SECTION 2

PERFORMANCE CHARACTERISTICS OF

SOUND-PROGRAMME CIRCUITS

Recommendation J.21

PERFORMANCE CHARACTERISTICS OF 15 kHz TYPE

SOUND-PROGRAMME CIRCUITS

Circuits for high-quality monophonic

and stereophonic transmissions

(Geneva, 1972; amended at Geneva, 1976 and 1980, and

at Melbourne, 1988)

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 meet the requirements laid out below.

1 Application

This Recommendation corresponds to CCIR Recommendation 505.
For the definition of absolute power, relative power and noise levels, see CCIR Recommendation 574.

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.

2 Interface characteristics

2.1 *Test conditions*

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 ?73 resistive.

2.2 *Impedance*

System input impedance 600 ?73, balanced System output impedance, provisionally Low, 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 ?73 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 § 3.2.2. This aspect needs further study.

2.3 *Levels*

Input maximum programme level +9 dBm0s Insertion gain (1 kHz at —12 dBm0) 0 dB Adjustment error, within ± | .5 dB Variation over 24 hours not to exceed ± | .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.

3 Overall performance

3.1 *Common parameters*

3.1.1 *Gain/frequency response*

Reference frequency 1 kHz (nominal value) The response shall be measured at —12 dBm0s

The gain/frequency response is given in Table 1/J.21.

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

H.T. [T1.21]

TABLE 1/J.21

The tolerance, permitted reactance and degree of unbalance need further study.

Frequency (kHz)	Response (dB)
{ 0.04 f < 0.125 }	+0.5 to —2.0
{ 0.125 f 10 | }	+0.5 to —0.5
{ 10 | < f 14 | }	+0.5 to —2.0
{ 14 | < f 15 | }	+0.5 to —3.0

tableau 1/J.21 [T1.21], p.

3.1.2 *Group delay variation*

Difference ?63τ, between the value of group delay at certain frequencies and the minimum value is given in Table 2/J.21. Between the points defined in Table 2/J.21, the tolerance limit varies linearly on a linear-delay/logarithmic frequency diagram.

H.T. [T2.21]

TABLE 2/J.21

kHz	?63τ (ms)
0.04	55
0.075	24
14 |	8
15 |	12

tableau 2/J.21 [T2.21], p.

3.1.3 *Noise*

The measurement to be made with an instrument conforming to CCIR Recommendation 468.

For radio-relay systems the requirements of Table 3/J.21 shall be met for at least 80% of the total time of any 30-day period. For 1% of the time an additional impairment of 4 dB, and for 0.1% of the time an additional impairment of 12 dB is acceptable.

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_0 400 Hz, *a* ≥" 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

Note — For digital systems appropriate values are under study. For further information see CCIR Report 647.

H.T. [T3.21]

TABLE 3/J.21

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
{ Idle channel noise (dBq0ps), max }	—42	—51
{ Programme-modulated noise (dBq0ps), max }	—30	—39

Tableau 3/J.21 [T3.21], p.

3.1.4 *Single tone interference*

Level of any individual tone:

(—73 + ?61) dBm0s

where ?61 is the weighting factor (positive or negative) as per CCIR Recommendation 468 at the particular frequency.

Administrations are urged to supply additional information on an appropriate value.

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.

3.1.5 *Disturbing modulation by power supply*

The level of the strongest unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains shall be less than —45 dBm0s with a test signal of 1 kHz at alignment level 0 dBm0s.

3.1.6 *Non-linear distortion*

3.1.6.1 *Harmonic distortion*

The total harmonic distortion (THD) shall be measured with the input signal at +9 dBm0s for frequencies up to 2 kHz at +6 dBm0s for frequencies above 2 kHz up to 4 kHz.

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 THD when measured with a true-RMS meter shall not be less than the following requirements shown in Table 4/5.21.

H.T. [T4.21]

TABLE 4/J.21

Input frequency (kHz)	Total harmonic distorsion	{ Second and third harmonic measured selectively }
{ 0.04 f < 0.125 }	1% | (—31 dBm0s)	0.7% (—34 dBm0s)
{ 0.125 f 2.0 }	0.5% (—37 dBm0s)	0.35% (—40 dBm0s)
{ 2.0 < f 4.0 }	0.5% (—40 dBm0s)	0.35% (—43 dBm0s)

tableau 4/J.21 [T4.21], p.

3.1.6.2 *Intermodulation*

With input signals at 0.8 kHz and 1.42 kHz, each at a level of +3 dBm0s, the third order difference tone at 0.18 kHz shall be less than 0.5% (—43 dBm0s).

Note — Attention is drawn to the fact that 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, assuming no pre-emphasis, the additional requirements of Table 5/J.21 apply.

H.T. [T5.21]

TABLE 5/J.21

{ 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) }	{

tableau 5/J.21 [T5.21], p.

3.1.6.3 *Distortion products measured by shaped noise*

Under study. See CCIR Report 640 (Kyoto, 1978).

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. the left and right channels of a stereo signal, 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.

3.1.8 *Intelligible cross-talk ratio*

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 values of Table 6/J.21.

H.T. [T6.21]
TABLE 6/J.21

Frequency (kHz)	Crosstalk attenuation (dB)
{ **0.04 <** *f* = 0.04 }	50
{ 0.04 < *f* < 0.05 }	{
Oblique straight-line segment on linear-decibel and logarithmic-frequency scales }	
{ 0.05 *f* 5**.** }	74
{ 5 | < *f* < 15**.** }	{
Oblique straight-line segment on linear-decibel and logarithmic-frequency scales }	
{ **15.** < *f* = 15**.** }	60

tableau 6/J.21 [T6.21], p.

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. (Administrations are invited to submit contributions on methods for measuring this parameter.)

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 at low frequencies (e.g. below about 100 kHz) on certain coaxial cable carrier systems. 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 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.

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 ± | .5 dB.

3.2 *Additional parameters for stereophonic programme transmission*

3.2.1 The difference in gain between A and B channels shall not exceed the values in Table 7/J.21.

H.T. [T7.21]
TABLE 7/J.21

Frequency (kHz)	Gain difference (dB)
{ 0.04 f < 0.125 }	1.5
{ 0.125 f 10 | }	0.8
{ 10 | < f 14 | }	1.5
{ 14 | < f 15 | }	3.0

tableau 7/J.21 [T7.21], p.

3.2.2 The phase difference between the A and B channels shall not exceed the values in Table 8/J.21.

H.T. [T8.21]
TABLE 8/J.21

Frequency (kHz)	Phase difference (degrees)
{ **0.04 <** *f* = 0.04 }	30
{ 0.04 < *f* < 0.2 }	{
Oblique straight-line segment on linear-degree and logarithmic-frequency scales }	
{ 0.2 *f* 4**.** }	15
{ 4**.** < *f* < 14**.** }	{
Oblique straight-line segment on linear-degree and logarithmic-frequency scales }	
{ **0.04 <** *f* = 14**.** }	30
{ 14**.** < *f* < 15**.** }	{
Oblique straight-line segment on linear-degree and logarithmic-frequency scale }	
{ **0.04 <** *f* = 15**.** }	40

tableau 8/J.21 [T8.21], p.

3.2.3 The cross-talk ratio between the A and B channels shall not be less than the following limits:

3.2.3.1 Intelligible cross-talk ratio, measured with sinusoidal test signal 0.04 to 15 kHz: 50 dB.

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 Table 9/J.21.

H.T. [T9.21]
TABLE 9/J.21

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

tableau 9/J.21 [T9.21], p.

3.3 *Additional requirements for digital systems*

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 offset. Recommendation O.33 recommends 1020 Hz.

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.

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 signal (f_o) is combined with the inband audio signals (f_i) or out-of-band interfering signals (f_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 f_i/f_d | values in Table 10/J.21 is sufficient.

H.T. [T10.21]
TABLE 10/J.21

	n = 2	*n* = 3		
f (kHz)	*9*	*13*	*7*	*11*
f (kHz)	*14*	*6*	*11*	*1*

tableau 10/J.21 [T10.21], p.

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 f_a/f_d | values in Table 11/J.21 is sufficient.

H.T. [T11.21]
TABLE 11/J.21

	$n = 1$	$n = 2$		
f (kHz)	*31*	*33*	*63*	*65*
f (kHz)	*1*			

tableau 11/J.21 [T11.21], p.

3.3.4 *Further parameters*

Characteristics for bit errors, clicks, jitter, etc. are under study. (See Study Programme 18A/CMTT and CCIR Report 647.)

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

PERFORMANCE CHARACTERISTICS OF 10 kHz TYPE

SOUND-PROGRAMME CIRCUITS

(The text of this Recommendation can be found in

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

Recommendation J.23

PERFORMANCE CHARACTERISTICS OF 7 kHz TYPE | (NARROW-BANDWIDTH)

SOUND-PROGRAMME CIRCUITS

’ | ’ | ’ |

Circuits of medium quality for monophonic transmission

(amended at Geneva, 1980 and at Melbourne, 1988)

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 below.

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.

2 Interface characteristics

This Recommendation corresponds to CCIR Recommendation 503. CCIR has agreed, at its XVIth Plenary Assembly, Dubrovnik, 1986, that CCIR Recommendation 504-2 will not be published in the next CCIR book.

For the definition of absolute power, relative power and noise levels, see CCIR Recommendation 574.

Sound-programme circuits of the 5 kHz type are widely used in North America.

6.4 kHz-type narrow bandwidth sound-programme circuits are still being used in some countries.

2.1 *Test conditions*

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 ?73 resistive.

2.2 *Impedance*

System input impedance 600 ?73, balanced System output impedance, provisionally Low, 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 ?73 max. (provisional value) within the nominal frequency range.

The tolerance, permitted reactance and degree of unbalance need further study.

2.3 *Levels*

Input maximum programme level +9 dBm0s Insertion gain (1 kHz at —12 dBm0) 0 dB
Adjustment error, within ± | .5 dB Variation over 24 hours not to exceed ± | .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.

3 Overall performance

3.1 *Common parameters*

3.1.1 *Gain/frequency response*

Reference frequency 1 kHz (nominal value) The response shall be measured at —12 dBm0s

The gain/frequency response is given in Table 1/J.23.

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

H.T. [T1.23]
TABLE 1/J.23

Frequency (kHz)	Response (dB)
0.05 *f* < 0.1	+1 to —3
{ 0.1 *f* 6.4 }	+1 to —1
{ 6.4 < *f* 7. }	+1 to —3

tableau 1/J.23 [T1.23], p.

3.1.2 *Group delay variation*

Difference ?63τ, between the value of group delay at certain frequencies and the minimum value is given in Table 2/J.23. Between the points defined in Table 2/J.23, the tolerance limit varies linearly in a linear-delay/logarithmic-frequency diagram.

H.T. [T2.23]
TABLE 2/J.23

Frequency (kHz)	?63τ (ms)
0.05	80
0.1	20
6.4	5
7.	10

tableau 2/J.23 [T2.23], p.

3.1.3 *Noise*

The measurement is to be made with an instrument conforming to CCIR Recommendation 468.

For radio-relay systems the requirements of Table 3/J.23 shall be met for at least 80% of the total time of any 30-day period. For 1% of the time an additional impairment of 4 dB, and for 0.1% of the time an additional impairment of 12 dB is acceptable.

H.T. [T3.23]

TABLE 3/J.23

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
{ Idle channel noise, max (dBq0ps) }	—44	—49
{ Programme-modulated noise, max (dBq0ps) }	—32	—37

tableau 3/J.23 [T3.23], p.

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_0 400 Hz, *a* ≥" 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.

Note — For digital systems appropriate values are under study. For further information see CCIR Report 647.

3.1.4 *Single tone interference*

Level of any individual tone:

(—73 + ?61) dBm0s

where ?61 is the weighting factor (positive or negative) 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.

3.1.5 *Disturbing modulation by power supply*

The level of the strongest unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains shall be less than —45 dBm0s with a test signal of 1 kHz at alignment level 0 dBm0s.

3.1.6 *Non-linear distortion*

Administrations are urged to supply additional information on an appropriate value.

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 THD when measured with a true-RMS meter shall not be less than the following requirements shown in Table 4/J.23.

H.T. [T4.23]

TABLE 4/J.23

Input frequency (kHz)	Total harmonic distorsion

0.05 f < 0.1
2%.
(—25 dBm0s)
}
{
0.1
f
2.0
}
	{
	
	1.4% (—28 dBm0s)

tableau 4/J.23 [T4.23], p.

Note — If THD cannot be measured directly, compliance is considered to be fulfilled if the second or third harmonics are measured selectively and a calculated value of k meets the requirement:

$$k = \sqrt{k\ \$\$Ei:2:2\_ + k\ \$\$Ei:2:3\_}$$

where k_2 is the second harmonic coefficient and k_3 is the third harmonic coefficient.

3.1.6.2 *Intermodulation*

With input signals of 0.8 kHz and 1.42 kHz, each at a level of +3 dBm0s, the third order difference tone at 0.18 kHz shall be less than 1.4% (—34 dBm0s).

3.1.6.3 *Distortion products measured by shaped noise*

Under study. See CCIR Report 640 (Kyoto 1978).

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.

3.1.8 *Intelligible cross-talk ratio*

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 values of Table 5/J.23.

H.T. [T5.23]
TABLE 5/J.23

108.if 508<53 .nr 80 53
108.if 508<53 .nr 80 53

Frequency (kHz)	Crosstalk attenuation (dB)
{ **0.5** *f* < 0.5 }	Slope 6 dB/octave
0.5 *f* 3.2	74
{ **0.5** *f* > 3.2 }	Slope —6 dB/octave

tableau 5/J.23 [T5.23], p.

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. (Administrations are invited to submit contributions on methods for measuring this parameter.)

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 at low frequencies (e.g. below about 100 kHz) on certain coaxial cable carrier systems. 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 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.

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.

3.2 *Additional parameters for stereophonic programme transmission*

Not applicable, this section concerns 15 kHz type sound-programme circuits (see Recommendation T.21).

3.3 *Additional requirements for digital systems*

3.3.1 If a test signal is harmonically related to the sampling frequency, measuring difficulties may arise. In this case the nominal 1 kHz test signal must be offset. The Recommendation O.33 recommends 1020 Hz.

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.

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 signal (f_o) is combined with the inband audio signals (f_i) or out-of-band interfering signals (f_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 f_i/f_d | values in Table 6/J.23 is sufficient.

H.T. [T6.23]

TABLE 6/J.23

	n = 2	n = 3		
f (kHz)	*5*	*7*	*3*	*5*
f (kHz)	*6*	*2*	*7*	*1*

tableau 6/J.23 [T6.23], p.

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 f_a/f_d | values in Table 7/J.23 is sufficient.

H.T. [T7.23]

TABLE 7/J.23

	n = 1	n = 2		
f (kHz)	*15*	*17*	*31*	*33*
f (kHz)	*1*			

tableau 7/J.23 [T7.23], p.

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).

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 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.

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.

Figure 1/J.31, (M), p.

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 ± | Hz and the programme-channel pilot is fed in as 16 | 00 ± 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, (M), p.

Figure 3/J.31, (M), p.

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*) 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 ± | 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, (M), p.

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 ± | .5 dB, or ± | .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 | Bm0 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

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dBm0(*t*) denotes that the level quoted is referred to a zero relative level point in a telephone channel.

and a test frequency is chosen which gives rise to an intermediate frequency 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.

H.T. [T1.31]

TABLE 1/J.31

Compressor characteristic

{ Programme signal level at the compressor input (dBm0) } Compressor gain (dB) (tolerance ± | .5 dB except at the point marked | where the tolerance is ± | .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 |

Tableau 1/J.31 [T1.31], p.

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 ± | 0%.

Note — Since the initial and final values of the compressor output voltage in the case of this compandor are not in a 1 | | 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 — ?63ps) 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: | (+- | 50 Hz

at 14 kHz: | (+- | 10 Hz

Attenuation for the midfrequencies

at 10 kHz: ≥" | 6 dB

at 14 kHz: ≥" | 2 dB

Note — The attenuation of these bandstop filters is sufficient without taking account of the compandor advantage.

The stopband attenuations should be maintained within ± | 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.

Figure 5/J.31, (M), p.

Figure 6/J.31, (M), p.

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^{D}lF261^{5}, as is documented by several Administrations (see CCIR Report 491 [5]). For the telephone service, the level with a probability of 10^{D}lF261^{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.

H.T. [T2.31]

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 | ua)			
}	—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			
2			
different stereophonic programmes			
2			
equal stereophonic programmes			
}	+3.5 +3.5 +3.5	{	
+14			
+18 programme channels only			
+22			
}			
10 programme channels			
{			
10			
different programmes			
5			
different stereophonic programmes			
2			
equal stereophonic programmes + 6 different monophonic programmes			
}	{		
+**4.5**			
+**4.5**			
+**4.5**			
}	{		
+15			
+19			
+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[D]lF261[5]).

a) Loading by one stereophonic programme is treated as loading by two identical monophonic programmes (worst case).

Tableau 2/J.31 [T2.31], p.

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 — ?63ps) dBm0s on the programme circuit 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 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 — ?63ps) dBm0s taking account of the amount of the narrow-band crystal stop-filter attenuation.

This value has been specified in Recommendation J.21 by CMTT. CCIR Report 493 [10] gives some additional information regarding the subjective impairments produced by interfering frequencies on a circuit using equipment conforming to Recommendation J.31.

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

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 — ?63ps) dBm0s mentioned above is met.

Figure 7/J.31, (M), p.

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.

Figure 8/J.31, (M), p.

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*

H.T. [T3.31]
TABLE A-1/J.31

Modulated programme frequencies }	{ Compandor control channel	Synchronizing pilot	
Channel A (inverted) 65**,** }	{ 81.39 . | | 83.18 kHz	84 kHz	
Channel B (erect) 88.04 kHz }	{ 84.82 . | | 86.61 kHz		

Tableau A-1/J.31 [T3.31], p.

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]).

H.T. [T4.31]

TABLE A-2/J.31

{		{	
		Channel A	Channel B
—∞	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 ± | ,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.

Tableau A-2/J.31 [T4.31], p.

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 ± | 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 Societ`a 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, 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, 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.

Recommendation J.32

CHARACTERISTICS OF
EQUIPMENT AND LINES USED FOR

SETTING UP 10 kHz TYPE SOUND-PROGRAMME CIRCUITS

(The text of this Recommendation can se found

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

Recommendation J.33

CHARACTERISTICS OF
EQUIPMENT AND LINES USED |
FOR SETTING UP

6.4 kHz TYPE SOUND-PROGRAMME CIRCUITS

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

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*

The performance characteristics of 6.4 kHz type sound-programme circuits are given in Recommendation J.23 (Yellow Book, 1980).
For the choice of groups and supergroups used, see Recommendation J.32.

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) be lower than that of telephone relative level (dBr) by 6.5 dB and emphasis network be applied.

In order to meet the requirement of —39 dBm0s noise level of 2500 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, Recommen dation 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 ± 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.

Figure A-1/J.33, (M), p.

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 sound-programme weighted network (0.05 to 6.4 kHz) 9.0 dB CCIR Recommendation 468 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]) 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 ± | 0^{D}lF261^{5} and the programme-channel pilot is fed in as 7833 1/3 Hz (stability better than ± | 0^{D}lF261^{5}) in the audio-frequency position.

Figure 1/J.34, (M), p.

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.

H.T. [T1.34]

TABLE 1/J.34

Channel range (kHz) Virtual carrier frequency | ua) (kHz) }	{
60 to 72 72 to 84 84 to 96 Inverted position 94**.5** } 96 to 108	Inverted position 70.5 Inverted position 82.25 { Inverted position 105.75

a) The carrier frequencies are multiples of 11.75 kHz and can be derived from a common generator frequency.

Tableau 1/J.34 [T1.34], p.

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 ± | .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 ± | .5 dB, but it is ± | .1 dB at programme signal levels at the compressor input of —∞, —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, 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, § 3.3.1, ITU, Geneva, 1978.