requirement for operation of the interface. The 00 state of bits 6 to 7 may be used if the transmitter does not support the indication of sampling frequency, the sampling frequency is unknown, or the sample frequency is not one of those that can be indicated in this byte. In the latter case for some sampling frequencies byte 4 may be used to indicate the correct value.

NOTE 4 When byte 1, bits 1 to 3 indicate single channel double sampling frequency mode then the sampling frequency of the audio signal is twice that indicated by bits 6 to 7 of byte 0.

5.5.1 Byte 1: Channel modes, user bits management

bits	3 2 1 0	Channel mode

states	0 0 0 0	Mode not indicated. Receiver default to two-channel mode. Manual override is enabled.
	1 0 0 0	Two-channel mode. Manual override is disabled.
	0 1 0 0	Single-channel mode (monophonic). Manual override is disabled.
	1 1 0 0	Primary-secondary mode, subframe 1 is primary. Manual override is disabled.
	0 0 1 0	Stereophonic mode, channel 1 is left channel. Manual override is disabled
	1 0 1 0	Reserved for user-defined applications.
	0 1 1 0	Reserved for user-defined applications.
	1 1 1 0	Single channel double sampling frequency mode. Sub-frames 1 and 2 carry successive samples of the same signal. The sampling frequency of the signal is double the frame rate, and is double the sampling frequency indicated in byte 0, but not double the rate indicated in byte 4, if that is used. Manual override is disabled. Vector to byte 3 for channel identification.
	0 0 0 1	Single channel double sampling frequency mode – stereo mode left. Sub-frames 1 and 2 carry successive samples of the same signal. The sampling frequency of the signal is double the frame rate, and is double the sampling frequency indicated in byte 0, but not double the rate indicated in byte 4, if that is used. Manual override is disabled.
	1 0 0 1	Single channel double sampling frequency mode – stereo mode right. Sub-frames 1 and 2 carry successive samples of the same signal. The sampling frequency of the signal is double the frame rate, and is double the sampling frequency indicated in byte 0, but not double the rate indicated in byte 4, if that is used. Manual override is disabled.
	1 1 1 1	Multichannel mode. Vector to byte 3 for channel identification.
	All other states of bits 0 to 3 are reserved and are not to be used until further defined.	

bits	**7 6 5 4**	**User bits management**
states	0 0 0 0	Default, no user information is indicated.
	1 0 0 0	192-bit block structure with user-defined content. Block start aligned with Channel Status block start.
	0 1 0 0	Reserved for the AES18 standard.
	1 1 0 0	User defined.
	0 0 1 0	User data conforms to the general user data format defined in IEC 60958-3.
	1 0 1 0	192-bit block structure as specified in AES52. Block start aligned with Channel Status block start.
	0 1 1 0	Reserved for IEC 62537
	All other states of bits 4 to 7 are reserved and are not to be used until further defined.	

5.5.2 Byte 2: Auxiliary bits, word length and alignment level

bits	2 1 0	Use of auxiliary bits
states	0 0 0	Maximum audio sample word length is 20 bits (default). Use of auxiliary bits not defined
	1 0 0	Maximum audio sample word length is 24 bits. Auxiliary bits are used for main audio sample data
	0 1 0	Maximum audio sample word length is 20 bits. Auxiliary bits in this channel are used to carry a single coordination signal. See note 1
	1 1 0	Reserved for user defined applications.
	All other states of bits 0 to 2 are reserved and are not to be used until further defined	

bits	5 4 3	Encoded audio sample word length of transmitted signal (See notes 2, 3, and 4)	
		Audio sample word length if maximum length is 24 bits as indicated by bits 0 to 2 above.	Audio sample word length if maximum length is 20 bits as indicated by bits 0 to 2 above.
states	0 0 0	Word length not indicated (default).	Word length not indicated (default).
	1 0 0	23 bits	19 bits
	0 1 0	22 bits	18 bits
	1 1 0	21 bits	17 bits
	0 0 1	20 bits	16 bits
	1 0 1	24 bits	20 bits
	All other states of bits 3 to 5 are reserved and are not to be used until further defined.		

bits	7 6	Indication of alignment level
states	0 0	Alignment level not indicated
	1 0	Alignment to SMPTE RP155, alignment level is 20 dB below maximum code.
	0 1	Alignment to EBU R68, alignment level is 18,06 dB below maximum code.
	1 1	Reserved for future use.

NOTE 1 The signal coding used for the coordination channel is described in annex B.

NOTE 2 The default state of bits 3 to 5 indicates that the number of active bits within the 20-bit or 24-bit coding range is not specified by the transmitter. The receiver should default to the maximum number of bits specified by the coding range and enable manual override or automatic set.

NOTE 3 The nondefault states of bits 3 to 5 indicate the number of bits within the 20-bit or 24-bit coding range which might be active. This is also an indirect expression of the number of LSBs that are certain to be inactive, which is equal to 20 or 24 minus the number corresponding to the bit state.

NOTE 4 Irrespective of the audio sample word length as indicated by any of the states of bits 3 to 5, the MSB is in time slot 27 of the transmitted subframe as specified in Part 3, 4.5.

5.5.3 Byte 3: Multichannel modes

bit	7	Multichannel mode
state	0	Undefined multichannel mode (default)
	1	Defined multichannel modes

The definition of the remaining bit states depends on the state of bit 7.

EITHER

bits	6 to 0	Channel number, when byte 3 bit 7 is 0
value	The channel number is the numeric value of the byte, plus one, with bit 0 as the least significant bit.	

OR,

bits	6 5 4	Multichannel mode, when byte 3 bit 7 is 1
states *note: LSB first*	0 0 0	Multichannel mode 0. The channel number is defined by bits 3 to 0 of this byte
	0 0 1	Multichannel mode 1. The channel number is defined by bits 3 to 0 of this byte
	0 1 0	Multichannel mode 2. The channel number is defined by bits 3 to 0 of this byte
	0 1 1	Multichannel mode 3. The channel number is defined by bits 3 to 0 of this byte
	1 1 1	User-defined multichannel mode. The channel number is defined by bits 3 to 0 of this byte
	All other states of bits 6 to 4 are reserved and are not to be used until further defined.	

bits	3 to 0	Channel number, when byte 3 bit 7 is 1
value	The channel number is the numeric value of these four bits, plus one, with bit 0 as the least significant bit.	

NOTE 1 The defined multichannel modes identify mappings between channel numbers and function. Some mappings may involve groupings of up to 32 channels by combining two modes.

NOTE 2 For compatibility with equipment that is only sensitive to the channel status data in one subframe the channel carried by subframe 2 may indicate the same channel number as channel 1. In that case it is implicit that the second channel has a number one higher than the channel of subframe 1 except in single channel double sampling frequency mode.

5.5.4 Byte 4: DARS, hidden information, multiple-rate sampling frequencies

bits	1 0	Digital audio reference signal
states	0 0	Not a reference signal (default)
	1 0	Grade 1 reference signal – see AES11
	0 1	Grade 2 reference signal – see AES11
	1 1	Reserved and not to be used until further defined

bit	2	Information hidden in PCM signal
	0	No indication (default)
	1	Audio sample word contains additional information in the least significant bits (see AES55).

bits	6 5 4 3	Sampling frequency
states	0 0 0 0	Not indicated (default)
	0 0 0 1	24 kHz
	0 0 1 0	96 kHz
	0 0 1 1	192 kHz
	0 1 0 0	384 kHz
	0 1 0 1	Reserved
	0 1 1 0	Reserved
	0 1 1 1	Reserved
	1 0 0 0	Reserved for vectoring
	1 0 0 1	22,05 kHz
	1 0 1 0	88,2 kHz
	1 0 1 1	176,4 kHz
	1 1 0 0	352,8 kHz
	1 1 0 1	Reserved
	1 1 1 0	Reserved
	1 1 1 1	User defined

bit	7	Sampling frequency scaling flag
state	0	No scaling (default)
	1	Sampling frequency is 1/1,001 times that indicated by byte 4 bits 3 to 6, or by byte 0 bits 6 to 7

NOTE 1 Bit 2 refers to information within the audio sample word, not in the auxiliary bits.

NOTE 2 When bit 2 is set to 1, processing of the audio signal (such as dithering, sample rate conversion and change in level) should be avoided. A receiver may also use this state as a hint that that it should look for extra information (such as MPEG surround sound, see ISO/IEC 23003 -1) in the least significant bits of the signal.

NOTE 3 The sampling frequency indicated in byte 4 is not dependent on the channel mode indicated in byte 1.

NOTE 4 The indication of sampling frequency, or the use of one of the sampling frequencies that can be indicated in this byte, is not a requirement for operation of the interface. The 0000 state of bits 3 to 6 may be used if the transmitter does not support the indication of sampling frequency in this byte, the sampling frequency is unknown, or the sampling frequency is not one of those that can be indicated in this byte. In the later case for some sampling frequencies byte 0 may be used to indicate the correct value.

NOTE 5 The reserved states of bits 3 to 6 of byte 4 are intended for later definition such that bit 6 is set to define rates related to 44,1 kHz, except for state 1000, and clear to defined rates related to 48 kHz. They should not be used until further defined.

5.5.5 Byte 5: Reserved

bits	7 to 0 Reserved
value	Set to logic 0 until further defined

5.5.6 Bytes 6 to 9: Alphanumeric channel origin

bits	7 to 0 Alphanumeric channel origin data
value (each byte)	7-bit data with no parity bit complying with ISO 646, International Reference Version (IRV). LSBs are transmitted first with logic 0 in bit 7. First character in message is byte 6. Nonprinted control characters, codes 01_{16} to $1F_{16}$ and $7F_{16}$, are not permitted. Default value is logic 0 (code 00_{16}).

NOTE ISO 646, IRV, is commonly identified as 7-bit ASCII

5.5.7 Bytes 10 to 13: Alphanumeric channel destination

bits	7 to 0 Alphanumeric channel destination data
value (each byte)	7-bit data with no parity bit complying with ISO 646, International Reference Version (IRV). LSBs are transmitted first with logic 0 in bit 7. First character in message is byte 10. Nonprinted control characters, codes 01_{16} to $1F_{16}$ and $7F_{16}$, are not permitted. Default value is logic 0 (code 00_{16}).

5.5.8 Bytes 14 to 17: Local sample address code

bits	7 to 0 Local sample address code
value (each byte)	32-bit binary value representing first sample of current block. Byte 14 is the least-significant byte. Default value is logic 0.

NOTE This is intended to be set to zero at the start of the recording, for example, and to have the same function as a recording index counter.

5.5.9 Bytes 18 to 21: Time-of-day sample address code

bits	7 to 0 Time-of-day sample address code
value (each byte)	32-bit binary value representing first sample of current block. Byte 18 is the least-significant byte. Default value is logic 0.

NOTE This is the time of day laid down during the source encoding of the signal and remains unchanged during subsequent operations. A value of all zeros for the binary sample address code is, for transcoding to real time, or to time codes in particular, to be taken as midnight (that is, 00 h, 00 min, 00 s, 00 frame). Transcoding of the binary number to any conventional time code requires accurate sample frequency information to provide a sample accurate time.

5.5.10 Byte 22: Reserved

bits	7 to 0	Reserved
	The bits in this byte are reserved and set to logic 0 until further defined	

NOTE Byte 22 was previously specified to carry a set of reliability flags. The definition of 'reliability' in this context was controversial and we are unaware of any application that ever used the feature.

5.5.11 Byte 23: Channel status data CRCC

bits	7 to 0	Channel status data cyclic redundancy check character (CRCC)
value	Generating polynomial is $G(x) = x^8 + x^4 + x^3 + x^2 + 1$. The CRCC conveys information to test valid reception of the entire channel status data block (bytes 0 to 22 inclusive). For serial implementations the initial condition of all ones should be used in generating the check bits with the LSB transmitted first. There is no default; this field shall always be coded with a correct CRCC. See 5.3.2 and annex C	

5.6 Channel Status when non-PCM audio is flagged

When the state of byte 0 bits 0 and 1 are both set to logic 1, the following bits of channel status may be implemented as for linear PCM audio – that is their interpretation may be independent of the state of byte 0 bit 1, The status bits listed in table 1 shall not be used for any other purpose pending further standardization.

Table 1 - Non-PCM audio, protected status bits

Byte	Bit	Function
0	5	Lock indication
0	6 to 7	Sampling frequency
1	4 to 7	User bits management
2	0 to 2	Use of auxiliary bits
3	0 to 7	Multichannel mode indications
4	3 to 7	Sampling frequency multipliers and scaling flag
23	0 to 7	Channel status data CRCC

6. Auxiliary bits

6.1 Availability of auxiliary bits

The four least-significant bits of the 24-bit audio sample word may be used for auxiliary purposes when the word length does not exceed 20 bits.

6.2 Use of auxiliary bits

When these bits are used for any purpose the transmitter shall indicate that use by encoding channel status in byte 2 bits 0, 1 and 2 (see 5.5.2).

NOTE A typical use is the addition of audio channels of limited bandwidth and resolution for co-ordination purposes. This is shown in annex B.

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.

NOTE AES-2id-2006 refers specifically to clause numbers in AES3-2003 rather than to the current multi-part series. Its information remains still valuable, and AES-2id will be updated in due course.

AES10-2008, *AES Recommended Practice for Digital Audio Engineering - Serial Multichannel Audio Digital Interface (MADI)*, Audio Engineering Society, New York, NY, USA.

AES47-2006, *AES standard for digital audio -- Digital input-output interfacing - Transmission of digital audio over asynchronous transfer mode (ATM) networks*, Audio Engineering Society, New York, NY, USA.

AES55-2007, *AES standard for digital audio engineering - Carriage of MPEG Surround in an AES3 bitstream*, 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 Technical Recommendation R68-1992, *Alignment level in digital audio production equipment and in digital audio recorders*. European Broadcasting Union, Geneva, Switzerland.

SMPTE RP155-2004, *Reference level for digital audio systems*. Society of Motion Picture and Television Engineers. New York, NY, US.

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.

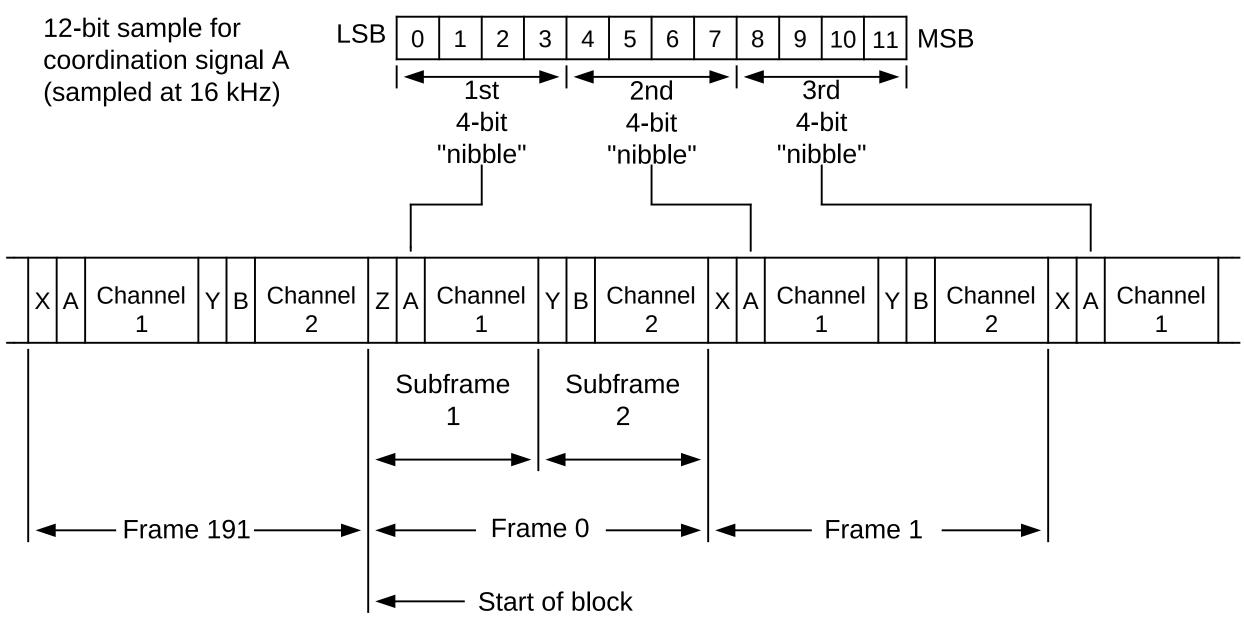
SMPTE 340-2008, *Format for Non-PCM Audio and Data in AES3 - ATSC A/52B Digital Audio Compression Standard for AC-3 and Enhanced AC-3 Data Types*. Society of Motion Picture and Television Engineers, White Plains, NY 10601, US.

ISO/IEC 23003-1 *Information Technology – MPEG audio technologies – Part 1: MPEG Surround,* ISO/IEC, Geneva, Switzerland.

Annex B (informative) Provision of additional, voice-quality channels

When a 20-bit coding range is sufficient for the audio signal, the 4 auxiliary bits may be used for a voice-quality coordination signal (talk back). This is signalled in byte 2 bits 0, 1 and 2 (see 5.5.2).

The voice-quality signal is sampled at exactly one-third of the sampling frequency for the main audio, coded uniformly with 12 bits per sample represented in 2's complement form. It is sent 4 bits at a time in the auxiliary bits of the interface subframes. One such signal may be sent in subframe 1 and another in subframe 2. The block start indication is used as a frame alignment word for the voice-quality signals. In the case of the transmission format specified in Part 3 the two subframes of frame 0 each contain the 4 LSBs of a sample of their respective voice-quality signal, as shown in figure B.1. Figure B.1 also shows two voice-quality signals, one in each subframe.



A nibble carrying A coordination signal
B nibble carrying B coordination signal
X,Y,Z preambles

Figure B.1 — Frame and block structure

Annex C (informative) Generation of CRCC (byte 23) for channel status

The channel status block format of 192 bits includes a cyclic redundancy check code (CRCC) occupying the last 8 bits of the block (byte 23). The specification for the code is given by the generating polynomial:

$$G(x) = x^8 + x^4 + x^3 + x^2 + 1$$

An example of a hardware realization in the serial form is given in figure B. l. The initial condition of all stages is logic 1.

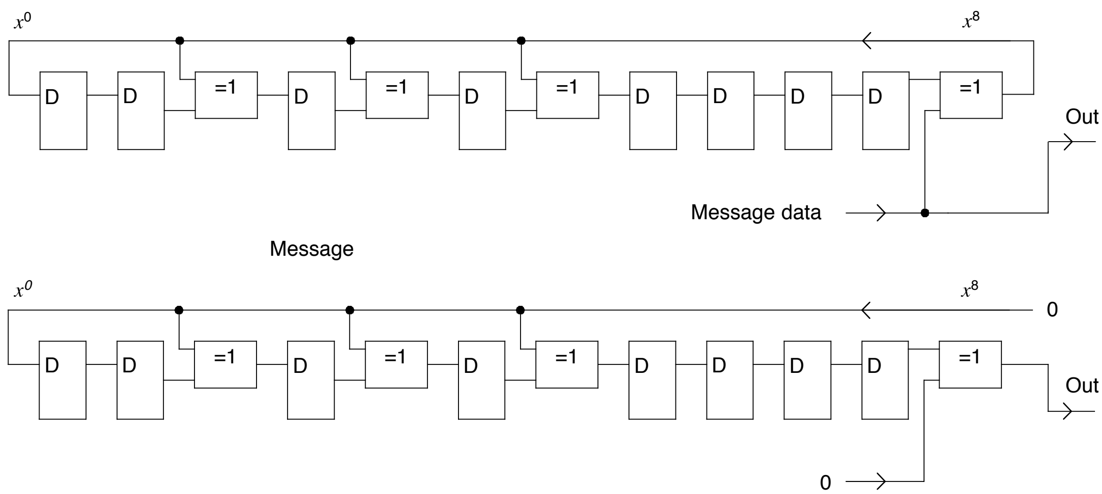


Figure C.1 — Flow diagram including exclusive-or gates

Two examples of channel status data and the resultant CRCC follow.

Example 1:

Byte	Bits set to logic 1
0	0 2 3 4 5
1	1
4	1

All other bits in channel status bytes 0 to 22 inclusive are set to logic 0:

Byte 23	Channel status data cyclic redundancy check character (CRCC)							
Bits	**0**	**1**	**2**	**3**	**4**	**5**	**6**	**7**
Channel status bits	*184*	*185*	*186*	*187*	*188*	*189*	*190*	*191*
value	1	1	0	1	1	0	0	1

Example 2:

Byte	Bits set to logic 1
0	0

All other bits in channel status bytes 0 to 22 inclusive are set to logic 0:

Byte 23	Channel status data cyclic redundancy check character (CRCC)							
Bits	**0**	**1**	**2**	**3**	**4**	**5**	**6**	**7**
Channel status bits	*184*	*185*	*186*	*187*	*188*	*189*	*190*	*191*
value	0	1	0	0	1	1	0	0

No particular level of implementation should be taken as implied by the examples given.