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

ITU-T Recommendation J.41

(Extract from the *Blue Book*)

NOTES

1 ITU-T Recommendation J.41 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.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.

TABLE 1/J.41

11 segment, 14 to 11 bit instantaneous companding A-law PCM for sound-programme signals (positive half only)[a)]

Normalized analogue input	Normalized analogue output	Compressed digital code	Segment No.	Effective resolution (bits)	11 bit coding Allocation of character signals Variant A[b)] 1	 2 3 4	 5 6 7 8 9 10	 11	 Variant B[c)] S	 X Y Z	 A B C D E F G
8160 to 8192	8176	895	1	9	0	1 1 1	1 1 1 1 1 1	1	0	1 1 0	1 1 1 1 1 1 1
4096 to 4128	4112	768					0 0 0 0 0 0	0			0 0 0 0 0 0 0
4080 to 4096	4088	767	2	10	0	1 1 0	1 1 1 1 1 1	1	0	1 0 1	1 1 1 1 1 1 1
2048 to 2064	2056	640					0 0 0 0 0 0	0			0 0 0 0 0 0 0
2040 to 2048	2044	639	3	11	0	1 0 1	1 1 1 1 1 1	1	0	1 0 0	1 1 1 1 1 1 1
1024 to 1032	1028	512					0 0 0 0 0 0	0			0 0 0 0 0 0 0
1020 to 1024	1022	511	4	12	0	1 0 0	1 1 1 1 1 1	1	0	0 1 1	1 1 1 1 1 1 1
512 to 516	514	384					0 0 0 0 0 0	0			0 0 0 0 0 0 0
510 to 512	511	383	5	13	0	0 1 1	1 1 1 1 1 1	1	0	0 1 0	1 1 1 1 1 1 1
256 to 258	257	256					0 0 0 0 0 0	0			0 0 0 0 0 0 0
255 to 256	255.5	255	6	14	0	0 1 0	1 1 1 1 1 1	1	0	0 0 1	1 1 1 1 1 1 1
128 to 129	128.5	128					0 0 0 0 0 0	0			0 0 0 0 0 0 0
127 to 128	127.5	127			0	1 0 0	1 1 1 1 1 1	X	0	0 0 0	1 1 1 1 1 1 1
0 to 1	0.5	0				0	0 0 0 0 0 0	0			0 0 0 0 0 0 0

X 11th bit freely available in variant A.

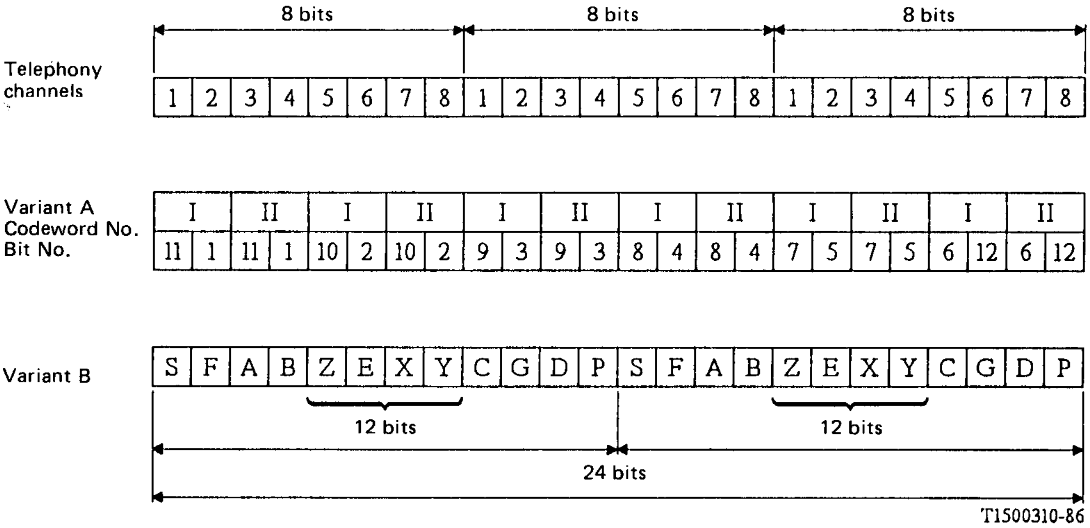
a) Character signals for the negative half are the same as those for the positive half except that the sign bits (bit 1 and S for variants A and B respe ctively) are inverted.

b) Variant A is presently used with digital equipment based on a 2048 kbit/s hierarchy. After coding and before the parity bit is inserted, bits 1 to 5 are inverted.

c) Variant B is presently used with digital equipment based on a 1544 kbit/s digital hierarchy. All bits, including the parity bit, are inverted and reformatted before transmission (see Figure 1/J.41).

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.



Variant A: BIT definitions:

1	Sign bit
2, 3, 4	Chord bits
5 to 11	Step bits
12	Parity bits

Variant B: BIT definitions:

S	Sign bit
X, Y, Z	Chord (since chord 1 1 1 is not used, and bits are inverted on the line, one of these bits will always be a one)
A to G	Step
P	Parity bit
	One of these 4 bits will always be a one (see chord above)

FIGURE 1/J.41

15 kHz sound programme channel bit sequences for transmission on A-law companded systems

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 Recommendations 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.

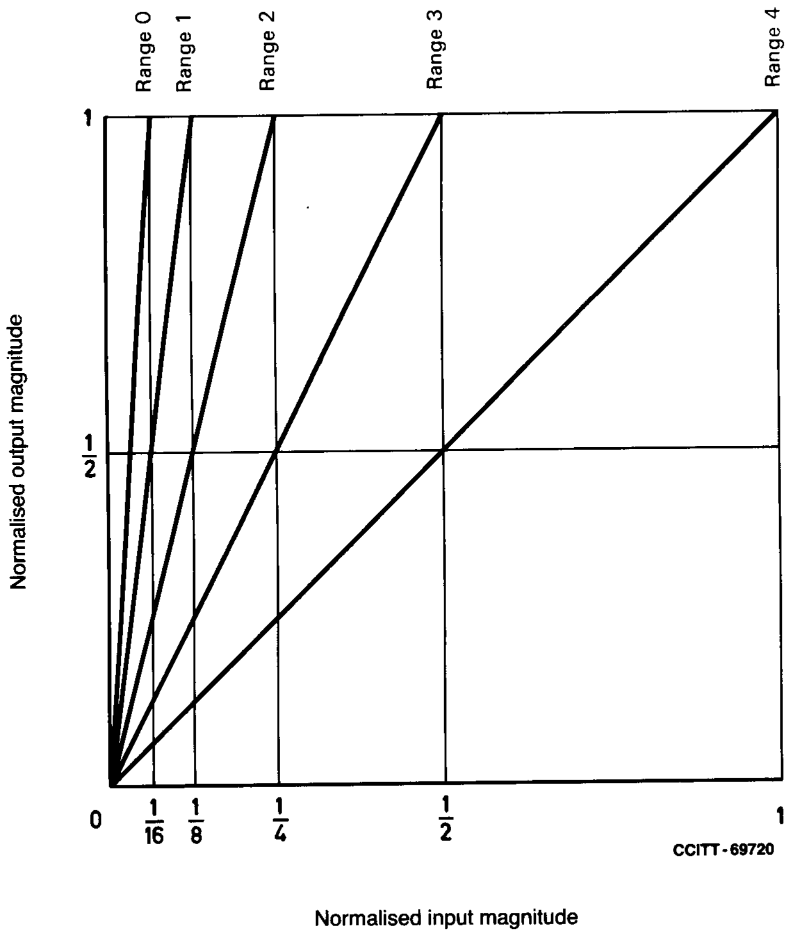


FIGURE 2/J.41
Near-instantaneous companding characteristic

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:

$$\begin{aligned}
R8 &= (R3 + R2 + R1) \text{ MOD } 2\\
R9 &= (R6 + R5 + R4) \text{ MOD } 2\\
R10 &= (R7 + R5 + R4 + R2 + R1) \text{ MOD } 2\\
R11 &= (R7 + R6 + R4 + R3 + R1) \text{ MOD } 2
\end{aligned}$$

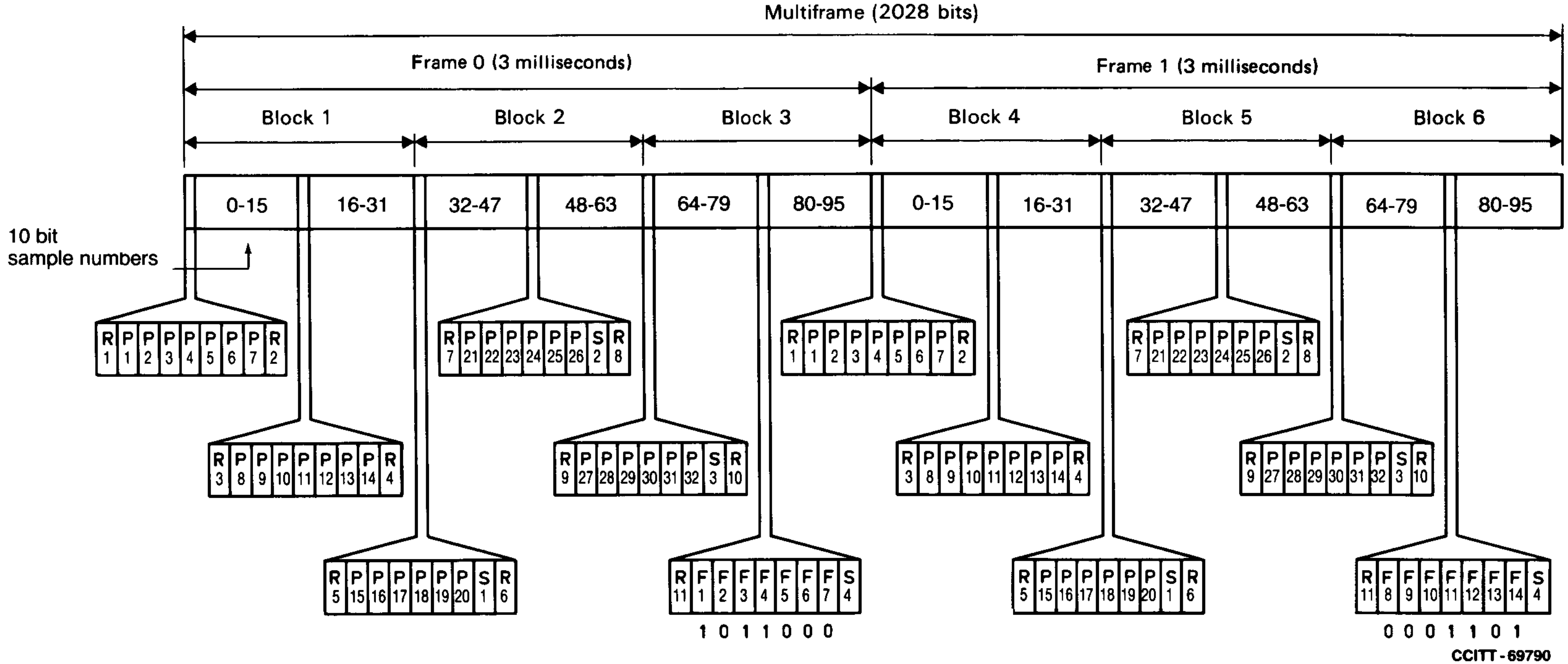
TABLE 2/J.41

Companding law – Two's complement coding

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

MSB Most significant bits.

LSB Less significant bits.



F Framing bit
P Sample parity bit
R Range code bit
S Signalling bit

FIGURE 3/J.41

Single channel frame format

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.

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.

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.

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

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 identification 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.

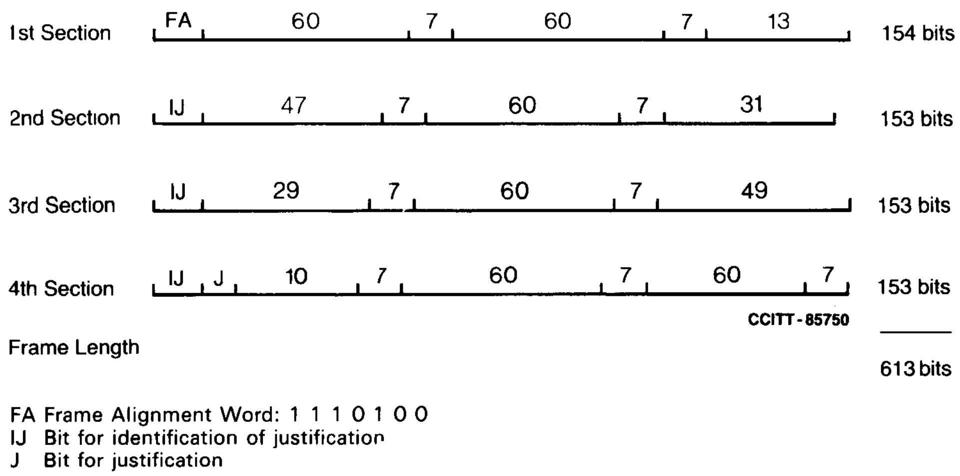


FIGURE 4/J.41

338 kbit/s to 384 kbit/s frame structure

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)

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[a] 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[b] 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:

$$\text{insertion loss} = 10 \log \frac{8.5 + \left(\frac{f}{1900}\right)^2}{1 + \left(\frac{f}{650}\right)^2} \quad (f \text{ in Hz with } f_0 = 1900 \text{ Hz}).$$

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.