INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

J.42

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

TELEVISION AND SOUND TRANSMISSION

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE MEDIUM QUALITY SOUND - PROGRAMME SIGNALS FOR TRANSMISSION ON 384 - kbit/s CHANNELS

ITU-T Recommendation J.42

(Extract from the *Blue Book*)

NOTES

1 ITU-T Recommendation J.42 was published in Fascicle III.6 of the *Blue Book*. This file is an extract from the *Blue Book*. While the presentation and layout of the text might be slightly different from the *Blue Book* version, the contents of the file are identical to the *Blue Book* version and copyright conditions remain unchanged (see below).

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

Recommendation J.42

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE MEDIUM QUALITY SOUND-PROGRAMME SIGNALS FOR TRANSMISSION ON 384-kbit/s CHANNELS

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

1 General

1.1 This Recommendation gives the characteristics of equipment for the coding of 7 kHz monophonic analogue sound-programme signals into a digital signal. Two monophonic digital signals can be combined to form a 384-kbit/s signal already specified in Recommendation J.41.

1.2 Equipment for coding of analogue sound-programme signals, as specified in this Recommendation, can be:

a) A stand alone encoder/decoder with a digital interface at 384 kbit/s. The encoder operation and the decoder operation may be performed in two separate equipments or in the same equipment.

b) A combined encoder-multiplex decoder-demultiplex with a digital interface at 1544 or 2048 kbit/s. The encoder-multiplex operation and the decoder-multiplex operation may be performed in two separate equipments or in the same equipment.

In case b) it is not mandatory to provide an external digital sound-programme access port at 384 kbit/s.

2 Transmission performance

The transmission performance per encoder/decoder pair shall be such that the limits specified in Recommendation J.23 (CCIR Recommendation 503) are not exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

3 Method of encoding

3.1 The recommended encoding laws are as specified in [1].

3.2 These encoding laws are based on a uniformly quantized, 14-bit per sample PCM technique with companding and employ either:

a) eleven-segment 14 to 11 bit instantaneous A-law companding, or

b) five-range 14 to 10 bit near-instantaneous companding.

3.3 Equipment characteristics common to both methods of encoding are:

Nominal audio bandwidth:	0.05 to 7 kHz
Audio interface:	see Recommendation J.23, § 2.
Sampling frequency:	16 $(1 \pm 5 \times 10^{-5})$ kHz.
Pre/de-emphasis:	Recommendation J.17 with 6.5 dB attenuation at 800 Hz.

Note – Pre-emphasis and de-emphasis are not used by the Administrations of Canada, Japan and the United States of America on their national circuits and on international circuits between each other, but are used on international circuits to other countries.

4 Equipment using instantaneous companding

4.1 *Coding table*

4.1.1 The coding law is specified in Table 1/J.41.

4.1.2 The allocation of character signals (PCM code words) is also given in Table 1/J.41. Two variants (A and B) of character signals are allowed.

Note – In the case of digital interconnection between variants A and B, the conversion from one set of character signals to the other set in Table 1/J.41, can be done without any performance degradation. In the case of analogue interconnection, a reduction in the S/N ratio, in the order of 3 dB, is expected.

4.2 *Bit rates*

Nominal source coding bit rate (16 kHz × 11 bit/sample)	176 kbit/s
Error protection (16 kHz × 1 bit/sample)	16 kbit/s
Transmission bit rate per sound-programme signal	192 kbit/s
Channel bit rate for 2 sound-programme signals	384 kbit/s

4.3 *Overload level*

The overload level for a sine-wave signal at zero dB insertion loss frequency (2.1 kHz) of the pre-emphasisis + 15 dBm0s.

4.4 *Digital signal format*

The character signal bit sequences for variants A and B, are shown in Figure 1/J.41.

4.4.1 *Variant A*

When transmitting two monophonic digital signals as one 384 kbit/s signal, with respect to the code word interleaving shown in Figure 1/J.41, the first two 12 bit code words are allocated to 7 kHz channel No. 1 and the second two 12 bit code words are allocated to 7 kHz channel No. 2.

4.4.2 *Variant B*

The 12 bit code word assignments when transmitting two monophonic digital signals as one 384-kbit/s signal is under study.

4.5 *Bit error protection*

One parity bit is added to each 11-bit character signal.

4.5.1 *Variant A*

The five most important bits of each sample are protected against errors by means of a parity bit. In the converter of the transmitting part, the parity bit is added as the 12th bit to each code word. Its value is fixed so that the 6 bit parity block always contains only an odd number of one values. In order that even bit error structures can also result in parity violations, the protected and unprotected bits of each code word are interleaved in ascending and descending sequence, as shown in Figure 1/J.41.

4.5.2 *Variant B*

The added parity bit shall be based on the 7 most significant bits of the 11-bit PCM word. These are bits S, X, Y, Z, A, B, C. The parity of "ones" bit shall be *even.* Since the chord bits (X, Y, Z) always contain a one, the minimum number of ones per sample is 2, resulting in a minimum ones density of 1/6.

4.5.3 *Error concealment*

If a parity violation is detected, an error concealment technique should be applied (for instance, replacement by interpolation, extrapolation or repetition). For multiple parity violation (error bursts), a muting technique should be applied.

4.6 *Digital interface at 384 kbit/s*

Under study (see Recommendations G.735 and G.737).

4.7 *Synchronization*

The coding equipment operates in synchronism with the clock of subsequent multiplex equipment or the network clock. In cases where the digital interface is provided, bit and byte (24 bit, as shown in Figure 1/J.41) timing information is required.

Variant A: A solution for synchronous access is given in the Recommandations G.735 and G.737.

Variant B: The solution for synchronous access is under study.

4.8 *Fault condition and consequent actions*

4.8.1 *Variant A*

Where a 384-kbit/s digital interface is provided, the same principles for fault conditions and subsequent actions as those outlined in Recommendation G.732, should be followed.

4.8.2 *Variant B*

Under study.

5 Equipment using near-instantaneous companding

5.1 *Introduction*

The equipment described in this section uses the near-instantaneous method of companding in the coding of medium quality sound-programme signals into digital form.

A two-stage process is used in the encoding equipment:

a) Conversion of a 7 kHz channel into a 169 kbit/s stream.

Note – The value of 169 kbit/s has been chosen to allow for the possible multiplexing of 12 channels into a 2048 kbit/s dedicated frame format.

b) Asynchronous insertion of two synchronous 169 kbit/s streams into a 384 kbit/s stream.

Note – The asynchronous insertion of two synchronous 169 kbit/s streams into a 384 kbit/s stream allows the use, at the encoder location, of a clock not necessarily synchronous to the network clock. It can be advantageous when the encoder equipment and the insertion equipment (see Recommendations G.735 and G.737) are located in different places, and when the transmission link between them is unidirectional,

and the reverse processes in the decoding equipment.

5.2 *Conversion from 7 kHz to 169 kbit/s and constitution of the 338-kbit/s signal*

5.2.1 *Overload level*

The overload level for a sine-wave signal at the zero dB insertion loss frequency (2.1 kHz) of the pre-emphasis circuit is + 12 dBm0s.

5.2.2 *Companding*

The same near-instantaneous companding procedure with a block of 32 samples (2 ms) as described in § 5.2.2 of Recommendation J.41, is used. The character signal is coded in 2’s complement form.

5.2.3 *Constitution of the 338-kbit/s signal*

Two 7-kHz channels (C1 and C2) are contained in one 338-kbit/s stream. The frame structure of the 338 kbit/s stream is defined in § 5.2.5 and in Figure 3/J.41. The following numbering of the samples within a given multiframe is defined as follows (see Figure 3/J.41):

Sample n of the multiframe is sample $(n - 96i)$ of frame i

$$0 \leq n \leq 191 \qquad\qquad i = 0 \text{ or } 1$$

Using the above notation, the following relationship between the bits of the 338 kbit/s multiframe and channels C1 and C2 can be defined:

Sample $2n$ of the multiframe corresponds to sample n of channel C1

Sample $(2n + 1)$ of the multiframe corresponds to sample n of channel C2

$$0 \leq n \leq 95$$

Range coding information associated with block $(2n - 1)$ of the multiframe is allocated to block n of channel C1 (derived from C1 samples in blocks $(2n - 1)$ and $(2n)$ of the multiframe).

Range coding information associated with block ($2n$) of the multiframe is allocated to block n of channel C2 (derived from C2 samples in blocks ($2n – 1$) and ($2n$) of the multiframe).

$$1 \leq n \leq 3$$

The range coding information and its protection, the sample format and the sample error protection are defined and transmitted as specified in this Recommendation and in §§ 5.2.3 to 5.2.5 of Recommendation J.41.

The criteria for loss and recovery of frame alignment at 338 kbit/s is defined in § 5.2.8 of Recommendation J.41.

5.3 *Conversion from 338 kbit/s to 383 kbit/s*

See Recommendation J.41, § 5.3.

5.4 *Digital interface at 384 kbit/s*

Under study.

5.5 *Fault conditions and consequent action*

Under study.

6 Digital interface between equipments using different coding standards

Under study

References

[1] CCIR Recommandation *Transmission of analogue high-quality sound-programme signals on mixed analogue and digital circuits using 384 kbit/s channels*, Vol. XII, Rec. 660, ITU, Geneva, 1986.