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(ex CMTT.724)

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TELEVISION AND SOUND TRANSMISSION

TRANSMISSION OF DIGITAL STUDIO QUALITY SOUND SIGNALS OVER H1 CHANNELS

ITU-T Recommendation J.57

(Formerly Recommendation ITU-R CMTT.724)

FOREWORD

The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the International Telecommunication Union. The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

ITU-T Recommendation J.57 (formerly Recommendation ITU-R CMTT.724) was elaborated by the former ITU-R Study Group CMTT. See Note 1 below.

NOTES

1 As a consequence of a reform process within the International Telecommunication Union (ITU), the CCITT ceased to exist as of 28 February 1993. In its place, the ITU Telecommunication Standardization Sector (ITU-T) was created as of 1 March 1993. Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector (ITU-R).

Conforming to a joint decision by the World Telecommunication Standardization Conference (Helsinki, March 1993) and the Radiocommunication Assembly (Geneva, November 1993), the ITU-R Study Group CMTT was transferred to ITU-T as Study Group 9, except for the satellite news gathering (SNG) study area which was transferred to ITU-R Study Group 4.

2 In this Recommendation, the expression “Administration” is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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TRANSMISSION OF DIGITAL STUDIO QUALITY SOUND SIGNALS OVER H1 CHANNELS *)

(1990)

The CCIR,

CONSIDERING

- (a) that the source encoding for digital sound signals in broadcasting studios is given in Recommendation 646;
- (b) that the 2-channel digital audio interface is specified in Recommendation 647;
- (c) that a format for digital studio-quality sound signal connections should be based on these Recommendations;
- (d) that the sound quality and the auxiliary information carried over the Recommendation 647 interface should be maintained as far as possible;
- (e) that the hierarchical bit rates for digital networks are given in CCITT Recommendation G.702;
- (f) that a digital hierarchy for interworking between networks using different transmission hierarchies is given in CCITT Recommendation G.802;
- (g) that the access levels for the H1 channels in the ISDN are given in CCITT Recommendation I.412;
- (h) that simple interworking between the different hierarchies is necessary;
- (j) that account must be taken of network impairments such as signal bit errors, error bursts and controlled slips;
- (k) that, for some applications, the introduction of excessive delay can cause operational problems,

RECOMMENDS

that for transmission of digital studio-quality sound signals the coding and multiplexing format given in Annex I should be used.

ANNEX I

CODING AND MULTIPLEXING FORMAT

1. Introduction

The transmission format is based on the 2-channel digital audio interface described in Recommendation 647, which should be read in conjunction with this text.

The 32 bits in each sub-frame of the Recommendation 647 interface are treated in the following way in the contribution format:

- Preamble (bit 0-3): not transmitted
- Bit 4-7: not transmitted
- Audio sample word (bit 8-27): companded
- Validity flag (bit 28): not transmitted
- User data (bit 29): transmitted transparently in H12 only
- Channel status (bit 30): subjected to data compression
- Parity bit (bit 31): not transmitted. Replaced by a parity bit for the audio sample word.

¹⁾ Formerly Recommendation ITU-R CMTT.724.

*) The United States of America withholds its approval of this Recommendation.

2. Compatibility between H11 and H12-based systems

The H12 level provides a total of 20 bits per sample, and H11 provides a total of 16 bits per sample. To simplify interworking between H11 and H12 channels, the companding of the audio signal is such that the samples are compressed for transmission in the H11 channel. In the H12 channel, extra bits may be conveyed, to improve resolution of the audio coding and provide a user data channel.

The essential data occupies the entire available capacity of the H11 channel, and the first 24 available octets of each frame of the H12 channel.

3. Coding law

Near instantaneous companding from 20 to 15 bits/sample is applied. A 1 ms companding block is used with 8 coding ranges.

The coding table is shown in Fig. 1a and the transmitted bits in Fig. 1b. Unused bits are set to 1 (one).

The 1 ms companding block introduces a fundamental delay of 2 ms per codec. In practice, the total delay will be slightly longer.

4. Sample error detection

One parity bit is applied to the 7 most significant bits of each transmitted sound sample such that the parity group is odd.

5. Sample interleaving

The companding block contains 96 sound samples (48 from each sound signal). The sound samples within the companding block are organized in 8 successive frames complying with CCITT Recommendation G.704. Each frame contains 6 samples from each sound signal, with the associated parity bits. Adjacent sound samples within the companding block are separated by four Recommendation G.704 frames. This is shown diagrammatically in Fig. 2. The first four frames carry all the odd-numbered samples from both sound signals; the second four frames carry all the even numbered samples.

In the event of an error burst corrupting consecutive bits for a period equivalent to up to four frames, the erroneous samples should be concealed by interpolation between adjacent samples (from part of the block unaffected by the error burst).

6. Signalling in parity [Chambers, 1985]

There are 96 parity bits per 1 ms companding block. Some extra data bits are carried by modifying the parity bits as follows:

6.1 *Scale factor transmission*

Each bit of each 3-bit scale factor word is carried in the parity of 8-sound samples defined in § 6.4, according to the following rule: a scale factor bit which is “0” causes the parity of the 8 samples to be unchanged; a scale factor bit which is “1” modifies the parity. At the decoder, a majority decision process is used to determine the scale factor bits and restore the original parity bit. The samples are then checked for errors in the normal way.

6.2 *Channel status*

Compressed channel status (see § 9) is carried in exactly the same manner as the scale factor bits.

6.3 *Framing signals*

Multiframe alignment signals (MFA) (see § 7) and frame slip detection signals (FSD) (see § 8) are carried by modifying the parity of single samples. These signals do not have the benefit of majority decision decoding, but are inherently predictable and may be decoded reliably.

6.4 *Signalling within the companding block*

The sample-interleaved companding block is represented in Fig. 3 by the rectangle at the bottom of the diagram. Each row within the rectangle represents a Recommendation G.704 frame, and is subdivided into 12 squares representing the 12 samples. The diagram further shows the modification of the parity bits associated with the 12 samples in each frame by the signalling in parity mentioned above.