



INTERNATIONAL TELECOMMUNICATION UNION

# ITU-T

TELECOMMUNICATION  
STANDARDIZATION SECTOR  
OF ITU

## J.55

(ex CMTT.718)

(06/90)

### TELEVISION AND SOUND TRANSMISSION

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## DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS ON DISTRIBUTION CIRCUITS USING 480 kbit/s (496 kbit/s) PER AUDIO CHANNEL

### ITU-T Recommendation J.55

(Formerly Recommendation ITU-R CMTT.718)

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## 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.55 (formerly Recommendation ITU-R CMTT.718) 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 Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector (ITU-R).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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Recommendation J.55<sup>1)</sup>

# **DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS ON DISTRIBUTION CIRCUITS USING 480 kbit/s (496 kbit/s) PER AUDIO CHANNEL**

(1990)

The CCIR,

CONSIDERING

- a) that the distribution of high-quality sound-programme signals from the studio to transmitters and users does not require further audio downstream processing;
- b) that in general more than one high-quality sound-programme signal has to be conveyed over distribution circuits as in the case of DBS;
- c) that a sampling frequency of 32 kHz is recommended for the digital transmission of high-quality sound-programme signals (Recommendation 606);
- d) that some digital broadcasting applications may require system performances that go beyond those offered by equipment complying with CCIR Recommendation 660;
- e) that the high-quality sound-programme signals should interface the ISDN at the H1 level as specified in CCITT Recommendation I.412,

UNANIMOUSLY RECOMMENDS

- 1. that for distribution applications where a sampling frequency of 32 kHz is used and where a dynamic range corresponding to more than 14 bits is required the coding method described in § 1 of Annex I should be used on links providing a BER of less than  $10^{-5}$ ;
- 2. that for transmission at H12 level, two stereophonic programmes or four monophonic programmes should be multiplexed according to the format described in § 2 of Annex I;
- 3. that for cases where a higher ancillary data capacity is required and dedicated 2048 kbit/s links are available, the coding method and multiplexing format described in Annex II should be used.

*Note* – Exchange of international digital sound-programme signals on networks with other hierarchical rates (1544 kbit/s in North America) shall be made using Recommendation 660.

<sup>1)</sup> Formerly Recommendation ITU-R CMTT.718.

## ANNEX I

DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS ON DISTRIBUTION  
CIRCUITS USING 480 kbit/s AUDIO CHANNEL**1. Coding characteristics****1.1 Sampling frequency**

The sampling frequency shall be 32 kHz. The sampling frequency tolerance shall be  $\pm 5 \times 10^{-5}$ , as specified in CCITT Recommendations G.732 and G.733 for primary PCM multiplex equipment. This sampling frequency is consistent with that indicated in CCIR Recommendation 606.

**1.2 Coding method**

16/14-bit floating point companding with a 2 ms coding block length (i.e. 64 consecutive samples per block) and a 3-bit scale factor (transmitted by signalling in parity).

The 16-bit samples of the sound signal are represented in a 2's complement format. The first bit of each word is the MSB (sign bit, 0 ~ +), and the last the LSB. Using a floating point system, the 16-bit samples are converted into 14-bit code words for transmission.

A 3-bit scale factor applying to a block of 64 samples indicates how many of the bits (0 . . . 7) following the sign bit ( $y_1$ ) in all sampled words have the same value as the sign bit (Fig. 1a)). The redundancy indicated by the scale factor does not need to be transmitted. Instead, the samples and their relevant information must be shifted towards the sign bits (floating-point system). This allows the 15th and 16th bits of the source code words to be transmitted in the case of low signal amplitudes. The bits marked Z1 to Z5 have not yet been assigned (Fig. 1b)).

At the receiving end the scale factor is used to shift the bits of the samples back to their original value. This yields 16-bit samples and limits the effects of unrecognized bit errors to the amplitude range indicated by the scale factor.

**1.3 Sample error-protection**

After having applied the floating-point technique for reducing the bits per sample from 16 to 14, a parity bit is calculated on the seven most significant bits of each sample, such that the number of "1"s in the group of the seven protected bits and the parity bit is odd. Thus, the best protection against clicks due to bit errors is guaranteed provided concealment in the form of the arithmetic mean value of the samples adjacent to the faulty sample is performed at the receiving end. If the concealment is done after the 14/16 bit reversion, errors in least significant bits can also be concealed in an optimum way.

**1.4 Ancillary data**

A data capacity of 4 kbit/s per channel is transmitted by signalling in parity.

**1.5 Signalling in parity [Chambers, 1985]**

Signalling in parity is achieved by transmitting the parity bits of an odd number of successive samples without inversion or after inversion, depending on the bit to be signalled. Inversion has to be done, if the bit to be transmitted is 1. The signalling technique used for the scale factor, its related parity bit, and the ancillary data is based on majority-decision logic at the receiver: for each channel, it processes twelve groups of five consecutive samples (three for the scale factor, one for its parity bit and eight for the ancillary data) to recognize simultaneously the odd or even parity of the sample and the data signalled in parity (see Fig. 2). A similar process is used to recognize the synchronization of the 2 ms frames through one group of four consecutive samples.

**1.6 Total bit rate**

With the parameters mentioned above the total bit rate required for one monophonic channel is 480 kbit/s [ $32 \text{ kHz} \times (14 + 1) \text{ bit}$ ].