SECTION 4

CHARACTERISTICS OF EQUIPMENTS FOR CODING ANALOGUE SOUND PROGRAMME SIGNALS

Recommendation J.41

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE HIGH 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 15 kHz monophonic analogue sound-programme signals into a digital signal of 384 kbit/s. For stereophonic operation, two monophonic digital codecs can be utilized. Two monophonic digital signals that form a stereophonic signal should be routed together over the same transmission systems (path) to avoid any difference in transmission delay.

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-demultiplex 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.

1.3 Two methods of encoding have been specified by the CMTT [1] and these form the basis for this Recommendation.

2 Transmission performance

The transmission performance per encoder/decoder pair shall be such that the limits specified in Recommendation J.21 (CCIR Recommendation 505) are not exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

Note — When transmitting stereophonic sound programme signals, it is necessary that the encoder and decoder are designed such that they will meet the specified requirements for phase difference.

In order to avoid any unnecessary complexity, the sampling of channels A and B should be performed simultaneously.

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.

For provisional rules for through connection between the two companding methods, see Note 4 in [1].

3.3 Other coding techniques which may be used by bilateral agreement of the Administrations concerned are also listed in Annex A. However, these techniques do not form part of this Recommendation.

3.4 Equipment characteristics common to both methods of encoding are:

Nominal audio bandwidth: 0.04 to 15 kHz.
Audio interface: see Recommendation J.21, § 2.
Sampling frequency (CCIR Recommendation 606): 32 $(1 \pm 5 \times 10^{-5})$ 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 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 in Table 1/J.41 can be implemented without any performance degradation. In the case of analogue interconnection, a small reduction in the S/N ratio, in the order of 3 dB, is expected.

4.2 *Bit rates*

Nominal source coding bit rate (32 kHz × 11 bits/sample) 352 kbit/s
Error protection 32 kbit/s
Transmission bit rate 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-emphasis is +15 dBm0s.

4.4 *Digital signal format*

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

4.5 *Bit error protection*

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

H.T. [T1.41]

TABLE 1/J.41
{
11 segment, 14 to 11 bit instantaneous companding A-law PCM for sound-programme signals (positive half only) | fR↑a↑)
}	

11 bit coding Allocation of character signals }	{

Normalized analogue input	Normalized analogue output	Compressed digital code	Segment No.	Effective resolution (bits)	. 1
Variant A | ua) 2 3 4 5 6 7 8 9 10 11 } Variant B | ub) X Y Z A B C D E F G }	. S	{			
8160 to 8192 1 1 1 1 1 1 1 } 1 1 1 } 4096 to 4128 0 0 0 0 0 0 0 }	8176. 0 4112.	895 1 1 1 1 1 1 1 1 1 0 768 0 0 0 0 0 0 0	1	9	0
4080 to 4096 1 1 1 1 1 1 1 } 1 1 0 } 2048 to 2064 0 0 0 0 0 0 0 }	4088. 0 2056.	767 1 1 1 1 1 1 1 1 0 1 640 0 0 0 0 0 0 0	2	10	0
2040 to 2048 1 1 1 1 1 1 1 } 1 0 1 } 1024 to 1032 0 0 0 0 0 0 0 }	2044. 0 1028.	639 1 1 1 1 1 1 1 1 0 0 512 0 0 0 0 0 0 0	3	11	0
1020 to 1024 1 1 1 1 1 1 1 } 1 0 0 } 512 to 516 0 0 0 0 0 0 0 }	1022. 0 514.	511 1 1 1 1 1 1 1 0 1 1 384 0 0 0 0 0 0 0	4	12	0
510 to 512 1 1 1 1 1 1 1 } 0 1 1 } 256 to 258 0 0 0 0 0 0 0 }	511. 0 257.	383 1 1 1 1 1 1 1 0 1 0 256 0 0 0 0 0 0 0	5	13	0

Tableau 1/J.41 [T1.41] (a l'italienne), p.

4.5.1 *Variant A*

The five most significant 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.

Figure 1/J.41, p.

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 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 high quality sound-programme signals into digital form.

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

a) Conversion of a 15 kHz channel into a 338 kbit/s stream.

Note — The value of 338 kbit/s has been chosen to allow for the possible multiplexing of 6 channels into a 2048 kbit/s dedicated frame format.

b) Asynchronous insertion of the 338 kbit/s stream into a 384 kbit/s stream.

Note — The asynchronous insertion of the 338 kbit/s stream 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 15 kHz to 338 kbit/s*

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*

Near-instantaneous companding is used to achieve a data rate reduction from 14 bits/sample to 10 bits/sample. The system codes a block of 32 samples into one of 5 gain ranges, according to the highest value sample in the block. The companding characteristic is shown diagramatically in Figure 2/J.41 and the parameters are specified in Table 2/J.41.

Figure 2/J.41, p.

5.2.3 *Range coding and protection*

Information defining the range used is transmitted over 3 successive blocks as a 7-bit word, increasing to 11 bits in a Hamming 7, 11 single error correcting code and distributed throughout the 3 blocks as follows:

The five possible values for each of the 3 range codes (one range code for each block in the 3 ms frame; see Figure 3/J.41), are:

Range 4 highest signal level

Range 3

Range 2

Range 1

Range 0 lowest signal level

Range codes generated in this way from three successive blocks are designated Ra, Rb and Rc. They are then used to compute a single 7-bit range code, R, as follows:

$$R = 25Ra + 5Rb + Rc + 1$$

R1 to R7 form the unsigned binary representation of this code which is transmitted LSB first (R1 to R7), followed by 4 protection bits R8 to R11 made up as follows:

R8 = (R7+R6+R5+R4+ R3+R2+R1) MOD 2

R9 = (R7+ R6+R5+R4 +R3+R2+R1) MOD 2

R10 = (R7 +R6 +R5+R4 +R3 +R2+R1) MOD 2

R11 = (R7+R6 +R5 +R4+R3 +R2 +R1) MOD 2

H.T. [T2.41]
TABLE 2/J.41
Companding law — Two's complement coding

Range Compressed digital code MSB LSB }	Normalized analogue input Effective Resolution	Normalized analogue output	{	
4	+8176 to +8192 0 to +16 —16 to 0 —8192 to —8176	+8184 | +8 | —8 | —8184 |	+511 (0111111111) 0 (0000000000) —1 (1111111111) —512 (1000000000)	10 bits
3	+4088 to +4096 0 to +8 —8 to 0 —4096 to —4088	+4092 | +4 | —4 | —4092 |	+511 (0111111111) 0 (0000000000) —1 (1111111111) —512 (1000000000)	11 bits
2	+2044 to +2048 0 to +4 —4 to 0 —2048 to —2044	+2046 | +2 | —2 | —2046 |	+511 (0111111111) 0 (0000000000) —1 (1111111111) —512 (1000000000)	12 bits
1	+1022 to +1024 0 to +2 —2 to 0 —1024 to —1022	+1023 | +1 | —1 | —1023 |	+511 (0111111111) 0 (0000000000) —1 (1111111111) —512 (1000000000)	13 bits
0	+511 to +512 0 to +1 —1 to 0 —512 to —511	+511.5 +0.5 —0.5 —511.5	+511 (0111111111) 0 (0000000000) —1 (1111111111) —512 (1000000000)	14 bits

MSB Most significant bits.

LSB Less significant bits.

Tableau 2/J.41 [T2.41], p.

Figure 3/J.41 (à l'italienne), p.

5.2.4 *Sample error protection*

32 bits per frame are used for sample error detection on the basis of 1 parity bit per 3 samples. Odd parity is employed, i.e., the total number of data bits set to state 1, in the protected samples, plus the parity bit is always an odd number. The distribution of the parity bits within the frame and the allocation of the parity bits to the samples is shown in Figure 3/J.41 and Table 3/J.41, respectively. Only the 5 most significant bits of the samples are protected. In order to ensure that, if two sequential bits are corrupted, the error can still be detected by the parity checking process, the protected and unprotected bits of each sample are interleaved in descending and ascending order, respectively: 1, *10* , 2, *9* , 3, *8* , 4, *7* , 5, *6* . LSB is transmitted first and the bits underlined are those protected by the parity check. Error concealment should be used and can be achieved, for example, by replacing an erroneous sample value by a sample value calculated by linear interpolation between adjacent correct samples, or by extrapolation of the previous sample if the following sample is itself in error.

H.T. [T3.41]

TABLE 3/J.41

Allocation of parity bits to the samples

Parity bit	Protects samples	Parity bit	Protects samples
1	3, 35, 66	17	14, 47, 78
2	8, 39, 71	18	18, 52, 83
3	12, 44, 75	19	23, 58, 89
4	17, 48, 79	20	27, 63, 95
5	21, 53, 84	21	15, 50, 80
6	26, 57, 88	22	22, 56, 85
7	31, 62, 92	23	29, 61, 91
8	19, 51, 82	24	0, 34, 65
9	24, 55, 86	25	5, 40, 70
10	28, 60, 90	26	10, 45, 74
11	32, 64, 94	27	7, 33, 68
12	2, 37, 69	28	13, 38, 76
13	6, 42, 73	29	16, 43, 81
14	11, 46, 77	30	20, 49, 87
15	4, 36, 67	31	25, 54, 93
16	9, 41, 72	32	1, 30, 59

This order has been chosen:

a) to spread each group of 3 protected samples as widely as possible;

b) to spread the 18 or 21 samples protected by each housekeeping word, with the maximum number of other samples between them.

Tableau 3/J.41 [T3.41], p.

5.2.5 *Single channel frame format*

Three 32 sample blocks, together with various housekeeping bits, form a single channel frame having a bit rate of 338 kbit/s and a duration of 3 ms. The number of bits per frame is therefore 3338 = 1014 bits, and these have been allocated as shown in Table 4/J.41. Figure 3/J.41 illustrates the frame arrangement for a single channel. Two frames are shown in Figure 3/J.41 and this format is referred to as a multiframe. Framing information is reversed, i.e. alternate bits in each frame of the multiframe.

5.2.6 *Two channels (stereo-pair) format*

Two separate 338 kbit/s streams are used to form a stereo-pair. Each of these bit streams is arranged as shown in Figure 3/J.41. The coders of the stereo-pair must be in synchronization. Care must be taken at the receiving end to compensate for any phase difference between the 2 channels.

5.2.7 *Synchronization of the 338-kbit/s stream*

The 338 kbit/s stream is synchronized to the coder sampling frequency.

H.T. [T4.41]
TABLE 4/J.41
Bit allocation in the frame

	Frame allocation (bits/frame)	Bit rate per channel (kbit/s)
Sample words	960	320.0
{ Range coding (including error protection) }	11	3.6
Sample word error protection	32	10.6
Signalling	4	1.3
Frame alignment	7	2.3
Total	1014	338.0

Tableau 4/J.41 [T4.41], p.

5.2.8 *Loss and recovery of frame alignment*

One of the following strategies is used:

a) Loss of single channel frame alignment shall occur if two or more consecutive frame alignment words are received incorrectly (for this purpose, bits F1 to F7, Frame 0, and bits F8 to F14, Frame 1, are both considered as frame alignment words: see Figure 3/J.41). An incorrect frame alignment signal is defined as one in which two or more bits are in error. Realignment shall be achieved when a single frame alignment signal is received correctly. If this word is a spurious code, a second attempt at realignment shall be made.

b) Only bits 1 to 10 of the 14 bit frame alignment word, derived from Frame 0 and Frame 1 (see Figure 3/J.41), are taken into account at the receiving end. Loss of frame alignment is assumed to have occurred when three consecutive frame alignment signals are received incorrectly in their predicted position. When frame alignment is assumed to have been lost, the automatic frame alignment recovery device will decide that alignment has been recovered when it registers two consecutive correct frame alignment signals.

5.3 *Conversion from 338 kbit/s to 384 kbit/s*

5.3.1 *Frame structure*

The frame structure (see Figure 4/J.41) with a nominal bit rate of 384 kbit/s and 613 bits in length is composed of:

— data input of 338 kbit/s;

— 63 redundancy bits for single error correction;

— bits for justification (J) and for identifiction of justification (IJ);

— the frame alignment (FA) signal.

The frame is arranged in 4 sections.

5.3.2 *Justification strategy*

The first bits of sections 2, 3 and 4 are used to identify justification.

The 462nd bit of the frame (second bit of the fourth section) is the justification bit.

In cases of justification, the justification bit may assume any value.

Where there is no justification, the position of the justification bit is occupied by an information bit.

On the basis of a majority criterion, the demultiplexer recognizes that justification has taken place, if two out of three justification identification bits are in state 1.

Figure 4/J.41, p.

5.3.3 *Error protection for the 338-kbit/s stream*

A redundancy of 7 bits is calculated every 60 bits (see Figure 4/J.41), to allow for the correction of a single error (Hamming code 67, 60) on reception of each group of 67 bits. The first bit transmitted in a group of 60 bits is considered as the most significant bit of the group for the computation of the redundancy. The first bit transmitted among the 7 redundancy bits represents the most significant bit of the remainder.

The polynomial generator is equal to $x^7 + x + 1$.

5.3.4 *Synchronization of the 384 kbit/s stream*

At the output of the coder, the 384 kbit/s stream is synchronously locked to the subsequent primary hierarchical level digital stream.

5.3.5 *Loss and recovery of frame alignment*

Loss of frame alignment is assumed to have occurred when three consecutive frame alignment signals are incorrectly received in their predicted position. When frame alignment is assumed to have been lost, the automatic frame alignment recovery device will decide that alignment has been recovered when it registers two consecutive correct frame alignment signals.

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 Recommendation *Transmission of analogue high-quality sound-programme signals on mixed analogue and digital circuits using 384 kbit/s channels* , Vol. XII, Rec. 660, UIT, Geneva, 1986.

ANNEX A
(to Recommendation J.41)

Coding methods for use by bilateral agreement

(see § 3.3 of this Recommendation)

H.T. [T5.41]
TABLE A-1/J.41

Nominal bandwidth Pre/de-emphasis Overload point (Note 3) Sampling frequency	0.04-15 (Note 1) (Note 2) +12 32	0.04-15 (Note 1) (None) +12 32	kHz — dBm0s kHz
Companding law Bit rate reduction	13 segments 14/10	7 segments 13/11	— bits

{
Finest resolution and corresponding noise
}
	14 | —66 |	13 —55	bits/sample dBq0ps

{
Coarsest resolution at +9 dBm0s/*f* 0 | ua) and corresponding noise
}
	8 | —30 |	10 —37	bits/sample dBq0ps

{
Resolution at +9 dBm0s/60 Hz and corresponding noise
}
	10 | —42 |	10 —37	bits/sample dBq0ps

Source coding Error protection Framing and signalling Service bit rate Transmission bit rate	320 | 16 | 0.66 336.66 336.66 | ub) 384	352 32 0 384 384	kbit/s kbit/s kbit/s kbit/s kbit/s
Proposed by	Italy	Japan	

a) *f* 0 = zero loss frequency of pre-emphasis.

b) Dedicated frame.

Note 1 — Performance characteristics for analogue 15 kHz type sound-programme circuits are given in Recommendation J.21 and the proposals are assumed to meet these requirements with at least three codecs in tandem.

Note 2 — The pre-emphasis used is:

insertion loss = 10 log

[unable to convert formula]

Note 3 — This is defined as the maximum r.m.s. level of sinusoidal signal which does not cause clipping: this value is independent of frequency if analogue peak limiter and pre-emphasis are removed and replaced by zero dB loss; with pre-emphasis the overload level is defined at the zero dB loss frequency (*f* 0). For detailed information, see Table I in CCIR Report 647.

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Tableau A-1/J.41 [T5.41], p.

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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-emphasis is +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

$$0 \; n \; 191 \quad i = 0 \text{ or } 1$$

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

$$0 \; n \; 95$$

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).

$$1 \; n \; 3$$

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.

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CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE HIGH QUALITY SOUND PROGRAMME SIGNALS FOR TRANSMISSION ON | 320 kbit/s CHANNELS

(Melbourne, 1988)

1 General

1.1 This Recommendation gives the characteristics of equipment for the coding of 15 kHz monophonic analogue sound-programme signals into a digital signal of 320 kbit/s. For stereophonic operation, two monophonic digital codecs can be utilized. Two monophonic digital signals that form a stereophonic signal should be routed together over the same transmission systems (path) to avoid difference in transmission delay.

1.2 Equipment for coding of analogue sound-programme signals can be:

a) A stand-alone encoder/decoder with a digital interface at 320 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-demultiplex operation may be performed in two separate equipments or in the same equipment.

In case b), it is not mandatory to provide an external access at 320 kbit/s.

2 Transmission performance

The transmission performance per encoder/decoder pair shall be such that the limits specified in Recommendation J.21 (CCIR Recommendation 505) are not exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

3 Method of encoding

3.1 The encoding method is based on a uniformly quantized 14-bit per sample PCM technique with differential 14- to 9.5-bit near instantaneous companding.

3.2 Fundamental characteristics of the equipment are:

Nominal audio bandwidth: 0.04 to 15 kHz.

Audio interface: see Recommendation J.21, § 2.

Sampling frequency

(CCIR Recommendation 606): 32 (1 + 5 × 10^{D}lF261^{5}) kHz,

Digital interfaces between Administrations which have adopted different systems should, if a bilateral agreement is not reached, operate at 384 kbit/s (H_0 channel) and carry signals encoded, according to Recommendation J.41, § 4. Any necessary transcoding will be carried out by Administrations using the system specified in this Recommendation.

Pre/de-emphasis: Recommendation J.17 with 6.5 dB attenuation at 800 Hz.

4 Characteristics of the equipment

4.1 *Introduction*

The equipment being described uses the differential near-instantaneous method of companding in the coding of high-quality sound-programme signals into digital form.

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

a) conversion of a 15 kHz channel into a 316 kbit/s stream;

b) asynchronous insertion of the 316 kbit/s stream into a 320 kbit/s stream;

Note — The asynchronous insertion of the 316 kbit/s stream into a 320 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 are located in different places, and when the transmission link between them is unidirectional,

and the reverse processes in the decoding equipment.

4.2 *Conversion from 15 kHz to 316 kbit/s*

4.2.1 *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 or +15 dBm0s.

4.2.2 *Companding*

Differential near-instantaneous companding is used to achieve a data rate reduction from 14 bits/sample to 9.5 bit/sample. The process of differential near-instantaneous companding is subdivided into the following stages:

a) near-instantaneous companding to achieve a data rate reduction from 14 bits/sample to 10 bits/sample as in § 5 of Recommendation J.41. The system coded a bloc of 32 samples into one of 5 gain ranges, according to the highest value sample in the block. The companding characteristic is shown diagramatically in Figure 1/J.43 and the parameters are specified in Table 1/J.43;

Figure 1/J.43, p.10

H.T. [T1.43]

TABLE 1/J.43

14 to 10 bit near-instantaneous companding law

Range Compressed digital code MSB LSB }	Normalized analogue input Effective resolution	Normalized analogue output	{	
4	+8176 to +8192 0 to +16 —16 to 0 —8192 to —8176	+8184 | +8 | —8 | —8184 |	+511 (0111111111) 0 (0000000000) —1 (1000000000) —512 (1111111111)	10 bits
3	+4088 to +4096 0 to +8 —8 to 0 —4096 to —4088	+4092 | +4 | —4 | —4092 |	+511 (0111111111) 0 (0000000000) —1 (1000000000) —512 (1111111111)	11 bits
2	+2044 to +2048 0 to +4 —4 to 0 —2048 to —2044	+2046 | +2 | —2 | —2046 |	+511 (0111111111) 0 (0000000000) —1 (1000000000) —512 (1111111111)	12 bits
1	+1022 to +1024 0 to +2 —2 to 0 —1024 to —1022	+1023 | +1 | —1 | —1023 |	+511 (0111111111) 0 (0000000000) —1 (1000000000) —512 (1111111111)	13 bits
0	+511 to +512 0 to +1 —1 to 0 —512 to —511	+511.5 +0.5 —0.5 —511.5	+511 (0111111111) 0 (0000000000) —1 (1000000000) —512 (1111111111)	14 bits

MSB Most significant bit.

LSB Least significant bit.

Tableau 1/J.43 [T1.43], p.11

b) division of a sequence of samples x (n) into two sequences one of which is a sequence of odd samples x (2 — 1) and the other is a sequence of even samples x (2n). Calculation of differential even samples ?63(2n) by the formula

c) additional near-instantaneous companding of the differential samples ?63(2n) to achieve a data rate reduction from 14 bits/sample to 9 bits/sample. The system codes a block of 16 even samples into one of 3 additional gain ranges, according to the highest value sample in the block. The companding characteristic is shown diagrammatically in Figure 2/J.43 and the parameters are specified in Table 2/J.43.

After multiplexing the odd samples x (2n — 1) represented by a compressed code of 10 bits per sample and the differential even samples ?63(2n) additionally represented by a compressed code of 9 bits per sample, an average of 9.5 bits per sample is obtained.

Figure 2/J.43, p.12

H.T. [1T2.43]

TABLE 2/J.43

14 to 9.0 bit near-instantaneous companding law

Range Compressed digital code MSB LSB }	Normalized input Effective resolution	Normalized output	{		
	2	+16 | 20 to +16 | 84	+16 | 52 |	+255 (011111111)	8 bits
		| 0 to | 64	| 32 |	0 (000000000)	
		{			
| (em64 to | 0					
}	| (em32 |	—1 (100000000)			
		—16 | 84 to —16 | 20	—16 | 52 |	—256 (111111111)	
4	1	| 8160 to | 8192	| 8176 |	+255 (011111111)	9 bits
		| 0 to | 32	| 16 |	0 (000000000)	
		{			
| (em32 to | 0					
}	| (em16 |	—1 (100000000)			
		| (em8190 to | (em8160	| (em8176 |	—256 (111111111)	
	0	| 4080 to | 4096	| 4088 |	+255 (011111111)	10 bits
		| 0 to | 16	| +8 |	0 (000000000)	
		{			
| (em16 to | 0					
}	| —8 |	—1 (100000000)			
		| (em4096 to | (em4080	| (em4088 |	—256 (111111111)	
	2	| 8160 to | 8192	| 8176 |	+255 (011111111)	9 bits
		| 0 to | 32	| 16 |	0 (000000000)	
		{			
| (em32 to | 0					
}	| (em16 |	—1 (100000000)			
		| (em8192 to | (em8160	| (em8176 |	—256 (111111111)	
3	1	| 4080 to | 4096	| 4088 |	+255 (011111111)	10 bits
		| 0 to | 16	| +8 |	0 (000000000)	
		{			
| (em16 to | 0					
}	| —8 |	—1 (100000000)			
		| (em4096 to | (em4080	| (em4088 |	—256 (111111111)	
	0	| 2040 to | 2048	| 2044 |	+255 (011111111)	11 bits
		| 0 to | +8	| +4 |	0 (000000000)	
		{			
| —8 to | 0					
}	| —4 |	—1 (100000000)			
		| (em2048 to | (em2040	| (em2044 |	—256 (111111111)	
	2	| 4080 to | 4096	| 4088 |	+255 (011111111)	10 bits
		| 0 to | 16	| +8 |	0 (000000000)	
		{			
| (em16 to | 0					
}	| —8 |	—1 (100000000)			
		| (em4096 to | (em4080	| (em4088 |	—256 (111111111)	
2	1	| 2040 to | 2048	| 2044 |	+255 (011111111)	11 bits
		| 0 to | +8	| +4 |	0 (000000000)	
		{			
| —8 to | 0					
}	| —4 |	—1 (100000000)			
		| (em2048 to | (em2040	| (em2044 |	—256 (111111111)	
	0	| 1020 to | 1024	| 1022 |	+255 (011111111)	12 bits
		| 0 to | +4	| +2 |	0 (000000000)	
		{			
| —4 to | 0					
}	| —2 |	—1 (100000000)			
		| (em1024 to | (em1020	| (em1022 |	—256 (111111111)	

Tableau 2/J.43 [1T2.43], p.13

H.T. [2T2.43]

TABLE 2/J.43 *(continuation)*

Range Compressed digital code MSB LSB }	Normalized input Effective resolution	Normalized output	{		
	2		2040 to	2048		2044		+255 (011111111)	11 bits
			0 to	+8		+4		0 (000000000)	
		{			
	—8 to	0					
}		—4		—1 (100000000)			
			(em2048 to	(em2040		(em2044		—256 (111111111)	
1	1		1020 to	1024		1022		+255 (011111111)	12 bits
			0 to	+4		+2		0 (000000000)	
		{			
	—4 to	0					
}		—2		—1 (100000000)			
			(em1024 to	(em1020		(em1022		—256 (111111111)	
	0		+510 to	+512		+511		+255 (011111111)	13 bits
			0 to	+2		+1		0 (000000000)	
		{			
	—2 to	0					
}		—1		—1 (100000000)			
		{			
	—512 to	—510					
}		—511		—256 (111111111)			
	2		1020 to	1024		1022		+255 (011111111)	12 bits
			0 to	+4		+2		0 (000000000)	
		{			
	—4 to	0					
}		—2		—1 (100000000)			
			(em1024 to	(em1020		(em1022		—256 (111111111)	
0	1		+510 to	+512		+511		+255 (011111111)	13 bits
			0 to	+2		+1		0 (000000000)	
		{			
	—2 to	0					
}		—1		—1 (100000000)			
		{			
	—512 to	—510					
}		—511		—256 (111111111)			
	0		+255 to	+256		+255.5	+255 (011111111)	14 bits
			0 to	+1		+0.5	0 (000000000)	
		{			
	—1 to	0					
}		—0.5	—1 (100000000)			
		{			
	—256 to	—255					
}		—255.5	—256 (111111111)			

MSB Most significant bit.

LSB Least significant bit.

Tableau 2/J.43 [2T2.43], p.13

4.2.3 *Range coding*

The five possible values of a gain range for a block of 32 samples and three possible values of an additional gain range for differential even samples of this block produce 15 possible values of a complex gain range which is represented by a four-bit code word. Complex range codes are shown in Table 3/J.43.

H.T. [T3.43]
TABLE 3/J.43

Basic Additional	0	1	2	3	4
0	1110	1101	1100	1011	1010
1	1001	1000	0111	0110	0101
2	0100	0011	0010	0001	0000

Tableau 3/J.43 [T3.43], p.

For error-protected transmission, two code words of the complex gain range (which correspond to two blocks) are combined into one 8-bit code word which is coded by a Hamming code (12,8). This code makes it possible to correct all singla errors in the code word of the complex gain range.

A code word of 12 bits comprising 8 bits of the gain range of two blocks and 4 check bits is transmitted in a cycle having a duration of 2 ms (ee Figure 3/J.43). The first 8 bits R1 to R8 correspond to two complex code words. The last four bits (R9 to R12) are check bits. They are determined as follows:

Modulo 2 addition is designated by $\oplus$ and inversion of bit R is designated by R .

Figure 3/J.43, p.

4.2.4 *Sample error protection*

The 5 most significant bits of 10-bit samples and 4 most significant bits of 9-digit samples are protected. One parity bit is generated for 5 most significant bits of each 10-digit sample. A parity bit is also generated for 4 most significant bits of each pair of 9-digit samples. A total of 24 bits are thus generated for a block of 32 samples. These 24 parity bits undergo error protection by means of a cyclic code (29,24). The code (29,24) is a shortened Hamming code (31,26). The polynomial generator of the code (29,24) is:

$$F(x) = x^5 + x^2 + 1 \tag{3}$$

To the receiving end only the check bits of the cyclic code (29,24) are sent, since 24 parity bits are reproduced according to the received sample. Thus, 5 protection bits correspond to a block of 32 samples, 10 protection bits for two blocks are transmitted in a cycle having a duration of 2 ms (see Figure 3/J.43).

In order to correct 8-bit error bursts, samples from four blocks are interleaved. Interleaving of samples from four blocks is shown in Table 6/J.43.

Note — Interleaving of samples from four adjacent blocks is an effective measure of error protection. Samples of a sound-programme signal are transmitted over the primary digital path in octets (8-bit words). Such samples interleaving ensures correction of erroneous octets.

4.2.5 *316 kbit/s channel frame*

The frame has a duration of 2 ms which corresponds to two 32-sample blocks. The frame duration of 2 ms equals to the multiframe duration of the primary digital multiplex equipment. Due to this coincidence of durations a possibility is provided to use the multiframe alignment signal of the primary digital multiplex equipment. With a digital rate of 316 kbit/s and a duration of 2 ms, the frame comprises 632 bits divided into 8 groups of 79 bits each. Bit allocation in the frame is shown in Table 4/J.43.

H.T. [T4.43]
TABLE 4/J.43
Bit allocation in the frame

	Frame allocation (bits/frame)	Bit rate per channel (kbit/s)
Samples	608	304
Range code	8	4
Check bits of a range code	4	2
Check bits of samples	10	5
Signalling and data bits	2	1
Total	632	316

Tableau 4/J.43 [T4.43], p.

The frame structure is shown in Figure 3/J.43 and Table 5/J.43. Table 6/J.43 shows the allocation of sample bits in a group, which provides for interleaving of samples from four blocks (see § 4.2.4 above) and interleaving of bits from different samples.

Note — As can ben seen from Table 6/J.43, an 8-bit error burst disintegrates into isolated single errors. For example, when errors occur in bits 1 to 8 of the first group (1 = 1) of the N-th frame, errors appear in the next four samples: the first sample of the first block frame N — 1 (n = 1, k = 1), the second sample of the second block of frame N — 1 (n = 2, k = 2), the second sample of the first block of frame N — 2 (n = 2, k = 1), the first sample of the second block of frame N — 2 (n = 1, k = 2). These isolated errors are corrected by means of interpolation.

H.T. [T5.43]

TABLE 5/J.43

316 kbit/s frame structure

Data type	Bit number in a group	Group number in a cycle
Sample bits	1-38; 41 to 78	1 to 8
{ Bits of the code words of the complex gain range of the 1st block (R1 to R4) }	39	1 to 4
{ Bits of the code words of the complex gain range of the 2nd block (R5 to R8) }	39	5 to 8
{ Check bits of two complex gain ranges (R9 to R12) }	79	2, 4, 6, 8
{ Check bits of the samples of the 1st block (R1 to R5) }	blanc 40 79	blanc 1, 3, 4 1, 3 |
{ Check bits of the samples of the 2nd block (R6 to R10) }	blanc 40 79	blanc 5, 7, 8 5, 7 |
Signalling and check bits (S)	40	2 | |
Data bits (D)	40	6 | |

Tableau 5/J.43 [T5.43], p.17

H.T. [T6.43]

TABLE 6/J.43

{ Bit number in sample n of block k } Bit number in group 1 of frame N }	{							
N — 1 n = 41—3	N — 2 n = 41—1	k = 1 n = 41—2	k = 2 n = 41	k = 1 n = 41—2	k = 2 n = 41	n = 41—3	n = 41—1	
1.6		1.6		1.6		1.6		1 to 8
2.7		2.7		2.7		2.7		9 to 16
3.8		3.8		3.8		3.8		17 to 24
4.9		4.9		4.9		4.9		25 to 32
5.10		5 |		5 |		5.10		33 to 38
	1.6		1.6		1.6		1.6	41 to 48
	2.7		2.7		2.7		2.7	49 to 56
	3.8		3.8		3.8		3.8	57 to 64
	4.9		4.9		4.9		4.9	65 to 72
	5.10		5 |		5 |		5.10	73 to 78

N Number of the current frame: N = 0, ± | , ± | , . | |

1 Number of the group in the frame: 1 = 1, 2, . | | , 8

k Number of the block in the frame: k = 1, 2 n Number of the sample in the block: $n = 1, 2, . | | , 32$

Tableau 6/J.43 [T6.43], p.18

4.2.6 *Synchronization of the 316 kbit/s stream*

The 316 kbit/s stream is synchronized to the coder sampling frequency.

4.2.7 *Frame alignment of the 316/s stream*

For the frame alignment the synchronizing properties of the Hamming code (12,8) are utilized and a special frame alignment signal is not employed. The signal R1-R12 is used as a frame alignment signal. In the frame alignment signal receiver the relationships (2) from § 4.2.3 are checked. The lock-in time of such a frame alignment signal is equal to the lock-in time of an 4-bit frame alignment signal.

4.3 *Asynchronous insertion of the 316 kbit/s signal into a 320 kbit/s stream*

4.3.1 *Frame structure of the 320 kbit/s signal*

The 320 kbit/s signal is composed of a data signal fo 316 kbit/s and a justification signal of 4 kbit/s. The 320 kbit/s stream is divided into groups of 80 bits, 79 bits being data bits and the 80th bit being the bit of the justification signal.

4.3.2 *Justification method*

A method of positive-negative justification with two-command control is used for the rate justification. The justification signal consists of justification commands and a data signal transmitted in the case of negative justification. The frame of the justification signal consists of 4 bits. The justification commands are transmitted by three bits 111 or 000. The same commands are used for frame alignment of the justification signal. The 4th bit in the frame is used to transmit a data signal in the case of negative justification.

4.3.3 *Allocation of the justification signal in the frame of the primary digital multiplex equipment*

Bits of the justification signal are allocated in the frames of the primary digital multiplex equipment, which comprise the frame alignment signal in the channel time slot 0.

In the frame of the primary digital multiplex equipment, which comprises the justification bit, this bit is the last of all bits of the 320 kbit/s signal which are allocated in the given frame, that is, the justification bit is the most remote bit from the frame alignment signal of the primary digital multiplex equipment.

4.4 *Digital interface between the encoder equipment and the insertion equipment*

Under study.

4.5 *Fault conditions and consequent actions*

Under study.

5 **Digital interface between equipments using different coding standards**

Under study.

Recommendation J.44

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE MEDIUM | QUALITY

SOUND-PROGRAMME SIGNALS FOR TRANSMISSION ON 320 kbit/s | CHANNELS

(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 320 kbit/s signal having a structure specified in Recommendation J.43.

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 320 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-demultiplex operation may be performed in two separate equipments or in the same equipment.

In case b) it is not mandatory to provide an external access at 320 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 exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

3 Method of encoding

3.1 The encoding method is based on a uniformly quantized 14-bit per sample technique with differential 14 to 9.5-bit near instantaneous companding

3.2 Fundamental characteristics of the equipment are:

Nominal audio bandwidth: 0.05 to 7 kHz.

Audio interface: see Recommendation J.23, § 2.

Sampling frequency: 16 (1 ± 5 × 10^{D}lF261^{5}) kHz.

Pre/de-emphasis: Recommendation J.17 with 6.5 dB attenuation at 800 Hz.

Digital interface between Administrations which have adopted different systems should, if a bilateral agreement is not reached, operate at 384 kbit/s (H_0 channel) and carry signals encoded according to Recommendation J.42, § 4. Any necessary transcoding will be carried out by Administrations using the system specified in this Recommendation.

4 Characteristics of the equipment

4.1 *Introduction*

The equipment described in this section uses the differential 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 158 kbit/s stream;

b) asynchronous insertion of two synchronous in-phase 158 kbit/s streams into a 320 kbit/s stream.

Note — The asynchronous insertion of two asynchronous in-phase 158 kbit/s streams into a 320 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 are located in different places, and when the transmission link between them is unidirectional,

and the reverse processes in the decoding equipment.

4.2 *Conversion from 7 kHz to 158 kbit/s and constitution of the 316 kbit/s signal*

4.2.1 *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 or +15 dBm0s.

4.2.2 *Companding*

The same differential near-instantaneous companding procedure with a block of 32 samples (2 ms), as described in § 4.2.2 of Recommendation J.43, is used.

4.2.3 *Range coding*

The same range coding for a block of 32 samples (2 ms), as described in § 4.2.3 of Recommendation J.43, is used.

4.2.4 *Sample error protection*

The same sample error protection for a block of 32 samples (2 ms), as described in § 4.2.4 of Recommendation J.43, is used.

4.2.5 *316 kbit/s channel frame*

Two 7 kHz channels (C1 and C2) are contained in one 316 kbit/s stream. The frame structure of the 316 kbit/s stream is described in § 4.2.5 of Recommendation J.43. The first block (k = 1) of each frame corresponds to channel C1 and the second block (k = 2) of each frame corresponds to channel C2.

4.3 *Asynchronous insertion of the 316 kbit/s signal into a 320 kbit/s stream*

See § 4.3 of Recommendation J.43.

4.4 *Digital interface between the encoder equipment and the insertion equipment*

Under study.

4.5 *Fault conditions and consequent actions*

Under study.

5 Digital interface between equipments using different coding standards

Under study.

SECTION 5

Section 5 has not yet been allocated.

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SECTION 6

CHARACTERISTICS OF CIRCUITS FOR TELEVISION TRANSMISSIONS

Former Recommendations J.61 and J.62 of Volume III-2 of the *Orange Book* have been cancelled. The corresponding CCIR Recommendations have been combined into CCIR Recommendation 567, which refers to all television standards and colour systems. This Recommendation 567 and some other texts from CCIR may be very useful for television transmissions via cable, and reference is given to the following CCIR Recommendations, published in Volume XII (of the XV Plenary Assembly of the CCIR), ITU, Geneva, 1982.

Recommendation J.61

TRANSMISSION PERFORMANCE OF TELEVISION CIRCUITS

DESIGNED FOR USE IN INTERNATIONAL CONNECTIONS

(Geneva, 1982)

(See CCIR Recommendation 567)

Recommendation J.62

SINGLE VALUE OF THE SIGNAL-TO-NOISE RATIO

FOR ALL TELEVISION SYSTEMS

(Geneva, 1982)

(See CCIR Recommendation 568)

Recommendation J.63

INSERTION OF TEST SIGNALS IN THE FIELD-BLANKING INTERVAL OF MONOCHROME AND COLOUR TELEVISION SIGNALS

(Geneva, 1982)

(See CCIR Recommendation 473)

Recommendation J.64

DEFINITIONS OF PARAMETERS FOR SIMPLIFIED AUTOMATIC MEASUREMENT OF TELEVISION INSERTION TEST SIGNALS

(Geneva, 1982)

(See CCIR Recommendation 569)

Recommendation J.65

STANDARD TEST SIGNAL FOR CONVENTIONAL LOADING OF A TELEVISION CHANNEL

(Geneva, 1982)

(See CCIR Recommendation 570)

Recommendation J.66

TRANSMISSION OF ONE SOUND PROGRAMME ASSOCIATED WITH ANALOGUE TELEVISION SIGNAL BY MEANS OF TIME DIVISION MULTIPLEX IN THE LINE SYNCHRONIZING PULSE

(Geneva, 1982)

(See CCIR Recommendation 572)

SECTION 7

GENERAL CHARACTERISTICS OF SYSTEMS FOR TELEVISION TRANSMISSION

OVER METALLIC LINES AND INTERCONNECTION WITH RADIO-RELAY LINKS

Recommendation J.73

USE OF A 12-MHz SYSTEM FOR THE SIMULTANEOUS TRANSMISSION OF

TELEPHONY AND TELEVISION

(amended at Geneva, 1964 and 1980)

The 12-MHz system on 2.6/9.5-mm coaxial cable pairs and the 12-MHz system on 1.2/4.4-mm coaxial pairs are defined in Recommendations G.332 [1] and G.345 [2] respectively.

Any 12-MHz system equipped for television transmission should be capable of transmitting the signals used in all the television systems defined in CCIR having a video bandwidth up to 5.5 MHz if necessary, by means of the switching (in terminal equipments only) of certain components.

1 Carrier frequency

The CCITT recommends the use of a carrier frequency of 6799 kHz with a tolerance of ± | 00 Hz for the transmission of all the television signals indicated above. The video band transmitted over the cable should be 5.5 MHz wide, whatever television system is to be used. The level recommended for this carrier has been defined for the interconnection points and is shown in Figures 1/J.73 and 2/J.73 (see Note 3 to these figures).

2 Modulation ratio

Amplitude modulation has to be used. The modulation ratio has to be higher than 100% (as indicated in Figure 3/J.73), so that, when the carrier is modulated by a signal corresponding to blanking level, its amplitude be equal to that of the carrier when it is modulated by a signal corresponding to the white level, assuming that the d.c. component is transmitted.

When a luminance bar (see CCIR Recommendation 567, Annex 1 to Part C, test signal element B2) is applied at a video junction point, the nominal peak voltage of the modulated carrier, at a point where the relative level for the television transmission is zero, should be as follows:

— for white or blanking level, 0.387 volt (i.e. the peak voltage of a sine-wave signal dissipating a power of 1 mW in a resistance of 75 ohms);

— for the synchronizing signals, 0.719 volt (i.e. the peak voltage of a sine-wave signal dissipating a power of 3.45 mW in a 75 ohm resistance).

Recommendations J.71 and J.72 of Volume III-2 of the *Orange Book* have been deleted.

Figure 1/J.73, p.

Figure 2/J.73, p.

Figure 3/J.73, p.

3 Vestigial-sideband shaping

The shaping of the vestigial-sideband signal has to be carried out entirely at the transmit point. Provisionally, the vestigial sideband should not exceed a width of 500 kHz. Figure 4/J.73 shows the frequency arrangement recommended for television transmission over the 12 MHz system.

Figure 4/J.73, p.

4 Relative power levels and interconnection at a frontier section

It is not possible to recommend relative power levels at the output of intermediate repeaters since they are very closely linked to the inherent design of each Administration's system.

When interconnection between two telephone systems is effected via a cable section that crosses a frontier, in accordance with Recommendation G.352 [3], each Administration should accept, on the receiving side, the level conditions which normally apply to the incoming system used in the other country. It may be possible to comply with this condition simply by insertion of a correcting network at the receiving end. The repeater section crossing the frontier, should then be less than 4.5 km long, the details being agreed directly between the Administrations concerned before the repeater stations are sited.

Where a line is to be used alternatively for ‘‘all-telephony’’ or for ‘‘telephony-plus-television’’, such a solution is not generally applicable. In this case, one of the frontier stations may act as a main station having the necessary types of pre-emphasis and de-emphasis networks to permit interconnection at flat points at the recommended levels. Figure 1/J.73 shows how this may be done in the general case and also shows how, at terminal stations, the same interconnections levels are used when connecting the line to telephony and television translating equipment.

However, if a common differential characteristic can be agreed for all types of 12-MHz line, then free interconnection of the full line-bandwidth becomes possible, both nationally (e.g. between working and spare lines) and internationally (between national systems of different designs). This method leads to the simpler interconnection arrangement of Figure 2/J.73.

In this arrangement, the circuit is always lined up for ‘‘all-telephony’’. For telephony-plus-television, the emphasis characteristic used for the ‘‘all-telephony’’ case is modified by the insertion, at the terminal equipment stations only, of differential pre-emphasis and de-emphasis networks additional to those used for ‘‘all-telephony’’ transmission.

5 Interference

Recommendation J.61 (equal to CCIR-Recommendation 567, Part D), indicates the overall values relative to the hypothetical reference circuit for television transmissions which are taken as objectives for design projects.

In the experience of certain Administrations, the weighted psophometric power can be distributed between the terminal equipment and the line in the ratio of 1 to 4.

In particular, the Administration of the Federal Republic of Germany uses, for the 12 MHz system, the following signal-to-weighted noise ratio:

— for terminal modulation equipment: 70 dB

— for terminal demodulation equipment: 64 dB

— for a line 840 km in length: 58 dB

These values result in a signal-to-noise ratio of 52 dB at the end of the reference circuit.

References

[1] CCITT Recommendation *12-MHz systems on standardized 2.6/9.5-mm coaxial cable pairs* , Vol. III, Rec. G.332.

[2] CCITT Recommendation *12-MHz systems on standardized 1.2/4.4-mm coaxial cable pairs* , Vol. III, Rec. G.345.

[3] CCITT Recommendation *Interconnection of coaxial carrier systems of different designs* , Vol. III, Rec. G.352.

Recommendation J.74

METHODS FOR MEASURING THE TRANSMISSION CHARACTERISTICS OF TRANSLATING EQUIPMENTS

1 No special measuring method is necessary for the carrier.

2 An oscilloscope can be used, for example, to measure the modulation ratio.

3 No special method is recommended for measuring pre-emphasis.

4 An oscilloscope can be used, for example, to measure the voltages at the input to the modulating equipment and the output from the demodulating equipment.

5 The following is an example of a method which can be used to measure the random noise at the modulator output:

The input and output video terminals of the modulator are closed with 75 ohm resistances and the modulator is set to give an output carrier power of 1 mW. The random noise power can then be measured with a selective measuring instrument, and the result is given relative to the video-frequency bandwidth for the television system concerned.

To measure noise produced by the demodulator, 1 mW of carrier power is sent to its input, and the random noise at the output is measured at the output terminals with a selective measuring instrument.

This method can also be used to measure parasitic noise having a recurrent waveform.

Note — Methods for measuring parasitic noise in television are being studied.

Recommendation J.75

INTERCONNECTION OF SYSTEMS FOR TELEVISION TRANSMISSION ON COAXIAL PAIRS AND ON RADIO-RELAY LINKS

1 Television transmission only

Direct video transmission over long, e.g. more than about 15 km, coaxial cables is unsatisfactory, because of the likelihood of picking up interference and the difficulties of low-frequency equalization; it is therefore necessary to transmit the television signal as a modulated carrier transmission, usually with a vestigial sideband.

On the other hand, the television signal can be transmitted directly in the baseband of a radio-relay system as a video signal. In general it is advantageous to do so, since this minimizes distortion and enables a better signal-to-noise ratio to be obtained as compared with a modulated signal with vestigial sideband, transmitted in the baseband. This procedure is recommended by the CCIR.

Interconnection between television channels on radio-relay and cable systems will therefore normally take place at video frequencies.

Levels and impedances at interconnection points should then conform to Recommendation J.61.

Exceptionally, in special cases, the video signal can be transmitted over short cables, or a vestigial-sideband television signal can be transmitted on short radio-relay links, to allow direct interconnection at line frequencies (radio-relay link baseband). Special arrangements may be necessary in such cases in respect of signal level, pre-emphasis and pilots, to maintain the recommended standard of transmission performance.

2 Telephony and television transmission, alternatively or simultaneously, on coaxial pairs or radio-relay links

2.1 *Interconnection between a coaxial cable system having alternative transmission of telephony and television and a radio-relay link with the same alternative transmission*

It is recommended that the following conditions should be met at the interconnection point:

— For telephony transmission, the frequency arrangements, the relative power levels of the telephone channels and the frequency of the pilots should be as indicated in Recommendation G.423 [1].

— For television transmission, interconnection should generally be made at video frequencies. Levels and impedances at interconnection points should then conform to Recommendation J.61.

2.2 *Interconnection between a coaxial system having simultaneous telephony and television transmission and a radio-relay link with the same simultaneous transmission*

On all radio-relay links designed for such simultaneous transmission, it is intended to transmit video-frequency television signals in the lower part of the baseband and telephony signals in the upper part. Since these arrangements are incompatible with those which are recommended by the CCITT for simultaneous telephony and television transmission on coaxial cables (Recommendation J.73), it will normally be possible to consider interconnection at video frequencies only for the television channel, and interconnection at group, supergroup, mastergroup or supermastergroup points for telephony.

However, by agreement between the Administrations concerned, direct interconnection may be achieved, in special cases, on a short system (on cable or radio), by using a frequency allocation recommended for the other type of system.

Reference

[1] CCITT Recommendation *Interconnection at the baseband frequencies of frequency-division multiplex radio-relay systems* , Vol. III, Fascicle III.2, Rec. G.423.

Recommendation J.77

CHARACTERISTICS OF THE TELEVISION SIGNALS

TRANSMITTED OVER 18 MHz AND 60-MHz SYSTEMS

(Geneva, 1980)

For television transmission on 18 MHz and 60 MHz systems, a modulation procedure has to be used which is independent of the structure of the signal to be transmitted. This is achieved by a reference carrier which defines the phase relationship between the transmit and receive side.

The transmission channel is capable of transmitting the signals used in all those television systems defined by the CCIR, in accordance with Report 624 [1].

The requirements to be met by the 18 MHz and 60 MHz transmission systems are to be found in Recommendations G.334 [2] and G.333 [3].

It is recommended that the following conditions be met:

1 Vestigial sideband shaping

The shaping of the vestigial sideband signal has to be carried out entirely at the transmit side. The vestigial sideband shall not exceed a width of 1 MHz, i.e. the width of the Nyquist slope shall not exceed 2 MHz.

2 Video pre-emphasis

Recommendation J.76 of Volume III-2 of the *Orange Book* has been deleted.

With regard to a more uniform loading of the coaxial line systems, it is recommended to use a video pre-emphasis network. The video pre-emphasis curve and the corresponding formula are shown in Figure 1/J.77. The video pre-emphasis amounts to 9 dB.

3 Nominal reference level of the modulated video signal

As a consequence of using a video pre-emphasis network, it is necessary to define a reference level at a suitable video frequency. It is recommended that this reference level be derived from the level of a single sideband measured after the Nyquist filter when a 1 kHz sine wave is transmitted, having a peak-to-peak amplitude of 0.7 volt at the video interconnection point. The reference level is this measured level plus 6 dB. The reference level is recommended to be +11 dBm0.

Figure 1/J.77, p.

4 Accuracy of carrier frequencies

The carrier frequency of the first modulation stage should have a tolerance not exceeding 11 Hz. Tolerances of the carrier frequencies for the higher modulation stages can be ignored if either Recommendation G.225 [4] is met, or if the carriers are derived from the relevant TV channel-pair pilots (see [5] and [6]).

5 Reference carrier

In order to enable accurate demodulation of the signal at the receive side, it is necessary to transmit a reference carrier.

The following characteristics are recommended:

— carrier frequency of the first modulation stage corresponding to the video frequency of 0 Hz;

— polarity negative, i.e. such that the amplitude of the modulated video signal is greater at black than at white;

— nominal power level: +10 dBm0, independent of signal level.

6 Low frequency suppression

In order to prevent disturbance of the reference carrier by the low frequency components of the video signal, it is necessary to reduce the level of the low frequency components. A low frequency suppression of 18 dB is recommended. The low frequency suppression curve and the corresponding formula are shown in Figure 1/J.77.

References

[1] CCIR Report *Characteristics of television systems* , Vol. XI, Report 624, ITU, Geneva, 1982.

[2] CCITT Recommendation *18-MHz systems on standardized 2.6/9.5-mm coaxial pairs* , Vol. III, Rec. G.334.

[3] CCITT Recommendation *60-MHz systems on standardized 2.6/9.5-mm coaxial cable pairs* , Vol. III, Rec. G.333.

[4] CCITT Recommendation *Recommendations relating to the accuracy of carrier frequencies* , Vol. III, Rec. G.225.

[5] CCITT Recommendation *60-MHz systems on standardized 2.6/9.5-mm coaxial cable pairs* , Vol. III, Rec. G.333, § 8.4, Note 2.

[6] CCITT Recommendation, *18-MHz systems on standardized 2.6/9.5-mm coaxial pairs* , Vol. III, Rec. G.334, § 9.4.2, Note.

Blanc

PART III

SUPPLEMENTS TO H AND J SERIES RECOMMENDATIONS

Blanc

MONTAGE: PAGE 234 = PAGE BLANCHE

Supplement No. 5

MEASUREMENT OF THE LOAD OF TELEPHONE CIRCUITS

UNDER FIELD CONDITIONS

(Referred to in Recommendations G.223 and H.51 this supplement is to be

found on page 295 of Fascicle III.2 of the *Red Book* , Geneva, 1985)

Supplement No. 12

INTELLIGIBILITY OF CROSSTALK BETWEEN TELEPHONE AND

SOUND-PROGRAMME CIRCUITS

(Referred to in Recommendation J.32; this supplement is to be found

on page 610 of Fascicle III.2 of the *Green Book* , Geneva, 1972.)

Supplement No. 16

OUT-OF-BAND CHARACTERISTICS OF SIGNALS APPLIED

TO LEASED TELEPHONE-TYPE CIRCUITS

(Referred to in Recommendation H.51; this Supplement is to be found on page 191

of
Fascicle III.4 of the *Red Book* , Geneva, 1985)