INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

J.43

TELEVISION AND SOUND TRANSMISSION

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

ITU-T Recommendation J.43

(Extract from the *Blue Book*)

NOTES

1 ITU-T Recommendation J.43 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.43

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE HIGH QUALITY SOUND PROGRAMME SIGNALS FOR TRANSMISSION ON 320 kbit/s CHANNELS[1)]

(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.20, § 2.
Sampling frequency (CCIR Recommendation 606):	$32\ (1 + 5 \times 10^{-5})$ kHz,
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.

1) 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.

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;

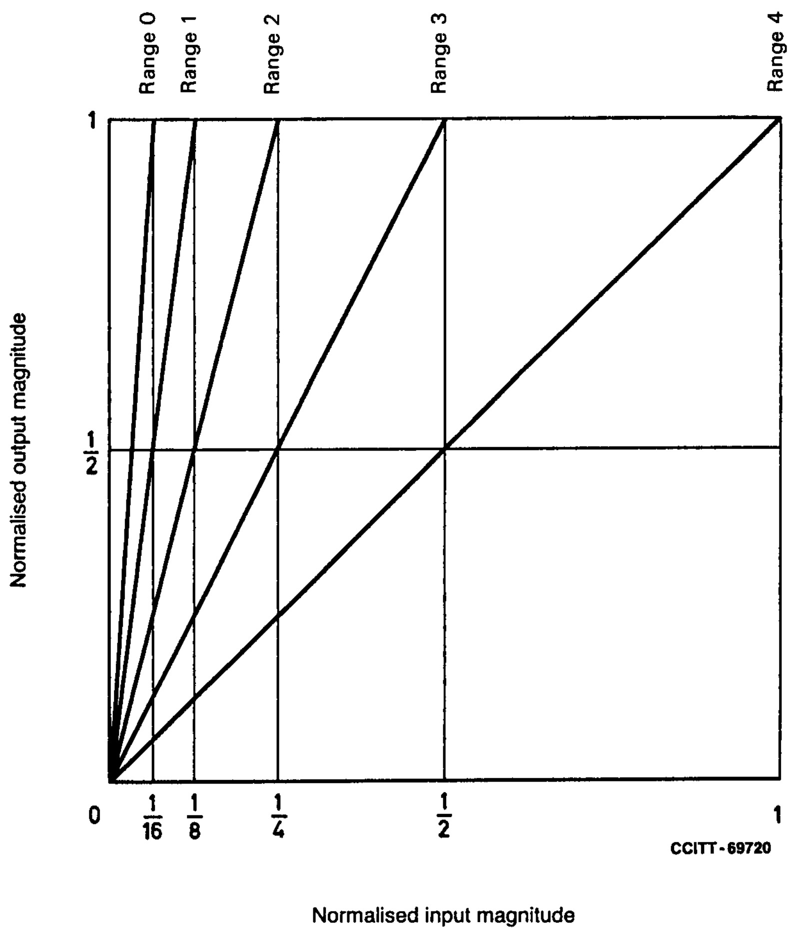

FIGURE 1/J.43

Companding characteristic

TABLE 1/J.43

14 to 10 bit near-instantaneous companding law

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	(1000000000)	
	−8192 to −8176	−8184	−512	(1111111111)	
3	+4088 to +4096	+4092	+511	(0111111111)	11 bits
	0 to +8	+4	0	(0000000000)	
	−8 to 0	−4	−1	(1000000000)	
	−4096 to −4088	−4092	−512	(1111111111)	
2	+2044 to +2048	+2046	+511	(0111111111)	12 bits
	0 to +4	+2	0	(0000000000)	
	−4 to 0	−2	−1	(1000000000)	
	−2048 to −2044	−2046	−512	(1111111111)	
1	+1022 to +1024	+1023	+511	(0111111111)	13 bits
	0 to +2	+1	0	(0000000000)	
	−2 to 0	−1	−1	(1000000000)	
	−1024 to −1022	−1023	−512	(1111111111)	
0	+511 to +512	+511.5	+511	(0111111111)	14 bits
	0 to +1	+0.5	0	(0000000000)	
	−1 to 0	−0.5	−1	(1000000000)	
	−512 to −511	−511.5	−512	(1111111111)	

MSB Most significant bit.

LSB Least significant bit.

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 $x(2\ n)$. Calculation of differential even samples $\Delta\,(2n)$ by the formula

$$\Delta(2n) = x(2n) - \frac{x(2n+1) + x(2n-1)}{2} \qquad (1)$$

c) additional near-instantaneous companding of the differential samples $\Delta\,(2n)$ 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 $\Delta\,(2n)$ additionally represented by a compressed code of 9 bits per sample, an average of 9.5 bits per sample is obtained.

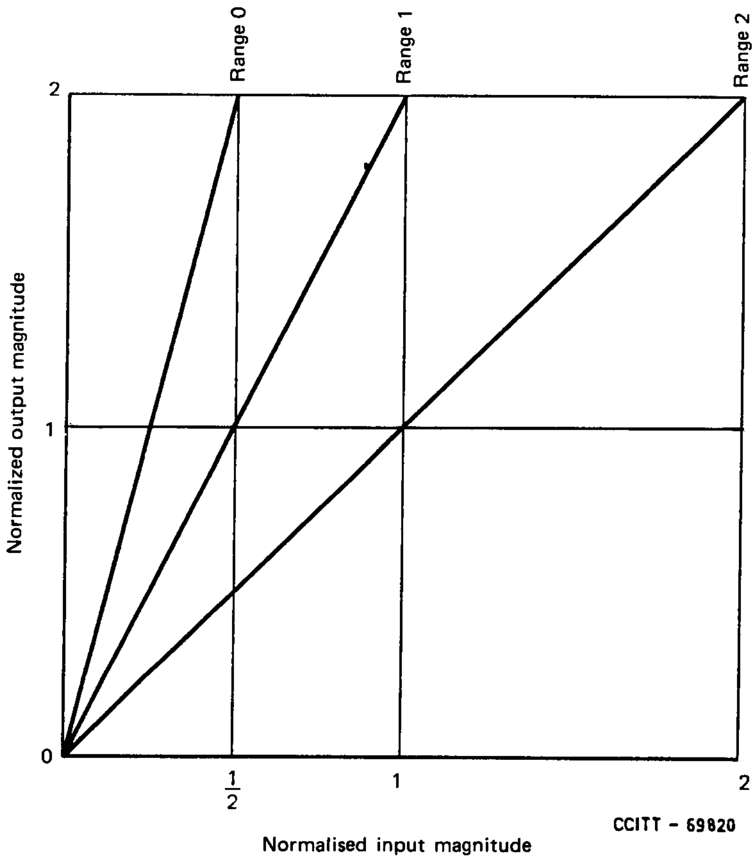

FIGURE 2/J.43

Companding characteristic

TABLE 2/J.43

14 to 9.0 bit near-instantaneous companding law

Range		Normalized input	Normalized output	Compressed digital code MSB LSB		Effective resolution
	2	+16 320 to +16 384	+16 352	+255	(011111111)	8 bits
		0 to +64	+32	0	(000000000)	
		−64 to 0	−32	−1	(100000000)	
		−16 384 to −16 320	−16 352	−256	(111111111)	
4	1	+8160 to +8192	+8176	+255	(011111111)	9 bits
		0 to +32	+16	0	(000000000)	
		−32 to 0	−16	−1	(100000000)	
		−8190 to −8160	−8176	−256	(111111111)	
	0	+4080 to +4096	+4088	+255	(011111111)	10 bits
		0 to +16	+8	0	(000000000)	
		−16 to 0	−8	−1	(100000000)	
		−4096 to −4080	−4088	−256	(111111111)	
	2	+8160 to +8192	+8176	+255	(011111111)	9 bits
		0 to +32	+16	0	(000000000)	
		−32 to 0	−16	−1	(100000000)	
		−8192 to −8160	−8176	−256	(111111111)	
3	1	+4080 to +4096	+4088	+255	(011111111)	10 bits
		0 to +16	+8	0	(000000000)	
		−16 to 0	−8	−1	(100000000)	
		−4096 to −4080	−4088	−256	(111111111)	
	0	+2040 to +2048	+2044	+255	(011111111)	11 bits
		0 to +8	+4	0	(000000000)	
		−8 to 0	−4	−1	(100000000)	
		−2048 to −2040	−2044	−256	(111111111)	
	2	+4080 to +4096	+4088	+255	(011111111)	10 bits
		0 to +16	+8	0	(000000000)	
		−16 to 0	−8	−1	(100000000)	
		−4096 to −4080	−4088	−256	(111111111)	
2	1	+2040 to +2048	+2044	+255	(011111111)	11 bits
		0 to +8	+4	0	(000000000)	
		−8 to 0	−4	−1	(100000000)	
		−2048 to −2040	−2044	−256	(111111111)	
	0	+1020 to +1024	+1022	+255	(011111111)	12 bits
		0 to +4	+2	0	(000000000)	
		−4 to 0	−2	−1	(100000000)	
		−1024 to −1020	−1022	−256	(111111111)	

TABLE 2/J.43 *(continuation)*

Range		Normalized input	Normalized output	Compressed digital code MSB LSB		Effective resolution
1	2	+2040 to +2048	+2044	+255	(011111111)	11 bits
		0 to +8	+4	0	(000000000)	
		−8 to 0	−4	−1	(100000000)	
		−2048 to −2040	−2044	−256	(111111111)	
	1	+1020 to +1024	+1022	+255	(011111111)	12 bits
		0 to +4	+2	0	(000000000)	
		−4 to 0	−2	−1	(100000000)	
		−1024 to −1020	−1022	−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)	
0	2	+1020 to +1024	+1022	+255	(011111111)	12 bits
		0 to +4	+2	0	(000000000)	
		−4 to 0	−2	−1	(100000000)	
		−1024 to −1020	−1022	−256	(111111111)	
	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.

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.

TABLE 3/J.43

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

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:

$$\begin{aligned}\overline{R}_9 &= R_1 \oplus R_2 \oplus R_3 \oplus R_7 \\ \overline{R}_{10} &= R_1 \oplus R_4 \oplus R_5 \oplus R_7 \oplus R_8 \\ \overline{R}_{11} &= R_2 \oplus R_4 \oplus R_6 \oplus R_7 \oplus R_8 \\ \overline{R}_{12} &= R_3 \oplus R_5 \oplus R_6 \oplus R_8\end{aligned} \tag{2}$$

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

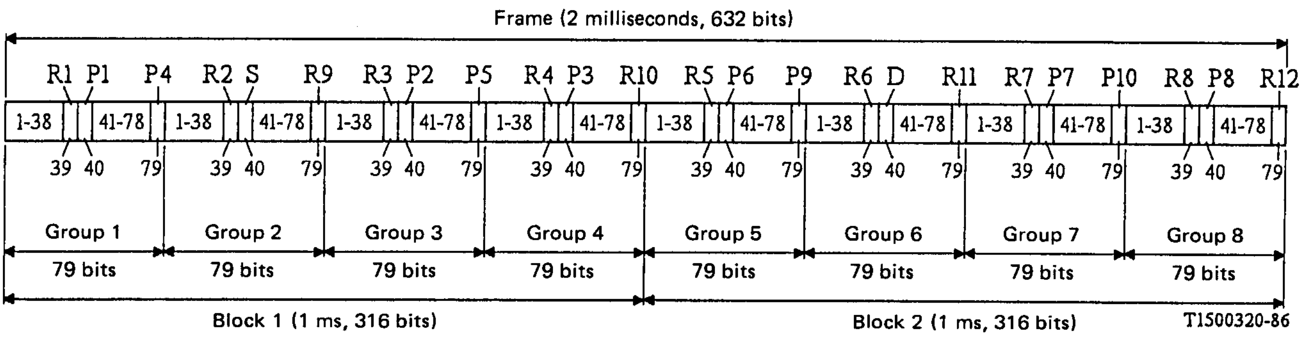

FIGURE 3/J.43

Single channel frame format

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.

TABLE 4/J.43

Bit allocation in the frame

	Frame allocation (bits/frame)	Bit rate per channel (Kbit/s)
Samples	608	304
Ranges 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

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 be 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.

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 sample of the 1st block (R1 to R5)	40 79	1, 3, 4 1, 3
Check bits of the samples of the 2nd block (R6 to R10)	40 79	5, 7, 8 5, 7
Signalling and check bits (S)	40	2
Data bits (D)	40	6

TABLE 6/J.43

Bit number in sample n of block k								Bit number in group 1 of frame N
N – 1				N – 2				
k = 1		k = 2		k = 1		k = 2		
n = 41 – 3	n = 41 – 1	n = 41 – 2	n = 41	n = 41 – 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, ± 2, . . .

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

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.