
Protocol Analysis at Digital Interfaces using the Audio Analyzers UPL and UPD

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Subject to change

Products:

Audio Analyzer UPL

Audio Analyzer UPD



ROHDE & SCHWARZ

1. Conclusion

For the transmission of digital audio data, different data formats are in use. The AES/EBU format, which is used in professional applications, as well as the S/P DIF format used in consumer electronics, transmit the audio data in time multiplex for both channels. Additional bits in the data stream cover complementary information, respectively are used for error detection. The generation and the analysis of those protocol data using the audio analyzers UPD or UPL is the topic of this application note. An extract of the AES3 standard informs about the basic definitions.

2. Protocol Analysis and Generation

In digital audio engineering, various data formats have been standardized for signal transmission. The professional audio standard is the so called AES/EBU interface which allows stereo signals with a resolution of up to 24 bits to be transmitted. The S/P DIF (Sony/Philips digital interface format) used in consumer electronics industry has a similar format, but uses different connector systems and electrical parameters.

The audio signal is transmitted in time multiplex for both stereo channels together with all the auxiliary information required for synchronization etc. The data format also provides capacity for further information and error checks. The basic structure of the standardized data format (AES3, see enclosed extract from the Standard) is briefly described in the following:

The digitized audio signal is transmitted at a fixed sampling rate, the standard sampling rates being 32 kHz, 44.1 kHz and 48 kHz depending on the desired quality of the transmitted audio frequency range. This sampling rate is used to transmit the left and the right audio channel in multiplex. A frame consists of a subframe for the right channel and a subframe for the left channel, each subframe having a length of 32 bits. Up to 24 bits are used for the audio signal proper, 4 bits form the preamble and are used for synchronization, 1 bit identifies the validity of the audio signal and 1 bit is used for the parity check of the transmitted word. There are two free bits in each subframe, the channel status bit and the user bit. These bits can be used to transmit auxiliary information, which is largely standardized for the channel status bit and freely selectable for the user bit.

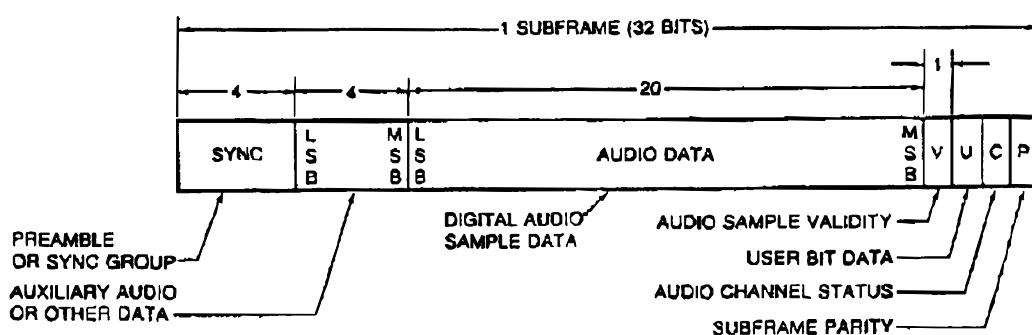


Fig. 1: Subframe format

These auxiliary information bits are transmitted with each audio sample, separately for the left and the right channel, the bit rate corresponding to the sampling rate of the audio signals, for instance 48 kbit/s at 48 kHz sampling rate.

For the channel status information, 192 successive channel status bits are combined and displayed in 24 words of 8 bits. The precise definition of the individual channel status words is given in the enclosed extract from the AES3 Standard. The total channel status information thus has a repetition rate of sampling rate divided by 192, ie with a sampling rate of 48 kHz the repetition rate is 250 Hz. The channel status information for the left and the right channel is always available simultaneously; such information can but need not be identical.

The individual bytes (words) of the channel status information have the following meaning (extract from AES3):

Byte	Bit 0	1	2	3	4	5	6	7
0	Use of channel status channel	Audio / nonaudio use	Audio signal emphasis			Locking of source sample frequency	Sampling frequency	
1	Channel mode				User bit management			
2	Use of auxiliary sample bits			Source word length and source encoding history		Reserved		
3	Future multichannel function description							
4	Digital audio reference signal		Reserved					
5	Reserved							
6	Alphanumeric channel origin data							
7								
8								
9								
10	Alphanumeric channel destination data							
11								
12								
13								
14	Local sample address code (32-bit binary)							
15								
16								
17								
18	Time-of-day sample address code (32-bit binary)							
19								
20								
21								
22	Reliability flags							
23	Cyclic redundancy check character							

Fig. 2: Channel status data format

In addition to the audio signal proper, the Audio Analyzers UPL and UPD is also able to generate and analyze the auxiliary data. Moreover, erroneous frames can be generated to test the compatibility of connected components (this feature is only available with UPD). It is possible to generate parity errors in a random sequence as well as block errors or checksum errors for cyclic redundancy checks (CRC).

Since the generation of special bit sequences with the assignment defined by the Standard is rather elaborate and ambiguous at the bit level, the UPL/UPD also provides the possibility of generating and analyzing these bit sequences in coded form already. Coding and decoding is made with the aid of so-called panel files, ie at the request of the user the control panel of the UPL/UPD generator and analyzer will be enhanced by the channel data display. These panel files are provided in the UPL/UPD for standard formats; the user can define and configure a control panel for the generation of channel status data in arbitrary formats. All data can of course also be displayed directly in binary or hexadecimal form.

The generation and the analysis of channel status data as well as the generation of the associated control panels is described in the following:

3. Generation of Channel Status Data

AES/EBU Protocol Definition

Upon selection of the standardized digital outputs, an additional PROTOCOL section is displayed on the generator panel. This section contains the commands for defining the protocol information (channel status data, user data, validity, parity).

Valid Chan	<p>Sets the validity identification within the AES/EBU data stream.</p> <p>NONE: no validity bit set at all</p> <p>1: validity bit set in channel 1 only (= left, A)</p> <p>2: validity bit set in channel 2 only (= right, B)</p> <p>1&2: validity bit set in both channels</p>
Parity (only with UPD)	<p>Defines whether the protocol is to be generated with parity errors.</p> <p>TRUE: all samples are generated with correct parity.</p> <p>WITH ERR: setting the error rate with the commands No. Trues, No. Falses and Offset.</p> <p>No. True: defines the number of samples with correct parity bit.</p> <p>Entry of numbers from 100 to 1,000,000</p> <p>Notes on parity error definition:</p> <p>First a number of samples with correct parity (Offset) is output. Then follows a cyclical sequence consisting of a number of samples with false parity bit (No. False) followed by a number of samples with correct parity bit (No. True). "True" and "False" are repeated alternately.</p> <p>The numbers refer to the sum of L and R samples.</p> <p>Applications:</p> <ul style="list-style-type: none"> • If the sum of No. True and No. False is 384, the positions of the parity errors within a block can be determined with the aid of Offset. • A very large Offset can be used to allow for proper settling of the test hardware. <p>No. False: defines the number of samples with false parity bit.</p> <p>Entry of numbers from 0 to 1,000</p> <p>With 0 setting, only correct parity bits are output.</p> <p>Offset: defines the number of samples with correct parity bit until the first parity error occurs.</p> <p>Entry of numbers from 0 to 1,000,000</p> <p>For more details, see under No. True.</p>
Block Err (only with UPD)	<p>Defines the interval (in blocks), in which errors in the block start preamble sequence are generated.</p> <p>Entry of numbers from 0 to 100,000</p> <p>0: no error, all block start preambles are correct.</p> <p>1 to 100,000: after the defined number of samples, a block start preamble is replaced by a normal "left" preamble (no block start).</p>
Sequ. Err (only with UPD)	<p>Defines the interval (in blocks) in which errors in the preamble sequence are generated.</p> <p>Entry of numbers from 0 to 10,000</p> <p>0: no error, all preambles are correct.</p> <p>1 to 10,000: a block with sequence error is generated at the defined block intervals. The 48th left preamble within this block is replaced by a right one. The sequence " L R R R L" is generated instead of the sequence " L R L R L". This inverted sample carries the status information of the associated left channel, only the preamble being inverted.</p>
Ch Stat L	<p>Defines the mode of generating the channel status data LEFT.</p>

ZERO: all channel status data bits are 0. (The operating mode (AES3, CRC, RAW) is defined by the command Ch Stat R).

FILE + AES3: UPL/UPD generates local time code and CRC, the other channel status data are defined by the following file.

PANEL + AES3: UPL/UPD generates local time code and CRC, the other channel data are set with the aid of the panel defined by the "Panelfile".

FILE+CRC:

PANEL + CRC: same as FILE + AES3 or PANEL + AES3, however local time code is not generated by UPL/UPD but set as default value.

FILE:

PANEL: same as FILE + AES3 or PANEL + AES3, but neither local time code nor CRC are generated by UPL/UPD (RAW mode).

The setting that can be made under this menu item affects Ch Stat R. Any settings under Ch Stat R which are not compatible with this setting will cause an error message and rejection of the setting. The PANEL can be assigned only once and the operating mode (AES3, CRC, RAW) must be the same for both channels.

BINARY: a window appears, where the single channel status bits can be input in binary mode. The SELECT key changes the bits from 0 to 1 and vice versa. Binary function can be used only for Ch Stat L or for Ch Stat R.

Filename

Definition of a file containing channel status data for LEFT.

Preset file type: *.pgc

Data are defined by one line. Keywords in the file define the bits and channel (left, right or both channels) to which data are assigned.

With this command, only those definitions containing data for the left channel or for both channels will be considered. The same file can thus be used for both channels.

Syntax:

Channel:

AES _ CHAN _ STAT or AES _ CHAN _ STAT _ BOTH is used to indicate the beginning of data for both channels. AES _ CHAN _ STAT _ RIGHT and AES _ CHAN _ STAT _ LEFT indicates the beginning of data for one channel.

These keywords may be used in any sequence and as often as desired.

Data:

Values are entered into the file with the following line:

Keyword BIT, followed by a range and the respective value.

Example: BIT: 12-15,
(BITs 12-15 are assigned "0001")

If the generator status data are to be taken from the analyzer, this can be defined by entering the keyword TRACK instead of a value.

Example: BIT: 12-15, TRACK

The time can be entered into the UPL/UPD by means of the keyword TIME instead of a value. The time is multiplied by the currently selected sampling rate to obtain the number of samples made since midnight.

Example: BIT: 112-143, TIME

Typical file: R&S_AES3.PGC

Ch Stat R

Defines the mode of generating the channel status data RIGHT. The setting that can be made depends on Ch Stat L. The PANEL can be assigned only once and the operating mode (AES3, CRC, RAW) must be the same for both channels.

For further details of the individual commands see Ch Stat L.

ZERO: All channel status data bits are 0. (Operating mode (AES, CRC, RAW) is defined by Ch. Stat L. Should Left also be set to ZERO, the operating mode is RAW).

EQUAL L: The two channels are equal, all definitions made for the left channel are copied to the right. The operating mode is defined by Ch Stat. L.

FILE+AES3: This selection is only possible if ZERO, FILE+AES3 or PANEL I AES3 has been selected for Ch Stat L.

PANEL + AES3: This selection is only possible if ZERO or FILE + AES3 has been selected for Ch Stat L.

FILE + CRC: This selection is only possible if ZERO, FILE + CRC or PANEL + CRC has been selected for Ch Stat L.

PANEL + CRC: This selection is only possible if ZERO or FILE + CRC has been selected for Ch Stat L.

FILE: This selection is only possible if ZERO, PANEL or FILE has been selected for Ch Stat L.

PANEL: This selection is only possible if ZERO or FILE has been selected for Ch Stat L.

BINARY: the selection is only possible if BINARY has not been selected for Ch Stat L

Filename Definition of a file containing channel status data for RIGHT.
Preset file type: *.pgc
The data are defined by one line. Keywords in the file define the bits and channel (left, right or both channels) to which the data are assigned (see also instructions for the left channel).
With this command, only those definitions containing data for the right channel or for both channels will be considered. The same file can thus be used for both channels.
Format: see under Ch Stat L

CRC Error (only with UPD) Defines the interval (in blocks) for generating errors in the CRC of the status data.
Entry of numbers from 0 to 100,000
0: no error, all CRCs are correct.
1 to 100,000: after the specified number of blocks a CRC error is simultaneously generated in both channels (CRC error: the correct CRC is replaced by the inverted bit pattern).

This command is only possible if the operating mode of the channel status data is CRC or AES3.

User Mode Defines the mode of generating user data.
ZERO: all user bits are initialized to be 0.
FILE DEF: user bits are output in the subsequent file according to definitions.
Note:
Changing the user bits briefly stops the generator and the analyzer.

Filename Definition of a file containing user data.
Preset file type: *.pgu

The file contains user data both for the left and for the right channel. The keyword AES _ USER _ DATA _ LEFT indicates the beginning of data for the left channel, while AES _ USER _ DATA _ RIGHT refers to the right channel.

In the following lines, values have to be entered as hexadecimal numbers without any further designation. Each line contains 32 bits.

The read-in user bits are cyclically repeated by the UPL/UPD, the cycle being equally long for the two channels. If a smaller number of bits is read in on one channel, zeros will be inserted. 6 bits (or a multiple thereof) yield corrected user data from the start of the block.
The maximum permissible number of user bits is 4096 word = 16384 bytes = 131072 bits per channel.

Example: AES_USER_DATA_RIGHT
0 x 55504420 # = ' U PD '

Panelfile Specification of a file containing the definitions of the user-programmable panel. This menu item is only displayed if a panel has been specified for Ch Stat L or Ch Stat R.

Preset file type: *.PPC

This file can be used to define an input panel tailored to the specific requirements. To this end, the respective commands, the permissible values and the target of the setting have to be entered into the file. 4 types of commands are permissible:

• **Selection commands**

A selection command is defined by entering the keyword SELECTION or EXTSELECT followed by parameters.

Example:

- SELECTION "Use", BIT:0, 0 = "CONS", 1 = "PROF"
(Bit 0 of the channel status data can be switched between ANLR TRACK, CONS and PROF with the aid of the menu line "Use".)
- SELECTION "Usermod", BIT:12-15, 0 = "notind", 3 = "USER"
(The four bits 12 to 15 of the channel status data can be switched between ANLR TRACK, not ind and USER with the aid of the menu line "Usermod".)

Note:

- Up to 12 normal selection commands and 3 extended commands are permissible.
- ANLR TRACK is automatically selected by the UPL/UPD and means copying of the corresponding bits of the analyzer.
- The BIT interval must not be greater than 32.
- Up to 8 selections per command (= SELECTION) are permissible. The selections can be assigned any value within the specified bit range.
- In the case of overlapping bit ranges, the previously defined bits will be overwritten by the latest setting.
- Up to 24 selections can be made with EXTSELECT.

• **Text commands**

A text command is defined by entering the keyword TEXT followed by parameters.

Example:

- TEXT " Origin", BIT:48-79
(The 32 bits 48 to 79 of the channel status data are filled by the text characters to be entered.)

Note:

- Up to four commands can be selected.
- The BIT must not be greater than 32, ie up to 4 text characters (ASCII) per command are permissible.
- If no text is input, the UPL/UPD will copy the corresponding contents from the analyzer.

• **Value commands (normal)**

A value command is defined by entering the keyword VALUE followed by parameters. Status bits may be specified by numbers. These numbers can be multiplied by a multiplier specified in the file.

Example:

- VALUE " Abs.Hour", BIT:144-175, MULT:SET_RATE M U LT: 3600
(The value defined here is entered into the 32 bits 144 to 175 of the channel status data, the number after MULT (3600 * and SET _ RATE = selected sampling rate) being multiplied by the value prior to the entry.)

Note:

- Up to 12 value commands are permissible.
- The BIT interval must not be greater than 32, the entry being limited to 31 bits.
- Upon panel input of -1, the corresponding bit range of the analyzer will be copied.
- The keywords MULT are optional; the keyword SET _ RATE may also be used as a multiplier.

• **Value commands (additive)**

Entry of the keyword ADDVALUE followed by parameters defines a value command which is added to already existing values.

Example:

- ADDVALUE " Abs.Min", BIT:144-175, MULT:2880000
(The value entered here is added to the 32 bits 144 to 175 of the channel status data, the number after MULT (288000 = 60 * 48000) being multiplied by the value prior to the addition.)

Note: - See under normal value commands.

Example: R&S _AES3.PPC for panel according to AES3 format
R&S _CONS.PPC for panel according to consumer format

The commands are displayed in the panel on the UPL/UPD in the following sequence:

1. SELECTION/EXTSELECT
2. VALUE / ADD VALUE
3. TEXT

In the file, the sequence may be as desired.

4. Protocol Analysis

Screen layout:

The screen (consisting of 16 lines of 50 characters each) is divided into two sections:

TOP: Invariable protocol elements:

- Validity:** Indicates the position of the validity bit in the respective channel.
- Parityerrors:** With the UPD, it indicates the sum of the previous parity errors; this number is reset to zero upon resetting the analyzer or pressing the start key.
With the UPL, it indicates if parity errors have occurred or not
- Change:** Indicates any changes in the channel status data.
NO: no change
LTC: changes only in the fields 'local time code' (bits 112 to 143) and CRC (bits 184 to 191).
YES: changes in any other bit position.
- Other:** Indicates whether the other channel is identical or different.
- Errors:** Indicates the errors occurred:
with UPL:
LOCK: PLL not locked
CONFIDEN: not decodeable (eye open less than half a bit)
CODING: Coding error detected
NONE: no error
with UPD:
UNEXP BB: unexpected preamble for block start (too early)
SQ BLOCK: no preamble (gap) for block start
NOT LOCK: no preamble for block start PREAMBLE: invalid preamble
SQ L/R: error in channel sequence (L/R)
RATE ERR: measured clock rate and set rate differ by more than 200 ppm from each other.
NONE: no error

BOTTOM: These 13 lines can be adapted to the currently used protocol with the aid of protocol control file.

Using UPD, the error counters Parityerror, CRC _ left and CRC _ right can be reset by pressing the start key or resetting the analyzer. Changing status bits are displayed in red. Bits that have not changed since the last output are displayed in green. Comments (PRINT command) are displayed in yellow.

Example:

Protocol Analysis: Channel Status Left

Text entered by UPD from here; user defined	Validity:	R:0	L:1	Parityerrors:-	
	Change:	YES	Other: EQUAL	Errors: NONE	
	Byte:	=====	AES3	=====	
	0:	Format:	prof	Mode:	audio
		Emph:	J.17	Source:	locked
		Rate:	48kHz		
	1:	Chanmod:	stereo	Usermod:	AES18
	2:	Auxmod:	20 no	Length:	23 R:0
	3-5:	Vector:	12	Grade:	2 R:0
	6-13:	Origin:	UPD_	Destin:	R&S_
	14-21:	Local:	01234567	Time:	12:45:00
	22:	Relia:	0-5:1 6-13:0	14-17:0	18-21:0
	23:	CRC L:		CRC R:	
	Measured sample rate: 48000.2				

This example is generated using the protocol file R&S _aes3.pac.

Source Selects the data to be displayed.
CHAN STAT L Channel status data left are displayed.
CHAN STAT R Channel status data right are displayed.
USER L User data left are displayed.
USER R User data right are displayed.

With CHAN STAT L or R, the following line is displayed:

Proto File Proto File selects the interpretation file for channel status data.
Selection from: file names, preset file type: *.pac

Description:
The protocol information is displayed in two sections; the upper two lines are displayed in a fixed format and contain information on validity, parity, other errors and differences between the two channels. The other lines are formatted according to the information contained in the file, the individual bits being assigned any output text. This file can be adapted by the user to suit any protocol.

Example: see R&S_AES3.PAC

File format: same as for USER DAT, see below

With USER DAT, the following lines are displayed:

FORMAT Format selects the interpretation mode.
Selection from: BINARY, HEX, ASCII, AES18
BINARY User data are displayed as a 01010 sequence. Display is block aligned.
HEX User data are displayed as a hexadecimal numerical sequence. Display is block aligned.
ASCII User data are displayed as text. Display is block aligned.
FILE Interpretation, block aligned

With FILE, the following line is displayed:

Proto File Proto File selects the interpretation file for user data.
Selection from: file names, preset file type: * PAU

File format:

The protocol file has the following structure:
One operation per line, followed by parameters which are separated by commas. The parameters allow the display of any bits (1 to 32) from the selected data, the output position being freely selectable. A line must not contain more than 255 characters.

Operations:

PRINT: test output (independent of data)

Example:

PRINT 26, 5, "Usermod:"
(the text "Usermod:" is entered in column 26, line 5)

VALUE: output of value, either as a HEX number (default) or as text provided that there is an assignment.
Example:

VALUE 17, 3, BIT:2-4, 0 = "not ind", 4 = "no empf", 6= "50/15 "
(in column 17, line 3 the contents of the three bits (2,3,4) is displayed as a hex number (or as text if the value is 0, 4 or 6))

VALUE 17,11, CRC_ L_ ERR
(in column 17, line 11 the contents of the CRC left error counter is displayed)

Note for output length:

If there are text assignments, the longest text determines the output length; in the absence of text assignments, the output length is determined by the number of bits to be displayed.

The following information can be used as a data source:

- 'BIT' followed by ':': statement of bits:
 - a number from 0 to 191: single bit
 - an interval (eg 4-9): combined sequence of bits, max. 32 bits permissible.
- 'CRC _ L _ ERR': internal error counter of CRC _ error left.
- 'CRC _ R _ ERR': internal error counter of CRC_ error right.
- 'MEASURED _ RATE': measured sampling rate (displayed as a 5.1-digit floating number (eg '48001.2')

BINARY: same as VALUE, but default output as bit pattern.

TEXT: text output (in ASCII) with number of letters being determined by the number of bits selected; for each TEXT operation, only 32 bits can be displayed; for longer text outputs several consecutive text lines have to be used.

Example:

TEXT 17,8, BIT:48-79
(in column 17, line 8 the contents of the 32 is shown as text)
Non-printable characters are shown as '?' or as '.' in the case of '0'.

TIME: time output (in the form of 12:45:56) of the selected number divided by the selected rate. Number/rate is interpreted as seconds since midnight.

Example:

TIME35,9, BIT:144-175, RATE:48000.0 TIME35,9, BIT: 144-175, RATE:SET _ RATE
(in column 35, line 9 the contents of the 32 bits is shown as time (eg 12:34:45)

RATE: should correspond to the sampling rate, may be stated as a floating number. The following specifications are possible as an alternative:

MEASURED_RATE: measured clock rate
SET_RATE: clock rate set on panel

5. Extract from AES3 Standard

AES Recommended practice for digital audio engineering-- Serial transmission format for two-channel linearly represented digital audio data

1 Scope

This document specifies a recommended interface for the serial digital transmission of two channels of periodically sampled and linearly represented digital audio data from one transmitter to one receiver.

It is expected that the format will be used to convey audio data that have been sampled at any of the sampling **frequencies recognized** by the *AES5 Recommended Practice for Professional Digital Audio Applications Employing Pulse-Code Modulation--Preferred Sampling Frequencies*.

The format is intended for use with shielded twisted pair cable of conventional design over distances of up to 100 m without transmission equalization or any special equalization at the receiver. Longer cable lengths may be used, but with a rapidly increasing requirement for care in cable selection and possible receiver equalization or the use of active repeaters, or both.

The document does not cover connection to any common carrier equipment, nor does it specifically address any questions about the synchronizing of large systems, although by its nature the format permits easy synchronization of receiving devices to the transmitting devices.

Specific synchronization issues are covered in AES 11.

In this interface specification, mention is made of an interface for consumer use. The two interfaces are not identical.

An engineering guideline document to accompany this interface specification is in preparation.

2 Interface format

2.1 Terminology

2.1.1 sampling frequency: Frequency of the samples representing an audio signal. When more than one audio signal is transmitted through the same interface, the sampling frequencies are identical.

2.1.2 audio sample word: Amplitude of a digital audio sample. Representation is linear in 2's complement binary form. Positive numbers correspond to positive analog voltages at the input of the analog-to-digital converter (ADC). The number of bits per word can be specified from 16 to 24 in two coding ranges (less than or equal to 20 bits and less than or equal to 24 bits).

2.1.3 auxiliary sample bits: 4 least significant bits (LSBs) which can be assigned as auxiliary sample bits and used for auxiliary information when the number of audio sample bits is less than or equal to 20.

2.1.4 validity bit: Bit indicating whether the audio sample bits in the subframe (time slots 4 to 27 or 8 to 27, depending on the audio word length as described in 2.2.1) are suitable for conversion to an analog audio signal.

2.1.5 channel status: Bits carrying, in a fixed format derived from the block (see 2.1.11), information associated with each audio channel which is decodable by any interface user.

2.1.6 user data: Channel provided to carry any other information.

2.1.7 parity bit: Bit provided to permit the detection of an odd number of errors resulting from malfunctions in the interface.

2.1.8 preambles: Specific patterns used for synchronization. There are three different preambles (see 2.4).

2.1.9 subframe: Fixed structure used to carry the information described in 2.2.2).

2.1.10 frame: Sequence of two successive and associated subframes.

2.1.11 block: Group of 192 consecutive frames. The start of a block is designated by a special subframe preamble (see 2.4).

2.1.12 channel coding: Coding describing the method by which the binary digits are represented for transmission through the interface.

2.2 Structure of format

2.2.1 Subframe format Each subframe is divided into 32 time slots, numbered from 0 to 31 (see figure 1).

Time slots 0 to 3 (preambles) carry one of the three permitted preambles (see 2.2.2 and 2.4; also see figure 2) .

Time slots 4 to 27 (audio sample word) carry the audio sample word in linear 2's complement representation. The most significant bit (MSB) is carried by time slot 27.

When a 24-bit coding range is used, the LSB is in time slot 4 [see figure 1(a)].

When a 20-bit coding range is sufficient, time slots 8 to 27 carry the audio sample word with the LSB in time slot 8. Time slots 4 to 7 may be used for other applications. Under these circumstances, the bits in time slots 4 to 7 are designated auxiliary sample bits [see figure 1(b)].

If the source provides fewer bits than the interface allows (either 20 or 24), the unused LSBs are set to logic 0.

Time slot 28 (validity bit) carries the validity bit associated with the audio sample word (see 2.5).

Time slot 29 (user data bit) carries 1 bit of the user data channel associated with the audio channel transmitted in the same subframe (see Section 3).

Time slot 30 (channel status bit) carries 1 bit of the channel status information associated with the audio channel transmitted in the same subframe (see Section 4) .

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

NOTE The preambles have even parity as an explicit property.

2.2.2 Frame format

A frame is uniquely composed of two subframes (see figure 2). The rate of transmission of frames corresponds exactly to the source sampling frequency.

The first subframe normally starts with preamble "X." However, the preamble changes to preamble "Z" once every 192 frames. This defines the block structure used to organize the channel status information. The second subframe always starts with preamble "Y."

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

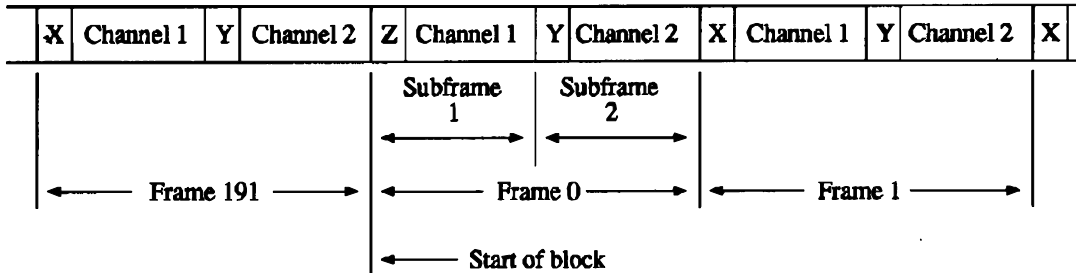


Figure 2. Frame format

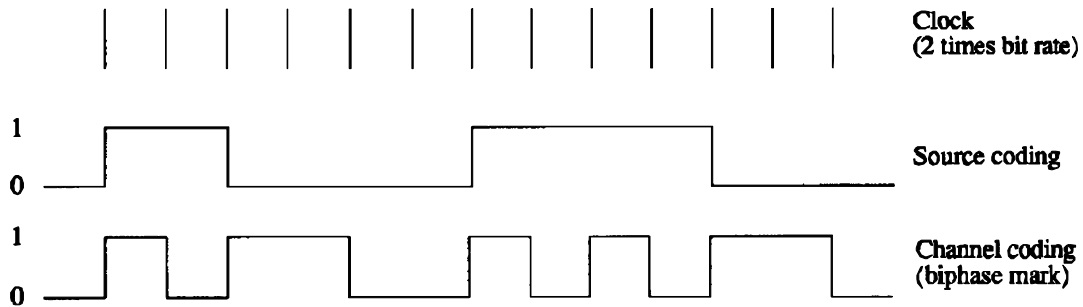


Figure 3. Channel coding

Like biphasic code, these preambles are dc free and provide clock recovery. They differ in at least two states from any valid biphasic sequence.

Figure 4 represents preamble "X."

NOTE: Owing to the even-parity bit in time slot 31, all preambles will start with a transition in the same direction (see 2.2.1). Thus only one of these sets of preambles will, in practice, be transmitted through the interface. However, it is necessary for either set to be decodable because a polarity reversal might occur in the connection.

2.5 Validity bit

The validity bit is logic 0 if the audio sample word is suitable for conversion to an analog audio signal, and it is logic 1 if it is not.

There is no default state for the validity bit.

3 User data format

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

Possible formats for the user data channel are indicated by the channel status byte 1, bits 4-7.

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

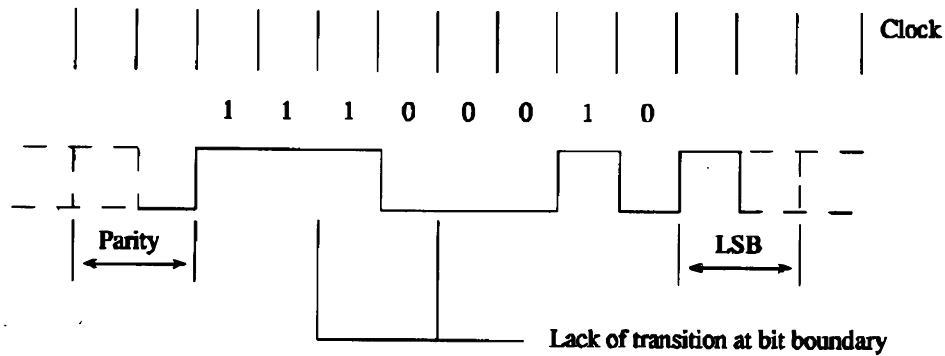


Figure 4 Preamble X (11100010).

4 Channel status format

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. Channel status information is organized in 192-bit blocks, subdivided into 24 bytes (see figure 5). The first bit of each block is carried in the frame with preamble "Z. "

Byte	Bit 0	1	2	3	4	5	6	7
0	Use of channel status channel	Audio / nonaudio use	Audio signal emphasis			Locking of source sample frequency	Sampling frequency	
1	Channel mode				User bit management			
2	Use of auxiliary sample bits			Source word length and source encoding history		Reserved		
3	Future multichannel function description							
4	Digital audio reference signal	Reserved						
5	Reserved							
6	Alphanumeric channel origin data							
7								
8								
9								
10	Alphanumeric channel destination data							
11								
12								
13								
14	Local sample address code (32-bit binary)							
15								
16								
17								
18	Time-of-day sample address code (32-bit binary)							
19								
20								
21								
22	Reliability flags							
23	Cyclic redundancy check character							

Figure 5. Channel status data format.

The specific organization follows, wherein the suffix 0 designates the first byte or bit.

Byte 0

bit 0	0 1	Consumer use of channel status block (see note). Professional use of channel status block.
bit 1	0 1	Normal audio mode. Nonaudio mode.
bits 2-4 bit state	234 000 100 110 111	Encoded audio signal emphasis. Emphasis not indicated. Receiver defaults to no emphasis with manual override enabled. No emphasis. Receiver manual override is disabled 50/15- μ s emphasis. Receiver manual override is disabled International Telegraph and Telephone Consultative Committee (CCITT) J. 17 emphasis (with 6.5 dB insertion loss at 800 Hz). Receiver manual override is disabled. All other states of bits 2-4 are reserved and are not to be used until further defined .
bit 5	1 0	Source sampling frequency unlocked. Default and source sampling frequency locked.
bits 6-7 bit state	67 00 01 10 11	Encoded sampling frequency. Sampling frequency not indicated. Receiver default to 48 kHz and manual override or auto set is enabled 48-kHz sampling frequency. Manual override or auto set is disabled. 44. 1-kHz sampling frequency. Manual override or auto set is disabled. 32-kHz sampling frequency. Manual override or auto set is disabled.

NOTE: The significance of byte 0, bit 0 is such that a transmission from an interface conforming to IEC 958 "consumer use" can be identified, and a receiver conforming only to IEC 958 "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) .

Byte 1

bits 0-3 bit state	0123 0000 0001 0010 0011 0100 0101 0110 1111	Encoded channel mode. Mode not indicated. Receiver default to two channel mode. Manual override is enabled. Two-channel mode. Manual override is disabled . Single-channel mode (monophonic). Manual override is disabled. Primary/secondary mode (subframe I is primary). Manual override is disabled . Stereophonic mode (channel 1 is left channel). Manual override is disabled . Reserved for user-defined applications. Reserved for user-defined applications. Vector to byte 3 . Reserved for future applications. All other states of bits 0- 3 are reserved and are not to be used until further defined .
bits 4-7 bit state	4567 0000 0001 0010 0011	Encoded user bits management. Default, no user information is indicated. 192-bit block structure. Preamble "Z" indicates the start of block. Reserved for the AES18 standard User defined All other states of bits 4 7 are reserved to be used until further defined

Byte 2

bits 0-2 bits state	<p>Encoded use of auxiliary sample bits</p> <p>012 000 Maximum audio sample word length is 20 bits (default). Use of auxiliary sample bits is not defined</p> <p>001 Maximum audio sample word length is 24 bits. Auxiliary sample bits are used for main audio sample data.</p> <p>010 Maximum audio sample word length is 20 bits. Auxiliary sample bits in this channel are used to carry a single coordination signal (see note 1)</p> <p>011 Reserved for user-defined applications .</p> <p>All other states of bits 0-2 are reserved to be used until further defined</p> <p>NOTE 1-The signal coding used for the coordination channel is described in Annex A</p>
bits 3-5 bit state	<p>Encoded audio sample word length of transmitted signal (see notes 2, 3, and 4).</p> <p>345 Audio sample word length if maximum length is 24 Audio sample word length if maximum length is 20 bits as indicated by bits 0-2 above.</p> <p>000 Word length not indicated (default) Word length not indicated (default)</p> <p>001 23 bits 19 bits</p> <p>010 22 bits 18 bits</p> <p>011 21 bits 17 bits</p> <p>100 20 bits 16 bits</p> <p>101 24 bits 20 bits</p> <p>All other states of bits 3-5 are reserved to be used until further defined</p>
bits 6-7	Reserved and are set to logic 0 until further defined
	<p>NOTE 2 The default state of bits 3-5 indicates that the number of active bits within the 20- 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 auto set.</p> <p>NOTE 3 The nondefault state of bits 3-5 indicates the number of bits within the 20- 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. The receiver should disable manual override and auto set for these bit states.</p> <p>NOTE 4 Irrespective of the audio sample word length as indicated by any of the states of bits 3-5, the MSB is in time slot 27 of the transmitted as specified in 2.2.1.</p>

Byte 3

bits 0-7 bit state	<p>Vectored target byte from byte 1</p> <p>Reserved for future use as multichannel function description. These bits are set to logic 0 at present</p>
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Byte 4

bits 0-1 bits state	<p>Digital audio reference signal (per AES11)</p> <p>01</p> <p>00 Not a reference signal (default)</p> <p>01 Grade 1 reference signal</p> <p>10 Grade 2 reference signal</p> <p>11 Reserved and not used until further defined</p>
bits 2-7	Reserved and are set to logic 0 until further defined

Byte 5

bits 0-7	Reserved and are set to logic 0 until further defined
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Bytes 6-9

bits 0-7 (each byte)	<p>Alphanumeric channel origin data. First character in message is byte 6.</p> <p>7-bit International Organization for Standardization (ISO) 646, American Standard Code for Information Interchange (ASCII), data with no parity bit. LSBs are transmitted first with logic 0 in bit 7. Nonprinted control characters (codes 01 to 1F hex and 7F hex) are not permitted. Default value is logic 0 (code 00 hex, ASCII null).</p>
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Bytes 10-13

bits 0-7 (each byte)	<p>Alphanumeric channel destination data. First character in message is byte 6.</p> <p>7-bit International Organization for Standardization (ISO) 646, American Standard Code for Information Interchange (ASCII), data with no parity bit. LSBs are transmitted first with logic 0 in bit 7. Nonprinted control characters (codes 01 to 1F hex and 7F hex) are not permitted. Default value is logic 0 (code 00 hex, ASCII null).</p>
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Bytes 14-17

bits 0-7 (each byte)	Local sample address code (32-bit binary with LSBs first). Value is of first sample of current block LSBs are transmitted first. Default value is logic 0. NOTE-This has the same function as a recording index counter.
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Bytes 18-21

bits 0-7 (each byte)	Time-of-day sample address code (32-bit binary with LSBs first). Value is of first sample of current block. LSBs are transmitted first. 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 (i.e., 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.
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Byte 22

bits 0-3	Flag used to identify whether the information carried by the channel status data is reliable. According to the following table, if reliable, the appropriate bits are set to logic 0 (default); if unreliable, the bits are set to logic 1
	Reserved and are set to logic 0 until further defined
bit 4	Bytes 0 to 5
bit 5	Bytes 6 to 13
bit 6	Bytes 14 to 17
bit 7	Bytes 18 to 21

Byte 23

bits 0-7	Channel status data cyclic redundancy check character (CRCC) 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. Default value is logic 0 for "minimum" implementation of channel status only (see 5.2.1) NOTE Annex B includes a diagram of the shift register circuit used to generate the code, two examples of channel status data, and the corresponding CRCC
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5. Interface format implementation

5.1 General

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.

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

5.2 Transmitter

Transmitters shall follow all the formatting and channel coding rules established in earlier sections of this specification including all notes therein. Along with the audio sample word, all transmitters shall correctly encode and transmit the validity bit, user bit, parity bit, and the three preambles. The channel status shall be encoded to one of the implementations given in 5.2.1, 5.2.2, and 5.2.3.

The following three implementations are defined: "minimum," "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 Section 4 shall remain unchanged.

5.2.1 "Minimum" implementation of channel status

The "minimum" implementation represents the lowest level of implementation of the interface that meets the requirements of this specification document. In the "minimum" implementation, transmitters shall encode and transmit channel status byte 0 bit 0 with a state of logic 1 signifying "professional use of channel status block." All other channel status bits of byte 0 to byte 23 inclusive shall be transmitted with the default state of all logic 0-5. In this circumstance, the receiver will adopt the default conditions specified in bytes 0 to 2.

If additional bytes of channel status (which do not fully comply with the "standard" implementation, see 5.2.2) are implemented as required by an application, the interface transmitter shall be classified as a "minimum" implementation of channel status.

It should be noted that the "minimum" implementation imposes severe operational restrictions on some receiving devices which may be connected to it. For example, receivers implementing byte 23 will normally show a cyclic redundancy check error when the default value of logic 0 is received as the CRCC. Also, reception of the default value for byte 0 bits 6-7 might cause improper operation in receiving devices not supporting manual override or auto set capabilities.

5.2.2 "Standard" implementation of channel status

The "standard" implementation provides a fundamental level of implementation which should prove sufficient for general applications in professional audio or broadcasting. In addition to conforming to the requirements described in 5.2.1 for the "minimum" implementation, a "standard" implementation interface transmitter 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.

5.2.3 "Enhanced" implementation of channel status

In addition to conforming to the requirements described in 5.2.2 for the "standard" implementation, the "enhanced" implementation shall provide further capabilities .

5.3 Receivers

Implementation in receivers is highly dependent on the application. Proper documentation shall be provided on the level of implementation of the interface receiver for decoding the transmitted information (validity, user, channel status, parity) and on whatever subsequent action is taken by the equipment of which it is a part.

6 Electrical requirements

6.1 General characteristics

The electrical parameters of the interface are based on those defined in CCITT V. 11 which allow transmission of balanced-voltage digital signals up to a few hundred meters distance.

In order to improve the balance of the transmitter or the receiver, or both, beyond that recommended by the CCITT, a circuit conforming to the general configuration shown in figure 6 may be used.

In this circuit, the series capacitors C_g and C_g isolate the transformers and prevent damage from connection to a source containing a dc voltage. In addition to achieving higher rejection of common-mode signals, the transformers reduce grounding and electromagnetic interference (EMI) problems. Although equalization may be used at the receiver, no equalization before transmission shall be permitted.

The interconnecting cable shall be balanced and screened (shielded) with a nominal characteristic impedance of 110 Ω at frequencies from 0.1 to 6.0 MHz .

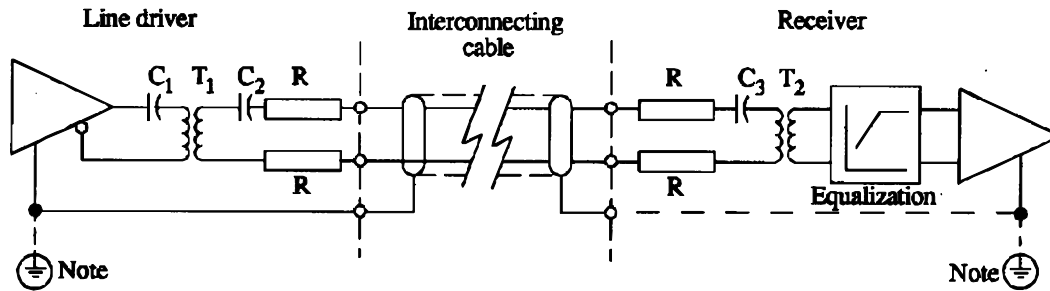
6.2 Line driver characteristics

6.2.1 Output impedance

The line driver shall have a balanced output with an internal impedance of $110\ \Omega + 20\%$, at frequencies from 0.1 to 6.0 MHz, when measured at the output terminals .

6.2.2 Signal amplitude

The signal amplitude shall be between 2 and 7 V peak to peak, when measured across a $110\text{-}\Omega$ resistor connected to the output terminals, without any interconnecting cable present.



Note - Signal ground may be further connected to external protective ground if national regulations require.

Figure 6 General circuit configuration.

6.2.3 Balance

Any common-mode component at the output terminals shall be more than 30 dB below the signal at frequencies from dc to 6 MHz.

6.2.4 Rise and fall times

The rise and fall times, determined between the 10% and 90% amplitude points, shall be between 5 ns and 30 ns when measured across a $110\text{-}\Omega$ resistor connected to the output terminals, without any interconnecting cable present.

NOTE: Operation toward the lower limit of 5 ns may improve the received signal eye pattern, but may increase EMI at the transmitter. Equipment must meet local regulations regarding EMI.

Data transitions shall occur within ± 20 ns of an ideal jitter-free clock measured at the half-voltage points.

NOTE: This specification applies only to the signal after channel coding. Tighter specifications apply to the audio sample clock.

6.3.1 Terminating impedance

The receiver shall present an essentially resistive impedance of $110\ \Omega + 20\%$ to the interconnecting cable over the frequency band from 0.1 to 6.0 MHz when measured across the input terminals. The application of more than one receiver to any one line might create transmission errors due to the resulting impedance mismatch.

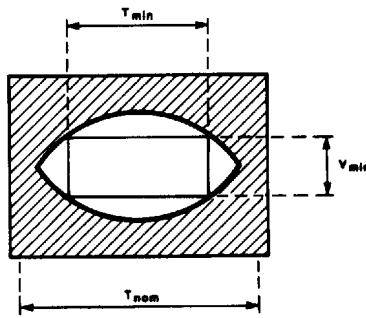
6.3.2 Maximum input signals

The receiver shall correctly interpret the data when connected directly to a line driver working between the extreme voltage limits specified in 6.2.2.

NOTE: The AES3-1985 specification for line driver signal amplitude was 10 V peak to peak maximum.

6.3.3 Minimum input signals

The receiver shall correctly sense the data when a random input signal produces the eye diagram characterized by a V_{min} of 200 mV and T_{min} of 50% of T_{nom} (see figure 7).



T_{min} : $0.5 \times T_{nom}$
 V_{min} : 200 mV
 T_{nom} : One-half the biphase symbol period

Figure 7, Eye diagram

6.3.4 Receiver equalization

Optional equalization can be applied in the receiver to enable an interconnecting cable longer than 100 m to be used. A suggested frequency equalization characteristic is shown in figure 8. The receiver shall meet the requirements specified in 6.3.2 and 6.3.3.

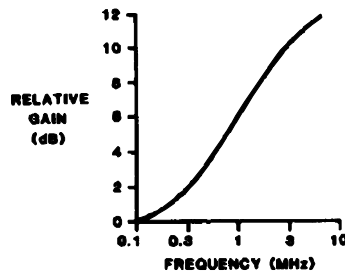


Figure 8. Suggested equalization characteristic for receiver

6.3.5 Common-mode rejection

There shall be no data errors introduced by the presence of a common-mode signal of up to 7 V peak at frequencies from dc to 20 kHz.

6.4 Connectors

The standard connector for both outputs and inputs shall be the circular latching three-pin connector described in IEC 268- 12. (This type of connector is normally called XLR.)

An output connector fixed on an item of equipment shall use male pins with a female shell. The corresponding cable connector shall thus have female pins with a male shell.

An input connector fixed on an item of equipment shall use female pins with a male shell. The corresponding cable connector shall thus have male pins with a female shell. The pin usage shall be:

- Pin 1 Cable shield or signal earth;
- Pin 2 Signal;
- Pin 3 Signal.

(Note that the relative polarity of pins 2 and 3 is not important in the digital case.)

Equipment manufacturers should clearly label digital audio inputs and outputs as such, including the terms "digital audio input" or "digital audio output" as appropriate .

In such cases where panel space is limited and the function of the connector might be confused with an analog signal connector, the abbreviation DI or DO should be used to designate digital audio inputs and outputs, respectively.

7 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 the most recent editions of the Standards listed below should be obtained.]

AES5-1984, AES Recommended Practice for Professional Digital Audio Applications Employing PulseCode Modulation Preferred Sampling Frequencies, Audio Engineering Society, New York, NY, USA (1984).

AES10-1991, AES Recommended Practice for Digital Audio Engineering--Serial Multichannel Audio Digital Interface (MADI), Audio Engineering Society, New York, NY, USA (1991) .

AES 11 - 1991, AES Recommended Practice for Digital Audio Engineering--Synchronization of Digital Audio Equipment in Studio Operations, Audio Engineering Society, New York, NY, USA (1991).

AES18-1992, 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 (1992).

CCIR Recommendation 647, A digital audio interface for broadcasting studios, Green Book, vol. 10 part 1, International Radio Consultative Committee (CCIR), Dubrovnik, Yugoslavia (1986).

CCITT Recommendation J.17, Pre-emphasis used on sound program circuits, International Telegraph and Telephone Consultative Committee (1972).

CCITT Recommendation V.11, Electrical characteristics for balanced double-current interchange circuits for general use with integrated circuit equipment in the field of data communications, International Telegraph and Telephone Consultative Committee (1976, 1980).

IEC 268-11, Sound system equipment, part 11: Application of connectors for the interconnection of sound system components, International Electrotechnical Commission, Geneva, Switzerland (1987).

IEC 268-12, Sound system equipment, part 12: Application of connectors for broadcast and similar use, International Electrotechnical Commission, Geneva, Switzerland (1987).

IEC 841, Audio recording--PCM encoder / decoder system, International Electrotechnical Commission, Geneva, Switzerland (1988).

IEC 908, Compact disc digital audio system, International Electrotechnical Commission, Geneva, Switzerland (1987).

IEC 958, Digital audio interface, International Electrotechnical Commission, Geneva, Switzerland (1989).

ISO 646: 1983, Information processing--ISO 7-bit coded character set for information interchange, International Organization for Standardization, Geneva, Switzerland (1983).

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