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Radiocommunication Sector of ITU

Recommendation ITU-R BS.647-3

(03/2011)

A digital audio interface for broadcasting studios

BS Series

Broadcasting service (sound)

International Telecommunication Union

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Note: *This ITU-R Recommendation was approved in English under the procedure detailed in Resolution ITU-R 1.*

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RECOMMENDATION ITU-R BS.647-3*

A digital audio interface for broadcasting studios

(1986-1990-1992-2011)

Scope

This Recommendation specifies an interface for the serial digital transmission of two channels of periodically sampled and linearly represented digital audio data to be used in broadcasting studios.

The ITU Radiocommunication Assembly,

considering

a) that there is a need in broadcasting studios to interconnect in the digital domain various pieces of digital sound equipment;

b) that there would be advantages if all equipment used the same interface connections;

c) that Recommendation ITU-R BS.646 – Source encoding for digital sound signals in broadcasting studios, defines the digital sound format used for sound and television broadcasting applications;

d) that the interface should make allowance for processing headroom;

e) that the interface should allow for auxiliary data of various kinds to be carried,

recommends

1 that the interface described in Annex 1 should be used for serial digital transmission of two channels of periodically sampled and linearly represented digital audio data in broadcasting studios. This same interface may be used as a transport of compressed audio signals and other user defined data;

2 that compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words “shall” or some other obligatory language such as “must” and the negative equivalents are used to express requirements. The use of such words shall in no way be construed to imply partial or total compliance with this Recommendation.

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* This Recommendation should be brought to the attention of the IEC and the Audio Engineering Society (AES).

Annex 1

Serial transmission format for two-channel linearly-represented digital audio data

The following five parts (Part 1 to Part 5) specify an interface for the serial digital transmission of two channels of periodically sampled and linearly represented digital audio data in broadcasting studios.

– Part 1 defines the terms applied for this specification.
– Part 2 defines the format for coding audio used for the audio content.
– Part 3 defines the format for coding metadata, or subcode, relating to the audio content and carried with it.
– Part 4 defines the format for transport of a digital audio interface.
– Part 5 specifies the physical and electrical parameters for different media.

Although this transmission specification is independent of sampling frequency the interface is primarily intended to be used at 48 kHz as this is the recommended sampling frequency for use in broadcasting applications (Recommendation ITU-R BS.646).The Annex does not cover connection to any common carrier equipment.

NOTE 1 – In this interface specification, mention is made of an interface for consumer use. The two interfaces are not identical.

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Part 1

Terminology

1 Introduction

This Part 1 defines the terms applied for this Recommendation.

2 Terminology

For the purpose of this Recommendation the following definitions of terms apply.

2.1 Sampling frequency

The frequency of the samples representing an audio signal.

2.2 Audio sample word

A series of binary digits representing the amplitude of an audio sample, also known as a PCM sample.

2.3 Auxiliary sample bits

The four least-significant bits (LSBs) of those allocated to audio 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.

2.4 Validity bit

A bit indicating whether the audio sample bits in the same subframe are suitable for direct conversion to an analogue audio signal.

2.5 Most-significant bit

In the context of this standard: the Most-significant bit (MSB) of an audio sample word, being the sign bit in the case of two's complement code.

2.6 Least-significant bit

In the context of this standard: the Least-significant bit (LSB) of an audio sample word.

2.7 Subframe

The smallest structural element in a transport defined in Part 4, carrying one PCM sample and ancillary information.

2.8 Channel status

Bits carrying, in a fixed format derived from the block information associated with each audio channel which is decodable by any interface user.

2.9 User data

Channel provided to carry any other information.

2.10 Metadata

Information relating to the audio content in the same channel.

2.11 Frame

Sequence of two successive and associated subframes.

2.12 Biphase-mark

Channel-coding (or line-coding) technique which minimizes DC content and maximizes clock-recovery energy relative to the original binary bitstream.

2.13 Even parity bit

A bit whose value is chosen such that the total number of ones in the field which includes it is even.

2.14 Preambles

Specific unique patterns used for synchronization, compatible with but not part of the biphase mark code. See § 6 of Part 4.

2.15 Block

Group of 192 consecutive frames with a defined start point. See § 6 of Part 4.

NOTE 1 – The start of a block is designated by a special subframe preamble. See §§ 5 and 6 of Part 4.

2.16 Channel coding/line coding

Coding describing the method by which the binary digits are represented for transmission through the interface, see biphase mark above.

2.17 Unit interval (UI)

Shortest nominal time interval in the coding scheme.

NOTE 1 – There are 128 UI in a sample frame. See Appendix A to Part 5.

2.18 Interface jitter

Deviation in timing of interface data transitions (zero crossings) when measured with respect to an ideal clock.

2.19 Intrinsic jitter

Output interface jitter of a device that is either free-running or synchronized to a jitter-free reference.

2.20 Jitter gain

Ratio, expressed in decibels, of the amplitude of jitter at the synchronization input of a device to the resultant jitter at the output of the device.

NOTE 1 – This definition excludes the effect of intrinsic jitter.

2.21 Frame rate

Frequency at which frames are transmitted.

Part 2

Audio content

1 Introduction

This Part 2 defines the format for coding audio used for the audio content.

2 Audio content

2.1 Audio content coding

The audio content shall be coded as linear PCM using 2's complement code.

2.2 PCM polarity

Positive analogue voltages shall be represented by positive binary numbers.

2.3 Coding precision options

The accuracy of the coding shall be between 16 and 24 bits, in two ranges for the purpose of indicating which length is in use in channel status data, 16 to 20 bits and 20 to 24 bits. See Part 3.

2.4 Intermediate coding precision

The interface permits maximum word lengths of either 20 or 24 bits. A source which provides fewer bits than this shall be justified to the MSB of the available word length and the unused LSBs shall be set to logic 0.

NOTE 1 – If a low-resolution signal were not so justified, then sign extension would be needed.

2.5 Non-audio content

The interface may alternatively carry data or audio which is compressed or in a different format in place of linear PCM audio, in either channel B or both channels. In such cases the validity bit shall be set independently in each channel and channel status encoded to indicate this. See Part 3.

NOTE 1 – Such use is not standardized here: provision is only made to protect standard equipment from such use.

2.6 DC content

The coded audio shall contain as little equivalent DC offset as possible, and in any case less than the analogue equivalent noise level.

3 Sampling frequency

3.1 Channel interdependency

The sampling frequency shall be the same in both channels.

3.2 Choice of sampling frequency

The recommended sampling frequency for use in broadcasting applications is 48 kHz as per Recommendation ITU-R BS.646.

4 Validity bit

4.1 Channel validity usage

The validity bit shall be set to logic 0 if the associated audio sample word is suitable for direct conversion to an analogue audio signal, and shall be set to logic 1 if it is not suitable. Where channel status indicates (in byte 0 bit 1 (see Part 3)) that the audio sample word is not in linear PCM form the validity bit shall be set to logic 1 in every subframe.

There is no default state for the validity bit.

4.2 Independent channel validity

Validity shall be set or reset for each and every sample independently in each channel.

5 Pre-emphasis

5.1 Pre-emphasis indication

The use of pre-emphasis, 50 µs pre-emphasis as per Recommendations ITU-R BS.450-3 or ITU-T J.17 pre-emphasis as per ITU-T Recommendation J.17, shall be indicated in channel status as defined in Part 3. Where no pre-emphasis is used, this may be indicated.

NOTE 1 – Positive indication is strongly preferred. The default value will normally be taken to indicate no pre-emphasis, but this condition is undefined.

Part 3

Metadata and subcode

1 Introduction

This Part 3 defines the format for coding metadata, or subcode, relating to the audio content and carried with it.

2 User data format

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

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

Known possible formats for the user data channel are indicated by the channel status byte 1, bits 4 to 7.

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

3 Channel status format

3.1 Channel status bit

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

NOTE 1 – 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.

3.2 Channel status block

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

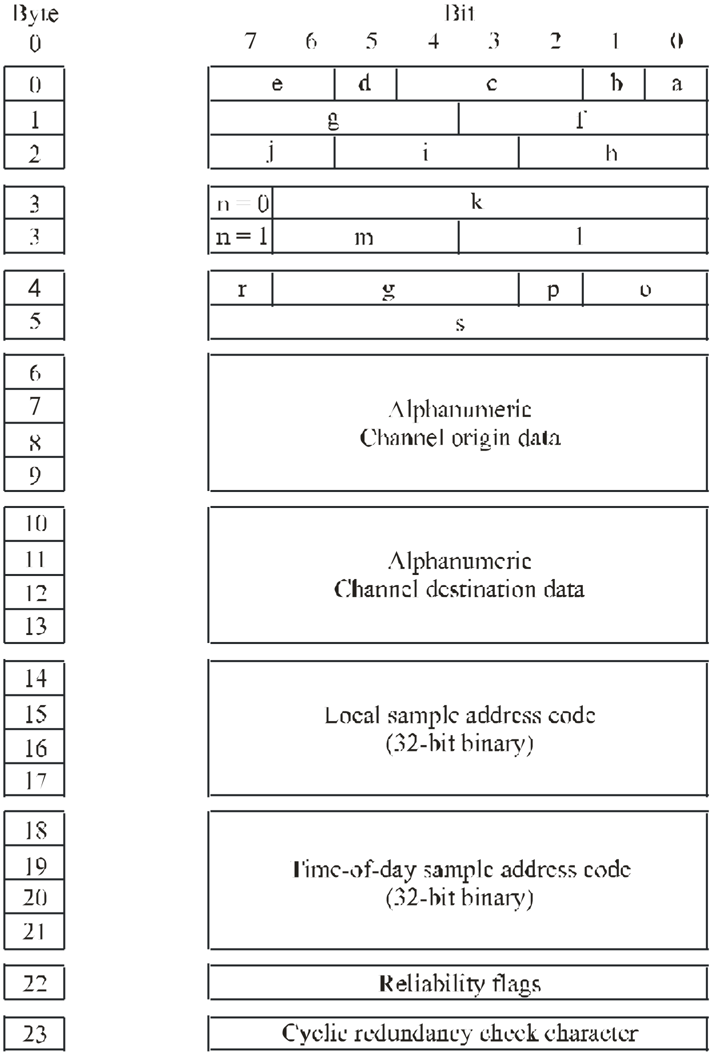
NOTE 1 – In Part 4, the frame that begins with preamble Z contains the first bit of a block in both channels. In other transports a "block start" flag is used to mark the first subframe in a block, and may be applied to each channel independently.

3.3 Channel status content

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

FIGURE 1

Channel status data format



Key:

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 aligment level
k Channel number
l Channel number
m Multichannel mode number
n Multichannel mode
o Digital audio reference signal
p Reserved but undefined
q Sampling frequency
r Sampling frequency scaling flag
s Reserved but undefined

BS.647-01

3.3.1 Byte 0: Basic audio parameters

Bit	0	Use of channel status block
State	0	Consumer use of channel status block (see Note 1)
	1	Professional use of channel status block

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

Bits	4 3 2	Audio signal emphasis
States	0 0 0	Emphasis not indicated. Receiver defaults to no emphasis with manual override enabled
	0 0 1	No emphasis. Receiver manual override is disabled
	0 1 1	50 µs + 15 µs emphasis, see Recommendation ITU-R BS.450. Receiver manual override is disabled
	1 1 1	ITU-T Recommendation J.17 emphasis (with 6.5 dB insertion loss at 800 Hz). Receiver manual override is disabled
	All other states of bits 2 to 4 are reserved and are not to be used until further defined	

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

Bits	7 6	Sampling frequency
States	0 0	Sampling frequency not indicated. Receiver default to interface frame rate and manual override or auto set is enabled
	1 0	48 kHz sampling frequency. Manual override or auto set is disabled
	0 1	44.1 kHz sampling frequency. Manual override or auto set is disabled
	1 1	32 kHz sampling frequency. Manual override or auto set is disabled

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

NOTE 2 – The indication that the audio sample words are not in linear PCM form requires that the validity bit be set for that channel. See § 2.5 in Part 2.

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

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

3.3.2 Byte 1: Channel modes, user bits management

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

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

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

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

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

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

NOTE 1 – The signal coding used for the coordination channel is described in Appendix A to Part 3.

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

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

NOTE 4 – Irrespective of the audio sample word length as indicated by any of the states of bits 3 to 5, the MSB is in time slot 27 of the transmitted subframe as specified in Part 4, § 2.5.

3.3.4 Byte 3: Multichannel modes

Bit	7	Multichannel mode
State	0	Undefined multichannel mode (default)
	1	Defined multichannel modes

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

Either:

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

or,

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

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

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

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

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

Bits	1 0	Digital audio reference signal
States	0 0	Not a reference signal (default)
	1 0	Grade 1 reference signal
	0 1	Grade 2 reference signal
	1 1	Reserved and not to be used until further defined

Bit	2	Information hidden in PCM signal
	0	No indication (default)
	1	Audio sample word contains additional information in the least-significant bits

Bits	6 5 4 3	Sampling frequency
States	0 0 0 0	Not indicated (default)
	0 0 0 1	24 kHz
	0 0 1 0	96 kHz
	0 0 1 1	192 kHz
	0 1 0 0	384 kHz
	0 1 0 1	Reserved
	0 1 1 0	Reserved
	0 1 1 1	Reserved
	1 0 0 0	Reserved for vectoring
	1 0 0 1	22.05 kHz
	1 0 1 0	88.2 kHz
	1 0 1 1	176.4 kHz
	1 1 0 0	352.8 kHz
	1 1 0 1	Reserved
	1 1 1 0	Reserved
	1 1 1 1	User defined

Bit	7	Sampling frequency scaling flag
State	0	No scaling (default)
	1	Sampling frequency is 1/1.001 times that indicated by byte 4 bits 3 to 6, or by byte 0 bits 6 to 7

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

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

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

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

NOTE 5 – The reserved states of bits 3 to 6 of byte 4 are intended for later definition such that bit 6 is set to define rates related to 44.1 kHz, except for state 1 000, and clear to defined rates related to 48 kHz. They should not be used until further defined.

3.3.6 Byte 5: Reserved

Bits	**7 to 0**	**Reserved**
Value	Set to logic 0 until further defined	

3.3.7 Bytes 6 to 9: Alphanumeric channel origin

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

NOTE 1 – ISO 646, IRV, is commonly identified as 7 bit ASCII.

3.3.8 Bytes 10 to 13: Alphanumeric channel destination

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

3.3.9 Bytes 14 to 17: Local sample address code

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

NOTE 1 – This is intended to be set to zero at the start of the recording, for example, and to have the same function as a recording index counter.

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

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

NOTE 1 – This is the time of day laid down during the source encoding of the signal and remains unchanged during subsequent operations. A value of all zeros for the binary sample address code is, for transcoding to real time, or to time codes in particular, to be taken as midnight (that is, 00 h, 00 min, 00 s, 00 frame). Transcoding of the binary number to any conventional time code requires accurate sample frequency information to provide a sample accurate time.

3.3.11 Byte 22: Reserved

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

NOTE 1 – Byte 22 was previously specified to carry a set of reliability flags. Use of this byte is deprecated and is now reserved.

3.3.12 Byte 23: Channel status data CRCC

Bits	7 to 0	Channel status data cyclic redundancy check character
Value	Generating polynomial is $G(x) = x^8 + x^4 + x^3 + x^2 + 1$ The cyclic redundancy check character (CRCC) conveys information to test valid reception of the entire channel status data block (bytes 0 to 22 inclusive). For serial implementations the initial condition of all ones should be used in generating the check bits with the LSB transmitted first. There is no default; this field shall always be coded with a correct CRCC. See § 3.5.2 and Appendix B to Part 3	

3.4 Channel status when non-PCM audio is flagged

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

TABLE 1

Non-PCM audio, protected status bits

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

3.5 Interface format implementation

3.5.1 Implementation levels

3.5.1.1 General

The following two implementations are defined: standard and enhanced. These terms are used to communicate in a simple manner the level of implementation of the interface transmitter involving the many features of channel status. Irrespective of the level of implementation, all reserved states of bits defined in § 3.3 shall remain unchanged.

3.5.1.2 Standard level

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

3.5.1.3 Enhanced level

In addition to conforming to the requirements described in § 3.5.1.2 for the standard implementation, the enhanced implementation shall provide further capabilities.

3.5.2 Transmitter requirement

Transmitters shall encode channel status to follow all the formatting and channel coding rules to one of the two specified implementation levels. All transmitters shall correctly encode and transmit channel status with the correct juxtaposition with respect to the Z preamble or block start (see Part 4).

3.5.3 Receiver requirement

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

NOTE 1– The purpose of the CRCC in byte 23 is to indicate corruption of the channel status block due to switching or editing effects (for example). Due consideration should be given to the implications of any action on downstream equipment and the associated system in general.

3.6 Interface format documentation

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

NOTE 1 – 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.

4 Auxiliary bits

4.1 Availability of auxiliary bits

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

4.2 Availability of auxiliary bits

When these bits are used for any purpose the transmitter shall indicate that use by encoding channel status in byte 2 bits 0, 1 and 2 (see § 3.3.3).

NOTE 1 – A typical use is the addition of audio channels of limited bandwidth and resolution for coordination purposes. This is shown in Appendix A to Part 3.

Appendix A
to Part 3

(Informative)

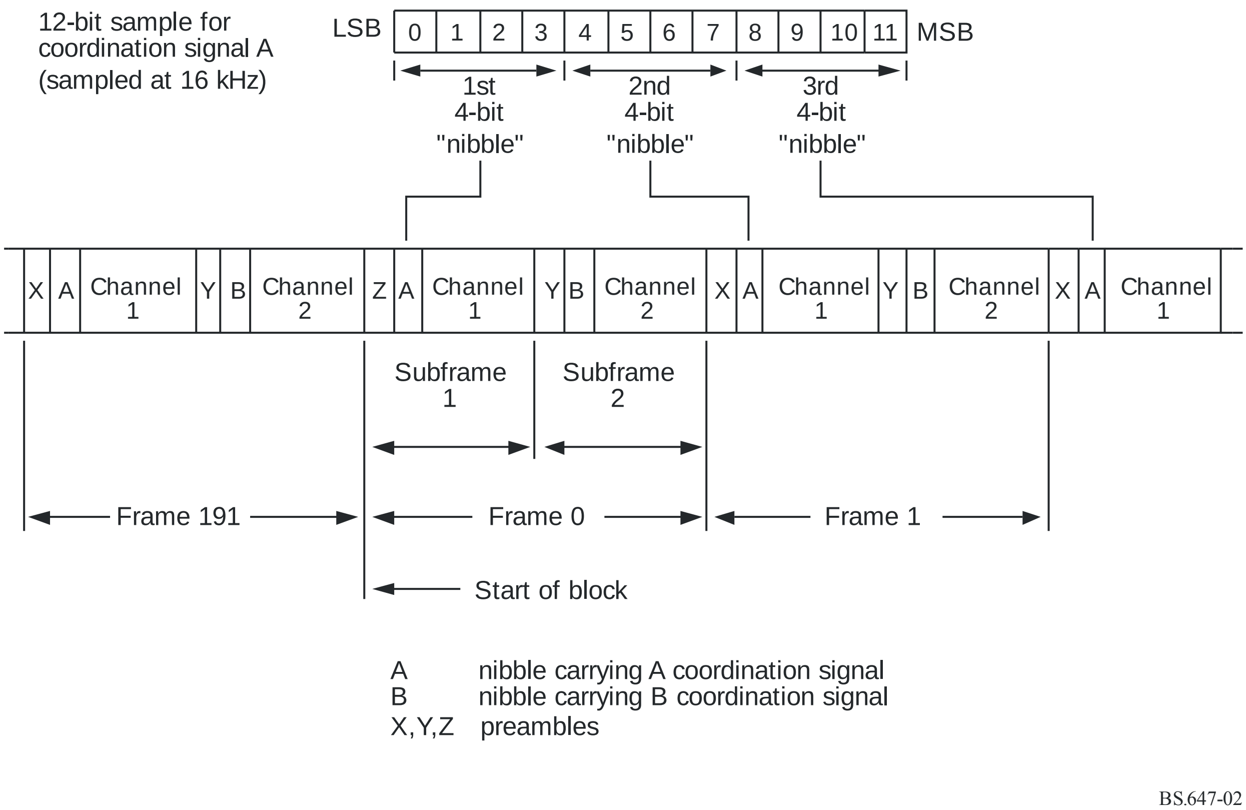
Provision of additional, voice-quality channels

When a 20 bit coding range is sufficient for the audio signal, the 4 auxiliary bits may be used for a voice-quality coordination signal (talk back). This is signalled in byte 2 bits 0, 1 and 2 (see § 3.3.3).

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

FIGURE 2

Frame and block structure



Appendix B
to Part 3

(Informative)

Generation of CRCC (byte 23) for channel status

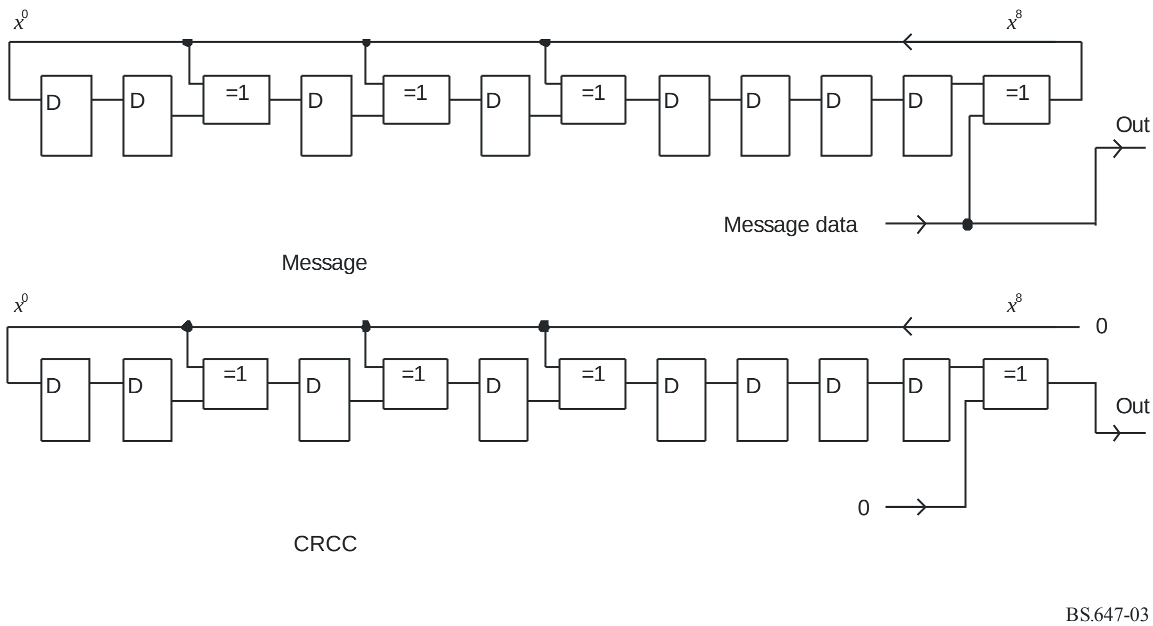
The channel status block format of 192 bits includes a cyclic redundancy check code (CRCC) occupying the last 8 bits of the block (byte 23). The specification for the code is given by the generating polynomial:

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

An example of a hardware realization in the serial form is given in Fig. 3. The initial condition of all stages is logic 1.

FIGURE 3

Frame and block structure



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

Example 1:

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

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

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

Example 2:

Byte	**Bits set to logic 1**
0	0

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

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

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

Part 4

Transport

1 Introduction

This Part 4 defines the format for transport of a digital audio interface.

2 Subframe

2.1 Subframe time slots

Each subframe shall be divided into 32 time slots, numbered from 0 to 31. See Fig. 4. Time slot 0 is transmitted first. Each time slot shall consist of 2 UI.

2.2 Preambles

Time slots 0 to 3, the preambles, shall comprise one of the three permitted preambles designated X, Y and Z. See §§ 5 and 6 and Fig. 7.

2.3 Audio data content

Time slots 4 to 27 shall contain the audio sample word, or some other data such as compressed audio, or some combination of audio and other data (see Part 2 and Part 3 clause 4).

2.4 Sample word orientation

The sample is carried LSB first.

2.5 MSB position

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

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

When a 20 bit coding range is sufficient, the LSB shall be in time slot 8. Time slots 4 to 7 may be used for other applications. Under these circumstances, the bits in time slots 4 to 7 are designated auxiliary sample bits (see Part 3).

2.6 Validity bit

Time slot 28 shall carry the validity bit associated with the audio sample word transmitted in the same subframe (see Part 2).

2.7 User data bit

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

2.8 Channel status bit

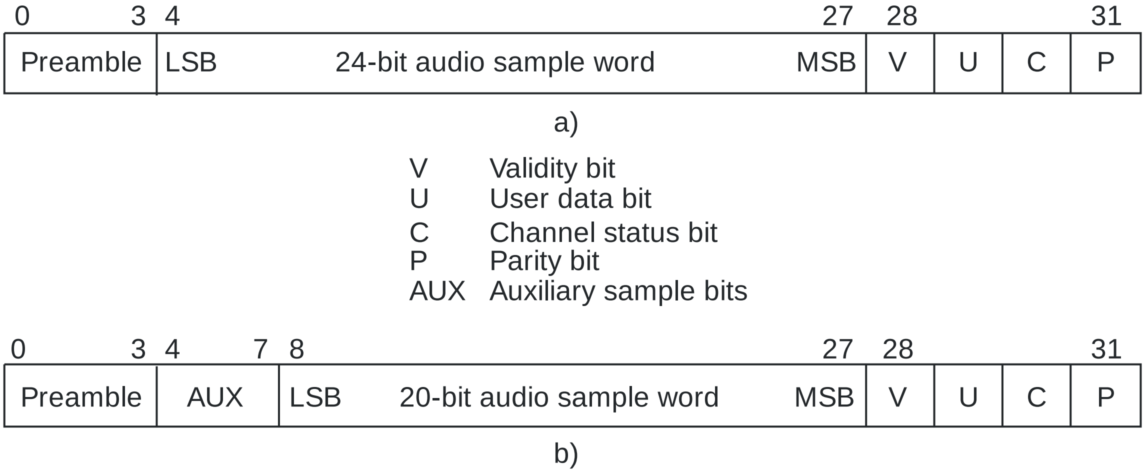
Time slot 30, the channel status bit, shall carry 1 bit of the channel status information associated with the audio channel transmitted in the same subframe (see Part 3).

2.9 Parity bit

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

FIGURE 4

Subframe format



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3 Frame

A frame shall comprise two subframes (see Fig. 5). Except where otherwise specified the rate of transmission of frames corresponds exactly to the source sampling frequency, and in the case of stereophonic signals the two subframes in a frame shall carry samples taken at the same instant.

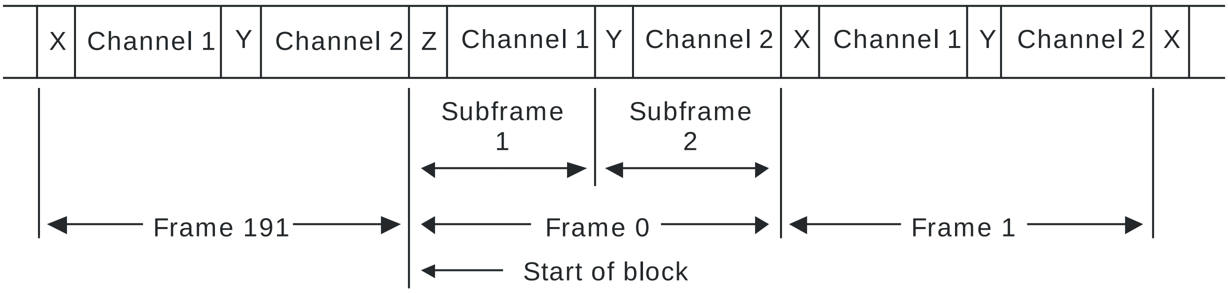
Examples include:

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

NOTE 1 – The modes of transmission are signalled by setting bits 0 to 3 of byte 1 of channel status (see Part 3).

FIGURE 5

Frame format



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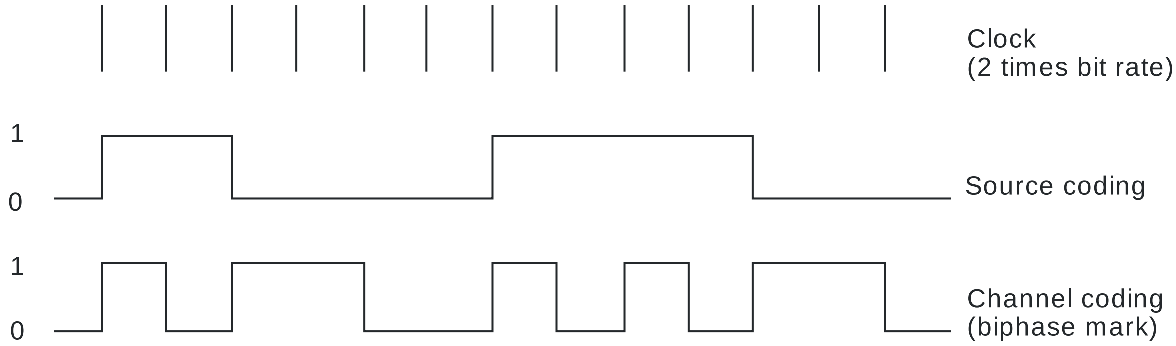
4 Channel coding (line coding)

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

Each bit to be transmitted shall be represented by a symbol comprising two consecutive binary states. The first state of a symbol shall always be different from the second state of the previous symbol. The second state of the symbol shall be identical to the first if the bit to be transmitted is logic 0 and it shall be different if the bit is logic 1. See Fig. 6. Each state shall occupy one unit interval (UI).

FIGURE 6

Channel coding



BS.647-06

NOTE 1 – Biphase-mark coding minimizes the direct-current (DC) component on the transmission line, facilitates clock recovery from the data stream, and makes the interface insensitive to the polarity of connections.

5 Preambles

5.1 Preamble time slots

Time slots 0 to 3 shall be encoded as preambles.

5.2 First subframe preamble

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

5.3 Second subframe preamble

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

NOTE 1 – Preambles are specific patterns providing synchronization and identification of the subframes and blocks. To achieve synchronization within one sampling period and to make this process completely reliable, these patterns violate the biphase-mark code rules, thereby avoiding the possibility of data imitating the preambles. The preambles have even parity as an explicit property.

5.4 Preamble codes

The form of the three types of preamble shall be as shown in the Table 2, represented by eight successive states, occupying four time slots. Figure 7 represents preamble X.

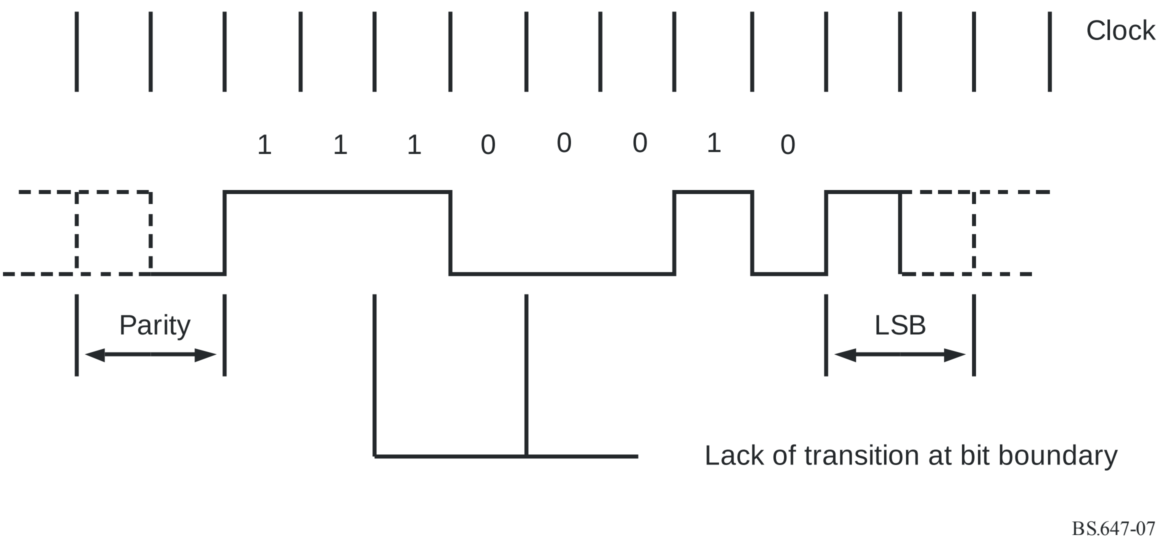
TABLE 2

Preamble codes

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

FIGURE 7

Preamble X (11100010)



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

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

NOTE 3 – The state is always inverted once per time slot plus once per data "one" bit. There are an even number of time slots in a subframe, and, owing to the even-parity bit in time slot 31 (see § 2.9), an even number of "one" bits, so the total number of inversions in any subframe is even. Hence, all preambles will start with the same state. Thus only one of these sets of preambles will, in practice, be transmitted through the interface. However, it is necessary for either set to be decodable in order to maintain immunity to polarity change.

6 Block

A sequence of 192 frames shall be designated a block. The first frame in this sequence shall contain a preamble type Z in place of the type X preamble. The subframes comprising this frame shall contain the first bit of the first byte of the channel status code described in Part 3.

Part 5

Physical and electrical parameters

1 Introduction

This Part 5 specifies the physical and electrical parameters for different media.

The transport format defined in Part 4 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 and at frame rates of up to 50 kHz. Longer cable lengths and higher frame rates 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. Provision is made in this Recommendation for adapting the balanced terminals to use 75 Ω coaxial cable, and transmission by fibre-optic cable is under consideration.

2 Common features

All interfaces shall be subject to the common jitter requirements in § 3. Other parameters shall comply with the transmission type specified.

The interface should use the balanced transmission format specified in Appendix B to Part 5. The interface may use one of the alternative transmission formats specified in subsequent appendices to Part 5.

3 Jitter

3.1 Output interface jitter

3.1.1 General

Jitter at the output of a device shall be measured as the sum of the jitter intrinsic to the device and jitter being passed through from the timing reference of the device.

3.1.2 Intrinsic jitter

The peak value of the intrinsic jitter at the output of the interface, measured at all the transition zero crossings shall be less than 0.025 UI when measured with the intrinsic-jitter measurement filter.

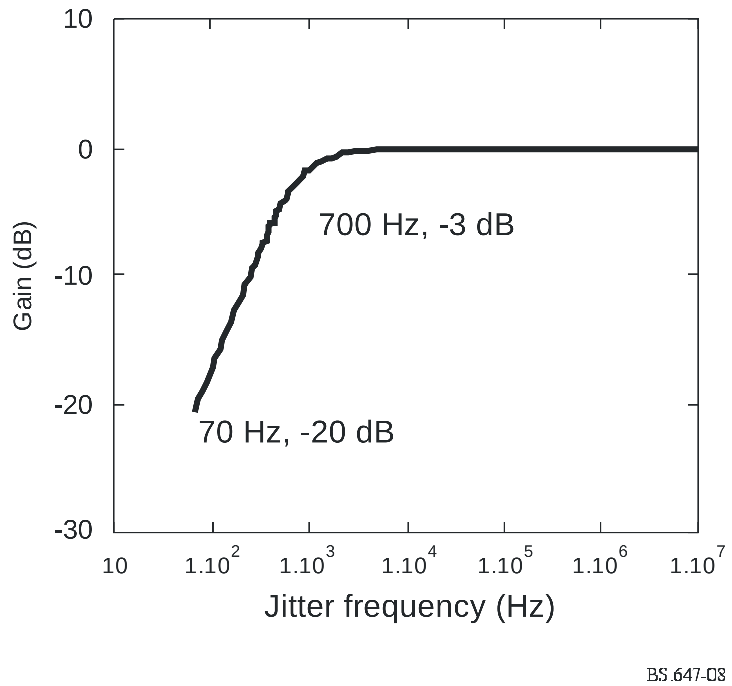
NOTE 1 – This jitter may be strongly asymmetric in character and the deviation from the ideal timing should meet the specification in either direction.

NOTE 2 – This requirement 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.

NOTE 3 – The intrinsic-jitter measurement-filter characteristic is shown in Fig. 8. It shows a minimum-phase high-pass filter with 3 dB attenuation at 700 Hz, a first order roll-off to 70 Hz and with a passband gain of unity.

FIGURE 8

Intrinsic-jitter measurement-filter characteristic



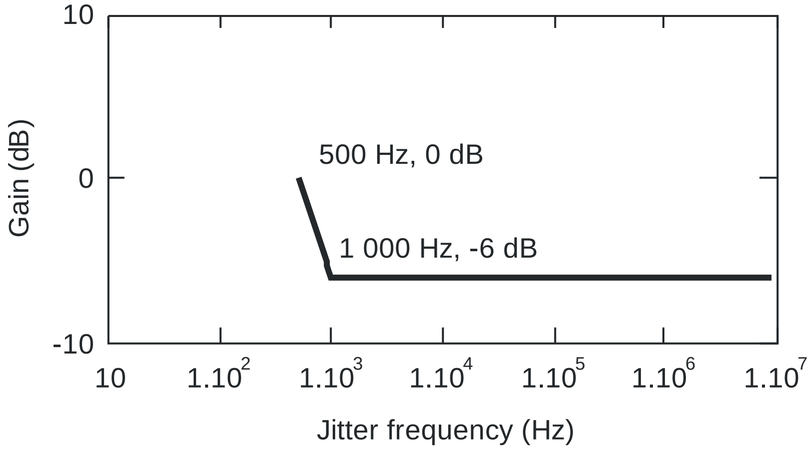
3.1.3 Jitter gain

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

NOTE 1 – If jitter attenuation is provided and it is such that the sinusoidal jitter gain falls below the jitter transfer function mask of Fig. 9 then the equipment specification should state that the equipment jitter attenuation is 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 9

Jitter transfer-function mask



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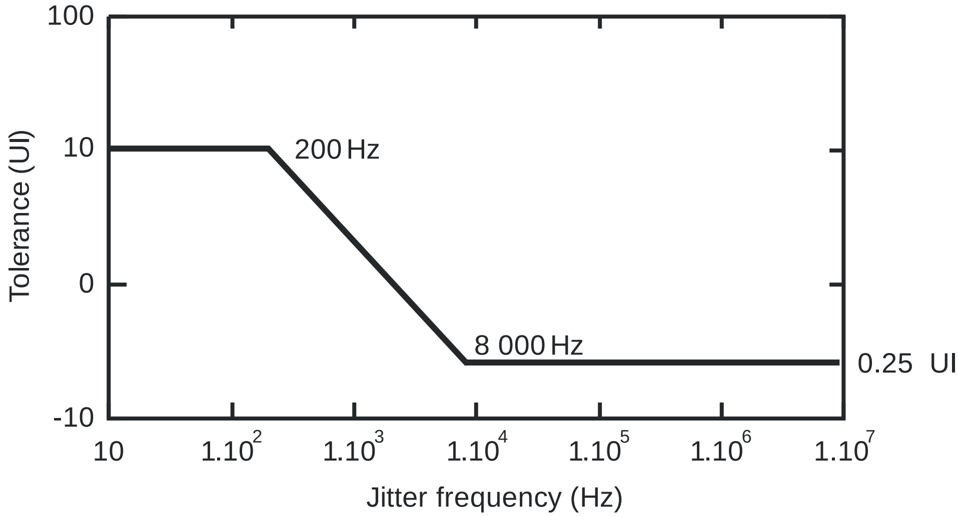
3.2 Receiver jitter tolerance

An interface data receiver should correctly decode an incoming data stream with any sinusoidal jitter defined by the jitter tolerance template of Fig. 10.

NOTE 1 – 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 10

Jitter tolerance template



BS.647-10

Appendix A
to Part 5

(Informative)

Symbol rates and UI

Demands on the performance of the interface are determined by the frame rate, which is in turn determined by the audio sampling frequency. A set of sampling frequencies referred to a basic rate of 48 kHz with options to use 44.1 kHz or 32 kHz is recommended. These basic rates may be scaled by certain multiples to achieve higher or lower sampling frequencies.

The following tables illustrate how the symbol rate at the interface, and the UI, change with different sampling-frequency multiples.

TABLE 3

Symbol rate (MHz) vs. sampling frequency

	Sampling frequency (F_s) kHz		
Multiple	**32**	**44.1**	**48**
0.25	1.024	1.411 2	1.536
0.5	2.048	2.822 4	3.072
1	4.096	5.644 8	6.144
2	8.192	11.289 6	12.288
4	16.384	22.579 2	24.576
8	32.768	45.158 4	49.152

TABLE 4

UI (ns) vs. sampling frequency

	Sampling frequency (F_s) kHz		
Multiple	**32**	**44.1**	**48**
0.25	976.56	708.62	651.04
0.5	488.28	354.31	325.52
1	244.14	177.15	162.76
2	122.07	88.58	81.38
4	61.04	44.29	40.69
8	30.52	22.14	20.35

NOTE 1 – As the sampling frequency is increased, the demand on jitter performance will also increase. For example: a sampling frequency of 8 * 48 kHz (384 kHz) will require an intrinsic jitter of 0.025 * 20.35 ns, or 0.51 ns (see § 3.1.2).

Appendix B
to Part 5

(Informative)

Balanced transmission

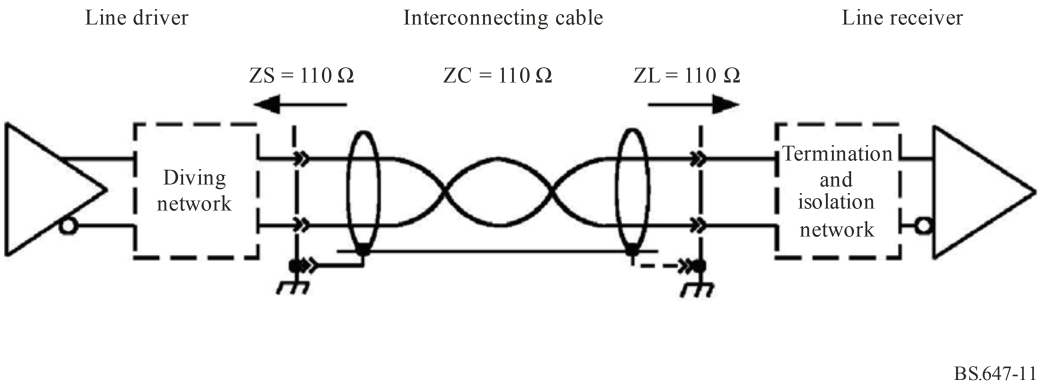
1 General characteristics

1.1 Configuration

A circuit conforming to the general configuration shown in Fig. 11 may be used.

FIGURE 11

Simplified example of the configuration of the circuit (balanced)



NOTE 1 – 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 over cables up to a few hundred metres in length.

1.2 Equalization

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 (about 6 MHz for 48 kHz).

1.3 Cable

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

The cable shall be one of the following types:

– shielded (screened) cable;

– unshielded (unscreened) twisted pair (UTP) structured wiring (Category 5 or better, see ISO/IEC 11801) (see Note 5);

– shielded (screened) twisted pair (STP) structured wiring (see ISO/IEC 11801).

The same cable type shall be used throughout any single interface connection, including patch leads.

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.

NOTE 4 – Care should be taken in design of the interface to provide adequate balance on the twisted pair within the Category 5 cable. Using RJ45 connectors, conventionally wired, current practice favours the use of pins 4 and 5 for Recommendation ITU-R BS.647 signals (separating them from ATM signals on the same cable, for example). Pins 3 and 6 are the preferred second pair. For full protection, the interface may have to withstand power voltages specified to support network equipment, and the use of transformers and blocking capacitors on the interface is strongly recommended.

NOTE 5 – UTP cable has been shown to offer transmission up to 400 m overall unequalized, or 800 m equalized, at 48 kHz frame rate.

2 Line driver characteristics

2.1 Output impedance

The line driver shall have a balanced output with an internal impedance of 110 Ω with a tolerance of 20%, at frequencies from 0.1 MHz to 128 times the maximum frame rate when measured at the output terminals.

2.2 Signal amplitude

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

NOTE 1 – A typical value is 4 V.

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 128 times the maximum frame rate when terminated in a floating load of 110 Ω.

2.4 Rise and fall times

The rise and fall times, determined between the 10% and 90% amplitude points, shall be between 0.03 UI and 0.18 UI when measured across a 110 Ω resistor connected to the output terminals, without any interconnecting cable present.

NOTE 1 – The minimum and maximum rise and fall times for a frame rate of 48 kHz are 5 ns and 30 ns respectively.

NOTE 2 – Operation toward the lower limit of 5 ns may improve the received-signal eye pattern, but may increase electromagnetic radiation at the transmitter. Care should be taken to meet local regulations regarding electromagnetic compatibility (EMC).

3 Line receiver characteristics

3.1 Terminating impedance

The receiver shall present an essentially resistive impedance of 110 Ω with a tolerance of 20% to the interconnecting cable over the frequency band from 0.1 MHz 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.

3.2 Maximum input signals

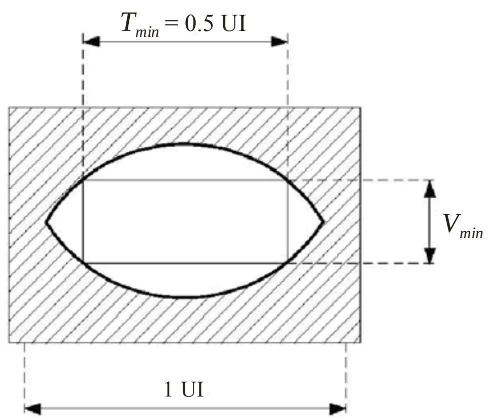
The receiver shall correctly interpret the data when connected directly to a line driver working between the extreme voltage limits specified in § 2.2.

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 Fig. 12.

FIGURE 12

Eye diagram, balanced receiver



T_{min} = 0.5 UI
V_{min} = 200 mV

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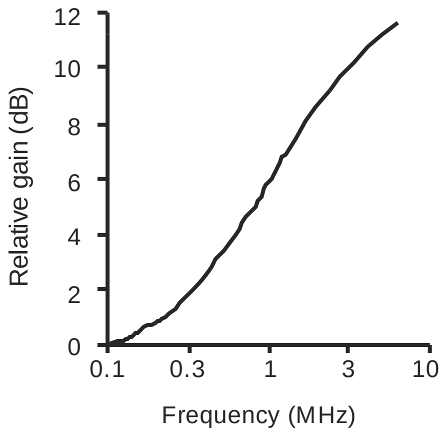
3.4 Receiver equalization

Equalization may be applied in the receiver to enable an interconnecting cable longer than 100 m to be used. A suggested frequency equalization characteristic for operation at frame rates of 48 kHz is shown in Fig. 13. The receiver shall meet the requirements specified in §§ 3.2 and 3.3.

FIGURE 13

Suggested equalizing characteristic for a receiver at 48 kHz frame rate



BS.647-13

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.

4 Connector

4.1 XLR connector

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

NOTE 1 – This type of connector is usually called XLR, or XLR-3.

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 2 – The channel coding means that the relative polarity of pins 2 and 3 is not important. See Part 4, clause 4. However it is recommended that relative polarity is preserved for these signal paths.

4.2 8-way modular connector

Where Category 5 structured cabling is used, the 8-way modular connector specified in IEC 60603-7 (sometimes called “RJ45”) is required. While the interface is by definition insensitive to polarity, for the purposes of constructing adaptors, XLR Pin 2 should be connected to RJ45 Pin 5 (or other odd-numbered pin), XLR Pin 3 should be connected to RJ45 Pin 4 (or even-numbered pin), consistent with using one of the four twisted pairs.

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 analogue signal connector, the abbreviation DI or DO should be used to designate digital audio inputs and outputs, respectively.

Appendix C
to Part 5

(Normative)

Coaxial transmission

The parameters set in this section apply to circuits where balanced equipment is adapted to coaxial cable. Other standards call for more stringent figures where conventional video equipment is used for Recommendation ITU-R BS.647 or less stringent figures where consumer equipment is connected over short distances using screened audio cable (IEC 60958-3).

1 Line driver characteristics

1.1 General

No equalization before transmission shall be permitted.

NOTE 1 – The specification for the line driver (also known as a generator or transmitter) is totally different from the balanced Recommendation ITU-R BS.647 electrical specification and is based on unbalanced coaxial-cable transmission consistent with conventional professional video practice.

1.2 Output impedance

The line driver shall have an unbalanced output circuit having a source impedance of 75 Ω and a return loss better than 15 dB over the frequency band from 0.1 MHz to 128 × frame rate (6.0 MHz in the case of 48 kHz).

1.3 Signal characteristics

The output signal characteristic shall be as shown in Fig. 14 and Table 5 when measured across a resistor connected to the output terminals. The resistor shall have a value of 75 Ω with a relative tolerance of ±1%.

FIGURE 14

Output signal waveform

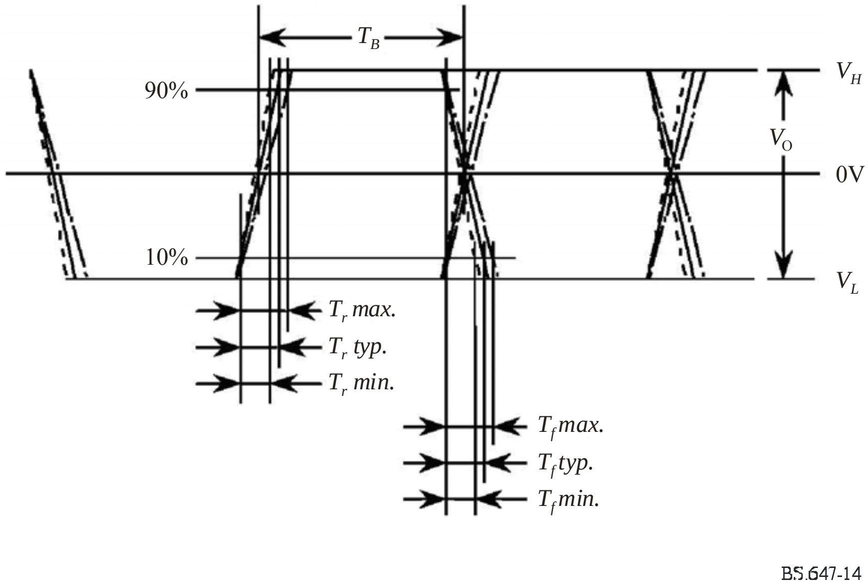


TABLE 5

Output signal characteristics

Parameter	Symbol	Minimum	Typical	Maximum	Unit
Output voltage	$V_O = V_H - V_L$	0.8	1.0	1.2	V
DC offset	$\lvert V_H + V_L \rvert$	–	–	< 50	mV
Rise time	T_r	0.185 (30 ns)	0.225 (37 ns)	0.27 (44 ns)	UI Note 6
Fall time	T_f	0.185 (30 ns)	0.225 (37 ns)	0.27 (44 ns)	UI Note 6
Bit width	T_B	–	1 (163 ns)	–	UI Notes 1, 6

NOTE 1 – Equal to 1/(128 × frame rate).

NOTE 2 – The output voltage is similar to typical analogue video signals.

NOTE 3 – Less DC offset provides a better result for long transmission.

NOTE 4 – The minimum value of the rise and fall times is chosen to restrict the bandwidth of the output signal. When this digital audio signal is fed to a conventional analogue video distribution amplifier (VDA), this specification prevents unnecessary phase distortion of the signal caused by limited bandwidth of the analogue VDA. High frame rates imply high video bandwidths free of phase distortion. Operation toward the lower limit may improve the received-signal eye pattern, but may increase EMC at the transmitter. Care should be taken to meet local regulations regarding EMC.

NOTE 5 – The maximum value of the rise and fall times is chosen with consideration given to the desirability of long-distance (1 000 m) transmission.

NOTE 6 – Figures in (brackets) represent time values where the frame rate is 48 kHz.

2 Coaxial cable characteristics

The interconnecting cable shall be coaxial and have a characteristic impedance of 75 Ω ± 3 Ω over the frequency band from 0.1 MHz to 128 × frame rate (6.0 MHz in the case of 48 kHz). It should be well screened.

3 Line receiver characteristics

3.1 General

Equalization may be used at the receiver.

NOTE 1 – The recovered signal integrity is determined by the signal condition at the end of the terminated cable and the receiver characteristics. Receiver characteristics such as threshold level, hysteresis level, input sensitivity, and so on, depend on the application. The application is defined in part by transmission distance, specific cable used, required noise margin, and performance of the downstream clock-recovery circuitry. If the intention is to preserve the integrity of the signal under various circumstances such that it be identical in all cases, then the requirement for the optimum receiver will differ in each case. Thus, this document establishes only the minimum requirements, rather than specifying the characteristic of every receiver.

3.2 Terminating impedance

The terminating impedance shall be a resistive impedance, at the cable connector, of 75 Ω with return loss of 15 dB or more over the frequency band from 0.1 to 128 × frame rate (6.0 MHz in the case of 48 kHz).

3.3 Maximum input signals

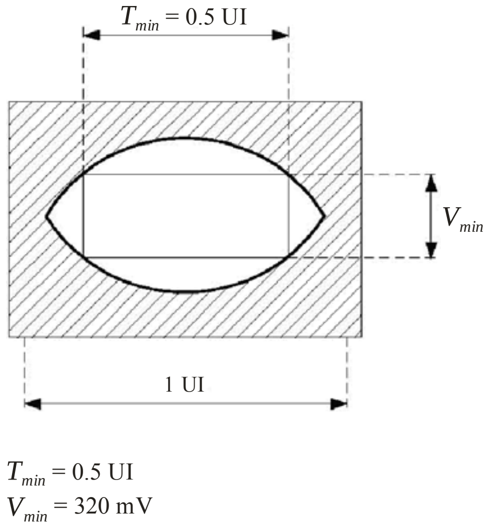
The receiver shall correctly interpret the data when connected directly to a line driver working between the extreme voltage limits specified in § 1.3.

3.4 Minimum input signals

The receiver shall correctly interpret the data when a random signal at the input connector produces the eye diagram characterized by a V_{min} of 320 mV and a T_{min} of 0.5 UI (see Fig. 15).

FIGURE 15

Eye diagram, coaxial receiver

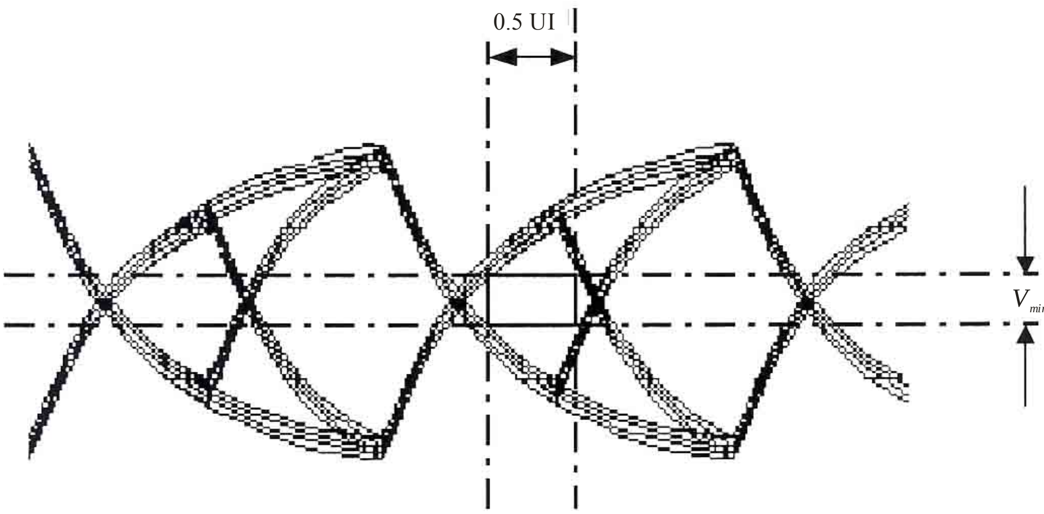


NOTE 1 – This specification is equivalent to one for minimum signal at the terminated BNC connector at the receiving end of the coaxial cable. It is written to maintain compatibility with existing equipment conforming to Appendix B to Part 5 when a resistive pad or transformer impedance-converter, adapting a BNC (75 Ω) connector to the type of XLR connector described in Appendix B to Part 5 (110 Ω), is used to connect the unbalanced coaxial cable to the balanced Recommendation ITU-R BS.647 input.

NOTE 2 – For transmissions beyond 1 000 m, experiments have found the necessity to use a receiver of high sensitivity that can reliably operate with an input signal eye diagram characterized by a V_{min} of 30 mV as shown in Fig. 16.

FIGURE 16

Eye pattern for long-distance transmission



4 Connector

The connector shall have mechanical characteristics conforming to type BNC as described in IEC 61169-8 (2007-2), Part 8.