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ITU-T

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STANDARDIZATION SECTOR
OF ITU

TELEVISION AND SOUND TRANSMISSION

GENERAL PRINCIPLES AND USER REQUIREMENTS FOR THE DIGITAL TRANSMISSION OF HIGH QUALITY SOUND PROGRAMMES

ITU-T Recommendation J.51

(Formerly "Recommendation ITU-R CMTT")

FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). 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, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation J.51 was revised by ITU-T Study Group 9 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 22nd of August 1994.

NOTE

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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SUMMARY

This Recommendation has been developed to help planners of circuits for digital sound-programme transmission. It refers to the coding form and the studio interface, and it lists different codecs recommended for use. Additionally, the Recommendation gives some guidance to designers of future systems on the parameters of a general nature which will need to be specified.

Recommendation J.51[1)]

GENERAL PRINCIPLES AND USER REQUIREMENTS FOR THE DIGITAL TRANSMISSION OF HIGH QUALITY SOUND PROGRAMMES

(1986; revised 1994)

The ITU-T,

considering

(a) that it is desirable to use common standards for the transmission of high-quality sound;

(b) that it is desirable to use as few code conversions as possible in the international digital sound exchange procedure;

(c) that to facilitate the exchange of signals, it is desirable to render the transmission interfaces as transparent as possible to the transmitted message content;

(d) that the network interface bit rates should take into account the hierarchical levels recommended by ITU-T;

(e) that it may be efficient to apply bit-rate reduction and error-protection methods to lower transmission costs;

(f) that quality and availability should not be restricted by the signal processing equipment or methods employed in the transmission circuit;

(g) that there is a need to clearly identify user’s requirements in the development of future systems and in the selection of an existing one for a specific application,

recommends

(1) that a sound programme originating in digital form should preferably be maintained in digital form for transmission;

(2) that the interface between the studio and the codec should, where possible, be in digital form and in accordance to ITU-R Recommendation BS.647;

(3) that parameters listed in Annex A should be used for guidance in the definition of future systems;

(4) that the selection of a transmission system should be made among those recommended by ITU-R for high-quality applications (Annex B lists such Recommendations existing at the end of 1993).

Annex A

Parameters to be used for the definition of future systems

(This annex forms an integral part of this Recommendation)

A.1 Sampling frequency

ITU-R Recommendation BS.646 specifies a sampling frequency of 48 kHz to be used in digital studios. Conversion of sampling frequency leads to a reduction of signal-to-noise ratio and erodes the system bandwidth; it also introduces an additional delay.

For contribution of sound-programme signals of digital studio quality it is recommended to maintain the 48 kHz sampling frequency.

[1)] Formerly Recommendation ITU-R CMTT.659.

For distribution or contribution, if it is not likely that post-production will take place, it may be acceptable to use a 32 kHz sampling frequency for transmission.

A.2 Dynamic range

As a minimum, the contribution system must provide a dynamic range equivalent to 16 bits per sample and if possible should be extendible to 18 bits per sample or more. The present trend seems to be in favour of a higher dynamic range, to reduce operational problems.

A.3 Transmission delay

The user requirements are known to be very short. For example, for monitoring off-air signals with headphones, only 5 to 10 ms may be acceptable. Since the actual delay will be the sum of the coding transmission and decoding delays, it may be difficult to meet this expectation. In that case, the only solution may be to adopt new procedures in the planning and operation of broadcast systems to allow for higher delays.

A.4 Ancillary data

The transmission system should provide capacity for programme associated data. The minimum capacity seems to be in the order of 2 kbit/s per programme. The maximum is defined by ITU-R Recommendation BS.647 (channel status and user data channels).

A.5 Error control

The system should provide error control taking into account the network constraints (e.g. random errors, burst errors, controlled slips, ATM cell losses, short interruptions, etc.).

A.6 ISDN compatibility

The transmission system should be compatible with the user-network interfaces and transmission channels being specified by ITU-T (e.g. H_1, H_0 and multiple B-channels).

In general, the encoding process should be locked to the network clock to avoid the use of justification techniques.

Annex B

Recommendations for digital transmission of high-quality sound programmes existing at the end of 1993

(This annex forms an integral part of this Recommendation)

Recommendation ITU-R	Maximum resolution (bits)	Sampling frequency (kHz)	Channel rate (kbit/s)
CMTT.724 (Note 1)	20	48	1920/1536 for a stereo channel
CMTT.718 (Note 2)	16	32	480/496
CMTT.660 (Note 3)	14	32	384
CMTT.719 (Note 3)	14	32	320

NOTES

1 This type is well-suited for contribution applications. Post-processing of the transmitted signal is possible without restriction, because samples are carried transparently from studio to studio.

2 This type of transmission link can be applied for distribution.

3 These systems are foreseen for use on mixed analogue-and-digital circuits.