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TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE

RED BOOK

VOLUME III – FASCICLE III.4

LINE TRANSMISSION OF NON-TELEPHONE SIGNALS

RECOMMENDATIONS OF THE H SERIES

TRANSMISSION OF SOUND-PROGRAMME AND TELEVISION SIGNALS

RECOMMENDATIONS OF THE J SERIES

VIIIth PLENARY ASSEMBLY
MALAGA-TORREMOLINOS, 8-19 OCTOBER 1984

Geneva 1985

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PRELIMINARY NOTES

1 The questions entrusted to each Study Group for the Study Period 1985-1988 can be found in Contribution No. 1 to that Study Group.

2 In this Fascicle, the expression "Administration" is used for shortness to indicate both a telecommunications Administration and a recognized private operating agency.

3 *Units*

The units used in this volume are in conformity with CCITT Recommendations B.3 and B.4 (Volume I).

The following abbreviations are used, particularly in diagrams and tables, and always have the following clearly defined meanings:

dBm the absolute (power) level in decibels;

dBm0 the absolute (power) level in decibels referred to a point of zero relative level;

dBr the relative (power) level in decibels;

dBm0p the absolute psophometric power level in decibels referred to a point of zero relative level.

PART I

Series H Recommendations

LINE TRANSMISSION OF NON-TELEPHONE SIGNALS

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LINES USED FOR THE TRANSMISSION OF SIGNALS OTHER THAN TELEPHONE SIGNALS, SUCH AS TELEGRAPH, FACSIMILE, DATA, ETC., SIGNALS[1)]

Part I contains two classes of Recommendations: those which define the characteristics of *transmission channels* (telephone-type, group, supergroup, etc., circuits) used only to transmit signals other than telephone signals, and those which define the characteristics of the *signals* used in such transmissions.

In this Part, "wideband" is used to qualify the transmission channels, and "wide-spectrum" the signals transmitted, so as to avoid any confusion between the transmission channels and the signals transmitted with regard to the frequency bands involved in transmission over group links, supergroup links, etc.

As far as possible, one should avoid specifying the characteristics of particular channels or signals in defining a new service and refer only to the characteristics of the channels mentioned in Section 1 of this Recommendation Series.

Section 6 of this Series is reserved for Recommendations concerning the characteristics of visual telephone systems.

Table 1 indicates the correspondence of Series H Recommendations to Recommendations of other Series.

TABLE 1

Series H Recommendations	*Recommendations of other Series*
H.12, § 1	M.1040 (Volume IV)
H.12, § 2	M.1025 (Volume IV)
H.12, § 3	M.1020 (Volume IV)
H.13	See Recommendation O.71 (Volume IV)
H.14, § 2	M.910 (Volume IV)
H.16	O.72 (Volume IV)
H.21	See also the Recommendations M.800 (Volume IV) and R.77 (Volume VII)
H.22	See also the Recommendation M.810 (Volume IV)
H.23	Extract of Recommendations R.31 and R.35 (Volume VII)
H.32	R.43 (Volume VII)
H.41	T.11 (Volume VII)
H.42	T.12 (Volume VII)
H.43	T.10 (Volume VII)
H.51	V.2 (Volume VIII)

1) Excluding the transmission of sound-programme and television signals, which is the subject of the Series J Recommendations.

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SECTION 1

LINES USED FOR THE TRANSMISSION OF SIGNALS OTHER THAN TELEPHONE SIGNALS, SUCH AS TELEGRAPH, FACSIMILE, DATA, ETC., SIGNALS

1.1 Characteristics of transmission channels used for other than telephone purposes

Recommendation H.11

CHARACTERISTICS OF CIRCUITS IN THE SWITCHED TELEPHONE NETWORK

(Mar del Plata, 1968)

The characteristics of these telephone circuits are in conformity with Recommendations G.151 [1], G.152 [2] and G.153 [3]. Audio-frequency circuits, the characteristics of which are in accordance with Recommendations G.124 [4], G.511 [5] and G.543 [6], may also be found.

Some information on the characteristics of communications established in the switched telephone network is given in [7] and [8].

References

[1] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151.

[2] CCITT Recommendation *Characteristics appropriate to long-distance circuits of a length not exceeding 2500 km*, Vol. III, Rec. G.152.

[3] CCITT Recommendation *Characteristics appropriate to international circuits more than 2500 km in length*, Vol. III, Rec. G.153.

[4] CCITT Recommendation *Characteristics of long-distance loaded-cable circuits liable to carry international calls*, Orange Book, Vol. III-1, Rec. G.124, ITU, Geneva, 1977.

[5] CCITT Recommendation *General characteristics of audio-frequency circuits*, Orange Book, Vol. III-1, Rec. G.511, ITU, Geneva, 1977.

[6] CCITT Recommendation *Specification for repeater sections of loaded telecommunication cable*, Orange Book, Vol. III-1, Rec. G.543, ITU, Geneva, 1977.

[7] *Results of measurements and observations on the stability of the overall loss of circuits in the international network*, Green Book, Vol. IV.2, Supplement No. 4.1, ITU, Geneva, 1973.

[8] *Characteristics of leased international telephone-type circuits*, Green Book, Vol. IV.2, Supplement No. 4.3, ITU, Geneva, 1973.

CHARACTERISTICS OF TELEPHONE-TYPE LEASED CIRCUITS

(Mar del Plata, 1968; amended at Geneva, 1972, 1976 and 1980)

1 Ordinary telephone-type circuits

Note 1 – The application of § 1 of this Recommendation to multiterminal leased circuits is intended only for radial networks in which these specifications are to be met between a designated central station and each of the outlying stations. It does not apply to multiterminal conference networks between any two stations.

Note 2 – The contents of § 1 correspond to part of Recommendation M.1040 [2].

1.1 *Scope of § 1 of the Recommendation and constitution of a leased circuit*

Paragraph 1 of this Recommendation details the characteristics of international leased circuits for telephony and other purposes that do not require the use of special-quality leased circuits conforming to §§ 2 or 3. These circuits are set up as described in Recommendation M.1050 [1].

1.2 *Characteristics*

1.2.1 *Nominal overall loss*

Because of the differing nominal level at renters' premises due to the various national practices, it is not normally possible to predict the nominal overall loss of the circuit at the reference frequency. Only exceptionally can a predetermined specified nominal overall loss at the reference frequency between renters' installations be offered and then only after prior consultation among the Administrations concerned.

For 4-wire circuits, the receiving relative level at the renters' premises should not be lower than −15 dBr. If a mean sending signal power of −15 dBm0 is assumed, the resulting minimum received power (−30 dBm) is sufficient for telephony and the other purposes for which circuits to this Recommendation are intended. Should these circuits be used for other purposes, higher receiving relative levels may be required in some circumstances.

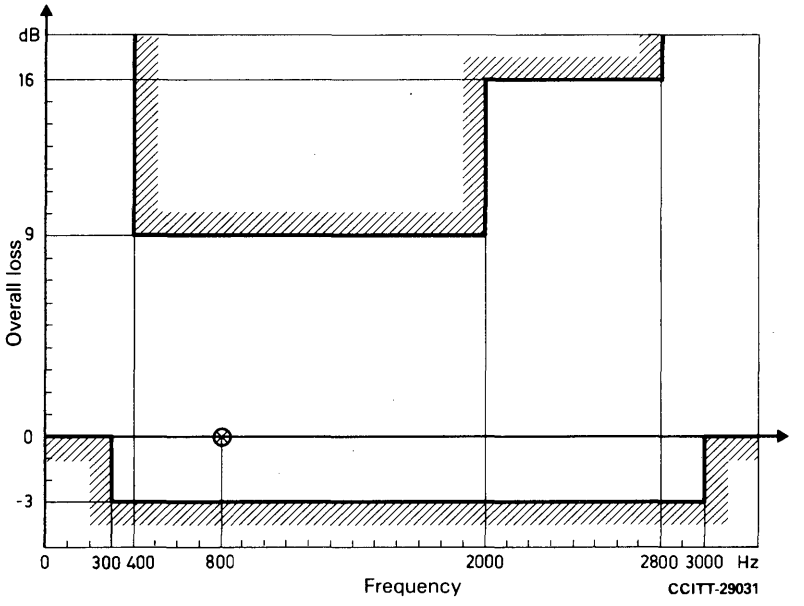
It should be noted that the overall loss in each direction of transmission may not have the same value.

1.2.2 *Loss/frequency distortion*

The limits for the overall loss relative to that at 800 Hz for the circuit between renters' installations are given in Figure 1/H.12.

1.2.3 *Random circuit noise*

The nominal level of the psophometric noise power at a renter's premises depends upon the actual constitution of the circuit, in partiular upon the length of frequency division multiplex carrier systems in the circuit. The provisional limit for leased circuits of distances greater than 10 000 km is −38 dBm0p. However, circuits of shorter length will have substantially less random noise (see also Annex A).



Note – At frequencies below 300 Hz and above 3000 Hz the loss shall not be less than 0.0 dB but is otherwise unspecified.

FIGURE 1/H.12

Limits for overall circuit loss relative to that at 800 Hz

2 Special quality leased circuits with basic bandwidth conditioning [1), 2)]

2.1 *Scope of § 2*

Paragraph 2 of this Recommendation deals with leased circuits for uses other than telephony – for example, data transmission.

The characteristics of §§ 2 and 3 are intended to ensure the provision of a circuit which will meet the requirements of digital transmission rates higher than those possible on a normal telephone-type circuit[3)]. § 2 is primarily intended for use with modems that contain equalizers.

2.2 *Characteristics*[4)]

Note – Except for the bandwidth characteristics (loss/frequency and group-delay distortions), the characteristics of § 2 are common to both types of "special quality leased circuits", that is, to §§ 2 and 3 of this Recommendation.

---

1) The application of § 2 of this Recommendation to multiterminal leased circuits is intended only for radial networks in which these specifications are to be met between a designated central station and each of the outlying stations. It does not apply to multiterminal conference networks between any two stations.

2) The contents of § 2 correspond to part of Recommendation M.1025 [3].

3) In order to ensure the proper operation of certain V Series modems operating at data signalling rates greater than 4800 kbit/s, it is necessary to specify improved and/or modified values for the following transmission system characteristics: random circuit noise, quantizing noise, harmonic distortion (intermodulation distortion). This subject requires further study.

4) Additionally, the characteristics for short breaks in transmission, phase hits, amplitude hits and low-frequency phase jitter are under study.

2.2.1 *Nominal overall loss*

Because of the differing nominal level at renters' premises due to the various national practices, it is not normally possible to predict the nominal overall loss of the circuit at the reference frequency. Only exceptionally can a predetermined specified nominal overall loss at the reference frequency between renters' installations be offered and then only after prior consultation among the Administrations concerned.

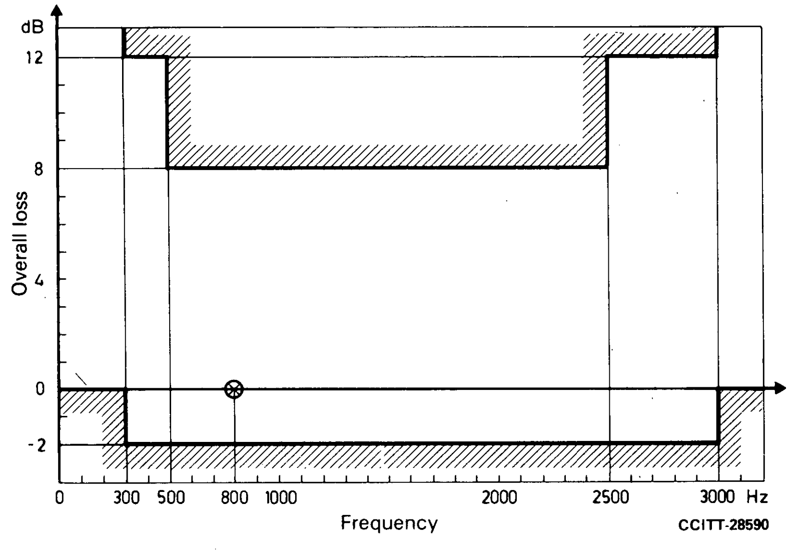
For 4-wire circuits, the value of the receiving relative level at the renters' premises should not be lower than -13 dBr.

For circuits intended to be used for data transmission using modems to V-series Recommendations, higher receiving relative levels may be required in some circumstances.

It should be noted that the overall loss in each direction of transmission may not have the same value.

2.2.2 *Loss/frequency distortion*[5)]

The limits for the overall loss relative to that at 800 Hz for the circuit between renters' installations are given in Figure 2/H.12.



Note — At frequencies below 300 Hz and above 3000 Hz, the loss shall not be less than 0.0 dB but is otherwise unspecified.

FIGURE 2/H.12

Limits for the overall loss of the circuit relative to that at 800 Hz

2.2.3 *Group-delay distortion*[5)]

The limits that apply to group-delay distortion are given in Figure 3/H.12 in which the limiting values over the frequency band are expressed as values relative to the minimum measured group delay.

2.2.4 *Variation of overall loss with time*

2.2.4.1 *Amplitude hits*

Where the circuit is to be used for data transmission using modems employing amplitude modulation techniques, for example modems to Recommendation V.29, amplitude hits may result in data errors. Using an instrument complying with Recommendation O.95, the number of amplitude hits greater than ± 2 dB should not exceed 10 in any 15-minute measuring period. (The value of ± 2 dB and the number of amplitude hits are provisional and subject to further study.)

5) It is expected that in most cases these "basic bandwidth" characteristics may be available without the addition of loss/frequency and/or group-delay equalization equipment.

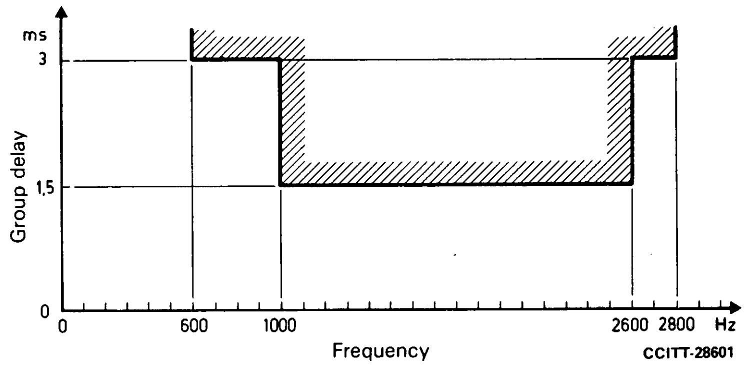


FIGURE 3/H.12

Limits for group delay relative to the minimum measured group delay in the 600-2800 Hz band

2.4.4.2 *Other variations*

For all circuits, variations with time of the overall loss at 800 Hz (including daily and seasonal variations but excluding amplitude hits) should be as small as possible but should not exceed ± 4 dB.

2.2.5 *Random circuit noise*

The nominal level of the psophometric noise power at the renter's premises depends upon the actual constitution of the circuit, in particular upon the length of frequency division multiplex carrier systems in the circuit. The provisional limit for special quality leased circuits of distances greater than 10 000 km is −38 dBm0p. However, circuits of shorter length will have substantially less random noise (see also Annex A).

Note – In the case of multiterminal circuits, the length to be taken into consideration is the sum of the lengths of the various sections of the link.

2.2.6 *Impulsive noise*

Impulsive noise should be measured with an instrument complying with Recommendation O.71 [4].

The number of impulsive noise peaks exceeding −21 dBm0 should not be more than 18 in 15 minutes.

2.2.7 *Phase jitter*

The value of phase jitter measured at a renter's premises depends upon the actual constitution of the circuit (for example, upon the number of modulation equipments involved). It is expected that any measurement of phase jitter using an instrument complying with Recommendation O.91 [5] will not normally exceed 10° peak-to-peak. However, for circuits of necessarily complex constitution and where 10° peak-to-peak cannot be met, a limit of up to 15° peak-to-peak is permitted.

2.2.8 *Quantizing noise (quantizing distortion)*

If any circuit section is routed over a PCM system, the signal will be accompanied by quantizing noise. The minimum ratio of signal-to-quantizing noise normally expected is 22 dB.

2.2.9 *Single tone interference*

The level of single tone interference in the band 300-3400 Hz shall not exceed a value which is 3 dB below the circuit noise objective indicated in Figure A-1/H.12.

2.2.10 *Frequency error*

The frequency error introduced by the circuit must not exceed ± 5 Hz. It is to be expected that in practice the error will be within closer limits than these.

2.2.11 *Harmonic distortion*

When a 700-Hz test frequency of −13 dBm0 is injected at the transmit end of a point-to-point circuit, the level of any individual harmonic frequency at the receiving end shall provisionally be at least 25 dB below the received level of the fundamental frequency.

3 Special quality leased circuits with special bandwidth conditioning [6), 7)]

3.1 *Scope of § 3*

Paragraph 3 of this Recommendation deals with leased circuits for uses other than telephony – for example, data transmission.

The characteristics of §§ 2 and 3 of this Recommendation are intended to ensure the provision of a circuit which will meet the requirements of digital transmission rates higher than those possible on a normal telephone-type circuit. § 3 is primarily intended for use with modems that do not contain equalizers.

3.2 *Characteristics*

Note – As discussed in § 2, except for the bandwidth characteristics (loss/frequency and group-delay distortions), the characteristics of § 2 are common to both types of "special quality leased circuits". Thus only the loss/frequency distortion and group-delay distortion will be described in this section.

3.2.1 *Nominal overall loss*

See § 2.2.1.

3.2.2 *Loss/frequency distortion*

The limits for the overall loss relative to that at 800 Hz for the circuit between renters' installations are given in Figure 4/H.12.

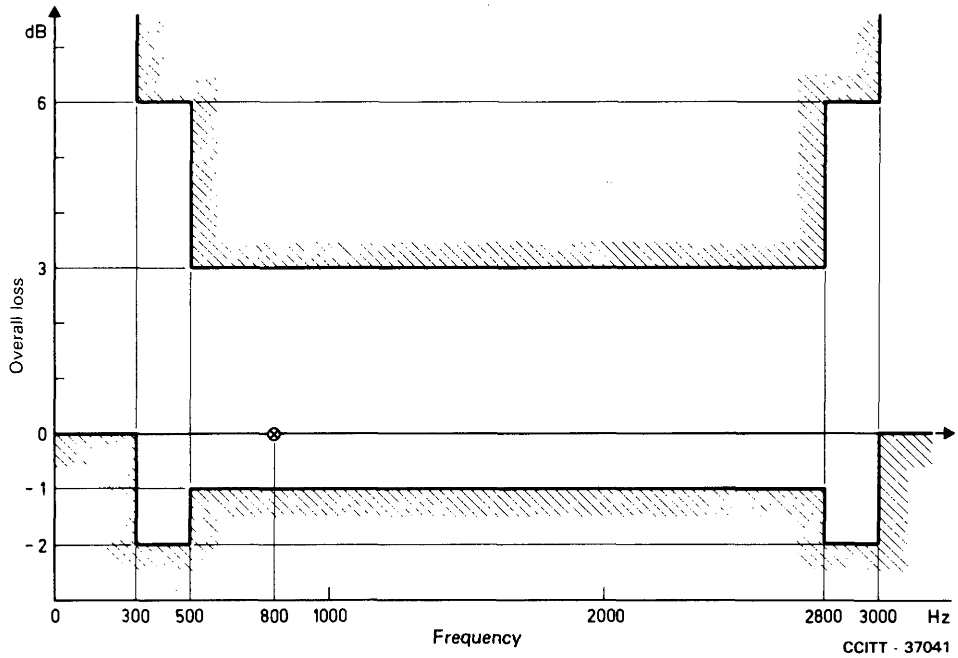
3.2.3 *Group-delay distortion*

The limits that apply to group-delay distortion are given in Figure 5/H.12 in which the limiting values over the frequency band are expressed as values relative to the minimum measured group delay.

The other characteristics correspond to these of §§ 2.2.4 to 2.2.11 of this Recommendation (see the Note to § 2.2).

6) The application of § 3 of this Recommendation to multiterminal leased circuits is intended only for radial networks in which these specifications are to be met between a designated central station and each of the outlying stations. It does not apply to multiterminal conference networks between any two stations.

7) The contents of § 3 correspond to part of Recommendation M.1020 [6].



Note — At frequencies below 300 Hz and above 3000 Hz, the loss shall not be less than 0.0 dB but is otherwise unspecified.

FIGURE 4/H.12

Limits for the overall loss of the circuit relative to that at 800 Hz

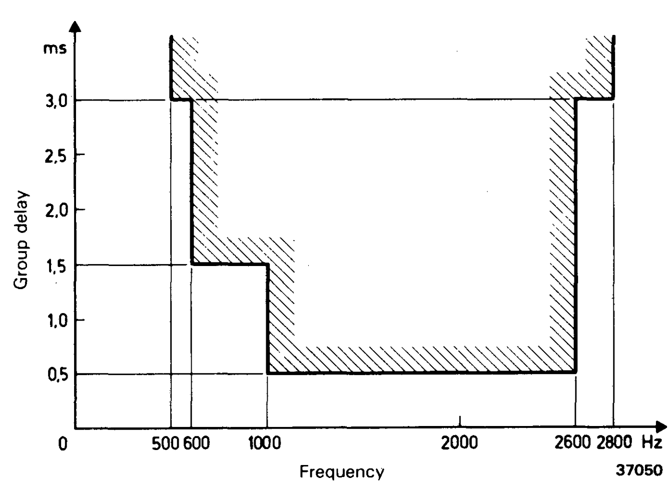


FIGURE 5/H.12

Limits for group delay relative to the minimum measured group delay in the 500-2800 Hz band

ANNEX A

(to Recommendation H.12)

Random circuit noise

Figure A-1/H.12 displays random noise versus length and is presented as a guide to the random noise performance which may be found on an international leased circuit.

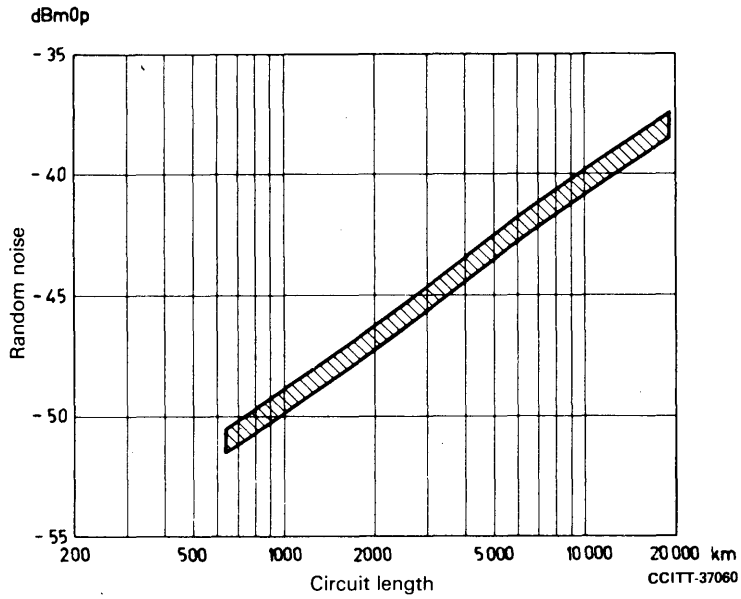


FIGURE A-1/H.12

Random noise circuit performance

Note – At the present time the section of the circuit provided by satellite (between earth stations) contributes approximately 10 000 pW0p (–50 dBm0p) of noise. Therefore, for the purpose of determining maintenance limits for noise measurements on leased circuits, the length of a circuit section routed on a communications satellite may be considered to be equivalent to 1000 km in Figure A-1/H.12.

References

[1] CCITT Recommendation *Lining up an international point-to-point leased circuit,* Vol. IV, Rec. M.1050.

[2] CCITT Recommendation *Characteristics of ordinary quality international leased circuits,* Vol. IV, Rec. M.1040.

[3] CCITT Recommendation *Characteristics of special quality international leased circuits with basic bandwidth conditioning,* Vol. IV, Rec. M.1025.

[4] CCITT Recommendation *Specification for an impulsive noise measuring instrument for telephone-type circuits,* Vol. IV, Rec. O.71.

[5] CCITT Recommendation *Essential clauses for an instrument to measure phase jitter on telephone circuits,* Vol. IV, Rec. O.91.

[6] CCITT Recommendation *Characteristics of special quality international leased circuits with special bandwidth conditioning,* Vol. IV, Rec. M.1020.

Recommendation H.13

CHARACTERISTICS OF AN IMPULSIVE NOISE MEASURING INSTRUMENT FOR TELEPHONE-TYPE CIRCUITS

(The text of this Recommendation can be found in Recommendation O.71 in Fascicle IV.4 of Volume IV.)

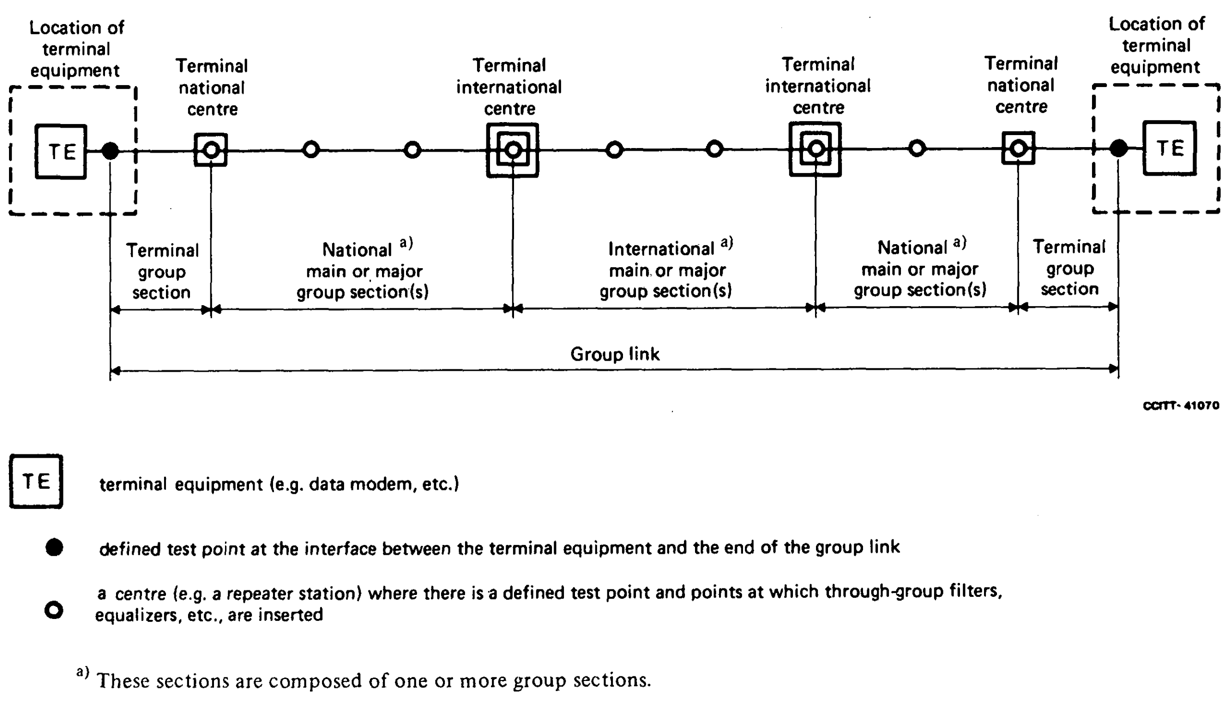
Recommendation H.14

CHARACTERISTICS OF GROUP LINKS FOR THE TRANSMISSION OF WIDE-SPECTRUM SIGNALS

(Mar del Plata, 1968; amended at Geneva, 1972, 1976, 1980 and 1984)

1 Constitution of a link, terminology, and scope of the Recommendation

A group link is composed of one or more group sections in tandem, generally prolonged at each end by "local lines" (denoted terminal group sections in Figure 1/H.14). These terminal group sections connect the group distribution frames of the terminal national centres with the equipment for sending and receiving wide-spectrum signals (modems, etc.) which may be situated either in the subscriber's premises or in any other place. In the latter case they are normally switched over the local telephone cable network, or sometimes over a special cable line or a radio-relay link. Only local lines carrying the 60-108 kHz wide-spectrum signal are termed "terminal group sections" and are included in the definition of a group link. The other case in which a baseband signal occupying a frequency band other than 60-108 kHz is transmitted over the local lines, the frequency translation to the 60-108 kHz band being made at the terminal national centres, is not dealt with in this Recommendation.



a) These sections are composed of one or more group sections.

FIGURE 1/H.14

Example of the constitution of a group link for wide-spectrum signal transmission

It should be noted that the group link does not comprise any terminal equipment (modems, etc.). Figure 1/H.14 illustrates these considerations.

2 Characteristics of corrected group links[1)]

The characteristics mentioned in §§ 2.1 and 2.2 below imply the use of a group pilot at 104.08 kHz. The use of a pilot in the middle of the group band requires different characteristics.

2.1 *Group-delay distortion*

The group-delay distortion over the band 68-100 kHz should not exceed 45 μs with respect to the value of the least group delay within that band. This value can be observed on a tandem connection of three group sections which includes two terminal group sections.

Note 1 – The following assumptions have been made:

1) The group-delay distortion of through-group connection equipment can be corrected so as not to exceed 15 μs over the band 68-100 kHz. It should be noted that through-group connection equipment comprises the group demodulating equipment, the through-group filter and the group modulating equipment (see Recommendation G.242 [2]). The equalization should be made in such a way that at least 6 group-delay maxima are obtained.
2) To respect these limits it may be necessary to avoid groups 1 and 5.
3) The use of a group containing the supergroup pilot should always be avoided.

In particular, group 3 should not be used when the supergroup pilot is at 411.920 or 411.860 kHz.

Note 2 – In certain cases where disturbing signals outside the basic group band have to be expected, additional filtering has to be provided in the local lines.

2.2 *Attenuation/frequency distortion*

The attenuation/frequency distortion of the whole link is given in Figure 2/H.14. It should be measured over the 60-108 kHz frequency range and equalized with a group link equalizer as necessary to meet the limits with respect to loss at 84 kHz.

2.3 *Carrier leaks*

The leak from a carrier in the 60-108 kHz band shall not exceed −40 dBm0.

Note 1 – Although this value is an objective, it may in some cases prove impossible to achieve owing to the composition of the link, which will generally involve the use of both old and new types of equipment. In any case, no carrier leak in the band 60-108 kHz should exceed −35 dBm0.

Note 2 – For group links used in data transmission with modems in line with Recommendation V.36 [3], problems may arise if the total carrier leak power exceeds −35 dBm0.

2.4 *Variations in level*

The following limits should not be exceeded:

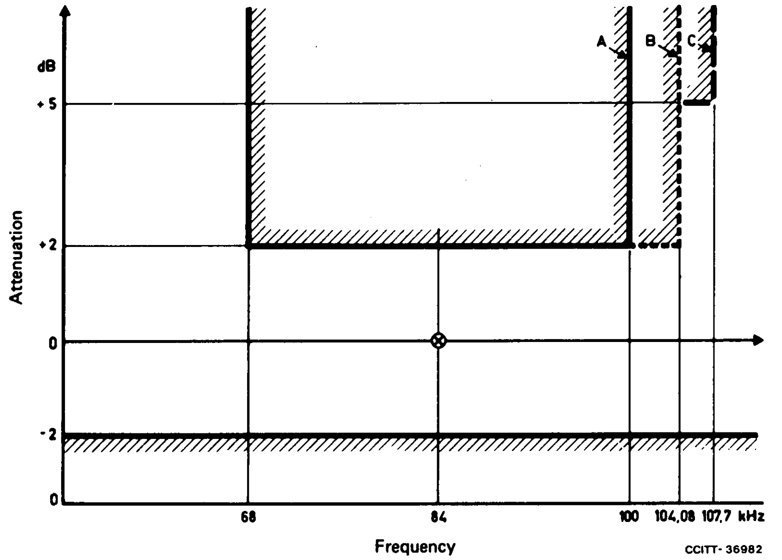
– short-term variations
  (for a few seconds) . . . . ± 3 dB
– long-term variations
  (during long periods, including seasonal and daily variations) . . . . ± 4 dB

relative to the nominal level.

2.5 *Background noise*

This can be expected to be substantially uniformly distributed over the group band, and to have a value calculated in accordance with Recommendations G.222 [4] and G.223 [5]. For an actual link, a margin should be allowed as indicated in Recommendation G.226 [6].

[1)] This § 2 corresponds to Recommendation M.910 [1].



A = These limits apply if the group reference pilot (104.08 kHz) is injected at an intermediate point on the link (e.g. the terminal national centre)

B = These limits apply if the group reference pilot is transmitted from end to end on the link (e.g. if it is injected by the terminal equipment)

C = These limits apply if the service channel is provided

Note 1 — If the service channel is provided, additional equalization may be needed and there will be no possibility of using simplified through-group filters.

Note 2 — 84 kHz is the reference frequency for the purposes of specifying and measuring attenuation/frequency distortion. This is not in contradiction with the use of the group reference pilot at 104.08 kHz.

FIGURE 2/H.14

Limits for attenuation/frequency distortion

2.6 *Impulsive noise*

Under study.

2.7 *Frequency error*

Maximum frequency error shall not exceed 5 Hz.

Note – According to Recommendation G.225 [7], this condition should readily be met in practice.

2.8 *Phase changes with time*

The interfering phase modulation and phase jitter should meet the conditions set forth in Recommendation G.229 [8].

2.9 *Power handling capability*

Applied signals should be within the limits given in Recommendation H.52.

3 Characteristics of uncorrected group links

This type of link may be used, for example, for data transmission, the equalization taking place in the terminal equipment.

As a general rule, a group pilot at 104.08 kHz should be used. The possible existence of pilot blocking filters has not been taken into account in the characteristics given here in § 3.

Only Groups 2 and 4 should be used.

As far as possible, these groups should alternate in such a way that the number of Group 2 sections and the number of Group 4 sections do not differ by more than one.

Annex A gives the hypothetical reference circuits for leased group links. These can be used to calculate attenuation distortion and group-delay distortion which may occur in practice rather than as guaranteed characteristics. Other characteristics are expected to be the same in the uncorrected group links as they are in the corrected ones (see §§ 2.3-2.9).

Impulse noise characteristics are under study.

ANNEX A

(to Recommendation H.14)

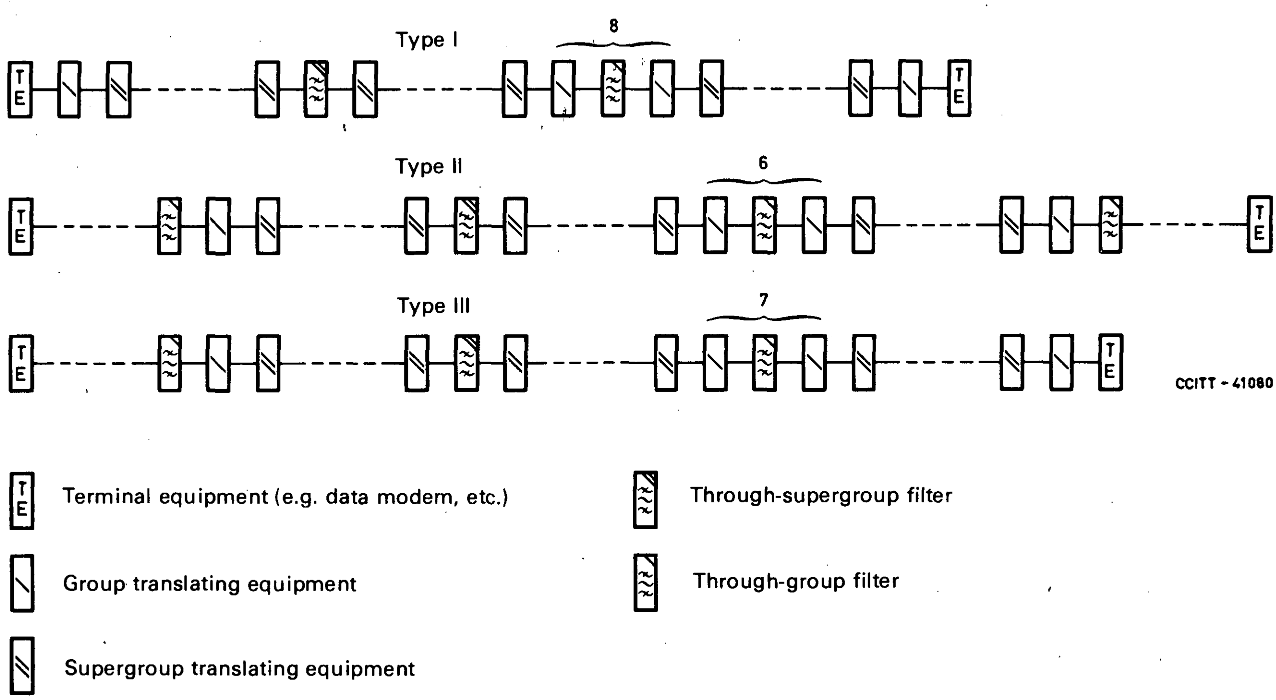
Hypothetical reference circuits for leased group links

A.1 *Hypothetical reference circuit*

Figure 3/H.14 represents a hypothetical reference circuit used to calculate the group-delay distortion.

Three types of circuit have been taken into account, all with 8 through-group filters.

Note – The number "8" chosen for through-group filters should be confirmed or amended after subsequent studies. It is possible that the choice of the number "8" corresponds to a case too unfavourable and that this number should be reduced.



Type I: Terminal equipment at both sides in the repeater station.
Type II: Terminal equipment at both sides at the renter's premises.
Type III: Terminal equipment at one side in a repeater station and at the other side at the renter's premises.

For information, Table 1/H.14 shows the number of different types of equipment found on the link for a range of through-group connections.

Note 1 — The through-group filters are all of the type recommended in Recommendation G.242 [2].

Note 2 — In Type II, through-group filters are planned to ensure the privacy of adjacent groups and to protect them from interference from the local network.

FIGURE 3/H.14

Hypothetical reference circuit for a leased group link

TABLE 1/H.14

Number of through-group connections	Type I	Type II	Type III
	0 1 2 3 4 5 6 7 8	0 1 2 3 4 5 6	0 1 2 3 4 5 6 7
Number of through-group filters	0 1 2 3 4 5 6 7 8	2 3 4 5 6 7 8	1 2 3 4 5 6 7 8
Number of through-supergroup filters	1 2 4 6 7 9 10 12 14	1 2 4 6 7 9 10	1 2 4 6 7 9 10 12
Number of group modulator and demodulator pairs	1 2 3 4 5 6 7 8 9	1 2 3 4 5 6 7	1 2 3 4 5 6 7 8
Number of supergroup modulator and demodulator pairs	2 3 5 7 8 10 11 13 15	2 3 5 7 8 10 11	2 3 5 7 8 10 11 13

A.2 *Attenuation distortion*

Under study.

A.3 *Group-delay distortion*

The group-delay distortion for a specific type of circuit (I, II or III) and for a specific number of sections can be calculated from the values in Table 2/H.14.

TABLE 2/H.14

Values for the mean curves of group-delay distorsion (in μs) as a function of frequency for various equipments

Frequencies (in kHz)	62	64	66	68	70	72	76	80	84	88	92	96	98	100	102	104	106
Through-group filter	250	104	60	37	25	16	7	2	0	2	7	16	25	37	60	104	250
Group translating equipment	8.0	6.7	5.6	4.8	4.0	3.3	1.8	0.6	0.0	0.0	0.6	1.2	1.7	2.2	2.8	3.6	4.6
Through-supergroup filter 2	0.0	0.1	0.2	0.3	0.5	0.6	0.9	1.2	1.6	2.0	2.4	2.9	3.2	3.5	3.8	4.2	4.6
Through-supergroup filter 4	4.7	4.2	3.7	3.3	2.9	2.6	2.0	1.6	1.2	0.9	0.6	0.4	0.3	0.2	0.2	0.1	0.0
Supergroup translating equipment 2	0.0	0.0	0.1	0.1	0.1	0.2	0.3	0.3	0.4	0.5	0.6	0.7	0.8	0.8	0.9	0.9	1.0
Supergroup translating equipment 4	1.0	0.9	0.9	0.8	0.8	0.7	0.6	0.5	0.4	0.3	0.3	0.2	0.1	0.1	0.1	0.0	0.0

Note 1 — The curves derived from these values are mean curves as far as through-group and through-supergroup filters are concerned.

Note 2 — The curves derived form these values for translating equipment are mean curves for the majority of the equipment in use. For the remaining equipment, the mean curve can be obtained be multiplying by two.

Note 3 — For group 2 in supergroup translating equipment for 120-channel systems, one can expect a curve equal to the curve in Table 2/H.14 multiplied by 3.

Note 4 — The group-delay distortion of the terminal extensions of group links has not been taken into account in Table 2/H.14. A provisional value for this group-delay distortion in each terminal group section is less than 2.5 μs.

Note 5 — The characteristic limits of group-delay distortion may be derived from the values given in Recommendations G.233 [9] and G.242 [2].

References

[1] CCITT Recommendation *Setting up and lining up an international leased group link for wide-spectrum signal transmission*, Vol. IV, Rec. M.910.

[2] CCITT Recommendation *Through-connection of groups, supergroups, etc.*, Vol. III, Rec. G.242.

[3] CCITT Recommendation *Modems for synchronous data transmission using 60-108 kHz group band circuits*, Vol. VIII, Rec. V.36.

[4] CCITT Recommendation *Noise objectives for design of carrier-transmission systems of 2500 km*, Vol. III, Rec. G.222.

[5] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*, Vol. III, Rec. G.223, § 4.

[6] CCITT Recommendation *Noise on a real link*, Vol. III, Rec. G.226.

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

[8] CCITT Recommendation *Unwanted modulation and phase jitter*, Vol. III, Rec. G.229.

[9] CCITT Recommendation *Recommendations concerning translating equipments*, Vol. III, Rec. G.233.

Recommendation H.15

CHARACTERISTICS OF SUPERGROUP LINKS FOR THE TRANSMISSION OF WIDE-SPECTRUM SIGNALS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1976)

1 Constitution of a link and terminology

The constitution and terminology for supergroup links are analogous to those for group links described in Recommendation H.14.

2 Characteristics of corrected supergroup links

2.1 *Group-delay distortion*

Provisionally, the rule $(1 + 2n)$ μs over the band 352-512 kHz is recommended as the limit for a supergroup link with n through-supergroup connection equipments (i.e. n modulating, demodulating and through-supergroup filter equipments). Supergroup links with corrected group-delay distortion should be restricted to supergroups 5, 6 and 7 in a mastergroup.

2.2 *Attenuation/frequency distortion*

Over the band 352-512 kHz the attenuation/frequency distortion should not exceed ± 2 dB with respect to the attenuation at 412 kHz.

Note – The reference frequency for the purposes of defining *distortion* should be 412 kHz even if the supergroup reference pilot used for *regulating* purposes is 547.92 kHz.

2.3 *Carrier leaks*

The leak from a carrier in the 352-512 kHz band shall not exceed −40 dBm0.

Note – Although this value is an objective, it may in some cases prove impossible to achieve owing to the composition of the link, which will generally involve the use of both old and new types of equipment. At all events, no carrier leak in the 352-512 kHz band should exceed −35 dBm0.

2.4 *Variations in level*

The following limits should not be exceeded:

– short-term variations
(for a few seconds) . . . . ± 3 dB

– long-term variations
(during long periods, including seasonal and daily variations) . . . . ± 4 dB

relative to the nominal level.

2.5 *Background noise*

This can be expected to be substantially uniformly distributed over the supergroup band, and to have a value calculated in accordance with Recommendations G.222 [1] and G.223 [2]. For an actual link, a margin should be allowed as indicated in Recommendation G.226 [3].

2.6 *Impulsive noise*

Under study.

2.7 *Frequency error*

Maximum frequency error shall not exceed 5 Hz.

Note – According to Recommendation G.225 [4], this condition should readily be met in practice.

2.8 *Phase changes with time*

Under study.

2.9 *Power handling capability*

Applied signals should be within the limits given in Recommendation H.53.

3 Characteristics of uncorrected supergroup links

Under study.

References

[1] CCITT Recommendation *Noise objectives for design of carrier-transmission systems of 2500 km*, Vol. III, Rec. G.222.

[2] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*, Vol. III, Rec. G.223, § 4.

[3] CCITT Recommendation *Noise on a real link*, Vol. III, Rec. G.226.

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

Recommendation H.16 [1]

CHARACTERISTICS OF AN IMPULSIVE-NOISE MEASURING INSTRUMENT FOR WIDEBAND DATA TRANSMISSION

(Geneva, 1972 and 1980)

The CCITT,

considering

that impulsive noise is of interest in wideband data transmission and that there is a need for a simple pulse counter suitable for field use,

provisionally *recommends*

that the instrument for impulsive-noise measurements should have the following characteristics:

1 Types of measurements

For the measurement of impulsive noise, the instrument should register a count whenever the instantaneous level applied to the input exceeds an adjustable threshold. This operation should be independent of the sense (or polarity) of the applied impulse.

For the measurement of circuit noise the instrument should provide means for indicating the average noise power.

2 Input impedance

The instrument should permit the measurements designated above on either balanced or unbalanced circuits at the nominal impedances which are used in wideband data transmission. On balanced circuits, the instrument should also be arranged to measure impulsive or circuit noise which is common to the two sides of the circuit with respect to earth.

Nominal input impedances should be provided as follows:

a) 75 ohms unbalanced;

b) 135 or 150 ohms balanced;

c) 135 or 150 ohms balanced with 20 000 ohms from each side of the circuit to a common 600 ohms which is returned to earth (the noise measurement is made across the 600-ohm resistor).

For the balanced input impedance [b) above] the balance of the input circuit in relation to earth should be such that when a 25-kHz sine wave, whose level is 70 dB higher than the instrument's threshold setting, is applied between the midpoint of the source impedance and the earth terminal of the instrument, the counter should not operate. Similarly, a 560-kHz sine wave, whose level is 42 dB higher than the threshold, when applied between the source impedance and the earth terminal of the instrument should not operate it. The above balance requirements shall hold for signal levels ranging up to 30 volts r.m.s.

Input arrangement c) above is provided for use in measuring impulsive and circuit noise which is common to the two sides of a balanced circuit with respect to earth.

3 Bandwidth and filter characteristics

For the condition of maximum bandwidth, the response should be within ± 1 dB of that at 25 kHz in the frequency range 275 Hz to 552 kHz, and should provide attenuation of at least 10 dB (with respect to that at 25 kHz) at frequencies below 50 Hz and above 1500 kHz.

[1] This Recommendation corresponds to Recommendation O.72.

Provision should be made for measurements on other specific bandwidths such as group or supergroup bands. These bandwidths may be obtained by means of plug-in or external filters, the characteristics of which should be as described in §§ 3.1 to 3.3 below.

3.1 For measurements on basic group-band circuits, the attenuation of the filter with reference to that at 84 kHz should lie on or within the limits shown in Figure 1/H.16.

3.2 For measurements on supergroup-band circuits, the attenuation of the filter with reference to that at 412 kHz should lie on or within the limits shown in Figure 2/H.16.

3.3 For measurements on baseband circuits with an upper frequency limit of 48 kHz, the attenuation of the filter with reference to that at 25 kHz should lie on or within the limits shown in Figure 3/H.16.

Note – When measuring in basic group and basic supergroup bands, one may use through-connection filters.

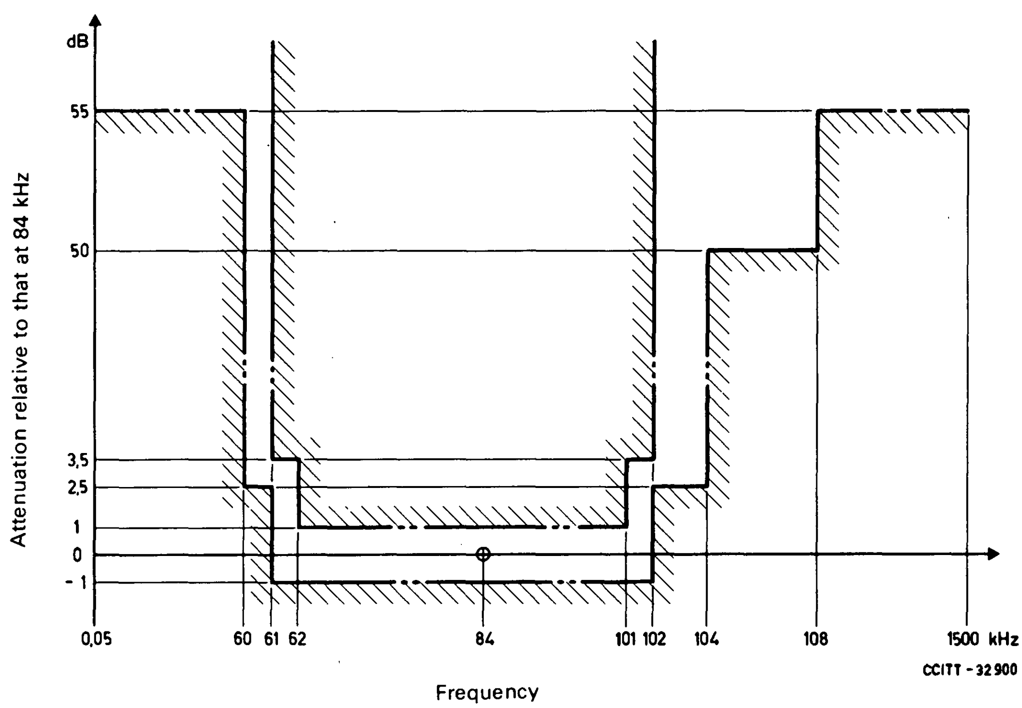


FIGURE 1/H.16

Permissible limits for the attenuation, relative to that at 84 kHz, of the filter for impulsive-noise measurements on a basic group band

4 Sensitivity and accuracy

For the measurement of impulsive noise, the threshold should be adjustable in steps of 1 dB for instantaneous levels from -60 to $+20$ dBm. For the measurement of circuit noise the sensitivity of the instrument should be -90 to $+10$ dBm at the calibration frequency. The accuracy of the instrument shall be $\pm$ 0.5 dB for any threshold setting or input polarity. The relative response to other signals should depend only on the attenuation characteristics for the maximum bandwidth or other selected bandwidths. The sensitivity of the instrument may be 30 dB less when used in the condition to measure circuit noise and impulsive noise common to the two sides of a balanced circuit with respect to earth.

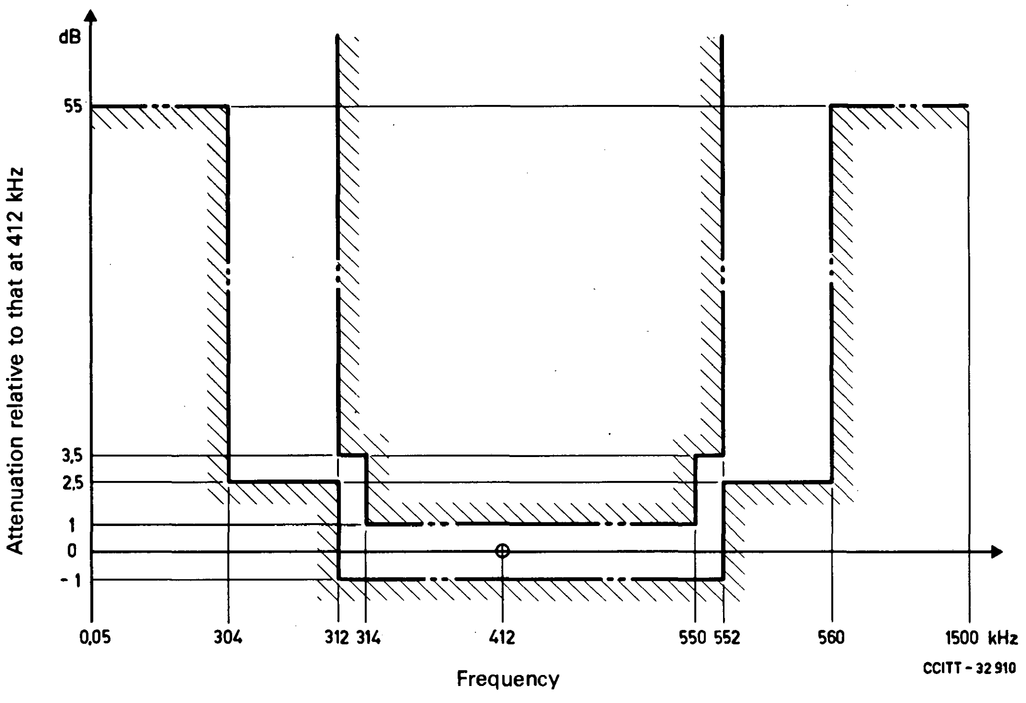


FIGURE 2/H.16

Permissible limits for the attenuation, relative to that at 412 kHz, of the filter for impulsive-noise measurements on a basic supergroup band

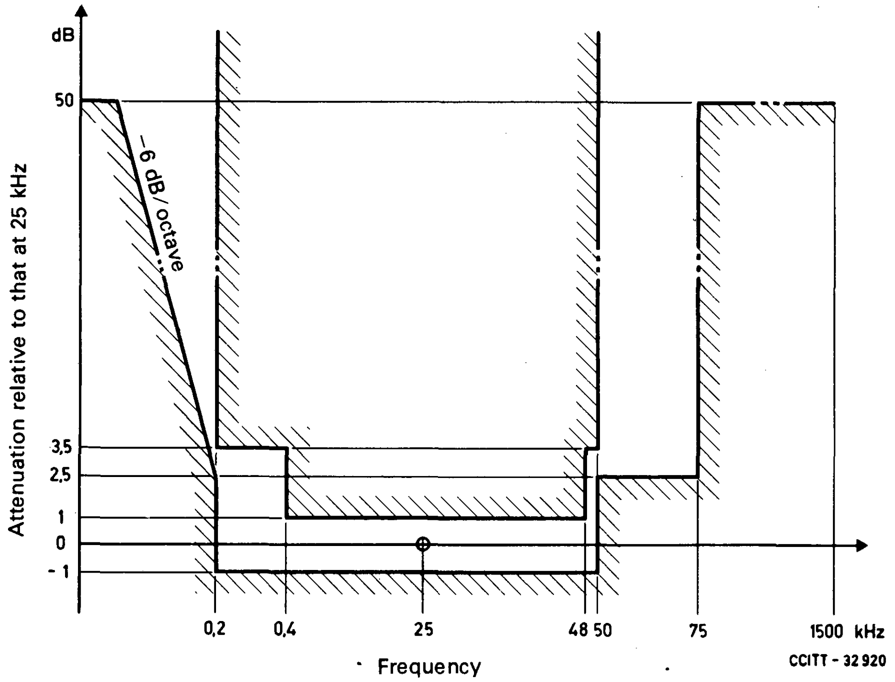


FIGURE 3/H.16

Permissible limits for the attenuation, relative to that at 25 kHz, of the filter for impulsive-noise measurements on a 48-kHz baseband circuit

5 Counting rate

Dead-time is defined as the time from the start of an impulse being registered until the counter is ready to register another impulse. A dead-time of 125 ± 25 ms shall be provided within the instrument.

Thus, the maximum counting rate is nominally eight impulses per second. The capacity of the counter shall be at least 999.

6 Calibration

Calibration should be possible from an internal signal or from the peaks of an externally applied sine-wave signal. For measurement of impulsive noise the calibration should be such that with the threshold adjusted to +3 dBm, the peaks of a 0 dBm sine wave will just operate the counter.

7 Timer

A built-in timer, continuously adjustable from 5 to 60 minutes, shall be provided. The accuracy shall be within ± 10% of the setting.

8 Temperature stability

All of the preceding clauses shall be satisfied when the ambient temperature varies between +10 °C and +40 °C.

1.2 Use of telephone-type circuits for voice-frequency telegraphy

Recommendation H.21

COMPOSITION AND TERMINOLOGY OF INTERNATIONAL VOICE-FREQUENCY TELEGRAPH SYSTEMS

(Mar del Plata, 1968)

Figure 1/H.21 illustrates the composition of an international voice-frequency telegraph (VFT) system and the terminology used.

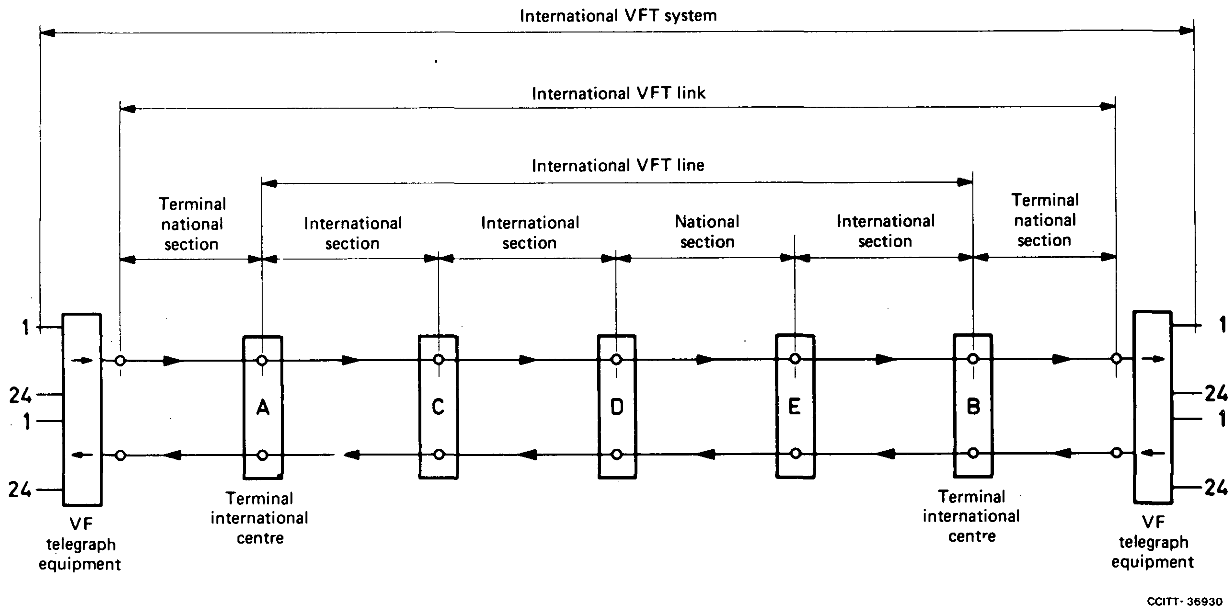
1 International voice-frequency telegraph system

This is the whole of the assembly of apparatus and lines including the terminal VFT equipment. In Figure 1/H.21 the system illustrated provides 24 duplex international telegraph circuits, but other numbers of telegraph circuits can be provided.

2 International VFT link
(sometimes referred to as the bearer circuit)

2.1 Four-wire telephone-type circuits are used for VFT links. The link comprises two unidirectional transmission paths, one for each direction of transmission, between the terminal VFT equipments.

2.2 The VFT link consists of an international telegraph line together with any terminal national sections connecting the international telegraph line to the VFT telegraph terminal equipment and may be constituted entirely on carrier channels (on symmetric pair, coaxial pair or radio-relay systems) or on audio-frequency lines or combinations of such lines.



(At the intermediate centres C, D and E and at the terminal international centres A and B, the signals transmitted are at audio frequencies. At these points it is possible to make measurements.)

FIGURE 1/H.21

The components of an international VFT system

2.3 The normal links for VF telegraphy have no terminating units, signalling equipment or echo suppressors.

3 International VFT line

3.1 The international VFT line may be constituted by using a channel in a carrier group or channels in tandem on a number of groups. National and international sections can be interconnected to set up an international telegraph line. See Figure 1/H.21, but note that § 3.2 below details the preferred method.

The international telegraph line could equally well be set up between, for example, only A and C or between C and D, in which case A and C, or C and D would be the terminal international centres.

3.2 Wherever possible an international telegraph line for a VFT link should be provided on channels of a single carrier group, thereby avoiding intermediate audio-frequency points. In some cases, such a group may not exist or, for special routing reasons, it may not be possible to set up the international telegraph line in the preferred way. In such cases, the international telegraph line will consist of channels in tandem on two or more groups with or without audio sections, depending on the line available and the routing requirements.

4 Terminal national sections connected to the international telegraph line

In many cases the VFT terminal equipment is remote from the terminal international centre of the international telegraph line (Figure 1/H.21), and such cases necessitate the provision of terminal national sections in order to establish international VFT links. These sections may be in short-distance local audio cables, amplified or unamplified, or may be routed in long-distance carrier groups or amplified audio plant as available.

TRANSMISSION REQUIREMENTS OF INTERNATIONAL VOICE-FREQUENCY TELEGRAPH LINKS (AT 50, 100 AND 200 BAUDS)

(Mar del Plata, 1968; amended at Geneva, 1972)

1 Links routed on carrier systems

Figure 1/H.21 shows the composition of an international circuit for voice-frequency (VF) telegraphy. The limits specified in the present Recommendation are based on the values between international terminal centres which are indicated in Recommendation G.151 [1] for an international telephone circuit and which are applied approximately to the international line in Figure 1/H.21. A slight increase has been made to certain characteristics to allow for unloaded national sections connecting the centres to the VF telegraph equipments since most telegraph installations belonging to public services are fairly close to the international maintenance centres.

1.1 *Nominal insertion loss at 800 Hz*

The nominal insertion loss of the link at 800 Hz depends on the nominal relative power levels at the extremities of the telegraph link. These levels will be those normally used in the national network of the countries concerned so that it is not possible to recommend a particular nominal value for the insertion loss.

The nominal relative power level at the input to the link and the absolute power level of the telegraph signals at this point must be such that the limits concerning the power level per telegraph channel at a zero relative point on carrier systems are respected.

1.2 *Variation of insertion loss with time*

In accordance with Recommendation M.160 [2]:

a) the difference between the mean value and the nominal value of the transmission loss should not exceed 0.5 dB;

b) the standard deviation from the mean value should not exceed 1 dB.

However, in the case of circuits set up wholly or partly on older equipment, where the international line consists of two or more circuit sections, a standard deviation not exceeding 1.5 dB may be accepted.

1.3 *Sudden variations of insertion loss and short interruptions*

Such defects of the transmission path impair the quality of the telegraph transmission and should be reduced to the minimum possible.

1.4 *Overall loss/frequency distortion*

The variation with frequency of the 600-ohm insertion loss of the link with respect to the loss at 800 Hz must not exceed the following limits:

1.4.1 *Links with 4-kHz sections throughout* (see Table 1/H.22)

TABLE 1/H.22

Frequency range (Hz)	Overall loss relative to that at 800 Hz
Below 300	Not less than −2.2 dB, otherwise unspecified
300- 400	−2.2 to +4.0 dB
400- 600	−2.2 to +3.0 dB
600-3000	−2.2 to +2.2 dB
3000-3200	−2.2 to +3.0 dB
3200-3400	−2.2 to +7.0 dB
Above 3400	Not less than −2.2 dB, otherwise unspecified

Overall loss limits are given in Table 1/H.22 and are shown hatched in Figure 1/H.22.

Note – The hatched limits in Figure 1/H.22 have been derived from the corresponding limits in Recommendation G.151 [1] by adding a margin to allow for the presence of unloaded national sections and also for the fact that the composition of the international line may be more complicated. This will permit the establishment of most international circuits for VF telegraphy without additional equalization.

In favourable cases it will be possible to respect the limits in the graph in Recommendation G.151 [1] which is shown as a broken line in Figure 1/H.22.

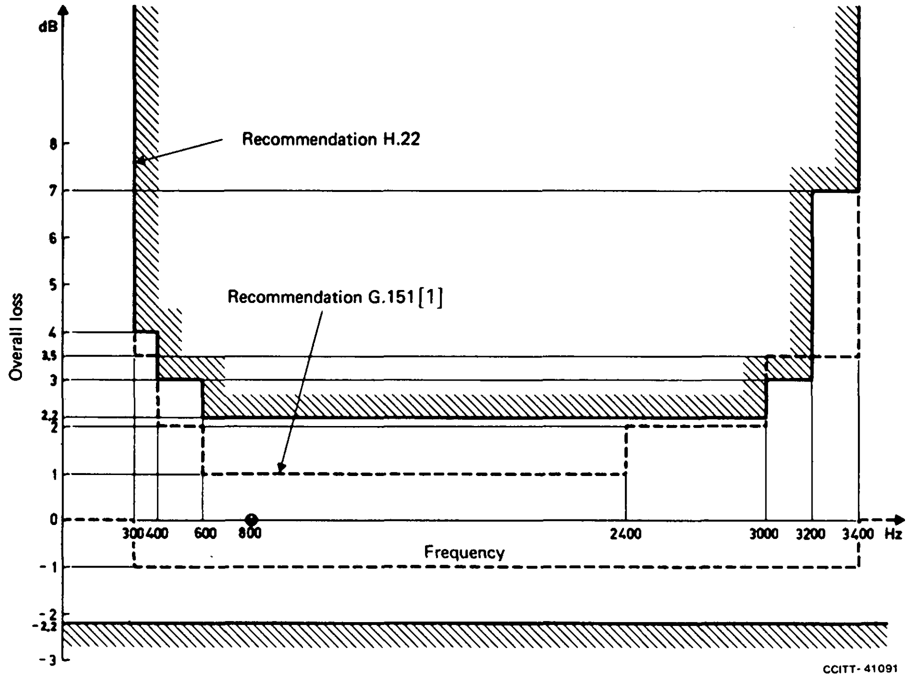


FIGURE 1/H.22

Variation with frequency of the overall loss relative to the value measured at 800 Hz with 4 kHz end-to-end sections

1.4.2 *Links with one or more 3-kHz sections* (see Table 2/H.22)

TABLE 2/H.22

Frequency range (Hz)	Overall loss relative to that at 800 Hz
Below 300	Not less than –2.2 dB, otherwise unspecified
300- 400	–2.2 to +4.0 dB
400- 600	–2.2 to +3.0 dB
600-2700	–2.2 to +2.2 dB
2700-2900	–2.2 to +3.0 dB
2900-3050	–2.2 to +6.5 dB
Above 3050	Not less than –2.2 dB, otherwise unspecified

1.5 *Noise*

1.5.1 *Uniform-spectrum random noise*

The mean psophometric noise power referred to a point of zero relative level should not exceed 80 000 pW0p (−41 dBm0p)[1].

Note – It was not possible to recommend a limit for the unweighted noise level. The CCITT psophometer with the telephone weighting network should continue to be the instrument used for specifying and measuring random noise power levels on telegraph links.

1.5.2 *Impulsive noise*

Impulsive noise should be measured with an instrument complying with Recommendation H.13 and used in the "flat" condition.

As a provisional limit for maintenance purposes, the number of impulsive noise peaks exceeding −18 dBm0 should not be more than 18 in 15 minutes.

Note – Final values are still under study.

1.6 *Crosstalk*

a) The crosstalk ratio between the go and return channels of the link should be at least 43 dB.

b) The crosstalk ratio between the link and other carrier circuits is restricted by the Recommendation cited in [3] to be not worse than 58 dB.

Crosstalk in any audio cables forming part of the terminal national sections should not normally significantly worsen the crosstalk ratio.

1.7 *Mean one-way propagation time*

The one-way propagation time referred to is the group delay as defined in [4] calculated at a frequency of about 800 Hz.

It should be noted that VFT links routed over high-altitude satellite communication systems introduce mean one-way propagation times in excess of 260 ms.

1.8 *Group-delay distortion*

Practical experience obtained up to the present shows that it is not necessary to recommend limits for group-delay distortion for 50-baud VFT links even when they are composed of several sections each provided on telephone channels of carrier systems. There is little practical experience with higher speed telegraph systems.

It may happen that under adverse conditions some telephone channels of the link are of insufficient quality to provide 24 telegraph channels. In such a case, a better combination of telephone channels must be chosen for the telegraph service.

1.9 *Frequency drift*

The frequency drift introduced by the link must not be greater than 2 Hz. According to Recommendation G.225 [5], this condition is fully met in practice even when the international line for VF telegraphy has the same composition as the 2500-km hypothetical reference circuit for the transmission system used.

1.10 *Interference caused by power supply sources*

When a sinusoidal measuring signal is transmitted over the link at a level of 0 dBm0, the level of the strongest unwanted side component should not exceed −45 dBm0 (see also Recommendation G.151 [1].

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[1] If recourse be had to synchronous operation, a higher noise level might be tolerated (such as −30 dBm0p for a particular telegraph system).

1.11 *Variation introduced by changeover to the reserve line or section*

1.11.1 *Change in insertion loss at 800 Hz*

Bearing in mind that the insertion loss of the normal line (or section) and the reserve line (or section) are both subject to variations with time, which in general will be uncorrelated, it is not possible to assign a limit to the change of insertion loss at 800 Hz introduced by the changeover procedure.

1.11.2 *Change in the insertion loss at other frequencies relative to that introduced at 800 Hz*

The insertion-loss distortion characteristic of the link when established over the normal route should be within 2 dB or less of that of the link when established over the reserve route. This limit applies over the frequency bands 300-3400 Hz or 200-3050 Hz as appropriate.

There should ordinarily be no difficulty in achieving the limit when only one portion of the link – for example, the international telegraph line or one section – has a reserve section. However, when two or more portions of the link are separately associated with reserve portions, it becomes difficult to ensure that all combinations of normal and reserve portions comply with the limit. In these circumstances, the best that can be done is to ensure that the insertion-loss characteristics of corresponding normal and reserve portions are as much alike as possible. Careful attention should be paid to the impedance of normal and reserve sections at the point where they are connected to the changeover apparatus so that errors due to changing mismatch losses are minimized. A suitable target would be for all impedances concerned to have a non-reactive return loss against 600 ohms exceeding 20 dB over the appropriate band of frequencies.

1.11.3 The nominal relative power level at 800 Hz of the normal and reserve lines or sections at the changeover points for a particular direction of transmission should be the same. This level will be that normally used in the national network of the country concerned.

2 Links via audio-frequency line plant

2.1 *Attenuation/frequency distortion*

Graph No. 6, Figure 2/H.22, shows the variations with frequency of the difference between the relative power levels at the origin and extremity of the link relative to the measured value at 800 Hz.

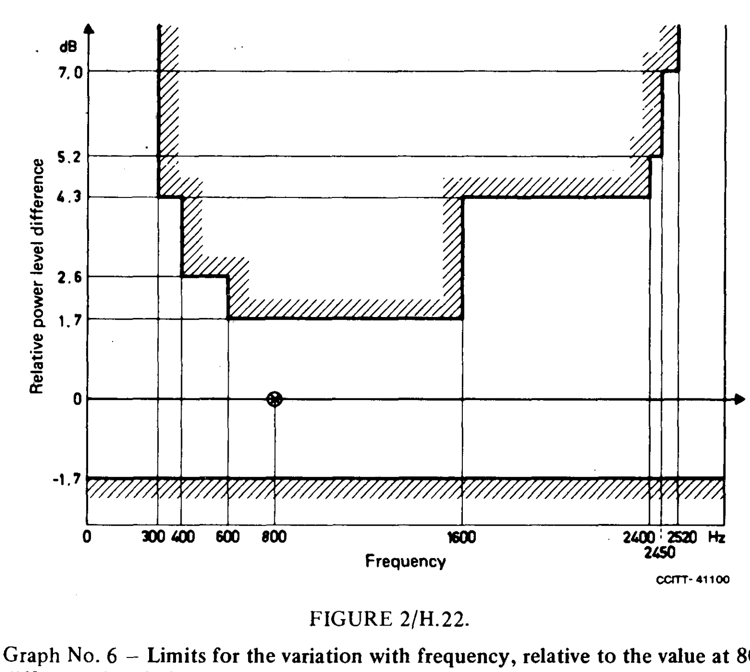

FIGURE 2/H.22.

Graph No. 6 – Limits for the variation with frequency, relative to the value at 800 Hz, of the difference in relative power levels (in dB) between the input and output of a link used for VF telegraphy (set up on a telephone circuit using the band 300-2600 Hz)

The permissible tolerances for the relative power level at the output of frontier repeaters are the same as those for 4-wire repeaters, if maintenance measurements are made by sending a power giving 1 mW at a zero relative level point (as found from the telephone circuit level diagram) to the input of the link for VF telegraphy. These tolerances are shown in Graph No. 7, Figure 3/H.22.

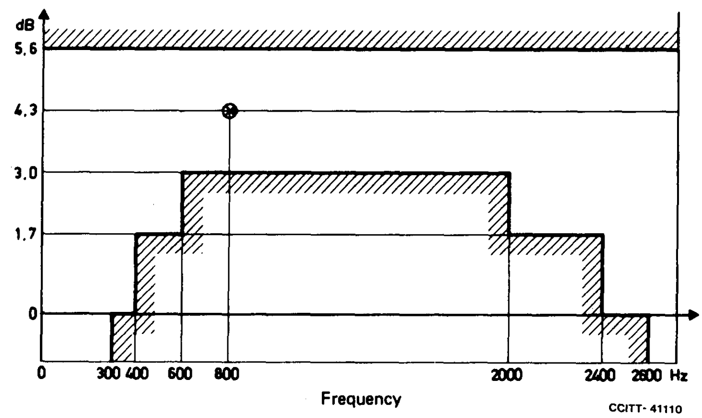

FIGURE 3/H.22.

Graph No. 7 – **Maintenance limits for the absolute power level (in dB) at the output of a frontier repeater (frontier side) for an international circuit with a bandwidth of 300-2600 Hz and used for VF telegraphy (to be measured with a sent power at the origin of the VFT link such as to give 1 mW at a zero relative level point, deduced from the level diagram of the telephone circuit)**

It does not appear necessary to fix particular limits for the variations with frequency of the level measured at the output of the frontier repeater since these may be calculated easily from the limits allowed for the relative power level.

2.2 *Level variations with time*

The relative power level at the point at the receiving end where the changeover between the VFT telegraph circuit and its reserve circuit takes place must be as constant as possible with time. Furthermore, any interruption in the circuit, even for a very short duration, spoils the quality of the telegraph transmission. Great care must therefore be taken when measurements are made on circuits and repeaters, when changing-over batteries, etc. To draw the attention of the staff to this matter, it is desirable for circuits used for VFT to be specially marked at the terminal stations and in the intermediate repeater stations.

2.3 *Freedom from modulation*

It is desirable to make special arrangements to avoid any modulation on the circuits and in the repeaters. Such modulation may in particular be caused by the variation in battery voltages or by the connection of equipment for sub-audio telegraphy to the cable pairs.

References

[1] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151.

[2] CCITT Recommendation *Stability of transmission*, Vol. IV, Rec. M.160.

[3] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151, § 4.

[4] CCITT Definition: *Group delay*, Terms and Definitions, Volume X.

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

BASIC CHARACTERISTICS OF TELEGRAPH EQUIPMENTS USED IN INTERNATIONAL VOICE-FREQUENCY TELEGRAPH SYSTEMS[1)]

(Mar del Plata, 1968; amended at Geneva, 1976)

1 Limiting power per channel

1.1 *Amplitude-modulated voice-frequency telegraph (AMVFT) systems at 50 bauds*

Administrations will be able to provide the telegraph services with carrier telephone channels permitting the use of 24 VFT channels (each capable of 50 bauds) on condition that the power of the telegraph channel signal on each channel, when a continuous marking signal (Z polarity) is transmitted, does not exceed 9 μW0.

For 18 telegraph channels only, the power so defined may be increased to 15 μW0 per telegraph channel so that even telephone channels with a relatively high noise level can then be used.

The power per telegraph channel should never exceed 35 μW0, however few channels there may be.

These limits are summarized in Table 1/H.23.

TABLE 1/H.23

Limiting power per telegraph channel when sending a continuous marking signal in AMVFT systems at 50 bauds

System	Limiting power at zero relative level point per telegraph channel when sending a continuous marking signal	
	μW0	dBm0
12 telegraph channels or less	35	−14.5
18 telegraph channels	15	−18.3
24 telegraph channels	9	−20.5

1.2 *Frequency-modulated voice-frequency telegraphy (FMVFT) systems at 50 bauds*

The mean power transmitted to line by 50-baud FMVFT systems is limited to 135 μW0 when all channels of the system are sending. This gives the limits shown in Table 2/H.23 for the mean permissible power per telegraph channel at a zero relative level point.

Some Administrations have bilateral agreements to reduce the total mean power level of FMVFT systems to −13 dBm0 (50 μW0). The CCITT encourages such reduction where feasible. The above Administrations have made their own determination of the feasibility of operating at the reduced level. As a guide, other Administrations may wish to use the suggested parameters provided by Study Group IX as given in Annex A to this Recommendation.

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1) This Recommendation reproduces, for information, some characteristics given in Recommendations R.31 [1] and R.35 [2].

TABLE 2/H.23

Normal limiting power per telegraph channel in 50-baud FMVFT systems

System	Permissible mean power at zero relative level point per telegraph channel	
	μW0	dBm0
12 telegraph channels or less	11.25	−19.5
18 telegraph channels	7.5	−21.3
24 telegraph channels	5.6	−22.5

2 Telegraph channel frequencies

For international VF 24-channel, 50-baud, non-synchronous telegraph systems the frequency series consisting of odd multiples of 60 Hz has been adopted, the lowest frequency being 420 Hz as shown in Table 3/H.23. In the case of frequency-modulated systems, these frequencies are the centre frequencies of the telegraph channels, the frequency of the signal sent to line being 30 Hz (or 35 Hz) above or below the centre frequency according to whether A or Z polarity is being sent.

TABLE 3/H.23

Telegraph channel position	Frequency (Hz)	Telegraph channel position	Frequency (Hz)
1	420	13	1860
2	540	14	1980
3	660	15	2100
4	780	16	2220
5	900	17	2340
6	1020	18	2460
7	1140	19	2580
8	1260	20	2700
9	1380	21	2820
10	1500	22	2940
11	1620	23	3060
12	1740	24	3180

In addition a pilot channel using a frequency of 300 Hz (or 3300 Hz) can be used. For details of the normal frequencies used in other types of telegraph system, see Recommendations R.37 [3], R.38 A [4] and R.38 B [5].

ANNEX A

(to Recommendation H.23)

Limits required by Study Group IX in respect of the bearer circuit for FMVFT if the total telegraph power is to be reduced to 50 microwatts (from 135 microwatts)

A.1 *Loss/frequency distortion*

The variation with frequency of the overall loss of the link with respect to the loss at 800 Hz should not exceed the limits given in Table A-1/H.23.

TABLE A-1/H.23

Frequency range (Hz)	Overall loss relative to that at 800 Hz
Below 300	Not less than –2.0 dB, otherwise unspecified
300- 500	–2.0 to +4.0 dB
500-2800	–1.0 to +3.0 dB
2800-3000	–2.0 to +3.0 dB
3000-3250	–2.0 to +4.0 dB
3250-3350	–2.0 to +7.0 dB
Above 3350	Not less than –2.0 dB, otherwise unspecified

A.2 *Random noise*

The mean psophometric noise power referred to a point of zero relative level should not exceed 32 000 pW0p (–45 dBm0p), using a psophometer in accordance with Recommendation P.53 [6].

A.3 *Impulsive noise*

The number of impulsive noise peaks exceeding –28 dBm0 should not be more than 18 in 15 minutes when measured with an impulsive noise counter in accordance with the Recommendation cited in [7].

References

[1] CCITT Recommendation *Standardization of AMVFT systems for a modulation rate of 50 bauds*, Vol. VII, Rec. R.31.

[2] CCITT Recommendation *Standardization of FMVFT systems for a modulation rate of 50 bauds*, Vol. VII, Rec. R.35.

[3] CCITT Recommendation *Standardization of FMVFT systems for a modulation rate of 100 bauds*, Vol. VII, Rec. R.37.

[4] CCITT Recommendation *Standardization of FMVFT systems for a modulation rate of 200 bauds with channels spaced at 480 Hz*, Vol. VII, Rec. R.38A.

[5] CCITT Recommendation *Standardization of FMVFT systems for a modulation rate of 200 bauds with channels spaced at 360 Hz usable on long intercontinental bearer circuits generally used with a 3-kHz spacing*, Vol. VII, Rec. R.38B.

[6] CCITT Recommendation *Psophometers (apparatus for the objective measurement of circuit noise)*, Vol. V, Rec. P.53. (Recommendation O.41.)

[7] CCITT Recommendation *Characteristics of an impulsive-noise measuring instrument for telephone-type circuits*, Orange Book, Vol. III-2, Rec. H.13, § h), ITU, Geneva, 1977.

1.3 Telephone circuits or cables used for various types of telegraph transmission or for simultaneous transmissions

Recommendation H.32 [1)]

SIMULTANEOUS COMMUNICATION BY TELEPHONY AND TELEGRAPHY ON A TELEPHONE-TYPE CIRCUIT [2)]

The CCITT,

considering

(a) that the use of a leased telephone-type circuit for simultaneous communication by telephony and telegraphy is envisaged in Recommendations D.1 [2] and H.32;

(b) that the CCITT has indicated conditions under which the simultaneous use of telephone-type circuits for telephony and telegraphy is technically tolerable;

(c) that standardization of the characteristics of equipment permitting simultaneous use of a telephone-type circuit for telephony and telegraphy is not justified, but that it is necessary to limit the power of the signals transmitted and to avoid the use of frequencies that will interfere with any telephone signalling equipment that may remain connected to the telephone-type circuit;

(d) that new demands for the allocation of particular frequencies for special purposes frequently arise and the number of frequencies used for any one purpose should not be unnecessarily extended;

(e) that the systems described below may be useful when the more modern systems advocated in Recommendation H.34 are not feasible,

unanimously recommends

(1) that, in the case of the simultaneous use of a telephone-type circuit for telephony and telegraphy, the resulting maximum permissible 1-minute mean power loading shall not exceed 50 μW0 (i.e. −13 dBm0);

(2) that, where frequency division multiplexing is employed, the general principle concerning the allocation of level to each type of service should be that the allowable mean signal power is proportional to the bandwidth assigned. This case is considered in more detail in Recommendation H.34, resulting in the total power of the telegraph signals being set at a level not exceeding 10 μW0 (i.e. approximately −20 dBm0);

(3) that there should not be more than three circuits of this type in a group of 12 telephone-type circuits and that the number of circuits of this type set up on a wideband carrier system should not exceed the number of supergroups in the system;

(4) that the telegraph signals transmitted must not interfere with any signalling equipment that may remain connected to the telephone-type circuit,

and notes

that some Administrations have permitted the use, for simultaneous telephony and telegraphy, of the frequencies 1680 Hz and 1860 Hz both for amplitude and for frequency modulation.

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1) Recommendation H.31, Volume III *Green Book* has been deleted.

2) Recommendation H.32 corresponds to Recommendation R.43 [1].

Note – If circuits equipped in accordance with the present Recommendation are used in a private network, it will be impossible to use push-button telephone sets or multifrequency signalling (e.g. Signalling System R2) in the network.

References

[1] CCITT Recommendation *Simultaneous communication by telephone and telegraph on a telephone-type circuit*, Vol. VII, Rec. R.43.

[2] CCITT Recommendation *General principles for the lease of international (continental and intercontinental) private leased telecommunication circuits*, Vol. II, Rec. D.1.

Recommendation H.34

SUBDIVISION OF THE FREQUENCY BAND OF A TELEPHONE-TYPE CIRCUIT BETWEEN TELEGRAPHY AND OTHER SERVICES

(Geneva, 1972, amended at Geneva, 1984)

1 General

The specific case considered here is that of frequency subdivision at nominally 2700 Hz, 2800 Hz and 2950 Hz of a 4-wire circuit into a main band (which can be used for telephony, data, phototelegraphy or facsimile transmission) and a secondary band, above the main band, reserved for frequency-modulation (FM) telegraphy.

The solution described in this Recommendation is recommended when the equipments are supplied by the Administration for use on leased circuits and when the derived voice channel is used on the public switched telephone network. It should be pointed out that, in accordance with the Recommendation cited in [1], Administrations are not obliged to accept any responsibility for the end-to-end quality of transmissions over connected circuits which comprise a private leased circuit network.

It is understood that any other system may be used on a leased circuit, provided the conditions concerning levels given in § 5 are observed; in this case, Administrations can give no guarantee concerning the quality of circuits, to the users of a leased circuit.

2 Main channel

With the upper part limited in this way, the main channel can be used for:

a) telephone calls of a reduced quality plus appropriate signalling system;

b) data transmission in accordance with Recommendations V.15 [2], V.16 [3], V.19 [4], V.20 [5], V.21 [6], V.22 [7], V.23 [8] V.26 [9] and V.26 *bis* [10];

c) data transmission in accordance with Recommendations V.27 [11], V.27 *bis* [12], V.27 *ter* [13];

d) data transmission in accordance with Recommendation V.29 [14];

e) facsimile transmission in accordance with Recommendation T.1 [15];

f) facsimile transmission in accordance with Recommendations T.2 [16] and T.3 [17] (Groups 1 and 2);

g) facsimile transmission in accordance with Recommendation T.4 [18] (Group 3).

For services b), c), f) and g) above, the combination of circuit and filters should be designed to keep the relative group-delay distortion and amplitude frequency response within the characteristics of the Recommendation M.1020 [21] circuit specification used, up to 100 Hz below the 3 dB filter cut-off for these services.

For service d) (the 2950 Hz filter), the group-delay distortion design limit should be under plus or minus 100 microseconds over the frequency band 550 Hz to 2850 Hz.

The characteristics of the main channel in a) are given in the case of a simple telephone circuit by Recommendation M.1040 [19] and for a private switched network by Recommendation G.171 [20].

The characteristics of the main channel in b) to g) are given in Recommendation M.1020 [21].

The characteristics of the telegraph channels are given in the R-Series Recommendations.

The level condition stated in § 5 must be complied with at all times.

With regard to service a), where applicable, account should be taken of the telephony impairment (about 2 dB) due to the limitation of the frequency band (see Recommendation G.113) [22].

With regard to services b) to g) inclusive, the subdivisions permit reliable data transmission in accordance with the following table:

Filter Stop [a)] Band Attenuation	Filter 3 dB Cutoff	Maximum Rate	Paragraph 2 Category
56 dB	2700 Hz	2400 bps	a, b and e
56 dB	2800 Hz	4800 bps	c, f and g
30 dB	2950 Hz	9600 bps	d

[a)] Stop Band operates from 100 Hz above the filter 3 dB point. For 9600 bps Alternative Voice and Data, the voice may require a 2.7 or 2.8 kHz-type filter.

3 Telegraph channels

The following are the preferred arrangements of telegraph channels in the secondary band in the case of a normal 300-3400 Hz telephone-type circuit for the three cases of subdivision:

Filter Type	2700 Hz	2800 Hz	2950 Hz
Option 1)	121, 122, 123, 124	122, 123, 124	123, 124
Option 2)	211, 123, 124	122, 212	212
Option 3)	211, 212		
Option 4)	406		

The numbering, modulation and other characteristics of the telegraph channel should comply with Recommendations R.35 [23], R.37 [24], R.38A [25] and R.70 *bis* [26], as far as possible, considering the reduced transmission level which may result in sub-standard performance.

Where the upper limit is reduced to 3050 Hz (as in telephone channels complying with Recommendation G.235 [27], it will only be possible to use two 120 Hz channels (Nos. 121 and 122) or one 240 Hz channel (No. 211) with the 2700 Hz subdivision.

With the same subdivision, the main channel may be used for:

- telephone calls,
- facsimile (including phototelegraphy),
- data,

and the secondary channel for:

- data transmission by telegraph channel.

However, private systems may be used depending on the characteristics of the portion of the band available, provided the conditions concernng levels given in § 5 are observed.

4 Filters

For the protection of telegraph channels for interference by speech or data transmission components in the upper frequency range, a filter must be used at the sending end with the nominal cut-off frequencies specified in § 2. These filters should be designed to minimize the impairment introduced to data transmission by the characteristic amplitude variations and delay distortion.

Note – This filter protects the telegraph channels from the signals transmitted on the main channel. The filters mentioned in Recommendation R.35 [23] to R.38A [25] for the protection in the opposite direction can be relied on; when the secondary channel is used for other purposes, special precautions should be taken to protect the main channel.

Sufficient protection of the main band from interference by the telegraph signals in the secondary band is assured by a similar filter at the receiving end. It is assumed that the telegraph channels are provided with filters so as to meet the provisions of Recommendations R.35 [23], R.37 [24] or R.38A [25].

5 Levels

The general principle concerning levels for each type of service is that the allowable mean signal power is proportional to the bandwidth assigned.

Of the maximum permissible 1-minute mean loading of 50 μW0 (–13 dBm0), 10 μW0 are allocated to the secondary band and the remaining 40 μW0 to the main band. In the case of telephony this implies that normal levels for speech and signalling can be retained (given in Recommendations G.223 [28] as 32 μW), and for the telegraph band one telegraph channel at –20 dBm0 or four channels at –26 dBm0.

6 Limitation of amplitude

It may be desirable to impose a limit on amplitude in the main band path so that the onset of nonlinearity in the common transmission path will not cause intermodulation and possible interference with the telegraph channels.

7 Network control

In many instances, Administrations may find it desirable to control the remote-end equipment for various purposes, such as choosing different filter cut-offs, reconfiguring the number and bandwidth of telegraph channels (as per § 3 above), switching from voice to data transmission remotely, etc. Furthermore, remote testing via loopbacks specified in Recommendation V.54 [29] may be desirable on such leased networks. For such purposes, a low-speed, narrow band control channel may be added. Such a channel shall operate in the controlled carrier mode, i.e. shall transmit a burst of tone only during a request for such a remote function. Such a channel shall not occupy any additional allocated bandwidth (i.e. under the modem reverse channel band) and its presence shall not cause interference to speech, telephone signalling, facsimile or data transmission, nor shall it disrupt any of the telephone channels. Furthermore, such a channel shall be equipped with sufficient protection by means of redundant coding to be able to distinguish between actual commands for remote control and random signals due to the overlapping spectra of speech, or data transmission originating from modems specified in §§ 2 b) to 2 g).

References

[1] CCITT Recommendation *General principles for the lease of international (continental and intercontinental) private leased telecommunication circuits*, Vol. II, Rec. D.1, § 5.8.

[2] CCITT Recommendation *Use of acousting coupling for data transmission*, Vol. VIII, Rec. V.15.

[3] CCITT Recommendation *Medical analogue data transmission modems*, Vol. VIII, Rec. V.16.

[4] CCITT Recommendation *Modems for parallel data transmission using telephone signalling frequencies*, Vol. VIII, Rec. V.19.

[5] CCITT Recommendation *Parallel data transmission modems standardized for universal use in the general switched telephone network*, Vol. VIII, Rec. V.20.

[6] CCITT Recommendation *300 bits per second duplex modem standardized for use in the general switched telephone network*, Vol. VIII, Rec. V.21.

[7] CCITT Recommendation *1200 bits per second duplex modem standardized for use on the general switched telephone network and on the leased circuits*, Vol. VIII, Rec. V.22.

[8] CCITT Recommendation *600/1200-baud modem standardized for use in the general switched telephone network*, Vol. VIII, Rec. V.23.

[9] CCITT Recommendation *2400 bits per second modem standardized for use on 4-wire leased telephone-type circuits*, Vol. VIII, Rec. V.26.

[10] CCITT Recommendation *2400/1200 bits per second modem standardized for use in the general switched telephone network*, Vol. VIII, Rec. V.26 *bis*.

[11] CCITT Recommendation *4800 bits per second modem with manual equalizer standardized for use on leased telephone-type circuits*, Vol. VIII, Rec. V.27.

[12] CCITT Recommendation *4800/2400 bits per second modem with automatic equalizer standardized for use on leased telephone-type circuits*, Vol. VIII, Rec. V.27 *bis*.

[13] CCITT Recommendation *4800/2400 bits per second modem standardized for use in the general switched telephone network*, Vol. VIII, Rec. V.27 *ter*.

[14] CCITT Recommendation *9600 bits per second modem standardized for use on point-to-point 4-wire leased circuits*, Vol. VIII, Rec. V.29.

[15] CCITT Recommendation *Standardization of phototelegraph apparatus*, Vol. VII, Rec. T.1.

[16] CCITT Recommendation *Standardization of Group 1 facsimile apparatus for document transmission*, Vol. VII, Rec. T.2.

[17] CCITT Recommendation *Standardization of Group 2 facsimile apparatus for document transmission*, Vol. VII, Rec. T.3.

[18] CCITT Recommendation *Standardization of Group 3 facsimile apparatus for document transmission*, Vol. VII, Rec. T.4.

[19] CCITT Recommendation *Characteristics of ordinary quality international leased circuits*, Vol. IV, Rec. M.1040.

[20] CCITT Recommendation *Transmission characteristics of leased circuits forming part of a private telephone network*, Vol. III, Rec. G.171.

[21] CCITT Recommendation *Characteristics of special quality international leased circuits with special bandwidth conditioning*, Vol. IV, Rec. M.1020.

[22] CCITT Recommendation *Transmission impairments*, Vol. III, Rec. G.113.

[23] CCITT Recommendation *Standardization of FMVFT systems for a modulation rate of 50 bauds*, Vol. VII, Rec. R.35.

[24] CCITT Recommendation *Standardization of FMFVT systems for a modulation rate of 100 bauds*, Vol. VII, Rec. R.37.

[25] CCITT Recommendation *Standardization of FMFVT systems for a modulation rate of 200 bauds, with channels spaced at 480 Hz*, Vol. VII, Rec. 38A.

[26] CCITT Recommendation *Numbering of international VFT channels*, Vol. VII, Rec. R.70 *bis*.

[27] CCITT Recommendation *16-channel terminal equipments*, Vol. III, Rec. G.235.

[28] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*, Vol. III, Rec. 223.

[29] CCITT Recommendation *Loop test devices for modems*, Vol. VIII, Rec. V.54.

1.4 Telephone-type circuits used for facsimile telegraphy

Recommendation H.41 [1)]

PHOTOTELEGRAPH TRANSMISSIONS ON TELEPHONE-TYPE CIRCUITS

Note – As far as carrier circuits are concerned, this Recommendation applies only to systems established on the basis of 12-channel groups; systems using 16-channel groups will form the subject of a later study.

When carrier circuits are used, frequency modulation offers advantages over amplitude modulation in that it does not overload carrier systems and avoids the influence of sudden level variations or noise. It is therefore to be preferred. The provisions of Recommendation T.1 [2] should be applied.

For these reasons, the CCITT

unanimously recommends

that phototelegraph transmissions over telephone circuits require that the following conditions be observed, according to the way in which the circuits are used for phototelegraphy:

1 Circuits permanently used for phototelegraphy

It seems that these circuits are few. In any case, they should even more easily meet the characteristics given in § 2 below.

2 Circuits used normally (and preferentially) for phototelegraphy

2.1 *Types of circuit to be used*

Two-wire circuits have no practical value for phototelegraphy because of feedback phenomena.

For the same reason, 4-wire circuits should be extended to the phototelegraph stations at the appropriate amplifier stations, the terminating units and echo suppressors always being disconnected.

The constitution of a phototelegraph circuit is given in Figure 1/H.41.

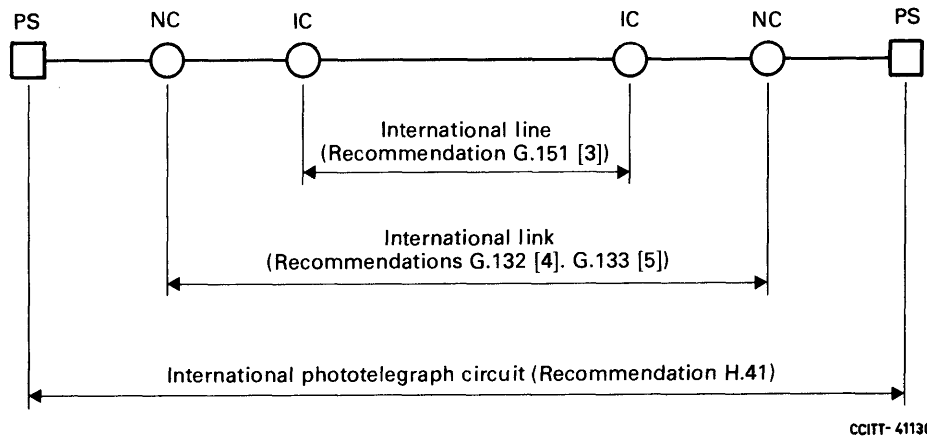
2.2 *Overall loss*

The same conditions apply to the overall transmission loss of 4-wire circuits used for phototelegraphy as apply, in general, for telephony.

2.3 *Sent signal power*

The emission voltage for the phototelegraph signal corresponding to maximum amplitude should be so adjusted that the maximum power level of the signal at the zero relative level point is −13 dBm0 for frequency-modulation phototelegraph transmissions and that the peak signal power level for amplitude-modulation phototelegraph transmissions in principle should be −3 dBm0. In the case of amplitude modulation, the level of the signal corresponding to black is usually about 30 dB lower than that of the signal corresponding to white.

1) Recommendation H.41 corresponds to Recommendation T.11 [1].



IC Terminal international centre
NC Terminal national centre
PS Phototelegraph station

Note — The phototelegraph circuit is set up on lines according to the terminology used by Study Group IV in Recommendations M.1010 [6] and M.1015 [7].

FIGURE 1/H.41

Constitution of a phototelegraph circuit

2.4 *Relative levels*

If phototelegraph transmissions take place simultaneously from a transmitting station to several receiving stations, arrangements shall be made at the junction point so that, on the circuits following the junction point, the same power levels are maintained as those prescribed for individual transmissions.

2.5 *Attenuation/frequency distortion*

The limits for attenuation/frequency distortion on international circuits used for phototelegraphy are given in Recommendation G.151 [3] concerning telephone circuits. The attenuation/frequency distortion between two terminal national centres shall therefore not exceed the limits indicated in Recommendation G.132 [4] and it will not normally be necessary to compensate the distortion of the lines linking the phototelegraph stations to the terminal national centres in order to obtain, for amplitude-modulated phototelegraph transmission, an attenuation/frequency distortion between phototelegraph stations of less than 8.7 dB in the wanted band.

2.6 *Variation of circuit overall loss with time*[2), 3)]

2.6.1 The objective is that:

2.6.1.1 the difference between the mean value and the nominal of the transmission loss value should not exceed 0.5 dB;

2.6.1.2 the standard deviation from the mean value should not exceed 1 dB.

However, in the case of circuits set up wholly or partly on older equipment, where the international line consists of two or more circuit sections, a standard deviation not exceeding 1.5 dB may be accepted.

2) See Recommendation M.160 [8] and [9].

3) The provisions specified under § 2.6 are provisional and need further study from the facsimile transmission point of view.

2.6.2 The method for achieving the above objective values is left to the discretion of Administrations (better maintenance, fitting of automatic regulators, etc.).

2.6.3 The assumption is made that these limits for the variation of loss with time of a single circuit may be compared to limits for loss measurements made on a set of circuits at a given time. Experience indicates that such a comparison has a practical validity although it has not been fully demonstrated at this time. Administrations are encouraged to use this Recommendation as giving currently practical limits for sets of circuits. This does not preclude the application of these limits to single circuits, should this prove practical at any time.

2.7 *Phase distortion*

Phase distortion limits the range of satisfactory phototelegraph transmissions. Differences between the group delays of a telephone circuit, in the interval of the phototelegraph transmission, should not exceed

$$\Delta t \leqslant \frac{1}{2 f_p}$$

f_p = maximum modulating frequency corresponding to the definition and scanning speed. (See Recommendation H.42.)

2.8 *Interference*

Interfering currents, whatever their nature, should not exceed the CCITT recommended limits for telephone circuits.

3 Telephone circuits rarely used for phototelegraphy

3.1 *Transmission characteristics*

It seems that the majority of the characteristics specified by the CCITT for modern telephone circuits are sufficient to permit phototelegraph transmissions on a circuit chosen at random in a group of circuits normally used for telephone working. However, it is not certain that such a circuit would have a sufficiently low phase distortion for such use, particularly channels 1 and 12 of a 12-circuit group, use of which is not advised. The influence of phase distortion is more noticeable in frequency modulation.

With amplitude modulation there is a further risk that phototelegraph transmissions will be subject to faulty modulation because the special precautions applied to circuits regularly used for phototelegraphy (see § 2.6 above) cannot be applied to circuits taken at random.

3.2 *Precautions concerning signalling*

As long as automatic switching for phototelegraph circuits is not envisaged, the signal receiver can be disconnected so that no signalling disturbances can occur even when frequency modulation is used. However, if frequency modulation is used for phototelegraph transmission and if it is impracticable to disconnect the signal receiver, then it would be desirable, in the case of the single-frequency system, that a blocking signal be transmitted along with the picture signal to operate the guard circuit and render the receiver inoperative.

It is also apparent that the frequency of such a blocking signal should lie well outside the range of frequencies involved in the picture transmission and the frequency and the level of the blocking signal must depend on the characteristics of the VF receiver (or receivers in the case of a tandem international connection), as designed by different Administrations to meet the specification to be prescribed for international signalling.

In the case of the two-frequency international signalling system, the CCITT has indicated its view that no interference will take place.

References

[1] CCITT Recommendation *Phototelegraph transmission on telephone-type circuits*, Vol. VII, Rec. T.11.

[2] CCITT Recommendation *Standardization of phototelegraph apparatus*, Vol. VII, Rec. T.1.

[3] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151.

[4] CCITT Recommendation *Attenuation distortion*, Vol. III, Rec. G.132.

[5] CCITT Recommendation *Group-delay distortion*, Vol. III, Rec. G.133.

[6] CCITT Recommendation *Constitution and nomenclature of international leased circuits*, Vol. IV, Rec. M.1010.

[7] CCITT Recommendation *Types of transmission on leased circuits*, Vol. IV, Rec. M.1015.

[8] CCITT Recommendation *Stability of transmission*, Vol. IV, Rec. M.160.

[9] *Statistical theory requirements*, Green Book, Vol. IV.2, Supplement No. 1.6, ITU, Geneva, 1973.

Recommendation H.42 [1)]

RANGE OF PHOTOTELEGRAPH TRANSMISSIONS ON A TELEPHONE-TYPE CIRCUIT

Note – In case of carrier circuits, this Recommendation applies only to systems established on the basis of 12-channel group links. Systems using 16-channel group links will be the subject of subsequent study.

(a) The differences between the group delays of the various frequencies and the width of the transmission band actually usable on a circuit for telephony give rise, when phototelegraph signals are started or stopped, to transient phenomena which limit the phototelegraph transmission speed.

(b) The range of phototelegraph calls of satisfactory quality, for a given transmission speed, depends especially on the constitution of the circuit, i.e. on:

- the loading and length, in the case of audio-frequency circuits;
- the number of 12-channel group links used in the case of carrier circuits, and on the choice of the carrier frequency for amplitude-modulated phototelegraph transmission, or on the mean frequency in the case of frequency modulation.

(c) Phototelegraph transmission of satisfactory quality requires that the limits of difference between the group delays in the transmitted frequency band, as shown in Figure 1/H.42, are not to be exceeded.

Note – The spot is assumed to have the same dimensions in both directions (square or circular).

(d) The CCITT has recommended group-delay distortion limits for international telephone circuits (see Recommendation G.133 [1]).

For these reasons, the CCITT

unanimously recommends

that, as regards the effect of phase distortion on phototelegraph transmission quality, the carrier frequency (where amplitude modulation is used) or the mean frequency (when frequency modulation is used) must be chosen in such a way that it is as near as possible to the frequency which has the minimum group delay on the telephone circuit.

1) This Recommendation corresponds to Recommendation T.12.

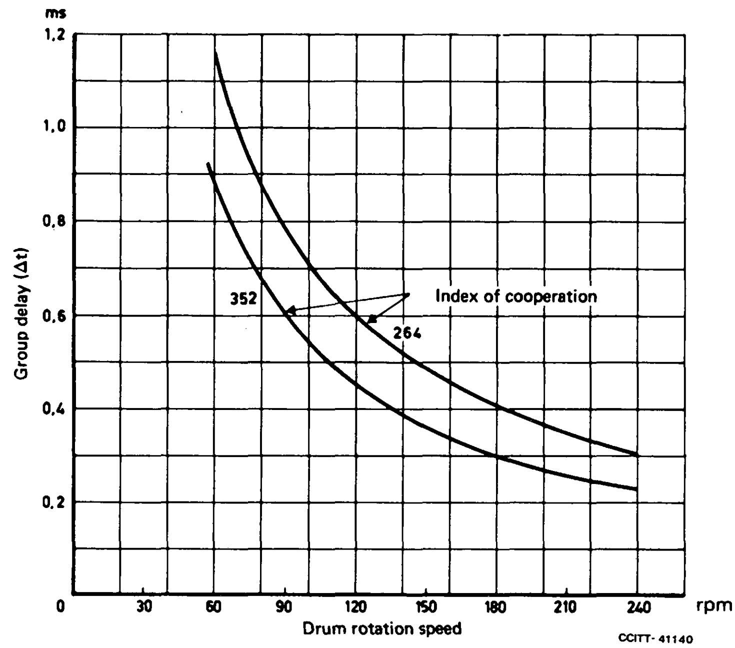


FIGURE 1/H.42
Permissible group-delay distortion in the transmitted frequency band as a function of the phototelegraph transmission speed

1 Circuits permanently used for phototelegraphy

1.1 It will generally be possible, by agreement between Administrations, to choose a circuit satisfying stricter limits than those specified above from the point of view of phase distortion.

1.2 Moreover, it will be possible to compensate phase distortions by inserting phase equalizers and to effect phototelegraph transmissions occupying the whole nominal band of the circuit.

2 Circuits normally (or preferentially) used for phototelegraphy

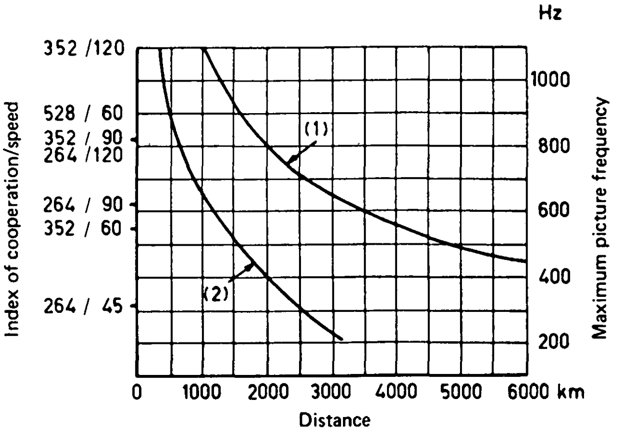
2.1 The greater the differences between the delays in the transmission intervals, the narrower should be the bandwidth chosen (leading to a lower phototelegraph definition or transmission speed).

2.2 Hence, audio-frequency circuits should in any case have only small loads.

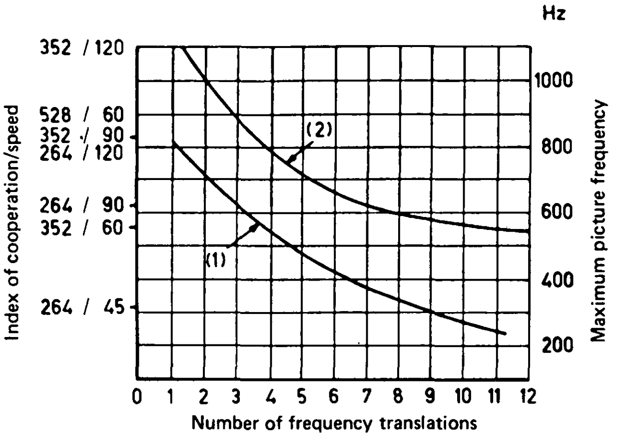
2.3 Phase distortion is well within the limits indicated above, in the case of carrier circuits, if a single modern-type carrier system is considered (and considering especially the telephone channels in the middle of a 12-channel group of such a system).

2.4 Nevertheless, it would be unjustifiable from the financial point of view to make the aforementioned recommendation concerning phase distortion stricter, simply with a view to the occasional use of only a few circuits for high-speed phototelegraph transmissions.

2.5 The curves of Figure 2/H.42 give information on the relative performances of amplitude- and frequency-modulated phototelegraph transmissions over audio-frequency and carrier telephone circuits.



a) *Audio circuits*



b) *Carrier circuits*

Curves (1) – AM carrier = 1300 Hz

Curves (2) – { FM = 1900 ± 400 Hz; AM carrier = 1900 Hz }

FIGURE 2/H.42

Range of phototelegraph transmissions

3 Telephone circuits rarely used for phototelegraphy

If phototelegraph connections are set up on circuits selected at random from modern-type groups of telephone circuits (for example, by automatic switching), a circuit may be taken which has too high a degree of phase distortion, particularly if it has been set up on channel 1 or 12 of a 12-channel group, the use of which is deprecated. It is impossible, in this case, to draw up general information on the range of phototelegraph transmissions; however, it will be possible to meet the conditions for a transmission of adequate quality if the phototelegraph connection comprises only one 12-channel group link and if transmission is effected in normal conditions as outlined in Recommendation T.1 [2].

References

[1] CCITT Recommendation *Group-delay distortion*, Vol. III, Rec. G.133.

[2] CCITT Recommendation *Standardization of phototelegraph apparatus*, Vol. VII, Rec. T.1.

Recommendation H.43 [1)]

DOCUMENT FACSIMILE TRANSMISSIONS ON LEASED TELEPHONE-TYPE CIRCUITS

(Geneva, 1964, amended at Mar del Plata, 1968, and at Geneva, 1972, 1976 and 1980)

1 Type of circuits to be used

The telephone-type circuits used should have characteristics as recommended in Recommendation H.12.

Note – If the leased circuit is used alternately for telephone conversation and facsimile transmission and if the latter is unidirectional, it is not necessary to provide for disabling echo suppressors located on the long-distance leased circuit. However, when such a circuit is used for the simultaneous operation in both directions, appropriate measures should be taken to disable echo suppressors before the actual facsimile transmission takes place.

2 Modulation

Equipment conforming to Recommendation T.2 [2] or Recommendation T.3 [3] may be used. In the case of Recommendation T.2 [2] equipment, either amplitude or frequency modulation may be chosen.

3 Power

The maximum power output of the transmitting apparatus into the line shall not exceed 1 mW whatever the frequency.

For frequency modulation equipment conforming to Recommendation T.2 [2] the level at the transmitter output shall be so adjusted that the level of the facsimile and control signals on the trunk circuit does not exceed –13 dBm0 regardless of the type of operation (duplex or simplex).

For amplitude modulation equipment conforming to Recommendation T.2 [2], higher black levels may be used provided the mean power in any hour, in one direction of transmission, does not exceed 32 μW (–15 dBm0) at a zero relative level point of the trunk circuit.

For equipment conforming to Recommendation T.3 [3], higher white levels may be used provided the mean power in any hour, in one direction of transmission, does not exceed 32 μW (–15 dBm0) at a zero relative level point of the trunk circuit.

4 Multipoint transmission

If facsimile transmissions take place simultaneously from a transmitting station to several receiving stations, arrangements shall be made at the junction points so that, on the circuits following the junction points, the same power levels are maintained as those prescribed for individual transmissions.

5 Phase distortion

Equipment conforming to Recommendation T.2 [2] should not require any special treatment. However, equipment conforming to Recommendation T.3 [3] may require phase distortion correction in some cases.

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1) Recommendation H.43 corresponds to Recommendation T.10 [1].

References

[1] CCITT Recommendation *Document facsimile transmission on leased telephone-type circuits*, Vol. VII, Rec. T.10.

[2] CCITT Recommendation *Standardization of Group 1 facsimile apparatus for document transmission*, Vol. VII, Rec. T.2.

[3] CCITT Recommendation *Standardization of Group 2 facsimile apparatus for document transmission*, Vol. VII, Rec. T.3.

1.5 Characteristics of data signals

Recommendation H.51 [1)]

POWER LEVELS FOR DATA TRANSMISSION OVER TELEPHONE LINES

(Mar del Plata, 1968; amended at Geneva, 1980)

The objectives in specifying data signal levels are as follows:

a) To ensure satisfactory transmission and to permit coordination with devices such as signalling receivers or echo suppressors, the data signal levels on international circuits should be controlled as closely as possible.

b) To ensure correct performance of multichannel carrier systems from the point of view of loading and noise, the mean power of data circuits should not differ much from the conventional value of channel loading (-15 dBm0 for each direction of transmission, see Note below). This conventional value makes allowance for a reasonable proportion P (dependent on the transmission systems and probably less than 50%; the value will have to be specified in subsequent studies) of the channels in a multichannel system being used for non-speech applications at fixed power levels at about -13 dBm0 for each direction of transmission.

   If the proportion of non-speech applications (including data) does not exceed the above value P, the mean power of -13 dBm0 for each direction of transmission would be allowable for data transmission also.

   However, assuming that the proportion of non-speech circuits is appreciably higher than P (due to the development of data transmission) on international carrier systems, a reduction of this power by 2 dB might be reasonable. (These values require further study).

   Note – The distribution of long-term mean power among the channels in a multichannel carrier telephone system (conventional mean value of -15 dBm0), probably has a standard deviation in the neighbourhood of 4 dB (see [2]).

c) It is probable that Administrations will wish to fix specific values for the signal power level of data modulators either at the subscribers' line terminals or at the local exchanges. The relation between these values and the power levels on international circuits depends on the particular national transmission plan; in any case, a wide range of losses among the possible connections between the subscriber and the input to international circuits must be expected.

d) Considerations a) to c) suggest that specification of the maximum data signal level only is not the most useful form. One alternative proposal would be to specify the nominal power at the input to the international circuit. The nominal power would be the statistically estimated mean power obtained from measurement on many data transmission circuits.

1) Recommendation H.51 corresponds to Recommendation V.2 [1].

For these reasons, the CCITT

unanimously recommends:

1 Data transmission over leased telephone-type circuits set up on carrier systems

1.1 The maximum power output of the subscriber's apparatus into the line shall not exceed 1 mW.

1.2 For systems transmitting tones continuously, e.g. frequency modulation systems, the maximum power level at the zero relative level point shall be −13 dBm0. When transmission of data is discontinued for any appreciable time, the power level should preferably be reduced to −20 dBm0 or lower.

1.3 For systems not transmitting tones continuously, e.g. amplitude-modulation systems, the signal characteristics *should meet all of the following requirements*:

- i) The maximum value of the 1-minute mean power shall not exceed −13 dBm0.
- ii) Provisionally, the maximum value of the instantaneous power shall not exceed a level corresponding to that of a sine wave signal of 0 dBm0. This limit should be confirmed or amended after further study.
- iii) Provisionally, the maximum signal power determined for a 10-Hz bandwidth centred at any frequency shall not exceed −10 dBm0. This limit should be confirmed or amended after further study.

Note 1 – It is estimated that the proportion of international circuits which are carrying data transmissions is approximately 20%. If the proportion should reach a high level (approximately 50% or even less in the case of high-usage systems), the limits now proposed would need to be reconsidered.

Note 2 – Supplement No. 16 gives information on the out-of-band power of signals applied to leased telephone-type circuits.

2 Data transmission over the switched telephone network

2.1 The maximum power output of the subscriber's equipment into the line shall not exceed 1 mW at any frequency.

2.2 For systems transmitting tones continuously, such as frequency- or phase-modulation systems, the output power level of the subscriber's equipment should be fixed at the time of installation to allow for loss between his equipment and the point of entry to an international circuit, so that the corresponding nominal level of the signal at the international circuit input shall not exceed −13 dBm0.

2.3 For systems not transmitting tones continuously, e.g., amplitude-modulation systems, the signal characteristics *should meet all of the following requirements* (see also Note 1 to § 1.3):

- i) The maximum value of the 1-minute mean power shall not exceed −13 dBm0.
- ii) Provisionally, the maximum value of the instantaneous power shall not exceed a level corresponding to that of a sinewave signal of 0 dBm0. This limit should be confirmed or amended after further study.
- iii) Provisionally, the maximum signal power determined for a 10-Hz bandwidth centred at any frequency shall not exceed −10 dBm0. This limit should be confirmed or amended after further study.

Note 1 – In practice, it is no easy matter to assess the loss between a subscriber's equipment and the international circuit, so that § 2 of the present Recommendation should be taken as providing general planning guidance.

Note 2 – In switched connections, the loss between subscribers' telephones may be high: 30 to 40 dB. The level of the signals received will then be very low, and these signals may suffer disturbance from, for example, the dialling pulses sent over other circuits.

If there is likely to be a heavy demand for international connections for data transmission over the switched network, some Administrations might want to provide special 4-wire subscriber lines. If so, the levels to be used might be those proposed for leased circuits.

References

[1] CCITT Recommendation *Power levels for data transmission over telephone lines*, Vol. VIII, Rec. V.2.

[2] *Measurement of the load of telephone circuits under field conditions*, Yellow Book, Supplement No. 5, ITU, Geneva, 1981.

Recommendation H.52

TRANSMISSION OF WIDE-SPECTRUM SIGNALS (DATA, FACSIMILE, ETC.) ON WIDEBAND GROUP LINKS

(Mar del Plata, 1968; amended at Geneva, 1972, 1976 and 1980)

Links meeting the provisions of Recommendation H.14 should be used.

1 Power level

1.1 The mean power level of the wideband signal over the range 60-108 kHz should not exceed $-15 + 10 \log_{10} 12 = -4$ dBm0.

1.2 In order to limit cross-modulation effects in wideband systems, the power level of any individual spectral component in the band 60-108 kHz should not exceed -10 dBm0 except for spectral components which are at multiples of 4 kHz (see the Recommendation cited in [1]).

With regard to its effect on non-telephone type signals, a discrete component is defined as a signal of sinusoidal form with a minimum duration of about 100 ms.

1.3 To protect the group or supergroup link pilots (used to establish wideband circuits) against other wide-spectrum signals (data, facsimile, etc.), it is recommended that the power spectrum emitted about the pilot frequency be limited in the equipment which transmits these signals (see Figure 1/H.52).

For continuous spectrum signals, the spectral density in the band $f_0 \pm 25$ Hz should not exceed -70 dBm0/Hz.

Other indications are given in the Recommendation cited in [2].

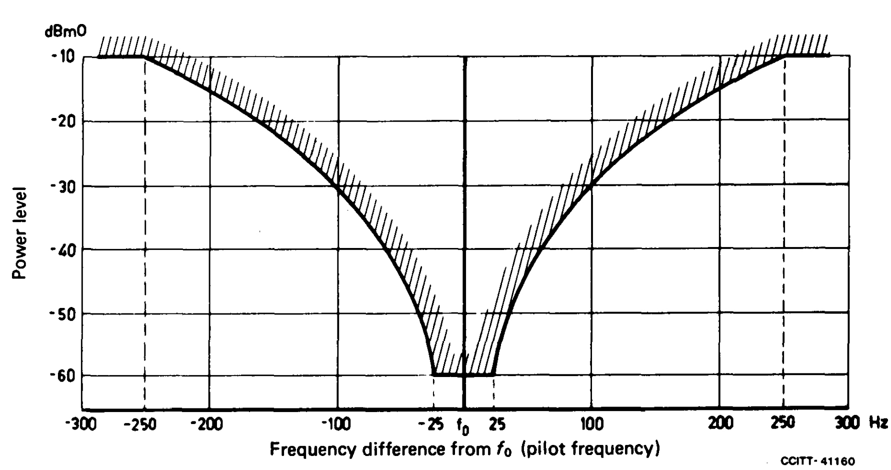


FIGURE 1/H.52

Maximum permissible level of discrete frequency components of wide-spectrum signals (group and supergroup signals) in the vicinity of group and supergroup pilot frequencies

2 Limitation of the power spectrum outside the band 60-108 kHz

The power level produced by the terminal equipment connected to the wideband group link shall not exceed −73 dBm0p in any 4-kHz band outside the range 60-108 kHz.

However, for the frequencies 48 and 56 kHz, with a precision of ± 1 Hz, an unweighted value of −50 dBm0 is permitted.

If the terminal equipment itself does not meet these conditions (e.g. a modem which just complies with the provisions of Recommendation V.35 [3]), an additional filter must be applied before the point of connection to the leased group link.

References

[1] CCITT Recommendation *Overall recommendations relating to carrier-transmission systems,* Vol. III, Rec. G.221, § 2.2.

[2] CCITT Recommendation *Pilots on groups, supergroups, etc.*, Vol. III, Rec. G.241, § 7.

[3] CCITT Recommendation *Data transmission at 48 kilobits per second using 60-108 kHz group band circuits,* Vol. VIII, Rec. V.35.

Recommendation H.53

TRANSMISSION OF WIDE-SPECTRUM SIGNALS (DATA, ETC.) OVER WIDEBAND SUPERGROUP LINKS

(Mar del Plata, 1968; amended at Geneva, 1972, 1976 and 1980)

Links meeting the provisions of Recommendation H.15 should be used.

1 Power level

1.1 The mean power level of the wideband signal over the range 312-552 kHz should not exceed $-15 + 10 \log_{10} 60 = +3$ dBm0.

1.2 In order to limit cross-modulation effects in wideband systems, the power level of any individual spectral component in the band 312-552 kHz should not exceed −10 dBm0, except for spectral components which are at multiples of 4 kHz, (see the Recommendation cited in [1]).

With regard to its effect on non-telephone type signals, a discrete component is defined as a signal of sinusoidal form with a minimum duration of about 100 ms.

1.3 In addition to § 1.2 above, the energy spectrum transmitted in the neighbourhood of the pilot frequencies should be limited in accordance with the Recommendation cited in [2].

2 Limitation of the power spectrum outside the band 312-552 kHz

The power level produced by the terminal equipment connected to the wideband supergroup link shall not exceed −73 dBm0p in any 4-kHz band outside the range 304-560 kHz.

If the terminal equipment itself does not meet these conditions, an additional filter must be applied before the point of connection to the leased supergroup link.

References

[1] CCITT Recommendation *Overall recommendations relating to carrier-transmission systems,* Vol. III, Rec. G.221, § 2.2.

[2] CCITT Recommendation *Pilots on groups, supergroups, etc.*, Vol. III, Rec. G.241, § 7.

SECTION 2

CHARACTERISTICS OF VISUAL TELEPHONE SYSTEMS

Recommendation H.100

VISUAL TELEPHONE SYSTEMS

(former Recommendation H.61, Geneva, 1980;
amended at Malaga-Torremolinos, 1984)

1 Definition

The **visual telephone service** is generally a two-way telecommunication service which uses a switched network of broadband analogue and/or digital circuits to establish connections among subscriber terminals, primarily for the purpose of transmitting live or static pictures.

Special application one-way systems, e.g. surveillance and some information retrieval systems, or a non-switched videoconference service, can be regarded as degenerate cases of the visual telephone service.

The visual telephone service also includes the associated speech.

2 Facilities to be offered

The design of the visual telephone service shall be such as to offer at least the following basic facilities:

a) Transmission of live pictures such as head and shoulders of one person or a small group of persons, with moderate definition.

b) Transmission of the associated speech.

c) Transmission of graphics information such as drawings and documents with high definition (e.g. 625 lines or 525 lines).

d) Video conference service, with or without the use of split-screen techniques.

The above-mentioned services shall, in general, be bi-directional, although uni-directional operation should be possible. Also, some of the facilities can be omitted, if not required, in order to minimize costs.

Note – At the subscriber terminal, the use of ancillary equipments, e.g. for document reproduction, video tape recordings, etc., shall be possible.

3 System parameters

3.1 *Picture standards*

3.1.1 The video standards of the subscriber sets shall be compatible with, readily convertible to, or identical to, the local broadcast television standards.

3.1.2 Two classes of picture standards are recommended for the visual telephone system. They are given in Table 1/H.100.

TABLE 1/H.100

Picture standards

Class	Items	The region to which the figures should be applied	
		Regions where TV broadcasting uses 25 pictures per second	Regions where TV broadcasting uses 30 pictures per second
a	Number of horizontal scanning lines	625	525
	Pictures per second	25 (2:1 interleaved)	30 (2:1 interleaved)
	Aspect ratio	4:3	4:3
	Video bandwidth	5 MHz	4 MHz
b	Number of horizontal scanning lines	313	263
	Pictures per second	25 (2:1 interleaved)	30 (2:1 interleaved)
	Aspect ratio	4:3	4:3
	Video bandwidth	1 MHz	1 MHz

Class *a* standards are identical to the local broadcast video standards and will, in most cases, give sufficient definition for real-time picture transmission of a group of people (e.g. for conferencing) and of graphics material.

Class *b* standards give sufficient definition for real-time transmission of a head and shoulder picture of one person or a small group. For the transmission of graphics information or other still pictures with high definition, a slow-scan technique has to be applied. For instance, a system using 625 or 525 horizontal scanning lines and 5, or less, pictures per second which gives a Class *a* definition in the 1 MHz bandwidth.

Further study is required to define slow scanning parameters.

4 Characteristics relating to split-screen techniques for Class *a* television conference systems [1)]

In television conference systems which use split-screen techniques to make more effective use of the picture area, the following features for the terminals and transmitted signals are recommended. Preferred seating arrangement for such systems are given in Annex A.

---

1) Split-screen techniques for systems using Class *b* standards require future study.

4.1 *Picture format*

The transmitted picture should be 4 : 3 aspect ratio, split into upper and lower halves corresponding to the groups of seats. Viewed from the camera system, the left-hand group should be in the upper half and the right-hand group in the lower half.

The split should occur at the end of lines 166 and 479 for 625-line television systems and at the end of line 142 in Field 1 and line 141 in Field 2 for 525-line television systems, as shown in Figure 1/H.100.

Before display, the receive equipment may discard half lines and first and last lines which are liable to be averaged during standards conversion or vertical aperture correction of mixed signals.

4.2 *Identification signal for split-screen system*

The identification signal for split-screen system should be inserted in the vertical blanking period, because the control is required for each television frame or field.

The line where the identification signal is inserted and its signal format are under study.

4.3 *Compatibility with non-split-screen systems*

The simplest kind of a video telephone terminal is composed of a single camera and other equipments. These terminals may be interconnected with split-screen system terminals. In that case, mechanical masks (if used) for the two split-screen displays (aspect ratio = 4 : 1.5) need to be removed, or if a display with 4 : 3 aspect ratio needs to be installed additionally.

4.4 *Cameras and displays arrangement*

The entrance pupils of the TV camera optical system should be as near as possible to the centre of the TV display showing remote conferees, in order to minimize errors in eye contact angle.

Unless means are employed to place these pupils in line with the display, e.g. by use of half-silvered mirrors, the camera system should be sited above the display and central to it.

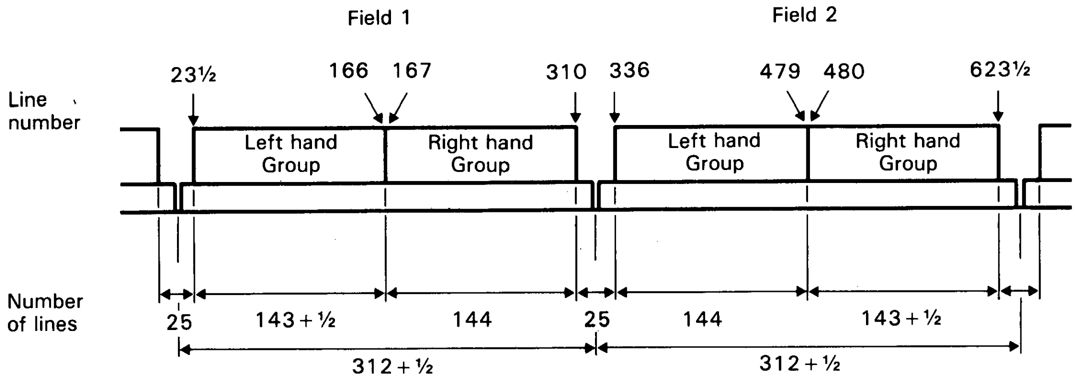
In order to keep the maximum horizontal errors as small as possible, the cameras used had better be in a cross-fire system, as for example in Figure A-1/H.100, and the camera/display assembly should be sited on the central axis of the terminal. However, in some cases, adoption of parallel-fire system as shown also in Figure A-1/H.100 is necessary due to a restriction in equipment arrangement.

Whether the two cameras are arranged in cross-fire or parallel-fire is left open to each Administration since the selection does not affect the interconnection of different systems.

4.5 *Picture processing methods at transmitting terminals*

In order to obtain the correct relationship between the signals from the two cameras for split-screen working, the cameras should be synchronized but the vertical drive pulses should be rephased. The drive to one should be advanced by one quarter of the vertical period while the drive to the other should be retarded by the same amount. This causes a central strip of the target of each camera tube to be used and so minimizes the effects of distortions in the corners of the targets. Figure B-1*a)*/H.100 illustrates the preferred method.

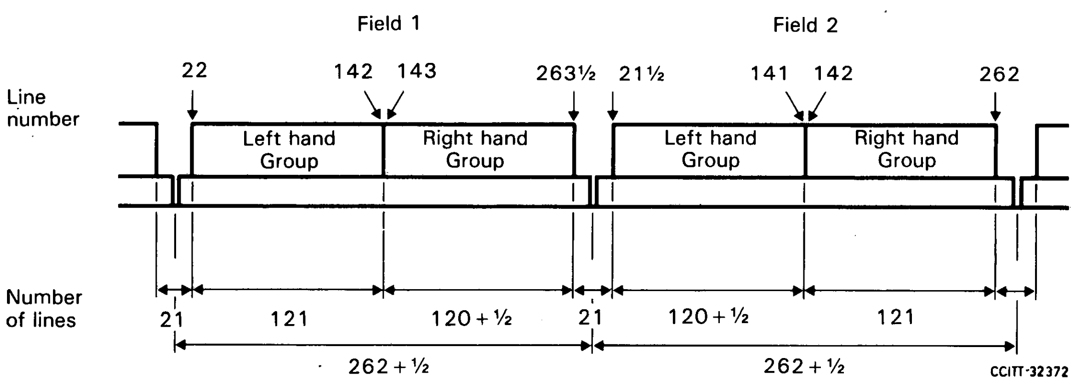
Alternative methods which are not recommended although they do not give rise to problems of end-to-end compatibility are compared in Annex B.



Left hand group:	first complete lines 24 and 336 last complete lines 166 and 479
Right hand group:	first complete lines 167 and 480 last complete lines 310 and 622

Lines 16-20 inclusive and 329-333 inclusive may contain identification, control or test signals.

a) 625-line television system



Left hand group:	first complete lines 22 (Field 1, 2) last complete lines 142 (Field 1), 141 (Field 2)
Right hand group:	first complete lines 143 (Field 1), 142 (Field 2) last complete lines 262 (Field 1, 2)

Lines 10-21 inclusive in Field 1 and 9½-21½ inclusive in Field 2 may contain identification, control or test signals.

b) 525-line television system

Note 1 – The method of defining the line number is following Fig. 2-1 of CCIR Report 624-2 for 625 system, and Fig. 2-3 for 525 system.

Note 2 – The notation which is used for line numbers is as follows. Line 23½ means that the picture starts (or finishes) half-way along line number 23. When totalling lines, a half-line is shown separately, e.g. 143 + ½.

FIGURE 1/H.100

Vertical format of split-screen video signal

ANNEX A

(to Recommendation H.100)

Seating arrangements when applying split-screen techniques for class *a* system

Preferred arrangements for video conferences using split-screen techniques are:

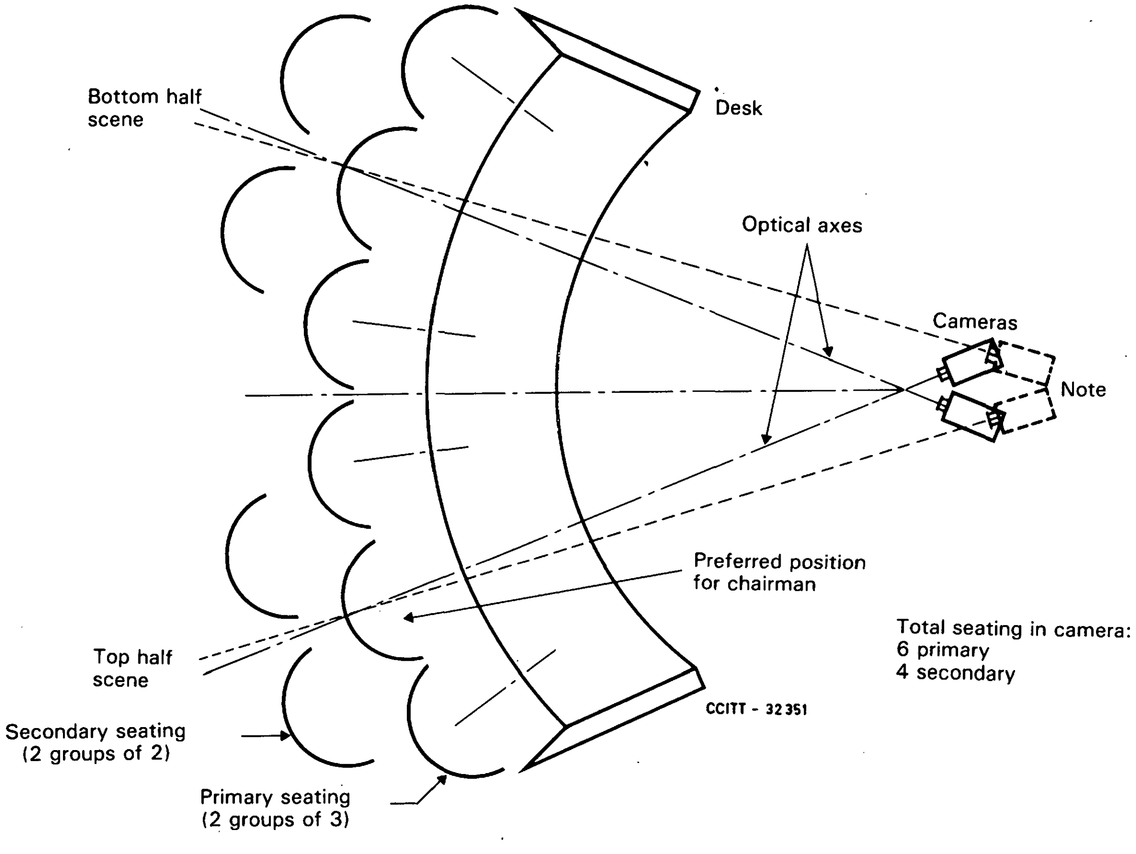
A.1 The conference terminal accommodation should be for 6 primary seats in two adjacent groups of 3 as shown in Figure A.1/H.100.

Provision for additional seating behind may be made, so long as allowance is made for the central gap between the two halves. For example, 4 additional persons may be seated on a second row as in the Figure.

A.2 The chairman's position should be in the centre of the left-hand group of seats (viewed from the camera) with user controls accessible from both this position and the one of the chairman's left.

Consequently, when split-screen pictures are displayed, stacked as received (i.e. shown as 3 over 3), the chairman's position is standardized as top centre.

The suite of 3 chairs containing the chairman's position should also be regarded as the primary position for occasions when only half of a studio is in use. Such standardization is necessary for connection of 3 studios in conference using time-division multiplex of pairs of TV signals to share a common trunk between two studios.



Note – Solid line cameras are for cross-fire. Dashed line cameras are for parallel-fire.

FIGURE A-1/H.100

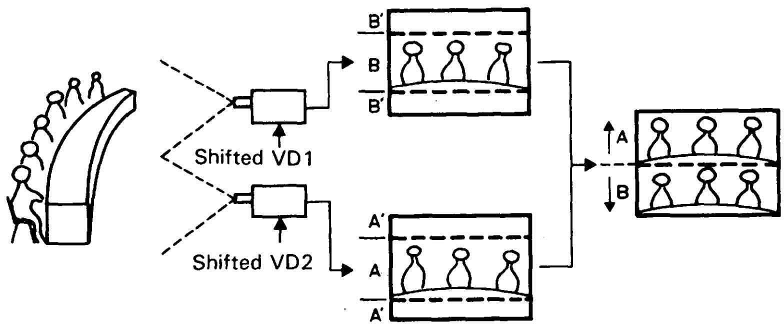
Studio plan view

ANNEX B

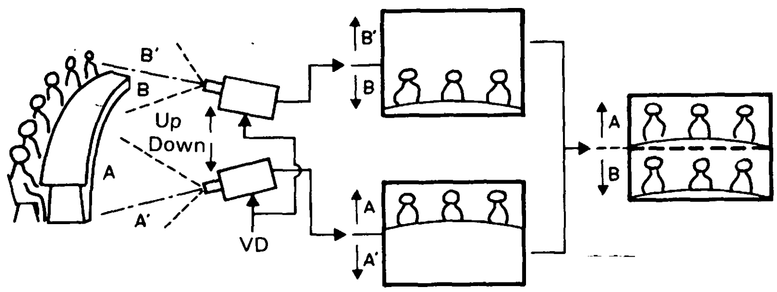
(to Recommendation H.100)

Picture processing methods in transmitting terminals

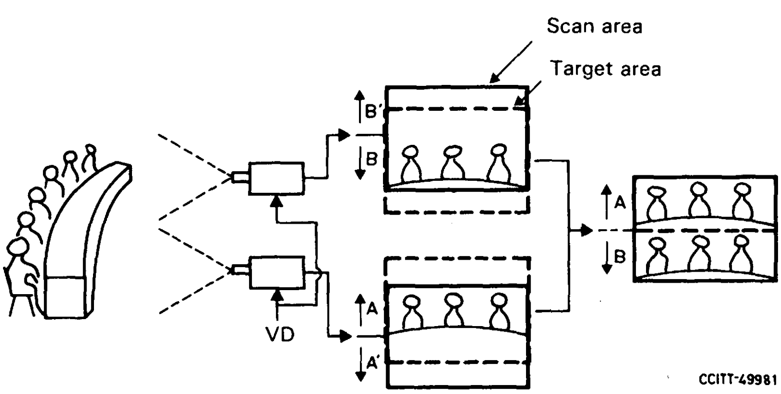
Alternative methods of obtaining the split-screen signal which are compatible with the recommended method and which might be useful for experiments and demonstrations are shown in *b)* and *c)* of Figure B-1/H.100. In method *b)*, the two cameras are directed upward and downward to pick up right and left halves of the conferencing room, respectively. Since circumferences of target and scanning areas are used, geometric and brightness distortions tend to occur. In method *c)*, vertical deflection currents are biased by the quantity corresponding to ± 1/4 of target height. Vertical deflection bias adjustment is needed every time cameras are exchanged. In method *a)*, the vertical driving pulses are phase-shifted by ± 1/4 V. The recommended method, *a)*, avoids the problems of methods *b)* and *c)*.



a) Vertical driving pulses are phase shifted



b) Cameras are directed upward and downward



c) Vertical deflection currents are biased

VD = vertical deflection

FIGURE B-1/H.100

Picture processing method at transmitting terminals

HYPOTHETICAL REFERENCE CONNECTIONS FOR VIDEOCONFERENCING USING PRIMARY DIGITAL GROUP TRANSMISSION

The CCITT,

considering

(a) that there is growing evidence of a customer demand for a videoconference service;

(b) that circuits to meet this demand can, at present, be provided effectively by digital transmission using the primary digital group;

(c) that switched digital transmission networks known as the Integrated Digital Network (IDN) and Integrated Services Digital Network (ISDN) are under study in Study Group XVIII, but the methods of exploiting these networks for the transmission of primary digital groups will not become clear until the studies have progressed further;

(d) that the existence of different digital hierarchies and different television standards in different countries complicates the problems of defining hypothetical reference connections;

(e) that a hypothetical reference connection may be used as a guide to simplify the problems of connections between countries with different television standards and digital hierarchies,

appreciating

that rapid advances are being made in research and development of video coding and bit-rate reduction techniques which may lead to further Recommendations being proposed for hypothetical reference connections for videoconferencing at bit rates which are multiples or sub-multiples of 384 kbit/s during subsequent study periods, so that this may be considered as the first of an evolving series of Recommendations,

noting

(a) that a hypothetical reference connection is a model in which studies relating to overall performance may be made, thereby allowing comparisons with standards and objectives; on this basis, limits for various impairments can be allocated to the elements of the connection;

(b) that such a model may be used:

- by an Administration to examine the effects on transmission quality of possible changes of impairment allocations in national networks,
- by the CCITT for studying the allocation of impairments to component parts of international networks,
- to test national rules for prima facie compliance with any impairment criteria which may be recommended by the CCITT for national systems;

(c) that hypothetical reference connections are not to be regarded as recommending particular values of impairments allocated to constituent parts of the connection, and they are not intended to be used for the design of transmission systems,

and recognizing

that the planning of the necessary transmission networks for a videoconference service will be facilitated if recommended hypothetical reference connections are available, even if only in a preliminary form without details of all transmission and switching arrangements, *recommends*

(1) that the hypothetical reference connection and means for digital transmission illustrated in Figures 1/H.110 and 2/H.110 shall be used as the model for studies of the overall performance of international videoconference connections, both intra-regional[1] and inter-regional[1], which are provided using minimum numbers of encoding and decoding equipments;

(2) that hypothetical reference connections of a more complex type, as, for example, those illustrated in Figure 3/H.110, being representative of many connections that may be employed in practice, should be studied further.

Note 1 – The hypothetical reference connection shown in Figure 1/H.110 contains the basic transmission elements, but is incomplete because switching has been excluded and the local ends and parts of the national network at each end of the connection have been left unspecified.

Note 2 – Because the arrangements of transmission systems interconnecting regions using different digital hierarchies have not yet been standardized, and because videoconferencing is likely to be a minority service in such transmission systems, it seems prudent to consider videoconference connections both where the primary hierarchical level on the inter-regional link is 1.5 Mbit/s and where it is 2 Mbit/s. In Figure 2b/H.110, the change between 2048 kbit/s and 1544 kbit/s transmission is placed at the 2048 kbit/s end of the long international network. The long distance part of the connection is thus operated at the lower bit rate. Where the international network is provided on a system which uses the 2048 kbit/s hierarchy, Figure 2c/H.110 maintains the efficiencies offered by the arrangement shown in Figure 2b/H.110, by making available the six vacated time slots for other use. Figure 2d/H.110 offers the possibility of improved picture quality compared with Figures 2b/H.110 and 2c/H.110 by making full use of the available 2048 kbit/s for the videoconferencing signal. This arrangement would require a 2048 kbit/s codec compatible with 525-line Video Standards, or the use of an external standards converter. This is for further study.

Note 3 – The lengths which have been assigned to the parts of the connections have been arbitrarily chosen, but have some consistency with existing CCITT and CCIR Recommendations. They are intended to be representative of long international connections, but not the longest possible. The lengths will likely require revision when studies on the error rates of digital paths have progressed to the stage when the error rates of the paths used in the connections can be predicted.

Note 4 – The propagation delay is one of the main factors to be studied based upon the structures and lengths of the connections in Figures 1/H.110, 2/H.110 and 3/H.110. However, in the absence of subjective test results, the specification of requirements for videoconferencing connections must await further study. This study and particularly operational experience are required to determine the extent to which Recommendation G.114, which applies to telephone connections, relates to videoconferencing connections.

Note 5 – In Figures 1/H.110 and 3/H.110, the codecs may be located anywhere within the international or national networks including the international gateway or the customer's premises.

Note 6 – The extensions beyond the codec shown as A or D in Figures 1/H.110 and 3/H.110 may include wideband analogue or high-speed digital transmission systems on terrestrial bearers. It is not expected that these transmission systems will have any significant influence on the quality of the picture or sound, or, on the propagation delay, other than that due to their length.

Note 7 – For inter-regional operation, television standards conversion between 525-line and 625-line video signals may be required. This conversion may be performed by the codecs themselves, or provided by external equipment.

Note 8 – The arrangements shown in Figure 2/H.110 provide for the simplest means of transmission. More complex means are possible and are not precluded.

Note 9 – The hypothetical reference connection shown in Figure 3/H.110 is of a more complex type than the connection shown in Figure 1/H.110, in that it includes codecs in cascade, and, possibly an external Television Standards Converter. The picture quality attainable with these more complex connections may be degraded with respect to that attainable using the connection illustrated in Figure 1/H.110. This and other aspects of the more complex connection must be studied further.

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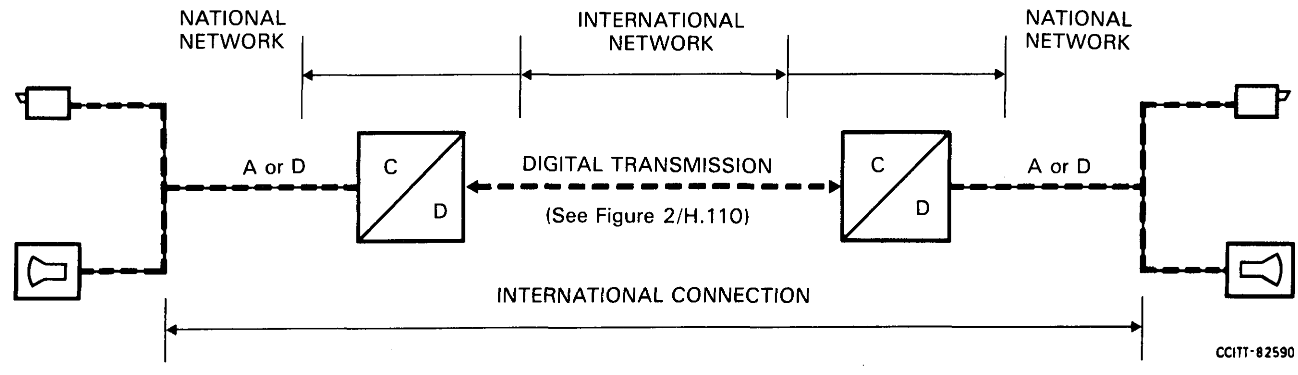
[1] The term "intra-regional" is used here to describe connections within a group of countries which share a common television scanning standard and a common digital hierarchy, and may or may not be in geographical proximity. The term "inter-regional" is used here to describe connections between groups of countries which have different television scanning standards and/or different digital hierarchies.



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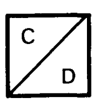
A or D – Wideband analogue or digital transmission or both all providing equivalent quality.

National option.

Digital transmission – Circuits for intra-regional or interregional digital transmission at the primary rate. This includes the international network and any national digital extensions thereof (see Figure 2/H.110).



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– Types of codec which may be used in the hypothetical reference connections are indicated below. Each can work with others of the same type and can interwork with other types shown, using a remultiplexer, if required. Codecs which perform these functions are described in Recommendation H.120.

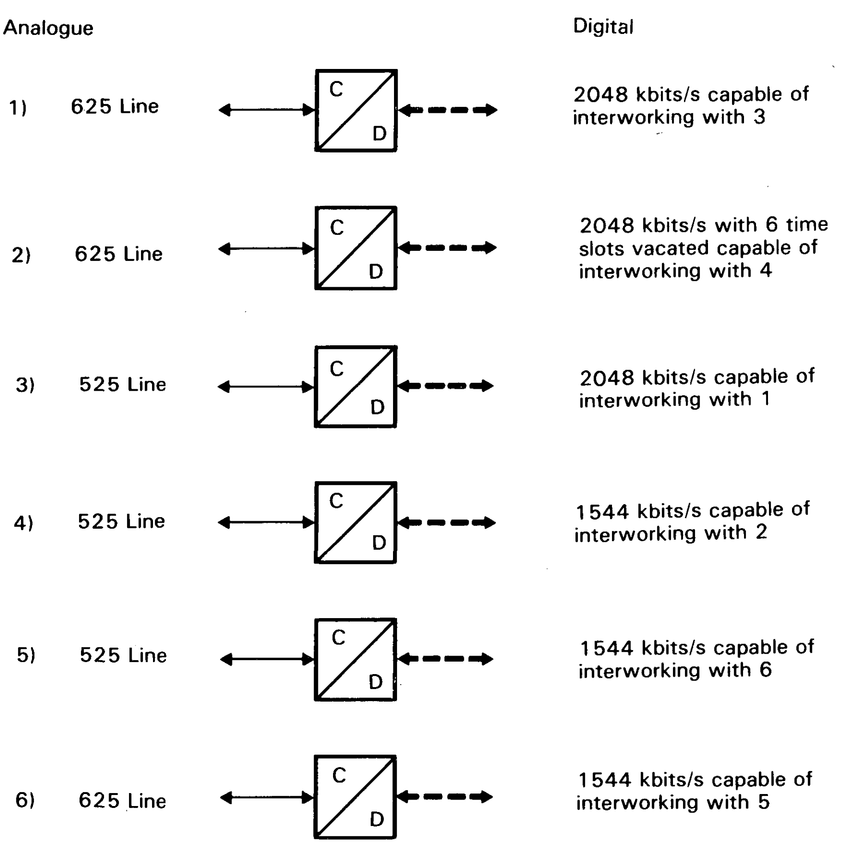

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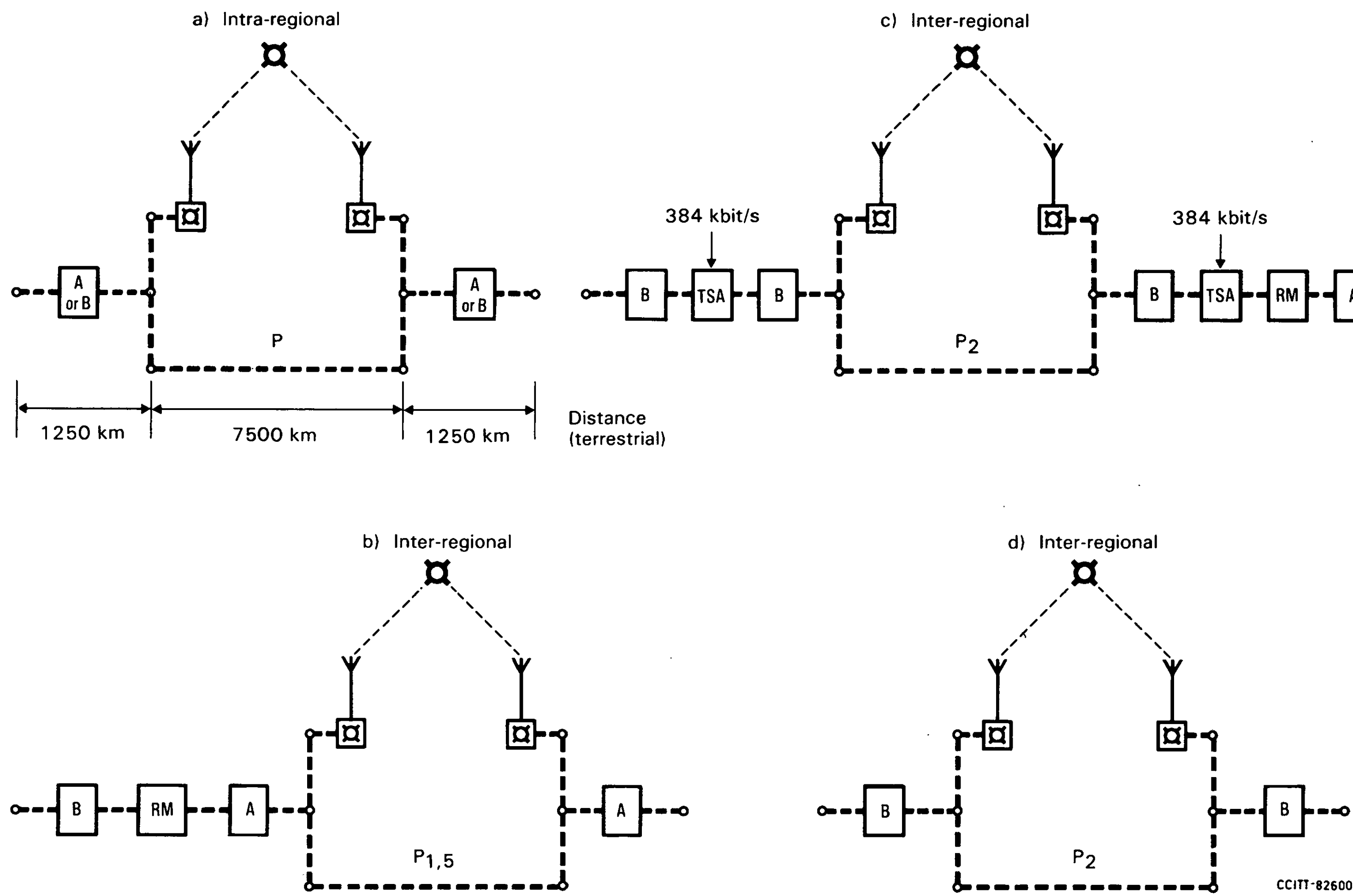
FIGURE 1/H.110

Hypothetical reference connection



Note – The distances shown in Figure 2a/H.110 are applicable to Figures 2b/H.110, 2c/H.110 and 2d/H.110. These distances refer to terrestrial transmission. Equivalent distances relating to satellite transmissions are for further study.

FIGURE 2/H.110

Means for digital transmission

Symbols for Figure 2/H.110

A — Termination of a 1544 kbit/s circuit with G.733 interface.

B — Termination of a 2048 kbit/s circuit with G.732 interface.

RM — Remultiplex Unit. This provides bit rate conversion between the 1544 kbit/s frame and 2048 kbit/s frame from which 6 timeslots have been vacated.

TSA — Optional Time Slot Access Unit. This provides means of inserting and extracting 384 kbit/s from the 2048 kbit/s frame which is not used for video conferencing.

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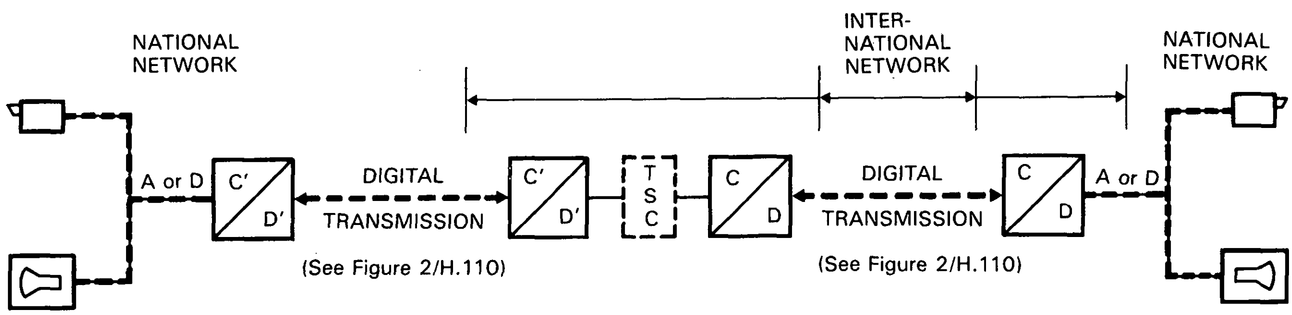
P — Primary level of digital hierarchy ($y + n \times 384$ kbit/s, where $n = 5$ or 4 and $y = 128$ or 8 kbit/s, respectively).

$P_{1.5}$ — 1544 kbit/s.

P_2 — 2048 kbit/s.



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Same symbols as Figure 1/H.110, and

C'/D' — Codecs in the HRC of Figure 3/H.110 which may be any compatible (with one another) combination of those defined as C/D in Figure 1/H.110 but not capable of interworking with the specific C/D codecs in Figure 3/H.110.

TSC — External Television Standards Converter. May or may not be required in the connection.

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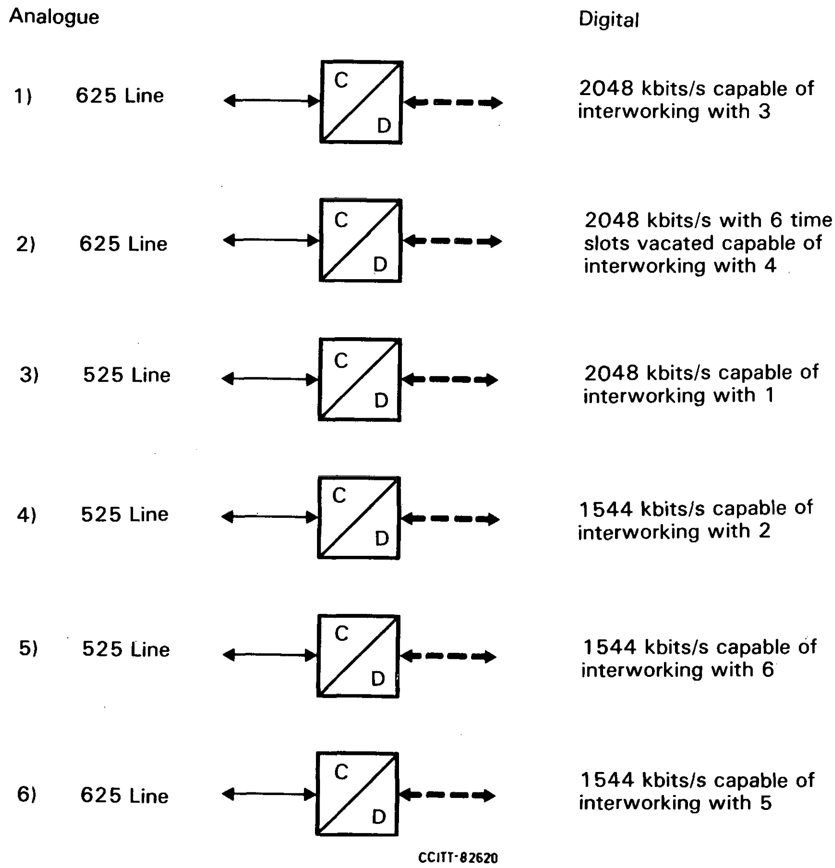
FIGURE 3/H.110

Complex hypothetical reference connection

Symbols for Figure 3/H.110:



Recommendation H.120

CODECS FOR VIDEOCONFERENCING USING PRIMARY DIGITAL GROUP TRANSMISSION

The CCITT,

considering

(a) that there is growing evidence of a customer demand for a videoconference serice;

(b) that circuits to meet this demand can, at present, be provided effectively by digital transmission using the primary digital group;

(c) that the existence of different digital hierarchies and different television standards in different parts of the world complicates the problems of specifying coding and transmission standards for international connections;

(d) that the eventual use of switched digital transmission networks should be taken into account,

appreciating

that rapid advances are being made in research and development of video coding and bit-rate reduction techniques which may lead to further Recommendations being proposed for videoconferencing at the bit rates which are multiples or submultiples of 384 kbit/s during subsequent study periods so that this may be considered as the first of an evolving series of Recommendations, *and noting*

that it is a basic objective of the CCITT to recommend unique solution for international connections as far as possible,

recommends

that the codecs having signal processing and interface characteristics described in Parts 1, 2 and 3 below, should be used for international videoconference connections.

Note – Codecs of types other than those described in this Recommendation are not precluded.

Introduction

Part 1 of this Recommendation specifies the codec, developed for operation with the 625-line, 50 field/s television standard and the 2048 kbit/s primary digital group. Its architecture has been chosen to permit variations in the detailed design of certain of the functional elements having the greatest influence on the picture quality. This enables future developments, aimed at improving the performance, to be incorporated without affecting the ability of different coders and decoders to interwork. For this reason, no details are given of such items as motion detectors or spatial and temporal filters. The Recommendation confines itself to the details necessary to enable a decoder correctly to interpret and decode the received signals.

The Annexes to Part 1 give details of some additional optional features which may be provided to supplement the basic design.

Part 2a describes a version of the codec for 525 line, 60 field/s and 1544 kbit/s operation which also provides automatic television standards conversion when connected to the version of the codec described in Part 1 via a re-multiplexing unit (to convert between Recommendations G.732 and G.733 frame structures) at the junction of the 2048 and 1544 kbit/s digital paths.

Other implementations of Part 2 are to be studied, for example:

- a version of the codec for 625-line, 50 field/s and 2048 kbit/s operation capable of interworking with the codec described in Part 3;
- a version of the codec for 525-line, 60 field/s and 2048 kbit/s operation capable of interworking with the codec described in Part 1.

Part 3 of the Recommendation is intended to describe a codec for intra-regional use in 525-line, 60 field/s and 1544 kbit/s regions. The detailed implementation is for further study.

The frame structures associated with the codecs described in this Recommendation are to be found in Recommendation H.130.

As the codecs are complex items using combined intraframe and interframe picture-coding techniques which tend to be known only to specialists, Appendix I is provided giving a brief outline of the principles involved in the codecs of Part 1 and Part 2a.

PART 1

(to Recommendation H.120)

A codec for 625-lines, 50 fields/s and 2048 kbit/s transmission for intra-regional[1] use and capable of interworking with the codec of Part 2a

1 Scope

This Recommendation defines 2048 kbit/s the essential features of a codec for the digital transmission, at 2048 kbit/s, of signals for videoconference or visual telephone service in accordance with Recommendation H.100. The video input to the coder and output from the decoder in a 625-line, 50 field/s signal, according to the "Class *a*" standard of Recommendation H.100, or alternatively, the 313-line, 50 field/s signal of the "Class *b*" standard. Provision is also made for a sound channel and optional data channels. A brief description of the operation of the codec is given in the Appendix to this Recommendation.

[1] The term "intra-regional" is used here to describe connections within a group of countries which share a common television scanning standard and a common digital hierarchy, and may or may not be in geographical proximity. The term "inter-regional" is used here to describe connections between groups of countries which have different television scanning standards and/or different digital hierarchies.

The Recommendation starts with a brief specification of the codec (§ 2) and a description of the video interface. This is followed by details of the source coder (§ 4) which provides analogue-to-digital conversion followed by recoding with substantial redundancy reduction in the face-to-face mode. The next § 5 deals with the video multiplex coder which inserts instructions and addresses into the digitized video signal to control the decoder so that it correctly interprets the signals received. § 6 is the transmission coder which arranges the various digital signals (video, sound, data, signalling) into a form compatible with Recommendation G.732 for transmission over 2048 kbit/s digital paths. § 7 describes optional forward error correction facilities. Provision is made in the digital frame structure for the inclusion of other optional facilities such as a graphics mode, encryption and multipoint conferencing. Details of such of these facilities as are at present available are given in the Annexes to this Recommendation.

2 Brief specification

2.1 *Video input/output*

The video input and output are standard 625-line, 50 field/s colour or monochrome television signals. The colour signals are in, or converted to, component form. Colour and monochrome operation are fully compatible.

2.2 *Digital output/input*

The digital output and input are at 2048 kbit/s, compatible with the frame structure of Recommendation G.732.

2.3 *Sampling frequency*

The video sampling frequency and the 2048 kHz network clock are asynchronous.

2.4 *Coding techniques*

Conditional replenishment coding supplemented by adaptive digital filtering, differential PCM and variable-length coding are used to achieve low bit-rate transmission.

2.5 *Audio channel*

An audio channel using 64 kbit/s is included. At present, coding is A-law according to Recommendation G.711, but provision is made for future use of more efficient coding.

2.6 *Mode of operation*

The normal mode of operation is full duplex.

2.7 *Codec-to-network signalling*

An optional channel for codec-to-network signalling is included. This conforms to emerging ideas in CCITT for switching 2-Mbit/s paths in the ISDN.

2.8 *Data channels*

Optional 2 × 64 kbit/s and 1 × 32 kbit/s data channels are available. These are used for video if not required for data.

2.9 *Forward error correction*

Optional forward error correction is available. This is required only if the long-term error rate of the channel is worse than 1 in 10^6.

2.10 *Additional facilities*

Provision is made in the digital frame structure for the future introduction of encryption, a graphic mode and multipoint facilities.

2.11 *Propagation delay*

When the coder buffer is empty and the decoder buffer full, the coder delay is less than 5 ms and the decoder delay is 130 ± 30 ms at 2 Mbit/s or 160 ± 36 ms when only 1.5 Mbit/s are in use[2].

3 Video interface

The normal video input is a 625-line, 50 field/s signal in accordance with CCIR Recommendation 472-1. When colour is being transmitted, the input (and output) video signals presented to the analogue/digital convertors (and from the digital/analogue convertors) are in colour-difference component form. The luminance and colour-difference components, E'_Y, $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ are as defined in CCIR Report 624-2. The analogue video input (and output) interface with the codec may be in the form of colour-difference components, colour components (R, G, B) or as a composite colour signal. The video interface is as recommended in CCIR Recommendation 567.

Optionally, any other video standard which can be converted to give 143 active lines per field may be used.

4 Source coder

4.1 *Luminance component or monochrome*

4.1.1 *Analogue-to-digital conversion*

The signal is sampled to produce 256 picture samples per active line (320 samples per complete line). The sampling pattern is orthogonal and line, field and picture repetitive. For the 625-line input, the sampling frequency is 5.0 MHz, locked to the video waveform.

Uniformly quantized PCM with 8 bits/sample is used.

Black level corresponds to level 16 (00010000).

White level corresponds to level 239 (11101111).

PCM code words outside this range are forbidden (the codes being used for other purposes). For the purposes of prediction and interpolation, the final picture element in each active line (i.e. picture element 255) is set to level 128 in both encoder and decoder.

In all arithmetic operations, 8-bit arithmetic is used and the bits below the binary point are truncated at each stage of division.

4.1.2 *Pre- and post-filtering*

In addition to conventional anti-aliasing filtering prior to analogue-to-digital conversion, a digital transversal filtering operation is carried out on the 625-line signal to reduce the vertical definition of the picture prior to conditional replenishment coding. As a result of this process, 143 active lines per field are used instead of the 287½ active lines of the 625-line signal, although the effective vertical definition is greater than one-half of that of a normal 625-line display. An interpolation process in the decoder restores the 625-line signal waveform.

4.1.3 *Conditional replenishment coding*

A movement detector identifies clusters of picture elements which are deemed to be moving. The basic feature is a frame memory which stores 2 fields of 143 lines, each line containing 256 addressable points. The memory is updated at the picture rate and differences between the incoming signal and the corresponding stored values are used to determine the moving area in the coder. A similar frame memory must exist at the decoder and be similarly updated under the control of addressing information received from the coder. It is not necessary to specify the techniques used for movement detection because they do not affect interworking, although they do affect the resultant picture quality.

Detected moving areas are transmitted by differential PCM with a maximum of 16 quantization levels. The first picture element in each moving area is transmitted by PCM. Variable-length coding is used on the DPCM code words.

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2) These are typical figures. The delays depend on the detailed implementation used.

The first picture element of each cluster and the complete PCM lines, when they are transmitted to provide systematic or forced updating, are coded in accordance with § 4.1.1.

4.1.3.1 *DPCM prediction algorithm*

The algorithm used for DPCM prediction is:

$$X = \frac{\Lambda + D}{2}, \text{ where X is the sample being predicted. (See Figure 1/H.120.)}$$

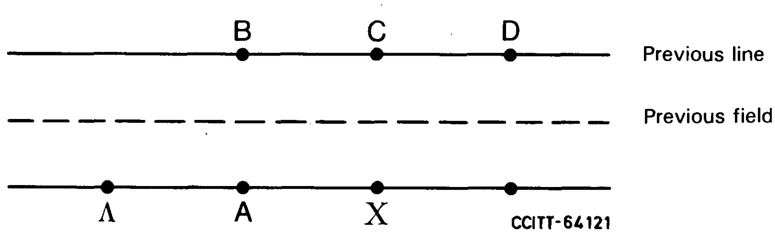


FIGURE 1/H.120
Identification of samples

For the purpose of prediction, line and field blanking are assumed to be at level 128 (out of 256).

4.1.3.2 *Quantization law and variable-length coding*

511 input levels are quantized to a maximum of 16 output levels. The quantizer does not assume the use of modulo 256 arithmetic.

The quantization law and associated variable-length codes which are used for both luminance and colour-difference picture elements in moving areas which are not horizontally subsampled are given in the following code table.

Code Table for Non-Horizontally-Subsampled Moving Areas

Input Levels	Output Levels	Variable-length Code	Code No.
−255 to −125	−141	1 0 0 0 0 0 0 0 0 1	17
−124 to − 95	−108	1 0 0 0 0 0 0 0 1	16
− 94 to − 70	− 81	1 0 0 0 0 0 0 1	15
− 69 to − 49	− 58	1 0 0 0 0 0 1	14
− 48 to − 32	− 39	1 0 0 0 0 1	13
− 31 to − 19	− 24	1 0 0 0 1	12
− 18 to − 9	− 13	1 0 1	10
− 8 to − 1	− 4	1 1	9
0 to 7	+ 3	0 1	1
8 to 17	+ 12	0 0 1	2
18 to 30	+ 23	0 0 0 1	3
31 to 47	+ 38	0 0 0 0 1	4
48 to 68	+ 57	0 0 0 0 0 1	5
69 to 93	+ 80	0 0 0 0 0 0 1	6
94 to 123	+107	0 0 0 0 0 0 0 1	7
124 to 255	+140	0 0 0 0 0 0 0 0 1	8

The End-of-Cluster code is 1 0 0 1 and is designated as Code Number 11. The End-of-Cluster code is omitted at the end of the last cluster in a line irrespective of whether it is a luminance cluster or a colour-difference cluster.

4.1.4 *Subsampling*

As the buffer fills, horizontal subsampling and field/field subsampling are introduced.

4.1.4.1 *Horizontal subsampling*

Horizontal subsampling is carried out only in moving areas. Normally, in this mode, only even elements are transmitted on even numbered lines and odd elements on odd numbered lines. This gives rise to a line quincunx pattern in moving areas.

Omitted elements are interpolated in the decoder by averaging the two horizontally adjacent elements.

Interpolated picture elements are placed in the frame stores. A moving area cluster will always start with a PCM value and finish with a transmitted DPCM picture element, even during subsampling. This means that in some instances, the transmitted cluster needs to be extended by one element in comparison with the moving area declared by the movement detector. At the end of the active line, however, this cannot occur as clusters must not extend into blanking, so cluster shortening by one element can be necessary.

Adaptive element subsampling allows the transmission of normally omitted elements, either to remove interpolation errors or, to provide a softer switch to subsampling and thus improve the picture quality. The signalling of the extra elements is achieved by using, on horizontally subsampled lines only, 8 quantizing levels for normally transmitted elements and the remaining 8 levels for the extra elements. Also, a cluster can finish either on a normally transmitted element or an "extra" element.

During horizontally subsampled lines, the following quantization law and variable-length code table will be used for both luminance and colour-difference samples in moving areas.

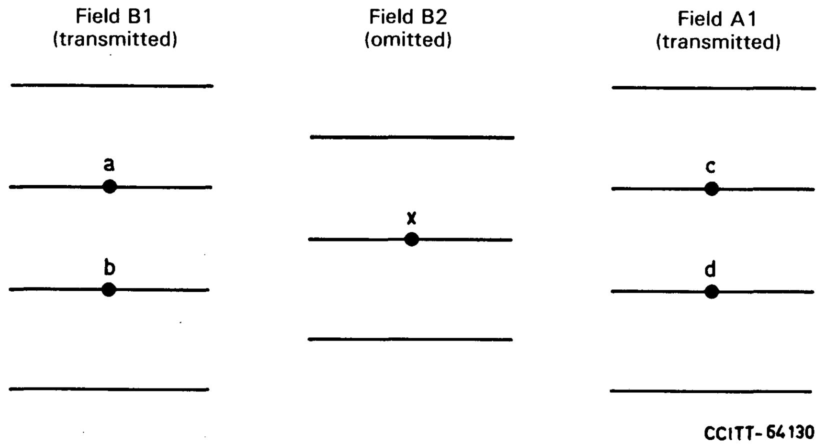
Quantisation		Variable-Length Codes			
Input Range	Output Levels	Normal Elements	Code No.	Extra Elements	Code No.
−255 to −41	−50	1 0 0 0 0 0 0 1	15	1 0 0 0 0 0 0 0 0 1	17
− 40 to −24	−31	1 0 0 0 0 1	13	1 0 0 0 0 0 0 0 1	16
− 23 to −11	−16	1 0 1	10	1 0 0 0 0 0 1	14
− 10 to − 1	− 5	1 1	9	1 0 0 0 1	12
0 to + 9	+ 4	0 1	1	0 0 0 1	3
10 to 22	+15	0 0 1	2	0 0 0 0 0 1	5
23 to 39	+30	0 0 0 0 1	4	0 0 0 0 0 0 0 1	7
40 to 255	+49	0 0 0 0 0 0 1	6	0 0 0 0 0 0 0 0 1	8

With regard to prediction, if element A is a non-transmitted element in a moving area, it is replaced by A_s (see Figure 1/H.120); if element D is part of a subsampled moving area, and not transmitted in the current frame, it is replaced by C.

4.1.4.2 *Field/field subsampling*

Either field can be omitted. In the omitted field, interpolation takes place only in those parts of the picture which are estimated to be moving. "Stationary" areas remain unchanged.

The estimated moving areas are formed from an OR function on the moving areas in the past and future fields, as shown:



x is a moving element if a OR b OR c OR d are moving.

For the purpose of field interpolation, PCM lines are considered as non-moving and field blanking is assumed to be at a level of 128 out of 256.

In the interpolator for monochrome or luminance signals, the operations $\frac{a+b}{2}$ and $\frac{c+d}{2}$ are carried out before the combined average is taken. Thus

$$x = \frac{\left[\frac{a+b}{2}\right] + \left[\frac{c+d}{2}\right]}{2}$$

The interpolated values are placed in the frame store.

4.2 *Colour-difference components*

4.2.1 *Analogue-to-digital conversion*

The signal is sampled to produce 52 samples per active line (64 samples per complete line). The sampling pattern is orthogonal and line-, field- and picture-repetitive. For the 625-line input, the sampling frequency is 1.0 MHz, locked to the video waveform.

The $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ samples are sited so that the centre of the first colour-difference sample on any line is co-sited with the centre of the third luminance sample (addressed as number 2). The $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ signals are stored and transmitted on alternate lines of the coded picture. The first active line of Field No. 1 contains $(E'_B - E'_Y)$ and the first active line of Field No. 2 contains $(E'_R - E'_Y)$. The colour difference signal not being transmitted during any line is obtained at the decoder by interpolation.

The vertical filtering (see § 4.2.2) is arranged so that the effective vertical positions of the colour-difference samples in each of the 286 active lines coincide with those of the corresponding luminance samples.

Uniformly quantized PCM with 8 bits/sample is used.

The $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ signals are quantized using ± 111 steps with zero signal corresponding to level 128. The analogue video signals are amplitude-limited so that the digitized signals do not go outside that range (corresponding to levels 16 to 239). The video levels are set so that a 100/0/75/0 colour bar signal (see CCIR Recommendation 471 for explanation of nomenclature) will occupy levels 17 to 239.

As for the luminance signal, forbidden PCM code words are available for purposes other than transmitting video sample amplitudes.

4.2.2 *Pre- and post-filtering*

In addition to conventional anti-aliasing filtering prior to analogue-digital conversion, a digital transversal filtering operation is carried out on the 625-line signal to reduce the vertical definition of the picture prior to conditional-replenishment coding. As a result of this process, 72 active lines of ($E'_R - E'_Y$) and 71 active lines of ($E'_B - E'_Y$) are used in Field No. 2 instead of 287½ active lines per field of a 625-line signal. Similarly, Field No. 1 contains 72 active lines of ($E'_B - E'_Y$) and 71 lines of ($E'_R - E'_Y$). An interpolation process in the decoder restores the 625-line signal waveforms.

4.2.3 *Conditional replenishment coding*

Coloured moving areas are detected, coded and addressed separately from the luminance moving areas, but the same principles are employed.

Detected moving areas are transmitted by differential PCM with a maximum of 16 quantization levels. The first picture element in each moving area is transmitted by PCM. Variable-length coding is used on the DPCM code words.

Complete PCM lines are transmitted to provide systematic and forced updating coincident with luminance PCM lines.

4.2.3.1 *DPCM prediction algorithm*

The algorithm used for colour-difference signals is:

$$x = A \text{ (see Figure 1/H.120)}$$

4.2.3.2 *Quantization law and variable-length coding*

As for luminance component (see §§ 4.1.3.2 and 4.1.4.1).

4.2.4 *Subsampling*

Horizontal subsampling is carried out in exactly the same way as for the luminance signal, including adaptive element subsampling.

Field/field subsampling of the colour-difference signals is also similar to that of the luminance signal. Either field can be omitted and, in the omitted field, interpolation takes place only in those parts of the picture which are estimated to be moving. Stationary areas remain unchanged.

The estimated moving areas are formed by an OR function on moving areas in past and future fields in the same manner as for luminance (§ 4.1.4.2).

For colour-difference signals, the interpolated value of x is $\left(\frac{a + c}{2}\right)$ or $\left(\frac{b + d}{2}\right)$ when x is in Field 1 or Field 2, respectively.

Both field and horizontal subsampling take place simultaneously with subsampling of the luminance signal and they are signalled to the decoder in the same way.

5 Video multiplex coding

5.1 *Buffer store*

The size of the buffer store is defined at the transmitting end only and is 96 kbit/s. Its delay is approximately equal to the duration of one picture (40 ms).

At the receiving end, the buffer must be of at least this length, but in some implementations of the decoder it may be longer.

5.2 *Video synchronization*

The method used for video synchronization permits the retention of the picture structure. The required information is transmitted in the form of Line Start and Field Start codes (LST and FST).

5.2.1 *Line start code*

The line start code includes a synchronization word, a line number code and a digit to signal the presence of element subsampling.

It has the format:

0 0 0 0 0 0 0 0 | 0 0 0 0 1 0 0 0 | "S" | 3-bit Line No. code |

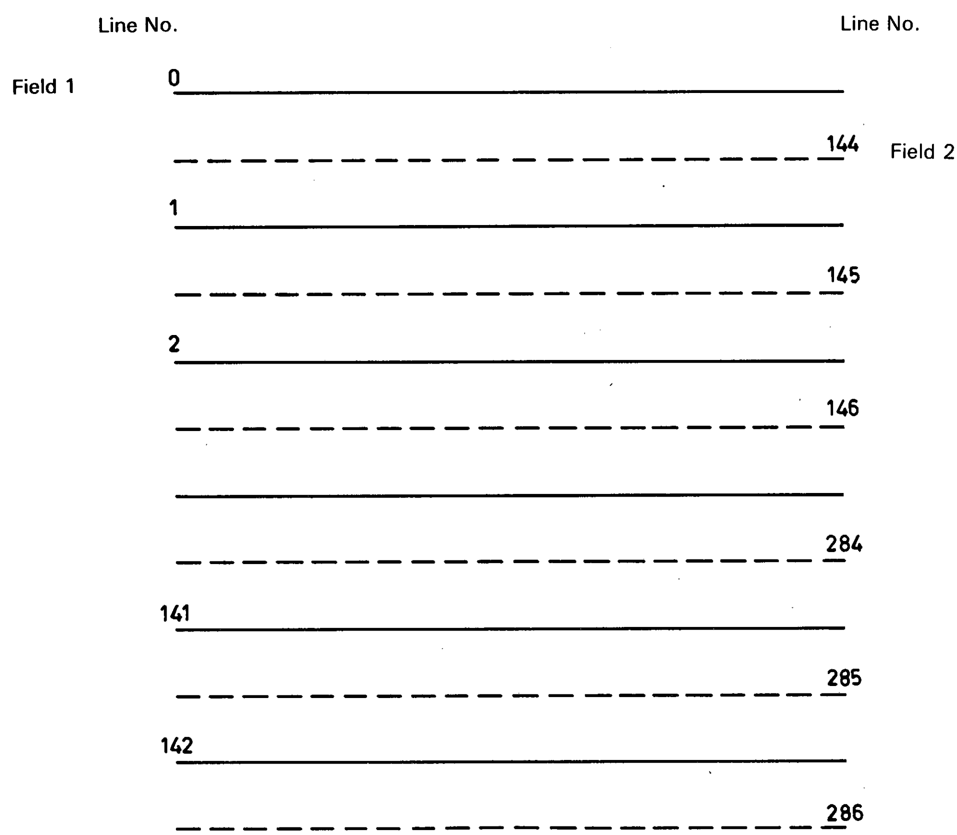
"S" is a 1 if horizontal subsampling occurs on the TV line following the line start code. "S" is a "don't care" condition on empty or PCM lines.

The line number code comprises the least three significant digits of the line number, where Line 0 = first active line of Field 1 and Line 144 = first active line of Field 2.

Lines numbered 143 and 287 are non-coded lines, used for field synchronization and line number continuity.

5.2.2 *Field start code*

There are two field start codes, FST-1 and FST-2, where the first line of the field following FST-2 is interlaced between the first two lines of the field following FST-1. FST-1 indicates the start of the first field, starting with line number 0. FST-2 indicates the start of the second field, starting with line number 144, as shown in the diagram below:



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Each field start code comprises a line start code, followed by an 8-bit word, followed by the line start code of the first line of the next field.

The field start code is:

LST					LST			
00000000	00001AAA	F	111	0000F11F	00000000	00001000	S	000

For FST-1, F = 1

For FST-2, F = 0

A = 0 for normal operation.

If required, A = 1 is used to signal that the buffer state is less than 6 kilobits (used in switched multipoint applications).

S is the subsampling digit as defined in § 5.2.1.

Field subsampling is signalled by two consecutive field start codes of the same number. For example:

FST – 1	field of data	FST – 1	field of data

signifies that field 2 has been omitted and that its moving areas must be interpolated as described in §§ 4.1.4.2 and 4.2.4.

5.3 *Addressing of moving areas*

The positions of the clusters of picture elements along each line which are deemed to form parts of moving areas are addressed by means of an address of the start of the cluster and an "End of Cluster" code (EOC).

The form of coding is:

LST	pcm value	8-bit address of the pcm picture element	variable-length dpcm coded moving area	EOC	pcm value	8-bit address	etc.

The PCM value is the amplitude of the first picture element of the cluster. When there is no colour-difference data, the EOC is omitted in the last luminance cluster of every line, i.e. both the LST and FST codes also signify end of cluster.

The EOC is 1001.

The address indicates the sample number along the line belonging to the first picture element of the cluster.

A cluster cannot start at the last element of the line, i.e. (11111111) is a forbidden cluster address, nor can it extend into line blanking even during subsampling.

The minimum gap between the end of one cluster and the start of the next is four picture elements, and the minimum length of a cluster is one picture element.

5.4 *Addressing of colour-difference data*

To permit the insertion of colour-difference data in a line containing moving picture elements, a colour escape code is inserted after the final luminance cluster in the line. This permits addresses to be re-used for colour clusters.

The escape code is 00001001 (an invalid PCM value) and follows the end of cluster code of the last luminance cluster (if any), otherwise it follows the line start code. This is followed by the addresses, variable-length codes and EOC codes of the subsequent coloured clusters, the sequence being terminated by the line start code of the next line.

The form of addressing of colour-difference moving areas is shown below:

...	VLC luminance picture element	EOC	00001001	pcm value of first colour-difference picture element	Address of first colour-difference cluster	VLC codes	...

...	EOC	pcm	Address	....	line start code	....

There are 52 colour-difference picture elements per line, the first of which is given an address with numerical value 4. The address range is therefore:

00000100 to 00110111.

A cluster cannot start at address (00110111) nor can it extend beyond this point, even during subsampling. The minimum gap between the end of one colour-difference cluster and the start of the next is 4 picture elements. The minimum cluster length is one picture element. Cluster bridging is not allowed between luminance and colour-difference clusters.

A monochrome decoder will discard the information between the colour escape code and the next line start code.

5.5 *PCM lines*

PCM lines are used for systematic or forced updating and are signalled as follows:

	Invalid pcm code	Invalid Cluster Address	pcm value of first picture element of line	254 × 8 bit pcm values	
LST	1 1 1 1 1 1 1 1	1 1 1 1 1 1 1 1	X X X X X X X X	X X ...	1 0 0 0 0 0 0 0

With monochrome, all 256 elements in the line are transmitted using S-bit PCM.

With PCM lines, the subsampling digit "S" is ignored at the receiver. PCM lines cannot be horizontally subsampled.

For the purpose of field interpolation, PCM lines are considered to be non-moving.

With colour signals, the colour-difference data will comprise 52 × 8-bit PCM values following the 256 × d-bit luminance elements. The colour escape code is not transmitted. A monochrome decoder will discard the colour-difference picture elements.

6 Transmission coding

The transmission coder assembles the video, audio, signalling and optional data channels into a 2048-kbit/s frame structure which is in accordance with Recommendation G.704. It also provides justification facilities to enable the video sampling frequency to be independent of the network clock.

6.1 *Serial data*

With all serialized data (video, audio and addressing), the most significant digit leads. Positive logic is used throughout.

6.2 *Audio*

The audio is coded into 64 bit/s using A-law PCM, as specified in Recommendation G.711.

In the coder, the delay difference between the coded audio and video when the buffer is empty should be within ± 5 ms. In the decoder, the delays must also be equalized, and the tolerance is under study.

The audio output should be muted in the event of loss of frame alignment.

6.3 *Transmission framing*

6.3.1 *General*

The frame structure is defined in Recommendation H.130 which specifies how the frame is structured and for what purposes the Time Slots are used. This information need not be repeated here.

Time Slot 2 (odd) is allocated to codec-to-codec signalling and the functions of the various bits are specified in the Recommendation on the frame structure. In most cases, the action to be taken by the encoder and/or decoder according to whether each of these bits is 0 or 1 is evident from the specified purpose of the bit. In the few cases where this is not so, additional information is given here.

6.3.2 *Use of certain bits in each octet in the odd frames of Time Slot 2*

Bits 1 and 2 are essential to the basic requirements of keeping the encoder and decoder operating in step.

Bits 3.7 to 3.13 provide facilities whose principal usefulness is likely to be multipoint conferencing. The methods for multipoint conferencing are still under study, but the frame structure and the codec have the capability of providing a number of special facilities which have been identified as necessary for multipoint.

Bit 1 – For clock justification

The frequency control arrangements are as follows:

The video sampling clock is locked to the line-scanning frequency of the incoming video signal, which has a permitted tolerance of $\pm$ 2 parts in 10^4.

The justification is controlled by a comparison frequency of (22 500/11) kHz, which is locked to the video clock.

The clock for the digital channel has a frequency of 2048 kHz $\pm$ 50 parts in 10^6.

The phase of the channel clock is compared with that of the comparison frequency and when the channel-clock phase exceeds that of the comparison frequency by 2 π radians, a 1 is transmitted. If the phase difference is less than 2 π radians, a 0 is transmitted.

Bit 2 – To signal buffer state

The degree to which the encoder buffer is filled, measured in increments of 1 K (1 K = 1024 bits), is signalled using an 8-bit binary code. The Most Significant Bit (MSB) is in frame 1 of the multiframe, the second MSB in frame 2, etc. The buffer state is sampled at the start of the multiframe in which its state is transmitted.

Bit 3.7 – Fast update request

On receipt of this bit set to 1, the transmitter buffer is forced to decrease its fill and stabilize to a state of less than 6 K by preventing coded picture elements from entering the buffer. Bit A is set to 1 in the next FST. The two following fields are treated as complete moving areas and the encoder uses an arrangement for control of the subsampling modes to make the buffer overflow condition unlikely.

Bit 3.9 – Advance warning of interruption

This bit (set to 1) is used to warn a decoder that its received signal may be interrupted after the start of the next supermultiframe for a period of not more than one second. On receipt of bit 3.9 set to 1; a decoder will display a still picture for a period of not more than one second, or until an FST code is received with bit A set to 1.

Bit 3.11 – Sound power signal

This bit is used to signal the sound power which is quantized to 8 bits and transmitted at supermultiframe rate. It is provided for use with encrypted multipoint. The precise form of the coding of these bits is under study.

Bit 3.13 – Data distribution

This bit is set permanently to 0 in all encoders. When a 1 is received from the network (introduced, for example, by a multipoint control unit), the encoder will vacate the same time slots in its outgoing signal as signalled on the incoming stream by the settings of the relevant bits 4 (which identify the use of the time slots). It will confirm the action by transmitting the same bit 4 settings as received. This function should be carried out within 10 supermultiframe periods.

7 Error correction

Provision is made for the optional use of forward error correction. This is required if the channel error rate is worse than 1 in 10^6 for significant periods of time. The error corrector used is a (4095, 4035) five-error correcting BCH[3)] code. The error-correcting decoder has the ability to correct up to 5 isolated errors and one burst of up to 16 errors in each block. At a channel error probability of 1×10^{-4}, the corrected error rate is 1.25×10^{-8}. The 60 parity bits which are required are obtained by removing the video from Time Slots 24 to 31 of frame number 15 of each multiframe.

Note – The question of whether error correction should be provided on the signal, on the link, or both, should be studied. Also under study is the question of whether the audio should be corrected by the same error corrector or whether a separate error-correcting codec should be used.

ANNEX A

(to Part 1 of Recommendation H.120)

Graphics option – 625 line

Under study.

ANNEX B

(to Part 1 of Recommendation H.120)

Encryption option – 625 line

Under study.

PART 2

(to Recommendation H.120)

Codec for inter-regional use not requiring separate television standards conversion

Part 2a – Version using 525-lines, 60 fields/s and 1544 kbit/s transmission capable of interworking with the codec of Part 1

1 Introduction

Part 2a indicates the changes and additions which must be made to the text of Part 1 in order to define the version of the codec for use with 525-line 60 fields/s television standards and transmission at 1544 kbit/s. The two versions are capable of interworking via a re-multiplexing unit which can convert the Recommendation G.733 compatible frame structure on one side to the Recommendation G.732 compatible frame structure (with 6 Time Slots empty) on the other.

The two versions of the codec are identical in most respects, the important differences (apart from the obvious ones arising from different input and output signals) being confined to the digital pre- and post-filters and the signals for the control of the buffers. Moreover, the detailed algorithms of the pre- and post-filters do not need to be specified to permit interworking. Only an outline of their mode of operation together with the few necessary specifications are therefore provided.

[3)] BCH = Bose, Chaudhuri and Hocquengham.

The Sections in Part 2 carry the same numbering as used in Part 1. Sections omitted in Part 2a can be assumed to be identical with the same-numbered Sections in Part 1, except in a few cases where references to 625-lines, 50 fields/s or 2048 kbit/s should be changed to 525-line, 60 field/s and 1544 kbit/s.

2 Brief specification

2.1 Video input/output – standard 525-line, 50 fields/s colour or monochrome television signals. The colour signals are in component form. Colour and monochrome operation are fully compatible.

2.2 Digital output/input – 1544 kbit/s, compatible with the frame structure of Recommendation G.733.

2.3 Video sampling frequency and 1544 kbit/s network clock are asynchronous.

2.7 An optional channel for codec-to-network signalling is included.

2.11 When the coder buffer is empty and the decoder buffer full, the coder delay is 31 ± 5 ms and the decoder delay is 176 ± 31 ms[4].

3 Video interface

The normal video input is a 525-line, 60 fields/s signal in accordance with CCIR Report 624-2. When colour is being transmitted, the input (and output) video signals are in component form. The luminance and colour-difference components, E'_Y, $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ are as defined in CCIR Report 624-2. The video interface is as recommended in CCIR Recommendation 567-1.

4.1.2 *Pre- and post-filtering*

4.1.2.1 *Spatial filtering*

A digital filter reduces the 242½ active lines-per-field of the 525-line signal to 143 lines-per-field, the same number as in the 625-line version of the codec. In the decoder, the digital post-filter uses interpolation to restore the signal to 525-lines per picture.

4.1.2.2 *Temporal filtering*

As in the 625 – line version, a recursive temporal pre-filter with non-linear transfer characteristics is used in the coder to reduce noise in the signal and increase coding efficiency. The frame store used in this filter can also be used as the storage element of a frame interpolator with variable coefficients which is used to reduce the transmitted frame rate to a value less than that of the input video signal. In 525-line to 525-line transmission, the transmitted frame frequency is about 29.67 Hz instead of the video rate of 29.97 Hz. In 525-line to 625-line transmission, the transmitted frame frequency is 25 Hz.

Because the (television) frames are leaving the coder more slowly than they are entering, the coding process is suspended for one frame every Nth input frame. N is approximately 100 for 525-line to 525-line operation and approximately 6 for 525-line to 625-line operation.

In the decoder, the digital post-filter incorporates a frame store in some versions of the 625-line codec where it is used in the line interpolation process. In the 525-line version, in addition to its use for line interpolation, it is used as a temporal interpolator with variable coefficients to provide an extra output frame during those periods when the decoding is temporarily suspended.

5.1.1 *Buffer control*

The amount to which the transmitting buffer is filled is used to control various coding algorithms (subsampling, etc.) and is signalled to the decoder to enable it correctly to interpret the received signals. In the 525-line codec, the transmission rate is less than the video input rate and hence the buffer tends to fill more rapidly than would be determined by the movement in the picture, only to empty again when the interpolator suspends the coding process.

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[4] These are typical figures. The delays depend on the detailed implementation used.

To avoid incorrect changes in coding algorithms, the buffer-state signal is modified to take account of the progressively changing coefficients of the interpolator in the pre-filter. The buffer then operates as though the data is coming from a video source whose frame rate is uniform and the same as the transmitted frame rate.

6.3 *Transmission framing*

The frame structure, compatible with Recommendation G.733 and also compatible with that of the 625-line version in Part 1, is given in Part 2 of Recommendation H.130.

ANNEX A

(to Part 2a) of Recommendation H.120)

Graphics option – 525 line

Under study.

ANNEX B

(to Part 2a) of Recommendation H.120)

Encryption option – 525 line

Under study.

PART 3

(to Recommendation H.120)

A Codec for 525-lines, 60 fields/s and 1544 kbit/s transmission for intra-regional use

Under study.

APPENDIX I

(to Recommendation H.120)

Outline description of operation of codecs in Part 1 and Part 2a

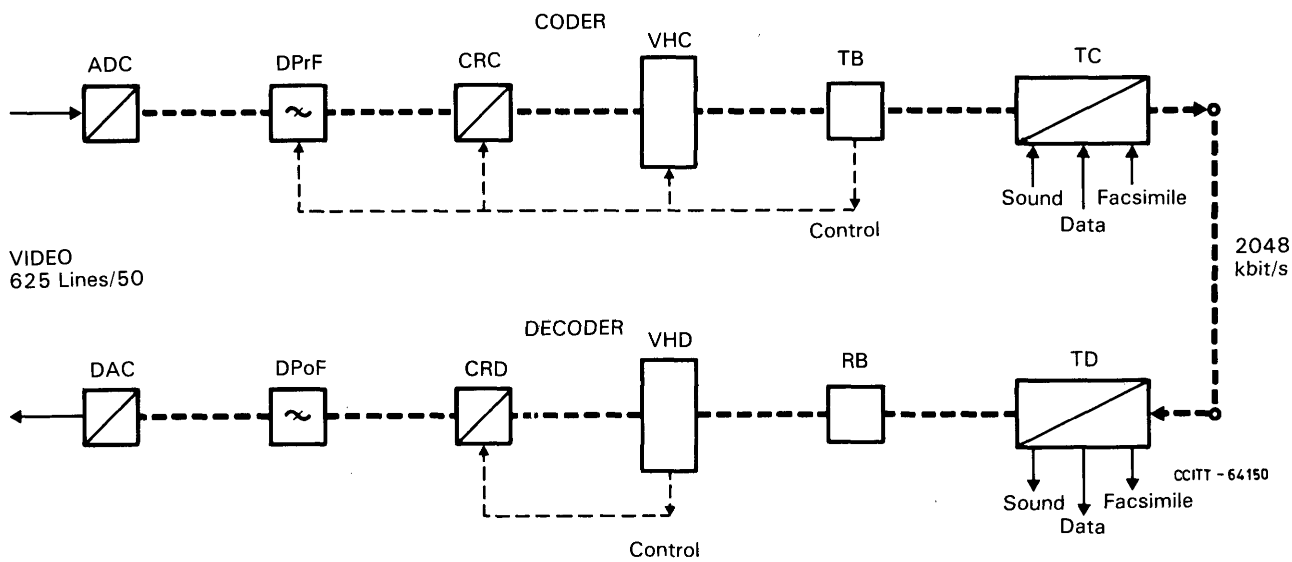
Since the conditional replenishment codec is a complex and unfamiliar item, this simplified outline of its method of operation is included to make the Recommendation more easily comprehensible. More complete descriptions are to be found in published papers [Duffy and Nicol, 1982] and [Nicol, Chiariglione and Schaefer, 1982].

A conditional replenishment codec operates by transmitting only those parts of a picture which differ significantly from one television frame to the next. This normally gives rise to data being generated in spurts separated by gaps in which no data are being generated. To match the non-uniform data generation to a channel transmitting at a uniform rate, a buffer is used to smooth out short-term fluctuations while, for longer-term variations, the coding algorithm is adaptively modified to change the rate of generation. In the event of too much data, caused for example by a lot of movement, the definition of the transmitted moving area is decreased, taking advantage of the reduced ability of the eye to perceive detail as the rate of movement increases. When little movement is present, the moving-area data are supplemented by data from non-moving areas in such a way that the whole picture is replenished over several picture periods. Picture stores are required at both transmitter and receiver and the objective is to make the content of the receiving store follow that of the transmitting store as closely as possible.

The codec can be regarded as comprising three basic sections: the source codec, the video multiplex codec and the transmission codec. Figure I.1/H.120 shows an outline of the arrangement.

In the source codec, the video signal is first digitized and optionally pre-filtered. When used, the pre-filter conditions the signal for further processing by reducing noise to improve the performance of the subsequent movement detector and to reduce the subjective effects of subsampling. The movement detector, in conjunction with the picture store, determines which areas in the picture are deemed to be moving. Noise introduces uncertainties in this decision and, when two or more groups of picture elements along a scanning line are deemed to be moving but are separated by small numbers of non-moving picture elements (probably caused by noise), the moving groups and separating elements are combined to form a single cluster, thus minimizing the addressing information which is required. Clusters of moving picture elements are then coded using DPCM followed by variable-length (entropy) coding where the shortest codes are allocated to the most frequently-occurring DPCM prediction errors.

The video multiplex codec adds to the video data line- and field-synchronization signals together with the addressing and other information (for example, whether PCM or DPCM is being transmitted) which must be transmitted in close association with the video to ensure that the decoder responds correctly.



Coder		Decoder	
ADC	Analogue-Digital Converter	DAC	Digital-Analogue Converter
DPrF	Digital Pre-Filter	DPoF	Digital Post-Filter
CRC	Conditional Replenishment Coder	CRD	Conditional Replenishment Decoder
VMC	Video Multiplex Coder	VMD	Video Multiplex Decoder
TB	Transmission Buffer	RB	Receiving Buffer
TC	Transmission Coder	TD	Transmission Decoder

FIGURE I-1/H.120

Outline block diagram of codec

The buffer, which strictly is part of the source coder, accepts the irregularly spaced bursts of data and delivers them at a uniform rate for transmission. The extent to which the buffer is filled is monitored and this is used to modify the rate of data generation by the source coder. It can reduce the data rate by modifying the pre-filter response and the thresholds in the movement detector, and by initiating element and field subsampling. On the other hand, if the buffer tends to empty, it may initiate the generation of complete PCM coded lines to provide systematic updating of the picture stores.

The transmission codec accepts the video data, adds a 64 kbit/s channel for sound, a 32 kbit/s channel for codec-codec signalling and optional additional data channels for facsimile, signalling or other data. It assembles the various signals into a frame structure, defined in Recommendation H.130 which is compatible with Recommendation G.732 and therefore suitable for transmission on 2048 kbit/s digital paths. In doing so, it provides the justification facilities to enable the clock for video processing to be independent of the network clock.

Bibliography

DUFFY (T. S.) and NICOL (R. C.): A codec for visual teleconferencing, *Communications 82,* IEE Conference Publication No. 209, 1982.

NICOL (R. C.), CHIARIGLIONE (L.) and SCHAWFER (P.): The development of the European Videoteleconference Codec, *Globecom 82,* IEEE global telecommunications conference, 1982.

Recommendation H.130

FRAME STRUCTURES FOR USE IN THE INTERNATIONAL INTERCONNECTION OF DIGITAL CODECS FOR VIDEOCONFERENCING OR VISUAL TELEPHONY

Introduction

Videoconferencing and visual telephony are new services which require greater bit rates than telephony. In the studies in CCITT on the ISDN and on international interworking, 384 kbit/s is emerging as an important channel capacity for wideband services. On this basis it is recommended that the videoconferencing and visual telephone services should be based on multiples of 384 kbit/s.

It is noted that both the 2048 kbit/s and 1544 kbit/s primary digital levels can be expressed by the formula $y + (n \times 384)$ kbit/s, where $n = 5$ or 4 and $y = 128$ or 8 kbit/s, respectively.

While this Recommendation covers only frame structures for transmission at the primary digital rates, it is not intended to suggest that transmissions using other frame structures or formats at primary rates or lower are precluded. In the future frame structures based on other multiples and/or sub-multiples of 384 kbit/s may also be considered.

PART 1

(to Recommendation H.130)

Characteristics of 2048 kbit/s (n = 5) frame structure for use in codecs described in Part 1 of Recommendation H.120

1 General characteristics

The multiplex structure described in this Recommendation is suitable for use on digital paths and connections which interconnect video codecs for videoconferencing or visual telehony using 2048 kbit/s transmission. The connections may either be direct or via higher-order digital multiplex equipment compatible with the primary PCM multiplex equipment defined in Recommendation G.732.

Some of the characteristics of this multiplex structure are identical to those in Recommendation G.732 and are covered by cross references to that Recommendation.

The main features of the multiplex structure are that it provides:

- one 64 kbit/s channel for frame alignment, alarm signals and other signals as required;
- one 64 kbit/s channel reserved for the transmission of the sound signal;
- one 32 kbit/s channel for codec-to-codec information;
- the option of one or two 64 kbit/s channels and/or one 32 kbit/s channel for stereophonic sound, facsimile, data, etc.;
- the possibility of end-to-end and subscriber-to-network signalling;
- the remaining capacity (between 1664 and 1888 kbit/s) is used for the encoded video signal.

1.1 *Fundamental characteristics*

The multiplex structure contains 32 time slots, each of 64 kbit/s.

1.2 *Bit rate*

The nominal bit rate is 2048 kbit/s. The tolerance on this rate is ± 50 parts per million (ppm).

1.3 *Timing signal*

The timing signal is a 2048-kHz signal from which the bit rate is derived. It should be possible to derive the timing signal from an internal source or from the network.

1.4 *Interfaces*

The interfaces should comply with Recommendation G.703.

2 Frame structure and time slot allocations

The frame structure is in accordance with Recommendatiion G.704, § 3.3. The time slot allocations within the frame are given in Table 1/H.130, two options are shown according to whether or not the network is switched (under control of signals within the frame structure).

3 Codec-to-codec information

This information is transmitted in the 32 kbit/s channel corresponding to odd frames of TS2 (frame parity is gained from the multiframe alignment in the 8th bit of alternate Time Slots 2, the frames are consecutively numbered 0 to 15, forming a multiframe).

The 32 kbit/s channel is structured in a multiframe and supermultiframe derived from 128 consecutive 256 bit frames. The multiframe is composed of 8 octets numbered 1, 3, 5, ..., 15, each from TS2 in an odd numbered 256 bit frame. The supermultiframe corresponds to 8 consecutive multiframes which are numbered 0, 1, 2, ..., 3.

The use of the bits in each octet in the odd frames is as follows:

- Bit 1 for clock justification
- Bit 2 for buffer state
- Bit 3 for coding mode identification; the 8 consecutive bits 3 of TS2 in a multiframe will carry the following information:

Bit 3.1 [1]	Codec facilities	(see below)
Bit 3.3	Colour transmission	(1 if provided)
Bit 3.5 [2]	Spare	(set to 0)
Bit 3.7	Fast update request	(1 if required)
Bit 3.9	Advance warning of interruption	(1 if required)

---

1) The notation used here should be interpreted as in the following examples: Bit 3.1 means Bit 3 (in TS2) of frame No. 1 in each multiframe: Bit 3.1.0 means Bit 3 (in TS2) of frame No. 1 in multiframe No. 0 of each supermultiframe.

2) The use of this bit for split-screen indication is under study.

Bit 3.11	Sound power signal, for use with encrypted multipoint	(under study)
Bit 3.13	Data distribution	(1 if required)
Bit 3.15	Spare	(set to 0)

Bit 3.1 is used to signal the availability of certain facilities in the decoder at supermultiframe rate, as follows:

Bit 3.1.0	Graphics (Mode 1)	(1 if provided)
Bit 3.1.1	High-quality speech	(1 if provided)
Bit 3.1.2	4 × 384 kbit/s capability	(1 if provided)
Bit 3.1.3	Encryption	(1 if provided)
Bit 3.1.4	System M	(1 if 525-line signal being coded)
Bit 3.1.5	Graphics (Mode 2)	(1 if provided)
Bit 3.1.6	Graphics (Mode 3)	(1 if provided)
Bit 3.1.7	Spare	(set to 0)

– Bit 4 to identify the use of time slots; the 8 consecutive bits 4 of TS2 in a multiframe will carry the following information:

Bit 4.1	TS2 (even) is used for video (0) or other (1)	
Bit 4.3	TS16 is used for video (0) or other (1)	
Bit 4.5	TS17 is used for video (0) or other (1)	
Bit 4.7	TS18 is used for video (0) or other (1)	
Bit 4.9	4 × 384 kbit/s working. Set to 1 in 1.5 Mbit/s codecs – Note 1	
Bit 4.11	Graphics transmission	(1 if required)
Bit 4.13	Error correction	(1 if required) – Note 2
Bit 4.15	Spare	(set to 0)

– Bit 5 for multipoint conferencing; provides a 4 kbit/s message channel (transparent through the codec) from customer to multipoint control unit, between control units and from customer to customer. (The message format and protocols are under study.)

  When the codec is not equipped with a message channel, bit 5 is used to signal split-screen: 1 = split-screen active, 0 = split-screen inactive.

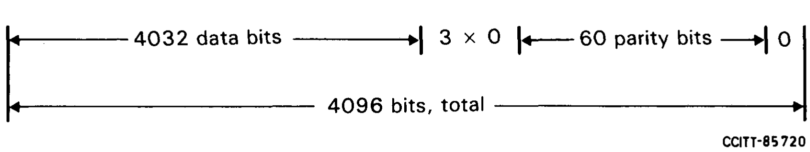
– Bit 6 Free (for possible national use) (set to 0)

– Bit 7 Free (for possible national use)

– Bit 8 For multiframe and supermultiframe alignment; the alues of bit 8 in each frame of the multiframe (multiframe and supermultiframe alignment patterns) should be as detailed in Table 2/H.130.

Note 1 – All 2 Mbit/s codecs will set Bit 4.9 permanently to 0. On receipt of Bit 4.9 set to 1, a 2 Mbit/s codec which allows 4 × 384 kbit/s working (i.e. Bit 3.1.2 set to 1) will vacate Time Slots 16, 26, 27, 28, 29, 30 and 31 in its transmitter and ignore them in the receiver.

Note 2 – When set to 1, the last 64 bits of each multiframe contain the error corrector parity bits. The multiframe then appears as follows:



The conditions signalled in bits 3 and 4 can only change at supermultiframe rate. The change at the decoder will take place at the start of the first supermultiframe following the one where the change in signalling has been detected. This procedure can be used to improve the resistance to transmission errors.

TABLE 1/H.130

Time-Slot Allocation in 32 Time-Slot Frame Structure of Recommendation G.704

	Bit rate (kbit/s)	Time-slot allocation (within the 256-bit frame) Non switched (i)	Switched (ii)
Frame Alignment, Network Alarms, etc.	as in G.704	0	0
Speech information	64	1	1
Codec-to-codec information	32	2	2
Signalling information (subscriber-network)	64	–	16
Fax, data, etc. (optional)	up to 2 × 64	17 and/or 18	17 and/or 18
Encoded video information (minimum)	(i) 27 × 64 (ii) 26 × 64	3 to 16 + 19 to 31	3 to 15 + 19 to 31

Note 1 – Frame alignment, network alarms, etc.

This information is transmitted in TS 0 with the same rules and characteristics as recommended in Recommendation G.704. Additionally, bit 8 in odd frames is used as a synchronization bit which is required when the codec is used with synchronous digital networks. On receipt of this bit set to zero, the transmission clock for the encoder will be derived from the incoming data stream. This bit is always set to one in the encoder.

Note 2 – Speech

Speech is transmitted at 64 kbit/s in TS 1. The coding law is the A-law of Recommendation G.711 or, for future applications, the law that will be recommended by CCITT for higher quality speech. In the case of stereophonic transmission, the second speech channel will be transmitted in TS 17.

Note 3 – Codec-to-codec information

This information requires a capacity of 32 kbit/s and is transmitted on odd frames of TS 2. The remaining 32 kbit/s capacity on the even frames of TE 2 will be used for encoded video or data transmission. The detailed *use* and structure of the 32 kbit/s channel for codec-to-codec information is described in § 3.

Note 4 – Signalling (subscriber-to-network)

A capacity of 16 kbit/s is considered adequate for videoconference as for basic access. The methods of switched access to the ISDN at 2048 kbit/s have not yet been formulated. Option (ii) avoids any problems in this respect, by leaving the whole of Time Slot 16 (64 kbit/s) clear of video information and available for subscriber signalling and call set-up information when switched access is required. For non-switched access, option (i) should be used.

Note 5 – Facsimile, data, etc.

When required, this information will be transmitted in TS 17 and/or 18.

Note 6 – Encoded video

A minimum of 26 × 64 kbit/s capacity is reserved for encoded video in TS 3 to 15 and 19 to 31. In addition, depending on applications, TS 2 (even frames), TS 16, 17 and 18 may also be used for video, providing a maximum of 29.5 × 64 kbit/s capacity; the available video bit-rate therefore lies between 1664 and 1888 kbit/s.

TABLE 2/H.130

Multiframe and Supermultiframe Alignment on Bit 8 of TS 2 (odd)

Multiframe alignment pattern (column 0, frames 1–13)

Frame								
1	1	1	1	1	1	1	1	1
3	1	1	1	1	1	1	1	1
5	1	1	1	1	1	1	1	1
7	0	0	0	0	0	0	0	0
9	0	0	0	0	0	0	0	0
11	1	1	1	1	1	1	1	1
13	0	0	0	0	0	0	0	0
15	1	1	1	0	0	1	0	*
Multiframe	0	1	2	3	4	5	6	7

Supermultiframe alignment pattern (frame 15, multiframes 0–6)

* Undefined (reserved for possible future use in a higher level framing structure).

PART 2

(to Recommendation H.130)

Characteristics of a 1544 kbit/s (n = 4) frame structure for use with codecs described in Part 2a) of Recommendation H.120

1 General characteristics

The multiplex structure described in this Recommendation is suitable for use on digital paths and connections which interconnect video codecs for videoconferencing or visual telephony using 1544 kbit/s transmission. The connections may either be direct or via higher-order digital multiplex equipment compatible with the primary PCM multiplex equipment defined in Recommendation G.733.

Some of the characteristics of this multiplex structure are identical to those in Recommendation G.733 and/or in Part 1 of this Recommendation; these are covered by cross references to the appropriate documents.

The main features of the multiplex structure are that it provides:

- one 8 kbit/s channel for frame alignment, alarm signals and other signals as required;
- one 64 kbit/s channel for the sound signal;
- one 32 kbit/s channel for codec-to-codec information;
- the option of one or two 64 kbit/s channels and/or one 32 kbit/s channel for auxiliary data services;
- the remaining capacity (between 1280 and 1440 kbit/s) is used for the encoded video signal.

1.1 *Fundamental characteristics*

The multiplex structure contains 24 time slots per frame, each of 64 kbit/s, plus one bit per frame for frame alignment and signalling. The number of bits per frame is 193 and the nominal frame repetition rate is 8000 Hz.

1.2 *Bit rate*

The nominal bit rate is 1544 kbit/s. The tolerance on this rate is ± 50 parts per million (ppm).

1.3 *Timing signal*

The timing signal is a 1544 kHz signal from which the bit rate is derived. It should be possible to derive the timing signal from an internal source or from the network.

1.4 *Interfaces*

The interfaces should comply with Recommendation G.703; the option of AMI or B8ZS should be provided as the interface code. Which of the two codes is used should be determined by bilateral agreement.

1.5 *Format restrictions enforced by the network*

As indicated in Recommendation G.703, runs of more than 15 "zeros" are forbidden in some networks; also, there must be, on average, at least three "ones" in every 24 digits. Provision is made by means of a scrambling system to ensure that forbidden patterns cannot occur.

2 Frame structure and time slot allocations

The basic frame structure follows Recommendation G.733 with changes in the Time Slot (TS) allocations. The time slots are numbered from 1 to 24, with the 193rd bit positioned between TS24 and TS1.

2.1 *Frame alignment*

The basic frame alignment is obtained bit No. 193, as in Recommendation G.733. The pattern transmitted is as follows:

Frame Number	Frame alignment signal	S-bit	Signalling bit
1	1	–	
2	–	0	
3	0	–	
4	–	0	
5	1	–	
6	–	1	A
7	0	–	
8	–	1	
9	1	–	
10	–	1	
11	0	–	
12	–	0	B

2.2 *Speech*

Speech is transmitted at 64 kbit/s in TS1. The coding law is the A-law of Recommendation G.711 or, for future applications, the law that will be recommended by CCITT for higher-quality speech. In the case of stereophonic transmission, the second speech channel will be transmitted in TS17.

2.3 *Codec-to-codec information*

This information is transmitted in the 32-kbit/s channel corresponding to the odd frames of TS2. The channel is structured in multiframes of 16 frames and supermultiframes of 8 multiframes is exactly the same way as in the 2-Mbit/s version in Part 1. Multiframe and supermultiframe alignment are obtained from bit 8 of TS2 (odd) in the same way as in Part 1.

The multiframe of TS2 for codec-to-codec signalling is quite independent of the basic 12-frame multiframe of Recommendation G.733.

2.4 *Signalling*

In the future, some 1.5 Mbit/s networks will allow the use of Bits A and B for signalling. This facility is not available on all networks.

2.5 *Facsimile, data, etc.*

When required, this information will be transmitted in TS16 and TS17 and TS2 (even).

2.6 *Encoded video*

A minimum of 20 × 64 kbit/s capacity is reserved for encoded video in TS3-15 and 18-24; depending on applications, TS2 (even), TS16 and 17 may also be used for video, providing a maximum of 22.5 × 64 kbit/s capacity. The available bit rate for video therefore lies between 1280 and 1440 kbit/s.

3 **Codec-to-codec information**

The structure of the multiframe and supermultiframe are exactly the same as in Part 1, except that each frame contains only 24 time slots as compared with 32 in the frames in Part 1.

The bit allocations (in TS2 (odd)) are identical with Part 1, with the following exceptions:

- Bit 3.1.2 is permanently set to 1
- Bit 4.9 is permanently set to 1
- Bit 6 is reserved for the transmission of encryption data (see Annex B to Part 2a of Recommendation H.120).
- Bit 7 if used for scrambler control (see § 4).

4 **Scrambling**

4.1 *General*

The bit sequence produced by a videoconference codec is not subject to any limitation on the bit patterns that are generated. Therefore, reversible processing has to be carried out at the output and input ports to ensure that the format restrictions specified for some 1544 kbit/s networks are not violated.

There are two typical constraints on the format:

1) There must not be runs of more than 15 consecutive "zeros".
2) The average density of "ones" must be at least 12.5%.

A classical self-synchronizing or reset scrambler, based on a maximum-length pseudo-random sequence, is incapable of guaranteeing that such a bit-sequence never occurs. It is however possible, by judicious choice of scrambler design, to minimize the number of violations of the above rules to such an extent that the residual violations can be removed by forcibly inserting "ones". The effect of this is to introduce transmission errors giving a residual bit-error-ratio of approximately 1×10^{-7}, which is imperceptible as far as the picture quality is concerned.

4.2 *Details of scrambling – first stage*

The scrambling sequence is applied to all 24 time slots but not to bit 139 nor to bit 7 of TS2 (odd).

Note – If data are inserted and/or extracted from TS2 (even), 16 or 17 within the network, the insertion/extraction equipments must ensure that the network constraints are not violated.

The 1544 Mbit/s serial data from the codec are first applied to the following scrambling sequence:

I N I N N I, where I = invert and
N = do not invert.

This sequence starts from the bit following bit 193, and is restarted every frame. Bit 193 and bit 7 of TS2 (odd) are not scrambled but the scrambling sequence is continuous through bit 7 of TS2 (odd).

4.3 *Details of scrambling – second stage*

Data scrambled by the above sequence are then checked for runs of more than 15 zeros. For signalling purposes, these data are considered to be in blocks of 385 bits. Each block starts with bit 8 of TS2 (odd) and ends with bit 6 of TS2 (odd). If a block of data preceding bit 7 of TS2 (odd) is found *not* to contain the string of data, 1 00000000 00000000 (i.e. no runs of 16 or more zeros), bit 7 of TS2 (odd) is set to one.

If a block of data signalling bit 7 of TS2 (odd) is found to contain the string of data, 1 00000000 00000001 (i.e. a run of 15 zeros), bit 7 of TS2 (odd) remains set to one, even if one or more subsequent runs of zeros within the same block reaches or exceeds 16. However, in such a case, the 16th zero(s) of the run(s) are set to one. As this is not signalled to the descrambler, it causes (a) single-bit transmission error(s).

Bit 7 of TS2 (odd) is set to zero only if the preceding block of data is found to contain the string, 1 00000000 00000000 (i.e. a run of 16 zeros or more), in which case the 16th zero is inverted to one and all subsequent strings of the form 1 00000000 0000000B within the same block have bit B inverted, except in the case where bit B = 1 before inversion, in which case it remains unchanged.

4.4 *Details of descrambler*

When bit 7 of TS2 (odd) is one, the preceding block of scrambled data is left unchanged. When bit 7 of TS2 (odd) is zero, the descrambler must detect all occurrences of the string 1 00000000 0000000B in the preceding block and invert the bit B. This can introduce transmission errors if the second or subsequent runs of zeros within the block (at the scrambler) contain 15 zeros.

The repetitive scrambling sequence, I N I N N I, is then applied to the data.

For the purpose of counting runs of zeros, at both the scrambler and descrambler, bit 7 of TS2 (odd) and bit 193 are both assumed to be zero. In the case where bit B would be on bit 193 or bit 7 of TS2 (odd), the string 1 00000000 000000B is used instead of 1 00000000 0000000B. Only bit B has to be within the block of data being considered. The preceding zeros may lie partially or completely within the peceding block.

When bit B is inverted, the "zeros" counter is reset to zero.

PART 3

(to Recommendation H.130)

Characteristics of a 1544 kbit/s (n = 4) frame structure for use with codecs described in Part 3 of Recommendation H.120

Under study.

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PART II

Series J Recommendations

SOUND-PROGRAMME AND TELEVISION TRANSMISSIONS

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SECTION 1

GENERAL RECOMMENDATIONS CONCERNING SOUND-PROGRAMME TRANSMISSIONS

Recommendation J.11

HYPOTHETICAL REFERENCE CIRCUITS FOR SOUND-PROGRAMME TRANSMISSIONS [1], [2], [3]

(Geneva, 1972; amended at Geneva, 1976)

Terrestrial systems and systems in the fixed-satellite service

The CCITT,

considering

(a) that there is a need to define a hypothetical reference circuit to enable design performance standards to be set;

(b) that the hypothetical reference circuit should allow the different types of sound-programme circuits to be compared on a common basis;

unanimously recommends

(1) that the main features of the hypothetical reference circuit for sound-programme transmissions over a terrestrial system (shown in Figure 1/J.11), which may be provided by either radio or cable, should be:

- the overall length between audio points (B and C) is 2500 km,
- two intermediate audio points (M and M') which divide the circuit into three sections of equal lengths,
- the three sections which are lined up individually and then inter-connected without any form of overall adjustment or correction;

(2) that the main features of the hypothetical reference circuit for sound-programme transmissions over a system in the Fixed-Satellite service (shown in Figure 2/J.11) should be:

- one link: earth station – satellite – earth station,
- one pair of modulation and demodulation equipments for translation from baseband to radio frequency, and from radio frequency to baseband, respectively.

---

[1] This Recommendation corresponds to CCIR Recommendation 502-2.

[2] The hypothetical reference circuits defined in this Recommendation should apply for both analogue and digital systems.

[3] For maintenance purposes there may be a need to define other circuits of which an illustration is shown in Annex A of this Recommendation.

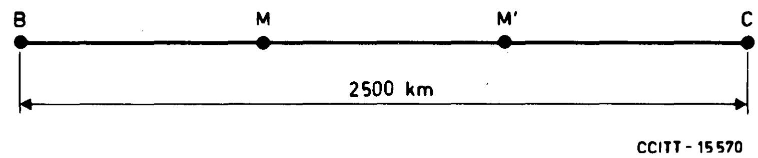


FIGURE 1/J.11

Hypothetical reference circuit for sound-programme transmissions over a terrestrial system

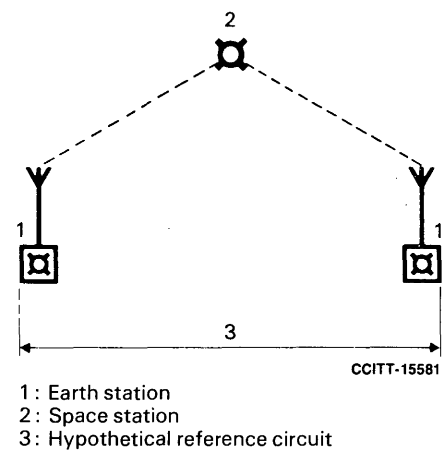


FIGURE 2/J.11

Hypothetical reference circuit for sound-programme transmissions over a system in the Fixed-Satellite Service

ANNEX A

(to Recommendation J.11)

Illustration of an international sound-programme connection

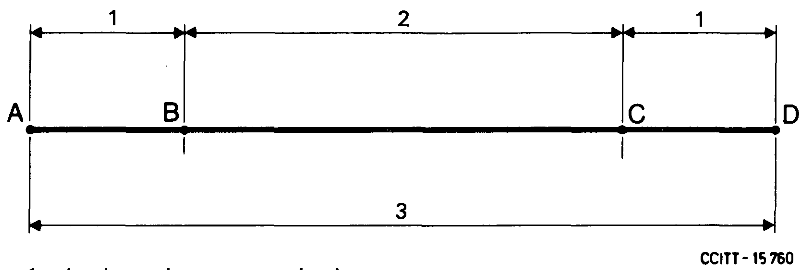
Figure A-1/J.11 illustrates a typical international sound-programme connection in which:

- point A, to be considered as the sending end of the international sound-programme connection, may be the point at which the programme originates (studio or outside location);
- point D, to be considered as the receiving end of the international sound-programme connection, may be a programme-mixing or recording centre or a broadcasting station;
- the local sound-programme circuit AB connects point A to the sending terminal station, point B, of the international sound-programme circuit BC;
- the local sound-programme circuit CD connects point C, the receiving terminal station of the international sound-programme circuit BC to the point D.

The hypothetical reference circuit must not be considered identical to any of the sound-programme circuits illustrated above or to those defined for maintenance purposes in Volume IV-3 of the CCITT Yellow Book. However, some of these circuits may display the same structure as the hypothetical reference circuit. Such types of circuits are:

- an international sound-programme connection comprising three audio sections;
- a single sound-programme circuit made up of three audio sections.

In this case, the performance standards set for the hypothetical reference circuit may be applied to these circuits.



1 — local sound-programme circuits
2 — international sound-programme circuit
3 — international sound-programme connection

FIGURE A-1/J.11

An international sound-programme connection

Recommendation J.12

TYPES OF SOUND-PROGRAMME CIRCUITS ESTABLISHED OVER THE INTERNATIONAL TELEPHONE NETWORK

(former Recommendation J.11; amended at Geneva, 1972 and 1980)

The CCITT recognizes the types of sound-programme circuits defined below.

Note – For the purposes of this Recommendation and other Recommendations in the Series J, sound-programme circuits have been classified in terms of the nominal effectively transmitted bandwidth. For convenience, the corresponding type of circuit from the administrative point of view (see Recommendation D.180 [1]) is given under each type of equipment in the following paragraphs.

1 15 kHz-type sound-programme circuit

This type of circuit is recommended for high-quality monophonic programme transmission and in certain arrangements is also recommended for stereophonic transmissions. This type of circuit corresponds to the "very wideband circuit" or "stereophonic pair", as appropriate, referred to in Recommendation D.180 [1].

The performance characteristics of 15 kHz-type sound-programme circuits suitable for both monophonic and stereophonic transmissions are defined in Recommendation J.21 and suitable equipment is specified in Recommendation J.31, for analogue transmission and in Recommendations J.41, G.735 and G.737 for digital transmission.

2 10 kHz-type sound-programme circuit

This type of circuit, previously known as the "normal programme circuit, type A", is recommended for monophonic transmission only. This type of circuit corresponds to the "wideband circuit" referred to in Recommendation D.180 [1]. The performance characteristics of 10 kHz-type sound-programme circuits are defined in Recommendation J.22 and suitable methods of provision are given in Recommendation J.32.

3 Narrow bandwidth sound-programme circuit (7 and 5 kHz-type sound-programme circuit)

These types of circuits are recommended:

- for setting up a large number of temporary sound-programme circuits for the transmission of commentaries and reports on events of large interest (e.g. sporting events); and
- for permanent sound-programme circuits which are used primarily for speech transmission or as connection between studio outputs and long-, medium- or short-wave broadcast-transmitter inputs.

The performance characteristics of narrow bandwidth sound-programme circuits are defined in Recommendation J.23, and as suitable equipment for 7 kHz-type circuit is specified in Recommendation J.34, for analogue transmission.

Note – These types of circuits fall within the category of "medium-band circuits" referred to in Recommendation D.180 [1] for tariff purposes.

4 Use of ordinary telephone circuits

For this type of transmission of special programmes such as speech, some operational aspects are given in Recommendation N.15 [2].

References

[1] CCITT Recommendation *International sound- and television-programme transmissions*, Vol. II, Rec. D.180.

[2] CCITT Recommendation *Maximum permissible power during an international sound-programme transmission*, Vol. IV, Rec. N.15.

Recommendation J.13

DEFINITIONS FOR INTERNATIONAL SOUND-PROGRAMME CIRCUITS

(former Recommendation J.12; amended at Geneva, 1972 and 1980)

Definition of the constituent parts of an international sound-programme connection

The following definitions apply to international sound-programme transmissions.

1 international sound-programme transmission

The transmission of sound over the international telecommunication network for the purpose of interchanging sound-programme material between broadcasting organizations in different countries. Such a transmission includes all types of programme material normally transmitted by a sound broadcasting service, for example, speech, music, sound accompanying a television programme, etc.

2 broadcasting organization (send)

The broadcasting organization at the sending end of the sound programme being transmitted over the international sound-programme connection.

3 broadcasting organization (receive)

The broadcasting organization at the receiving end of the sound programme being transmitted over the international sound-programme connection.

4 international sound-programme centre (ISPC)

A centre at which at least one international sound-programme circuit terminates and in which international sound-programme connections can be made by the interconnection of international and national sound-programme circuits.

The ISPC is responsible for setting up and maintaining international sound-programme links and for the supervision of the transmissions made on them.

5 international sound-programme connection

5.1 The unidirectional path between the broadcasting organization (send) and the broadcasting organization (receive) comprising the international sound-programme link extended at its two ends over national sound-programme circuits to the broadcasting organizations (see Figure 2/J.13).

5.2 The assembly of the "international sound-programme link" and the national circuits between the broadcasting organizations, constitutes the "international sound-programme connection". Figure 3/J.13 illustrates, by way of example, an international sound-programme connection as it might be encountered in practice.

6 **international sound-programme link** (Figure 2/J.13)

The unidirectional path for sound-programme transmissions between the ISPCs of the two terminal countries involved in an international sound-programme transmission. The international sound-programme link comprises one or more international sound-programme circuits interconnected at intermediate ISPCs. It can also include national sound-programme circuits in transit countries.

7 **international sound-programme circuit** (Figure 1/J.13)

The unidirectional transmission path between two ISPCs and comprising one or more sound-programme circuit sections (national or international), together with any necessary audio equipment (amplifiers, compandors, etc.).

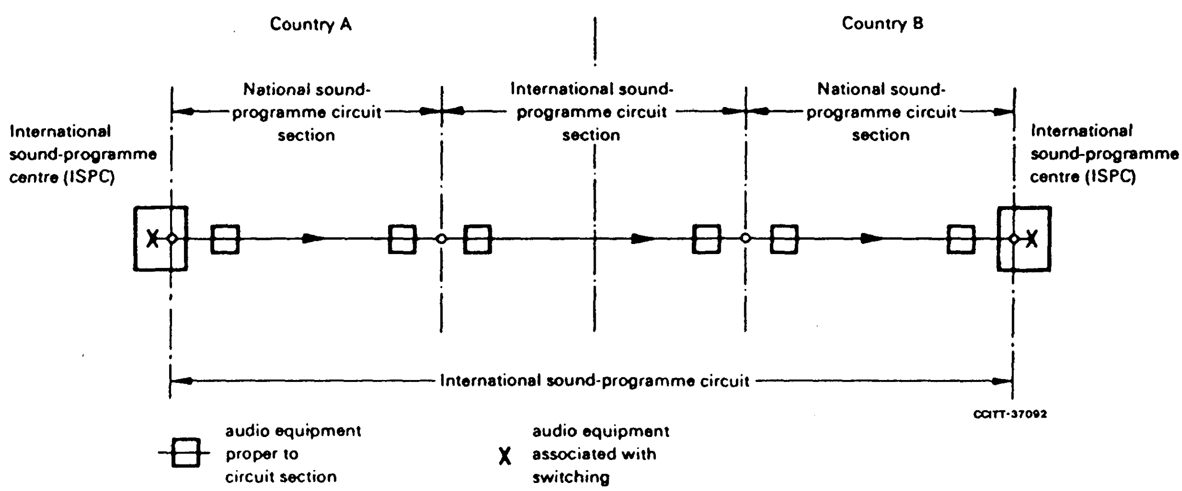


FIGURE 1/J.13

An international sound-programme circuit composed of two national and one international sound-programme circuit-section

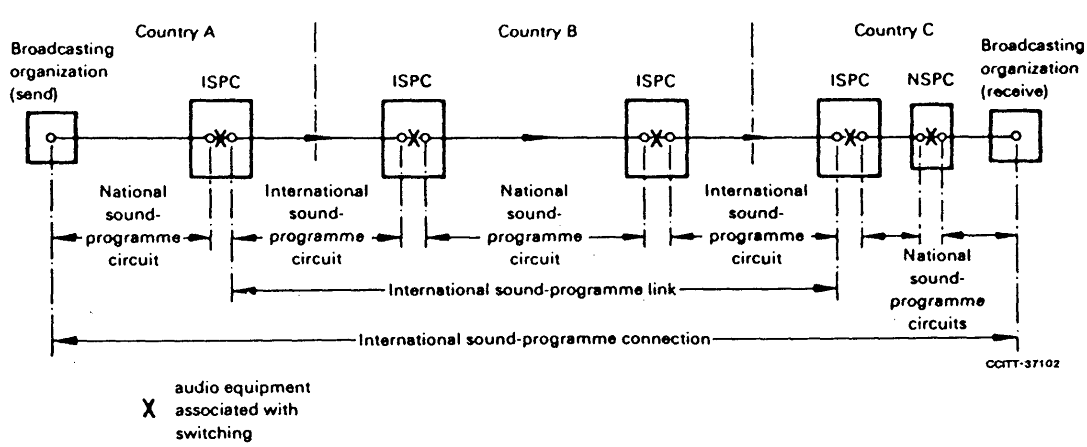
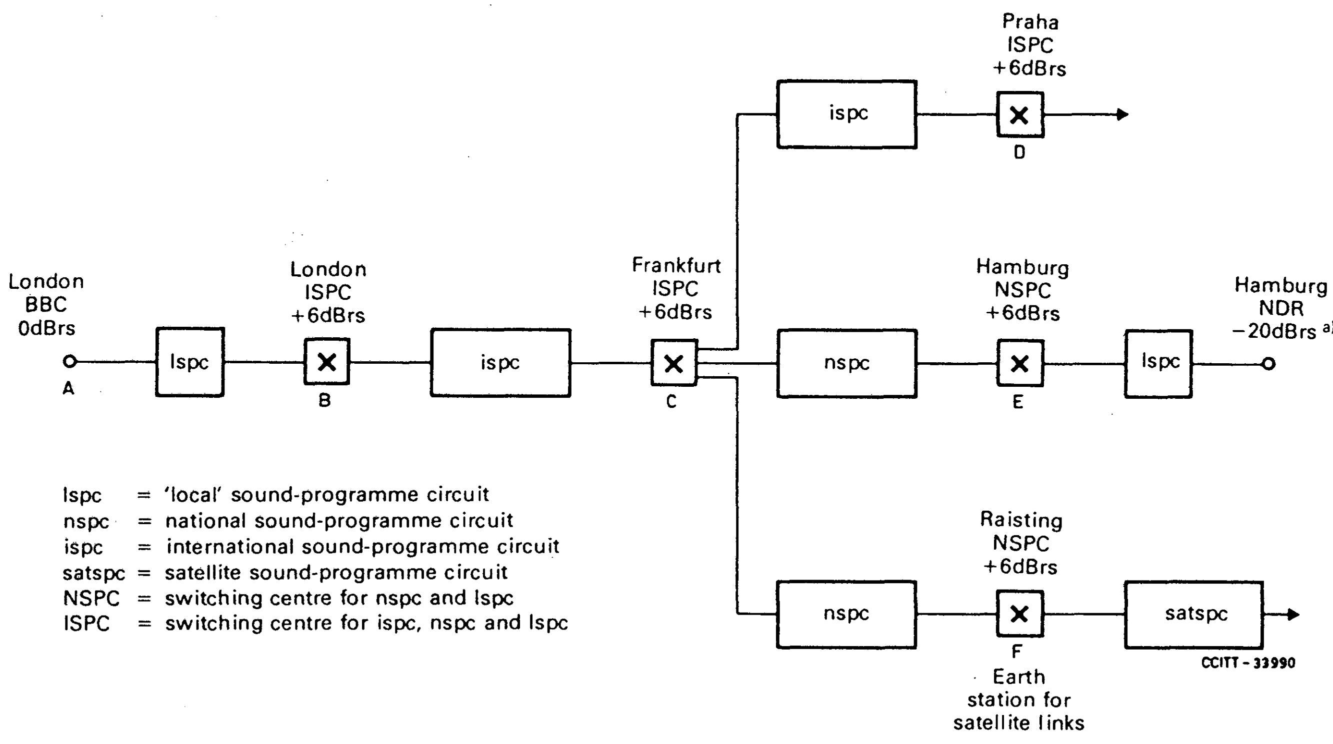


FIGURE 2/J.13

An international sound-programme link composed of international and national sound-programme circuits and extended on a national sound-programme circuit at each end to form an international sound-programme connection



Note – Maximum level of sound programme signals: +9dBm0s (this means +9dBms at a 0dBrs relative level point and +15dBms at a +6dBrs relative level point respectively). The value of +9dBms corresponds to a peak voltage of 3.1 V which is the maximum value of a sine-wave signal of 2.2 V r.m.s.

a) Other values can be chosen by the relevant Administration on a national basis.

FIGURE 3/J.13

Diagram of an international sound-programme circuit

8 **sound-programme circuit-section** (Figure 1/J.13)

Part of an international sound-programme circuit between two stations at which the programme is transmitted at audio frequencies.

The normal method of providing a sound-programme circuit section in the international network will be by the use of carrier sound-programme equipment. Exceptionally sound-programme circuit sections will be provided by other means, for example, by using amplified unloaded or lightly loaded screened-pair cables or by using the phantoms of symmetric-pair carrier cables.

9 **national circuit**

The national circuit connects the ISPC to the broadcasting authority; this applies both at the sending and at the receiving end. A national circuit may also interconnect two ISPCs within the same country.

10 **effectively transmitted signals in sound-programme transmission**

For sound-programme transmission, a signal at a particular frequency is said to be effectively transmitted if the nominal overall loss at that frequency does not exceed the nominal overall loss at 800 Hz by more than 4.3 dB. This should not be confused with the analogous definition concerning telephony circuits given in [1].

For sound-programme *circuits*, the overall loss (relative to that at 800 Hz) defining effectively transmitted frequency is 1.4 dB, i.e. about one-third of the allowance.

Reference

[1] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151, § 1, Note 1.

RELATIVE LEVELS AND IMPEDANCES ON AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

(former Recommendation J.13; amended at Geneva, 1972, 1976 and 1980)

1 Level adjustment on an international sound-programme connection

The CCITT recommends the use of the *constant voltage* method. If, to a zero relative level point of the international sound-programme connection, a zero absolute voltage level is applied (sine-wave signal of 0.775 volts r.m.s.) at the reference frequency 0.8 or 1 kHz, the absolute voltage level at the output of each sound-programme circuit (Points B, C, D ... F of Figure 3/J.13) should be +6 dB (i.e. 1.55 volts r.m.s.). Therefore these points have to be regarded as relative level points of +6 dBrs according to Recommendations J.21, J.22 and J.23.

The zero relative level point is, in principle, the origin of the international sound-programme connection (Point A in Figure 3/J.13). Different conventions may be agreed between the telephone Administration and the broadcast organization within a country, provided that the levels on the international sound-programme link are unchanged.

A zero relative level point is, in principle, a point at which the sound-programme signals correspond exactly with those at the origin of the international sound-programme connection. At a point of zero relative levels, signals have been controlled in level by the broadcasting organization such that the peak levels very rarely exceed +9 dB relative to the peak values reached by a sine-wave signal of 0.775 volts r.m.s. (for a 600-ohm resistor load, when levels are expressed in terms of dBm).

In a Draft New Recommendation Study Group 10/CCIR has defined test signals to be used on international sound-programme connections based on existing CCITT Recommendations. These definitions are given in Annex A for information.

2 Diagram of signal levels on an international sound-programme connection

All signal levels are expressed in terms of r.m.s. values of sine-wave signals with reference to 0.775 volts.

The voltage level diagram for an international sound-programme connection, however made up, should be such that the voltage levels shown are not such as to exceed the maximum undistorted power which an amplifier can deliver to a sound-programme link when a peak voltage (i.e. +9 dB) is applied to a zero relative level point on the international sound-programme connection.

With these conditions, +6 dB is the nominal voltage level at the output of the terminal amplifiers of the sound-programme circuits making up the international sound-programme link (Points B, C, D ... F of Figure 3/J.13).

Considering the fact,

- that rare excursions of the permitted maximum signal level may occur, and
- that adjustment errors and maintenance tolerance have to be taken into account.

Sound-programme circuits need a definite overload margin. The amount of this margin is still under study.

If a sound-programme circuit which is part of the international sound-programme link is set up on a group in a carrier system, the objective for a new design of equipment is that the relative level of the sound-programme circuit, with respect to the relative level of the telephone channel, should be chosen such that the mean value and the peak value of the load presented by the sound-programme channel should be no higher than that of the telephone channels which are replaced by the sound-programme channel. The effects of pre-emphasis and compandors should, where present, be taken into consideration.

It is recognized that this condition may not be observed in all cases, particularly in certain existing types of equipments. It is recommended that in those cases the zero relative level points of the sound-programme circuit and of the telephone channels should coincide.

It might be as well, however, if the equipment could, where possible, tolerate a maximum difference of ± 3 dB between the relative levels of the sound-programme and telephone transmissions, so that the best adjustment can be obtained, depending on any noise or intermodulation present, but at the same time observing the constraints imposed by the considerations on loading.

Note – The relative level at which the modulated sound-programme signal is applied to the group link are given in Recommendations J.31 for 15 kHz-type circuits, J.34 for 7 kHz-type circuits and in the Annex to Recommendation J.22 for 10 kHz-type circuits.

3 Definitions and abbreviations for new sound-programme signals

Definitions and symbols are in current use to define relative levels for telephony. However, additional definitions and symbols are necessary for the absolute and relative levels in respect of sound-programme signals. The corresponding definitions and symbols for telephony and sound-programme signals are given below.

3.1 dBm0 [1)]

The absolute signal power level, in decibels, referred to a point of zero relative level.

3.2 dBr [1)]

The relative power level, in decibels.

3.3 dBm0s

The absolute signal power level, in decibels, referred to a point of zero relative sound-programme level.

3.4 dBrs

The relative (power) level, in decibels, with respect to sound-programme signals. (This abbreviation is only applicable at points in a sound-programme circuit where the signals can nominally be related to the input by a simple scaling factor.)

Note – For information the opinion of Study Group 10/CCIR to the use of level definitions is given in Annex B.

ANNEX A

(to Recommendation J.14)

TEST SIGNALS TO BE USED ON INTERNATIONAL SOUND-PROGRAMME CONNECTIONS [2)]

The CCIR,

CONSIDERING

(a) that many impairments in international programme exchange on sound-programme connections are to be attributed to different national test signal definitions;

(b) that some existing definitions are to be found in different Recommendations of the CCITT and the CCIR;

(c) that for clarification a list of those definitions should be available;

UNANIMOUSLY RECOMMENDS

that for an international sound-programme connection only the test signals defined below should be used.

---

1) These symbols traditionally relate to telephony relative levels.

2) Under these conditions a peak programme meter will indicate levels not exceeding the level of the permitted maximum signal (PMS).

A.1 *Alignment signal (AS)*

Sine-wave signal at a frequency of 1 kHz, which is used to align the international sound-programme connection. The signal level corresponds to 0 dBm0s (i.e. 0.775 volts r.m.s. at a zero relative level point). In accordance with CCITT Recommendation N.13, the period of sending the alignment signal should be kept as short as possible – preferably to less than 30 seconds.

A.2 *Measurement signal (MS)*

Sine-wave signal at a level 12 dB below the alignment signal level which should be used for long-term measurements and measurements at all frequencies. See CCITT Recommendations N.12, N.13, N.21 and N.23.

A.3 *Permitted maximum signal (PMS)*

Sine-wave signal at 1 kHz, 9 dB above the alignment signal level, equivalent to the permitted maximum programme-signal level. The sound-programme signal should be controlled by the sending broadcaster so that the amplitudes of the peaks only rarely exceed the peak amplitude of the PMS sine-wave test signal.[3)]

A numerical example may serve to clarify this definition. The alignment signal has a r.m.s. voltage of 0.775 V and a peak amplitude of 1.1 V at a zero relative level point. The instantaneous peak amplitude of the sound-programme signal at this point should only rarely exceed 3.1 V.

Although it is intended that the peaks of the sound-programme signal should not exceed the permitted maximum signal level, an overload margin must be provided so that rare excursions of the sound-programme signal above the permitted maximum signal level may be tolerated.

ANNEX B

(to Recommendation J.14)

Opinion of Study Group 10 for a draft amendment to CCIR Recommendation 574-1

Logarithmic quantities and units
(Volume XIII, page 85)

Study Group 10, taking into account the need to introduce unified methods of measurements for audio channels adopted by all sound and television broadcasting organizations and Administrations members of the CCIR, suggests introduction of the term "dBu" as the measuring unit for absolute voltage level.

Up to today only the term "dBm" was used for this purpose, and this fact has often caused confusion since CCIR Recommendation 574-1 states that "dBm" should be used as a measuring unit only for absolute power level (see Recommendation 574-1, § 2.10).

During the Interim Meeting of CCIR, Geneva, 1983, and together with the third meeting of IWP 10/6, Study Group 10 decided to prepare a draft amendment of Recommendation 574-1 (Annex I) which is intended to be discussed by the Joint CCIR/CCITT Study Group for Vocabulary (CMV), during the next study period. If the term "dBu" is accepted by the CMV it might be necessary to define related units.

3) Under these conditions a peak programme meter will indicate levels not exceeding the level of the permitted maximum signal (PMS).

LINING-UP AND MONITORING AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

(former Recommendation J.14; amended at Geneva, 1972 and 1980)

For the alignment of international sound-programme connections Study Group CMTT is prepared to recommend a

Three-level test signal

This proposal is based on the test signal definitions given in Annex A of Recommendation J.14 and specifies a test signal generator which should be used on sound-programme circuits generally. This specification is reproduced in Annex A. Furthermore in Annex B. A common alignment procedure for Peak Programme Meters and VU-meters using the three-level test signal is given. From this information it can be seen what indicators will be produced by the three-level test signal on the different types of Peak Programme Meters and Volume Meters.

To comply with the provisions of Recommendation J.14, the lining-up and monitoring of an international sound-programme connection should ensure that, during the programme transmission, the peak voltage at a zero relative level point will not exceed 3.1 volts, which is that of a sinusoidal signal having an r.m.s. value of 2.2 volts. The methods for achieving this condition as well as the relevant performance requirements are given in Recommendations N.10 to N.18 (see references [1] to [8]).

Some indication of the volume or of the peaks of the signals during programme transmission may be obtained by monitoring at the studio, in the repeater stations, or at the transmitter. One of the instruments, the characteristics of which are summarized in Table 1/J.15, may be used.

Since there is no simple relation between the readings given by two different instruments for all types of programme transmitted, it is desirable that the broadcast organization controlling the studio and the telephone Administration(s) controlling the sound-programme circuit should use the same type of instrument so that their observations are made on a similar basis.

In general the telephone Administration and the broadcast organization of a country agree to use the same type of instrument. It is desirable to reduce to a minimum the number of different types of instrument and to discourage the introduction of new types which only differ in detail from those already in service. The unified use of the peak indicator specified in reference [9] is under study.

During programme transmission, the signal level at the output of the last amplifier controlled by the sending broadcast organization (Point A of Figure 3/J.13) should be monitored to see that the meter deflection of the measuring instrument is always lower than the peak voltage for the overall line-up, allowance being made for the peak factor of the programme involved.

It should be remembered that the amplitude range from a symphony orchestra is of the order of 60 to 70 dB, while the specification for sound-programme circuits is based on a range of about 40 dB. Before being passed to the sound-programme circuit, therefore, the dynamic ratio of the studio output needs to be compressed.

TABLE 1/J.15

Principal characteristics of the various instruments used for monitoring the volume or peaks during telephone conversations or sound-programme transmission

Type of instrument	Rectifier characteristic (see Note 3)	Time to reach 99% of final reading (milliseconds)	Integration time (milliseconds) (see Note 4)	Time to return to zero (value and definition)
(1) vu meter (United States of America) (see Note 1)	1.0 to 1.4	300	165 (approx.)	equal to the integration time
(2) Peak indicator for sound-programme transmissions used by the British Broadcasting Corporation (BBC Peak Programme Meter) (see Note 2)	1		10 (see Note 5)	3 seconds for the pointer to fall 26 dB
(3) Maximum amplitude indicator used by the Federal Republic of Germany (type U 21)	1	around 80	5 (approx.)	1 or 2 seconds from 100% to 10% of the reading in the steady state
(4) OIRT – Programme level meter: type A sound meter type B sound meter		for both types: less than 300 ms for meters with pointer indication and less than 150 ms for meters with light indication	 10 ± 5 60 ± 10	for both types: 1.5 to 2 seconds from the 0 dB point which is at 30% of the length of the operational section of the scale
(5) E.B.U. standard peak programme meter (see Note 7)	1	–	10	2.8 seconds for the pointer to fall 24 dB

Note 1 – In France a meter similar to the one defined in line (2) of the table has been standardized.

Note 2 – In the Netherlands a meter (type NRU-ON301) similar to the one defined in line (4) of the table has been standardized.

Note 3 – The number given in the column is the index *n* in the formula $V_{(output)} = [V_{(input)}]^n$ applicable for each half-cycle.

Note 4 – The "integration time" was defined by the CCIF as the "minimum period during which a sinusoidal voltage should be applied to the instrument for the pointer to reach to within 0.2 neper or nearly 2 dB of the deflection which would be obtained if the voltage were applied indefinitely". A logarithmic ratio of 2 dB corresponds to 79.5% and a ratio of 0.2 neper to 82%.

Note 5 – The figure of 4 milliseconds that appeared in previous editions was actually the time taken to reach 80% of the final reading with a d.c. step applied to the rectifying integrating circuit. In a new and somewhat different design of this programme meter using transistors, the performance on programme remains substantially the same as that of earlier versions and so does the response to an arbitrary, quasi-d.c. test signal, but the integration time, as defined in Note 4, is about 20% greater at the higher meter readings.

Note 6 – In Italy a programme meter with the following characteristics is in use:

Rectifier characteristic: 1 (see Note 3)
Time to reach 99% of final reading: approx. 20 ms
Integration time: approx. 1.5 ms
Time to return to zero: approx. 1.5 s from 100% to 10% of the reading in the steady state.

Note 7 – This meter is intended specifically for use in monitoring sound signals transmitted internationally, and therefore incorporates a scale conforming to CCITT Recommendation N.15[5], calibrated in dB from −12 to +12 relative to a level marked "TEST" corresponding to 0 dBm at a zero relative level point. In addition to the normal mode of operation having the characteristics shown above, the meter may be operated temporarily in a "slow" mode facilitating the comparison of observations made at widely separate points. The peak values indicated in this mode have no absolute significance, and may only be used for such comparisons.

References

[1] CCITT Recommendation *Limits for international sound-programme links and connections*, Vol. IV, Rec. N.10.

[2] CCITT Recommendation *Essential transmission performance objectives for international sound-programme centres (ISPC)*, Vol. IV, Rec. N.11.

[3] CCITT Recommendation *Measurements to be made during the line-up period that precedes a sound-programme transmission*, Vol. IV, Rec. N.12.

[4] CCITT Recommendation *Measurements to be made by the broadcasting organizations during the preparatory period*, Vol. IV, Rec. N.13.

[5] CCITT Recommendation *Maximum permissible power during an international sound-programme transmission*, Vol. IV, Rec. N.15.

[6] CCITT Recommendation *Identification signal*, Vol. IV, Rec. N.16.

[7] CCITT Recommendation *Monitoring the transmission*, Vol. IV, Rec. N.17.

[8] CCITT Recommendation *Monitoring for charging purposes, releasing*, Vol. IV, Rec. N.18.

[9] IEC Publication 268-10A.

ANNEX A

(to Recommendation J.15)

Report 820 (MOD I)

A.1 The text below is proposed for insertion immediately following § 3.2:

"3.3 *A three-level test signal*

As means to assist in identification of various test levels experienced in practice, a new test signal is proposed in the draft new Recommendation in Annex A. The test signal comprises a 1 kHz tone presented cyclically at +9 dBm0s, 0 dBm0s and −12 dBm0s; and a stored speech message peaking to the permitted maximum programme level.

3.4 *Preferred monitoring instrument*

The CMTT considers that programme meters conforming to IEC Publications 268-10 and 268-10-A are suitable for measurements of levels in international sound-programme connections."

A.2 The following text is proposed:

PROPOSAL FOR A NEW RECOMMENDATION

THREE-LEVEL TEST SIGNAL FOR THE ALIGNMENT OF INTERNATIONAL SOUND-PROGRAMME CONNECTIONS

The CCIR,

CONSIDERING

(a) that the sinusoidal test signals specified in the CCITT Recommendations of the J and N series are each restricted to a single level value only;

(b) that no information can be derived from these test signals about the relationship which exists at the point of input between the actual level measured and the permitted maximum programme level;

(c) that many impairments in international programme exchanges can be traced to misinterpretations of the single-level test signals;

[UNANIMOUSLY] RECOMMENDS

1. that a three-level sinusoidal test signal at a reference frequency of 1 kHz should be used to check the alignment of international sound-programme connections;

2. that the following three levels should be allocated to such a test signal:

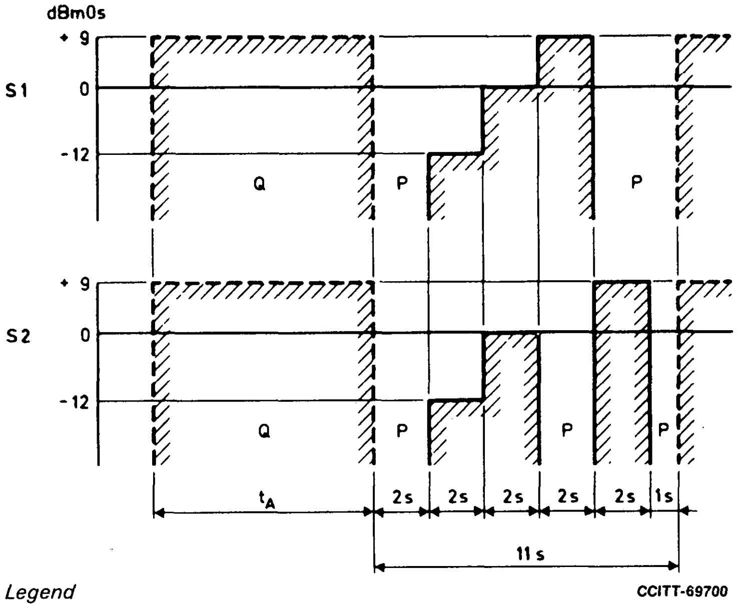
+9 dBm0s permitted maximum programme level;[1]

0 dBm0s alignment level;[1]

−12 dBm0s measuring level;[1]

3. that for monophonic and stereophonic connections the three levels of such a signal should be repeated cyclically as specified in the format shown in Figure A-1/J.15;

4. that this three-level test signal should be combined with a station announcement.



Legend

Cycle duration t_A + 11 s

Q = station announcement

S 1 = left stereo information or monophonic information

S 2 = right stereo information

P = signal pauses

t_A = duration of the station announcement

Note – t_A varies depending on the length of the message.

FIGURE A-1/J.15

Format for the three-level test signal for sound-programme connections

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[1] The terms used to describe the test signal levels are confirmed by Study Group 10 of CCIR (see draft Recommendation AB/10 in interim booklet Document 10/180).

ANNEX B

(to Recommendation J.15)

A common alignment procedure for PPM and VU Meters using the three-level tone test signal

B.1 Broadcasters have evolved, over a period of forty years, procedures for using both types of meter to control programme levels, procedures which are satisfactory to the organizations using them, so that they produce neither over-modulation, leading to distortion, nor under-modulation, leading to impairment from noise.

Although different kinds of programme material deflect the two meters differently, the organizations using them have evolved techniques that produce satisfactory level control and artistic balance within the programme.

B.2 The sensitivity of Peak Programme Meters is such that a sine wave signal at the Alignment Level, 0 dBm0s, indicates "Test" on an EBU PPM (this corresponds to "4" on the BBC PPM and "−9" on the PPMs of the Federal Republic of Germany and OIRT, see Figure B-1/J.15).

B.3 The sensitivity of the VU Meter is such that a sine wave signal at the Alignment Level, 0 dBm0s, produces nearly full indication, 0 VU in Australia and North America and +2 VU in France, see Figure B-1/J.15.

B.4 The PPM reads "quasi-peak", that is, its peak indication on programme signals reads a little lower than true peaks. Operators are instructed to make the programme peaks give the same indication as a sinusoidal tone at +9 dBm0s[2)]. The true peaks of the programme are higher than indicated by up to 3 dB. When, additionally, operator errors are taken into account, the true peaks of the programme signal may reach the amplitude of a sinusoidal tone at +15 dBm0s.

B.5 The VU meter indicates the mean level of the programme, which is generally much lower than the true peak. Operators are instructed to make programmes peak generally to the 0 VU reading. Experience has shown that the true programme peaks are higher than indicated by between +6 dB and +13 dB, depending on the programme material. When, additionally, operator errors are taken into account, the true peaks of the signal may be up to 16 dB higher than indicated, corresponding to the peak amplitude of a sinusoidal tone at +16 dBm0s, or alternatively +14 dBm0s when application of the Alignment Level signal results in +2 VU indication.

B.6 Thus although the dynamic characteristics of the two meters are different, the highest peak levels encountered after control using either meter are very similar.

B.7 Thus an international connection between broadcasters will be correctly aligned when a sinusoidal signal at Alignment Level, 0 dBm0s, produces the indication appropriate to that level at both the sending and receiving ends of the circuit, regardless of the type of meter employed.

To avoid any confusion between Alignment Level and other levels that might be used, it is recommended that the Three-Level Tone Test Signal described in Document CMTT/87(Rev.1) be used for the alignment of an international sound-programme connection.

The diagram, Figure B-1/J.15, illustrates the indications given by a number of programme level meters when the Three-Level Tone Test Signal is applied to them.

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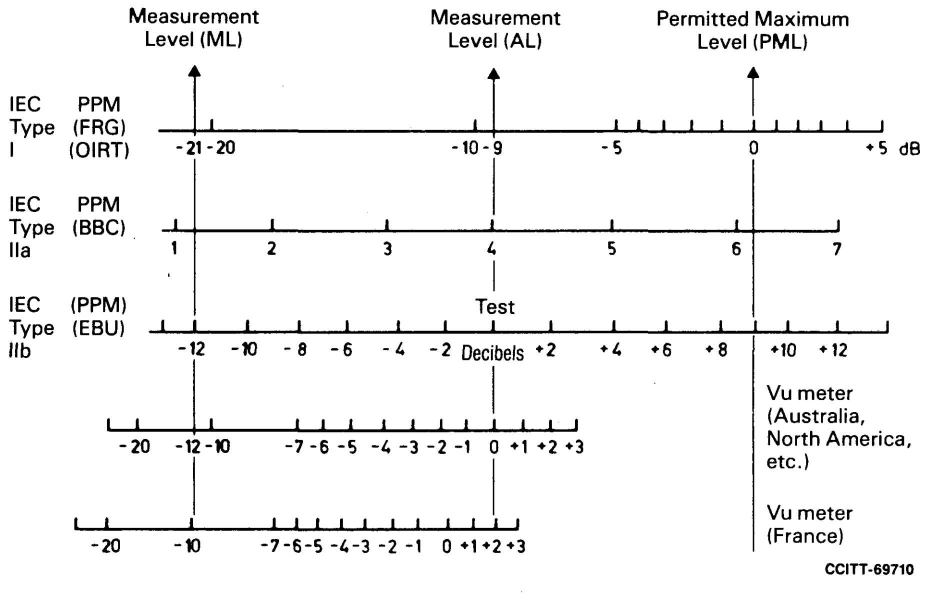
2) In some organizations, +8 dBm0s.



Note – Meter indications are schematic – not to scale.

FIGURE B-1/J.15

Indications produced by various types of Programme Meter with the Three-Level Tone Test Signal

Recommendation J.16

MEASUREMENT OF WEIGHTED NOISE IN SOUND-PROGRAMME CIRCUITS

(Geneva, 1972; amended at Geneva, 1976 and 1980)

The noise objectives for sound-programme circuits are defined in terms of psophometrically weighted noise power levels at a zero relative level point. Psophometric weighting is used to ensure that the objectives and the results of measurements are directly related to the disturbing effect of the noise on the human ear. The psophometric weighting for sound-programme circuits consists of two operations:

– a frequency-dependent weighting of the noise signal, and

– a weighting of the time function of the noise signal to take account of the disturbing effect of noise peaks.

To achieve results which are comparable, it is recommended that for the measurement of noise in sound-programme circuits a measuring set should be used which conforms to the characteristics laid down in CCIR Recommendation 468-3, which is reproduced at the end of this Recommendation.

Annex A gives symbols and definitions used in noise measurements.

ANNEX A

(to Recommendation J.16)

Symbols and definitions used in noise measurements

A clear distinction should be made between measurements performed with equipment conforming to the Recommendation cited in [1] and those with equipment conforming to CCIR Recommendation 468-3.

It is recommended that the definitions and symbols in Table A-1/J.16 be used.

TABLE A-1/J.16

Definitions and symbols for the specification of noise measured on sound-programme circuits

Definitions	Symbols
Unweighted noise level, measured with a quasi-peak measuring instrument complying with CCIR Recommendation 468-3 and referred to a point of zero relative sound-programme level	dBq0s
Weighted noise level, measured with a quasi-peak measuring instrument complying with CCIR Recommendation 468-3 and referred to a point of zero relative sound-programme level	dBq0ps

CCIR RECOMMENDATION 468-3 [1)]

MEASUREMENT OF AUDIO-FREQUENCY NOISE VOLTAGE LEVEL IN SOUND BROADCASTING

(Question 50/10)

(1970 - 1974 - 1978 – 1982)

The CCIR,

CONSIDERING

(a) that it is desirable to standardize the methods of measurement of audio-frequency noise in broadcasting, in sound-recording systems and on sound-programme circuits;

(b) that such measurements of noise should provide satisfactory agreement with subjective assessments,

UNANIMOUSLY RECOMMENDS

that the noise voltage level be measured in a quasi-peak and weighted manner, using the measurement system defined below:

1. Weighting network

The nominal response curve of the weighting network is given in Fig. 1 b which is the theoretical response of the passive network shown in Fig. 1 a. Table I gives the values of this response at various frequencies.

1) This Recommendation is also of interest to the CMTT.

The permissible differences between this nominal curve and the response curve of the measuring equipment, comprising the amplifier and the network, are shown in the last column of Table I and in Fig. 2.

Note 1. – When a weighting filter conforming to § 1 is used to measure audio-frequency noise, the measuring device should be a quasi-peak meter conforming to § 2. Indeed, the use of any other meter (e.g. an r.m.s. meter) for such a measurement would lead to figures for the signal-to-noise ratio that are not directly comparable with those obtained by using the characteristics that are described in the present Recommendation.

Note 2. – The whole instrument is calibrated at 1 kHz (see § 2.6).

TABLE I

Frequency (Hz)	Response (dB)	Proposed tolerance (dB)
31.5	−29.9	±2.0
63	−23.9	±1.4 (1)
100	−19.8	±1.0
200	−13.8	±0.8 (1)
400	− 7.8	±0.5 (1)
800	− 1.9	±0.3 (1)
1 000	0	±0.2
2 000	+ 5.6	±0.5
3 150	+ 9.0	±0.5 (1)
4 000	+10.5	±0.5 (1)
5 000	+11.7	±0.5
6 300	+12.2	0
7 100	+12.0	±0.2 (1)
8 000	+11.4	±0.4 (1)
9 000	+10.1	±0.6 (1)
10 000	+ 8.1	±0.8 (1)
12 500	0	±1.2 (1)
14 000	− 5.3	±1.4 (1)
16 000	−11.7	±1.6 (1)
20 000	−22.2	±2.0
31 500	−42.7	+2.8 (1) / −∞

(1) This tolerance is obtained by a linear interpolation on a logarithmic graph on the basis of values specified for the frequencies used to define the mask, i.e., 31.5, 100, 1000, 5000, 6300 and 20 000 Hz.

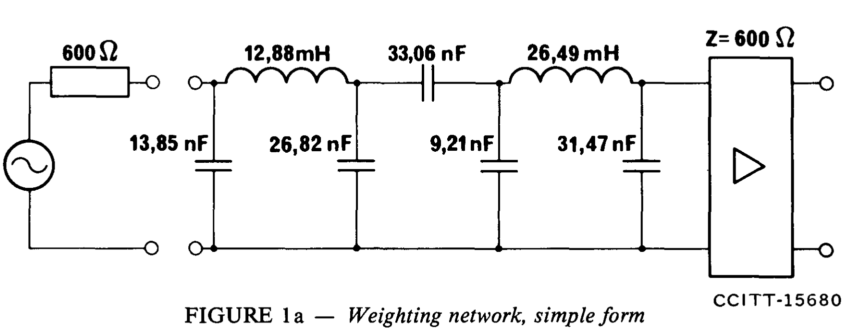


FIGURE 1a — *Weighting network, simple form*

(A constant-resistance realization is described in Annex I)

(A tolerance of at most 1% on the component values and a *Q*-factor of at least 200 at 10 000 Hz are sufficient to meet the tolerances given in Table I.)
(The difference between the responses at 1000 Hz and 6300 Hz may be adjusted more precisely by a small adjustment of the 33.06 nF capacitor.)

2. Characteristics of the measuring device

A quasi-peak value method of measurement shall be used. The required dynamic performance of the measuring set may be realized in a variety of ways (see Note). It is defined in the following sections. Tests of the measuring equipment, except those for § 2.4, should be made through the weighting network.

Note. – After full wave rectification of the input signal, a possible arrangement would consist of two peak rectifier circuits of different time constants connected in tandem [CCIR, 1974-78].

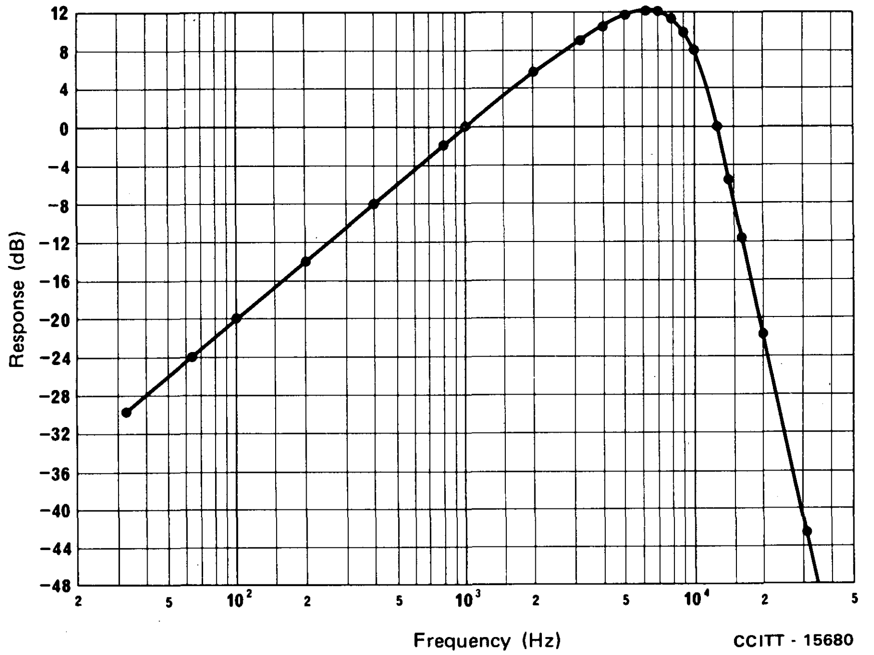


FIGURE 1*b* – *Frequency response of the weighting network shown in Fig. 1 a*

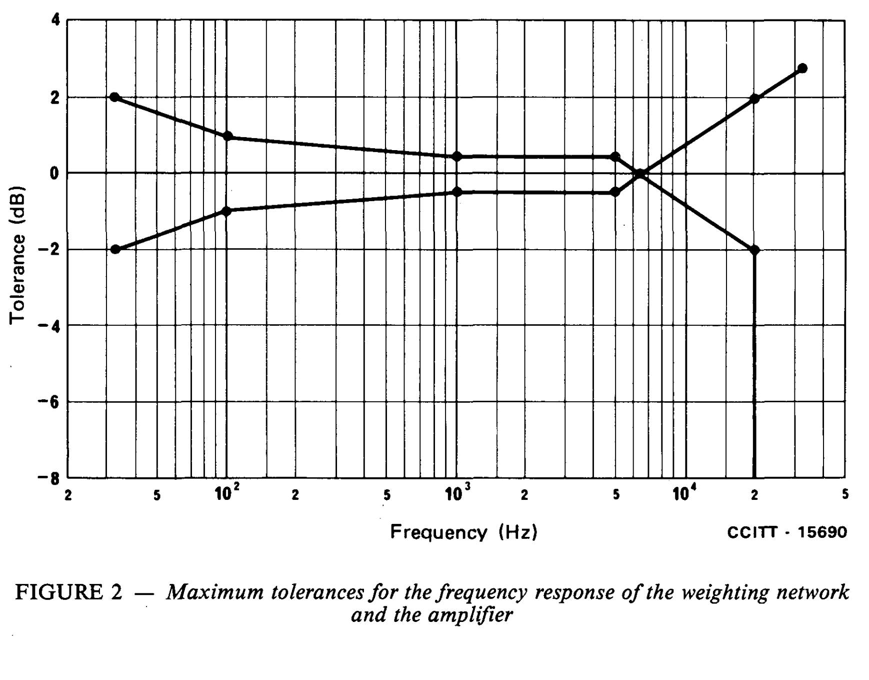


FIGURE 2 — *Maximum tolerances for the frequency response of the weighting network and the amplifier*

2.1 *Dynamic characteristic in response to single tone-bursts*

Method of measurement

Single bursts of 5 kHz tone are applied to the input at an amplitude such that the steady signal would give a reading of 80% of full scale. The burst should start at the zero-crossing of the 5 kHz tone and should consist of an integral number of full periods. The limits of reading corresponding to each duration of tone burst are given in Table II.

The tests should be performed both without adjustment of the attenuators, the readings being observed directly from the instrument scale, and also with the attenuators adjusted for each burst duration to maintain the reading as nearly constant at 80% of full scale as the attenuator steps will permit.

TABLE II

Burst duration (ms)	1(1)	2	5	10	20	50	100	200
Amplitude reference steady signal reading (%)	17.0	26.6	40	48	52	59	68	80
(dB)	−15.4	−11.5	−8.0	−6.4	−5.7	−4.6	−3.3	−1.9
Limiting values:								
– lower limit (%)	13.5	22.4	34	41	44	50	58	68
(dB)	−17.4	−13.0	−9.3	−7.7	−7.1	−6.0	−4.7	−3.3
– upper limit (%)	21.4	31.6	46	55	60	68	78	92
(dB)	−13.4	−10.0	−6.6	−5.2	−4.4	−3.3	−2.2	−0.7

(1) The Administration of the U.S.S.R. intends to use burst durations ≥5 ms.

2.2 *Dynamic characteristic in response to repetitive tone-bursts*

Method of measurement

A series of 5 ms bursts of 5 kHz tone starting at zero-crossing is applied to the input at an amplitude such that the steady signal would give a reading of 80% of full scale. The limits of the reading corresponding to each repetition frequency are given in Table III.

The tests should be performed without adjustment of the attenuators but the characteristic should be within tolerance on all ranges.

TABLE III

Number of bursts per second	2	10	100
Amplitude reference steady signal reading (%)	48	77	97
(dB)	−6.4	−2.3	−0.25
Limiting values:			
– lower limit (%)	43	72	94
(dB)	−7.3	−2.9	−0.5
– upper limit (%)	53	82	100
(dB)	−5.5	−1.7	−0.0

2.3 *Overload characteristics*

The overload capacity of the measuring set should be more than 20 dB with respect to the maximum indication of the scale at all settings of the attenuators. The term "overload capacity" refers both to absence of clipping in linear stages and to retention of the law of any logarithmic or similar stage which may be incorporated.

Method of measurement

Isolated 5 kHz tone-bursts of 0.6 ms duration starting at zero-crossing are applied to the input at an amplitude giving full scale reading using the most sensitive range of the instrument. The amplitude of the tone-bursts is decreased in steps by a total of 20 dB while the readings are observed to check that they decrease by corresponding steps within an overall tolerance of ± 1 dB. The test is repeated for each range.

2.4 *Reversibility error*

The difference in reading when the polarity of an asymmetrical signal is reversed shall not be greater than 0.5 dB.

Method of measurement

1 ms rectangular d.c. pulses with a pulse repetition rate of 100 pulses per second or less are applied to the input in the unweighted mode, at an amplitude giving an indication of 80% of full scale. The polarity of the input signal is reversed and the difference in indication is noted.

2.5 *Overswing*

The reading device shall be free from excessive overswing.

Method of measurement

1 kHz tone is applied to the input at an amplitude giving a steady reading of 0.775 V or 0 dB (see § 2.6). When this signal is suddenly applied there shall be less than 0.3 dB momentary excess reading.

2.6 *Calibration*

The instrument shall be calibrated such that a steady input signal of 1 kHz sine-wave at 0.775 V r.m.s., having less than 1% total harmonic distortion, shall give a reading of 0.775 V, 0 dB. The scale should have a calibrated range of at least 20 dB with the indication corresponding to 0.775 V (or 0 dB) between 2 and 10 dB below full scale.

2.7 *Input impedance*

The instrument should have an input impedance $\geqslant 20$ kΩ and if an input termination is provided then this should be 600 Ω ± 1%.

3. Presentation of results

Noise voltage levels measured according to this Recommendation are expressed in units of dBqps.

Note 1 – If, for technical reasons, it is desirable to measure unweighted noise, the method described in Annex II should be used.

Note 2 – The influence of the weighting network on readings obtained with different spectra of random noise is discussed in Report 496.

REFERENCE

CCIR Document

[1974-78]: 10/28 (United Kingdom).

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ANNEX I

CONSTANT REGISTANCE REALIZATION OF WEIGHTING NETWORK

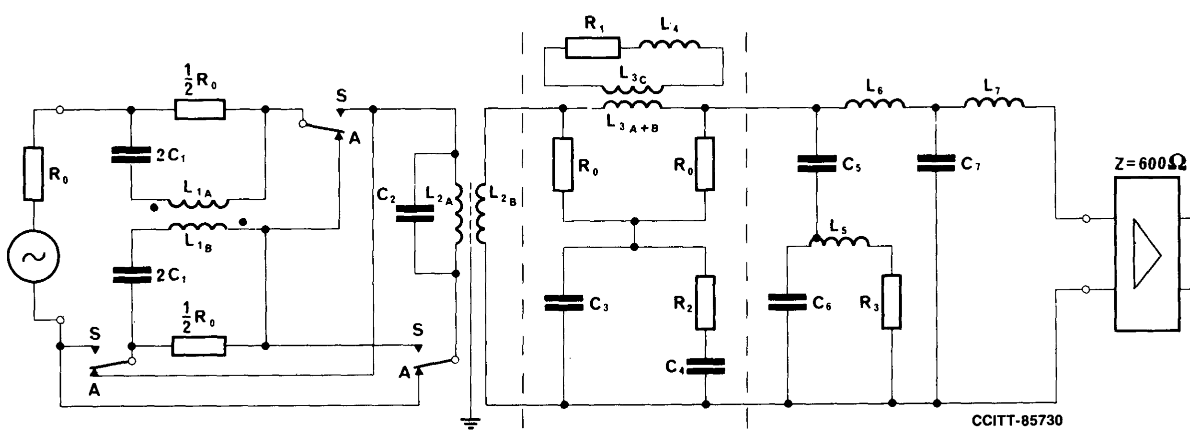

FIGURE 3

Constant resistance realization of weighting network

R (Ω)	C (nF)
R_0: 600	$2C_1$: 83.7
½ R_0: 300	C_2: 35.28
R_1: 912	C_3: 38.4
R_2: 3340	C_4: 7.99
R_3: 941	C_5: 23.8
	C_6: 13.94
	C_7: 35.4

	L (mH)
L_1:	12.70 (for both windings in series)
L_2:	15.06 (for each of two windings separated by electrostatic shield)
L_{3A+B}:	16.73 (two equal windings in series)
L_{3C}:	4.18 (one winding, turns half L_{3A+B}, can have large d.c. resistance, absorbed in R_3)
L_4:	20.1 (can have large d.c. resistance, absorbed in R_3)
L_5:	31.5 (with tap 20.1 at 0.798 of total turns)
L_6:	13.29
L_7:	8.00

A: unbalanced

S: balanced

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AUSTRALIAN BROADCASTING COMMISSION Engineering Development Report No. 106 – Constant resistance realization of CCIR noise weighting network, Recommendation 468.

ANNEX II

UNWEIGHTED MEASUREMENT

It is recognized that unweighted measurements outside the scope of this Recommendation may be required for specific purposes. A standard response for unweighted measurements is included here for guidance.

Frequency response

The frequency response shall be within the limits given in Fig. 4.

This response serves to standardize the measurement and ensure consistent readings of noise distributed across the useful spectrum. When out-of-band signals, e.g. carrier leaks, are present at a sufficient amplitude, they may produce readings that are inconsistent between measuring equipments whose responses are different but still fall within the tolerance template of Fig. 4.

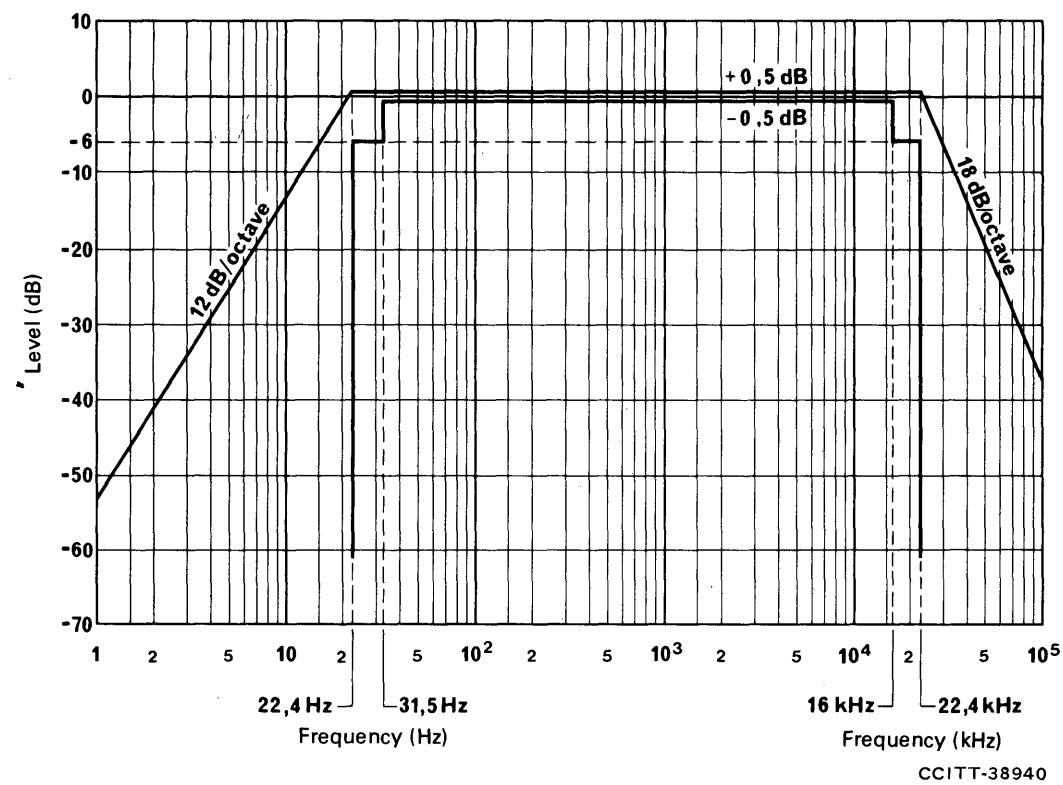

FIGURE 4

BIBLIOGRAPHY

CCIR Document

[1978-82]: 10/76 (CMTT/14) (Canada).

PRE-EMPHASIS USED ON SOUND-PROGRAMME CIRCUITS

(Geneva, 1972)

The noise spectrum in group links is usually uniformly distributed, i.e. all parts of the frequency band are equally disturbed by the noise signal. Sound-programme signals, on the other hand, are not of uniform distribution. The mean power density of the signals tends to decrease towards higher frequencies. Furthermore, the sensitivity of the receiving part (consisting of the radio receiver, the loudspeaker and the human ear) in respect of noise is very dependent on the frequency. (This can be seen from the psophometric weighting curve which is a measure of the sensitivity of the complete receiving part.)

Taking these three facts together it appears to be advantageous to use pre-emphasis on sound-programme circuits set up on carrier systems.

The advantages which could be gained by using different pre-emphasis curves are rather small. It is recommended, therefore, that a single pre-emphasis curve should be used whenever pre-emphasis is applied to sound-programme circuits in group links.

It is further recommended that the pre-emphasis attenuation curve should be that given by the following formula:

$$\text{Insertion loss between nominal impedances} = 10 \log_{10} \frac{75 + \left(\frac{\omega}{3000}\right)^2}{1 + \left(\frac{\omega}{3000}\right)^2} \text{ (dB)}$$

where ω is the angular frequency corresponding to frequency f. Some values are given in Table 1/J.17.

TABLE 1/J.17

f(kHz)	Insertion loss (dB)
0	18.75
0.05	18.70
0.2	18.06
0.4	16.48
0.8	13.10
2	6.98
4	3.10
6.4	1.49
8	1.01
10	0.68
∞	0

The de-emphasis network should have a complementary curve.

The pre-emphasis curve calculated from this formula passes through the following points:

The measured pre-emphasis and de-emphasis curves should not depart by more than ± 0.25 dB from the theoretical curves when the measured levels at 800 Hz are made to coincide with the theoretical levels.

Note – The formula given above defines only the "insertion-loss/frequency" characteristic. The level at which the modulated programme signal is different for the various types of sound-programme equipments and it depends on the modulation method and the type of compandors used. This information is given in the appropriate Recommendations (J.31, J.34, J.41).

Recommendation J.18

CROSSTALK IN SOUND-PROGRAMME CIRCUITS SET UP ON CARRIER SYSTEMS

(Geneva, 1972; amended at Geneva, 1980)

This Recommendation outlines the principles followed by the CCITT in determining what limits are appropriately set for sources of crosstalk affecting sound-programme circuits and other principles which Administrations might apply to ensure that the objectives for intelligible crosstalk in sound-programme circuits are achieved in practice.

1 The causes of crosstalk arising in the transmission parts of telecommunications networks occur in:

a) frequency translating equipments at all levels, viz. audio, group, supergroup, and higher order translating equipments;

b) group, supergroup, etc., through-connection equipments (i.e. filter characteristics);

c) transmission systems, both the line (including repeater) and station equipments.

Different crosstalk mechanisms, e.g. inductive, capacitive and other couplings, intermodulation involving continuous fixed-frequency tones such as pilots, etc., operate in these equipments and systems. A particular channel may thus be disturbed by intelligible crosstalk from a number of potential disturbing sources.

However, because of the interconnections which occur at distribution points along the length of a sound-programme circuit, the same disturbing and disturbed signals are rarely involved in more than one exposure.

2 Only the more important crosstalk mechanisms are the subject of Recommendations (e.g. coaxial and balanced pair cable repeater section FEXT limits of the Series J Recommendations, Section 3); the limits are such that at least the objectives for intelligible crosstalk ratio between *telephone* circuits (generally 65 dB, Recommendation G.151 [1]) may be met. In some cases it is practicable to take into account the more stringent objectives for *sound-programme* circuits (Recommendations J.21, J.22 and J.23). Certain crosstalk mechanisms, because they are not significant for telephony (e.g. near-end crosstalk limits for cable repeater sections), are not the subject of Recommendations; nevertheless, they may be significant in relation to sound-programme circuit objectives.

In principle, a probability of exposure can be attributed to each source of crosstalk, not all potential sources exerting their influence in every case. Given the respective probabilities and distributions, the risk of encountering low values of crosstalk attenuation could be calculated.

Without carrying out this analysis it is estimated that the risk of encountering adverse systematic addition for some sources is small and the allocation of the complete overall objective to a single source of crosstalk as the minimum value of crosstalk attenuation appears justifiable. For other sources, particularly where the equipments involved are specifically intended for sound-programme transmission, it is appropriate to require some higher minimum attenuation values so as to allow for some adverse addition (Recommendation G.242 [2] specifying through-connection filter discrimination requirements against out-of-band components in the band occupied by sound-programme circuits is an example).

3 For these reasons meeting intelligible crosstalk objectives on sound-programme circuits in practice depends on:

a) reasonable care in the allocation of plant for sound-programme circuits, so that the principal crosstalk mechanisms, a single exposure to any of which may itself suffice to exceed the objective, are avoided.

   Among these mechanisms are:

   – far-end and near-end crosstalk at certain frequency bands in line-repeater sections (e.g. the lowest and highest frequency bands of coaxial systems);

   – systematic addition of near-end crosstalk between go and return channels of a group link;

b) readiness to change allocated plant in the few cases where crosstalk is excessive because of systematic addition of two or more disturbing sources.

4 The CCITT limits agreed for crosstalk ratios between bands potentially occupied by sound-programme circuits are in terms of effects at single frequencies. The following factors need to be taken into account when assessing from such limits the probability of encountering intelligible crosstalk into real sound-programme circuits:

a) no methods of assessing the subjective effects of intelligible crosstalk in the bands occupied by sound-programme circuits have as yet been standardized;

b) the intelligibility of crosstalk can be affected by:

   – the use of emphasis in the disturbed circuit;

   – noise masking effects;

   – modulation arrangements (e.g. double sideband) in the disturbed circuit;

   – frequency offsets and inversions;

   – the use of compandors;

c) the mechanisms most liable to cause excessive intelligible crosstalk are, in general, highly frequency-dependent. These cases are those readily prevented by selective plant allocation advocated in § 3 above;

d) crosstalk attenuation can, as a rule, be characterized by a mean value and a standard deviation; the mean value is usually several decibels higher than the worst value, which occurs with only a very small probability.

5 Go-return crosstalk

The assumptions made in the course of the CCITT study of go-return crosstalk in sound-programme circuits, and which served as the basis for the crosstalk limits prescribed in respect of group and higher-order translation equipments (Recommendation G.233 [3]) are given in the following:

a) the nominal maximum distance of the exposure to go-return crosstalk of two sound-programme circuits occupying opposite directions of the same group link is 560 km, i.e. 2/9 of the hypothetical reference circuit distance;

b) the equipments assumed to contribute to such go-return crosstalk are:

   – 560 km of line;

   – one pair of channel translations;

   – one pair of group translations;

   – three pairs of higher-order translations;

   – two through connections.

The corresponding calculation is given in the Annex.

It was considered that the contribution of the line to go-return crosstalk can be limited to the range of values indicated in the Annex, given that precautions outlined in § 3 above are exercised.

It is possible that, in the study of new transmission systems, the CCITT will be able to take such account of sound-programme circuit crosstalk objectives that these precautions may be relaxed somewhat. This study is in progress in the CCITT with respect to 60 MHz systems.

ANNEX A

(to Recommendation J.18)

Calculations of overall go-return crosstalk between two sound-programme circuits occupying opposite directions of the same group link

Equipment	Crosstalk ratio limit (dB)	Crosstalk power per exposure in the disturbed circuit arising from a signal of 0 dBm0 on the disturbing circuit (pW)	Number of exposures	Total crosstalk power (pW)	Crosstalk ratio (dB)
Line	80 to 85 (single homogeneous section)	10 to 3	2 (2/9 h.r.c)	20 to 6	77 to 82
Channel translation	85	3	2	6	82
Group translation	80	10	2	20	77
Supergroup and higher translations	85	3	6	18	77.5
Through filters (cabling)	85	3	2	6	82
Totals (without compandors)				70 to 56	71.5 to 72.5
Totals (with programme-circuit compandors with a minimum companding advantage of 10 dB)				7 to 6	81.5 to 82.5

References

[1] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151.

[2] CCITT Recommendation *Through-connection of groups, supergroups, etc.*, Vol. III, Rec. G.242.

[3] CCITT Recommendation *Recommendations concerning translating equipments*, Vol. III, Rec. G.233.

Recommendation J.19 [1)]

A CONVENTIONAL TEST SIGNAL SIMULATING SOUND-PROGRAMME SIGNALS FOR MEASURING INTERFERENCE IN OTHER CHANNELS [2)]

(Geneva, 1980)

The CCITT,

considering

(a) that on FDM systems non-linear crosstalk may cause mutual interference between the several types of transmission channels;

(b) that the interference depends on the total loading of the FDM system;

(c) that the interference in a channel can be measured as a noticeable deterioration of the signal-to-noise ratio;

(d) that for setting realistic performance limits of interference, a conventional test signal imitating the sound-programme channel loading is desirable,

unanimously recommends

that for simulating sound-programme signals a conventional test signal with the following parameters should be used:

(1) a uniform spectrum energizing signal covering the frequency band up to at least 15 kHz shall be shaped according to the nominal insertion loss/frequency shown in Table 1/J.19 and Figure 1/J.19;

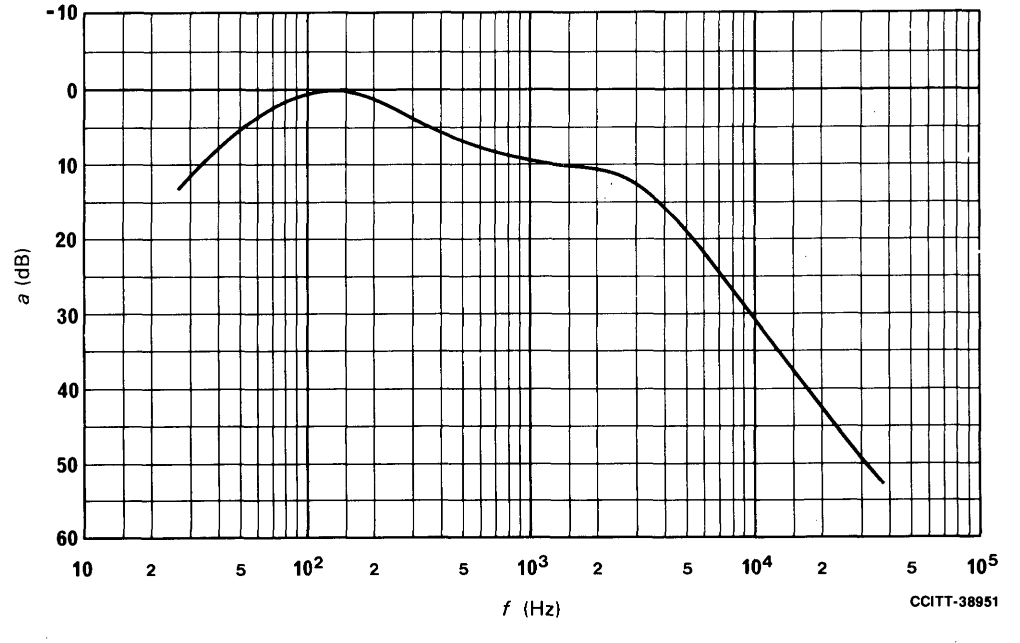

FIGURE 1/J.19

Insertion local frequency

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1) This Recommendation corresponds to CCIR Recommendation 571-1.

2) For the definitions of absolute power, relative power and noise levels, see CCIR Recommendation 574-1.

(2) the conventional test signal can be produced from a Gaussian white noise generator associated with a shaping network conforming with Figure 2/J.19;

(3) the total test signal power applied to a sound-programme circuit under test shall be cyclically changed in level according to Table 2/J.19.

Note – This Recommendation is derived from studies given in Report 497-3.

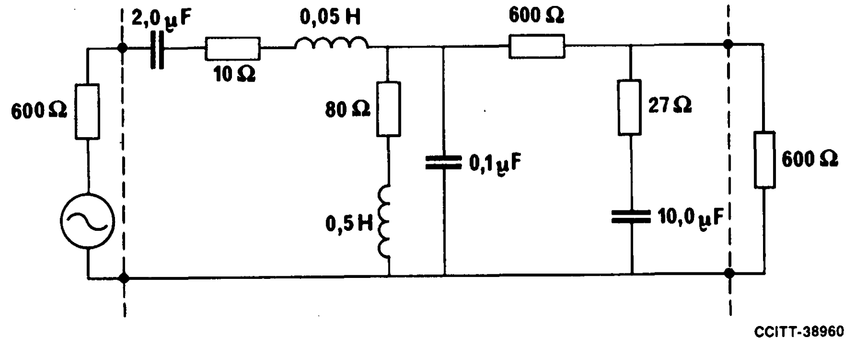

FIGURE 2/J.19

TABLE 1/J.19

Frequency (Hz)	Relative insertion-loss (dB)	Tolerance (±dB)
31.5	10.9	0.5
63	3.4	0.3
100	0.4	0.2
(122)	(0.0)	(0)
200	1.5	0.2
400	5.7	0.3
800	8.7	0.3
1 000	9.2	0.3
2 000	10.6	0.5
3 150	13.0	0.5
4 000	15.7	0.5
5 000	18.8	0.5
6 300	22.5	0.5
7 100	24.6	0.5
8 000	26.6	0.5
9 000	28.6	0.5
10 000	30.4	1.0
12 500	34.3	1.0
14 000	36.3	1.0
16 000	38.6	1.0
20 000	42.5	1.0
31 500	50.4	1.0

TABLE 2/J.19

Step	Level	Time for which signal is applied
1	−4 dBm0s	4 s
2	+3 dBm0s	2 s
3	no signal	2 s

ANNEX A

(to Recommendation J.19)

Study Group XV of the CCITT had put some questions as regards Recommendation 571 and the CMTT has worked out their answers. As those questions and the answers may be helpful for anyone who applies the conventional test signal for carrying out measurements of any kind, they are given below:

Question:

(a) For the measurement of crosstalk from a sound-programme circuit to a telephone circuit, could the signal described in CCIR Recommendation 571 be used, considering the different bandwidth and possible frequency shift?

Reply:

– The intelligible crosstalk ratio is based on selective measurements in the telephone circuit when the sinusoidal signals are transmitted in the sound-programme circuit within the frequency range of 0.3 to 3.4 kHz. In Recommendation J.21 a minimum ratio of 65 dB is defined.

– The unintelligible crosstalk ratio should be ascertained by measuring the increase of noise in the telephone circuit by loading the disturbing sound-programme with the simulated test signal defined in CCIR Recommendation 571. As for this increase no tolerable values are recommended up to now, the CMTT proposes such values based on a maximum noise contribution produced by interference of −65 dBm0p. Depending on the basic noise level in the telephone circuit the following increased values can be tolerated:

TABLE A-1/J.19

Basic noise level (dBm0p)	−75	−70	−65	−60	−55	−50
Tolerable increase of noise level (dB)	10.4	6.2	3	1.2	0.4	0.1

Question:

(b) What is the equivalent value for 65 dB ratio (given in Recommendations J.21, J.22 and J.23) using sinusoidal tones, when measuring with the recommended new test signal?

Reply:

The answer to this question is included in the proposal for the measurement of the ratio for the total crosstalk caused by intermodulation given in the answer to Question *(a)*.

Question:

(c) Can the signal defined in Table 2/J.19 from the point of view of the mean loading it would impose on transmission systems and in the light of Recommendations N.12 and N.13 be regarded as acceptable for unrestricted use over complete sound-programme circuits of any constitution?

Reply:

The conventional test signal simulating sound-programme signals defined in CCIR Recommendation 571/Recommendation J.19 in all aspects can be regarded as acceptable for unrestricted use over sound-programme circuits of any constitution.

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SECTION 2

PERFORMANCE CHARACTERISTICS OF SOUND-PROGRAMME CIRCUITS

Recommendation J.21 [1)]

PERFORMANCE CHARACTERISTICS OF 15 kHz TYPE SOUND-PROGRAMME CIRCUITS [2)]

Circuits for high-quality monophonic and stereophonic transmissions

(Geneva, 1972; amended at Geneva, 1976 and 1980)

The CCITT,

considering

(a) that it is necessary to set transmission standards for sound-programme circuits;

(b) that quality requirements for the hypothetical reference circuit are established for analogue sound programmes;

(c) that advantage should be taken of the technical evolution made possible by the introduction of digital techniques, particularly for mixed analogue and digital circuits,

recommends

that, with due regard to the application constraints, equipment for new circuits shall meet the requirements laid out in Annex A.

ANNEX A

(to Recommendation J.21)

A.1 *Application*

The Recommendation applies to homogeneous analogue or mixed analogue-and – digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation J.11.

For estimation of the performance of circuits shorter or longer than the HRC, see CCIR Recommendation 605.

1) This Recommendation corresponds to CCIR Recommendation 505-2.

2) For the definition of absolute power, relative power and noise levels, see CCIR Recommendation 574-1.

Note 1 – For all-digital circuits a separate Recommendation might be envisaged after further study.

Note 2 – For further work, CCIR Report 496 may be consulted. This Report also draws attention to certain differences between CCIR and OIRT Recommendations.

A.2 *Interface characteristics*

A.2.1 *Test conditions*

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 Ω resistive.

A.2.2 *Impedance*

System input impedance	600 Ω, balanced[3]
System output impedance, provisionally	Low-Z, balanced

The open-circuit output level shall not decrease more than 0.3 dB within the nominal frequency range, if the output is terminated by the specified test load.

The reactive part of the source impedance must be restricted to 100 Ω max. (provisional value) within the nominal frequency range.

This clause alone would however not rule out a large difference in the reactive parts of the output impedances of a stereophonic pair, and this in turn could lead to difficulties in meeting § A.3.2.2. This aspect needs further study.

A.2.3 *Levels*

Input maximum programme level	+9 dBm0s
Insertion gain (1 kHz at −12 dBm0)	0 dB
Adjustment error	±0.5 dB
Variation over 24 hours not to exceed	±0.5 dB
Relative level (see Recommendation J.14)	+6 dBrs

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organizations to insert additional trimming attenuators.

A.3 *Overall performance*

A.3.1 *Common parameters*

A.3.1.1 *Gain/frequency response*

Reference frequency	1 kHz (nominal value)
The response shall be measured at	−12 dBm0s

Frequency (kHz)	Response (dB)
0.04 ... < 0.125	+0.5 ... −2.0
0.125 ... 10	+0.5 ... −0.5
>10 ... 14	+0.5 ... −2.0
>14 ... 15	+0.5 ... −3.0

[3] The tolerance, permitted reactance and unbalance need further study.

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

A.3.1.2 *Group delay variation*

Difference between the value of group delay at the following frequencies and the minimum value:

kHz	$\Delta\tau$ (ms)
0.04	55
0.075	24
14	8
15	12

Between the points defined above, the tolerance limit varies linearly in a linear-delay/logarithmic-frequency diagram.

A.3.1.3 *Noise*

The measurement to be made with an instrument conforming to CCIR Recommendation 468.

For radio-relay systems the requirements shall be met for at least 80% of the total time of any 30-day period. For 1% of the time a 4 dB worse, and for 0.1% of the time a 12 dB worse value is acceptable.

Idle channel noise, max. −42 dBq0ps
Programme-modulated noise, max. −30 dBq0ps

Programme-modulated noise can only occur on sound-programme circuits which are equipped with compandors (for example types of circuits corresponding to Recommendation J.31).

This noise value may be measured with the aid of an auxiliary sinusoidal test signal +9 dBm0s/60 Hz which has to be suppressed by a high-pass filter ($f_0 \leqslant$ 400 Hz, $a \geqslant$ 60 dB/60 Hz) before the measuring set.

CCIR Report 493 indicates that if a compandor is used, an improved signal-to-noise ratio is necessary to avoid objectionable effects with some programme material.[4)].

Note – For digital systems appropriate values are under study. For further information see CCIR Report 647-2.

A.3.1.4 *Single tone interference*

Level of any individual tone $\leqslant (-73 - \psi)$ dBm0s

Where ψ is the weighting as per CCIR Recommendation 468 at the particular frequency.

For sound-programme transmissions over carrier systems, occurrence of carrier leaks can be expected. For this reason, stop filters may be provided in the carrier frequency path which can be switched in, if required, to suppress the tones otherwise audible in the upper frequency range from 8 to 15 kHz. For a hypothetical reference circuit, a 3 dB bandwidth of less than 3% for stop filters, referred to the mid-frequency, is recommended. The use of stop filters influencing frequencies below 8 kHz should be avoided.

4) Administrations are urged to supply additional information on an appropriate value.

A.3.1.5 *Disturbing modulation by power supply*

The ratio of the level of a sine-wave test signal applied to the sound-programme circuit to the highest level unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains rectifiers shall be ⩾ 45 dB. The value required with higher frequency a.c. primary sources and when inverters and d.c.-d.c. converters are used, has to be determined. (Study Programme 17F/CMTT; see also CCIR Report 495-1, Vol. XII, Geneva 1974.)

A.3.1.6 *Non-linear distortion*

A.3.1.6.1 *Harmonic distortion*

The total harmonic distortion (THD) shall be measured with the input signal at +9 dBm0s at frequencies below 4 kHz; and at +6 dBm0s for 4 kHz and higher.

The duration for which a single-tone is to be transmitted at these levels should be restricted in accordance with Recommendations N.21 and N.23.

The performance when measured with a true-RMS meter shall not be less than the following requirements:

Input tone (kHz)	THD	Second and third harmonic measured selectively
0.4 ... <0.125	1% (−31 dBm0s)	0.7% (−34 dBm0s)
0.125 ... 7.5	0.5% (−37 dBm0s)	0.35% (−40 dBm0s)

A.3.1.6.2 *Intermodulation*

With input signals at 0.8 kHz and 1.42 kHz, each at +3 dBm0s, the third order difference tone at 0.18 kHz shall be less than the following values:

0.5% (−43 dBm0s)

Note – Attention is drawn to the fact in transmission systems using compandors, a 3rd order difference-tone may occur which exceeds the specified limit of 0.5%. This may occur when the difference between the two fundamental frequencies is less than 200 Hz. Thus, the components due to 3rd order distortion will have frequencies which correspond to the difference between the two test frequencies. However, in these cases the subjective masking is such that a distortion up to 2% is acceptable.

For 15 kHz systems intended for baseband transmissions on physical circuits only, and on modulation equipment in local loops, the following additional requirements apply:

Input signals at +3 dBm0s each	Maximum difference-tone level at 1.6 kHz
5.6 kHz and 7.2 kHz	0.5% (−43 dBm0s) (second order)
4.2 kHz and 6.8 kHz	0.5% (−43 dBm0s) (third order)

A.3.1.6.3 *Distortion products measured by shaped noise: under study. See CCIR Report 640.*

A.3.1.7 *Error in reconstituted frequency* (applies only to FDM systems)

Not to be greater than 1 Hz.

Note – A maximum error of 1 Hz is in principle acceptable where there is only a single transmission path between the signal source and the listener.

Where the broadcast network can involve two or more parallel paths, e.g., commentary and separate sound channels, or radio broadcast from different transmitters on the same frequency, unacceptable beats may occur unless zero error can be assured. This is under study.

A.3.1.8 *Intelligible cross-talk ratio*

A.3.1.8.1 The intelligible near-end and far-end cross-talk ratios between sound-programme circuits, or from a telephone circuit (disturbing) into a sound-programme circuit (disturbed) shall be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit, and shall not be less than the following values:

Frequency (kHz)	Crosstalk attenuation (dB)
0.04	50
0.04 ... 0.05	oblique straight-line segment on linear-decibel and logarithmic-frequency scales
0.05 ... 5	74
5 ... 15	oblique straight-line segment on linear-decibel and logarithmic-frequency scales
15	60

A.3.1.8.2 The near-end and far-end cross-talk attenuations between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) shall be at least 65 dB.

Note 1 – It is understood that this value is defined between the relative levels applicable to telephone circuits. An explanation for the relation between the relative levels for sound-programme circuits and telephone circuits is given in Recommendation J.22, Annex B.

Note 2 – The attention of Administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or in the low frequency range (e.g. below about 100 kHz) in certain carrier systems on coaxial cables. If sub-standard is to be avoided, such systems or parts of systems must not be used for setting up programme channels.

Note 3 – When 4000 pW0p or more noise is continuously present in the telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk ratio of 58 dB between a sound-programme circuit and a telephone circuit is acceptable.

Note 4 – The attention of Administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk limits between two sound-programme circuits, simultaneously occupying the go and return channels respectively, of a carrier system (the most economical arrangement) because in those circumstances they occupy the same position in the line-frequency band (see Recommendation J.18).

Note 5 – The value indicated is based on the assumption that sine wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6 – The effect of cross-talk from a sound-programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound-programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

A.3.1.9 *Amplitude linearity*

When a 1 kHz input signal is stepped from −6 dBm0s to +6 dBm0s, or vice versa, the output level shall change accordingly by 12 ± 0.5 dB.

A.3.2 *Additional parameters for stereophonic programme transmission*

A.3.2.1 The difference in gain between A and B channels shall not exceed the following values:

Frequency (kHz)	Gain difference (dB)
0.04 ... < 0.125	1.5
0.125 ... 10	0.8
> 10 ... 14	1.5
> 14 ... 15	3.0

A.3.2.2 The phase difference between the A and B channels shall not exceed the following values:

Frequency (kHz)	Phase difference (degrees)
0.04	30
0.04 ... 0.2	oblique straight-line segment on linear-degree and logarithmic-frequency scales
0.2 ... 4	15
4 ... 14	oblique straight-line segment on linear-degree and logarithmic-frequency scales
14	30
14 ... 15	oblique straight-line segment on linear-degree and logarithmic-frequency scale
15	40

A.3.2.3 The cross-talk ratio between the A and B channels shall not be less than the following limits:

A.3.2.3.1 Intelligible cross-talk ratio, measured with sinusoidal test signal 0.04 to 15 kHz: 50 dB.

A.3.2.3.2 Total cross-talk ratio predominantly caused by intermodulation: 60 dB.

This value is ascertained by loading one of the two channels with the sound-programme simulating signal defined in CCIR Recommendation 571. In the other channel, the noise contribution due to intermodulation shall not be higher than −51 dBq0ps.

This leads to an increase of noise depending on the idle channel noise value. The tolerable increase is given in the table below.

Idle channel noise (dBq0ps)	−60	−57	−54	−51	−48	−45	−42
Tolerable increase of noise (dB)	9,5	7	4,8	3	1,8	1,0	0,5

A.3.3 *Additional requirements for digital systems*

A.3.3.1 If a test signal is harmonically related to the sampling frequency measuring difficulties may arise. In that case the nominally 1 kHz test signal must be slightly adjusted. No exact preferred frequency can be stated at present, but 1004 Hz and 1020 Hz are examples of frequencies certain Administrations use.

A.3.3.2 *Unbalance of the limitation level*

The difference between those levels which lead to a limitation of the positive or negative half-wave of the test signal shall not exceed 1 dB.

A.3.3.3 *Intermodulation with the sampling signal*

Intermodulation products (f_d) caused by non-linearities may occur in the sound-channel when the sampling signal (f_o) is combined with the inband audio signals (f_i) or out-of-band interfering signals (f_a).

A.3.3.3.1 *Inband intermodulation*

The following combination rule applies: $f_d = f_o - nf_i$.

Only values with $n = 2$ or 3 are of importance.

The level difference between a 0 dBm0s signal (f_i) and the intermodulation products (f_d) shall not be less than 40 dB.

A restriction to the following f_i/f_d values is sufficient:

	$n = 2$		$n = 3$		
f_i	9	13	7	11	kHz
f_d	14	6	11	1	kHz

A.3.3.3.2 *Out-of-band intermodulation*

The following combination rule applies: $f_d = nf_o \pm f_a$.

Only values with $n = 1$ or 2 are of importance.

The level difference between a 0 dBm0s signal (f_a) and the intermodulation products (f_d) shall not be less than 60 dB.

A restriction to the following f_a/f_d values is sufficient:

	$n = 1$		$n = 2$		
f_a	31	33	63	65	kHz
f_d	1				kHz

A.3.3.4 *Further parameters*

Characteristics for bit errors, clicks, jitter, etc. are under study. (See Study Programme 18A/CMTT and CCIR Report 647-2.)

Note – The CCIR has issued Recommendation 572 which deals with the transmission of one sound-programme associated with an analogue television signal by means of time-division multiplex in the line synchronizing pulse. The system recommended is a digital one, using pulse code modulation. A sound-programme bandwidth of 14 kHz is provided.

Biblography

CCIR Document (1978-1982): CMTT/68 (OIRT).

Recommendation J.22[1)]

PERFORMANCE CHARACTERISTICS OF 10 kHz TYPE SOUND-PROGRAMME CIRCUITS[2)]

(former Recommendation J.21; amended at Geneva, 1972, 1976 and 1980)

The CCITT,

considering

(a) that 10 kHz-type circuits are not quite adequate for modern high-quality transmission;

(b) that such circuits are more than adequate for medium-quality transmission;

(c) that they have been extensively used in the past;

(d) that some Administrations may for various reasons still wish to continue to provide circuits of this type;

(e) the necessity to reduce the number of different standards in the interest of not forcing the industry to diversify production and other efforts unnecessarily;

(f) that such reduction as in (e) would be expected to assist in keeping system costs low,

concludes

that the 10 kHz-type circuit is non-preferred for new systems in general, but may still be justified in certain cases (for example extension of already implemented systems),

recommends

(1) that whenever possible, 7 kHz-type or 15 kHz-type sound-programme circuits be considered for new systems rather than the 10 kHz-type;

(2) however, when it is necessary to choose the 10 kHz-type for new systems, the circuit shall meet requirements set forth in Annexes A and B of this Recommendation.

ANNEX A

(to Recommendation J.22)

Performance characteristics of 10 kHz-type sound-programme circuits

A.1 *Application*

The requirements below apply to new 10 kHz-type analogue monophonic circuits and shall be met by the hypothetical reference circuit (HRC) defined in Recommendation J.11.

For estimation of the performance of circuits shorter or longer than the HRC see CCIR Recommendation 605.

Note 1 – For real audio-frequency circuits it may not be possible to meet the requirement for intelligible cross-talk. When this occurs, Administrations concerned have to agree on an appropriate relaxed value.

1) This Recommendation corresponds to CCIR Recommendation 504-2. CMTT at its interim meeting agreed that this Recommendation will not be published in the next CCIR book.

2) For the definition of absolute power, relative power and noise levels, see CCIR Recommendation 574-1.

Note 2 – For real circuits involving frequency division multiplex systems (FDM) it may not be possible to meet the requirements for intelligible cross-talk and noise. When this occurs Administrations concerned have to agree on appropriate relaxed values. See Annex B for further information.

A.2 *Interface characteristics*

A.2.1 *Test conditions*

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 ohm resistive.

A.2.2 *Impedance*

System input impedance	600 ohm balanced[3)]
System output impedance, provisionally	Low-Z, balanced

The open-circuited output level shall not decrease more than 0.3 dB within the nominal frequency range, if the output is terminated by the specified test load.

Note – The reactive part of the source impedance must be restricted to 100 ohm maximum (provisional value) within the nominal frequency range.

A.2.3 *Levels*

Input maximum programme level	+9 dBm0s
Insertion gain (1 kHz at −12 dBm0s)	0 dB
Adjustment error	±0.5 dB
Variation during 24 hours not to exceed	±0.5 dB
Relative level (see Recommendation J.14)	+6 dBrs

A.3 *Performance of the HRC*

A.3.1 *Gain/frequency response*

Reference frequency 1 kHz

Frequency (Hz)	Relative level (dB)
50 ... < 100	+1.7 to −4.3
100 ... < 200	+1.7 to −2.6
200 ... < 6 000	±1.7
6 000 ... < 8 000	+1.7 to −2.6
8 000 ... < 10 000	+1.7 to −4.3

3) The tolerance and permitted reactance need further study.

A.3.2 *Group delay variation*

Difference between the value of group delay at the following frequencies and the minimum value:

f (Hz)	$\Delta\tau g$ (ms)
50	$\leqslant 80$
100	$\leqslant 20$
10 000	$\leqslant 8$

Between the points defined above the tolerance limit varies linearly in a linear-delay/logarithmic-frequency diagram.

A.3.3 *Maximum weighted noise level*

The measurement to be made with an instrument conforming to CCIR Recommendation 468-3:

−39 dBq0ps.

Note 1 — The weighting as per Recommendation P.53B (Green Book) is now obsolete, and further use of instruments conforming to that specification is deprecated.

Note 2 — Unweighted measurements are not considered to be meaningful.

Note 3 — For circuits on carrier systems, it is not always possible, in the absence of special precautions, to meet the limits recommended in this paragraph (see Annex B).

A.3.4 *Single tone interference*

Under study. (The subjective assessment of single tone interference on high quality circuits will be carried out by a method described in CCIR Report 623).

A.3.5 *Disturbing modulation by power supplies*

The ratio of the level of a sine wave test signal applied to the sound-programme circuit to the highest level unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains rectifiers shall be $\geqslant$ 45 dB. The value required with higher frequency a.c. primary sources and when inverters and d.c.-d.c. converters are used, has to be determined. (Study Programme 17F/CMTT; see also CCIR Report 495-1, Volume XII, Geneva, 1974.)

A.3.6 *Non-linear distortion*

As measured with fundamental signals at +9 dBm0 the total harmonic distortion (THD) shall be:

f (Hz)	THD
0.05 ... < 0.01	$\leqslant$3% (−21 dBm0)
0.01 ... 10	$\leqslant$2% (−25 dBm0)

Note – Precautions should be taken in the measurement of harmonic distortion on circuits equipped with pre-emphasis networks. (See Recommendation N.21.)

A.3.7 *Accuracy of reconstituted frequency*

The difference between the initial and reconstructed frequencies should not exceed 1 Hz except when the broadcast network is composed of two or more parallel paths, e.g. commentary and separate sound channels, or a programme is broadcast from different transmitters on the same frequency. Any difference between the initial and reconstituted frequencies within such networks can give rise to unacceptable beats on the received programme. Therefore when these circumstances apply it is recommended that sound-programme circuits should introduce no frequency error in the reconstituted frequencies.

A.3.8 *Intelligible cross-talk*

A.3.8.1 The near-end and far-end cross-talk attenuations between two sound-programme circuits, or between a telephone circuit (disturbing circuit) and a sound-programme circuit (disturbed circuit) shall be at least 74 dB.

A.3.8.2 The near-end and far-end cross-talk attenuations between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) shall be at least 65 dB.

Note 1 – It is understood that this value is defined between the relative levels applicable to telephone circuits. An explanation for the relation between the relative levels for sound-programme circuits and telephone circuits is given in Annex B.

Note 2 – The attention of Administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or in the low frequency range (e.g. below about 100 kHz) in certain carrier systems on coaxial cables. If sub-standard performance is to be avoided, such systems or parts of systems must not be used for setting up programme channels.

Note 3 – When 4000 pW0p or more noise is continuously present in a telephone channel (this may be the case in satellite systems, for example) a reduced cross-talk attenuation of 58 dB from the sound-programme circuit into the telephone circuit is acceptable.

Note 4 – The attention of Administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk limits between two sound-programme circuits simultaneously occupying the go and return channels respectively, of a carrier system (the most economical arrangement), because in those circumstances they occupy the same position in the line-frequency band. (See Recommendation J.18.)

Note 5 – The value indicated is based on the assumption that sine wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6 – The effect of cross-talk from a sound-programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound-programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

A.3.9 *Amplitude linearity*

When a 1 kHz input signal is stepped from -6 dBm0 to $+6$ dBm0, the output level shall increase by 12 ± 0.5 dB.

ANNEX B

(to Recommendation J.22)

Values of noise expected in practice on 2500 km circuits

Estimated noise power levels

The following table shows the noise values arising when sound-programme circuits (using pre-emphasis and de-emphasis in accordance with Recommendation J.17) are set up in place of three telephone channels, each of which conforms to the general noise objectives given in Recommendation G.222. The assumptions made for the purpose of the noise calculations are shown at the end of this Annex.

TABLE A-1/J.22

	One-minute mean value	
	for not more than 20% of a month	for not more than 0.1% of a month
Noise power level weighted with the network of CCITT Recommendation P.53 B (Green Book) [a]	—44.5 dBm0ps	—37.5 dBm0ps

[a] This network is now obsolete.

Note – The increased noise level shown as occurring for less than 0.1% of a month, applies when the carrier circuit is established over a radio-relay system.

When 10 kHz-type sound-programme circuits which include emphasis and de-emphasis networks, are set up on a carrier system it is recommended that, for reasons of overload, the relative level at 1000 Hz on such a circuit at a zero-relative level point (deduced from the level diagram of telephone circuits set up on the same 12-circuit group) should lie between a maximum of -1.5 dB and a minimum of -4.5 dB.

The level of -1.5 dB could be considered as normal. A further 3 dB adjustment to permit a decrease down to -4.5 dB should, however, be included to cover the case of exceptional overloading if operational experience shows that in fact this is necessary.

Note – Certain problems connected with the use of pre-emphasis on carrier systems have not yet been satisfactorily resolved. These are:

– the limitation of the level of testing tones, which is of concern to CCITT Study Group IV;

– the effect of pre-emphasis on the harmonic distortion requirements which the programme circuit should meet at high frequencies [4].

Use of compandors

Provided the compressor and the expandor are of the same make, it is possible to obtain overall transmission characteristics as regards noise, which conform to the CCITT Recommendations for the 2500 km hypothetical reference circuit, without introducing other factors that might impair transmission performance. The CCITT is now examining recommendations on the compressor and the expandor, considered separately, so as to achieve the same result.

[4] Measurements of harmonic distortion on programme circuits having pre-emphasis must be treated with reserve. This point is being studied by the CCITT.

Assumptions and conventional terms

The expression dBm0ps is used to indicate noise power levels in a sound-programme circuit which are psophometrically weighted and measured in decibels relative to 1 mW at a point of zero relative level in that circuit. The CCITT practice is to quote noise level for sound-programme circuits relative to "peak programme" or "maximum voltage" which is defined as a voltage of 2.2 V r.m.s. (measured at the terminals of an impedance of 600 ohms) at a point of zero relative level, i.e. 9 dB above telephone test level, and the signal-to-noise ratio objective of 57 dB is thus equivalent to a noise power level of −48 dBm0ps.

The value for not more than 20% of a month was calculated for 10 kHz-type circuits on the following assumptions:

– Noise on one telephone channel (including the multiplex equipment) according to Recommendation G.222, weighted for telephony:	−50 dBm0ps
– Bandwidth correction for 10 kHz:	+ 5 dB
– Suppression of weighting for telephony (in the case of a uniform-spectrum noise):	+ 2.5 dB
– Improvement due to pre-emphasis[5] (see Recommendation J.17):	− 9 dB
– Effect of the relative level shifted by −1.5 dB at 800 Hz:	+ 1.5 dB
– Weighting for sound-programme transmissions:	+ 5.5 dB
Total	−44.5 dBm0ps

The value for not more than 0.1% of a month was calculated on the basis of the noise variations to be expected on a radio-relay link used mainly for providing telephone circuits and conforming with Recommendation G.222.

Recommendation J.23

PERFORMANCE CHARACTERISTICS OF NARROW-BANDWIDTH SOUND-PROGRAMME CIRCUITS[1), 2), 3), 4)]

Circuits of medium quality for monophonic transmission

(amended at Geneva, 1980)

The CCITT,

considering

(a) that it is necessary to set transmission standards for sound-programme circuits;

(b) that quality requirements for the hypothetical reference circuit are established for analogue sound programmes;

(c) that advantage should be taken of the technical evolution made possible by the introduction of digital techniques, particularly for mixed analogue and digital circuits,

recommends

that, with due regard to the application constraints, equipment for new circuits shall meet the requirements laid out in Annex A.

5) Set to have zero loss at 800 Hz.

1) This Recommendation corresponds to CCIR Recommendation 503-2. CMTT has agreed at its interim meeting that CCIR Recommendation 504-2 will not be published in the next CCIR book; consequent changes have been proposed to Recommendation 503-2.

2) For the definition of absolute power, relative power and noise levels, see Recommendation 574-1.

3) Sound-programme circuits of the 5 kHz-type are widely used in North America.

4) 6.4 kHz-type narrow bandwidth sound-programme circuits are still being used in some countries.

ANNEX A

(to Recommendation J.23)

A.1 *Application*

The Recommendation applies to homogeneous analogue or mixed analogue-and-digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation J.11.

For estimation of the performance of circuits shorter or longer than the HRC, see CCIR Recommendation 605.

Note 1 – For all-digital circuits a separate Recommendation might be envisaged after further study.

Note 2 – For further work, CCIR Report 496 may be consulted. This Report also draws attention to certain differences between CCIR and OIRT Recommendations.

A.2 *Interface characteristics*

A.2.1 *Test conditions*

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 Ω resistive.

A.2.2 *Impedance*

System input impedance	600 Ω, balanced[5]
System output impedance, provisionally	Low-*Z*, balanced

The open-circuit output level shall not decrease more than 0.3 dB within the nominal frequency range, if the output is terminated by the specified test load.

The reactive part of the source impedance must be restricted to 100 Ω max. (provisional value) within the nominal frequency range.

A.2.3 *Levels*

Input maximum programme level	+9 dBm0s
Insertion gain (1 kHz at −12 dBm0s)	0 dB
Adjustment error	±0.5 dB
Variation over 24 hours not to exceed	±0.5 dB
Relative level (See Recommendation J.14)	+6 dBrs

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organizations to insert additional trimming attenuators.

A.3 *Overall performance*

A.3.1 *Common parameters*

A.3.1.1 *Gain/frequency response*

Reference frequency	1 kHz (nominal value)
The response shall be measured at	−12 dBm0s

[5] The tolerance and permitted reactance need further study.

5 kHz systems		7 kHz systems	
Frequency (kHz)	Response (dB)	Frequency (kHz)	Response (dB)
0.07 ... <0.2	+1 ... −3	0.05 ... <0.1	+1 ... −3
0.2 ... 4	+1 ... −1	0.1 ... 6.4	+1 ... −1
>4 ... 5	+1 ... −3	>6.4 ... 7	+1 ... −3

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

A.3.1.2 *Group delay variation*

Difference between the value of group delay at the following frequencies and the minimum value:

5 kHz systems		7 kHz sytems	
kHz	$\Delta\tau$ (ms)	kHz	$\Delta\tau$ (ms)
0.07	$\leqslant 60$	0.05	$\leqslant 80$
5	$\leqslant 15$	0.1	$\leqslant 20$
		6.4	$\leqslant 5$
		7	$\leqslant 10$

Between the points defined above, the tolerance limit varies linearly in a linear-delay/logarithmic-frequency diagram.

A.3.1.3 *Noise*

The measurement to be made with an instrument conforming to CCIR Recommendation 468.

For radio-relay systems the requirements shall be met for at least 80% of the total time of any 30-day period. For 1% of the time a 4 dB worse, and for 0.1% of the time a 12 dB worse value is acceptable.

		Analogue system types	
		5 kHz	7 kHz
Idle channel noise, max.	(dBq0ps)	−32	−44
Programme-modulated noise, max.	(dBq0ps)	–	−32

Programme-modulated noise can only occur on sound-programme circuits which are equipped with compandors (for example types of circuits corresponding to Recommendation J.31).

This noise value may be measured with the aid of an auxiliary sinusoidal test signal +9 dBm0s/60 Hz which has to be suppressed by a high-pass filter ($f_0 \leqslant 400$ Hz, $a \geqslant 60$ dB/60 Hz) before the measuring set.

CCIR Report 493 indicates that if a compandor is used, an improved signal-to-noise is necessary to avoid objectionable effects with some programme material.[6)]

Note – For digital systems appropriate values are under study. For further information see CCIR Report 647-2.

A.3.1.4 *Single tone interference*

Level of any individual tone $\leqslant (-73 - \psi)$ dBm0s

Where ψ is the weighting as per CCIR Recommendation 468 at the particular frequency.

For sound-programme transmissions over carrier systems, occurrence of carrier leaks can be expected. For this reason, stop filters may be provided in the carrier frequency path which can be switched in, if required, to suppress the tones otherwise audible in the upper frequency range from 8 to 15 kHz. For a hypothetical reference circuit, a 3 dB bandwidth of less than 3% for stop filters, referred to the mid-frequency, is recommended. The use of stop filters influencing frequencies below 8 kHz should be avoided.

A.3.1.5 *Disturbing modulation by power supply*

The ratio of the level of a sine-wave test signal applied to the sound-programme circuit to the highest level unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains rectifiers shall be $\geqslant$ 45 dB. The value required with higher frequency a.c. primary sources and when inverters and d.c.-d.c. converters are used, has to be determined. (CCIR Study Programme 17F/CMTT; see also CCIR Report 495-1, Vol. XII, Geneva 1974.)

A.3.1.6 *Non-linear distortion*

A.3.1.6.1 *Harmonic distortion*

The total harmonic distortion (THD) shall be measured with the input signal at +9 dBm0s.

The duration for which a single-tone is to be transmitted at this level should be restricted in accordance with Recommendations N.21 and N.23.

The performance when measured with a true-RMS meter shall not be less than the following requirements:

Input tone (kHz)		THD
5 kHz-type circuits	7 kHz-type circuits	
0.07 ... <0.1 0.1 ... 2.5	0.05 ... <0.1 0.1 ... 3.5	2% (−25 dBm0) 1.4% (−28 dBm0)

A.3.1.6.2 *Intermodulation*

With input signals at 0.8 kHz and 1.42 kHz, each at +3 dBm0s, the third order difference tone at 0.18 kHz shall be less than the following values:

5 kHz systems	*7 kHz systems*
1.4% (−34 dBm0s)	1.4% (−34 dBm0s)

6) Administrations are urged to supply additional information on an appropriate value.

A.3.1.6.3 Distortion products measured by shaped noise: under study. See CCIR Report 640.

A.3.1.7 *Error in reconstituted frequency* (applies only to FDM systems)

Not to be greater than 1 Hz.

Note – A maximum error of 1 Hz is in principle acceptable where there is only a single transmission path between the signal source and the listener.

Where the broadcast network can involve two or more parallel paths, e.g., commentary and separate sound channels, or radio broadcast from different transmitters on the same frequency, unacceptable beats may occur unless zero error can be assured. The CCITT is studying methods of effecting this in all recommended systems.

A.3.1.8 *Intelligible cross-talk ratio*

A.3.1.8.1 The intelligible near-end and far-end cross-talk ratios between sound-programme circuits, or from a telephone circuit (disturbing) into a sound-programme circuit (disturbed) shall be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit, and shall not be less than the following values:

5 kHz and 7 kHz systems	
Frequency (kHz)	Crosstalk attenuation (dB)
<0.5	Slope 6 dB/octave
0.5 ... 3.2	74
>3.2	Slope -6 dB/octave

A.3.1.8.2 The near-end and far-end cross-talk attenuations between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) shall be at least 65 dB.

Note 1 – It is understood that this value is defined between the relative levels applicable to telephone circuits. An explanation for the relation between the relative levels for sound-programme circuits and telephone circuits is given in Recommendation J.22, Annex B.

Note 2 – The attention of Administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or in the low frequency range (e.g. below about 100 kHz) in certain carrier systems on coaxial cables. If sub-standard is to be avoided, such systems or parts of systems must not be used for setting up programme channels.

Note 3 – When 4000 pW0p or more noise is continuously present in the telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk ratio of 58 dB between a sound-programme circuit and a telephone circuit is acceptable.

Note 4 – The attention of Administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk limits between two sound-programme circuits, simultaneously occupying the go and return channels respectively, of a carrier system (the most economical arrangement) because in those circumstances they occupy the same position in the line-frequency band (see Recommendation J.18).

Note 5 – The value indicated is based on the assumption that sine wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6 – The effect of cross-talk from a sound-programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound-programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

A.3.1.9 *Amplitude linearity*

When a 1 kHz input signal is stepped from −6 dBm0s to +6 dBm0s, or vice versa, the output level shall change accordingly by 12 ± 0.5 dB.

A.3.2 *Additional parameters for stereophonic programme transmission*

(not applicable)

A.3.3 *Additional requirements for digital systems*

A.3.3.1 If a test signal is harmonically related to the sampling frequency, measuring difficulties may arise. In that case the nominally 1 kHz test signal must be slightly adjusted. No exact preferred frequency can be stated at present, but 1004 Hz and 1020 Hz are examples of frequencies certain Administrations use.

A.3.3.2 *Unbalance of the limitation level*

The difference between those levels which lead to a limitation of the positive or negative half-wave of the test signal shall not exceed 1 dB.

A.3.3.3 *Intermodulation with the sampling signal*

Intermodulation products (f_d) caused by non-linearities may occur in the sound-channel when the sampling signal (f_o) is combined with the inband audio signals (f_i) or out-of-band interfering signals (f_a).

A.3.3.3.1 *Inband intermodulation*

The following combination rule applies: $f_d = f_o - n f_i$.

Only values with $n = 2$ or 3 are of importance.

The level difference between a 0 dBm0s signal (f_i) and the intermodulation products (f_d) shall not be less than 40 dB.

A restriction to the following f_i/f_d values is sufficient:

	$n = 2$		$n = 3$		
f_i	5	7	3	5	kHz
f_d	6	2	7	1	kHz

A.3.3.3.2 *Out-of-band intermodulation*

The following combination rule applies: $f_d = nf_o \pm f_a$.

Only values with $n = 1$ or 2 are of importance.

The level difference between a 0 dBm0s signal (f_a) and the intermodulation products (f_d) shall not be less than 60 dB.

A restriction to the following f_a/f_d values is sufficient:

	$n = 1$		$n = 2$		
f_a	15	17	31	33	kHz
f_d	1				kHz

A.3.3.4 *Further parameters*

Characteristics for bit errors, clicks, jitter, etc. are under study. (See Study Programme 18A/CMTT and CCIR Report 647.)

Bibliography

CCIR Document [1978-1982]: CMTT/68 (OIRT).

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SECTION 3

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP SOUND-PROGRAMME CIRCUITS

Recommendation J.31

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 15 kHz TYPE SOUND-PROGRAMME CIRCUITS

(Geneva, 1972; amended at Geneva, 1976 and 1980)

It is recognized that the overall objective given in Recommendation J.21 can be met by many different types of systems and that some solutions may be preferable to others for national networks, the choice depending on the particular requirements of an Administration.

It is, however, a basic objective of the CCITT to standardize a single solution to be adopted for international circuits. Furthermore, several Administrations have indicated that a single solution for international circuits will considerably ease the problem of providing these circuits.

The CCITT therefore recommends for international circuits the use of the solution described in § 1 below, in the absence of any other arrangement between the interested Administrations, including if necessary the Administrations of the transit countries. Other solutions which have been considered and are capable of meeting the recommended characteristics of Recommendation J.21 are described in Annexes A, B and C.

The characteristics of the group links, which have to be used in any case, are given in § 2 below.

1 Characteristics of an equipment allowing two 15 kHz type carrier-frequency sound-programme circuits to be established on a group

Introduction

An equipment allowing the establishment of 15 kHz type sound-programme circuits (in accordance with Recommendation J.21) on carrier telephone systems which conform to the noise objectives in Recommendation G.222 [1] is defined here. The use of this equipment does not cause either a mean or a peak load higher than that of the telephone channels which it replaces [1]. The two sound-programme circuits set up on one group can be used either as two independent monophonic circuits or as a pair of circuits for stereophonic transmissions.

The following, covering frequency position, pre-emphasis, compandor and programme-channel pilot, are to be considered as integral parts of the Recommendation, forming the complete definition of the equipment covered by this Recommendation.

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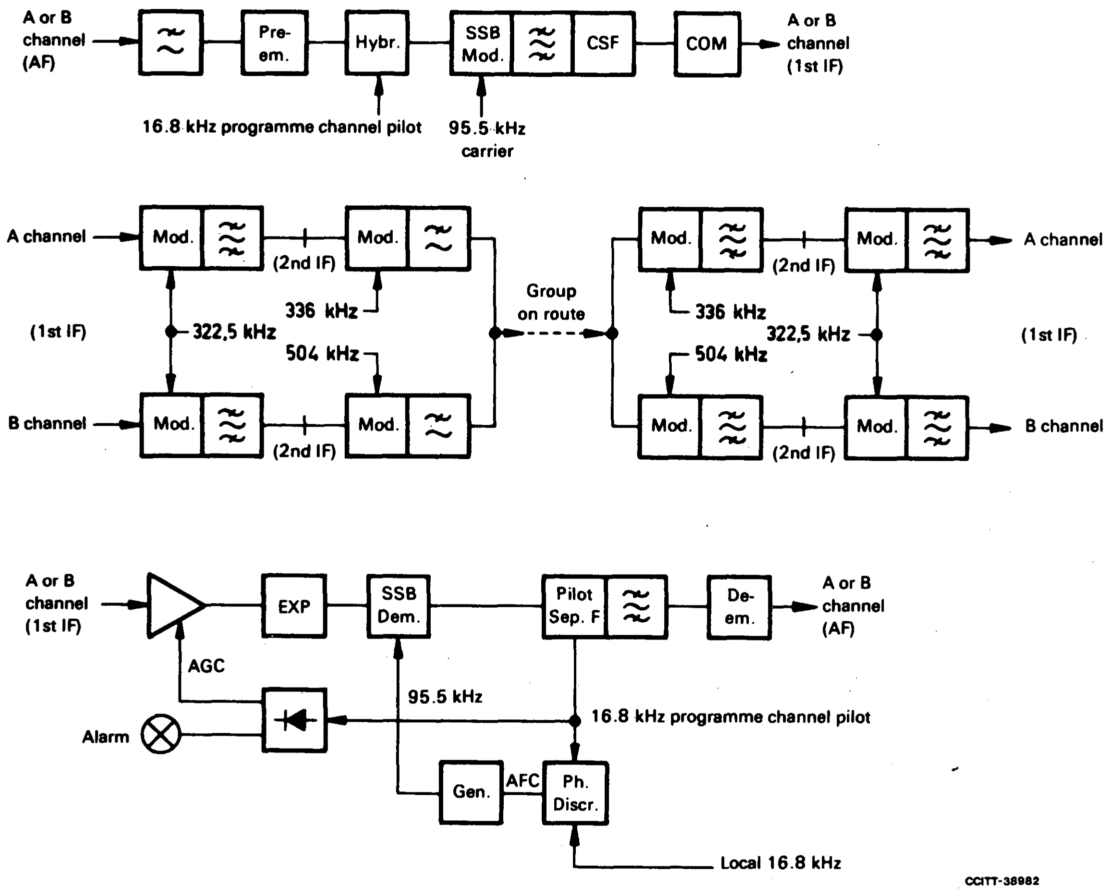
1) This is the objective given in Recommendation J.14 for new design of equipment.

The block schematic of a suitable equipment is given in Figure 1/J.31.



Pre-em.	= pre-emphasis	Pilot Sep. F.	= pilot separating filter
Hybr.	= hybrid	De-em.	= de-emphasis
Mod.	= modulator	Ph. Discr.	= phase discriminator
SSB mod.	= single sideband modulator	AFC	= automatic frequency control
CSF	= carrier suppression filter	AGC	= automatic gain control
EXP	= expander	IF	= intermediate frequency
COM	= compressor	AF	= audio frequency
SSB dem.	= single sideband demodulator		

FIGURE 1/J.31

First modulation, auxiliary modulations and demodulation of the two-channel programme system

1.1 *Frequency position in the basic group 60-108 kHz*

The frequency position in the basic group is shown in Figure 2/J.31. For both programme channels, the tolerance on the virtual carrier frequency is ± 3 Hz and the programme-channel pilot is fed in as 16 800 ± 0.1 Hz in the audio-frequency position.

Note – Programme channel B can be replaced by telephone channels 1 to 6.

1.2 *Intermediate frequency position* (see 1st IF in Figure 3/J.31)

Figure 3/J.31 gives an example of a modulation scheme which is suitable for deriving the line frequency positions shown in Figure 2/J.31, and in which two intermediate frequency stages are used. It is recommended that the first intermediate frequency (1st IF) be identical for each of the sound-programme channels A and B, and the inverted sideband be used based on suppressed carrier of 95.5 kHz.

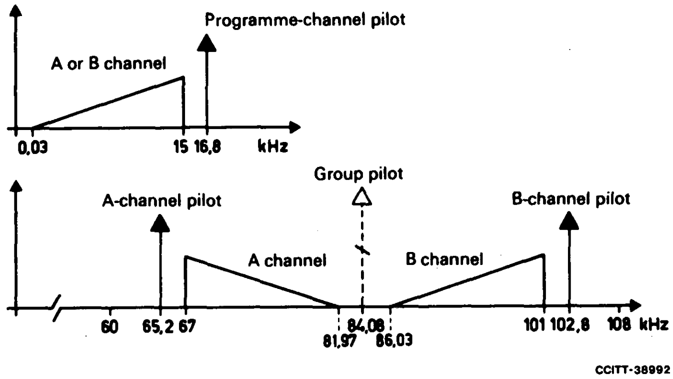


FIGURE 2/J.31

Line-frequency positions of the two-programme channels in the group

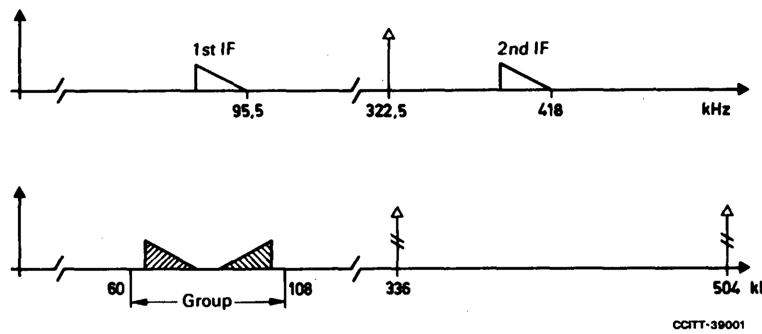


FIGURE 3/J.31

Modulation scheme for the two-channel programme system

It is possible to interconnect sound-programme channels at the 1st IF, but each of the two programme channels must be individually connected. At the intermediate frequency point the sound-programme signal has already been pre-emphasized and compressed, and sound-programme circuits may thus be interconnected at the 1st IF without introducing additional compandors.

The relative level at the interconnection point is similar to the relative level in the carrier telephone system in the basic group at the receiving end (−30.5 dBr). The absolute level is determined by the pre-emphasis and compressor; the long-term mean power of the sound signal (A or B channel) is about 250 μW0.

The nominal impedance chosen in this example is 150 ohms balanced with a 26 dB return loss.

The programme channel pilot is through connected at 95.5 − 16.8 = 78.7 kHz, at a level of −12 dBm0 in the absence of a programme signal.

Special through-connection filters for the sound-programme channel are not required. The bandpass filters at the output of the second modulation stage (receiving end) have sufficient stopband rejection.

1.3 *Pre-emphasis and de-emphasis*

Pre-emphasis and de-emphasis should be applied before the compressor and after the expander respectively in accordance with Recommendation J.17, the 800 Hz attenuation of the pre-emphasis being set to 6.5 dB.

1.4 *16.8 kHz pilot signal*

At the sending end the 16.8 kHz pilot signal is fed in after the pre-emphasis and before the following modulator and compressor with a level of −29 dBm0 ± 0.1 dB. In the absence of a programme signal, this pilot level is increased by 17 dB by the compressor to −12 dBm0(*t*)[2)] on the carrier transmission path. After having passed through the expander, the pilot is branched off for control purposes after the demodulator and before the de-emphasis via a 16.8 kHz bandpass filter and is then suppressed in the transmission channel.

The control functions of the pilot are as follows: frequency and phase correction of the demodulator and compensation of the transmission loss deviations between compressor and expander. In view of the need to transmit stereophonic signals, the phase control should be sufficiently accurate so that the phase difference between the two channels does not exceed 1° even if the frequencies corresponding to the frequencies of the received pilots are in error by ± 2 Hz due to the carrier system.

1.5 *Compandor*

1.5.1 As shown in Figure 4/J.31 the compressor characteristic has a transition from the range of constant gain at low input levels to a range of constant loss at high input levels. Table 1/J.31 indicates the precise dependence of the compressor amplification as a function of the input level. The compressor and expander are controlled by the r.m.s. value of the sum of the voltages of programme and pilot signals.

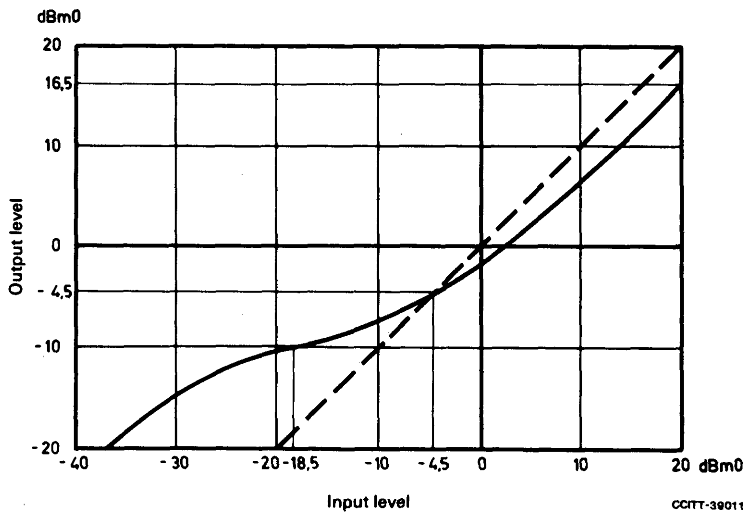


FIGURE 4/J.31

Characteristic of the compressor

In Table 1/J.31, the compressor is pre-loaded by the pilot; in the absence of both programme and pilot, the gain of the compressor reaches the value of 22 dB.

The amplification of the expander is complementary to that of the compressor. The tolerance should also be ± 0.5 dB, or ± 0.1 dB as shown in Table 1/J.31.

1.5.2 The attack and recovery times of the compressor are measured in 12 dB steps (see Recommendations G.162 [2] and O.31 [3]) between the point of the unaffected level of −4.5 dBm0 and the level of −16.5 dBm0 and vice versa. In order to obtain as pronounced an envelope as possible in the oscillogram, the pilot is disconnected during this measurement and a test frequency is chosen which gives rise to an intermediate frequency

2) dBm0(*t*) denotes that the level quoted is referred to a zero relative level point in a telephone channel.

that is approximately in the middle of the IF band. The attack and recovery times of the compressor are, as in Recommendation G.162 [2], the times between the instant when the output voltage of the compressor is suddenly changed and the instant when, after the sudden change, the output voltage passes the arithmetic mean value between initial and final values.

TABLE 1/J.31

Compressor characteristic

Programme signal level at the compressor input (dBm0)	Compressor gain (dB) (tolerance ±0.5 dB except at the point marked * where the tolerance is ±0.1 dB)
−∞	+17.0 *
−40.0	+16.9
−35.0	+16.5
−30.0	+15.6
−25.0	+13.2
−20.0	+ 9.7
−15.0	+ 6.0 *
−10.0	+ 2.7
− 5.0	+ 0.2
− 4.5	0.0
0.0	− 1.3
+ 3.0	− 2.0 *
+ 5.0	− 2.3
+10.0	− 2.9
+15.0	− 3.2
+20.0	− 3.5

The nominal values of the times so measured are:

- attack time: 1 ms;
- recovery time: 2.8 ms.

The subject of tolerances for these values is a matter for further study.

The transient behaviour of the expander is observed with the compressor and expander interconnected. If the same steps are then applied to the compressor input, the signal at the expander output should not deviate from the final steady-state value by more than ± 10%.

Note – Since the initial and final values of the compressor output voltage in the case of this compandor are not in a 1:2 ratio because of the curved characteristic, the arithmetic means here are not 1.5 and 0.75, respectively, as in the case of the telephone compandor.

1.6 *Impedance at audio points*

The audio input-impedance should be 600 ohms balanced with a minimum return loss of 26 dB.

1.7 *Attenuation/frequency distortion due to the sending and receiving equipments*

The total attenuation distortion introduced by a sending and a receiving equipment should not exceed the following ranges:

40 to 125 Hz: +0.5 to −0.7 dB

125 Hz to 10 kHz: +0.3 to −0.3 dB

10 to 15 kHz: +0.5 to −0.7 dB

relative to the gain at 800 or 1000 Hz.

1.8 *Suppression of carrier leaks at 10 kHz and 14 kHz*

Since, according to Recommendation H.14 [4], carrier leaks may be of the order of −40 dBm0 and that Recommendation J.21, § 3.1.6 requires a suppression to (−73 − Δps) dBm0s for single-tone interference, narrow-band crystal stop-filters should be available for insertion if required, and should have the following specifications:

1-dB bandwidth of the stopband

at 10 kHz: ⩽ ± 150 Hz

at 14 kHz: ⩽ ± 210 Hz

Attenuation for the midfrequencies

at 10 kHz: ⩾ 36 dB

at 14 kHz: ⩾ 22 dB

Note – The attenuation of these bandstop filters is sufficient without taking account of the compandor advantage.

The stopband attenuations should be maintained within ± 2 Hz referred to the above midfrequencies, in order to allow for the normal frequency variation of the carrier leaks.

In order to be able to use crystal bandstop filters of a simple design, it is recommended to assign them not to the AF position but to the corresponding IF position, additional allowance having to be made for the carrier frequencies used in the terminal equipment:

10 kHz corresponding to 85.5 kHz and

14 kHz corresponding to 81.5 kHz.

Note – Contribution COM XV-No. 31 (Study Period 1973-1976) from the Federal Republic of Germany gives details of the calculation and numerical data for a possible filter characteristic.

1.9 *Interconnection*

When sound-programme circuits employing equipment in conformity with this Recommendation are interconnected, it is recommended that, where possible, the through connection should be performed either in the group-frequency position or in the position of the 1st IF. As described in § 1.2 above, interconnection in these positions will exclude unnecessary compandor stages from the through connection.

1.10 *Equalizers for gain and phase difference*

In order to be able to meet the quality parameters specified in Recommendation J.21, § 3.1.3, for monophonic and §§ 3.2.1 and 3.2.2 for stereophonic sound-programme transmissions, gain and phase-difference equalizers in the group-frequency position have to be assigned to the sound-programme channel equipment before the hybrid at the receiving end. These equalizers can be switched in steps and their characteristics are adapted to the typical distortions by making them fan-shaped.

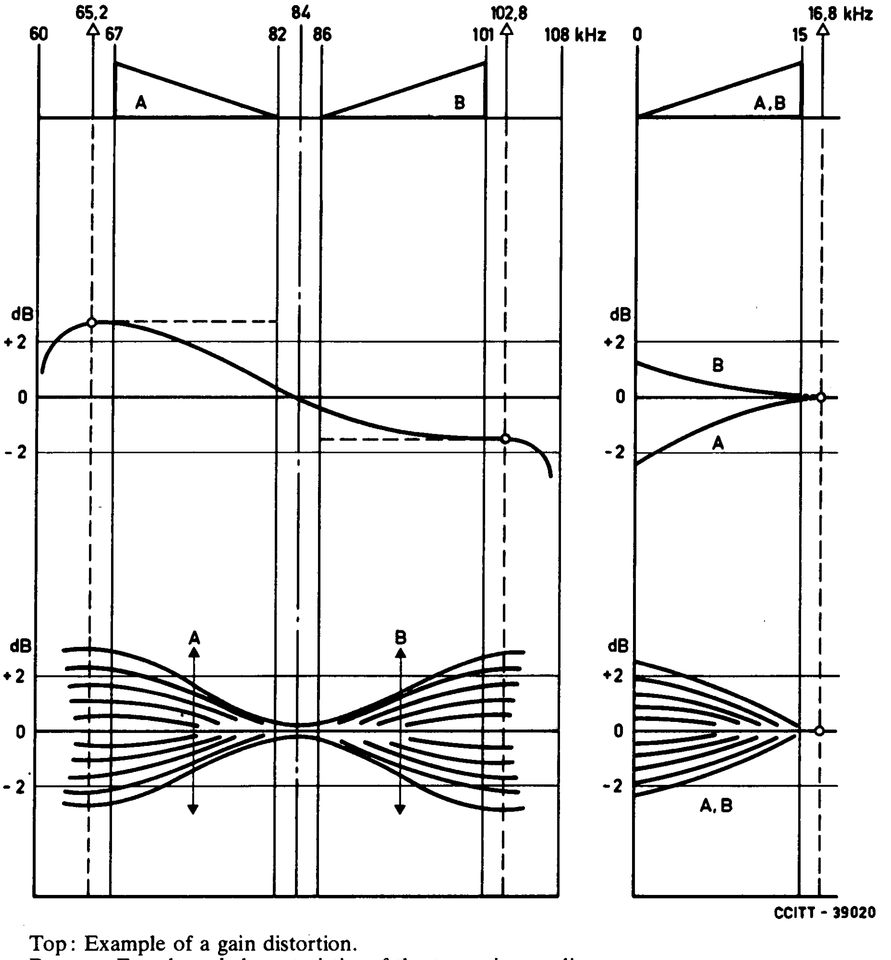
The gain equalizers are required to compensate for the frequency-dependent gain distortions in the lower and upper frequency ranges of the group on which the sound-programme channels are established. By means of the phase-difference equalizers, the phase distortion occurring in the group is increased in the upper or lower half of the group-frequency band to such an extent that a characteristic which is skew-symmetric about the centre frequency of the group is obtained, i.e. phase coincidence between the sound-programme channel positions.

Figures 5/J.31 and 6/J.31 show the effectiveness of the gain and phase-difference equalizers within the frequency band of the group and their effects on gain and phase-difference of the sound-programme channels in the AF position. Here, allowance is made for the fact that deviations at the pilot frequency of 16.8 kHz in the AF position are always automatically adjusted to zero by means of the pilot regulation.

In order to facilitate international cooperation in determining the optimum equalizer setting within a very short time, the lining-up procedure and arrangement of measuring equipment detailed below is recommended.

At the sending end, this arrangement consists of a signal generator with a high level accuracy and a very low output impedance, which produces the measuring frequencies of 0.525 kHz (= 1/32) and 8.4 kHz (= 1/2) derived from the pilot frequency of 16.8 kHz. The two measuring frequencies should be transmitted simultaneously over both sound-programme channels, individually or at automatically alternating 3.9-second intervals. In the latter case the clock is obtained by a further division of 0.525 kHz by 2^{12}.

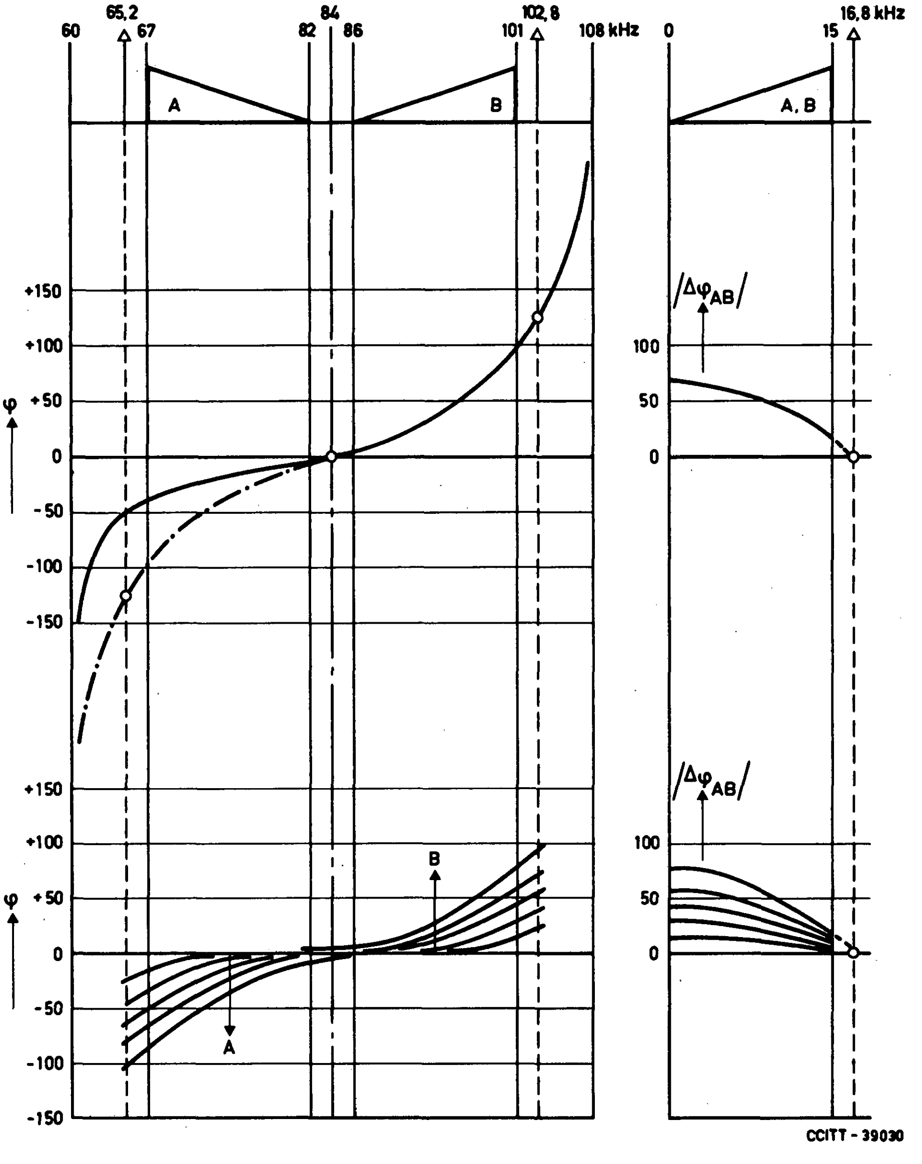
At the receiving end, use is made of a receiver having a calibrated measuring instrument which indicates the level in each of the two sound-programme channels and the phase-difference between them derived from the level of the voltage difference in the two channels. The received measuring frequency is indicated by a lamp. Since the frequency-dependent characteristic of the so-called fan equalizer used for gain and phase-difference equalization is defined for the individual steps, it is possible to confine oneself to the two measuring frequencies considered to be sufficiently representative when determining the optimum equalizer setting.



Top: Example of a gain distortion.
Bottom: Fan-shaped characteristics of the two gain equalizers.

FIGURE 5/J.31

Principle of gain equalization in the group-frequency position and its effect on the sound-programme channels in the AF position, allowance being made for the pilot regulation



Top: Example of phase symmetry distortion. Ideal skew-symmetric phase characteristic shown.
Bottom: Fan-shaped characteristics of the phase-symmetry equalizers.

FIGURE 6/J.31

Principle of phase-symmetry equalization in the group-frequency position and its effect on the phase difference between the sound-programme channels in the AF position, allowance being made for the pilot phase regulation

1.11 *Usable power reserve*

1.11.1 *Audio-frequency parts of the equipment* (before pre-emphasis and after de-emphasis):

1.11.1.1 *Peak power level*

The equivalent power level of the peak of sound-programme signals, when they are controlled in accordance with Recommendations J.14 and J.15 so as to have a quasi-peak power of +9 dBm0s, exceeds a level of about +12 dBm0s with a probability of 10^{-5}, as is documented by several Administrations (see CCIR Report 491-2 [5]). For the telephone service, the level with a probability of 10^{-5}, i.e. the level of +12 dBm0s, should be respected in any case.

1.11.1.2 *Margin against saturation*

A margin of 3 dB should be maintained between the peak power level in § 1.11.1.1 and the overload point, to allow for level variations.

1.11.1.3 *Overload point, definitions*

First definition – The **overload point** or overload level of an amplifier is at that value of absolute power level at the output, at which the absolute power level of the third harmonic increases by 20 dB when the input signal to the amplifier is increased by 1 dB.

This first definition does not apply when the test frequency is so high that the third harmonic frequency falls outside the useful bandwidth of the amplifier. The following definition may then be used:

Second definition – The overload point or overload level of an amplifier is 6 dB higher than the absolute power level in dBm, at the output of the amplifier, of each of two sinusoidal signals of equal amplitude and of frequencies A and B respectively, when these absolute power levels are so adjusted that an increase of 1 dB in both of their separate levels at the input to the amplifier causes an increase, at the output of the amplifier, of 20 dB in the intermodulation product of frequency 2A-B.

1.11.1.4 *Value of the overload point*

The overload point of these audio-frequency parts therefore should be higher than +15 dBm0s.

1.11.2 *Carrier-frequency parts of the programme modulating equipment (between compressor and telephone multiplex and between telephone multiplex and expander)*

The overload point, as defined in § 1.11.1.3 should have a margin of 2 dB against the equivalent peak power value of a group channel (+19 dBm0). The overload point of these carrier-frequency parts therefore should be higher than +21 dBm0.

1.11.3 *Complete equipment, back to back*

Test measurements should be possible without degradation visible on an oscilloscope:

– with one or two sine-wave test signals of any frequency with peak power levels up to +12 dBm0s,

– with tone pulses of any frequency with levels up to 0 dBm0s.

1.12 *Loading of groups and supergroups*

Table 2/J.31 gives some observed figures for the loading of groups and supergroups in the most essential cases.

2 Characteristics of a group link used to establish two 15-kHz type carrier-frequency sound-programme circuits

The lining-up of international group links is described in Recommendation M.460 [9] in which information is given on the attenuation/frequency characteristics which should be obtained. To comply with the attenuation/frequency characteristics of sound-programme circuits in accordance with Recommendation J.21, it may be necessary to include a small amount of additional equalization.

TABLE 2/J.31

Loading of groups and supergroups in the case of sound-programme transmission with the carrier programme system recommended in CCITT Recommendation J.31, § 1

	n_m (dBm0)	n_p (dBm0)
Group		
12 telephone channels (as in Recommendation G.223 [6])	−4	+19
1 programme channel only	−6	+12
1 programme channel + 6 telephone channels	−3.5	+12 programme channel only
2 programme channels (different monophonic programmes)	−3	+13
1 stereophonic pair [a]	−3	+17
2 programme channels (identical monophonic programmes)	−3	+17
Supergroup		
60 telephone channels (as in Recommendation G.223 [6])	+3	+21
4 programme channels in 2 groups + 36 telephone channels:		
4 different programmes	+3,5	+14 programme channels only
2 different stereophonic programmes	+3.5	+18 programme channels only
2 equal stereophonic programmes	+3.5	+22 programme channels only
10 programme channels		
10 different programmes	+4	+15
5 different stereophonic programmes	+4	+19
2 equal stereophonic programmes + 6 different monophonic programmes	+4	+22

n_m Long-term mean power level [7].

n_p Equivalent peak power level [8] (= level of equivalent sine-wave whose amplitude is exceeded by the peak voltage of the multiplex signal only with a bilateral probability of 10^{-5}).

a) Loading by one stereophonic programme is treated as loading by two identical monophonic programmes (worst case).

Group links for programme transmission have to meet special requirements concerning carrier leaks and other interfering frequencies so that programme transmission conforms to the standard as defined in Recommendation J.21.

The basic requirement is that interfering frequencies appearing in the programme bands have to be suppressed to $(-73 - \Delta ps)$ dBm0s on the programme circuit[3]. For frequencies corresponding to audio frequencies above 8 kHz, additional suppression is possible by special spike filters in the terminal equipment of the programme circuit.

Group links to be used for programme transmission according to Recommendation J.21 and using programme terminal equipment according to Recommendation J.31 have to meet, therefore, the following requirements:

a) Carrier leaks[4] at 68, 72, 96 and 100 kHz and any single-tone interference signal falling outside the band of frequencies used for sound-programme transmission including the pilots (see Figure 2/J.31) should not be higher than −40 dBm0. This allows the necessary suppression to $(-73 - \Delta ps)$ dBm0s taking account of the amount of the narrow-band crystal stop-filter attenuation.

3) This value has been specified in Recommendation J.21 by CMTT. CCIR Report 493-2 [10] gives some additional information regarding the subjective impairments produced by interfering frequencies on a circuit using equipment conforming to Recommendation J.31.

4) Having the frequency precision of carriers.

b) Carrier leaks at 76, 80, 88 and 92 kHz and any other single-tone interference signal falling within the band of frequencies used for sound-programme transmission including the pilots (see Figure 2/J.31), should not be higher than:

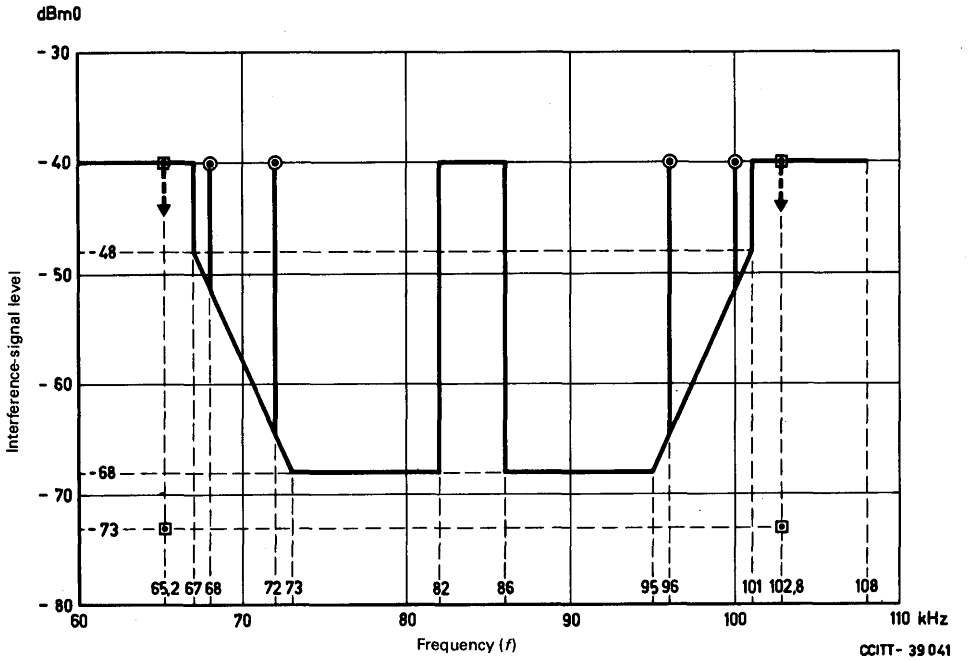
- for frequencies between 73 kHz and 95 kHz: −68 dBm0,
- for frequencies at 67 kHz and 101 kHz: −48 dBm0.

In the bands 67 to 73 kHz and 95 to 101 kHz the requirement is given by straight lines (linear frequency and dB scales) interconnecting the requirements given above[5].

It is necessary to consider whether additional requirements for the characteristics of group links for 15-kHz sound-programme transmission are needed beyond those covered in Recommendation M.460 [9] (for example, group delay distortion in the case of stereophonic transmission bearing in mind the possibility of changeover to stand-by paths).

The above requirements are illustrated in Figure 7/J.31.

Note – Figure 8/J.31 gives the permissible level of single-tone interference for the systems described in Annexes A, B and C, such that the basic requirement of (−73 − Δps) dBm0s mentioned above is met.



The continuous curve represents the general requirements for single-frequency interfering tones, with the following exceptions:

◉ carrier-leak frequencies at which the requirements are relaxed to −40 dBm0 are shown thus.

▣ at frequencies of A- and B-channel pilots, 65.2 and 102.8 kHz ±300 Hz interfering signals should be at least 40 dB below the lowest possible level of the pilots (i.e. −29 dBm0 −3.5 dB when compressor input signal is large).

FIGURE 7/J.31

Mask for the carrier leaks and any other tone-interference signal falling within the group band

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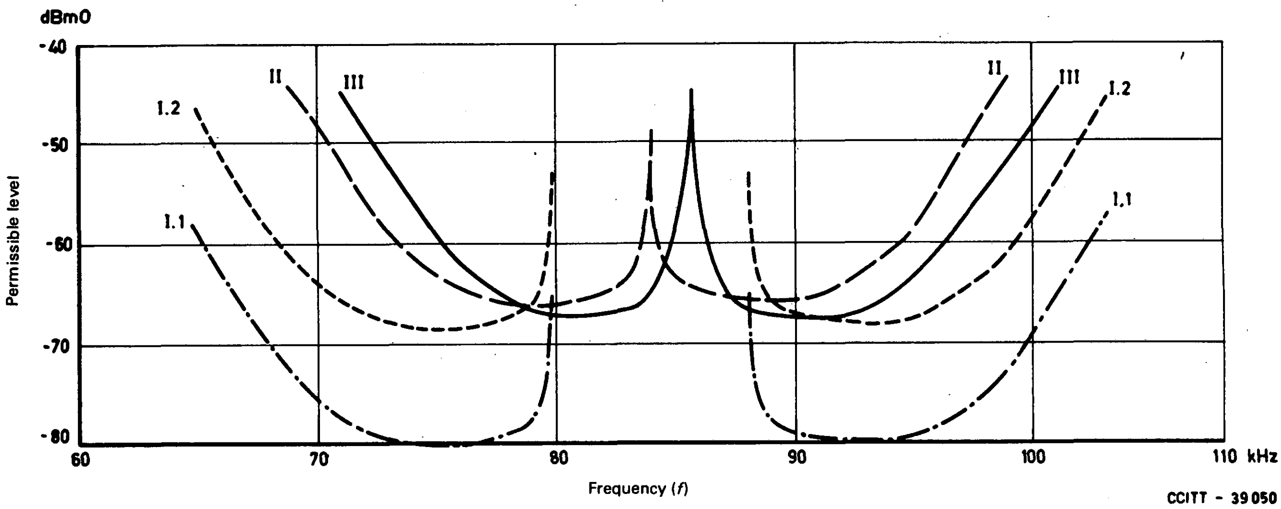
5) These values are still under study. It has been assumed that the compandor gives a subjective improvement of at least 12 dB. CMTT is asked to confirm that this assumption is valid.



Curve I.1: requirement for the system of Annex A, *without* compandor gain.
Curve I.2: requirement for the system of Annex A, *with* compandor gain.
Curve II: requirement for the DSB system of Annex B.
Curve III: requirement for the system of Annex C.

FIGURE 8/J.31

Permissible level of a single-frequency interference on the group link

ANNEX A

(to Recommendation J.31)

Single sideband system

(Contribution of the N.V. Philips Telecommunicatie Industrie)

This Annex concerns a single-sideband sound-programme transmission equipment incorporating pre- and de-emphasis combined with a compandor characterized by a separate FM control channel.

The equipment operates on group links of carrier telephone systems.

Both peak and average loads to the group are compatible with those of the replaced telephone channels.

A.1 *Frequency allocation in the group*

TABLE A-1/J.31

	Modulated programme frequencies	Compandor control channel	Synchronizing pilot
Channel A (inverted)	65 . . . . . 79.96 kHz	81.39 . . . 83.18 kHz	84 kHz
Channel B (erect)	88.04 . . . 103 kHz	84.82 . . . 86.61 kHz	

Channels A and B (see Table A-1/J.31) can be used for independent monophonic sound-programme circuits or combined into a stereophonic pair. Either channel A or B can be deleted and substituted by the corresponding telephone channels.

Group pilots at 84.08, 84.14 and 104.08 kHz and telephone channels 1 and 12 are compatible with this frequency allocation.

A.2 *Pre-emphasis*

Pre-emphasis takes place before compression by means of a network according to Recommendation J.17. The insertion loss at 800 Hz is 6.5 dB.

A.3 *Compandor*

A.3.1 *Steady-state characteristics*

The compandor has a separate frequency-modulated control channel containing the information on the degree of compression, as indicated in Table A.2/J.31.

For the lowest programme levels, the total improvement in signal-to-noise ratio will be 19.8 dB (when weighting by means of a psophometer according to the Recommendation cited in [11]).

TABLE A-2/J.31

Compressor input level (dBm0)[a]	Compressor gain (dB)	Control channel frequency (kHz)	
		Channel A	Channel B
$-\infty$	17	81.39	86.61
−40	17	81.39	86.61
−35	16.9	81.40	86.60
−30	16.7	81.41	86.59
−25	15.9	81.43	86.57
−20	13.5	81.52	86.48
−15	9.5	81.70	86.30
−10	4.8	81.94	86.06
− 5	0	82.24	85.76
0	− 4.9	82.56	85.44
+ 5	− 9.6	82.90	85.10
+10	−11.8	83.18	84.82
+15	−11.8	83.18	84.82

[a] The relative level at the compressor input to be considered is 6.5 dB higher than that corresponding to an 800 Hz audio-frequency test-tone. With pre-emphasis and compressor, an audio input level of e.g. +6.5 dBm0s at 800 Hz will thus give rise to a compressor input level of 0 dBm0 and hence to a group level of −4.9 dBm0(*t*).

The level in the control channel is −17 dBm0(*t*).

The expander gain tracks that of the compressor with a tolerance of ±0.5 dB.

dBm0(*t*) denotes that the level quoted is referred to a 0 relative level point in a telephone channel.

dBm0s denotes that the level quoted is referred to the sound-programme circuit.

A.3.2 *Transient behaviour of the compressor*

Considering a 12 dB level step at the compressor input from −17 dBm0 to −5 dBm0 (point of unaffected level), the compressor attack time is defined as the time interval needed for the compressor output voltage to reach the arithmetical mean between initial and final values.

Taking the sudden level variation in the opposite direction yields the definition of the compressor recovery time.

The nominal values of attack and recovery time are respectively 2.4 and 4 ms.

A.3.3 *Transient behaviour of the expander*

With compressor and expander interconnected and when applying at the compressor input sudden level variations from -17 dBm0 to -5 dBm0 and vice versa, the expander output voltage should not deviate by more than 10% from the steady-state values.

A.4 *Synchronizing pilot*

A synchronizing pilot at 84 kHz with a level of -20 dBm0(t) is used in order to reduce frequency and phase errors due to the group link.

Frequency offset is reduced by a factor of 21.

At the transmitting and receiving terminals, the modulating and demodulating carriers should be phase-coherent with the synchronizing pilot in such a way that a frequency offset of 2 Hz does not give rise to a phase difference between the two channels of the stereophonic pair exceeding 1°.

ANNEX B

(to Recommendation J.31)

Double-sideband system

(Contribution of L.M. Ericsson, ITT and Telettra)

B.1 *Frequency allocation*

Double-sideband modulation of a carrier frequency of 84.080 kHz. The sidebands are located in the band 69.080-99.080 kHz. The carrier is reduced in level, so that it can be used in the normal way for a group pilot.

B.2 *Pre-emphasis*

The pre-emphasis curve given in Recommendation J.17 should be used.

B.3 *Compandors*

Compandors are not an integral part of these systems.

B.4 *Levels of programme signal in carrier system*

The levels are such that a sine wave of 800 Hz applied at the audio input with a level of 0 dBm0s will appear at the group output, having been through a pre-emphasis network, as two sideband frequencies each with a level of $+2$ dB compared to the relative level of the telephone channels, that is $+2$ dBm0(t). This level should be adjustable over a range of about $\pm$ 3 dB.

B.5 *Group regulation*

Normal group regulation is available using 84.080 kHz. This frequency had the normal level and tolerances for a pilot as given in the Recommendation cited in [12].

B.6 *Carrier regeneration*

Different versions of this system rely respectively on the correct phase of the group pilot or on the use of an auxiliary pilot above the programme band (16.66 kHz or 16.8 kHz, for example, has been proposed for national systems); a frequency of 16.8 kHz should be reconsidered for international use; the sending terminal should, where necessary, be adapted to meet the needs of the receiving terminal in either respect. The level of any auxiliary pilot should not exceed -20 dBm0(t), i.e. referred to the telephone channel level in the group.

ANNEX C

(to Recommendation J.31)

Transmitting of six sound-programme circuits on a supergroup link

(Contribution of Società Italiana
Telecomunicazioni Siemens SpA)

A system for setting up on group links one monophonic programme circuit or two circuits combined into a stereophonic programme, is described in Contribution COM XV-No. 151 (Study Period 1973-1976) and is widely used in Italy.

A new type of equipment for the transmission of six programme channels allocated in the band of a basic supergroup has been developed and successfully adopted experimentally.

The essential characteristic of this system is the utilization of a single sideband, modulated in amplitude, with a suppressed carrier of 86 kHz and a synchronous demodulation using a 16.8-kHz pilot in order to have no errors in the transmitted frequencies and no errors in the phase relation between the signals A and B for stereophonic programmes.

The carrier of 86 kHz is suitable for allocating the programme signal to that sideband which is unaffected by telephone carrier leaks and for avoiding intelligible crosstalk between telephone and programme channels.

The single-sideband modulation employs the phase-shift technique. By means of this the programme channel is allocated either to the lower sideband between 71 and 86 kHz or to the upper sideband between 86 and 101 kHz.

In a second modulation procedure the six sound-programmes are allocated to the band of the basic supergroup 312-552 kHz with the carriers 346 kHz, 382 kHz, 418 kHz, 454 kHz, 490 kHz and 526 kHz.

The measurements carried out show that the system complies with the values recommended in Recommendation J.21 for the high-quality circuits with equipments whose price renders the system economical, even for distances of some hundreds of kilometres.

References

[1] CCITT Recommendation *Noise objectives for design of carrier-transmission systems of 2500 km*, Vol. III, Rec. G.222.

[2] CCITT Recommendation *Characteristics of compandors for telephony*, Vol. III, Rec. G.162.

[3] CCITT Recommendation *Specification for an automatic measuring equipment for sound-programme circuits*, Vol. IV, Rec. O.31.

[4] CCITT Recommendation *Characteristics of group links for the transmission of wide-spectrum signals*, Vol. III, Rec. H.14.

[5] CCIR Report *Characteristics of signals sent over sound-programme circuits*, Vol. XII, Report 491-2, ITU, Geneva, 1982.

[6] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*, Vol. III, Rec. G.223.

[7] *Ibid.*, § 1.

[8] *Ibid.*, § 6.2.

[9] CCITT Recommendation *Bringing international group, supergroup, etc., links into service*, Vol. IV, Rec. M.460.

[10] CCIR Report *Compandors for sound-programme circuits*, Vol. XII, Report 493-3, ITU, Geneva, 1982.

[11] CCITT Recommendation *Psophometers (apparatus for the objective measurement of circuit noise)*, Green Book, Vol. V, Rec. P.53, Part B, ITU, Geneva, 1973.

[12] CCITT Recommendation *Pilots on groups, supergroups, etc.*, Vol. III, Rec. G.241, §§ 2 and 3.

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 10 kHz TYPE SOUND-PROGRAMME CIRCUITS

(former Recommendation J.22; amended at Geneva, 1964, and Mar del Plata, 1968)

10 kHz type sound-programme circuits can be provided in wideband cables by the following methods:

1 Special pairs for sound broadcasting

If a broadcast programme is to be distributed to a number of intermediate points along the line (if this includes carrier telephone systems), it may be necessary to use a pair of conductors with a special screen for programme transmissions; or it may happen that it is preferable to transmit the broadcast programme over the carrier system itself or on the phantom of the unloaded pairs.

It should be remembered, however, that interstice pairs in a coaxial cable are principally intended for the maintenance and supervision of the telephone carrier system routed over the coaxial pairs.

2 10-kHz sound-programme circuits routed over channels of a carrier telephone system in cable

It is recommended to use the frequency band corresponding to three telephone channels of a carrier system to form a 10 kHz type sound-programme circuit. One such assembly of three channels may be used in this matter in a 12-circuit group.

The CCITT has already recommended the position defined below as Position I for this assembly of three channels to provide programme transmission in basic group B.

Position I: Frequency band used: 84-96 kHz

Virtual carrier frequency: 96 kHz

The CCITT no longer recommends the use of Position II, defined in the old Recommendation (*Red Book*, Volume III), in the international service.

The CCITT recommends also the following frequency arrangements in the basic Group B:

Position III: Frequency band used: 84-96 kHz

Virtual carrier frequency: 95.5 kHz

This position may be adopted whether a compandor is used or not.

Supplement No. 12 of the *Green Book* [1] indicates the improvement in crosstalk to be expected from offset of the carrier frequency and in particular from the use of Position III.

Note – Some Administrations use a pilot inserted in the audio-frequency part of the sound-programme modulation equipment for the purpose of regulating the equivalent and supervising the link as a whole.

While, generally speaking, the provision of automatic group regulation should suffice to ensure satisfactory stability of the equivalent, a pilot like the one suggested by one of these Administrations might be useful when compandors (which increase variations of the equivalent) are used, or when the switching of sound-programme circuits to RF is envisaged or when frequency synchronization is required between the ends of the circuit.

With the limit that has been given in Recommendation J.14 for the "peak voltage" transmitted by one such assembly of three channels, these assemblies (used for sound-programme transmissions) may be placed in any basic group (or in all the basic groups) of a supergroup (or in all supergroups) of a carrier system on coaxial cable.

The CCITT has not limited the possible positions (in the basic supergroup) of the groups over which 10-kHz type sound-programme circuits can be routed, but it can be said that the basic groups (in a supergroup) which appear most appropriate for such circuits are groups 2, 3 and 4. These groups are subject to less attenuation distortion at the edges (produced by certain filters in the supergroup) than groups 1 and 5. The most appropriate supergroups in which to place the sound-programme circuits are those which are transmitted on the coaxial cable with the lowest carrier frequencies, because the frequency deviation (due to instability of the frequency generators) on the channels of these groups will be proportionately lower than the deviation on channels in supergroups transmitted at a high frequency. Supergroup 2 (the basic supergroup) has the additional advantage of having one stage of modulation less than the other supergroups.

In the case of a carrier system on symmetric pairs, it may be necessary to make a special choice of the group of the system and the pairs to be used in order that the conditions concerning crosstalk for the complete sound-programme circuit will be satisfied. (See Recommendations J.18 and J.22.)

Where a sound-programme circuit is used in circumstances requiring zero error of the reconstituted frequencies (see CCITT Recommendation J.22) it is recommended that a frequency reference pilot be transmitted by the send programme equipment. The frequency of this pilot should be the same as the virtual carrrier frequency (i.e. corresponding to zero audio frequency in the sound-programme signal) and its level should be -30 ± 1 dBm0. In the receive programme equipment this frequency reference pilot should be used to synchronize the carrier frequencies used in the translation process in such a way that no errors occur in the reconstituted frequencies.

3 Use of phantom circuits on unloaded symmetric pairs equipped with carrier systems

Experience has shown that the phantom circuits of symmetric pairs in cables equipped with carrier systems may allow transmission (as defined in Recommendation J.22, § 1) from 50 Hz to 10 000 Hz. These circuits have the advantage that derivations at various repeater stations of the carrier system can easily be made, thus allowing the distribution of a radio programme or the picking up of a supplementary programme at various points along the line.

When such phantom circuits are used over long distances, it may be necessary to provide manual or automatic regulation to compensate for changes of attenuation with time.

4 Use of the band of frequencies below 12 kHz

The use of phantom circuits (see § 3 above) naturally depends on a multiple twin or a star quad cable being available. If only a pair cable is available, a possible solution would be to place the sound-programme transmission in the frequency band below 12 kHz, i.e. below the frequency band used for the carrier telephone channels; but this solution involves difficulties with filters or with crosstalk balancing frames, if any exist.

Reference

[1] *Intelligibility of crosstalk between telephone and sound-programme circuits*, Green Book, Vol. III-2, Supplement No. 12, ITU, Geneva, 1973.

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 6.4 kHz-TYPE SOUND-PROGRAMME CIRCUITS [1)]

The CCITT recommends that, when an Administration wishes to provide a sound-programme circuit transmitted on a carrier system using a frequency band corresponding to two telephone channels, the circuit should occupy the frequency range 88 kHz to 96 kHz in the basic 12-channel group B frequency band and the virtual carrier frequency within this range should be 96 kHz, or as an alternative, 95.5 kHz [2)].

If there is an arrangement between interested Administrations, including if necessary the Administration of transit countries, a solution allowing the establishment of up to four 6.4 kHz-type sound-programme circuits in a basic group as described in Annex A may be used.

ANNEX A

(to Recommendation J.33)

Four 6.4 kHz type sound-programme circuits in a basic group

(Contribution by the PTT of China)

A.1 *Frequency position and modulation scheme*

In order that the requirements of the performance characteristics of adjacent basic groups, supergroups, etc., through-connection equipments are not more stringent than those for the 15 kHz-type sound-programme circuits, the band of four 6.4 kHz programme frequencies in a group should be within the range of 65.3 to 102.7 kHz band.

In order that modulation procedure is the same as that of 15 kHz-type sound-programme circuits, three level modulations are adopted. Modulation procedure and frequency position are shown in Figure A-1/J.33. All the carriers and pilots are derived from 12 kHz basic frequency.

A.2 *Emphasis network and compandor*

In order that the signal mean load of four 6.4 kHz-type sound-programme circuits in telephone circuits is less than -3 dBm0, and the peak value load less than $+19$ dBm0, it is necessary that the programme relative level (dBrs) should be lower than that of telephone relative level (dBr) by 6.5 dB and emphasis network should be applied.

In order to meet the requirement of -39 dBm0s noise level of 2,500 km hypothetical reference circuits defined in Recommendation J.23 (Yellow Book, 1980) in addition to the emphasis network, compandor should also be applied.

6.4 kHz system applies emphasis network as described in Recommendation J.17. At 0.8 kHz, the insertion loss of pre-emphasis is 6.5 dB, while the insertion gain of de-emphasis is 6.5 dB.

6.4 kHz system applies the same compandors as 15 kHz system does. (See Figure 4/J.31, Recommendation J.31.)

A.3 *Pilot*

To ensure the stability of instertion loss and deviation of frequency required in programme circuits, a 7.5 kHz pilot at a level of -29 dBm0 $\pm$ 0.1 dB is inserted after pre-emphasis and before modulator in transmission path.

The pilot, after demodulator in receiving path, is derived so as to regulate frequencies and levels.

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1) The performance characteristics of 6.4 kHz-type sound-programme circuits are given in Recommendation J.23 (Yellow Book, 1980).

2) For the choice of groups and supergroups used, see Recommendation J.32.

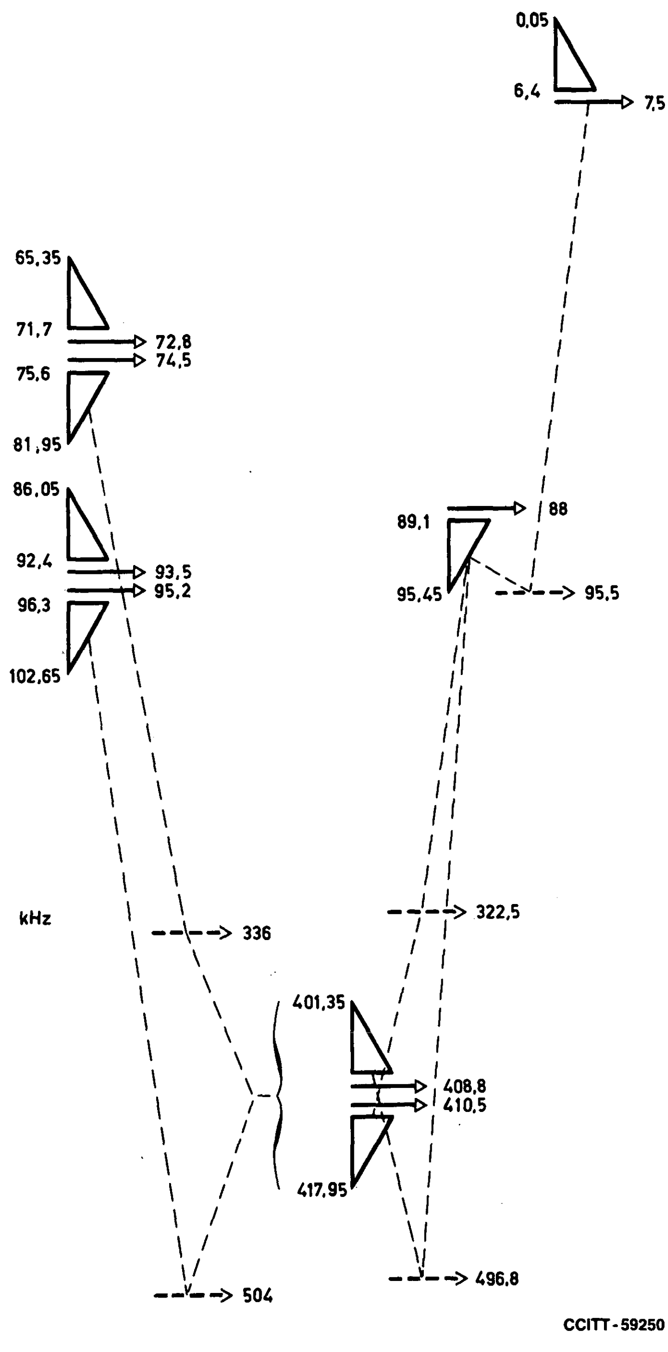


FIGURE A-1/J.33

Frequency position of four 6.4 kHz-type sound-programme channels in a group

A.4 *Noise*

Weighted noise of telephone channel hypothetical reference circuits	−50 dBm0p
Due to:	
telephone weighting network loss	2.5 dB
bandwidth expanding from 3.1 kHz to 6.4 kHz	3.2 dB
CCIR Recommendation 468-3 sound-programme weighted network (0.05 to 6.4 kHz)	9.0 dB
CCIR Recommendation 468-3 quasi-peak value measurement	5 dB
Sum (noise of hypothetical reference circuit without emphasis and compandor)	−30.3 dBq0ps
Variation of weighted noise level within the range of 0.05 to 6.4 kHz band due to de-emphasis (6.5 dB/800 Hz)	− 3 dB
Variation of noise level due to expandor	−12 dB
Noise of weighted hypothetical reference circuit of 6.4 kHz type programme channels (with emphasis and compandor)	−45.3 dBq0ps

There is about 6 dB safety margin compared with −39 dBq0ps for 6.4 kHz type programme circuits described in Recommendation J.23.

A.5 *Summary*

In a group, four 6.4 kHz sound-programme channels (A, B, C and D) can be established, and A (or D) can be replaced by three telephone channels, A + B (or C + D) can be replaced by one 15 kHz sound-programme channel or by six telephone channels.

This system meets every requirement of 6.4 kHz-type sound-programme circuits described in Recommendation J.23 (Yellow Book, 1980). There is no risk of overload in a group even when four programme channels transmit the same programme simultaneously.

Recommendation J.34

CHARACTERISTICS OF EQUIPMENT USED FOR SETTING UP 7 kHz TYPE SOUND-PROGRAMME CIRCUITS

(Geneva, 1980)

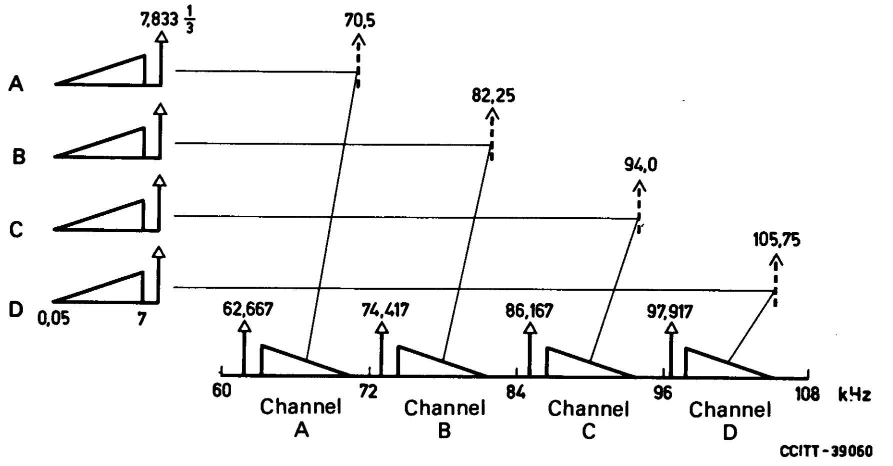
Introduction

An equipment allowing the establishment of 7 kHz type sound-programme circuits (in accordance with CCIR Recommendation 503-1 [1]) on carrier telephone systems which conform to the noise objectives in Recommendation G.222 [2] is defined here. The use of this equipment does not cause either a mean or a peak load higher than that of the telephone channels which it replaces. The sound-programme circuits set up on one group can be used only as monophonic circuits.

The following recommendations, covering frequency position, pre-emphasis, compandor and programme-channel pilot, are to be considered as integral parts of the Recommendation, forming the complete definition of the equipment covered by this Recommendation.

1 Frequency position in the basic group 60-108 kHz

The frequency position in the basic group is shown in Figure 1/J.34. For the programme channels, the stability of the virtual carrier frequency is $\pm 10^{-5}$ and the programme-channel pilot is fed in as 7833 1/3 Hz (stability better than $\pm 10^{-5}$) in the audio-frequency position.



Note – The carrier frequencies are multiples of 11.75 kHz and can be derived from a common generator frequency.

FIGURE 1/J.34

Frequency allocation for four 7 kHz type sound-programme channels set up on one group

Note 1 – Programme channel D can be replaced by telephone channels 1 to 3; programme channel C by telephone channels 4 to 6; programme channel B by telephone channels 7 to 9; programme channel A by telephone channels 10 to 12.

Note 2 – The use of programme channel D is only compatible with group pilots at 84.14 and 84.08 kHz, but not at 104.08 kHz. Moreover, this channel cannot be used in Group 3 of a supergroup with a 411.92 kHz pilot or a 411.86 kHz pilot.

The frequency positions are as shown in Table 1/J.34.

TABLE 1/J.34

Channel range (kHz)	Virtual carrier frequency [a)] (kHz)
60 to 72	70.5 Inverted position
72 to 84	82.25 Inverted position
84 to 96	94 Inverted position
96 to 108	105,75 Inverted position

[a)] The carrier frequencies are multiples of 11.75 kHz and can be derived from a common generator frequency.

2 Pre-emphasis and de-emphasis

Pre-emphasis and de-emphasis should be applied before the compressor and after the expander respectively in accordance with Recommendation J.17, the 800 Hz attenuation of the pre-emphasis being set to 6.5 dB.

3 7833 1/3-Hz pilot signal

At the sending end, the 7833 1/3-Hz pilot signal is fed in after the pre-emphasis and before the following modulator and compressor with a level of -29 dBm0 $\pm$ 0.1 dB (the relative level at this point being defined under the assumption that the compressor is switched off and replaced by 0 dB loss). In the absence of a programme signal, this pilot level is increased by 14 dB by the compressor to -15 dBm0 on the carrier transmission path. After having passed through the expander, the pilot is branched off for control purposes after the demodulator and before the de-emphasis via a 7833 1/3-Hz bandpass filter and is then suppressed in the transmission channel.

The control functions of the pilot are frequency regeneration of the demodulator and compensation of the transmission loss deviations between compressor and expander. The frequency regeneration of the demodulator should be sufficiently accurate so that the frequency offset between the audio-frequency (AF) programmes at the transmit end and at the receive end is less than 0.6 Hz even if the frequency offset of the group connection is 2 Hz.

4 Compandor

The characteristic of the compressor is the same as in Recommendation J.31, § 1.5.1 with the only exception that the output level is decreased by 3 dB. The maximum compressor gain is 14 dB, the minimum compressor gain is -6.5 dB. With an input level of -18.5 dBm0, its output level is -13 dBm0.

The tolerance of the compressor gain is $\pm$ 0.5 dB, but it is $\pm$ 0.1 dB at programme signal levels at the compressor input of $-\infty$, -15 and $+3$ dBm0 (in agreement with Table 1/J.31).

The amplification of the expander is 3 dB larger than that given in Recommendation J.31, § 1.5.1.

5 Attenuation/frequency distortion due to the sending and receiving equipments

The total attenuation/frequency distortion introduced by a sending and a receiving equipment should not exceed the following preliminarily recommended ranges:

0.05 to 0.1 kHz: $+0.7$ to -1.0 dB

0.1 to 6.4 kHz: $+0.5$ to -0.5 dB

6.4 to 7 kHz: $+0.7$ to -1.0 dB

relative to the gain at 800 or 1000 Hz.

Note – These values are still under study. Three carrier sections with two intermediate audio points according to the hypothetical reference circuit (h.r.c.) (Recommendation J.11) should comply with the CCIR Recommendation cited in [3].

6 Suppression of carrier leaks

Carrier leaks which, after demodulation, fall into the AF programme band should have a level lower than -68 dBm0 in the carrier frequency position.

A carrier leak at, and residuals from pilots in the vicinity of, 64 kHz with a level above -68 dBm0 will generate an intolerable single-tone interference at 6.5 kHz in channel A. If required, it may be suppressed sufficiently with a lowpass filter at the AF output of channel A. Then this channel can be used for a 5 kHz type sound-programme circuit.

References

[1] CCIR Recommendation *Performance characteristics of narrow-bandwidth sound-programme circuits*, Vol. XII, Rec. 503-1, ITU, Geneva, 1978.

[2] CCITT Recommendation *Noise objectives for design of carrier-transmission systems of 2500 km*, Vol. III, Rec. G.222.

[3] CCIR Recommendation *Performance characteristics of narrow-bandwidth sound-programme circuits*, Vol. XII, Rec. 503-1, § 3.3.1, ITU, Geneva, 1978.

SECTION 4

CHARACTERISTICS OF EQUIPMENTS FOR CODING ANALOGUE SOUND PROGRAMME SIGNALS (FOR TRANSMISSION ON 384 kbit/s CHANNELS)

Recommendation J.41

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

(See Notes 1 and 2)

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 either:

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 505-2) are not exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

Note 1 – ISDN broadband channels, channel structures and connection types, each at 384 kbit/s, have been recommended in Recommendations I.211, I.412 and I.340, respectively.

Note 2 – Other systems are proposed and are the subject of Question 2/XV, for example, characteristics of equipment for coding of analogue sound-programme signals at 316 kbit/s insertion into a 320 kbit/s channel and multiplexing of six high quality sound-programme channels within 2048 kbit/s.

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.	EFF. RESOLUTION (BITS)	11 bit coding Allocation of character signals Variant A* 1	2 3 4	5 6 7 8 9 10	11	Variant B** 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	0 0 1	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 0 0

X = 11th bit freely available in variant A.

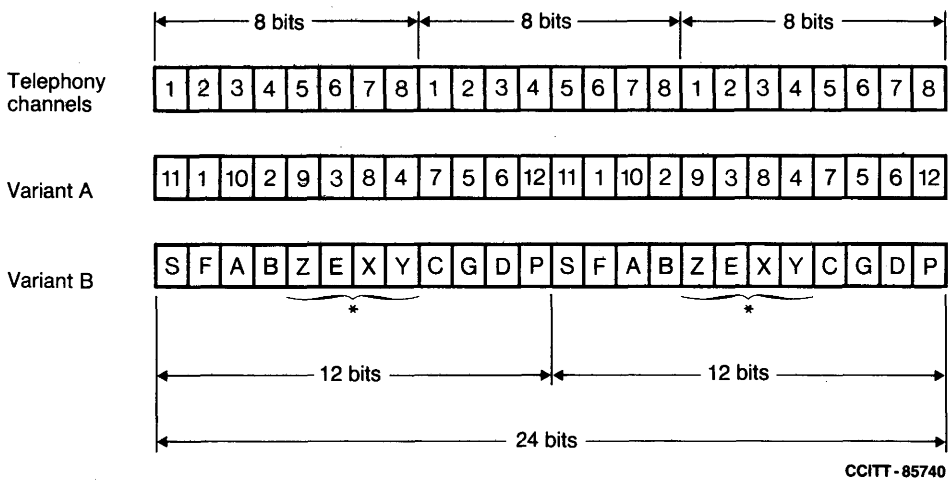
* Variant A is presently used with digital equipment based on a 2048 kbit/s hierarchy. After coding and before the parity bit is inserted, bits 1 to 5 are inverted.

** 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).

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 respectively) are inverted.

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 it 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.



Notes – Variant A: BIT Definitions:
1 Sign bit
2, 3, 4 Chord bit
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 the Annex 1 to Question 23/XVIII.

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. (See Annex 2 to Question 2/XV).

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.

It should be noted that a coding method having many fundamental parameters in common with this method has been proposed. This involves conversion of a 15 kHz channel into a 316 kbit/s stream and insertion of the 316 kbit/s stream into a 320 kbit/s stream (see Question 2/XV, Annex 3).

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.

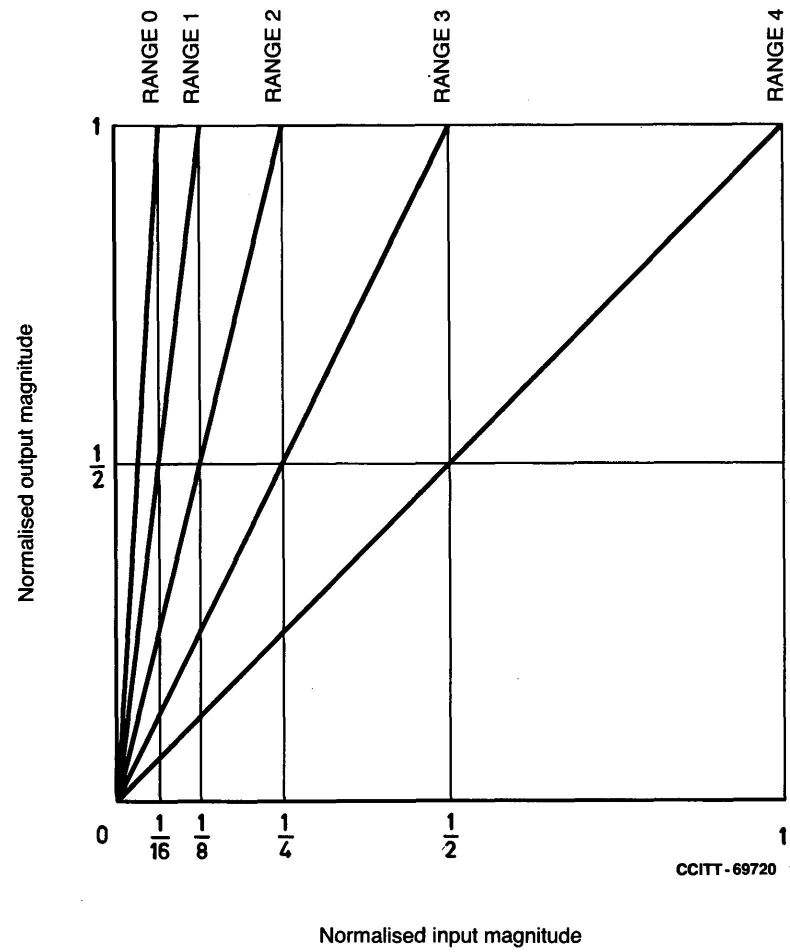


FIGURE 2/J.41
Near-instantaneous companding characteristic

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 msec 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:

$$R8 = (R3+R2+R1) \text{ MOD } 2$$

$$R9 = (R6+R5+R4) \text{ MOD } 2$$

$$R10 = (R7+R5+R4+R2+R1) \text{ MOD } 2$$

$$R11 = (R7+R6+R4+R3+R1) \text{ MOD } 2$$

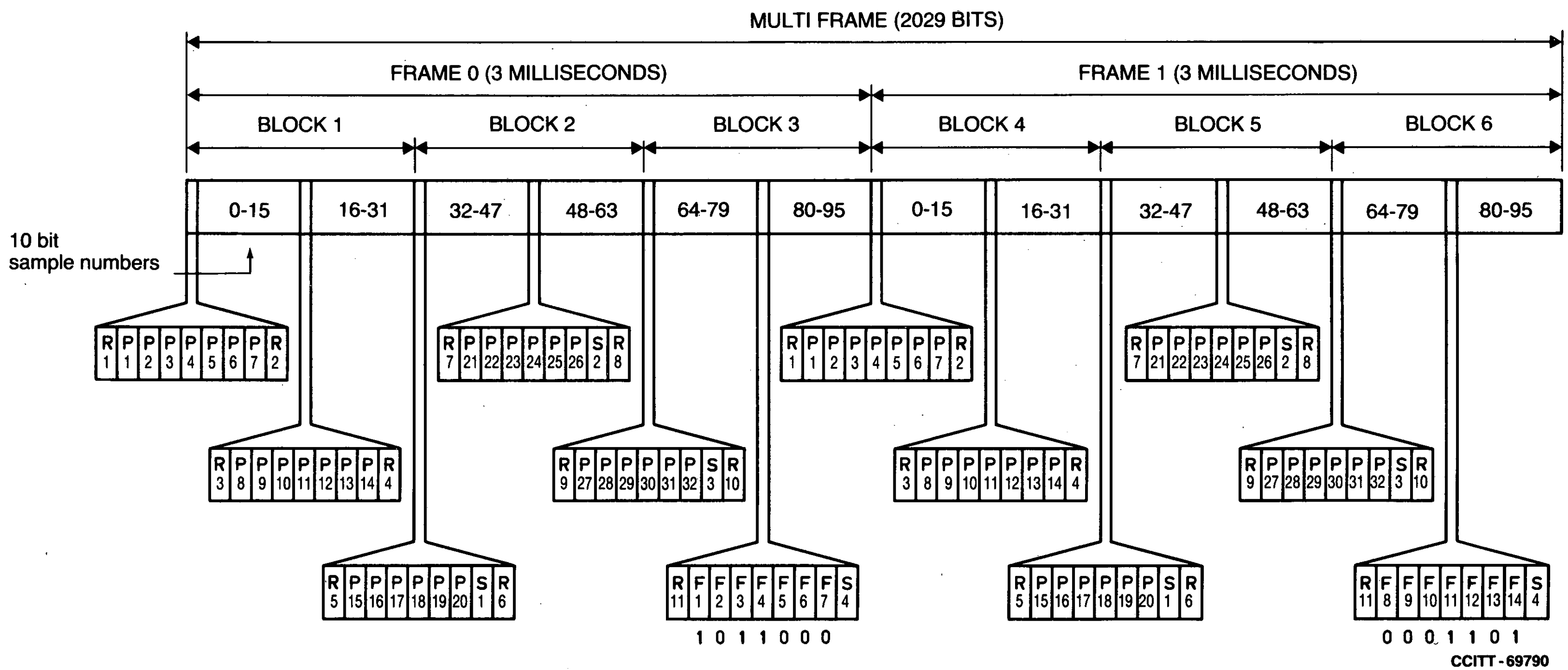
TABLE 2/J.41

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 msec. 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 (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*

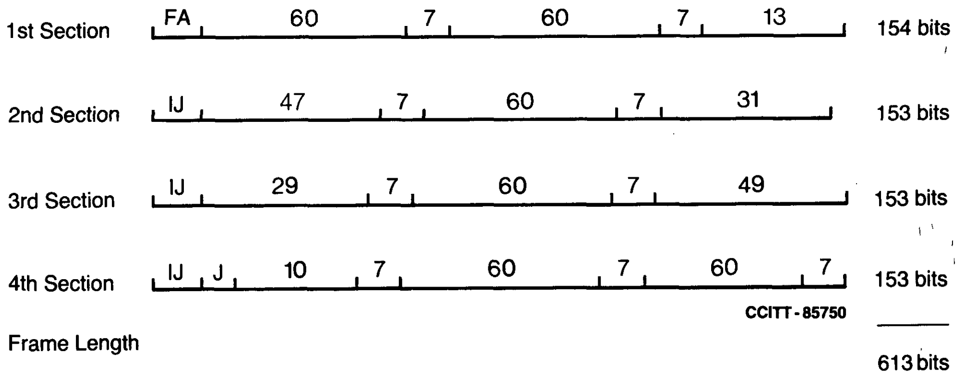
The first bits of sections 2, 3 and 4 are used to identify justification.

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.



FA: Frame Alignment Word: 1 1 1 0 1 0 0
IJ: bit for identification of justification
J: bit for justification

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.

Reference

[1] Document CMTT/133 (Booklet of the Interim Meeting of CMTT) *Transmission of analogue high-quality sound-programme signals on mixed analogue and digital circuits using 384 kbit/s channels,* Draft Recommendation AB/CMTT, November 1983.

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* 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** 384	352 32 0 384 384	kbit/s kbit/s kbit/s kbit/s kbit/s
Proposed by	Italy	Japan	

* f_0 = zero loss frequency of pre-emphasis.

** 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 647-2.

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

(See Notes 1 and 2)

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 either:

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 503-2) 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 1 – ISDN broadband channels, channel structures and connection types, each at 384 kbit/s, have been recommended in CCITT Recommendations I.211, I.412 and I.340.

Note 2 – Other systems are proposed and are the subject of Question 2/XV, for example, characteristics of equipment for coding of analogue sound-programme signals at 316 kbit/s insertion into a 320 kbit/s channel and multiplexing of twelve medium quality sound-programme channels within 2048 kbit/s.

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, the first 12 bit code word is allocated to 7 kHz channel No. 1 and the second 12 bit code word is 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, 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 Annex 1 to Question 23/XVIII.

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. (See Annex 2 to Question 2/XV.)

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.

It should be noted that a coding method having many fundamental parameters in common with this method has been proposed. This involves conversion of two 7 kHz channels into a 316 kbit/s stream and insertion of the 316 kbit/s stream into a 320 kbit/s stream (see Question 2/XV, Annex 3).

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

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

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

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

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

$$0 \leqslant n \leqslant 95$$

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).

$$1 \leqslant n \leqslant 3$$

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.

Reference

[1] Document CMTT/133 (Booklet of the Interim Meeting of CMTT) *Transmission of analogue high-quality sound-programme signals on mixed analogue and digital circuits using 384 kbit/s channels*, Draft Recommendation AB/CMTT, November 1983.

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 567-1, 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 567-1)

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 473-3)

Recommendation J.64

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

(Geneva, 1982)

(See CCIR Recommendation 569-1)

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 [1)]

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 ± 100 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).

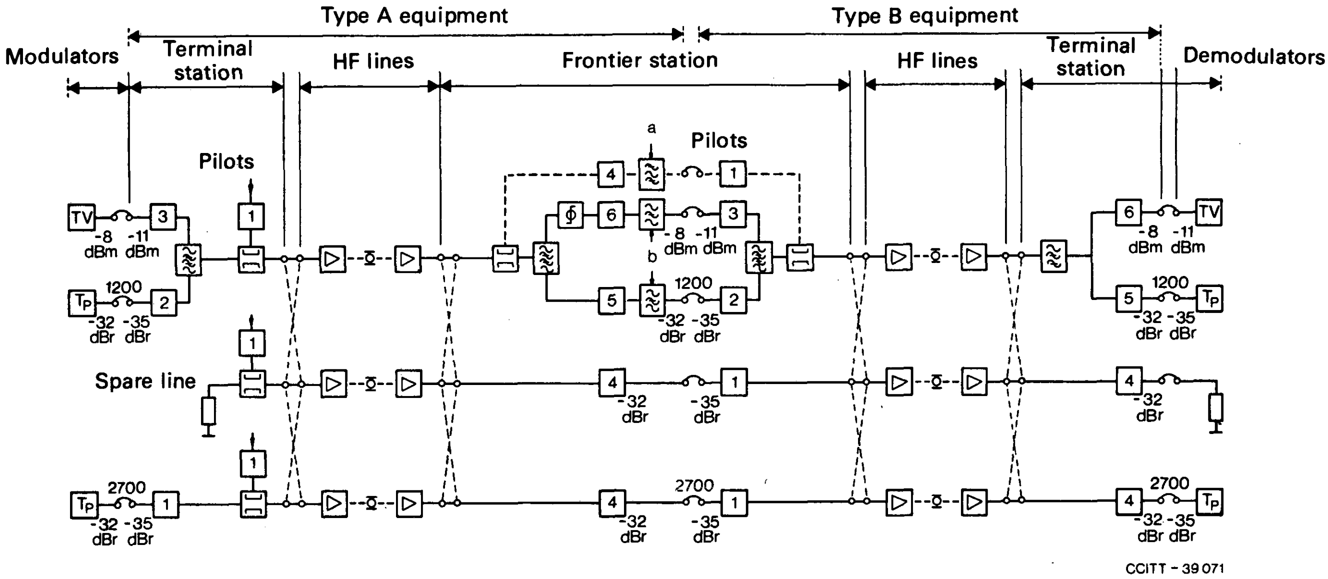
2 Modulation ratio

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 is 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 567-1, 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).

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



1 4 pre-emphasis (respectively de-emphasis) for telephony 2700 channels

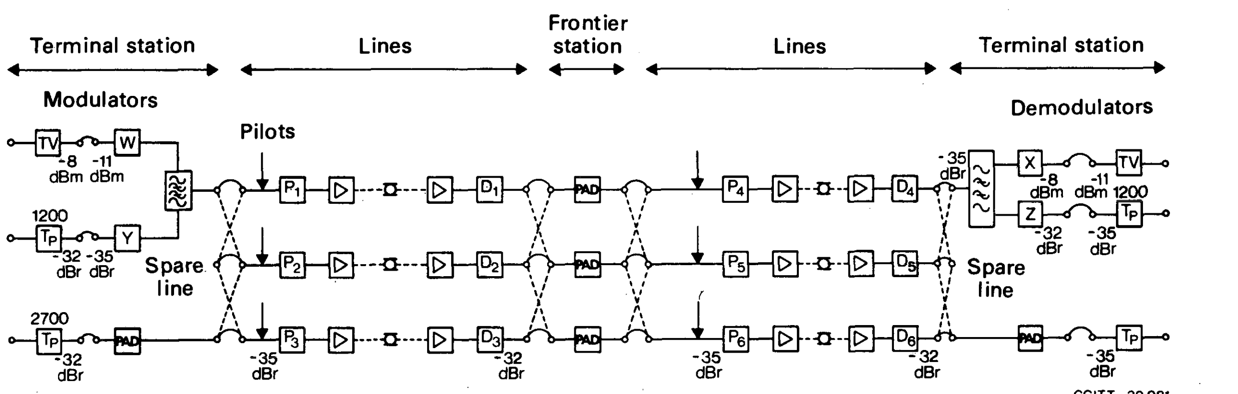
2 5 pre-emphasis (respectively de-emphasis) for telephony 1200 channels } simultaneous transmission

3 6 pre-emphasis (respectively de-emphasis) for television

a pilot pass-filter
b pilot stop-filter

FIGURE 1/J.73

General case of interconnection of 12-MHz lines



Complement of line

Nominal relative levels on the line for each telephone channel in the "all-telephony" case

W X differential pre-emphasis and de-emphasis for television

Y Z differential pre-emphasis and de-emphasis for telephony 1200 channels } common for all types of 12 MHz line

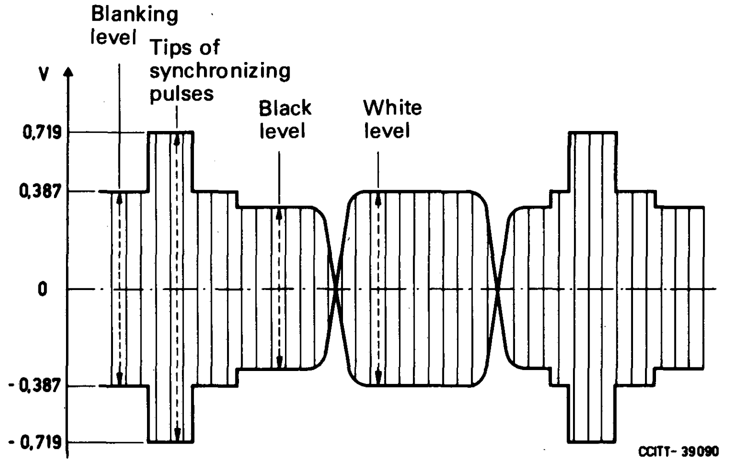
P_1 D_1 pre-emphasis and de-emphasis for telephony, 2700 channels, particular to each design of 12 MHz line

Notes to Figures 1/J.73 and 2/J.73

1. Interconnection of pilots, e.g. blocking and re-injecting or by-passing, should be agreed between Administrations.
2. The level of the line pilots is fixed at −10 dBm0 for the all-telephony case. When the line is used to transmit telephony and television simultaneously, different values of pre-emphasis may be required; although the absolute levels of the pilots will remain the same, they may no longer be at −10 dBm0.
3. The television levels shown are those of the modulated carrier, relative to the white or blanking level (0dBm) of the idealized reference signal described in § 2 of this Recommendation. This means that the television levels are indicated in dBm values.
4. The characteristics of the filters in Figure 1/J.73 (used for separating and combining the telephony and television bands so that the necessary arrangements for pre-emphasis and de-emphasis can be made) must be agreed between Administrations.

FIGURE 2/J.73

Showing use of differential emphasis networks to simplify interconnection of 12 MHz lines of different designs



Note – The voltages shown are the values measured at a zero relative level point for television transmission in the 12 MHz system.

FIGURE 3/J.73
Envelope of carrier modulated by Test Signal No. 2

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.

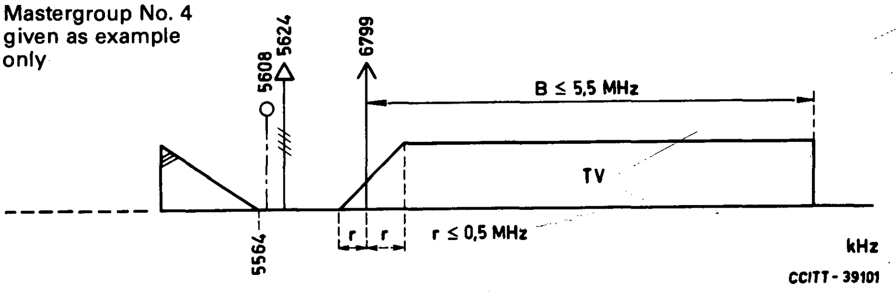


FIGURE 4/J.73
Frequency allocation for television on a 12 MHz system

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 567-1, 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.

– 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, Rec. G.423.

Recommendation J.77 [1)]

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 624-2 [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.

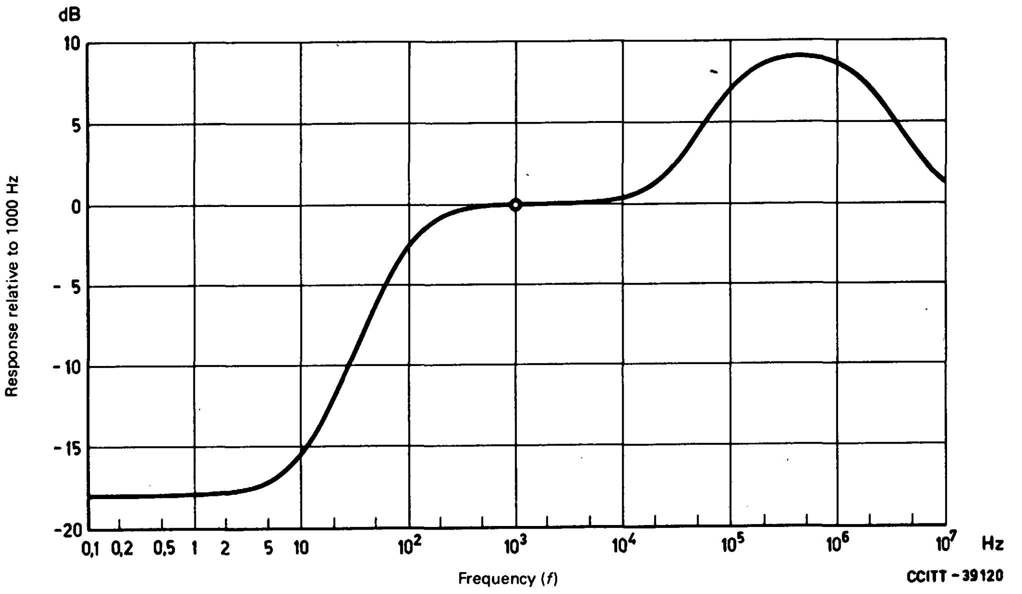
2 Video pre-emphasis

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.

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



Video pre-emphasis: $10 \log_{10} (1 + a) + 10 \log_{10} \left[1 + \frac{a}{\left(\frac{Q}{V}\right)^2 + 1} \right]$

where

$V = \frac{f}{f_0} - \frac{f_0}{f}$ $\qquad Q = 14.5$

$a = 7$ $\qquad f_0 = 450$ kHz

Low frequency suppression: $-10\ _{10}\ \frac{b^2 + (2\pi\tau f)^2}{1 + (2\pi\tau f)^2}$

where

$b = 8$
$\tau = 14$ ms

FIGURE 1/J.77

Frequency response of video pre-emphasis and low frequency suppression relative to the value at 1 kHz

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 624-2, 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.

PART III

SUPPLEMENTS TO H AND J SERIES RECOMMENDATIONS

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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 in Fascicle III.2)

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 Volume III of the *Green Book*, Geneva, 1972.)

Supplement No. 16

OUT-OF-BAND CHARACTERISTICS OF SIGNALS APPLIED TO LEASED TELEPHONE-TYPE CIRCUITS

(Geneva, 1980; referred to in Recommendation H.51)

Joint Working Party LTG collected for information some data on the out-of-band power of signals applied to leased telephone-type circuits.

The following constitutes a summary of the data collected so far.

1 Out-of-band components accompanying signals applied to voice-band leased circuits (Contribution of the United Kingdom Post Office)

In the United Kingdom it is considered essential to restrict the level of out-of-band components accompanying voice-band signals for the following purposes:

1) to allow the harmonious coexistence (as far as possible) in the local paired network of an increasing variety of services, e.g., subscriber carrier systems, visual telephone, data, etc., all of which are susceptible to crosstalk from wanted (or unwanted) signals applied to other pairs in the local network;
2) to reduce interference with adjacent channels where voice-band signals are extended over normal carrier telephone systems;
3) to reduce the amount of out-of-band interference thrown back into voice-band signals when these are extended over PCM systems.

The out-of-band components considered can originate in several ways: for example, as out-of-band components generated along with the voice-band signals themselves, such as harmonics, or as insufficiently suppressed by-products of encoding processes.

Taking into account the effect of the factors described above and the characteristics of the signals concerned, limits have been derived for the spectral distribution of energy for the out-of-band components with which voice-band attachments must comply before they are allowed to be connected to the network. The same limits serve to indicate the level of unrelated out-of-band signals which may be presented to the receiving attachment. The limits currently used in the United Kingdom are shown in Figure 1 and these illustrate one way of specifying limiting values derived from such studies.

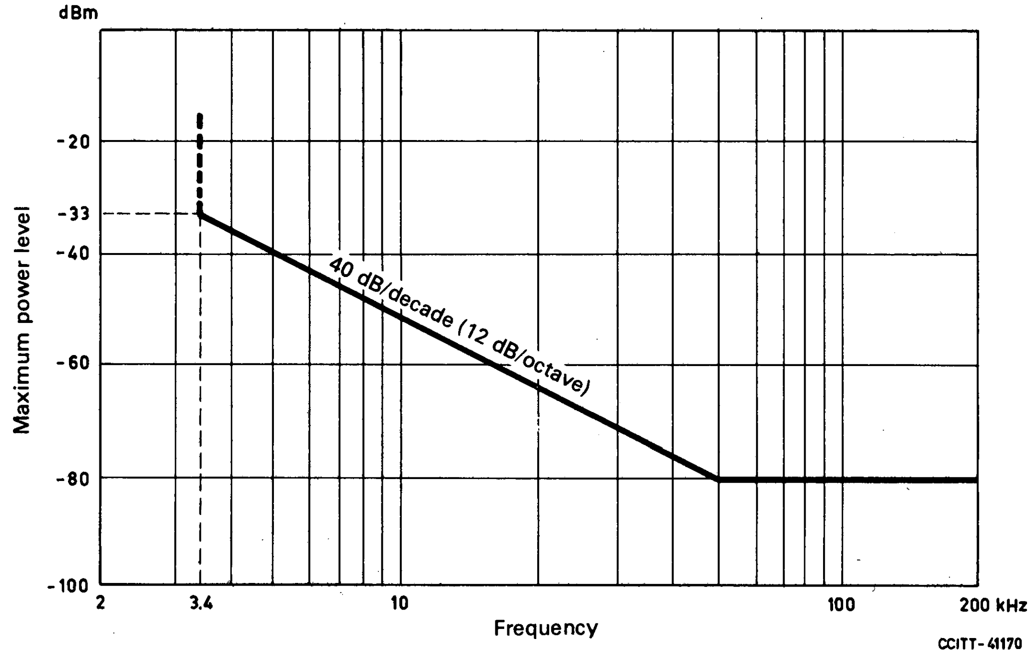

FIGURE 1

Maximum power level of individual spectral components above 3.4 kHz of the output signal from apparatus connected to voice-band circuits

As an example of a particular study (but one which is not claimed as exhaustive), consideration is given here to the implicit interaction of PCM channelling equipment with some other services and systems, arising from the limits given in the Recommendation cited in [1].

Some comments arising from this example are also included.

It is assumed that the audio input/output terminals of PCM equipment may be connected:

i) to a local distribution pair,

ii) to another PCM multiplex, or

iii) to an FDM multiplex.

The connection may be permanent (in the case of private circuits) or switched for the duration of a call. In the case of the distribution pair this may be carrying at the same time an HF service such as a 1 + 1 subscriber carrier system.

2 Rule adopted on the French network concerning limitation of the out-of-band power for transmission of signals of services other than telephony (Contribution of the French Administration)

The rule at present applied on the French network regarding the out-of-band send power spectrum of signals of services other than telephony (facsimile, phototelegraphy, data, telegraphy, etc.) transmitted on telephone-type circuits is as follows:

P_{0-4} the power of the signal transmitted to the line by the subscriber equipment in the 0-4 kHz band

P_{4-8} the power in the 4-8 kHz band

P_{8-12} the power in the 8-12 kHz band

$P_{4n-4(n+1)}$ the power in the band 4n-4 (n+1) kHz.

The power spectrum measured at the sending end, in the most unfavourable case, i.e., for the signal transmitted to the line with the widest spectrum, must be such that:

$$10 \log_{10} \frac{P_{0-4}}{P_{4-8}} \geqslant 20 \text{ dB}$$

$$10 \log_{10} \frac{P_{0-4}}{P_{8-12}} \geqslant 35 \text{ dB}$$

$$10 \log_{10} \frac{P_{0-4}}{P_{4n-4(n+1)}} \geqslant 55 \text{ dB}$$

for n integer $\geqslant 3$.

These conditions are shown in Figure 2.

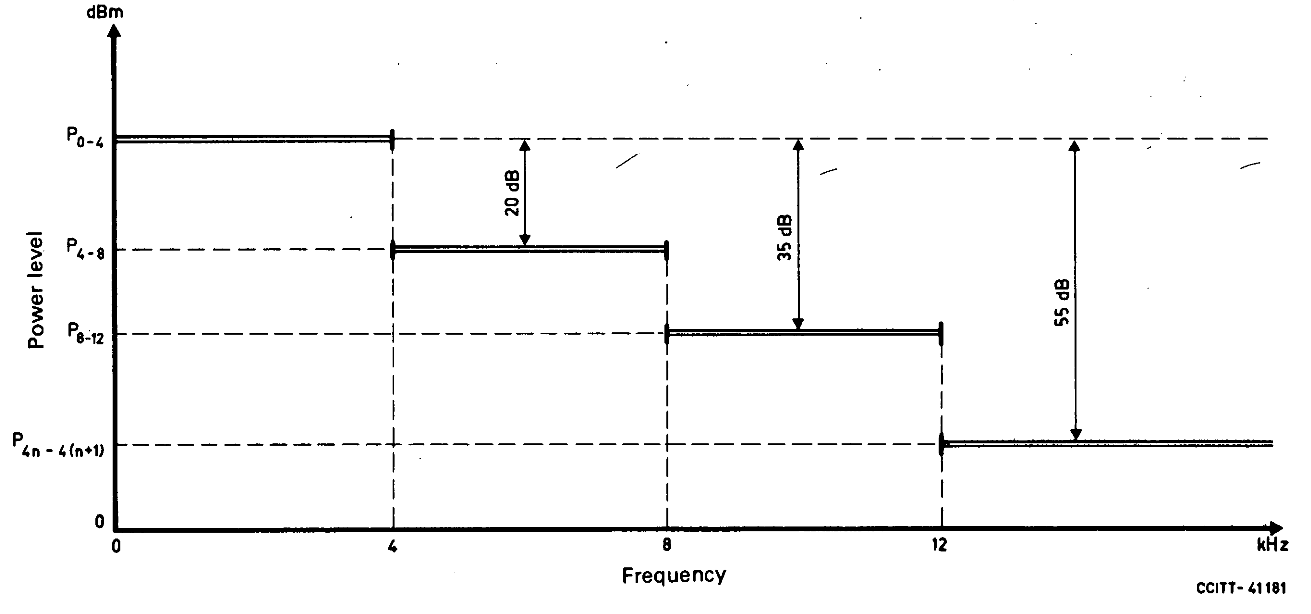

FIGURE 2

3 **Limitation of the out-of-band power spectrum of signals transmitted over telephone-type circuits** (Contribution of the Nippon Telegraph and Telephone Public Corporation)

3.1 *General*

Recommendation V.15 [2] gives limits for signal power outside the 0-4 kHz band from acoustic coupling equipments for data transmission.

NTT considers that these limits can also be complied with by terminal equipments other than acoustic coupling equipments.

Therefore, the rule applied by NTT for both digital and analogue circuits in this connection is based on Recommendation V.15 [2].

3.2 *The rule*

The rule is as follows:

The signal power outside the 0-4 kHz band shall not exceed the following values:

p – 20 dB in the 4-8 kHz band

p – 40 dB in the 8-12 kHz band

p – 60 dB in any 4 kHz band above 12 kHz

where p is the signal power in the 0-4 kHz band.

3.3 *Remark*

NTT is of the opinion that the rule under discussion should be basically in accordance with Recommendation V.15 [2].

References

[1] CCITT Recommendation *Performance characteristics of PCM channels at audio frequencies*, Vol. III, Rec. G.712, §§ 5.1, 6.1, 6.2, 7.1 and 7.2.

[2] CCITT Recommendation *Use of acoustic coupling for data transmission*, Vol. VIII, Rec. V.15.

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