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SMPTE 338M, Television — Format for Non-PCM Audio and Data in AES3 — Data Types

SMPTE 339M, Television — Format for Non-PCM Audio and Data in AES3 — Generic Data Types

SMPTE 340M, Television — Format for Non-PCM Audio and Data in AES3 — ATSC A/52 (AC-3) Data Type

SMPTE 341M, Television — Format for Non-PCM Audio and Data in AES3 — Captioning Data Type

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SMPTE 330M-2000, Television — Unique Material Identifier (UMID)

SMPTE 331M-2000, Television — Element and Metadata Definitions for the SDTI-CP

SMPTE 332M-2000, Television — Encapsulation of Data Packet Streams over SDTI (SDTI-PF)

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Approved SMPTE Recommended Practices

The Society approved three SMPTE Recommended Practices:

RP 203-2000, Real-Time Opportunistic Data Flow Control in an MPEG-2 Transport Emission Multiplex

RP 204-2000, SDTI-CP MPEG Decoder Templates

RP 205-2000, UMID Applications

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—Carlos V. Girod, Jr., P.E., Director of Engineering

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PROPOSED SMPTE STANDARD

for Television —

Format for Non-PCM Audio and Data in an AES3 Serial Digital Audio Interface

1 Scope

This standard specifies an interface format for the transport of non-PCM audio and data in professional applications using the AES3 serial digital audio interface. This standard includes both physical and logical specifications, based on the existing AES3 format, to allow exchange of non-PCM data between different devices. The standard accommodates multiple non-PCM audio and data formats and allows carriage of multiple data streams within a single interface. This standard provides means for carrying time code or time alignment information so that the information conveyed over this interface may be synchronized with information content delivered over other interfaces.

2 Normative references

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

AES3-1992, Digital Audio Engineering — Serial Transmission Format for Two-Channel Linearly Represented Digital Audio Data

SMPTE 338M, Television — Format for Non-PCM Audio and Data in AES3 — Data Types

3 Definition

3.1 **frame frequency:** The frequency at which AES3 frames occur. When linear PCM audio is carried within the AES3 interface, the frame frequency is equivalent to the sampling rate of the linear PCM audio.

4 Introduction

The AES3 standard is widely used in industry to convey linear PCM audio between digital audio devices. However the AES3 standard is limited to two channels of audio. Significant issues arise when multiple AES3 channels are used to convey greater than two channels of associated audio. This standard defines a method in which the existing AES3 format is modified to convey non-PCM data, including non-PCM audio bit streams which are typically, but not necessarily, bit-rate reduced. This allows a single audio program of more than 2 channels or multiple audio programs, each potentially consisting of more than 2 channels, to be carried over a single AES3 interface.

This standard specifies a modification to the logical portion of the AES3 standard and is compatible with the existing AES3 standard for transport of linear PCM audio. Therefore, this standard can facilitate the interconnection of equipment which may be capable of working with either linear PCM or non-PCM audio and data. This method may allow some existing equipment which is capable of recording linear PCM to also record non-PCM data. Independent use of the AES3 channels is also supported by this standard to allow both one channel of linear PCM audio and non-PCM data to be carried within a single AES3 signal.

This standard accommodates methods of synchronization both for reconstruction of the original source audio signals coded within non-PCM audio streams and for time alignment with other information streams such as an associated video stream. Synchronization methods for specific non-PCM bitstreams are dependent on the data type carried within the bitstream and are beyond the scope of this standard. However, other standards and recommended practices may contain important and necessary information regarding synchronization requirements for specific data types. In general, it is required to refer to such information in order to properly transmit and receive non-PCM streams using this standard. References to documents containing synchronization requirements for specific data types can be found in SMPTE 338M.

NOTE – Because of the wide variety of data types that may potentially be conveyed according to this standard, no global synchronization requirements are specified in this standard. However, synchronization of non-PCM data content, both in terms of the relationship of the coded audio sampling rate to the AES3 frame frequency (when conveying non-PCM audio) and in terms of time synchronization to other information streams, is very important to the proper use of this standard. Additionally, synchronization requirements for specific data types may impose buffering requirements on devices that support these data types. Therefore, it is required that additional documents containing synchronization requirements for specific data types be referenced in order to maintain compatibility with these data types.

A dedicated data type, the time stamp data type, is included to support synchronization methods. Many data types may make use of information contained in the time stamp data bursts, which can include SMPTE 12M time code information, to maintain time synchronization with other information streams. References for documents describing the time stamp data type can be found in SMPTE 338M.

This standard is applicable to professional audio equipment only. Existing standards (IEC 61937) cover transport of non-PCM data in the consumer environment. Some interoperability between professional and consumer equipment is accommodated by this standard, and specific compatibility requirements are described.

4.1 Overview

The logical format of the AES3 interface consists of a sequence of subframes. Each subframe is intended to convey one linear PCM sample, and contains 32 time slots, each of which (excluding the four time slots used for synchronization purposes) can carry a single bit of information. A pair of subframes, each containing the PCM word of one audio channel, make up an AES3 frame containing two PCM words, one from channel 1 and one from channel 2. A sequence of 192 frames makes up a block. The 192 channel status bits for each channel during a block make up the 192-bit (24-byte) channel status word for that channel. The standard usage of the 32 AES3 time slots is modified when conveying non-PCM data. This usage is shown in table 1.

The non-PCM data steams to be conveyed are formed into data bursts, each consisting of a preamble containing information about the burst followed by a data payload. The data bursts are placed in the audio sample word/aux data fields of AES3 subframes in one of two modes. In the frame mode, the data space from each subframe within an AES3 frame is combined to allow up to 48 bits of data to be placed in each frame. In the subframe mode, each channel is treated independently and data are not shared across subframes within a frame. In this mode, each subframe may contain either linear PCM audio or non-PCM data. This allows the AES3 interface to simultaneously convey two linear PCM channels, or one linear PCM channel and one set of data bit streams, or two sets of data bit streams.

Table 1 – AES3 subframe bit field usage for non-PCM data

Bit locations	AES3 field designations	Non-PCM data usage
0 – 3	Sync preamble	As per AES3
4 – 7	Aux data field or 4 LSBs of audio sample word	Non-PCM data in 24-bit mode
8 – 27	Audio sample word	Non-PCM data
28	Validity (V) bit	As per AES3
29	User data (U) bit	As per AES3
30	Channel status (C) bit	Bytes 0, 1, 2, 23 defined, other bytes undefined
31	Parity (P) bit	As per AES3

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Data bursts are tagged with a number indicating to which data stream they belong. Up to seven different non-PCM data streams, along with an additional stream type dedicated for time stamp data bursts, may be time multiplexed together to form a set of data bit streams. In the subframe mode, this allows up to 14 independent non-PCM data streams to be multiplexed within a single AES3 interface.

Data bursts are placed in the audio sample word/aux data fields of AES3 subframes using either 16, 20, or 24 bits of the available space within each subframe. While the 24-bit mode allows more efficient use of the AES3 data capacity, the 16- and 20-bit modes may be desired when interfacing with existing equipment that may be limited to 16- or 20-bit operation. Annex B contains additional comments regarding the use of 20- and 24-bit data modes.

5 Interface format

5.1 Detailed specification

The logical interface format shall be as defined in AES3 except as noted in this standard.

The electrical and mechanical characteristics of the interface shall conform to either AES3 or ANSI/SMPTE 276M.

When in the subframe mode and one channel is used to convey linear PCM, the channel conveying linear PCM shall be used in accordance with AES3.

Non-PCM data shall be placed in the available AES3 data space in bursts as described in this standard. The non-PCM data shall occupy some or all of bit locations 4-27 of the AES3 subframe. Unused bit locations within a subframe or in subframes between bursts shall be set to 0.

5.2 Channel status word

For AES3 channels that convey non-PCM data byte 0, byte 1, byte 2, and byte 23 of the channel status word shall be used as described in this standard. The usage of the remaining bytes of the channel status word is undefined for channels that convey non-PCM data. It is recommended that each bit of the undefined channel status bytes be set to 0.

NOTE – AES3 defines three implementation types in regard to use of channel status features: minimum, standard, and enhanced. For compatibility with existing implementations of AES3, it is recommended that the standard implementation be used. The usage of channel status bytes 0, 1, 2, and 23 as defined in this standard is consistent with the standard implementation as described in AES3 although specific interpretations of some bit fields differ from the AES3 specification.

Byte 0 bit 0 of the channel status word shall be set to 1, indicating professional use of the channel status block. Consumer use of the AES3 bit stream is not considered in this standard.

Byte 0 bit 1 shall be set to 1, indicating nonaudio mode.

Byte 0 bits 2-4 shall be set to 000.

Byte 0 bit 5 shall be set according to AES3 paragraph 4, but shall indicate the source frame frequency lock status. The source frame frequency shall be interpreted as the source rate from which the AES3 interface frame rate is derived. This bit shall not necessarily be used to indicate that the source sampling rates of audio signals encoded within non-PCM audio streams in the AES3 signal are locked to the AES3 frame rate, although such use may be specified by certain data steam types.

Byte 0 bits 6-7 shall be set according to AES3 paragraph 4, but shall indicate the frame frequency of the AES3 interface. These bits shall not necessarily be used to indicate the source sampling rate of audio signals encoded within non-PCM audio streams in the AES3 signal, although such use may be specified by certain data stream types.

NOTE - State 00 (interpreted as frame frequency not indicated) shall be allowed by this standard, but it is recommended that the actual frame frequency be indicated.

Table 2 provides a summary of the setting of byte 0 of the channel status word when conveying non-PCM data.

Table 2 – Channel status bits in byte 0

Bit(s)	Value	Comments
Bit 0	1	Professional use of channel status block
Bit 1	1	Nonaudio mode
Bits 2-4	000	Emphasis not indicated
Bit 5	—	Frame frequency lock status
Bits 6, 7	—	Indicates frame frequency per AES 3

Byte 1 bits 0-3 shall be set to 0000. Byte 1 bits 4-7 shall be set according to AES3 paragraph 4.

Table 3 provides a summary of the setting of byte 1 of the channel status word when conveying non-PCM data.

Table 3 – Channel status bits in byte 1

Bit(s)	Value	Comments
Bits 0-3	0000	Encoded channel mode not indicated
Bits 4-7	—	Encoded user bits management per AES 3

Byte 2 bits 0-2 shall be set according to AES3 paragraph 4 to indicate the auxiliary sample bit usage. In this context, audio sample word length shall refer to the non-PCM data word length (the number of bits used to convey non-PCM data) as defined by the data_mode parameter (see 6.1.3.2). In the case that multiple non-PCM data streams are carried within the AES3 interface and multiple data_mode words exist, the data mode corresponding to the largest non-PCM data word length shall be used as the reference.

Byte 2 bits 3-5 shall be set according to AES3 paragraph 4, but shall indicate the non-PCM data word length (the number of bits used to convey non-PCM data) as defined by the data_mode parameter (see 6.1.3.2). In the case that multiple non-PCM data streams are carried within the AES3 interface and multiple data_mode words exist, the data mode corresponding to the largest non-PCM data word length shall be used as the reference.

NOTE – State 000 (interpreted as non-PCM data word length not indicated) shall be allowed by this standard, but it is recommended that the actual non-PCM data word length be indicated. Annex B contains additional comments regarding the use of these bits for the 20- and 24-bit data modes.

Byte 2 bits 6-7 shall be set to 00.

Table 4 provides a summary of the setting of byte 2 of the channel status word when conveying non-PCM data.

Byte 23 bits 0-7 shall be set according to AES3 paragraph 4 to indicate a valid CRCC value for the channel status block (the default state of 0 is not allowed). This usage is summarized in table 5.

Table 4 – Channel status bits in byte 2

Bit(s)	Value	Comments
Bits 0-2	—	Auxiliary sample bit usage per AES3
Bits 3-5	—	Non-PCM data word length per AES3
Bits 6, 7	00	Reserved

Table 5 – Channel status bits in byte 23

Bit(s)	Value	Comments
Bits 0–7	—	CRCC word per AES3

5.3 Sample rate synchronization

This standard places no requirement on synchronization between the AES3 interface rate (frame frequency) and sample rates of the audio coded within non-PCM data streams. However, other standards or recommended practices may specify a fixed relationship between the AES3 interface and the coded audio sample rate for certain data types. References to documents containing sample rate synchronization requirements for specific data types can be found in SMPTE 338M (see note in clause 4).

6 Data burst format

The non-PCM data streams to be conveyed shall be formed into data bursts consisting of data words in a continuous sequence of AES3 frames. Each data burst shall consist of a burst_preamble followed by a burst_payload. When multiple streams are present, bursts from each stream shall be placed in the AES3 stream in a time-division multiplexed fashion.

6.1 burst_preamble

The burst_preamble shall occur at the beginning of each data burst and shall be followed by the burst_payload. The burst_preamble shall occupy 16, 20, or 24 bits in each of 4 consecutive subframes in one of two manners according to whether the frame or subframe mode is in use. The preamble shall consist of four words designated as Pa, Pb, Pc, Pd. The contents of these four words shall be as specified in table 6. When placed into an AES3 subframe, the MSB of a preamble word shall be placed into time slot 27 of that subframe. The LSB of each preamble word shall be placed into time slot 12, 8, or 4 depending on the data mode. In the 16-bit mode time slots, 11-8 shall be set to 0 for each subframe containing a preamble word. In the 16- and 20-bit modes, it is recommended that time slots 7-4 also be set to 0 for each subframe containing a preamble word.

Table 6 – Preamble words

Preamble word	Contents	
Pa	Sync word 1	= 0xF872 (16-bit mode) = 0x6F872 (20-bit mode) = 0x96F872 (24-bit mode)
Pb	Sync word 2	= 0x4E1F (16-bit mode) = 0x54E1F (20-bit mode) = 0xA54E1F (24-bit mode)
Pc	burst_info value	
Pd	length_code (unsigned integer), equal to the number of data bits in the burst_payload	

6.1.1 Frame mode

The 4 preamble words shall be contained in 2 sequential frames. The frame beginning the data burst shall contain preamble word Pa in the Ch1 subframe, and Pb in the Ch2 subframe. The next frame shall contain Pc in Ch1 and Pd in Ch2.

6.1.2 Subframe mode

The 4 preamble words shall be contained in 4 sequential subframes of the individual channel (Ch1 or Ch2) being employed to convey the non-PCM data. The subframe (of the channel being used) beginning the data burst shall contain preamble word Pa, the next subframe (of the channel) in the burst shall contain Pb, etc.

6.1.3 burst_info

The burst_info value shall contain information about the contents of the burst_payload as specified in table 7. Bit 15, bit 19, or bit 23 of burst_info shall be considered the MSB depending on the data mode (16, 20, or 24 bit). Note that the burst_info MSB will always be located in time slot 27 of an AES3 subframe.

6.1.3.1 data_type

The 5-bit data_type field shall indicate the type of data contained in the burst_payload. The MSB of the data_type field shall be placed in bit 4, 8, or 12 of the burst_info word depending on the data mode. Note that the data_type MSB will always be located in time slot 16 of an AES3 subframe.

Data_type values shall apply to individual data bursts only. The data type may vary between differing stream numbers when multiple data streams are carried within the AES3 interface.

Supported data types and the mapping of data_type values to specific data types are defined in SMPTE 338M. Additional standards may contain specific data type dependent formatting requirements. References to documents containing formatting requirements for specific data types can be found in SMPTE 338M.

Table 7 – burst_info

Bit(s)			Value
24-bit mode	20-bit mode	16-bit mode	
0-3	—	—	Reserved
4-7	0-3	—	Reserved
8-12	4-8	0-4	data_type (5-bit unsigned integer = 0-31)
13-14	9-10	5-6	data_mode
15	11	7	error_flag 1 indicates data burst may contain errors, 0 indicates data may be valid.
16-20	12-16	8-12	data_type_dependent
21-23	17-19	13-15	data_stream_number

6.1.3.2 data_mode

The 2-bit data_mode field shall indicate the mode in which the data for the burst_payload is placed in AES3 subframes as specified in table 8. The MSB of the data_mode field shall be placed in bit 6, 10, or 14 of the burst_info word depending on the data mode. Note that the data_mode MSB will always be located in time slot 18 of an AES3 subframe.

In each data mode, the burst_payload data words shall occupy the subframe time slots as indicated in table 8. In the 16- and 20-bit modes, unused time slots shall contain the value 0.

Data-mode values shall apply to individual data bursts only. The data mode may vary between consecutive data bursts of a given data_stream_number, or between data bursts of differing stream numbers when multiple data streams are carried within the AES3 interface.

Table 8 – data_mode

Value	Data mode	burst_payload position
0	16-bit mode	Subframe time slots 27-12
1	20-bit mode	Subframe time slots 27-8
2	24-bit mode	Subframe time slots 27-4
3	Reserved	N/A

NOTE – Annex B contains additional comments regarding the use of the 20- and 24-bit data modes.

6.1.3.3 error_flag

The error_ flag bit shall provide an error indication for the data in the burst_payload. If the data in the burst_payload is known to be error free or if it is unknown whether the data contains errors, then the value of this bit shall be set to a 0. If the data in the burst_payload is known to contain errors, this bit may be set to a 1. Note that the error_flag bit will always be located in time slot 19 of an AES3 subframe.

6.1.3.4 data_type_dependent

The data_type_dependent field shall contain 5 bits whose meaning is dependent on the value of data_type. Specific coding for this field may be found in other standards or recommended practices applying to specific data types. References to documents containing descriptions of this field for specific data types can be found in SMPTE 338M.

6.1.3.5 data_stream_number

The 3-bit data_stream_number shall indicate the number of the data stream to which the burst belongs. The MSB of the 3-bit data_stream_number shall be placed in bit 15, 19, or 23 of the burst_info word depending on the data mode. Note that the data_stream_number MSB will always be located in time slot 27 of an AES3 subframe.

Each independent data stream shall use a unique value for data_stream_number. Eight data stream numbers (0-7) are available. Data stream number 7 is reserved for the time stamp data type. All time stamp data bursts shall be encoded with a data_stream_number set to 7. Data stream numbers 0-6 are available for all data types except the time stamp data type. Therefore, up to 7 independent data streams may be time multiplexed in the AES3 interface when in the frame mode. In the subframe mode, each AES3 channel shall be treated independently and the requirement for unique data stream numbers for each data stream shall apply only within a given AES3 channel. In this mode, up to 14 independent data streams (7 in each channel) may be time multiplexed in the AES3 interface.

NOTE – Individual time stamp data bursts apply to specific data bursts of other data types. Although all time stamp data bursts are identified as data stream number 7, they should not be considered as a single stream of related time stamp values. When time code information is carried within time code data bursts, multiple time code streams may be conveyed within data bursts identified as data stream number 7. Other standards or recommendations contain further information regarding the time stamp data type. References to documents containing information about the time stamp data type can be found in SMPTE 338M.

6.1.4 length_code

The length_code shall indicate the length of the burst_payload in bits. The length_code shall occupy 16, 20, or 24 bits of an AES3 subframe depending on the data mode. The length_code MSB shall always be located in time slot 27 of an AES3 subframe.

The burst_ payload field is limited in size based on the data mode: from 0 to 65,535 bits in the 16-bit mode, from 0 to 1,048,575 bits in the 20-bit mode, and from 0 to 16,777,215 bits in the 24-bit mode. The size of the burst_preamble is not counted in the value of length_code.

6.2 burst_payload

The burst_payload shall be segmented into data words and placed in a continuous sequence of AES3 frames in one of two modes:

6.2.1 Frame mode

In the frame mode, both AES3 channels shall be utilized to carry one set of non-PCM data streams. The available data space from each subframe within an AES3 frame shall be combined when packing data bursts into a continuous sequence of frames. This mode will allow up to 32, 40, or 48 data bits to be placed in a single AES3 frame depending on the data_mode setting in the burst_preamble.

Considering the burst_payload as a serial stream of bits, the first bit of the first data word of the payload in a burst shall occupy the MSB bit position of subframe 1 (time slot 27) and the last bit of the first data word shall occupy the LSB bit position (according to the data_mode setting) of subframe 2. The last data bits of the burst_payload may occupy only a fraction of the last frame. Any unused bits in the last frame shall be set to 0.

6.2.2 Subframe mode

In the subframe mode, each AES3 channel shall be utilized independently to carry either one set of non-PCM data streams or linear PCM audio. The subframe from each AES3 channel within a frame shall be considered independently when packing data bursts into a continuous sequence of frames. This mode will allow up to 16, 20, or 24 data bits per channel to be placed in a single AES3 frame depending on the data_mode setting in the burst_preamble.

Considering the burst_payload as a serial stream of bits, the first bit of the first data word of the payload in a burst shall occupy the MSB bit position of the subframe (time slot 27) and the last bit of the first data word shall occupy the LSB bit position (according to the data_mode setting) of the subframe. The last data bits of the burst_payload may occupy only a fraction of the last frame. Any unused bits in the last frame shall be set to 0.

The channel status words for each channel shall be treated independently when in the subframe mode. Any channel conveying non-PCM data shall set the channel status bits according to this standard. Any channels conveying linear PCM audio shall set the channel status bits according to AES3.

6.3 Burst spacing

There shall not be a sequence of 4096 or more AES3 frames (in frame mode) or subframes (in subframe mode) which contain at least one data burst, without the beginning of at least one of the data bursts preceded by four AES3 subframes which have subframe contents in time slots 8-27 of all 0s. This requirement ensures that there are occurrences of an extended sync code of 0, 0, 0, 0, Pa, Pb.

Data bursts from a given non-PCM data stream shall be placed in the AES3 interface in consecutive order. If multiple non-PCM data streams are placed in the AES3 interface (or in an individual channel in the subframe mode), the data bursts from each stream shall be interleaved in a time multiplexed manner. This standard places no other requirements on the multiplexing method as long as the above requirements are met for each stream. Other standards or recommended practices may define specific multiplexing techniques which may be data type dependent.

This standard makes no requirement for placing data bursts at fixed intervals or locations within the AES3 stream, although such features may be made use of for conveying synchronization information. Other standards or recommended practices may specify data type dependent fixed reference locations and repetition rates for such purposes. References to documents containing burst spacing requirements for specific data types can be found in SMPTE 338M. The time_stamp data type may also be used for conveying specific synchronization information for some data types. These synchronization methods may place requirements on burst spacing (see note in clause 4).

6.4 Data type dependent fields

The format of data contained within the data_type_specific and burst_payload fields is dependent on the data_type field and is beyond the scope of this standard. Specific coding for these fields may be found in other standards or recommended practices applying to specific data types. References to documents containing formatting information on

these fields for specific data types can be found in SMPTE 338M.

7 Consumer format compatibility

This standard shall apply to professional equipment only. Some users in a professional environment may wish to have compatibility with consumer devices. For explicit compatibility with consumer devices, professional equipment should implement interfaces or specific interface modes conforming to the consumer format specification IEC 61937. However, it may be desired for some professional equipment to be compatible at the level of bit stream formatting while still utilizing the AES3 interface in the professional mode. This clause lists specific compatibility requirements and issues to consider in terms of bit stream formatting.

7.1 Receiving devices

The consumer bit stream format can, in general, be considered a subset of the professional bit stream format defined by this standard. Professional devices implementing this standard should be capable of reading consumer burst preambles and properly extracting consumer data bursts carried in the AES3 interface. This does not guarantee professional receivers can properly receive and decode all data types. Some consumer data types may be undefined by this standard. In practice, professional receivers should discard any data bursts containing undefined data types. Refer to 7.3 for additional data type dependent compatibility issues.

7.2 Source devices

In order for professional devices to produce AES3 output bit streams compatible with the consumer format, data bursts should be formatted in the following manner:

All data bursts shall be restricted to the 16-bit frame mode (data_mode=0, both AES3 subframes used).

At least one bit stream of data_stream_number=0 shall be present. This bit stream shall contain coded audio information considered to be a main audio service. Consumer devices may not be capable of receiving bit streams with data stream numbers greater than 0.

Some professional data types defined by this standard may be undefined in the consumer format. Consumer devices can be anticipated, but not guaranteed, to ignore these data types.

Consumer devices should be capable of reading burst preambles and properly extracting data bursts formatted according to the above recommendations. This does not guarantee consumer receivers can properly receive and decode all data types. Professional devices should also take into account consumer synchronization requirements for certain data types which may affect the ability of consumer devices to receive and decode the encoded data. Refer to 7.3 for data type dependent compatibility issues. In addition, many consumer receivers rely on channel status information for detection of non-PCM coded data. Consumer receivers may not be capable of detecting bit streams in AES3 signals where the channel status is set according to this standard.

7.3 Data type dependent issues

For data types that are common to both professional and consumer specifications, specific data type dependent issues may still affect bit stream compatibility between professional and consumer devices. For some data types, the data_type_dependent field may differ between professional and consumer specifications. Such differences may prohibit proper exchange of coded data and/or synchronization information. The coding of the burst_payload may also differ for some data types. In addition, synchronization methods required by the consumer format for some data types may place restrictions on data burst spacing which, in some cases, may differ from the allowed professional formats. Data type specific formatting issues are beyond the scope of this standard. Compatibility with specific data types may be addressed by other standards and recommended practices. References to documents containing formatting requirements for specific data types can be found in SMPTE 338M.

Annex A (informative)
Autodetection of audio/data mode

The AES3 interface can convey either PCM audio, non-PCM data, or both in separate channels. Receiving devices capable of receiving AES3 streams containing PCM and non-PCM data may wish to know whether the AES3 information is to be considered PCM audio, non-PCM data, or both. This information is best conveyed by setting bit 1 of the channel status word to indicate data. In some applications, it may be useful for receivers to be able to determine whether the AES3 contents are PCM audio or non-PCM data, without referring to bit 1 of the channel status word. This may be done quite reliably by recognizing that the sync code formed by the first two words of the preamble (Pa, Pb) are unlikely to occur very often in natural PCM audio. By looking for an extended sync code consisting of six words (4 zeros followed by Pa, Pb which, in the case of the 16-bit mode is 0x0000, 0x0000, 0x0000, 0x0000, 0xF872, 0x4E1F), the probability of a false occurrence of sync will be vanishingly small. The decision process which may be followed is shown in figure A.1. In this diagram, the mode of the receiver can be switched between PCM and DATA. The SYNC function is meant to indicate whether, in a span of 4096 AES3 frames, the extended sync code is found. Note that if the AES3 stream goes idle (all zeros), the autodetector will go into PCM mode, and only switch back to DATA mode when a data burst appears. If this behavior is undesirable, it can be prevented by inserting null data bursts at least once every 4096 AES3 frames.

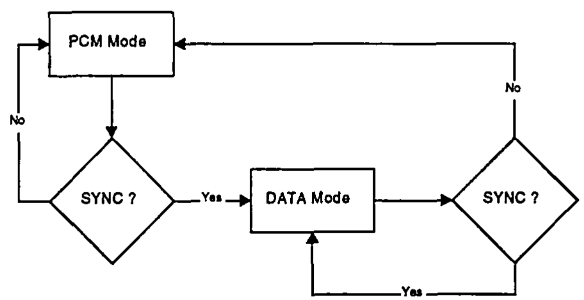

Figure A.1 – PCM-DATA automode detection

Annex B (informative)
Equipment compatibility issues

B.1 Data mode

This standard defines 20- and 24-bit data modes so that efficient use can be made of the available data space in the AES3 interface. Users should be aware that many digital audio devices restrict their operation to 16 bits. Such devices may truncate 20- or 24-bit data words to 16 bits for storage and/or transmission. Other devices may accommodate 20-bit data words, but not 24 bits. It is recommended that users consider whether such devices may be possible receivers of non-PCM data signals when determining data modes. In general, it is recommended that the 16-bit data mode be used in cases where the extra data bandwidth of the 20- and 24-bit modes is not needed.

Users should also be aware that some existing devices make use of the channel status bits for encoded audio word length to determine stored/transmitted data word sizes. If the 20- or 24-bit modes are used, it is recommended to set the encoded audio word length channel status bits according to the data mode as described in 5.2. Use of the default audio word length setting may result in improper truncation of the non-PCM data words. Receiving devices, when receiving non-PCM data, should determine the data mode directly from the data_mode field in the data burst preamble rather than from the encoded audio word length channel status bits.

B.2 Channel status usage

Many existing digital audio devices do not store/transmit channel status information that is present on received digital audio inputs. Other existing devices only store a limited amount of channel status information. Such devices, while not strictly compatible with this standard, still may be useful for conveying non-PCM information formatted according to this standard. For this reason, it is recommended that receiving devices not rely on channel status fields as the sole source of information as to whether non-PCM data are present and for determining non-PCM format information such

as data mode. For example, the nonaudio channel status bit (bit 1) may be used as the primary means of determining the presence of non-PCM data, but an autodetection technique, such as that described in annex A, may be used to confirm the channel status indication. This may allow proper reception of non-PCM data output from noncompliant devices.

B.3 Non-PCM data modification

Whenever possible, receiving devices should take into account the fact that non-PCM data words present in input signals may have become modified or corrupted in several ways, especially if devices not strictly compliant with this standard are involved in the storage and/or transmission of the signal. Possible modifications include truncation of 20- and 24-bit non-PCM data words to 16 or 20 bits, momentary disruptions in the bit stream due to signal switching or synchronization devices, errors propagated from storage devices or transmission links, and other forms of PCM type processing performed by equipment not compliant with this standard (dithering, gain changes, sample rate conversion, effects processing, etc.). No specific capability for detection of these kinds of bit stream modifications is provided by this standard; however, certain encoded data types may provide such capability (CRC, checksums, parity, etc.). It is recommended that receiving devices make use of these detection features to provide operator feedback.

Annex C (informative)
Bibliography

ANSI/SMPTE 276M-1995, Television — Transmission of AES-EBU Digital Audio Signals Over Coaxial Cable

SMPTE 12M-1999, Television, Audio and Film — Time and Control Code

IEC 61937, Digital Audio — Interfaces for Non-Linear PCM Encoded Audio Bitstreams Applying IEC 60958

ITU-T Rec. H.2220.0 | ISO/IEC 13818-1, Information Technology — Generic Coding of Moving Pictures and Associated Audio Information: Systems

PROPOSED SMPTE STANDARD

SMPTE 338M

for Television —
Format for Non-PCM Audio and Data in AES3 — Data Types

1 Scope

This standard describes the data_type field defined in SMPTE 337M. This field describes data types that may be carried in an AES3 digital audio interface according to SMPTE 337M. This standard defines supported data types, but does not cover formatting that may be required for each data type. References are included for additional standards that describe data type specific formatting requirements.

2 Normative references

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

SMPTE 337M, Television — Format for Non-PCM Audio and Data in an AES3 Serial Digital Audio Interface

SMPTE 339M, Television — Format for Non-PCM Audio and Data in AES3 — Generic Data Types

SMPTE 340M, Television — Format for Non-PCM Audio and Data in AES3 — ATSC A/52 (AC-3) Data Type

SMPTE 341M, Television — Format for Non-PCM Audio and Data in AES3 — Captioning Data Type

3 Introduction

SMPTE 337M describes general formatting requirements when carrying non-PCM data in an AES3 digital audio bit stream. Data are formatted into data bursts each consisting of a burst_preamble and a burst_payload. The data_type field in the burst_preamble of each data burst defines the type of non-PCM data carried within the burst_payload of the data burst. Each data type includes additional formatting requirements not defined in SMPTE 337M.

This standard maps the value of the data_type field to specific data types and references additional standards that contain data type specific formatting requirements and information. No specific formatting information is contained in this standard.

Some data types are mapped directly to data_type values. Specific references are included for these data types.

4 Data types

Table 1 defines the data_type field described in SMPTE 337M.

5 Data type references

This clause provides references to additional standards that describe specific data types and formatting requirements for these data types.

5.1 Data type 0 – Null data

Reference: SMPTE 339M, Generic data types.

5.2 Data type 1 – ATSC A/52 (AC-3) (audio)

Reference: SMPTE 340M, ATSC A/52 (AC-3) data type.

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Table 1 – Data type field

data_type value	Data type
0	Null data
1	ATSC A/52 (AC-3) data (audio)
2	Time stamp data
3	Reserved
4	Reserved MPEG-1 layer 1 data (audio)
5	Reserved MPEG-1 layer 2 or 3 data or MPEG-2 data without extension (audio)
6	Reserved MPEG-2 data with extension (audio)
7	Reserved
8	Reserved MPEG-2 layer 1 data low-sampling frequency (audio)
9	Reserved MPEG-2 layer 2 or 3 data low-sampling frequency (audio)
10–26	Reserved
27	Reserved SMPTE KLV data
28	Reserved Dolby E data (audio)
29	Captioning data
30	User defined data
31	Reserved

5.3 Data type 2 – Time stamp

Reference: SMPTE 339M, Generic data types.

5.4 Data type 29 – Captioning data

Reference: SMPTE 341M, Captioning data type.

5.5 Data type 30 – User data

Reference: SMPTE 339M, Generic data types.

PROPOSED SMPTE STANDARD

SMPTE 339M

for Television —

Format for Non-PCM Audio and Data in AES3 — Generic Data Types

1 Scope

This standard specifies data type specific format requirements for several types of data bursts that may be carried within an AES3 interface according to SMPTE 337M. Included are descriptions of the data type, the format of the burst_payload for the data type, the coding of data type dependent fields in the burst_preamble, and additional data burst and bit stream formatting requirements not defined in SMPTE 337M. This includes specific synchronization methods which may affect formatting.

This standard covers generic data types which include null data, time stamp data, and user defined data.

2 Normative references

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

SMPTE 337M, Television — Format for Non-PCM Audio and Data in an AES3 Serial Digital Audio Interface

SMPTE 338M, Television — Format for Non-PCM Audio and Data in AES3 — Data Types

3 Null data (data_type = 0)

The null data type is provided so that the preamble sync codes may be intentionally inserted into the data stream. Null data bursts may be of use in receiving devices to potentially enhance reliable autodetection of whether or not the subframe contains PCM audio or non-PCM data.

3.1 Null data burst_preamble

The burst_preamble for a null data burst shall set the length_code, error_flag, and data_type_dependent values to 0. The data_stream_number shall be set to any valid number other than 0x7. The data_type shall be set to 0.

3.2 Null data burst_payload

The null data burst shall contain no burst_payload (a payload of length 0).

4 Time stamp (data_type = 2)

The time stamp data type is used to convey time synchronization information associated with a specific data burst. This information shall include SMPTE 12M time code information and/or specific delay information. The use of time stamp bursts is optional. When present, the time stamp information shall apply to the data burst which immediately follows in the AES3 interface. When time code information is present, the time stamp may contain date and time zone information coded in the time code user (binary) groups according to SMPTE 309M.

4.1 Time stamp burst_preamble

The burst_preamble for a time stamp data burst shall include a data_type_dependent field set to a value of 0. The data_stream_number shall be set to 0x7. The data_mode parameter shall be set to 0. The data_type shall be set to 2.

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NOTE – When time code information is conveyed within time stamp data bursts, data stream number 7 should not be interpreted as a single data stream of time code information. Time stamp data bursts are independent and apply only to the data burst that immediately follows in the AES3 interface; therefore, time code information from multiple time code sources may be contained within data stream number 7. Within an individual data burst, time code information from at most one time code source will be present.

4.2 Time stamp burst_payload

The time stamp burst_payload shall always be packed in the 16-bit data mode and shall have a length of 6-9 16-bit words. The burst_payload shall be in the format shown in table 1.

In table 1, numbers in brackets (e.g., [63]) refer to the corresponding bit number of the SMPTE 12M time code word. These bits shall correspond to user group (Usr8-Usr1), color frame flag (cf), drop frame flag (df), and unassigned bits from a SMPTE 12M time code word. Flag bit f1 (in word 5) shall be set to a 1 if the information in these bit fields has been copied from a source of SMPTE 12M time code. If this information has not been provided, flag bit f1 shall be set to a 0, and these bit fields shall be set to 0.

The SMPTE 12M time code hours address field is denoted by the bit fields H20-H1 corresponding to the 12M BCD coded hours bits (e.g., H20 corresponds to 20s of hours). The minutes, seconds, and frames address fields are denoted in a similar fashion by the bit fields M40-M1, S40-S1, and F20-F1, respectively.

Word 4 carries a value designated sample number. This word is provided for use when the time stamp is followed by (and applies to) a data burst containing low bit rate encoded audio. The sample number field allows the time code value to be associated with a particular audio sample that is represented in a burst of encoded audio. When the time stamp is followed by a data burst that does not contain encoded audio, this field is not used but must still be present. In this case, it shall be set to a value of 0.

In word 4, bit fields s15-s0 denote a sample number to which the 12M time code applies. The sample number is an unsigned integer that shall indicate the linear PCM sample to which the time code information applies. When the time stamp data burst is associated with a data burst containing coded audio, the sample number shall be relative to the linear PCM samples represented by the coded audio data burst as illustrated in figure 1. For example, if an audio data burst represents 1536 linear PCM samples and the sample number is set to 1007, then the 12M time code information applies to sample 1007 of the 1536 samples. If the sample number contained in a time stamp is greater than the number of samples represented by the corresponding audio data burst, then the sample number shall correspond to a sample represented by a subsequent data burst of the same stream number.

Table 1 – Time stamp burst_payload

Time stamp payload word		MSB							Bit number								LSB
		15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
0	Usr8, Usr7, hours	[63]	[62]	[61]	[60]	[55]	[54]	[53]	[52]	[59]	[58]	H20	H10	H8	H4	H2	H1
1	Usr6, Usr5, minutes	[47]	[46]	[45]	[44]	[39]	[38]	[37]	[36]	[43]	M40	M20	M10	M8	M4	M2	M1
2	Usr4, Usr3, seconds	[31]	[30]	[29]	[28]	[23]	[22]	[21]	[20]	[27]	S40	S20	S10	S8	S4	S2	S1
3	Usr2, Usr1, cd, df, frames	[15]	[14]	[13]	[12]	[7]	[6]	[5]	[4]	[11]	[10]	F20	F10	F8	F4	F2	F1
4	Sample number	s15	s14	s13	s12	s11	s10	s9	s8	s7	s6	s5	s4	s3	s2	s1	s0
5	Reserved, flags	R	R	R	R	R	R	R	R	R	f2	a3	a2	a1	a0	f1	[10]
6	User private (optional)	U	U	U	U	U	U	U	U	U	U	U	U	U	U	U	U
7	User private (optional)	U	U	U	U	U	U	U	U	U	U	U	U	U	U	U	U
8	Delay (optional)	D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0

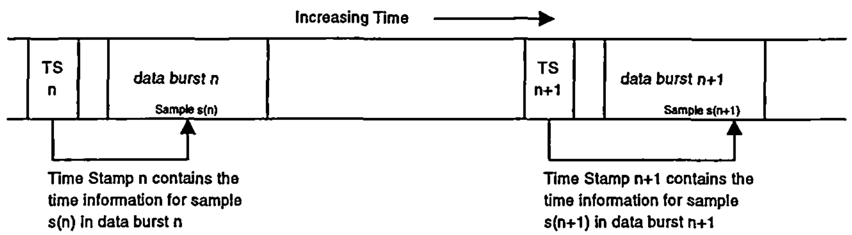

Figure 1 – Sample number indication in time stamp data bursts

For example, if an audio data burst represents 1536 linear PCM samples and the sample number is set to 1546, then the 12M time code information applies to sample 10 of the 1536 samples represented by the next data burst of the same stream number. In such cases, the same 12M time code information may be repeated more than once with different sample numbers in each case. In all cases, the sample number shall be accurate to within ± 0.5 ms of the ideal value.

In word 5, bit fields a3-a0 denote a 4-bit frame rate code. The frame rate code shall be encoded as shown in table 2. The drop-frame flag bit (bit 10 of the SMPTE 12M time code word) is provided in word 5 if the timing source is SMPTE 12M time code; its presence in bit 6 of word 3 is conditional on the value of the f1 flag bit. The presence of all 12M time code fields in words 0-5 shall be indicated by the flag bit f2 in word 5. If this bit is set to 0, 12M time code information is present. If this bit is set to 1, 12M time code information is not present and all bit fields in words 0-5 other than f2 are not defined. Reserved bits (R) in word 5 shall not be used and shall be set to 0.

Words 6 and 7 are optional words that may contain user private data; however, their presence is required when the optional delay field (word 8) is present.

Word 8 is optional. If present, it shall contain a delay indication denoted by bit fields D15-D0. The delay field is a signed integer that indicates the offset, in terms of AES3 frames, of the reference point of the corresponding data burst from the defined reference position for that data burst. A positive value shall indicate the reference point is advanced (in time) from the reference position, while a negative value shall indicate the reference point is delayed (in time) from the reference position. For instance, a delay setting of +2037 indicates that the reference point of the corresponding data burst is present in the AES3 bit stream 2037 frames ahead of the reference position for the burst. An exception is the value 0x8000 which shall be used to indicate no delay information is provided even though word 8 is present. This gives a range of ± 32767 AES3 frames, equivalent to approximately ± 682 ms with an AES3 reference sample rate of 48 kHz.

Table 2 – Frame rate code

Frame rate code a3	a2	a1	a0	Frame rate
0	0	0	0	Not indicated
0	0	0	1	24 ÷1001 (23.98)
0	0	1	0	24
0	0	1	1	25
0	1	0	0	30 ÷1001 (29.97)
0	1	0	1	30
0	1	1	0	50
0	1	1	1	60 ÷1001 (59.94)
1	0	0	0	60
–	–	–	–	Reserved
1	1	1	1	Reserved

The definitions of the reference point and reference position are relative to the data_type setting of the corresponding data burst. The definitions may vary between data types and may not exist for some data types, in which case the meaning of the delay field is undefined.

5 User defined (data_type = 30)

The user data type is provided for the transmission of arbitrary user data.

5.1 User data burst_preamble

The data_type_dependent field is undefined for the user data type. The data_stream_number shall be set to any valid number other than 0x7. The data_type shall be set to 30.

5.2 User data burst_payload

The user data burst may be of any length and data mode. The contents of the burst_payload are undefined.

Annex A (informative)
Bibliography

AES3-1992, Digital Audio Engineering – Serial Transmission Format for Two-Channel Linearly Represented Digital Audio Data

SMPTE 12M-1999, Television, Audio and Film — Time and Control Code

SMPTE 309M-1999, Television — Transmission of Date and Time Zone Information in Binary Groups of Time and Control Code

PROPOSED SMPTE STANDARD

SMPTE 340M

for Television —

Format for Non-PCM Audio and Data in AES3 — ATSC A/52 (AC-3) Data Type

1 Scope

This standard specifies data type specific format requirements for AC-3 data bursts carried within an AES3 interface according to SMPTE 337M.

2 Normative references

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

ATSC A/52, Digital Audio Compression (AC-3) Standard

SMPTE 337M, Television — Format for Non-PCM Audio and Data in an AES3 Serial Digital Audio Interface

SMPTE 338M, Television — Format for Non-PCM Audio and Data in AES3 — Data Types

SMPTE 339M, Television — Format for Non-PCM Audio and Data in AES3 — Generic Data Types

3 burst_preamble

The AC-3 data type is used to convey non-PCM audio streams encoded according to ATSC A/52 (ITU-R BS.1196).

3.1 data_type

The data_type shall have a value of 1.

3.2 data_type_dependent

The burst_preamble for an AC-3 data burst shall include a data_type_dependent field encoded as shown in table 1.

rep_rate_flag — The repetition rate flag shall be set to 0 if the AC3 data burst is placed in the AES3 interface such that the reference point of the data burst (as defined in 4.3) occurs at the AC-3 standard repetition rate (as defined in 4.4). The flag shall be set to 1 if the reference point does not occur at the AC-3 standard repetition rate. This flag is intended to be set to the same state for all data bursts of a given AC-3 data stream to indicate whether data bursts for the stream occur at the standard repetition.

Table 1 – Values of data_type_dependent field for AC-3 data type

data_type_dependent bit number	Meaning
0–2	Reserved, should be set to 000
3	Repetition rate flag (rep_rate_flag)
4	Not full service flag (not_full_svc)

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not_full_svc — This is a 1-bit field which indicates whether or not this audio service is a full service suitable for presentation, or whether this audio service is only a partial service which should be combined with another audio service before presentation. This bit shall be set to a 0 if this audio service is sufficiently complete to be presented to the listener without being combined with another audio service (for example, a visually impaired service which contains all elements of the program; music, effects, dialog, and the visual content descriptive narrative). This bit shall be set to a 1 if the service is not sufficiently complete to be presented without being combined with another audio service (e.g., a visually impaired service which only contains a narrative description of the visual program content and which needs to be combined with another audio service which contains music, effects, and dialog).

data_stream_number — The data_stream_number shall be set to any valid number other than 0x7.

4 burst_payload

4.1 AC-3 burst_payload

The AC-3 data stream consists of a sequence of AC-3 sync frames as defined in ATSC A/52. Each AC-3 sync frame represents 1536 encoded audio samples. AC-3 sync frame boundaries naturally occur at a frequency of exactly once every 1536 AES3 frames. The burst_payload of each AC-3 data burst shall contain one complete AC-3 sync frame. The length of the AC-3 burst_payload will depend on the encoded bit rate (which determines the AC-3 sync frame length).

4.2 AC-3 sampling frequency

When AC-3 data are conveyed by the AES3 interface, the AES3 frame frequency shall be equal to the sampling frequency of the AC-3 encoded audio. When more than one coded AC-3 bit stream is transmitted through the same interface, the audio sampling frequencies shall be identical. Bits 24-27 of the channel status word shall indicate the sampling frequency.

4.3 AC-3 reference point

The reference point of an AC-3 data burst is defined as bit 0 of the burst_payload; i.e., the first bit of the encoded AC-3 frame contained within the data burst.

4.4 AC-3 standard repetition rate

The AC-3 standard repetition rate is defined as 1536 AES3 frames. Data bursts for an AC-3 bit stream shall be considered as occurring at the standard repetition rate if the reference points for consecutive data bursts (of the same data stream number) are spaced 1536 AES frames apart.

4.5 AC-3 standard decode latency (professional)

The AC-3 reference decode latency for professional applications is defined as the time equivalent to 1536 AES3 frames at the current AES3 frame frequency (e.g., 32 ms at 48-kHz frame frequency). This means that a reference decoder would output the first PCM sample encoded in an AC-3 frame exactly 1536 sample times after the first bit of the frame is received by the decoder.

4.6 AC-3 reference position

AC-3 data bursts may occupy a reference position within an AES3 signal that is associated with a companion video signal. An AC-3 burst shall be defined as occupying the reference position if the reference point of the AC-3 burst is located in the AES3 interface such that the AC-3 stream can be decoded by a reference decoder and be presented at the intended time (e.g., in lip sync) with the associated video signal.

AC-3 bursts may have associated time stamp information in time stamp data bursts preceding the AC-3 data burst in the AES3 bit stream. These time stamps may include an optional delay value. For AC-3 data bursts that occupy a reference position, the delay shall be defined as 0. For AC-3 data bursts that do not occupy a reference position, the delay shall indicate the offset of the data burst (in AES3 frames) from the reference position.

4.7 AC-3 data burst synchronization

AC-3 data streams that include data bursts occurring at the standard repetition rate and reference position may be decoded by a decoder with minimal buffering and synchronization capabilities. However, in some cases, it may not be possible or desired to include data bursts at the standard repetition rate.

For instance, when multiple AC-3 streams are carried within the AES-3 interface, conflicts between data burst spacing may exist when one or more data bursts attempt to occupy the same reference positions. The combination of the rep_rate_flag and the delay field within time stamp bursts may be used to allow data bursts to be freely moved in the AES-3 interface when such conflicts exist. For AC-3 data streams with the rep_rate_flag set to 1, data burst spacing may vary dynamically between data bursts for that stream. The delay field in the time stamp burst is one method that may be used to indicate the offset for each data burst. SMPTE 12M time code information with an audio sample number is another method of specifying the absolute time of the audio encoded within the AC-3 frames.

The delay field may also be used to indicate data bursts that have been delayed due to equipment processing. For instance, an AC-3 bit stream produced by an encoder may contain data bursts occurring at the standard repetition rate, but still contain time stamp data bursts with constant delay value indicating the delay incurred by the encoding process. This delay value may be used, e.g., by MPEG-2 multiplexing equipment, to determine the audio delay of a companion AC-3 encoder.

Annex A (informative)
Bibliography

AES3-1992, Digital Audio Engineering — Serial Transmission Format for Two-Channel Linearly Represented Digital Audio Data

SMPTE 12M-1999, Television, Audio and Film — Time and Control Code

ITU-R BS.1196 (10/95), Audio Coding for Digital Terrestrial Television Broadcasting

PROPOSED SMPTE STANDARD

SMPTE 341M

for Television — Format for Non-PCM Audio and Data in AES3 — Captioning Data Type

1 Scope

This standard specifies data type specific format requirements for caption data bursts carried within an AES3 interface according to SMPTE 337M.

2 Normative references

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

SMPTE 337M, Television — Format for Non-PCM Audio and Data in an AES3 Serial Digital Audio Interface

SMPTE 338M, Television — Format for Non-PCM Audio and Data in AES3 — Data Types

EIA-708-A-1998, Digital Television (DTV) Closed Captioning

3 Preamble

The data_type shall be set to 29. Multiple formats of captioning data may be supported by this data type. The exact format of the captioning payload depends on the value of the data_type_dependent field. The data_stream_number shall be set to any number other than 0x7. The data_mode parameter shall be set to a value of 0 (16-bit mode).

3.1 Caption data_type_dependent field

The burst_preamble for a caption data burst shall set the data_type_dependent value to match the format of the caption data payload (see table 1).

Table 1 – Value of data_type_dependent value for captioning data

data_type_dependent value	Caption data payload format
0	Reserved
1	EIA-708-A caption distribution packet
2–31	Reserved

3.2 Caption data payload

Multiple formats of caption data may be carried. The following clause specifies payload formats that have been defined for carriage in this standard.

3.2.1 EIA-708-A caption distribution packet (data_type_dependent = 1)

When the data_type_dependent value = 1, the caption data payload shall be formatted in the form of a caption distribution packet as specified in EIA-708-A. There shall be exactly one caption distribution packet carried in a single payload.

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Annex A (informative)
Bibliography

AES3-1992, Digital Audio Engineering — Serial Transmission Format for Two-Channel Linearly Represented Digital Audio Data

PROPOSED SMPTE RECOMMENDED PRACTICE

RP 65
Revision of RP 65-1995

Motion-Picture Enlargement/Reduction Ratios

1 Scope

This practice specifies the enlargement/reduction ratios to be used in copying motion pictures from one film size to another while maintaining the aspect ratio and composition of the original film.

This practice also specifies the dimensions of the projectable image area on the resulting copy, and gives the dimensions for a matte which produces an opaque border on the final projection copy. Also specified are dimensions for camera viewfinder marks which can be used in composing the original photography for the copy format.

2 Ratios and dimensions

The enlargement/reduction ratios and dimensions shall be as given in the tables. All table dimensions are in inches. The centers of the original image, the projectable image area, the matte, the viewfinder marks, and the copy image shall all be coincident with the center of the copy format as given in the appropriate referenced standard. When a matte dimension is not given, no matte is required in that direction since the copied image fills the standard image dimension of the copy format.

Table 1 – Reduction from 35-mm to 16-mm

From 35-mm projection format to 16-mm projection format		1.37:1 1.37:1	1.66:1 1.66:1	1.85:1 1.85:1
Reduction ratio		0.461 ± 0.002	0.461 ± 0.002	0.461 ± 0.002
Projectable image on 16-mm copy	Width Height	0.380 ref 0.278 nom	0.380 ref 0.229 nom	0.380 ref 0.206 nom
Matte on 16-mm copy	Height		$0.229\ ^{+0.005}_{-0.0}$	$0.206\ ^{+0.005}_{-0.0}$
Viewfinder marks for original	Width Height	0.825 ref 0.602 ref	0.825 ref 0.497 ref	0.825 ref 0.446 ref

Table 2 – Enlargement from 16-mm to 35-mm

From 16-mm projection format to 35-mm projection format		1.33:1 1.33:1
Enlargement ratio		2.105 ± 0.002
Projectable image on 35-mm copy	Width Height	0.800 nom 0.602 ref
Matte on 35-mm copy	Width	$0.800\ ^{+0.010}_{-0.0}$
Viewfinder marks for original	Width Height	0.380 ref 0.286 ref

Table 3 – Enlargement from 35-mm to 70-mm

From 35-mm projection format to 70-mm projection format		1.85:1 1.85:1	2.39:1 anamorphic 2.39:1 flat
Enlargement ratio		1.951 ± 0.002	1.159 ± 0.002
Horizontal decompression			$2.00\ {}^{+0.0}_{-0.03}$
Projectable image on 70-mm copy	Width Height	1.610 nom 0.870 ref	1.912 ref 0.800 nom
Matte	Width	$1.610\ {}^{+0.010}_{-0.0}$	
	Height		$0.800\ {}^{+0.010}_{-0.0}$
Viewfinder marks for original	Width Height	0.825 ref 0.446 ref	0.825 ref 0.690 ref

Table 4 – Reduction from 35-mm anamorphic to 16-mm anamorphic

From 35-mm projection format to 16-mm projection format		Anamorphic 2.39:1 anamorphic
Reduction ratio		0.414 ± 0.002[1)]
Projectable image on 16-mm copy	Width Height	0.342 nom 0.286 ref
Matte	Width	$0.342\ {}^{+0.005}_{-0.0}$
Viewfinder marks for original	Width Height	0.825 ref 0.690 ref

[1)] Some reductions from 35-mm anamorphic to 16-mm anamorphic use the 35-mm to 16-mm reduction ratio given in table 1 (i.e., 0.461) and the 16-mm copy is projected using the standard 16-mm projectable area of 0.380 in by 0.286 in, thus projecting only 0.620 in of the vertical information rather than the usual 0.690 in projected in 35-mm. Such practice is, therefore, deprecated.

Annex A (informative)
Additional data

A.1 Choice of enlargement/reduction ratios

Enlargement/reduction ratios have been calculated, insofar as possible, to fit the entire projectable image area on the original film to the standard projectable image area of the copy format. By so doing, all of the image originally meant to be seen by the audience will be seen when projecting the enlarged or reduced copy. The aspect ratio and the size of the exposed image outside the projectable image area may differ in the original format and the copy format, resulting in images which do not completely fill the standard image areas of the copy format. These unfilled areas of the copy may occur at the image top and bottom, the sides, or both.

A.2 Projection aperture

Because the unfilled areas described in A.1 can occur, it is recommended that the projector used for showing the copy be fitted with a suitable projection aperture. The dimensions given for the projectable image area can be used as a reference to design a projector aperture suitable to the image size on the final projection copy.

A.3 Matte

It is recognized that the enlarged or reduced copy may not be projected with a projection aperture designed specifically for the image size on the copy. Additionally, clear film outside the projected area may cause flare in projection even with a proper aperture. Therefore, a matte of the dimensions given in the tables may be required at some point in the duplication process so as to produce an essentially opaque border surrounding the image area on the final projection copy. The matte is intended to be slightly larger than the appropriate projectable image area so that it will not be seen on the screen when the preferred projector aperture is used, but if the copy is projected without the preferred projector aperture, essentially the same projected image area will be seen.

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Annex B (informative)
Bibliography

ANSI/SMPTE 152-1994, Motion-Picture Film (70-mm) — Projectable Image Area

ANSI/SMPTE 195-1993, Motion-Picture Film (35-mm) — Motion-Picture Prints — Projectable Image Area

ANSI/SMPTE 215-1995, Motion-Picture Film (65-mm) — Camera Aperture Image

ANSI/SMPTE 233-1998, Motion-Picture Film (16-mm) — Projectable Image Area and Projector Usage

SMPTE 7-1999, Motion-Picture Film (16-mm) — Camera Aperture Image and Usage

SMPTE 59-1998, Motion-Picture Film (35-mm) — Camera Aperture Images and Usage