IEC 60958-4

Edition 2.1 2008-07

INTERNATIONAL STANDARD

Digital audio interface –
Part 4: Professional applications

IEC 60958-4:2003+A1:2008(E)

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Part 4: Professional applications

INTERNATIONAL
ELECTROTECHNICAL
COMMISSION

PRICE CODE **CE**

ICS 33.160.30

ISBN 2-8318-9703-3

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DIGITAL AUDIO INTERFACE –

Part 4: Professional applications

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International Standard IEC 60958-4 has been prepared by IEC technical committee 100: Audio, video and multimedia systems and equipment.

This consolidated version of IEC 60958-4 consists of the second edition (2003) [documents 100/643/FDIS and 100/669/RVD] and its amendment 1 (2008) [documents 100/1330/FDIS and 100/1355/RVD].

The technical content is therefore identical to the base edition and its amendment and has been prepared for user convenience.

It bears the edition number 2.1.

A vertical line in the margin shows where the base publication has been modified by amendment 1.

This publication has been drafted in accordance with the ISO/IEC Directives, Part 2.

The main changes with respect to the previous edition (1999) are listed below.

– The scope specifies the professional application of IEC 60958-1 (generalities have been removed to an introduction).
– A clause on terms and definitions has been added.
– In Table 1, expanded channel status assignments have been added and channel status definitions expanded to accommodate extended sampling frequencies, indication of alignment level and multi-channel options.
– Figure 1 and associated text has been revised to be more generalized. Three notes on cable performance factors have been added.
– The impedance specification is now dependent on maximum frame rate.
– The common-mode balance specification is now dependent on maximum frame rate
– The impedance specification is now dependent on maximum frame rate.

IEC 60958 consists of the following parts under the generic title *Digital audio interface*:

Part 1: General

Part 3: Consumer applications

Part 4: Professional applications

The committee has decided that the contents of the base publication and its amendments will remain unchanged until the maintenance result date indicated on the IEC web site under "http://webstore.iec.ch" in the data related to the specific publication. At this date, the publication will be

• reconfirmed,
• withdrawn,
• replaced by a revised edition, or
• amended.

A bilingual version of this publication may be issued at a later date.

INTRODUCTION

The interface specified in this standard is primarily intended to carry monophonic or stereophonic programmes at a 48 kHz sampling frequency and with a resolution of up to 24 bits per sample. It may alternatively be used to carry signals sampled at other rates such as 32 kHz, 44,1 kHz, or 96 kHz. Note that conformity to this interface specification does not require equipment to utilize these rates and also that the capability of the interface to indicate other sample rates does not imply that it is recommended that equipment supports these rates. To eliminate doubt, equipment specifications should define the supported sampling frequencies.

The format is intended for use with shielded twisted-pair cables over distances of up to 100 m without transmission equalization or any special equalization at the receiver and at frame rates of up to 50 kHz. Longer cable lengths and higher frame rates may be used with cables better matched for data transmission, or with receiver equalization, or both.

In both cases, the clock references and auxiliary information are transmitted along with the audio data. Provision is also made to allow the interface to carry non-audio data.

DIGITAL AUDIO INTERFACE –

Part 4: Professional applications

1 Scope

This International Standard specifies the professional application of the interface for the interconnection of digital audio equipment defined in IEC 60958-1.

2 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

IEC 60268-12:1987, *Sound system equipment – Part 12: Application of connectors for broadcast and similar use*

IEC 60958-1, *Digital audio interface – Part 1: General*

IEC 60958-3, *Digital audio interface – Part 3: Consumer applications*

ISO/IEC 646:1991, *Information technology – ISO 7-bit coded character set for information interchange*

ITU-T Recommendation J.17:1988, *Pre-emphasis used on sound-programme circuits*

ITU-T Recommendation V.11:1996, *Electrical characteristics for balanced double-current interchange circuits operating at data signalling rates up to 10 Mbit/s*

3 Terms and definitions

The terms and definitions given in IEC 60958-1 apply to this part of IEC 60958.

4 Interface format

4.1 General

The interface format as defined in IEC 60958-1 shall be used.

For historical reasons, preambles “B”, “M” and “W”, as defined in 4.3 of IEC 60958-1, shall, for use in professional applications, be referred to as “Z”, “X” and “Y”, respectively.

4.2 Validity bit

For this standard, the validity bit shall be used to indicate whether the main data field bits in the sub-frame are suitable for conversion to an analogue audio signal using linear PCM coding.

5 Channel status

5.1 General

The channel status for each audio signal carries information associated with that audio signal; thus it is possible for different channel status data to be carried in the two sub-frames 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 pre-emphasis.

Channel status information is organized in a 192-bit block, subdivided into 24 bytes, numbered 0 to 23 (see Table 1). The first bit of each block is carried in the frame with preamble “Z”.

The individual bits of a block are numbered 0 to 191.

The primary application is indicated by channel status bit 0.

For the professional applications described here, this first channel status bit equals “1”.

NOTE For consumer digital audio equipment, this first channel status bit equals “0”, and this part of IEC 60958 does not apply.

Secondary applications may be defined within the framework of these primary applications.

5.2 Professional linear PCM application

The specific organization of the channel status data is defined in this clause and summarized in Table 1.

The significance of byte 0, bit 0 is such that a transmission from an interface conforming to IEC 60958-3 (“consumer use”) can be identified. Also, a “professional use” transmission, defined in this part of IEC 60958, can be correctly identified by a “consumer use” receiver. Connection of a “consumer use” transmitter with a “professional use” receiver or vice versa might result in unpredictable operation. Thus, the byte definitions in this clause apply only when bit 0 = “1” and bit 1 = “0” (professional linear PCM use of the channel status block).

Table 1 – Channel status data format for professional linear PCM application

Byte									
0		a = “1”	b = “0”	C			d	e	
	Bit	0	1	2	3	4	5	6	7
1		f				g			
	Bit	8	9	10	11	12	13	14	15
2		h			i			j	
	Bit	16	17	18	19	20	21	22	23
3		k							n=”0”
		l				m			n=”1”
	Bit	24	25	26	27	28	29	30	31
4		o		p	q				r
	Bit	32	33	34	35	36	37	38	39
5		Reserved but undefined at present							
	Bit	40	41	42	43	44	45	46	47
6		Alphanumeric channel origin data							
	Bit	48	49	50	51	52	53	54	55
7		Alphanumeric channel origin data							
	Bit	56	57	58	59	60	61	62	63
8		Alphanumeric channel origin data							
	Bit	64	65	66	67	68	69	70	71
9		Alphanumeric channel origin data							
	Bit	72	73	74	75	76	77	78	79
10		Alphanumeric channel destination data							
	bit	80	81	82	83	84	85	86	87
11		Alphanumeric channel destination data							
	bit	88	89	90	91	92	93	94	95
12		Alphanumeric channel destination data							
	bit	96	97	98	99	100	101	102	103
13		Alphanumeric channel destination data							
	bit	104	105	106	107	108	109	110	111
14		Local sample address code (32-bit binary)							
	bit	112	113	114	115	116	117	118	119
15		Local sample address code (32-bit binary)							
	bit	120	121	122	123	124	125	126	127
16		Local sample address code (32-bit binary)							
	bit	128	129	130	131	132	133	134	135
17		Local sample address code (32-bit binary)							
	bit	136	137	138	139	140	141	142	143
18		Time of day code (32-bit binary)							
	bit	144	145	146	147	148	149	150	151
19		Time of day code (32-bit binary)							
	bit	152	153	154	155	156	157	158	159
20		Time of day code (32-bit binary)							
	bit	160	161	162	163	164	165	166	167
21		Time of day code (32-bit binary)							
	bit	168	169	170	171	172	173	174	175
22		Reliability flags							
	bit	176	177	178	179	180	181	182	183
23		Cyclic redundancy check character							
	bit	184	185	186	187	188	189	190	191

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

Byte 0

Bit 0	Use of channel status block	
State	“1”	Professional use of channel status block (note 1)
Bit 1	Linear PCM identification	
State	“0”	Audio sample word represents linear PCM samples (note 1)
	“1”	Audio sample word used for purposes other than linear PCM samples

NOTE 1 The functions of channel status bits 0 and 1 are defined in IEC 60958-1.

Bits 2 to 4	Encoded audio signal pre-emphasis.	
Bit	2 3 4	
State	“0 0 0”	Pre-emphasis not indicated. Receiver defaults to no pre-emphasis with manual override enabled.
	“1 0 0”	No pre-emphasis. Receiver manual override is disabled.
	“1 1 0”	50 µs/15 µs pre-emphasis. Receiver manual override is disabled.
	“1 1 1”	ITU-T Recommendation J.17 pre-emphasis (with 6,5 dB insertion loss at 800 Hz). Receiver manual override is disabled.
	All other states of bits 2 to 4 are reserved and shall not be used until further defined.	
Bit 5	Lock indication	
State	“0”	Default, lock condition not indicated.
	“1”	Source sampling frequency unlocked.
Bits 6 to 7	Encoded sampling frequency	
Bit	6 7	
State	“0 0”	Sampling frequency not indicated. Receiver defaults to 48 kHz and manual override or auto set is enabled.
	“0 1”	48 kHz sampling frequency. Receiver manual override or auto set is disabled.
	“1 0”	44,1 kHz sampling frequency. Receiver manual override or auto set is disabled.
	“1 1”	32 kHz sampling frequency. Receiver manual override or auto set is disabled.

NOTE 2 The indication of sampling frequency, or the use of one of the sampling frequencies that can be indicated in this byte, is not a requirement for operation of the interface. The 00 state of bits 6 to 7 may be used if the transmitter does not support the indication of sampling frequency, if the sampling frequency is unknown, or if the sample frequency is not one of those that can be indicated in this byte. In the latter case for some sampling frequencies byte 4 may be used to indicate the correct value.

NOTE 3 When bits 8 to 11 in byte 1 indicate single-channel double-sampling frequency mode, the sampling frequency of the audio signal is twice that indicated by bits 6 to 7 in byte 0.

Byte 1

The six modes of transmission are signalled by setting bits 8 to 11 of byte 1 of channel status.

- *Two-channel mode:* In two-channel mode, the samples from both channels are transmitted in consecutive sub-frames. Channel 1 is in sub-frame 1 and channel 2 is in sub-frame 2.
- *Stereophonic mode:* In stereophonic mode, the interface is used to transmit stereophonic signals, and the two channels are presumed to have been simultaneously sampled. The left, or “A”, channel is in sub-frame 1 and the right, or “B”, channel is in sub-frame 2.
- *Single-channel mode (monophonic):* In monophonic mode, the transmitted bit rate remains at the normal two-channel rate and the audio sample word is placed in sub-frame 1. Time slots 4 to 31 of sub-frame 2 either carry the bits identical to sub-frame 1 or are set to logical “0”. A receiver normally defaults to channel 1, unless manual override is provided.
- *Primary/secondary mode:* In some applications requiring two channels where one of the channels is the main or primary channel while the other is a secondary channel, the primary channel is in sub-frame 1 and the secondary channel is in sub-frame 2.
- *Multichannel mode:* The one or two channels carried on the interface are part of a larger group. Channel identification within this group is in byte 3.
- *Single-channel double-sampling frequency mode:* Sub-frames 1 and 2 carry successive samples of the same signal. The sampling frequency of the signal is double the frame repetition rate and is double the sampling frequency indicated in byte 0 (but not double the rate indicated in byte 4, if that is used). Manual override is disabled.

Bits 8 to 11	Encoded channel mode	
Bit	8 9 10 11	
State	“0 0 0 0”	Mode not indicated. Receiver defaults to two-channel mode and manual override is enabled.
	“0 0 0 1”	Two-channel mode. Receiver manual override is disabled.
	“0 0 1 0”	Single-channel mode (monophonic). Receiver manual override is disabled.
	“0 0 1 1”	Primary/secondary mode (sub-frame 1 is primary). Receiver manual override is disabled.
	“0 1 0 0”	Stereophonic mode (sub-frame 1 is left channel). Receiver manual override is disabled.
	“0 1 0 1” and “0 1 1 0”	Reserved for user-defined applications.
	“0 1 1 1”	Single-channel double-sampling frequency mode – vector to byte 3 for channel identification.
	“1 0 0 0 “	Single-channel double-sampling frequency mode – stereophonic left.
	“1 0 0 1”	Single-channel double-sampling frequency mode – stereophonic right.
	“1 1 1 1”	Multichannel mode. Vector to byte 3.

All other states of bits 8 to 11 are reserved and shall not be used until further defined.

Bits 12 to 15	Encoded user bits management	
Bit	12 13 14 15	
State	"0 0 0 0"	Default, user data format is undefined.
	"0 0 0 1"	192-bit block structure. Preamble "B" indicates the start of the block.
	"0 0 1 0"	Reserved for the AES18 standard.
	"0 0 1 1"	User-defined.
	"0 1 0 0"	User data conforms to the general user data format as defined in IEC 60958-3.
	"0 1 0 1"	Reserved for metadata as described in AES52.
	All other states of bits 12 to 15 are reserved and shall not be used until further defined.	

Byte 2

Bits 16 to 18	Encoded use of auxiliary sample bits	
Bit	16 17 18	
State	"0 0 0"	Maximum audio sample word length is 20 bits (default). Use of auxiliary sample bits is not defined.
	"0 0 1"	Maximum audio sample word length is 24 bits. Auxiliary sample bits are used for main audio sample data.
	"0 1 0"	Maximum audio sample word length is 20 bits. Auxiliary sample bits in this channel are used to carry a single co-ordination signal.
	"0 1 1"	Reserved for user-defined applications.
	All other states of bits 16 to 18 are reserved and shall not be used until further defined.	

Bits 19 to 21	Encoded audio sample word length of transmitted signal		
Bit	19 20 21	Audio sample word length, if maximum length is 24 bits (indicated by bits 16 to 18 above).	Audio sample word length, if maximum length is 20 bits (indicated by bits 16 to 18 above).
State	"0 0 0"	Word length not indicated (default).	Word length not indicated (default).
	"0 0 1"	23 bits	19 bits
	"0 1 0"	22 bits	18 bits
	"0 1 1"	21 bits	17 bits
	"1 0 0"	20 bits	16 bits
	"1 0 1"	24 bits	20 bits
	All other states of bits 19 to 21 are reserved and shall not be used until further defined.		

NOTE 4 The default state of bits 19 to 21 indicates that the number of active bits within the 20-bit or 24-bit coding range is not specified by the transmitter. The receiver should default to the maximum number of bits specified by the coding range and enable manual override or auto set.

NOTE 5 The non-default state of bits 19 to 21 indicates the number of bits within the 20-bit or 24-bit coding range which might be active. This is also an indirect expression of the number of LSBs that are certain to be inactive, which is equal to 20 or 24 minus the number corresponding to the bit state. The receiver should disable manual override and auto set for these bit states.

NOTE 6 Irrespective of the audio sample word length as indicated by any of the states of bits 19 to 21, the MSB is in time slot 27 of the transmitted sub-frame as specified in 3.2.1 of IEC 60958-1.

Bits 22 and 23	Indication of alignment level	
Bit	22 23	
State	“0 0”	Alignment level not indicated (default).
	“0 1”	Alignment level is 20 dB below maximum code (refer to SMPTE RP155).
	“1 0”	Alignment level is 18,06 dB below maximum code (refer to EBU R68).
	“1 1”	Reserved for future use.

Byte 3

Bit 31	Multichannel mode control bit	
State	“0”	Undefined multichannel mode (default).
	“1”	Defined multichannel modes.

The definition of the remaining bit states depends on the state of bit 31.

When bit 31 is 0:

Bits 24 to 31 Channel number

Bit	24	25	26	27	28	29	30	31
State	X	X	X	X	X	X	X	0
	LSB						MSB	

The channel number is the value of the byte plus one.

When bit 31 is 1:

Bits 24 to 31 Channel number and multichannel number.

Bit	24	25	26	27	28	29	30	31
State	X	X	X	X	Y	Y	Y	1
	LSB			MSB				

The channel number is one plus the numeric value of the bits shown by “X” taken as a binary number. The bits shown by “Y” define the multichannel mode as follows.

Bits 28 to 30	Multichannel mode number	
Bit	28 29 30	
State	“0 0 0”	Multichannel mode 0. The channel number is defined by bits 24 to 27.
	“1 0 0”	Multichannel mode 1. The channel number is defined by bits 24 to 27.
	“0 1 0”	Multichannel mode 2. The channel number is defined by bits 24 to 27.
	“1 1 0”	Multichannel mode 3. The channel number is defined by bits 24 to 27.
	“1 1 1”	User defined multichannel mode. The channel number is defined by bits 24 to 27.

All other states of bits 28 to 30 are reserved and are not to be used until further defined.

Byte 4

Bits 32 to 33	Digital audio reference signal	
Bit	32 33	
State	"0 0"	Not a reference signal (default).
	"0 1"	Grade 1 reference signal.
	"1 0"	Grade 2 reference signal.
	"1 1"	Reserved and not used until further defined.

Bit 34	Reserved and set to "0" until further defined.

Bits 35 to 38	Sampling frequency	
Bit	35 36 37 38	
State	"0 0 0 0"	Not indicated (default)
	"1 0 0 0"	24 kHz
	"0 1 0 0"	96 kHz
	"1 1 0 0"	192 kHz
	"0 0 1 0"	Reserved.
	"1 0 1 0"	Reserved.
	"0 1 1 0"	Reserved.
	"1 1 1 0"	Reserved.
	"0 0 0 1"	Reserved (for vectoring)
	"1 0 0 1"	22,05 kHz
	"0 1 0 1"	88,2 kHz
	"1 1 0 1"	176,4 kHz
	"0 0 1 1"	Reserved.
	"1 0 1 1"	Reserved.
	"0 1 1 1 "	Reserved.
	"1 1 1 1"	User defined.

Bit 39		Sampling frequency scaling flag
State	"0"	No scaling (default)
	"1"	Sampling frequency is 1/1.001 times that indicated either in bits 35 to 38 or bits 6 to 7.

Byte 5

Bits 40 to 47	Reserved and set to "0" until further defined

Bytes 6 to 9

Alphanumeric channel origin data. First character in message is byte 6.

Bits 48 to 79	7-bit ISO/IEC 646 (ASCII) data with no parity bit. LSBs are transmitted first with “0” in bit 7. Non-printing control characters (codes 01 to 1F hex and 7F hex) are not permitted. Default value is “0” (code 00 hex, ASCII null).

Bytes 10 to 13

Alphanumeric channel destination. First character in message is byte 10.

Bits 80 to 111	7-bit ISO/IEC 646 (ASCII) data with no parity bit. LSBs are transmitted first with “0” in bit 7. Non-printing control characters (codes 01 to 1F hex and 7F hex) are not permitted. Default value is “0” (code 00 hex, ASCII null).

Bytes 14 to 17

Local sample address code (32-bit binary with LSBs first). Value is of first sample of current block.

Bits 112 to 143	LSBs are transmitted first. Default value is “0”.

NOTE 7 This has the same function as a recording index counter and increments by 192 for each successive block, unless a discontinuity or edit occurs.

Bytes 18 to 21

Time-of-day sample address code (32-bit binary with LSBs first). Value is of first sample of current block.

Bits 144 to 175	LSBs are transmitted first. Default value is “0”.

NOTE 8 This is the time of day laid down during the source encoding of the signal. It remains unchanged during subsequent operations, and increments by 192 for each successive block, unless a discontinuity or edit occurs. 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 m, 00 s, 00 frame). Transcoding of the binary number to any conventional time code requires accurate sample frequency information to provide a sample time accurate to ±1 sample period.

Byte 22

Flag used to identify whether the information carried by the channel status data is reliable. According to the following list, if data is reliable the appropriate bits are set to “0” (default); if the data is unreliable, the bits are set to “1”.

Bits 176-179	Reserved, and set to “0” until further defined.
Bit 180	Bytes 0 to 5.
Bit 181	Bytes 6 to 13.
Bit 182	Bytes 14 to 17.
Bit 183	Bytes 18 to 21.

Byte 23

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 “1”s should be used in generating the check bits with the LSBs transmitted first. Default value is logical “0” for “minimum” implementation of channel status only (see 7.2.1).

6 User data

6.1 General

The default value of the user bits is “0”.

6.2 Application

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 12 to 15.

7 Implementation

7.1 General

To promote compatible operation between items of equipment built to this standard, 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.

7.2 Transmitter

Transmitters shall follow all the formatting and channel coding rules established in this standard. 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 7.2.1, 7.2.2, and 7.2.3.

These three implementations are defined as 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 clause 4 shall remain unchanged.

7.2.1 Minimum implementation of channel status

The minimum implementation represents the lowest level of implementation of the interface that meets the requirements of this standard. In the minimum implementation, transmitters shall encode and transmit channel status byte 0 bit 0 with a state of logical “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 logical “0”s. In this circumstance, the receiver shall adopt the default conditions specified in bytes 0 to 2.

If additional bytes of channel status (which do not fully conform to the standard implementation, given in 7.2.2) 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 that may be connected to it. For example, receivers implementing byte 23 will normally show a cyclic redundancy check error when the default value of logical "0" is received as the CRCC. Also, reception of the default value for byte 0 bits 6 to 7 might cause improper operation in receiving devices that do not support manual override or auto-set capabilities.

7.2.2 Standard implementation of channel status

The standard implementation provides a fundamental level of implementation that should prove sufficient for general applications in professional audio or broadcasting. In addition to conforming to the requirements described in 7.2.1 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 standard.

7.2.3 Enhanced implementation of channel status

The enhanced implementation shall correctly encode and transmit other channel status bits in addition to conforming to the requirements described in 7.2.2 for the standard implementation.

7.3 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 response is made by the equipment of which it is a part.

8 Electrical requirements

8.1 General

The type of transmission line and timing accuracy of the transmitted signal wave form shall meet the required quality or purpose of use.

8.2 Balanced line

8.2.1 General characteristics

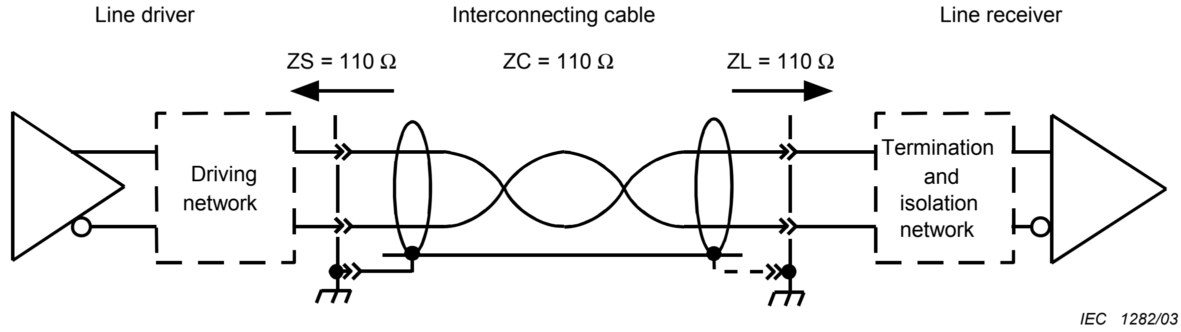
The electrical parameters of the interface are based on those defined in ITU-T Recommendation V.11 which allow transmission of balanced-voltage digital signals up to a few hundred meters in length.

In order to improve the balance of the transmitter or the receiver, or both, beyond that recommended by the ITU-T, a circuit conforming to the general configuration shown in Figure 1 may be used.

Although equalization may be used at the receiver, there shall be no equalization before transmission.

The frequency range used to qualify the interface electrical parameters is dependent on the maximum data rate supported. The upper frequency is 128 times the maximum frame rate.

The interconnecting cable shall be balanced and screened (shielded) with a nominal characteristic impedance of 110 Ω at frequencies from 100 kHz to 128 times the maximum frame rate.



NOTE 1 Holding closer tolerances for the characteristic impedance of the cable, and for the driving and terminating impedances, can increase the cable lengths for reliable transmission and for higher data rates.

NOTE 2 Closer tolerances for the balance of the driving impedance, the terminating impedance, and for the cable itself can reduce both electromagnetic susceptibility and emissions.

NOTE 3 Using cable having lower loss at higher frequencies can improve the reliability of transmission for greater distances and higher data rates.

Figure 1 – Simplified example of the configuration of the circuit (balanced)

8.2.2 Line driver characteristics

8.2.2.1 Output impedance

The line driver shall have a balanced output with an internal impedance of 110 Ω with a relative tolerance of ±20 %, at frequencies from 100 kHz to 128 times the maximum frame rate, when measured at the output terminals.

8.2.2.2 Signal amplitude

The signal amplitude shall lie between 2 V and 7 V peak-to-peak, when measured across a resistor of 110 Ω with a relative tolerance of ±1 % connected to the output terminals, without any interconnecting cable present.

8.2.2.3 Balance

Any common-mode component at the output terminals shall be more than 30 dB below the signal at frequencies from 100 kHz to 128 times the maximum frame rate.

8.2.2.4 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 Ω 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. IEC/CISPR standards and local regulations regarding EMI should be taken into account.

8.2.2.5 Output interface jitter

Output jitter is a combination of jitter intrinsic to the device and jitter being passed through from the timing reference of the device.

8.2.2.5.1 Intrinsic jitter

The peak intrinsic output jitter measured at all the transition zero crossings shall be less than 0,025 unit interval (UI) (see UI definition in IEC 60958-1) when measured with the intrinsic jitter measurement filter.

NOTE This applies both when the equipment is locked to an effectively jitter-free timing reference (which may be a modulated digital audio signal) and when the equipment is free-running.

The intrinsic jitter measurement filter is shown in Figure 2. It is a minimum-phase high-pass filter with a 3 dB frequency of 700 Hz, a first order roll-off to 70 Hz and with a pass-band gain of unity.

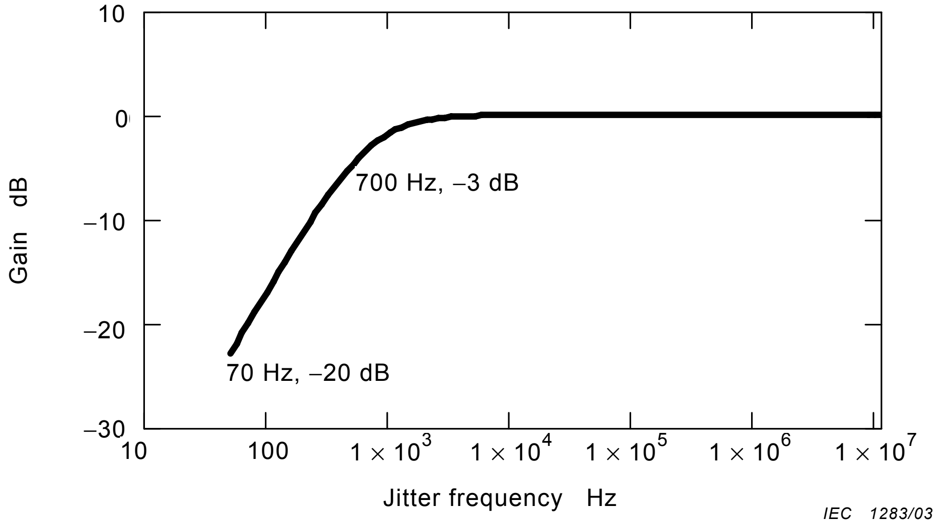


Figure 2 – Intrinsic jitter measurement filter

8.2.2.5.2 Jitter gain or peaking

The sinusoidal jitter gain from any timing reference input to the signal output shall be less than 2 dB at all frequencies.

NOTE It is recommended that, where jitter attenuation is provided, it should be such that the sinusoidal jitter gain falls below the jitter attenuation mask shown in Figure 3. It is desirable that the equipment specification states whether the equipment does or does not have jitter attenuation within this specification. (The mask imposes no additional limit on low-frequency jitter gain. The limit starts at the input jitter frequency of 500 Hz where it is 0 dB, and falls to –6 dB at and above 1 kHz.)

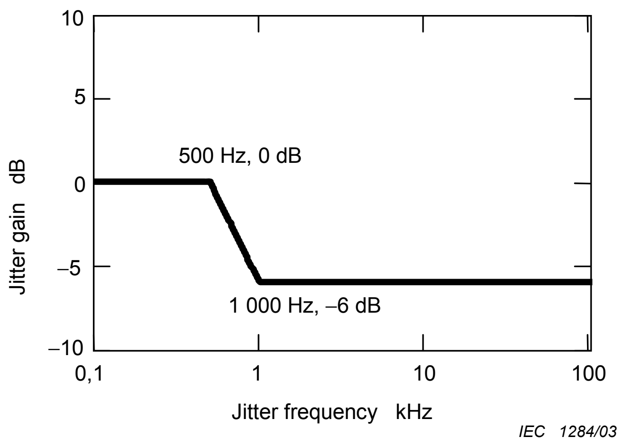


Figure 3 – Jitter attenuation mask (optional)

8.2.3 Line receiver characteristics

8.2.3.1 Terminating impedance

The receiver shall present a substantially resistive impedance of 110 Ω with a relative tolerance of ±20 % to the interconnecting cable over the frequency band from 100 kHz to 128 times the maximum frame rate 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.

8.2.3.2 Maximum input signals

The receiver shall correctly interpret the data when presented with a signal of which the peak-to-peak voltage, measured in accordance with 8.2.2.2, is 7 V.

NOTE The first edition of the IEC 60958 specification for balanced line driver amplitude was 10 V peak-to-peak maximum.

8.2.3.3 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 0,5 UI (see Figure 4).

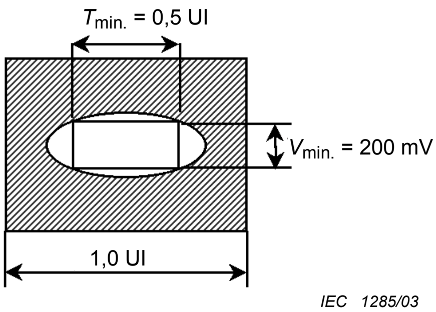


Figure 4 – Eye diagram

8.2.3.4 Receiver equalization

Equalization may be applied in the receiver to enable an interconnecting cable longer than 100 m to be used. A suggested equalizing characteristic for operation at a frame rate of 48 kHz is shown in Figure 5. The receiver shall still meet the requirements specified in 8.2.3.2 and 8.2.3.3.

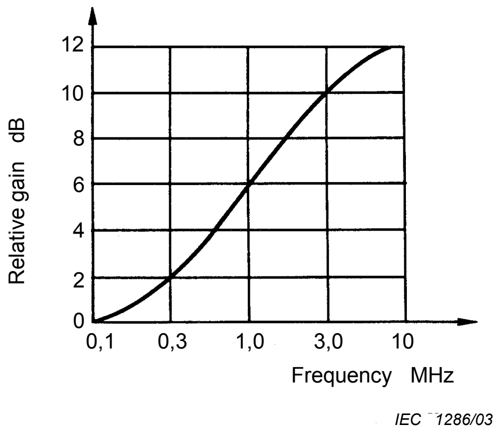


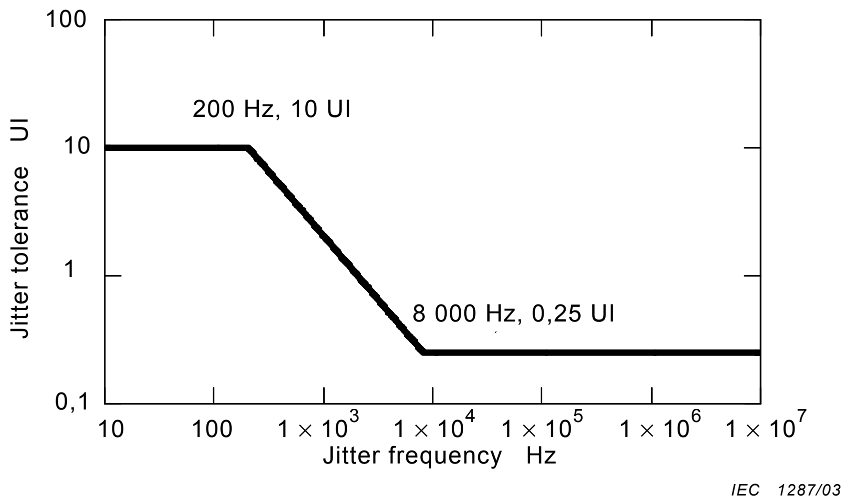
Figure 5 – A suggested equalizing characteristic for the receiver operating at a frame rate of 48 kHz

8.2.3.5 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 d.c. to 20 kHz.

8.2.3.6 Receiver jitter tolerance

An interface data receiver should correctly decode an incoming data stream with any sinusoidal jitter defined by the jitter tolerance template shown in Figure 6.



NOTE The template requires a jitter tolerance of 0,25 UI peak to peak at high frequencies, increasing with the inverse of frequency below 8 kHz to level off at 10 UI peak to peak below 200 Hz.

Figure 6 – Receiver jitter tolerance template

8.2.4 Connectors

The standard connector for both outputs and inputs shall be the circular latching three-pin connector described in IEC 60268-12 (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 receptacles 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.

The relative polarity of pins 2 and 3 shall not affect operation of the interface.

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

8.3 Unbalanced coaxial cables

Besides the transmission over balanced lines (clause 8.2) professional applications use unbalanced coaxial cables for transmissions over long distances (e.g. 1000 m) and/or for use at high sample rates to ensure the highest possible transmission quality. Signal characteristics, jitter, etc. (excluding voltage and impedance) are the same as described in clause 8.2 so the signal can be converted from balanced to unbalanced and vice versa with transformers. A relevant document to be considered is AES-3id-2001. This contains additional information as electrical requirements, cable and equalizer characterization and circuit implementations.

Bibliography

SMPTE Recommended Practice RP155-1997, *Audio levels for digital audio records on digital television tape recorders*

EBU Technical Recommendation R68-1992, *Alignment levels in digital audio production equipment and in digital audio recorders*

AES3-2003, *AES standard for digital audio – Digital input-output interfacing – Serial transmission format for two-channel linearly represented digital audio data*

AES-3id-2001(r2006), *AES information document for digital audio engineering – Transmission of AES3 formatted data by unbalanced coaxial cable*

AES52-2006, *AES standard for digital audio engineering – Insertion of unique identifiers into the AES3 transport stream*

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