



Recommendation ITU-R BS.647-3
(03/2011)

**A digital audio interface
for broadcasting studios**

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Broadcasting service (sound)



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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.

* 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 μ s 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. 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

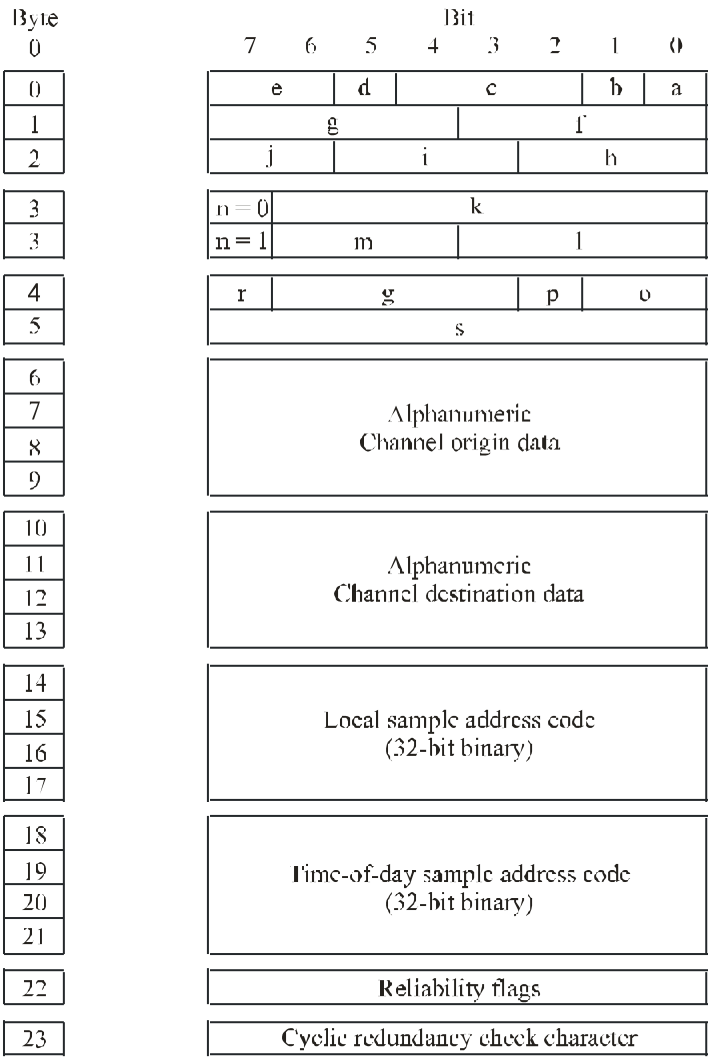
Channel status information shall be organized in 192 bit blocks, subdivided into 8 bit bytes numbered from 0 to 23. The transmission format shall mark every 192nd frame to show that it carries the first bit of a block. Within each byte, the bits are numbered from 0 to 7, 0 being the first bit transmitted, 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 | j | Indication alignment level |
| b | Linear PCM identification | k | Channel number |
| c | Audio signal pre-emphasis | l | Channel number |
| d | Lock indication | m | Multichannel mode number |
| e | Sampling frequency | n | Multichannel mode |
| f | Channel mode | o | Digital audio reference signal |
| g | User bits management | p | Reserved but undefined |
| h | Use of auxiliary sample bits | q | Sampling frequency |
| i | Source word length | r | Sampling frequency scaling flag |
| | | s | Reserved but undefined |

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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 μ s + 15 μ s 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	