INTERNATIONAL TELECOMMUNICATION UNION

CCITT

THE INTERNATIONAL
TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE

BLUE BOOK

VOLUME III – FASCICLE III.1

GENERAL CHARACTERISTICS OF INTERNATIONAL TELEPHONE CONNECTIONS AND CIRCUITS

RECOMMENDATIONS G.101-G.181

IXTH PLENARY ASSEMBLY
MELBOURNE, 14-25 NOVEMBER 1988

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PRELIMINARY NOTES

1 In this fascicle, the expression "Administration" is used for shortness to indicate both a telecommunication Administration and a recognized private operating agency.

2 Recommendations G.106, G.107 and G.108 of the *Red Book* have been transferred to Fascicle II.3 of the *Blue Book* under the numbers E.800, E.810 and E.830 respectively.

3 In this fascicle the use of "Reference Equivalent" (RE) and "Correction Reference Equivalent" (CRE) has been discontinued and only Loudness Rating (LR) values are cited in the Recommendations. However, a full discussion can be found in the *Red Book*, Fascicle III.1, on the relation between REs, CREs, and LRs.

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Note – This index provides a partial list of Recommendations pertaining to transmission performance. It is not a complete index of all performance Recommendations, and reference should be made to the Series P Recommendations for information regarding transmission quality.

TABLE 1

Summary of main characteristics specified by the CCITT for international telephone circuits [a] and international connections

(This very condensed table is not a Recommendation, and reference should be made to the complete Recommendations)

		For an international circuit (1)	For a complete connection or for its parts (2)
Loudness ratings		G.111, § 2	For the connection and for the national systems G.111, § 1; G.121
Nominal 4-wire (transmission plan, see G.101)		0.0 dB G.101 – Digital 0.5 dB G.101 – Analogue Echo effects (G.131, § 2)	Four-wire chain national circuits: G.101, § 2.2, G.121, G.122
Transmission stability		G.131, § 1	Balance return loss of national network (G.122)
Band of frequencies effectively transmitted	Limits in Hz	At least 300-3400 (G.151, § 1)	Four-wire chain of 6 circuits: 300-3400 (G.132)
	Additional attenuation at limits of frequency	9 dB (G.151, § 1 and G.132)	9 dB (G.151, § 1 and G.132)
Attenuation/frequency distortion		G.151, § 1; Figure 1/G.151	Objective for 12 circuits (Figure 1/G.132) For data: see H.12
Group delay (t)		G.114	For the connection (G.114) $t \leqslant 150$ ms, without reservation $t \leqslant 400$ ms, acceptable with conditions For data: see H.12
Phase distortion (from the group delay t)		$t_m - t_{min} \leqslant 30$ ms [b] $t_M - t_{min} \leqslant 15$ ms [b] (G.133)	For the 4-wire chain (G.133) $t_m - t_{min} \leqslant 60$ ms $t_M - t_{min} \leqslant 30$ ms For each national 4-wire chain: (G.133) $t_m - t_{min} \leqslant 15$ ms $t_M - t_{min} \leqslant 7.5$ ms
Variation of overall loss with time		Mean deviation from nominal $\leqslant \pm 0.5$ dB Std. dev $\leqslant 1$ dB or 1.5 dB (G.151, § 3)	Extension circuits: as (1) (G.151) For data: see H.12
Linear crosstalk between different circuits (near- or far-end crosstalk ratio Δ)		$\Delta \geqslant 65$ dB (G.151, § 4, Notes 1 and 3)	Extension circuits: as (1) (G.151)
Near-end crosstalk ratio between the two directions of transmission		Ordinary circuits: $\geqslant 43$ dB (G.151, § 4) With speech concentrator: $\geqslant 58$ dB With echo suppressor: $\geqslant 55$ dB (G.151, § 4) (Note 4)	Extension circuits: as (1) (G.151)

TABLE 1 *(concluded)*

<table>
<tr><th colspan="2"></th><th>For an international circuit
(1)</th><th>For a complete connection
or for its parts
(2)</th></tr>
<tr><td colspan="2">Circuit noise</td><td>See Table 1 *bis*</td><td></td></tr>
<tr><td colspan="2">VF impedance of the channel translating equipment</td><td></td><td>Nominal value 600 ohms
(G.232, § 11.2)</td></tr>
<tr><td colspan="2">Frequency difference at two ends of a carrier circuit</td><td>$\leqslant$ 2 Hz (G.135, G.225)</td><td>G.135, G.225</td></tr>
<tr><td rowspan="4">Power at zero relative level point</td><td>Telephony, mean power in busy hour</td><td colspan="2">Speech currents, etc. 22 μW [c] (G.223)
Electric signals + tones 10 μW (G.223)
(see G.224 for the power of signalling pulses)</td></tr>
<tr><td>Voice frequency telegraphy.
Maximum power per channel for VFT systems having:
24 channels
18 channels
12 channels or less</td><td>Amplitude modulation. Power when sending continuous mark [H.23, § 2, a)]

9 μW
15 μW
35 μW</td><td>Frequency modulation mean power [H.23, § 2, b)]

5.6 μW
7.5 μW
11.25 μW</td></tr>
<tr><td>Private wire telegraphy and telephony</td><td colspan="2">Telegraphy level $\leqslant$ −13 dBm0 (H.32) [d]</td></tr>
<tr><td>Phototelegrapy</td><td colspan="2">Amplitude modulation −3 dBm0. Frequency modulation −13 dBm0 (H.41)</td></tr>
<tr><td colspan="2" rowspan="2">Maximum power for data transmission over leased circuits
(H.51, § 1) [d]</td><td colspan="2">1 mW on subscriber's line
Frequencies $\geqslant$ 2400 Hz, see G.224</td></tr>
<tr><td colspan="2">Frequency modulation: −13 dBm0 or −20 dBm0
Amplitude modulation: $\geqslant$ −13 dBm0</td></tr>
<tr><td colspan="2" rowspan="2">Maximum power for data transmission over circuits in the switched network
(H.51, § 2) [d]</td><td colspan="2">1 mW on subscriber's line
Frequencies: $\geqslant$ 2400 Hz, see G.224</td></tr>
<tr><td colspan="2">Frequency or phase modulation: $\geqslant$ 13 dBm0
Amplitude modulation: $\geqslant$ −13 dBm0</td></tr>
</table>

m = nominal minimum frequency effectively transmitted.

M = nominal maximum frequency effectively transmitted.

min = frequency corresponding to minimum group-delay time.

[a] Unless otherwise indicated, circuits for voice-frequency telegraphy or phototelegraphy have the same characteristics.

[b] These values apply to the chain of international circuits.

[c] Calculation target value or conventional value for a hypothetical reference circuit.

[d] This Recommendation contains restrictions of use. See also Recommendation H.34.

SUMMARY TABLES

TABLE 1 *bis*

Summary of noise objectives specified by the CCITT and the CCIR for telephone circuits

(This very condensed table is not a Recommendation,
and reference should be made to the complete Recommendations)

Types of systems				General objectives						
				Cable [a)] or radio-relay link			Single-hop satellite link	Submarine cable [a)]	All systems	
Telephone circuits considered [b)]				National 4-wire extension circuits and international circuits from 250 to 2500 km	Circuits of 5000 km	Circuits [c)] from 2500 to about 25 000 km	Circuits from 7500 to about 15 000 km	Circuits from 2500 to about 25 000 km	Chain of six international circuits	
Recommendations of the CCITT				G.152 G.212 [d)] G.222 G.226	G.215	G.153		G.153	G.143	G.143
Recommendations of the CCIR				391, 392 393, 395 396, 397			352 353			
Hypothetical reference circuit (HRC) or typical circuit considered				HRC of 2500 km [e)] or similar real circuit	Circuit of 5000 km [e)]	Circuit of 7500 km [e)]	Basic HRC of at least 7500 km		Chain of about 25 000 km	Chain of more than 25 000 km
Recom-mended objectives	Psopho-metric power	Hourly mean	Total power	10 000 pW					50 000 pW	
			Terminal equipment	25 000 pW	2500 pW0p				About 5 000 to 7 000 pW0p	
			Line	75 000 pW i.e. 3 pW/km	7500 pW0p 2 pW0p/km	15 000 pW [f)] 2 pW/km or better [g)]	10 000 pW [f)]	1 pW/km [g)]	About 1.5 pW/km	1 pW/km for each section longer than 2500 km
		For one minute exceeded during 20% of the month	Line	10 000 pW0p	5000 pW0p		10 000 pW [f)]			
		% of a month during which the psophometric power for one minute due to the line indicated can be exceeded	47 500 pW 50 000 pW 63 000 pW	0.1	0.1	0.3 [f)]	0.3 [f)]			
Unweighted power		% of the month during which 10^6 pW (5 ms) can be exceeded		0.1		0.3 [f)]	0.3 [f)]			

TABLE 1 *bis (concluded)*

Special objectives								
In national networks	Radio-relay links					Tropospheric radio-relay links in special conditions	Open-wire lines	
Noise due to the national transmission systems	Circuits not very different from HRC 280 < L < 2500 km	Composition of links very different from HRC: 50 ⩽ L < 280 km	280 < L ⩽ 840 km	840 < L ⩽ 1670 km	1670 < L ⩽ 2500 km	One or two circuits at most in one world connection	Up to 2500 km	More than 2500 km
G.123							G.311	G.153
CCIR Recommendations	395	395	395	395	395	396; 392		
Total length L in km of the longline FDM carrier systems in the national chain						HRC of 2500 km [e)]	HRC of 2500 km [e)]	Circuit of 10 000 km
(4000 + 4L) pW or (7000 + 2L) pW [h)]							20 000 pW [i)]	50 000 pW [i)]
							2500 pW	
	3 L pW	(3 L + 200) pW		(3 L + 400) pW	(3 L + 600) pW		17 500 pW	
	3 L pW	(3 L + 200) pW		(3 L + 400) pW	(3 L + 600) pW	25 000 km		
	$\frac{L}{2500} \times 0.1$	$\frac{280}{2500} \times 0.1$	$\frac{L}{2500} \times 0.1$	$\frac{L}{2500} \times 0.1$	$\frac{L}{2500} \times 0.1$	0.5		
						0.5		

[a] For these systems, it is sufficient to check that the objectives for the hourly mean is attained.

[b] Special objectives for telegraphy are indicated in Recommendations G.143, G.153 and G.442. Objectives for data transmission are shown in Recommendations G.143 and G.153.

[c] For some very large countries, refer to Recommendation G.222, § 3.

[d] See, in this Recommendation, the details of the hypothetical reference circuits to be considered.

[e] The objectives for line noise, in the same column, are proportional to the length in the case of shorter lengths.

[f] Provisionally.

[g] Objective 3 pW/km for the worst circuits; if a real circuit has more than 40 000 pW, it should be be equipped with a compandor.

[h] For planning purposes.

[i] Except in extremely unfavourable climatic conditions.

General comment – All the values mentioned in this table refer to a point of zero relative level of a telephone circuit set up on the system under consideration (of the first circuit, for the chain). Furthermore (G.123), the psophometric e.m.f. of noise induced by power lines should not exceed 1 mV at the "line" terminals of the subscriber's station. The mean value of the busy-hour noise power through a 4-wire national exchange: $\leq$ 200 pWp. Limits of unweighted noise through exchange: 100 000 pW.

TABLE 2

Summary of main characteristics specified by the CCITT for carrier terminal equipments

(This very condensed table is not a Recommendation, and reference should be made to the complete Recommendations)

	Systems wholly in cable (G.232)	Systems on open-wire lines	
		3-channel (G.361)	12-channel (G.232)
Level of carrier leak on the line: a) within the 60-108 kHz band – per channel – per group [a] b) outside the 60-108 kHz band	 ⩽ −26 dBm0 ⩽ −20 dBm0 ⩽ −50 dBm0	 ⩽ −17 dBm0 ⩽ −14.5 dBm0	 ⩽ −26 dBm0 ⩽ −20 dBm0 ⩽ −50 dBm0
Attenuation/frequency distortion	Figures 1/G.232 and 2/G.232		
Group delay	Table 1/G.232		
Non-linear distortion	Figure 3/G.232		
Amplitude limiting	Definition (G.232, § 8)		
Crosstalk ratio	⩾ 65 dB for intelligible crosstalk (G.232, § 9) ⩾ 60 dB for unintelligible crosstalk between adjacent channels (G.232, § 9)		
Near-end crosstalk ratio (A) between HF points	⩾ 47 dB without echo suppressors (G.232, § 9) ⩾ 62 dB with echo suppressors (G.232, § 9)		
Near-end crosstalk ratio (X) between audio points	⩾ 53 dB without echo suppressors (G.232, § 9) ⩾ 68 dB with echo suppressors (G.232, § 9)		
Relative levels	G.232, § 11; Table 2/G.232		
Impedance	600 Ω (G.232, § 12)		
Protection and suppression of pilots	G.232, § 13		

[a] When part of the group is transmitted over open-wire lines (see Recommendation G.232, § 5.1).

Note – See Recommendations G.234 and G.235 for 8-channel and 16-channel equipments, respectively.

TABLE 3

Summary of main characteristics specified by the CCITT for groups and supergroups

(This very condensed table is not a Recommendation,
and reference should be made to the complete Recommendations)

	Group		Supergroup
Ratio between wanted component and the following components, defined in G.242, § 1.2: – intelligible crosstalk [a] – unintelligible crosstalk [a] – possible crosstalk – harmful out-of-band – harmless out-of-band	at 84 kHz (G.242) (dB) 70 70 35 40 17		at 412 kHz (G.242) (dB) 70 70 35 40 17
Additional suppression to safeguard pilot frequencies (G.243)			at least 40 dB at 308 kHz ± 8 Hz at least 20 dB at 308 and 556 kHz ± 40 Hz (relative to 412 kHz value)
Additional suppression to safeguard additional measuring frequencies (G.243)			at least 20 dB at 308 and 556 kHz ± 20 Hz at least 15 dB at 308 and 556 kHz ± 50 Hz (relative to 412 kHz) (see also Figure 1/G.243)
Range of insertion loss over the passband for through-connection equipments	± 1 dB relative to 84 kHz (G.242)		± 1 dB relative to 412 kHz ⩽ 3 dB for SG 1 and 3 (G.242)
Range of insertion loss over 10 °C and 40 °C for through-connection equipments	± 1 dB to 84 kHz relative to the insertion loss at 25 °C (G.242)		± 1 dB at 412 kHz relative to the insertion loss at 25 °C (G.242)
Pilot frequency for (G.241)	Frequency (kHz) [b]	Accuracy (Hz)	Absolute power level at zero relative level point (for tolerances, see G.241) (dBm0)
– Basic group B [c]	84.080 84.140 104.080	± 1 ± 3 ± 1	−20 −25 −20
– Basic supergroup	411.860 411.920 547.920	± 3 ± 1 ± 1	−25 −20 −20

[a] For telephony (G.242).

[b] See Recommendation G.241 for use of these frequencies.

[c] Also applies to 8-channel groups (G.234).

TABLE 3 *bis*

Summary of main characteristics specified by the CCITT for mastergroups, supermastergroups and 15-supergroup assembly

	Mastergroup	Supermastergroup	15-supergroup assembly
Ratio between wanted component and the following components defined in G.242, § 1.2	at 1552 kHz (G.242) (dB)	at 11 096 kHz (G.242) (dB)	at 1552 kHz (G.242) (dB)
– intelligible crosstalk [a]	70	70	70
– unintelligible crosstalk [a]	70	70	70
– possible crosstalk	35	35	35
– harmful out-of-band	40	40	40
– harmless out-of-band	17	17	17
Variation of insertion loss in passband of through-connection equipment	± 1 dB with respect to value at 1552 kHz (G.242)	± 1.5 dB with respect to value at 11 096 kHz ± 1 dB in each mastergroup (G.242)	± 1.5 dB with respect to value at 1552 kHz ± 1 dB in each supergroup (G.242)
Variation of insertion loss between 10 °C and 40 °C of through-connection equipment	± 1 dB at 1552 kHz relative to insertion loss at 25 °C (G.242)	± 1 dB at 11 096 kHz relative to insertion loss at 25 °C (G.242)	± 1 dB at 1552 kHz relative to insertion loss at 25 °C (G.242)
Relative levels at distribution frames (G.233)	(dBr)	(dBr)	(dBr)
– transmit	−36	−33	−33
– receive	−23	−25	−25 or −33
Return loss at modulator input (G.233)	(dB) ⩾ 20	(dB) ⩾ 20	(dB) ⩾ 20
Mastergroup, supermastergroup or 15-supergroup assembly pilots (G.241) in:	Frequency (kHz)	Frequency (Hz)	Level (for tolerances, see G.241) (dBm0)
– basic mastergroup	1 552	± 2	−20
– basic supermastergroup	11 096	± 10	−20
– basic 15-supergroup assembly	1 552	± 2	−20

[a] For telephony (G.242).

TABLE 4

Summary of characteristics specified by the CCITT for carrier systems on open-wire lines

(This very condensed table is not a Recommendation,
and reference should be made to the complete Recommendations)

	Systems acting on each pair		
	3-circuit systems	8-circuit systems G.314 [c]	12-circuit systems
Line frequencies – for a single system	Figure 1/G.361; (see also G.361 §§ 1.1, 2.1, 2.2)	Figure 1/G.314	Figure 1/G.311 or Figure 2/G.311
– for several systems on the same route	Figure 1/G.361	[G.314, c)]	See Figure 3/G.311 and Figure 4/G.311 for examples
Pilots – frequency	16.110 and 31.110 kHz or 17.800 kHz [a] (G.361, § 1.3)	[G.314, d)]	(G.311, § 5)
– level	-15 dBm0		-20 dBm0 [b]
Terminal equipment and intermediate repeater output. Relative level per channel at 800-Hz equivalent frequency	$\leqslant$ 17 dBr (G.361, § 1.2)	$\leqslant$ 17 dBr [G.314, b)]	$\leqslant$ 17 dBr ± 1 dBr (terminal equipment) $\leqslant$ 17 dBr ± 2 dBr (intermediate repeater equipment) (G.311, § 3)
Frequency accuracy of pilot and carrier frequency generators	2.5×10^{-5} (G.361, §§ 1.3 and 1.8)	1×10^{-5} [G.314, d)]	5×10^{-6} (G.311, § 6)

a) Used only by agreement between Administrations.

b) Provisional Recommendation.

c) For text of this Recommendation, see *Orange Book*, Volume III-1, Geneva, 1976.

TABLE 5

Summary of characteristics specified by the CCITT for carrier systems on symmetric-pair cables [a)]

(This very condensed table is not a Recommendation,
and reference should be made to the complete Recommendations)

	System			
	1, 2 or 3 groups	4 groups	5 groups	2 supergroups
Line frequencies	Figure 2*a*)/G.322	Figure 2*b*)/G.322 Scheme 1 Scheme 1 *bis* [b)]	Figure 2*c*)/G.322 Scheme 2 Scheme 2 *bis* [b)]	Figure 4/G.322 Schemes 3 and 4 Scheme 3 *bis* [b)]
Relative level at repeater output [c)] (low-gain systems) (G.322, § 2.2.1)	−11 dBr	−14 dBr	−14 dBr	−14 dBr
Relative level at repeater output [c)] (valve-type systems) [G.324, B, b)] [d)] – nominal value – tolérance	+4.5 dBr ± 2 dB	+1.75 dBr ± 2 dB	+1.75 dBr ± 2 dB	+1.75 dBr ± 2 dB
Return loss of repeater and line impedances [G.322, § 1.5)]	$\leqslant 0.15\sqrt{\frac{f_{max}}{f}}$ or $\leqslant 0.25$	$\leqslant 0.15\sqrt{\frac{f_{max}}{f}}$ or $\leqslant 0.10$	$\leqslant 0.15\sqrt{\frac{f_{max}}{f}}$ or $\leqslant 0.10$ (paper-insulated cables) $\leqslant 0.15\sqrt{\frac{f_{max}}{f}}$ or $\leqslant 0.17$ (cable types II *bis* and III *bis* [b)], G.611)	
Relative level at repeater input [c)]	$\geqslant -56.5$ dBr [G.324, B, b)]			
Pilots	For alternate methods, see Figure 5/G.322			60 kHz ± 1 Hz and 556 kHz ± 3 Hz (G.322, § 1.4.2)
Monitoring frequencies (low-gain systems)	(G.322, § 2.2.2)			
Harmonic distortion (low-gain systems)	See Table 1/G.322			
Harmonic distortion (valve-type systems)	See Table 1/G.324 [d)]			

[a)] For 12 + 12 systems, see Recommendations G.325 and G.327.

[b)] Used only by agreement between Administrations.

[c)] Not applicable to power-fed repeaters.

[d)] For text of this Recommendation, see *Orange Book*, Volume III-1, Geneva, 1976.

TABLE 6

Summary of characteristics specified by the CCITT for carrier systems on 2.6/9.5-mm coaxial cables

(This very condensed table is not a Recommendation, and reference should be made to the complete Recommendations.)

	System			
	2.6 MHz [a)] (1)	4 MHz (2)	12 MHz (3)	60 MHz (4)
Line frequencies	Figure 1/G.337 [d)] and Figure 1/G.338 [d)]	Figure 1/G.338 [d)] and Figure 3/G.332	Figure 1/G.332 to Figure 4/G.332	Figure 1/G.333 and Figure 2/G.333
Pilot frequencies				
– line-regulating pilots	60 kHz ± 1 Hz or 308 kHz ± 3 Hz 2604 kHz ± 30 Hz [G.337 [d)], A, b)]	60 kHz ± 1 Hz or 308 kHz ± 3 Hz 4092 kHz ± 40 Hz and see G.338 [d)], b) 1)	4287 kHz ± 49.2 Hz for valve-type systems [G.339 [d)], b) 1)] 12 435 kHz ± 124.3 Hz transistorized systems (G.332, § 2.1)	4287 kHz ± 42.9 Hz 12 435 kHz ± 124.3 Hz 22 372 kHz ± 223.7 Hz 40 920 kHz ± 409.2 Hz (G.333, § 2.1)
– auxiliary line-regulating pilots	[G.337 [d)], A, b)]	[G.338 [d)], b) 1)]	308 kHz ± 3 Hz and 12 435 kHz ± 124.3 Hz for valve-type systems [G.339 [d)] b) 1)] 308 kHz ± 3 Hz and 4287 kHz ± 42.9 Hz for transistorized systems (G.332, § 2.1)	
Frequency comparison pilots				
– national	as (2)	60 or 308 kHz, 1800 kHz [b)] [G.338 [d)], b) 2)]	300 or 308 kHz (G.332, § 2.2)	
– international	as (2)	1800 kHz [G.338 [d)], b) 2)]	308 and 1800 kHz 300 kHz [b)], 808 kHz [b)] and 1552 kHz [b)] (G.332, § 2.2)	4200 or 8316 kHz (G.333, § 2.2)
Additional measuring frequencies	[G.337 [d)], A, c)]	[G.338 [d)], b) 4)]	(G.332, § 2.3) and [G.339 [d)], b) 3)]	(G.333, § 2.3)
Level of line-regulating pilots and additional measuring frequencies				
– adjustment value	as (2)	– 10 dBm0 ± 0.5 dB [G.338 [d)], b)] – 1.2 Nm0 for some systems [G.338 [d)], b)]	– 10 dBm0 ± 0.5 dB [G.332, b) 1)] – 1.2 Nm0 for valve-type systems [G.339 [d)], b)]	as (2)
– error in the level	as (3)	as (3)	±0.1 dB (G.332, § 2.1)	as (3)
– variation with time	as (3)	as (3)	± 0.3 dB (G.332, § 2.1)	as (3)

TABLE 6 *(Concluded)*

	System			
	2.6 MHz [a] (1)	4 MHz (2)	12 MHz (3)	60 MHz (4)
Impedance match between repeaters and line N (as defined in G.332, § 3)	$N \geqslant 40$ dB for $f < 300$ kHz [G.338 [d], e)] $N \geqslant 45$ dB for $f > 300$ kHz [G.338 [d], e)]		$N \geqslant 48$ dB for $300 \leqslant f \leqslant 5564$ kHz [valve-type systems G.339 [d], e)] $N \geqslant 48$ dB for $f = 300$ kHz and $N \geqslant 55$ dB for $f \geqslant 800$ kHz (transistorized systems G.332, § 5)	$N = 65$ dB [c] (G.333, § 5)
Relative level on line			[G.332, f)] and [G.339 [d], f)	(G.333, § 6)

a) Use of the 6-MHz system for telephony is specified otherwise (see G.337 [d], B).

b) Only used by agreement between Administrations.

c) The value of 65 dB is valid for telephone transmission.

d) For the text of Recommendations G.337, G.338 and G.339, see *Orange Book*, Volume III-1, Geneva, 1976.

TABLE 7

Summary of characteristics specified by the CCITT for carrier systems on 1.2/4.4-mm coaxial cables

(This very condensed table is not a Recommendation, and reference should be made to the complete Recommendations.)

	Systems			
	1.3 MHz	4 MHz	6 MHz	12 MHz
Line frequencies	Figure 1/G.341	Schemes 1 and 2 of Figure 1/G.343	Schemes 1, 2 and 3 of Figure 1/G.344	(G.345)
Pilot frequencies – line-regulating pilots	1364 kHz ± 13.6 Hz (G.341, § 2.1)	See G.343, § 2.1 and Scheme 1 [G.338, b) [c] 1]; Scheme 2 (G.332. § 2.1)	308 kHz ± 3 Hz (G.344)	The provisions of this Recommendation are those appearing in Recommendation G.332 (see the preceding Table 6), with the exception of the matching
– auxiliary line-regulating pilots	60 kHz ± 1 Hz or 308 kHz ± 3 Hz (G.341, § 2.1)	4287 kHz ± 42.8 Hz [a] (G.343, § 2.1)	4287 kHz ± 42.8 Hz [b] 6200 kHz ± 62 Hz (G.344, § 2.1)	
– frequency comparison pilots	60 kHz or 308 kHz (G.341, § 2.2)	Scheme 1 [G.338 [c], b) 2)] and Scheme 2 (G.332, § 2.2)	Schemes 1 and 2 [G.338 [c], b) 2)] Scheme 3 (G.332, § 2.2)	
Additional measuring frequencies	(G.341, § 2.3)	(G.343, § 2.3)	(G.344, § 2.3)	
Level of line-regulating pilots and additional measuring frequencies – adjustment value	−10 dBm0 or −1.2 Nm0 for some systems (G.341, § 2)	−10 dBm0 (G.343, § 2)	−10 dBm0 (G.344, § 2)	
– tolerances		(G.343, § 2)	(G.344, § 2)	
Impedance match between repeaters and line	$N \geq 54$ dB for a 6-km repeater section $N \geq 52$ dB for an 8-km repeater section (G.341, § 5)	$N \geq 50$ dB for $f = 60$ kHz $N \geq 57$ dB for $f \geq 300$ kHz (4-km repeater section G.343, § 5)	$N \geq 60$ dB for $f \geq 300$ kHz $N = 50$ dB for $f = 60$ kHz (3-km repeater section, G.344, § 5)	$N = 63$ dB for a 2-km repeater section (G.345)
Relative levels on line and interconnection	(G.341, § 6)	−9 dBr at 4028 kHz or −8.5 dBr at 4287 kHz (G.343, § 6)	−17 dBr (G.344, § 5)	(G.332, § 6)

[a] Only used by agreement between Administrations.

[b] Only used by agreement between Administrations.

[c] For text of this Recommendation, see *Orange Book*, Volume III-1, Geneva, 1976.

TABLE 8

Summary of main characteristics specified by the CCITT
for international circuits for programme transmissions

(This very condensed table is not a Recommendation,
and reference should be made to the complete Recommendations)

	Type of circuits [a), b)]			
	15 kHz [c), d)]	10 kHz [g)]	5 kHz [h)]	7 kHz
Frequency band effectively transmitted by the complete link (kHz) Additional attenuation at these limits	0.04 to 15 1.5 dB at 0.04 kHz 3 dB at 15 kHz (J.21) [d)]	0.05 to 10 4.3 dB J.22	0.07 to 5 3 dB J.23	0.05 to 7 3 dB J.23
Attenuation/frequency distortion	±0.5 dB; 0.125 to 10 kHz (J.21, § A.3.1.1)			
Group delay at frequency $f(\tau f)$ relative to the minimum value of group delay	15 kHz ⩽ 12 ms 14 kHz ⩽ 8 ms 0.075 kHz ⩽ 24 ms 0.04 kHz ⩽ 55 ms (J.21)	10 000 Hz ⩽ 8 ms 1 000 Hz ⩽ 20 ms 50 Hz ⩽ 80 ms (J.22, § A.3.2)	0.07 kHz ⩽ 60 ms 5 kHz ⩽ 15 ms J.23	0.5 kHz ⩽ 80 ms 0.1 kHz ⩽ 20 ms 6.4 kHz ⩽ 5 ms 7 kHz ⩽ 10 ms J.23
Maximum absolute voltage level at a sound-programme zero relative level point	+9 dB (J.14) – Peak voltage 3.1 V (Figure 3/J.13)			
Definition of zero relative level at a point in a carrier circuit)	Level to give no greater load than that for the telephone channels replaced (J.31, § 2)	As for Telephony, is within ±3 dB (J.14)		
Nominal relative voltage level at the input and output of the circuit defined in J.13	6 dB (J.14)			
Variation of relative level with time	⩽ ± 0.5 dB (daily variation) (J.21, § A.2.3)	⩽ ± 0.5 dB (daily variation) (J.22, § A.2.3)	⩽ ± 0.5 dB (daily variation) (J.23, § A.2.3)	
Intelligible crosstalk attenuation (near-end or far-end ratio)	0.04 kHz ⩾ 50 dB 0.5 kHz ⩾ 74 dB 5 kHz ⩾ 74 dB 15 kHz ⩾ 60 dB (J.21, § A.3.1.8)	Between 2 programme transmission circuits or telephony into sound programme ⩾ 74 dB Sound programme into telephony: ⩾ 65 dB (J.22 and J.23 respectively) [e)]		

TABLE 8 *(cont.)*

	Type of circuits [a), b)]			
	15 kHz [c), d)]	10 kHz [g)]	5 kHz [h)]	7 kHz
Circuit noise including nonlinear crosstalk) [f)]	Level ⩽ −47 dBm0ps (new weighting network according to J.16)	Psophometric voltage at the end of 1) cable circuit ⩽ 6.2 mV 2) open-wire circuit ⩽ 15.6 mV		

a) Characteristics applicable to the hypothetical reference circuits, defined in Recommendation J.11.

b) Types of circuits described in Recommendation J.12.

c) For the additional characteristics specified by the CCITT for 15-kHz stereophonic sound-programme circuits (see Recommendation J.21.

d) See CCIR Recommendations 505.

e) Special precautions needed for crosstalk between the two directions of transmission (see Recommendations J.18 and J.22).

f) Measures taken to reduce the effects of noise in a group link (see Recommendation J.17).

g) See CCIR Recommendation 504.

h) See CCIR Recommendation 503.

TABLE 9

Summary of main characteristics of analogue signals at audio frequencies, at terminals of PCM equipments

(This very condensed table is not a Recommendation, and reference should be made to the complete Recommendations)

Analogue characteristics measured at input and output parts [a), b)]	Test signal			
	Signal	Frequency range	Power level, x (dBm0)	
Attenuation/frequency distortion			Preferred value: −10 alternative: 0	Figure 1/G.712
Envelope-delay distortion			0	Figure 2/G.712
Idle channel noise: – weighted – single frequency – due to receiving equipment				 −65 dBm0p −50 dBm0p −75 dBm0p
Image frequency	sine wave	> 4 kHz	x	< x − 25 dBm0
Level of out-of-band image signals	sine wave	300-3400 Hz	0	< −25 dBm0
Intermodulation products: – $2f_1 - f_2$ – any intermodulation: project	 two sine wave sine wave sine wave	 f_1 and f_2 (Hz) 300-3400 Hz 50 Hz	 $-21 < x < -4$ −9 −23	 < x − 35 dBm0 < −49 dBm0
Variation of gain: – with input level (reference = gain at input level of −10 dBm0) – with time (stability)	 white noise sine wave sine wave	 700-1100 Hz 700-1100 Hz	 $-55 < x < -10$ $-10 < x < 3$ $-55 < x < 3$	 Figure 7*a)*/G.712 Figure 7*b)*/G.712 Figure 7*c)*/G.712 ± 0.2 dB in 10 minutes ± 0.5 dB in one year
Crosstalk: – interchannel – go-return	 sine wave white noise sine wave	 700-1100 Hz 300-3400 Hz	 0 0	 < −65 dBm0 < −60 dBm0 > 60 dB
Distortion	Gaussian noise sine wave	 700-1100 Hz	$-55 < x < 3$ $-45 < x < 0$	Figure 5/G.712 Figure 6/G.712

a) Parameters of input and output ports:

- 600 ohms balanced, 4-wire ports;
- return loss better than 20 dB over frequency range 300-3400 Hz (provisional recommendation).

b) For correct application to the equipments, see § 1 of Recommendation G.712.

PART I

Recommendations G.100 to G.181

GENERAL CHARACTERISTICS OF INTERNATIONAL TELEPHONE CONNECTIONS AND CIRCUITS

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SECTION 1

GENERAL CHARACTERISTICS FOR INTERNATIONAL TELEPHONE CONNECTIONS AND INTERNATIONAL TELEPHONE CIRCUITS

1.0 General

Recommendation G.100

DEFINITIONS USED IN FASCICLE III.1

(Melbourne, 1988)

Introduction

The definitions given below have been found to be useful in the study of telephone connections and telephone circuits.

The detailed definitions appearing in Recommendation G.102 are referred to, but not reproduced.

The definitions of specialized terms which are not mentioned here can be found in:

- Recommendation G.106, for availability and reliability;
- Recommendation G.117 as concerns unbalance about earth;
- Annex A to Recommendation G.111 as concerns speech transmission performance;
- Paragraph 1.6 of this fascicle for echo suppressors, echo cancellers, compandors, etc.

1 General terms

1.1 **hypothetical reference connection (HRX)**

F: communication fictive de référence

S: conexión fictiva de referencia (CFR)

A hypothetical connection of defined structure, length and performance in a telecommunication network for analogue or digital (or mixed) signal transmission, to be used as a model in which studies relating to overall performance may be made, thereby allowing comparisons with standards and objectives.

1.2 **input/output** (Recommendations G.111, G.121, etc.)

F: entrée/sortie

S: entrada/salida

Terms used to indicate the direction of transmission at an interface of an equipment item. These terms avoid the ambiguity encountered in the use of “transmit/receive” or “send/receive”.

1.3 **relative level (at a point on a circuit)**

F: niveau relatif (en un point d'un circuit)

S: nivel relativo (en un punto de un circuito)

The expression 10 $\log_{10}$ (P/P_0) dBr where P represents the power of a test signal of 1000 Hz at the point concerned and P_0 the power of that signal at the *transmission reference point.*

Note – This quantity is independent of P_0, it is a composite gain (level difference). For further details, see Recommendation G.101, § 5.3.2.

1.4 **transmission reference point**

F: point de référence pour la transmission

S: punto de referencia para la transmisión

A hypothetical point at or near to the sending end of each channel (preceding the virtual switching point specified by the CCITT), used as the "zero relative level point" in the computation of nominal relative levels.

1.5 **return loss**

F: affaiblissement d'adaptation

S: pérdida de retorno

Quantity characterizing the degree of match between two impedances, Z_1 and Z_2. It is given by the expression:

$$L_R = 20 \log_{10} \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| \text{ dB.}$$

2 Transmission performance objectives

2.1 **performance objective**

F: objectif pour la qualité de fonctionnement

S: objetivo de calidad de funcionamiento

(Defined in Recommendation G.102.)

2.2 **design objective**

F: objectif pour les projects

S: objetivo de diseño

(Defined in Recommendation G.102.)

2.3 **commissioning objective**

F: objectif pour la mise en service

S: objetivo de puesta en servicio inicial

(Defined in Recommendation G.102.)

2.4 **limits for maintenance purposes (maintenance limits)**

F: limites de maintenance

S: límites de mantenimiento

(Defined in Recommendation G.102.)

3 Transmission impairments

3.1 group-delay distortion

F: distorsion de temps de propagation de groupe

S: distorsión por retardo de grupo

The difference between group delay at a given frequency and minimum group delay, in the frequency band of interest.

3.2 quantizing distorsion unit (qdu)

F: unité de distorsion de quantification (udq)

S: unidad de distorsión de cuantificación (udc)

(For this concept see Recommendation G.113.)

4 Propagation time, echo and stability

4.1 balance return loss

F: affaiblissement d'équilibrage

S: atenuación de equilibrado

At a 4-wire terminating set ("hydrid"), that portion of the *semi-loop loss* which is attributable to the degree of match between the impedance, Z_2, connected to the 2-wire line terminals, and the balance impedance, Z_B. It is given approximately by the expression:

$$L_{BR} = 20 \log_{10} \left| \frac{Z_2 + Z_B}{Z_2 - Z_B} \right| \quad \text{dB}$$

Note – Under most circumstances the expression given is sufficiently accurate. However, for some worst case evaluations, the exact expression must be used. The exact expression is:

$$L_{BR} = 20 \log_{10} \left| \frac{Z_0 + Z_B}{2Z_0} - \frac{Z_2 + Z_0}{Z_2 - Z_B} \right| \quad \text{dB}$$

where Z_0 is the 2-wire input impedance. (If $Z_0 = Z_B$ the two expressions become identical.)

4.2 echo

F: écho

S: eco

Unwanted signal delayed to such a degree that, for instance in telephony, it is perceived as distinct from the wanted signal (i.e. the signal directly transmitted).

Note 1 – Distinction is made between *talker echo* and *listener echo.*

Note 2 – An echo is usually considerably attenuated with respect to the wanted signal.

4.3 echo balance return loss

F: affaiblissement d'équilibrage pour l'écho

S: atenuación de equilibrado para el eco

Balance return loss averaged with $1/f$ power weighting over the telephone band, in accordance with Recommendation G.122, § 4.

4.4 echo control device

F: dispositif de réduction de l'écho

S: dispositivo de control de eco

A voice-operated device placed in the 4-wire portion of the circuit and used for reducing the effect of echo.

Note – This reduction is in practice carried out either by subtracting an estimated echo from the circuit echo (i.e. cancelling it) or by introducing loss in the transmission path to suppress the echo (echo suppression).

4.5 echo loss, L_{ECHO}

F: affaiblissement d'écho A_{ECHO}

S: atenuación de eco, A_{ECO}

Semi-loop loss averaged with $1/f$ power weighting over the telephone band, in accordance with Recommendation G.122, § 4.

Note 1 – In cases where a point *t* (2-wire point) exists, the echo loss is approximately equal to the sum of the transmission losses *a-t* and *t-b* and the *echo balance return loss.* (Points *a* and *b* are shown in Recommendation G.122.)

Note 2 – Distinction may be made between the echo loss of a given piece of equipment and that of a national system (cf. Note 2 to definition in § 4.11).

4.6 talker echo loudness rating (of an international connection)

F: l'équivalent à la sonie pour l'écho pour la personne qui parle (d'une communication internationale)

S: índice de sonoridad del eco para el hablante (en una conexión internacional)

The sum of the sending loudness rating, receiving loudness rating of the talker's national system, twice the loss of the international chain and the *echo loss* (*a-b*) of the listener's national system, as defined at the virtual switching point. (Points *a* and *b* are shown in Recommendation G.122.)

4.7 listener echo (receive end echo)

F: l'écho à la réception

S: eco para el oyente (eco en la recepción)

Echo produced by double reflected signals and disturbing the listener, receiving voice-band data equipment, etc.

Note 1 – The term "received end echo" is a term preferred by some Administrations.

Note 2 – With small delay against the wanted signal (less than about 3 ms) listener echo may cause *hollowness* in telephony. In transmission of voice-band data signals, listener echo may cause bit errors and, in any case, reduces the margin against other disturbances.

4.8 listener echo loss (receive echo loss)

F: affaiblissement de l'écho à la réception

S: atenuación para el oyente (atenuación de eco en la recepción)

Degree of attenuation of the double reflected signal with respect to the wanted signal. In terms of the absolute losses of both signals, the listener echo loss is (see Figure 1/G.100): $LE = L_2 - L_1$.

Note – For practical purposes the listener echo loss is equal to the *open-loop loss* (valid if the latter exceeds 8 dB). The listener echo loss characterizes the degree of disturbance by *hollowness,* as well as the disturbing effect on voice-band data modem receivers.

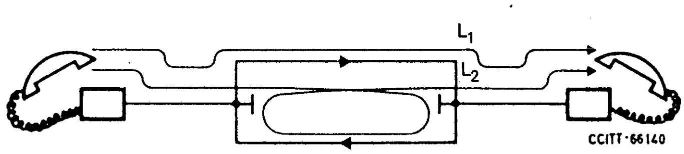


FIGURE 1/G.100

4.9 **hollowness**

F: son caverneux

S: cavernosidad

Distortion in telephony caused by double reflected signals and subjectively perceived as a "hollow sound", i.e. as if the talker would speak into some hollow vessel.

Note – Hollowness is to be distinguished from *listener echo.*

4.10 **open-loop loss (OLL)**

F: affaiblissement en boucle ouverte

S: atenuación en bucle abierto (ABA)

In a loop formed by a 4-wire circuit (or a cascade connection of two or more 4-wire circuits) and terminated by 2-wire ends (i.e. having "4-wire terminating sets", or hybrids, at both ends), the loss measured by breaking the loop at some point, injecting a signal and measuring the loss incurred in traversing the open loop. All impedance conditions should be preserved while making the measurement. See Figure 2/G.100.

Note 1 – In practice the OLL is equal to the listener echo loss.

Note 2 – The OLL is also equal to the sum of the two *semi-loop losses* associated with a loop.

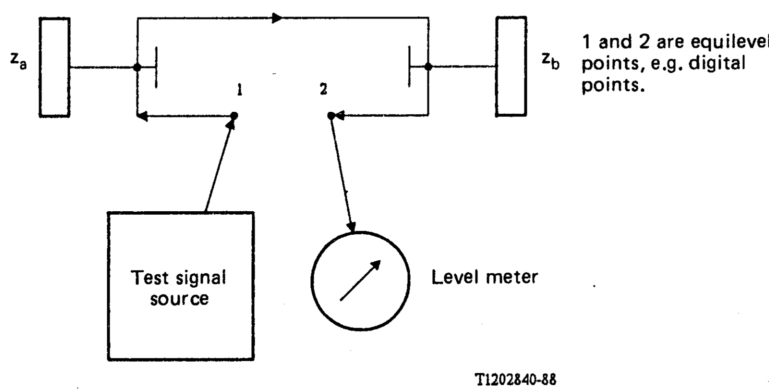


FIGURE 2/G.100

4.11 **path a-t-b (transmission loss of . . .); semi-loop loss**

F: affaiblissement du trajet a-t-b; affaiblissement en demi-boucle

S: atenuación del trayecto a-t-b; atenuación en semibucle

The transmission loss between the points "a" and "b" of the 4-wire termination (as defined at the virtual switching points) independent of whether there exists or not a physical point "*t*".

4.11.1 *Possible alternative to the definition in § 4.11*

semi-loop loss

F: affaiblissement en demi-boucle

S: atenuación en semibucle

In an arrangement comprising a 4-wire circuit (or a cascade connection of several 4-wire circuits) with unwanted coupling between the go and return direction at the ends of the circuit – usually via a 4-wire terminating set, or via acoustical coupling – the loss measured between the input and output. See Figure 3/G.100.

Note 1 – The semi-loop loss is an important quantity in determining *echo balance return loss, echo loss, listener echo loss* (see also *open-loop loss*).

Note 2 – Distinction may be made between the semi-loop loss of a given piece of equipment and the semi-loop loss of a national system. The latter is measured at equi-level points in an ISC which serves as a national gateway exchange.

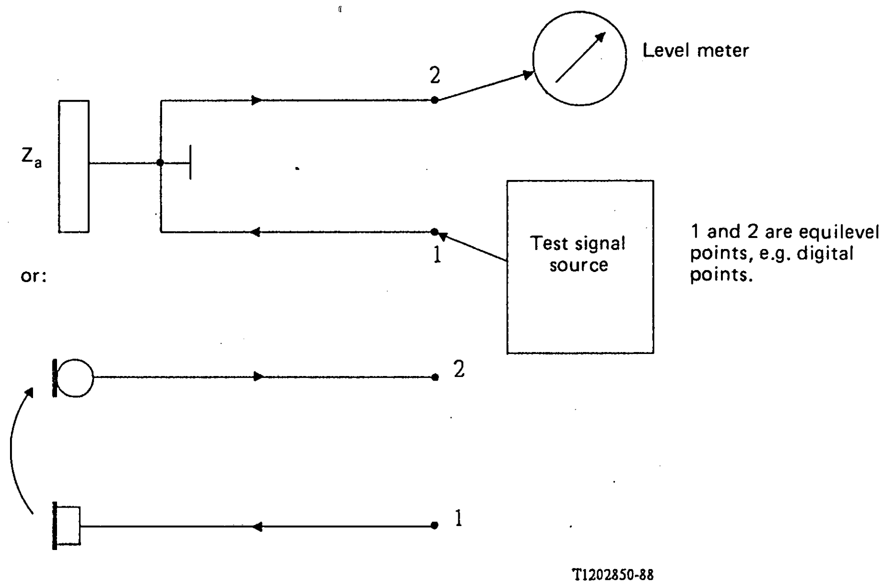


FIGURE 3/G.100

4.12 **stability loss**

F: affaiblissement pour la stabilité

S: atenuación para la estabilidad

The lowest value of the semi-loop loss in the frequency band to be considered.

4.13 **talker echo**

F: écho pour la personne qui parle

S: eco para el hablante

Echo produced by reflection near the listener's end of a connection, and affecting the talker.

4.14 **test balance return loss (TBRL)**

F: affaiblissement d'équilibrage en position de mesure

S: atenuación de equilibrado en posición de medida (AEPM)

The *balance return loss* measured against a test impedance (i.e. in this case the impedance Z_2 – cf. definition of *balance return loss* – is a specified test impedance).

Note – The TBRL characterizes the precision of the balance network.

4.15 **mean one-way propagation time**

F: temps de propagation moyen dans un sens

S: tiempo medio de propagación en un sentido

In a connection, the mean of the propagation times in the two directions of transmission.

Note – The use of this concept is explained in Recommendation G.114.

5 Equipment

5.1 R or T pads (in telephone extension)

F: compléments de ligne R ou T (dans un système national)

S: atenuadores R o T (en la prolongación telefónica)

The R or T pad represents the transmission loss between the 0 dBr points at the digital/analogue codec and the 2-wire side of the 2-wire/4-wire terminating unit or the same in the reversed direction, respectively.

Note – The transmission loss introduced by the combination of the R and T pads in the subject of CCITT Recommendations.

Recommendation G.101

THE TRANSMISSION PLAN[1)]

(Geneva, 1964; amended at Mar del Plata, 1968, Geneva, 1972, 1976 and 1980; Malaga-Torremolinos, 1984)

1 Principles

The transmission plan of the CCITT established in 1964 was drawn up with the object of making use, in the international service, of the advantages offered by 4-wire switching. It is referred to in the Recommendations appearing in Part I, Section 1 of the Series G Recommendations. However, the recommendations in the plan are to be considered as met if the use of technical means other than those described below gives an equivalent performance at the international exchange.

Recommendations G.121 and G.122 describe the conditions to be fulfilled by a national network for this transmission plan to be put into effect.

Note 1 – From the point of view of the transmission plan, no distinction is made between intercontinental circuits and other international circuits.

Note 2 – Short trans-frontier circuits are not covered by this plan and should be the subject of agreement between the Administrations concerned.

Note 3 – The Appendix to the present Section 1 of the Series G Recommendations contains the justification for the values of corrected reference equivalents appearing in Recommendations G.111 and G.121.

2 Definition of the constituent parts of a connection

2.1 *The international chain of circuits and the national systems*

A complete **international telephone connection** consists of three parts, as shown in Figure 1/G.101. The division between these parts is determined by the *virtual analogue switching points* in the originating/terminating international switching centres (ISCs). These are theoretical points with specified relative levels (see Figure 2/G.101 and §§ 5.1 and 5.2 of this Recommendation).

1) This Recommendation is partly reproduced in Recommendation Q.40 [1].

The three parts of the connection are:

– Two national systems, one at each end. These may comprise one or more 4-wire national trunk circuits with 4-wire interconnection, as well as circuits with 2-wire connection up to the local exchanges and the subscriber sets with their subscriber lines.

– An international chain made up of one or more 4-wire international circuits. These are interconnected on a 4-wire basis in the international centres which provide for transit traffic and are also connected on a 4-wire basis to national systems in the international centres.

An international 4-wire circuit is delimited by its virtual analogue switching points in an international switching centre.

Note 1 – In principle the choice of values of the relative levels at the virtual analogue switching points on the side of a national system is a national matter. In practice, several countries have chosen −3.5 dBr for receiving as well as for sending. These are theoretical values; they need not actually occur as any special equipment item; however, they serve to determine the relative levels at other points in the national network. If, for instance, the loss "*t-b*" or "*a-t*" is 3.5 dB (as is the case in several countries, cf. Table A-1/G.121), then it follows that the relative levels at point *t* are 0 dBr (input) and −7 dBr (output).

Note 2 – The virtual analogue switching points may not be the same as the points at which the circuit terminates physically in the switching equipment. These latter points are known as the circuit terminals; the exact position of these terminals is decided in each case by the Administration concerned.

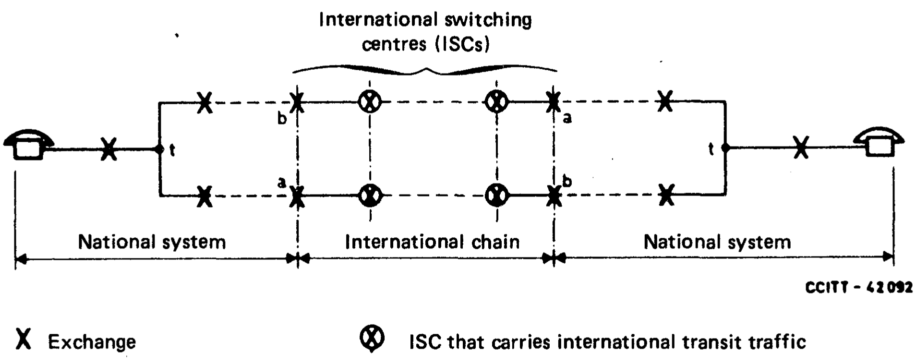


FIGURE 1/G.101

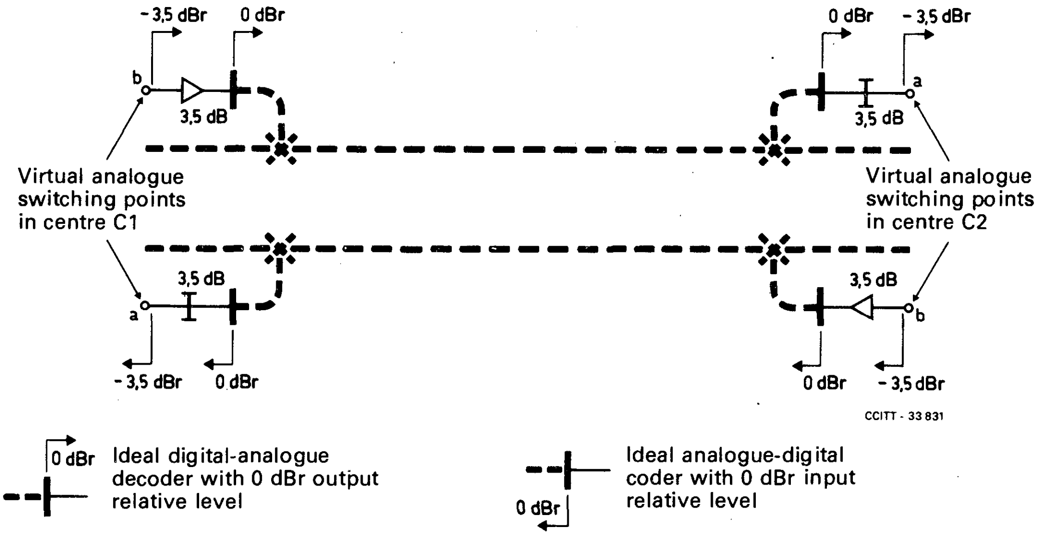
Definition of the constituent parts of an international connection

2.2 *National extension circuits: 4-wire chain*

When the maximum distance between an international exchange and a subscriber who can be reached from it does not exceed about 1000 km or, exceptionally, 1500 km, the country concerned is considered as of average size. In such countries, in most cases, not more than three national circuits are interconnected on a 4-wire basis between each other and to international circuits. These circuits should comply with the recommendations of Subsection 1.2.

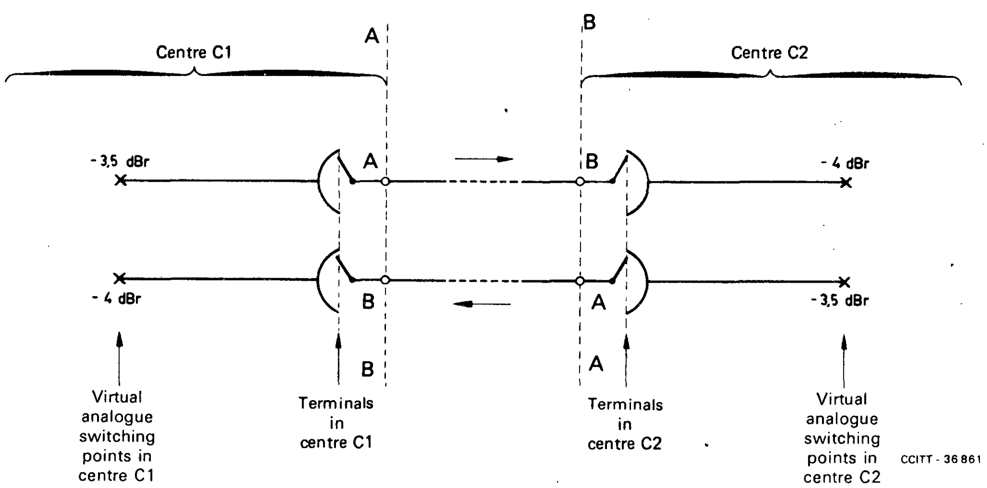
In a large country, a fourth and possibly a fifth national circuit may be included in the 4-wire chain, provided it has the nominal transmission loss and the characteristics recommended for international circuits used in a 4-wire chain (see Recommendation G.141, § 1, § 4 of this Recommendation and the Recommendations in Subsection 1.5).

Note – The abbreviation "a **4-wire chain**" (see Figure 3/G.101) signifies the chain composed of the international chain and the national extension circuits connected to it, either by 4-wire switching or by some equivalent procedure (as understood in § 1 above).



Note – Ideal coders and decoders are assumed to show a relation between analogue and digital signals and vice versa exactly in accordance with the appropriate tables for A-law or μ-law of Recommendation G.711 [2].

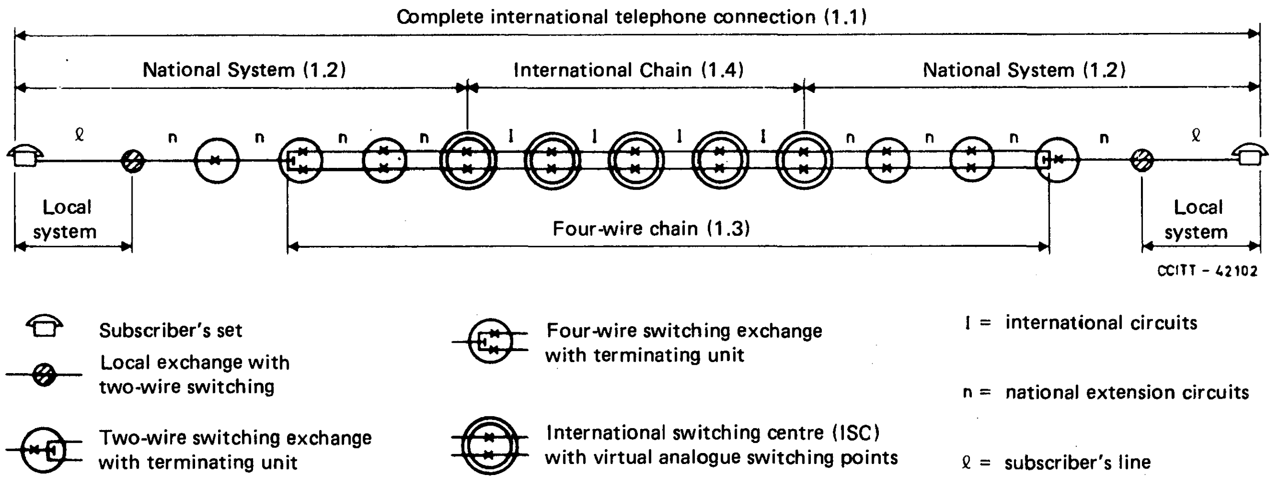
a) *Definition of virtual analogue switching points for a digital international circuit between digital international centres*



b) *Definition of virtual analogue switching points for an analogue international circuit between analogue international centres*

FIGURE 2/G.101

Definitions for international circuits



Note – The arrangement shown for the national systems are examples only. The numbers given in brackets refer to the Subsections of Section 1 (Fascicle III.1) in which recommendations may be found relevant to that part of the connection. In addition, the circuits making up this chain must individually meet the requirements of Subsection 1.5.

FIGURE 3/G.101

An international connection to illustrate the nomenclature adopted

3 Number of circuits in a connection

3.1 *National circuits*

It seems reasonable to assume that in most countries any *local exchange* can be connected to the international network by means of a chain of four (or less) national circuits. Five national circuits may be needed in some countries, but it is unlikely that any country may need to use more than five circuits. Hence the CCITT has reached the conclusion that four circuits is a representative figure to assume for the great majority of international connections.

In most modern national networks, the four circuits will probably include three 4-wire amplified circuits (usually set up on FDM carrier systems) and one 2-wire circuit, probably unamplified. However, cases in which local exchanges are be reached by four amplified circuits, among them usually at least one PCM circuit, are becoming more and more frequent. All these circuits may be 4-wire circuits.

3.2 *International circuits*

According to the International Telephone Routing Plan (Recommendation E.171), the number of international circuits is restricted to four.

3.3 *Hypothetical reference connections*

See Recommendation G.103.

3.4 Tables 1/G.101, 2/G.101 and 3/G.101 give the percentage relative and cumulative frequencies of the number of circuits encountered in an international connection calculated from a survey of about 270 million international telephone connections taken in 1973. These tables take traffic weighting into account.

TABLE 1/G.101

Relative frequencies of the number of circuits in the two national extensions and the international chain (expressed as percentages)

Number of circuits	Originating LE-CT3	International CT3-CT3′	Terminating CT3′-LE′
1	33.8	95.1	32.9
2	38.9	4.5	39.5
3	20.2	0.3	20.4
4	6.0	–	6.1
5	1.0	–	1.0

Note – The relative frequencies of 6 and 7 circuits in the originating national system are 0.005% and 0.0005% respectively. The relative frequencies of 4, 5 and 6 international circuits are 0.03%, 0.00007% and 0.00009% respectively.

The means and modal numbers of national circuits are both equal to 2. This applies to both originating and terminating national extensions. The mean number of international circuits is 1.1 and the modal number is 1.

TABLE 2/G.101

Relative and cumulative frequency of the total number of circuits between local exchanges (expressed as percentages)

Number of circuits LE to LE′	Relative frequency (%)	Cumulative frequency (%)
3	10.61	10.61
4	25.44	36.05
5	28.77	64.82
6	20.39	85.20
7	10.08	95.29
8	3.60	98.89
9	0.93	99.81
10	0.17	99.98
11	0.02	100.00

Note – The relative frequencies of connections with 12, 13 and 14 circuits are 0.0012%, 0.000088% and 0.0000049% respectively. The mean value is equal to 5.1 and the modal value is equal to 5.

4 Incorporation of unintegrated digital processes

4.1 *General*

The worldwide telephone network is now undergoing a transition from what is predominantly analogue operation to mixed analogue/digital operation. In the longer term, it is possible to foresee a continued transition to predominantly digital operation.

Figure 4/G.101 is intended to demonstrate how unintegrated analogue/digital PCM processes can occur in the international nework by illustrating a possible stage in the development of a national network as it progresses from all-analogue to all-digital. As indicated, subnetworks could arise in the country in which the transmission systems and the telephone exchanges are all-digital and fully integrated. Such subnetworks (referred to as "digital cells" by some) will require analogue/digital conversion processes in order to interface into the remainder of the network. Furthermore, some of the trunk-junctions (toll connecting trunks) and trunk-circuits (intertoll trunks) may be provided in some countries by 7-bit PCM systems, serving analogue exchanges. Conversely, some digital exchanges may have to switch analogue circuits. Manual assistance switchboards, PBXs and subscribers' multiplex systems using PCM digital techniques are also allowed for. Naturally, any of the circuits indicated as 7-bit PCM could be either analogue or 8-bit PCM; but one of the worst cases is illustrated.

TABLE 3/G.101

Relative and cumulative frequency of the number of circuits in the 4-wire chain (expressed as percentages)

Number of circuits in the 4-wire chain	Relative frequency (%)	Cumulative frequency (%)
1	2.65	2.65
2	14.16	16.81
3	27.49	44.30
4	26.43	70.73
5	17.28	88.01
6	8.33	96.34
7	2.83	99.18
8	0.70	99.88
9	0.11	99.99
10	0.0065	100.00

Note – The relative frequencies of 4-wire chains comprising 11 and 12 circuits are estimated to be 0.000475% and 0.0000322% respectively. The mean value is equal to 3.8 and the modal value is equal to 4.

Notes to Tables 1/G.101, 2/G.101 and 3/G.101

1 – The basic information, displayed in Table 1/G.101, derives from an analysis of the routing details of about 270 million telephone connections in 1973 conducted under the auspices of CCITT Study Group XIII in which 23 countries participated. LE signifies "local exchange".

2 – Table 2/G.101 is derived from Table 1/G.101 on the assumption that the three distributions of Table 1/G.101 are uncorrelated.

3 – Table 3/G.101 is derived from Table 1/G.101 on the basis of the following assumptions:

- Of all the international traffic handled by primary centres, 30% originates from (or terminates at) local exchanges co-sited with the primary centre. The remaining 70% involves a trunk junction between the local exchange and the primary centre.
- In the case of routing over 1 national circuit, 50% of those circuits are assumed to be 4-wire and 4-wire switched at the CT3 and thus to be included in the 4-wire chain. The other 50% are assumed to be 2-wire switched at the CT3, and thus do not participate in the 4-wire chain. This is assumed to be the case for both national extensions, independently.
- Any national routing involving 5 to 7 national circuits will incorporate a 2-wire switched trunk-junction.
- All the other routings (i.e. involving 2 to 4 national circuits) will be regarded as being with or without 2-wire switched trunk-junctions in the ratio 7:3.
- The routings in the two countries are uncorrelated.

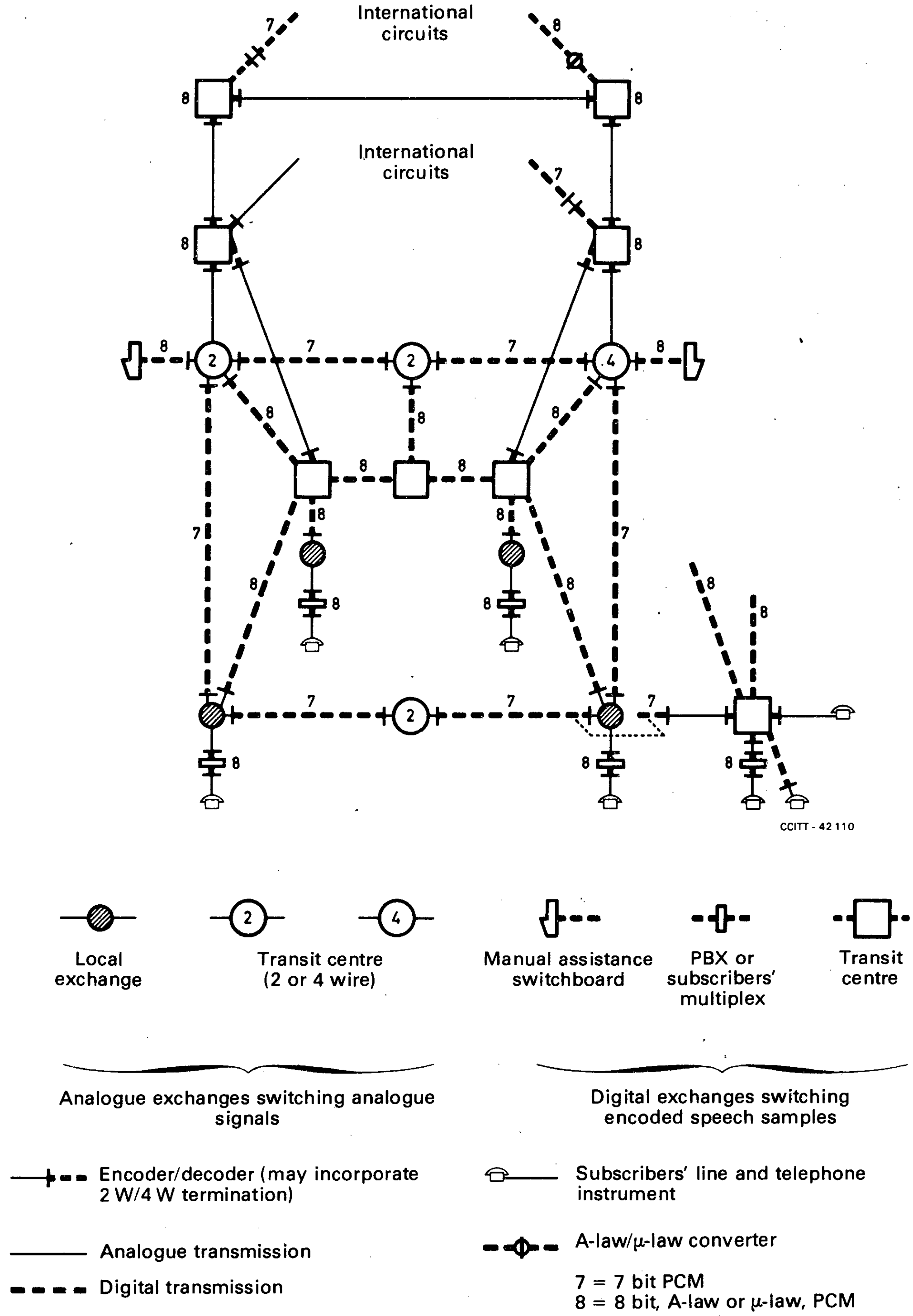


FIGURE 4/G.101
A possible intermediate stage of development in a national network

With regard to 7-bit PCM, it should be noted that such systems are not recommended by the CCITT. The only recommended analogue/digital (A/D) conversion processes for telephone services are 8-bit PCM processes (reference: CCITT Recommendation G.711 [2]). There are in some countries 7-bit PCM systems in operation which have been designed and installed prior to the appearance of Recommendation G.711 and, as existing systems, they should be taken into account, notwithstanding the fact that such systems are of a provisional nature as they will likely be removed from service as soon as their practical usefulness comes to an end.

In view of the foregoing, international telephone connections may for some time include one national 7-bit PCM trunk-junction (toll connecting trunk) or exceptionally two such 7-bit PCM circuits. In addition, international satellite circuits using 7-bit PCM coding may be encountered as well as A-law/μ-law conversion processes and digital pads.

The mixed analogue/digital period is expected to last a considerable number of years. Consequently, it will be necessary to ensure that transmission performance in this period will be maintained at a satisfactory level.

4.2 *Types of telephone circuits*

In the mixed analogue/digital period, international circuits could, in particular, consist of the types indicated in Figure 5/G.101. In all cases, the virtual analogue switching points are identified (conceptually) and the relative levels at these points, specified.

Although the circuit types shown in Figure 5/G.101 are classed as international circuits, the configurations involved could also occur in national telephone networks. However, in national networks the relative levels at the virtual analogue switching points of the circuits could be different from those indicated for international circuits.

The Type 1 circuit in Figure 5a)/G.101 represents the case where digital transmission is used for the entire length of the circuit and digital switching is used at both ends. Such a circuit can generally be operated at a nominal transmission loss of 0 dB as shown because of the transmission properties exhibited by such circuits (e.g., relatively small loss variations with time).

The Type 2 circuit in Figure 5b)/G.101 represents the case where the transmission path is established on a digital transmission channel in tandem with an analogue transmission channel. Digital switching is used at the digital end and analogue switching at the analogue end.

It might be possible, in some cases, to operate Type 2 circuits with a nominal loss of 0 dB in each direction of transmission. For example, where the analogue portion could be provided with the necessary gain stability and where the attenuation distortion would permit such operation.

The Type 3 circuit in Figure 5c)/G.101 represents the case where the transmission path is established over a tandem arrangement consisting of digital/analogue/digital channels as shown. Digital switching is assumed at both ends.

The Type 4 circuit in Figure 5d)/G.101 represents the case where the transmission path is established over a tandem arrangement consisting of analogue/digital/analogue channels as shown. Analogue switching is assumed at both ends.

The Type 5 circuit in Figure 5e)/G.101 represents the case where analogue transmission is used for the entire length of the circuit and analogue switching is used at both ends.

International circuits of this type are usually operated at a loss L, where L is nominally = 0.5 dB between virtual analogue switching points.

Note – General remarks concerning the allocation of losses in the mixed analogue/digital circuits

In circuit types 2, 3 and 4, the pads needed to control any variability in the analogue circuit sections (arising from loss variations with time or attenuation distortion) are shown in a symmetrical fashion in both directions of transmission. However, in practice, such arrangements may require nonstandard levels at the boundaries between circuit sections. Administrations are advised that should they prefer to adopt an asymmetric arrangement, e.g., by putting all the loss into the receive direction at only one end of a circuit (or circuit section); then, provided that the loss is small, e.g., a total of not more than 1 dB, there is no objection on transmission plan grounds.

The small amount of asymmetry that results in the international portion of the connection will be acceptable, bearing in mind the small number of international circuits encountered in most actual connections.

As far as national circuits are concerned, Administrations may adopt any arrangements they wish provided that the requirements of Recommendation G.121, § 2.2, are complied with.

In some cases transmultiplexers may be used, in which case the circuits may not be available at audio-frequency at the point at which a pad symbol is used in the diagrams of Figure 5/G.101. Should the variability of the analogue portions merit additional loss, the precise way in which this loss can be inserted into the circuits is a matter for Administrations to decide bilaterally.

4.3 *Number of unintegrated PCM digital processes*

Restrictions due to transmission impairments

In the mixed analogue/digital period, it may be necessary to include a substantial number of unintegrated digital processes in international telephone connections. To ensure that the resulting transmission impairments (quantizing, attenuation and group-delay distortion) introduced by such processes do not accumulate to the point where overall transmission quality can be appreciably impaired, it is recommended that the planning rule given in Recommendation G.113 § 3 be complied with. The effect of this rule is to limit the number of unintegrated digital processes in both the national and international parts of telephone connections.

In the case of all-digital connections, transmission impairments can also accumulate due to the incorporation of digital processes (e.g., digital pads). The matter of accumulating such impairments under all-digital conditions is also dealt with in Recommendation G.113 § 3.

4.4 *Transmission of analogue and digital data*

In the mixed analogue/digital period, the presence in telephone connections of analogue/digital converters, encoding law converters, digital pads, or other types of digital processes, would not preclude the transmission of analogue data. However, on overall digital connections, digital type data could be adversely affected by devices such as encoding law converters and digital pads, since they involve signal recoding processes. Consequently, for the transmission of digital data, arrangements should be made to switch-out or bypass any device whose operation entails the recoding of digital data signals.

4.5 *General principle*

It is recognized that in the mixed analogue/digital period, there could be a considerable presence of unintegrated digital processes in the worldwide telephone network. Consequently, it is important that the incorporation of these processes should take place in such a way that when integration of functions can occur, unnecessary items of equipment would not remain in the all-digital network.

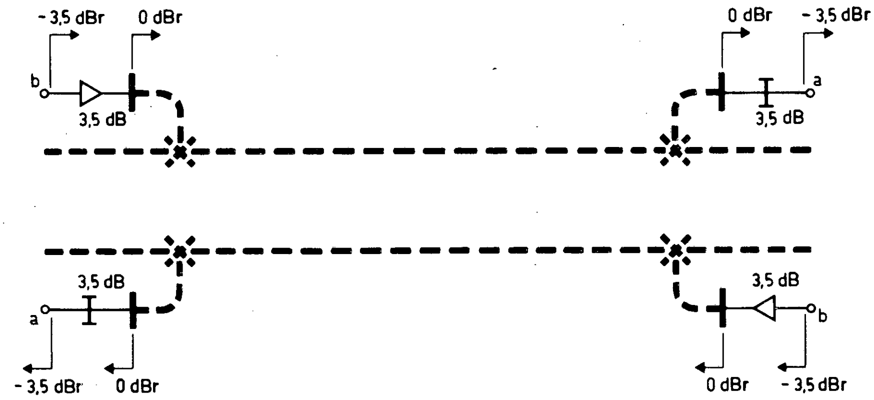
5 **Conventions and definitions**

5.1 *Virtual analogue switching points*

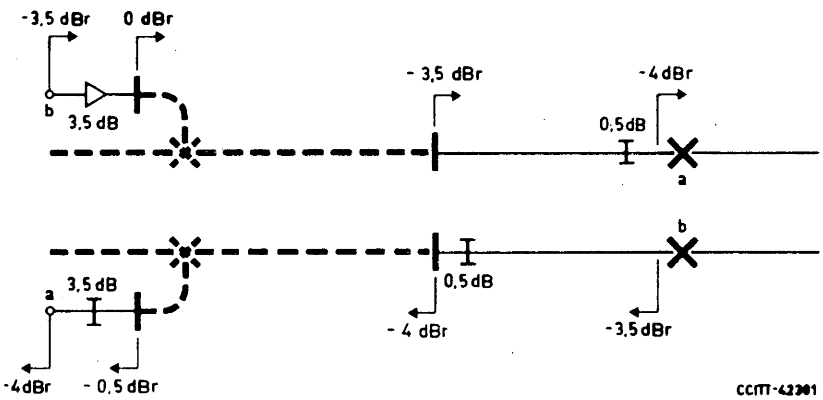
The concept "virtual switching points" has been useful in making transmission studies with regard to all-analogue connections. For example, these points have been used to define the boundary between international circuits as well as between international circuits and national extensions. The "virtual switching points" also provided convenient locations to which transmission quantities could be referred.

The incorporation of digital encoding processes into the worldwide telephone network no longer makes it possible, in all cases, to determine theoretical points which correspond to the "virtual switching points" of all-analogue connections. Since it would be desirable, in mixed analogue/digital connections to have analogous points, the concept of "virtual analogue switching points" has been adopted. This concept postulates the existence of ideal codecs through which the desired analogue points could be derived.

The term "virtual analogue switching points" is also used for all-analogue situations and replaces the older term "virtual switching points".

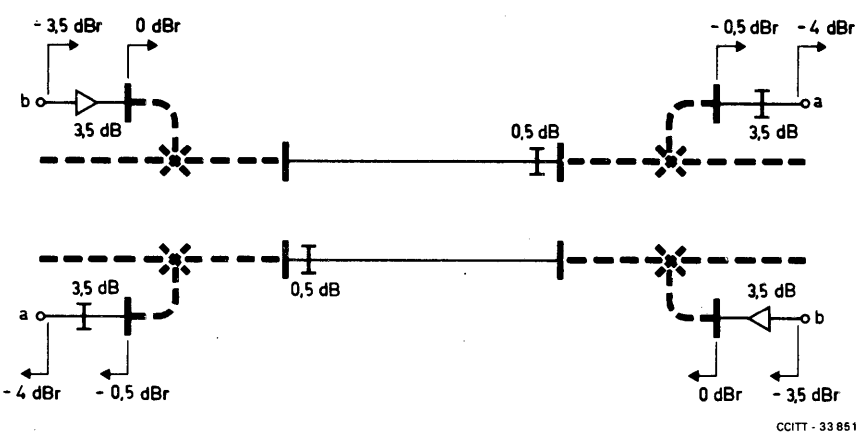


a) *Type 1 circuit – All digital circuit with digital switching at both ends*



Note – Pads required if the analogue circuit section introduces significant amounts of attenuation distortion or variation with time.

b) *Type 2 circuit – Digital/analogue circuit with digital switching at one end and analogue switching at the other end*



Note – Pads required if the analogue circuit section introduces significant amount of attenuation distortion or variation with time.

c) *Type 3 circuit – Digital/analogue/digital circuit with digital switching at each end*

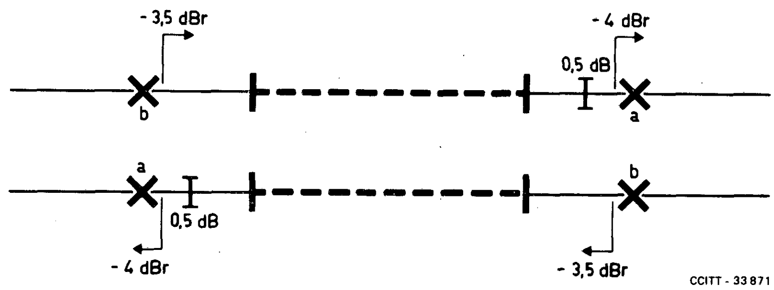
Note — Conventions and symbols adopted for "real" and ideal codecs:

— Ideal coders and decoders are assumed to show a relation between analogue and digital signals and vice versa exactly in accordance with the appropriate tables for A-law or μ-law of Recommendation G.711.

— "Real" coders and decoders are assumed to be such that the performance characteristics of an encoder/decoder pair between audio frequency ports will meet the requirements of Recommendation G.712.

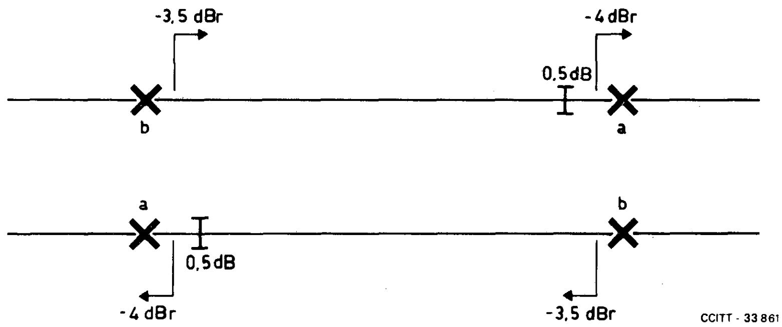
— The symbol for a "real" codec does not include a relative level for the analogue input or output port. If it is desired to specify the relative level, then this should be done by denoting the relative level on the analogue transmission side of the codec. This will avoid any possible confusion with the symbol for an ideal codec.

FIGURE 5/G.101
Types of international circuits

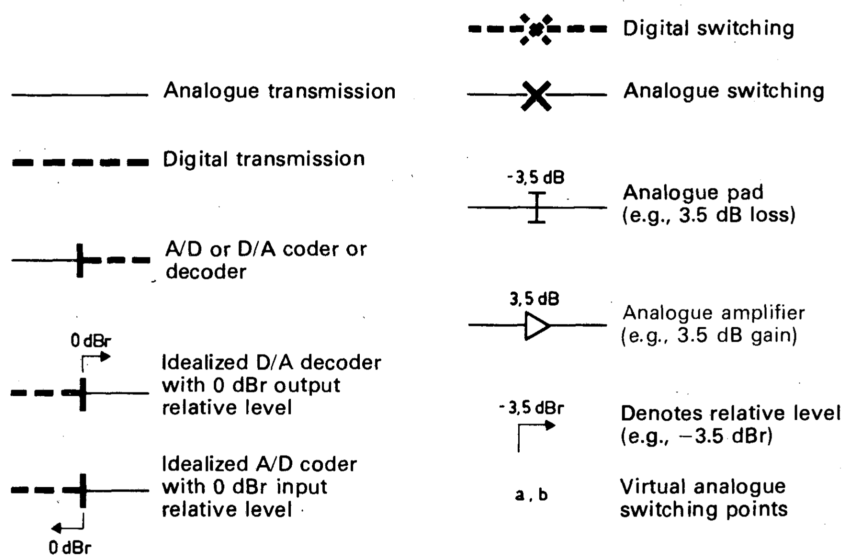


Note – Pads required if the analogue circuit sections introduce significant amount of attenuation distortion or variation with time.

d) *Type 4 circuit – Analogue/digital/analogue circuit with analogue switching at each end*



e) *Type 5 circuit – All analogue circuit with analogue switching at both ends*



Note – The pad symbols in the circuits are not intended to imply that real attenuators are needed. They are a convention of transmission planning engineers.

FIGURE 5/G.101 (end)

Types of international circuits

5.2 *Relative level specified in the virtual analogue switching points of international circuits*

The virtual analogue switching points of an international 4-wire telephone circuit are fixed by convention at points of the circuit where the nominal relative levels at the reference frequency are:

- sending: −3.5 dBr;
- receiving: −4.0 dBr, for analogue or mixed analogue/digital circuits;
  −3.5 dBr for digital circuits or for the very short circuits mentioned under Note 3 below.

The nominal transmission loss of this circuit at the reference frequency between virtual analogue switching points is therefore 0.5 dB for analogue and 0 dB for digital circuits.

Note 1 – See the definition in § 5.3 below. The position of the virtual analogue switching points is shown in Figure 2/G.101, and in Figure 1/G.122.

Note 2 – Since the 4-wire terminating set forms part of national systems and since its actual attenuation may depend on the national transmission plan adopted by each Administration, it is no longer possible to define the relative levels on international 4-wire circuits by reference to the 2-wire terminals of a terminating set. In particular, the transmission loss in terminal service of the chain created by connecting a pair of terminating sets to a 4-wire international circuit cannot be fixed at a single value by Recommendations. The virtual analogue switching points of circuits might therefore have been chosen at points of arbitrary relative level. However, the values adopted above are such that in general they permit the passage from the old plan to the new to be made with the minimum amount of difficulty.

Note 3 – If a 4-wire analogue circuit forming part of the 4-wire chain contributes negligible delay and variation of transmission loss with time, it may be operated at zero nominal transmission loss between virtual analogue switching points. This relaxation refers particularly to short 4-wire tie-circuits between switching centres – e.g., circuits between two international switching centres in the same city.

5.3 *Definitions*

5.3.1 **transmission reference point**

F: point de référence pour la transmission

S: punto de referencia para la transmisión

A hypothetical point used as the zero relative level point in the computation of nominal relative levels. At those points in a telephone circuit the nominal mean power level (−15 dBm) defined in Recommendation G.223 [3] shall be applied when checking whether the transmission system conforms to the noise objectives defined in Recommendation G.222 [4].

Note – For certain systems, e.g. submarine cable systems (Recommendation G.371 [5]), other values apply.

Such a point exists at the sending end of each channel of a 4-wire switched circuit preceding the virtual switching point; on an international circuit it is defined as having a signal level of +3.5 dB above that of the virtual switching point.

In frequency division multiplex equipment, a hypothetical point of flat zero relative level (i.e. where all channels have the same relative level) is defined as a point where the multiplex signal, as far as the effect of intermodulation is concerned, can be represented by a uniform spectrum random noise signal with a mean power level as defined in Recommendation G.223 [6]. The nominal mean power level in each telephone channel is −15 dBm as defined in Recommendation G.223 [3].

5.3.2 **relative (power) level**

F: niveau relatif de puissance

S: nivel relativo (de potencia)

5.3.2.1 *Basic significance of relative level in FDM systems*

The relative level at a point in a transmission system characterizes the signal power handling capacity at this point with respect to the conventional power level at a zero relative level point[2)].

[2)] Taking into account such aspects as (basic) noise, intermodulation noise, peak power, etc. (see Recommendation G.223).

If, for example, at a particular point an FDM system designed for a large number of channels the mean power handling capacity per telephone channel corresponds to an absolute power level of S dBm, the relative level associated with this point is (S + 15) dBr. In particular, at 0 dBr point, the conventional mean power level referred to one telephone channel is −15 dBm.

5.3.2.2 *Definition of relative level, generally applicable to all systems*

The relative level at a point on a circuit is given by the expression 10 $\log_{10} (P/P_0)$ dBr, where P represents the power of a sinusoidal test signal at the point concerned and P_0 the power of that signal at the transmission reference point. This is numerically equal to the composite gain (definition in *Yellow Book*, Fascicle X.1) between the transmission reference point and the point concerned, for a nominal frequency of 1000 Hz. For example, if a reference signal of 0 dBm at 1000 Hz is injected at the transmission reference point, the level at a point of x dBr will be x dBm (apparent power $P_x = 10^{x/10}$mW). In addition, application of a digital reference sequence (DRS, § 5.3.3) will give a level of x dBm at a point of x dBr. The voltage of a 0 dBm0 tone at any voiceband frequency at a point of x dBr is given by the expression:

$$\mathrm{V} = \sqrt{10^{x/10} \times 1 \text{ W} \times 10^{-3} \, |Z_R|_{1000}} \text{ volts}$$

where $|Z_R|_{1000}$ is the modulus of the nominal impedance of the point at a nominal frequency of 1000 Hz.

Note 1 – The nominal reference frequency of 1000 Hz is in accordance with Recommendation G.712, § 16. For existing (analogue) transmission systems, one may continue to use a reference frequency of 800 Hz.

Note 2 – The relative levels at particular points in a transmission system (e.g. input and output of distribution frames or of equipment like channel translators) are fixed by convention, usually by agreement between manufacturers and users.

The recommendations of the CCITT are elaborated in such a way that the absolute power level of any testing signal to be applied at the input of a particular transmission system, to check whether it conforms to these recommendations, is clearly defined as soon as the relative level at this point is fixed.

Note 3 – The impedance Z_R may be resistive or complex; in the latter case the power P_x is an apparent power.

Note 4 – It is assumed that between the virtual analogue switching points of a circuit, established over international transmission systems, only points of equal relative level are interconnected in those systems, so that the transmission loss of the circuit will be equal to the difference in relative levels at the virtual analogue switching points (see § 5.2 of this Recommendation).

5.3.2.3 *Relation between corrected send reference equivalents, loudness ratings and relative levels*

The relationship between the 0 dBr point and the level of $T_{\max}$ in PCM encoding/decoding processes standardized by the CCITT is set forth in Recommendation G.711 [2]. In particular, if the minimum nominal corrected send reference equivalent (CSRE) of local systems referred to a point of 0 dBr of a PCM encoder is not less than 3.5 dB, or the minimum nominal send loudness rating (SLR) under the same conditions is not less than −1.5 dB, and the value of $T_{\max}$ of the process is set at +3 dBm0 (more accurately 3.14 dBm0 for A-law and 3.17 for μ-law), then in accordance with § 3 of Recommendation G.121, the peak power of the speech will be suitably controlled.

5.3.2.4 *Compatibility of relative levels of analogue and digital systems*

When the signal load is controlled as outlined in § 5.3.2.3, points of equal relative levels of FDM and PCM circuits may be directly connected together and each will respect the other's design criteria. This is of particular importance when points in the two multiplex hierarchies are connected together by means of transmultiplexers, codecs or modems.

5.3.2.5 *Determination of relative level*

Figure 6/G.101 illustrates the principle of how the relative level at the input and output analogue points of a "real" codec can be determined.

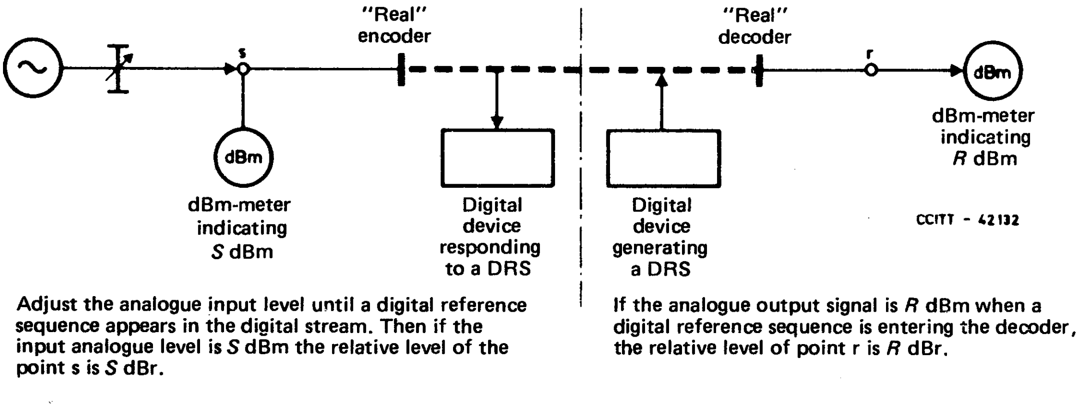


FIGURE 6/G.101

Set-up for determining the relative level at the input and output analogue points of a "real" codec using digital reference sequences

When using Figure 6/G.101 to determine the relative levels of a "real" codec with non-resistive impedances at the analogue input and output ports, the following precautions must be observed:

i) the test frequency should be 1000 Hz with a suitable offset;

ii) the power at points *s* and *r* is expressed as apparent power, i.e.

$$\text{Apparent power level} = 10 \log_{10} \left[\frac{(\text{Voltage at point})^2 \times 10^3}{(\text{Modulus of nominal impedance at 1000 Hz})\ (1\ \text{W})} \right] \text{dBm}$$

iii) point *r* is terminated with the nominal design impedance of the decoder to avoid significant impedance mismatch errors.

Note – Precautions ii), iii) above are, of course equally applicable to the case of resistive input and output impedances and would generally be observed by conventional test procedures. Standardizing the reference frequency as in i) above is, however, essential for complex impedances because of the variation of nominal impedance with the test frequency.

5.3.2.6 *Relative level of a point in a digital link*

The relative level to be associated with a point in a digital path carrying a digital bit stream generated by a coder lined-up in accordance with the principles of § 5.3.2.3 above is determined by the value of the digital loss or gain between the output of the coder and the point considered. If there is no such loss or gain the relative level at the point considered is, by convention, said to be 0 dBr.

The equivalent absolute power level of a digital link may be established as in Figure 7/G.101 by using an ideal decoder. The relative level at a point X in the bit stream can then be assigned by comparing the power at the output of the ideal decoder with that at the analogue zero relative level point originating the digital signal.

5.3.3 **PCM digital reference sequence (DRS)**

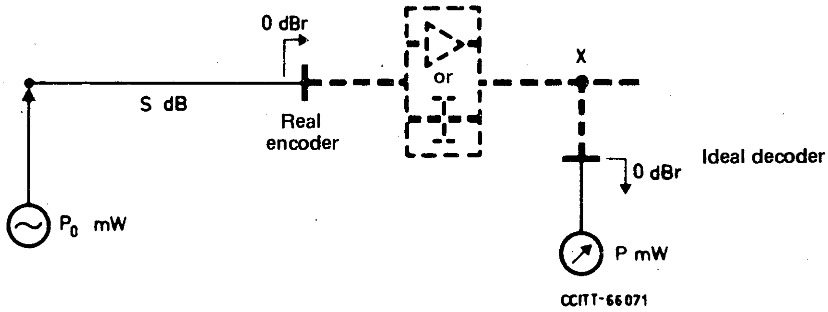
F: séquence numérique de référence MIC

S: secuencia de referencia digital MIC (SRD)

5.3.3.1 A PCM digital reference sequence is one of the set of possible PCM code sequences that, when decoded by an ideal decoder, produces an analogue sinusoidal signal at the agreed test reference frequency (i.e. a nominal 800 or 1000 Hz signal suitably offset) at a level of 0 dBm0.

Conversely an analogue sinusoidal signal at 0 dBm0 at the test reference frequency applied to the input of an ideal coder will generate a PCM digital reference sequence.

Some particular PCM digital reference sequences are defined in Recommendation G.711 [2] in respect to A-law and μ-law codecs.



Procedure

An analogue input signal is applied to the coder with a level of P_0 mW at the 0 dBr point. If this signal results in an analogue signal of P mW at the output of the ideal decoder then:

$$\text{Relative level at point X} = 10 \log_{10} \left(\frac{P}{P_0}\right) \text{ dBr}$$

Note – It is understood that the signal is always within the dynamic range of the conversion process.

FIGURE 7/G.101

Procedure for determining the relative level of a point in a digital link

5.3.3.2 In studying circuits and connections in mixed analogue/digital networks, use of the digital reference sequence can be helpful. For example, Figure 8/G.101 shows the various level relationships that one obtains (conceptually) on a Type 2 international circuit where one end terminates at a digital exchange and the other end at an analogue exchange. In the example of Figure 8/G.101, the analogue portion is assumed to require a loss of 0.5 dB and that provision for this loss is made by introducing a 1.0 dB pad (0.5 dB for each direction of transmission) in the receive direction at the analogue exchange. This has been deliberately chosen to illustrate the utility of the concept of a digital reference sequence.

Figure 8/G.101 gives an example where all the analogue loss is introduced in the output direction at the analogue exchange. In this case the relative levels at the various codecs can be derived from either the DRS or the transmission reference point at the input of the international circuit with no ambiguity.

If, however, in Figure 8/G.101 the analogue circuit section is lined up so as to give an overall loss in the direction $b_1 - a_2$, care must be taken in the use of the DRS. In this case the 0 dBm0 sinusoidal reference signal and DRS may result in different levels at the point a_2. Account should be taken of this effect when designing lining-up procedures for mixed analogue/digital circuits.

As a general principle, the relative levels on a mixed analogue/digital circuit should be referred to the transmission reference point at the input of the circuit.

5.3.4 **circuit test access point**

The CCITT has defined circuit test access points as being "4-wire test-access points so located that as much as possible of the international circuit is included between corresponding pairs of these access points at the two centres concerned". These points, and their relative level (with reference to the transmission reference point), are determined in each case by the Administration concerned. They are used in practice as points of known relative level to which other transmission measurements will be related. In other words, for measurement and lining-up purposes, the relative level at the appropriate circuit test access point is the relative level with respect to which other levels are adjusted.

5.3.5 *Measurement frequency*

For all international circuits 800 Hz is the recommended frequency for single-frequency maintenance measurements. However, by agreement between the Administrations concerned, 1000 Hz may be used for such measurements.

A frequency of 1000 Hz is in fact now widely used for single-frequency measurements on some international circuits.

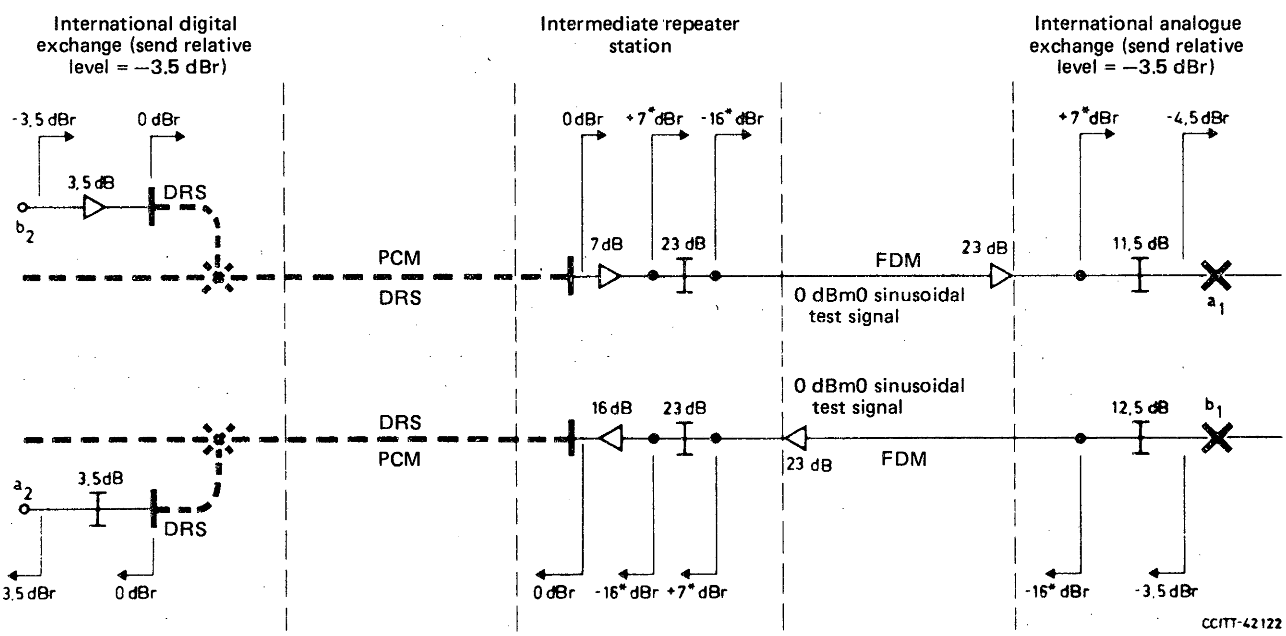
Multifrequency measurements made to determine the loss/frequency characteristic will include a measurement at 800 Hz and the frequency of the reference measurement signal for such characteristics can still be 800 Hz.

Note 1 – Definitions of §§ 5.3.1 and 5.3.2 are used in the work of Study Group XII. Definitions of §§ 5.3.4 and 5.3.5, taken from Recommendations M.565 [7] and M.580 [8], are included for information.

Note 2 – In order to take account of PCM circuits and circuit sections, the nominal frequencies 800 Hz and 1000 Hz are in fact offset by appropriate amounts to avoid interaction with the sampling frequency. Details can be found in Supplement No. 3.5 to Volume IV [9].

5.4 *Interconnection of international circuits in a transit centre*

In a transit centre, the virtual analogue switching points of the two international circuits to be interconnected are considered to be connected together directly without any additional loss or gain. In this way a chain of international circuits has a nominal transmission loss in transit equal to the sum of the individual circuit losses.



DRS Digital reference sequence
PCM PCM channel
FDM FDM channel
* one of the set of VF relative levels cited in [10] for the purpose of illustration
—●— Multiplex VF input/output point

Transmission loss: $b_2 - a_1 = 1.0$ dB
$b_1 - a_2 = 0$ dB

Note – For meaning of other symbols, see legend for Figure 5/G.101.

FIGURE 8/G.101

Use of a digital reference sequence in the design and line-up of a Type-2 international circuit

References

[1] CCITT Recommendation *Transmission Plan*, Vol. VI, Rec. Q.40.

[2] CCITT Recommendation *Pulse Code Modulation (PCM) of Voice Frequencies*, Vol. III, Rec. G.711.

[3] CCITT Recommendation *Assumption for the Calculation of Noise on Hypothetical Reference Circuits for Telephony*, Vol. III, Rec. G.223, § 1.

[4] CCITT Recommendation *Noise Objectives for Design of Carrier-Transmission Systems*, Vol. III, Rec. G.222.

[5] CCITT Recommendation *Carrier Systems for Submarine Cable*, Vol. III, Rec. G.371.

[6] CCITT Recommendation *Assumption for the Calculation of Noise on Hypothetical Reference Circuits for Telephony*, Vol. III, Rec. G.223, § 2.

[7] CCITT Recommendation *Access points for international telephone circuits*, Vol. IV, Rec. M.565.

[8] CCITT Recommendation *Setting-Up and Lining-Up an International Circuit for Public Telephony*, Vol. IV, Rec. M.580.

[9] *Test frequencies on circuits routed over PCM systems*, Vol. IV, Supplement No. 3.5.

[10] CCITT Recommendation *12-Channel Terminal Equipments*, Vol. III, Rec. G.232, § 11.

Recommendation G.102

TRANSMISSION PERFORMANCE OBJECTIVES AND RECOMMENDATIONS

(Geneva, 1980)

1 General

The CCITT has drawn up (or is in the process of studying) Recommendations concerning transmission impairments and their permissible magnitude with the object of achieving satisfactory performance of the network. Such impairments include for example:

a) loudness rating (LR) and loss,

b) noise,

c) attenuation distortion,

d) crosstalk,

e) single tone interference,

f) spurious modulation,

g) effects of errors in digital systems.

Some Recommendations state objectives for an impairment with the implicit assumption that other impairments are at their maximum value (e.g. noise and loss).

In many instances the objectives are based primarily on telephony; this however may require special measures to be applied when other, more demanding services (e.g. sound-programme transmission) are to be incorporated within the network or constituent parts thereof.

The following distinctions may be made between different types of objectives:

1) performance objectives for networks,

2) performance objectives for circuits, transmission and switching equipment,

3) design objectives for transmission and switching equipment,

4) commissioning objectives for circuits, transmission and switching equipment,

5) maintenance/service limits for circuits, transmission and switching equipment.

2 Explanation of a performance objective

The performance objective for a measurable transmission impairment for networks, entire connections, national systems forming part of international connections, international chains of circuits, individual circuits etc. often describes in statistical terms (mean value, standard deviation, or probability of exceeding stated value, etc.) the value to be aimed at in transmission network and systems planning. It describes the performance which, based for example on subjective or other performance assessment tests, it is desirable to aim at in order to offer the user a satisfactory service.

The items (circuits, systems, equipments) making up the network are normally assumed to have a performance related to that recommended by the performance objectives. Traffic weighting will, in some cases, be applied to calculations.

A powerful set of tools which may be used in analyses concerning network objectives and compliance therewith are the hypothetical reference connections described in Recommendation G.103.

3 Explanation of a design objective

The "design objective" for a measurable transmission impairment (e.g. noise, error-rate, attenuation-distortion) for an item of equipment (e.g. a line system, a telephone exchange) is its value when the item is operating in certain electrical/physical environments which might be defined by such parameters as power supply voltage, signal load, temperature, humidity, etc. Some of these parameters may be the subject of CCITT Recommendations and some may not, and it is for the Administrations to assign values to them when they prepare specifications. A suitable allowance may also be made for aging. The most adverse combination of the specified parameters is often assumed.

The purpose of a "design objective" is to provide a basis for the design of an item with respect to the quantity concerned. The significance of the design objective for an item, and examples of the relative frequency of impairment values, are illustrated in Figures 1/G.102 and 2/G.102 respectively.

Design objectives will in many cases directly form the basis of a specification clause for the development and/or the purchase of equipments.

A powerful set of tools used in connection with applying design objectives are the hypothetical reference (HR) circuits and hypothetical reference (HR) digital paths (see relevant Recommendations in the G.100 and G.700 Series).

4 Explanation of a commissioning objective

The conditions encountered on real circuits and installed equipment may differ from the assumptions valid for the HR circuits and for the design of equipment. Therefore the performance to be expected at the time of commissioning cannot be deduced uniquely from Recommendations relating to HR circuits. Suitable allowances may have to be made for such matters as circuits being made up of equipments of different design, line systems differing substantially in length from a homogeneous section, etc. (see for example Recommendation G.226 [1] for noise on real links).

Commissioning objectives are not normally the subject of CCITT Recommendations.

5 Explanations of limits for maintenance purposes

In service, the performance of an item or assembly of items may deteriorate for various reasons: aging, excessive loading, excessive environmental conditions, operations errors, components failures, etc. and there is an economic penalty in service costs if such deterioration is always to be kept negligibly small. Therefore design objectives are chosen to confer as great a margin as possible to assure a satisfactory in-service performance.

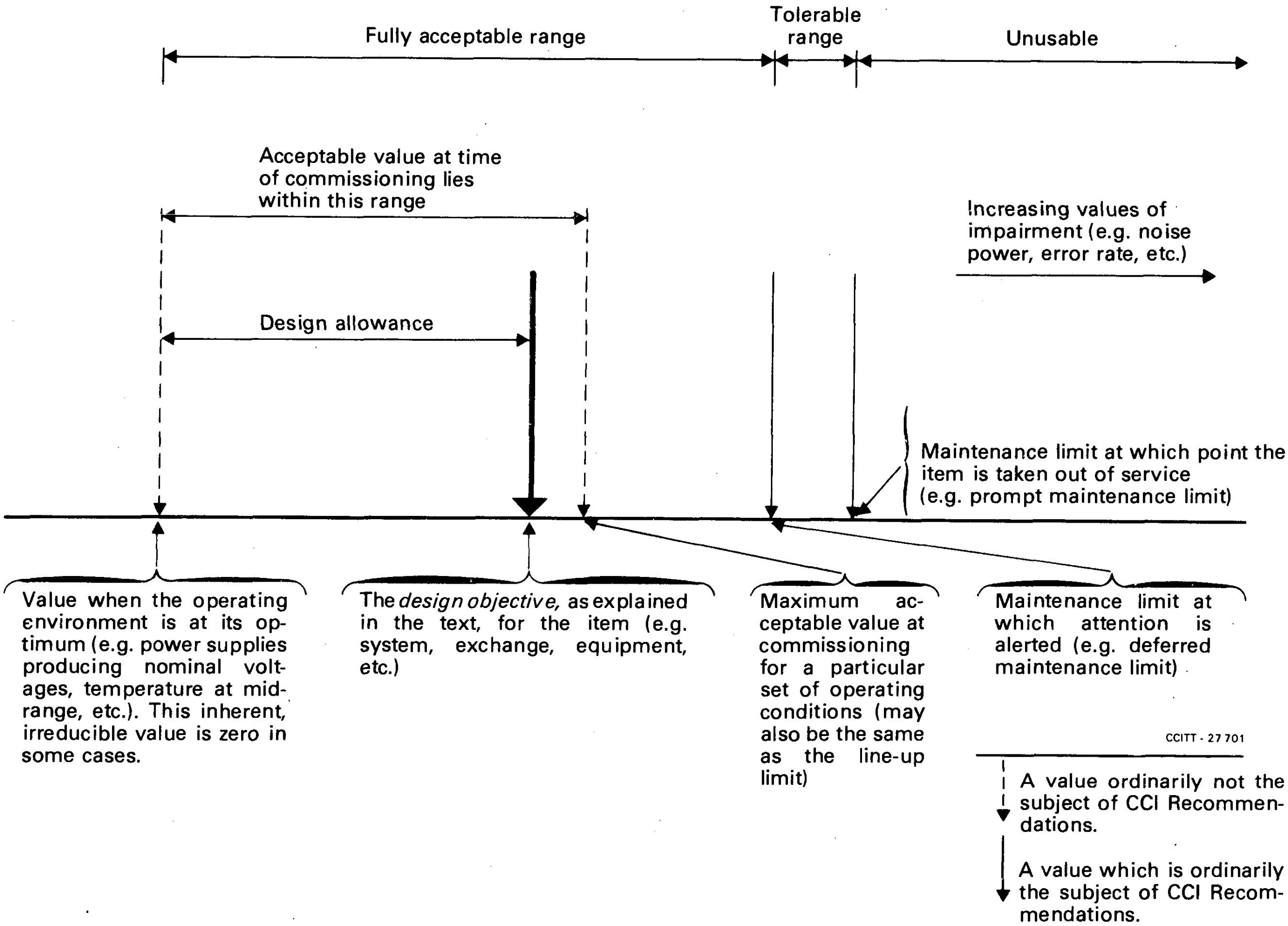


FIGURE 1/G.102

Illustration of the significance of design objective for an item

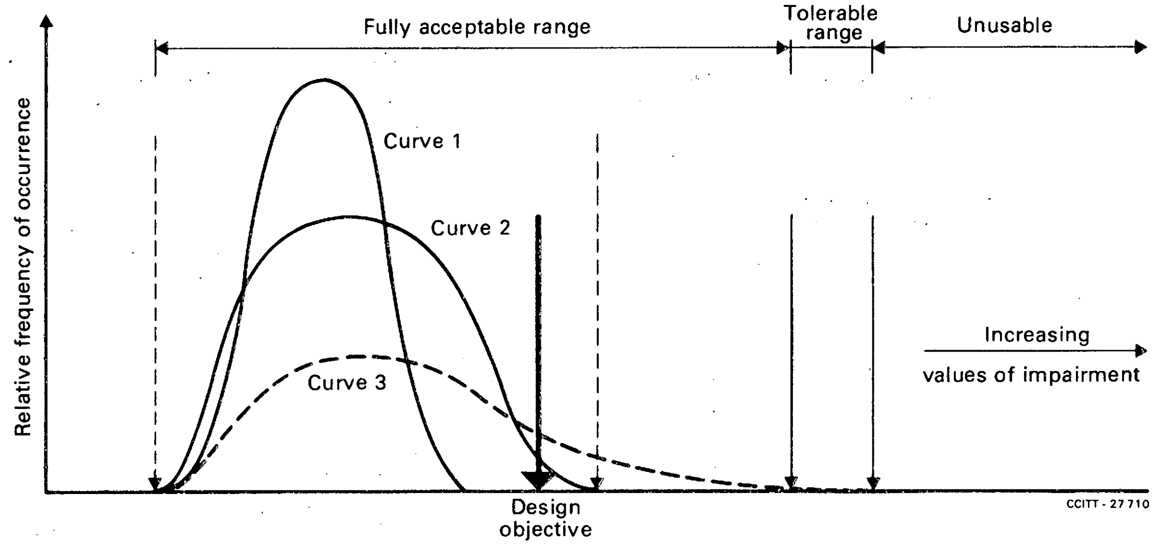
With transmission impairments, there is often no value which represents a clear boundary between "tolerable" and "unusable" performance and in practice a range of impairments in excess of those provided by design objectives will give satisfactory service to customers. This is the case for telephony but for other services may be different.

Nevertheless it is often expedient to define a particular value of impairment above which the item is deemed to be "unusable" and at which the item will be withdrawn from service at the first opportunity so that remedial action can be taken to restore the performance to comply with some defined limit (e.g. limit for prompt maintenance action).

It is often useful to define a performance limit at which attention is alerted but (perhaps) no action is taken immediately (e.g. limit for deferred maintenance action).

These limits are usually independent of the type of service carried by that particular entity. However, it is sometimes necessary to define a performance limit for a particular type of service, beyond which the customer is no longer offered a satisfactory service quality. This limit may differ for various services; some may coincide with a prompt maintenance limit (service limit).

These limits (and others, if necessary) would fall above the design objective. These limits are illustrated in Figure 1/G.102 and a generic title for them is "maintenance limits".



Such curves may be obtained for ensembles of items of equipment at the time of commissioning.
Alternatively curves may be plotted representing the performance of an item during its lifetime.

Curve 1 – Example of relative frequency of occurrence of impairments at time of commissioning in which the design value is met with some margin. A similar distribution might be achieved in service throughout the lifetime of an item of equipment if the effect of environmental conditions etc. is negligible. An example might be the attenuation distortion of transformers.

Curve 2 – Example of the relative frequency of occurrence of impairments at time of commissioning in which the design value is exceeded with some agreed probability because the item of equipment is used in a way which is more demanding than that in the design objectives. An example might be the effect of a repeater spacing of a radio or line system greater than anticipated.

Curve 3 – Example of the relative frequency of occurrence of impairments in service in which the working environment has parameters more onerous than or additional to those specified. Examples might be the effect of excessive loading, component failure or operational errors.

FIGURE 2/G.102

Examples of the relative frequency of impairment values

Reference

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

Recommendation G.103

HYPOTHETICAL REFERENCE CONNECTIONS

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

This Recommendation mainly deals with the analogue network, Recommendation G.104 deals with the wholly digital network and § 4 of this Recommendation deals with the transitional problems when some digital circuits are introduced into the analogue network. Ultimately, it is envisaged that all reference connections, whether they refer to analogue or digital systems, will be combined within one Recommendation.

1 Purpose

A hypothetical reference connection for transmission impairment studies is a model in which the impairments contributed by circuits and exchanges are described.

Such a model may be used by an Administration:

- to examine the effect on transmission quality of possible changes of routing structure, noise allocations and transmission losses in national networks, and
- to test national planning rules for *prima facie* compliance with any statistical impairment criteria which may be recommended by the CCITT for national systems.

For these purposes, several models are desirable. The three hypothetical reference connections described below should encompass most of the studies required to be undertaken.

Hypothetical reference connections are *not* to be regarded as recommending particular values of loss or noise or other impairments, although the various values quoted are in many cases recommended values. Hypothetical reference connections are *not* intended to be used for the design of transmission systems.

2 Composition of hypothetical reference connections

2.1 The composition of the various connections is defined in Figures 1/G.103, 2/G.103 and 3/G.103.

Figure 1/G.103 – The longest international connection with the maximum number of international and national circuits expected to occur in practice. Such a connection would typically have high corrected reference equivalents and high noise contributions, and the noise contribution from international circuits may be significant. The attenuation distortion, group delay, and group-delay distortion would also all be extremely high. Such connections are rare.

Figure 2/G.103 – An international connection of moderate length (say, not longer than 2000 km) comprising the most frequent number of international and national circuits. In such a connection, the noise contribution of the national systems would be expected to predominate. Such a connection is used in a large proportion of international calls.

Figure 3/G.103 – An international connection comprising the practically maximum number of international circuits and the least number of national circuits. Such connections are numerous.

2.2 *The following General Remarks apply to Figures 1/G.103, 2/G.103 and 3/G.103*

2.2.1 The hypothetical reference connections show the international circuits connected together at 0 dBr and −0.5 dBr virtual switching points instead of −3.5 dBr and −4 dBr points. This was felt to be more directly useful to those who might have to use the reference connections in their studies.

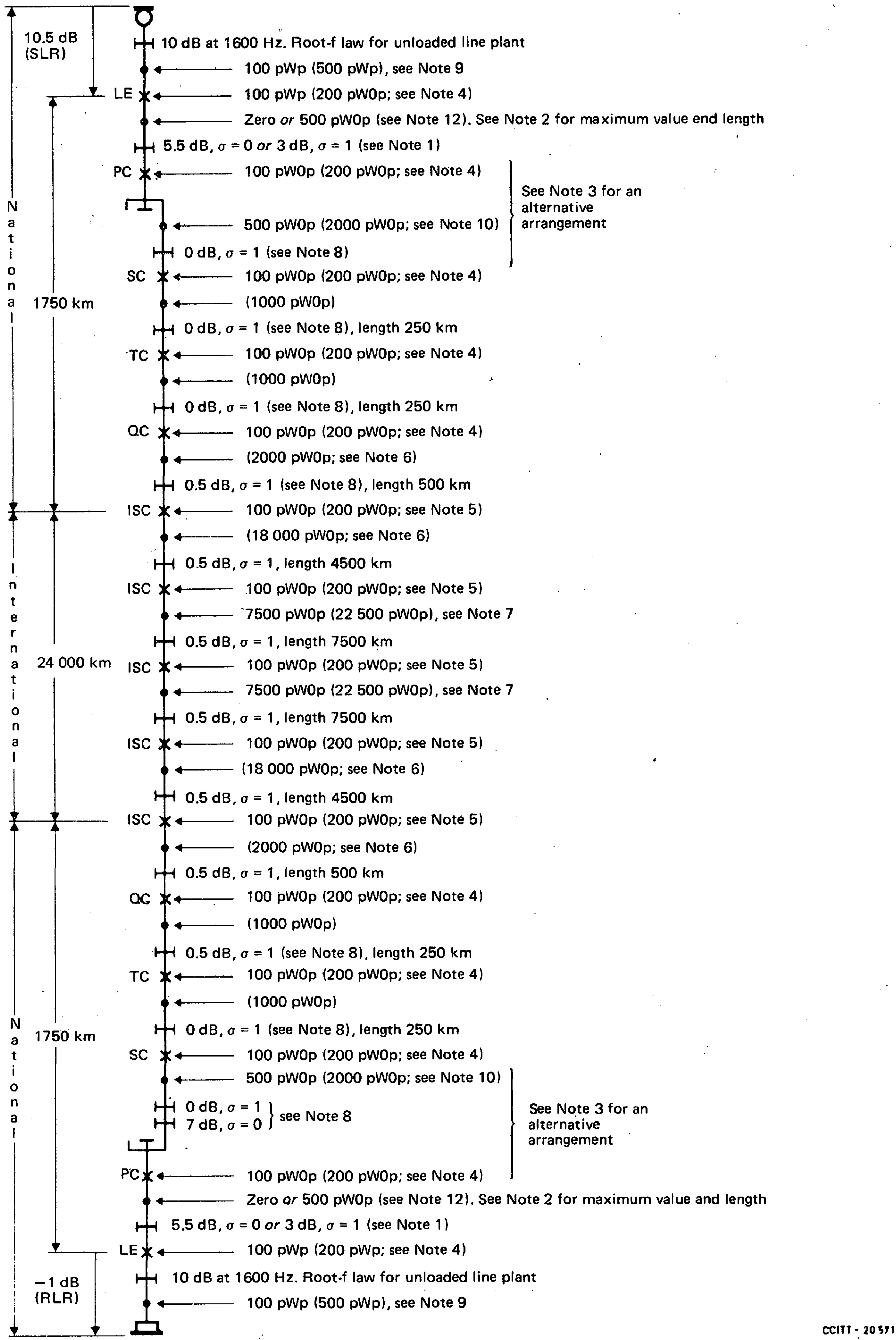
It might be felt that it is somewhat inconsistent that the hypothetical reference connections do not use "conventional" −3.5/−4 dBr virtual switching points. However, if the reference connections are drawn using that convention, the noise power figures appearing on the diagram can no longer be the familiar ones that appear elsewhere in other Recommendations. Annex A gives further explanations.

2.2.2 The nomenclature is based on the international routing plan recommended in Recommendation E.171, i.e. ISC = International Switching Centre (formerly referred to as CT3), ITC = International Transit Centre.

2.2.3 In each case only one direction of transmission is shown.

2.2.4 The design objectives for the mean noise powers are indicated according to current recommendations. For long-distance carrier circuits they are proportional to length, the appropriate noise power rate, 4 pW/km or 1 pW/km, being used according to whether the basic hypothetical reference circuit is one 2500 km long or 7500 km long.

2.2.5 The abbreviation pW0p stands for picowatts psophometric referred to a point of zero relative level. In the case of exchange noise, the point referred to is considered to be in the circuit immediately downstream, of the exchange. The noise powers for circuits are referred to points of zero relative level in the circuits themselves and not to some point on the connection.



Legends for Figures 1/G.103, 2/G.103 and 3/G.103

SLR	sending loudness rating
RLR	receiving loudness rating
LE	local exchange
PC	primary centre
SC	secondary centre
TC	tertiary centre
QC	quaternary centre
ISC	international switching centre

FIGURE 1/G.103

The longest international connection likely to occur in practice

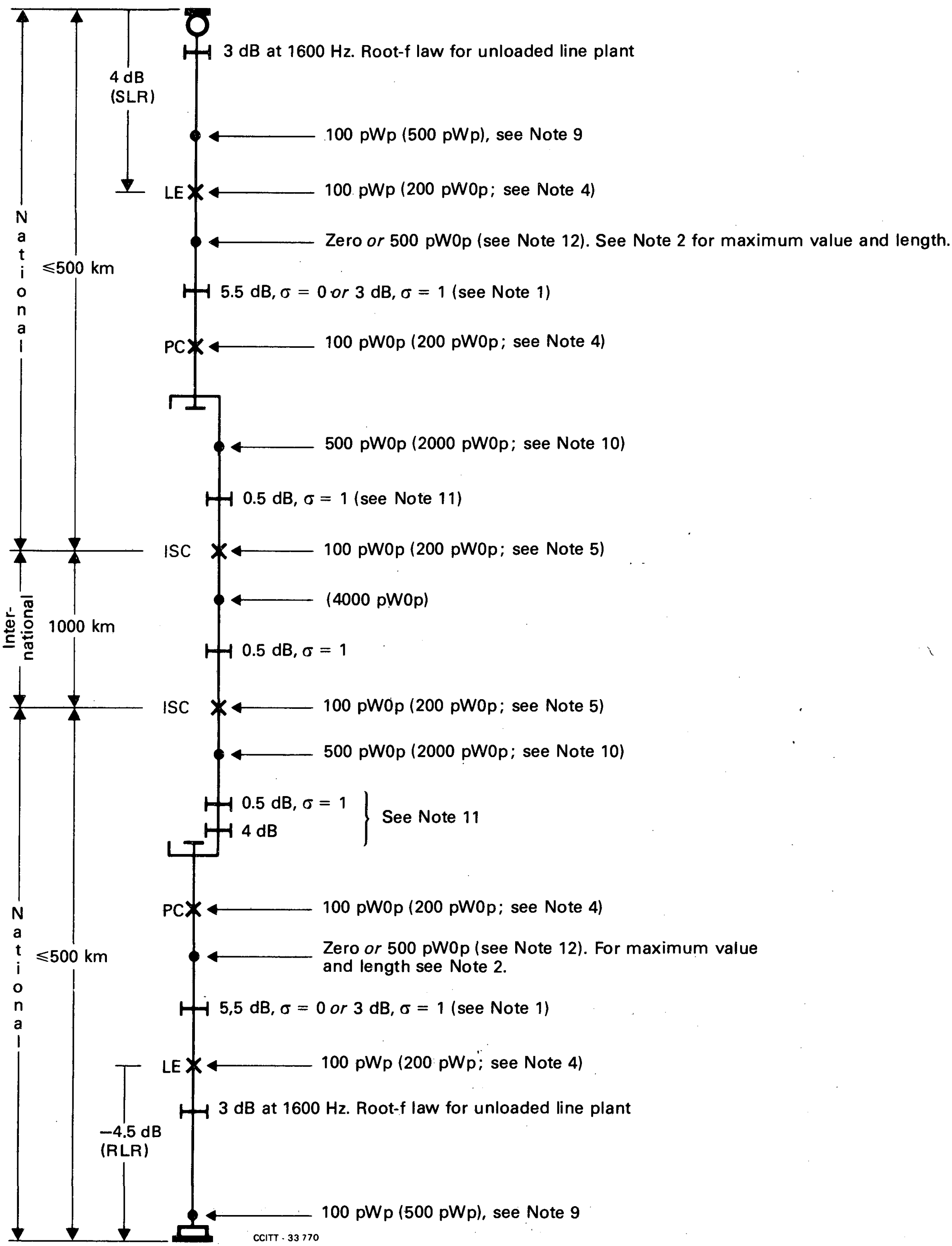


FIGURE 2/G.103

An example of an international connection of moderate length with only one international circuit

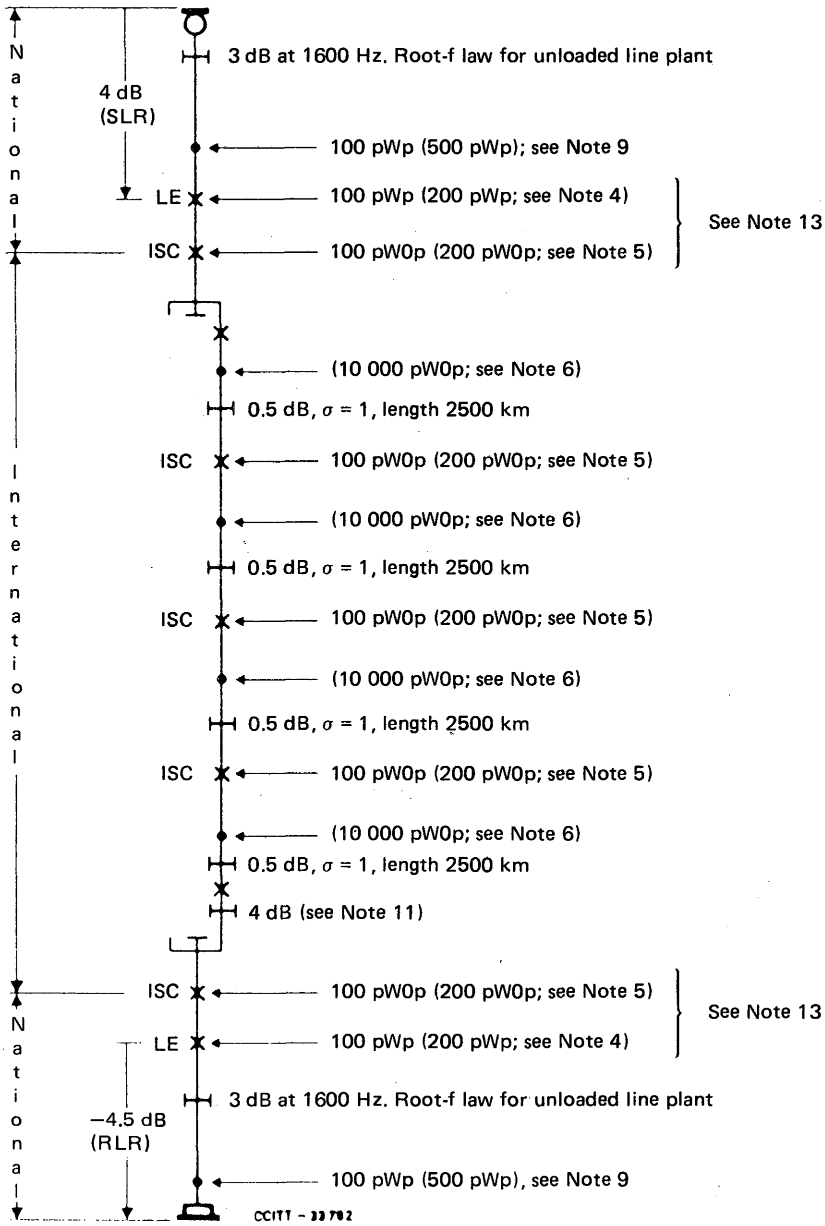


FIGURE 3/G.103

An example of an international connection of 4 international circuits between subscribers situated near the terminal ISCs

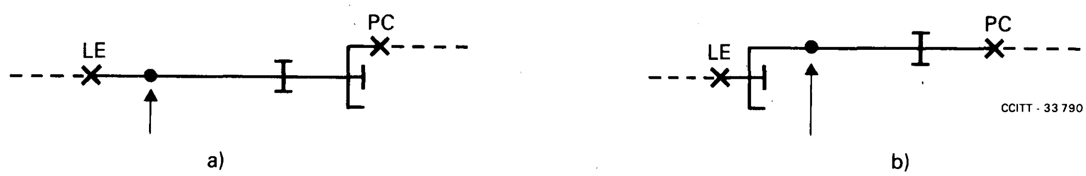
Notes to Figures 1/G.103, 2/G.103 and 3/G.103

Note 1 – For circuits on physical line plant the LR may be taken to have a nominal maximum value of 6 dB with $\sigma = 0$. This value was arrived at in the following way: Recommendation G.121 gives a 97% limit on 20 dB send loudness rating (SLR) referred to a point of −3.5 dBr on the international circuit at the IC. Referring this to a zero relative level point at the input to the chain of national and international circuits (i.e. to the primary centre) gives 16.5 dB. Reference [3] indicates that a 10.5 dB send loudness rating (SLR) is typical for maximum local lines, thus leaving 6 dB for the circuit from the local exchange to the primary centre, switching losses being included (see general remark 2.2.10).

For FDM or TDM short-distance carrier circuits which are 2-wire switched at the primary centre, the nominal value of the circuit loss may be taken as 3 dB with $\sigma = 1$. This loss is equal to the LR of the circuit; its loss-distortion effect is estimated by including an additional long-distance circuit in the connection (Recommendation G.111, § A.3.2). This circuit may for instance be provided on a PCM system using either 7-bit encoding ($\mu = 100$ or $A = 87.6$) or 8-bit encoding ($\mu = 225$ or $A = 87.6$). Although only 8-bit coding is recommended by CCITT, a nonrecommended 7-bit coding is used in some countries.

Note 2 – For FDM or TDM short-distance carrier circuits not exceeding about 250 km, the maximum value of noise power may be taken to be 1000 pW0p. See Recommendation G.123.

Note 3 – The following arrangements may be encountered if 4-wire switching (space-division or time-division) is used at the primary centre. Clearly in principle the terminating set may be at any point between the 2-wire switch and the 4-wire switch, although in practice it is ordinarily associated with one or the other.



If arrangement b) is adopted, then the minimum loss *a-t-b* (called for in accordance with Recommendation G.122) must still be assured, irrespective of whether the national transmission plan uses the 3.5 + 0 + 0 + 0 or 2.5 + 0.5 + 0.5 + 0.5 basis, since there could now be an extra circuit in the 4-wire chain. Where an additional 0.5 dB is needed, this could in principle either be introduced by changing the loss of the tertiary centre/ISC circuit from 0 to 0.5 dB, or by allocating it to the PC/LE circuits. Such arrangements may be encountered at either end of the connection.

Note 4 — The value of 200 pW0p as the design objective for the maximum noise power in a national 4-wire automatic exchange is taken from Recommendation G.123, § 3. The same value, i.e. an absolute noise power of 200 pWp, has provisionally been assumed for national 2-wire exchanges. No assumption has been made concerning the position of any national zero relative level point.

Note 5 – The value of 200 pW0p as the design objective for the maximum noise power in an international exchange is that recommended in Recommendation Q.45 [4].

Note 6 – The noise value corresponds to a design objective of 4 pW0p/km for the most adverse noise power during the busy hour.

Note 7 — The average value of 7500 pW0p for the ISC/ISC circuits assumes that 1 pW/km is the average value for line noise power. For the worst circuit, 3 pW/km is the design objective leading to the limit of 22 500 pW0p. Companders would be used to improve noise only if it exceed 40 000 pW0p (see Recommendation G.143).

Note 8 – Both countries are assumed to have the 3.5 + 0 + 0 + 0 dB type of plan. The nominal value of the pad in the receiving direction at the primary centre includes the loss of the terminating unit (see General Remark 2.2.10).

Note 9 – The average value of 100 pWp, for subscriber line noise is considered to be typical and is used by at least one Administration as an objective for maximum noise at the receiver.

Note 10 – The maximum value of 2000 pW0p provides for a circuit length of about 500 km with some margin.

Note 11 — Both countries are assumed to have the 2 + 0.5 + 0.5 + 0.5 dB type of plan. The nominal value of the 4 dB pad in the receiving direction at the switching centre includes the loss of the terminating unit (see General Remark 2.2.10).

Note 12 – The noise power level may be taken as negligible if the circuit is provided on physical line plant. A mean value of 500 pW0p is appropriate if the circuit is provided on a *FDM* or *TDM* short-distance carrier system.

Note 13 — The local exchange and primary centre are assumed to be both co-sited with the ISC.

2.2.6 The pad symbols represent the nominal loss of the particular channel or circuit, and the relative position of the noise generator, and the pad indicates that if the noise is to be referred to the receiving end of a circuit it must be modified by the power ratio corresponding to the loss of the pad.

If it is required to refer the noise powers to some particular point on the connection (for example, the receiving local exchange or the point of zero relative level on the first international circuit) then the rule to be applied is as follows:

If a noise power level at a point *A* is to be referred to a point *B* downstream of its position, it is obtained by augmenting the level at point *B* by the sum of the losses that is imagined to be traversed from *A* to *B*. If it is to be referred to a point *C* upstream of its position, it is obtained by diminishing the level at point *C* by the sum of all the losses that is imagined to be traversed from *A* to *C*.

2.2.7 The nominal terminal loss of the connection [i.e. the normal overall loss less the sum of the transit losses (via net losses) of the individual circuits] is shown as one pad associated with the extreme right-hand circuit in the 4-wire chain. This artifice enables the noise powers to be indicated as if they were injected at zero relative level points on the individual circuits as explained in Annex A.

2.2.8 Information concerning the distributions of attenuation distortion and group-delay distortion is to be found in Annex A of Recommendation G.113. Calculated values of some possible combinations of basic transmission impairments are given in Supplement No. 20, *Red Book*, Fascicle III.1.

Recommendation G.114 gives information concerning group delay.

2.2.9 The standard deviation of transmission loss of circuits is in accord with the objectives of Recommendation G.151 § 3 and also with the results obtained in practice and specified in [1].

2.2.10 "Circuit" in these reference connections is defined in the sense of Recommendation M.700 [2] as the whole of the line and the equipment proper to the line; it extends from the switches of one exchange to the switches of the next. In this way switching and exchange cabling losses are included in the values of transmission loss assigned to the circuits, together with the loss (or gain) introduced by the transmission system. If it is required to separately distinguish exchange losses, an additional pad symbol of appropriate value may be used.

It should also be noted that, according to this convention, the 3.5-dB loss ordinarily assigned to a terminating set does not figure explicitly in 2-wire/4-wire circuits; its value is also included in the loss assigned to the circuit.

3 Number of modulation and demodulation equipments

For the study of transmission performance, the longest international connection expected to occur (see Figure 1/G.103) may be considered to have the following arrangement of modulator/demodulator pairs in the 4-wire chain.

TABLE 1/G.103

	Number of modulator/demodulator pairs in a wholly analogue 4-wire chain		
	Eight national circuits	Circuits between ISCs	Total
Channel	8	4	12
Group	12	10	22
Supergroup	16	20	36

Of the 12 channel modulator/demodulator pairs a maximum of three may be of the special type which provide more than 12 telephone circuits per group.

4 Developments arising from the introduction of PCM digital processes

The worldwide telephone network is undergoing a transition from what is largely an analogue network to a mixed analogue/digital network. Looking farther into the future, this transition is expected to continue and result in a network that would be predominantly digital. Background on this transitional process is given in Recommendations G.101, § 4.1 and G.104.

With reference to the hypothetical reference connections of Figures 1/G.103, 2/G.103 and 3/G.103, the configurations used concerning numbers of circuits and numbers of exchanges should also be appropriate for network conditions in the mixed analogue/digital period. However, for transmission studies pertaining to mixed analogue/digital connections, account must also be taken of all unintegrated digital processes that might be present. Such unintegrated digital processes could have an important effect on overall transmission performance particularly with regard to such parameters as quantizing distortion (Recommendation G.113), and transmission delay. Guidance is provided on the use of appropriate hypothetical reference connections for a mixed analogue/digital network in Annex B.

Where the worldwide network becomes all-digital, many of the transmission impairments that were present in the mixed analogue/digital period, due to the incorporation of unintegrated digital processes, would be eliminated. However, certain processes might remain which could introduce transmission penalties. These are the processes which operate on the basis of recoding the bit stream as is done, for example, in the case of digital pads. Although the accumulated transmission impairments introduced by such processes may be well within recommended limits, the resulting loss of bit integrity could be an important disadvantage. This is particularly true in the case of services requiring the preservation of bit integrity on an end-to-end basis. Consequently, processes of this type should be avoided where possible, or appropriate arrangements made to circumvent them, where services requiring bit integrity are to be carried over the affected connections.

ANNEX A

(to Recommendation G.103)

An explanation of how hypothetical reference connections can be drawn assuming all send switching levels are 0 dBr

A.1 Consider the connection shown in Figure A-1/G.103 in which 3 circuits with losses of 1 dB, 6 dB and 2 dB are connected together by exchanges with actual send switching levels of −2, +1 and −3 dBr.

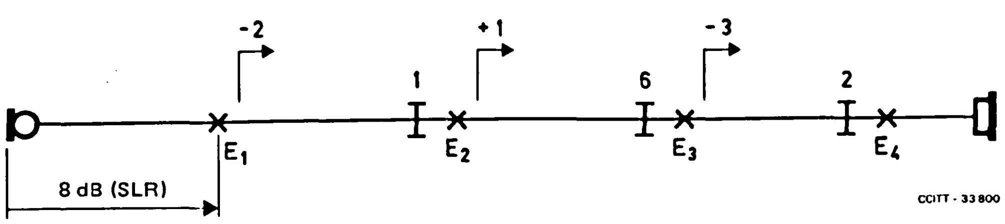


FIGURE A-1/G.103
Connection with various send switching levels

A.2 We assume that noise powers of these circuits are N_1, N_2 and N_3 pW0p respectively. Figure A-2/G.103 shows these noise powers entering their circuits via appropriately valued pads chosen to take cognizance of the switching level concerned and dispense with the arrow symbols.

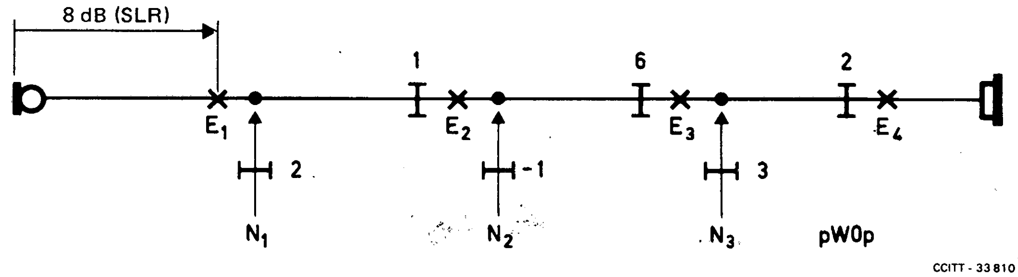


FIGURE A-2/G.103
The noise powers added

A.3 We note that N_1 traverses a total of 11 dB to reach E_4, N_2 a total of 7 dB, and N_3 a total of 5 dB. Also the difference between the accumulated send loudness rating (SLR) at each exchange and the corresponding circuit noise level is 6 dB (for N_1), 10 dB (for N_2) and 12 dB (for N_3). Hence we may redraw the connection reallocating the losses as shown in Figure A-3/G.103 in which all send switching levels are 0 dBr and all the other conditions are met as well.

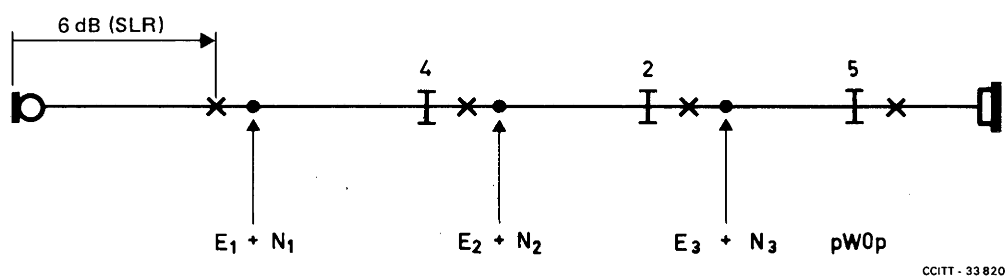


FIGURE A-3/G.103
All send switching levels are 0 dBr

A.4 Since the relative level of the immediate downstream circuit at each switch point is now arranged to be 0 dBr, the exchange noise powers can be added as is done in the hypothetical reference connections in Recommendation G.103.

ANNEX B

(to Recommendation G.103)

Guidance on hypothetical reference connections for a mixed analogue/digital connection

This annex provides guidance on a method to model a mixed analogue/digital network. For simplicity and for ease of comparison with an all-analogue network, retention of the network configurations now given in Figures 1/G.103 to 3/G.103 is appropriate. Figures 1/G.103 and 2/G.103, in particular, represent respectively, examples of the longest, though infrequent, type of connection and a connection of moderate length which occurs most frequently. The three connections provide an adequate range of connection types for most purposes but some guidance is desirable with respect to the selection of the circuits and exchanges which should be analogue and those which should be digital. This choice may depend on the matter under study. Two examples are designated for each of the connections: one which maximizes the number of digital processes and one which would be more representative of an evolving network. The worst case situation can be represented by making all of the exchanges digital and leaving all of the circuits analogue. A set of more representative connections is obtained by defining islands of digital connectivity such that the number of independent digital processes in each connection is approximately one-half of the maximum. For the representative connections all exchanges are assumed to be digital. In addition, the specific circuits designated in Table B-1/G.103 are also assumed to be digital with digital connection to the digital switches at each end of the circuit. This has the effect of creating "digital islands" with integrated digital processes, such that each island may be regarded as a single digital process.

TABLE B-1/G.103

Assumed digital circuits (listed from top to bottom)		
Figure 1/G.103	Figure 2/G.103	Figure 3/G.103
PC to SC TC to QC 1st ISC to 2nd ISC 4th ISC to 5th ISC QC to TC SC to PC	PC to ISC ISC to PC	LE to ISC 2nd ISC to 4th ISC [a)] ISC to LE

a) Single digital island.

Note – For an explanation of abbreviations, see Figure 1/G.103.

References

[1] CCITT *Green Book*, Vol. IV.2, Section 4, Supplements, ITU, Geneva, 1973.

[2] CCITT Recommendation *Definitions for the maintenance organization*, Vol. IV, Rec. M.700.

[3] CCITT manual *Transmission planning of switched telephone networks*, ITU, Geneva, 1976.

[4] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45.

Recommendation G.105

HYPOTHETICAL REFERENCE CONNECTION FOR CROSSTALK STUDIES

(Geneva, 1980)

1 Purpose

This Recommendation gives guidance concerning the application of Recommendation P.16 [1] in the general switched telephone network and recommends the structure and parameters of a hypothetical reference connection specifically designed for crosstalk studies.

2 General remarks

2.1 *Accuracy of fundamental data*

2.1.1 There is always some degree of uncertainty in applying to real telephone conversation the results of tests in which subjects were asked to listen attentively to see if they were able to detect the presence of intelligible crosstalk. Furthermore, this type of test cannot be expected to indicate reliably the extent to which a subscriber's confidence in the privacy of his own conversation is undermined by overhearing another conversation. Hence in general the aim should be to reduce the risk of potentially intelligible crosstalk as much as possible.

2.1.2 In applying the calculation method given in Recommendation P.16 [1], errors can occur if the distributions of crosstalk attenuations and loudness ratings are skew, rather than normal, or are truncated by test acceptance procedures. This arises because we are generally seeking low probabilities of encountering intelligible crosstalk which are highly dependent on the tails of distributions being accurately defined. One way of avoiding this difficulty is to apply Monte-Carlo methods as described, for example, in the CCITT manual cited in [2], taking care to make enough iterations to secure the necessary accuracy.

2.1.3 Considerable care must be taken to obtain representative values of the loss and noise in crosstalk paths being studied. In particular, errors arising from small changes in mean values can easily result in the calculated probability of overhearing being in error by a factor of 10 or more (see, for example, [3]).

2.2 *Effect of line and room noise*

2.2.1 The masking effect of line noise is another aspect which is important and raises some difficulties. On the one hand if, for the purpose of establishing crosstalk limits, the level of line noise is assumed to be negligible, unrealistic demands may be placed on the crosstalk attenuation required to be introduced by items of plant. On the other hand, if it is assumed that circuits and exchanges in service introduce noise power levels comparable with their design objectives, e.g. the well known 4 pW0p/km, the incidence of overhearing may be unacceptably high, particularly when the network is lightly loaded so that noise power levels can be expected to be at their lowest.

As in many transmission studies, a compromise has to be made somewhere between these extremes. In some cases, it may be necessary to rely on measurements of noise power levels on established plant during light and busy traffic periods. However, it must not be overlooked that limits devised now must, if possible, take the future into account. It is a wise principle that the successful performance of equipment in one part of the network should not be dependent upon adventitious imperfections of other parts of the network, particularly if such imperfections are likely to be eliminated or reduced in the future, e.g. by new designs of local exchange or by the extensive use of digital long-distance transmission systems.

2.2.2 Unlike line noise the effect of room noise can be reduced by a determined listener. Hence Recommendation P.16 [1] recommends that negligible room noise be assumed when deriving a design objective for equipment.

2.3 *Probabilities and distributions involved*

2.3.1 When constructing the distribution of crosstalk attenuation introduced by equipment and cables, it is appropriate to consider only the worst (acceptable) values. For example, in a 10-pair cable only the worst disturber for each pair should be taken into account, i.e. 10 values. This distribution should not be diluted by the other 80 better values. In the busy period the worst potential disturber of a particular pair can be relied upon to be activated.

2.3.2 In respect of intelligible crosstalk between local calls established in the same local exchange network, the probability of a potentially disturbing subscriber making a call at the same time as the disturbed subscriber can be significantly low certainly in the case of residential subscribers, although this is probably not the case for business subscribers and PBXs. Information concerning this topic and showing how to calculate the probabilities concerned will be found in [4].

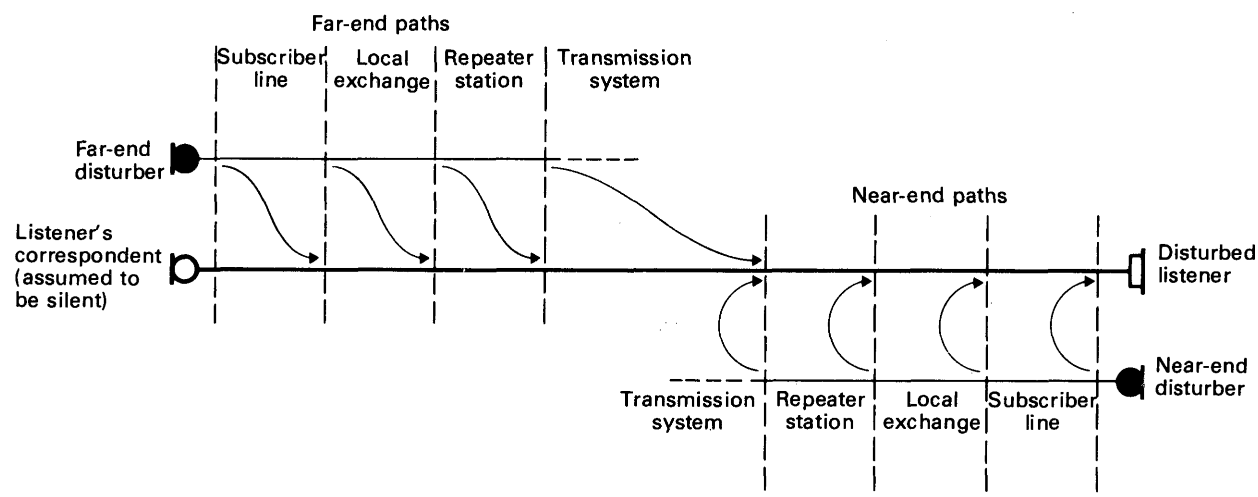
2.3.3 Multiple entries into a telephone connection of intelligible crosstalk signals all at significant levels and all derived from one source is so unlikely an event that it may be ignored for the purposes of deriving design limits. Hence the crosstalk mechanism of interest is assumed to be the dominant one when deriving limits, and all other sources are deemed to be negligible, and may thus attract the whole of the allowance.

However, when a network performance objective for crosstalk has to be divided among the exchanges and circuits making up the connection, it may be necessary to give some consideration to the number of potential crosstalk paths from different sources. For example, crosstalk limits may be assigned to complete paths through an exchange and to complete junction or trunk circuits. Thus, on simple other-exchange connections (ignoring, for the moment, crosstalk arising within local cables) there are three dominant sources of crosstalk, and if, for example, the aim were to be not greater than 1 in 100 for such connections, the probability of overhearing from each source should be reduced to 1 in 300 (assuming equal probabilities and no correlation between the sources).

Figures 1/G.105 and 2/G.105 illustrate some crosstalk paths of significance.

3 Hypothetical reference connections for crosstalk

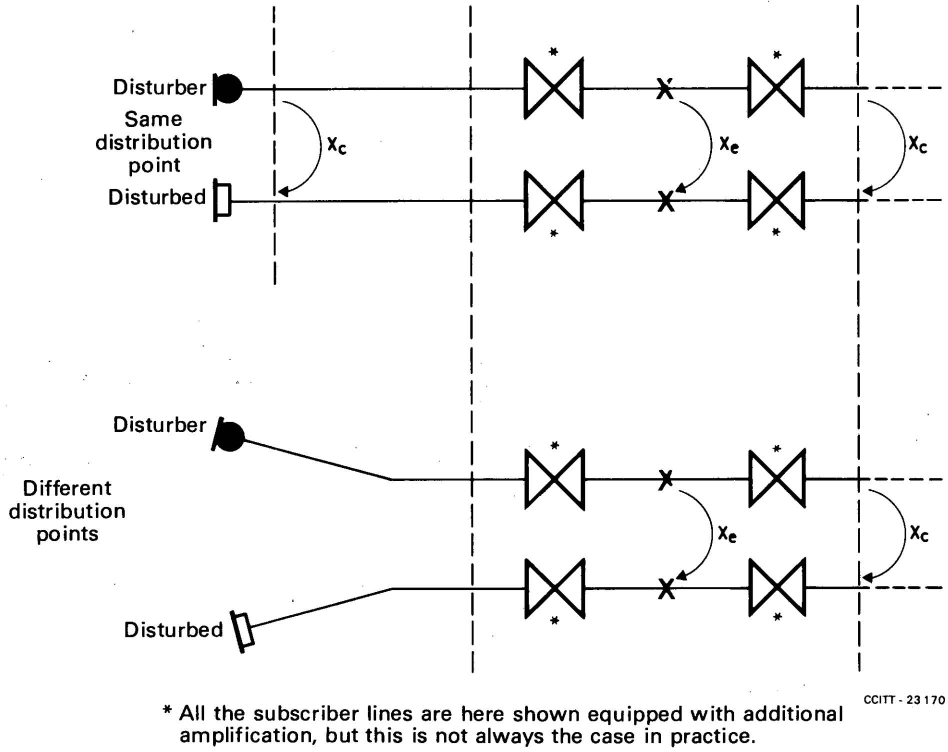
Figure 3/G.105 illustrates the essential elements of two hypothetical reference connections appropriate to crosstalk studies in respect of telephone circuits and exchanges. It will be observed that the connections are much simpler than the corresponding ones in Recommendation G.103 used for studying noise and loss. It would be inappropriate to study the risk of potentially intelligible crosstalk between a pair of 12-circuit connections of near maximum length and noise, in order to arrive at, for example, a limit for channel equipment crosstalk, because the majority use of the channel equipment bought and installed to the specification is in much simpler, quieter, and more numerous connections.



Note – Individual crosstalk limits for "repeater stations" (e.g. multiplexing equipment) and "transmission systems" are not the subject of this Recommendation which only deals with subscriber lines, exchanges, and interexchange circuits. In particular, limits recommended for circuits would be apportioned by the competent CCI Study Group(s).

FIGURE 1/G.105

Some far-end and near-end crosstalk paths of significance when considering potentially intelligible overhearing between telephone connections



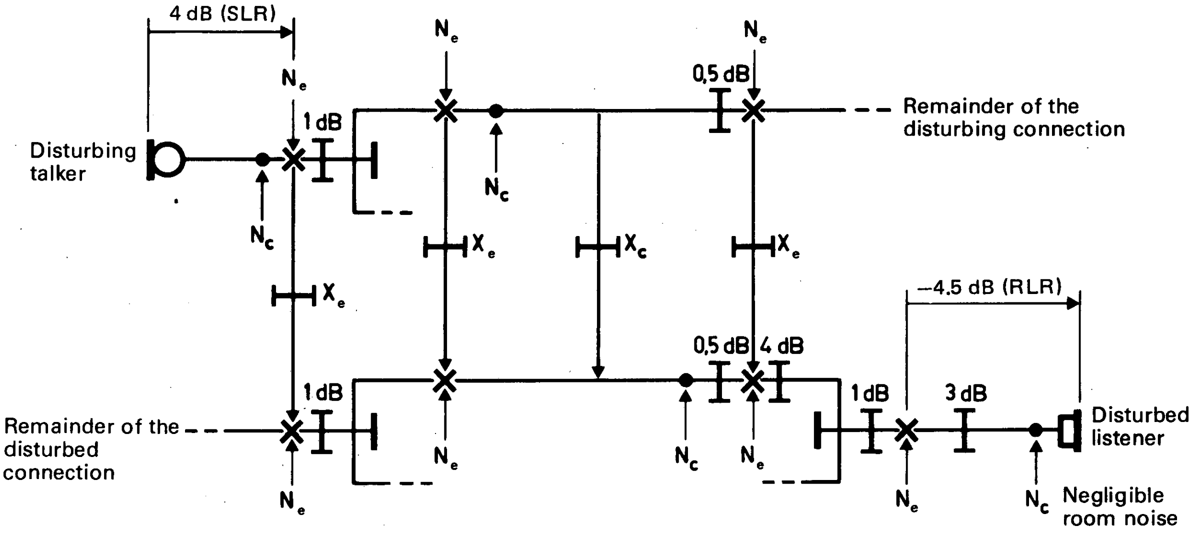
Note 1 – On own-exchange calls, overhearing between customers served by different distribution points may be assumed to be due only to exchange crosstalk or to crosstalk arising within local cables (near-end or far-end) on the far side of the exchange switching equipment. For other-exchange calls, the crosstalk paths are assumed to occur within the exchange and between junction or trunk circuits.

Note 2 – In the case of overhearing between customers served by the same distribution point, it should also be assumed that crosstalk can arise within the local cable (near-end crosstalk) or other permanently connected equipment. The particular customers who are unfavourably located in this respect will depend to a great extent on the type of local telephone circuits in use. When current-regulated telephones are used, customers on limiting length local lines are most at risk because the sensitivities of the telephone instrument are highest on these lines.

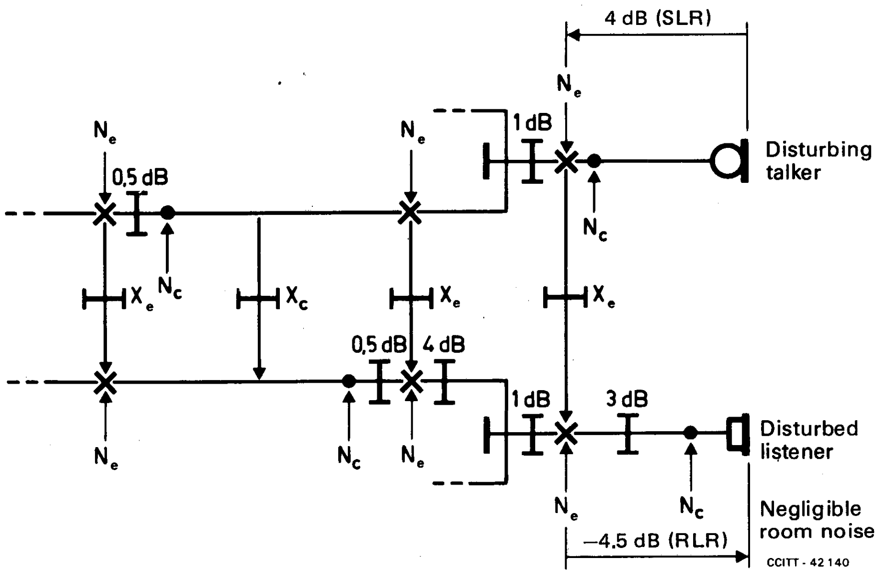
Note 3 – The effect of additional exchange amplification sometimes associated with long lines must be included where appropriate.

FIGURE 2/G.105

Some hypothetical crosstalk reference paths for studying crosstalk in the local exchange network



a) *Far-end crosstalk paths*



b) *Near-end crosstalk paths*

Note 1 — The disturbed connection is taken to be a very simple one, the disturbed listener being connected to a local exchange co-sited with the trunk exchange (e.g. the first ISC or the national primary centre).

Note 2 – Suitable values for the various circuit and exchange noise powers are

Circuit noise (N_c):	subscribers local line:	100 pWp
	4-wire circuit:	500 pW0p
	(Satellite circuit:	10 000 pW0p)
Exchange noise (N_e):	local exchange:	50 pWp or pW0p (as appropriate)
	4-wire exchange:	100 pW0p

Note 3 – In accordance with the convention adopted in Recommendation G.103, the send switching level at all exchanges is shown as 0 dBr. In practice, other values of relative level are encountered and must be taken into account in the study.

Note 4 – Only one crosstalk mechanism is assumed to be dominant at any one time.

FIGURE 3/G.105

Hypothetical reference connections for crosstalk between switched telephone connections

References

[1] CCITT Recommendation *Subjective effects of direct crosstalk; Thresholds of audibility and intelligibility*, Vol. V, Rec. P.16.

[2] CCITT Manual *Transmission planning of switched telephone networks*, ITU, Geneva, 1976.

[3] *Social Crosstalk in the Local Area Network*, Electrical Communication (ITT), Vol. 49, No. 4, pp. 406-417, 1974.

[4] LAPSA (P. M.): Calculation of multidisturber crosstalk probabilities, *Bell System Technical Journal*, Vol. 55, No. 7, September 1976.

1.1 General recommendations on the transmission quality for an entire international telephone connection

Recommendation G.111

LOUDNESS RATINGS (LRs) IN AN INTERNATIONAL CONNECTION

(Geneva, 1964; amended at Mar del Plata, 1968, Geneva, 1972, 1976 and 1980; Malaga-Torremolinos, 1984 and Melbourne, 1988)

Preamble

Paragraphs 1 to 5 of this Recommendation apply in general to all-analogue, mixed analogue/digital and all-digital international telephone connections. However, where recommendations are made on specific aspects in § 6 for mixed analogue/digital or all-digital connections, § 6 will govern.

In the international transmission plan, the loudness rating (LR) between two subscribers is not strictly limited; its maximum value results from the various Recommendations indicated below.

The CCITT,

considering

(a) that loudness ratings (LRs) as defined in Recommendation P.76 have been determined by subjective tests described in Recommendation P.78 and that the difference between the values thus determined in various laboratories (including the CCITT Laboratory) are smaller than for reference equivalents;

(b) that for planning purposes, LRs are defined by objective methods as described in Recommendations P.65, P.64 and P.79;

(c) that the conversion formulae from reference equivalents (REs) and corrected reference equivalents (CREs) (see Annex C) are not accurate enough to be applied to specific sets; that therefore, the Administrations who still rely on values of reference equivalents (determined in the past in the CITT Laboratory) for the type of sets used, will need to find recommended values of corrected reference equivalents in CCITT documentation, *recommends*

that the values given below, either in terms of LR should be used to verify that international telephone connections provide an adequate loudness of received speech.

that Administrations employing CREs should preferably translate the LRs of this Recommendation into their national CREs by the methods given in Annex C or, as a second choice, apply the values given in Volume III of the *Red Book*.

Note 1 – The main terms used in this Recommendation are defined and/or explained in Annex A.

Note 2 – For many telephone sets using carbon microphones, the SLR and STMR values can only be determined with limited accuracy.

1 Nominal LRs of the national systems

1.1 *Definition of* nominal LRs of the national systems

Send and receive loudness ratings, SLRs and RLRs respectively, may in principle be determined at any interface in the telephone network. When specifying SLRs and RLRs of a national system, however, the interface is chosen to lie at the international exchange.

An increasing number of international systems will be connected to national systems via a *digital* interface, where by definition the relative levels are 0 dBr. Therefore, in this Recommendation and in Recommendation G.121, the SLRs and RLRs of the *national systems* are referred to a *0 dBr point* at the international exchange. (See Recommendation G.101, § 5). This convention is applied both for digital and analogue interconnections between the national and international systems (unless otherwise specified in particular cases).

If these interconnections are made on an analogue basis, however, the actual relative levels at the interface may be chosen by the Administration concerned. Thus, if the standardized relative levels at the analogue interface are S dBr and Q dBr for the (national) sending and receiving systems respectively, the relation between the actual LRs at the interface and a 0 dBr point are

SLR (Interface) $= SLR - S$

RLR (Interface) $= RLR + Q$

(see Figure 1/G.111).

Moreover, for transmission planning purposes, the concept of the virtual analogue switching point (VASP) has often been used. The VASPs generally have no physical existence but have been found to be convenient when studying all-analogue and mixed analogue/digital connections. If the international section is analogue, or mixed analogue/digital, the relative levels at the VASP are by convention:

$S = -3.5$ dBr

$Q = -4.0$ dBr.

Note 1 – $Q = -4.0$ dBr corresponds to a 0.5 dB nominal loss between the VASPs of the international circuit. However, if a single international circuit is used only for comparatively short and straightforward international connections, this loss may be increased if the use of echo control devices can thereby be avoided. See Recommendation G.131, § 2.1. Thus, in such cases the value of Q will be decreased accordingly.

Note 2 – If the international analogue circuit exhibits an appreciable attenuation distortion with frequency, the overall loudness rating (OLR) of the international connection may increase slightly more than the nominal loss between the VASPs. See § A.4.2.

The concept of VASP has also been used when the international circuit was digital. The convention is then:

$S = -3.5$ dBr

$Q = -3.5$ dBr.

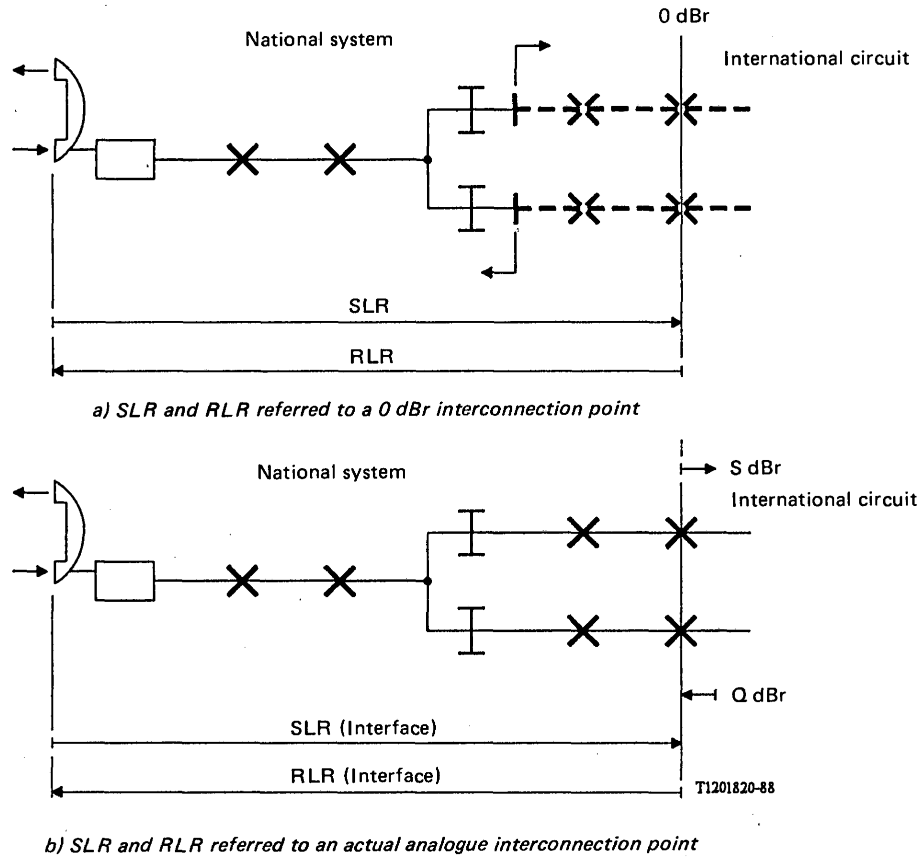


FIGURE 1/G.111

Definition of SLR and RLR reference points for a national system

1.2 *Recommended values*

Recommendation G.121 gives objectives for the nominal SLR and RLR of national systems.

2 Nominal overall loss of the international chain

The nominal loss between the virtual switching points of each international analogue circuit should in principle be 0.5 dB at 1020 Hz. However, some circuits can be operated with higher losses (see Recommendation G.131, § 2.1) and certain analogue circuits may be operated at zero loss (see Note 3 of Recommendation G.101, § 5). Digital circuits are used with a nominal transmission loss of 0 dB (see § 6).

As far as transmission is concerned, there is no strict limit on the number of international analogue circuits which may be interconnected in tandem, provided each of them has a nominal loss, between the virtual switching points, of 0.5 dB in the transit condition and provided there is a 4-wire interconnection. Naturally, the fewer the number of interconnected circuits the better the transmission performance is likely to be (see Recommendation G.101, § 3).

Note – Information on the actual number of circuits which are found in international connections is given in Recommendation G.101, § 3.

3 LRs and directional effects in a complete connection

3.1 *Nominal LRs for each transmission direction*

Paragraphs A.3 and A.4 of Annex A show how to calculate OLR, the overall loudness rating, of a complete connection. The nominal OLR of an international connection is the sum of:

- the nominal SLR, send loudness rating, of the national sending system (see Recommendation G.121, § 4, and Annex A);
- the nominal CLR, circuit loudness rating, of the international chain (see Annex A);
- the nominal RLR, receive loudness rating, of the national receiving system (see Recommendation G.121, § 4, and Annex A).

3.2 *Traffic-weighted mean values of OLRs*

For connections under practical conditions a suitable value of OLR seems to be 10 dB in most cases.

Note – For connections totally free from echo and sidetone problems, investigations have shown the optimum OLR to be somewhat lower, about 5 dB, but the optimum is rather flat so that moderate deviations from the given value have little subjective effect. (However, the "preferred OLR" in a particular application will to some extent depend on what subscribers have become used to. Thus, in some analogue PBXs, internal calls have had a very low OLR. Replacing such a PBX by a digital PBX having a higher OLR might cause some subscriber comments on "low speech levels". Some Administrations have solved this problem by means of a manual volume control in the receive part of the telephone set, the total range of variation being in the order of 10 to 12 dB. Note that in mobile telephony a common practice is to include a volume control which affects both the receive and send sides but in opposite directions.)

The long-term objective for the traffic-weighted mean value should lie in the range of 8 to 12 dB.

An objective for the mean value is necessary to ensure that satisfactory transmission is given to most subscribers.

Note 1 – The long-term values cannot be attained at this time and an appropriate short-term objective for OLR is a range of 8 to 21 dB.

Note 2 – The 0.5 dB transmission loss of each analogue circuit in the international chain (see § 2 above) has been allowed for by noting that the average number of international circuits encountered in international connections is 1.1. (See Recommendation G.101 § 3.)

As a result the ranges mentioned above do not include allowances for connections between countries which:

- involve more than one 0.5 dB international circuit;
- involve a single international circuit which has a higher loss than 0.5 dB as permitted by Recommendation G.131, § 2.1.

Note 3 – Recommendation G.121, § 1 gives values for national systems based on the overall objectives of this Recommendation.

Note 4 – The ranges stated for OLR are for planning and do not include measuring and manufacturing tolerances.

Note 5 – Besides loudness, other important factors have to be considered in transmission planning. Sidetone, echo and stability problems may cause degradation of the overall speech quality in a connection. Thus, it is important to adopt an adequate *impedance strategy* in the national transmission plan to avoid harmful mismatches in the network. (An example is given in Supplement 10 of Fascicle VI.1.)

3.3 *Difference in transmission loss between the two directions of transmission*

In an international connection between local exchanges the contribution to the asymmetry introduced by the two national systems is limited by the provisions of Recommendation G.121, § 2.2. The international circuits could, in practical circumstances outlined in the General Remarks in Recommendation G.101, § 4 introduce additional asymmetry. This additional asymmetry will be acceptably small.

4 Variation in time and effect of circuit noise

4.1 *Variations in time*

The LR values calculated for national systems (Recommendation G.121, § 4) do not cover variations in time of the loss of various parts of the national system. Recommendation G.151, § 3 gives the objectives recommended by the CCITT for transmission loss variations on international circuits and national extension circuits as compared with the nominal values.

4.2 *Effect of circuit noise*

See Recommendation G.113.

5 Practical limits of the OLR between two operators or one operator and one subscriber

The same loudness rating limits as between two subscribers should apply.

6 Incorporation of PCM digital processes in international connections

6.1 *Connections with a digital 4-wire chain extending to the local exchanges*

As the national network develops, an international telephone connection might have the configuration indicated in Figure 2/G.111, in which the analogue/digital interface occurs at the local exchange. In such a connection, the nominal transmission loss introduced by the 4-wire chain of national and international digital circuits is 0 dB. Consequently, the 4-wire chain generally does not contribute to the control of stability and echo. However, part of the loss required to control stability and echo is at the local exchange, as indicated by the R and T pads, the remainder being provided by the balance return loss at the 2-wire/4-wire terminating unit (see also Recommendation G.122).

Values of R and T are discussed in Recommendation G.121, § 6, where it is concluded that values can be chosen to cater for the national losses and levels, provided that the CCITT Recommendations for international connections are always met. For example, the sum of R and T will need to be at least so high that the requirements of Recommendation G.122 are met. This should be especially noted in cases when stability balance return losses approach 0 dB at the 2-wire/4-wire terminating unit. Examples of values for R and T that have been adopted by some Recommendations are given in Annex C to Recommendation G.121.

Other transmission considerations to be taken into account in the planning of connections involving 4-wire local exchanges in a mixed analogue/digital network include system loading and crosstalk.

Figure 2/G.111 also shows R and T as analogue pads. This need not always be the case since under some conditions it might be more practical or necessary to introduce the required loss by means of digital pads. However, if digital pads are used, their detrimental effect on digital data or other services requiring end-to-end bit integrity must be taken into account as indicated in Recommendations G.101, § 4.4 and G.103, § 4.

6.2 *Mixed analogue/digital connections*

To provide satisfactory transmission on international connections in the mixed analogue/digital period, it is likely that existing national transmission plans will have to be amended or new ones developed to provide for appropriate national extensions. All the relevant CCITT Recommendations should be complied with. The Recommendations concerning national extensions with 4-wire chains extending 4-wire local exchanges are given in Recommendation G.121, § 6.

Thus, the transmission planning of transition phases should preferably not involve any degradation of the quality previously experienced.

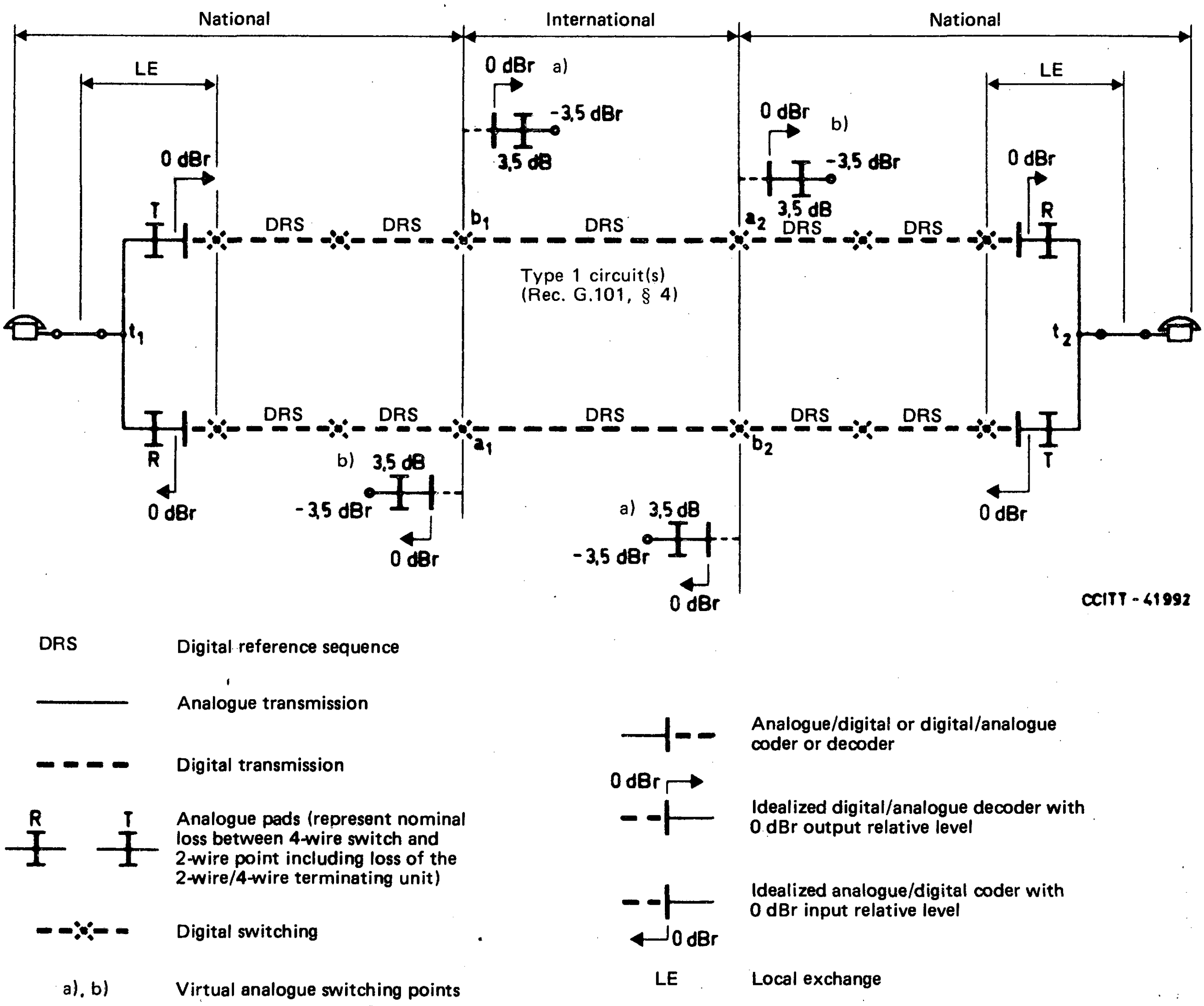


FIGURE 2/G.111

Example of an international connection in which the digital 4-wire chain extends to a 4-wire local exchange with 2-wire analogue subscriber lines

ANNEX A

(to Recommendation G.111)

Explanations related to Recommendations G.111, G.121, G.122 G.131, G.134: properties and uses of loudness ratings

Note – The CCITT *definitions* of loudness ratings can be found in Volume V.

A.1 *General explanations of loudness rating terms as used in the Series G Recommendations*

A.1.1 *Loudness rating (LR)*

As used in the Series G Recommendations for planning; loudness rating is an *objective* measure of the loudness loss, i.e. a weighted, electro-acoustic loss between certain interfaces in the telephone network. (The nature of the weighting will be dealt with later.) If the circuit between the interfaces is subdivided into sections the sum of the individual section LRs is equal to the total LR.

How to determine and to apply LRs in the Series G Recommendations is described in §§A.3 and A.4. The methods are sufficiently accurate for all practical purposes. (Fundamentally, loudness ratings are based on subjective methods as described in Recommendations P.76 and P.78. However, subjectively measured values, in general, vary too much with time and test teams to be really useful for transmission planning.)

In loudness rating contexts, the subscribers are represented from a measuring point of view by an artificial mouth and an artificial ear respectively, both being accurately specified.

A.1.2 *Overall loudness rating (OLR)*

The loudness loss between the speaking subscriber's mouth and the listening subscriber's ear via a connection.

A.1.3 *Send loudness rating (SLR)*

The loudness loss between the speaking subscriber's mouth and an electric interface in the network. [The loudness loss is here defined as the weighted (dB) average of driving sound pressure to measured voltage.]

A.1.4 *Receive loudness rating (RLR)*

The loudness loss between an electric interface in the network and the listening subscriber's ear. [The loudness loss is here defined as the weighted (dB) average of driving e.m.f. to measured sound pressure.]

A.1.5 *Circuit loudness rating (CLR)*

The loudness loss between two electrical interfaces in the network (via a circuit), each interface terminated by its nominal impedance which may be complex. [The loudness loss is here approximately equivalent to the weighted (dB) average of the composite electric loss.]

Note – Junction loudness rating (JLR) is a special case of CLR, the terminations being 600 ohms resistive.

A.1.6 *Sidetone loudness losses*

A.1.6.1 *Talker's sidetone, sidetone masking rating (STMR)*

The loudness loss between a subscriber's mouth and his (earphone) ear via the *electric* sidetone path (see Recommendation P.10 for a full definition).

A.1.6.2 *Listener's sidetone rating (LSTR)*

The loudness loss between a Hoth-type room noise source and the subscriber's (earphone) ear via the *electric* sidetone path (see Recommendation P.10 for a full definition).

A.1.7 *Echo loudness losses*

A.1.7.1 *Talker echo loudness rating (TELR)*

The loudness loss of the speaker's voice sound reaching his ear as a delayed echo.

A.1.7.2 *Listener echo loudness rating (LELR)*

The difference in loudness loss between the speaker's direct voice sound and its delayed echo reaching the listening subscriber's ear.

A.1.8 *Crosstalk receive loudness rating (XRLR)*

The loudness loss from a disturbing electric interface to the disturbed subscriber's ear via the crosstalk path.

A.2 *Psycho-acoustic model for loudness ratings*

By the fundamental definition of loudness ratings, a *flat loss* (i.e. a loss constant with frequency) introduced in a path increases the loudness rating by the same amount. When evaluating the influence of a frequency-dependent loss, however, one needs a psycho-acoustic model of how the brain interprets loudness impressions. Therefore, a short description will be given of a simple model found adequate for loudness rating planning considerations. (See Recommendation P.79 for more complete explanations.)

The ear can be thought of as a bank of bandpass filters approximately equally spaced on a logarithmic frequency scale. If the sound signal in a certain band exceeds the threshold of hearing, the corresponding filter produces an output. All filter outputs are then added to create an impression of loudness, the rule of addition depending on the sound level.

For very *low* sound levels (near the threshold of hearing) the filter outputs are added on a power basis. For *normal* speech sound levels, the loudness measure can be described as obtained neither as power nor voltage addition but rather as the sum of the *logarithm* of the filter outputs. The procedure can be described by Equation (A.2-1) which covers sound levels from very low to normal. (This algorithm is in effect the same as the one given in Recommendation P.79, only written in a slightly different form.)

$$LR = L_0 - \frac{10}{m} \log_{10} \left\{ \sum_{i=1}^{N} K_i \cdot 10^{-0.1 m L_i} \right\} \qquad \text{(A.2-1)}$$

where

L_0 is a constant (for instance, L_0 is equal to 0 for CLR, LELR), depending on the particular LR in question.

N is the number of equivalent bandpass filters, the index i refers to filter No. i at frequency f_i. (Usually, the "filters" are chosen with a 1/3-octave spacing in the frequency scale. The appropriate frequency range to consider will be discussed later.)

L_i is the loss at f_i of the path under study. (Provided the sound level at that frequency is above the threshold of hearing.)

m (the "loudness growth factor") is a constant depending on the sound level:

m = 0.2 for normal speech levels,

m = 0.5 for "lower" sound levels (corresponding to voltage addition),

m = 1 for very low sound levels, near the threshold of hearing (corresponding to power addition).

$m = 0.2$ is applicable for OLR, SLR, RLR, JLR, CLR and sidetone phenomena, while $m = 0.5$ and 1 are appropriate for echo and cross-talk.

K_i is the weighting coefficient at f_i. The K_i's have the general property that their sum is equal to 1 in the frequency range considered:

$$\sum_{i=1}^{N} K_i = 1 \tag{A.2-2}$$

The K_i's are determined by the following factors:

a) voice spectrum of the "average" speaker;

b) hearing acuity of the "average" listener;

c) frequency response of the "nominal" path typical for the particular LR in question.

The shape of the K_i-weighting is not very critical. For transmission planning, most often a flat weighting will do. This topic is treated below in §§ A.3 and A.4.

Equation (A.2-1) can be applied in various loudness-related rating calculations. Examples may be found in Supplement No. 19, Volume V.

What frequency range should be used in the computations? For LR planning purposes, only that frequency range should be considered in which the transmission is assured. In general, this means from 300 Hz to 3400 Hz for international calls. However, for very weak speech sounds, such as just discernable crosstalk, the proper band for computation is narrower, in the order of 500 Hz to 2000 Hz. This is because the human hearing acuity falls off at the band edges for low level sounds.

Note – The K_i's are different for the 300-3400 Hz and the 500-2000 Hz bands.

It is immediately apparent again from Equations (A.2-1) and (A.2-2) that a flat loss of L dB will increase the LR by the same amount. It also turns out that if the spread in the L_i-values is *moderate*, Equation (A.2-1) can be simplified to:

$$LR = L_0 + \sum_{i=1}^{N} K_i \cdot L_i \tag{A.2-3}$$

This linear approximation is the reason why the total loudness rating of a connection can be computed by simply adding the loudness ratings of its parts. The procedures to follow will be discussed in § A.4. [A rule of thumb: if $m = 0.2$ and the spread in L_i is less than 10-15 dB, Equation (A.2-3) can be applied.]

A.3 *Measurement of loudness ratings of telephone sets*

The loudness ratings of telephone sets are determined objectively by special measuring instruments conforming to Recommendations P.64, P.65 and P.79 with regard to the physical implementation and computational algorithm respectively. For analogue sets, the measurement set-up must provide a representative current feeding bridge and may or may not include different lengths of (artificial) unloaded subscriber lines. The parameters usually measured are SLR, RLR and STMR.

The results should not be applied directly for transmission planning, however, before some precautions are observed regarding bandwidth and terminating impedances.

Commercial instruments following Recommendation P.79 use a measuring band of 200 to 4000 Hz or even 100 to 8000 Hz. This is a good deal wider than the band for which CCITT Recommendations specify an assured transmission, namely 300 to 3400 Hz. (See for instance Recommendations G.132 and G.151.) Thus, in a national system which may be included in an international connection one has to consider the analogue telephone set being somewhat less loud than the P.79-measured values.

Also note that the P.64-P.79 loudness rating measurements are specified to be made with a terminating impedance of 600 ohms. This is most often not the impedance appearing in the 2-wire part of the network. For various reasons, many Administrations now specify a complex nominal impedance. Thus, there will be a mismatch effect.

For SLR and RLR an investigation has been made for a range of typical analogue telephone set sensitivity and impedance characteristics as well as nominal impedances. The result is that, with sufficient practical accuracy, 1 dB should be added to the measured values of SLR and RLR of *analogue* telephone sets in the LR planning of networks which can be included in an international connection. Thus, with the designation SLRw and RLRw for the measured values:

$$\begin{aligned} SLR &= SLRw + 1 \\ RLR &= RLRw + 1 \end{aligned} \qquad \text{(A.3-1)}$$

Note that the same correction also applies when an unloaded subscriber cable is included in the P.79 measurements.

For *digital* sets, however, the correction is *not* needed because the coded and filters in the set limit the band to a certain extent.

In the following the designations SLR and RLR always refer to planning values. Specifically, SLR(Set) and RLR(Set) refer to the telephone set itself without subscriber cable, and including the 1 dB correction in the analogue case.

Parameters of further interest to the planner are of course the telephone set input impedance Z_c and/or its return loss against the nominal circuit impedance.

Note that for STMR measurements the line terminating impedance must be so specified that it represents realistic network conditions, i.e. a termination not necessarily 600 ohms.

In addition to straightforward STMR measurements, it is useful to determine the so-called "no sidetone line impedance" Z_{s0}, or equivalent sidetone balance impedance. Knowing Z_{s0} in addition to SLR and RLR, the transmission planner is able to estimate the sidetone performance better under the widely varying conditions which may occur in the network. See § A.4.3 for further details. (Note that Z_{s0} may vary with the line current.)

Listener sidetone may cause some subscriber difficulties when modern, high-sensitivity sets having linear microphones are used in noisy environments. To get a quantitive understanding of the problem, the set sending sensitivity curves for both direct (speech) sound and diffuse (room noise) sound should be measured. (See the *Handbook on Telephonometry* [4] and Recommendation P.64 for details.) The result is preferably persented as the difference:

$$DELSM = S_s\,(\text{diffuse}) - S_s\,(\text{direct}) \qquad \text{(A.3.2)}$$

(See § A.4.3.3.)

Note 1 – DELSM is fairly constant with frequency. (The diffuse field sensitivity measurements should be made with an obstacle resembling the human head in front of the handset microphone. The present practice is to use the LR artificial mouth as such an obstacle. However, the detailed measurement procedure is under study.)

Note 2 – The actual shape of the frequency-dependent K_i-weighting in the P.79 algorithm as used for telephone set measurements is of no immediate concern to the transmission planner. However, the P.79 weighting seems not to represent "ordinary people's" speech and hearing too well. Therefore, if one tries to analyze attenuation distortion and bandwidth limitation effects on loudness only, P.79 results must be interpreted with caution.

Note 3 – Up to now, when making national transmission plans, most Administrations have used other forms of objective measuring instruments to characterize the telephone sets. Translating such a transmission plan into terms of loudness ratings means a corresponding conversion of the "old" telephone set data. This should be done by actually *measuring* the loudness ratings of typical examples of the sets in use. (There is too much uncertainty in general conversion formulas to obtain LRs from RE, CRE, OREM-B, IEEE-Objective LR, etc.)

A.4 *Application of loudness ratings in the Series G Recommendations*

A.4.1 *General remarks*

Theoretically, one could determine the total attenuation/frequency response between the input and the output ports and compute the LR in question by an algorithm as given in § A.2. However, for transmission planning it is far more convenient to evaluate the LR of the *individual* parts. This is especially true for the present situation with a proliferation of different types of telephone sets allowed in most Administrations' networks. Therefore, in what follows the telephone set influence on loudness ratings will be characterized by its SLR and/or RLR value(s).

Most important in transmission planning for loudness performance is to have *consistent* rules – even if they are simple. To strive for high precision in the computations is rather illusory. For example, the subscriber may control the subjective loudness quite substantially with his handset: voluntarily by pressing it more or less strongly to his ear (10 dB range?) and involuntarily by moving the microphone away from its optimum position.

A.4.2 *Normal speech transmission*

Figure A-1/G.111 depicts a speech connection between two subscribers, consisting of several cascaded parts.

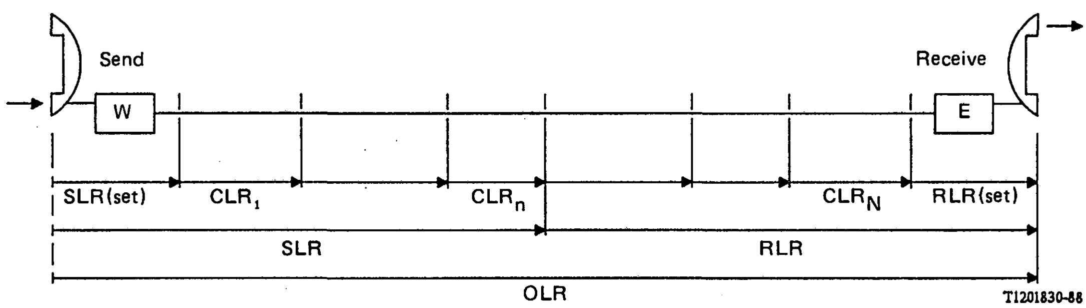


FIGURE A-1/G.111

LRs in a normal speech connection

The send and receive loudness ratings of the telephone sets themselves are designated as SLR(set) and RLR(set) respectively and the circuit loudness ratings as CLR_n. (For explanations, see § A.1.) Then, at interface $i = n$ in the direction from S to R we have:

$$\begin{aligned} SLR &= SLR(\text{set}) + \sum_{i=1}^{n} CLR_i \\ RLR &= RLR(\text{set}) + \sum_{i=n+1}^{N} CLR_i \\ OLR &= SLR + RLR \end{aligned} \tag{A.4-1}$$

SLR(set) and RLR(set) are determined (measured) according to the principles described in § A.3.

When the circuit loss is constant with frequency the CLR is, of course, equal to the composite loss at the reference frequency 1020 Hz, using the nominal impedances appropriate to the particular interfaces. Thus, normally the CLRs are equal to the *difference in relative levels* between the respective interfaces. (The exception occurs when the circuit includes an interface having a "jump" in the relative level. See Recommendation G.121, § 6.3 for a discussion.)

If the attenuation distortion is noticeable, the CLR is equal to the *average loss* over the frequency band 300 Hz to 3400 Hz on a logarithmic frequency scale, i.e. a flat K_i-weighting in Equation (A.2-3) and with the constant $L_0 = 0$. [If the attenuation distortion is exceptionally high Equation (A.2-1) should be used with $m = 0.2$] The loss is to be measured or computed as a *voltage loss*, corrected by a frequency-independent term, i.e. the loss is equal to the sum of the composite loss at 1020 Hz and the voltage loss deviation from the value at 1020 Hz. (This practice is in accordance with Recommendation G.101, § 5.3.2.2).

Note 1 – Some Administrations may instead, want to use the so-called composite loss distortion as a basis for computing the CLR of a circuit in their national transmission planning. Morever, the various aspects of the complete user end-to-end loss distortion is being studied by Study Group XII.

When the loss is determined by measurement it should be under nominally matched impedance conditions. In practice, this means *either* between two physical impedances as is the case for 600 ohms measurements *or* between a low-impedance generator and a high-impedance indicator. Either method can be used, depending on what is most practical. The measurement results do not differ very much. In the latter case, a 6 dB correction must of course be applied.

It is interesting to note that, for *unloaded subscriber cable* sections, the CLRs are equal to the composite loss at the reference frequency 1020 Hz with sufficient accuracy for planning purposes, that is, they are equal to the difference in relative levels at the interfaces. (It turns out that, from a loudness point of view, the lower losses at frequencies below 1020 Hz compensate the higher losses at frequencies above 1020 Hz).

Note 2 – In the particular case of a subscriber cable, the telephone set and the exchange may have different nominal input impedances. Strictly speaking, one should then consider "insertion loss" rather than "composite loss" as the basis for CLR, as a zero length line should be associated with $CLR = 0$. However, the impedance mismatch between set and exchange impedances usually does not result in a significant composite loss at 1020 Hz. Therefore, the designation "composite loss" may also be used in this case.

The CLR per kilometer of an unloaded subscriber cable can also be estimated from the cable characteristics by the following expression:

$$\dot{CLR} = K\sqrt{R \cdot C} \qquad \text{(A.4-2)}$$

where

R is the cable resistance in ohms/km

C is the cable capacitance in nF/km

K is a constant, the value of which is dependent on the cable termination:

$K = 0.014$, if $Z_0 = 900$ ohms resistive

$K = 0.015$, if $Z_0 = 600$ ohms resistive

$K = 0.016$, if Z_0 is a complex impedance.

Note 3 – "Complex impedance" means here such 3- or 2-element RC impedances as have been chosen by Administrations to resemble the image impedance of unloaded cables.

Note 4 – Equation (A.4-2) gives the image attenuation at about 800 Hz for $K = 0.014$, and at about 1020 Hz for $K = 0.016$. Some Administrations have been using the cable image attenuation at a certain frequency (for instance 1600 Hz) as a measure of the permissible attenuation in the subscriber network. However, the same numerical value should not be used automatically as the permissible CLR when transforming the transmission plan into terms of loudness ratings.

Note 5 – Most often the errors in CLR when using Equation (A.4-2) are less than 0.4 dB.

Most modern channel equipment, including digital exchanges, can be considered as having essentially flat attenuation/frequency characteristics when estimating CLRs. An example of a more pronounced channel attenuation distortion can be found in Recommendation G.132, dealing with attenuation/frequency distortion limits for 12 4-wire circuits in tandem. Assuming a maximum attenuation variation curve just touching the *upper* corners in Figure 1/G.132, a calculation shows that the attenuation distortion contributes 2.4 dB to the CLR which is to be added to the loss value at 1020 Hz (i.e. about 0.2 dB per circuit.)

Note 6 – An $OLR = 9$ dB may be considered as being well within the optimum range for connection loudness. Interestingly, at that value the average acoustic loss from the speaker's mouth to the listener's ear is about 0 dB, taken over a logarithmic frequency scale.

A.4.3 *Sidetone*

A.4.3.1 *General remarks*

As mentioned above, the sidetone quantities STMR and LSTR refer specifically to the signals reaching the ear via the *electric* sidetone path.

A.4.3.2 *Talker's sidetone STMR*

STMR can be *measured* as discussed in § A.3, using the actual terminating impedances occurring in the network.

In many circumstances it may be more convenient to *compute* STMR from telephone set data and network data.

For transmission planning purposes, one can use the telephone set loudness ratings and the balance return loss between the line impedance and the sidetone balance impedance. In practice, the following algorithm is generally sufficiently accurate:

$$STMR = SLR\,(\text{set}) + RLR\,(\text{set}) + A_m - 1 \qquad \text{(A.4-3)}$$

where

SLR (set), RLR (set) refer to the telephone set as before. A_m is a weighted mean of the sidetone balance return loss A_{rst}:

$$A_m = -\frac{10}{m}\log_{10}\left\{\sum_{i=1}^{N} K_i - 10^{-0.1mA_{rst}}\right\} \qquad \text{(A.4-4)}$$

where:

$m = 0.2$; the K_i's are found in Table A-1/G.111; and

$$A_{rst} = 20\log_{10}\left|\frac{Z_c + Z_{s0}}{2Z_c} \cdot \frac{Z + Z_c}{Z - Z_{s0}}\right| \qquad \text{(A.4-5)}$$

Here,

Z_c is the input impedance of the set

Z_{s0} is the sidetone balance impedance of the set (equivalent)

Z is the impedance of the line, "seen" by the set when the connection is established.

Note 1 – A_{rst} is about equal to the return loss between Z_{s0} and Z.

Note 2 – When the *actual* telephone set send and receive sensitivity curves as functions of frequency are known, it is possible to closely simulate STMR measurements by a more elaborate algorithm (Recommendation P.79, § 8).

As can be seen from Table A-1/G.111 and Figure A-2/G.111 there is very little emphasis on the lower frequencies in the STMR weighting. This is because the "human sidetone" path via head bone conduction dominates over the electric path in that frequency range.

Note 3 – $STMR = 7$ or 8 dB is well within the preferred range of talker's sidetone. At that value the average acoustic loss from the talker's mouth to his ear via the electric sidetone is typically about 0 dB (averaging done with the K_i weighting given in Table A-1/G.111).

TABLE A-1/G.111

STMR weighting

i	F_i (kHz)	K_i
1	0.2	0
2	0.25	0.01
3	0.315	0.02
4	0.4	0.03
5	0.5	0.04
6	0.63	0.05
7	0.8	0.08
8	1	0.12
9	1.25	0.12
10	1.6	0.12
11	2	0.12
12	2.5	0.12
13	3.15	0.12
14	4	0.05

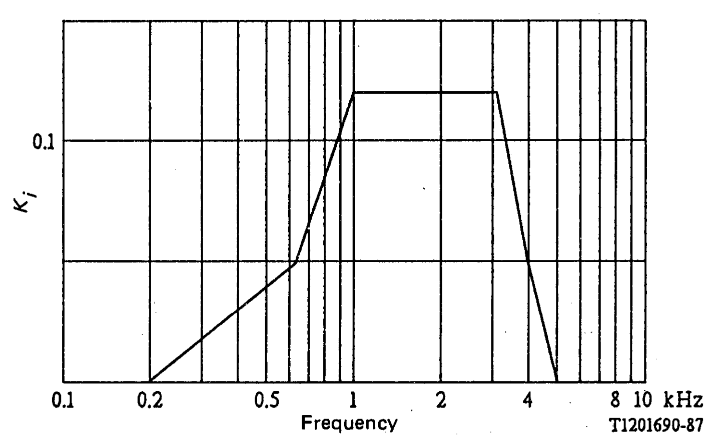


FIGURE A-2/G.111

Sidetone weighting K_i as given in Table A-1/G.111

A.4.3.3 *Listener's sidetone rating (LSTR)*

High room noise at the listening subscriber's premises disturbs the received speech in several ways:

a) By noise picked up by the "free" ear. This disturbance can be disregarded here because the brain has a stereophonic analysis ability to "switch off" irrelevant signals coming from the wrong direction.

b) By noise leaking past the ear at the handset ear.

c) By noise picked up by the handset microphone and transmitted to the handset ear via the electric sidetone path.

In practice, the phonemena under c) often are the most troublesome. (Of course, they are also the only ones within the control of the transmssion planner.)

Investigations have shown that, at low frequencies, the earcap leakage dominates over the electric sidetone path in much the same way as bone conduction does for the talker's sidetone. Therefore, the same K_i-weighting as for STMR can be applied. (At least if the ear-phone cap is not too awkwardly shaped.) Thus, the LSTR may be computed from STMR and the weighted mean of DELSM, the difference between diffuse and direct sound sensitivity curves of the set (see § A.3):

$$\begin{aligned} LSTR &= STMR + D \\ D &= -\sum_{i=1}^{N} K_i \cdot (DELSM)_i \end{aligned} \qquad \text{(A.4-6)}$$

Note 1 – For modern telephone sets with linear microphones, D is in the order of 1.5 to 4 dB. The value of D is, to some extent, dependent on the handset geometric shape but not on the room noise level. Sets with carbon microphones, however, typically have a sensitivity threshold, making them somewhat less susceptible to room noise. Their D-value is in the order of 6 to 8 dB at 60 dBA room noise. However, some modern designs using linear microphones (notably headsets) also incorporate a sensitivity threshold making them less susceptible to room noise.

Note 2 – Physically, above 800 to 1000 Hz, the earcap shields the listening ear from a direct pick-up of room noise but the electric path provides an indirect contribution. Under conditions of high room noise (60 dBA or higher) and high loss connections, the listener's sidetone rating should be greater than 13 dB. This corresponds approximately to the earcap having an equivalent room noise shielding effect of 5 to 6 dB at the higher frequencies.

A.4.4 *Echo and crosstalk*

A.4.4.1 *General remarks*

Echo and crosstalk sounds are much less loud than normal speech. Therefore, the "loudness growth factor", m in the evaluation algorithm (Equation A.2-1) should be chosen higher than 0.2. Experience has shown the following procedure to be appropriate:

The total loudness rating path under consideration is divided into parts, whose loudness ratings are added. The parts are:

1) send and receive circuits of the telephone set(s),

2) the purely electric circuits.

For the telephone set(s), the normal SLR and RLR values are used. For the electric circuits, the loudness loss is evaluated with $m = 0.5$ or 1, corresponding to voltage or power addition. (Which m-value and which frequency range to use will be given below for each application.)

The electric circuit loudness loss LC is computed according to Equation (A.2-1) with a flat weighting over the (logarithmic) frequency band 300 to 3400 Hz. The logarithmic band is divided into $(N-1)$ equal sections, i.e. by N points.

$$LC(m) = -\frac{10}{m} \log_{10} \sum_{i=1}^{N} K_i \cdot 10^{-0.1mL_i} \qquad \text{(A.4-7)}$$

where

$$K_1 = K_N = \frac{1}{2(N-1)}$$

$$K_i = \frac{1}{N-1};\ i = 2 \cdot \cdot \cdot (N-1) \qquad \text{(A.4-8)}$$

If the summation (or integration) is done on a linear frequency scale Equation (A.4-7) transforms into

$$LC(m) = \frac{1}{m} C - \frac{10}{m} \log_{10} \int_{300}^{3400} 10^{-0.1mL(f)} \frac{1}{f} \mathrm{d}f \qquad \text{(A.4-9)}$$

where

$$C = 10 \log_{10} \left\{ \ln \left(\frac{f_2}{f_1} \right) \right\} \qquad \text{(A.4-10)}$$

Thus, if $f_1 = 300$ Hz, $f_2 = 3400$ Hz, then $C = 3.9$ dB (A.4-11)

and if $f_1 = 500$ Hz, $f_2 = 2000$ Hz, then $C = 1.4$ dB (A.4-12)

A.4.4.2 *Talker echo loudness rating (TELR)*

Following the principles given in § A.4.4.1 we have

$$TELR = SLR(\text{set}) + RLR(\text{set}) + L_e \qquad \text{(A.4-13)}$$

where SLR (set), + RLR (set) refer to the telephone set involved.

The echo loss L_e is computed according to Equation (A.4-7) or (A.4-8) with $m = 1$ and $f_1 = 300$ Hz, $f_2 = 3400$ Hz.

$$L_e = LC(m = 1) \qquad \text{(A.4-14)}$$

Note 1 – For *TELR* = 9 dB, the echo of the speaker's voice would reach his ear with about 0 dB loss averaged over a logarithmic frequency scale.

Note 2 – The value of L_e computed by this method is identical to the value obtained using the method given in Recommendation G.122, § 4.2.

Note 3 – The difference between talker's sidetone and talker echo is that the latter of course is associated with delay. Recent investigations indicate that, at about 2-4 ms delay, the effect of talker echo begins to be clearly distinguishable from even a strong talker's sidetone. To avoid subscriber annoyance from echo, the echo needs more suppression than sidetone signals, all the more so, the longer the delay is. The problem is under study in Question 9/XII.

Note 4 – For circuits terminated by a digital 4-wire telephone set, an echo path is introduced by the acoustic path from earphone to microphone. In this case the echo path loss [L_i or $L(f)$ in Equation (A.4-7) and (A.4-9) respectively] includes the acoustic path as well as the send and receive characteristics of the handset. It is practical to relate a weighted measure of the echo path loss to the 0 dBr 4-wire points, using Equation (A.4-7) or (A.4-9) with $m = 1$. This weighted measure is designated AEL (0).

A.4.4.3 *Listener echo loudness rating (LELR)*

LELR is a weighted average of the listener echo LE over the frequency band 300 to 3400 Hz. The weighting should be done according to Equations (A.4-6) or (A.4-8) with $m = 0.5$.

Note – In North American practice a term WEPL, "weighted echo path loss", is used. When one computes WEPL, the factor $m = 0.5$ but the weighting is flat over a *linear* frequency scale. In general, LELR and WEPL do not differ very much numerically.

A.4.4.4 *Crosstalk receive loudness rating (XRLR)*

The harmful effect of crosstalk is of course directly related to the actual speech level in the disturbing channel. Unfortunately, there is no firm relation between send loudness rating (SLR) and speech level in telephone networks, as investigations have shown. Therefore, it would be misleading to include SLR in a crosstalk loudness rating. Expected speech levels (mean and standard deviation) have to be estimated from other network data. The problem is dealt with in Recommendation P.16.

Following the principles given in § A.4.4.1 we have:

$$XRLR = RLR(\text{set}) + L_x \qquad \text{(A.4-15)}$$

where RLR(set) refers to the telephone set involved.

The crosstalk L_x is computed according to Equation (A.4-9) or (A.4-8) with $m = 1$, $f_1 = 500$ Hz, $f_2 = 2000$ Hz.

$$L_x = LC(m = 1) \qquad \text{(A.4-16)}$$

Note – In practice the crosstalk value at around 1020 Hz has been found to represent L_x fairly well (see Recommendation G.134, § A.3.1).

ANNEX B

(to Recommendation G.111)

Recommended values and limits of the loudness ratings for circuits in international connections

The connection configuration is shown in Figure B-1/G.111 and the LR values in Table B-1/G.111.

The interfaces between the national and international sections are assumed to be at relative level of 0 dBr, as is the case for digital interconnections. The relation between LRs at a 0 dBr point and at a virtual analogue switching point (VASP) is discussed in § 1.1. See also Table D-1/G.111.

Note – The long-term traffic weighted mean values of LRs should be the same for each *main* type of subscriber categories, such as urban, suburban and rural. Considering the mean value only for the *whole* country in the transmission plan might lead to a discrimination against some important customer groups.

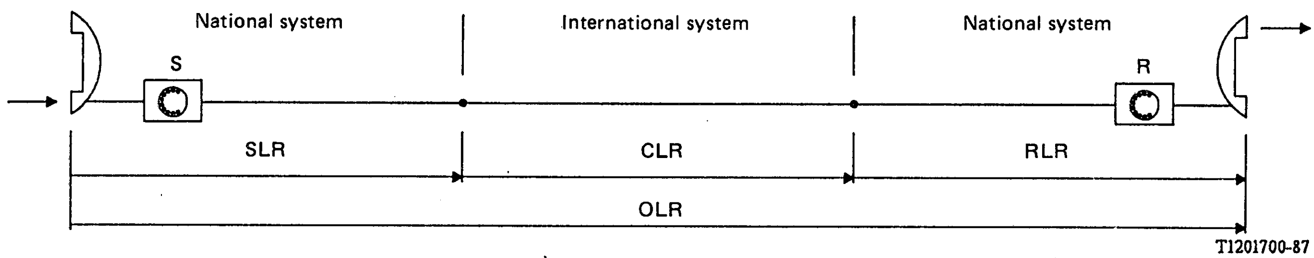


FIGURE B-1/G.111

Designation of LRs in an international connection

TABLE B-1/G.111

LR values as cited in Recommendations G.111 and G.121

	SLR	CLR	RLR	OLR
Traffic-weighted mean values:				
long term	7-9 [b]	0-0.5 [e]	1-3 [b], [f]	8-12 [a], [e], [f]
short term	7-15 [b]	0-0.5 [e]	1-6 [b], [f]	8-21 [a], [c], [f]
Maximum values for an average-sized country	16.5 [c]		13 [c]	
Minimum value	– 1.5 [d]			

[a] Recommendation G.111, § 3.2.

[b] Recommendation G.121, § 1.

[c] Recommendation G.121, § 2.1.

[d] Recommendation G.121, § 3.

[e] When the international chain is digital, CLR = 0. If the international chain consists of one analogue circuit, CLR = 0.5, and then OLR is increased by 0.5 dB. (If the attenuation distorsion with frequency of this circuit is pronounced, the CLR may increase by another 0.2 dB. See § A.4.2).

[f] See also the remarks made in § 3.2.

ANNEX C

(to Recommendation G.111)

Translation of LR values into CRE values

A full discussion can be found in Annex D, on the general relations between reference equivalents (REs), corrected reference equivalents (CREs), and loudness ratings (LRs). Strictly speaking, one should make a distinction between:

a) CREs as derived by computation from subjective REs,

b) R25 equivalents measured subjectively,

c) Objective R25 equivalents (OR25Es) measured objectively.

However, Administrations seem to use the term CRE for all three categories, and this practice has been adopted here.

The relation between CREs and LRs can be written as follows:

$$SCRE = SLR_w + x$$

$$RCRE = RLR_w + y$$

(The index w here indicates a measurement according to Recommendation P.79, wideband, 0.2-4 kHz).

In Recommendation G.111, of the *Red Book*, Fascicule III.1, we find

$$x = 5; \; y = 5$$

However, these values are only general averages. Administrations should determine x and y by *actual objective LR measurements* on those typical sets which have been assigned CRE values in their national networks. Large variations may be found for specific sets, compared to the general averages.

ANNEX D

(to Recommendation G.111)

Justification for the values of LR appearing in Recommendations G.111 and G.121

D.1 *General*

D.1.1 *General principles*

When redrafting Recommendations G.111 and G.121 on the basis of CRE in 1980, the following two principles had been observed:

a) Administrations which used planning methods based on reference equivalents should not have serious difficulties in applying the new Recommendations.

b) The transmission performance provided for subscribers should not deteriorate.

When recommending LR values in Recommendations G.111 and G.121 *Red Book* version, it was not possible to strictly apply this principle because:

– the difference $CRSE - SLR$ depends on the type of handset used;

– in any case, the sending and receiving differences for various types of sets may vary, since different values of RE may be found in various laboratories or with different testing teams.

To satisfy principle b) above, it was agreed to take $SLR = CSRE - 5$ and $RLR = CRRE - 5$ dB which are the means (over a variety of types of sets) of the differences found in the CCITT Laboratory during a certain period. This indicates that transmission performance will be safeguarded as a whole, but certain Administrations may encounter difficulties to meet recommended values of LR.

D.1.2 *Optimum values*

The conversion of "preferred values" formerly expressed as RE is not clear.

On the basis of the information available in 1984 [1] an overall LR of 5 dB was recommended, but it was realized that a larger value might be preferable in the presence of echoes.

D.1.3 *Addition of LRs in the case of analogue subscribers' stations*

Let us define the national system for CREs (see *Red Book* version of this Recommendation, §§ A.3.3 and A.3.4). The overall CRE of a connection is:

$$Y = CSNRE + CRNRE + X + D_0 + A \qquad \text{(D-1)}$$

where $CSNRE = CSRE + b + c$ (sending) and $CRNRE = CRRE + b + c$ (reception),

where

$CSRE$ and $CRRE$ relate to the local systems,

b is the CRE of a trunk junction,

c is the total of the losses (at 800 or 1000 Hz) of long-distance national circuits, exchanges and 2-wire/4-wire terminating unit,

X is the total loss of international circuits,

D_0 and A (ADE) are defined in Annex B, *Red Book* version.

Similarly, the overall LR will be:

$$Z = SNLR + RNLR + X + D'_0 + A' \qquad \text{(D-2)}$$

with

$$SNLR = SLR + b' + c \qquad \text{(D-3)}$$

where

D'_0 is negligible and

A', b' are virtually equal to A, b (cf. Annex B, *Red Book* version).

If it is assumed (§ D.1.1 above) that $SLR = CSRE - 5$, $RLR = CRRE - 5$, and $D'_0 = -4$ (since the Recommendations were originally applied to old-type subscriber's stations), then $Z = Y - 6$ dB is obtained.

In fact, the recommended values were derived from $Z = Y - 5$ dB, which is not a significant difference, but the recommendations for the national system are a little more stringent, because the ADE of national long distance circuits was included in the national system.

D.2 *LRs recommended in 1988*

D.2.1 The maximum values and the minimum for sending have been retained; other values differ from those recommended in 1984, as explained below.

D.2.2 *Optimum value*

Values directly determined in terms of overall LR (Recommendations P.78 or P.79) during conversation tests are available as follows:

British Telecom [1], in the presence of room noise, found a maximum mean opinion score (MOS) for OLR = 3 dB and a minimum difficulty percentage for OLR = 7.2 dB. It was proposed to adopt 5 dB as the optimum value and an almost equally good performance was found in a range from 1 to 10 dB.

NTT [2] found values between 4 and 6 dB according to noise conditions; an optimum OLR = 5.34 dB is used in the OPINE model.

According to the TRANSRAT model [3], maximum MOS is obtained for $L_e = 7.5$ (corresponding to $L'_e = 8.5$ in Supplement No. 3, § 1, of Volume V, where L_e = OLR(EARS). There are reasons to think that L_e is higher than OLR (See Recommendation P.79) by a few dB, so that this should not differ significantly from the above values; this point is being studied under Question 7/X.II.

In any case, such maxima are very flat and there is evidence that higher values would apply in the presence of echoes. It may be provisionally concluded that to obtain the best performance, OLR (See Recommendation P.79) should not exceed about 10 dB, but should not be much smaller.

D.2.3 *Traffic weighted mean values*

An optimum OLR of 10 dB was adopted and it was subdivided between sending and receiving in the same manner as for the LRs of digital subscriber's sets (the latter being referred to a 0 dBr point). This gives the long-term objectives.

The values of A (see § D.1.3) used previously, which took into account both effects of attenuation distortion on loudness and naturalness of speech, were replaced by a fixed allowance of 2 dB (1 dB in each national system, see § A.3) when analogue subscriber's stations are used. This, combined with a small margin in the previous Recommendation version (see § D.1.3 of this Annex), made it possible to increase the traffic-weighted means for sending by about 4 dB and to keep the same overall values.

D.3 *Conclusion*

Table D-1/G.111 recapitulates the values of LR recommended in 1984 and those which are recommended now.

TABLE D-1/G.111

Values (dB) of sending, receiving, circuit and overall loudness rating cited in Recommendations G.111 and G.121

		Recommended in 1984			Recommended in 1988					
		SLR	RLR	OLR	SLR		RLR		CLR	OLR
		VASP	VASP		0 dBr	VASP	0 dBr	VASP		
Optimum value				≈ 5						≈ 10
Traffic-weighted mean values:										
long-term objective	(minimum)	6.5	−2.5	8	7	10.5	1	−3	(Note 1)	8
	(maximum)	8	−1	11	9	12.5	3	−1	(Note 1)	12
short-term objective	(maximum)	14	2.5	20.5	15	18.5	6	2	(Note 1)	21
Maximum values for an average-sized country		20	9		16.5	20	13	9	$n \times 0.5$ (Note 2)	
Minimum for sending		2			−1.5	2				

Note 1 – *CLR* = 0 for a digital international circuit, 0.5 dB for an analogue one. The average number of international circuits is about 1.

Note 2 – n is the number of analogue international circuits.

Note 3 – The VASPs are defined in Recommendation G.101.

References

[1] CCITT Contribution COM XII-97 (British Telecom), Study Period 1981-1984.

[2] OSAKA (S.) and KAKEHI (N.): Objective model for evaluating telephone transmission performance, *Review of the Electric Communication Laboratories*, Vol. 34, No. 4, pp. 437-444, 1986.

[3] HATCH (R. W.) and SULLIVAN (J. L.): Transmission rating models for use in planning of telephone networks, *Conference Record NTC 76*, pp. 23.2-1 to 23.2-5, Dallas, 1976.

[4] CCITT *Handbook on Telephonometry*, ITU, Geneva 1987.

Recommendation G.113

TRANSMISSION IMPAIRMENTS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984 and Melbourne, 1988)

1 Transmission impairment

1.1 The objectives for the attenuation distortion of a maximum-length 4-wire chain are given in Recommendation G.132 and those of the signal-independent noise performance of such maximum-length connections are given in § 2 of this Recommendation. Bearing in mind that less complicated connections (which are more numerous) will have less attenuation distortion and less noise, then the maximum, average and minimum values of loudness rating recommended in Recommendation G.121 will ensure an adequate transmission performance on international connections.

1.2 Should values of attenuation distortion or noise greatly different from those recommended by the CCITT for systems and equipments be contemplated, then guidance concerning possible changes in transmission performance can be found in Recommendation P.11 and Annexes [1], with some indication of possible trade-offs between them.

2 Network performance objective for circuit noise on complete telephone connections

The CCITT recommends that the network performance objective for the mean value, expressed in decibels and taken over a large number of worldwide connections (each including four international circuits), of the distribution of one-minute mean values of signal-independent noise power of the connections, should not exceed −43 dBm0p referred to the input of the first circuit in the chain of international circuits.

3 Transmission impairments due to digital processes

The incorporation of unintegrated digital processes in international telephone connections, particularly during the mixed analogue/digital period, can result in an appreciable accumulation of transmission impairments. It is, therefore, necessary to ensure that this accumulation does not reach a point where it can seriously degrade overall transmission quality.

3.1 *Quantizing distortion*

From the point of view of quantizing distortion, it is recommended that no more than 14 units of quantizing distortion (qdu) should be introduced in an international telephone connection.

For telephone connections which incorporate unintegrated digital processes, it is permissible to simply add the units of quantizing distortion that have been assigned to the individual digital processes to determine the total or overall quantizing distortion. Some sources of quantizing distortion and the units tentatively assigned to them are given in § 3.2.

By definition, an average 8-bit codec pair (A/D and D/A conversions, A-law or μ-law) which complies with Recommendation G.711 introduces 1 quantizing distortion units (1 qdu). An average codec pair produces about 2 dB less quantizing distortion than the limits indicated in Recommendation G.712. This would correspond to a single-to-distortion ratio of 35 dB for the sine-wave test method and approximately 36 dB for the noise test method. (A total of fourteen 8-bit PCM processes each of which just comply with the limits for signal-to-distortion ratio in Recommendation G.712 would be unacceptable). The same principle should be applied when proposing planning values of quantizing distortion units for other digital processes.

In principle, the number of units for other digital processes are determined by comparison with an 8-bit PCM codec pair such that the distortion of the digital process being evaluated is assigned *n* quantizing distortion units if it is equivalent to *n* unintegrated 8-bit PCM process in tandem. Several methods of comparison are possible; these include objective measurements (or equivalent analysis), subjective tests, and data tests in which the effect on the bit error ratio at the output of a voice-band data modem receiver is used as a criterion.

At the present time no objective measurement capability exists which can produce results (e.g. SNR) that correlate closely with results obtained from subjective measurement of the effect of many of the digital processes now being studied on speech performance. Therefore, the number of units of quantization distortion for digital processes should, in general, be determined by subjective measurement methods, such as those found in Recommendation P.81. In some instances the number of units of quantization distortion for a digital process can be determined without subjective measurement by decomposing a digital process into two or more parts and allocating to the parts suitable fractions of the total number of units assigned to the digital process. However, while this method may be considered an objective method for determining the qdu assignments for the parts, it uses as a starting point a subjectively determined value. Furthermore, except for relatively simple digital processes where the decomposition is uncomplicated, this method may not be reliable and should be used with care.

Planning rules should be aplicable to all signals transmitted in the voice-frequency band. Therefore, in general, both speech quality and data performance must be considered. Speech quality should be evaluated by subjective tests and data performance should be evaluated by objective measurements which provide estimates of the expected bit error ratio and signalling performance. At present, however, because of the lack of an objective method for evaluating the effect of digital processes on voice-band data performance, the planning rule in this Recommendation is limited to voice connection planning purposes only. § 4 discusses some of the problems associated with developing a planning rule for connections carrying voice-band data and other non-speech signals. Such a rule would be based on a unit reflecting the contribution digital processes make to the impairment or impairments that affect voice-band data modems and/or signalling systems. Such a unit does not exist yet.

Note – The effect of quantizing distortion on speech transmission is under study in Question 18/XII and the effect of quantizing distortion on data transmission is under study in Question 25/XII.

3.2 *Sources of quantizing distortion*

The units of quantizing distortion (qdu) tentatively assigned to a number of digital processes are given in Table 1/G.113. Background information on these assignments is given in Supplement Nos. 21 and 22, *Red Book*, Fascicles III.1 and III.2, respectively and in the notes associated with Table 1/G.113.

Conceptually the number of qdu assigned to a particular digital process should reflect the effect of only the quantization noise produced by the process on speech. In practice the qdu must be determined from subjective measurements of real or simulated processes, where subjects will be exposed to not only the quantization noise but other impairments produced by the digital process tested.

Therefore, the subjective test results will be biased by these other impairments if the levels of these other impairments differ to a greater or lesser extent from the levels produced by PCM (the reference). Such biases will cause the derived qdu to not be a true measure of the effect of quantization distortion. The qdu assignment will instead reflect the effect of all the impairments on speech quality. Thus, to reduce the chance for such a bias to occur when determining the qdu assignments for digital processes, it is important to design the subjective test so as to:

1) minimize the contributions of impairments other than quantization distortion to the subjective test results, or

2) equalize the levels of these other impairments in the test and reference conditions.

3.3 *Effect of random bit errors*

The effect of random bit errors is under study in Question 25/XII.

3.4 *Attenuation distortion and group-delay distortion*

The provisional recommendation made in § 3.1 specifies that the total quantizing distortion introduced by unintegrated digital processes in international telephone connections should be limited to a maximum of 14 units. It is expected that if this provisional recommendation is complied with, the accumulated attenuation distortion and the accumulated group-delay distortion introduced by unintegrated digital processes in such connections would also be kept within acceptable limits.

Note – The relationships among limitations imposed by quantizing distortion, attenuation distortion and group-delay distortion are under study in Study Group XII.

3.5 *Provisional planning rule*

As a consequence of the relationship indicated in § 3.4 above concerning quantizing distortion, attenuation distortion and group-delay distortion, it is possible to recommend a provisional planning rule governing the incorporation of unintegrated digital processes in international telephone connections. This provisional planning rule is in terms of units of transmission impairment which numerically are the same as the units of quantizing distortion allocated to specific digital processes as indicated in Table 1/G.113. The provisional planning rule is as follows:

The number of units of transmission impairment in an international telephone connection should not exceed: 5 + 4 + 5 = 14 units.

Under the above rule, each of the two national portions of an international telephone connection are permitted to introduce up to a maximum of 5 units of transmission impairment and the international portion up to a maximum of 4 units.

Note – It is recognized that in the mixed analogue/digital period, it might for a time not be practical for some countries to limit their national contributions to a maximum of 5 units of transmission impairment. To accommodate such countries, a temporary relaxation of the provisional planning rule is being permitted. Through this relaxation, the national portion of an international telephone connection would be permitted to introduce up to 7 units of transmission impairment. Theoretically, this could result in international telephone connections with a total of 18 qdu of transmission impairment. Such connections would introduce an additional transmission penalty insofar as voice telephone service is concerned. Administrations which find it indispensable to have a national allowance of more than 5 units (but no more than 7 units) should ensure that not more than a small percentage of traffic on national extensions exceeds 5 units.

3.6 *Limitations of the provisional planning rule*

In § 3.5, it is assumed that for estimating the transmission impairment due to the presence of unintegrated digital processes in international telephone connections, the units of transmission impairment correspond to the units of quantizing distortion and that the simple addition of such units would apply.

For international telephone circuits that include tandem digital processes in an all-digital environment, adding the individual units of quantizing distortion might not accurately reflect the accumulated quantizing distortion (and, consequently, the accumulated units of transmission impairment). This could be the case since the individual amounts of quantizing distortion power produced by the individual digital processes might not be uncorrelated and, therefore, the addition of individual units of quantizing distortion might, under some circumstances, indicate totals that could be different from those actually in effect. This is explained in some detail in Supplement No. 21, *Red Book*, Fascicle III.1.

Although the 5 + 4 + 5 = 14 rule given in § 3.5 might under some conditions provide only approximate results, the rule, nevertheless, is considered to be suitable for most planning purposes particularly in cases involving unintegrated digital processes. Examples of tandem digital processes which are explicitly taken into account in Table 1/G.113 are A-μ-A code conversion, μ-A-μ code conversion, and PCM-ADPCM-PCM conversion.

TABLE 1/G.113

Planning values for quantizing distorsion

(Speech service only; see § 4 for voiceband data considerations)
(see Notes 1, 11 and 12)

Digital process	Quantizing distortion units (qdu)	Notes
Processes involving A/D conversion		
8-bit PCM codec-pair (according to Recommendation G.711 [2], A- or μ-law)	1	2, 3
7-bit PCM codec-pair (A- or μ-law)	3	3, 4, 5
Transmultiplexer pair based on 8-bit PCM, A- or μ-law (according to Recommendation G.792)	1	3
32 kbit/s ADPCM (with adaptative predictor) (combination of an 8-bit PCM codec pair and a PCM-ADPCM-PCM tandem conversion)	3.5	6
Purely digital processes		
Digital loss pad (8-bit PCM, A- or μ-law)	0.7	7
A/μ-law or μ/A-law converter (according to Recommendation G.711 [2])	0.5	10
A/μ/A-law tandem conversion	0.5	
μ/A/μ-law tandem conversion	0.25	
PCM to ADPCM to PCM tandem conversion (according to Recommendation G.721)	2.5	8, 9
8-7-8 bit transcoding (A- or μ-law)	3	9

Note 1 – As a general remark, the number of units of quantizing distortion entered for the different digital processes is that value which has been derived at a mean Gaussian signal level of about –20 dBm0. The cases dealt with in Supplement 21 (at the end of this fascicle) are in accordance with this approach.

Note 2 – By definition.

Note 3 – For general planning purposes, half the value indicated may be assigned to either of the send or receive parts.

Note 4 – This system is not recommended by CCITT but is in use by some Administrations in their national networks.

Note 5 – The impairment indicated for this process is based on subjective tests and was provided by Study Group XII.

Note 6 – For this 32 kbit/s ADPCM process a value of 3.5 units was derived by Study Group XII from subjective measurements on a combination of an 8-bit PCM codec par and a PCM/ADPCM/PCM conversion according to Recommendation G.721.

Note 7 – The impairment indicated is about the same for all digital pad values in the range 1-8 dB. One exception is the 6 dB A-law pad which introduces negligible impairment for signals down to about –30 dBm0 and thus attracts 0 units for quantizing distorsion.

Note 8 – The value of 2.5 units was derived by subtracting the value for an 8-bit PCM codec pair from the 3.5 units determined subjectively for the combination of an 8-bit PCM code pair and a PCM/ADPCM/PCM conversion. Multiple synchronous digital conversions, such as PCM/ADPCM, PCM/ADPCM/PCM, are assigned a value of 2.5 units.

Note 9 – This process might be used in a digital speech interpolation system.

Note 10 – The qdu contribution made by coding law converters (e.g., μ-law to A-law) are assigned to the international part.

Note 11 – The qdu assignments to these digital processes reflect, to the extent possible, only the effect of quantization distortion on speech performance. Other impairments, such as circuit noise, echo and attenuation distorsion also affect speech performance. The effect of these other impairments must therefore be taken into account in the planning process.

Note 12 – The qdu impairments in this table are derived under the assumption of negligible bit error.

4 Effect of transmission impairments on voiceband data performance

The effect of transmission impairments on voiceband data performance is under study in Question 25/XII. Some information provided by one Administration is available in Annex 4 to the Question.

Just as speech quality is affected by the transmission impairments found on telephone connections, so too is voiceband data quality. Many different impairments are present on a connection; some are steady-state impairments (e.g. loss, noise, quantization distortion, phase jitter, harmonic and intermodulation distortions, envelope delay distortion, echo, and attenuation distortion) while others are transient (e.g. impulse noise, phase or gain hits, and dropouts) and may tend to occur infrequently. Both steady-state and transient impairments can affect speech and voiceband data. However, the transient impairments almost always have a bigger impact on data than on speech. This is also true of some of the steady-state impairments, e.g. phase jitter and envelope delay distortion. Because of this, planning rules for circuits carrying speech usually concentrate on controlling the steady-state impairments, and less attention is paid to the transient impairments. If new planning rules are to be created with the intent of controlling the buildup of the impairments that are important to voiceband data, then these new rules will have to treat the transient as well as the steady-state impairments.

The extent to which certain impairments affect voice-band data depend upon the modem speed, modulation used and other characteristics such as whether the modem contains an equalizer to correct for envelope delay distortion. Low speed modems, operating at 1200 bit/s or less can usually tolerate a poorer SNR than higher speed modems. They also tend to be less sensitive to envelope delay distortion than the higher speed modems. Modems operating at 4800 bit/s and higher will usually contain an envelope delay distortion equalizer to minimize the effect of envelope delay distortion on the performance. Transients affect all modems, to a greater or lesser extent depending on many factors.

Two other factors influencing how impairments impact on voice-band data performance are:

a) whether error detection and/or correction techniques are employed, and

b) how the information to be sent is encoded.

If error correction is not used then error causing impairments will cause errors in the output data. However, if error correction is used then the impact of error causing impairments will only reduce the data throughput rate. Depending on how customer information is coded, errors can have more or less serious effects. For example, the loss of a letter in a word, because of a bit error in the 8 bits representing the letters of the alphabet, is probably less important than an error in the 8 bits used to convey information about the size, shape or location of a graphical symbol in an image.

Bit compression techniques such as ADPCM (according to Recommendation G.721) have a very significant effect on high speed ($\geq$ 4800 bit/s) modem performance (see Annex C).

From the point of view of developing a simple planning rule which can be used to assess the effects of digital processes on voice-band data performance, several points are important:

1) Impairments (especially transients) other than those customarily measured for speech performance are important for measuring voice-band data performance.

2) A simple meaure of the steady-state impairments (e.g. signal-to-total noise ratio) may not prove to be a satisfactory basis for a voice-band data planning rule. A planning rule may have to take the transient impairments into account.

3) Modem type and speed must be taken into account. Thus, unlike the planning rules for speech, rules for voice-band data may turn out to be modem-specific.

4) The type of data service may influence the extent to which certain kinds of data errors and, thus, certain impairments are important. Therefore the planning rules may be service-specific.

5) Only an objective measurement method taking these first four points into account is likely to provide a successful basis for deriving useful planning rules.

6) Such a measurement method does not exist at present.

Therefore, until much more progress has been made in determining what impairments affect voice-band data performance, how to measure these impairments, what levels of these impairments are important, and how the differences in modem type, speed and other characteristics can be accounted for, this Recommendation must be limited in its application to speech services only.

ANNEX A

(to Recommendation G.113)

Information for planning purposes concerning attenuation distortion and group-delay distortion introduced by circuits and exchanges in the switched telephone network

A.1 The information given in Tables A-1/G.113 to A-6/G.113 is derived from measurements[1] on modern equipment. The performance of actual connections in the switched telephone network can be expected to be worse than would be calculated from the tabulated data because of:

- mismatch and reflexion;
- unloaded subscribers' lines;
- loaded trunk-junctions with a low cutoff frequency;
- older equipment.

TABLE A-1/G.113

Two-wire local and primary exchanges

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	1.69	1.20	0.56	0.07
300	0.63	0.81	0.28	0.05
400	0.30	0.43	0.23	0.05
600	0	0.28	0.11	0.03
800	0	0	0.05	0.02
1000	−0.05	0.11	0.03	0.01
2000	−0.04	0.35	0	0
2400	−0.29	0.45	0	0
2800	−0.45	0.50	0	0
3000	−0.24	0.65	0	0
3400	−0.29	0.63	0	0

Note – The group-delay distortion may be taken to be with respect to about 2000 Hz.

[1] Supplied by AT&T, Telecom Australia, Italy, British Telecom, NTT and Switzerland.

TABLE A-2/G.113

Four-wire exchanges

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value (dB)	Standard deviation (dB)	Mean value (ms)	Standard deviation (ms)
200	0.32	0.14	0.40	0.02
300	0.16	0.28	0.14	0.02
400	0.13	0.21	0.14	0.03
600	0.02	0	0.07	0.02
800	0	0	0.03	0.01
1000	0	0	0.02	0.01
2000	0.01	0.14	0	0
2400	0.06	0.21	0	0
2800	0.02	0.02	0	0
3000	0.10	0.07	0	0
3400	0.20	0.50	0	0

Note – The group-delay distortion may be taken to be with respect to about 2000 Hz.

TABLE A-3/G.113

Trunk junctions

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	4.29	1.95	3.05	0.36
300	0.86	0.49	1.42	0.18
400	0.36	0.31	0.78	0.09
600	0.09	0.17	0.34	0.06
800	0	0.03	0.16	0.02
1000	−0.03	0.04	0.08	0.02
2000	0.14	0.20	0.02	0.01
2400	0.33	0.29	0.06	0.03
2800	0.58	0.35	0.18	0.06
3000	0.88	0.55	0.31	0.11
3400	2.21	1.06	0.92	0.26

Note 1 – The group-delay distortion may be taken to be with respect to about 1500 Hz.

Note 2 – The sample of trunk junctions included those on metallic lines, FDM and PCM systems.

Note 3 – PCM circuits may exhibit a somewhat lower attenuation distortion at 2000 Hz than that indicated above.

Note 4 – The values for trunk junctions are inclusive of 2-wire/4-wire terminations.

TABLE A-4/G.113

Circuits provided on a direct 12-channel group

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	1.56	0.92	5.42	0.22
300	0.39	0.43	2.97	0.35
400	0.11	0.30	1.45	0.22
600	0.05	0.18	0.76	0.10
800	0	0	0.44	0.05
1000	−0.01	0.11	0.26	0.02
2000	−0.03	0.19	0.01	0.01
2400	0.04	0.21	0.06	0.02
2800	0.13	0.33	0.21	0.04
3000	0.16	0.43	0.45	0.04
3400	1.03	0.56	1.97	0.20

Note 1 – The group-delay distortion may be taken to be with respect to about 1800 Hz.

Note 2 – The data relates to 4 kHz channel translating equipment, the principal source of distortion in telephone circuits provided on direct 12-channel groups, i.e., circuits with only one circuit-section.

TABLE A-5/G.113

Circuits provided on a direct 16-channel group

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	2.80	1.63	9.74	0.40
300	0.04	0.19	4.39	0.27
400	−0.07	0.20	2.49	0.09
600	0.02	0.09	1.02	0.56
800	0	0	0.47	0.35
1000	0.09	0.08	0.19	0.28
2000	0.06	0.12	0.03	0.14
2400	0.03	0.14	0.36	0.31
2800	0.03	0.16	1.59	1.06
3000	−0.01	0.28	4.29	0.38

Note 1 – The group-delay distortion may be taken to be with respect to about 1200 Hz.

Note 2 – The data relates to 3-kHz FDM channel translating equipment, the principal source of distortion in telephone circuits provided on direct 16-channel groups, i.e., circuits with only one circuit-section.

TABLE A-6/G.113

Circuits comprising three circuit-sections (4 kHz + 3 kHz + 4 kHz)

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	5.92	2.09	20.58	0.51
300	0.82	0.64	10.33	0.56
400	0.15	0.47	5.39	0.32
600	0.12	0.27	2.54	0.58
800	0	0	1.35	0.36
1000	0.07	0.17	0.71	0.28
2000	0	0.29	0.05	0.14
2400	0.11	0.33	0.48	0.31
2800	0.29	0.49	2.01	1.06
3000	0.31	0.67	5.19	0.38

Note 1 – This table has been derived from Tables A-4/G.113 and A-5/G.113, and relates to international circuits in which the middle section is routed on 3-kHz spaced channel equipment, e.g., a submarine circuit-section.

Note 2 – The group-delay distortion may be taken to be with respect to about 1400 Hz.

A.2 The reference frequency for attenuation distortion is 800 Hz. The reference frequency for group-delay distortion (i.e. the frequency at which the group delay is a minimum) has been estimated in each case.

A.3 In the results for circuits no allowance has been made for line signalling terminations although in some cases these distortions are included in the data for exchanges.

References

[1] CCITT Recommendation *Effect of transmission impairments*, Vol. V, Rec. P.11 and Annexes.

[2] CCITT Recommendation *Pulse code modulation (PCM) of voice frequencies*, Vol. III, Rec. G.711.

ANNEX B

(to Recommendation G.113)

Effect of transmission impairments on voiceband data

(from AT&T)

B.1 *Introduction*

The present transmission plan for international connections provides guidance for the control of transmission performance, primarily to permit satisfactory transmission of speech signals. The significant impairments and their effect on speech signals are described in Recommendation P.11. These impairments include loudness loss, circuit noise, sidetone loudness loss, room noise, attenuation distortion, talker echo, listener echo, quantizing distortion and phase jitter. Other Recommendations involving data performance on leased circuits include H.12, M.1020 and M.1025.

The use of international connections for the transmission of non-speech signals such as voiceband data creates the need for increasing the scope of the transmission plan to include guidance on the control of additional impairments. The significant impairments for voiceband data include impulse noise, envelope delay distortion, phase jitter, non-linear distortion, tone-to-noise ratio, frequency shift, gain transients and phase transients. The following sections provide information on these impairments based on AT&T's experience. All the parameter values quoted are illustrative minimum end-to-end performance objectives of the *pre-divested* AT&T public switched network. Typical values obtained on the network are much better than the minimum objectives. These minimum values are considered to be consistent with satisfactory modem performance at speeds up to 4.8 kbit/s. More stringent minimum objectives are considered necessary for satisfactory performance at higher speeds such as 9.6 kbit/s. The parameter values shown are for illustration only and do not represent a proposed Recommendation.

B.2 *Impulse noise*

Impulse noise is defined as any excursion of the noise waveform on a channel which exceeds a specified level threshold. Impulse noise is evaluated on channels by couting the number of excursions during a predetermined time interval. In order to minimize contributions due to thermal noise, the minimum threshold is normally set 12 to 18 dB above the r.m.s. value of the noise. The impulse noise level is designated to be that threshold at which the average counting rate is equal to one per minute.

The measuring instruments used to count noise impulses may employ either electromechanical or electronic counters. In some sets, the maximum counting rate is controlled to be seven per second.

The contribution of impulse noise to error rate becomes significant when the noise peaks reach a level 3 to 12 dB below the r.m.s. data signal level depending upon: the type of modulation used by the data modems, the speed of transmission in bits per second, and the magnitudes of other transmission impairments on the channel. The minimum impulse noise objective is that no more than 15 counts in 15 minutes are to be tallied at a level above threshold which is 6 dB below the received data level. Control is exercise through engineering rules and limits on measured impulse noise levels.

Since most impulse noise originates as transients from the operation of relays and other switching equipment, engineering rules and mitigative measures are aimed at shielding low-level carrier signals from the radiation associated with these transients.

B.3 *Envelope delay (group delay)*

Envelope delay is defined as the derivative with respect to frequency of the phase characteristic of the channel. Measuring this derivative is impractical, so it is approximated by a difference measurement. There are numerous envelope delay measuring sets in use employing various frequency widths for this difference measurement. The AT&T standard is 166-2/3 Hz. In test results, these differences show up as varying resolution of ripples in the envelope delay characteristic. Narrow frequency widths yield higher resolution but reduced accuracy.

The frequency of minimum envelope delay in telecommunication channels is usually in the vicinity of 1800 Hz. Therefore, envelope delay measurements are usually normalized to zero at 1800 Hz. Departure from zero at other frequencies is referred to as envelope delay distortion. Envelope delay distortion gives rise to intersymbol interference in data transmission which causes errors and increased sensitivity to background noise.

In the network, envelope delay is controlled primarily in the design of channel bank filters and other apparatus. Typical minimum objectives for envelope delay distortion are 800 μsec maximum in the band from 1004 to 2404 Hz and 2600 μsec maximum in the band 604 to 2804 Hz.

B.4 *Phase jitter*

Phase jitter is defined as unwanted angular modulation of a transmitted signal. Its most commonly observed property is that it perturbs the zero crossings of a signal. Since noise also perturbs the zero crossings of a signal, it usually causes readings on a phase jitter measuring set even though no incidental modulation may be present.

Phase jitter impairs data transmission by reducing data receiver margin to other impairments. Phase jitter is controlled by the design of transmission equipment. Although specific sources of phase jitter, such as primary carrier frequency supplies, have been located in the field, the corrective techniques have usually required design changes in specific equipment. The end-to-end minimum objective for phase jitter is 10 degress peak-to-peak for the frequency band of 20 to 300 Hz and 15 degrees peak-to-peak for the band of 4 to 300 Hz.

B.5 *Non-linear distortion*

Non-linear elements in transmission equipment give rise to harmonic and intermodulation distortion which are more generally referred to as non-linear distortion. Non-linear distortion measurements are made usually in terms of intermodulation distortion measurements.

Non-linear distortion can be broadly defined as the generation of signal components from the transmitted signal that add to the transmitted signal usually in an undesired manner. The non-linear distortion of concern here is that found within an individual voice channel. It should not be confused with the intermodulation noise caused by non-linearities in the multiplex equipment and line amplifiers of a frequency division multiplex system. Although these non-linearities can contribute to the non-linear distortion at voice frequencies, their contribution is usually negligible.

Non-linear distortion is commonly measured and identified by the effect it has on certain signals. For example, if the signal is a tone having frequency A, the non-linear distortion appears as harmonics of the input, i.e. it appears as tones at $2A$, $3A$, etc. Since most of the distortion product energy usually occurs as the second and third harmonics, distortion is often quantified by measuring the power of each of these harmonics and is called second and third harmonic distortion. If the amount of non-linear distortion is measured by the power sum of all the harmonics, the result is called total harmonic distortion. These distortion powers are not meaningful unless the power of the wanted signal (the fundamental) is known, so measurements are usually referred to the power of the fundamental and termed second, third, or total harmonic distortion.

Historically, two different methods of measuring non-linear distortion on voiceband channels have been used: the signal-tone method and the 4-tone method. However, the single-tone method is no longer used.

For the 4-tone method, four equal level tones are transmitted as two sets of tones at a composite signal power of data level (−13 dBm0). One set consists of tones at 856 and 863 Hz (a 7-Hz spacing). A second set uses frequencies of 1374 and 1385 Hz (an 11-Hz spacing). The frequency spacing within each set of tones is not critical but should be different for each set. Let these four tones be called A_1, A_2, B_1, and B_2. The second order products $(A + B)$ fall at $A_1 + B_1$, $A_1 + B_2$, $A_2 + B_1$ and $A_2 + B_2$. If the spacing between A_1 and A_2 is the same as that between B_1 and B_2 then $A_1 + B_2 = A_2 + B_1$ and these two components will add on a voltage basis and give an erroneous reading.

The third order products ($2B - A$) fall at $2B_1 - A_1$, $2B_1 - A_2$, $2B_2 - A_1$, $2B_2 - A_2$, $B_1 + B_2 - A_1$ and $B_1 + B_2 - A_2$. The receiver uses 50-Hz wide filters to select the $A + B$, $B - A$, and $2B - A$ products. R_2 is the ratio of the power of the received composite fundamentals to the power average of the $A + B$ and $B - A$ products. R_3 is the ratio of received composite fundamentals to the $2B - A$ products.

An advantage of the 4-tone method, the method currently used in AT&T, is that the 4-tone test signal has an amplitude density function quite similar to that of a data signal. However, because of the relatively wide (50 Hz) passband of the receiver filters, the measurements with the 4-tone method are more affected by circuit noise.

The intermodulation products arising from non-linear distortion add to the wanted signal and interfere with it much as noise does. The intermodulation products are more damaging than noise, however, and the ratio of fundamental to second- or third-order products should be in the range of 25 to 38 dB, depending upon the type of data transmission, for satisfactory operation.

Non-linear distortion is controlled primarily in the design of equipment. However, such things as aging vacuum tubes in older equipment and poor alignment of PCM channel banks can cause this distortion to increase over its design limits. The overall customer-to-customer minimum objective for non-linear distortion using the 4-tone method of measurement is 27 dB minimum for R_2 and 32 dB minimum for R_3.

B.6 *Tone-to-noise ratio*

For voice transmission, the noise that is heard during the quiet intervals of speech is most important and this is what the standard message circuit noise measurement evaluates. For data transmission, the noise on the channel during active transmission and corresponding signal-to-noise ratio is important. In systems using compandors or quantizers, the noise increases during active transmission. In order to measure this noise, a -16, -13, or -10 dBm0 tone is transmitted from the far end of the channel under test and then filtered out ahead of the noise measuring set. The filter used to remove the tone is a narrow notch filter centered at the frequency of the tone. This type of measurement is also referred to as noise-with-tone. Test equipment is now available which uses 1004 Hz as the tone for this measurement.

Noise, of course, can cause errors in data transmission and a tone signal-to-noise ratio objective of at least 24 dB should be maintained for satisfactory performance. Noise is controlled in the design of transmission equipment, in the engineering of transmission systems (by such factors as repeater spacing), and in the maintenance of these systems.

B.7 *Frequency shift*

When a tone experiences a change in frequency as it is transmitted over a channel, the channel is said to have frequency shift or offset. Frequency shift can be measured by using frequency counters at both ends of a channel. When the input frequency differs from the output frequency, the difference is the frequency shift on the channel.

In modem telecommunication equipment, the frequency shift, if any at all, is usually on the order of 1 Hz or less. Some older carrier systems may have substantial amounts of offset, e.g. 15 to 20 Hz.

Frequency shift is important in systems which use narrowband receiving filters such as telegraph multiplexers and remote meter reading equipment. When systems using these types of transmission experience frequency shift, the received signals fall outside the bandwidth of the filters. Frequency shift can occur on facilities which use single sideband suppressed carrier transmission. Within AT&T, frequency shift is controlled by means of the frequency synchronization network. The minimum objective for frequency shift is ± 5 Hz.

B.8 *Gain and phase transients*

Gain and phase changes that occur very rapidly may be encountered on telecommunication channels. Some of the more common causes of these phenomena are automatic switching to standby facilities or carrier supplies, patching out working facilities to perform routine maintenance, fades or path changes in microwave facilities, and noise transients coupled into carrier frequency sources. The channel gain and phase (or frequency) shift may return to its original value in a short time or remain at the new values indefinitely.

Gain changes are typically detected by changes in an automatic gain control circuit and phase changes by means of a phase locked loop. In order to provide protection against the test set detectors falsely operating on peaks of uncorrelated noise (impulse noise), a guard interval of 4 ms is designed into the gain or phase peak indicating instrument. Unfortunately, such a guard interval will also effectively make out true phase hits shorter than 4 ms that are not also accompanied by a peak amplitude excursion. The risk is considered justified at this time when one compares the known relative frequencies of occurrence of phase jumps to those of impulse noise.

Instrument used to measure gain and phase hits, as the rapid gain and phase changes are usually called, do so by monitoring the magnitude and phase of a sinusoidal tone. Hits are recorded and accumulated on counters with adjustable threshold levels. Gain hit counters typically accumulate events exceeding thresholds of 2, 3, 4 and 6 dB although they do not distinguish an increase from a decrease of magnitude. Similarly, phase hit counters accumulate changes at thresholds from 5 to 45 degrees in 5-degree steps. They respond to any hits equal to or in excess of the selected threshold. A switch which removes the impulse noise blanking feature under the user's discretion may be desirable when impulse phase hit activity is suspected. The wide variety in hit waveforms, the effect of noise on measurements, and the allowable tolerances in thresholds and measurement circuitry, will generally contribute to different hit counts even on instruments of identical design. This variability will lead to some confusing among those testing with hit counters of different manufacturers. An alternative specification of the entire hit counting circuitry is under further investigation by the Institute of Electrical and Electronic Engineers.

Gain hits begin to cause errors in high-speed data transmission when their magnitude is on the order of 2 to 3 dB. Phase hits begin to cause errors when their magnitude is about 20 to 25 degrees. The end-to-end minimum objective for gain hits is to have no more than eight gain hits exceeding 3 dB in 15 minutes; the minimum objective for phase hits is to have no more than eight phase hits in 15 minutes at a threshold of 20 degrees. A dropout is defined as a decrease in level greater than or equal to 12 dB lasting at least 4 ms. The minimum objective for dropouts is to have no more than two dropouts per hour.

ANNEX C

(to Recommendation G.113)

Adaptive differential pulse code modulation (ADPCM) performance impact on voiceband data

(From AT&T)

(According to G.721)

Abstract

This Annex is mainly based on an AT&T Bell Laboratories paper given at the "IEEE Global Telecommunications Conference" 2-5 December, 1985. It is provided to support Recommendation G.113 as applied to voiceband data performance. The results indicate that, assigning a data qdu value to equipment using 32 kbit/s ADPCM (Recommendation G.721) would be a difficult task since the performance is strongly dependent on the modem speed and type.

The Annex reports on the results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing asynchronously tandemed 32 kbit/s ADPCM (Recommendation G.721) systems interspersed with simulated analogue impairments. A representative sample of 4.8 kbit/s transmision, and two 9.6 kbit/s devices were tested: an experimental design of the CCITT V.32 standard operating at 9.6 kbit/s for a full duplex modem, and another currently available 9.6 kbit/s product (similar to a V.29 modem). The results of the testing indicate that 4.8 kbit/s voiceband data transmission will perform adequately through asynchronous tandemed ADPCM systems, but that 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same conditions.

C.1 *Introduction*

It is possible to use adaptive differential pulse code modulation (ADPCM) at bit rates lower than 64 kbit/s per channel with, in many cases, less than proportional decrease in analogue transmission performance. Therefore, the use of a 32 kbit/s ADPCM algorithm on voice grade channels would essentially double the channel capacity of the associated facilities.

With the potential economic benefit due to increased capacity also comes the expectation of ensuing degradation of individual channel performance. Our results show that high speed voiceband data (e.g. 4.8 kbit/s or greater) would incur significant performance penalties with this new technology in place.

In this Annex we report on the results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing concatenated CCITT Standard 32 kbit/s ADPCM (Recommendation G.721) systems [1] interspersed with simulated analogue impairments. The channel configurations are designed to be representative of actual topologies possible on the public switched network with ADPCM systems in place. Asynchronously tandemed [2)] ADPCM hardware contained in these test channels range in number from zero to seven while the interspersed analogue impairments are obtained by allocating parameters from impairment distributions measured in the end office connections study (EOCS) [2], loop studying 1970 [3], and 1980 Loop Surveys. We also tested performance using connections with asynchronously tandemed 64 kbit/s PCM systems, implemented in D4 channel banks, to compare with ADPCM configurations that showed particularly poor performance, so that it could be determined whether the ADPCM algorithm or simply the PCM coding was at root.

Modems used for the testing were of the high speed type. We tested a representative sample of 4.8 kbit/s transmission (V.29 type), and two 9.6 kbit/s modems: an experimental design of the V.32 modem standard for a full duplex modem, and another currently available device (V.29 type). All of these devices are 2-2 wire modems which are, or will be, marketed for use on the public switched network.

The results of our testing indicate that 4.8 kbit/s voiceband data transmission will perform adequately through multiple asynchronous tandeming of ADPCM systems, but that, 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same configurations.

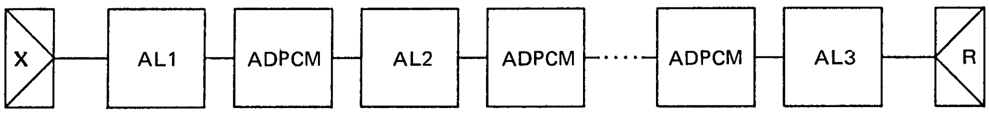
C.2 *Test condition architecture*

It is known that ADPCM algorithm precision is to a great extent dependent on the nature of the signal which is to be encoded and transmitted. Signals with little or no stochastic components, such as pure tones, traverse these systems very well, with little or no distortion. On the other hand, high speed voiceband data signals which inherently have a large stochastic component and substantial bandwidth are significantly affected by ADPCM coding. Due to this, our test condition architecture examines these high speed modem types. We have furthermore tried to efficiently limit the quantity of testing required by using a universal architecture template for all our studies.

C.2.1 *4.8 kbit/s half-duplex*

Figure C-1/G.113 shows the test configuration architecture for 4.8 kbit/s half-duplex testing. The configuration is shown terminated on both ends with modems. The sequence of additional apparatus on the chart begins from the left with simulated analog impairments (AL1) representative of analog loop and access trunk (AT). Then the long haul segment consists of an ADPCM system, one 500 mile equivalent L-carrier analog link (AL2) followed by from 1 to 6 ADPCM's respectively. This structure is representative of an interexchange portion consisting of multiple links and models the segment as if all analog impairments occur early in the segment. Although this placement of the analog impairments is somewhat conservative, it is conterbalanced by the fact that the impairments are those of a single L-carrier link and is a good approximation of reality given the constraint of using a single impairment simulator for the long haul part. Finally egress to the receiver proceeds through another analog impairment simulator (AL3) representative of analog trunk and loop. Interpersing analog impairments with ADPCMs in this manner for the connection is more representative of actual network topologies and applications than simply lumping all analog impairments in one place.

[2)] Asynchronous tandeming takes place when a previously ADPCM coded signal is decoded to its analogue version and then recoded in a subsequent ADPCM system.



Modem	Access	LH			Egress	Modem
	Loop&AT	ADPCM	1 LMX	n(ADPCM)	AT&Loop	# tests
4.8 kbit/s	85	None	85	None	85	1
	None	ADPCM	None	1-6 ADPCM	None	6
	None	PCM	None	1-6 PCM	None	6 AN
	μ, 85	ADPCM	μ, 85	1-6 ADPCM	μ, 85	48
	μ, 85	PCM	μ, 85	1-6 PCM	μ, 85	48 AN

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FIGURE C-1/G.113

Test condition architecture for 4.8 kbit/s modem

It is clearly necessary to determine, for this configuration, the type and actual values of the analogue impairments to be dialed into simulators AL1, AL2 and AL3. Using a network performance modeling tool the results of the end office connections study (EOCS), and the assumption that high speed data customers connect to the network via data jacks, we derived the end-to-end mean (M) and 85th percent conditions of the major subset of impairments for switched network channels. Note that although we refer to the channel with each impairment at the 85 percent level as the 85th percentile channel, in fact it is somewhat worse because all impairments at 85% in one channel simultaneously would actually appear less than 15% of the time. Nevertheless, we then allocated these end-to-end values to the analogue impairment simulators. The results of this allocation, the impairment types, and the end-to-end values are shown in Table C-1/G.113. The values designated are allocated from the end-to-end mean (M), while the values designated "85" are allocated from the 85% end-to-end impairment values. The discussion of Figure C-1/G.113 can now be completed by describing the various values of analogue impairments as well as type and number of digital equipment present. The first configuration shows no ADPCMs but contains the allocated impairments from the 85th percent channel. Next, for additional reference, we tested six channels containing from 2 to 7 ADPCMs only, with no analogue impairments. Another six channels were to be tested as necessary with only PCM devices asynchronously tandemed, if and only if the previous corresponding ADPCM tests showed poor performance. Finally, the important tests with both analog impairments allocated to the simulators from the mean (μ) and 85th percent channel with from 2 to 7 ADPCMs (or PCMs as necessary) were performed.

C.2.2 *9.6 kbit/s full and half-duplex*

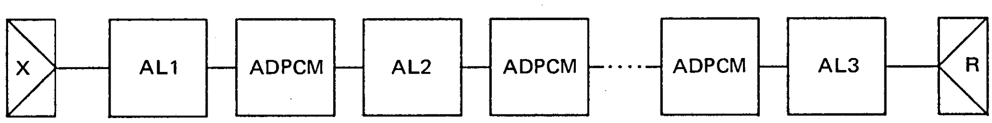
Here the test configuration architecture template is shown with a chart in Figure C-2/G.113. An experimental implementation of the V.32 modem standard for a 9.6 kbit/s full-duplex modem was tested under identical values of analogue impairments as those used for the 4.8 kbit/s modem. Although the channel segments have the same representation in the template, we only tested from 1 to 3 ADPCMs in the long haul segment. The simulated full-duplex operation was tested with the opposite channel excited with data, a signal-to-listener echo ratio of 12 dB, and a listener echo delay of 25 ms, in line with tests previously reported to Study Group XVIII [4]. For these tests Table C-1/G.113 again has the relevant values for the analogue impairment simulators.

Also shown are three tests of another 9.6 kbit/s half-duplex modem with ADPCMs only. This modem is specifically designed for use on the public switched network and represents expected performance of the most currently available 9.6 kbit/s technology.

TABLE C-1/G.113

EOCS derived test conditions

	AL1	AL2	AL3	E-E
Impairment	μ/85	μ/85	μ/85	M/85
Loss (dB)	11.0/11.4	1.1/1.7	11.0/11.4	23.0/24.5
C-notch noise (dBmC)	32.0/35.6	37.5/38.5	24.0/27.6	29.4/31.0
Slope (dB)	1.5/3.0	0.0/0.2	1.5/3.0	2.9/6.1
Env. delay distortion (μs)	226/388	632/755	226/388	1084/1535
2nd intermod. (dB)	66.0/50.2	58.4/53.8	66.0/50.2	52.7/46.3
3rd intermod. (dB)	74.0/53.0	56.9/50.3	74.0/53.0	51.7/44.3
Phase jitter (p-p)	0.5/0.7	1.9/3.7	0.5/0.7	3.5/5.1
Level (dBm)				−27.0/28.5
S/N (dB)				31.6/28.5



Modem	Access	LH			Egress	Modem
	Loop&AT	ADPCM	1 LMX	n(ADPCM)	AT&Loop	# tests
V.32	85	None	85	None	85	1
9.6 kbit/s	None	ADPCM	None	1-3 ADPCM	None	3
V.32	None	PCM	None	1-3 PCM	None	3 AN
	μ, 85	ADPCM	μ, 85	1-3 ADPCM	μ, 85	24
	μ, 85	PCM	μ, 85	1-3 PCM	μ, 85	24 AN

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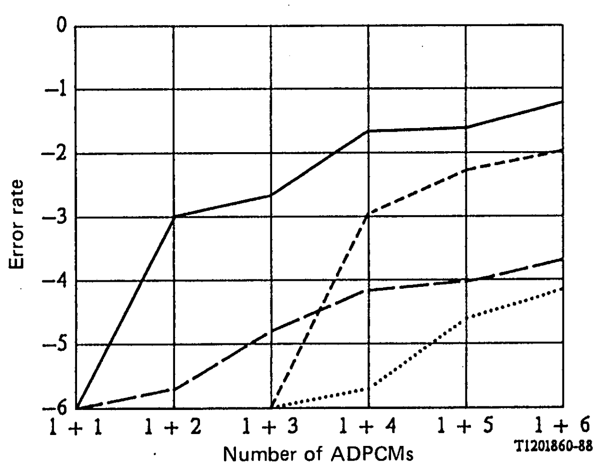
FIGURE C-2/G.113

Test condition architecture for 9.6 kbit/s modems

C.2.3 *4.8 kbit/s ADPCM performance*

For 4.8 kbit/s transmission, the salient results are shown in Figure C-3/G.113. We have plotted four curves on the axes: two 1000-bit block error rates (BLER) and two bit error rates (BER), one each for the mean and 85% EOCS channels. The abscissa counts the number of asynchronously tandemed ADPCMs in the connection. Due to the architecture of the tests these are ennumerated as 1 + *n*. The "1" represents the ADPCM between AL1 and AL2 while *n* is the number of ADPCM systems between AL2 and AL3.

We see clearly from the graphs that all the error performance measures degrade as the number of asynchronously tandemed ADPCMs increases, and that performance on the 85% channel, containing worse values of analog impairments, is inferior to the mean channel results. We assume an acceptance limit for modem accuracy behaviour of a BER < 10^{-5} on 85% of channels and a BLER < 10^{-2} on 85% channels. Hence, if we focus on the 85% channel from EOCS, we see that 4.8 kbit/s performance will be at acceptable limits if the number of ADPCMs is between 4 and 5 for BLER and between 3 and 4 for BER. More recent results imply that for some modems the BER criteria is marginal with 3 in tandem and only 2 would be acceptable. We know of course that the BER criterion is stricter than the BLER limit because bit errors represent a greater burst phenomenon which is to a large extent ameliorated by the use of block transmission implemented with an error detection/correction protocol. Nevertheless, we tested and present both results because customer data communication applications will dictate which measure is more relevant.



------ Log (BLER) mean channel

.... Log (BER) mean channel

———— Log (BLER) 85% channel

— — — Log (BER) 85% channel

BLER Block error rate

BER Bit error rate

FIGURE C-3/G.113

ADPCM performance (mean and 85% channels) with a 4.8 kbit/s modem

C.2.4 *V.32 modem-ADPCM performance*

The outcomes of tests on the experimental testbed representing a 9.6 kbit/s device conform to Recommendation V.32 is shown in Figure C-4/G.113. Note that we have àgain plotted four performance curves. As before, performance of the 85th percent channel is inferior to that of the mean channel. If we now focus on the 85th percent channel BLER, we see that the acceptable performance limit occurs between 2 and 3 asynchronously tandemed ADPCMs, while for BER the number is somewhere between 0 and 1. Which performance measure is appropriate depends on customer application. We are here observing that a larger stochastic component of the data signal implies poorer error performance of the modem. In this case the use of 9.6 kbit/s shows a definite degradation in performance over the same topology with 4.8 kbit/s devices.

It is also interesting to see if changing the position of segments with poorer impairment values effects modem performance. Figure C-5/G.113 shows a graph of three BLER curves for V.32 modems where we have taken the allocated 85th percent segment first on access, then on the long-haul part, and finally on the egress of the test channel, the other segments being at the allocated mean values of impairments. First, note that these curves fall between the full 85th percent channel and the mean channel in performance. Next, note that there does appear to be a mild dependence on the location of the more severe impairment values. Worse impairments close to the transmitter appear to have a more destructive effective on modem BLER performance than if they appear closer to the receiver. This means that analogue impairments on access are probably more significant in affecting modem error rates than those in the long-haul network or egress. The observed effect is mild, however, probably because the impairment values of the allocated 85th percent segments are really not much poorer than those for the allocated mean segments.

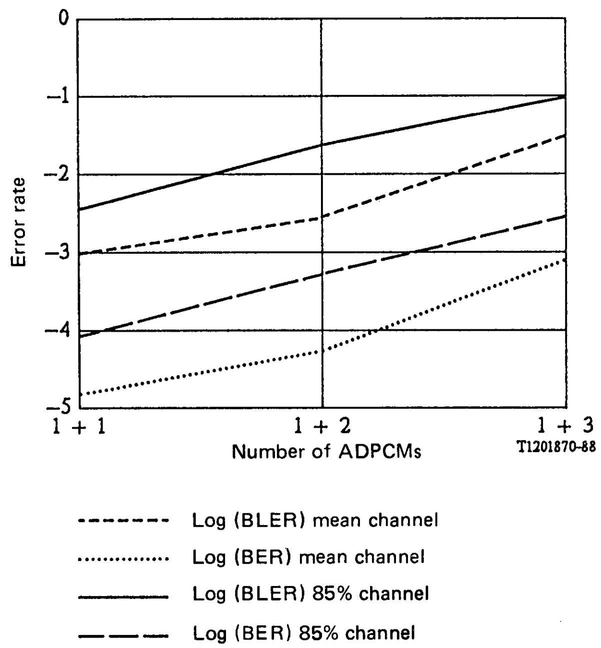


FIGURE C-4/G.113

ADPCM performance (mean and 85% channels) with a V.32 modem

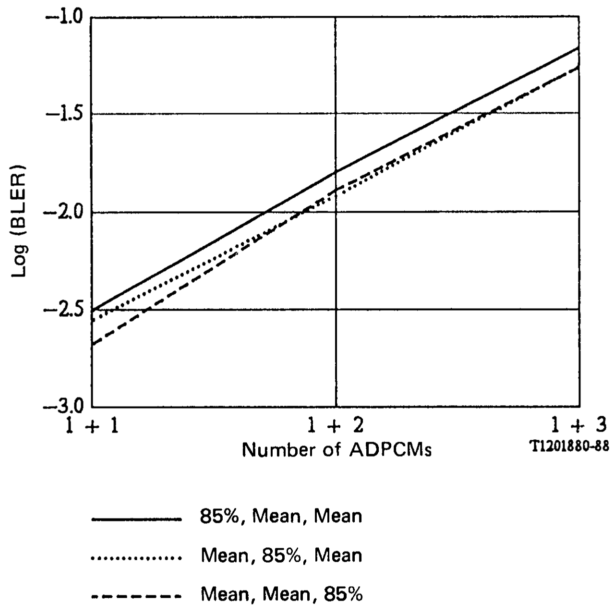


FIGURE C-5/G.113

ADPCM performance (impairment position study) with a V.32 modem

C.2.5 *9.6 kbit/s – ADPCM performance*

As a final test of modem performance, we have subjected another 9.6 kbit/s device, utilizing more traditional technology, to a sequence of asynchronously tandemed ADPCMs. This modem is a 2-wire device advertised by the vendor for use on the public switched network at signalling rates to 9.6 kbit/s. We have tested the device performance with no analogue impairments at all in the test channel. During the course of the empirical determination, it was discovered that the modem start sequence and the ADPCM algorithm interacted to prevent commencement of communication between transmitter and receiver. It was therefore necessary to test by allowing modem training to occur on an ordinary PCM channel after which ADPCMs were cut in to observe performance. Similar availability problems would also probably occur for any speed modem whose start-up training sequence is similar to that of this 9.6 kbit/s product.

Figure C-6/G.113 shows the performance results for this modem. Without analog impairments the number of ADPCMs may simply be ennumerated sequentially. The BLER outcome indicates that between 0 and 1 ADPCM encoding is all that can meet our performance criterion. For BER it appears, again by our normal criterion, that ADPCM is incompatible with proper operation of the modem. Since it is expected that many modem vendors will, or have already, announced high speed 2-wire devices for use on the public switched network, the presence of ADPCM on these channels is likely to cause performance problems for those devices which are similar to the one tested for training, modulation, and detection.

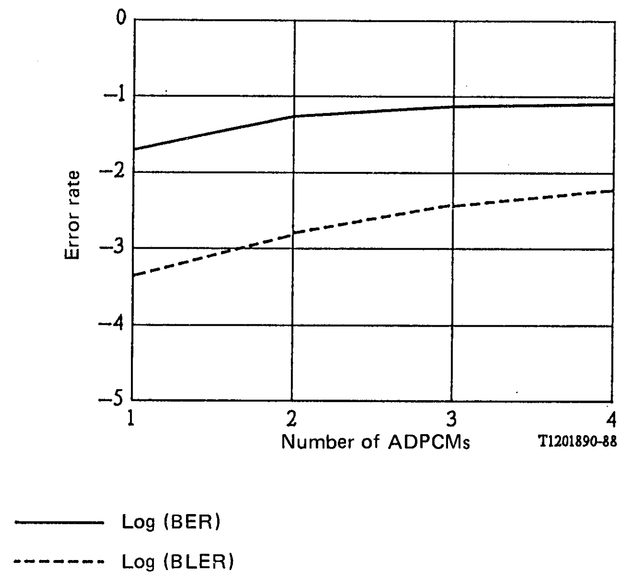


FIGURE C-6/G.113

9.6 kbit/s ADPCM performance (no analogue impairments) with a 9.6 kbit/s modem

C.3 *Conclusions*

In this Annex we have reported on the architecture, laboratory apparatus, and results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing asynchronously tandemed ADPCM systems interspersed with simulated analogue impairments. The results are compactly displayed in Table C-2/G.113 which shows that communication at 4.8 kbit/s may proceed through more asynchronous tandemed ADPCMs than in the case of using 9.6 kbit/s devices. Furthermore, communication at 9.6 kbit/s can be unacceptable when a BER criterion is applied, but sometimes acceptable when a BLER criterion is applicable. Clearly the appropriate criterion depends on the data communication user's application.

TABLE C-2/G.113

Number of allowed ADPCMs on EOCS 85% channel

Modem	BER = 10^{-5}	BLER = 10^{-2}
4.8 kbit/s (V.29)	3/4 [a)]	4/5
V.32	0/1	2/3
9.6 kbit/s	0	0/1

[a)] More recent results imply the range is 2/4.

References

[1] Draft-Proposed American National Standard 32 kbit/s ADPCM Algorithm and Line Format, Committee T1, Subcommittee T1Y1, Document No. T1Y1, LB 85-01, 28 March, 1985.

[2] CAREY, (M. B.), CHEN, (H.-T.), DESCIOUX, (A.), INGLE, (J. F.) and PARK, (K. I.): 1982/83 End Office Connection Study: Analog voice and Voiceband Data Transmission Performance Characterization of the Public Switched Network, *AT&T Bell Lab. Tech. J.*, 63, 2059, 1984.

[3] MANHIRE, (L. M.) Physical and Transmission Characteristics of Customer Loop Plant, *BSTJ*, 57, 1/70.

[4] KALB, (M.), MORTON (C. H.) and SHYNK, (JU. J.): DATACAL – A Voiceband Data Communication Connection Performance Model, *Proc. of the Second International Network Planning Symposium, University of Sussex*, Brighton, UK, 21-25 March, 1983.

Recommendation G.114

MEAN ONE-WAY PROPAGATION TIME

(Geneva, 1964; amended Mar del Plata, 1968, Geneva, 1980; Malaga-Torremolinos, 1984 and Melbourne, 1988)

The times in this Recommendation are the means of the propagation times in the two directions of transmission in a connection. When opposite directions of transmission are provided by different media (e.g. a satellite channel in one direction and a terrestrial channel in the other) the two times contributing to the mean may differ considerably.

1 Limits for a connection

It is necessary in an international telephone connection to limit the propagation time between two subscribers. As the propagation time is increased, subscriber difficulties increase, and the rate of increase of difficulty rises. Relevant evidence is given in references [1] to [10], particularly with regard to b) below.

As a network performance objective, the CCITT therefore *recommends* the following limitations on mean one-way propagation times when echo sources exist and appropriate echo control devices, such as echo suppressors and echo cancellers, are used:

a) 0 to 150 ms, acceptable.

Note – Echo suppressors specified in Recommendation G.161 of the Blue Book [11] may be used for delays not exceeding 50 ms (see Recommendation G.131, § 2.2).

b) 150 to 400 ms, acceptable, provided that increasing care is exercised on connections when the mean one-way propagation time exceeds about 300 ms, and provided that echo control devices, such as echo suppressors and echo cancellers, designed for long-delay circuits are used;

c) above 400 ms, unacceptable. Connections with these delays should not be used except under the most exceptional circumstances.

Until such time as additional, significant information permits Administrations to make a firmer determination of acceptable delay limits, they should take full account of the documents referred to under References in selecting, from alternatives, plans involving delays in range b) above.

Note 1 – The above values refer only to the propagation time between two subscribers. However, for other purposes (e.g. in Recommendation G.131) the mean one-way propagation time of an echo path is to be estimated. The values in § 2 may be used in such estimations.

Note 2 – There is good evidence that echo cancellers fitted at both ends of a long-delay connection generally yield superior performance over current types of echo suppressors. (For further details, see § 2.2 of Recommendation G.131.)

Note 3 – It should be noted that although an echo suppressor and an echo canceller on the same connection are compatible (they can satisfactorily interwork), the full benefits of echo cancellers are only experienced when both ends are so equipped. In particular, an Administration unilaterally replacing its echo suppressors with echo cancellers will cause little benefit to its own subscriber on international connections if the echo suppressor still remains at the other end.

Note 4 – Available experimental data (Annex A) has indicated that connections with delays somewhat greater than 400 ms may be acceptable provided that echo cancellers conforming to the specifications of Rec. G.165, or other echo control devices with equivalent performance, are used. However, the use of connections with delays greater than 400 ms is not recommended at present and is under study in Question 27/XII.

Note 5 – The use of equipment that introduces clipping, noise contrast, low echo return loss enhancement or other impairments that may degrade echo performance (such as may be the case with hands free telephones, especially in a changing noise environment) may have to be controlled to achieve acceptable transmission quality on connections with delays in the range from 150 to 400 ms. This subject is under study in Question 11/XII.

2 Values for circuits

In the establishment of the general interconnection plan within the limits in § 1 the one-way propagation time of both the national extension circuits and the international circuits must be taken into account. The propagation time of circuits and connections is the aggregate of several components; e.g. group delay in cables and in filters encountered in FDM modems of different types. Digital transmission and switching also contribute delays. The conventional planning values given in § 2.1 may be used to estimate the total propagation time of specified assemblies which may form circuits or connections.

2.1 *Conventional planning values of propagation time*

Provisionally, the conventional planning values of propagation time in Table 1/G.114 may be used.

2.2 *National extension circuits*

The main arteries of the national network should consist of high-velocity propagation lines. In these conditions, the propagation time between the international centre and the subscriber farthest away from it in the national network will be as follows:

a) in purely analogue networks, the propagation time will probably not exceed:

$$12 + (0.004 \times \text{distance in kilometres}) \text{ ms.}$$

Here the factor 0.004 is based on the assumption that national trunk circuits will be routed over high-velocity plant (250 km/ms). The 12 ms constant term makes allowance for terminal equipment and for the probable presence in the national network of a certain quantity of loaded cables (e.g. three pairs of channel translating equipments plus about 160 km of H 88/36 loaded cables). For an average size country (see Figure 2/G.103) the one-way propagation time will be less than 18 ms;

b) in mixed analogue/digital networks, the propagation time can generally be estimated by the equation given for purely analogue networks. However under certain unfavourable conditions increased delay may occur compared with the purely analogue case. This occurs in particular when digital exchanges are connected with analogue transmission systems through PCM/FDM equipments in tandem, or transmultiplexers. With the growing degree of digitization the propagation time will gradually approach the condition of purely digital networks;

TABLE 1/G.114

Transmission medium	Contribution to one-way propagation time	Remarks
Terrestrial coaxial cable or radio relay system; FDM and digital transmission	4 μs/km	Allows for delay in repeaters and regenerators
Optical fibre cable system; digital transmission	5 μs/km	Allows for delay in repeaters and regenerators
Submarine coaxial cable system	6 μs/km	
Satellite system – 14 000 km altitude – 36 000 km altitude	 110 ms 260 ms	Between earth stations only
FDM channel modulator or demodulator	0.75 ms [a)]	Half the sum of propagation times in both directions of transmission
FDM compandored channel modulator or demodulator	0.5 ms [b)]	
PCM coder or decoder	0.3 ms [a)]	
PCM/ADPCM/PCM transcoding	0.5 ms	
Transmultiplexer	1.5 ms [c)]	
Digital transit exchange, digital-digital	0.45 ms [d)]	
Digital local exchange, analogue-analogue	1.5 ms [d)]	
Digital local exchange, analogue subscriber line-digital junction	0.975 ms [d)]	
Digital local exchange digital subscriber line-digital junction	0.825 ms [d)]	
Echo cancellers	1 ms [e)]	

a) These values allow for group-delay distortion around frequencies of peak speech energy and for delay of intermediate higher order multiplex and through-connecting equipment.

b) This value refers to FDM equipments designed to be used with a compandor and special filters.

c) For satellite digital communications where the transmultiplexer is located at the earth station, this value may be increased to 3.3 ms.

d) These are mean values: depending on traffic loading, higher values can be encountered, e.g. 0.75 ms (1.950 ms, 1.350 ms or 1.250 ms) with 0.95 probability of not exceeding. (For details, see Recommendation Q.551.)

e) Echo cancellers, when placed in service, will add a one-way propagation time of up to 1 ms in the send path of each echo canceller. This delay excludes the delay through any codec in the echo canceller. No significant delay should be incurred in the receive path of the echo canceller.

c) in purely digital networks between exchanges (e.g. an IDN), the propagation time as defined above will probably not exceed:

$$3 + (0.004 \times \text{distance in kilometers}) \text{ ms.}$$

The 3 ms constant term makes allowance for one PCM coder or decoder and five digitally switched exchanges.

Note – The value 0.004 is a mean value for coaxial cable systems and radio-relay systems; for optical fibre systems 0.005 is to be used;

d) in purely digital networks between susbscribers (e.g. an ISDN), the delay of c) above has to be increased by up to 3.6 ms if burst-mode (time compression multiplexing) transmission is used on 2-W local subscriber lines.

2.3 *International circuits*

International circuits[1] will use high-velocity transmission systems, e.g. terrestrial cable or radio-relay systems, submarine systems or satellite systems. The planning values of § 2.1 may be used.

The magnitude of the mean one-way propagation time for circuits on high altitude communication satellite systems makes it desirable to impose some routing restrictions on their use. Details of these restrictions are given in Recommendation Q.13 [12]. (See also Annex A below.)

ANNEX A

(to Recommendation G.114)

Long propagation delay and echo related considerations for telephone circuits

A.1 *Introduction*

International connections (see Figure 1/G.103 or Figure 1/G.104) comprising submarine cables, may involve a maximum one-way transmission delay of about 170 ms. This Annex addresses the basic issues of national and international connections which inherently entail comparatively larger one-way transmission delays.

A one hop satellite connection even with an ISL (Inter-Satellite Link) of moderate length introduces one-way transmission delay within the recommended limit of 400 ms. However, a careful analysis of the additional probable delay contributions by digital signal processing (e.g. TDMA, DSI, DCME, 16 kbit/s and 32 kbit/s low bit rate encoding, bit-regeneration, packet-switching, etc.), among other sources, has led to the notion that the recommended limit of 400 ms mean one-way propagation delay may be unnecessarily restrictive.

In light of recent technical improvements in echo-control techniques, it is feasible to consider an extension to this limit. Administrations are encouraged to take note of the continuing nature, as well as need, of further investigations in this area.

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[1] For short nearby links, telecommunications cables operated at voice frequencies may also be used in the conditions set out in the introduction to Sub-section 5.4 of Fascicle III.2.

In order to analyse this problem further, consider that two distinct types of effects must be considered in connection with the mean one-way propagation time; namely, echo-related speech quality impairments and pure (transit) delay related conversational difficulty. Echo control devices, i.e. echo suppressors and especially echo cancellers, can be suitably employed for overcoming the former effect.

The 4-wire circuits provides a close approximation to echo-free connections, assuming minimum acoustic coupling across the handset. In the long run with expansion of the ISDN implementation, use of 4-wire circuits is expected in grow. However, 2-wire circuits and their accompanying hybrid connection, as well as other componentes causing echo, are still likely to be present in vaying degrees during the foreseeable future. Thus, the use of modern echo cancellers in satellite circuits is currently regarded as the most effective method for overcoming the echo problem, provided that the characteristics of the echo path to be modeled by the echo canceller are linear and time invariant, or varying only slowly compared with the convergence speed of the echo canceller.

A brief discussion of delay measurements, their effect on circuit quality and the subscriber reaction are provided below.

A.2 *Effect of long transmission delays on the subscriber*

A.2.1 *Early measurements*

Figure A-1/G.114 shows the effect of long transmission delay on the difficulty of conversation experienced by the subscriber. Curve 1 is the result of investigations in 1964 and 1965 [5, 8 *et al.*] where the performance of the first operational satellite Early Bird was tested in circuits between France, the United Kingdom, the United States and the Federal Republic of Germany. The circuits were equipped with early versions of various echo suppressors, had a certain amount of noise power (about 20 000 pW0p), and had different bandwidths on the TAT-3 cable route (230-3200 Hz) as opposed to the satellite (170-3400 Hz). Curve 1 (F/P) shows the same interview results on the basis of a fair-or-poor opinion rating by the subscribers.

From curve 1 it can be seen that, at about 400 ms of delay, more than 50% of the subscribers have difficulties with the conversation. A 40% value of difficulty corresponds to a delay of about 300 ms. On the other hand, the percentage of fair-or-poor opinions of the subscribers is about 15% lower than the percentage of difficulties. This may result from the fact that some of the inquired customers, in spite of the difficulties they had, found the received speech quality good or excellent.

On the basis of these observations, 300 ms of delay was selected as the threshold of difficulty and 400 ms as the maximum allowable delay in international connections for telephony in earlier versions of Rec. G.114.

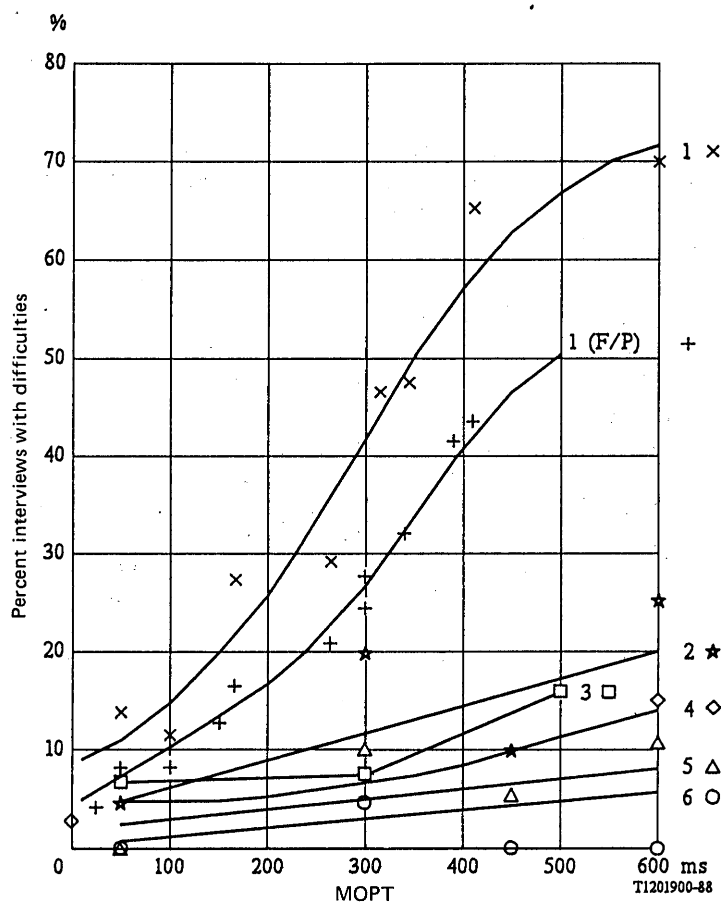
In addition to these results, other ealier results exist. Williams and Moye [30, 31] investigated the effect of unsuppressed echo on conversations over simulated telephone links with different values of echo return loss and with flat or shaped echo-path frequency characteristics.

Curves 2, 5 and 6 show the results for connections with echo return losses of 37 dB (shaped), 37 dB (flat) and 50 dB (flat or shaped). Curve 4 shows laboratory test results [32] or simulated connections equipped with echo suppressors and with an echo return loss of about 20 dB. These test results were obtained using a linear time invariant echo path.

Figure A-1/G.114 also includes some recent results obtained from circuits with long delay but which were equipped with modern echo cancellers with an echo return loss of about 18 dB [29] (see § A.2.3).

From curves 2 to 6 (which obtained better methods of echo control or high echo return loss values) it can be seen that the influence of longer propagation delay on the difficulties of conversations is much smaller than indicated by curve 1, which used earlier versions of echo suppressors.

Other investigations summarized in [33] which were obtained from circuits having only pure transmission delay (i.e. echo free 4-wire circuits), have shown that mean one-way propagation delays up to 600 ms appear to have no significant influence on the subjective judgements of telephone subscribers.



Test conditions:

Curve No.	Echo control	Kind of test	ERL (dB)	Line noise	Room noise [dB (A)]	Reference
1	ES	International field test (1964/65)		20 000 pW0p	–	/5, 8/
2	–	Laboratory tests (1970)	37 (shaped)	–61.5 dBmp	50	/30, 31/
3	EC	National field test (1987)	18	–	–	/29/
4	ES	Laboratory tests (1970)	> 20	–50 dBm0p	50	/32/
5	–	Laboratory tests (1970)	37 (flat)	–61.5 dBmp	50	/30, 31/
6	–	Laboratory tests (1970)	50	–61.5 dBmp	50	/30, 31/

ES Echo suppressor

EC Echo canceller

FIGURE A-1/G.114

Effect of long mean one-way propagation times (MOPT) on the difficulty of conversion

A.2.2 *Later measurements*

Following technical advancement, design developments and performance enhancements of echo cancellers [16-19], experiments were conducted by Helder and Lopiparo [20], DiBiaso [21], Post and Silverthorn [22], and others to evaluate the subjective performance of echo suppressors and echo cancellers on satellite and terrestrial facilities in the U.S., Canada and other domestic satellite networks.

Helder and Lopiparo [20] reported results of testing of certain terrestrial, half-hop satellite[2)], and one-hope satellite circuits in the U.S. in 1976 and 1977. DiBiaso's report [21] is based on a study of tests and subjective evaluation of echo control methods performed during 1975-77 by the American Telephone and Telegraph Company (AT&T) and others using the U.S. domestic satellite system (COMSTAR), together with conventional analog echo suppressors (ES), digital echo suppressors (DES) [23] and experimental echo cancellers (EC) [24-25], and examining the cases of terrestrial, half-hop satellite, one-hope satellite and two-hop satellite connections, respectively. A detailed account of these test results is provided elsewhere [26]. A summary of these test results, represented in terms of the percent of calls rated unacceptable for the various cases mentioned above, is reproduced here in Figure A-2/G.114. The graph demonstrates the improvement possible through the use of the digital echo suppressor and echo canceller in the half-hop and one-hop satellite connections, respectively, to yield performances in these two cases practically equivalent to the terrestrial circuits with echo suppressors. Basically, similar conclusions were reached by using somewhat different criteria for performance and quality; e.g. percent of calls terminated early or percent of calls replaced, or percent of calls needing operator assistance.

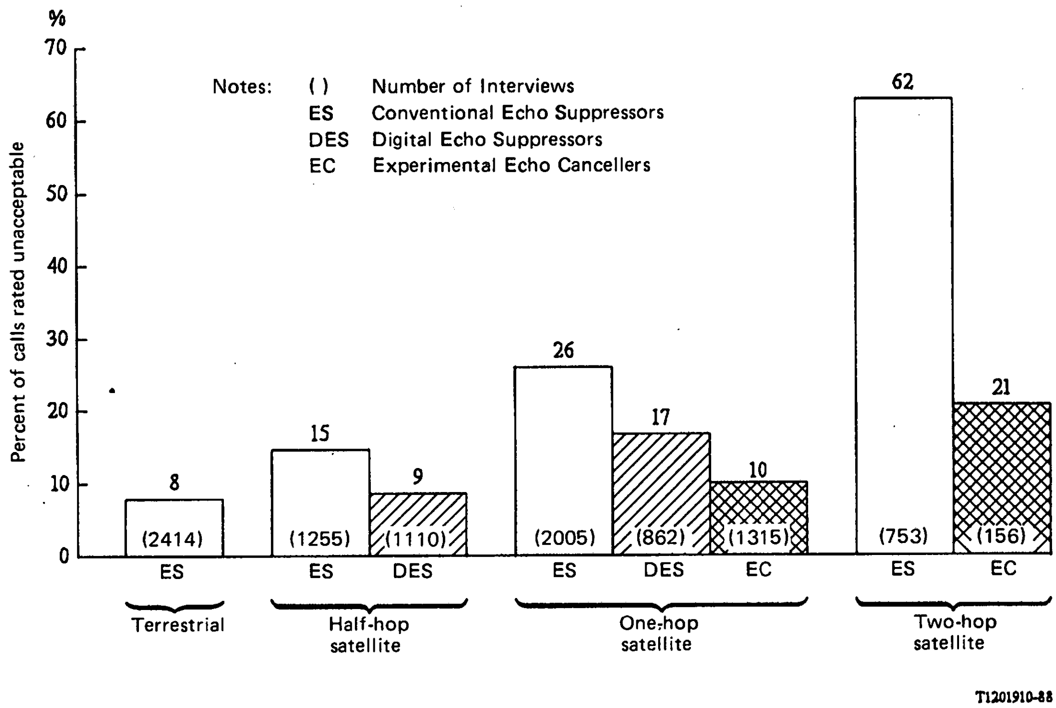


FIGURE A-2/G.114

Domestic satellite user reaction test results comparing echo control methods

2) Half-hop connection refers to the situation when the forward link is via satellite but the return link is terrestrial (or vice-versa).

In 1978, Post and Silverthorn [22] performed an evalutation of nine experimental conditions characterized by generically different methods of echo control on the Trans-Canada Telephone System (TCTS) satellite and certain terrestrial links. Figure A-3/G.114 provides a partial summary of their results in terms of percent of interviews that judged the terrestrial, echo canceller-equipped satellite (S/EC) and echo suppressor-equipped satellite (S/ES) circuits as excellent, good, fair, or poor as regards to quality. Figure A-4/G.114 provides a summary of analogous test results as derived from similar domestic and international satellite and terrestrial networks [22]. These results serve to illustrate the near equivalence of the performance of satellite circuits equipped with echo cancellers and long-haul terrestrial circuits with echo suppressors. These results also demonstrate the poorer performance of echo suppressors as compared to echo cancellers in the satellite link. Consequently, echo suppressors are not considered optimal for satellite links and only echo cancellers are recommended to be employed. For terrestrial applications, the improvement resulting from the use of echo cancellers is expected to be only marginal; and system economy may still justify the use of echo suppressors in the terrestrial links.

The above observations confirm the conclusion that the difficulties experienced by telephone users of satellite networks is primarily due to echo related impairments associated with the long propagation delay. This impairment can be sufficiently reduced with the use of echo cancellers to yield a performance for one-hop satellite connections practically equivalent to that of terrestrial connections [27-28].

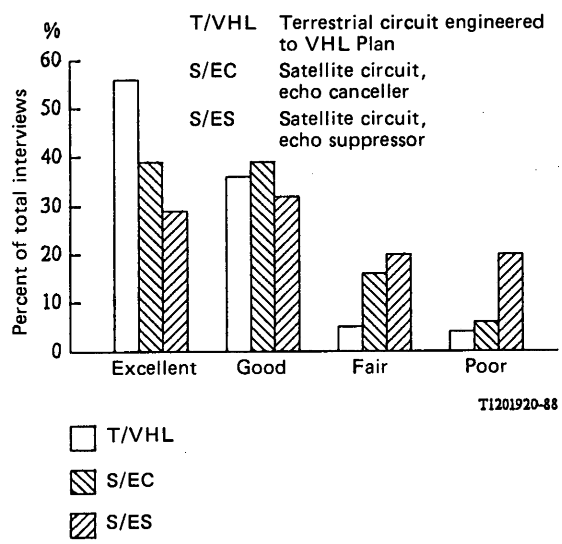


FIGURE A-3/G.114

Distribution of responses for Toronto-Halifax calls

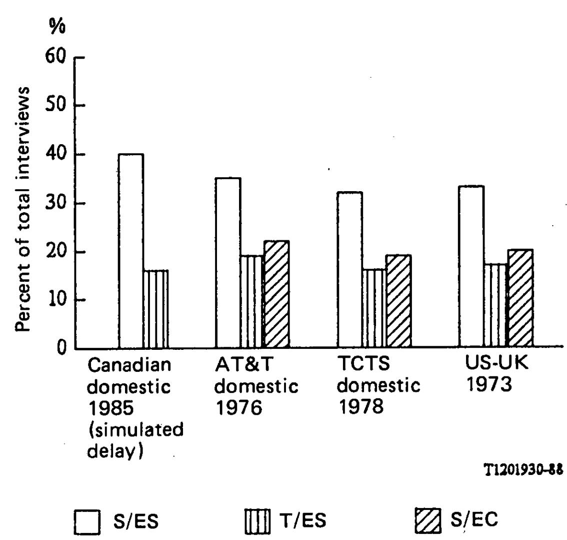


FIGURE A-4/G.114

Customer opinion tests on satellite calls from 1965 to 1978

A.2.3 *Recent and future measurements*

In 1987, Communications Satellite Corp. (COMSAT) of the U.S.A. performed a series of tests to determine the effectiveness of echo cancellers in terrestrial and satellite circuits, using echo cancellers conforming to Rec. G.165 and a callback interview procedure as per Rec. P.77, Annex A. Details of the procedure were presented recently [29] and a summary of the results is shown in Figure A-1/G.114, curve 3 giving a plot of the percent difficulty as a function of mean one way propagation time. A one way delay value of 45 ms over terrestrial circuits was taken as a reference, and the effect of increasing the delay value to 300 ms and 500 ms over terrestrial and satellite links was evaluated.

It was concluded on the basis of the COMSAT results that no significant difference between 45 ms and 300 ms delays resulted for the "percent difficulty" score. At a 500 ms delay, the percent difficulty score approximately doubled (from 7.3% to 15.8%), but this value is still considerably smaller than earlier results of over 60% [13].

The above results support the view that connections with delays somewhat greater than 400 ms may be accepted provided that echo cancellers conforming to the specifications of Recommendation G.165 or other echo control devices with equivalent performance are used. This may permit accommodation of signal processing and Inter Satellite Links (ISL) with moderate angular separations, without causing any significant or noticeable degradations.

Further tests, measurements and evaluation of subjective performance using state-of-the-art echo cancellers in modern satellite connections should prove to be useful to determine what, if any, additional improvements over these results are likely or achievable.

A.3 *Summary and conclusions*

The transmission impairments associated with long delay circuits are best analysed by separating the echo-induced degradation and the subjective difficulty due to pure delay. Clearly, as shown by the tests cited above, echo suppressors (with fixed break-in sensitivity) used in satellite circuits are far less efficient than echo cancellers. The effectiveness of echo cancellers in removing the echo effect and the associated impairments is sufficient to yield high or acceptable performance in a long delay satellite circuit. Further improvement in the performance of echo cancellers and the associated satellite circuits are continuing. Thus, under these conditions the dominant impairments are associated with the pure delay component.

A number of recent works and continuing interest in the area indicate the possibility of developing and utilizing even more improved and efficient echo cancellers. VLSI fabrication of echo cancellers is also a viable option and this is expected to lead to a significantly lower cost for equipping long delay telephone circuits. Thus, with the use of such suitable devices, the comparatively larger pure delay in international connections is not expected to cause the degree of degradation in the channel quality or efficiency as was experienced in earlier tests without echo control or with echo suppressors with fixed break-in sensitivity. Appropriate use of echo cancellers has been shown to indeed provide international or national satellite connections yielding quality and performance practically equivalent to the terrestrial connections for telephony. These results only refer to electric echo and additional studies are necessary to determine the effect of acoustic echo (see Note 5 of Question 27/XII).

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Recommendation G.117

TRANSMISSION ASPECTS OF UNBALANCE ABOUT EARTH (DEFINITIONS AND METHODS)

(Geneva, 1980; amended at Malaga-Torremolinos, 1984 and Melbourne, 1988)

1 Objective

This Recommendation gives a comprehensive set of prescriptive measurements of various balance parameters for one-port and two-port networks. These are intended for use either in the field or in the factory with relatively simple test apparatus (e.g. standard transmission oscillators, level measuring sets), and a special test bridge. Measuring arrangements for assessing the degree of unbalance are covered in Recommendation O.121 [1], which are consistent with this Recommendation.

The definitions and methods are so devised that the results obtained from separately-measured (or specified) items of equipment (e.g. feeding-bridges, cable pairs, audio inputs to channel translating equipment, etc.) can be meaningfully combined though not necessarily by simple decibel addition. This allows the performance of a tandem connection of such items to be predicted or at least, bounds determined for that performance. Performance in this sense means those features affected by unbalanced conditions, e.g. level of impulsive noise, sensitivity to longitudinal exposure, crosstalk ratios, etc.

2 Principles of the scheme of nomenclature

Many different terms have been used throughout the literature concerning unbalance about earth, some conflicting, or in other respects inadequate. The descriptive titles of the quantities given in this Recommendation are based on the following principles which have been adopted:

a) Mode *conversion*, e.g. a poor (unbalanced) termination will develop an unwanted transverse signal when excited by a longitudinal signal. The measure of this effect is here termed *longitudinal conversion ratio*, and when expressed in transmission units *longitudinal conversion loss*, or LCL.

b) When a two-port is involved where for example an excitation at one port produces a signal at the other port, then the designation will include the word *transfer*, for example *longitudinal conversion transfer ratio* and the corresponding *loss*, LCTL.

c) The impedance of the longitudinal path presented by a test object is a key parameter. The term *longitudinal impedance ratio* and the corresponding decibel expression, *longitudinal impedance loss*, are used to characterize the particular measurement defined.

d) Active devices which are sources of signals (e.g. an oscillator, the output port of an amplifier) are additionally characterized by the amount of unwanted longitudinal signal that is present in the output. The key word *output* is now included, to give *longitudinal output voltage*, and the corresponding *longitudinal output level*. When such unwanted signals are expressed as a proportion of the wanted (transverse) signal the key phrase is *output signal balance ratio*, the decibel expression of which is *output signal balance*.

e) Devices which continuously respond to signals (e.g. level-measuring sets, the input port of an amplifier) and which can in principle respond to unwanted longitudinal signals by reason of internal mechanisms (i.e. even if their input impedances were perfectly balanced) are characterized by measures containing the words *input interference*. These measures are *input longitudinal interference ratio* and the corresponding decibel expression *input longitudinal interference loss*. The long-established and well-defined *common-mode rejection ratio* is maintained. The term *sensitivity coefficient* is avoided, since this is widely used in the Directives [2] and the work of Study Group V with a rather specialized meaning.

f) When a two-port network is involved, the input and output signals may not be the same, for example, they may have different levels, frequencies (FDM modems) or structure (PCM multiplex equipments). These aspects should be taken into account when formulating proposals for the item under test.

g) In the case of receiving devices in which the operation is not a linear continuous function of the level of the input signal (e.g. a group-delay measuring set or a data modem) the key principle is the *threshold* level of the interference; this is the level at or above which an unacceptable amount of degradation of performance or misoperation occurs. Thus *longitudinal interference threshold voltage* and the corresponding *levels* are obtained.

3 Summary of the descriptive terms used

3.1 *One-port networks*

a) transverse reflexion factor (transverse return loss: TRL),

b) transverse conversion ratio (loss: TCL),

c) longitudinal conversion ratio (loss: LCL),

d) longitudinal impedance ratio (loss: LIL),

e) transverse output voltage (level: TOL),

f) longitudinal output voltage (level: LOL).

(Voltages e) and f) are unwanted signals uncorrelated to the wanted signals.)

3.2 *Two-port networks*

3.2.1 *Separate measurement*

For each port taken separately the one-port measures:

a) transverse reflexion factors (transverse return losses: TRL),

b) transverse conversion ratio (loss: TCL),

c) longitudinal conversion ratios (losses: LCL),

d) longitudinal impedance ratios (losses: LIL),

e) transverse output voltage (levels: TOL),

f) longitudinal output voltage (levels: LOL).

3.2.2 *Measurement combined*

In addition the following transfer parameters are for each of the two directions of transmission:

a) transverse transfer ratios (losses: TTL),

b) transverse conversion transfer ratios (losses: TCTL),

c) longitudinal transfer ratios (losses: LTL),

d) longitudinal conversion transfer ratios (losses: LCTL).

3.3 *Signal generating devices*

a) Output signal balance ratio (losses: OSB).

This is in addition to the six one-port measures listed in § 3.1.

3.4 *Signal receiving devices*

a) Input longitudinal interference ratio (loss: ILIL).

b) Longitudinal interference threshold voltage (level).

These are in addition to the six one-port measures listed in § 3.1. If the wanted signal is longitudinal (e.g. as in a signalling system) and the interfering voltage transverse, replace the word *longitudinal* with *transverse* in the descriptive terms.

4 Definitions and measuring techniques based on idealized measuring arrangements

The illustrated definitions in this section assume ideal test bridges (with lossless infinite-inductance centre-tapped coils), zero impedance voltage generators and infinite-impedance voltmeters.

An important aspect of this set of mutually consistent measurements is that the test bridge provides simultaneously defined reference terminations of Z ohms for the transverse paths, and $Z/4$ ohms for the longitudinal paths. From this starting point, the performance of cascaded items, each measured in the prescribed fashion, can be calculated. This takes account of the fact that the cascaded items do not, in general, exhibit the reference impedances provided by the test conditions.

It simplifies the mathematical treatment if the reference impedance is nonreactive and this also accords with the important objective of being able to use readily-available transmission test-apparatus to obtain field and factory measurement results.

The ideal test bridge configuration used in the following pages is shown in Figure 1/G.117.

The transverse and longitudinal sources E_T and E_L are activated as required by the particular measurement being made. In Figure 6/G.117, neither source is active, and the bridge then provides only passive terminations of Z and $Z/4$.

Note – It would have been in keeping with traditional transmission theory for the parameters to be defined in terms of half the open-circuit e.m.f. However, to harmonize with Recommendation O.121, this Recommendation defines some parameters in terms of V_{T1}. If the input impedance of the device under test is nominally equal to the driving device, then the two methods are equivalent.

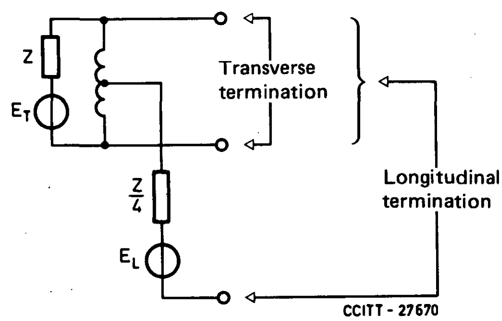
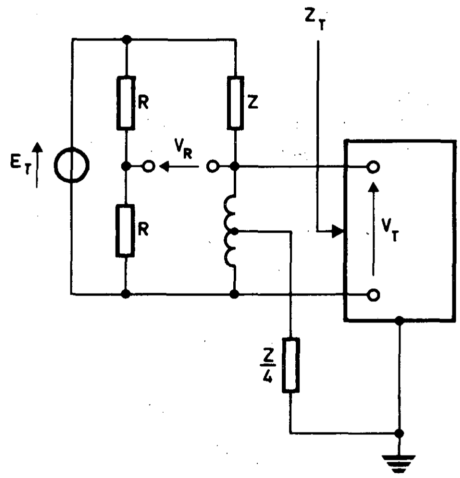


FIGURE 1/G.117

4.1 *One-port networks*

4.1.1 *Transverse reflexion factor (return loss)* (see Figure 2/G.117)



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$$\text{Transverse reflexion factor } \rho = \frac{Z - Z_T}{Z + Z_T} = \frac{\text{reflected voltage}}{\text{forward voltage}} = \frac{2V_R}{E_T}$$

and

$$\text{Transverse return loss (TRL)} = 20 \log_{10}\left|\frac{1}{\rho}\right| = 20 \log_{10}\left|\frac{E_T}{2V_R}\right| \text{ dB.}$$

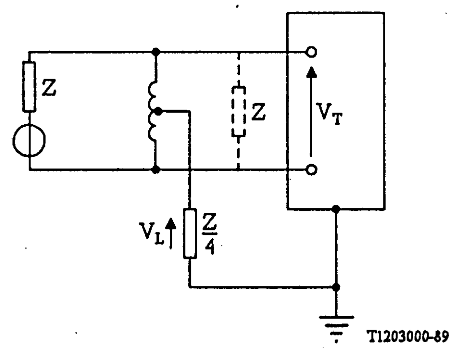
Note 1 – **The value of *R* is (theoretically) irrelevant. The potential divider across the zero-impedance generator is only needed to derive half the generator voltage, which is numerically equal to the forward voltage needed for the definition.**

Note 2 – **Conventional return-loss measuring bridges do not terminate the longitudinal path with *Z*/4. This is unimportant when the return loss is some 20 dB or so less than the longitudinal conversion loss of the test object. In this case the reflected power is substantially greater than the power diverted to the longitudinal path, and there is negligible error.**

Note 3 – If Z_T is known then clearly $\rho = 1 - \frac{2V_T}{E_T}$ is not needed. If V_T is measured ρ can be calculated from the expression $\rho = 1 - \frac{2V_T}{E_T}$, which is however somewhat inconvenient for high values of return loss.

FIGURE 2/G.117

4.1.2 *Transverse conversion ratio (loss)* (see Figure 3/G. 117)



$$\text{Transverse conversion ratio, } k = \frac{V_L}{V_T}$$

and

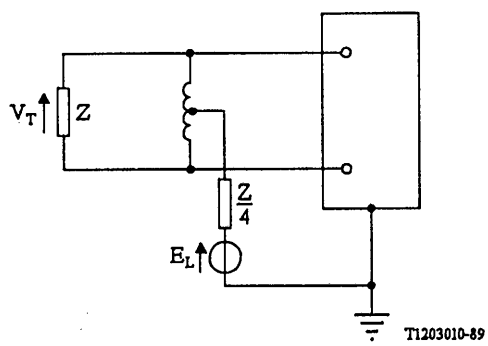
$$\text{Transverse conversion loss (TCL)} = 20 \log_{10}\left|\frac{1}{k}\right| = 20 \log_{10}\left|\frac{V_T}{V_L}\right| \text{ dB.}$$

Note 1 – In the case where the network is linear passive and bilateral, the transverse conversion loss (TCL) is equal to half the longitudinal conversion ratio *c*. However, this relationship is not true for other network arrangements.

Note 2 – The dotted component is needed for a two-terminal device which, when in use, only bridges the transmission circuit and will not be explicitly referred to again.

FIGURE 3/G.117

4.1.3 *Longitudinal conversion ratio (loss)* (see Figure 4/G.117)



$$\text{Longitudinal conversion ratio, } c = \frac{V_T}{E_L}$$

and

$$\text{Longitudinal conversion loss (LCL)} = 20 \log_{10}\left|\frac{1}{c}\right| = 20 \log_{10}\left|\frac{E_L}{V_T}\right| \text{ dB.}$$

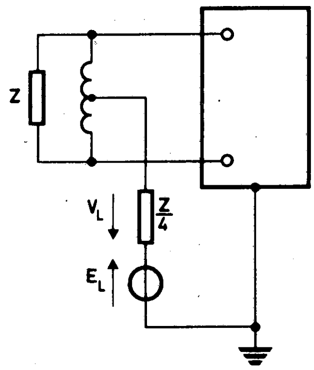
Note 1 – This measure is variously referred to in other Recommendations as:

a) Longitudinal balance
b) Degree of unbalance
c) Unbalance
d) Degree of longitudinal balance
e) Signal balance ratio
f) Impedance unbalanced to earth.

Note 2 – The longitudinal conversion ratio is applicable to any one-port, even to those which are sources of signals (e.g.: oscillator output terminals). In such cases the transverse voltage V_T must be measured selectively if it is required to measure this loss in respect of a signal generator in operation. See § 5.2.

FIGURE 4/G.117

4.1.4 *Longitudinal impedance ratio (loss)* (see Figure 5/G.117)



$$\text{Longitudinal impedance ratio, } q = \frac{E_L}{V_L}$$

and

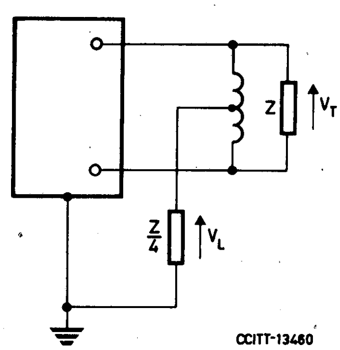
$$\text{Longitudinal impedance loss (LIL)} = 20 \log_{10} |q| = 20 \log_{10} \left| \frac{E_L}{V_L} \right| \text{ dB.}$$

Note 1 – This is an additional measure that is needed if the performance of a cascade of items is to be predicted.

Note 2 – In the case of test-objects which are virtually earth free (e.g.: double-insulated, portable test apparatus with no deliberate connection to earth) the value of V_L will be very small and the corresponding ratio (and loss) will be very large. In such cases the coupling introduced between longitudinal and transverse paths will be very small and the effect is not important.

FIGURE 5/G.117

4.1.5 *Transverse and longitudinal output voltages (levels)* (see Figure 6/G.117)



$$\text{Transverse output voltage} = V_T$$
$$\text{Transverse output level (TOL)} = 20 \log_{10} \left| \frac{V_T}{1 \text{ volt}} \right| \text{ dBV.}$$
$$\text{Longitudinal output voltage} = V_L$$
$$\text{Longitudinal output level (LOL)} = 20 \log_{10} \left| \frac{V_L}{1 \text{ volt}} \right| \text{ dBV.}$$

Note 1 – These measures relate to unwanted signals uncorrelated to the wanted signal. For example, a d.c. signalling system in the longitudinal path may deliver unwanted transverse signals. Similarly the output of an amplifier may deliver an unwanted longitudinal "hum" signal, or a cable pair may deliver unwanted longitudinal signals arising from induction or radiation.

Note 2 – Other reference voltages than 1 volt may be used, for exemple 0.775 V for 1 mW at 600 Ω (with the designation dB [3]).

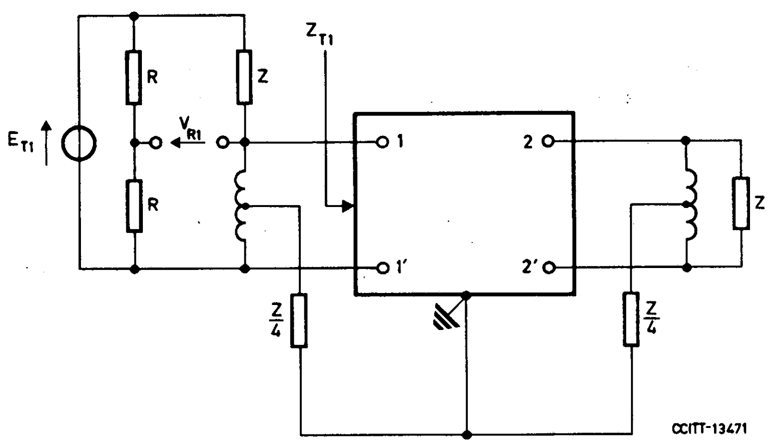
FIGURE 6/G.117

4.2 *Two-port networks*

These follow similar principles to those defined for one-port networks but now signals can be transferred from one port to the other. The two ports are distinguished by the subscripts 1/1′ for one end and 2/2′ for the other. There are two types of measurements:

- those in which the excitation and response are at the same side of the network; these are as already defined for a one-port but will carry a single subscript 1/1 or 2/2′ as appropriate;
- those in which the excitation and response are at opposite sides of the network. The designation will contain the word transfer and the symbol two subscripts, the order of which indicates the direction of transmission.

4.2.1 *Transverse reflexion factors (return losses)* (see Figure 7/G.117)



$$\text{Transverse reflexion factor at port 1/1} = \rho_1 = \frac{Z - Z_{T1}}{Z + Z_{T1}} = \frac{2V_{R1}}{E_{T1}}$$

and

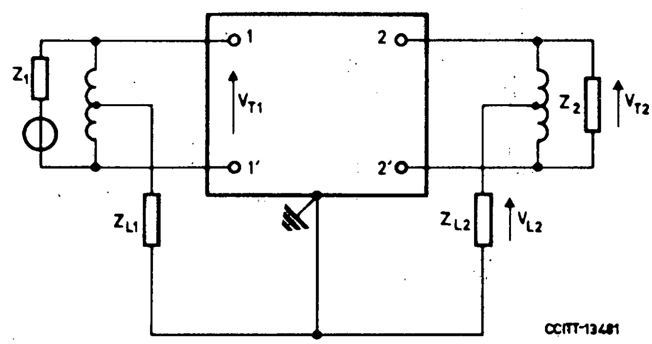
$$\text{Transverse return loss at port 1/1' (TRL}_1) = 20 \log_{10}\left|\frac{1}{\rho_1}\right| = 20 \log_{10}\left|\frac{E_{T1}}{2V_{R1}}\right| \text{ dB}$$

and similarly for port 2/2′ (TRL_2).

Note – Z_{T1} is the impedance presented by port 1/1′ when port 2/2′ is terminated with a test-bridge as shown.

FIGURE 7/G.117

4.2.2 *Transverse transfer ratios (losses) and conversion transfer ratios (losses)* (see Figure 8/G.117)



$$\text{Transverse transfer ratio 1 to 2} = g_{12} = \frac{V_{T2}}{V_{T1}}$$

and

$$\text{Transverse transfer loss 1 to 2 } (\text{TTL}_{12}) = 20 \log_{10} \left| \frac{1}{g_{12}} \right| = 20 \log_{10} \left| \frac{V_{T1}}{V_{T2}} \right| \text{ dB.}$$

$$\text{Transverse conversion transfer ratio 1 to 2} = t_{12} = \frac{V_{L2}}{V_{T1}}$$

and

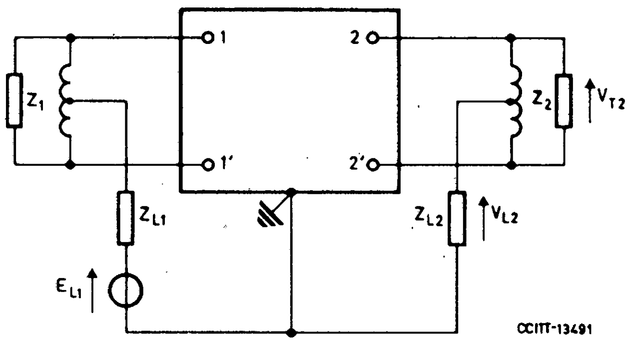
$$\text{Transverse conversion transfer loss 1 to 2 } (\text{TCTL}_{12}) = 20 \log_{10} \left| \frac{1}{t_{12}} \right| = 20 \log_{10} \left| \frac{V_{T1}}{V_{L2}} \right| \text{ dB.}$$

Interchanging 1 and 2 gives the definition for the transfer ratios TCTL for the other direction of transmission.

Note – Z_1 and Z_2 are the terminating impedances connected to the input and/or output port respectively of the item under test. Z_1 and Z_2 are generally within ± 25 percent of the nominal impedance of the port to which they are connected. If measurements are made via high-impedance input ports, an additional impedance Z_1 should be connected to the input port 1/1. The longitudinal impedances Z_{L1} and Z_{L2} are nominally equal to $Z_1/4$ and $Z_2/4$ respectively. Different values, however, may be used. This may be necessary to more properly simulate operating conditions of the time under test. In such cases the value of Z_{L1} or Z_{L2} shall be specified by the Recommendation covering the item under test.

FIGURE 8/G.117

4.2.3 *Longitudinal transfer ratios (losses) and conversion transfer ratios (losses)* (see Figure 9/G.117)



$$\text{Longitudinal transfer ratio 1 to 2} = m_{12} = \frac{V_{L2}}{E_{L1}}$$

and

$$\text{Longitudinal transfer loss 1 to 2 (LTL}_{12}) = 20 \log_{10} \left| \frac{1}{m_{12}} \right| = 20 \log_{10} \left| \frac{E_{L1}}{V_{L2}} \right| \text{ dB.}$$

$$\text{Longitudinal conversion transfer ratio 1 to 2} = h_{12} = \frac{V_{T2}}{E_{L1}}$$

and

$$\text{Longitudinal conversion transfer loss 1 to 2 (LCTL}_{12}) = 20 \log_{10} \left| \frac{1}{h_{12}} \right| = 20 \log_{10} \left| \frac{E_{L1}}{V_{T2}} \right| \text{ dB.}$$

Interchanging ports 1/1′ and 2/2′ gives the definitions for the transfer ratios and losses LTL_{21} and $LCTL_{21}$ for the other direction of transmission.

Note 1 – This measure is referred to in other Recommendations as *impedance imbalance to earth.*

Note 2 – It would have been more in keeping with traditional transmission theory if these quantities were defined in terms of *half* the open-circuit e.m.f. However, the CCITT Recommendations concerning balance parameters involving a longitudinal excitation are already in terms of the open-circuit e.m.f. It is not thought useful to introduce a 6-dB "discrepancy" between existing practice and these new definitions.

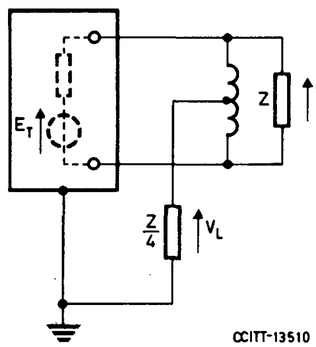
Note 3 – Z_1 et Z_2 are the impedances connected in parallel to the input and/or output port respectively of the item under test. Z_1 and Z_2 are generally within ± 25 percent of the nominal impedance of the port to which they are connected. If measurements are made via high-impedance input ports, an additional impedance Z_1 should be connected between ports 1/1′. The longitudinal impedances Z_{L1} and Z_{L2} are nominally equal to $Z_1/4$ or $Z_2/4$ respectively. Different values, however, may be used. This may be necessary to properly simulate operating conditions of the item under test. In such cases the value Z_{L1} and/or Z_{L2} shall be specified by the Recommendation covering the item under test.

FIGURE 9/G.117

4.3 *Signal generating devices*

In addition to the six one-port measures already defined, an additional measure is required to control the amount of unwanted signal correlated with the wanted signal delivered by the device to the circuit it is connected to. This special measure is the output signal balance ratio (loss).

4.3.1 *Output signal balance ratio (loss)* (see Figure 10/G.117)



$$\text{Output signal balance ratio, } b = \frac{V_L}{V_T}$$

and

$$\text{Output signal balance loss (OSB)} = 20 \log_{10} \left| \frac{1}{b} \right| = 20 \log_{10} \left| \frac{V_T}{V_L} \right| \quad \text{dB.}$$

Note 1 – This measure is a generalized version of the quantities referred to as the unbalance of ouput e.m.f.

Note 2 — This measure is also related in a somewhat indirect and complicated fashion to the sensitivity coefficients for electromagnetic and electrostatic induction defined in [2], if the cable pair is considered as a simultaneous source of a transverse signal correlated with the induced longitudinal voltages.

Note 3 – The test object itself provides the source of signal. Hence a separate generator is not required.

Note 4 – The definition relates particularly to generators of transverse signals (e.g.: transmission oscillators) but can be readily extended to cover the case of a longitudinal signal generator (e.g.: a low-frequency signalling system using the earthed-phantom). In this case the ratio could be inverted so that the decibel expression remains positive.

Note 5 – The other quantities (return loss, longitudinal conversion loss, longitudinal impedance loss and the uncorrelated transverse and longitudinal output voltages) must be measured selectively in order that their values in working conditions be obtained.

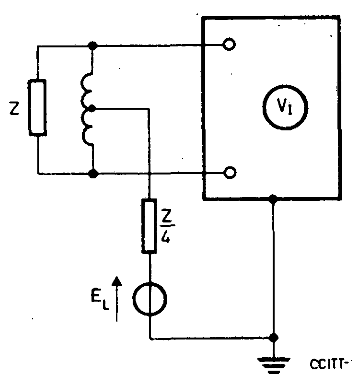
FIGURE 10/G.117

4.4 *Signal receiving devices*

In addition to the six one-port measures already defined, additional measures are required for signal receiving devices to control their sensitivity to unwanted signals. Two cases are important. Firstly, there are receiving devices in which the response is a linear, continuous function of the wanted signal level, e.g. the indication of a level-measuring set. In this case unwanted signals give rise to *inaccuracy.*

In the other kind of receiver such as data modems, group-delay distortion measuring sets, signalling receivers, unwanted signals cause errors or *misoperation.* Two additional measures are defined.

4.4.1 *Input longitudinal interference ratio (loss)* (see Figure 11/G.117)



$$\text{Input longitudinal interference ratio} = s = \frac{V_I}{E_L}$$

and

$$\text{Input longitudinal interference loss} = 20 \log_{10}\left|\frac{1}{s}\right| = 20 \log_{10}\left|\frac{E_L}{V_I}\right| \text{ dB},$$

in which V_I is the voltage indicated by the measuring set being tested.

Note 1 – This is a generalized version of the quantities referred to as the receiver signal balance ratio (Recommendation O.41 [4]).

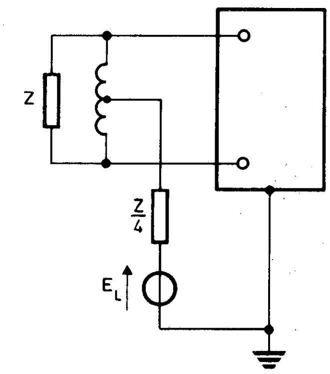
Note 2 – The measuring instrument itself provides one of the voltages required by the definition.

Note 3 – This measure is related to the well-known *common-mode rejection ratio* but not in any simple fashion. In particular it is not 6 dB different. This is because when the longitudinal rejection ratio is measured, the input transverse terminals are short-circuited and there is no transverse signal to generate any additional longitudinal signal via the unbalance of the input impedance. See § 5.3 for further explanation.

Note 4 – The concept could be extended to cover receivers which respond linearly to longitudinal signals, and here it is transverse signals that interfere. The designation would then be input *transverse* interference ratio (loss) with a correspondingly different circuit arrangement.

FIGURE 11/G.117

4.4.2 *Longitudinal interference threshold voltage (level)* (see Figure 12/G.117)



$$\text{Longitudinal interference threshold voltage} = E_L$$

and

$$\text{Longitudinal interference threshold level} = 20 \log_{10} \left| \frac{E_L}{1 \text{ volt}} \right| \text{ dBV},$$

in which E_L is the voltage at which misoperation of the test device just occurs.

Note 1 – Other reference voltages than 1 volt may be used, for example, 0.775 V for 1 mW into 600 Ω (with the designation dB [3]).

Note 2 – "Misoperation" or the amount of degradation of performance would have to be defined. For a data modem it might have to be in terms of error ratio.

Note 3 – The threshold voltage may be specified as an rms value, or as an impulsive voltage as measured by an impulsive counter, or in terms of its waveshape (e.g.: square, triangular).

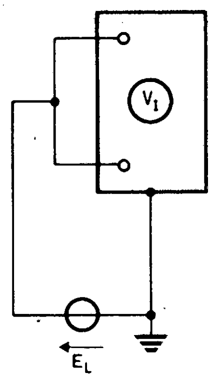
Note 4 – The concept could be extended to cover unwanted transverse signals affecting the operation of longitudinal receivers, with appropriate changes to the testing circuit and designation.

FIGURE 12/G.117

5 Other measurement definitions

5.1 *Common-mode rejection ratio*

This is another quantity that is appropriate to signal receivers and is measured in accordance with the principle shown in Figure 13/G.117, the input terminals being short-circuited and then energized together.



$$\text{Common-mode rejection ratio} = \left|\frac{E_L}{V_I}\right|$$

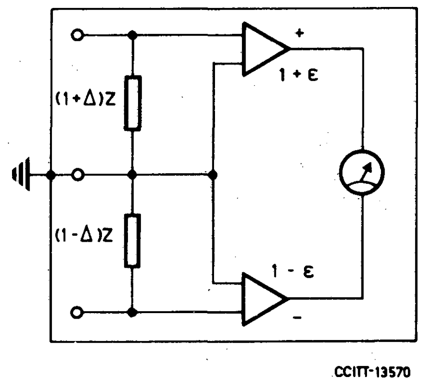
and

$$\text{Common-mode rejection} = 20 \log_{10} \left|\frac{E_L}{V_I}\right| \text{ dB.}$$

Note – V_I is the voltage indicated by the measuring set being tested.

FIGURE 13/G.117

It is clear that this measure is similar to the input longitudinal interference ratio but since there is no transverse signal (by reason of the short circuit) no longitudinal/transverse conversion mechanism within the test-object is excited. In general, there is no simple relationship between the two measures, as can be seen from the generalized measuring instrument illustrated in Figure 14/G.117, in which the input impedance is unbalanced and the gain ratios of the two halves of the differential amplifier are also slightly different. Provided the value for ε is as in Figure 14/G.117 and $\Delta \ll 1$, the various balance parameters are as indicated. This assumes the common mode rejection ratio is not twice the input longitudinal interference ratio, i.e. there is not a 6-dB difference between their decibel values.



$$\text{Common mode rejection ratio} = 2\epsilon$$

$$\text{Input longitudinal interference ratio} = \epsilon + \frac{\Delta}{2} \quad (\epsilon, \Delta \ll 1)$$

$$\text{Longitudinal impedance ratio} = 0.5 \quad (\Delta \ll 1)$$

$$\text{Longitudinal conversion ratio} = \frac{\Delta}{2} \quad (\Delta \ll 1)$$

FIGURE 14/G.117

A measuring set in which there is both a passive unbalance and an internal active unbalance

References

[1] CCITT Recommendation *Measuring arrangements to assess the degree of unbalance about earth*, Vol. IV, Rec. O.121.

[2] CCITT *Directives concerning the protection of telecommunication lines against harmful effects from electricity lines*, Chapter XVI, ITU, Geneva, 1978.

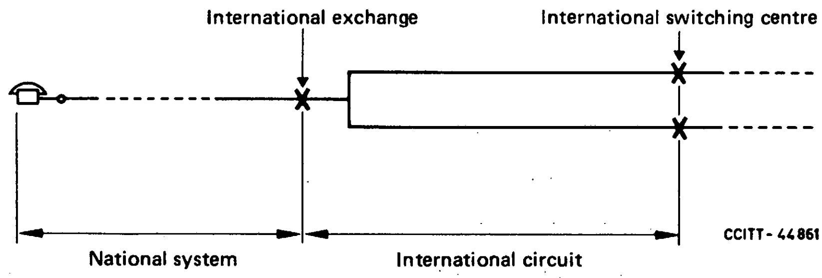
[3] CCITT Recommendation *Logarithmic quantities and units*, Vol. XIII, Rec. 574, ITU, Geneva, 1986.

[4] CCITT Recommendation *Specification for a psophometer for use on telephone-type circuits*, Vol. IV, Rec. O.41.

1.2 General characteristics of national systems forming part of international connections

The following subsection groups together the Recommendations which national systems must conform to if international communications are to be of reasonable quality.

The principles of these Recommendations also apply in cases where an international circuit is 2-wire switched at one end in an international centre. This case may arise while the CCITT transmission plan is being implemented. The figure below illustrates the arrangement.



Recommendation G.120

TRANSMISSION CHARACTERISTICS OF NATIONAL NETWORKS [1)]

1 Application of CCITT Recommendations on telephone performance to national networks

The different parts of a national network provided by both analogue and digital transmission systems to be used for an international connection should meet the following general recommendations:

1.1 The national sending and receiving systems should satisfy the limits recommended in:

- Recommendation G.121 as regards loudness rating (LR);
- Recommendation G.133 as regards group-delay distortion;
- Recommendation G.122 as regards balance return loss and transmission loss;
- Recommendation G.123 for circuit noise.

Note – Reference should also be made to Recommendations P.12 [2] and G.113.

1.2 Long-distance trunk circuits forming part of the main arteries of the national network should be high-velocity propagation circuits which enable the limits fixed in Recommendation G.114 to be respected. They should conform to Recommendations G.151 and G.152.

Loaded-cable circuits should conform to Recommendation G.124 [3] and carrier circuits to Recommendation G.123.

1.3 National trunk circuits should have characteristics enabling them to conform to Recommendations G.131, G.132 and G.134 as regards the other characteristics of the 4-wire chain constituted by the international telephone circuits and the national trunk extension circuits.

1) Former Recommendation P.21 [1].

1.4 International centres should satisfy Recommendations Q.45 [4], Q.45 *bis*, Q.551, Q.552 and Q.553.

National automatic 4-wire centres should observe the noise limits specified in Recommendation G.123, § 3.

Manual telephone trunk exchanges should satisfy Recommendation P.22 [5].

Information on the transmission performance of automatic local exchanges is given in the CCITT manual cited in [6].

2 National transmission plan

Every Administration is free to choose whatever method it considers appropriate for specifying transmission performance and to adopt the appropriate limits to ensure satisfactory quality for national calls, it being understood that in addition the Recommendation relating to loudness ratings (LRs) (Recommendation G.121) should be satisfied for international calls.

Note – To meet this twofold condition with respect to national and international calls, each Administration has to draw up a national transmission plan, i.e. it must specify limits for each part of the national network.

The manual cited in [6] contains descriptions of the transmission plans adopted by various countries and also some indications concerning the methods that can be used to establish such a plan.

In particular, Annexes A and B to Recommendation G.111 contain useful information for Administrations who wish to apply the LE method to their national connections.

References

[1] CCITT Recommendation *Application of CCITT Recommendations on telephone performance to national networks*, Red Book, Vols. V and V *bis*, Rec. P.21, ITU, Geneva, 1962 and 1965; amended at Mar del Plata, 1968, to become Rec. P.20 (G.120) *Transmission characteristics of national networks*, White Book, Vol. V (Vol. III), ITU, Geneva, 1969.

[2] CCITT Recommendation *Articulation reference equivalent (AEN)*, Yellow Book, Vol. V, Rec. P.12, ITU, Geneva, 1981.

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

[4] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45.

[5] CCITT Recommendation *Manual trunk exchanges*, Orange Book, Vol. V, Rec. P.22, ITU, Geneva, 1977.

[6] CCITT manual *Transmission planning of switched telephone networks*, ITU, Geneva, 1976.

Recommendation G.121

LOUDNESS RATINGS (LRs) OF NATIONAL SYSTEMS

Preamble

Paragraphs 1 to 5 of this Recommendation apply in general to all analogue, mixed analogue/digital and all digital international telephone connections. However, where recommendations are made on specific aspects in § 6 for mixed analogue digital or all-digital connections, § 6 will govern.

All sending and receiving LRs in this Recommendation are "nominal values" as explained in § 4 of this Recommendation and are referred to the corresponding virtual analogue switching points of an international circuit at the international switching centre unless otherwise stated.

The definition of the virtual analogue switching points of international circuits can be found in Figure 1/G.111.

The CCITT,

considering

(a) that loudness ratings (LRs) as defined in Recommendation P.76 have been determined by subjective tests described in Recommendation P.78 and that the difference between the values thus determined in various laboratories (including the CCITT Laboratory) are smaller than for Reference Equivalents;

(b) that for planning purposes, LRs are defined by objective methods as described in Recommendations P.65, P.64 and P.79;

(c) that the conversion formulae from Reference Equivalents and corrected reference equivalents (CREs) (see Annex C to Recommendation G.111) are not accurate enough to be applied to specific sets; that therefore, the Administrations who still rely on values of Reference Equivalents (determined in the past in the CCITT Laboratory) for the type of the sets they use need to find recommended values of CREs in CCITT documentation,

recommends

(1) that the values given below in terms of LR should be used by Administrations to verify that their national systems meet the general objectives resulting from Recommendation G.111,

(2) that Administrations employing CREs should preferably translate the LRs of this Recommendation into their national CREs by the methods given in Annex C to Recommendation G.111 or, as a second choice, apply the values given in Volume III of the *Red Book*.

Note 1 – The main terms used in this Recommendation are defined and/or explained in Annex A to Recommendation G.111.

Note 2 – For many telephone sets using carbon microphones, the SLR and STMR values can only be determined with limited accuracy.

1 Nominal LRs of the national systems

1.1 *Definition of nominal LRs of the national systems*

Send and Receive Loudness Ratings, SLRs and RLRs respectively, may in principle be determined at any interface in the telephone network. When specifying SLRs and RLRs of a national system, however, the interface is chosen to lie at the international exchange.

An increasing number of international systems will be connected to national systems via a *digital* interface where by definition the relative levels are 0 dBr. Therefore, in this Recommendation and in Recommendation G.111 the SLRs and RLRs of the *national systems* are referred to a *0 dBr exchange test point* at the international exchange. See Recommendation G.101, § 5. This convention is applied both for digital and analogue interconnections between the national and international systems (unless otherwise specified in particular cases).

However, the concept of "virtual analogue switching point", VASP, has also been used in the planning of all-analogue, mixed analogue-digital and digital systems. If the connection to the international circuit is made on an analogue basis the *actual* relative levels at the interface may of course be chosen by the Administration concerned. For a discussion of these matters, see Recommendation G.111, § 1.1.

In this Recommendation, values at the VASP are also given.

1.2 *Traffic-weighted mean values of the distribution of send and receive loudness ratings, SLRs and RLRs*

An objective for the mean value is necessary to ensure that satisfactory transmission is given to most subscribers. Transmission would not be satisfactory if the maximum values permitted in § 2 were consistently used for every connection.

An appropriate subdivision of the overall loudness requirement is obtained by the following long-term objectives referred to a 0 dBr international switching point.

SLR: 7 to 9 dB

RLR: 1 to 3 dB

and at the VASP

SLR: 10.5 to 12.5

RLR: −3 to −1

Note 1 – In some networks the long-term values cannot be attained at this time and appropriate short-term objectives are at 0 dBr

SLR: 7 to 15 dB

RLR: 1 to 6 dB

and at the VASP

SLR: 10.5 to 18.5 dB

RLR: −3 to 2 dB

Note 2 – In some networks the actual traffic distribution is known only incompletely. In such cases, subscribers generating heavy traffic, like PBXs, should be given special consideration.

Note 3 – The long-term traffic weighted mean values of LRs should be the same for each *main* type of subscriber categories, such as urban, suburban and rural. Only considering the mean value for the *whole* country in the transmission plan might lead to a discrimination of some important customer groups.

Note 4 – The ranges stated for SLR and RLR are for planning and do not include measuring and manufacturing tolerances.

Note 5 – Some Administrations have found it advantageous in some circumstances to include a manual volume control in the receive part of the digital telephone set. See the remarks made in Rec. G.111, § 3.2.

2 Maximum Send and Receive Loudness Ratings, SLR and RLR

2.1 *Values for each direction of transmission*

The maximum SLRs and RLRs given below in Table 1/G.121 mainly apply when the national system is predominantly analogue. When modernizing networks by digital techniques, efforts should be made to avoid having those maximum values for the national system.

TABLE 1/G.121

Nominal maximum LRs recommended for national systems

Country size [a]	No. of nat. [b] circuits in the 4-w chain	0 dBr point		VASP	
		SLR	RLR	SLR	RLR
Average	Up to 3	16.5	13	20	9
Large	4	17	13.5	20.5	9.5
Large	5	17.5	14	21	10

[a] See Recommendation G.101, § 2.2.

[b] Analogue or mixed analogue/digital.

Note – When comparing these maximum values of LRs with LRs determined for existing networks some discrepancies may be found. If the actual LRs are greater by 2 or even 3 dB this is no cause for concern. On the other hand, if a margin of 2 or 3 dB seems to appear, the permissible attenuation for subscriber lines should not automatically be increased. The first step should instead be to use the margin to improve the traffic-weighted mean values referred to in § 1.2.

2.2 *Difference in transmission loss between the two directions of transmission in national systems*

It has been found pratical to introduce a certain difference in loss between the directions 4-wire-to-2-wire and 2-wire-to-4-wire. As can be seen from Figure 1/G.121 this difference is equal to $D_o = (R - T)$ dB referred to the 0 dBr 4-wire reference points. Referred to the VASPs as in Figure 1/G.122 the difference is $D_v = (R - T - 7)$ dB. For international transmission compatibility it is desirable that Administrations choose approximately the same value of these differences. Table C-1/G.121 indicates that $R = 7$, $T = 0$ dB are the most common pad values, giving $D_o = 7$, $D_v = 0$ on the average. For planning of new networks, these are the preferred values. Thus, the difference in loss between the two directions of transmission on an international connection should not exceed 8 dB, preferably not 6 dB.

The following points should be noted:

1) Bearing in mind that most Administrations allocate the losses of their national extension circuits in much the same sort of way connections set up in practice should not exhibit differences much in excess of 3 dB.

2) As far as speech transmission is concerned, from the studies carried out by several Administrations in 1968-1972, it is clear that for connections with overall LRs falling within the range found in practice, no great disadvantage attaches to any reasonable difference in LR between the two directions of transmission.

3) When devising national transmission plans, Administrations should take into account the needs of data transmission between modems complying with the pertinent Recommendations.

3 Minimum SLR

Administrations must take care not to overload the international transmission systems if they reduce the attenuation in their national trunk network.

Provisionally a nominal minimum value of $SLR = -1.5$ dB referred to a 0 dBr point or 2 dB referred to the send virtual analogue switching point of the international circuit is recommended in order to control the peak value of the speech power applied to international transmission systems. It should be noted that the imposition of such a limit does not serve to control the long-term mean power offered to the system.

In some countries a very low sending loudness rating value may occur if unregulated telephone sets are used. Furthermore, the speech power applied to the international circuits by operators' sets must be controlled so that it does not become excessive.

4 Determination of nominal Loudness Ratings

Loudness Ratings and their properties and uses are explained in Annex A to Recommendation G.111. There it is described how a particular LR of a national system may be determined as a sum of the individual LRs of its parts. Also, rules are given for how to obtain the individual LRs of these parts, i.e. for telephone sets, subscriber lines, junctions, channel equipment, etc.

Note that Send and Receive Loudness Ratings of *analogue telephone sets* are measured under specified conditions which do not exactly correspond to those valid for a national system which is part of an international connection. The measurements are done with a terminating impedance of 600 ohms resistive and over a much wider bandwidth (100-8000 Hz or 200-4000 Hz) than the assured bandwidth of the international connection (300-3400 Hz).

Therefore, to avoid confusion, measured values of Send and Receive Loudness Ratings of *analogue* telephone sets are designated by the index "w" (for wideband). To get the proper values of SLR and RLR for *planning* international connections, 1 dB should be added to the measured values in order to compensate for bandwidth and impedance mismatch effects. Thus,

$$SLR = SLR_w + 1$$

$$RLR = RLR_w + 1$$

A *digital* telephone set, however, does not need these corrections because the codec and filters in the set limit the band anyhow.

In general, the loudness loss between *two electrical interfaces*, the Circuit Loudness Rating CLR, is equal to the corresponding difference in relative levels. (Unless an interface with a "jump" in relative level is included in the path. See § 6.3.)

"Nominal value" here signifies a "reasonable engineering average" for typical conditions as exemplified in what follows, excluding "worst cases".

With regard to circuits and other items of equipment, variations with time, temperature etc. are not included in the nominal CLRs, Circuit Loudness Ratings.

For telephone sets, most Administrations today have to accept a large variety of types which comply with some national specification having rather wide limits. The requirements for SLR and RLR usually refer to a measuring setup with a variable artificial line terminated by a feeding bridge and a nominal impedance which may be complex or, most often, 600 ohms.

The specification is often drawn up in the form of upper and lower limits for SLR_w and RLR_w as functions of line length (or possibly line current). The "nominal" SLR_w and RLR_w of telephone set plus subscriber line may then be interpreted as the arithmetic mean between the upper and lower limit curves.

In practice, the subjective quality impression of the overall loudness changes rather insignificantly for fairly large variations of OLR around the optimum value and it is unlikely that sets with worst possible LRs are associated with limiting line lengths. Therefore, rather wide manufacturing tolerances, commonly about ± 3 dB, can be accepted for the individual set SLR (set) and RLR (set). (SLR (set) and RLR (set) refer to set measurements without the subscriber line but as function of line current, including the 1 dB bandwidth correction.)

Note however, that the *sum* of SLR (set) + RLR (set) for an individual 2-wire telephone set must be controlled more carefully so that is does not decrease below a certain minimum value. The reason is that, under certain circumstances, subscribers react very unfavourably to strong sidetone and talker echo. Both effects depend directly on this LR sum in addition to the unavoidable network impedance variations. This minimum limit is often translated into a minimum limit for STMR as measured against a specified impedance. See § 5 for a discussion.

5 Sidetone

5.1 *General*

Especially for those connections approaching the limits for high Loudness Ratings and/or noise, further transmission impairments should be avoided. One important precaution is to ensure that an adequate *sidetone* performance is maintained for the various circuit combinations occurring in the telephone system. ("Adequate" is in most cases to be interpreted as a sufficiently high sidetone loss.)

For 2-wire telephone sets, the sidetone performance is basically dependent on set sensitivity and impedance variation limits as explained in Annex A to Recommendation G.111. Thus, a national transmission plan should not only give rules for allocation of losses in the network but also provide an appropriate impedance strategy to follow. (An example is given in Supplement No. 10 of Vol. VI.)

Note that for sidetone evaluations one has to consider the line impedance "seen" by the 2-wire telephone set in the actual, *complete* connection. In modern system configurations this impedance cannot always be simulated by an artificial line terminated by a simple R-C network. Either one has to use a more elaborate measuring setup or resort to computations from known data of the circuits involved. (A number of computer programs exists which can be employed for such purposes.)

Of special interest is the fact that a 4-wire link inserted in a 2-wire connection may cause large impedance variations. As this is a common network practice – for instance digital exchanges – a simplified calculation method is discussed in Annex B.

Ideally, a 2-wire telephone set could be designed to have an adaptive sidetone balancing function, thus widening the acceptable range of line impedances. Such costly techniques are very exceptional, however, and should not be prescribed for the "standard" sets to be used in the network. A possible, cheaper alternative is to design a set with a Z_{so} varying in a predetermined manner with the line feeding current. (Z_{so} = equivalent sidetone balance impedance.) However, the best strategy is to control the impedances in the network. Thus, the use of complex nominal input impedances to exchanges is tending rather to reduce the range of impedances seen from the set.

Digital telephone sets are of course connected 4-wire to the digital network and thus there exists no near-end impedance mis-match to produce a sidetone effect. Instead, a small, internal feedback from send to receive is introduced. For judging the overall transmission quality the far-end effects have to be considered. However, those effects caused by impedance mis-matches and/or acoustic echoes can have a substantial influence.

Under some difficult transmission circumstances, analog telephone sets are also 4-wire connected to the network. This applies for (analog) mobile and maritime services and, in the past, for some exceptionally large, private networks.

5.2 *Talker's sidetone STMR*

STMR, the sidetone masking rating, is explained in Annex A.1 to Recommendation G.111. How to determine STMR is described in Annexes A.3 and A.4 to Recommendation G.111. See also Annex B to Recommendation G.121 and Recommendations P.76 and P.79.

In a face-to-face conversation there is a certain airpath feedback from the talker's mouth to his ear, partly via room reflexions. Using the handset in a telephone conversation the electric sidetone path should provide about the same feedback, the acceptable range being rather large. Unfortunately, in many present 2-wire connections the impedance deviations from the ideal are so large that the electric sidetone feedback becomes too strong, i.e. STMR too low. This causes the speaker to lower his voice and/or move the earphone away from his ear, thus impairing the acoustic transmission quality.

The following values are given as a guide for transmission planning.

For 2-wire telephone sets:

$STMR$ = 7 – 12 dB:	Preferred range.
$STMR$ = 20 dB:	Upper limit, above which the connection feels dead.
$STMR$ = 3 dB:	Lower limit, acceptable only for low-loss connections, i.e. low OLR.
$STMR$ = 1 dB:	Lowest (short-term) limit for exceptional cases, such as very short subscriber lines.

For digital (4-wire) telephone sets:

$STMR$ = 15 ± 5 dB:	Preferred range for near-end, introduced sidetone (far-end effects disregarded).

Note 1 – When $STMR$ = 7 or 8 dB, this corresponds to the average acoustic loss from the talker's mouth to his ear via the electric sidetone path being about 0 dB in typical cases.

Note 2 – STMR has to be determined for the *complete* connection. (See the comments made in § 5.1.)

Note 3 – In the presence of high room noise, requirements on LSTR may be the controlling factor.

Note 4 – If the reflected electric signal has a noticeable delay it is interpreted as an echo rather than sidetone, which means it needs more suppression to avoid subscriber dissatisfaction. See Recommendations G.122 and G.131. (Recent investigations indicate that at a delay of 2-4 ms, the echo begins to be clearly distinguishable from even a strong "normal" sidetone.) The problem is under study in Question 9/XII.

5.3 *Listener's sidetone LSTR*

LSTR, the listener's sidetone rating is defined in Annex A.1 to Recommendation G.111. How to determine LSTR is described in Annexes A.3 and A.4 to Recommendation G.111.

The presence of a listener's sidetone means that room noise is picked up by the handset microphone and transmitted to the handset ear via the electric sidetone path. LSTR is a measure of how well this room noise sidetone is suppressed. Too low values of LSTR means that the room noise will be *amplified* at the handset ear. This is obviously very disturbing for subscribers in noisy environments, especially for high-loss connections.

Note – High noise gives the impression of lower received speech levels.

For a particular telephone set there is a fixed relation between the talker's and the listener's sidetone, STMR and LSTR respectively. For sets with linear microphones LSTR is typically between 1.5 and 4 dB higher than STMR, independent of the noise level. For carbon microphone sets the difference is dependent on the room noise level, a threshold effect being noticeable. For 60 dB(A) room noise (Hoth-type) the difference is in the order of 6 to 8 dB. (For other noise levels and some handset designs the difference can be as high as 15 dB.)

In general, subscribers prefer sets with linear microphones because the sound quality is much superior. However, when replacing old carbon microphone sets in noisy environments with modern linear sets, care must be taken to ensure that the LSTR-value is sufficiently high. (However, some linear microphone sets do include a noise threshold function.)

The following value should be striven for in modern telephone systems:

$$LSTR > 13 \text{ dB}$$

Note 1 – $LSTR$ = 13 dB corresponds approximately to that of the earcap of the handset functions as a shield for the room noise with an average attenuation of 5 or 6 dB. (For the higher frequencies; the lower frequencies leak past the earcap.)

Note 2 – LSTR has to be determined for the *complete* connection. (See the comments made in § 5.1.)

6 Incorporation of PCM digital processes in national extensions

6.1 *Effect on national transmission plans*

The incorporation of PCM digital processes into national extensions might require that existing national transmission plans be amended or replaced with new ones.

The national transmission plans to be adopted should be compatible with existing national analogue transmission plans and also capable of providing for mixed analogue/digital operation. In addition, the plans should be capable of providing for a smooth transition to all-digital operation.

Thus, the transmission planning of transitional phases should preferably not involve any degradation of the quality previously experienced.

6.2 *Transmission loss considerations*

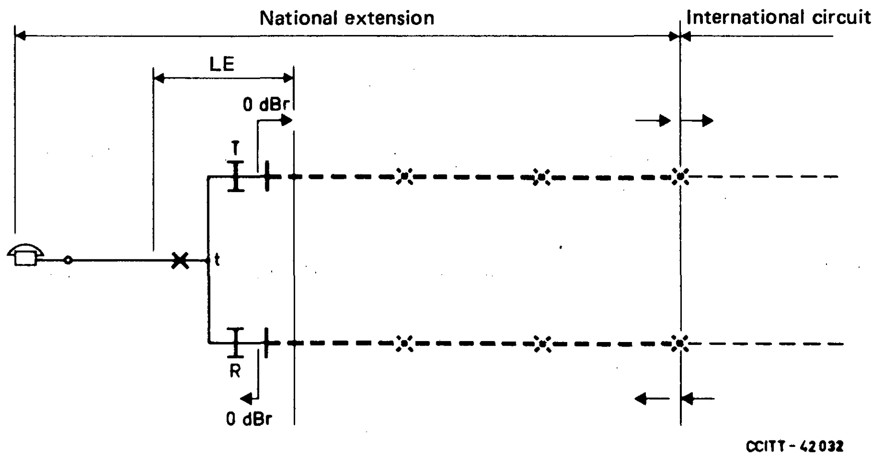
Where the national portion of the 4-wire chain is wholly digital between the local exchange and the international exchange, the transmission loss which the extension must contribute to the maintenance of stability and the control of echo on an international connection can be introduced at the local exchange. The manner in which the required loss should be introduced is to be governed by the national transmission plan adopted. Three of possibly many different configurations of such national extensions are shown in Figure 1/G.121.

In case 1 and 2 of Figure 1/G.121, the R pad represents the transmission loss between the 0 dBr point at the digital/analogue decoder and the 2-wire side of the 2-wire/4-wire terminating unit. Similarly, the T pad represents the transmission loss between the 2-wire side of the 2-wire/4-wire terminating unit and the 0 dBr point at the analogue/digital coder. In practice there can be levels other than 0 dBr and hence consequential changes in the R and T pad-values.

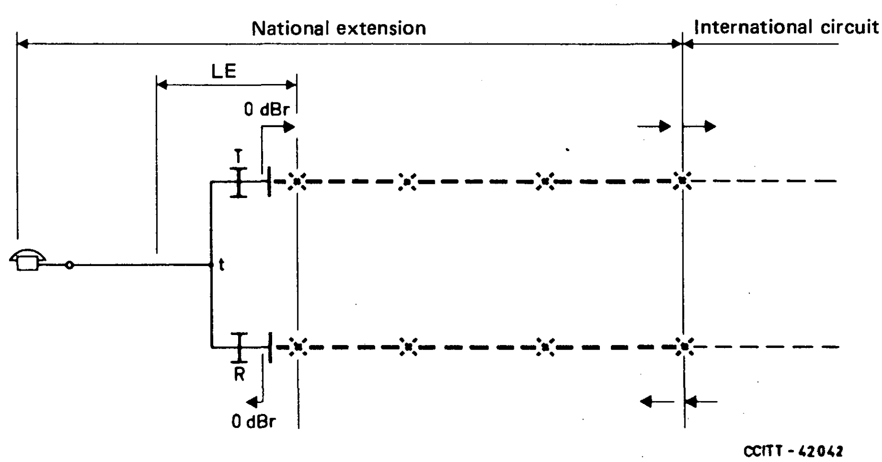
The individual values of R and T can be chosen to cater for the national losses and levels, provided that the CCITT Recommendations for international connections are always met. It is recognized that for evolving networks, the values of R and T may not be the same as the values appropriate to the all digital 4-wire national chain. However, for the case of an all-digital national chain, the choice of values of R and T is particularly important in determining the performance in respect of echo and stability. For example, if the balance return loss at the 2-wire/4-wire terminating unit can approach 0 dB under worst case terminating conditions, then the sum of R and T needs to be at least so high that the requirements of Recommendation G.122 are to be met. Examples of the values for R and T that have been adopted by some Administrations are given in Annex C to Recommendation G.121.

In case 2 of Figure 1/G.121, it is possible with a sufficiently high balance return loss to comply with the Recommendations concerning loudness ratings, stability, and echo without requiring a particular value for the sum of the R and T pad values. However it will still be necessary to comply with the provisions concerning differential loss (§ 6.4 of this Recommendation) which in turn implies that

$$R - T = 3 \text{ to } 9 \text{ dB}$$

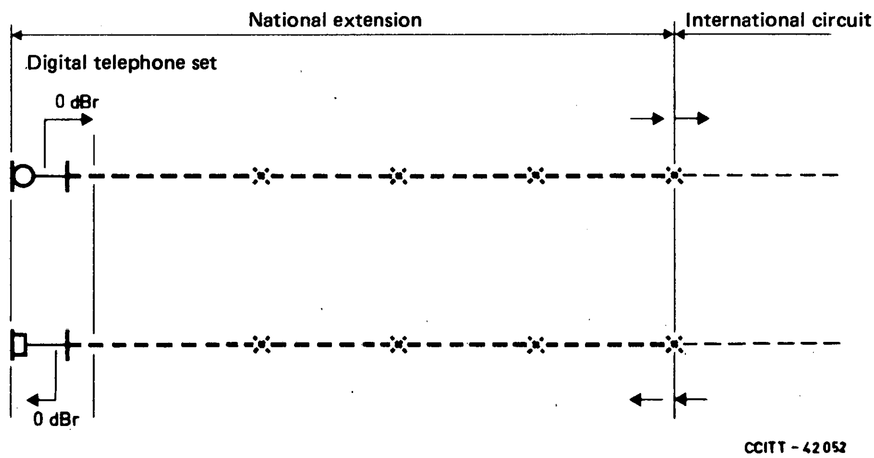


a) Case 1 – Two-wire analogue switching at local exchange and 2-wire analogue subscriber line



Note – No 2-wire switch point between the subscriber's local line and the terminating unit at the local exchange.

b) Case 2 – Four-wire digital switching at the local exchange but 2-wire analogue subscriber lines



c) Case 3 – Four-wire switching at the local exchange, 4-wire digital subscriber line and digital telephone set

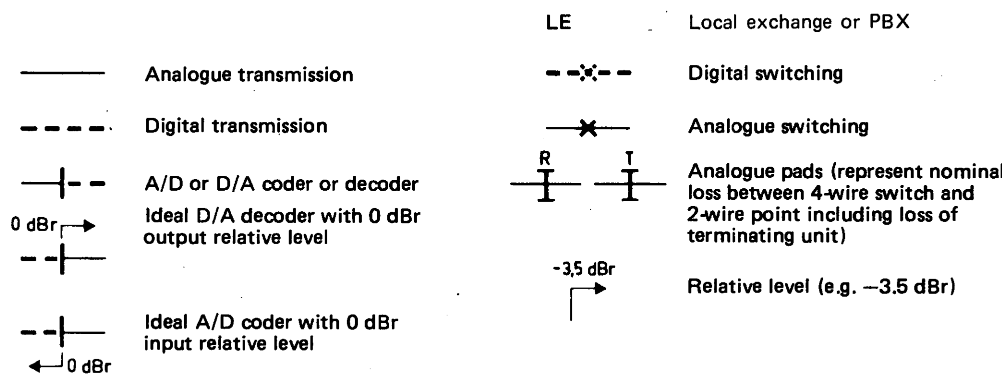


FIGURE 1/G.121

Examples of national extensions in which the digital 4-wire chain extends to a 4-wire local exchange

However, a local exchange designed on these principles and which is at the end of a national extension containing asymmetric analogue portions cannot take the whole of the asymmetry allowance.

The R and T pads shown in Figure 1/G.121 are also shown as analogue pads. This type of pad might not necessarily be introduced under all conditions. In some situations it might be more practical to introduce the required loss at the local exchange, or at some other point of the national extension, by means of digital pads. However, if digital pads are used, their detrimental effect on digital data or other services requiring end-to-end bit integrity must be taken into account as indicated in Recommendation G.101, § 4.4 and G.103, § 4.

For speech, the quantizing distortion will increase. See Recommendation G.113, § 4. The concept of relative levels is also affected by a digital pad. See § 6.3.

The arrangement in case 3 of Figure 1/G.121 assumes 4-wire digital switching at the local exchange in combination with a 4-wire digital local line and a 4-wire "digital telephone set".

Stability and echo on international connections are governed by Recommendation G.122.

6.3 *The designation of relative levels and digital pads*

"Relative level" (expressed in dBr) is a useful concept in transmission planning by which one can determine gain or loss between points in a system as well as signal handling requirements for transmission equipment. The general definitions are found in Recommendation G.101. To clarify further the use of relative levels in Recommendations G.111 and G.121 some special aspects will be discussed here.

The relative level at a point of a circuit is in principle determined by comparison with the "transmission reference point", TRP, for that circuit, a *hypothetical* point used as the zero relative level point. Such a point exists at the sending end of each channel of a 4-wire switched circuit preceding the international exchange.

When the international connection is *digital* by means of a conventional PCM system, the transmission reference point is equal to the digital exchange test point i.e. the digital bit stream is associated with a relative level of 0 dBr. The power handling capacity of the digital bit stream is interpreted as the clipping level of a sinusoidal signal when introduced via an ideal codec: +3.14 dBm for the A-law, +3.17 for the μ-law (see Recommendation G.101, §§ 5.3.2.4 to 5.3.3.2).

When the international connection is established by an *analogue* (FDM) system, the transmission system would be designed to handle a power load of -15 dBm per channel at the transmission reference point if this existed in physical form. Thus, when the transmission system has a (nominal) power handling capacity of $(-15 + S)$ dBm at the actual international interconnection point the relative level at that point is $+S$ dBr.

In normal network situations, the relative level at a certain point is numerically equal to the "composite gain" between that point and the transmission reference point for the circuit concerned at the reference frequency 1020 Hz. For instance, for analogue international connections the sending relative level at VASP, the virtual analogue switching point, is -3.5 dBr (by definition). The loss of the international circuit is 0.5 (as recommended by the CCITT) and thus the relative level at the receive VASP in the other country is -4 dBr.

Likewise, in normal network cases, circuits are interconnected with matching power handling capabilities.

Thus digital (PCM) bit streams not subjected to digital gain or loss are always associated with a relative level of 0 dBr.

In some exceptional cases however, the rules relating relative level to "composite loss" and "power handling capacity" do not apply exactly. For practical reasons some types of interfaces will have "jumps" in relative levels because two (or more) different transmission reference points occur in tandem.

One example is digital gain or loss introduced in the send direction. Following the definition given in Recommendation G.101, § 5.3.2.6 there will be a jump in relative level as illustrated in Figure 2/G.121 at point B. The loss between points A and B is T dB but the difference in relative level is 0 dB.

Another example is to be found in certain international connections which include several 4-wire (analogue or mixed analogue-digital) systems in cascade between the VASPs. If there are no such circuits, for stability reasons the loss is then made equal to $n \cdot 0.5$ dB.

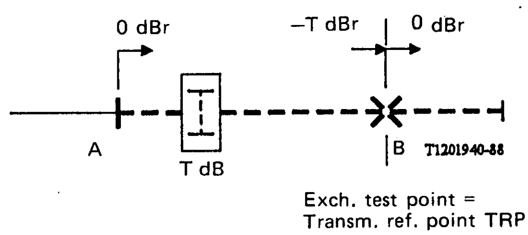


FIGURE 2/G.121

Exemple of a jump in relative level at an interface. (Point B)

Note 1 – The "power handling capacity" refers to a *nominal* load, not to the *actual* load which the system is subjected to. For instance, for an analogue system at the TRP the nominal load of −15 dBm corresponds to 0.032 mW of which 0.010 mW is considered to originate from signalling and tones, 0.022 mW from speech, carrier leaks and voice telegraphy. The nominal speech load at the TRP thus is −16.6 dBm taken as an average with time from a batch of channels during a busy hour. The actual average speech level may very well differ from this value. This is of course even more probable for an individual channel. (However, the aim should always be for the actual load to be close to the nominal load for which the transmissions system gives optimum performance.)

Note 2 – For many reasons, digital gain or loss should be used only exceptionally in a network.

Note 3 – If digital gain or loss is introduced the firm relations between relative level and power handling capacity may be lost. For instance, in an arrangement in accordance with Figure 2/G.121 the actual possible maximum peak level to the right of point B (i.e. at 0 dBr) will be T dB lower than + 3.14 dBm. Likewise, to the left of point B (i.e. at $-T$ dBr) the noise threshold level will be T dB higher than in a normal PCM system.

ANNEX A

(to Recommendation G.121)

Evaluation of the nominal differences of loss between the two directions of transmission

A.1 Consider an international connection between primary centres in two Administrations, established over one international circuit as shown in Figure A-1/G.121.

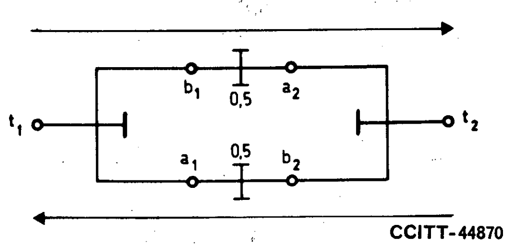


FIGURE A-1/G.121

The nominal overall losses in each of the two directions of transmission are:

$$1 \rightarrow 2 = t_1b_1 + 0.5 + a_2t_2 \quad \text{(dB)}$$

and

$$2 \rightarrow 1 = t_2b_2 + 0.5 + a_1t_1 \quad \text{(dB)}$$

where a and b are defined as in Recommendation G.122, so that the difference between the two directions is:

$$(t_1b_1 - a_1t_1) - (t_2b_2 - a_2t_2) = d_1 - d_2$$

in which d signifies $d_1 = t_1b_1 - a_1t_1$ or $d_2 = t_2b_2 - a_2t_2$.

Note – As long as the 2-wire nominal impedance are resistive there is no problem in defining "loss". The modern trend is toward using complex nominal impedances, however, and then some conventions have to be observed. In Recommendation Q.551, § 1.2.3 – § 1.2.5 is prescribed how to measure digital exchanges with analogue parts. In short, the rules are:

a) The equipment (circuit) is measured under nominally matched impedance conditions for the analogue ports. During the measurements, the 4-wire loop must be broken in the return direction. (In practice, this means *either* between two physical impedances as is the case for 600 ohms measurements *or* between a low-impedance generator and a high-impedance indicator. Either method can be used, depending on what is most practical. The measurement results do not differ very much.) Note when the second method is used, a 6 dB correction must be applied.

b) The nominal loss is the composite loss at the reference frequency 1020 Hz (i.e. the voltage loss corrected by 10 times the logarithm of the impedance ratio).

c) The attenuation distortion as a function of the frequency f is 20 times the logarithm of the ratio of the voltage at 1020 Hz to the voltage at f.

ANNEX B

(to Recommendation G.121)

Transmission considerations for a 4-wire loop inserted in a 2-wire circuit

B.1 *General*

A 4-wire loop normally exhibits a considerable change of phase as a function of frequency. Thus, it may have a large influence on the attenuation distortion and the impedances when inserted in a 2-wire circuit because of the reflexions encountered. In what follows exact expressions will be given for loss and impedance together with an approximate rule useful for estimating certain sidetone effects.

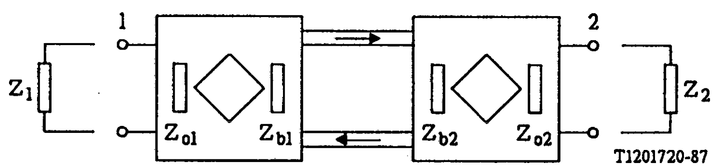


FIGURE B-1/G.121

A 4-wire loop inserted in a 2-wire connection

In Figure B-1/G.121 is shown a 4-wire loop with 2-wire ports Nos. 1 and 2. The following designations are used.

Terminating impedances: Z_1 and Z_2.

2-wire input impedances (4-wire loop open): Z_{o1} and Z_{o2}.

Balance impedances: Z_{b1} and Z_{b2}.

Loss and phase shift under matched load conditions, i.e. $Z_1 = Z_{o1}$ and $Z_2 = Z_{o2}$;

from port 1 to port 2 (4-wire loop open from port 2 to 1): L_1 dB, B_1 deg;

from port 2 to port 1 (4-wire loop open from port 1 to 2): L_2 dB, B_2 deg.

We now define the following (complex) factors:

$$C_1 = 10^{-L_1/20} \cdot (\cos B_1 - j \sin B_1)$$

$$C_2 = 10^{-L_2/20} \cdot (\cos B_2 - j \sin B_2)$$

$$C_{r1} = \frac{2Z_{o1}}{Z_{o1} + Z_{b1}} \cdot \frac{Z_1 - Z_{b1}}{Z_1 + Z_{o1}}$$

$$C_{r2} = \frac{2Z_{o2}}{Z_{o2} + Z_{b2}} \cdot \frac{Z_2 - Z_{b2}}{Z_2 + Z_{o2}} \qquad \text{(B-1)}$$

$$C_{b1} = \frac{Z_{o1} - Z_{b1}}{Z_{o1} + Z_{b1}}$$

$$C_{b2} = \frac{Z_{o2} - Z_{b2}}{Z_{o2} + Z_{b2}}$$

The balance return losses at port 1 and 2 are:

$$L_{br1} = -20 \log_{10} \left| C_{r1} \right|; \; L_{br2} = -20 \log_{10} \left| C_{r2} \right| \qquad \text{(B-2)}$$

Note that the balance return losses may become *negative* for some terminations. Therefore, a few comments will be given on this aspect as some peculiar circuit configurations can be encountered during the setup of a call.

The minimum balance return loss at a port with (2-wire) input impedance Z_o and balance impedance Z_b occurs when the terminating impedance is a *pure reactance*, the value of which depends on Z_o and Z_b. (Thus in general, neither the open- or the short-circuit condition!)

The minimum balance return loss value is:

$$(L_{br})_{min} = -20 \log_{10} \left\{ \frac{1}{\cos V} + \sqrt{(1 - S)^2 + (\tan V - T)^2} \right\} \qquad \text{(B-3)}$$

where

$$\left. \begin{aligned} V &= \text{phase angle of } (Z_o) \\ S + jT &= \frac{2Z_o}{Z_o + Z_b} \end{aligned} \right\} \qquad \text{(B-4)}$$

A case of special interest is when by design Z_o is made identical with Z_b. Then Equation (B-4) transforms into:

$$\begin{aligned} &(L_{br})_{min} = -20 \log_{10} \left\{ \tan \tfrac{1}{2} (90^\circ - |V|) \right\} \\ &(Z_o = Z_b) \end{aligned} \qquad \text{(B-5)}$$

This minimum occurs when the terminating impedance is a pure reactance jX of *opposite* sign to the reactance of Z_o and has the value:

$$\left| X \right| = \left| Z_o \right| \qquad \text{(B-6)}$$

Note 1 – In general, the more reactive Z_o and Z_b are, the lower will the minimum balance return loss be when unfortunate terminations are met within the network. For instance, if Z_o and Z_b would be exactly matched to the unloaded subscriber cable characteristic impedance angle of $-45°$, $(L_{br})_{min}$, equals -7.7 dB. Thus, extremely reactive values of Z_o and Z_b should be avoided.

Note 2 – For *normal* cases encountered in the network the terminations, as well as the balancing networks, most often have a negative reactive component. The balance return loss and the return loss also do not differ very much numerically.

Note 3 – In many practical cases open- and short-circuit conditions represent "worst cases".

B.2 *Attenuation*

According to the CCITT convention for loss with complex, nominal impedances, the loss from port 1 to port 2 with the 4-wire loop closed is

$$L_{12} = L_1 + 20 \log_{10} \left| \frac{Z_2\,(1.02\text{ kHz})}{Z_1\,(1.02\text{ kHz})} \right| + 20 \log_{10} \left| \frac{Z_{o1} + Z_1}{2Z_{o1}} \right| + \\ + 20 \log_{10} \left| \frac{Z_{o2} + Z_2}{2Z_2} \right| + 20 \log_{10} \left| 1 - C_1 \cdot C_2 \cdot C_{r1} \cdot C_{r2} \right| \qquad \text{(B-7)}$$

The sum of the first four terms represents the loss which would be measured with the 4-wire loop broken in the return direction from port 2 to port 1. The second term is a correction for the terminating impedances being unequal. (Assuming Z_1 and Z_2 are the nominal, reference impedances.) The third and fourth terms represent mis-match effects.

Finally, the fifth term shows the ripple effects due to loop phase shift and non-perfect balancing at the ports, i.e. Z_{b1} not being equal to Z_1 and Z_{b2} not to Z_2.

B.3 *Impedance*

When the 4-wire loop is closed the input impedance at port 1 is:

$$Z_{in1} = Z_{o1} \frac{(Z_{o1} + Z_{b1}) + 2Z_{b1} \cdot C_1 \cdot C_2 \cdot C_{r2}}{(Z_{o1} + Z_{b1}) - 2Z_{o1} \cdot C_1 \cdot C_2 \cdot C_{r2}} \qquad \text{(B-8)}$$

A measure of the deviation of Z_{in1} from the nominal 2-wire input impedance Z_{o1} can be had from the return loss:

$$L_{r1} = 20 \log_{10} \left| \frac{Z_{in1} + Z_{o1}}{Z_{in1} - Z_{o1}} \right| \qquad \text{(B-9)}$$

Using Eq. (B – 8) we get

$$L_{r1} = L_1 + L_2 + L_{br2} + 20 \log_{10} \left| 1 - C_1 \cdot C_2 \cdot C_{b1} \cdot C_{r2} \right| \qquad \text{(B-10)}$$

Note 1 – The last term in Equation (B-10) represents a (high-periodicity) ripple. However, often it is not very large. If $Z_o = Z_b$ it is zero!

Note 2 – If the loop loss $(L_1 + L_2)$ is low, the effective input impedance at one port can be appreciably affected by conditions at the other.

B.4 *Sidetone considerations*

Sidetone effects can be most critical for subscribers very close to a digital exchange, i.e. with zero line length. Therefore, we will here study this case in some detail.

If a subscriber is connected directly to port 1 in Figure B-1/G.121, Equation (B-8) can be used to compute the impedance Z the telephone set sees at its terminals. Then the sidetone balance return loss A_{rst} and its weighted mean value A_m is calculated as is shown in Annex A.4.3 to Recommendation G.111, using the telephone set input impedance Z_c and its equivalent sidetone balance impedance Z_{so}. Finally, the talker's and the listener's sidetones, STMR and STLR respectively, are obtained using the value of A_m in Equation (A.4-3) in Annex A to Recommendation G.111.

The procedure just described is somewhat tedious as it involves the exact computation of the 2-wire impedance of the closed 4-wire loop. To give a rapid indication of the magnitude of sidetone effects the following simplified method can be used.

The sidetone mis-match effects are considered as the superposition of two "echo" effects, namely:

a) The sidetone balance return loss A_{rst1} between the telephone set and the *nominal* input impedance Z_{o1} of the (near-end) port to which the set is connected. The weighted mean value A_{m1} is computed using Equation (A.4-3) in Annex A to Recommendation G.111.

b) The far-end port impedance mis-balancing translated to the near-end part i.e. the return loss L_{r1} as given by Equation (C-10)[1] is used to compute a mean value A_{m2} by means of Equation (A.4-3) in Annex A to Recommendation G.111.

Finally, the two "sidetone echoes" are added on a power basis to give a new weighted mean value:

$$A_m = -10 \log_{10} \left\{ 10^{\frac{-A_{m1}}{10}} + 10^{\frac{-A_{m2}}{10}} \right\}$$

Note – The far-end impedance mis-match effects will of course be interpreted not as a sidetone but as an echo if the round trip delay is long. The change from sidetone to echo perception might begin at a delay of about a few milliseconds. (This problem is under study in Question 9/XII.) Long-delay echoes are far more noticeable than sidetone.

ANNEX C

(to Recommendation G.121)

Examples of values of R and T pads adopted by some Administrations

This annex gives the values of R and T pads that have been adopted by some Administrations for their digital networks. The values given are those appropriate for digital connections between subscribers with existing analogue 2-wire subscriber lines on digital local exchanges. It is recognized that different values may be appropriate for connections in the evolving mixed analogue/digital network.

These values are given as guidance to developing countries who are considering the planning of new networks. If similar values are adopted for new networks then, in association with adequate echo and stability balance return losses, there are unlikely to be difficulties in meeting the requirements of Recommendation G.122.

[1] Ignoring the last term.

Some Administrations consider losses in terms of the input and output relative levels. These values can be derived from Table C-1/G.121 by using the relationship given in Figure C-1/G.121.

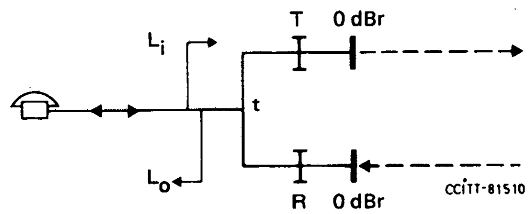


FIGURE C-1/C.121

Relation between relative levels and R- and T-pads

In this circuit, it is assumed that the relative levels of the encoder input and the decoder output are 0 dBr, that the T-pad represent all the loss between the 2-wire point, t, and the ecoder input, and that the R-pad represents all the loss between the decoder output and t. Accordingly, the relation between relative levels and losses is:

$$L_i = T, L_o = -R$$

Note – The modern trend is to use a complex nominal impedance at the 2-wire port. See the note in Annex A.1 for how "loss" should be interpreted in such a case.

In exceptional cases, some of the R and T losses may be achieved by digital pads. See § 6.2 and § 6.3 for a discussion.

In general, the range of input levels has been derived assuming that speech powers in the network are close to the conventional load assumed in the design of FDM systems. However, actual measurements reveal that this load is not being attained (see Supplement No. 5 to Fascicle III.2 of the *Yellow Book*). For this reason, it may be that there is some advantage in adopting different input (and output) levels for future designs of exchange. However, any possible changes need to take into account:

i) the range of speech powers encountered on an individual channel at the exchange input and the subjective effects of any peak clipping, noting that any impairment is confined to that channel;

ii) levels of non-speech analogue signals (e.g. from data modems or multifrequency signalling devices) particularly from customers on short exchange lines;

iii) the need to meet the echo and stability requirements of Recommendation G.122, particularly when the sum of R and T is less than 6 dB;

iv) the need to consider the difference in loss between the two directions of transmission, as required by § 6.3 of Recommendation G.121.

At this stage Administrations should note that there may be some advantage in considering a range of level adjustment for future designs of digital local exchange.

TABLE C-1/G.121

Values of R and T for various countries

	Connection type					
	Own exchange		Local via digital junctions (digital trunks)		Trunk via digital trunk exchange	
	R dB	T dB	R dB	T dB	R dB	T dB
Germany (F.R.) (For subscribers on short lines: R = 10 dB, T = 3 dB)	7	0	7	0	7	0
Australia	6	0	6	0	6	0
Austria	7	0	7	0	7	0
Belgium	7	0	7	0	7	0
Canada	0	0	3	0	6	0
Denmark	6	0	6	0	6	0
Spain	7	0	7	0	7	0
United States	0	0	3	0	6	0
Finland	7	0	7	0	7	0
France	7	0	(not used)	(not used)	7	0
India	6	0	6	0	6	0
Italy	7	0	7	0	7	0
Japan	4	0	8	0	8	0
The Netherlands	4.5	1.5	4.5	1.5	4.5 (National) 10.5 (International)	1.5
Norway	5	2	5	2	5	2
United Kingdom (Values shown are for median lines; additional loss is introduced on short local lines in both directions of transmission)	6	1	6	1	6	1
Sweden	5	0	5	0	5 (National) 7 (International)	0 (National) 0 (International)
USSR	7	0	7	0	7	0
Yugoslavia	7	0	7	0	7	0
New Zealand	7	0.5	7	0.5	7	0.5

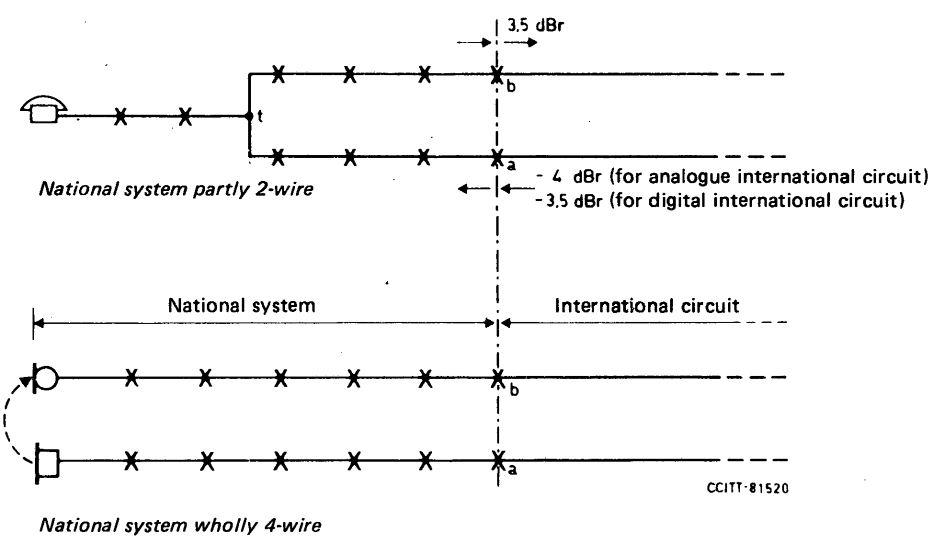
INFLUENCE OF NATIONAL SYSTEMS ON STABILITY, TALKER ECHO, AND LISTENER ECHO IN INTERNATIONAL CONNECTIONS

(Geneva, 1964; amended at Mar del Plata, 1968, Geneva, 1972, 1976 and 1980; Malaga-Torremolinos, 1984 and Melbourne, 1988)

1 Introduction

The information provided in this Recommendation applies to all national systems.

Representations of a national system which extend up to the virtual analogue switching points are shown in Figure 1/G.122.



Note – a, b are the virtual analogue switching points of the international circuit.

FIGURE 1/G.122

National system representation and virtual analogue switching point definition

The transmission loss introduced between *a* and *b* by the national system, referred to as the loss (*a-b*), is important from three points of view:

a) it contributes to the margin that the international connection has against oscillation during the setting-up and clearing-down of the connection. A minimum loss over the band 0-4 kHz is the characteristic value;

b) it contributes to the margin of stability during a communication. Again, a minimum loss in the band 0-4 kHz is the characteristic value, but in this case the subscribers' apparatuses (telephone, modem, etc.) are assumed to be connected and in the operating condition;

c) it contributes to the control of echoes and, in respect of the subjective effect of talker echo, a weighted sum of the loss (*a-b*) over the band 300-3400 Hz is the characteristic value.

In addition, echoes circulating in any of the 4-wire loops in the national system or in the international 4-wire chain, give rise to listener echo, which can affect voice-band analogue data transmission.

The requirements stated in this Recommendation represent network performance objectives.

2 Loss (a-b) to avoid instability during set-up, clear-down and changes in a connection

2.1 Instability should be avoided during all normal conditions of set-up, clear-down and other changes in the composition (e.g. call-transfer) of a complete connection. To ensure adequate stability of international connections the distribution (taken over many actual calls) of the loss (*a-b*) during the worst situation should be such that the risk of a loss (*a-b*) of 0 dB or less does not exceed 6 in 1000 calls when using the calculation method applied in § 2.2. This requirement should be observed at any frequency in the band 0 to 4 kHz.

Note 1 – The signalling and switching systems have an influence on the loss (*a-b*) under set-up and clear-down conditions. For example, in some systems 4-wire registers control the set-up and do not establish the 4-wire path until the answer signal is successfully received. In others, circuits are released immediately if busy conditions are encountered. In these circumstances the risk of oscillation does not arise.

Note 2 – Recommendation Q.32 gives information on methods of securing an adequate loss (*a-b*) of an incoming national system before the called-subscriber answers (i.e. while ringing tone is transmitted) or if busy or number unobtainable conditions are encountered.

Note 3 – If there are no such arrangements as described in Notes 1 or 2 above then in general it would be safe to assume that there is no balance return loss provided by the called local telephone circuit (if 2-wire). In this case the necessary loss (*a-b*) must be provided by the transmission losses in the national system.

Note 4 – The stability of international telephone connections at frequencies outside the band of effectively transmitted frequencies (i.e. below 300 Hz and above 3400 Hz) is governed by the following transmission losses at the frequencies of interest:

- the balance return loss at the terminating units;
- the transmission losses of the terminating units;
- the transmission losses of the 4-wire circuits.

Note 5 – Conditions which only last for a few tens of milliseconds can be left out of consideration because in such a short time oscillations cannot build up to a significant level.

2.2 The limit recommended in § 2.1 may be met, for instance, by imposing the following simultaneous conditions on the national network:

1) The sum of the nominal transmission losses in both directions of transmission *a-b* and *t-b* measured between the 2-wire input of the terminating set *t*, and one or other of the virtual switching points on the international circuit, *a* or *b* should not be less than $(4 + n)$ dB, where n is the number of analogue or mixed analogue-digital 4-wire circuits in the national chain.
2) The stability balance return loss at the terminating set *t*, should have a value not less than 2 dB for the terminal conditions encountered during normal operation.
3) The standard deviation of variations of transmission loss of a circuit should not exceed 1 dB (see Recommendation G.151, § 3).

In a calculation to verify if these values are acceptable, it may be assumed that (see [1]):

- there is no significant difference between nominal and mean value of the transmission losses of circuits;
- variations of losses for both directions of transmission of the same circuit are fully correlated;
- distributions are Gaussian.

For the loss (a-b), it then results:

Mean value: $2 + 4 + n = 6 + n$ dB

Standard deviation: $\sqrt{4n}$ dB

With $n = 4$ the mean value becomes 10 dB and the standard deviation 4 dB, resulting in a probability for values lower than 0 dB of $6 \cdot 10^{-3}$.

Note – There is no need for the two quantities a-t and t-b to be equal, so that differential gain can be used in the national network. This practice may be needed to meet the requirements of Recommendation G.121, § 2, but it implies that the transmission loss in terminal service of the 4-wire chain plus the terminating sets may be different according to the direction of transmission. The choice of the nominal value of the transmission loss t-b should in all cases be made with an eye to Recommendation G.121, § 3 dealing with the minimum sending reference equivalent to be imposed in each national chain, to avoid any risk of overloading in the international network.

3 Unweighted loss (a-b) on established connections

3.1 The objective is that the risk of the loss (a-b) reaching low values at any frequency in the range 0-4 kHz should be as small as practicable. This requires restrictions on the distribution of values of stability loss (a-b) for the population of actual international calls established over the national system. Such a distribution can be characterized by a mean value and a standard deviation.

The objective will be obtained by a national system sharing a mean value of at least $(10 + n)$ dB together with a standard deviation not larger than $\sqrt{6.25 + 4n}$ dB in the band 0-4 kHz; where n is the number of analogue or mixed analogue-digital 4-wire circuits in the national chain. Other distributions are acceptable as well, as long as they yield equivalent or better results calculated according to the convention of [1].

Note 1 – See Note in § 2.2.

Note 2 – In the more conventional case of a of Figure 1/G.122, the loss (a-b) is calculated as the sum of circuit losses, terminating loss and stability balance return loss. In fact the loss (a-b) at a given frequency is the sum of the circuit losses at that frequency plus the balance return loss at the same frequency. For planning purposes, it may be assumed that the stability loss is equal to or greater than the sum of the stability balance return loss plus the sum of the circuit losses at the reference frequency. This follows from the observation that the least circuit loss typically occurs in the vicinity of the reference frequency.

Note 3 – Wholly digital circuits may be assumed to have a transmission loss with mean value and standard deviation equal to zero. Voice coders in circuits or in exchanges are expected to offer smaller variations in transmission loss than carrier circuits. For the variations in transmission loss of a coder-decoder combination, standard deviations in the order of 0.4 dB have been reported.

Note 4 – The subscriber's apparatus (telephone, modem, etc.) in the local telephone circuit is assumed to be "off hook" or equivalent, and thus providing balance return loss.

Note 5 – In practice, the distribution of stability balance return loss is distinctly skew, most of the standard deviations being provided by values above the mean. It could be unduly restrictive to assume a normal distribution.

Note 6 – The CCITT manual cited in [3] describes some of the methods proposed, and in some cases successfully applied, by Administrations to improve balance return losses.

3.2 The distribution of stability loss (a-b) recommended in § 3.1 above could, for example, be attained if, in addition to meeting the conditions of § 2.2 the mean value of the stability balance return loss at the terminating set is not less than 6 dB and the standard deviation not larger than 2.5 dB.

4 Echo loss (a-b) on established connections

4.1 In order to minimize the effects of echo on international connections it is recommended that the distribution of echo loss (*a-b*) for the population of actual international calls established over the national system should have a mean value of not less than $15 + n$ dB with a standard deviation not exceeding $\sqrt{9 + 4n}$, where n is the number of analogue or mixed analogue-digital 4-wire circuits in the national chain.

Other distributions are acceptable as well, as long as they yield equivalent or better results calculated according to the convention of Supplement No. 2.

Note 1 – Echo suppressors and cancellers according to Recommendations G.164 and G.165, typically require 6 dB of signal loss (*a-b*) for the *actual* signal converging the canceller or being controlled by the suppressor. This signal loss (*a-b*) is the ratio of incident to reflected signal power on the return path. The value of signal loss (*a-b*) will depend both upon the loss (*a-b*) frequency response and the signal spectrum. Therefore, it is desirable from a performance point of view that the stability loss (*a-b*) during an established connection should be at least 6 dB, since this will ensure proper operation for any signal (frequency spectrum) in the band 0-4 kHz.

However it may not be practical to always achieve this level of performance, especially at the higher frequencies characteristic of voice-band data signals. For speech, typically the speech signal loss (*a-b*) will be at least 6 dB if the echo loss is 6 dB. However, for voice-band data signals a higher echo loss is required to ensure a data signal loss (*a-b*) of 6 dB. For some data signals an echo loss of at least 10 dB is required. It should be noted that some modems operating half-duplex on satellite circuits equipped with echo cancellers may require proper operation of the canceller to prevent long delay echoes that exceed the receiver squelch period from causing data transmission problems.

Note 2 – See Note 2 in § 3.1. In a similar manner for planning purposes it can be assumed that the echo loss is equal to or greater than the sum of the echo balance return loss and the circuit losses at the reference frequency.

Note 3 – Recommendation G.131 provides guidance on the application of echo control devices.

4.2 The echo loss (*a-b*) is derived from the integral of the power transfer characteristic $A(f)$ weighted by a negative slope of 3 dB/octave starting at 300 Hz, extending to 3400 Hz, as follows:

$$\text{Echo loss } L_e = 3.85 - 10 \log_{10} \left[\int_{300}^{3400} \frac{A(f)}{f} \mathrm{d}f \right] \text{ dB}$$

where

$$A(f) = 10^{\frac{-Lab(f)}{10}} \text{ with } L_{ab} = \text{loss } (a\text{-}b)$$

Note 1 – The above method replaces an earlier method in which the echo loss of the path *a-t-b* was provisionally defined as the expression in transmission units of the unweighted mean of the power ratios in the band 500-2500 Hz. The new method has been found to give better agreement with subjective opinion for individual connections. However, study has shown that echo path loss distributions for large samples of actual connections calculated by the two methods have almost identical means and standard deviations. Therefore, data gathered by the older method is still considered useful in planning studies.

Note 2 – Evidence was presented which showed that a white noise signal is not necessarily optimum to measure the residual echo level after cancellation, because an echo cancellor does not converge to quite the same condition as it does with a real speech signal. It may be better to use the conventional telephone signal (Recommendation G.227 [5]) or better still, an artificial speech signal (see [6]). A good compromise is the weighted noise signal based on the principle recommended above.

Note 3 – Improved balance return losses at *t* can be obtained when the local exchange uses 4-wire switching and the local line is permanently associated with the 2-wire/4-wire conversion unit and its balance network (see Recommendation Q.552 for examples). When there is 2-wire switching a compromise balance network must be used.

Note 4 – There is evidence that a 4-wire handset telephone in normal use can contribute significant acoustic echo to the communication. Hence in some circumstances (low transmission loss, long delay times) echo control devices may be needed.

4.3 An example of how the recommendation quoted in § 4.1 above can be achieved would be for the mean value of the sum of the transmission losses *a-t* and *t-b* not to be less than $(4 + n)$ dB with a standard deviation from the mean not exceeding $2\sqrt{n}$ dB, accompanied by an echo balance return loss at the terminating set *t*, of not less than 11 dB with a standard deviation not exceeding 3 dB.

5 Effects of listener echo (receive end echo) [1)]

5.1 *General*

It has been assumed in §§ 1 to 4 that only one 2-wire-4-wire-2-wire loop (further referred to as loop) occurs in a connection. Consequently, the requirements in §§ 1 to 4 are valid for that case, i.e. they refer to the "semi-loop" seen directly from the VASPs (virtual analogue switching points). However, in mixed analogue/digital connections several loops may occur when 4-wire digital exchanges (including PABXs) are connected 2-wire to other exchanges. Such loops have typically low loss and short delay times (at most a few milliseconds). Signals reflected twice, i.e. at both hybrids that terminate a loop, would therefore contribute to listener echo. These listener echo signals:

- can lead to objectionable "hollowness" in voice communications, and
- can impair the bit error ratio of received voice-band data signals.

In general it has been found that for satisfactory transmission, data modem receivers require higher values of listener echo loss (in the band 500-2500 Hz) than speech (in the band 300-3400 Hz).

The effect of listener echo is characterized by the difference in level between the direct signal and the multiple reflected signal. In Figure 2/G.122 the loss of the direct path is assumed to be S dB, whereas the loss along the path of the reflected signal is L dB. The listener echo loss (LE) then is $L - S$ dB. It can be seen from Figure 2/G.122 that if only two reflections occur (only double-reflected signals), the listener echo loss LE equals the loss around the loop (open-loop loss, OLL), as all other losses are incurred equally by the direct and the reflected signals.

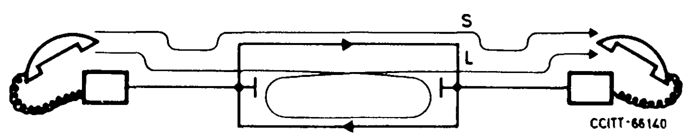


FIGURE 2/G.122

Listener echo

It should be noted that usually the listener echo will consist of a series of signals being reflected two times, four times, etc. and hence LE and OLL are in principle not equal. In practice however LE and OLL may be taken as equal when OLL exceeds about 8 dB.

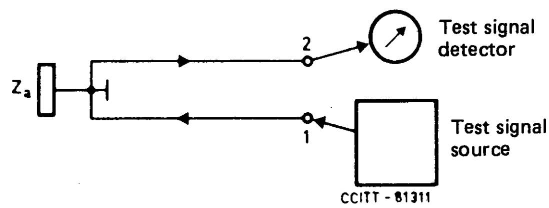
The loss around the loop can be measured by breaking the loop at some point, injecting a signal and measuring the loss incurred in traversing the open loop. All impedance conditions of the closed loop and at the 2-wire ends should be preserved whilst making the measurement. The measured quantity is the open-loop loss (OLL).

For practical purposes, semi-loop measurements may be more easily carried out, especially in the case of 4-wire exchanges with 2-wire circuit terminations, since it is sometimes difficult to maintain a connection through a 4-wire exchange and interrupt one direction of transmission. Figure 3/G.122 explains the notion of the semi-loop loss (SLL).

---

1) The use of "listener echo" in this context might be misleading. It could be substituted by a more appropriate term. The term "received end echo" is a term preferred by some Administrations.

The sum of the two semi-loop losses of a 2-wire/4-wire/2-wire device is equal to its open-loop loss (and hence very nearly to its listener echo loss) – again assuming that impedance conditions at the 2-wire ends are preserved whilst making the measurements.



Note – 1 and 2 are equi-level points (e.g. digital points).

FIGURE 3/G.122

Measuring the semi-loop loss

5.2 *Limitation of listener echo*

5.2.1 *Voice-band data transmission*

The minimum values for the listener echo loss are under study. However, the following consideration provides an example and may serve as an indication of what values of OLL might be required by existing types of modems with a bit rate of up to 2.4 kbit/s, in order to obtain high quality data transmission:

- a complete connection should not contain more than five (exceptionally seven) physical loops;
- loops with very high OLL (exceeding, e.g. 45 dB) need not be included in the number of loops in the connection;
- the OLL of each loop at any frequency in the band 500-2500 Hz, should not be less than the values indicated in Table 1/G.122 (based on $OLL = 18 + 10 \log m$, where m = total number of loops).

TABLE 1/G.122

In one national system		Maximum total number of loops in international connection
Number of national loops	OLL of each loop	
1	22 dB	3
2	25 dB	5
3	26.5 dB	7

5.2.2 *Voice transmission*

Voice performance in the presence of listener echo can be quantified in terms of a weighted value of OLL over the voice-frequency band of 300-3400 Hz. Two weighting functions have been defined in Supplement No. 3 in Volume V.

Using the information given in Recommendation P.11 appropriate values of OLL may be derived as a function of loop round-trip delay for satisfactory voice transmission. These values are presently under study.

ANNEX A

(to Recommendation G.122)

Measurement of stability loss (*a-b*) and echo loss (*a-b*)

The stability loss (*a-b*) and the echo loss (*a-b*) as defined in §§ 3.1 and 4.1 respectively may be measured by apparatus at the international centre in accordance with the principle of Figure A-1/G.122.

In respect of the echo measurement, the combined response of the send and receive filters must be such that the definition given in § 4.2 of the text is effectively implemented, e.g. such that the difference between a measured echo loss and one calculated from the loss/frequency characteristic does not exceed 0.25 dB.

The allocation of the total response between send and receive is not critical and any reasonable division may be used provided that:

– excessive interchannel interference is avoided in national transmission systems due to an unrestricted spectrum of the transmitted signal;

– unwanted signals that may give rise to errors, e.g. hum, circuit noise, carrier leak signals, are prevented from entering the receiver.

Appropriate arrangements (not shown) are needed for automatic or manual access to the 4-wire switches at the international centre and also to ensure that due account is taken of the transmission levels at the actual switching points.

As far as the stability measurement is concerned, if a sweep oscillator is used, attention must be paid to the risks of engendering false operation of national signalling systems.

For both measurements anomalous results may be obtained if echo suppressors are encountered in the national extension.

To measure the echo loss (*a-b*), the output of the send filter is first connected to the input of the receive filter and the appropriate level set and noted. The apparatus is then connected as in Figure A-1/G.122 and the new reading on the meter noted. The loss so indicated is the echo loss (*a-b*).

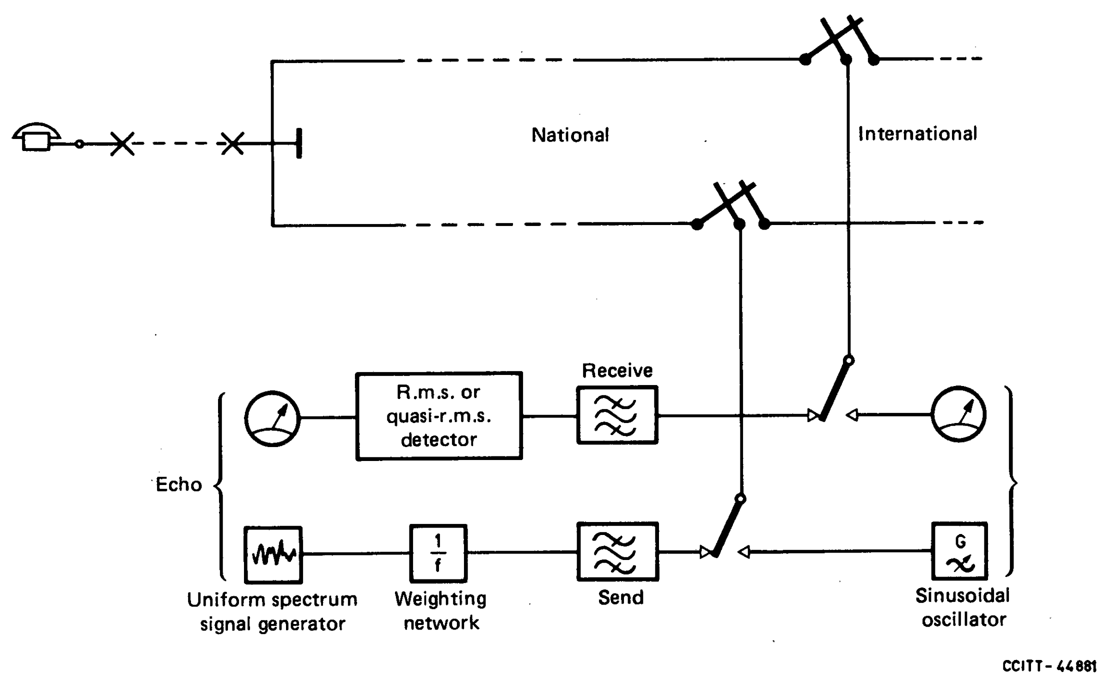


FIGURE A-1/G.122

Principle of measuring the transmission loss of the path *a-t-b* from the points of view of stability and of echo

ANNEX B

(to Recommendation G.122)

Explanation of terms associated with the path *a-t-b*

(Contribution of British Telecom and Australia)

B.1 *Return loss*

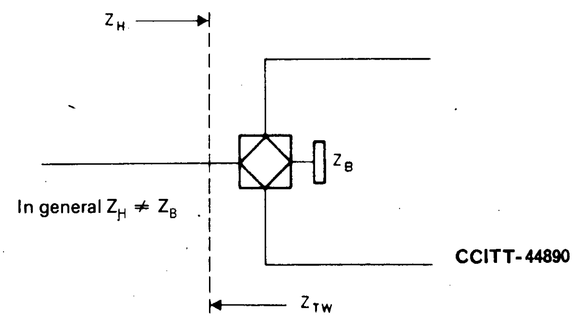
This is a quantity associated with the degree of match between two impedances and is given by the expression:

$$\text{Return loss of } Z_1 \text{ versus } Z_2 = 20 \log_{10} \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| \text{ dB}$$

The use of the expression "return loss" should be confined to 2-wire paths supporting signals in the two directions simultaneously.

B.2 *Balance return loss*

A clear definition is given in the preamble of Recommendation G.122. Figure B-1/G.122 illustrates the definition.



$$\text{Balance return loss} = 20 \log_{10} \left| \frac{Z_B + Z_{TW}}{Z_B - Z_{TW}} \right| \text{ dB.}$$

FIGURE B-1/G.122

The 2-wire portion must be in the condition appropriate to the study, e.g., if speech echo is being studied the telephone set must be in the speaking condition.

In the particular case (which occurs very often) in which the impedances presented by each of the paths in the 4-wire portion is also Z_B (e.g. 600 ohms) then the terminating set presents an impedance of the 2-wire point which is substantially equal to Z_B. Figure B-2/G.122 illustrates this case.

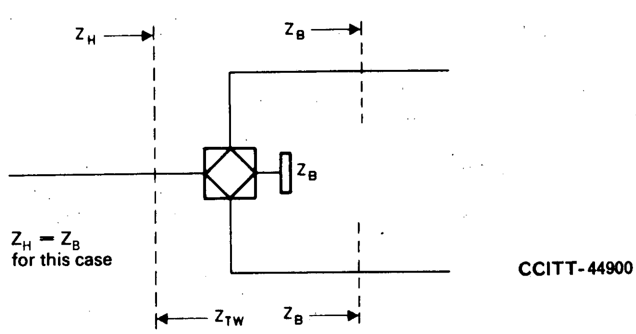


FIGURE B-2/G.122

The term "balance return loss" (*not* return loss) should always be used for the contribution to the loss of the path *a-t-b* attributable to the degree of match between Z_B and Z_{TW}.

B.3 *Transmission loss of the path a-t-b*

There is room for confusion here because the concept can be applied to arrangements in which there is no physical point "*t*" at all, e.g. as in some laboratory simulations of long connections in which echo is introduced by a controlled unidirectional path bridging the two 4-wire paths. The point "*t*" is necessary in the Recommendation because practical public switched telephone networks are being dealt with.

Thus in general two cases arise.

Case 1: There does exist a point "*t*" (Figure B-3/G.122).

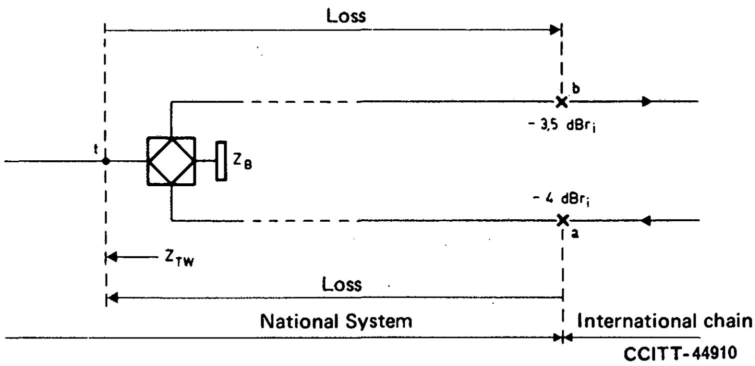


FIGURE B-3/G.122

The transmission loss of the path *a-t-b* may be calculated from

$$\text{loss }(a\text{-}t) + 20 \log_{10} \left| \frac{Z_B + Z_{TW}}{Z_B - Z_{TW}} \right| + \text{loss }(t\text{-}b)$$

The diagram is drawn in terms of the virtual switching points of the international circuit with their associated relative levels. The subscript *i* in the abbreviation dBr_i signifies that these relative levels are with respect to a 0 dBr point of the international circuit.

It is clear that any other convenient pair of relative levels (differing by 0.5 dB in the correct sense) can be used in practice, e.g., the actual switching levels used in an international centre.

Case 2: There does not exist any "*t*" (Figure B-4/G.122).

This relates particularly to laboratory testing arrangements.

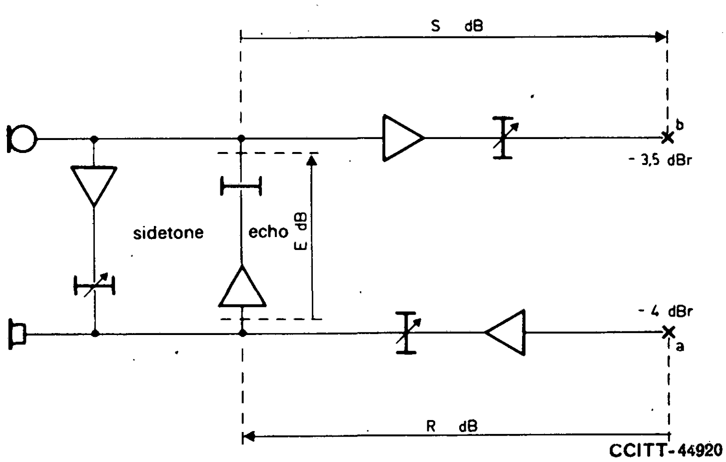


FIGURE B-4/G.122

In this case the loss of the path a-"t"-b may be calculated from: $(R + E + S)$ dB (assuming acoustic feedback at the 4-wire telephone to be negligible).

In both cases the loss of "the path a-t-b" can in principle be directly measured by the principles described in Annex A, i.e. by injecting a signal at a and measuring the result at b, so that one may properly say for all cases

$$\left\{ \begin{array}{c} \text{transmission loss} \\ \text{of the path } a\text{-}t\text{-}b \end{array} \right\} \equiv \left\{ \begin{array}{c} \text{transmission loss} \\ \text{between } a \text{ and } b \end{array} \right\}$$

or, more shortly

$$\text{loss } (a\text{-}t\text{-}b) \equiv \text{loss } (a\text{-}b)$$

B.4 *Stability and echo losses*

So far the quantities dealt with are functions of frequency and yield a graph of attenuation/frequency distortion. When it is required to characterize such a graph with a single number, additional qualifying indications are, for example, stability loss (a-b) and echo loss (a-b).

The text of this Recommendation gives the definitions of these single-number descriptions thus: the stability loss (a-b) is the least value (measured or calculated) in the band 0-4 kHz (see §§ 2.1 and 3.1), and the echo loss (a-b) is a weighted integral of the loss/frequency function over the band 300-3400 Hz, as defined in § 4.2.

When the echo-path loss/frequency characteristic is available in graphical or tabular form, alternative techniques for the calculation of echo loss (a-b) are preferable to that suggested for the field measurement given in Annex A.

Note – When evaluating echo loss from graphical or tabulated data, sufficient frequency points should be taken to ensure that the influence of the shape of the amplitude/frequency characteristic is adequately preserved. The more irregular the shape, the more points should be taken for a given accuracy.

Graphical data (trapezoidal rule)

If the loss/frequency characteristic of the echo-path is available in graphical form (or the data were suitably measured) the echo loss may be calculated by using the trapezoidal rule as follows:

1) Divide the frequency band (300 to 3400 Hz) into N sub-bands of equal width on a log-frequency scale.

2) Read off the echo loss at each of the $N + 1$ frequencies at the edges of the N sub-bands, and express it as an output/input power ratio, A_i.

3) Calculate the echo loss using the formula:

$$L_e = -10 \log_{10} \left[\frac{1}{N} \left(\frac{A_0}{2} + A_1 + A_2 \ldots + A_{N-1} + \frac{A_N}{2} \right) \right]$$

Tabulated data

When the loss/frequency data are only available at $N + 1$ discreet frequencies, which are nonuniformly spaced on a log-frequency scale, proceed as follows:

An approximation to the formula for echo loss (a-b) given in the text is:

$$L_e = 3.24 - 10 \log_{10} \sum_{i=1}^{N} (A_i + A_{i-1}) (\log_{10} f_i - \log_{10} f_{i-1})$$

where

A_0 is the output/input power ratio at frequency of $f_0 = 300$ Hz,

A_i the ratio at frequency f_i, and

A_N the ratio at frequency $f_N = 3400$ Hz.

Note 1 – The approximation involved is to assume that within the sub-band f_{i-1}, to f_i, the power ratio is constant and has the value $A(f) = (A_i + A_{i-1})/2$.

Note 2 – The constant 3.24 in the approximate formula arises from a combination of the constant 3.85 in the definition and other constants produced by the approximation.

The sum of product terms in the approximation formula may be conveniently calculated as illustrated by the following example:

TABLE B-1/G.122

f_i (Hz) (1)	$\log_{10} f_i$ (2)	$\log_{10} f_i - \log_{10} f_{i-1}$ (3)	loss (dB) (4)	ratio A_i (5)	$A_i + A_{i-1}$ (6)	(3) × (6) (7)
300	2.477		∞	0		
		0.222			0.124	0.0275
500	2.699		9.05	0.124		
		0.204			0.402	0.0820
800	2.903		5.56	0.278		
		0.097			0.636	0.0617
1000	3.000		4.46	0.358		
		0.176			0.838	0.1475
1500	3.176		3.19	0.48		
		0.125			0.970	0.1213
2000	3.301		3.09	0.49		
		0.097			0.881	0.0855
2500	3.398		4.08	0.391		
		0.079			0.571	0.0451
3000	3.477		7.45	0.180		
		0.055			0.180	0.0099
3400	3.532		∞	0		
					Total	0.5804

$$L_e = 3.24 - 10 \log 0.5804 = 5.6 \text{ dB}$$

B.5 *Overall loudness rating of the echo path* (Talker echo loudness rating, TELR)

Recommendation G.131 is concerned with complete talker echo paths and it is convenient to characterize this path in terms of loudness rating (LR). By convention we may regard the echo balance return loss as the contribution it makes to the overall loudness rating (OLR) of the mouth-ear echo path. Naturally, as indicated in § 2 of the text, the echo loss (*a-b*), when this is already known, may be used instead of the sum of three quantities: the LR (*a-t*), the echo balance return loss at *t* (averaged according to § 2) and the LR (*t-b*).

Hence the nominal overall loudness rating of the echo path may be calculated as illustrated in Figure B-5/G.122.

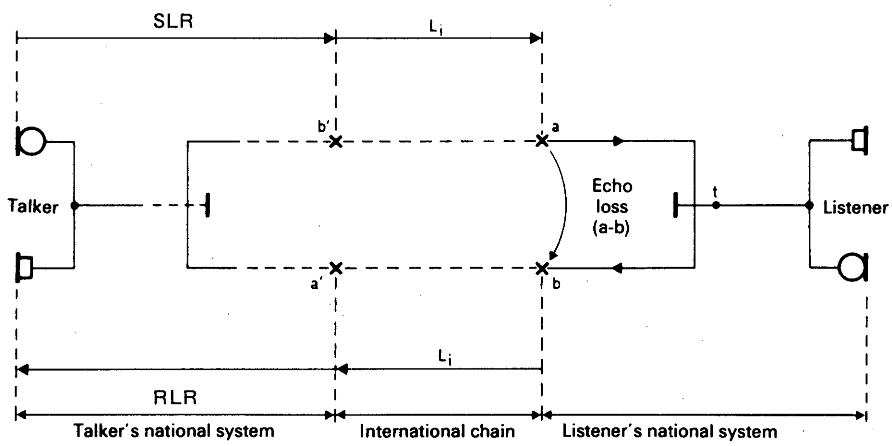


FIGURE B-5/G.122

Overall Loudness Rating of the echo path (Talker echo loudness rating, TELR), see Annex A/G.111

= SLR + RLR of the talker's national system,

\+ twice the LR of the international chain (i.e.: $2L_i$),

\+ the echo loss (*a-b*) of the listener's national system (i.e. averaged according to this Recommendation).

B.6 *Résumé of useful terms*

return loss – Relates to a 2-wire bidirectional circuit; classical definition.

balance return loss – Proportion of the loss at the *a-t-b* path attributable to the degree of match between the 2-wire impedance and the balance impedance at the terminating unit. Applicable only if there is a point "*t*".

transmission loss of the path *a-t-b* – Can be regarded as the loss (*a-b*), whether there exists a physical point "*t*" or not.

stability loss (*a-b*) – The least value of the loss (*a-b*) in the band 0 to 4 kHz.

echo loss (*a-b*) – The loss (*a-b*) averaged according to the definition in § 2 of the text.

echo balance return loss – A balance return loss averaged according to § 2 of the text.

overall loudness rating of the echo path (Talker echo loudness rating, TELR) – The sum of the send loudness rating and receive loudness rating of the talker's national system, twice the LR of the international chain, and the echo loss (*a-b*) of the listener's national system.

References

[1] *Calculations of the stability of international connections established in accordance with the transmission and switching plan*, CCITT Green Book, Vol. III-2, Supplement No. 1, ITU, Geneva, 1973.

[2] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232, § 2.

[3] CCITT manual *Transmission planning of switched telephone networks*, ITU, Geneva, 1976.

[4] CCITT Recommendation *Reduction of the risk of instability by switching means*, Vol. VI, Rec. Q.32.

[5] CCITT Recommendation *Conventional telephone signal*, Vol. III, Rec. G.227.

[6] CCITT Question 8/XII, Annex 2, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

Recommendation G.123

CIRCUIT NOISE IN NATIONAL NETWORKS

(Geneva, 1964; amended at Mar del Plata, 1968, Geneva, 1972, 1976 and 1980 and Melbourne 1988)

1 Noise induced by power lines

The network performance objective for the psophometric e.m.f. of the noise produced by magnetic and/or electrostatic induction from all the power lines affecting one or more parts of a chain of telephone lines[1] joining a subscriber's set to its international centre should not exceed 1 millivolt, this being the value at the line[1] terminals of the subscriber's set (when receiving), it being assumed that the telecommunication installations inserted in that chain are balanced to earth as perfectly as possible, in conformity with the most modern equipment construction.

It should be noted that, even in the case of perfectly balanced lines[1], the insertion of equipment having too great a degree of unbalance to earth may cause unacceptable noise at the terminals of a subscriber's receiver.

In every national network, it is usually possible, in practice, to find switching centres such that some of the lines[1] that terminate at those centres (lines[1] in cable, conforming to CCITT specifications) are free from noise arising from neighbouring power lines. It is then sufficient to determine the psophometric e.m.f.s arising from all the power lines[1] affecting one or more parts of the chain of lines[1] joining such a centre to the subscriber's set.

2 Noise contributed by transmission systems

2.1 *Analogue systems*

2.1.1 *Very-long-distance circuits* (about 2500-25 000 km)

If an extension circuit more than 2500 km long is used in a large country, it will have to meet all the recommendations applicable to an international circuit of the same length (Recommendation G.153). This implies that the equipment design objective for the line noise in channels used to provide these circuits should not exceed 2 pW0p/km.

[1] "Line" as used in this § 1 should be understood as meaning subscriber's line, trunk junction or trunk circuit.

2.1.2 *Circuit ranging in length from very short distances up to 2500 km*

These circuits should meet the requirements of Recommendation G.152. This implies that according to the noise objectives of Recommendation G.222 [1] the accumulated line noise should correspond to an average of not more than 3 pW0p/km and the noise power produced by the various modulating equipments should meet the provisions of the Recommendation cited in [2].

Taking account of the particular structure of a real circuit the pertinent Recommendations CCITT/G.226 [3] (for cable systems) or CCIR/395 [4] (for radio-relay systems) must be applied when assessing its noise performance.

Note 1 – The permissible noise contributions from equipment do not depend on whether the circuits form part of the international 4-wire chain or are connected to it by 2-wire switching. However, the circuit noise powers assume that the hypothetical reference connections of Recommendation G.103 are, or will be in future, reasonably typical of connections. They also assume that the total length of circuits connecting the local exchange to the primary centre is not excessive. The attention of Administrations is drawn to a conclusion of studies carried out by the CCITT during the 1964-1968 Study Period, that if the additional percentage of "poor or bad" opinions on the quality of connections due to noise introduced by the circuits connecting the local exchange to the primary centre is not to exceed one half of that caused by the presence in the connection of all other sources of circuit noise, then the noise contributed by each one of these circuits should be limited to about 500 pW0p (mean for all the channels of the system during any hour).

Note 2 – Under the above conditions and assuming the maximum noise values permitted for pairs of channel modulators (200 pW0p), group modulators (80 pW0p) and supergroup modulators (60 pW0p), a total noise power of 500 pW0p will not be exceeded by a circuit connecting the local exchange to the primary centre (Figure 1/G.103) when its length is less than about 50 to 100 km.

Note 3 – In the case that those circuits are operated with compandors conforming to Recommendation G.162, the permitted noise powers are to be understood inclusive of the effect of the compandor gain.

2.2 *Digital system*

Circuits provided by PCM systems which accord with the G.700 Series of Recommendations, in particular Recommendation G.712 [5], will have an acceptable noise performance which is substantially independent of their length.

2.3 *Mixed circuits*

The noise value in a circuit provided by both analogue and digital transmission systems depends on the whole length of analogue sections and of the number of codecs in a circuit.

Noise limits and measurement methods for a mixed circuit are studied under Questions 26/XII, 16/IV and 18/IV.

3 Noise in a national 4-wire automatic exchange[2)]

3.1 *Definition of a* **connection through an exchange**

Noise conditions in a national 4-wire automatic exchange are defined by reference to a "connection" through this exchange. By "connection through an exchange" is to be understood the pair of wires corresponding to a direction of transmission and connecting the input point of a circuit incoming in the exchange to the output point of a different circuit outgoing from the exchange. These input or output points are those defined in Recommendation Q.45 (points A and D of Figure 1/Q.45 [8]) and are not necessarily the same as the text access points defined in Recommendation M.640 [9].

[2)] In accordance with Recommendation Q.31 [6], the limits are the same as in Recommendation Q.45 [7].

3.2 *Equipment design objective for the mean noise power during the busy-hour*

The mean of the noise over a long period during the busy-hour should not exceed the following values:

1) Psophometrically weighted noise: -67 dBm0p (200 pW0p),

2) Unweighted noise: -40 dBm0 (100 000 pW0) measured with a device with a uniform response curve throughout the band 30-20 000 Hz.

Note – A sufficient variety of connections should be chosen to ensure that the measurements are representative of the various possible routes through the exchange.

3.3 *Equipment design objective for the impulsive noise during the busy-hour*

Noise counts should not exceed 5 counts in 5 minutes at the threshold level of -35 dBm0 (see the Recommendation cited in [10] for measurement procedure).

Note – Figure 3/Q.45 [11] shows the maximum number of impulsive noise counts acceptable in a 5-minute period.

4 **Noise allocation for a national system** (guide for planning purposes)

The noise powers indicated in the following text are nominal values.

Network planning should be such that the noise power entering the international network and attributable to national sending systems meets the limits of the following rule:

The psophometric noise power introduced by the national sending system at a point of zero relative level on the first international circuit must not exceed either $(4000 + 4L)$ or $(7000 + 2L)$ pWp, whichever is less, and where L is the total length in kilometres of the long-distance FDM carrier systems in the national chain. The corresponding quantities referred to the send virtual switching point are $(1800 + 1.8L)$ and $(3100 + 0.9L)$ pWp.

The derivation of this rule is explained in Annex A.

Note – A problem, which has already arisen in some national networks, as regards the receiving direction, is that when losses are reduced the circuit noise becomes more noticeable, particularly during periods of no conversation. This is particularly relevant in the case of large countries in which the noise contribution from line systems is high. Hence if an Administration complies with a recommendation concerning national noise power levels and then subsequently improves transmission, perhaps by introducing 4-wire switching in lower-order exchanges, it may find itself in a worse situation as regards noise. It follows that it is important to preserve a proper balance between noise and loss.

ANNEX A

(to Recommendation G.123)

Noise allocation for a national system

A.1 It is desirable that the noise power arising in national networks be limited in terms of the level appearing at the virtual switching points – the agreed interface between the national and the international network. In order to do this, some particular distribution of losses within the national network must be assumed. The solution is to adopt an agreed reference connection in order to specify maximum noise power levels from national sources referred to the virtual switching point of the international circuit.

A.2 Having regard to the way in which national networks are constructed, it is appropriate to express the noise allowance in the form $A + BL$ where A is a fixed allowance resulting from noise in exchanges and from short-haul multiplex systems, B is an allowance for a noise rate per unit length from long-haul multiplex systems and L is the total length of these latter systems in the national portion of the international connection. Two such expressions are necessary, one for countries of average size and another for large countries (in the sense of Recommendation G.121).

A.3 This approach is comparatively straightforward in the national sending system and serves to limit the amount of noise injected into the international connection.

A.4 *Average-sized countries* (i.e. not greater than 1500 km from the CT3 to the most remote local exchange)

The relevant hypothetical reference chain for the national sending system is given in Figure A-1/G.123[3]. The circuit between the local exchange and the primary centre is assumed to be routed on an FDM carrier system of length not exceeding 250 km and operated at a nominal loss of 3 dB. The noise power on this circuit is taken to be the maximum value of 2000 pW0. The circuit between the primary centre and the secondary centre is also assumed to be routed on an FDM carrier system of the same type.

The line noise power rate of the two long-distance trunk circuits is assumed to be 4 pW/km and the total line length of these two circuits ($L_1 + L_2$ in Figure A-1/G.123) approaches the limit of 1500 km arbitrarily defining "a country of average size" in Recommendation G.121. It is thus assumed that the distance covered by the two short-haul systems is a very small proportion of the total length of the complete national sending system.

Each exchange is assumed to contribute 200 pWp in accordance with § 3 of the text, or Q.31 [6].

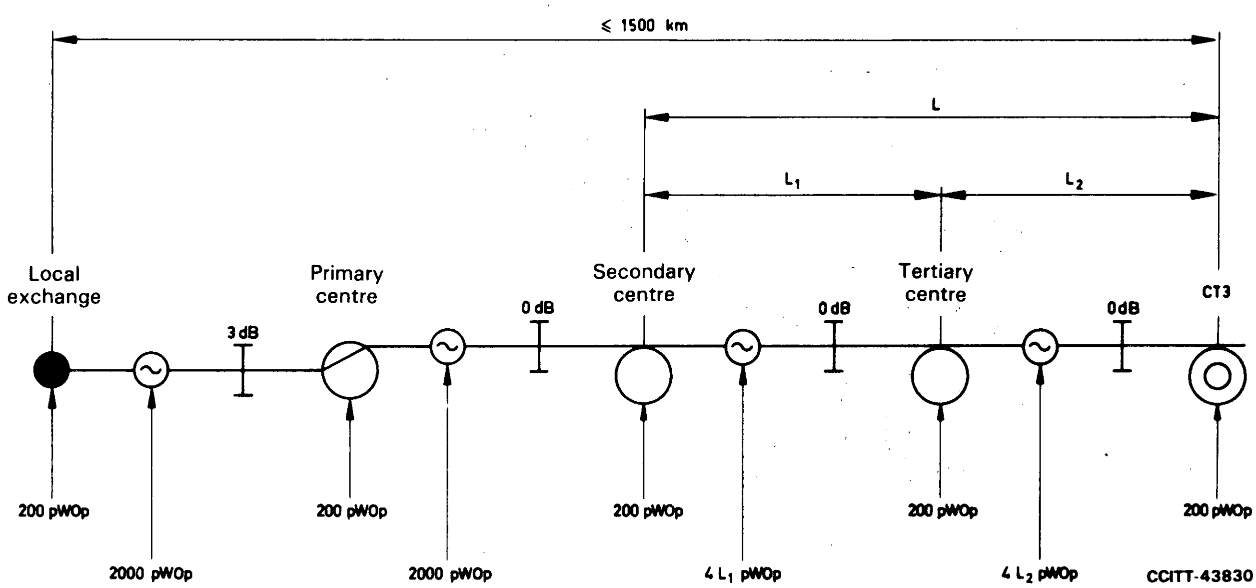


FIGURE A-1/G.123

The total noise power level referred to a point of zero relative level on the first international circuit at the CT3 is (moving from right to left and adding in each successive noise contribution encountered):

$$200 + 4L_2 + 200 + 4L_1 + 200 + 2000 + 200 + \tfrac{1}{2}(2000) + \tfrac{1}{2}(200) = 3900 + 4L \text{ pW0}$$

where $L = L_1 + L_2$. This may be conveniently rounded off to $4000 + 4L$ pW0.

This expression is valid for L not exceeding 1500 km leading to, at that distance, 10 000 pW0.

[3] *Note by the CCITT Secretariat* – The noise values shown in this figure are maximum values; see also the corresponding element of Figure 1/G.103.

A.5 *Large countries*

When L is in excess of 1500 km the additionnal long-distance circuits in the national network should in principle be engineered to international standards, and in particular some large countries have found it necessary to plan national systems with noise power rates lower than 4 pW/km.

A convenient value to assume is 2 pW/km; this is in rough agreement with the practice of one such large country and is also in line with Recommendation G.153.

The rule for large countries has been established as shown in Figure A-2/G.123 in which the 4000 + $4L$ rule is shown passing through the point (1500 km, 10 000 pW). A line with a slope of 2 pW/km is constructed to pass through the same point and its intercept is seen to be 7000 pW. Hence the rule for large countries is 7000 + $2L$ pW0. (The 0.5-dB nominal loss of the last national circuit has been ignored for simplicity's sake.)

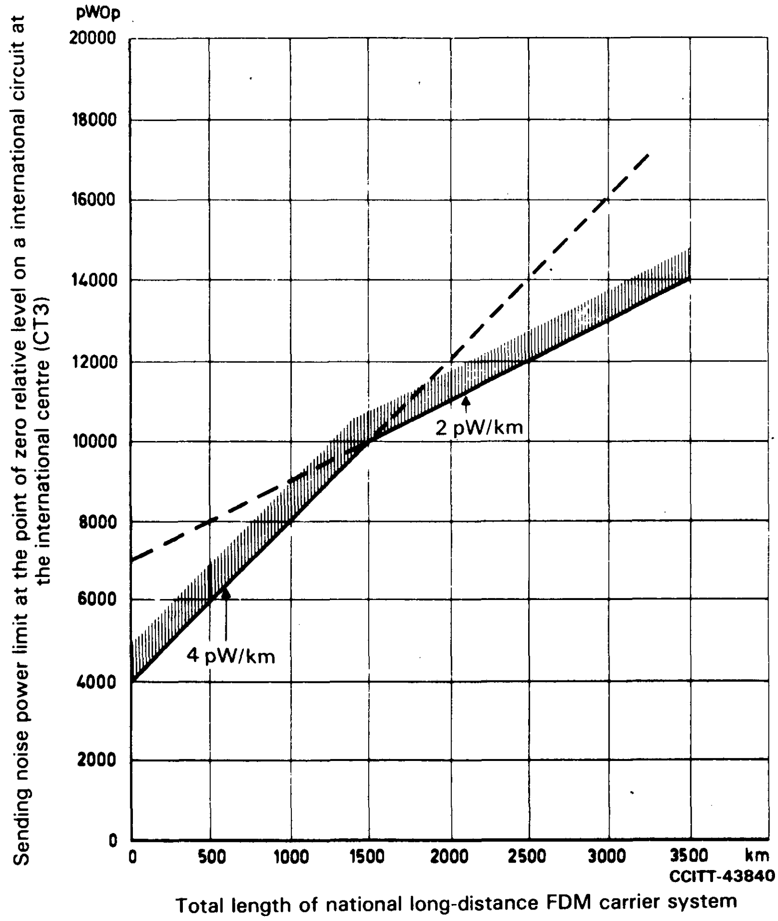


FIGURE A-2/G.123

References

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

[2] *Ibid.*, § 4.

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

[4] CCIR Recommendation *Noise in the radio portion of circuits to be established over real radio-relay links for FDM telephony*, Vol. IX, Rec. 395, ITU, Geneva, 1986.

[5] CCITT Recommendation *Performance characteristics of PCM channels between 4-wire interfaces at voice frequencies*, Vol. III, Rec. G.712.

[6] CCITT Recommendation *Noise in a national 4-wire automatic exchange*, Vol. VI, Rec. Q.31.

[7] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45.

[8] *Ibid.*, Figure 1/Q.45.

[9] CCITT Recommendation *Four-wire switched connections and four-wire measurements on circuits*, Yellow Book, Vol. IV, Rec. M.640, ITU, Geneva, 1981.

[10] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45, Annex A.

[11] *Ibid.*, Figure 3/Q.45.

Recommendation G.125

CHARACTERISTICS OF NATIONAL CIRCUITS ON CARRIER SYSTEMS

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

Carrier circuits which are likely to form part of international connections should meet the requirements of Recommendation G.132 as far as attenuation distortion is concerned. The circuits should transmit all types of signal (e.g. speech, data, facsimile) which might normally be expected, according to Recommendations over this part of the connection.

Recommendations relating to the noise performance of national circuits are now to be found in Recommendation G.123 (circuit noise in national networks).

1.3 General characteristics of the 4-wire chain formed by the international circuits and national extension circuits

This subsection gives the overall characteristics recommended for the 4-wire chain defined in Recommendation G.101, § 2.

Recommendation G.131

STABILITY AND ECHO

(Geneva, 1964; amended at Mar del Plata, 1968, and Geneva, 1972, 1976, and 1980; Malaga-Torremolinos, 1984 and Melbourne, 1988)

1 Stability of telephone transmission

The nominal transmission loss of international circuits having been fixed, the principal remaining factors which affect the stability of telephone transmission on switched connections are:

– the variation of transmission loss with time and among circuits (Recommendation G.151, § 3);
– the attenuation distortion of the circuits (Recommendation G.151, § 1);
– the distribution of stability balance return losses (Recommendation G.122, §§ 2 and 3).

The stability of international connections has been calculated and the results are displayed graphically in Figure 1/G.131, which shows the proportion of connections (out of all the possible connections) likely to exhibit a stability of less than or equal to 0 dB or 3 dB as a function of the number of all analogue circuits comprising the 4-wire chain and the mean values of stability balance return loss that may be assumed. Of course the proportion of connections actually established which exhibit a stability lower than or equal to the values considered will be very much smaller.

Note – If digital circuits are included in the 4-wire chain, the stability is likely to be better than shown in Figure 1/G.131, as these circuits will exhibit a lower transmission loss variability than is assumed in that figure.

When interpreting the significance of the curves showing the proportion of calls likely to have a stability of 3 dB or less it should be borne in mind that the more complicated connections will undoubtedly incorporate a circuit equipped with an echo suppressor or canceller, in which case the stability during conversation is very much higher.

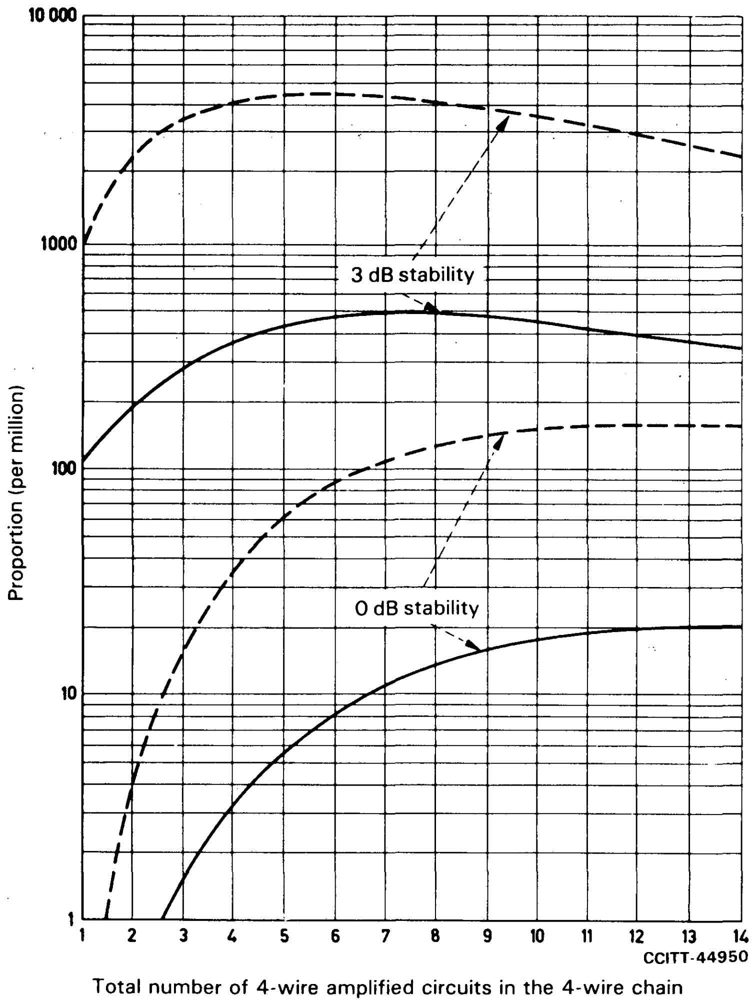


FIGURE 1/G.131

Proportion of possible connections with a stability equal to or less than 0 dB or 3 dB

The simplifying assumptions underlying the calculations are:

a) National circuits are added to the international chain in compliance with Recommendation G.122.

b) The standard deviation of transmission loss among analogue international circuits routed on groups equipped with automatic regulation is 1 dB. This accords with the assumptions used in Recommendation G.122. The results of the 10th series of tests by Study Group IV indicate that this target is being approached in that 1.1 dB was the standard deviation of the recorded data and the proportion of unregulated international groups in the international network is significantly decreasing.

c) The variations of transmission loss in the two directions of transmission are perfectly correlated.

d) The departure of the mean value of the transmission loss from the nominal value is zero. As yet there is little information concerning international circuits maintained between 4-wire points.

e) No allowance has been made for the variations and distortions introduced by the national and international exchanges.

f) The variation of transmission loss of circuits at frequencies other than the test frequency is the same as that at the test frequency.

g) No account has been taken of attenuation distortion. This is felt to be justifiable because low values of balance return loss occur at the edges of the transmitted band and are thus associated with higher values of transmission loss.

h) All distributions are Gaussian.

Bearing in mind these assumptions, the conclusion is that the Recommendations made by the CCITT are self-consistent and that if these Recommendations are observed and the maintenance standard set for variation of loss among circuits is achieved, there should be no instability problems in the transmission plan. It is also evident that those national networks which can exhibit no better stability balance return loss than 3 dB mean, 1.5 dB standard deviation are unlikely to seriously jeopardize the stability of international connections as far as oscillation is concerned. However, the near-singing distortion and echo effects that may result give no grounds for complacency in this matter.

Details of the calculations are set out in [1].

2 Limitation of echoes

The main circuits of a modern telephone network providing international communications are high-velocity carrier circuits on symmetric, coaxial or optical fibre pairs or radio-relay systems. Echo control devices such as echo suppressors and echo cancellers are not normally used except on connections involving very long international circuits. There is often no general need for echo control devices in national networks but they may be required for the inland service in large countries. Echo control devices may also be needed on loaded-cable circuits (low-velocity circuits) used for international calls.

Echoes may be controlled in one of two ways: either the overall loss of the 4-wire chain of circuits may be adjusted so that echo currents are sufficiently attenuated (which tacitly assumes a particular value for the echo return loss) or an echo control device can be fitted.

2.1 *Transmission loss adjustment*

The curves of Figure 2/G.131 indicate the minimum value of the overall loudness rating (OLR)[1)] in the echo path that must be introduced if no echo suppressor is to be fitted. The OLR is shown as a function of the mean one-way propagation time. Supplement No. 2, at the end of this fascicle, explains how these curves have been derived and Annex A to this Recommendation gives an example of their application.

The solid curves are applicable to a chain of analogue circuits which are connected together 4-wire. However, they may also be used for circuits connected together 2-wire if precautions have been taken to ensure good echo return losses at these points (i.e. averaged in accordance with Recommendation G.122) for example, a mean value of 27 dB with a standard deviation of 3 dB.

Note – This value is only sufficient to assure average echo losses (a-b) of $(15 + n)$ dB, as currently called for in Recommendation G.122 § 4.1.

The dashed curve is applicable to fully digital connections with analogue subscriber lines (such as shown in Figure 2/G.111), and, under certain assumptions (see Supplement No. 2), to fully digital connections with digital subscriber lines (such as shown in *b)* of Figure 1/G.104. In the latter case the echo path includes the acoustical path between earpiece and mouthpiece of the handset.

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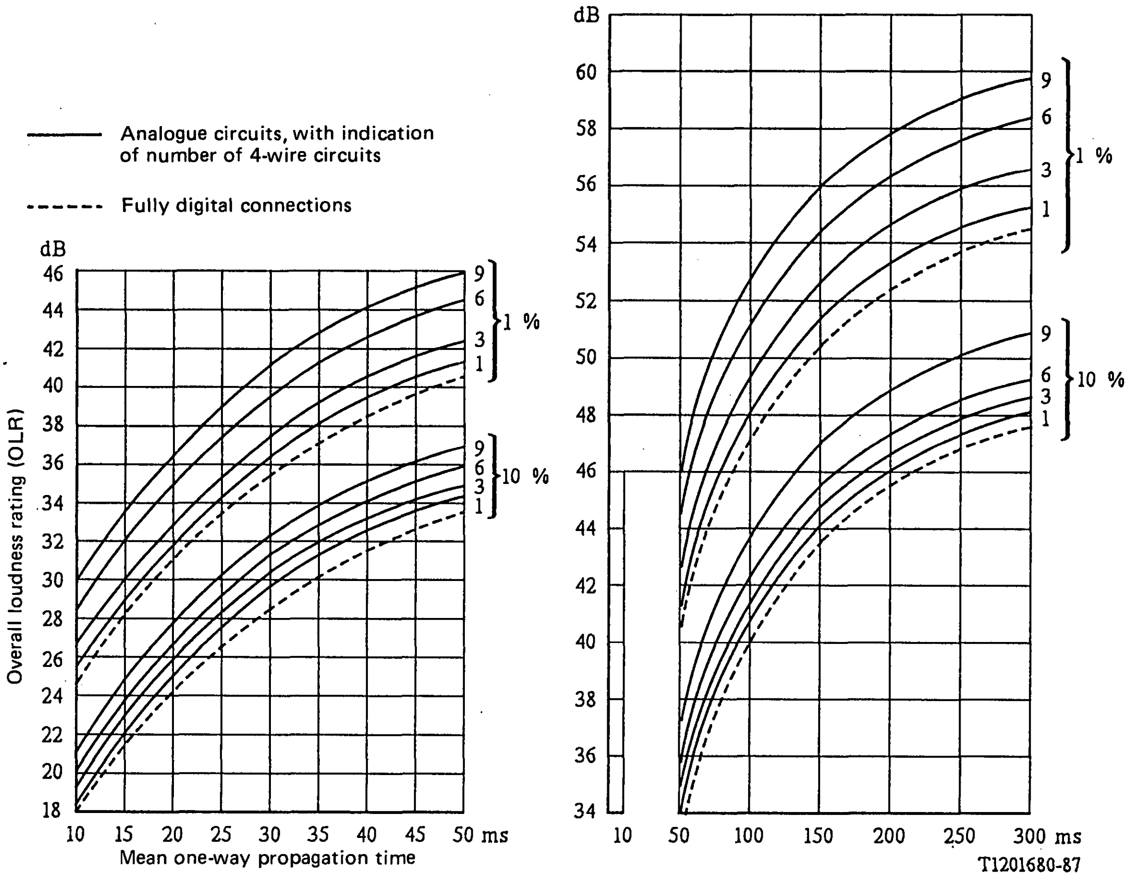
1) While Figure 2/G.131 is based on nominal values of LR of trunk junction and trunk circuits, it refers to minimum SLR and RLR values of subscriber systems.

When an international circuit is used only for comparatively short and straightforward international connections the nominal transmission loss between virtual analogue switching points may be increased in proportion to the length of the circuit according to the following rule, if the use of echo control devices can thereby be avoided:

- up to 500 km route distance: 0.5 dB;
- between 500 km and 1000 km route distance: 1.0 dB;
- for every additional 500 km or part thereof: 0.5 dB.

However, such a circuit may not form part of multicircuit connections unless the nominal transmission loss is restored to 0.5 dB.



Note 1 – The percentages refer to the probability of encountering objectionable echo.

Note 2 – The overall loudness rating of the echo path is here defined as the sum of:

- the loudness rating in the two directions of transmission of the local telephone system of the talking subscriber (assumed to have minimum values of loudness rating);
- the loudness rating in the two directions of transmission of the chain of circuits between the 2-wire end of the local telephone system of the talking subscriber and the 2-wire terminals of the 4 wire-2 wire terminating set at the listener's end;
- the mean value of the echo balance return loss at the listener's end.

FIGURE 2/G.131

Echo tolerance curves

2.2 *Echo control devices*

The preferred type of echo suppressor is a terminal, differential, half-echo suppressor operated from the far end. There are several types of half-echo suppressor in use in the international network, one suitable only for use in connections with mean one-way propagation times not exceeding 50 ms, referred to as a short-delay echo suppressor, and the others suitable for use in connections with any mean one-way propagation time, especially times well over 50 ms, referred to as a long-delay echo suppressor like those used on circuits routed on communication-satellite systems. The characteristics of the short-delay echo suppressors are given in [2]. The characteristics of echo suppressors which can be used on connections with either short or long propagation times are given in [3] and in Recommendation G.164 (echo suppressors with new functions). Another type of echo control can be obtained by echo cancellers. The characteristics are given in Recommendation G.165.

From subjective test information received, it is concluded that:

1) Echo cancellers in accordance with Recommendation G.165 provide superior speech transmission performance (at the 0.05 confidence level) to that provided by:

   a) echo suppressors according to Recommendation G.161 (*Orange Book*);

   b) echo suppressors according to Recommendation G.164 with fixed break-in differential sensitivity, FBDS;

   Note – Two Administrations have the view that echo cancellers according to Recommendation G.165 and echo suppressors according to Recommendation G.164 with adaptative break-in differential sensitivity (ABDS) provide about the same performance when the echo path loss is considerably above the lower end of its range; calculations based on Recommendation G.122, § 2 and assuming a minimum echo loss of 6 dB, indicate that the majority of echo path losses will be greater than the minimum value.

2) echo suppressors in accordance with Recommendation G.164 with ABDS provide superior speech transmission performance to that provided by echo suppressors with FBDS.

3) echo control devices of different types (i.e. echo suppressors or cancellers in accordance with the series G Recommendations) placed at opposite ends of a connection will operate compatibly. In this case the subjective quality perceived at one end is almost uniquely dependent on the performance of the echo control device installed at the opposite end.

Note 1 – Regional satellite circuits routed in parallel with terrestrial circuits, without perceivable echo, will benefit from the use of echo control devices of the best quality. Otherwise any degradation of the normal quality by routing over the satellite circuit may be found objectionable by the subscriber.

Note 2 – Bilateral agreement between Administrations may facilitate the introduction in the network of echo control devices of better quality.

2.3 *Rules governing the limitation of echoes*

The rules given below are subdivided into ideal rules and practical rules. It is recognized that no practical solution to the problem could comply with rules so exclusive and inflexible as the ideal rules. Practical rules are suggested in the hope that they will ease the switching and economic problems. They should not be invoked unless the ideal rules cannot reasonably be complied with.

2.3.1 *Rules for connections without echo control devices*[2)]

2.3.1.1 *Ideal rule – Rule A*

For a connection between any pair of local exchanges in different countries, the probability of incurring the opinion "unsatisfactory" due to talker echo shall be less than 1%, when minimum practical nominal send and receive loudness ratings are assumed for the talker's telephone and line.

Note – Calls between a given pair of local exchanges may encounter different numbers of 4-wire circuits, according to the routing discipline and time of day. Figure 2/G.131 permits compliance with this rule to be assessed for the separate parts of the total traffic which encounter 1, 2, 3 ... 9 4-wire circuits, under certain conventional assumptions. (See Supplement No. 2 at the end of this fascicle.)

[2)] The rules in this Recommendation have been updated (to include echo cancellers) and regrouped, compared with previous versions of Recommendation G.131. The letters indicating the rules are the same as in previous versions of Recommendation G.131 in order to provide a degree of continuity.

2.3.1.2 *Practical rule – Rule E*

For connections involving the longest national 4-wire extensions of the two countries, a probability of incurring an "unsatisfactory" opinion due to echo not of 1% (Rule A) but of 10% can, by agreement between the Administrations concerned, be tolerated. This Rule E[3] is valid only in those cases where it would otherwise be necessary, according to Rule A[3], to use an echo control device solely for these connections, and where there is no need for echo control devices on connections between the regions in the immediate neighbourhood of the two international centres concerned.

2.3.2 *Rules for connections with echo control devices*

2.3.2.1 *Ideal rules*

2.3.2.1.1 *Rule B*

1) Not more than the equivalent of one full echo suppressor (i.e. two half-echo suppressors) should be included in any connection needing an echo suppressor. When there is more than one full echo suppressor the conversation is liable to be clipped; lockout can also occur.
2) Circuits equipped with echo cancellers (Recommendation G.165) can be connected together in tandem without echo performance degradation.
3) A circuit equipped with echo suppressors (Recommendation G.164) can be connected with another circuit equipped with echo cancellers (Recommendation G.165) without additional performance degradation.

   Note – The overall performance will not be better than that provided by the poorer performing device.

2.3.2.1.2 *Rule D*

The half-echo suppressors should be associated with the terminating sets of the 4-wire chain of the complete connection. This:

- reduces the chance of speech being multilated by the echo suppressors because the hangover times can be very short;
- reduces the change of ineffective echo canceller operation as end delays are short and minimum required echo losses can be assured.

2.3.2.2 *Practical rules*

2.3.2.2.1 *Rule F*

If, as is appreciated, Rule D above cannot be complied with, the echo control device may be fitted at the international exchange or at an appropriate national transit centre. However, each echo control device should be located sufficiently near to the respective subscribers for the end delays not to exceed the maximum value recommended in Recommendation G.161, (*Orange Book*) and Recommendations G.164 and G.165 of this fascicle. For countries of average size, this will normally mean that the originating and terminating control devices will be in the countries of origin and destination of the call.

2.3.2.2.2 *Rule G*

In isolated cases a full short-delay echo suppressor may be fitted at the outgoing end of a transit circuit (instead of two half-echo suppressors at the terminal centres) provided that neither of the two hangover times exceeds 70 ms. This relaxation may reduce the number of echo suppressors required and may also simplify the signalling and switching arrangements. It is emphasized that full echo suppressors must not be used indiscriminately; the preferred arrangement is two half-echo suppressors as near the terminating sets as possible. A full echo suppressor should be as near to the "time-centre" of the connection as possible, because this will require lower hangover times.

Whether a full long-delay echo suppressor or canceller can be used in this circumstance is under study.

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[3] Recommendation Q.115 [4] is a study of the application of Rules A and E to the United Kingdom-European network relations.

2.3.2.2.3 *Rule K*

On a connection that requires an echo suppressor, up to the equivalent of two full echo suppressors (e.g. three half-echo suppressors or two half-echo suppressors and a full one) may be permitted. Every effort should be made to avoid appealing to this relaxation because the equivalent of two or more full echo suppressors, with long hangover times, on a connection can cause severe clipping of the conversation and considerably increases the risk of lockout. This rule does not apply to echo cancellers (see Rule B).

2.3.2.2.4 *Rule L*

In general it will not be desirable to switch out (or disable) the intermediate echo suppressors when a circuit equipped with long-delay echo control devices is connected to one with short-delay echo suppressors. However, it would be desirable to switch out (or disable) the intermediate echo suppressors if the mean one-way propagation time of that portion of the connection which would now fall between the terminal half-echo suppressors is not greater than 50 ms, since the different types are likely to be compatible. An intermediate echo canceller need not be switched out.

2.3.3 *General rules*

2.3.3.1 *Ideal rule – Rule C*

Connections that do not require echo control devices should not be fitted with them, because they increase the fault rate and are an additional maintenance burden.

2.3.3.2 *Practical rules*

2.3.3.2.1 *Rule H*

In exceptional circumstances, such as breakdown, an emergency route may be provided. The circuits of this route need not be fitted with echo control devices if they are usable without them for a short period. However, if the emergency routing is to last more than a few hours, echo control devices must be fitted according to Rules A to E above.

2.3.3.2.2 *Rule J*

It is accepted that a connection that does not require an echo control devices may in fact be unnecessarily equipped with one or two half-echo suppressors, or a full echo suppressor or echo cancellers. (The presence of an echo suppressor in good adjustment on a circuit with modest delay times can hardly be detected and in the case of echo cancellers it may improve the overall performance of the connection.)

Where a terminating international exchange is accessible from an originating international exchange by more than one route, and

1) at least one route requires echo suppressors, and at least one route does not; and
2) the originating exchange is unable to determine which route is to be used;

echo control devices should be connected in all cases.

2.3.3.2.3 *Rule M*

It has been found in actual practice that echo can be made tolerable by providing loss in the circuit if the one-way propagation time (delay) of the echo is less than about 25 ms. For delays longer than this, too much circuit loss is needed to attenuate echo, and echo control devices are required.

Note – The equivalent of this rule is stated in Recommendation G.161, § B.b. (*Orange Book*). This rule has not been expressed in earlier versions of Recommendation G.131.

2.4 *Insertion of echo control devices in a connection*

Ways of inserting echo control devices in a connection which have been considered are the following:

1) provide a pool of echo control devices common to several groups of circuits, and arrange for an echo control device to be associated with any circuit that requires one (see Recommendation Q.115 [4]);

2) arrange for the circuits to be permanently equipped with echo control devices but switch them out (or disable them) when they are not required (see [5]);

3) divide the circuits of an international route into two groups, one with and one without echo control devices and route the connection over a circuit selected from the appropriate group according to whether the connection merits an echo control device. However, it is recognized that circuits may not be used efficiently when they are divided into separate groups. This must be borne in mind;

4) conceive schemes in which the originating country and the terminal country are divided into zones at increasing mean radial distances from the international centre and determine the nominal lengths of the national extensions by examining routing digits and circuits-of-origin.

Whichever method is used, due regard must be paid to the last sentence of § 2.1 above. Methods of achieving the required reduction of circuit losses are under study by the CCITT. The nature and volume of the traffic carried by a particular connection will also influence the economics of the methods and hence the choice among them.

The CCITT is currently studying what recommendations are necessary to ensure that the insertion of echo control devices in international connections complies, overall, with the practical rules given above.

It should be appreciated that different continents need not use the same method although the methods must be compatible to permit intercontinental connections. There appears to be no great difficulty in arranging this.

2.5 *Speech processing devices*

Some speech processing devices, such as speech interpolation devices, have an inherent echo-suppressor function. However, such devices may only suppress echo during the single talk mode and not during double talking conditions (see Recommendation G.164, § 1.7) unless they are equipped to perform full echo-suppressor functions. When devices without full echo control are connected in tandem with echo cancellers, performance degradation due to echo may occur during double talking conditions as the intermediate echo canceller will not be effective during double talk.

ANNEX A

(to Recommendation G.131)

Application of Recommendation G.131, § 2

Recommendation G.131, § 2.3.1.1, Rule A, requires, for each pair of countries, an assessment of echo conditions for each possible pair of local exchanges to ascertain whether the plot of corrected reference equivalent of echo path against mean one-way propagation time for that pair of exchanges, lies above or below the appropriate 1% line in Figure 2/G.131.

The variables in the problem are indicated in Table A-1/G.131 and illustrated for all analogue connections in Figure A-1/G.131 and for all digital connections in Figure A-2/G.131.

For a given pair of exchanges, all eight items are known or can be estimated. A plot of overall loudness rating [1) + 2) + 3) + 4) of Table A-1/G.131] as a function of mean one-way propagation time [5) + 6) + 7) of Table A-1/G.131] on Figure 2/G.131 may be assessed in relation to the 1% curve, for a given number of analogue circuits in the 4-wire chain for fully analogue connections and mixed analogue/digital connections or, for fully digital connections using the appropriate curve.

TABLE A-1/G.131

Quantities needed for echo assessment

Overall loudness rating of the echo path, made up of the sum of:

1) the minimum of the sum of the values of the sending and receiving loudness ratings of the local system of country A (talker end));
2) the nominal loudness rating from, and to, the virtual analogue switching points (a_A and b_A) of the chain of national circuits in country A, connecting the local exchange to the international exchange;
3) the nominal loudness rating in each direction of transmission of the international chain;
4) the echo loss (a_B-b_B) of the national system of country B (listener end).

Mean one-way propagation time, made up of half the sum of the propagation times of:

5) the paths from the telephone set in country A, to and from the virtual analogue switching points a_A and b_A;
6) the two directions of transmission of the international chain;
7) the path a_B-b_B of country B.

In addition, there will be needed for fully analogue or mixed analogue/digital connections:

8) the number of analogue circuits in the 4-wire chain (see Figure 3/G.101).

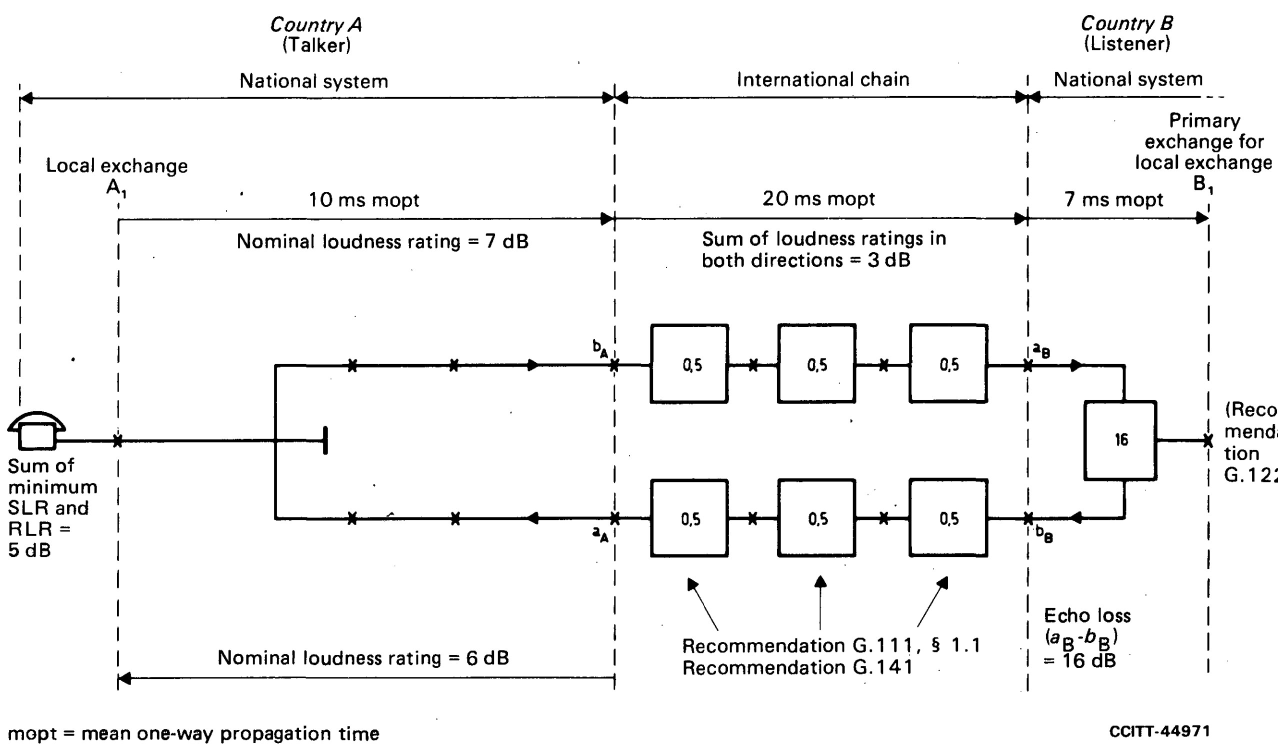

FIGURE A-1/G.131

Example of application of Figure 2/G.131 to fully analogue connections

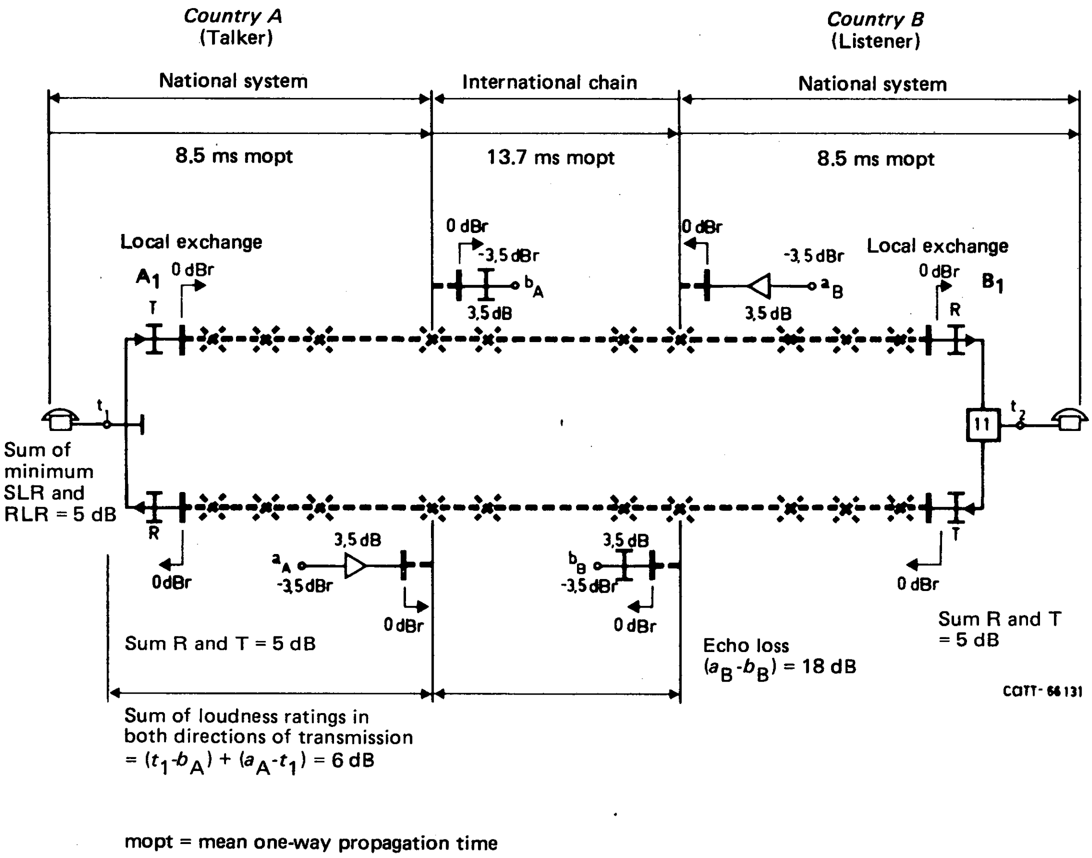

FIGURE A-2/G.131

Example of application of Figure 2/G.131 to fully digital connections with analogue 2-wire subscriber lines

A.1 *Full analogue connections* (Figure A-1/G.131)

For the purpose of this Recommendation, it may be assumed that the principal reflection at the listener's end occurs at the 4-wire/2-wire terminating set, which may be assumed to be located at the primary exchange associated with the listener's local exchange. The components of 4) of Table A-1/G.131 are then the losses a_B-t and t-b_B, plus the echo balance return loss at the 2-wire port of the terminating set. This return loss will be the mean overall, of the off-hook subscriber's lines, which may be presented to the 2-wire port of the terminating set by the listener's local exchange. (Figure 2/G.131 assumes that the standard deviation of the return loss is 3 dB.) If the mean value is not known, it may be assumed that 4) of Table A-1/G.131 is in accordance with Recommendation G.122, § 4, viz., a mean value of $(15 + S)$ dB where S is the sum of the nominal losses in the two directions of transmission of the circuits in the listener's national 4-wire chain (S is assumed to be 1 dB in this case).

For a given pair of local exchanges, successive connections may encounter different numbers of 4-wire circuits, and the total traffic may be regarded as a number of packets of various proportions encountering from one to nine 4-wire circuits. Each "packet" may be tested with the aid of Figure 2/G.131 and the results combined in order to assess whether Rule A is complied with for the totality of traffic.

Figure A-1/G.131 shows, as an example, an application of Recommendation G.131, § 2, where a listener's a-t-b path is assumed to be in accordance with Recommendation G.122. For simplicity, it is assumed that 100% of the traffic encounters the given conditions. Values for the example are as follows:

Talker's country A

Distance from local exchange A_1 to international exchange	1600 km
Mean one-way propagation time from local exchange A_1 to international exchange	11 ms [4]
Simultaneous-minimum sending and receiving loudness rating (sum) of the local system	5 dB
Loudness rating from local exchange to international exchange (b_A)	7 dB [5]
Loudness rating from international exchange to local exchange (a_A)	6 dB [5]
Number of 4-wire circuits	2

International chain A to B

Number of circuits	3 [6]
Distance	3200 km
Mean one-way propagation time	17 ms [4]
Sum of loudness ratings in both directions $2 \times 3 \times 0.5$ dB	3 dB

Listener's country B

Mean echo loss (a_B-b_B) = (15 + 1) dB	16 dB (Rec. G.122)
Distance from international exchange to primary exchange associated with local exchange B_1 (i.e. point of principal reflection)	1120 km
Mean one-way propagation time corresponding to above distance	16 ms [4]
Number of 4-wire circuits	1

Total number of 4-wire circuits = 2 + 3 + 1 = 6

Total mean one-way propagation time = 11 + 17 + 16 = 44 ms . . . . (A-1)

Total loudness rating of the echo path = 5 + 7 + 6 + 3 + 16 = 37 dB . . . . (A-2)

If (A-1) and (A-2) are plotted on Figure 2/G.131, the point lies below the 1% line for six 4-wire circuits, indicating a probability of more than 1% of incurring an "unsatisfactory" opinion. The conclusion also applies to other possible numbers of 4-wire circuits.

A.2 *Fully digital connections* (Figure A-2/G.131)

It may be assumed that the principal reflection at the listener's end occurs at the 4-wire/2-wire terminating set, which is located at the listener's local exchange. The components of 4) of Table A-1/G.131 are then the losses a_B-t and t-b_B plus the echo balance return loss at the 2-wire port of the terminating set. This return loss will be the mean, overall, of the off-hook subscriber's lines, which may be presented to the 2-wire port of the terminating set by the listener's local exchange. (Figure 2/G.131 assumes that the standard deviation of the return loss is 3 dB.) If the mean value is not known, it may be assumed that it is in accordance with Recommendation G.122, § 4.3, viz., a mean value of 11 dB.

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[4] Assuming a velocity of propagation for the transmission systems of 250 km/ms, 3 FDM channel modulators and demodulators of 1.5 ms each for talker's country A and the international chain of circuits A to B, and a 12 ms constant for listener's country B (see Recommendation G.114).

[5] It is assumed that the loaded trunk-junction introduces an additional 1 dB (in each direction) when changing from nominal transmission loss to loudness rating.

[6] An unusually large number, chosen only to illustrate the principle of addition of loss.

In order to apply Figure A-2/G.131 the value of n is not required in this case (as the digital circuits in the 4-wire chain do not contribute to the overall circuit loss variability). However, the number of digital exchanges has an effect on the propagation time, for instance, in accordance with Table 1/G.114, that each digital transit exchange adds 0.45 ms to the mean one-way propagation time of the connection.

Figure A-2/G.131 shows an example where the sum of the R and T pads is either 6 or 7 dB. Values for the example are as follows:

Talker's country A

Distance from local exchange A_1 to international exchange	1600 km
Mean one-way propagation time from local exchange A_1 to international exchange	8.5 ms [7)]
Simultaneous-minimum sending and receiving loudness rating (sum) of the local system	5 dB
Sum of loudness ratings in both directions of transmission $(t_1\text{-}b_A) + (a_A\text{-}t_1)$	6 dB

International chain A to B

Distance	3200 km
Mean one-way propagation time	13.7 ms [8)]
Loudness rating of international chain	0 dB

Listener's country B

Distance from local exchange B_1 to international exchange	1600 km
Mean one-way propagation time	8.5 ms [7)]
Mean echo loss $(a_B\text{-}b_B) = (11 + 7)$ dB	18 dB
Total mean one-way propagation time = 8.5 + 13.7 + 8.5 = 30.7 ms	(A-3)
Total loudness rating of the echo path = 5 + 6 + 0 + 18 = 29 dB	(A-4)

If (A-3) and (A-4) are plotted on Figure 2/G.131, the point lies below the 1% line (and also the 10% line) for fully digital connections, indicating a propability of more than 1% incurring an "unsatisfactory" opinion.

Conclusion

a) An echo control device should be used on the connection; or

b) the loss in the echo path should be increased (but the limitations of Recommendation G.121 must be observed).

Note – It should be noted, when contemplating to increase the loss in the echo path, that digital pads placed in digital circuits need to be switched out for digital data signals (but not for voiceband data signals) as they destroy the bit transparency for such signals.

A.3 *Mixed analogue/digital connections*

The examples given in Figures A-1/G.131 and A-2/G.131 allow the construction of mixed analogue/digital connection models by combining the appropriate elements of the two figures. The quantities stated in Table A-1/G.131 can be calculated with these models. (Quantity 8) of this table (number of circuits) should now be taken as the number of analogue circuits in the 4-wire chain (thus not including the digital circuits). The appropriate solid curve in Figure 2/G.131 will approximate the required echo tolerance curve with good accuracy.

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7) Assuming a velocity of propagation for the transmission systems of 250 km/ms, 4 exchange delays of 0.45 ms each and 0.3 ms delay in the coder or decoder. (In practice a local digital exchange will contribute a little more than 0.45 ms, see Recommendation G.114.)

8) Assuming a velocity of propagation for the transmission systems of 250 km/ms and 2 exchange delays of 0.45 ms each.

Note – In mixed analogue/digital networks the propagation time can become larger than in purely analogue or digital networks. The latter occurs in particular when digital exchanges are connected with analogue transmission systems through PCM/FDM equipments in tandem or transmultiplexers. Many different configurations may arise.

References

[1] *Calculation of the stability of international connection established in accordance with the transmission and switching plan*, Green Book, Vol. III, Supplement No. 1, ITU, Geneva, 1973.

[2] CCITT Recommendation *Definitions relating to echo suppressors and characteristics of a far-end operated, differential, half-echo suppressor*, Blue Book, Vol. III, Rec. G.161, Section B, ITU, Geneva, 1964.

[3] CCITT Recommendation *Echo-suppressors suitable for circuits having either short or long propagation times*, Orange Book, Vol. III, Rec. G.161, Sections B and C, ITU, Geneva, 1977.

[4] CCITT Recommendation *Control of echo suppressors*, Vol. VI, Rec. Q.115.

[5] CCITT – *Insertion and disablement of echo suppressors*, Blue Book, Volume VI.1, Question 2/XI, Annex 3, ITU, Geneva, 1966.

Recommendation G.132

ATTENUATION DISTORTION

(Geneva, 1964; Mar del Plata, 1968; Geneva, 1972 and Melbourne, 1988)

The network performance objectives for the variation with frequency of transmission loss in terminal condition of a worldwide 4-wire chain of 12 circuits (international plus national extensions), each one routed over a single group link, are shown in Figure 1/G.132, which assumes that no use is made of high-frequency radio circuits or 3-kHz channel equipment.

Note 1 – The design objectives contained in the Recommendation cited in [1], for carrier terminal equipments are such that for a chain of 6 circuits (international and national extensions) in tandem, each circuit being equipped with one pair of channel translating equipments, the attenuation distortion would in most cases be less than 9 dB between 300 and 3400 Hz. For the case of 12 circuits in tandem it can be expected that in most cases the attenuation distortion will not exceed 9 dB between about 400 and 3000 Hz. As far as the international chain is concerned, see Recommendation G.141, § 1.

Note 2 – It is only in a small proportion of international connections that the 4-wire chain will in fact comprise 12 circuits.

Note 3 – Limits given in Figure 1/G.132 should be met also for mixed connections using the analogue-digital equipments. Probably, the number of analogue-digital equipment (pair codecs) for the mixed connections with 12 circuits does not exceed 6 (see Recommendation G.103, Annex B).

It should be recognized that a connection containing six coder-decoder pairs where each pair just meets the attenuation distortion requirements found in Recommendation G.712 will not meet the attenuation distortion requirement found in Recommendation G.132 for 3400 Hz.

However, it is likely that real coder-decoder pairs will have attenuation distortion performance better than in Recommendation G.712, so for practical purposes the likelihood of not complying with Recommendation G.132 is very small.

Note 4 – Studies are being carried out by Study Group IV and Study Group XII about how well this objective is being met in practice, about the expectation with which it should be met in future (taking account of Note 2 and Note 3 and about any possible consequential need for notifications to Recommendations referring to equipments.

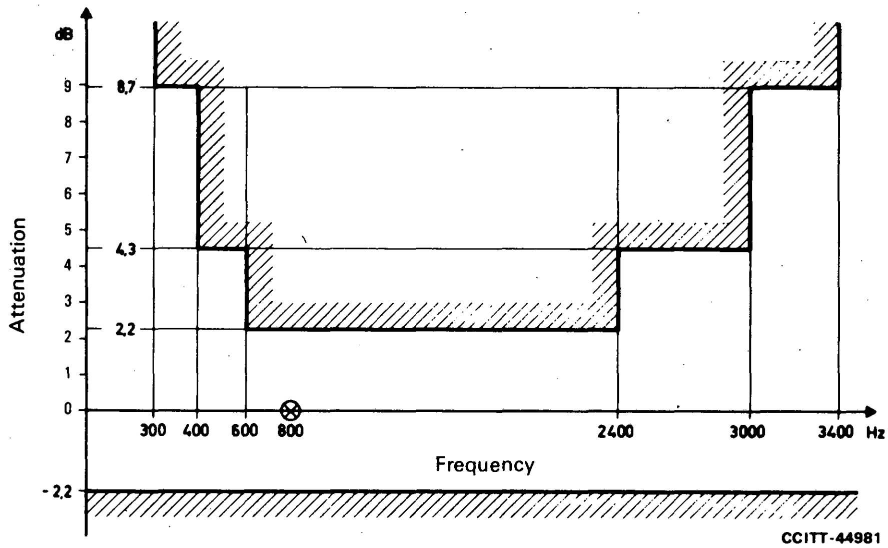

FIGURE 1/G.132

Permissible attenuation variation with respect to its value measured at 800 Hz (objective for worldwide 4-wire chain of 12 circuits in terminal service)

Reference

[1] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232, § 1.

Recommendation G.133

GROUP-DELAY DISTORTION

(Geneva, 1964; amended at Geneva, 1980)

The network performance objectives for the permissible differences for a worldwide chain of 12 circuits each on a single 12-channel group link, between the minimum group delay (throughout the transmitted frequency band) and the group delay at the lower and upper limits of this frequency band are indicated in the Table 1/G.133.

Group-delay distortion is of importance over a band of frequencies where the attenuation is of importance, i.e. at which the attenuation is less than 10 dB relative to the value at 800 Hz. This will normally be the case for frequencies higher than about 260-320 Hz and lower than about 3150-3400 Hz respectively for the lower and upper limit of the frequency band as indicated in Table 1/G.133.

TABLE 1/G.133

	Lower limit of frequency band (ms)	Upper limit of frequency band (ms)
International chain	30	15
Each of the national 4-wire extensions	15	7.5
On the whole 4-wire chain	60	30

Note – Limits given in Table 1/G.133 should be met both for analogue circuits and mixed circuits with analogue and digital sections.

LINEAR CROSSTALK[1)]

(Geneva, 1964; amended at Mar del Plata, 1968)

1 Linear crosstalk between different 4-wire chains of circuits (analogue and mixed)

As a network performance objective, the signal-to-crosstalk ratio which may exist between two 4-wire chains of circuits comprising international and national circuits is restricted by Recommendation G.151, § 4.1, as regards circuits, and by Recommendation Q.45 [1], as regards international centres.

2 Linear crosstalk between go and return channels of the 4-wire chain of circuits (analogue and mixed)

As a network performance objective, the signal-to-crosstalk ratio between the two directions of transmission of a 4-wire chain of circuits is restricted by Recommendation G.151, § 4.2, as regards circuits and by Recommendation Q.45 [1] as regards international centres.

ANNEX A

(to Recommendation G.134)

Methods for measuring crosstalk in exchanges, on international telephone circuits and on a chain of international telephone circuits

A.1 The method used for measuring crosstalk will depend on the type of crosstalk. In general one or the other of the following two situations will be encountered:

a) crosstalk in an exchange arising mainly from a single source or from several nearby sources;

b) crosstalk measured at the end of a circuit or chain of circuits and which is the result of multiple sources of crosstalk occurring at points along the circuit or chain of circuits. The total crosstalk will depend on the relative phases of the individual contributions and may accordingly vary greatly with frequency. On long circuits or chains of circuits, difficulties may arise when making crosstalk measurements at a single frequency owing to small variations in the frequency of the master oscillators supplying translating equipment at various points along the circuit or chain of circuits.

A.2 Available methods for measuring crosstalk are as follows[2)]:

a) single-frequency measurements (e.g. at 800 Hz or 1000 Hz);

b) measurements made at several frequencies (e.g. at 500, 1000 and 2000 Hz), the results being averaged on a current or voltage basis;

c) measurements made using a uniform spectrum random noise or closely spaced harmonic series signal shaped in accordance with a speech power density curve. Such measurements should be made in accordance with the Recommendation cited in [3];

d) voice/ear tests, in which speech is used as the disturbing source and the crosstalk is measured by listening and comparing its level with a reference source whose level can be adjusted by some form of calibrated attenuating network.

1) Recommended methods for the measurement of crosstalk are described in Annex A.

2) It is a question here of the measurement of the frequency (or frequencies) to be used; the measure of the crosstalk for a given frequency is described in [2]

A.3 Pending further study, the following methods are provisionally recommended for "type tests" and "acceptance tests" involving crosstalk measurement.

A.3.1 *Crosstalk in exchanges*

Crosstalk should be measured at 1100 Hz which, in the experience of some Administrations, is equivalent to a measurement made with a conventional telephone signal generator (Recommendation G.227 [4]) and a psophometer.

A.3.2 *Crosstalk on an international telephone circuit or chain of international telephone circuits*

Crosstalk should be measured using a uniform spectrum random noise or closely spaced harmonic series signal shaped in accordance with the speech power density curve of Recommendation G.227 [4]. The measurements should be made in accordance with the Recommendation cited in [3].

Note 1 – In cases of difficulty with A.2.a) and A.2.b), voice/ear tests are recommended.

Note 2 – In the case of telephone circuits used for voice-frequency telegraphy the near-end signal-to-crosstalk ratio between the two directions of transmission should be measured at each of the telegraph channel carrier frequencies, i.e. at each odd multiple of 60 Hz from 420 Hz to 3180 Hz inclusive. However, difficulty can arise in practice because of the effect mentioned in A.1.b) above.

References

[1] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45.

[2] *Measurement of crosstalk*, Green Book, Vol. IV.2, Supplement No. 2.4, ITU, Geneva, 1973.

[3] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232, § 9.2.

[4] CCITT Recommendation *Conventional telephone signal*, Vol. III, Rec. G.227.

Recommendation G.135

ERROR ON THE RECONSTITUTED FREQUENCY

(Mar del Plata, 1968)

As the channels of any international telephone circuit should be suitable for voice-frequency telegraphy, the network performance objective for the accuracy of the virtual carrier frequencies should be such that the difference between an audiofrequency applied to one end of the circuit and the frequency received at the other end should not exceed 2 Hz, even when there are intermediate modulating and demodulating processes.

To attain this objective, the CCITT recommends that the channel and group carrier frequencies of the various stages should have the accuracies specified in the corresponding clauses of Recommendation G.225 [1].

Experience shows that, if a proper check is kept on the operation of oscillators designed to these specifications, the difference between the frequency applied at the origin of a telephone channel and the reconstituted frequency at the other end hardly ever exceeds 2 Hz if the channel has the same composition as the 2500-km hypothetical reference circuit for the system concerned.

Calculations indicate that, if these recommendations are followed, in the 4-wire chain forming part of the hypothetical reference connection defined in Figure 1/G.103 [1)] there is about 1% probability that the frequency difference between the beginning and the end of the connection will exceed 3 Hz and less than 0.1% probability that it will exceed 4 Hz.

1) In fact, the chain considered for these calculations comprised 16 (instead of 12) modulator-demodulator pairs to allow for the possibility that submarine cables with equipments in conformity with Recommendation G.235 [2] might form part of the chain. No allowance was made, however, for the effects of Doppler frequency-shift due to inclusion of a non-stationary satellite; values for this shift are given in CCIR Report 214 [3].

The CCITT notes that in mixed circuits having several digital sections the requirements concerning frequency error are met more easily since digital systems do not change the frequency of an audio frequency channel.

References

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

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

[3] CCIR Report *The effects of doppler frequency-shifts and switching discontinuities in the fixed satellite service,* Vol. IV, Report 214, ITU, Geneva, 1986.

1.4 General characteristics of the 4-wire chain of international circuits; international transit

Recommendation G.141

ATTENUATION DISTORTION

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

1 Attenuation distortion

1.1 *All-analogue conditions*

The design objectives recommended for carrier terminal equipment by the Recommendation cited in [1] are such that for a chain of six circuits, each equipped with a single pair of channel translating equipments in accordance with that Recommendation, the network performance objective for the attenuation distortion given by Figure 1/G.132 will in most cases be met. The distortion contributed by the seven international centres is thereby included.

Note – To assess the attenuation distortion of the international chain, the limits indicated for international circuits in Recommendation G.151, § 1 must not be added to the limits for international centres mentioned in Recommendation Q.45 [2]. In fact, on the one hand, some exchange equipment would be counted twice if this addition were made; on the other, the specification limits of Recommendation Q.45 [2] apply to the worst possible connection through an international exchange, while the maintenance limits of Recommendation G.151, § 1 apply to the poorest international circuit. The specifications of the various equipments are such that the mean performance will be appreciably better than could be estimated by the above-mentioned addition.

1.2 *Mixed analogue/digital conditions*

In the mixed analogue/digital period, it is expected that the attenuation/frequency characteristics of the analogue carrier terminal equipment that is to be used in international telephone connections will continue to be governed by existing Recommendations that are relevant to this type of circuit.

Where unintegrated PCM digital processes are to be included in international telephone connections, it is recommended that the attenuation/frequency characteristic of the bandpass filters associated with such processes should comply with the more stringent version of Figure 1/G.712 [3]. The latter Recommendation applies specifically to cases where integrated PCM digital processes are associated with trunk junctions (toll connecting trunks), trunk circuits (intertoll trunks), and international circuits.

With regard to the incorporation of unintegrated PCM digital processes in local telephone networks, the required attenuation/frequency characteristics of the bandpass filters involved are still under study.

References

[1] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232, § 1.

[2] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45.

[3] CCITT Recommendation *Performance characteristics of PCM channels between 4-wire interfaces at voice frequencies*, Vol. III, Rec. G.712, Figure 1/G.712.

Recommendation G.142

TRANSMISSION CHARACTERISTICS OF EXCHANGES

(Geneva, 1980; amended at Melbourne, 1988)

This Recommendation consists of two parts. The first part, § 1, is concerned with the voice-frequency transmission characteristics of international analogue exchanges. The information involved is encompassed within Recommendation Q.45 [1]. The second part, § 2, is concerned with the voice-frequency transmission considerations that should be taken into account in the design of digital exchanges and their incorporation into the network. The digital exchanges referred to include local exchanges and transit exchanges (national and international). The transmission considerations relate primarily to the properties which digital exchanges should possess to enable them to operate under different and changing network conditions with respect to the content of analogue, mixed analogue/digital and all-digital plant.

Detailed transmission characteristics for digital exchanges are contained in Recommendations Q.551, Q.552, Q.553 and Q.554 (Fascicle VI.5).

1 International analogue exchange

The commissioning objectives for the transmission requirements to be respected by an international analogue exchange are included in Recommendation Q.45 or Q.45 *bis*.

2 Digital exchanges

2.1 *Digital processes – Effect on transmission*

Digital (TDM) exchanges, to varying degrees, are required to include such digital processes as analogue-to-digital coders, digital-to-analogue decoders and digital recoding processes, examples of which are companding law converters and digital pads. The extent to which such digital processes might be included in a digital exchange is determined by the network environment in which the exchange is to operate (i.e., all-analogue, mixed analogue/digital or all-digital).

Digital processes such as those referred to above, attract transmission penalties. These penalties can be expressed in terms of "units of transmission impairment".

A limit is placed on the permissible accumulation of units of transmission impairment in an international telephone connection. Details of the planning rule resulting from this limit and the penalties introduced by individual digital processes are given in Recommendations G.101, § 4 and G.113, § 3.

In accordance with Recommendation G.113, § 3 it is provisionally recommended that no more than 14 units of transmission impairment be permitted to accumulate in an international connection. Of these 14 units, a maximum of 5 units could be introduced by each national extension and a maximum of 4 units by the international portion. Since one 8-bit PCM codec pair (coder and decoder) introduces 1 unit of transmission impairment, it is clear that unintegrated PCM digital processes involving analogue/digital conversions, (e.g. codecs) or digital processes involving the recoding of information (e.g. digital pads) should not be allowed to proliferate in an uncontrolled fashion. Figure 1/G.142 shows some of the transmission paths that might be established through a digital exchange and the "units of transmission impairment" attributable to the digital processes in these paths.

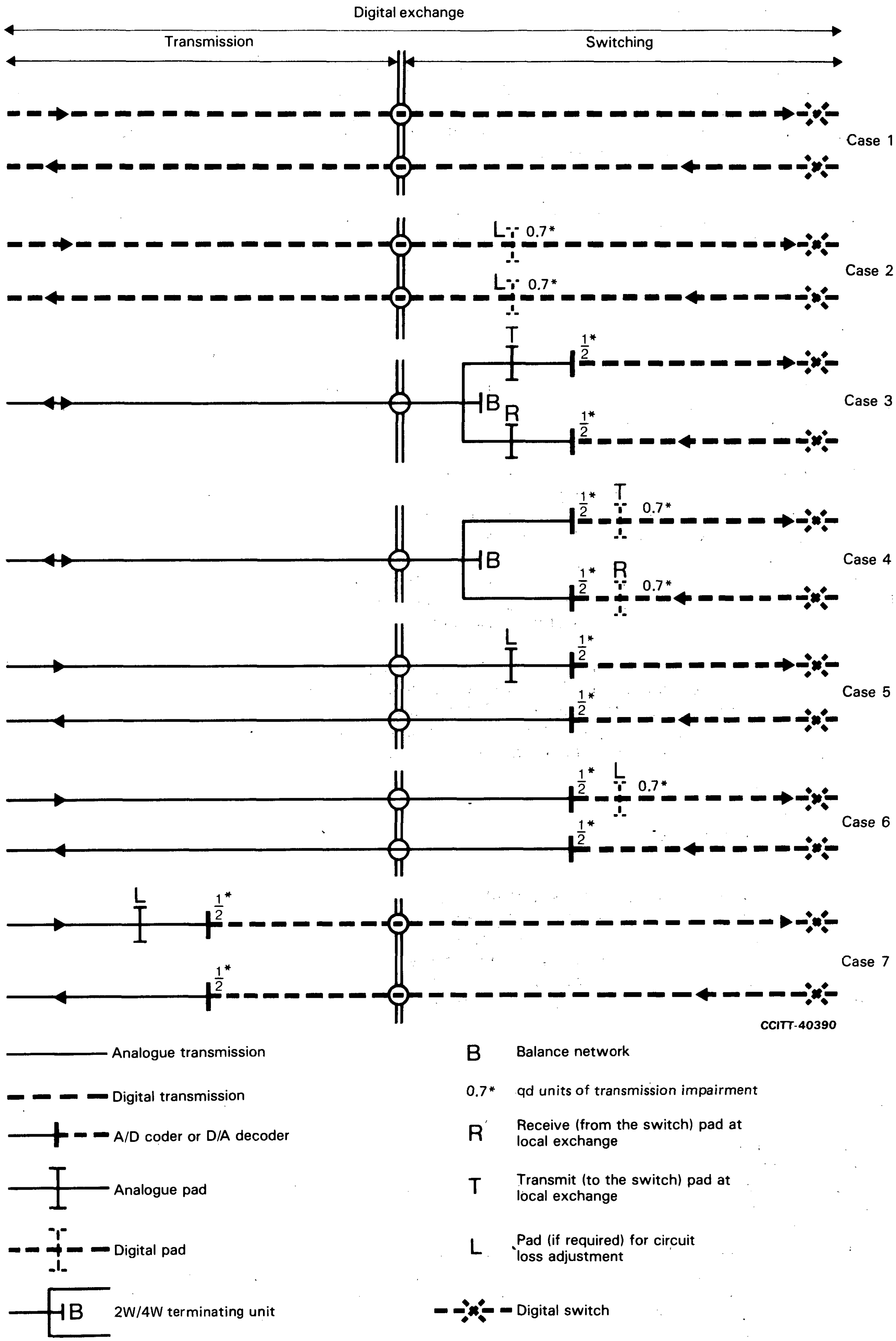

FIGURE 1/G.142

Transmission paths at digital exchanges

2.2 *Transmission loss through a digital exchange*

The 4-wire digital switching function at a digital exchange should introduce a nominal transmission loss of 0 dB. Thus, in Figure 1/G.142 (Case 1) if a 0 dBm0 sinusoidal test signal is introduced at the analogue terminals of an ideal coder connected to the input of a digital switch, a Digital Reference Sequence (DRS) should be transmitted unaltered through the switch and produce a 0 dBm0 sinusoidal signal at the analogue terminals of a decoder connected to the output of the digital switch.

Except for the transmission loss considered above (and perhaps the possible loss due to exchange wiring) all transmission losses which are to be introduced by a digital exchange, either in a digital or analogue form, are to be governed by the applicable transmission plan (see § 2.4 below).

2.3 *Relative levels*

On digital paths within an all-digital network, relative levels have no real meaning or use. However, as long as a substantial portion of the worldwide telephone network is of an analogue nature, it is necessary and useful to assign relative levels to digital exchanges.

The relative levels assigned to a digital exchange are applicable at the virtual analogue switching points of the exchange. The virtual analogue switching points are theoretical points as explained in Recommendation G.101, § 5.1. The concept of applying relative levels at the virtual analogue switching points of a digital exchange is dealt with in Recommendations G.101, § 4.2 and G.101, § 5.2.

In accordance with Recommendation G.101, § 5.2 the send relative level at an international digital exchange should be −3.5 dBr. In the case of digital exchanges in national extensions, the send relative levels should be governed by the applicable national transmission plan.

With regard to the receive relative level at a digital exchange, this level is related to the transmission loss of the circuits terminating at the exchange. In the case of an international digital exchange, it is desirable to have the receive relative level at −3.5 dBr to avoid having to introduce digital pads. But see the general Note in Recommendation G.101, § 4.2 for exceptions. In the case of national extensions, the receive relative levels, as in the case of the send relative levels, are to be determined on the basis of the applicable national transmission plan.

2.4 *Echo and stability control*

The overall echo and stability losses presented by a national extension are a function of the relevant transmission losses and, in the case of the use of 2-wire conversion circuits, the balance return loss introduced by the 2-wire/4-wire conversion circuit. Both contributions need to be considered in the design of digital local exchanges where there is generally scope for improving the echo and stability losses. Such improvements are likely to be needed as connections in digital networks will tend to have lower losses and longer delays than analogue connections with a consequent worsening in echo performance.

2.4.1 *Transmission loss contribution*

The requirements for controlling stability and echo on international connections under all-digital or mixed analogue/digital network conditions are dealt with in Recommendation G.122. In accordance with the latter Recommendation, the national extensions are to be mainly responsible for effecting this control. Arrangements for doing so are dealt with in Recommendation G.121, § 6.

Recommendation G.121, § 6 provides the framework within which individual national transmission plans are to provide for the necessary features to effect the required control. In the case of a digital 4-wire national extension (i.e., all-digital down to the local exchange but with 2-wire analogue subscriber lines), the control can be effected entirely at the local exchange. Where the national extension is to be of a mixed analogue/digital nature, the control under some national transmission plans might be distributed among the different parts of the national extension but the main burden would in general still lie with the local exchange. Figure 1/G.142 contains examples of some of the different arrangements that might be encountered at a digital exchange.

The arrangement in Case 1 of Figure 1/G.142 deals with the termination of a digital circuit at what might be a national or international digital exchange. In this particular case, the circuit is to be operated without introducing additional loss at the exchange.

The arrangement in Case 2 of Figure 1/G.142 also deals with the termination of a digital circuit at a national or international digital exchange. However, in this case, the relevant transmission plan requires that loss should be associated with the circuit at the exchange through the medium of digital pads. See § 2.6 below regarding the use of digital pads.

The arrangement in Case 3 of Figure 1/G.142 deals with the termination of a 2-wire subscriber's line at a digital local exchange. The pads designated R and T are pad symbols intended to represent loss or level adjustment made in the analogue portion. Recommendation G.121, § 6 is concerned with the appropriate choice of values for R and T.

The arrangement in Case 4 of Figure 1/G.142 is similar to that of Case 3 except that the losses R and T are shown as being provided in the digital portion. See § 2.6 below regarding the use of digital pads.

The arrangement in Cases 5, 6 and 7 of Figure 1/G.142 deals with the termination of analogue circuits at a national or international digital exchange. In Case 5, an analogue pad (L) is used to develop the required loss of the circuit in accordance with the relevant transmission plan. Case 6 is similar to Case 5 except that a digital pad (L) is used to develop the required circuit loss. Case 7 is also similar to Case 5 except that the analogue pad (L) as well as the A/D coder and D/A decoder are provided as part of the transmission equipment associated with the circuit rather than by equipment that is built-in as part of the switching system. Although not shown in Figure 1/G.142, the A/D coders, the D/A decoders, the 2-wire/4-wire terminating units and the pads involved in Cases 2, 3 and 4 can also be provided as part of the transmission equipment on the transmission side of the exchange rather than by equipment that is built-in as part of the switching system.

2.4.2 *Balance return loss contribution*

The contribution of balance return loss to the overall echo and stability losses is illustrated in Cases 3 and 4 of Figure 1/G.142 which show the situation of 2-wire local lines terminating on a digital local exchange. The achieved balance return loss is determined by the match between the impedance presented by the 2-wire local line and customer terminating apparatus and the balancing impedance chosen for the digital exchange line card.

In many designs of digital local exchange there is no 2-wire switch and the 2-wire line is permanently connected to the line card. This arrangement has significant advantages for balance return loss as there is likely to be a significant reduction in the range of impedances presented to any single line card. It is then possible to choose a line card balancing impedance more closely matched to the local line impedances and obtain an improvement in balance return loss compared with the conventional compromise impedances.

The optimum balancing impedance will not be the same for all Administrations as it needs to take into account the local cable types used together with the range of customer apparatus impedances. It is possible that the use of different exchange balancing impedances for different local line classes will give an improvement in performance at the expense of some increase in network Administration. In general it has been found that the use of balancing networks which resemble the impedance presented by local cable give the optimum performance. Examples of balancing impedances adopted by a number of Administrations are given in Recommendation Q.552.

Further improvement in balance return loss is possible where the impedance of the customer apparatus can be influenced by the Administration. Telephone instruments with an input impedance close to the impedance of the local cable can result in an improvement in the balance return loss at the digital local exchange in the order of 10 dB on short local lines.

2.5 *Local transmission*

On local calls between subscribers served by the same digital local exchange, the switching of 2-wire subscriber lines such as those shown in Figure 1/G.142, Case 3, results in an equipment arrangement which takes on the appearance of a voice-frequency repeater – see Figure 2/G.142. As is well known, such an arrangement must include sufficient loss around the loop to provide for an adequate margin of stability. To provide for this loss, some 2-wire to 2-wire attenuation may be acceptable in some cases. The attenuation might be supported by the national transmission plan, as it provides adequate loudness rating distribution for local calls. However, in cases where the 2-wire to 2-wire attenuation is to be comparable to that generally prevailing at an analogue exchange, i.e., approximately 0 dB, adequate balance return losses must be provided at the 2-wire/4-wire junctions. This could entail increasing the existing values of balance return loss at these points. Methods for doing this are under study by Study Group XII.

Increasing the balance return losses as referred to above should also be beneficial to the control of echo and stability in national connections beyond the local exchange as well as on international connections.

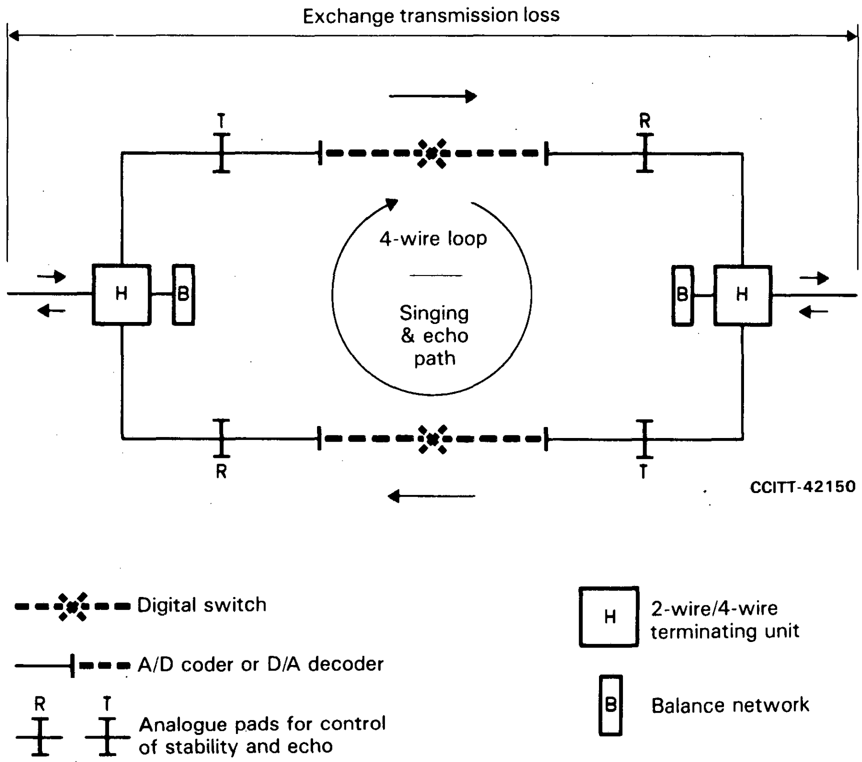

FIGURE 2/G.142

Configuration of digital local exchange on 2-wire to 2-wire connections

2.6 *Sidetone and input impedance*

Digital local exchanges can have a significant influence on the sidetone performance of telephone instruments, particularly those instruments on relatively short local lines. The reason for this can be seen in Figure 2/G.142 where the impedance presented by the exchange to the local line is a function of the input impedance of the line card and the characteristics of the singing and echo path within the exchange.

For optimum sidetone performance on short local lines the input impedance of the exchange line card should be close to the anti-sidetone impedance of the telephone instrument. In the case where the telephone instrument is designed to give good sidetone performance on long local lines this anti-sidetone impedance is likely to be close to the characteristic impedance of the 2-wire local cable. This would lead to the digital local exchange also presenting an impedance close to that of the 2-wire local cable.

On longer local lines the exchange impedance will have less effect on the sidetone performance as the impedance presented to the telephone is masked by the local cable impedance.

The final choice of exchange impedance needs to take into account a number of factors:

- telephone set impedance and sensitivity characteristics;
- local line network characteristics;
- digital exchange current feeding arrangements,

the objective being that the customer should not see a worsening in sidetone performance when connected to a digital exchange. The impedance chosen by a number of Administrations are given in Recommendation Q.552 and it is clear that there is a considerable difference between the impedances which reflects the differences between the national networks.

2.7 *Digital pads*

The use of a digital pad to produce the required transmission loss in a digital path attracts a transmission penalty. This penalty has to come out of the allowance of "units of transmission impairment" allotted to the national and international portions of international connections – see Recommendation G.113, § 3. Additionally, since digital pads involve the use of digital recoding processes, the use of such pads in paths where bit integrity must be preserved is unattractive. This can be an important consideration where multipurpose networks are contemplated. Consequently, if digital pads must be introduced, arrangements should be made to switch them out or to bypass them.

2.8 *Transmission delay*

Transmission delays through digital exchanges could be significant. For example, such delays could have the effect of decreasing the length of connections on which echo control devices (e.g., echo suppressors or echo cancellers) should be applied. Transmission delays at digital local exchanges (or at digital PBXs) could in some cases also affect the impedance match between subscriber lines and the exchange (or PBX) in a way that could adversely affect subscriber sidetone. Transmission delays through digital exchanges should, therefore be minimized. See Recommendation G.114, § 2 for details of the delay introduced by various items of digital equipment and systems.

For transmission delays that might be encountered at digital exchanges; see Recommendation Q.551.

Reference

[1] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45.

Recommendation G.143

CIRCUIT NOISE AND THE USE OF COMPANDORS

(Geneva, 1964; amended at Mar del Plata, 1968; Geneva, 1972 and 1980 and Malaga-Torremolinos, 1984)

1 Noise objectives for telephony

1.1 *Principle*

Taking into account the network performance objectives for noise allowed in national networks (Recommendation G.123), it is desirable that the circuit performance objective for the mean psophometric power in any hour of the total noise generated by a chain of six international circuits, some of which may exceed 2500 km in length, on a connection used for international telephone calls, should not exceed 50 000 picowatts referred to a zero relative level point of the first circuit in the chain (level −43 dBm0p).

Of course, a lower value of the total noise may be expected when the international chain consists of only a small number of international circuits, not exceeding 2500 km in length and conforming to Recommendation G.152 (in particular, the circuit performance objective for the noise of such circuits is that the mean psophometric power in any hour does not exceed 10 000 pW at a zero level point on the circuit, level −50 dBm0p).

However, as connections longer than 25 000 km will be set up, the CCITT recommends, as an objective, that on sections longer than 2500 km used for international traffic, line equipment be supplied with a circuit performance objective for noise not greatly exceeding L picowatts on a circuit L km long (see [1]). There is obvious advantage in working to the same standard on short sections when this can reasonably be done.

Note 1 – Noise objectives for maintenance purposes are the subject of Recommendation M.580 [2]. Table 4/M.580 of that Recommendation is reproduced here:

TABLE 4/M.580

Maintenance noise objectives for public telephone circuits

Distance (km)	< 320	321 to 640	641 to 1600	1601 to 2500	2501 to 5000	5001 to 10 000	10 001 to 20 000
Noise (dBm0p)	−55	−53	−51	−49	−46	−43	−40

Note 2 – Strictly speaking, the noise objective for communication-satellite systems (see Recommendation G.153, § 3) cannot be expressed in the form of a given number of picowatts per km. See also the Note of Recommendation M.580 [2].

1.2 *Noise produced by equipment*

The equipment design objective for noise produced by the modulating equipment in the international chain of circuits in the longest hypothetical reference connection (see Figure 1/G.103) can be estimated on the assumption that such equipment comprises:

– 6 channel-modulation pairs, or 8 to 10 if 3-kHz-spaced channel equipment is used on transoceanic routes;
– 12 to 14 group-modulation pairs;
– 18 to 24 supergroup-modulation pairs;

for all of which a total circuit performance for the combined psophometric power of 5000 to 7000 pW0p (at a point of zero relative level on the first circuit of the international chain of 4-wire circuits) is a generous assumption.

The equipment design objective of −67 dBm0p for the hourly-mean psophometric power level at each international switching point quoted in Recommendation Q.45 [3] is equivalent to about 2000 pW0p at a point of zero relative level on the first circuit in the 4-wire chain.

It may thus be seen that the equipment design objective for the noise produced by the equipment does not constitute a large part of the network performance objective for the total noise generated by the international chain.

1.3 *Division of the overall circuit performance objective for noise*

The land sections in the international chain, set up on cable carrier systems or on radio-relay links, should in principle afford circuits of the quality defined above. In practice, by agreement between Administrations, the circuit performance objective for noise could be shared between the submarine and overland systems in such a way that the submarine cable systems contribute at a somewhat lower rate, e.g. 1 pW/km, and the overland systems contribute at a somewhat higher rate, e.g. a maximum of 2 pW/km. This result may be achieved either by setting up special systems, or by a proper choice of channels in systems designed to the 3 pW/km objective.

Note – In some countries, overland systems forming part of a circuit substantially longer than 2500 km (e.g. 5000 km or more) have been constructed with the same circuit performance objective for noise as the submarine cable system, i.e. 1 pW/km.

1.4 *Circuits operated with speech concentrators* [1)]

It would be desirable for all the circuits making up a group for use with a concentrator system to have approximately the same noise power level under operating conditions.

2 Use of syllabic compandors [2), 3), 4)]

For many years, international (and national) circuits will continue to be provided on existing transmission systems which have been designed to other standards, e.g. 4 pW/km, as given in Recommendation G.152. Furthermore, the circuit noise produced by transmission systems can increase above the values originally achieved because of ageing effects, and changes of system loading. There is therefore a need for a simple practical criterion that can be applied for planning purposes to an international circuit to determine if, as far as noise power is concerned, it is suitable for establishing multicircuit worldwide telephone connections or whether it can be made suitable by fitting compandors [2)].

It is recommended that, for the present, the systematic use of compandors conforming to Recommendation G.162 in the long-distance national and international network be restricted.

Compandors conforming to Recommendation G.166 may be used in the network provided planning is done to minimize the number of compandored circuits in tandem. It is desirable to have at most one compandored circuit in a connection. Preliminary results obtained by one Administration indicate that for voice operation no more than three compandored circuits in tandem should be allowed. Some high speed modems (9.6 kbit/s) may experience difficulty on a connection with even one compandored circuit. To ensure compliance not more than one compandored circuit should be used in the international segment. Additional information is required before a firm planning rule can be established including possible application in national extensions on circuits with moderate noise levels.

It must be pointed out that the action of a compandor doubles the effect of any variations in the transmission loss occurring in that part of the circuit which lies between the compressor and the expander and for this reason compandors, if needed, should be fitted at the ends of circuit sections provided by inherently stable line transmission systems such as submarine cable systems.

The following planning rule is recommended by the CCITT as a guide for deciding whether an international circuit requires a compandor:

If the hourly-mean psophometric circuit noise power level of an international circuit substantially longer than 2500 km (e.g. 5000 km or more) is less than −44 dBm0p (at a point of zero relative level on the circuit) no compandor is necessary.

If the circuit noise power level is −44 dBm0p (40 000 pW0p) or greater, a compandor should be fitted.

It is, of course, to be understood that circuits of length 2500 km or less will always meet the appropriate general noise objectives (Recommendation G.222 [4]) without the need for compandors.

Note 1 – This rule has been devised to make possible the planning of the international telephone network, using presently available circuits. It should in no way be interpreted as relaxation of the design objectives recommended in § 1 of this Recommendation, nor should it be applied for maintenance purposes (see Note 1 of § 1.1 above).

Note 2 – The compandors used should conform to the limits proposed in Recommendation G.162 or in Recommendation G.166.

Note 3 – In accordance with the Recommendation cited in [5], circuits with a noise power level of −37 dBm0p or worse are removed from service.

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1) For example, TASI (Time Assignment Speech Interpolation) of CELTIC (Concentrateur exploitant les temps d'inoccupation des circuits); see Recommendation G.163.

2) The instantaneous compandors that are associated with certain transmission systems are considered to be an integral part of these systems.

3) For characteristics of syllabic compandors for telephony on high capacity long distance systems, see Recommendation G.166.

4) See Annex A for further considerations relating to the use of syllabic compandors.

3 Noise limits for telegraphy

Noise limits for telegraphy are given in Recommendation H.22 [6].

4 Noise limits for data transmission

The following objectives are acceptable for data transmission at data signalling rates not exceeding 1200 bit/s. It is expected that the values actually experienced on many circuits and connections will be better than the following limits.

4.1 *Leased circuits for data transmission*

A reasonable limit for uniform spectrum random noise for a data transmission *leased* circuit, assuming that plant liable to impulsive noise interference is avoided, and as high a modulation rate as possible is to be used without significant error rate, would appear to be −40 dBm0p.

4.2 *Switched connections*

For switched connections a limit of, say, −36 dBm0p without compandors may be taken for intercontinental circuits on which compandors may be used.

ANNEX A

(to Recommendation G.143)

Additional considerations relating to the use of syllabic compandors

(The following information was available from Study Group XII)

This annex addresses compandor advantage in § A.1, followed by a recommendation of the permissible advantage limits for planning purposes (§ A.2). A requirement of circuit stability between compressor and expander is given in § A.3, and §§ A.4 and A.5 deals with aspects of system loading and companded circuits in tandem.

A.1 *Compandor advantage*

To define **compandor advantage**, assume:

a) an international circuit not equipped with compandors and contributing N dBm0 of noise to the overall end-to-end connection (including typical national extensions) and meeting the noise objectives of Recommendation G.152 or Recommendation G.153, and

b) the same international circuit equipped with compandors and connected to typical national extensions, yielding the noise performance subjectively equivalent to or better than that of the circuit described in a), while contributing N' dBm0 of noise in between compressor and expander.

Then the compandor advantage for the international circuit of b) is defined as $(N' - N)$ dB.

A.2 *Compandor advantage limit*

For planning purposes, the compandor advantage defined in § A.1 should not exceed 10 dB.

Note – It should be emphasized that this value applies to the international portion of the connection only. Other portions of the connection could permit a higher value when selected with due regard to the effect it has on the total noise of the end-to-end connection during the presence of the signal.

A.3 *Circuit stability*

The international circuit between compressor and expander should have an insertion loss which, when considered over a long period of time, has a standard deviation not exceeding 0.75 dB.

A.4 *Circuit loading*

It is generally advisable to select the unaffected level of the compandor equal to -10 dBm0. However, if Administrations mutually desire to operate at a different value of unaffected level, it should be selected such that it results in a system loading which minimizes total distortion due to noise, intermodulation, or other load-dependent characterisics and should always be dictated by the allowable compandor advantage limit.

A.5 *Compandored circuits in tandem*

The following paragraphs apply to circuits fitted with compandors according to Recommendation G.162.

Results of experiments with compandored circuit links in tandem show that two compandored links in tandem can produce a noticeable degradation only if the second link exceeds, by a considerable margin, the recommended compandor advantage limit of 10 dB. The experiment was admittedly designed to uncover gross effects by limiting the subjective judgement to only seven persons per test condition.

The conclusion drawn was that two links in tandem, each of which is limited to 10 dB compandor advantage, will not pose a restriction to users. This however, does not constitute sufficient guidance for application for the number of compandored links permissible in an end-to-end international connection.

References

[1] CCITT *Red Book*, Vol. V *bis*, Annexes B and C, ITU, Geneva, 1965.

[2] CCITT Recommendation *Setting-up and lining-up an international circuit for public telephony*, Vol. IV, Rec. M.580.

[3] CCITT Recommendation *Transmission characteristics of an international exchange*, Vol. VI, Rec. Q.45.

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

[5] CCITT Recommendation *Setting-up and lining-up an international circuit for public telephony*, Vol. IV, Rec. M.580, § 6.

[6] CCITT Recommendation *Transmission requirements of international voice-frequency telegraph links (at 50, 100 and 200 bauds)*, Vol. III, Rec. H.22.

1.5 General characteristics of international telephone circuits and national extension circuits

Recommendation G.151

GENERAL PERFORMANCE OBJECTIVES APPLICABLE TO ALL MODERN INTERNATIONAL CIRCUITS AND NATIONAL EXTENSION CIRCUITS

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

1 Attenuation distortion

The circuit performance objectives for attenuation distortion of international circuits and national extension circuits should individually be such that the network performance objectives of Recommendation G.132 are complied with. Recommendation G.232 [1] gives equipment design objectives.

It follows from the Recommendations mentioned above that, as a rule, the frequency band effectively transmitted by a telephone circuit, according to the definition adopted by the CCITT (i.e. the band in which the attenuation distortion does not exceed 9 dB compared with the value for 800 Hz), will be a little wider than the 300-3400 Hz band, and for a single pair of channel terminal equipments of this type, the attenuation distortion at 300 Hz and 3400 Hz should never exceed 3 dB and in a large number of equipments should not average more than 1.7 dB (see Graphs A and B in Figure 1/G.232 [2]). Even more complex circuits, and circuits using terminal equipments with 3-kHz-channel spacing in accordance with Recommendation G.235 [3], should satisfy the limits in Figure 1/G.151; to ensure that these limits are respected, equalizers are inserted, if necessary, when the circuits are set up (Recommendation M.580 [4]).

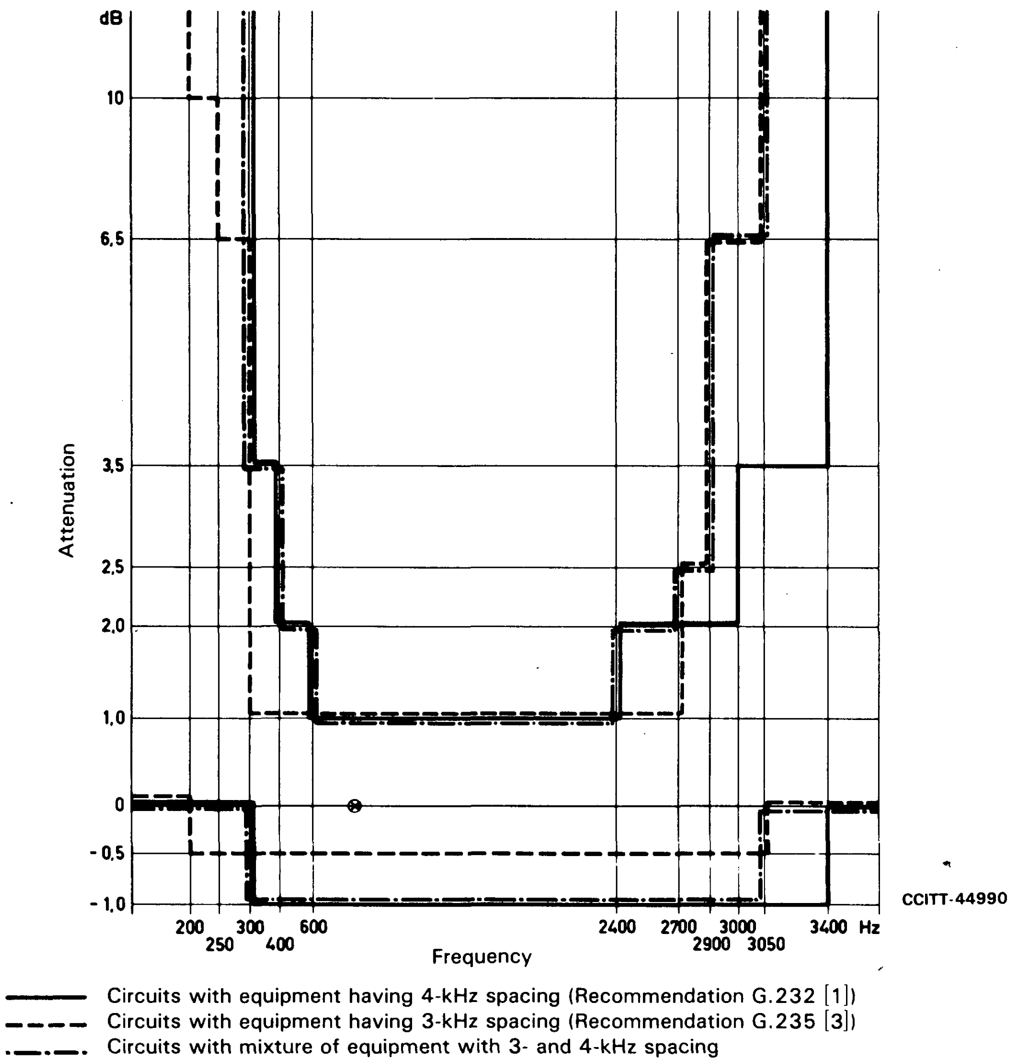

FIGURE 1/G.151

Line-up limits of circuits with 3-kHz and 4-kHz channel equipment

Note 1 – The CCITT examined the possibility of recommending a specific frequency below 300 Hz as the lower limit of the frequency band effectively transmitted, taking the following considerations into account:

1) The results of subjective tests carried out by certain Administrations show that it is possible to improve transmission quality if the lower limit of the transmitted frequency band is reduced from 300 Hz to 200 Hz. These tests show a definite increase in the loudness of the received speech, and also in the quality of the transmission as judged by opinion tests; the improvement in articulation is, on the other hand, very slight.

2) However, such an extension would probably have the following disadvantages:

a) it would slightly increase the cost of equipment;

b) it would introduce some difficulties in balancing the terminating sets at the ends of the 4-wire chain, if it were desired to use 4-wire circuits without exceeding the values of nominal transmission loss recommended in the new transmission plan;

c) it would increase the possible susceptibility to interference, whether as subjective noise or as disturbances interfering with carrier equipment (see the Recommendation cited in [5]) or affecting compandor gain;

d) the additional energy transmitted in consequence of extending the band could increase the loading of carrier systems;

e) the out-of-band signalling systems recognized by the CCITT could not be used.

In view of the above, the CCITT has issued the aforementioned Recommendations concerning signals transmitted at frequencies between 300 and 3400 Hz.

Note 2 – In applying the Recommendations, Administrations may mutually agree to transmit signals at frequencies below 300 Hz over international circuits. Every Administration may, of course, decide to transmit signals at frequencies below 300 Hz over its national extension circuits, provided it is still able to apply the CCITT transmission plan to international communications.

2 Group delay

The group-delay performance objectives of international circuits and national extension circuits should be such that the network performance objectives of Recommendations G.114 and G.133 are met.

3 Variations of transmission loss with time

The CCITT recommends the following circuit performance objectives [objective a) has been used to assess the stability of international connections – see Recommendation G.131, § 1]:

a) The standard deviation of the variation in transmission loss of a circuit should not exceed 1 dB. This objective can be obtained already for circuits on a single group link equipped with automatic regulation and should be obtained for each national circuit, whether regulated or not. The standard deviation should not exceed 1.5 dB for other international circuits.

b) The difference between the mean value and the nominal value of the transmission loss for each circuit should not exceed 0.5 dB.

4 Linear crosstalk [1)]

4.1 *Between circuits*

The circuit performance objective for the near-end or far-end crosstalk ratio (intelligible crosstalk only) measured at audio-frequency at trunk exchanges between two complete circuits in terminal service position should not be less than 65 dB.

Note 1 – When a minimum noise level of at least 4000 pW0p is always present in a system (e.g. this may be the case in satellite systems, for example) a reduced crosstalk ratio of 58 dB between circuits is acceptable.

Note 2 – Coaxial pair cables complying with Recommendations G.622 [6] and G.623 [7] already allow this condition to be fulfilled if it is assumed that the frequency bands for which crosstalk is caused by the cable and those for which crosstalk is due to the equipments are not the same. On the other hand FDM systems on symmetric pair cables do not always allow a limit more stringent than 58 dB to be met.

Note 3 – In cases where the length of a homogeneous section of a real transmission system substantially exceeds the length of a homogeneous section of the HRC, the 65 dB limit may not be met in all cases for all the channels in the system.

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1) The methods recommended for measuring crosstalk are described in Annex A to Recommendation G.134.

4.2 *Between the go and return channels of a 4-wire circuit*

4.2.1 *Ordinary telephone circuit* (see Note 1 below)

Since all ordinary telephone circuits may also be used as VF telegraph bearers, the circuit performance objective for the near-end crosstalk ratio between the two directions of transmission should be at least 43 dB.

4.2.2 *Circuits used with a speech concentrator*

For circuits and circuit sections used to interconnect terminal speech concentrator equipments, near-end crosstalk between any two channels will appear in the form of crosstalk between circuits and hence the circuit performance objective for the total near-end crosstalk ratio introduced between speech concentrators should not be less than 58 dB. (See Notes 2 and 4 below.)

4.2.3 *Circuits used with modern echo suppressors, for example high-altitude satellite circuits*

The circuit performance objective for the near-end crosstalk ratio of any circuit equipped with terminal far-end operated, half-echo suppressors of modern design should not be less than 55 dB. This is to avoid nullifying the effect of the suppression loss introduced by modern echo suppressors. (See Notes 2, 3 and 4 below.)

Note 1 – Telephone circuits which are not equipped with (or used in conjunction with) modern echo suppressors designed for long propagation times are referred to in § 4.2.1 above. Circuits which can form part of switched connections with a long propagation time and which then lie between terminal half-echo suppressors of modern design should, wherever possible, conform to the higher standards given in this § 4.2.3.

Note 2 – The channel-translating equipment provides the principal go-to-return crosstalk path on circuits or circuit-sections routed on carrier systems with modern translating and line transmission equipment (but see Note 4 below). It should be noted that crosstalk paths between the high-frequency input and the high-frequency output and also between the voice-frequency input and the voice-frequency output on channel-translating equipments contribute to the go-to-return crosstalk ratios of circuits and circuit sections. Both these paths must be taken into account when considering circuits or circuit sections used between terminal speech concentrator equipments or modern echo suppressors. The following cases arise:

Speech concentrators

Both the high-frequency path and the voice-frequency path contribute to the crosstalk ratio.

Echo suppressors

1) A circuit comprising one circuit section between far-end operated, half-echo suppressors: the high-frequency path is dominant.
2) A circuit comprising more than one circuit section between the suppressors: at points where channel-translating equipments are connected together at voice-frequency. The voice frequency crosstalk path of one equipment is effectively in parallel with the high-frequency crosstalk path of the other, so that both must be taken into account.
3) More than one circuit between the suppressors: this occurs when intermediate adjacent half-echo suppressors are switched out (or disabled) and the go-to-return crosstalk arises in a fashion analogous to that described in 2) above, circuits replacing circuit sections.

Note 3 – If channel equipments just conforming to the Recommendation cited in [8] are used on a circuit comprising three circuit sections, then assuming r.m.s. addition of crosstalk paths the crosstalk ratio would be approximately 60 dB.

Note 4 – If channel equipments used on a circuit comprising three circuit sections just comply with the Recommendation cited in [9], then the least go-to-return crosstalk ratio, assuming r.m.s. addition of the various paths, would be approximately 56 dB which is 2 dB less than is required for speech concentrators in § 4.2.2 above. However, the assumptions are most pessimistic and there is not likely to be any difficulty in practice. The limit for echo suppressor in § 4.2.3 above is complied with.

Note 5 – Some types of symmetrical-pair line transmission systems introduce significantly low go-to-return crosstalk ratios on the derived circuits and wherever possible such systems should not be used to provide circuits or circuit sections for use with speech concentrators or modern echo suppressors.

Note 6 – Some attention must be given to the unbalance of the audio parts of FDM channel equipments if the crosstalk of 65 dB is not to be diminished by crosstalk in station cabling due to unbalanced cable terminating equipment.

5 Nonlinear distortion

Experience has shown that telephone circuits set up on systems for which the CCITT has issued recommendations (the elements of which systems, taken separately, meet the relevant nonlinearity requirements) are equally suitable, as far as nonlinearity is concerned, for telephone and voice-frequency telegraph transmission.

Note – In carrier telephone circuits, the nonlinear distortion produced by the line amplifiers and by modulation stages other than the channel-translating equipment can be ignored. Hence the above remarks are applicable to circuits of any length.

6 Error on the reconstituted frequency

See Recommendation G.135.

7 Interference at harmonics from the mains and other low frequencies

Signals carried by transmission systems are sometimes modulated by interfering signals from mains frequency power supplies, induced voltages caused by railway traction currents and from other sources. This unwanted modulation can take the form of amplitude or phase modulation or a combination of both. This interference may be characterized by the level of the strongest unwanted side component when a sine wave signal is applied with a power of 1 mW at the point of zero relative level (0 dBm0) on a telephone circuit. The circuit performance objective for the maximum admissible level of the unwanted side components on a complete telephone circuit should then not exceed −45 dBm0 (i.e. the minimum side component attenuation should be 45 dB). This circuit performance objective should apply to all low frequency interfering signals up to about 400 Hz.

Note 1 – This level was found to be acceptable for circuits for FM and AM VF-telegraphy, facsimile transmission, speech, telephone signalling and data transmission.

Note 2 – For limits applicable to sound-programme circuits, see the Recommendation cited in [10].

Note 3 – The main causes of interference due to power sources are:

a) residual ripples at the terminals of d.c. supply which are directly transmitted to equipments through the power-fed circuits;

b) the a.c. to the dependent power-fed stations in some systems, which interferes through the power-separating filter or through the iron tapes of coaxial pairs;

c) the induction voltages in the d.c. supply line to power-fed dependent stations in some systems;

d) the amplitude and phase unwanted modulations of the various carriers due to cause a) which are increased in the frequency-multiplying equipments.

Note 4 – The effect of the modulation process is that an input signal of frequency f Hz will produce, for example, corresponding output signals at frequencies f, $f \pm 50$, $f \pm 100$, $f \pm 150$ Hz, etc.

8 Single tone interference in telephone circuits

The single tone interference level in a telephone circuit should not be higher than −73 dBm0p (provisional value, pending the conclusion of studies by Study Group XII). Psophometric weighting should only be accounted for when the frequency of the interference is well defined.

References

[1] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232.

[2] *Ibid.*, Figure 1/G.232, Graphs A and B.

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

[4] CCITT Recommendation *Setting-up and lining-up an international circuit for public telephony*, Vol. IV, Rec. M.580.

[5] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232, § 6.

[6] CCITT Recommendation *Characteristics of 1.2/4.4-mm coaxial cable pairs*, Vol. III, Rec. G.622.

[7] CCITT Recommendation *Characteristics of 2.6/9.5-mm coaxial cable pairs*, Vol. III, Rec. G.623.

[8] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232, § 9.1.

[9] *Ibid.*, § 9.3.

[10] CCITT Recommendation *Performance characteristics of 15-kHz type sound-programme circuits*, Vol. III, Rec. J.21, § 3.1.7.

Recommendation G.152

CHARACTERISTICS APPROPRIATE TO LONG-DISTANCE CIRCUITS OF A LENGTH NOT EXCEEDING 2500 km

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

This Recommendation applies to all modern international circuits not more than 2500 km in length. It also applies to national trunk circuits in an average-size country, and which may be used in the 4-wire chain of an international connection.

It is understood that, should an extension circuit more than 2500-km long be used in a large country, it will have to meet all the recommendations applicable to an international circuit of the same length.

1 Circuits on land or submarine cable systems or on line-of-sight radio-relay systems

The circuits in question are mostly set up in cable or radio-relay link carrier systems, such that the noise objectives of Recommendation G.222 [1] are applicable to a circuit with the same make-up as the hypothetical reference circuit 2500-km long.

A consequence of Recommendation G.222 [1] is that, for a circuit L-km long ($L \leqslant 2500$ km), the circuit performance objective for the mean psophometric noise power during any hour should be of the order of 4 L picowatts, excluding very short circuits and those with a very complicated composition, this latter case being dealt with in Recommendation G.226 [2].

2 Circuits on tropospheric-scatter radio-relay systems

The CCIR has defined a hypothetical reference circuit and fixed circuit performance objectives in its Recommendations 396 [3] and 397 [4] respectively.

3 Circuits on open-wire carrier systems

The Recommendation cited in [5] contains relevant noise objectives.

Note – Recommendation M.580 [6] deals with noise objectives for maintenance purposes. See Note 1 of Recommendation G.143, § 1.1.

References

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

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

[3] CCIR Recommendation *Hypothetical reference circuit for trans-horizon radio-relay systems for telephony using frequency-division multiplex*, Vol. IX, Rec. 396, ITU, Geneva, 1986.

[4] CCIR Recommendation *Allowable noise power in the hypothetical reference circuit of trans-horizon radio-relay systems for telephony using frequency-division multiplex*, Vol. IX, Rec. 397, ITU, Geneva, 1986.

[5] CCITT Recommendation *General characteristics of systems providing 12 carrier telephone circuits on an open-wire pair*, Vol. III, Rec. G.311, § 8.

[6] CCITT Recommendation *Setting-up and lining-up an international circuit for public telephony*, Vol. IV, Rec. M.580.

Recommendation G.153

CHARACTERISTICS APPROPRIATE TO INTERNATIONAL CIRCUITS MORE THAN 2500 KM IN LENGTH

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

These circuits should meet the general requirements set forth in Recommendation G.151 and should, in addition, according to the kind of system on which they are set up, meet the particular provisions of §§ 1, 2, 3 and 4 below.

Note 1 – Some circuits which do not meet the noise objectives specified in the present Recommendation can nevertheless be used for telephony (if they are fitted with compandors), telegraphy or data transmission (§§ 2, 3 and 4 of Recommendation G.143; Table 1/G.153 summarizes these Recommendations).

Note 2 – Recommendation M.580 [1] deals with noise objectives for maintenance purposes. See Note 1 of Recommendation G.143, § 1.1).

1 Circuits more than 2500 km in length on cable or radio-relay systems, with no long submarine cable section

In many cases circuits of this kind, between 2500 km and about 25 000 km long will, throughout most of their length, be carried in land-cable systems or radio-relay systems already used to give international circuits not more than 2500 km long, and designed on the basis of the objectives already recommended for such systems in Recommendation G.222 [3].

Moreover, it is unlikely that the number of channel demodulations will exceed that envisaged in the corresponding part of the longest international connection referred to in Recommendation G.103. There will also be cases where it will be possible to establish such circuits on systems designed on the basis of national hypothetical reference circuits of the type referred to in the Recommendation cited in [4]. This being so, the CCITT issues the following recommendations:

1.1 *Variations in transmission loss with time*

Automatic level adjustment should be used on each group link on which the circuit is routed. In addition, all possible steps should be taken to reduce changes of transmission loss with time.

1.2 *Performance objectives for circuit noise*

It is provisionally recommended that systems to provide such international circuits not more than 25 000 km long should be designed on the basis of the noise objectives at present recommended for 2500-km hypothetical reference circuits.

TABLE 1/G.153

Noise objectives or limits [a] for very long circuits providing various services [b]

Psophometric power		Type of objective or limit	
pW0p	dBm0p	For a connection, a chain of circuits, or a leased circuit	For a circuit which may form part of a switched connection
40 000	−44		Limit for a telephone circuit used without a compandor (Recommendation G.143, § 2)
50 000	−43	Objective for a chain of 6 international circuits, obtained in practice by a combination of circuits with circuit performance objectives of 1, 2 or 4 pW/km (Recommendation G.143, § 1)	
80 000	−41	Limit for FM VF telegraphy, in accordance with CCITT standards (Recommendation H.22 [2])	
100 000	−40	Limit for data transmission over a leased circuit (Recommendation G.143, § 4.1)	
250 000	−36		Acceptable for data transmission over the switched network (Recommendation G.143, § 4.2). A circuit exceeding this limit without a compandor cannot be used in a chain of 6 telephone circuits, even if it is equipped with a compandor (Recommendation G.143, § 2)
10^6	−30	Tolerable for a certain system of synchronous telegraphy (Recommendation H.22 [2])	

a) Only the mean psophometric power over one hour has been indicated, referred to a point of zero relative level of the international circuit, or of the first circuit of the chain.

b) The noise limits are determined according to the minimum performance requirements of each service. The noise objectives are commissioning objectives for various transmission systems.

Whenever possible lower noise objectives should be sought and it is recognized that in some large countries systems forming part of a circuit substantially longer than 2500 km (e.g. 5000 km) are constructed according to the principles referred to in the Recommendation cited in [4]. Alternatively lower noise figures can be obtained by a suitable choice of telephone channels making up the circuits. Provisionally the short-term noise performance objectives for circuits of this kind of length up to about 7500 km are as follows:

The one-minute mean noise power shall not exceed 50 000 pW (−43 dBm0p) for more than 0.3% of any month and the unweighted noise power, measured or calculated with an integrating time of 5 ms, shall not exceed 10^6 pW (−30 dBm0) for more than 0.03% of any month. It is to be understood that these objectives are derived pro rata from the objectives for circuits of 2500 km length (Recommendation G.222 [3]); for lengths between 2500 and 7500 km proportionate intermediate values should apply.

The CCITT is not yet able to recommend objectives for short-term noise performance on circuits of the above type which exceed 7500 km in length.

2 Circuits more than 2500 km with a long submarine cable section

2.1 *Attenuation distortion*

A circuit of this kind may, for reasons of economy, comprise terminal equipments with carriers spaced 3 kHz apart, in accordance with Recommendation G.235 [5].

If terminal equipment be used with carrier spacing of 4 kHz, it must at least meet the requirements of Recommendation G.232 [6]. Some countries use improved terminal equipment in circuits permanently used for intercontinental operation.

2.2 *Performance objectives for circuit noise attributable to the submarine cable section*

2.2.1 *Without compandor*

The circuit performance objective for the mean noise per hour of a very long submarine-cable system designed for use without compandors and with no restrictions for telephony, voice-frequency telegraphy and data transmission should not exceed 3 pW/km on the worst channel. The circuit performance objective for the mean noise power for each direction of transmission, extended over all the channels used for the longest circuits, should not exceed 1 pW/km.

Note – However, it would be desirable that the circuits in a group to be operated with a speech concentrator system [1)] should all have more or less the same noise level.

2.2.2 *With compandor*

At present, the CCITT does not propose to study systems which, by relying on the *systematic* use of compandors, have noise objectives which are greatly different from those of § 2.2.1 above.

2.3 *Performance objectives for circuit noise attributable to other sections*

The other sections of the circuit should comply with the recommendations given in § 1 of this Recommendation.

3 Circuits on communication-satellite systems

The CCIR and the CCITT are considering the extent to which circuits set up on communication-satellite systems may be integrated into the worldwide network; some of the limitations on the use of such circuits are outlined in Recommendation Q.13 [7].

The CCIR has made recommendations as far as circuit noise is concerned and has defined a hypothetical reference circuit (CCIR Recommendation 352 [8]) and the allowable noise power in this reference circuit (CCIR Recommendation 353 [9]).

4 Circuits more than 2500 km in length set up on open-wire lines

Paragraph 4 is not published in this Book, but can be found under Part D of Recommendation G.153, *Orange Book*, ITU, Geneva, 1977.

References

[1] CCITT Recommendation *Setting-up and lining-up an international circuit for public telephony*, Vol. IV, Rec. M.580.

[2] CCITT Recommendation *Transmission requirements of international voice-frequency telegraph links (at 50, 100 and 200 bauds)*, Vol. III, Rec. H.22.

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

1) See footnote 2) in Recommendation G.143, § 2.

[4] *Ibid.*, § 3.

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

[6] CCITT Recommendation *12-channel terminal equipments*, Vol. III, Rec. G.232.

[7] CCITT Recommendation *The international routing plan*, Vol. VI, Rec. Q.13.

[8] CCIR Recommendation *Hypothetical reference circuits for telephony and television in the fixed satellite service*, Vol. IV, Rec. 352, ITU, Geneva, 1986.

[9] CCIR Recommendation *Allowable noise power in the hypothetical reference circuit for frequency-division multiplex telephony in the fixed satellite service*, Vol. IV, Rec. 353, ITU, Geneva, 1986.

1.6 Apparatus associated with long-distance telephone circuits

Recommendation G.161

ECHO-SUPPRESSORS SUITABLE FOR CIRCUITS HAVING EITHER SHORT OR LONG PROPAGATION TIMES

(See Vol. III of the *Orange Book*, ITU, Geneva, 1977).

Recommendation G.162

CHARACTERISTICS OF COMPANDORS FOR TELEPHONY

(Geneva, 1964; amended at Mar del Plata, 1968)

These characterisics are applicable to compandors of modern design for use either on very long international circuits or on national and international circuits of moderate length.

Some of the clauses given below specify the joint characteristics of a compressor and an expander in the same direction of transmission of a 4-wire circuit. The characteristics specified in this way can be obtained more easily if the compressors and expanders are of similar design; in certain cases close cooperation between Administrations may be necessary.

It should also be noted that the equipement produced so far for circuits of moderate length may be completely satisfactory for those circuits and yet not quite meet the clauses of this Recommendation.

1 Definition and value of the unaffected level

The unaffected level is the absolute level, at a point of zero relative level on the line between the compressor and the expander of a signal at 800 Hz, which remains unchanged whether the circuit is operated with the compressor or not. The unaffected level is defined in this way in order not to impose any particular values of relative level at the input to the compressor or the output of the expander.

The unaffected level should be, in principle, 0 dBm0. Nevertheless, to make allowances for the increase in mean power introduced by the compressor, and to avoid the risk of increasing the intermodulation noise and the overload which might result, the unaffected level may, in some cases, be reduced by perhaps as much as 5 dB. However, this reduction of unaffected level entails a diminution of the improvement in signal-to-noise ratio provided by the compandor. This possible reduction should be made by direct agreement between the Administrations concerned. No reduction is necessary, in general, for systems with less than 60 channels.

Note – The increase in the mean power in the transmitted band determined by the compressor in the telephone channel depends on the value of the unaffected level, the attack and recovery times, the distribution of the speech volumes and the mean power level of transmitted speech. When 0 dBm0 is adopted for the unaffected level, it appears that the effective increase in the mean power level is of the order of 2 or 3 dB.

2 Ratio of compression and expansion

2.1 *Definition and preferred value of the* **ratio of compression**

The ratio compression of a compressor is defined by the formula:

$$\alpha = \frac{n_e - n_{e0}}{n_s - n_{s0}}$$

where:

n_e is the input level;

n_{e0} is the input level corresponding to 0 dBm0;

n_s is the output level;

n_{s0} is the output level corresponding to an input level of n_{e0}.

The preferred value of α is 2, though lower values are permissible, provided sufficient noise improvement is obtained. The value shall not exceed 2.5 for any level of input signal and at any temperature between +10 °C and +40 °C.

2.2 *Definition and preferred value of the* **ratio of expansion**

The ratio of expansion of an expander is defined by the formula:

$$\beta = \frac{n'_s - n'_{s0}}{n'_e - n'_{e0}}$$

where:

n'_e is the input level;

n'_{e0} is the input level corresponding to 0 dBm0;

n'_s is the output level;

n'_{s0} is the output level corresponding to an input level of n'_{e0}.

The preferred value of β is 2, though lower values are permissible, provided sufficient noise improvement is obtained. The value shall not exceed 2.5 for any level of input signal and at any temperature between +10 °C and +40 °C.

2.3 *Range of level*

The range of level over which the recommended value of α and β should apply should extend at least:

from +5 to −45 dBm0 at the input of the compressor, and

from +5 to −50 dBm0 at the nominal output of the expander.

2.4 *Variation of compressor gain*

The level at the output of the compressor, measured at 800 Hz, for an input level of 0 dBm0, should not vary from its nominal value by more than ± 0.5 dB for a temperature range of +10 °C to +40 °C and a deviation of the supply voltage of ± 5% from its nominal value.

2.5 *Variation of expander gain*

The level at the output of the expander, measured at 800 Hz for an input level of 0 dBm0, should not vary from its nominal value by more than ± 1 dB for a temperature range of +10 °C and +40 °C and a deviation of the supply voltage of ± 5% from its nominal value.

Note – It is desirable, especially for compandors intended for very long circuits, to set stricter limits than the values of +0.5 dB and ± 1 dB given under § 2.4 and § 2.5; +0.25 dB and +0.5 dB respectively are preferable.

2.6 *Conditions for stability*

The insertion of a compandor shall not appreciably reduce the margin of stability. To ensure this, for the combination of an expander and a compressor on the same 4-wire circuit and at a given station, the error of the output level of the compressor with respect to any value of expander input level shall not exceed + 0.5 dB. This error is referred to the level obtained at the compressor output when the input level is 0 dBm0. This limit shall be observed at all frequencies between 200 and 4000 Hz within the temperature range +10 °C to +40 °C. No negative limit is specified for the error. In this test an attenuator shall be inserted between the expander and the compressor, the value of which is to be set in accordance with the following Note 1.

Note 1 – This Note concerns the influence of a compandor on the loop gain of a 4-wire circuit and on the margin of stability.

In examining this problem, a connection was considered made up of three 4-wire circuits, *AB*, *BC* and *CD*, which link the terminal stations *A* and *D* (at which the terminating sets are located) through the intermediate stations *B* and *C*. It is assumed that the circuit *BC* is equipped with compandors. It is desired to determine the tolerances for the gain of the combination of expander and compressor at *C* in order to limit the reduction in the margin of stability caused by their insertion. To facilitate study of this question it is assumed that, in normal use, the expander output and compressor input are points of the same relative level.

The following expression then gives the loss between the output of the expander at *C* and the input of the compressor at *C*:

$$a_s = a_0 + a_r + a_x + a_y$$

where

a_0 is the nominal transmission loss of the chain of circuits between the 2-wire terminals at *A* and *D*;

a_r is the balance return loss at the terminating set at *D*;

a_x is the departure of transmission loss of channel *CD* from its nominal value;

a_y is the departure of transmission loss of channel *DC* from its nominal value.

The two latter values may be positive or negative.

It may be concluded that, in order that the measurement of the gain of the combination of an expander and a compressor at the same station may satisfactorily determine the total effect on the margin of stability, the following conditions must be observed:

The expander must be connected to the compressor via an attenuator, the loss of which should cover the entire range of values for a_s which actually occur when there is a risk of instability. To take account of all practical conditions, it would probably be necessary to consider a very wide range.

However, considering only the important example of a terminal compandor and zero balance return loss, then $a_s = a_0$ and this is the value which is generally recommended for the loss of the attenuator between expander and compressor in this test.

Nevertheless, when it is possible to determine the exact values of a_r, a_x and a_y, corresponding to the most probable condition of instability, the exact value of a_s can be specified.

It has been assumed that the expander output and the compressor input are normally points of the same relative level. If this is not the case, and if the relative level at the expander output is a_c dB higher than the relative level at the compressor input, the loss in the attenuator should be increased by a_c (which may be positive or negative).

Note 2 – Cross-connection between the control circuits of the compressor and expander may have advantages from the point of view of circuit echoes; hence, its use should be allowed. On the other hand its use, which has some disadvantages from the point of view of signalling-to-voice break-in, will certainly be confined to exceptional cases. In consequence, there seems no need for any special recommendations on the subject.

2.7 *Tolerances on the output levels of the combination of compressor and expander in the same direction of transmission of a 4-wire circuit*

The compressor and expander are connected in tandem. A loss (or gain) is inserted between the compressor output and expander input equal to the nominal loss (or gain) between these points in the actual circuit in which they will be used. Figure 1/G.162 shows, as a function of level of 800-Hz input signal to the compressor, the permissible limits of difference between expander output level and compressor input level. (Positive values indicate that the expander output level exceeds the compressor input level.)

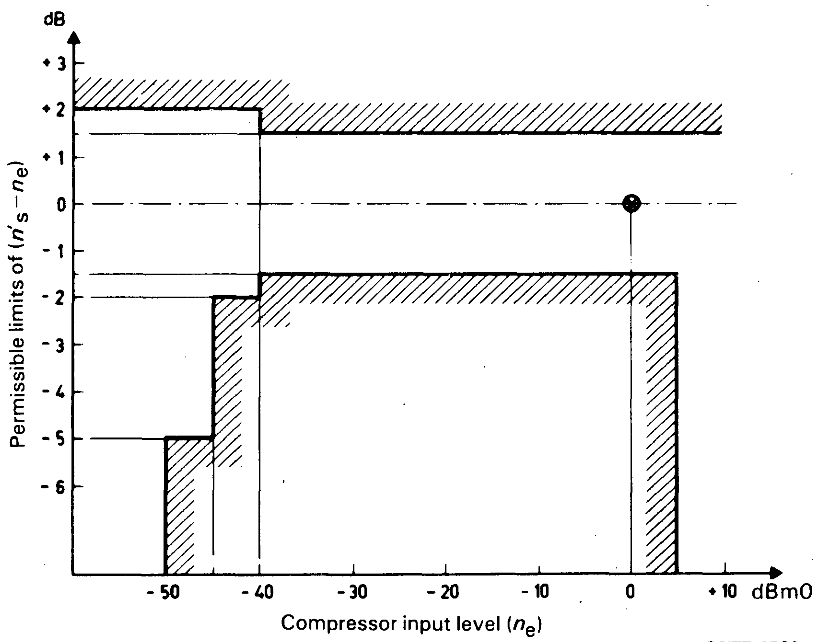

FIGURE 1/G.162

The limits shall be observed at all combinations of temperature of compressor and temperature of expander in the range +10 °C to +40 °C. They shall also be observed when the test is repeated with the loss (or gain) between the compressor and expander increased or decreased by 2 dB.

Note – The change of gain (or loss) of 2 dB mentioned in § 2.7 above is equal to twice the standard deviation of transmission loss recommended as an objective for international circuits routed on single group links in Recommendation G.151, § 3.

3 Impedances and return loss

The nominal value of the input and output impedances of both compressor and expander should be 600 ohms (nonreactive).

The return loss with respect to the nominal impedance of the input and the output of both the compressor and the expander should be no less than 14 dB over the frequency range 300 to 3400 Hz and for any measurement level between +5 and −45 dBm0 at the compressor input or the expander output.

4 Operating characteristics at various frequencies

4.1 *Frequency characteristic with control circuit clamped*

The control circuit is said to be clamped when the control current (or voltage) derived by rectification of the signal is replaced by a constant direct current (or voltage) supplied from an external source. For purposes here, the value of this current (or voltage) should be equal to the value of the control current (or voltage) obtained when the input signal is 0 dBm0 at 800 Hz.

For the compressor and the expander taken separately, the variations of loss or gain with frequency should be contained within the limits of a diagram that can be deduced from Figure 1/G.132 by dividing the tolerance shown by 8, the measurement being made with a constant input level corresponding to a level of 0 dBm0.

These limits should be observed over the temperature range +10 °C and +40 °C.

4.2 *Frequency characteristic with control circuit operating normally*

The limits given in § 4.1 should be observed for the compressor when the control circuit is operating normally, the measurement being made with a constant input level corresponding to a level of 0 dBm0.

For the expander, under the same conditions of measurement, the limits can be deduced from Figure 1/G.132 by dividing the tolerances shown by 4.

These limits should be observed over the temperature range +10 °C to +40 °C.

5 Nonlinear distortion

5.1 *Harmonic distortion*

Harmonic distortion, measured with an 800-Hz sine wave at a level of 0 dBm0, should not exceed 4% for the compressor and the expander taken separately.

Note – Even in an ideal compressor, high output peaks will occur when the signal level is suddenly raised. The most severe case seems to be that of voice-frequency signalling, although the effect can also occur during speech. It may be desirable, in exceptional cases, to fit the compressor with an amplitude limiter to avoid disturbance due to transients during voice-frequency signalling.

5.2 *Intermodulation tests*

It is necessary to add a measurement of intermodulation to the measurements of harmonic distortion whenever compandors are intended for international circuits (regardless of the signalling system used), as well as in all cases where they are provided for national circuits over which multi-frequency signalling, or data transmission using similar types of signals, is envisaged.

The intermodulation products of concern to the operation of multi-frequency telephone signalling receivers are those of the third order, of type $(2f_1 - f_2)$ and $(2f_2 - f_1)$, where f_1 and f_2 are two signalling frequencies.

Two signals at frequencies 900 Hz and 1020 Hz are recommended for these tests.

Two test conditions should be considered: the first, where each of the signals at f_1 and f_2 is at a level of –5 dBm0 and the second, where they are each at a level of –15 dBm0. These levels are to be understood to be at the input to the compressor or at the output of the expander (uncompressed levels).

The limits for the intermodulation products are defined as the difference between the level of either of the signals at frequencies f_1 or f_2 and the level of either of the intermodulation products at frequencies $(2f_1 - f_2)$ or $(2f_2 - f_1)$.

A value for this difference which seems adequate for the requirements of multi-frequency telephone signalling (including end-to-end signalling over three circuits in tandem, each equipped with a compandor) is 26 dB for the compressor and the expander separately.

Note 1 – These values seem suitable for Signalling System No. 5, which will be used on some long international circuits.

Note 2 – It is inadvisable to make measurements on a compressor plus expander in tandem, because the individual intermodulation levels of the compressor and of the expander might be quite high, although much less intermodulation is given in tandem measurements since the characteristics of compressor and expander may be closely complementary. The compensation encountered in tandem measurements on compressor and expander may not be encountered in practice, either because there may be phase distortion in the line or because the compressor and expander at the two ends of the line may be less closely complementary than the compressor and expander measured in tandem.

Hence the measurements have to be performed separately for the compressor and the expander. The two signals at frequencies f_1 and f_2 must be applied simultaneously, and the levels at the output of the compressor or expander measured selectively.

6 Noise voltages

The effective value of the sum of all noise voltages referred to a zero relative level point, the input and the output being terminated with resistances of 600 ohms, shall be less than or equal to the following values:

– at the output of the compressor:	(10 mV unweighted	−38 dBm0)
	(7 mV weighted	−41 dBm0p)
– at the output of the expander:	(0.5 mV weighted	−84 dBm0p)

It is not considered useful to specify a value of unweighted noise voltage for the expander.

7 Transient response

The overall transient response of the combination of a compressor and expander which are to be used in the same direction of transmission of a 4-wire circuit fitted with compandors shall be checked as follows:

The compressor and expander are connected in tandem, the appropriate loss (or gain) being inserted between them as in § 2.7.

A 12-dB step signal at a frequency of 2000 Hz is applied to the input of the compressor, the actual values being a change from −16 to −4 dBm0 for attack, and from −4 to −16 dBm0 for recovery. The envelope of the expander output is observed. The overshoot (positive or negative), after an upward 12-dB step expressed as a percentage of the final steady-state voltage, is a measure of the overall transient distortion of the compressor-expander combination for attack. The overshoot (positive or negative) after a downward 12 dB step, expressed as a percentage of the final steady-state voltage is a measure of the overall transient distortion of the compressor-expander combination for recovery. For both these quantities the permissible limits shall be ± 20%. These limits shall be observed for the same conditions of temperature and of variation of loss (or gain) between compressor and expander as for the test in § 2.7.

In addition, the attack and recovery times of the compressor alone shall be measured as follows:

Using the same 12-dB steps as above for attack and recovery respectively, the attack time is defined as the time between the instant when the sudden change is applied and the instant when the output voltage envelope reaches a value equal to 1.5 times its steady-state value. The recovery time is defined as the time between the instant when the sudden change is applied and the instant when the output voltage envelope reaches a value equal to 0.75 times its steady-state value.

The permissible limits shall be not greater than:

- 5 ms for attack,
- 22.5 ms for recovery.

The following additional test shall be used to check the effect of the compandor on certain signalling systems which may be sensitive to envelope distortion immediately following the sudden application of a sinusoidal signal.

The overall transient response of the combination of a compressor and expander which are to be used in the same direction of transmission on a 4-wire circuit is measured with an "infinite" upward input step, i.e. with a signal applied after a period with no input.

The level of the signal to be applied is −5 dBm0.

Provided the measurement is effected with an interval of at least 50 ms between the pulses, the limits shown by an unbroken line in Figure 2/G.162 should be observed for the overshoot of the final voltage V_1; in most cases an attempt should be made if possible to observe the narrower limits, indicated in the figure by a broken line.

These limits shall be observed for the same conditions of temperature loss (or gain) between compressor and expander as for the tests with 12-dB steps.

Note 1 – The tests of transient distortion described involve the measurement of the overshoot or undershoot of the envelope of the applied sinusoidal signal. It may happen that, due to small unbalances in the variable loss device, very-low-frequency components of the control current appear at the output. These are not a modulation of the signal frequency, but they produce an unsymmetrical waveform and render it difficult to determine the overshoot or undershoot of the envelope. While it is undesirable that these low-frequency components should be so large as to increase significantly the risk of overload of the line equipment, they are of no importance for speech transmission and will not affect tuned signalling receivers. However, it is desirable to consider whether these components may affect the guard circuits of some signalling receivers. If so, it may be necessary to specify a maximum value for these components and to include an appropriate test in this Recommendation.

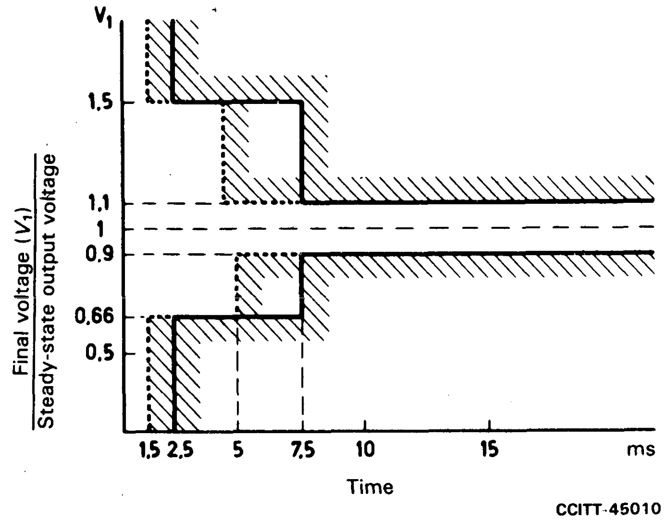

FIGURE 2/G.162

To simplify the measurement of the true envelope amplitude in the presence of these unbalance components, it is admissible and convenient to insert at the input to the measuring oscillograph a high-pass filter having a cut-off frequency of about 300 Hz. However, a filter which is effective in removing unbalance components may itself introduce additional transient distortion in the signal envelope. To avoid this difficulty, the following method of calculation may be adopted which does not require a filter.

If at any instant the amplitude of the envelope in a positive direction is $+E_1$, and in the negative direction is $-E_2$ then the two-envelope amplitude is given by

$$\frac{1}{2}\left[(+E_1) - (-E_2)\right] \equiv \frac{1}{2}\left[\,|E_1| + |E_2|\,\right]$$

and the unbalance component is given by

$$\frac{1}{2}\left[(+E_1) + (-E_2)\right] \equiv \frac{1}{2}\left[\,|E_1| - |E_2|\,\right]$$

This method is not only simple and free of the transient distortion problem which occurs with a filter, but it also provides direct information on the unbalance which, as indicated above, may be important.

Note 2 – The time constants of the expander control circuit should in principle be equal to those of the compressor control circuit so as to avoid any overshoot (positive or negative) in the transient response.

Note 3 – If an Administration prefers to use a direct method of measuring expander attack and recovery times, the following might be adopted;

To define the attack and recovery times of the expander, a sudden change in level from −8 to −2 dBm0 should be applied to its input for measurement of the attack time, and from −2 to −8 dBm0 for measurement of the recovery time. The attack time is represented by the time between the moment when the abrupt variation is applied and the moment when the output voltage reaches a value x times its final value. The recovery time is represented by the time between the moment when the abrupt variation is applied and the moment when output voltage reaches a value y times its final value. The times thus measured should lie between the same limits as those shown for the compressor. Bearing in mind detailed differences in the construction of the various compandors now in use, specific figures for x and y cannot be given. Hence, each Administation will have to determine the correct values of x and y for the type of compandor concerned.

For an ideal expander, 0.57 and 1.51 are valid for x and y; by way of example, the Italian Administation has found 0.65 for x and 1.35 for y for a certain type of construction.

Some Administrations have said that it might be preferable to specify fixed values of x and y, for all types of expander, leaving Administrations free to choose the limit values for attack and recovery times according to the different types of expander. Values of 0.75 and 1.5 are proposed for x and y in this method of measurement.

Note 4 – The "infinite" step transient response measurements refer to a compressor-expander combination connected in tandem; moreover, several Administrations have investigated the possibility of meeting the limits shown in the Figure 2/G.162, even for a chain of three compandors in tandem, by bringing also the channel modulating and demodulating equipment into the connection. This modem equipment may cause an undesirable transient phenomenon in the step at the expander output; this phenomenon, and the intermodulation of the third order associated with it, may influence the multi-frequency signalling.

Recommendation G.163

CALL CONCENTRATING SYSTEMS

(Mar del Plata, 1968)

1 Characteristics

The characteristics of the TASI system which is now in operation on submarine cable systems are given in references [1] and [2].

The characterisitics of the CELTIC system are given in reference [3].

ATIC (Time Assignment with Sample Interpolation) is a time assignment system for pulse code transmission. A description of the basic function is given in reference [4] and another article on its statistical efficiency is quoted in reference [5].

Note – The use of these concentrating systems involves various restrictions; for example, they may call for a special signalling system and they increase system loading (see the Recommendation cited in [6]).

2 Possibility of interconnection

To ensure satisfactory speech quality when call concentrating systems of the TASI type are operated in tandem, it is necessary that each concentrator introduce only a very small speech impairment at the peak of the busy hour. The present TASI concentrators were designed with the objective that the average speech lost during the peak of the busy hour will be approximately 0.5%. In addition, the interpolation process in TASI is designed so that there is a very small probability that the amount of speech lost in any speech spurt will be greater than the length of an average syllable (about 250 ms). Subjective tests [7] have been made on individual working TASI systems and the results, obtained by interviewing customers, show that the impairment due to a properly loaded and maintained TASI is essentially undetectable by the customer. No such tests have been carried out on call concentration systems in tandem.

Because of the subjective problems involved, estimates made of the speech impairment that would result from tandem call concentration systems must be qualitative without subjective tests. The probability of excessive clipping, even in a system of three concentrators in tandem with each having the same busy hour, can be kept to a satisfactory level by arranging the system so that the impairment introduced by each concentrator is small, as in the case of the present TASI system. If the tandem concentrators are located in different time zones or in areas with different peak traffic hours, the lighter loaded concentrators will cause negligible additional impairment.

Assuming that present and future concentrators will be operated and designed so as to meet the criterion of very small speech impairment during the peak of the busy hour, it is recommended that no restrictions be imposed on tandem operation of concentrators at this time. In addition, it is recommended that no test on tandem operation should be made until tandem operation of concentrators is a reality. At such time, tests could be made under working conditions to determine the effects of tandem concentrators on speech and to establish whether any adjustment of the ratio of number of simultaneous calls to the number of channels would be required to keep speech clipping to a negligible amount.

The estimated probability that the forward-transfer pulse for the CCITT No. 5 Signalling System will be clipped for a certain length of time in one, two, and three TASIs in tandem can be found in [8].

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[1] FRASER (J. M.), BULLOCK (D. B.) and LONG (N. G.): Overall characteristics of a TASI system, *Bell System Technical Journal*, Vol. XLI, No. 4, July 1962.

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[3] DAYONNET (F. D.), JOUSSET (A.) and PROFIT (A.): Le CELTIC: Concentrateur exploitant les temps d'inactivité des circuits, *L'Onde Electrique*, Vol. XLII, No. 426, pp. 675-687, September 1962.

[4] LYGHOUNIS (E.): Il sistema A.T.I.C., *Telecommunicazioni*, No. 26, pp. 21-29, March 1968.

[5] BONATTI (M.) and MOTOLESE (F.): Probabilità di attività delle giunzioni di un doppio fascio telefonico, *Telecommunicazioni*, No. 23, pp. 24-28, June 1967.

[6] CCITT Recommendation *FDM carrier systems for submarine cable*, Vol. III, Fascicle III.2, Rec. G.371, § 3.

[7] HELDER (G. K.): Customer evaluation of telephone circuits with delay, *B.S.T.J.*, Vol. XLV, No. 7, September 1966.

[8] *TASI characteristics affecting signalling*, Green Book, Vol. VI.4, Supplement No. 2, ITU, Geneva, 1973.

Recommendation G.164

ECHO SUPPRESSORS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984 and at Melbourne, 1988)

1 General

1.1 *Application*

This Recommendation is applicable to the design of echo suppressors used on international telephone connections which have:

1.1.1 mean one-way propagation times between subscribers of up to the maximum regarded as acceptable in Recommendation G.114. (The design of the echo suppressor should not impose any lower limit of delay on its use);

1.1.2 a level of circuit noise entering the send-in port (S_{in}) or receive-in port (R_{in}) of up to -40 dBm0p;

1.1.3 round trip end delays between the receive-out port (R_{out}) and S_{in} port of the echo suppressor of up to 24 ms (including all transmission and switching plant).

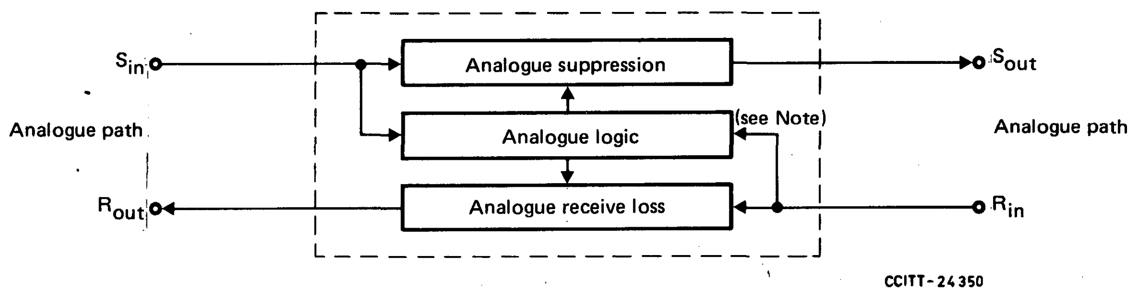
Note – Recommendation G.161 [1] refers to 25 ms. The value of 24 ms, a multiple of 2, is used in this Recommendation as being more applicable to the design of digital echo suppressors;

1.1.4 a loss of the echo path in dB (see the Recommendation cited in [2] that is likely to be such that the minimum loss from R_{out} to S_{in} of the echo suppressor will be equal to the difference between relative levels at these two ports plus 6 dB.

Echo suppressors must be designed to perform in a satisfactory manner under all the conditions described above.

1.2 *Design features*

Echo suppressors conforming to the characteristics given in this Recommendation are terminal, half-echo suppressors having differential operation and a break-in algorithm which incorporates a partial break-in state. They may be further characterized by whether the transmission paths, the logic functions and the speech processing (suppression and receive loss) use analogue or digital techniques. The combinations of these which are most likely to be practicable and to which this Recommendation is particularly addressed are shown in Figures 1/G.164, 2/G.164, 3/G.164 and 4/G.164 as Types A, B, C and D. All the requirements of this Recommendation apply equally to Types A, B, C and D except where noted.



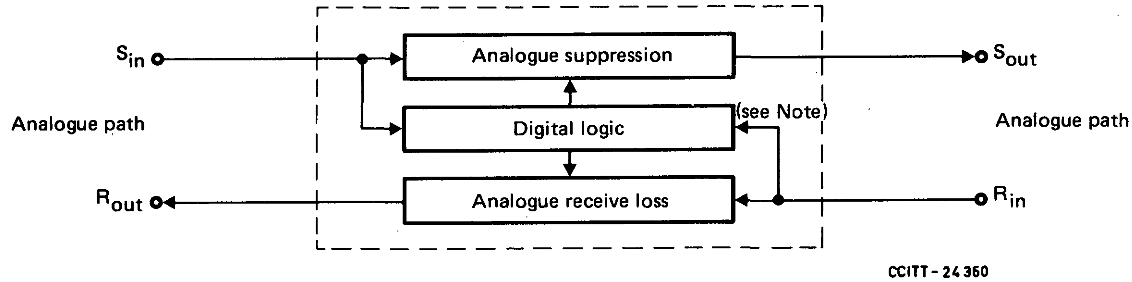
Abbreviations: Receive-in port (R_{in}) Send-in port (S_{in})

Receive-out port (R_{out}) Send-out port (S_{out})

Note – **This input may be connected to either side of the receive loss, depending on the logic circuitry.**

FIGURE 1/G.164

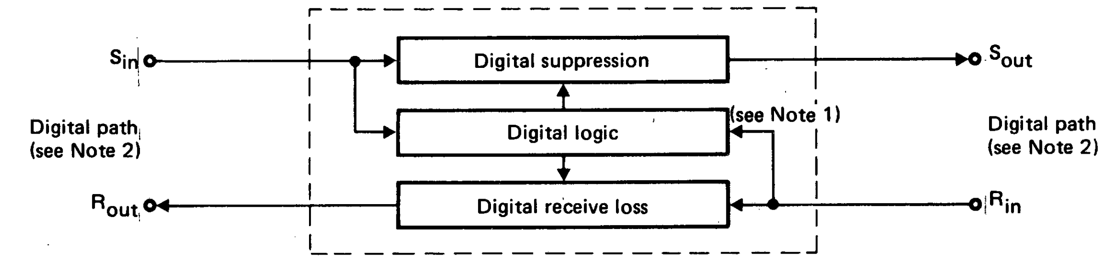
Type A echo suppressor



Note – **This input may be connected to either side of the receive loss, depending on the logic circuitry.**

FIGURE 2/G.164

Type B echo suppressor

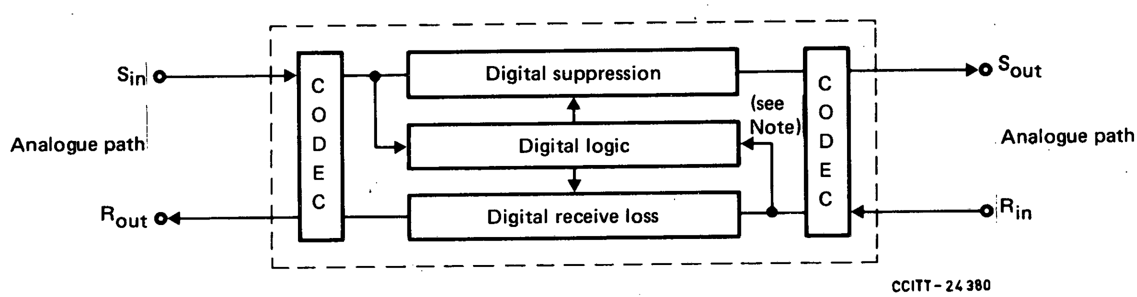


Note 1 – **This input may be connected to either side of the receive loss, depending on the logic circuitry.**

Note 2 – **The digital path may be at any digital interface, i.e. 64 kbit/s, 1544 or 2048 kbit/s or at any higher order interface.**

FIGURE 3/G.164

Type C echo suppressor



Note – **This input may be connected to either side of the receive loss, depending on the logic circuitry.**

FIGURE 4/G.164

Type D echo suppressor

1.3 *Variants*

1.3.1 Recommendation G.161 [1] is still applicable for the design of analogue echo suppressors. Analogue echo suppressors must conform to either Recommendation G.164 in its entirety or Recommendation G.161 [1] in its entirety.

1.3.2 This Recommendation is applicable to echo suppressors that employ fixed differential sensitivity, see § 3, and those that employ adaptive differential sensitivity, see § 4.

1.4 *Compatibility*

It is necessary for all echo control devices used on international connections to be compatible with each other. Echo suppressors designed according to this Recommendation will be compatible with each other, with echo suppressors conforming to Recommendation G.161 [1] and with echo cancellers designed to Recommendation G.165. Compatibility is defined as follows:

Given:

1) that a particular type of echo control device (say Type I) has been designed so that satisfactory performance is achieved when any practical connection is equipped with a pair of such devices, and

2) that another particular type of echo control device (say Type II) has been likewise designed;

then Type II is said to be compatible with Type I if it is possible to replace an echo control device of one type with one of the other type, without degrading the performance of the connection to an unsatisfactory level.

In this sense compatibility does not imply that the same test apparatus or methods can necessarily be used to test both Type I and Type II echo control devices.

1.5 *Need for test methods*

Objective test methods are very important to permit measurement of essential operating characteristics of echo suppressors. Suitable test methods are therefore given in § 6 of this Recommendation. Echo suppressors must operate properly in response to speech signals. Because of the difficulty of defining a speech test signal, the following tests are type tests and rely on the use of sine wave signals for convenience and repeatability. These tests should be performed on echo suppressors only after the design has been shown to properly operate on speech input signals.

1.6 *Enabling/disabling*

Each echo suppressor should be equipped with:

a) a facility which provides for enabling or disabling by an externally derived ground (earth) from the trunk circuit. The enabler should function to permit or prevent normal echo suppressor operation. Certain Type C echo suppressors may be disabled directly by a digital signal.

   Some digital data signals may require type C echo suppressors to provide 64 kbit/s bit sequence integrity in the externally disabled state.

b) a tone disabler which functions to prevent the introduction of the suppression and receive loss when specified disabling tone signals are transmitted through the suppressors. Thus it should disable for specified tones but should not disable on speech. (See § 5.)

1.7 *Explanatory notes*

1.7.1 When an echo suppressor is in its suppression mode, it places a large loss in the return path which, besides suppressing echo, prevents the speech of the second party of the conversation from reaching the first party when both parties are talking simultaneously (termed "double-talking"). To reduce this effect (called "chopping") during double-talking, the echo suppressor must be able to operate in a second mode when both parties are talking simultaneously. The terminology usually used is that the second party must be able to "break-in" or remove suppression when the second party interrupts during an utterance by the first party.

1.7.2 The result of break-in is to transform the circuit from one permitting speech in one direction to one permitting speech on both directions simultaneously, and a necessary consequence of this action is to permit echo to return unsuppressed. To reduce the amount of echo returned during break-in, loss is inserted in the receive path. This, of course, attenuates the received speech. If the break-in action is adjusted to minimize the echo, the speech of one or both double-talking parties will still be chopped to some extent as the control of the echo suppressor transfers from one party to the other. The basic requirements in the design of an echo suppressor are therefore two:

1) to provide adequate suppression of echo when speech from one talker only is present;

2) to provide ease and unobtrusiveness of break-in during double-talking.

The second requirement involves two mutually exclusive functions:

a) avoidance of chopping of double-talking speech;

b) elimination of echo during and after double-talking.

1.7.3 A differential circuit is used to detect the condition when break-in should take place. The level of the speech in the send path is compared with the level of the speech in the receive path to determine whether the send speech is the echo of the first party, or speech of the second party. Echo is reduced in level by the echo path loss and is delayed by twice the propagation time between the echo suppressor and the points of reflection. (The round trip delay in the echo path is called "end-delay".) The minimum echo path loss and the maximum end-delay must be considered in the design of the differential circuit.

1.7.4 Echo suppressors with fixed differential sensitivity are designed such that if speech in the send path is below the level of the expected echo (considering the minimum echo path loss), suppression will not be removed. If speech in the send path is above the level of the expected echo, break-in will occur and the suppression will be removed.

1.7.5 Echo suppressors with adaptive differential sensitivity are designed to adapt to the actual echo path loss on the connection (which is usually substantially higher than the minimum value, see Recommendation G.122, § 2). Speech in the send path is thus more often above the level of the expected echo, and break-in occurs more easily. The adaptation time is typically less than one second and adaptation is stopped or slowed down during double-talking. The adaptive function reduces the degradation in the send path due to speech chopping.

1.7.6 Break-in hangover is used to minimize chopping of double-talking speech. A two step process is recommended as a protection against false break-in due to echo or to impulse noise:

a) The state of partial break-in is entered initially. This state is characterized by short break-in hangover times. The receive loss may or may not be inserted but, if used, must have an equally short break-in hangover time.

b) After the signal conditions producing break-in have persisted for some time, the full break-in state is entered. Receive loss must be inserted and longer break-in hangover times applied.

2 Definitions relating to echo suppressors

2.1 echo suppressor

F: suppresseur d'écho

S: supresor de eco

A voice-operated device placed in the 4-wire portion of a circuit and used for inserting loss in the transmission path to suppress echo. The path in which the device operates may be an individual circuit path or a path carrying a multiplexed signal.

2.2 full echo suppressor

F: suppresseur d'écho complet

S: supresor de eco completo

An echo suppressor in which the speech signals on either path control the suppression loss in the other path.

2.3 half-echo suppressor

F: demi-suppresseur d'écho

S: semisupresor de eco

An echo suppressor in which the speech signals of one path control the suppression loss in the other path but in which this action is not reciprocal.

2.4 differential echo suppressor

F: suppresseur d'écho différentiel

S: supresor de eco diferencial

An echo suppressor whose operation is controlled by the difference in level between the signals on the two speech paths.

2.5 **partial break-in echo suppressor**

F: suppresseur d'écho à intervention partielle

S: supresor de eco con intervención parcial

An echo suppressor which includes partial and full break-in functions.

2.6 **adaptive break-in echo suppressor**

F: suppresseur d'écho à intervention adaptable

S: supresor de eco con intervención adaptativa

An echo suppressor in which the break-in differential sensitivity is automatically adjusted according to the attenuation of the echo path.

2.7 **suppression loss**

F: affaiblissement de blocage

S: atenuación para la supresión

The specified minimum loss which an echo suppressor introduces into the send path (of the echo suppressor) to reduce the effect of echo currents.

2.8 **receive loss**

F: affaiblissement à la réception

S: atenuación en la recepción

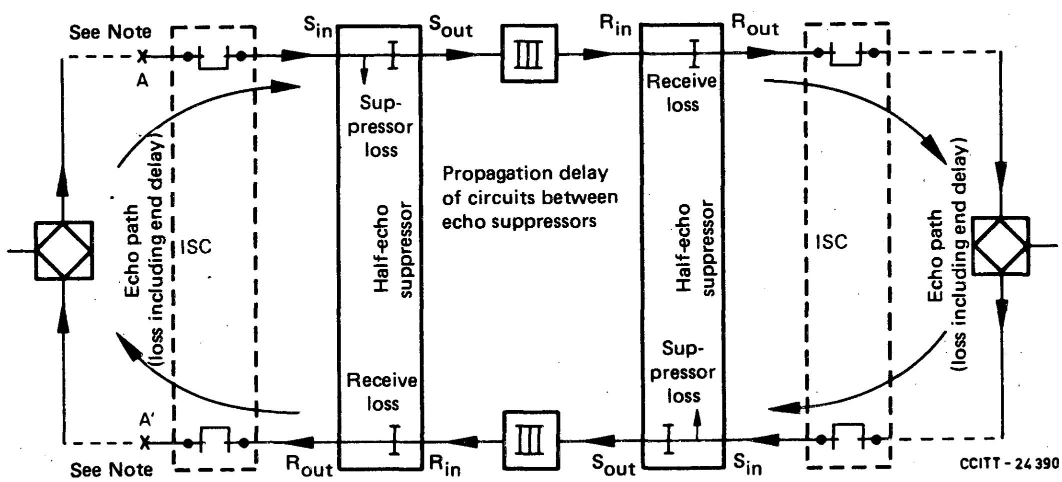
The specified loss which an echo suppressor introduces into the receive path (of the echo suppressor) to reduce the effect of echo currents during break-in.

2.9 **terminal echo suppressor** (see Figure 5/G.164)

F: suppresseur d'écho terminal

S: supresor de eco terminal

An echo suppressor designed for operation at one or both terminals of a circuit.



ISC International switching centre

Note – **In some applications the echo suppressor is inserted at point A, A'.**

FIGURE 5/G.164

2.10 **suppression operate time**

F: temps de fonctionnement pour le blocage

S: tiempo de funcionamiento para la supresión

The time interval between the instant when defined test signals, applied to the send- and/or receive-in ports, are altered in a defined manner and the instant when the suppression loss is introduced into the send path of the echo suppressor.

2.11 **suppression hangover time**

F: temps de maintien pour le blocage

S: tiempo de bloqueo para la supresión

The time interval between the instant when defined test signals applied to the send- and/or receive-in ports are altered in a defined manner, and the instant when the suppression loss is removed from the send path.

2.12 **partial break-in**

F: intervention partielle

S: intervención parcial

A temporary condition of break-in which exists at the onset of break-in. This state is characterized by a short break-in hangover time. The receive loss may be inserted during partial break-in provided it also has the short break-in hangover time.

2.13 **partial break-in operate time**

F: temps de fonctionnement pour l'intervention partielle

S: tiempo de funcionamiento para la intervención parcial

The time interval between the instant when defined test signals, applied to the send- and/or receive-in ports, are altered in a defined manner such as to remove suppression and the instant when suppression is removed. Insertion of loss in the receive path may occur at the same time or slightly after removal of suppression.

2.14 **full break-in**

F: intervention totale

S: intervención total

A stable condition of break-in which follows the partial break-in condition once it has been determined, with high probability, that the signal causing break-in is speech. This state is characterized by the insertion of receive loss and longer break-in hangover times.

2.15 **full break-in operate time**

F: temps de fonctionnement pour l'intervention totale

S: tiempo de funcionamiento para la intervención total

The time interval between the instant when defined test signals, applied to the send- and/or receive-in ports, are altered in a defined manner such as to remove suppression and extend the hangover time and the instant when the extended hangover time is applied. Removal of suppression occurs at the same time as for partial break-in. Insertion of loss in the receive path may occur at the same time or slightly after removal of suppression.

2.16 **break-in hangover time**

F: temps de maintien pour l'intervention

S: tiempo de bloqueo para la intervención

The time interval between the instant when defined test signals, applied to the send- and/or receive-in ports, are altered in a defined manner such as to restore suppression and the instant when suppression is restored. The hangover time for removal of loss in the receive path may be longer than that for restoration of suppression.

2.17 **differential sensitivity**

F: sensibilité différentielle

S: sensibilidad diferencial

The difference, in dB, between the relative level of the test signals applied to the send path and receive path when break-in occurs.

3 Characteristics of echo suppressors with fixed break-in differential sensitivity

3.1 *Transmission performance*

The performance characteristics apply, unless otherwise noted, when steady state signals are separately applied to the send and receive paths.

The limits on transmission characteristics specified below shall be observed over the temperature range +10 °C to 40 °C and over the power supply variations permitted by individual Administrations.

Echo suppressors of Types A, B and D are placed in the voice-frequency portion of a 4-wire circuit which is nominally of 600-ohms impedance. The send (transmit or office-to-line) and the receive (line-to-office) paths are at different relative levels in different national networks; two such sets of levels are:

1) send, −16 dBr; receive, +7 dBr;
2) send, −4 dBr; receive, +4 dBr.

Test tone frequencies are 800 Hz or 1000 Hz, nominal. To avoid submultiples of the 8000-Hz sampling frequency, test tone frequencies should fall within the ranges 804 to 860 Hz and 1004 to 1020 Hz respectively.

3.1.1 *Type A and B echo suppressors*

3.1.1.1 *Insertion loss*

The insertion loss at 800 Hz (or 1000 Hz) of an echo suppressor in an unoperated condition shall be 0 ± 0.3 dB, for test tone levels < 0 dBm0.

3.1.1.2 *Attenuation distortion*

The attenuation distortion shall be such that if Q dB is the loss at 800 Hz (or 1000 Hz), the loss shall be within the range $(Q + 0.3)$ dB to $(Q - 0.2)$ dB at any frequency in the band 300-3400 Hz, and at 200 Hz within the range of $(Q + 1.0)$ dB to $(Q - 0.2)$ dB.

3.1.1.3 *Delay distortion*

The delay distortion shall not exceed 30 μs measured between any two frequencies in the band 1000-2400 Hz and 60 μs in the band 500-3000 Hz.

3.1.1.4 *Impedance*

The values of impedance and return loss shall apply to all states of operation of the echo suppressors.

1) The nominal value of the inputs and outputs shall be 600 ohms (nonreactive).
2) The return loss with respect to the nominal impedance shall not be less than 20 dB from 300-600 Hz nor less than 25 dB from 600-3400 Hz.
3) The impedance unbalance to earth of each port shall not be less than 50 dB over the frequency range 300 to 3400 Hz.

3.1.1.5 *Overload*

The insertion loss at 800 Hz (or 1000 Hz) shall not increase by more than 0.2 dB for test tone levels from 0 to +5.0 dBm0.

3.1.1.6 *Harmonic distortion*

The total harmonic distortion power, for a pure 800 Hz (or 1000 Hz) sine wave at a level of 0 dBm0, shall not exceed −34 dBm0.

3.1.1.7 *Intermodulation*

For frequencies $f_1 = 900$ Hz and $f_2 = 1020$ Hz applied simultaneously each at a level of −5 dBm0, the difference between the output levels of either frequency f_1 or f_2 and the level of either of the intermodulation products at $(2f_1 - f_2)$ or $(2f_2 - f_1)$ should be at least 45 dB. When speech compressors are used to provide loss during break-in, this requirement is reduced during the break-in mode to 26 dB for the receive path (W-state receive path).

3.1.1.8 *Transient response*

If loss devices which are inserted in the receive path operate at a syllabic rate, the transient performance of such devices should conform to Recommendation G.162 which deals with the overall transient response of compandors.

3.1.1.9 *Noise*

The mean weighted psophometric power introduced by an echo suppressor shall not exceed −70 dBm0p. The mean unweighted noise power in a band of 300-3400 Hz introduced by an echo suppressor shall not exceed −50 dBm0.

3.1.1.10 *Crosstalk*

When an echo suppressor is installed in a working circuit, the crosstalk attenuation between the send path and the receive path (and conversely) shall be such that the signal power in the disturbed path due to crosstalk from the disturbing path shall not exceed −65 dBm0 for any sinusoidal signal in the disturbing path having a power of +5 dBm0 or less and within the band 300-3400 Hz.

3.1.1.11 *Spurious outputs produced by the echo suppressor*

The various operations of the echo suppressor must not result in any appreciable spurious outputs such as internally generated impulses due to transient conditions. In particular these must not be of such magnitude as would be likely to falsely operate the suppression or break-in feature of any other echo suppressor that might be in the connection. Consideration must include that of multilink connections having several pairs of echo suppressors in tandem.

To prevent false operation of other echo suppressors in a built-up connection, the zero-to-peak voltage of any transient output produced in the receive or transmit paths (terminated in 600 ohms) due to echo suppressor operation caused by signals in the opposite path should not exceed 20 mV at a point of zero relative level (−34 dBV0) after first filtering the transient to a 500 to 3000 Hz bandwidth. Additionally, the duration of any such transient should be such that it is not audible in the presence of normal levels of noise (e.g. −50 dBm0p).

3.1.2 *Type C echo suppressor*

3.1.2.1 *General*

An echo suppressor of Type C inserted into a digital transmission path between codecs meeting the performance characteristics of Recommendation G.712 [3] should not alter such performance.

3.1.2.2 *Group delay*

The group delay through the echo suppressor shall not exceed 0.25 ms.

3.1.2.3 *Effect of digital loss pads*

Digital loss pads inserted into the receive path during the break-in mode may increase the quantizing distortion. Type C echo suppressors, which maintain signalling bit integrity for channel associated signalling for systems in accordance with Recommendation G.733 [4] by bypassing the least significant bit, are likely to exhibit a greater increase in quantizing distortion during the break-in mode than Type C echo suppressors used in systems with common channel signalling. See Footnote c) to Table 1/G.164.

3.1.2.4 *Effect of instantaneous digital compressors*

When an instantaneous compressor is employed in the receive path of the suppressor during break-in, it shall not produce distortion exceeding the following limits:

a) *Harmonic distortion*

With a sinusoidal input signal of 0 dBm0 at any frequency between 300 Hz and 1 KHz, the third harmonic distortion produced should not exceed -30 dBm0.

b) *Intermodulation distortion*

With an input signal of two equal amplitude sinusoids at $f_1 = 900$ and $f_2 = 1020$ Hz at levels of -3 to -35 dBm0, the distortion products at $(2f_1 - f_2)$ and $(2f_2 - f_1)$ should not exceed a level of -16 dB relative to the output level of each tone. For input levels below -35 dBm0 this ratio should be at least -20 dB.

3.1.3 *Type D echo suppressors*

3.1.3.1 *General*

The performance characteristics of Recommendation G.712 [3] apply for the codecs.

3.1.3.2 *Group delay*

The group delay shall not exceed that of the codecs alone by more than 0.25 ms.

3.1.3.3 *Effect of digital loss pads*

Digital loss pads inserted into the receive path during the break-in mode may increase the quantization distortion over the limits specified in Recommendation G.712 [3]. See Footnote c) to Table 1/G.164.

3.1.3.4 *Effect of instantaneous digital compressors*

See § 3.1.2.4.

3.2 *Characteristics with steady-state input signals applied independently to the send and receive paths*

3.2.1 The action of an echo suppressor with fixed differential sensitivity which incorporates the general features described in § 1 is explained below with the aid of the idealized operational diagram shown in Figure 6/G.164. The significant combinations of input signals are represented by the areas X, Y, Z, W and V.

3.2.2 The area X corresponds to the absence of any appreciable signal on either the send or the receive path. The area Y corresponds to the presence of signals only on the send path. The area Z represents those combinations of signal levels for which the echo suppressor should provide suppression in the send path. The area W corresponds to break-in when the suppression should be absent. The area V corresponds to hysteresis that is provided to ensure that the break-in condition is retained when the signal on the send path has fallen slightly below the minimum level at which break-in would be initiated; the area V therefore represents a bistable condition. Table 1/G.164 shows the losses that should be inserted in the two paths, when each of the five areas X, Y, Z, W and V is occupied continuously. The right hand column of the table refers to tests described in § 6. Figure 7/G.164 shows the boundaries for the receiving loss C, that should be inserted in the receive path during break-in. The information given in Figures 6/G.164 and 7/G.164 and in Table 1/G.164 applies for steady-state signals with the inter-area boundaries being crossed very slowly.

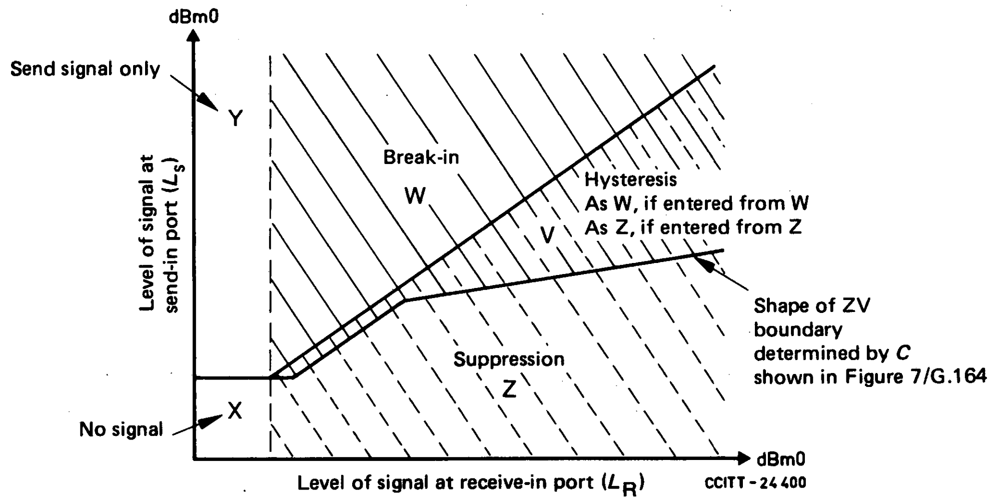


FIGURE 6/G.164

Conceptual diagram showing operational states of echo suppressors with fixed differential sensitivity under ideal conditions

TABLE 1/G.164

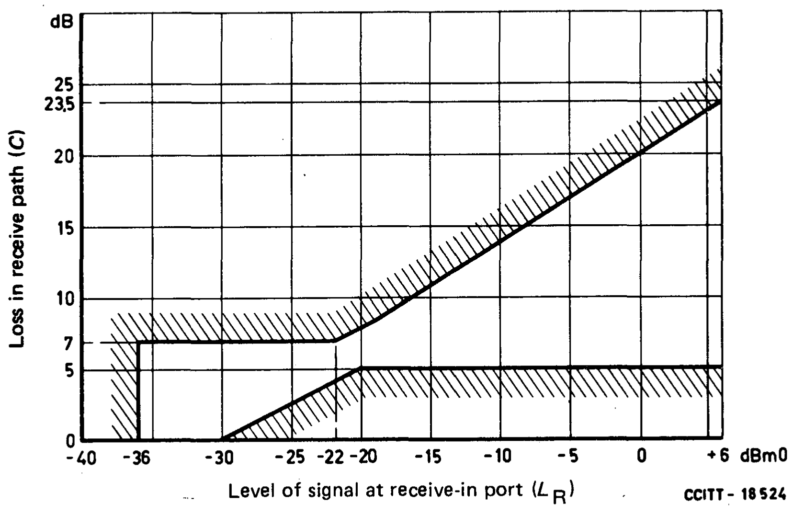
Key to operational diagram Figure 6/G.164

Area	Loss in send path (dB)	Loss in receive path (dB)	Test No.
X	0	0	1
Y	0	0 [b]	2
W	0	Within limits for *C* shown in Figure 7/G.164 [c]	2
Z	50 minimum [a]	0	1
V	As W, if entered from W As Z, if entered from Z		

[a] When echo suppressors are used on low noise circuits, suppression of the far end noise may be objectionable due to noise contrast. Two administrations have shown that this impairment may be reduced by the insertion of noise, equivalent to far end noise, during suppression.

[b] When the loss in the receive path is provided by a speech compressor, the loss should be zero for receive signals $\leqslant -36$ dBm0.

[c] Information given in Supplement No. 21 at the end of this fascicle indicates that for A-law encoded telephone signals, the additional quantizing distortion due to a fixed digital loss pad is minimum for a loss value of 6 dB. For high level receive signals this will also apply to loss values which are an integer multiple of 6 dB. For μ-law encoded telephone signals the additional quantizing distortion is practically independent of the digital loss pad value.



Note – **The recommended values are those enclosed in the nonshaded area.**

FIGURE 7/G.164

Recommended loss C, to be inserted in receive path during break-in

3.2.3 The features shown in Figure 6/G.164 are concerned only with characteristics that can be determined without knowledge of, or access to, the internal circuits of echo suppressors. These characteristics are determined by application of test signals to the external terminals of the echo suppressor and observation of its state by external measurements. Test methods for measurements to verify compliance with the requirements are given in § 6.

3.2.4 The signal levels that define the various thresholds are given in Table 2/G.164.

3.2.4.1 The nominal suppression threshold is -31 dBm0 when there is essentially no speech in the send path. The release from suppression is also nominally -31 dBm0 but can be as much as 3 dB below the suppression threshold. When signals above the threshold exist in both the send and receive paths, the intent of the requirement is that the echo suppressor be in the suppress (Z) state if $L_R \geqslant L_S$, should transfer to the break-in (W) state for $L_S \geqslant L_R$ and should revert to the suppression state for $L_R \geqslant L_S + C$. Tolerances are provided to account for filter, power supply and temperature variations.

3.2.4.2 The frequency response limits of the suppression control path are given in Figure 8/G.164. The frequency response limits of the break-in control paths are given in Figure 9/G.164. It is desirable to provide such filtering in echo suppressors. However, this is difficult to implement in the case of Types C and D. Therefore, for these types, this filtering may be omitted where Administrations can ensure that any interfering signals are at such a low level that they do not adversely affect echo suppressor operation. Tests 1 and 3 of § 6 can be used to measure the frequency responses.

TABLE 2/G.164

Inter-area threshold levels

Boundary	Symbol of threshold	At 1000 Hz (see Note 1) dBm0 at 20 ± 5 °C	At 1000 Hz (see Note 1) dBm0 between 10 and 40 °C	Variation with frequency	Test No.
Suppression					
X to Z	Txz	$-33 \leq Txz \leq -29$ for $L_S = -40$	$T'xz = Txz \pm 1$		1
Z to X	$Tzx_{max.}$ $Tzx_{min.}$	$Txz - 0$ dB $Txz - 3$ dB	$T'xz - 0$ dB $T'xz - 3$ dB	Figure 8/G.164	1
Break-in					
V to W (previous input Z)	Tvw	$L_R - 3 \leq L_S \leq L_R$ (see Notes 3, 4, 5 and 6) $(-26.5 \leq L_R \leq +3)$		$T'vw = Tvw \pm 1.5$ dB between 500 and 3000 Hz (see Note 2)	3
V to Z (previous input W)	$Tvz_{max.}$ $Tvz_{min.}$	$Tvw - C + 2$ dB (see Notes 3, 4 and 5) $Tvw - C - 3$ dB $(-26.5 \leq L_R \leq +3)$		$T'vz = Tvz \pm 1.5$ dB between 500 and 3000 Hz (see Note 2)	3

L_S Level (dBm0) at send-in port.

L_R Level (dBm0) at receive-in port.

C Le loss inserted in the receiving path during break-in. This characteristic must comform with the limits shown in Figure 7/G.164.

Note 1 – The test frequency is 1004 to 1020 Hz to avoid submultiples of the 8000 Hz sampling frequency.

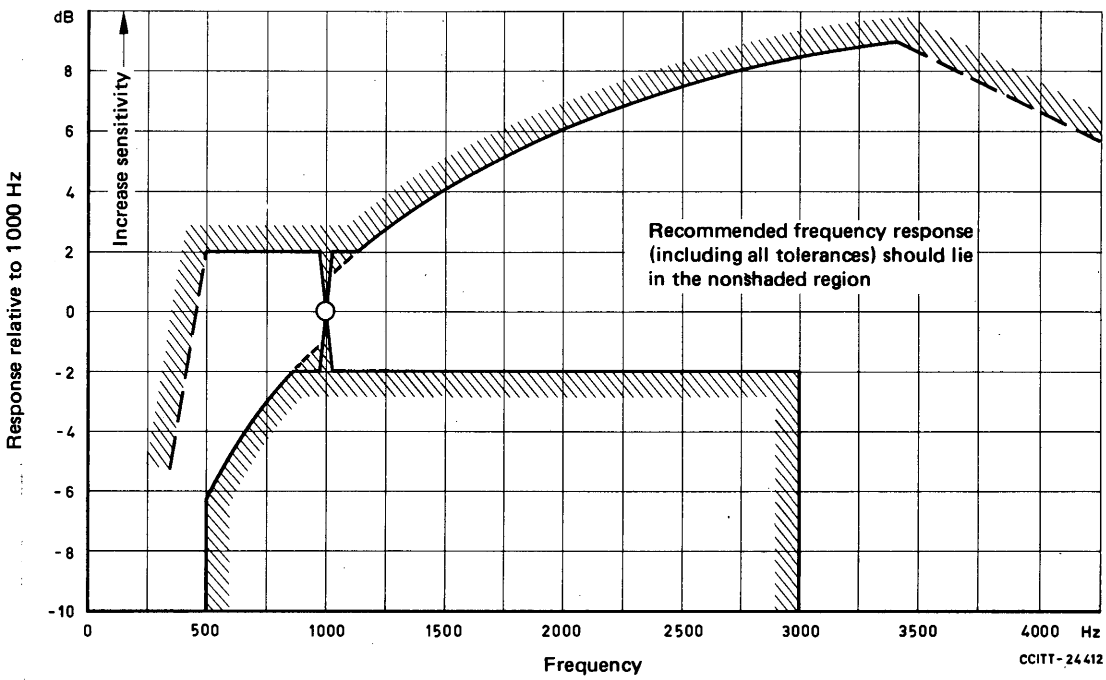
Note 2 – Tolerances in the attenuation/frequency characteristics of the two filters of the break-in detector must be taken into account, but it is desirable that the break-in threshold should be as independent of frequency as possible; a tolerance of ± 1.5 dB should apply if L_S and L_R are varied together over the frequency range 500-3000 Hz.

Note 3 – This excludes tolerances due to codecs (± 0.5 dB in Recommendation G.712 [3]).

Note 4 – The Tvw and Tvz tolerance limits may occasionally be exceeded by up to 1 dB in the range $-26.5 \leq L_R \leq +3$ dBm0 due to quantizing effects. This can, in theory, cause false retention of break-in when using steady state test signals (see test 8). This does not occur for speech signals.

Note 5 – The limiting values of the Tvw and Tvz thresholds combined with small values of echo path loss and small values of C can, in theory, cause oscillation between suppression and break-in for tests using low-level steady state signals. This has not been observed on existing echo suppressors and does not occur for speech signals.

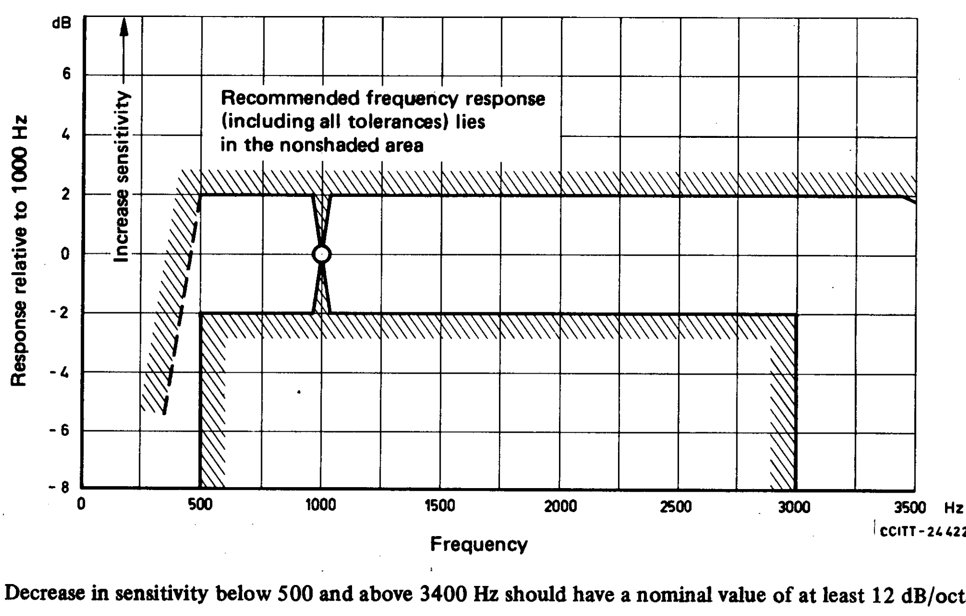
Note 6 – The fixed Tvw threshold, symbolizing differential sensitivity having a nominal value of 0 dB, ensures against false break-in due to echo for a minimum echo path loss of 6 dB (see § 1.1.4).



Note – Decrease in sensitivity below 500 and above 3400 Hz should have a value of at least 12 dB/octave.

FIGURE 8/G.164

Recommended frequency response of suppression control path of echo suppressor



Note – Decrease in sensitivity below 500 and above 3400 Hz should have a nominal value of at least 12 dB/octave.

FIGURE 9/G.164

Recommended frequency response of each control path of break-in detector of echo suppressor

k193.3 *Dynamic characteristics when signals are applied, removed or changed in the send and receive paths independently*

3.3.1 The dynamic characteristics can be specified by stating the time that elapses when the conditions of the signals pass from a point in one area to one in another before the state appropriate to the second area is established (Figure 6/G.164 and Figure 12/G.164). When passing from X to Z, this is termed the suppression operate time and when passing in the opposite direction it is termed suppression hangover time. When passing from the Z area through V to W (or Y) it is termed the break-in operate time and when passing from W through V to Z it is termed the break-in hangover time. The V/W and V/Z boundaries may, in practice, be crossed at any angle; the requirements in Table 3/G.164 deal with vertical and horizontal directions.

3.3.2 The suppression (X/Z) operate time should be nearly constant for the sudden application of any signal in the receive path greater than the threshold (−31 dBm0) in the absence of any appreciable signal in the send path. Similarly, for transitions from suppression to break-in for L_R constant (Z/V/W), the operate times shown in Table 3/G.164 should in general apply to the complete range of possible signal pairs (L_R and L_S) and not just to the two pairs shown in Table 3/G.164.

3.3.3 The hangover times shown in Table 4/G.164 should in general apply whenever suppression or break-in has occurred irrespective of the levels of the causative signals.

3.3.4 When sudden changes are made in the levels of sinusoidal test signals at a frequency of 1000 Hz, the times of operation given in Table 3/G.164 apply and the recommended values of hangover given in Table 4/G.164 apply. The right-hand part of each table refers to tests described in § 6.

3.3.5 The operate times of the receive pad in the Y/W transition is not separately stated or tested, but should be within the limits allowed for the suppression operate time.

3.4 *Performance under conditions of small echo-path loss and when end-delay may be present*

The foregoing requirements apply when the echo suppressor is tested under conditions such that the signals in the send and receive paths are independent. In practice, satisfactory performance must also be maintained when the send path is connected to the receive path through an echo path that may have end-delay and low loss. Three features of the dynamic performance must be checked under these conditions. § 6 describes test arrangements suitable for measuring these conditions. The three conditions are described as follows:

3.4.1 An echo (leakage through the echo path) must not cause false operation of the break-in condition when the echo-path loss is low and the end-delay is zero. The trouble could be caused by inappropriate design of the control path time constants. When a signal is suddenly applied to R_{in}, this trouble would show itself as a temporary false operation of the break-in condition, persisting for the duration of the break-in hangover time (see Test No. 7).

3.4.2 If insufficient protection against end-delay is incorporated in the echo suppressor, the break-in circuit may operate on the trailing edge of the echo. This can occur with the sudden removal of a signal at R_{in} when the echo-path loss is low and the end-delay is large (see Test No. 7).

3.4.3 In certain designs it can happen that the hysteresis represented by the bistable area V (see Figure 6/G.164) is excessive in relation to the amount of loss inserted in the receive path. This can result in the false retention of break-in by echo occurring under the following conditions: A steady-state signal is present at R_{in} port and is coupled to S_{in} port via the echo path. A signal of sufficient amplitude and duration to cause break-in is then applied to S_{in} port. Upon cessation of this signal, the echo of the receive signal falsely maintains the break-in condition (see Test No. 8).

TABLE 3/G.164

Operate times

Boundary	Initial signals (see Note)		Final signals (see Note)		Recommended value (ms)	Test No.	Excursion (see Figure 12/G.164)	Test circuit (Figure number)	Oscilloscope trace (Figure number)
	Send L_S (dBm0)	Receive L_R (dBm0)	Send L_S (dBm0)	Receive L_R (dBm0)					
Suppression X/Z	−40 −40	−40 −40	−40 −40	−25 −11	⩽ 2	4	a ⟶ b a ⟶ d	14/G.164	15/G.164
Break-in Z/V/W L_S constant	−15 −15 −15	−10 − 5 0	−15 −15 −15	−25 −25 −25	24-36	5	h ⟶ i g ⟶ i f ⟶ i	14/G.164	16/G.164
Break-in Z/V/W L_R constant	−40 −40	−25 −15	−19 − 9	−25 −15	Partial: ⩽ 2 Full: 6-10	6	b ⟶ k c ⟶ j	17/G.164	17/G.164

Note – See also § 3.3.2.

TABLE 4/G.164

Hangover times

Boundary	Initial signals		Final signals		Recommended value (ms)	Test No.	Excursion (see Figure 12/G.164)	Test circuit (Figure number)	Oscilloscope trace (Figure number)
	Send L_S (dBm0)	Receive L_R (dBm0)	Send L_S (dBm0)	Receive L_R(dBm0)					
Suppression Z/X	−40 −40	−25 −11	−40 −40	−40 −40	24-36	4	b $\longrightarrow$ a d $\longrightarrow$ a	14/G.164	15/G.164
Break-in W/V/Z L_R constant	−19 − 9	−25 −15	−40 −40	−25 −15	Partial: ⩽ 26 Full: 48-66	6	k $\longrightarrow$ b j $\longrightarrow$ c	17/G.164	18/G.164

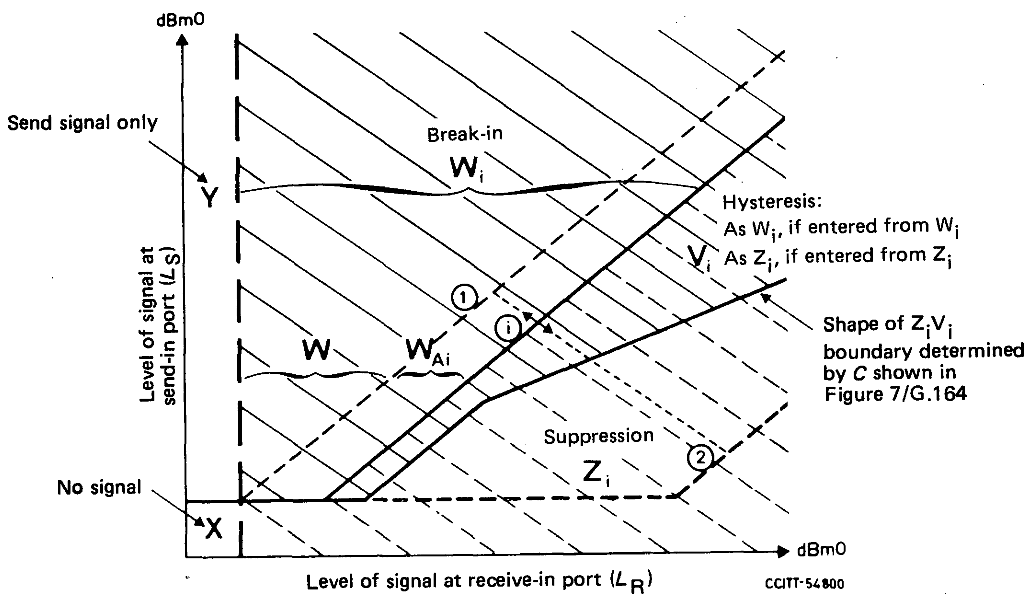
4 Characteristics of echo-suppressor with adaptive break-in differential sensitivity

4.1 *Provisions of § 3*

The provisions of § 3 apply to echo suppressors with adaptive break-in differential sensitivity when $a_x = 0$ (see below). Tests 1 through 8 (see § 6) must be performed and the requirements in Tables 1/G.164 and 2/G.164 apply only when $a_x = 0$.

4.2 *Characteristics of the adaptive function with signals applied independently to the send and receive paths*

4.2.1 The action of the adaptive function is explained below with the aid of the idealized operational diagram shown in Figure 10/G.164.



Note 1 – $W_i = W + W_{Ai}$, where W_{Ai} may vary with time

Note 2 – Line (i) ($L_S = L_R - a_x$) may vary between line (1) ($L_S = L_R$) and line (2) ($L_S = L_R - a_{x\max}$).

FIGURE 10/G.164

Conceptual diagram showing operational states of adaptive break-in echo suppressors under ideal conditions

4.2.2 The adaptive function automatically adjusts the differential sensitivity to the echo path loss. The adaptation characteristic a_x is used to describe this change in sensitivity, and the Tvw threshold (see Table 2/G.164) becomes the $T\mathrm{v_iw_i}$ threshold given by

$$L_R - a_x - 3 \leqslant L_S \leqslant L_R - a_x,$$

a_x is such that, for echo path losses of 6 dB, it equals 0 and the equation reduces to:

$$L_R - 3 \leqslant L_S \leqslant L_R$$

as given in Table 2/G.164.

4.2.3 When the echo suppressor is in the suppression mode (area Z_i, Figure 10/G.164) the adaptive function will cause rapid convergence of a_x to the value of the echo path loss a_E minus 6 dB, i.e.:

$$a_x = L_R - L_S - 6 \text{ dB}.$$

a_x may be quantized with up to approximately 3 dB steps. In this case the value of a_x after convergence shall be:

$$a_E - 9 \leqslant a_x \leqslant a_E - 6$$

except that $a_x \geqslant 0$. This equation shall hold for a_E at least up to 26 dB i.e. for $a_{x\,max} \geqslant 20$ dB (see Table 5/G.164). The rate of convergence is given in Table 6/G.164.

TABLE 5/G.164

Loss value of a_x after convergence in Z_i state

Echo path loss, a_E (dB)	a_x (dB)
$\leqslant 6$	0
7	0 to 1
8	0 to 2
9	0 to 3
10	1 to 4
.	.
.	.
x	x-9 to x-6
.	.
.	.
.	.
26	17 to 20
.	.
.	.
.	.
$a_{x\,max} + 6$	$a_{x\,max} - 3$ to $a_{x\,max}$
$a_{x\,max} + 7$	$a_{x\,max} - 2$ to $a_{x\,max}$
$a_{x\,max} + 8$	$a_{x\,max} - 1$ to $a_{x\,max}$
$\geqslant a_{x\,max} + 9$	$a_{x\,max}$

TABLE 6/G.164

Rate of change of adaptation characteristic a_x

Operational states (see Figure 10/G.164)		Variations of adaptation characteristic a_x	Rate of change	Test No.
Z_i		Adapting to the echo return loss (increasing or decreasing) ($a_x \rightarrow a_E - 6$ dB)	> 4 dB/s (See Note 1)	10 b)
Y		Storing the last value	–	–
W_i	W	Storing the last value	–	–
	W_{Ai}	Storing the last value or decreasing to minimum possible value causing suppression	(See Note 2)	10 a)
X		Clearing the last value ($a_x \rightarrow 0$ dB)	> 4 dB/s	10 c)
V_i		As Z_i, if entered from Z_i As W_i, if entered from W_i		

Note 1 – Rates of adaptation on speech for a_x of approximately 10 dB/s have been shown to be subjectively acceptable.

Note 2 – If a_x is decreased in the W_{Ai} region, the rate of change should not exceed the rate of adaptation for a_x in the Z_i region.

4.2.4 The break-in mode of the echo suppressor (area W_i) is divided into two sub-areas W and W_{Ai}.

4.2.4.1 In the W area the last value of a_x should be stored.

4.2.4.2 In the W_{Ai} area two different strategies are possible. The first is to store the last value of a_x. The second is to permit a_x to decrease toward zero. The rate of change of a_x should preferably be slower than the rate of adaptation (see Table 6/G.164, Note 2). Experience has shown that these two strategies perform very similarly when the echo suppressor is operating on speech rather than test sine waves.

4.2.5 When no speech is present (X area), a_x should decrease to zero (see Table 6/G.164).

4.2.6 Tests 9 and 10 in § 6, may be used to measure the dynamic characteristics of the adaptive function.

5 Characteristics of echo-suppressor tone disablers

5.1 *General*

Each echo suppressor should be equipped with a tone disabler which functions to prevent the introduction of the suppression and receive loss when data or other specified tone signals are transmitted through the suppressor. Thus it should disable for specified tones but should not disable on speech. The tone disabler should detect and respond to a disabling signal which may be present in the send or receive path.

5.2 *Disabling characteristics* (see Figure 11/G.164)

The disabling tone transmitted is 2100 Hz ± 15 Hz at a level of −12 ± 6 dBm0. The frequency of the tone applied to the disabler is 2100 Hz ± 21 Hz (see Recommendation V.21 [5]). The disabling channel bandwidth should be chosen wide enough to encompass this tone (and possibly other disabling tones used within national networks). At the same time, the disabling channel bandwidth should be such that, in conjunction with guard action and timing, adequate protection is provided against false operation of the disabler by speech signals. The disabling channel sensitivity (threshold level) should be such that the disabler will operate on the lowest expected power of the disabling tone. The band characteristics shown in Figure 11/G.164 will permit disabling by the 2100-Hz disabling tone as well as others used in North America. The figure indicates that in the frequency band 2079 Hz to 2121 Hz disabling *must* be possible whilst in the band 1900 Hz to 2350 Hz it *may* be possible.

Providing that only the recommended 2100-Hz disabling tone is used internationally, interference with signalling equipment will be avoided. Unintentional disabling of the echo suppressor by signalling tones is not considered detrimental, since the echo suppressor serves no needed functions during the time when signalling tones are present on the circuit.

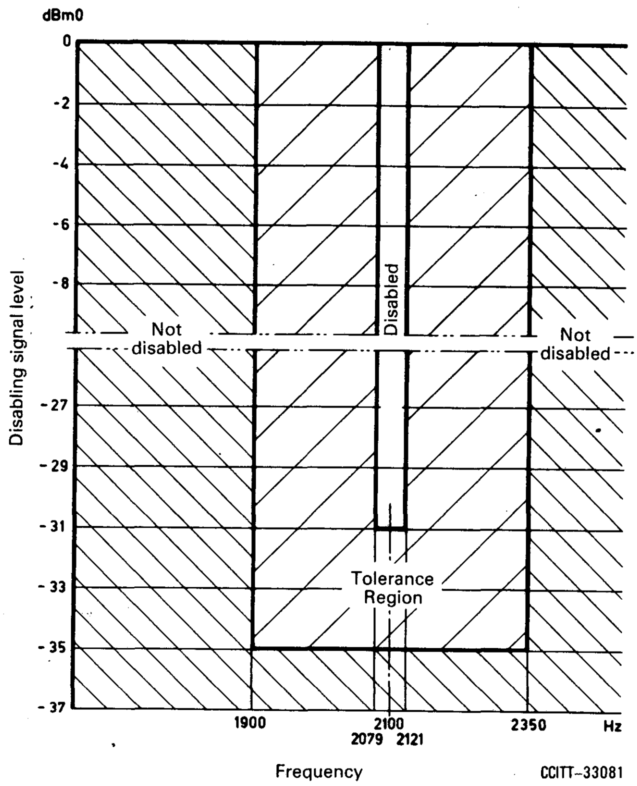


FIGURE 11/G.164

Required disabling band characteristics

5.3 *Guard band characteristics*

Energy in the voice band, excluding the disabling band, must be used to oppose disabling so that speech will not falsely operate the tone disabler. The guard band should be wide enough and with a sensitivity such that the speech energy outside the disabling band is utilized. The sensitivity and shape of the guard band must not be such that the maximum idle or busy circuit noise will prevent disabling. In the requirement, white noise is used to simulate speech and circuit noise. Thus, the requirement follows:

Given that white noise (in a band of approximately 300-3400 Hz) is applied to the tone disabler simultaneously with a 2100-Hz signal, the 2100-Hz signal is applied at a level 3 dB above the midband disabler threshold level. The white noise energy level required to inhibit disabling should be no greater than the level of the 2100-Hz signal and no less than a level 5 dB below the level of the 2100-Hz signal. As the level of the 2100-Hz signal is increased over the range of levels to 30 dB above the midband disabler threshold level, the white noise energy level required to inhibit disabling should always be less than the 2100-Hz signal level.

5.4 *Holding-band characteristics*

The tone disabler, after disabling, should hold in the disabled state for tones in a range of frequencies. The bandwidth of the holding mode should encompass all present or possible future data frequencies. The release sensitivity should be sufficient to maintain disabling for the lowest level data signals expected, but should be such that the disabler will release for the maximum idle or busy circuit noise. Thus the requirement follows:

The tone disabler should hold in the disabled mode for any single-frequency sinusoid in the band from 390-700 Hz having a level of −27 dBm0 or greater, and from 700-3000 Hz having a level of −31 dBm0 or greater. The tone disabler should release for any signal in the band from 200-3400 Hz having a level of −36 dBm0 or less.

5.5 *Operate time*

The operate time must be sufficiently long to provide talk-off protection, but less than the CCITT recommended limit of 400 ms. Thus the requirement is that the tone disabler operate within 300 ± 100 ms after receipt of the sustained disabling signal having a level in the range between a value 3 dB above the midband disabler threshold level and a value of 0 dBm0.

5.6 *False operation due to speech currents*

It is desirable that the tone disabler should rarely operate falsely on speech. To this end, a reasonable objective is that, for an echo suppressor installed on a working circuit, usual speech currents should not on the average cause more than 10 false operations during 100 hours of speech. In addition to the talk-off protection supplied by the disabling channel bandwidth, by guard band operation and by the operate time, talk-off protection can be supplied by recycling. That is, if speech which simulates the disabling signal is interrupted because of inter-syllabic periods, before disabling has taken place the operate timing mechanism should reset. However, momentary absence or change of level in a true disabling signal should not reset the timing.

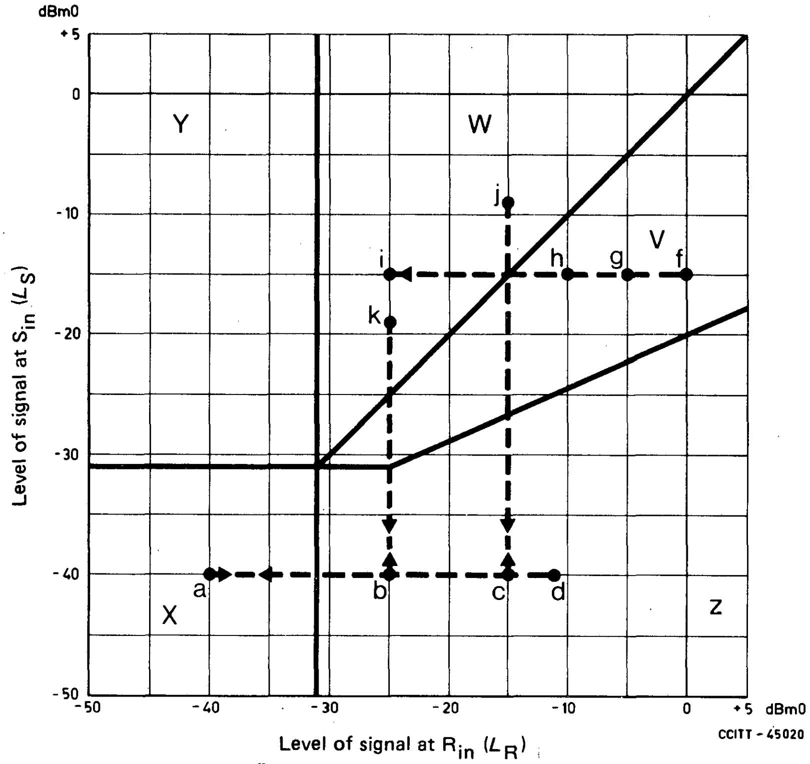
5.7 *Release time*

The disabler should not release for signal drop-outs less than the CCITT recommended value of 100 ms. To cause a minimum of impairment upon accidental speech disabling, it should release within 250 ± 150 ms after a signal in the holding band falls at least 3 dB below the maximum holding sensitivity.

6 **Test arrangements to measure essential operating characteristics of echo suppressors**

6.1 *General considerations*

6.1.1 An echo suppressor with sinusoidal signals applied to its S_{in} and R_{in} ports will assume one of a number of states depending on the relative levels of the two signals. Any given combination of levels of the two input signals may be represented by a point on a typical operational diagram (see, for example, Figure 12/G.164). Each area on this diagram corresponds (under steady conditions) to a particular state identified by the losses in the two speech paths and the internal organization of its logic.



Note – The boundaries shown are typical. The lower boundary of the V region is shown for the maximum loss *C* allowed by Figure 7/G.164.

FIGURE 12/G.164

Operational diagram showing levels used in dynamic tests (see Table 3/G.164 and 4/G.164)

6.1.2 The tests described here assume the use of analogue test signals. In the case of Type C echo suppressors, codecs meeting Recommendation G.712 [3] will be required to interface the suppressor to the analogue test equipment. When tests are performed on Types C and D echo suppressors, due account must be made for the added propagation delays due to the codecs when measuring operate times by the observation of output signals. Further, in level measurements due account must be made for codec tolerances. Frequencies which are submultiples of the sampling frequency may give misleading results and should be avoided in these tests. Note that if external filtering is required to meet the requirements of § 3.2.4.2, it should be included when these tests are performed.

6.1.3 The *static* characteristics of an echo suppressor are specified by stating the inter-area boundaries and the losses in the two speech paths when signals pass slowly from one area to another.

The *dynamic* characteristics of both echo suppressors with fixed and adaptive differential sensitivity are specified by stating the time that elapses when a signal passes suddenly from a point in one area to one in a second area before the state appropriate to the second area is established.

All characteristics unique to echo suppressors with adaptive differential sensitivity are dynamically tested.

The various tests described in § 6 are summarized in Table 7/G.164.

TABLE 7/G.164

Recommended tests for echo suppressors

Test	Characteristic measured	Block diagram (Figure)	Oscilloscope trace (Figure)	Type of echo suppressor: N: non-adaptive, A: adaptive
1	Suppression threshold and loss	13/G.164	–	N, A
2	Y/W threshold and receive loss	13/G.164	–	N, A
3	Break-in differential sensitivity	13/G.164	–	N, A
4	Suppression operate and hangover times	14/G.164	15/G.164	N, A
5	Break-in L_S constant	14/G.164	16/G.164	N, A
6	Partial and full break-in L_R constant	17/G.164	18/G.164	N, A
7	False break-in protection	19/G.164	–	N, A
8	Test for excessive hysteresis	20/G.164	21/G.164	N, A
9	Adaptive break-in differential sensitivity	22/G.164	23/G.164	A
10	a) Rate of decrease of a_x in the W_{Ai} state b) Rate of increase of a_x in the Z_i state c) Rate of clearing of a_x in the X state	22/G.164 25/G.164 27/G.164	24/G.164 26/G.164 28/G.164	A

6.1.4 The descriptions of the test circuits presented here are given so as to indicate a possible method for the application of the appropriate test signals. Other techniques for producing these signals (for example, the use of separate sine wave generators for send and receive) may be employed. Although the test frequency is nominally 1000 Hz, a frequency in the range of 1004-1020 Hz should be chosen to avoid a submultiple of the sampling frequency.

6.2 *Measurement of static characteristics*

The static characteristics measured are losses in the send and receive paths and the inter-area threshold levels (Tables 1/G.164 and 2/G.164). The equipment required is:

– one oscillator with 600-ohm balanced output impedance;
– two 600-ohm balanced attenuators;
– one 600-ohm mixing pad;
– two level-measuring sets with 600-ohm balanced input impedance.

The diagram of connections is shown in Figure 13/G.164.

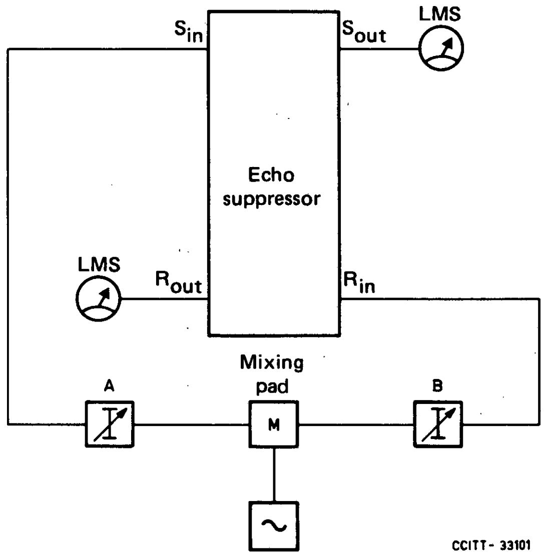


FIGURE 13/G.164

Test circuit for measurement of static characteristics (Tests 1, 2 and 3)

6.2.1 *Test No. 1 – Suppression threshold and loss*

1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4).
2) Adjust A and B so that $L_S = L_R = -40$ dBm0.
3) Note that no loss is inserted in the send and receive paths. Requirement: See Table 1/G.164 (X area).
4) Increase L_R until suppression occurs and note the value of L_R and the suppression loss. Requirement: $-33 \leqslant (L_R = Txz) \leqslant -29$ dBm0 (see Table 2/G.164).
5) Decrease L_R until suppression releases and note the value of L_R. Requirement: $Txz - 3 \leqslant L_R \leqslant Txz$ (see Table 2/G.164).
6) Set the oscillator to appropriate frequencies to check for conformity within the bounds shown in Figure 8/G.164 and repeat steps 2 to 5.

6.2.2 *Test No. 2 – Y/W threshold and receive loss in break-in state*

1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4).
2) Adjust A so that $L_S = +3$ dBm0.
3) Adjust B so that L_R varies over the range $-40 \text{ dBm0} \leqslant L_R \leqslant L_S$. Operation within the boundaries of Figure 7/G.164 is observed by monitoring $L_{Rin} - L_{Rout}$ which equals loss C. Y/W threshold occurs where $C > 0$ dB.

   Note – Record values of C as a function of L_R for use in Test No. 3, step 5.

6.2.3 *Test No. 3 – Break-in differential sensitivity*

1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4).
2) Adjust A so that $L_S' = -40$ dBm0.
3) Adjust B so that $L_R = -26.5$ dBm0.
4) Increase L_S until suppression is removed and loss is inserted in the receive path. Note the value of L_S. Requirement: see Tvw, Table 2/G.164.

5) Decrease L_S until suppression is inserted and loss is removed from the receive path. Note the value of L_S. Requirement: see *Tvz*, Table 2/G.164.

6) Increase L_R in appropriate steps up to +3 dBm0 and repeat steps 4 and 5.

7) Set the oscillator to appropriate frequencies to check for the conformity within the bounds shown in Figure 9/G.164 and repeat steps 2 to 6.

6.3 *Measurement of dynamic characteristics when L_S and L_R are applied independently*

The dynamic characteristics measured are the suppression and break-in operate and hangover times (Tables 3/G.164 and 4/G.164). The equipment required is:

- one oscillator with 600-ohm balanced output impedance, set to 1000 Hz;
- three 600-ohm balanced attenuators;
- three 600-ohm mixing pads;
- two tone-burst generators, the ON and OFF periods of which must be independently variable from zero to at least 200 ms each, and which are capable of being held manually in either state. The input and output impedance in both states must be 600 ohms. One tone-burst generator is driven by the other and has 100 ms delay such that it turns ON 100 ms after the other turns ON;
- two 600-ohm terminating resistors;
- one dual beam oscilloscope, preferably with long persistence screen.

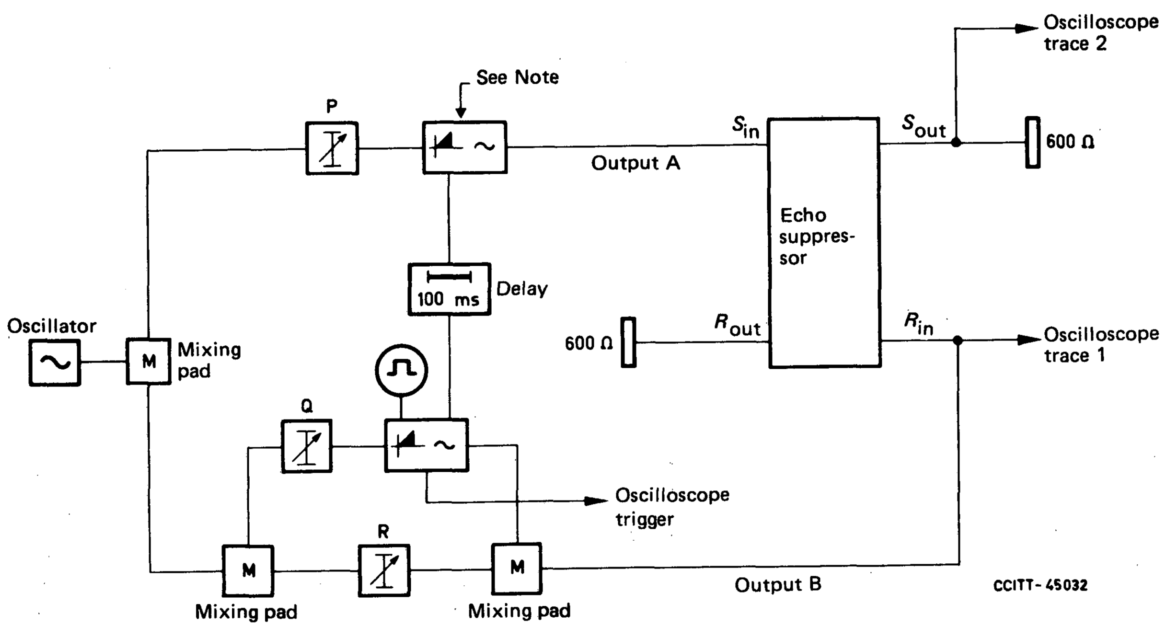
Note – If the ON or OFF periods of the tone pulses are not stated then the value of 200 ms for either should be assumed. Refer to Tables 3/G.164 and 4/G.164 for appropriate performance requirements for Test Nos. 4, 5 and 6.

6.3.1 *Tests in which L_S is maintained constant*

6.3.1.1 *Test No. 4 – Suppression operate and hangover times*

1) Adjust attenuators P, Q and R shown in Figure 14/G.164 to produce the L_R and L_S values of Tables 3/G.164 and 4/G.164.

2) Read times as shown in Figure 15/G.164.



Note – For suppression operate and hangover times, this modulator is maintained in the conducting state.

FIGURE 14/G.164

Test circuit for the measurement of dynamic characteristics with constant L_S [Suppression (Test No. 4) and break-in (Test No. 5)]

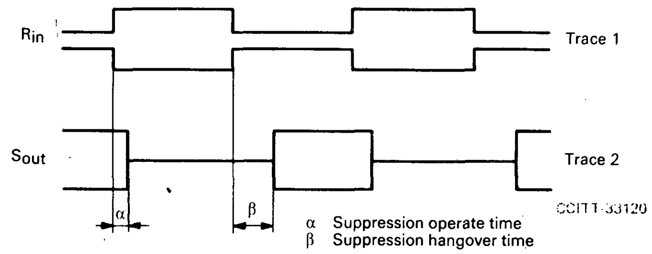


FIGURE 15/G.164

Trace for suppression operate and hangover times

6.3.1.2 *Test No. 5 – Break-in operate time, L_S constant*

In this test, L_R is decreased while a constant L_S is maintained, and a break-in operate time is measured. Since break-in hangover with L_S constant is difficult to measure (due to the difficulty of ensuring a return to the Z state), it is not possible to distinguish between partial and full break-in. This is not considered to be important for break-in with L_S constant.

1) Adjust attenuators P, Q and R shown in Figure 14/G.164 to produce the L_R and the L_S values of Table 3/G.164.
2) Read times as shown in Figure 16/G.164.

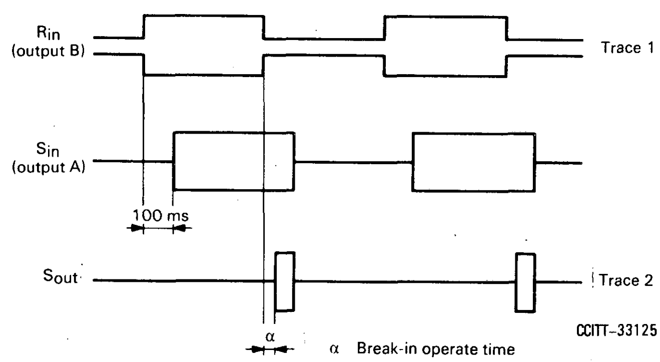


FIGURE 16/G.164

Trace for break-in operate time, L_S constant

6.3.2 *Test in which L_R is maintained constant*

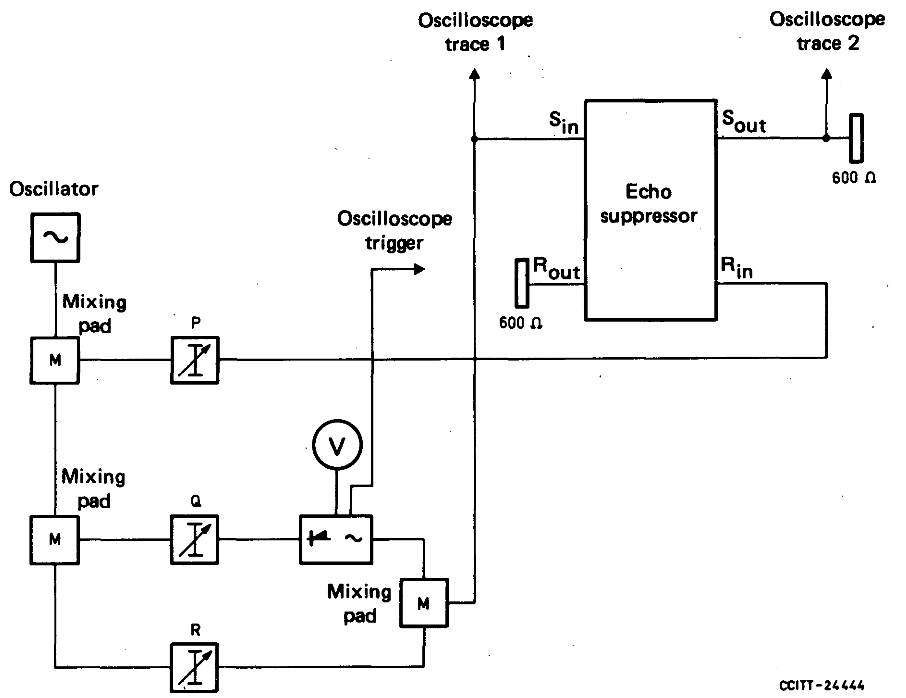
6.3.2.1 *Test No. 6 – Partial and full break-in operate and hangover times, L_R constant*

The equipment required is the same as for Test Nos. 4 and 5, set up according to Figure 17/G.164. In this test L_R is kept constant, L_S is increased, and the partial and full break-in operate and hangover times are measured. To test for partial and full break-in, the duration of time L_S is in the ON state must be varied.

1) Set oscillator to 1000 Hz (for tolerances, see § 6.1.4).
2) Adjust attenuator P of Figure 17/G.164 to produce $L_R = -25$ dBm0.
3) Adjust attenuators Q and R of Figure 17/G.164 to produce $L_S = -40$ dBm0 in the OFF state and $L_S = -19$ dBm0 in the ON state.
4) Starting with a 0 ms duration ON state for L_S, increase the duration of the ON state until partial break-in occurs. Partial break-in is characterized by the short operate and hangover times given in Tables 3 and 4/G.164. Note the oscilloscope traces in a) of Figure 18/G.164 for the definitions of the times.

5) Continue to increase the duration of L_S ON until full break-in, characterized by the extended operate and hangover times of Tables 3/G.164 and 4/G.164 occurs. Note the oscilloscope traces in b) of Figure 18/G.164 for the definitions of the times.

6) Repeat steps 3 to 5 for other pairs of levels given in Tables 3/G.164 and 4/G.164. Note that for all values of $L_R > -26.5$ dBm0 and L_S increasing from below threshold to a value $> L_R$, partial and full break-in should occur.



Note – Variable element V allows the ON and OFF times of the toneburst generator to be separately varied from 0 to 100 ms.

FIGURE 17/G.164

Test circuit for measurement of dynamic characteristics with constant L_R [break-in, Z/V/W, (Test No. 6)]

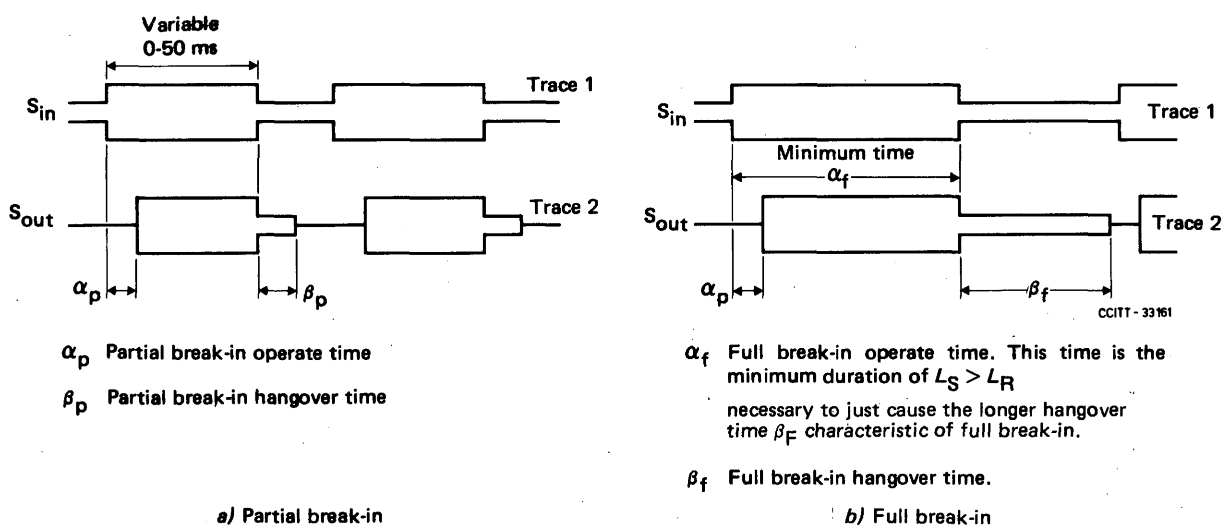


FIGURE 18/G.164

Trace for partial and full break-in operate and hangover times, L_R constant

6.4 *Measurement of echo-suppressor operation when the S_{in} is connected to R_{out} port through an echo path that may include delay as well as loss*

In this test, the echo suppressor is checked for false break-in on returning echo.

6.4.1 *Test No. 7 – False operation of break-in*

The diagram of connections is shown in Figure 19/G.164, and the equipment required is:

– one oscillator with 600-ohm balanced output impedance;
– three 600-ohm balanced attenuators;
– one 600-ohm terminating resistor;
– two 600-ohm mixing pads;
– one tone-burst generator;
– one audio-frequency delay device variable in the range 0-24 ms;
– one dual beam oscilloscope.

1) Set oscillator to 1000 Hz, and delay element to zero delay (for tolerances, see § 6.1.4).
2) Adjust X so that the total loss of echo path (*a-t-b*) is equal to the difference in test levels on the send and receive path, plus 6 dB.
3) Adjust Y so that the OFF signal is -26 dBm0.
4) Adjust Z so that the ON signal is -20 dBm0.
5) While the pulsed signal is applied to R_{in}, check for absence of signal on Trace 2 of the oscilloscope, indicating correct operation.
6) Reduce X until false break-in occurs, and note that the decrease in echo path loss is not less than 2 dB.
7) Repeat steps 4, 5 and 6 with signals at R_{in} of -10 and 0 dBm0 when the pulse generator is ON.
8) Repeat steps 2, 4 and 7 with signals at R_{in} of -40 dBm0 when the pulse generator is OFF.
9) Repeat steps 2 to 8 with the delay set to 24 ms.

Note that false break-in should not occur for *any* pulsed pair of signal levels at R_{in} with the delay set at up to 24 ms, and the echo path loss 6 dB or greater.

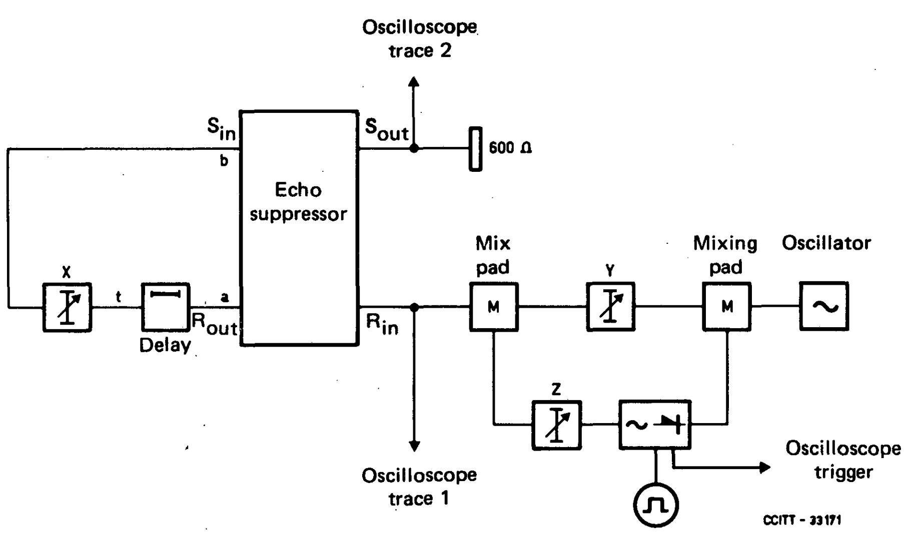


FIGURE 19/G.164

Test circuit for false break-in

6.4.2 *Test No. 8 – False retention of break-in due to provision of excessive hysteresis*

The diagram of connections is shown in Figure 20/G.164 and the equipment required is:

- one oscillator with 600-ohm balanced output impedance;
- three 600-ohm balanced attenuators;
- two 600-ohm mixing pads;
- one 600-ohm terminating resistor;
- one tone-burst generator;
- one amplifier (used as buffer);
- one dual beam oscilloscope.

1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4).
2) Adjust Q so that the path loss between R_{out} and S_{in} is equal to the difference in test levels at these points plus 6 dB.
3) Adjust R so that $L_R = -28$ dBm0.
4) Adjust P so that $L_S = (L_R + 3)$ dBm0.
5) Check that the signal on trace 2 of the oscilloscope is proper (see Figure 21/G.164) denoting non-occurrence of false retention of break-in.
6) Repeat steps 3 to 5 for values of L_R of -16 and 0 dBm0.

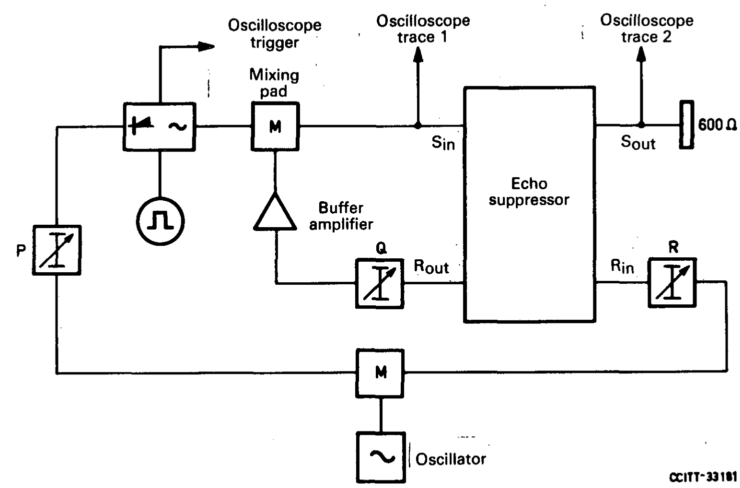


FIGURE 20/G.164

Test circuit for false retention of break-in due to provision to excessive hysteresis.

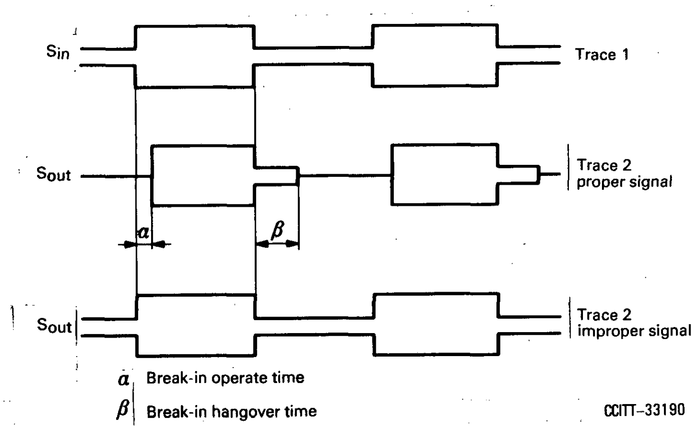


FIGURE 21/G.164

Traces for false retention of break-in due to provision of excessive hysteresis

6.5 *Measurements of the specific dynamic characteristics of adaptive break-in echo suppressors*

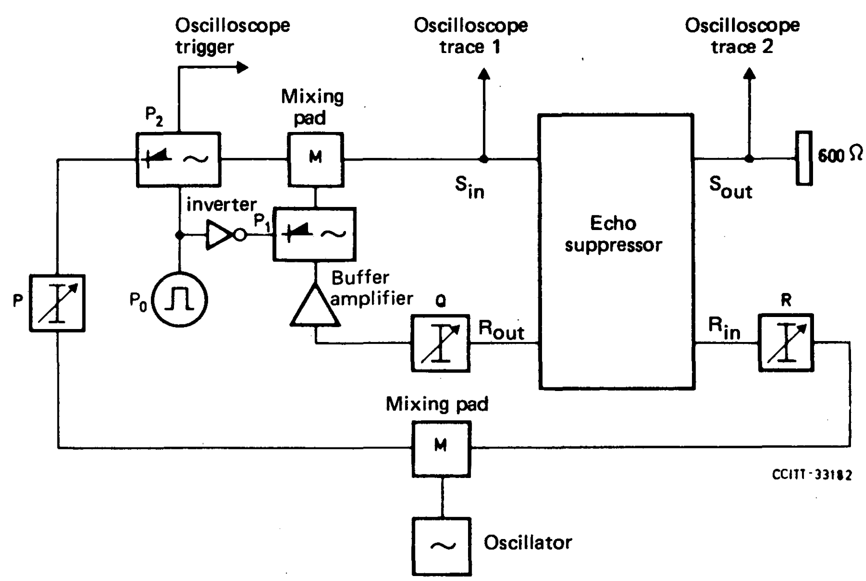
6.5.1 *Test No. 9 – Adaptive break-in differential sensitivity*

The connection diagram is shown in Figure 22/G.164 and the equipment required is:

– one oscillator with 600-ohm balanced impedance;
– three 600-ohm balanced attenuators;
– one 600-ohm terminating resistor;
– two 600-ohm mixing pads;
– two tone-burst generators with period variable up to 10;
– one inverter;
– one amplifier (used for buffer);
– one dual beam oscilloscope.

1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4);
2) Adjust R so tha $L_R = 0$ dBm0;
3) Adjust Q so that the attenuation between R_{out} and S_{in} is equal to the difference in test levels at these points plus 6 dB ($a_E = 6$ dB);
4) With P initially set to at least 55 dB, reduce P to increase L_S until suppression is removed. On trace 2 of the oscilloscope (see Figure 23/G.164) verify that $T\mathrm{v_iw_i}$ satisfies $L_R - a_E + 3 < L_S < L_R - a_E + 6$;
5) Repeat steps 2 to 4 for $L_R = -8$ dBm0;
6) Repeat steps 2 to 4 for $L_R = -15$ dBm0;
7) Repeat steps 2 to 6 for $a_E = 15$ dB;
8) Repeat steps 2 to 5 for $a_E = 24$ dB;
9) Repeat steps 2 to 4 for $a_E = 26$ dB;

Explanation: Test No. 9 checks that the minimum range of a_x is at least 20 dB ($a_{x\,max} > 20$ dB).

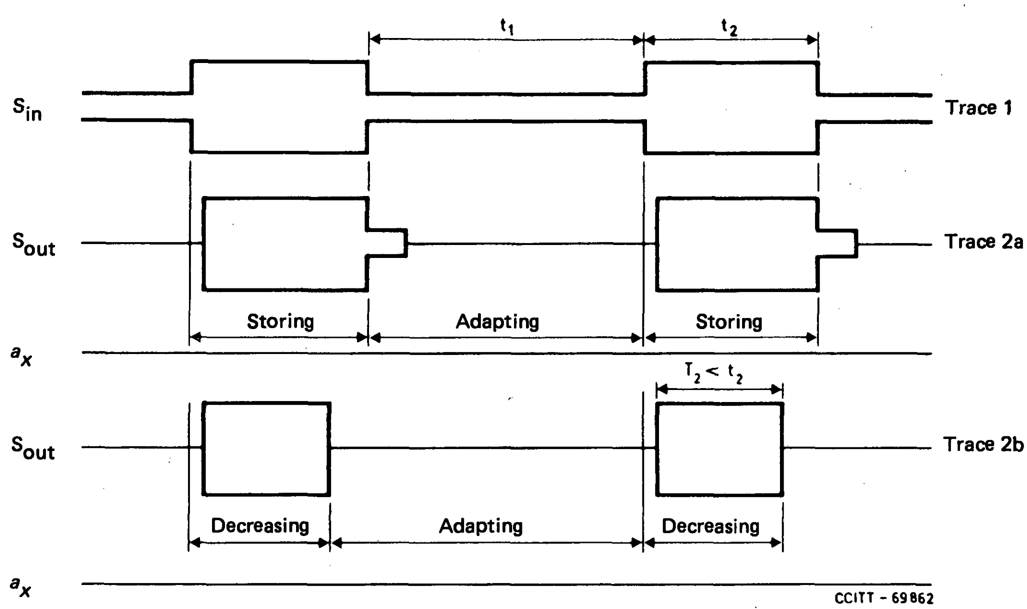


Note 1 – $P_0 = P_2 = \overline{P}_1$ (relationship between waveforms P_0, P_1 and P_2).

Note 2 – This diagram may also be used in Test No. 10a, see § 6.5.2.1.

FIGURE 22/G.164

Test circuit for measurement of adaptive break-in differential sensitivity and rate of decrease of a_x in the W_{Ai} area



Note 1 – Initially set t_1 to five seconds and t_2 to approximately one second.

Note 2 – A smaller value of t_1 may be used depending on the rate of increase and amplitude of variation of a_x in the Z_i state.

Note 3 – Two variants of Trace 2 are possible:

a) For $a_E = 6$ dB, break-in hangover time «β» (see Figure 21/G.164) must be observed after the W/Z transition (Trace 2a).

b) For $a_E > 6$ dB, time «β» must be observed after the W_{Ai}/Z_i transition if a_x is stored in the W_{Ai} state (Trace 2a) (see Table 6/G.164), but not observed if a_x is decreased and the duration t_2 is long enough (Trace 2b).

FIGURE 23/G.164

Traces for measurement of adaptive break-in differential sensitivity

6.5.2 *Test No. 10 – Measurement of rates of change for a_x*

6.5.2.1 *Measurement of the rate of decrease of a_x in the W_{Ai} state, Test No. 10 a*

The connection diagram (Figure 22/G.164) and the required equipment are the same as in Test No. 9.

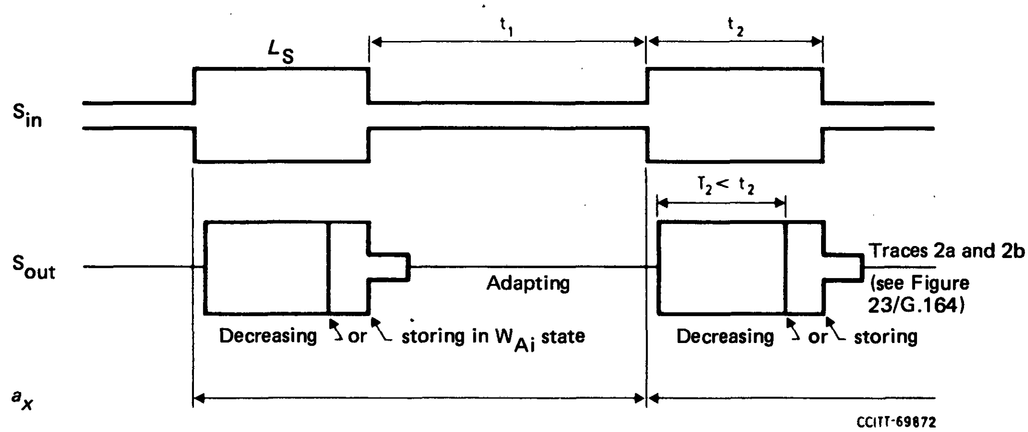
1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4.)
2) Adjust R so that $L_R = 0$ dBm0
3) Adjust Q so that $a_E = 20$ dB [after convergence in state Z_i, a_x must be equal to 14 dB nominally (a_{xC})]
4) Using P, increase L_S until suppression is removed and loss C inserted in the receive path (see Figure 7/G.164). Check that Tv_iw_i satisfies $-17 \leqslant L_S \leqslant -14$ (dBm0)
5) Repeat step 4 to obtain traces of Figure 24/G.164. When break-in elapses (end of T_2) before end of t_2, the echo suppressor makes a_x decrease in the W_{Ai} area. Then measure T_2.
6) The end of break-in occurs when a_x has decreased to threshold level a_{xE} where

$$L_R - C - L_S - 3 \leqslant a_{xE} \leqslant L_R - C - L_S.$$

Check that the theoretical decreasing speed of a_x in the W_{Ai} state is approximately given by:

$$V = \frac{a_{xC} - a_{xE}}{T_2} \quad \text{dB/s}$$

where $a_{xC} = 14$ dB.



Note 1 – t_1 could be as long as approximately 3.5 s in duration depending on the rate of increase of a_x in the Z_i state.
Note 2 – The rate of decrease of a_x in the W_{Ai} state may be very slow. It may be necessary for t_2 to be increased in order to be able to observe the wave form of Trace 2b.

FIGURE 24/G.164

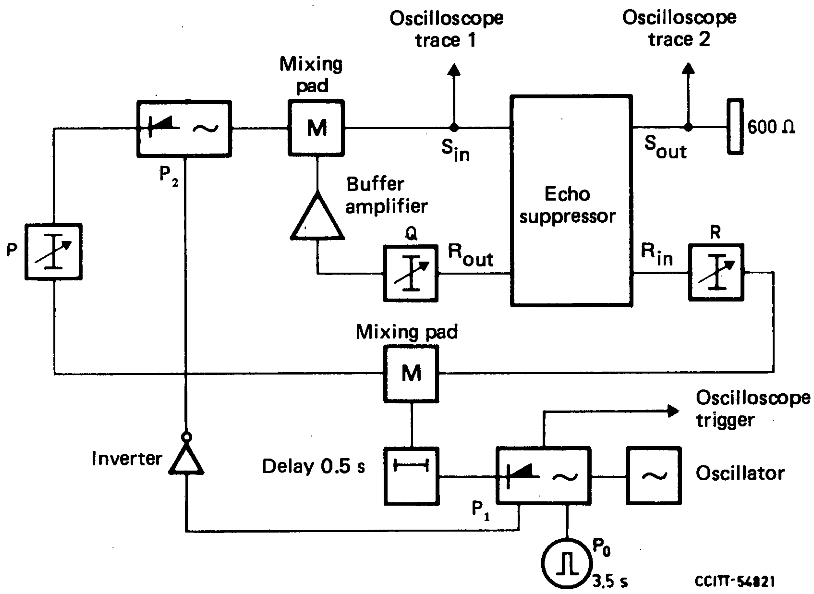
Traces for measurement of the rate of decrease of a_x in the W_{Ai} state

6.5.2.2 *Measurement of the rate of increase of a_x in the Z_i state (see Figure 25/G.164 and Figure 26/G.164), Test No. 10 b*

The connection diagram is shown Figure 25/G.164 and the equipment required is:

– one oscillator with 600-ohm balanced impedance;
– three 600-ohm balanced attenuators;
– two 600-ohm mixing pads;
– one 600-ohm terminating resistor;
– two tone-burst generators with period variable up to 10 s;
– one inverter;
– one amplifier (used as a buffer);
– one audio-frequency delay device with 500 ms delay;
– one dual beam oscilloscope.

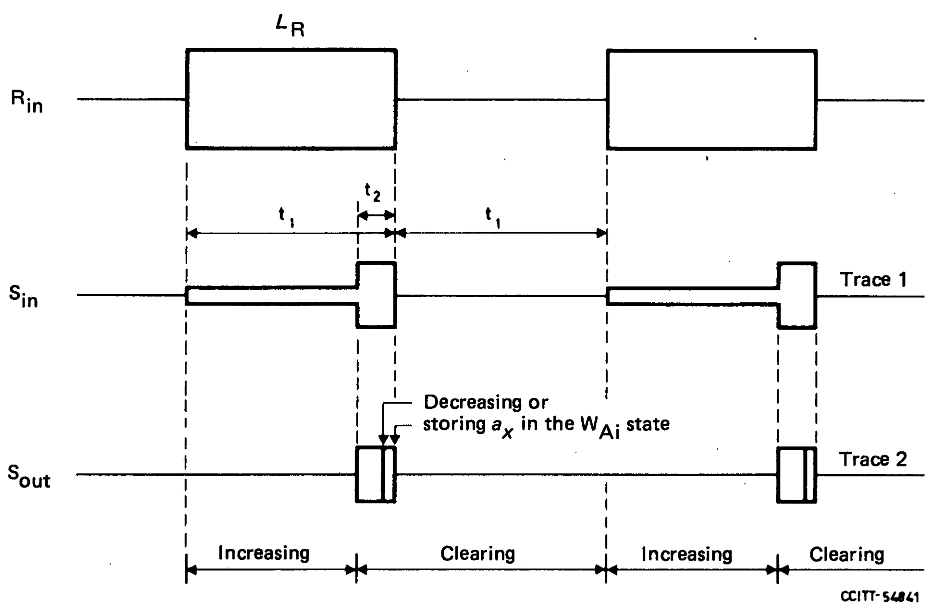
1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4);
2) Adjust R so that L_R = 0 dBm0;
3) Adjust Q so that a_E = 20 dB;
4) Adjust P so that L_S = −12 dBm0;
5) Adjust the tone "ON" and "OFF" periods of P_0 to 3.5 s;
6) Check that t_1 and t_2 are respectively equal to 3.5 and 0.5 s;
7) Check that break-in occurs on trace 2 of the oscilloscope (see Figure 26/G.164).



Note – $P_0 = P_1 = \overline{P}_2$ (relationship between waveforms P_0, P_1 and P_2)

FIGURE 25/G.164

Test circuit for measurement of the rate of increase of a_x in the Z_{Ai} state



Note – Either trace is possible depending on the algorithm used in the W_{Ai} state.

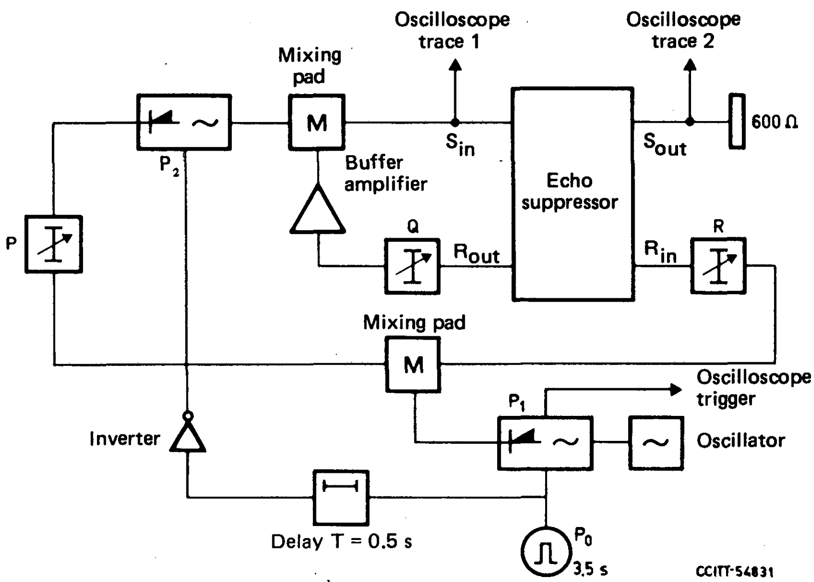
FIGURE 26/G.164

Traces for measurement of the rate of increase of a_x in the Z_i state

6.5.2.3 *Measurement of the rate of clearing of a_x in the X state (see Figures 27/G.164 and Figure 28/G.165), Test No. 10 c*

The diagram of connection is shown in Figure 27/G.164 and the equipment required is the same as for Test No. 10 b.

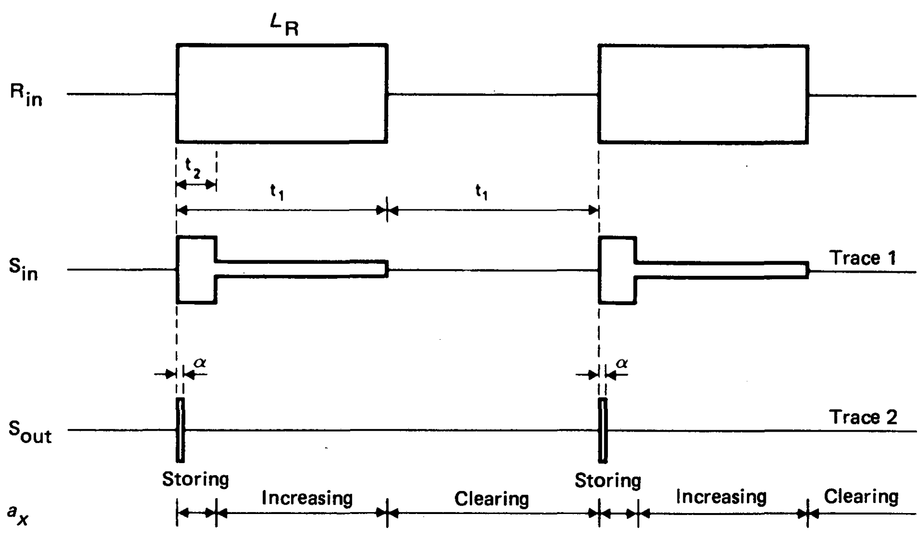
1) Set the oscillator to 1000 Hz (for tolerances, see § 6.1.4);
2) Adjust Q so that $a_E = 20$ dB;
3) Adjust R so that $L_R = 0$ dBm0;
4) Adjust P so that $L_S = -12$ dBm0;
5) Adjust the tone "ON" and "OFF" periods of P_0 to 3.5 s;
6) Check that t_1 and t_2 are respectively equal to 3.5 and 0.5 s;
7) Check that break-in does not occur on trace 2 of the oscilloscope.



Note – $P_0 = P_1$; $P_2 = \overline{P_1 \times \delta(T)}$ (relationship between waveforms P_0, P_1 and P_2)

FIGURE 27/G.164

Test circuit for measurement of the rate of P_2 clearing of a_x in the X state



Note – A possible presence of L_S out during $\alpha < 2$ ms does not affect the result of the test.

FIGURE 28/G.164

Traces of measurement of the rate of clearing of a_x in the X state

References

[1] CCITT Recommendation *Echo suppressors suitable for circuits having either short or long propagation times*, Orange Book, Vol. III, Rec. G.161, ITU, Geneva, 1977.

[2] CCITT Recommendation *Influence of national networks on stability and echo in international connections*, Orange Book, Vol. III, Rec. G.122, Part B, b), ITU, Geneva, 1977.

[3] CCITT Recommendation *Performance characteristics of PCM channels between 4-wire interfaces at voice frequencies*, Vol. III, Rec. G.712.

[4] CCITT Recommendation *Characteristics of primary PCM multiplex equipment operating at 1544 kbit/s*, Vol. III, Rec. G.733.

[5] CCITT Recommendation *300-baud modem standardized for use in the general switched telephone network*, Vol. VIII, Rec. V.21.

Recommendation G.165

ECHO CANCELLERS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984 and at Melbourne, 1988)

1 General

1.1 Echo cancellers are voice operated devices placed in the 4-wire portion of a circuit (which may be an individual circuit path or a path carrying a multiplexed signal) and are used for reducing the echo by subtracting an estimated echo from the circuit echo. They may be characterized by whether the transmission path or the subtraction of the echo is by analogue or digital means (see Figures 1/G.165, 2/G.165 and 3/G.165).

1.2 This Recommendation is applicable to the design of echo cancellers using digital or analogue techniques, and intended for use in an international circuit. Echo cancellers designed to this Recommendation will be compatible with each other and with echo suppressors designed in accordance with Recommendations G.161 [1] and G.164. Compatibility is defined in Recommendation G.164, § 1.4. Freedom is permitted in design details not covered by the requirements.

Echo cancellers may be used for purposes other than network echo control on international circuits, e.g. in active 2-wire/4-wire hybrids or 2-wire repeaters, but this Recommendation does not apply to such echo cancellors.

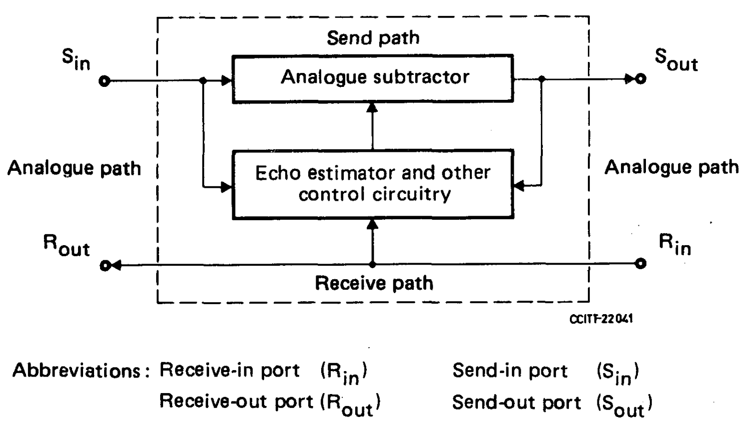
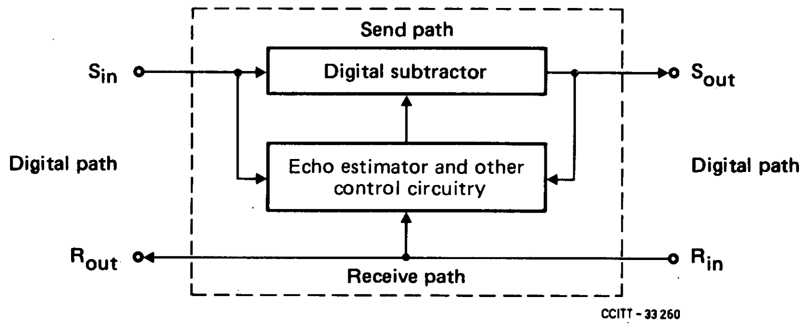


FIGURE 1/G.165

Type A echo canceller



Note – Functionally, a type C digital echo canceller (DEC) interfaces at 64 kbit/s. However, 24 or 30 digital echo cancellers for example may be combined corresponding to the primary digital hierarchy levels of 1544 kbit/s or 2048 kbit/s respectively.

FIGURE 2/G.165

Type C echo canceller

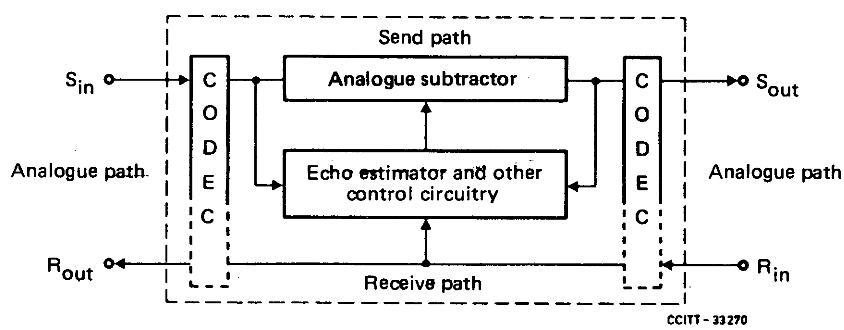


FIGURE 3/G.165

Type D echo canceller

2 Definitions relating to echo cancellers[1)]

In the definition and text, L will refer to the relative power level of a signal, expressed in dBm0 and A will refer to the attenuation or loss of a signal path expressed in dB.

2.1 **echo canceller** (see Figure 4/G.165)

F: annuleur d'écho

S: compensador de eco; cancelador de eco

A voice operated device placed in the 4-wire portion of a circuit and used for reducing near-end echo present on the send path by subtracting an estimation of that echo from the near-end echo.

1) These definitions assume that nonlinear processing, e.g. centre clipping, is not present in the send or receive paths unless otherwise specified and that the signal at S_{in} is purely echo.

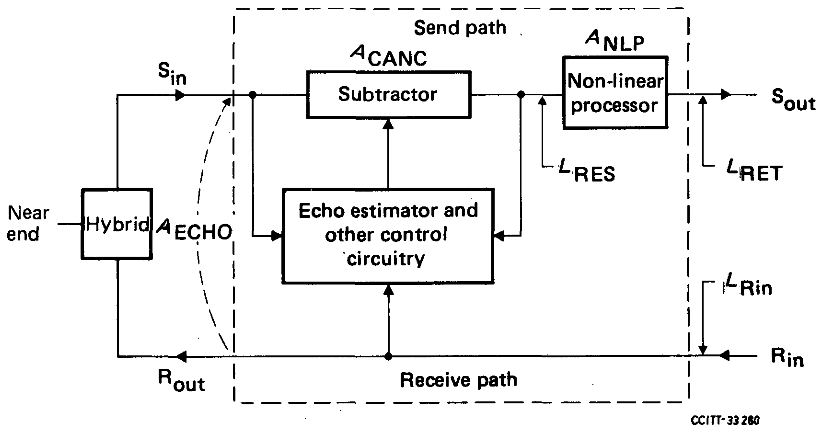


FIGURE 4/G.165

Echo canceller

2.2 **echo loss (A_{ECHO})**

F: affaiblissement d'écho (A_{ECHO})

S: atenuación del eco (A_{ECO})

The attenuation of a signal from the receive-out port (R_{out}) to the send-in port (S_{in}) of an echo canceller, due to transmission and hybrid loss, i.e. the loss in the echo path.

Note – This definition does not strictly adhere to the echo loss definition given in Recommendation G.122, § 2.2 which applies to loss of the *a-t-b* path viewed from the virtual switching point of the international circuit. The echo canceller may be located closer to the echo reflection point.

2.3 **cancellation (A_{CANC})**

F: annulation (A_{NL})

S: compensación; cancelación (A_{COMP})

The attenuation of the echo signal as it passes through the send path of an echo canceller. This definition specifically excludes any nonlinear processing on the output of the canceller to provide for further attenuation.

2.4 **residual echo level (L_{RES})**

F: niveau d'écho résiduel (N_{RES})

S: nivel de eco residual (N_{RES})

The level of the echo signal which remains at the send-out port of an operating echo canceller after imperfect cancellation of the circuit echo. It is related to the receive-in signal L_{Rin} by

$$L_{RES} = L_{Rin} - A_{ECHO} - A_{CANC}$$

Any nonlinear processing is not included.

2.5 **nonlinear processor (NLP)**

F: processeur non linéaire (PNL)

S: procesador no lineal (PNL)

A device having a defined suppression threshold level and in which:

a) signals having a level detected as being below the threshold are suppressed, and

b) signals having a level detected as being above the threshold are passed although the signal may be distorted.

Note 1 – The precise operation of a nonlinear processor depends upon the detection and control algorithm used.

Note 2 – An example of a nonlinear processor is an analogue centre clipper in which all signal levels below a defined threshold are forced to some minimum value.

2.6 **nonlinear processing loss (A_{NLP})**

F: affaiblissement par traitement non linéaire (A_{TNL})

S: atenuación por procesamiento (o tratamiento) no lineal (A_{PNL})

Additional attenuation of residual echo level by a nonlinear processor placed in the send path of an echo canceller.

Note – Strictly, the attenuation of a nonlinear process cannot be characterized by a loss in dB. However, for purposes of illustration and discussion of echo canceller operation, the careful use of A_{NLP} is helpful.

2.7 **returned echo level (L_{RET})**

F: niveau de retour d'écho (N_{RET})

S: nivel del eco devuelto (N_{DEV})

The level of the signal at the send-out port of an operating echo canceller which will be returned to the talker. The attenuation of a nonlinear processor is included, if one is normally present. L_{RET} is related to L_{Rin} by

$$L_{RET} = L_{Rin} - (A_{ECHO} + A_{CANC} + A_{NLP}).$$

If nonlinear processing is not present, note that $L_{RES} = L_{RET}$.

2.8 **combined loss (A_{COM})**

F: affaiblissement combiné (A_{COM})

S: atenuación combinada (A_{COMB})

The sum of echo loss, cancellation loss and nonlinear processing loss (if present). This loss relates L_{Rin} to L_{RET} by:

$$L_{RET} = L_{Rin} - A_{COM}, \text{ where } A_{COM} = A_{ECHO} + A_{CANC} + A_{NLP}.$$

2.9 **convergence**

F: convergence

S: convergencia

The process of developing a model of the echo path which will be used in the echo estimator to produce the estimate of the circuit echo.

2.10 **convergence time**

F: temps de convergence

S: tiempo de convergencia

For a defined echo path, the interval between the instant a defined test signal is applied to the receive-in port of an echo canceller with the estimated echo path impulse response initially set to zero, and the instant the returned echo level at the send-out port reaches a defined level.

2.11 **leak time**

F: temps de fuite

S: tiempo de fuga

The interval between the instant a test signal is removed from the receive-in port of a fully-converged echo canceller and the instant the echo path model in the echo canceller changes such that, when a test signal is reapplied to R_{in} with the convergence circuitry inhibited, the returned echo is at a defined level.

This definition refers to echo cancellers employing, for example, leaky integrators in the convergence circuitry.

3 Characteristics of echo cancellers

3.1 *General*

This Recommendation is applicable to the design of echo cancellers. The echo cancellers are assumed to be "half" echo cancellers, i.e. those in which cancellation takes place only in the send path due to signals present in the receive path.

3.2 *Purpose, operation and environment*

Echo, in any 2-wire or combination 2- and 4-wire telephone circuit, is caused by impedance mismatches. An echo canceller can be used to reduce this echo to tolerable levels.

The echo present at the send-in port of an echo canceller is a distorted and delayed replica of the incoming speech from the far end, i.e. the echo is the incoming speech as modified by the echo path. The echo path is commonly described by its impulse response (see Figure 5/G.165). This response of a typical echo path shows a pure delay t_r, due to the delays inherent in the echo path transmission facilities, and a dispersed signal due to band limiting and multiple reflections. The sum of these is the echo path delay, t_d. The values of delay and dispersion will vary depending on the properties of the echo paths, e.g. they may vary for different national networks. It is assumed that the echo paths are basically linear and not continuously varying[2)], e.g. have no phase roll (see Recommendation G.164). In addition, the loss of the echo path in dB (see § 2.2 above) is likely to be such that the minimum loss from R_{out} to S_{in} of the echo canceller will be equal to the difference between relative levels at these two ports plus 6 dB. Echo cancellers designed to this Recommendation will perform properly for echo loss (A_{ECHO}) of 6 dB or greater. For (A_{ECHO}) less than 6 dB they may also work but with degraded performance. It is not possible to quantify this degraded performance.

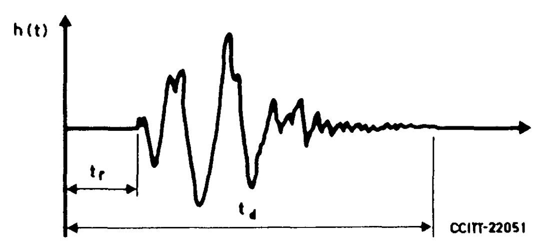


FIGURE 5/G.165

Example of an impulse response of an echo path

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2) Echo cancellers designed specifically for echo paths which are nonlinear and/or time variant are likely to be much more complex than those not so designed. It is felt that insufficient information exists to include such echo cancellers in this Recommendation. Echo cancellers conforming to this Recommendation are adaptive and will cope with slowly varying echo paths when only receive speech is present.

An echo canceller must be able to synthesize a replica of the echo path impulse response. Many echo cancellers model the echo path using a sampled data representation, the sampling being at the Nyquist rate (8000 Hz). Such an echo canceller, to function properly, must have sufficient storage capacity for the required number of samples[3]. Typically, too few storage locations will prevent adequate synthesis of all echo paths: too many storage locations will create undesirable additional noise due to the unused locations which, because of estimation noise, are generally not zero. It should be recognized that an echo canceller introduces an additional parallel echo path. If the impulse response of the echo path model is sufficiently different from the echo path impulse response, the total returned echo may be larger than that dùe to the echo path only.

The echo paths change as the echo canceller is used in successive connections. When speech first arrives at R_{in}, the echo canceller must adapt or converge to the new echo path, and it is desirable that this be fairly rapid, e.g. about one-half second. Also the residual echo should be small regardless of the level of the receive speech and the characteristics of the echo path. Some Administrations feel that a slightly higher residual echo level may be permitted provided it is further reduced using a small amount of nonlinear processing (see § 5).

When there is receive speech and the near party begins to double talk, an echo canceller may interpret the transmit signal as a new echo signal and attempt to adapt to it. This can seriously degrade the subjective quality of the connection. Not only is the echo cancellation reduced but distortion of the double talking speech may occur as the echo canceller dynamically attempts to adapt. Two common approaches are taken as a solution. The first is to use algorithm which causes slow adaptation during periods of double talk. The second is to employ a double talk detector, similar to that used in echo suppressors. The echo canceller double talk detector, however, generally should favour break-in at the expense of false operation on echo. This differs from the double talk detector in an echo suppressor.

Thus, echo cancellers have the following fundamental requirements:

1) rapid convergence;

2) subjective low returned echo level during single talk;

3) low divergence during double talk.

When echo cancellers are located on the subscriber side of the international signalling equipment, signalling tones do not pass through the cancellers so no special action is necessary. When cancellers are on the international side of the signalling equipment they are normally disabled by the switch during the active signalling exchange intervals in order to prevent distortion of the signalling tones by the echo canceller. When signalling tones simultaneously appear at the canceller receive and send ports (double talk) the receive signal will be processed through the echo path model contained in the canceller. The signal estimate produced by the canceller may sufficiently distort the send side signal so that it will not be properly recognized by the signalling receive unit (Note 1). An echo canceller must de disabled during the transmission of the CCITT No. 6 and No. 7 continuity check signal (Note 2). If an echo canceller conforming to Recommendation G.165 is located on the international side of CCITT No. 5 signalling units an enabled canceller, it will interfere with the continuously compelled signalling exchange CCITT No. 5 unless additional special precautions are taken. See Recommendation Q.115 for details.

Note 1 – For some echo cancellers this problem may not occur when the send and receive frequencies are different.

Note 2 – CCITT Recommendation Q.271 on CCITT No. 6 and Recommendation Q.724 on CCITT No. 7 both include the following statement: “As the presence of active echo suppressors in the circuit would interfere with the continuity check, it is necessary to disable the suppressors during the check and to re-enable them, if required, after the check has been completed.”

[3] Echo cancellers having storage capacities of 16 ms to 40 ms have been successfully demonstrated. Maximum echo path delay t_d, in the network in which the canceller will be used will determine the required storage capacity.

3.3 *External enabling/disabling*

An option should be included in the echo canceller to provide for enabling or disabling by an externally derived ground (earth) from the trunk circuit. The enabler should function to permit or prevent normal echo canceller operation. Certain type C echo cancellers may be disabled directly by a digital signal. Some digital data signals may require Type C echo cancellers to provide 64 kbit/s bit sequence integrity in the externally disabled state.

3.4 *Tests and requirements for performance with inputs signals applied to the send and receive paths*

3.4.1 *Transmission performance*

The appropriate transmission performance requirements of Recommendation G.164 also apply to echo cancellers except as noted below.

3.4.1.1 *Delay distortion – Type A*

The delay distortion relative to the minimum delay shall not exceed the values given in Table 1/G.165.

TABLE 1/G.165

Frequency band (Hz)	Delay distortion (μs)
500- 600	300
600-1000	150
1000-2600	50
2600-3000	250

3.4.1.2 *Attenuation distortion – Type A*

The attenuation distortion shall be such that if Q dB is the attenuation at 800 Hz (or 1000 Hz) the attenuation shall be within the range (Q + 0.5) dB to (Q − 0.2) dB at any frequency in the band 300-3400 Hz and at 200 Hz, within the range of (Q + 1.0) dB to (Q − 0.2) dB.

3.4.1.3 *Group delay – Type C*

The group delay in the send path should be kept to a minimum and should not exceed 1 ms. No significant delay should occur in the receive path.

Note – The creation of frame slips in the echo path can lead to an occasional degradation of the echo cancellation. If a delay is necessary to synchronize the digital send and receive paths, the global admissible delay on the send path, including the group delay mentioned above, must not exceed 1 ms and on the receive path 250 μs.

3.4.1.4 *Group delay – Type D*

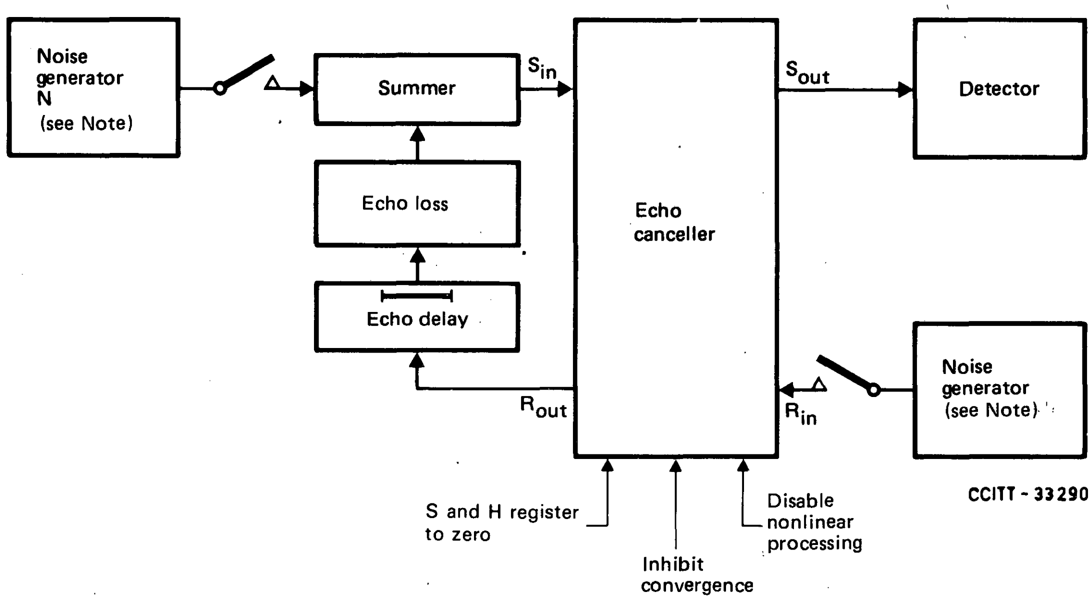
The group delay in the send and receive paths shall meet the requirements of § 3.4.1.3 for Type C echo cancellers with the addition of the delay allowed for codecs as given in Recommendation G.712.

3.4.2 *Echo canceller performance*

The performance requirements which follow are for echo cancellers which include nonlinear processors (see Annex A for echo cancellers which do not include a nonlinear processor).

In the tests, it is assumed that the nonlinear processor can be disabled, that the echo path impulse response store (H register) can be cleared (set to zero) and that adaptation can be inhibited.

The requirements are described in terms of tests made by applying signals to R_{in} and S_{in} of an echo canceller, and measuring the S_{out} signals. The test set-up is as shown in Figure 6/G.165. The ports are assumed to be at equal relative level points. Band-limited noise is used as the receive input test signal. The echo loss is independent of frequency.



Note – The requirements in § 3.4.2 are based on the use of band-limited white noise (300-3400 Hz) as the test signal. Noise shaped in accordance with Recommendation G.227 may also be used. However, the applicability of the requirements in § 3.4.2 requires confirmation and is under study.

The use of alternative test signals more representative of real speech and possible changes in test procedures and requirements are also under study.

FIGURE 6/G.165

Test for echo canceller performance

The primary purpose of an echo canceller is to control the echo of a speech stimulus signal. This is done by synthesizing a replica of the echo path impulse response and using it to generate an estimate of the echo which is subtracted from the actual circuit echo. The synthesis must be accomplished using a speech input signal. Because of the difficulty of defining a speech test signal, the following tests are type tests and rely upon the use of a band-limited noise test signal primarily for measurement convenience and repeatability. These tests should be performed on an echo canceller only after the design has been shown to properly synthesize a replica of the echo path impulse response from a speech input signal and its corresponding echo. Speech signals are not used in the tests in this section. Additionally, the nonlinear processor in the echo canceller should be designed to minimize and potentially avoid the perceptible effects of double-talk clipping and noise contrast [see Recommendation G.164, Table 1, Note a)]. Tests to ensure proper operation are under study.

3.4.2.1 *Test No. 1 – Steady state residual and returned echo level test*

This test is meant to ensure that the steady state cancellation (A_{CANC}) is sufficient to produce a residual echo level which is sufficiently low to permit the use of nonlinear processing without undue reliance on it.

The H register is initially cleared and a receive signal is applied for a sufficient time for the canceller to converge producing a steady state residual echo level.

Requirement (provisional)

With the H register initially set to zero, the nonlinear processor disabled for all values of receive input signal level such that $L_{Rin} \geqslant -30$ dBm0 and $\leqslant -10$ dBm0 and for all values of echo loss $\geqslant 6$ dB and echo path delay, $t_d \leqslant \Delta$ ms[4], the residual echo level should be less than or equal to that shown in Figure 7/G.165. When the nonlinear processor is enabled, the returned echo level must be less than -65 dBm0.

Note – Recommendation G.113 allows for up to 5 PCM codecs in the echo path. Meeting the requirement of Figure 7/G.165 under those conditions has not been verified. This is under study.

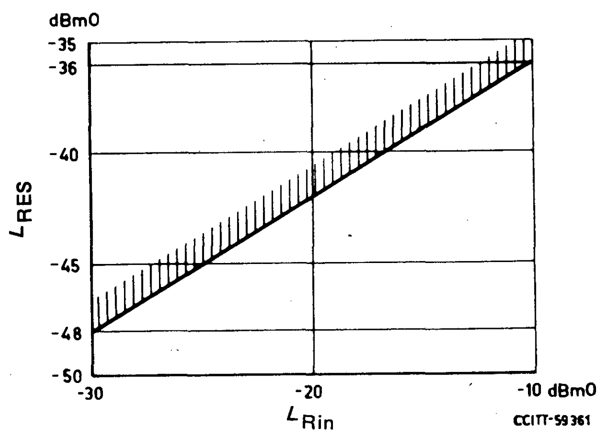


FIGURE 7/G.165

3.4.2.2 *Test No. 2 – Convergence test*

This test is meant to ensure that the echo canceller converges rapidly for all combinations of input signal levels and echo paths and that the returned echo level is sufficiently low. The H register is initially cleared and adaption is inhibited. The double talk detector, if present, is put in the double talk mode by applying signals to S_{in} and R_{in}. The signal at S_{in} is removed and simultaneously adaption is enabled. The degree of adaption, as measured by the returned echo level, will depend on the convergence characteristics of the echo canceller and the double talk detection hangover time.

[4] Different echo cancellers may be designed to work satisfactorily for different echo path delays depending on their application in various networks. Thus Δ, whenever it appears in this Recommendation, represents the echo path delay, t_d, for which the echo canceller is designed.

The test procedure is to clear the H register and inhibit adaption. Signal N is applied at a level -10 dBm0 and a signal is applied at R_{in}. Then N is removed and simultaneously adaption is enabled (see Figure 8/G.165). After 500 ms inhibit adaption and measure the returned echo level. The nonlinear processor should be enabled.

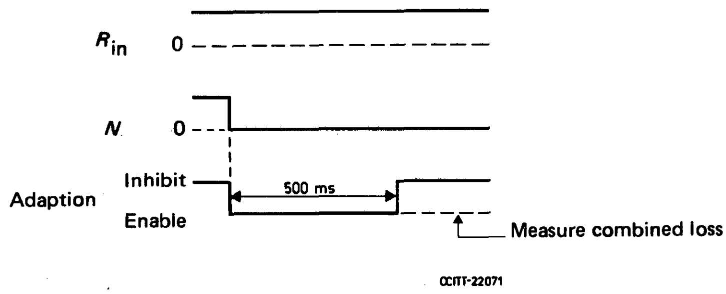


FIGURE 8/G.165

Requirement

With the H register initially set to zero, for all values $L_{\text{Rin}} \geqslant -30$ dBm0 and $\leqslant -10$ dBm0 and present for 500 ms and for all values of echo loss $\geqslant 6$ dB and echo path delay, $t_d \leqslant \Delta$ ms, the combined loss ($A_{\text{COM}} = A_{\text{ECHO}} + A_{\text{CANC}} + A_{\text{NLP}}$) should be $\geqslant 27$ dB.

3.4.2.3 *Test No. 3 – Performance under conditions of double talk*

The two parts of this test are meant to test the performance of the canceller under various conditions of double talk. The tests make the assumption that, upon detection of double talk, measures are taken to prevent or slow adaption in order to avoid excessive reduction in cancellation.

3.4.2.3.1 Test No. 3 a is meant to ensure that the double talk detection is not so sensitive that echo and low level near-end speech falsely cause operation of the double talk detector to the extent that adaption does not occur. The test procedure is to clear the H register; then for some value of echo delay and echo loss, a signal is applied to R_{in}. Simultaneously (see Figure 9/G.165) an interfering signal which is sufficiently low in level to not seriously hamper the ability of the echo canceller to converge, is applied at S_{in}. This signal should not cause the double talk detector to be activated, and adaption and cancellation should occur. After 1 s the adaption is inhibited and the residual echo measured. The nonlinear process should be *disabled.*

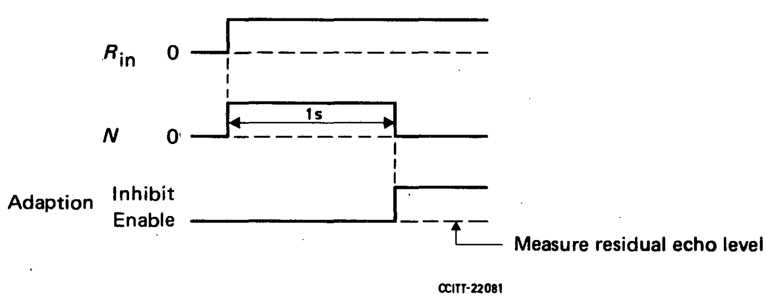


FIGURE 9/G.165

Requirement

With the H register initially set to zero for all values of $L_{Rin} \geqslant -25$ dBm0 and $\leqslant -10$ dBm0, $N = L_{Rin} - 15$ dB, $A_{ECHO} \geqslant 6$ dB and echo path delay, $t_d \leqslant \Delta$ ms, convergence should occur within 1.0 s and L_{RES} should be $\leqslant N$.

3.4.2.3.2 Test No. 3 b is meant to ensure that the double talk detector is sufficiently sensitive and operates fast enough to prevent large divergence during double talking.

The test procedure is to fully converge the echo canceller for a given echo path. A signal is then applied to R_{in}. Simultaneously (see Figure 10/G.165) a signal N is applied to S_{in} which has a level at least that of R_{in}. This should cause the double talk detector to operate. After any arbitrary time, $\delta t > 0$, the adaption is inhibited and the residual echo measured. The nonlinear processor should be disabled.

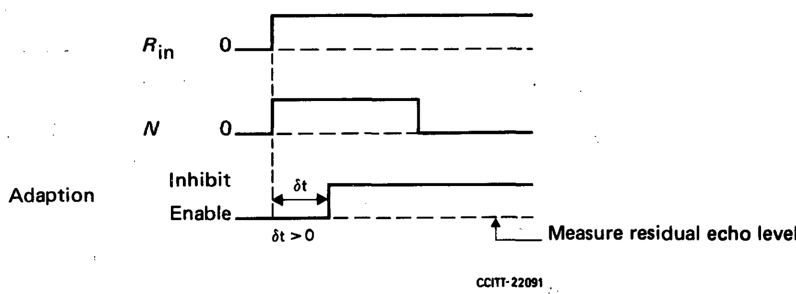


FIGURE 10/G.165

Requirement

With the echo canceller initially in the fully converged state for all values of $L_{Rin} \geqslant -30$ dBm0 and $\leqslant -10$ dBm0, and for all values of $N \geqslant L_{Rin}$ and for all values of echo loss $\geqslant 6$ dB and echo path delay $t_d \leqslant \Delta$ ms, the residual echo level after the simultaneous application of L_{Rin} and N for any time period should not increase more than 10 dB over the steady state requirements of Test No. 1.

3.4.2.4 *Test No. 4 – Leak rate test*

This test is meant to ensure that the leak time is not too fast, i.e. that the contents of the H register do not go to zero too rapidly.

The test procedure is to fully converge the echo canceller for a given echo path and then to remove all signals from the echo canceller. After two minutes the contents of the H register are frozen, a signal applied to R_{in} and the residual echo measured (see Figure 11/G.165). The nonlinear process is used in normal operation, it should be *disabled.*

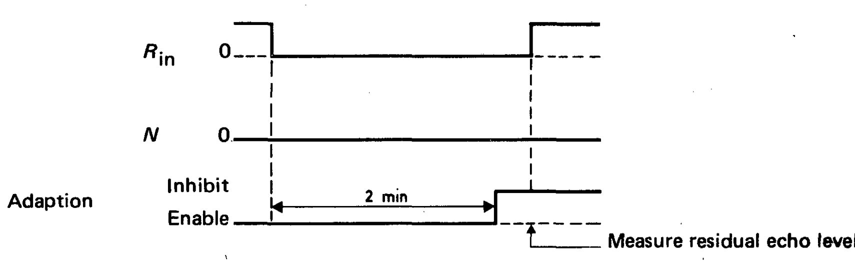


FIGURE 11/G.165

Requirement

With the echo canceller initially in the fully converged state for all values of $L_{Rin} \geqslant -30$ dBm0 and $\leqslant -10$ dBm0, two minutes after the removal of the R_{in} signal, the residual echo level should not increase more than 10 dB over the steady state requirement of Test No. 1.

3.4.2.5 *Test No. 5 – Infinite return loss convergence test*

This test is meant to ensure that the echo canceller has some means to prevent the unwanted generation of echo. This may occur when the H register contains an echo path model, either from a previous connection or the current connection, and the echo path is opened (circuit echo vanishes) while a signal is present at R_{in}.

The test procedure is to fully converge the echo canceller for a given echo path. The echo path is then interrupted while a signal is applied to R_{in}. 500 ms after interrupting the echo path the returned echo signal at S_{out} should be measured (see Figure 12/G.165). The nonlinear processor should be *disabled.*

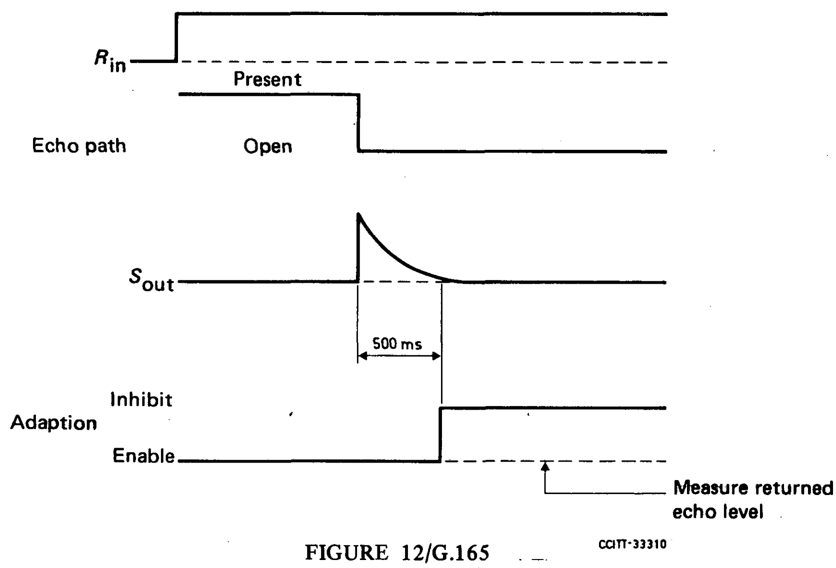


FIGURE 12/G.165

Requirement (provisional)

With the echo canceller initially in the fully converged state for all values of echo loss $\geqslant$ 6 dB, and for all values of $L_{Rin} \geqslant -30$ dBm0 and $\leqslant -10$ dBm0, the returned echo level at S_{out}, 500 ms after the echo path is interrupted, should be $\leqslant -37$ dBm0.

3.4.2.6 *Test No. 6 – Stability test*

Under study.

4 Characteristics of an echo canceller tone disabler

4.1 *General*

To ensure proper operation of all currently specified V-series modems, the echo cancellers covered by this Recommendation should be equipped with a tone detector that conforms to this section. This tone detector responds to a disabling signal which is different from that used to disable the echo suppressor as described in Recommendation G.164, § 5 and consists of a 2100 Hz tone with periodic phase reversals inserted in that tone. The tone disabler should respond only to the specified in-band signal. It should not respond to other in-band signals, e.g. speech, or a 2100 Hz tone without a phase reversal. The tone disabler should detect and respond to a disabling signal which may be present in either the send or the receive path.

The requirements for echo canceller disabling to ensure proper operation with ATME No. 2 equipment that transmits the 2100 Hz tone with phase reversals could be met by using either the tone disabler specified in this section, or the echo suppressor tone disabler specified in Recommendation G.164, § 5. However, use of the Recommendation G.164, § 5 disabler does not assure proper operation with all currently specified V-series modems.

The term disabled in this section refers to a condition in which the echo canceller is configured in such a way as to no longer modify the signals which pass through it in either direction. Under this condition, no echo estimate is subtracted from the send path, the non-linear processor is made transparent, and the delay through the echo canceller still meets the conditions specified in § 3.4.1. However, no relationship between the circuit conditions before and after disabling should be assumed. For one thing, the operation of echo cancellers with tonal inputs (such as the disabling tone) is unspecified. Additionally, the impulse response stored in the echo canceller prior to convergence (and prior to the disabling tone being sent) is arbitrary. This can lead to apparent additional echo paths which, in some echo canceller implementations, remain unchanged until the disabling tone is recognized. Also note that echo suppressors could be on the same circuit and there is no specified relationship between their delay in the enabled and disabled states. In spite of the above, it is possible, for example, to measure the round-trip delay of a circuit with the disabling tone but the trailing edge of the tone burst should be used and sufficient time for all devices to be disabled should be allotted before terminating the disabling tone and starting the timing.

It should be noted that this condition does not necessarily fulfil the requirements for 64 kbit/s bit sequence integrity, for which case other means of disabling in line with Recommendation G.165, § 3.4 will apply.

A reference tone disabler is described in Annex B.

4.2 *Disabler characteristics*

The echo canceller tone disabler requires the detection of a 2100 Hz tone with phase reversals of that tone. The characteristics of the transmitted signal are defined in Recommendation V.25. Phase variations in the range of 180° ± 25° must be detected while those in the range of 0° ± 110° must not be detected.

The frequency characteristics of the tone detector are the same as the characteristics of the echo suppressor tone detector given in Recommendation G.164, § 5.2.

The dynamic range of this detector should be consistent with the input levels as specified in Recommendation V.2 and H.51 with allowances for variation introduced by the public switched telephone network.

4.3 *Guardband characteristics*

Similar to that defined in Recommendation G.164, § 5.3, consistent with the dynamic range given in § 4.2 above with the following exception. The detector should operate perfectly with white noise less than or equal to 11 dB below the level of the 2100 Hz signal. No definitive guidelines can be given for the range between 5 and 11 dB because of the variations in the test equipment used. In particular, performance may vary with the peak-to-average ratio of the noise generator used. As a general guideline, however, the percentage of correct operation (detection of phase variations of 180° ± 25° and non-detection of phase variations of 0° ± 110°) should fall by no more than 1% for each dB reduction in signal-to-noise below 11 dB. The Administration of the Federal Republic of Germany mentions the possibility of designing a detector capable of operating perfectly at 5 dB signal-to-noise ratio.

4.4 *Holding-band characteristics*

Same as defined in Recommendation G.164, § 5.4.

4.5 *Operate time*

The operate time must be sufficiently long to provide immunity from false operation due to voice signals, but not so long as to needlessly extend the time to disable. The tone disabler is required to operate within one second of the receipt of the disabling signal.

4.6 *False operation due to speech currents*

Same as in Recommendation G.164, § 5.6.

4.7 *False operation due to data signals*

It is desirable that the tone disabler should rarely operate falsely on data signals from data sets that would be adversely affected by disabling of the echo canceller. To this end, a reasonable objective is that, for an echo canceller installed on a working circuit, usual data signals from such data sets should not, on the average, cause more than 10 false operations during 100 hours of data transmissions.

4.8 *Release time*

Same as in Recommendation G.164, § 5.7.

4.9 *Other considerations*

Both the echo of the disabling tone and the echo of the calling tone may disturb the detection of the echo canceller disabling tone. As such, it is not recommended to add the receive and transmit signal inputs together to form an input to a single detector.

Careful attention should be given to the number of phase reversals required for detection of the disabling tone. Some Administrations favour relying on 1 to improve the probability of detection even in the presence of slips, impulse noise, and low signal-to-noise ratio. Other Administrations favour relying on 2 to improve the probability of correctly distinguishing between non-phase-reversed and phase-reversed 2100 Hz tones.

5 **Nonlinear processors for use in echo cancellers**

5.1 *Scope*

For the purpose of this Recommendation the term "nonlinear processor" is intended to mean only those devices which fall within the definition given in § 2.5 and which have been proven to be effective in echo cancellers. It is possible to implement such nonlinear processors in a number of ways (centre clippers being just one example), with fixed or adaptive operating features, but no recommendation is made for any particular implementation. General principles and guidelines are given in § 5.2. More detailed and concrete information requires reference to specific implementations. This is done in Annex C for the particular case of a "reference nonlinear processor". The use of this term denotes an implementation given for guidance and illustration only. It does not exclude other implementations nor does it imply that the reference nonlinear processor is necessarily the most appropriate realization on any technical, operational or economic grounds.

5.2 *General principles and guidelines*

5.2.1 *Function*

5.2.1.1 *General*

The nonlinear processor is located in the send path between the output of the subtractor and the send-out port of the echo canceller. Conceptually, it is a device which blocks low level signals and passes high level signals. Its function is to further reduce the residual echo level (L_{RES} as defined in § 2.4) which remains after imperfect cancellation of the circuit echo so that the necessary low returned echo level (L_{RET} as defined in § 2.7) can be achieved.

5.2.1.2 *Network performance*

Imperfect cancellation can occur because echo cancellers which conform to this Recommendation may not be capable of adequately modelling echo paths which generate significant levels of nonlinear distortion (see § 3.2). Such distortion can occur, for example, in networks conforming to Recommendation G.113 in which up to five pairs of PCM codecs (conforming to Recommendation G.712) are permitted in an echo path. The accumulated quantization distortion from these codecs may prevent an echo canceller from achieving the necessary L_{RET} by using linear cancellation techniques alone. It is therefore recommended that all echo cancellers capable only of modelling the linear components of echo paths but intended for general network use should incorporate suitable nonlinear processors.

5.2.1.3 *Limitations*

This use of nonlinear processors represents a compromise in the circuit transparency which would be possible by an echo canceller which could achieve the necessary L_{RET} by using only modelling and cancellation techniques. Ideally, the non-linear processor should not cause distortion of near-end speech. In practical devices it may not be possible to sufficiently approach this ideal in this case it is recommended that nonlinear processors should not be active under double talk or near-end single-talk conditions. From this it follows that excessive dependence must not be placed on the nonlinear processor and that L_{RES} must be low enough to prevent objectionable echo under double-talk conditions.

5.2.1.4 *Data transmission*

Nonlinear processors may affect the transmission of data through an enabled echo canceller. This is under study.

5.2.2 *Suppression threshold*

5.2.2.1 *General*

The suppression threshold level (T_{SUP}) of a nonlinear processor is expressed in dBm0 and is equal to the highest level of a sine-wave signal at a given moment that is just suppressed. Either fixed or adaptive suppression threshold levels may be used.

5.2.2.2 *Fixed suppression threshold*

With a fixed suppression threshold level the appropriate level to use will depend upon the cancellation achieved and the statistics of speech levels and line conditions found in the particular network in which the echo canceller is to be used. It is therefore recommended that the actual level should be field selectable to permit the user to adjust it for the actual network environment. Values of fixed suppression threshold levels to be used are under study – see Notes 1 and 2.

Note 1 – As an interim guide, it is suggested that the suppression threshold level should be set a few decibels above the level that would result in the *peaks* of L_{RES} for a "2σ-talker" and a "2σ-echo return loss" being suppressed.

Note 2 – Results of a field trial reported by one Administration indicated that a fixed suppression threshold level of -36 dBm0 gave a satisfactory performance. A theoretical study, by another Administration, of an echo path contianing five pairs of PCM codecs showed that for an L_R of -10 dBm0, the quantization noise could result in an L_{RES} of -38 dBm0.

5.2.2.3 *Adaptive suppression threshold*

A good compromise can be made between using a high T_{SUP} to prevent it being exceeded by loud talker residual echo and using a low T_{SUP} to reduce speech distortion on break-in by making T_{SUP} adaptive to the actual circuit conditions and speech levels. This may be achieved in a number of ways and no recommendation is made for any particular implementation. General guidelines applicable to the control algorithm and suppression threshold levels are under study.

5.2.3 *Control of nonlinear processor activation*

5.2.3.1 *General*

To conform to the recommendation made in § 5.2.1.3, it is necessary to control the activation of the nonlinear processor so that it is not active when near-end speech is likely to be present. When "active", the nonlinear processor should function as intended to reduce L_{RES}. When "inactive", it should not perform any nonlinear processing on any signal passing through the echo canceller.

5.2.3.2 *Control guidelines*

It is recommended that the following two guidelines should govern control of the activation of a nonlinear processor. First, because they are intended to further reduce L_{RES}, they should be active when L_{RES} is at a significant level. Second, because they should not distort near-end speech, they should be inactive when near-end speech is present. Where these two guidelines conflict the control function should favour the second.

5.2.3.3 *Static characteristics*

A conceptual diagram showing the two operational states of a nonlinear processor is shown in Figure 13/G.165. The L_S L_R plane is divided into two regions, W and Z by the threshold WZ. In the W region the nonlinear processor is inactive while in the Z region it is active. Proper control of the nonlinear processor to ensure operation in the appropriate region requires recognition of the double-talk condition or the presence of near-end speech. Imperfect detection of double-talk combined with a high suppression threshold level will result in distortion of near-end speech. The echo canceller then exhibits some of the characteristics of an echo suppressor. A low suppression level will permit easy double-talking, even if a detection error is made because the near-end speech will suffer only a low level of non-linear distortion. If the suppression threshold level is too low then peaks of residual echo may be heard.

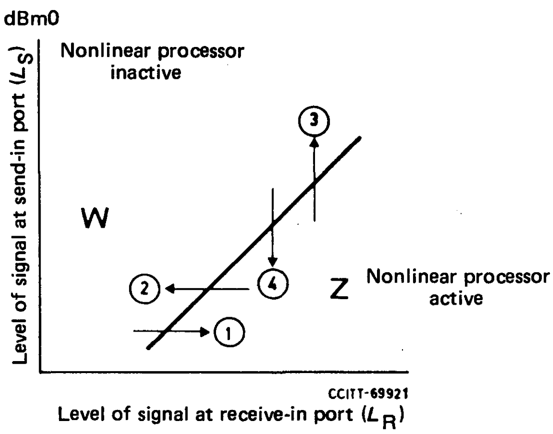


FIGURE 13/G.165

Nonlinear processor operating regions

5.2.3.4 *Dynamic characteristics*

The dynamic characteristics can be specified by stating the time that elapses when the signal conditions pass from a point in one area to a point in the other area before the state appropriate to the second area is established. Four such transitions are shown by arrows in Figure 13/G.165.

Transition No. 1 – W to Z, L_S constant, L_R increasing

In this case the L_S signal occurred first and the L_R is increasing to a sufficiently high level to override the L_S signal in the control path and cause the nonlinear processor to change from the inactive to the active state. Since this will cause distortion of the L_S signal (near talker speech in this case) the action should not be initiated too quickly.

Transition No. 2 – Z to W, L_S constant, L_R decreasing

In this case the L_R signal has overriden the L_S signal in the control path and the nonlinear processor is in the active state. The L_R signal is now decreasing. The nonlinear processor should remain in the active state sufficiently long to prevent echo, which is stored in the echo path, from being hear by the far talker.

Transition No. 3 – Z to W, L_R constant, L_S increasing

This transition is replicating the onset of double talk. As soon as possible after the L_S signal is detected the nonlinear processor should be switched to the inactive state in order to minimise any distortion of the near talker speech.

Transition No. 4 – W to Z, L_R constant, L_S decreasing

In this case L_S has been recognised but is decreasing. Any action which is taken should favour continuing to permit the L_S signal to pass. This implies there should be some delay in switching the nonlinear processor back to the active state.

5.2.4 *Frequency limits of control paths*

Under study.

Note – Depending on the particular implementation of the nonlinear processor, the considerations and frequency response limits given in Recommendation G.164, § 3.2.4.2 for the suppression and break-in control paths of echo suppressors may also be applicable to similar control paths used in nonlinear processors. These control paths may include the activation control and adaptive suppression threshold level control.

5.2.5 *Signal attenuation below threshold level*

The attenuation of signals having a level below that of the suppression threshold level of a nonlinear processor in the active state must be such that the requirements of § 3.4.2.1 are met.

5.2.6 *Testing of nonlinear processors*

The nonlinear processor may be considered as a special case of an echo suppressor which is limited to suppressing only low level signals. The types of test required to determine the nonlinear processor performance characteristics are very similar to the echo suppressor tests given in Recommendation G.164. However, depending on the specific implementation of a nonlinear processor, the transitions between areas W and Z of Figure 13/G.165 may not be as sharply defined as is the case for echo suppressors. Signals observed at the send-out port of the echo canceller may be distorted for short periods when transitions between the W and Z operating regions occur. Although Recommendation G.164 may be used as a guide to the testing of nonlinear processors it may be necessary to introduce unique test circuit modifications in order to make measurements on some specific nonlinear processor implementations. No recommendation can be given for a universal test circuit appropriate for all nonlinear processor implementations.

ANNEX A

(to Recommendation G.165)

Echo cancellers without nonlinear processing

It may be possible to implement echo cancellers without the inclusion of nonlinear processing. For these echo cancellers the total echo loss is provided by echo cancellation. The achievable echo cancellation is limited by the characteristics of the echo path and by the method of implementing the echo canceller. In particular, if one pair of codecs conforming to Recommendation G.712 is used in the echo path or in the echo canceller, the maximum echo cancellation (considering quantizing errors in the echo canceller and other impairments) is that shown by the solid line in Figure A-1/G.165.

Echo cancellers conforming to the solid line in Figure A-1/G.165 have been tested and found to provide acceptable performance in Japan. Other tests, however, suggest that the echo cancellation required in echo cancellers for general application is at least that shown by the broken line in Figure A-1/G.165. Further study is needed. Pending the results of that study, echo cancellers which do not include nonlinear processors are not yet recommended for general application.

All the provisions and tests in the body of Recommendation G.165 apply to these echo cancellers except as follows:

a) § 3.4.2.1: the residual echo level requirement is that shown by the solid line of Figure A-1/G.165.

b) For all other tests, any reference to non-linear processing should be ignored.

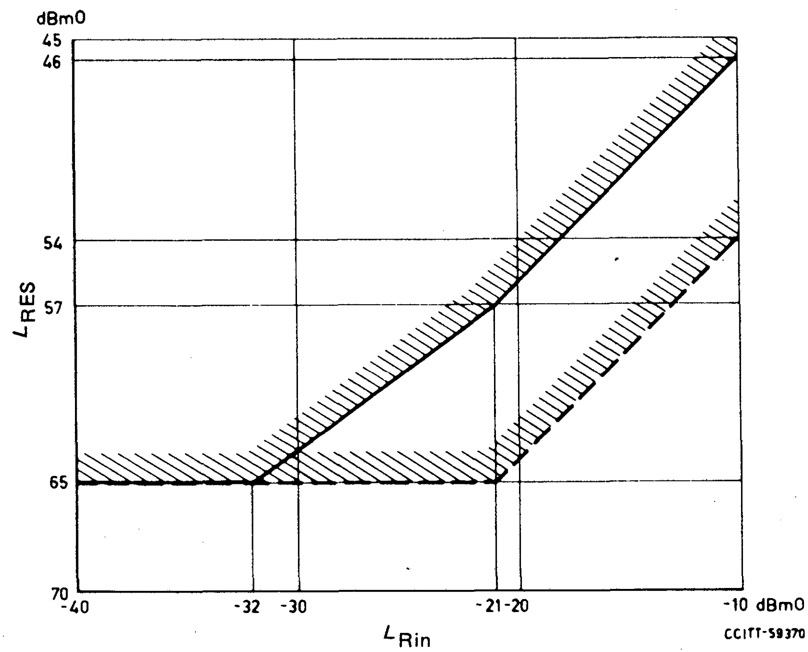


FIGURE A-1/G.165

ANNEX B

(to Recommendation G.165)

Description of an echo canceller reference tone disabler

B.1 *General*

This annex describes the characteristics of an echo canceller reference tone disabler. The use of the term *reference* denotes a disabling implementation given for guidance only. It does not exclude alternative implementations of a tone disabler which responds to the signal as defined in Recommendation V.25, and which also meets all of the criteria for reliability of operation and protection from false operation by speech signals.

B.2 *Disabler characteristics*

The echo canceller reference tone disabler described in this annex detects a 2100 Hz tone with periodic phase reversals which occur every 450 ± 25 ms. The characteristics of the transmitted signal are defined in Recommendation V.25.

B.2.1 *Tone detection*

The frequency characteristics of the tone detector used in this reference tone disabler are the same as the characteristics of the echo suppressor tone detector given in Recommendation G.164, § 5.2, except that the upper limit of the dynamic range is −6 dBm0.

B.2.2 *Phase reversal detection*

The reference tone disabler responds to a signal which contains phase reversals of 108° ± 10° at its source (as specified in Recommendation V.25) when this signal has been modified by allowable degradations caused by the network, e.g. noise, phase jitter, etc. This disabler is insensitive to phase jitter of ± 15° peak-to-peak in the frequency range of 0-120 Hz. This accommodates to the phase jitter permitted by Recommendations H.12 and G.229. In order to minimize the probability of false disabling of the echo canceller due to speech currents and network-induced phase changes, this reference tone disabler does not respond to single phase changes of the 2100 Hz tone in the range 0° ± 110° occurring in a one second period. This number has been chosen since it represents the approximate phase shift caused by a single frame slips in a PCM system.

B.3 *Guardband characteristics*

Meet requirements in Recommendation G.164, § 5.3.

Note – The possibility of interference during the phase reversal detection period has been taken into account. One potential source of interference is the presence of calling tone as specified in Recommendation V.25. If the calling tone interferes with the detection of the phase reversal, the entire disabling detection sequence is restarted, but only one time. Recommendation V.25 ensures at least one second of quiet time between calling tone burst.

B.4 *Holding-band characteristics*

Meet requirements in Recommendation G.164, § 5.4.

B.5 *Operate time*

The reference tone disabler operates within one second of the receipt, without interference, of the sustained 2100 Hz tone with periodic phase reversals, having the level in the range −6 to −31 dBm0. The one second operate time permits the detection of the 2100 Hz tone and ensures that two phase reversals will occur (unless a slip or impulse noise masks one of the phase reversals).

B.6 *False operation due to speech currents*

Meets requirements in Recommendation G.164, § 5.6.

B.7 *False operation due to data signals*

Meets the requirement in Recommendation G.165, § 4.7. To this end, the tone disabler circuitry becomes inoperative if one second of clear (i.e. no phase reversals or other interference) 2100 Hz tone is detected. The detected circuit remains inoperative during the data transmission and only becomes operative again 250 ± 150 ms after a signal in the holding band falls at least 3 dB below the maximum holding sensitivity. Thus the possibility of inadvertent disabling of the echo canceller during data transmission is minimized.

B.8 *Release time*

Meets the requirements in Recommendation G.164, § 5.7.

ANNEX C

(to Recommendation G.165)

Description of a reference nonlinear processor

C.1 *General*

This annex, which is for the purposes of illustration only and not intended as a detailed design (see § 5.1), describes a reference nonlinear processor based upon concepts that are as simple as possible but having included in it a sufficient number of features to give guidance for a wide range of possible implementations. To this end two variants of the reference nonlinear processor are included. Both are based on a centre clipper having either of the idealized transfer functions illustrated in Figure C-1/G.165. The suppression threshold level (determined, in this case by the clipping level) in the first variant is adaptive, adaptation being by reference to L_R. Activation control is by reference to the difference between L_R and L_S. In the second variant the suppression threshold is fixed. It is assumed that the reference nonlinear processor is used in an echo canceller which can achieve a cancellation of the linear components of any returned echo of at least N dB. The value of N is under study.

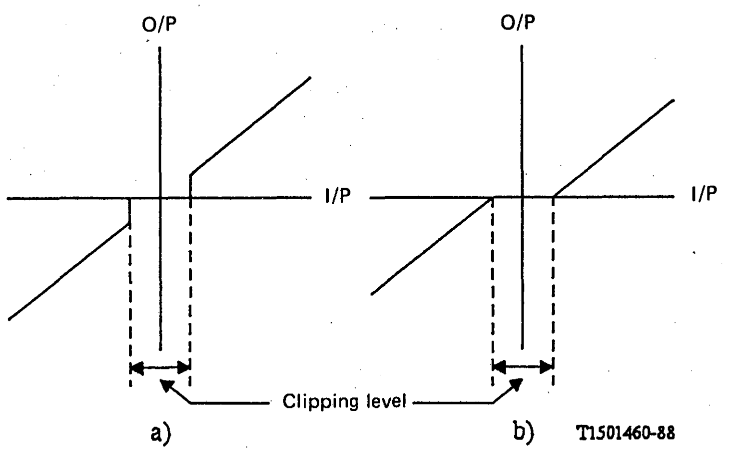


FIGURE C-1/G.165

Two examples of idealized centre clipper transfer functions

C.2 *Suppression threshold* (T_{SUP})

Adaptive $T_{SUP} = (L_R - x \pm 3)$ dBm0 for $-30 \leq L_R \leq -10$ dBm0

Fixed $T_{SUP} = x'$ dBm0

Note – Values of x and x' are under study. Values of 18 for x and -36 for x' have been suggested by confimation is required that these values are appropriate for use in all networks.

C.3 *Static characteristics of activation control*

$T_{WZ} = (L_R - y \pm 3)$ dBm0 for $-30 \leq L_R \leq -10$ dBm0

Note 1 – T_{WZ} is as defined in § 5.2.3.3.

Note 2 – The value of y may be different for each variant, and this is under study. Values of x dB in the case of the adaptive T_{SUP} and ≥ 6 dB for y in the case of the fixed T_{SUP} seem reasonable.

C.4 *Dynamic characteristics of activation control*

Dynamic characteristics of the activation control are given in Table C-1/G.165 and C-2/G.165. Also see Figure 13/G.165.

C.5 *Frequency limits of control paths*

See Recommendation G.165, § 5.2.4.

C.6 *Testing*

Tables C-1/G.165 and C-2/G.165 indicate, by reference to Recommendation G.164 how the dynamic performance of nonlinear processor activation control may be checked using sine wave signals. Figures C-2/G.165 and C-3/G.165 show the traces obtained on an oscilloscope for these tests.

TABLE C-1/G.165

Nonlinear processor hangover times

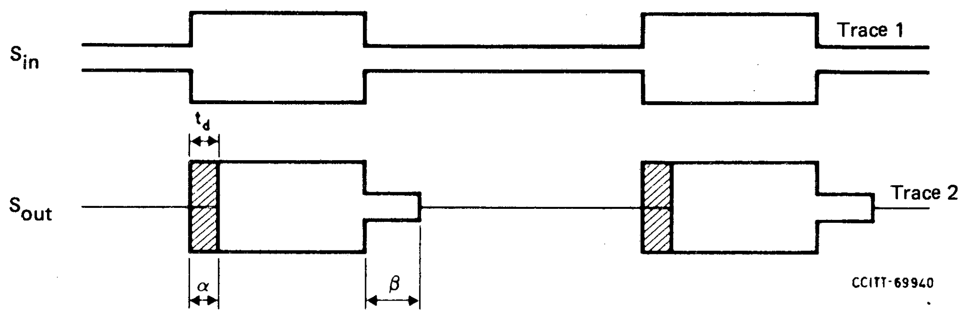
Boundary		Initial signal		Final signal		Recommended value (ms)	Test No. (Rec. G.164)	Excursion (see Figure 13/G.165)	Test circuit, Figure:	Oscilloscope trace
		Send L_S (dBm0)	Receive L_R (dBm0)	Send L_S (dBm0)	Receive L_R (dBm0)					
Z/W	Fixed	−25	−10	−25	−30	15-64	5	Transition 2	14/G.164	Trace 1 and trace 2 of Figure C-3/G.165 (β)
	Adaptive	−55 −40 −30	−20 −15 − 5	−55 −40 −30	−40 −40 −30	Δ[a]				
W/Z	Fixed	−15	−25	−40	−25	16-120	6	Transition 4	17/G.164	Trace 1 and trace 2 of Figure C-2/G.165 (β)
	Adaptive	−40 −40 −25	−50 −30 −15	−55 −55 −40	−50 −30 −15	30-50				

[a] Δ is defined in § 3.4.2.1 [footnote [4]].

TABLE C-2/G.165

Nonlinear processor operate times

Boundary		Initial signal		Final signal		Recommended value (ms)	Test No. (Rec. G.164)	Excursion (see Figure 13/G.165)	Test circuit, Figure:	Oscilloscope trace
		Send L_S (dBm0)	Receive L_R (dBm0)	Send L_S (dBm0)	Receive L_R (dBm0)					
W/Z	Fixed	−25	−30	−25	−10	16-120	4	Transition 1	14/G.164	Trace 2 of Figure C-3/G.165 (α)
	Adaptive	−55 −40 −30	−40 −40 −30	−55 −40 −30	−20 −15 − 5	15-75				
Z/W	Fixed	−40	−25	−15	−25	⩽ 1	6	Transition 3	17/G.164	Trace 2 of Figure C-2/G.165 (α)
	Adaptive	−55 −55 −40	−50 −30 −15	−40 −40 −25	−50 −30 −15	⩽ 5				



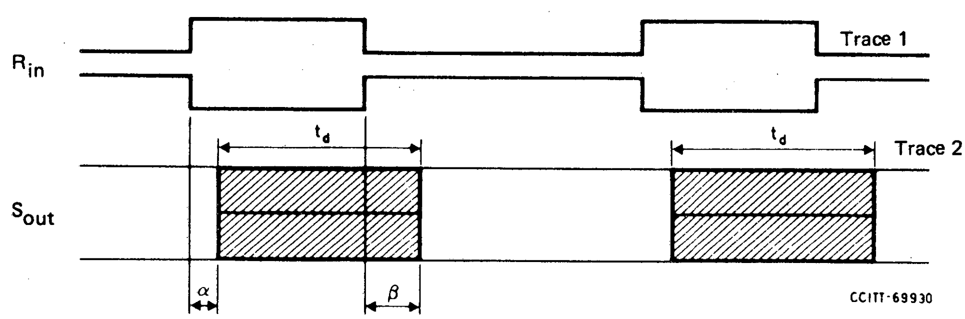
α Operate time

β Hangover time

t_d Time interval in which the distorted signal may be observed

FIGURE C-2/G.165

Traces for NLP operate and hangover times, L_R constant



α Operate time

β Hangover time

t_d Time interval in which the distorted signal may be observed

FIGURE C-3/G.165

Traces for NLP operate and hangover times, L_S constant

Reference

[1] CCITT Recommendation – *Echo suppressors suitable for circuits having either short or long propagation time*, Orange Book, Volume III.1, Recommendation G.161, ITU, Geneva, 1977.

Recommendation G.166

CHARACTERISTICS OF SYLLABIC COMPANDORS FOR TELEPHONY ON HIGH CAPACITY LONG DISTANCE SYSTEMS

(Malaga-Torremolinos, 1986; amended at Melbourne, 1988)

Compandors adhering to Recommendation G.162, *Yellow Book*, were intended for use in small capacity network systems and their use in large capacity network long-distance systems is not recommended. Compandors adhering to this Recommendation are intended for use in large capacity long-distance systems. Their use on small capacity network systems is optional. They are not intended for use in subscriber applications such as mobile communication systems.

1 General

1.1 Syllabic compandors are devices in which gain variations occur at a rate comparable to the syllabic rate of speech. A compandor consists of a combination of a compressor at one point in a communication path, for reducing the amplitude range of signals followed by an expander at another point for a complementary increase in the amplitude range. The compandor enhances the subjective speech performance primarily due to two actions. The compressor increases the average speech level of weaker signals prior to entering a communication path where increased noise is expected to be encountered. The expander, in returning the speech signal to its original dynamic range provides a subjective enhancement to the communication path by attenuating the noise perceived by the listening party during silences. For a further description of compandor operation see Annex A.

1.2 This Recommendation does not specify the detector characteristics, e.g., peak, r.m.s. or average.

The performance recommended may not be sufficient to ensure compatibility between compandors conforming to this Recommendation but which are of different design. Before using compressors and expanders of different design origins at opposite ends of the same circuit, Administrations should test them for compatibility. The tests should take account of the sensitivity of compandor performance to the characteristics of the test signal.

1.3 The use of a number of syllabic compandors on circuits carried on the same FDM carrier may result in a changed load being presented to the FDM system. The FDM system operating parameters could, therefore, require appropriate adjustment as a function of the load.

1.4 It should be noted that the subjective enhancement which occurs on speech, when syllabic compandors are used, does not apply to transmission of non-speech signals which may experience a signal-to-noise degradation on syllabic compandored circuits.

1.5 Some of the clauses given below specify the joint characteristics of a compressor and an expander in the same direction of transmission of a 4-wire circuit. The characteristics specified in this way can be obtained more easily if the compressors and expanders are of similar design; in certain cases close cooperation between Administrations may be necessary. Application rules for syllabic compandors address this issue.

2 Definitions

2.1 unaffected level

The unaffected level is the absolute level, at a point of zero relative level on the line between the compressor and the expander of a signal at 800 Hz, which remains unchanged whether the circuit is operated with the compressor or not. The unaffected level is defined in this way in order not to impose any particular values of relative level at the input to the compressor or the output of the expander.

To make allowances for the increase in mean power introduced by the compressor, and to avoid the risk of increasing the intermodulation noise and the overload which might result, the unaffected level must be adjusted taking into account the capacity of the system. (See Reference [1], Chapter II, Annex 4, for detailed discussion of this adjustment.)

2.2 **ratio of compression**

The ratio of compression of a compressor is defined by the formula:

$$\alpha = \frac{L_{1\,\mathrm{CIN}} - L_{2\,\mathrm{CIN}}}{L_{1\,\mathrm{COUT}} - L_{2\,\mathrm{COUT}}}$$

where

$L_{1\,\mathrm{CIN}}$ and $L_{2\,\mathrm{CIN}}$ are any two different compressor input levels within the compressor operating range.

$L_{1\,\mathrm{COUT}}$ and $L_{2\,\mathrm{COUT}}$ are the compressor output levels corresponding to input levels $L_{1\,\mathrm{CIN}}$ and $L_{2\,\mathrm{CIN}}$ respectively.

2.3 **ratio of expansion**

The ratio of expansion of an expander is defined by the formula:

$$\beta = \frac{L_{1\,\mathrm{EOUT}} - L_{2\,\mathrm{EOUT}}}{L_{1\,\mathrm{EIN}} - L_{2\,\mathrm{EIN}}}$$

where

$L_{1\,\mathrm{EIN}}$ and $L_{2\,\mathrm{EIN}}$ are any two different expander input levels within the expander operating range.

$L_{1\,\mathrm{EOUT}}$ and $L_{2\,\mathrm{EOUT}}$ are the expander output levels corresponding to input levels $L_{1\,\mathrm{EIN}}$ and $L_{2\,\mathrm{EIN}}$ respectively.

3 Characteristics of syllabic compandors

3.1 *Unaffected level*

A nominal value of −10 dBm0 for the unaffected level is recommended for high capacity systems. However, Administrations are free to mutually negotiate a different unaffected level to allow optimal loading of their transmission systems. Such variation is expected to be in the range −10 to −24 dBm0. The loading effects of pilot tones should be considered.

3.2 *Ratio of compression* α

The compandor compression ration α should be 2 over the range of level specified in § 3.4 and over the temperatura range +10 °C to +40 °C. The difference between the measured level and the calculated level at the output of the compressor assuming a value of exactly 2 should not exceed ± 0.25 dB.

3.3 *Ratio of expansion* β

The compandor expansion ratio β should be 2 over the range of level specified in § 3.4 and over the temperature range +10 °C to +40 °C. The difference between the measured level and the calculated level at the output of the expander assuming a value of exactly 2 should not exceed ± 0.4 dB.

3.4 *Range of level*

Under study

The range of level over which the recommended value of α and β should apply, should extend at least:

from +5 to −60 dBm0 at the input of the compressor, and

from +5 to −65 dBm0 at the nominal output of the expander.

3.5 *Variation of compressor gain*

The level at the output of the compressor, measured at 800 Hz, for an input level equal to the unaffected level, should not vary from its nominal value by more than ± 0.25 dB for a temperature range of +10 °C to +40 °C and a deviation of the supply voltage of ± 5% from its nominal value.

3.6 *Variation of expander gain*

The level at the output of the expander, measured at 800 Hz for an input level equal to the unaffected level, should not vary from its nominal value by more than ± 0.5 dB for a temperature range of +10 °C to +40 °C and a deviation of the supply voltage of ± 5% from its nominal value.

3.7 *Tolerances on the output levels of the combination of compressor and expander in the same direction of transmission of a 4-wire circuit*

The compressor and expander are connected in tandem. A loss (or gain) is inserted between the compressor output and expander input equal to the nominal loss (or gain) between these points in the actual circuit in which they will be used. Figure 1/G.166 shows, as a function of level of 800 Hz input signal to the compressor, the permissible limits of difference between expander output level and compressor input level. (Positive values indicate that the expander output level exceeds the compressor input level.)

The limits shall be observed at all combinations of temperature of compressor and temperature of expander in the range +10 °C to +40 °C. They shall also be observed when the test is repeated with the loss (or gain) between the compressor and expander increased or decreased by 2 dB and the measurement corrected by ± 4.0 dB, assuming a β of 2.00.

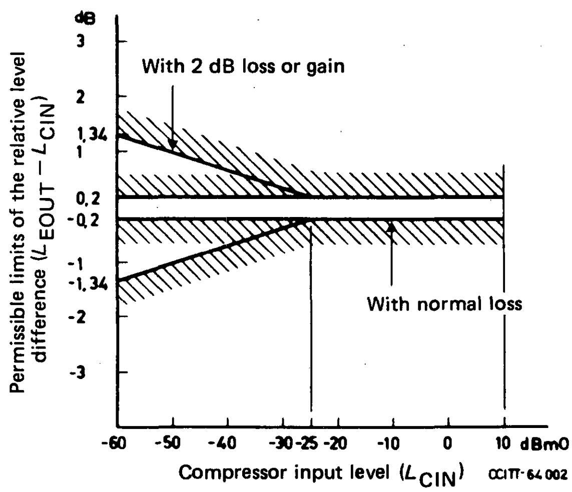


FIGURE 1/G.166

Tolerances on the output levels of the combination of compressor and expander

3.8 *Conditions for stability*

See descriptions given in § 2.6 of Recommendation G.162, Volume III of the *Yellow Book*, ITU, Geneva, 1981, § 2 of Recommendation G.143, *Red Book*, and Reference [1].

The limits shall be observed at all combinations of temperature of compressor and temperature of expander in the range +10 °C to +40 °C. They shall also be observed when the test is repeated with the loss (or gain) between the compressor and expander increased or decreased by 2 dB.

Note – The change of gain (or loss) of 2 dB mentioned in § 3.7 above is equal to twice the standard deviation of transmission loss recommended as an objective for international circuits routed on single group links in Recommendation G.151, § 3.

4 Impedances and return loss

The nominal value of the input and output impedances of both compressor and expander should be 600 ohms (nonreactive).

The return loss with respect to the nominal impedance of the input and the output of both the compressor and the expander should be no less than 20 dB over the frequency range 300 to 3400 Hz and for any measurement level between +5 and −60 dBm0 at the compressor input or the expander output.

5 Operating characteristics at various frequencies

5.1 *Frequency characteristic with control circuit clamped*

The control circuit is said to be clamped when the control current (or voltage) derived by rectification of the signal is replaced by a constant direct current (or voltage) supplied from an external source. For purposes here, the value of this current (or voltage) should be equal to the value of the control current (or voltage) obtained when the input signal is set to the unaffected level.

For the compressor and the expander taken separately, the variations of loss or gain with frequency should be contained within the limits of a diagram that can be deduced from Figure 1/G.132 by dividing the tolerance shown by 8, the measurement being made with a constant input level corresponding to the unaffected level.

5.2 *Frequency characteristic with control circuit operating normally*

The limits given in § 5.1 should be observed for the compressor when the control circuit is operating normally, the measurement being made with a constant input level corresponding to the unaffected level.

For the expander, under the same conditions of measurement, the limits can be deduced from Figure 1/G.132 by dividing the tolerances shown by 4.

These limits should be observed over the temperature range +10 °C to +40 °C.

6 Nonlinear distortion

6.1 *Harmonic distortion*

The total harmonic distortion, measured with an 800 Hz sine wave at the unaffected level, should not exceed 0.5% for the compressor and the expander taken separately.

Note – Even in an ideal compressor, high output peaks will occur when the signal level is suddenly raised. The most severe case seems to be that of voice-frequency signalling, although the effect can also occur during speech. It may be desirable, in exceptional cases, to fit the compressor with an amplitude limiter to avoid disturbance due to transients during voice-frequency signalling.

6.2 *Intermodulation tests*

It is necessary to add a measurement of intermodulation to the measurements of harmonic distortion whenever compandors are intended for international circuits (regardless of the signalling system used), as well as in all cases where they are provided for national circuits over which multi-frequency signalling, or data transmission using similar types of signals, is envisaged.

The intermodulation products of concern to the operation of multi-frequency telephone signalling receivers are those of the third order, of type $(2f_1 - f_2)$ and $(2f_2 - f_1)$, where f_1 and f_2 are two signalling frequencies.

Two signals at frequencies 900 Hz and 1020 Hz are recommended for these tests.

Two test conditions should be considered: the first, where each of the signals at f_1 and f_2 is at a level of −5 dBm0 and the second, where they are each at a level of −15 dBm0. These levels are to be understood to be at the input to the compressor or at the output of the expander (uncompressed levels).

The limits for the intermodulation products are defined as the difference between the level of either of the signals at frequencies f_1 or f_2 and the level of either of the intermodulation products at frequencies $(2f_1 - f_2)$ or $(2f_2 - f_1)$.

A value for this difference which seems adequate for the requirements of multi-frequency telephone signalling (including end-to-end signalling over three circuits in tandem, each equipped with a compandor) is 32 dB for the compressor and the expander separately.

Note 1 – These values seem suitable for Signalling System No. 5, which will be used on some long international circuits.

Note 2 – It is inadvisable to make measurements on a compressor plus expander in tandem, because the individual intermodulation levels of the compressor and of the expander might be quite high, although much less intermodulation is given in tandem measurements since the characteristics of compressor and expander may be closely complementary. The compensation encountered in tandem measurements on compressor and expander may not be encountered in practice, either because there may be phase distortion in the line or because the compressor and expander at the two ends of the line may be less closely complementary than the compressor and expander measured in tandem.

Hence the measurements have to be performed separately for the compressor and the expander. The two signals at frequencies f_1 and f_2 must be applied simultaneously, and the levels at the output of the compressor or expander measured selectively.

7 Noise

The effective value of the sum of all noise referred to a zero relative level point, the input and the output being terminated with resistances of 600 ohms, shall be less than or equal to the following values:

- at the output of the compressor: −45 dBm0p
- at the output of the expander: −80 dBm0p.

8 Transient response

The overall transient response of the combination of a compressor and expander which are to be used in the same direction of transmission of a 4-wire circuit fitted with compandors shall be checked as follows:

The compressor and expander are connected in tandem, the appropriate loss (or gain) being inserted between them as in § 3.7.

A 12-dB step signal at a frequency of 2000 Hz is applied to the input of the compressor, the actual values being a change from −16 to −4 dBm0 for attack, and from −4 to −16 dBm0 for recovery. The envelope of the expander output is observed. The overshoot (positive or negative), after an upward 12-dB step expressed as a percentage of the final steady-state voltage, is a measure of the overall transient distortion of the compressor-expander combination for attack. The overshoot (positive or negative) after a downward 12-dB step, expressed as a percentage of the final steady-state voltage is a measure of the overall transient distortion of the compressor-expander combination for recovery. For both these quantities the permissible limits shall be ± 20%. These limits shall be observed for the same conditions of temperature and of variation of loss (or gain) between compressor and expander as for the test in § 3.7.

In addition, the attack and recovery times of the compressor alone shall be measured as follows:

Using the same 12-dB steps as above for attack and recovery respectively, the attack time is defined as the time between the instant when the sudden change is applied and the instant when the output voltage envelope reaches a value equal to 1.5 times its steady-state value. The recovery time is defined as the time between the instant when the sudden change is applied and the instant when the output voltage envelope reaches a value equal to 0.75 times its steady-state value.

The permissible limits shall be:

- 3 ms minimum, 5 ms maximum for the attack time, and
- 13.5 ms minimum, 22.5 maximum for the recovery time.

ANNEX A

(to Recommendation G.166)

Compandor enhancement characteristics

The improvement which the compandor makes available is based on the fact that interference is most objectionable during quiet speech or pauses, but is masked by relatively loud speech. While it will not be necessary, therefore, to alter the performance of the system for speech signals at a high level, an improvement has to be provided when the signal level is low. This noise reduction can be arranged by introducing loss at the receiving end of the circuit during periods when the signal is faint or absent. The loss so introduced will affect the noise or crosstalk which has crept in along the route, so that the interference is reduced by the amount of this loss. However, the desired signals are also affected, and in order that the speech level finally received shall be unchanged by the insertion of the compandor, an equal amount of gain has to be introduced at the sending end. The overall equivalent of the circuit is thereby kept constant, and also the low level signals are raised above the background of interference on the line.

The above-mentioned condition must not, however, be allowed to persist when high-level signals have to be transmitted, or overloading could occur in the line amplifiers along the route. The function of the compandors is to introduce the required amounts of gain and loss automatically in just such a way that the overall circuit equivalent remains unchanged irrespective of the speech level, while the signal-to-noise ratio is increased for low-level signals. This is shown schematically in the level diagram of Figure A-1/G.166. For one particular level, called the *unaffected* level *X*, the use of the compandor at no point introduces gain or loss, and the signal passes at an unchanged level throughout the system, as shown by (1), (2), (3).

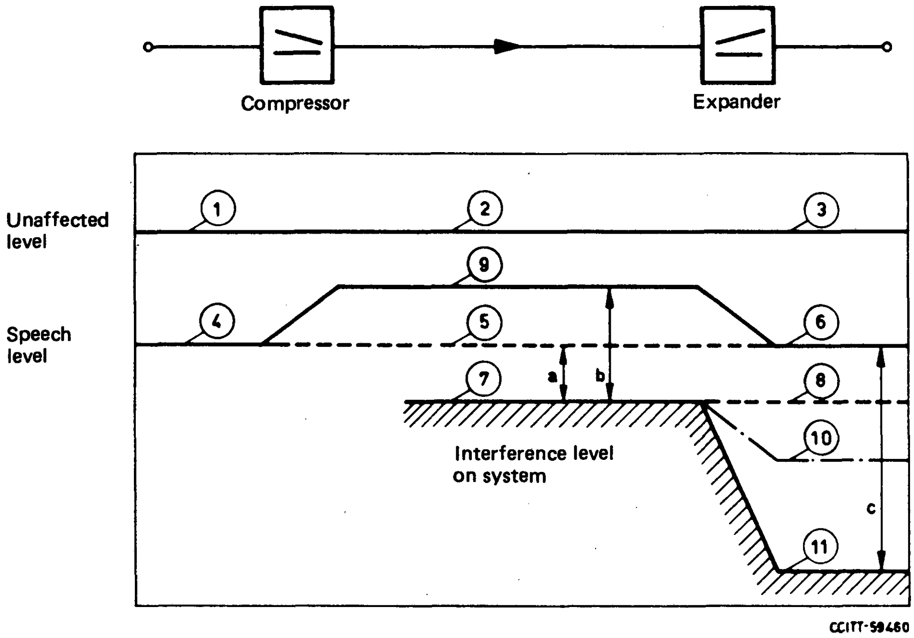
Any given level of speech (4) would also normally (i.e. without compandors) pass at an unchanged level through the system as shown at (4), (5), (6). If we suppose that the level of interference on the system (noise, crosstalk, etc.) is that shown by (7), the signal/interference ratio is then given by *a*, and the interference level appearing at the output is that shown by (8), during both speech and pauses.

By the introduction of the compandor, however, the incoming speech level (4) is raised to (9), thereby giving a signal/interference ratio within the system of *b*. The level of the speech is restored to (6) at the receiving end, and the corresponding interference level *during speech* is shown at (10). However, as stated earlier, of even greater significance is the interference level during pauses, which is that shown at (11). Thus the effective ratio between speech signals and interference heard *during pauses* has the value shown by *c*.

The part of the compandor at the sending end is called the compressor, because the range of levels of the incoming speech signals is compressed. The unaffected level recommended by the CCITT for high capacity systems is -10 dBm0. However, Administrations may mutually negotiate a different unaffected level to permit optimal loading of their transmission systems. The unaffected level is expected to range from -10 to -24 dBm0. The selected unaffected level will affect the mean power per channel.

The part of the compandor at the receiving end is called the expander, and the same level remains unchanged.

It will be seen from the foregoing that, when compandors are required, one compandor has to be inserted at each end of the telephone circuit in the voice-frequency 4-wire path, with the compressor in the sending channel and the expander in the receiving channel.



All levels are referred to a point of zero relative level.
Full lines show system performance with compandors; dashed lines without.
a Signal/interference without compandor.
b Increased signal/interference obtained during speech due to use of compressor.
c Ratio between speech and interference heard during pauses, due to use of entire compandor. For a theoretical 2:1:2 compandor, this has a value 2 *b*.

FIGURE A-1/G.166

Level chart for transmission system with compandors

Reference

[1] CCITT Manual *Transmission planning of switched telephone networks*, ITU, Geneva, 1976.

1.7 Transmission plan aspects of special circuits and connections using the international telephone connection network

Recommendation G.171

TRANSMISSION PLAN ASPECTS OF PRIVATELY OPERATED NETWORKS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984 and Melbourne, 1988)

1 General

This Recommendation primarily concerns privately switched networks for telephony. In certain circumstances these networks may be suitable for the transmission of analogue encoded data signals but no special arrangements have been made to ensure satisfactory performance in this respect. Although digital facilities on a portion of a circuit or digital switches may be employed, §§ 1-9 of this Recommendation mainly covers analogue interconnection of circuits and switches. §§ 10 and 11 cover some aspects of all digital connections.

It should be noted that not all Administrations provide such a facility. Others permit interconnection between private telephone networks and the public telephone network. In this latter case assurance cannot always be given that transmission performance conforming to CCITT standards will be obtained. In a similar manner the interconnection of multiple private networks may result in connections with degraded transmission performance.

It is not intended that this Recommendation should prevent the making of bilateral agreements for special network configurations. In such circumstances it is suggested that the network plans given here be used as a guide to permissible alternative arrangements.

The transmission plan described in this Recommendation is similar to that of the switched public network and therefore it is desirable that several other Recommendations such as G.151 be complied with where possible and appropriate. In this respect, it is noted that some requirements in Recommendation G.151 are more stringent than those contained in this Recommendation (e.g. attenuation distortion), and some impairments which are more important for voice-band data are covered in G.151 but are not included in this Recommendation.

A major consideration in the private plan is that typically, a PBX functions both in the role of a local exchange and a tandem centre and therefore it is necessary to use a technique such as pad switching to achieve the appropriate connection loss.

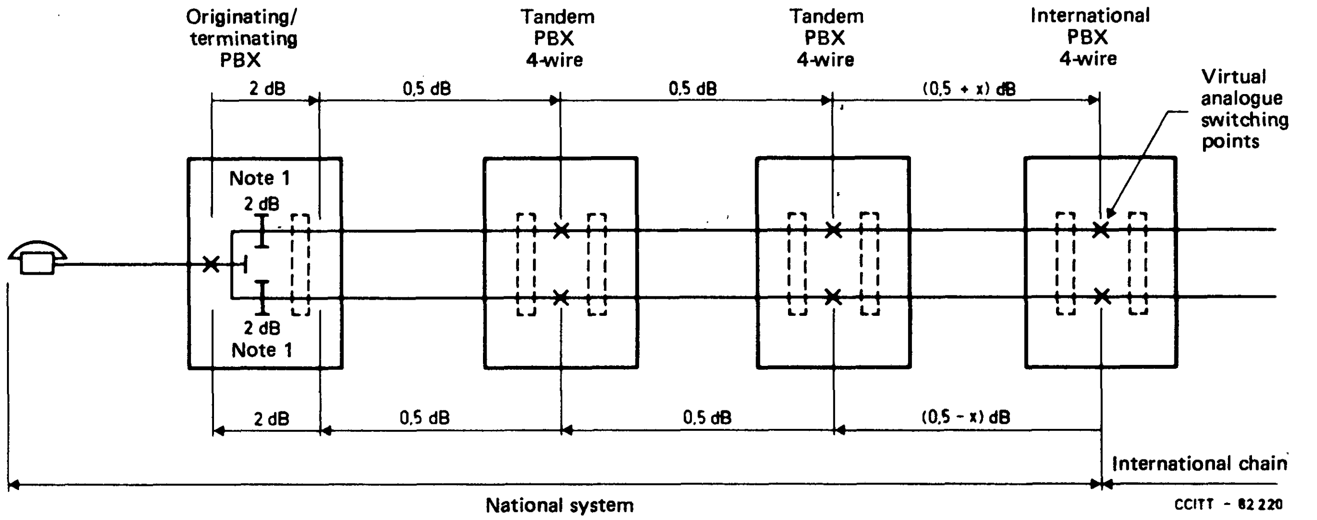
The network configurations discussed in this Recommendation may also be implemented by replacing some or all of the PBXs with switching capability dedicated to a private user that is located on the premises of the telephone Administration rather than on the customers' premises.

Recommendation M.1030 provides information on the maintenance of international leased circuits forming part of private switched networks. Recommendation Q.8 describes signalling systems to be used for international leased circuits.

2 Network configurations

2.1 *Preferred 4-wire network configurations*

The preferred network configurations are shown in Figure 1/G.171 and Figure 2/G.171. Four-wire PBXs are used in conjunction with low loss 4-wire circuits. The loss plans shown are for illustration and are based on the national plans discussed in Recommendation G.121. For convenience the later figures will only use the variable loss plan for illustration. It should be noted that the fixed loss plan without modification, (Figure 2/G.171) is only suitable when the national system is limited in size at most to 1000 to 1500 km.



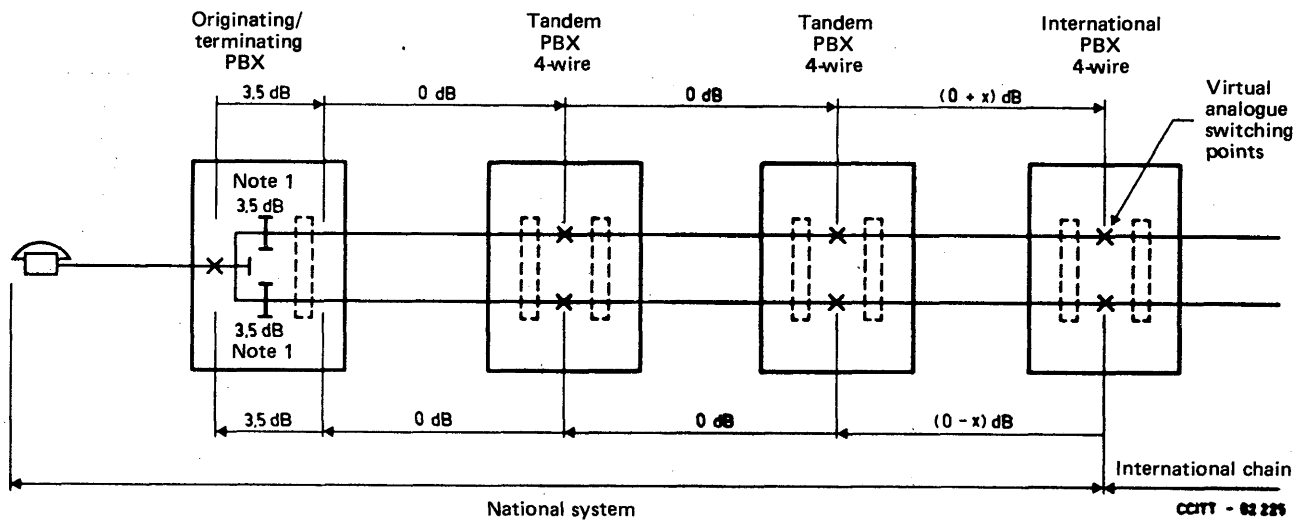
Note 1 – 2 dB switchable pad or equivalent. The pad is switched in on terminating/originating connections and out on tandem connections.

Note 2 – The value of x is the loss necessary to convert between actual and virtual analogue switching points.

Note 3 – Denotes echo canceller or half echo suppressor that may be fitted.

FIGURE 1/G.171

4-wire network configuration variable loss



Note 1 – 3.5 dB switchable pad or equivalent. The pad is switched in on terminating/originating connections and out on tandem connections.

Note 2 – The value of x is the loss necessary to convert between actual and virtual analogue switching points.

Note 3 – Denotes echo canceller or half echo suppressor that may be fitted.

FIGURE 2/G.171

4-wire network configuration fixed loss

At each PBX a switchable pad or equivalent is used in such a manner that the pad is "out" of the circuit when the PBX switch is in the tandem mode but is "in" the circuit at an originating/terminating PBX. This allows a flexible configuration of PBXs while maintaining control on echo loss and overall loudness rating. The PBX terminating the international chain is referred to as the International PBX (IPBX). Conceptually the virtual analogue switching points are located at the IPBX.

It should be noted that typically short PBX subscriber lines may need more loss in the connections to meet the Recommendations on send and receive LR at the virtual analogue switching points. This will of course depend on the send and receive LR of the telephone and subscriber line. It may also be necessary to add loss on intra-PBX calls.

2.2 *Allowed network configuration using 2-wire circuits*

The configuration shown in Figure 3/G.171 allows for the use of 2-wire circuits. This is not desirable and should be avoided. If used, 2-wire circuits should only be deployed between an originating/terminating PBX and the first tandem PBX. A 2-wire circuit may be all 2-wire or consist of a mix of 2-wire and 4-wire segments.

The use of 2-wire circuits may require special loss control at the connecting tandem PBX. If the stability/echo requirements of §§ 5 and 6 cannot be met otherwise, it will be necessary to switch the pad or equivalent loss into the tandem connection to the 2-wire circuit. This would require special translation and control at the tandem PBX to identify 2-wire trunks not consistent with the stability/echo requirements. If this is not possible the added loss is required on all tandem connections, causing a degradation in overall loudness rating.

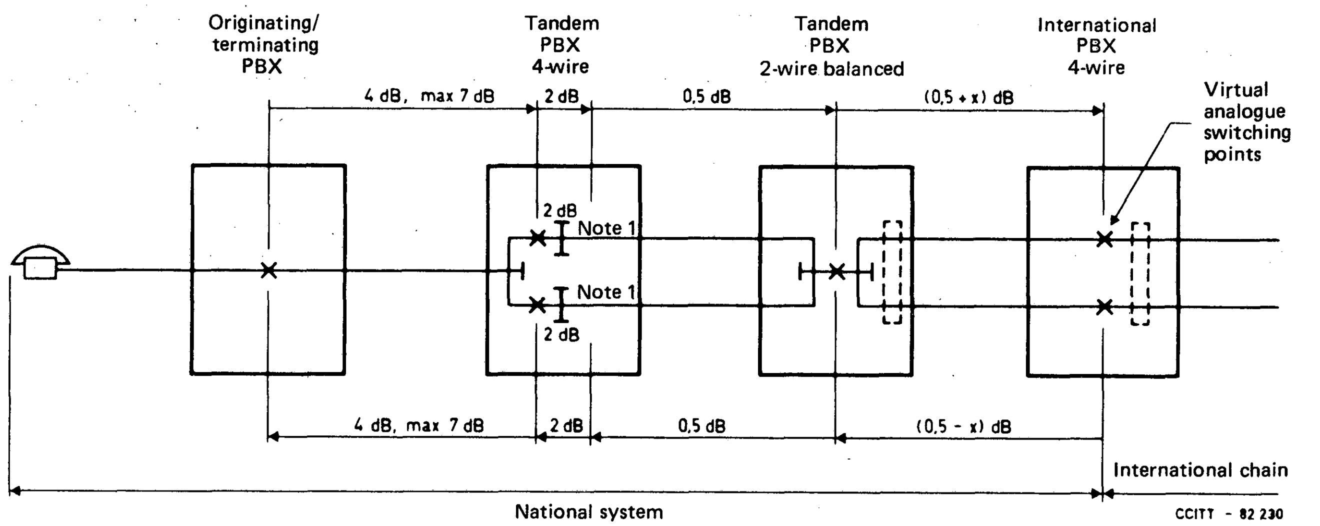
2.3 *Balanced 2-wire tandem PBXs*

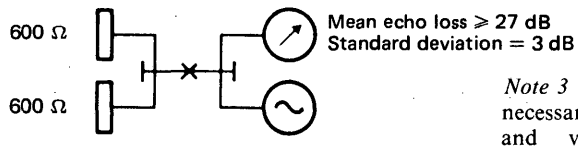
As shown in Figure 3/G.171, 2-wire PBXs may be used in the tandem mode if the collection of interconnected 4-wire interfaces meet balance requirements as shown in Note 2. With a mean echo loss of 27 dB and a standard deviation of 3 dB, the effects of echo at the PBX are negligible with respect to the principal echo at the originating/terminating PBX or at the tandem PBX connected to a 2-wire circuit. Recommendation G.131 refers to these balance values in reference to tandem 2-wire switches. It is provisionally recommended, that at most three 2-wire PBXs be contained in a single national extension. This would correspond to a 2-wire terminating/originating PBX with two additional balanced 2-wire tandem PBXs.

As shown in Figure 4/G.171, the IPBX may be 2-wire. The virtual analogue switching points are adjacent to the 2-wire/4-wire terminating unit on the 4-wire side. If the PBX is used for tandem switching it must be balanced and pad switching or equivalent should be employed as previously described.

2.4 *Network constraints*

To control loss, distortion, noise and delay a maximum of seven circuits in tandem is recommended from originating to terminating PBX. Allocation of the number of circuits between national and international chains should remain flexible and should be done on an individual network basis subject to the seven circuit maximum. There should, however, be a maximum of five tandem circuits in a connection in any single national extension.





Note 1 – 2 dB switchable pad or equivalent. The pad is switched in the tandem connection to a 2-wire circuit if the stability balance requirements of §§ 5 and 6 cannot be met otherwise.

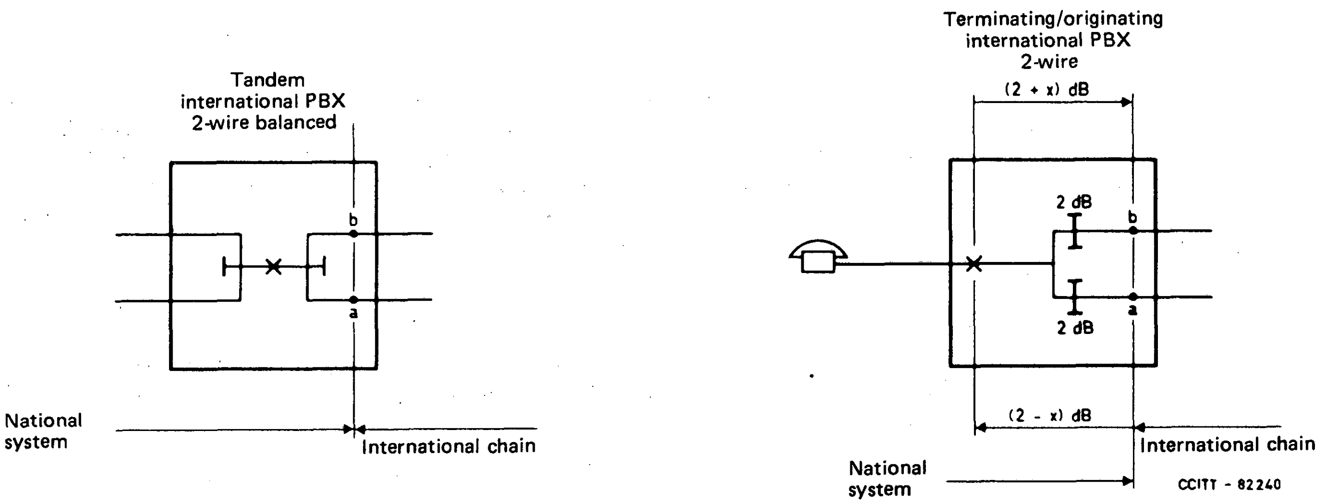
Note 2 – Two-wire balanced PBX.

Note 3 – The value of x is the loss necessary to convert between actual and virtual analogue switching points.

Note 4 – Denotes echo canceller or half echo suppressor that may be fitted.

FIGURE 3/G.171

Network configuration using 2-wire circuits



Note – The value of x is the loss necessary to convert between actual and virtual analogue switching points.

FIGURE 4/G.171

International PBX 2-wire

Recommendation G.114 on mean one-way propagation time, should be observed. In particular, at most one satellite circuit should be present in a connection. If it is not possible to adhere to this constraint assurance cannot be given that transmission conforming to CCITT standards will be obtained.

The arrangements shown in the Figures 1/G.171 to 4/G.171, are suggested methods of meeting the Recommendations on stability, echo and CRE (LR) as given in §§ 5, 6 and 7. Other approaches achieving the same performance are acceptable.

3 Nominal transmission loss of international circuits

3.1 *Four-wire circuits*

Recommendation G.111 is applicable to this type of circuit and therefore the normal transmission loss at the reference frequency between the virtual analogue switching points will be 0.5 dB for circuits employing analogue transmission. An indication of the locations of the virtual analogue switching points is also given in Recommendation G.111 and conceptually these will be at the private exchange on which the circuit terminates. Four-wire circuits do not contain 2-wire circuit sections.

3.2 *Two-wire presented circuits*

This nomenclature is intended to cover circuits which are not available with a 4-wire interface (e.g., circuits between 2-wire switching nodes).

For the purposes of this Recommendation the location of the virtual analogue switching points for this type of circuit can be considered as being adjacent to the 2-wire/4-wire terminating unit (4-wire side). It can then be treated in the same way as a 4-wire circuit. (See Figure 4/G.171.)

Note 1 – The real loss of the circuit between actual switching points at the reference frequency cannot be exactly specified without prior knowledge of the switching levels.

Note 2 – Differences between the two directions of transmission in the real loss of the circuit may occur. The annexes to Recommendation G.121 examine this effect in some detail.

Note 3 – A circuit is defined as the complete transmission path between the switch points of the two private exchanges concerned.

Note 4 – Actual transmission loss will differ from the nominal values and will vary with time. For all circuits, variations with time of the overall loss at the reference frequency (including daily and seasonal variations but excluding amplitude hits) should be as small as possible but should not exceed ± 4 dB.

4 Nominal transmission loss of national circuits

4.1 *Four-wire circuits*

The nominal loss at the reference frequency should be 0.5 dB between actual switching points. This includes 4-wire circuits terminated on balanced 2-wire PBXs. The loss of the circuit between the actual and virtual analogue switching points of the IPBX depends upon the PBX transmission level used in the national plan.

4.2 *Two-wire circuits*

Two-wire circuits may contain mixed 2-wire/4-wire segments. The nominal loss at the reference frequency should not exceed 7 dB, and should preferably be lower, for example 4 dB.

Note 1 – Certain national arrangements in large countries may employ a nominal loss in excess of 0.5 dB on 4-wire circuits or may employ a distance dependent loss in order to improve talker echo performance without use of echo control devices. This approach is acceptable if the Recommendations on LR of § 7 are satisfied.

Note 2 – Since leased circuits may contain circuit sections routed in local unloaded distribution cable pairs, care will be needed to ensure that there is an adequate stability bearing in mind the relative gain introduced by unloaded cable pairs.

Note 3 – Loss variation should be controlled as described for international circuits.

5 Stability

5.1 *National 2-wire circuits/2-wire presented circuits*

Two-wire presented circuits are 4-wire circuits terminated on 2-wire PBXs. Provisionally the nominal loss around any 4-wire loop should not be less than 6 dB at any frequency in the band 0 to 4 kHz, for all the terminal conditions encountered in normal operation (e.g. including the idle state and the set-up phase of the connection).

5.2 *Terminating systems for international circuits*

National terminating systems which interface with international circuits should comply with the stability requirements of Recommendation G.122. In the case of 2-wire presented international circuits, the virtual analogue switching points can be considered as being adjacent to the 2-wire/4-wire terminating unit (4-wire side). (See Figure 4/G.171.)

During the set-up and clear-down of a call the loss between virtual analogue switching points (*a-b*) must satisfy that of Recommendation G.122, § 1.

The signalling system has an influence on the loss under set-up conditions as explained in Recommendation G.122. If the requirement cannot be met with the configurations described herein, it will be necessary to increase either the switched or fixed losses.

During an established communication, the suggested configurations of Figures 1/G.171, 2/G.171 and 3/G.171 provide for compliance with Recommendation G.122 as follows. Assuming that the PBX subscriber lines have a distribution of stability balance return loss equivalent or superior to that of public subscriber lines and that the distribution has a mean value of 6 dB and a standad deviation of $\sqrt{6.25}$ dB, then the distribution of stability of loss (*a-b*) is consistent with the recommended distribution of Recommendation G.122, § 1 using the same assumptions as contained in that Recommendation.

Note – In order to obtain the recommended value of stability on 2-wire presented low-loss (e.g. 3 dB) circuits, it will be necessary for the 2-wire/4-wire terminating units to be located at the private exchanges. This may not be necessary on circuits with a higher nominal loss. The CCITT manual cited in [1] gives guidance on this topic.

6 Echo

6.1 *Terminating systems for international circuits*

National terminating systems which interface with international circuits should comply with the echo loss (*a-b*) requirements of Recommendation G.122, § 2 and the requirements of Recommendation G.131, § 2 for the control of echo.

During an established communication, the suggested configuration of Figures 1/G.171, 2/G.171 and 3/G.171 provide for compliance with Recommendation G.122, § 2 as follows. Assuming that PBX subscriber lines have a distribution of echo balance return loss equivalent or superior to that of public subscriber lines and that the distribution has a mean value of 11 dB with a standard deviation of 3 dB, then the distribution of echo loss (*a-b*) is consistent with the recommended distribution of Recommendation G.122, § 2 using the same assumptions as contained in that Recommendation.

6.2 *Echo control devices*

When echo control devices (e.g., echo suppressors or echo cancellers) are necessary it is preferable that they be located at the private exchange. This minimizes end delay and also allows disabling of the device during tandem operation, if necessary. In addition, some signalling systems require local disabling of echo control devices during certain signalling phases. The echo control device (echo canceller or far-end operated half-echo suppressor) for the international circuit would be located at the PBX terminating the international chain, since this same PBX typically could originate/terminate traffic or tandem switch to many trunks without echo control. However, if connecting national circuits introduces enough delay to warrant echo control, then echo control devices would also be provided on these circuits.

If far-end operated half-echo suppressors are used, intermediate suppressors should be disabled. This is not necessary for echo cancellers since tandem operation does not cause degraded performance. In either case the functioning echo control device on the connection is effectively moved closer to the PBX subscriber line, further reducing end delay. The echo control devices are located in the 4-wire portion of the network and between the first hybrid and the international chain. However, the devices may be located at the international centre when the previously described performance factors can still be satisfactorily controlled and there is a maintenance and/or cost advantage for such location.

The loss of circuits fitted with echo control devices should be 0 dB.

Echo suppressors and cancellers according to Recommendation G.164 and Recommendation G.165, typically require 6 dB of signal loss (*a-b*) for the *actual* signal converging the canceller or being controlled by the suppressor. Therefore it is desirable from a performance point of view that the stability loss (*a-b*) during an established connection should be at least 6 dB, since this will ensure proper operation for *any* signal (frequency spectrum) in the band 0-4 kHz. However this may not be economically achievable. The spectrum of a typical speech signal and return path is such that if the *echo* loss (*a-b*) is at least 6 dB, then the signal loss (*a-b*) for the speech signal is expected to be at least 6 dB and the echo control devices should operate properly. However, the spectrum of some voice-band data signals and of the return path is such that an echo loss (*a-b*) of at least 10 dB is required to ensure that the signal loss (*a-b*) for the actual data signal is 6 dB. (Modems operating half-duplex on satellite circuits may require echo protection for proper operation.) Therefore, when an echo control device is located at a PBX, the echo loss at the 4-wire terminals of the device looking towards the subscriber line should be at least 6 dB for 99.5% of the connections and 10 dB for 95% of the connections for all network configurations during an established communication. This is not a new requirement in that the values are consistent with the recommended distribution for echo loss independent of the number of circuits between the echo control device and the subscriber line, assuming the distribution is Gaussian, which is a conservative assumption.

The suggested configuration of Figures 1/G.171, 2/G.171 and 3/G.171 provides for compliance with the minimum echo loss Recommendations. Using these configurations there is always a loss pad or equivalent between the echo control device and the 2-wire termination. Then, under the conditions described in § 6.1, the distribution of echo loss at the terminals of the echo control device is consistent with the recommended distribution.

If the private network uses echo suppressors and connects to a public network using echo cancellers, then difficulty in canceller convergence may be experienced when the suppressor is in the tail path of the canceller. However, performance will then be determined by the echo control devices at each end of the connection.

7 Loudness ratings (LRs) of extension circuits

7.1 *Loading*

Administrations must ensure that the technical arrangements that they authorize in respect of operating levels, sensitivities, etc. for private networks are not in conflict with the design criteria of the international transmission system. Attention is drawn to Recommendation G.121, § 3, which specifies a nominal minimum value of 2 dB sending LR referred to the virtual analogue switching point.

7.2 *Sending LR*

The maximum sending LR of the telephone and PBX subscriber line circuit (that portion analogous to the local telephone circuit in the public network) should not exceed 10.5 dB. This value is in accord with the example of a maximum local telephone circuit used in Figure 1/G.103. In practice, it is to be expected that most sending LR values will be considerably lower than this limit.

Administrations should attempt to choose values such that they comply with the preferred long-term objective of Recommendation G.121, § 1 (value referred to the virtual analogue switching point).

7.3 *Receiving LR*

The maximum receiving LR of the telephone and PBX subscriber line circuit (that portion analogous to the local telephone circuit in the public network) should not exceed 4 dB. This value is in accord with the example of a maximum local telephone circuit used in Figure 1/G.103. In practice it is to be expected that most receiving LR values will be considerably lower than this limit although due account must be taken of the need to preserve adequate margins against excessive noise, crosstalk and sidetone.

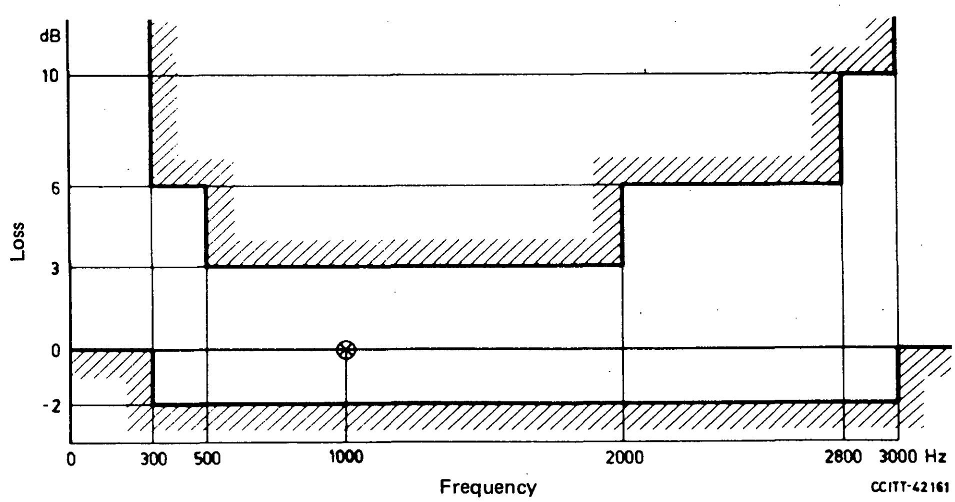
Administrations should attempt to choose values such that they comply with the preferred long-term objective of Recommendation G.121, § 1 (values referred to the virtual analogue switching point).

The sending LR and receiving LR for all connections should be such that there is compliance with Recommendation G.111, § 3.2 on overall LR.

8 **Loss/frequency distortion**

8.1 *Four-wire circuits*

The loss/frequency distortion of each 4-wire circuit should not exceed the limits shown in Figure 5/G.171. These limits are also applicable to the 4-wire portion of the circuit if it is terminated in a 2-wire switching node (see § 2).



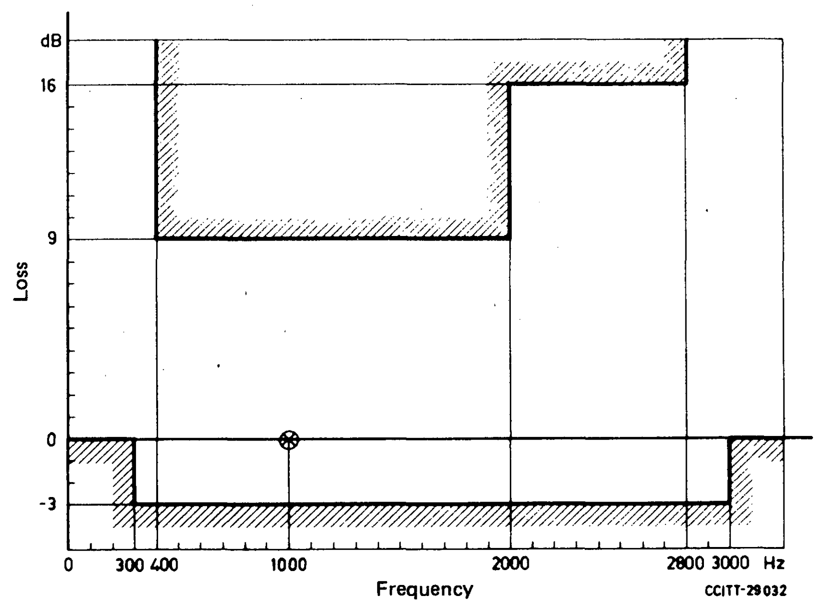
Note – The figures of 300 Hz and 3 kHz for the limitation of edge-band gain are provisional because Recommendation G.232 [2] permits a wider frequency range for FDM terminal equipment.

FIGURE 5/G.171

Limits for overall loss of the circuit relative to that at 1000 Hz for 4-wire circuits

8.2 *Two-wire presented circuits*

The loss/frequency distortion of each 2-wire circuit should not exceed the limits shown in Figure 6/G.171.



Note – The figures of 300 Hz and 3 kHz for the limitation of edge-band gain are provisional because Recommendation G.232 [2] permits a wider frequency range for FDM terminal equipment.

FIGURE 6/G.171

Limits for overall loss of the circuit relative to that at 1000 Hz for 2-wire presented circuits

9 Noise

The requirements of the relevant Recommendations should be met in respect of noise by each of the circuit sections and Recommendations G.123 and G.143, § 1 gives some general guidance on system noise characteristics. The nominal level of random noise power at the private exchange will depend upon the actual constitution of the circuit but should not exceed –38 dBm0p (provisional maintenance limit for circuits longer than 10 000 kilometres). In practice circuits of shorter length will exhibit substantially less random noise. Figure 7/G.171 serves as a guide to the expected performance.

Circuits having sections routed via communication satellites, designed according to Recommendation G.153, may be assessed in respect of noise performance by ascribing a nominal 1000 km of circuit length for the satellite path. It should be noted, however, that although such an allowance is appropriate for most satellites carrying international traffic, there may be certain locations where noise levels in excess of this value may be found.

10 Digital interconnection

In a digital private network of digital PBXs digitally interconnected a principal issue is the loss plan. In order to achieve transparent digital connections in the PBX network, the loss between digital interfaces should be 0 dB. However, it is necessary to insert loss in the PBX associated with the interconnection of digital and analogue interfaces. If digital loss is introduced between digital interfaces, it is desirable that options be made available to bypass the digital pad so that transparent connections can be provided. The bypass of digital pads may require special signalling arrangements.

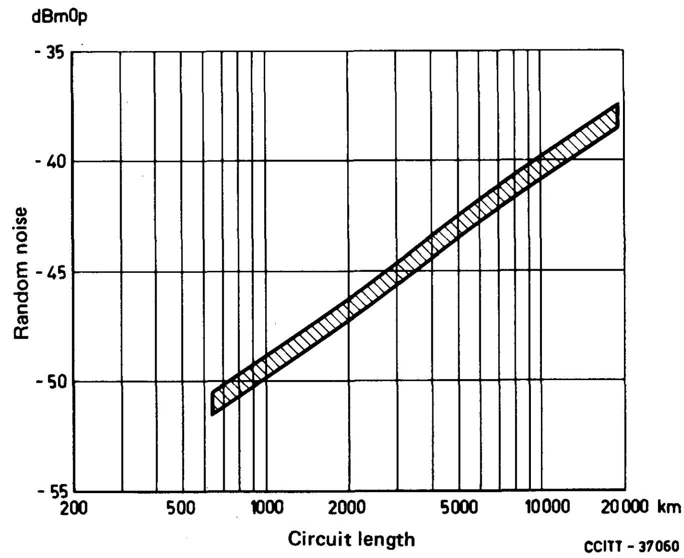


FIGURE 7/G.171

Random noise circuit performance

In a private network there are normally several different categories of analogue interfaces on the digital PBX. Such interfaces correspond to on-premise stations, off-premise stations, analogue tie trunks to the other PBXs and analogue connections to the PSN. The port-to-port loss matrix in the digital PBX between the combination of analogue and digital interfaces in conjunction with the loss of any analogue facilities and lines make up the overall loss plan. Different PBXs using the same port-to-port loss matrix are compatible for use in the same network consistent with the overall loss plan.

Because of the many different types of cross connections the overall loss plan represents a compromise, with optimum performance not being achieved on all connections. The loss plan should provide for acceptable send, receive, and overall loudness rating as discussed in § 7.3 of this Recommendation for all connection types. The annex to this Recommendation describes, as an example, one loss plan developed to meet this objective. At the current time the annex does not deal with digital interfaces to digital telephones.

11 Interconnection to the public switched telephone network

It is not always possible to assure that transmission performance meets CCITT standards when interconnecting private networks to the public switched network. This situation is complicated by the many different types of connections that are possible. In analogue networks a common interconnection problem is an increase of loss and a degradation in overall loudness rating. Relative level requirements at the digital exchange as described in Recommendation Q.552 (§ 2.2.4) should be complied with. For digital networks it is possible to make the interconnection more nearly transparent. The following guidelines apply to digital networks:

i) The preferred interconnnection between digital PBX and digital end-office employs digital facilities with a transparent connection at the end office. The loss preferably should be moved to the private network at the digital/analogue conversion point or digital telephone.

ii) Encoding and decoding levels in the private network should be consistent with the national plan and provide loudness ratings as in Recommendation G.121.

iii) A synchronization plan for the private network should be compatible with the national synchronization strategy.

When an analogue or digital private network is interconnected to the public switched network and an international connection is established, the national extension consists of the PSTN and connected private network. All requirements on the national extension should be complied with in this configuration. In particular the loudness rating requirements of Recommendation G.121 and the echo and stability requirement of Recommendation G.122 should be complied with. The PSTN with the private network connected should meet the echo and stability requirements at the virtual analogue switching point.

Control of delay and talker-echo performance can create problems on interconnections of private networks and the PSN. First, since it is likely that echo control devices in the private network will be in tandem with such devices in the PSN, echo cancellers should be used in the private network to prevent impairment. Further specific areas of concern are as follows:

i) Talker echo performance on calls where echo control is not normally provided in the PSN. The additional private network delay may result in unacceptable performance on a significant proportion of calls (Rule M, Recommendation G.131).

ii) The additional private network delay may result in the end-delay limits for existing PSN echo control devices being exceeded. In order to control these factors it may be necessary to deploy echo cancellers in the private network on the interconnecting circuit, particularly to control reflections back to the PSN. The delay limits and/or echo control strategy used in the private network should ensure acceptable talker-echo performance in accordance with the rules in Recommendation G.131, § 2.3. In addition, the private network delay limits should be as low as practicable to minimize overall connection delays in accordance with Recommendation G.114.

Other parameters essential to an acceptable overall performance include quantizing distortion, sidetone, noise, attenuation distortion, group delay distortion, crosstalk, error rate, jitter and wander. It is not practicable to provide private network limits for these parameters consistent with the national extension over-all requirements for all configurations. It is important that the constituent parts of the private network be designed in accordance with the relevant CCITT Recommendation covering these parameters.

References

[1] CCITT Manual *Transmission planning switched telephone networks,* ITU, Geneva, 1976.

[2] CCITT Recommendation *12-channel terminal equipments,* Vol. III, Rec. G.232.

ANNEX A

(to Recommendation G.171)

Digital private network loss plan/performance

A.1 *Introduction*

In the United States, the electronic industries association (EIA) is working on a proposed standard [1] for the port-to-port losses of a digital PBX. This contribution describes the loss plan and some network performance results which form the rationale for the plan. The intent of the contribution is only to provide background material to assist in the study of the question. In particular the information can help in developing an extension to Recommendation G.171, to cover the agreed upon first priority of work on a digital private network loss plan including interconnection to public networks.

A.2 *Digital private networks*

The loss plan for digital private networks is patterned after the predivestiture AT&T public switched digital network. The latter specifies a fixed 6 dB local exchange-to-local exchange loss for most connections [2]. Moreover, it was planned to function harmoniously with the existing extensive analogue network, such that hybrid connections would provide quality performance.

The end-to-end loss for digital private networks is 12 dB for connections terminated in on-premises stations (ONS) at both ends. The 6 dB difference between such connections and public switched digital networks connections was proposed to compensate for the difference bewtween the public network average subscriber line loss (approximately 4 dB) and a private network on-premises station average line loss (approximately 1 dB). Thus, 3 dB of the 12 dB end-to-end loss is assigned to each line, at each end of the connection (see Figure A-1/G.171).

When an off-premises station (OPS) is used instead of an ONS, then the 3 dB allocation to the subscriber line is dropped, because an OPS line has loss comparable to a regular subscriber line or is designed for via net loss (VNL) + 4 dB loss. Thus, if both ends of a digital private network connection terminate in OPS lines, the end-to-end loss will be 6 dB. If one end of the connection is terminated in OPS and the other in ONS, then the end-to-end connection loss will be 9 dB.

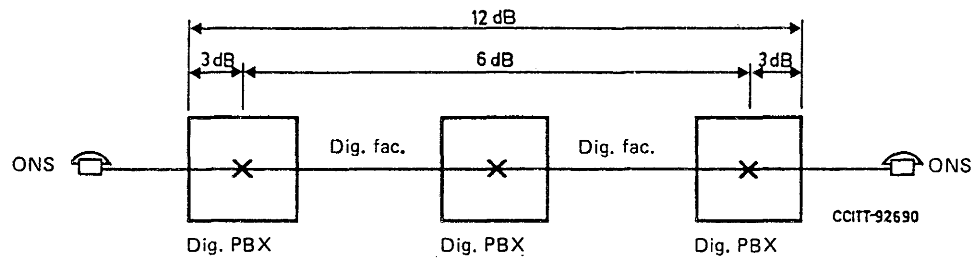


FIGURE A-1/G.171

Digital private network connection

A.3 *The EIA loss plan*

The EIA loss plan for digital PBXs was proposed to implement the loss plan of digital private networks. It also provides flexibility to digital PBXs to operate in a mixed analogue/digital environment, and to interconnect private and public networks.

The EIA loss plan for digital PBXs is presented in Table A-1/G.171. The Table shows the loss to be inserted by the PBX between various interfaces, in the two transmission directions. The method of implementation of loss (analogue, digital or both) is not specified. However, in an end-to-end digital connection all the loss is inserted at the terminating PBX, except for the 3 dB loss allocated to an originating ONS. As an example, to achieve the 12 dB loss value shown in Figure A-1/G.171, each end-PBX has to insert 3 dB loss in the transmit direction, and 9 dB loss (6 + 3) in the receive direction (this is specified by cell 1-D); but, the middle PBX should not insert any loss (cell 4-D).

As stated before, the Table also specifies loss values to interface with analogue private network facilities, and with the public network through analogue or digital facilities. Loss values are also specified for satellite PBX trunks. Satellite PBX trunks are in general short. Therefore, by specifying a separate set of loss values for interconnections with such trunks, it is possible to obtain better overall performance. Where analogue facilities are used, it is assumed that they are designed to VNL.

TABLE A-1/G.171

Digital PBX loss plan

	A ONS		B OPS		C A/TT		D D/TT		E S/ATT S/DTT	
	↑	↓	↑	↓	↑	↓	↑	↓	↑	↓
1 ONS	6	6	3	3	3	3	3	9	3	3
2 OPS	3	3	0	0	2	2	0	6	2	2
3 A/TT	3	3	2	2	0	0	−3	3	0	0
4 D/TT	9	3	6	0	3	−3	0	0	6	0
5 S/ATT S/DTT	3	3	2	2	0	0	0	6	0	0
6 A/CO	0	0	0	0	0/2	0/2 (Note 2)	−3/0	3/6 (Notes 1, 2)	0	0
7 D/CO	3	3	0	0	2	2	0/−3	6/3 (Note 3)	0	0
8 A/TO	6	6	3	3	0	0	−3	3	3	3
9 D/TO	9	3	6	0	3	−3	0	0	6	0

(Values in dB)

PBX Interfaces

ONS Line interface to on-premises line

OPS Line interface to off-premises line

A/TT Analogue trunk interface to tie trunk

D/TT Digital trunk interface to digital tie trunk, combination tie trunk or any other tie trunk with a digital termination at the muPBX

S/ATT Analogue trunk interface to analogue satellite PBX tie trunk

S/DTT Digital trunk interface to digital satellite PBX tie trunk

PBX Interfaces (cont.)

A/CO Analogue trunk interface to analogue Central Office (CO) trunk

D/CO Digital trunk interface to digital CO Trunk, combination CO trunk or any other CO trunk with a digital termination at the muPBX

A/TO Analogue trunk interface to analogue Toll Office (TO) trunk

D/TO Digital trunk interface to digital TO trunk, combination TO trunk, or any other TO trunk with a digital termination at the muPBX

Note 1 – The −3/3 dB value pair should be provided for connections between an A/CO port and a D/TT port serving as the interface to a combination tie trunk to a satellite PBX.

Note 2 – It is desirable that the low-loss option (0/0 or −3/3) be used when the muPBX-CO trunk loss is greater than or equal to 2 dB *and* the ERL ⩾ {18,13} *and* SRL ⩾ {10,6} measured into a 900 ohm +2.16 μF termination at the CO. [The notation {M,L} signifies that the median value is M and the lower limit is L.]

Note 3 – The 0/6 dB loss pair shall always be provided. The −3/3 dB loss pair is a desirable option to be used for internetwork applications in which no significant configuration will encounter echo, stability, or overload problems because of the reduced loss. With the −3/3 dB loss pair, subscriber station DTMF signals transmitted through the DCO into the private network might experience nonrecoverable digit mutilation in secondary signalling applications (DTMF signalling after the connection has been established, e.g., order entry) because of the 3 dB gain.

A.4 *Grade-of-service study results in support of the EIA PBX loss plan*

Since the loss plan for private digital networks is patterned after the AT&T public switched digital network loss plan, it is expected that the performance of private digital networks will be comparable to that of the public switched digital network. The end office-to-end office 6 dB loss for public switched digital connections is a compromise value. As Figure A-2/G.171 shows, the optimum loss value is a function of the connection length (which determines the round trip delay of reflected signals). But, the 6 dB compromise loss will provide quality performance for most connections. Equivalently, the Grade-of-Service (GOS) of a connection will depend on the end-to-end loss. The 12 dB end-to-end loss for digital private networks (or the 3 dB allocation to each ONS loop) is supported by the following GOS results.

Figure A-3b/G.171 shows the loss-noise-echo GOS[1] for private digital network connections with ONS at both ends, as a function of loss. Loss is presented in terms of the variable P value assigned to each ONS, in addition to a fixed end-to-end 6 dB loss (see Figure A-3a/G.171). Three connection lengths are considered: short, medium, and long, having lengths of 45 miles, 250 miles, and 1820 miles, respectively. The echo return loss used in this simulation has a mean value of 12 dB and a standard deviation of 3. As the results show, the optimal value of P increases with the length of the connection.

[1] GOS is shown in terms of the mean percent good-or-better rating using the Cavanaugh, Hatch, Sullivan model.

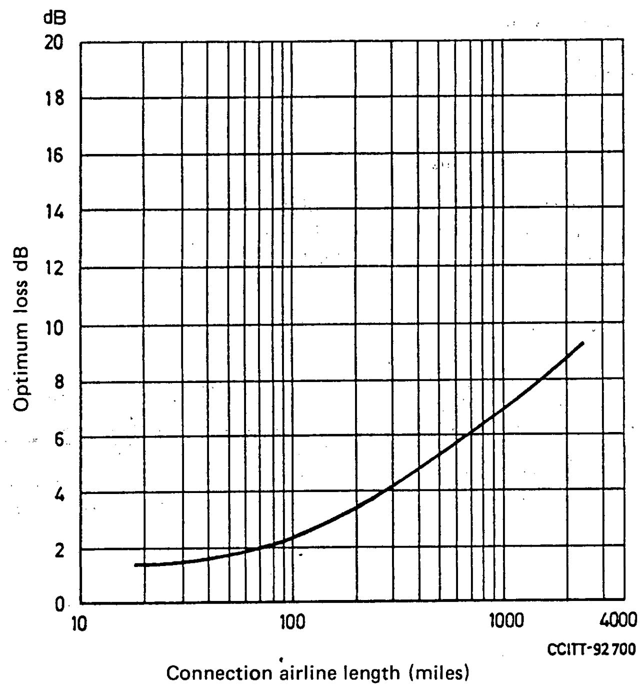


FIGURE A-2/G.171

Switched digital network optimum loss
(from Ref. 2)

Figure A-4a/G.171 is similar to Figure A-3a/G.171 except that one ONS is replaced with an OPS. The PBX on which the OPS homes, does not have a "P" loss associated with the OPS line. Also, the echo return loss parameters are different. Figure A-4b/G.171 shows the loss-noise-echo GOS results for the connection of Figure A-4a/G.171, for the three private network lengths, as perceived by the ONS customer (the GOS for the OPS customer will be different). The dependency of the GOS results on the value of P is similar to those of Figure A-3b/G.171. For P = 6 dB, the performance on the long connection (see Figure A-4b/G.171) will be close to optimal. However, the performance on the other connections will start to deteriorate. The deterioration will be significant when both ends of a connection terminate in ONS (see Figure A-3b/G.171). Since connections with ONSs are the prevailing type, a value of P = 3 dB provides the best compromise, and is the value selected.

The P = 3 dB value is also used when the interconnecting facilities are analogue and as the following example shows, it is a good compromise for interconnections of private and public networks. Figure A-5a/G.171 is such an example. Figure A-5b/G.171 displays the corresponding GOS results as perceived by the ONS customer.

A.5 *Conclusions*

A loss plan for digital private networks, which is implemented through the proposed EIA PBX loss plan, is discussed. It is shown through GOS results that the loss plan represents a good compromise and provides a high level of performance and flexibility for various connection types.

Simulated connections
Digital
ONS → ONS

Tie trunk lengths

Short	....	45 miles
Medium	....	250 miles
Long	....	1820 miles

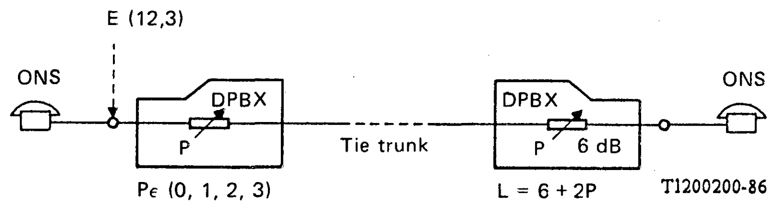


FIGURE A-3a/G.171

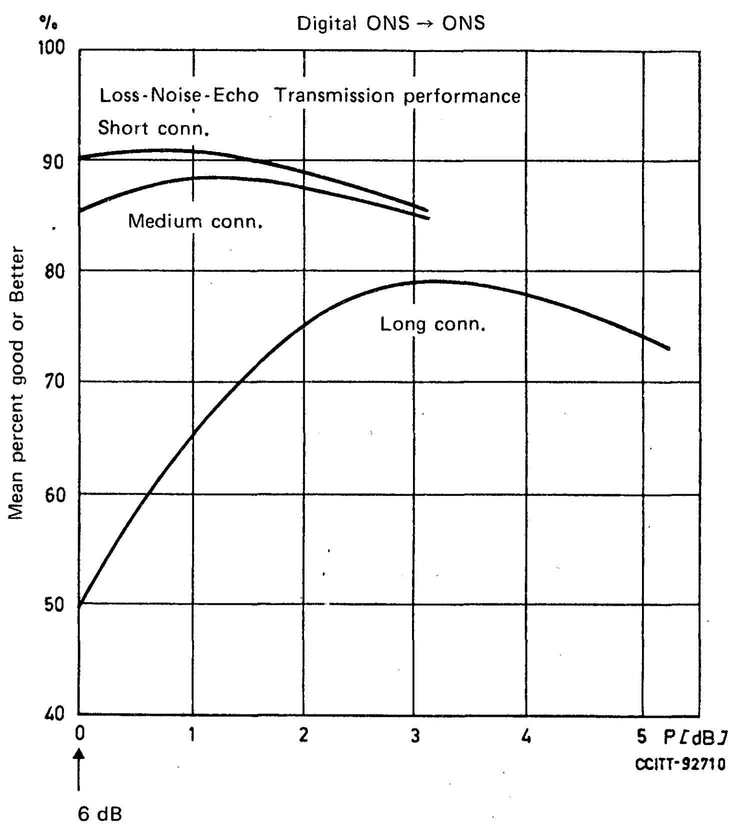


FIGURE A-3b/G.171

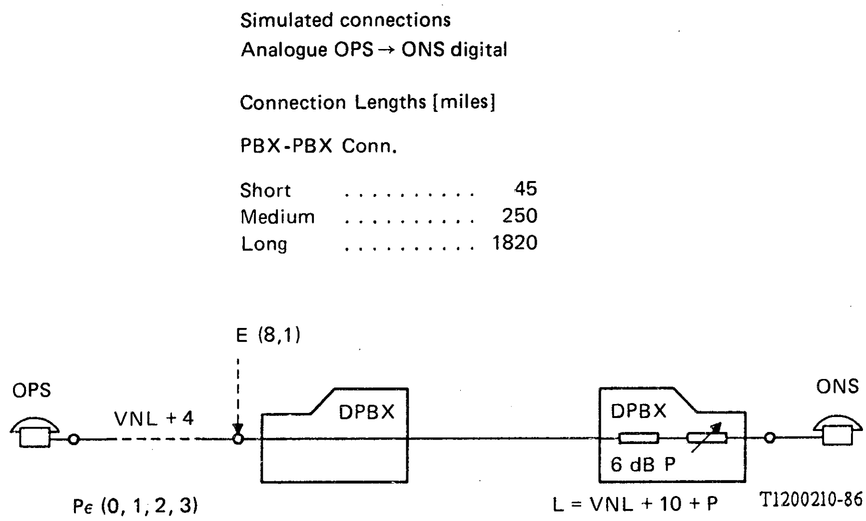


FIGURE A-4a/G.171

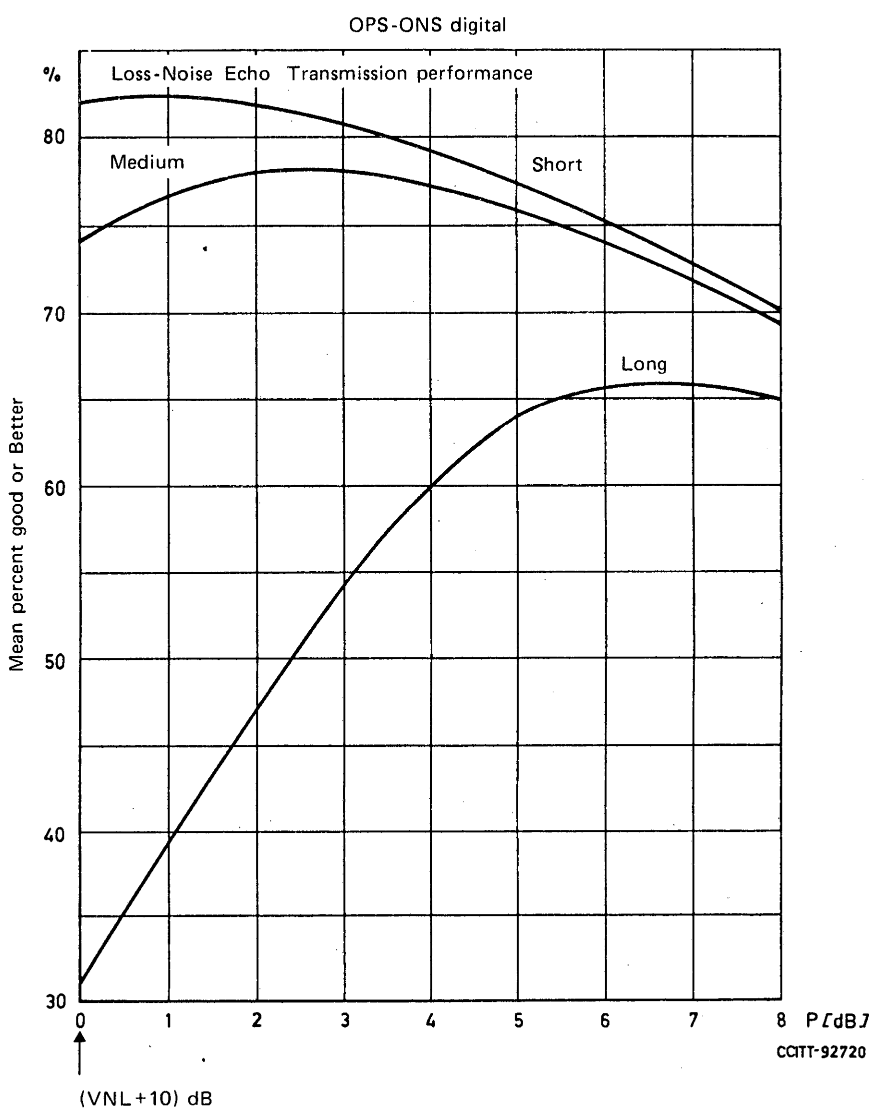


FIGURE A-4b/G.171

Simulated connections
Analogue
Public network → Private network

Connection Lengths [miles]

PBX-PBX conn.

Short	45
Medium	250
Long	1820

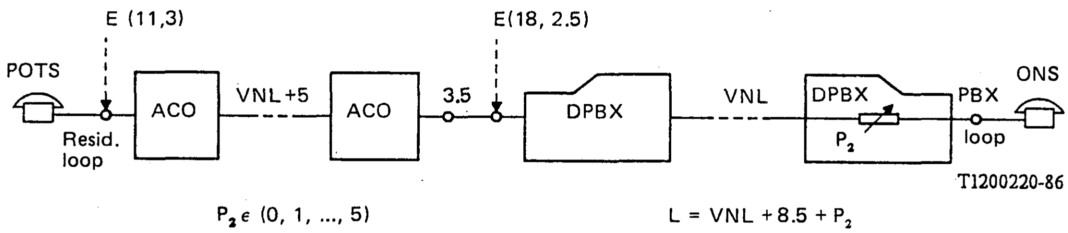


FIGURE A-5a/G.171

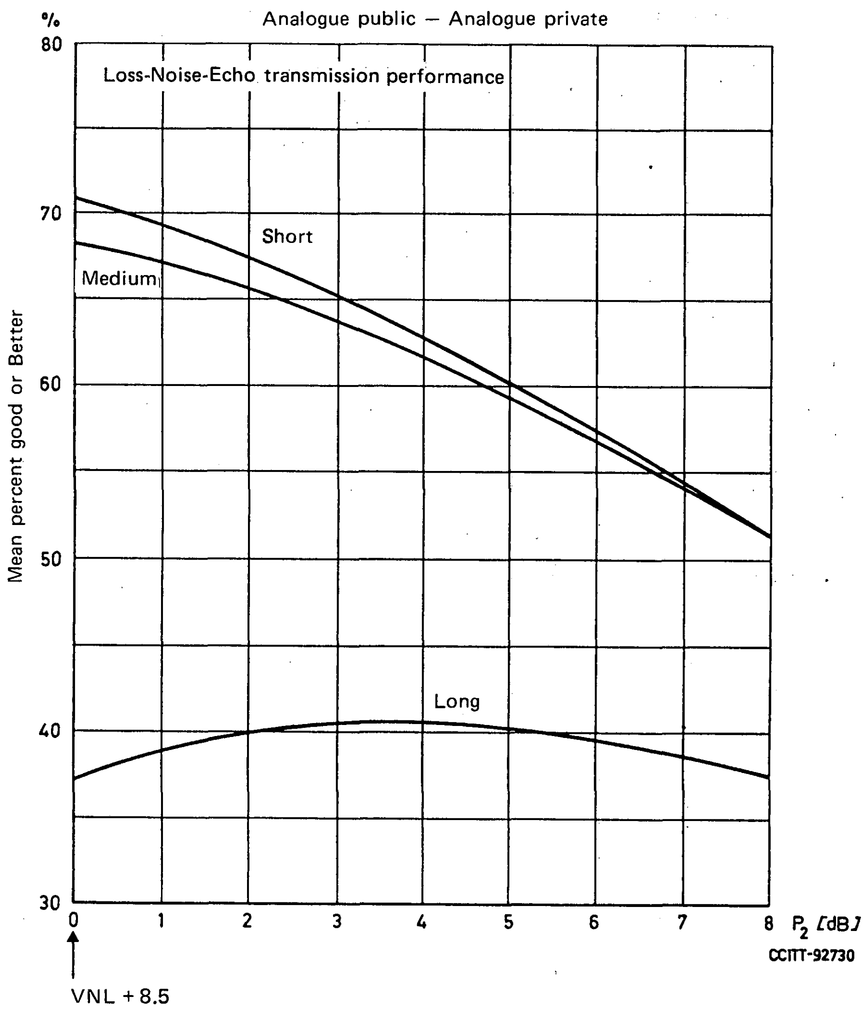


FIGURE A-5b/G.171

References

[1] EIA PN-1378, Private branch exchange (PBX) switching equipment for voiceband applications.

[2] AT&T, Notes on the network, 1980.

Recommendation G.172

TRANSMISSION PLAN ASPECTS OF INTERNATIONAL CONFERENCE CALLS

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

The transmission considerations given here are applicable to conference calls set up and operated in accordance with Recommendation E.151.

1 In order to respect CCITT Recommendations concerning loudness ratings on international connections, high quality bridging equipment shall be used. This equipment shall be designed to provide a nominal transmission loss of 0 dB in the direction from whichever participant is for the moment active (speaking) to all inactive (listening) participants. This loss shall be measured between equal level switching points of national circuits or virtual switching points of international circuits.

Note – Some conference bridges employ the use of automatic gain control (AGC) to minimize the contrast that exists between the speech levels of participants on connections having different losses, and the above consideration does not apply for such bridges. The transmission consideration for bridges with AGC is a subject for future study.

2 A modern conference bridge shall be used which employs techniques to avoid excessive transmission impairment from the accumulation of noise and echo at the output of the bridge in a multiport conference arrangement.

In a conference connection with two bridges: one bridge has N1 ports including a talker and the other bridge has N2 ports, noise increases as the number of ports is increased, according to the approximate rule: 10 log (N1 + N2 – 1).

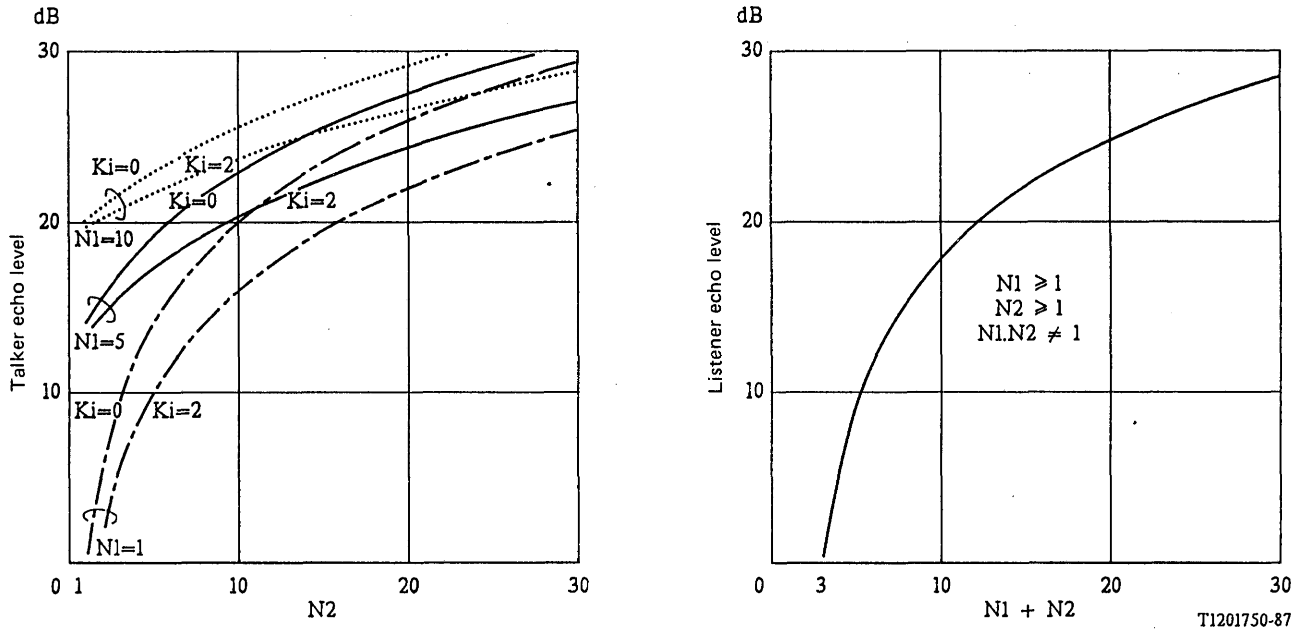
Talker and listener echoes also increase as the number of ports is increased as shown in Figure 1/G.172.

The multi-bridge configuration thus highlights the need for noise and echo control.

Note 1 – For example, a conference bridge which provides voice-activated switched loss or its equivalent may be used. In such a bridge, 15 dB of loss would be connected in each input to the bridge when the customer on that path is inactive. When a participant becomes active the loss is switched from his talking path to his listening path. This differential action protects the talker from echo and prevents a reduction of singing margin when the switch is operated. The loss which normally exists in the transmit path attenuates weak input signals such as noise before they enter the bridge. With this arrangement the level of the total signal reflected back to any active port will be the sum of the individual reflections from all other ports diminished by 30 dB.

This bridge can be equipped with about 30 ports.

Note 2 – A description of a conference bridge employing voice-activated switched loss is available in Annex 2 to Question 6/XVI in Volume III-3 of the *Green Book*. The transmission requirements contained in that annex could be used for the design of bridging equipment. Requirements for the design of bridging equipment using other techniques to control level contrast and noise and echo accumulation are the subject of future study.



a) Talker echo level

b) Listener echo level

Ki Transmission loss between bridges
N1 Number of #1 bridge's subscribers
N2 Number of #2 bridge's subscribers

FIGURE 1/G.172

Increased talker and listener echo levels in a conference with two bridges

3 Optimum operation of a conference bridge is obtained when its location is close to the center of the connection. This tends to equalize loss from the bridge to all conference locations on the connection, thus minimizing level contrast. Thus bridging equipment for international calls should be at high order transit centers.

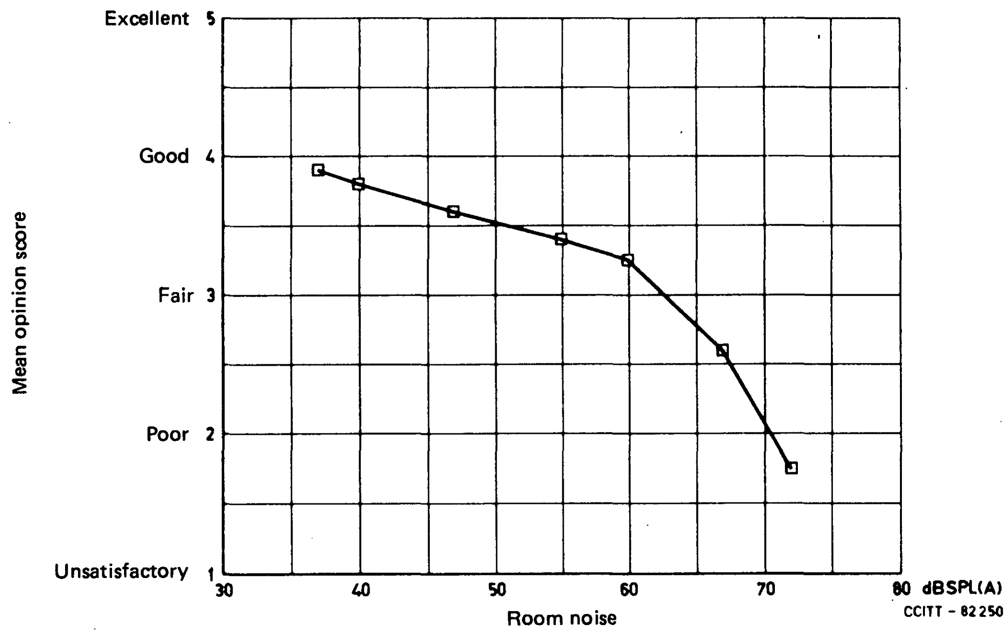
4 Bridging equipment should be 4-wire presented and 4-wire switched on both national and international circuits, wherever possible.

5 Attention is called to Recommendation G.114 concerning mean one-way propagation time which recommends that connections with delays in excess of 400 ms should not be used except under the most exceptional circumstances. To comply with this recommendation, care should be taken in the selection of a connection diagram so as to avoid the use of more than a single satellite circuit. For some conferences, using a single star network, this could influence the choice of location for the conference bridge. For other conferences, the use of a multiple star network could be selected with a single satellite circuit, equipped with appropriate echo suppressors, linking the conference bridges.

6 The conference connections should be carefully chosen so as to minimize the number of voice-activated switched loss devices in tandem to no more than two per conference leg. This includes customer premises conference equipment (such as loudspeaking telephones) and network equipment (such as echo suppressors), but excludes the bridging equipment.

7 Whenever the conference involves a single person at a location using a subscriber handset the room noise should be limited to about 60 dBSPL(A) [1)] at the user position to provide good quality transmission. Figure 2/G.172 shows the mean opinion score of transmission quality versus room noise [1]. Failure of the customer to comply with this guideline may cause the conference to be unacceptable.

8 When a conference involves more than a single person at each location it may be desirable to use conference rooms equipped with microphones and loudspeakers. To assure an adequate signal-to-noise ratio and freedom from the effects of conference room reverberation, the microphone and loudspeaker placement guidelines contained in Supplement No. 4, Volume V should be followed [2)].



Note – This curve represents the opinion of customers listening over the telephone in room noise from 37 dBSPL(A) to 72 dBSPL(A). Each point is an average over all values of speech level and circuit noise given at that room noise level.

FIGURE 2/G.172

Relationship of transmission quality to room noise

Reference

[1] *Guidelines for improving telephone communications in noisy room environments*, Bell System Technical Reference, PUB 42902, February 1980, American Telephone and Telegraph Co.

1) Sound pressure level relative to 20 μPa and using the A-weighting. See Recommendation P.54 for information concerning sound level measurements.

2) Another problem associated with hands-free conferencing is the likelihood of acoustic feedback between loudspeaker and microphone. While this feedback is today generally controlled using voice-activated switched loss in the conference room terminal equipment, note is taken of the fact that Study Group XV has proposed new studies to determine how to use echo cancellers to control the acoustic feedback.

1.8 Protection and restoration of transmission systems

Recommendation G.180

CHARACTERISTICS OF N + M TYPE DIRECT TRANSMISSION RESTORATION SYSTEMS FOR USE ON DIGITAL SECTIONS, LINKS OR EQUIPMENT

(Melbourne, 1988)

1 General

Transmission restoration functions are often implemented in the modern telecommunication networks to improve the availability and quality of service, by minimizing the effects or potential effects of a transmission failure, and to make the maintenance operations easier.

The terminology and general principles of transmission restoration are described in Recommendation M.495. The functional organization for automatic transmission restoration is described in Recommendation M.496.

2 Object of Recommendation

This Recommendation specifies the characteristics of equipment for N + M type direct transmission restoration systems (protection link switching) for digital transmission links (see Recommendation G.701). The general arrangement of a system for N + M direct transmission restoration is shown in Figure 1/G.180. This Recommendation refers to the equipment labelled as RSE (Restoration Switching Equipment) and RSCE (Restoration Switching Control Equipment).

This Recommendation is intended also to cover the case where the signals at the interfaces T belong to different hierarchical levels. In this case, each access at one side can be a group of accesses as indicated in the example of Figure 2/G.180. The left part of this figure refers to the particular case where the restored path is not on a complete link but just through a multiplex equipment.

Note – The equipment specified in this Recommendation can possibly be used also for N + M automatic or semi-automatic transmission rerouting (protection network switching) but generally this type of restoration function is implemented by different equipment, often incorporating also other functions (such as, for example, automatic digital distribution frames). This type of equipment is under study.

Three types of direct transmission restoration systems are considered by this Recommendation:

The first one should permit routing of any one of N normal links on to any one of M restoration links.

The second type should permit the interconnection of any of the N accesses to any one of the N + M links.

The third one should permit routing of any one of N normal links to a single restoration link (in many cases this type can be considered just a special case of the first type).

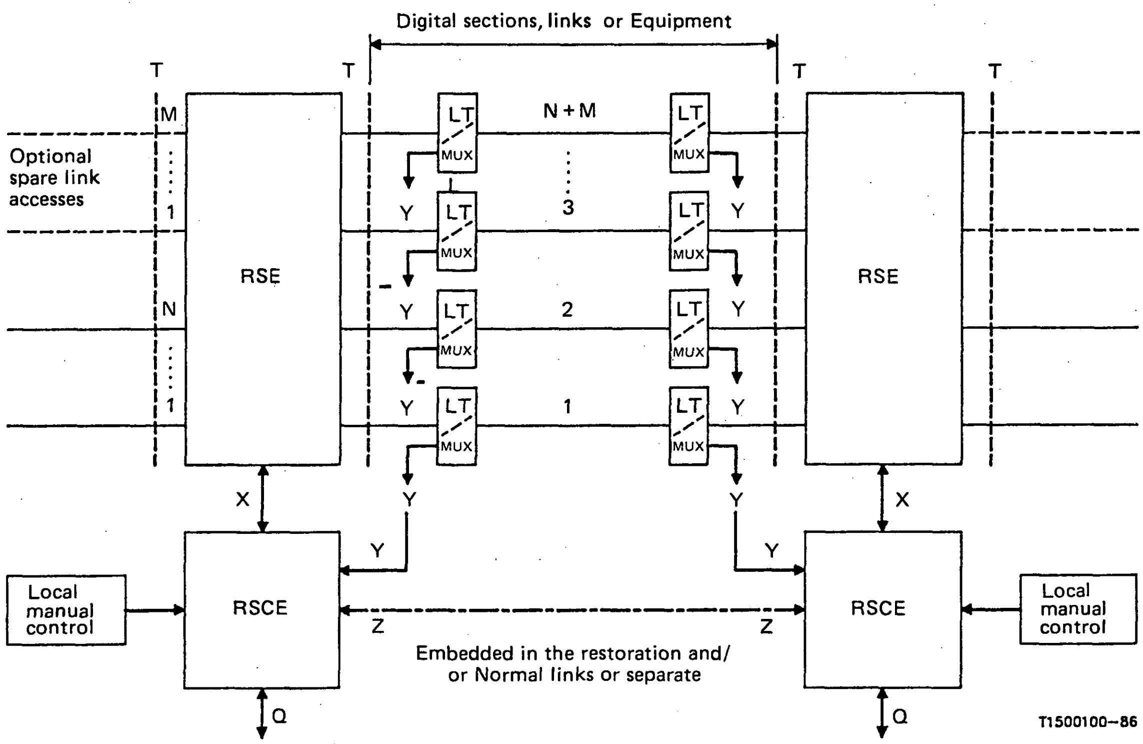
For all the types two options exist:

a) to switch the two directions of transmission independently; and

b) to switch the two directions of transmission simultaneously.

This Recommendation does not cover the restoration systems fully embedded in transmission systems and the 1 + 1 systems where the switching occurs at the receive end only (see Recommendation G.181).

The hierarchical levels at interfaces T are those specified in Recommendation G.702 (hierarchy levels 1 and up).



RSE Restoration switching equipment

RSCE Restoration switching control equipment

LT/MUX Line terminal or multiplex

-------- This link can be replaced by connection in the THN

X Non-standardized (or standard Q) interface. If RSE and RSCE are combined interface X does not exist.

Y Interface to be standardized in the future (possibly standard Q)

Z Interface possibly to be standardized

Q Optional standard Q interface to the TMN (telecommunication management network) or to a control centre

T Transmission path interfaces

Note 1 – The failing of a section or link can possibly be detected in the RSE and the information transferred to the RSCE. In this case interface Y may not exist.

Note 2 – The interworking with the TMN via interface Q is at present beyond the scope of the Recommendation.

Note 3 – The N + M digital links or sections can be in the same cable or follow different routes.

Note 4 – Spare link accesses may be used to convey low priority traffic.

Note 5 – The interface between the RSCE and local manual control is not standardized and it is not covered by this Recommendation.

FIGURE 1/G.180

N + M direct transmission restoration system

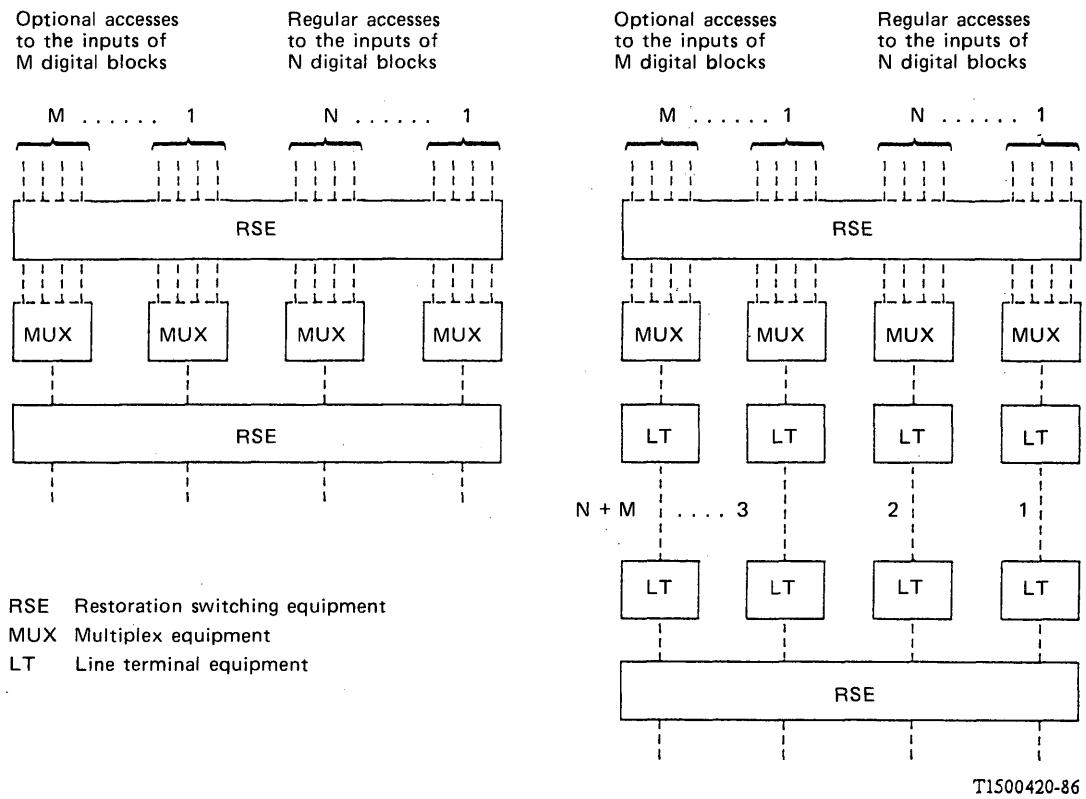


FIGURE 2/G.180

Examples of restoration systems where the hierarchical levels at the two ends are different

3 RSE specifications

Two types of RSE are considered by this Recommendation: the "regenerative" and the "non-regenerative" ones.

The first type, where the digital signal undergoes a complete process of retiming and reshaping, makes the RSE a digital equipment and it is sometimes considered to be advantageous, e.g. from the station cabling point of view.

The second type, where the output signal is proportional to the input signal (except for minor distortion) is considered to be useful in some circumstances, e.g. from a reliability and cost point of view.

3.1 *Interfaces*

3.1.1 *Transmission path interfaces* (T)

The relevant parameters and the recommended values are listed in the uppermost part of Table 1/G.180 for the non-regenerative RSE and in the uppermost part of Table 2/G.180 for the regenerative RSE.

TABLE 1/G.180

Provisionally recommended values for the interface and transfer characteristics of non-regenerative RSE

INTERFACE	Nominal impedance	As stated in Rec. G.703
	Return loss at the input port (with the output port terminated on the nominal impedance)	z dB above the values stated in Rec. G.703 (Note 1)
	Accepted levels	The output levels considered in Rec. G.703
TRANSFER	Transfer function between the input and the output of the RSE (terminated on the nominal impedances) (Note 2)	$< x$% of the interconnecting pair loss and phase distortion allowed in Rec. G. 703 for the relevant hierarchical level, or the complement thereof, plus or minus y dB flat loss. It is assumed that the loss vs frequency distortion approximates to the $\sqrt{f}$ law (Notes 3 and 4)
	Crosstalk attenuation	$> Y_1$ dB from any channel $> Y_2$ dB multi-channel interference evaluated on a voltage-sum basis. These values apply up to a frequency value equal to the nominal bit-rate (Note 5)

Note 1 – The value for z is under study. A possible value is $z = 6$ dB.

Note 2 – As a corresponding requirement is under study to permit the connection of test equipment in protected monitoring points, the relevant specification could alternatively be adopted.

Note 3 – The values of x and y are under study. A proposal states: $x = 10\%$ and $y = 0.5$ dB.

Note 4 – A delay limit will be also considered in future if benefit is expected from that.

Note 5 – The Y_1 and Y_2 limits are under study. $y_1 = 40$ and $y_2 = 30$ have been proposed as compromise values among different proposals. Different limits could possibly be adopted for RSE having a different configuration (e.g. N + 1 or N + M).

3.1.2 *Control interfaces*

The only control interface of the RSE is X. This interface is not at present specified by the CCITT. However in the future, it may be specified as a Q interface (see Recommendation G.771).

If the interface X is not standardized, the separation between the RSE and RSCE (and consequently between §§ 3 and 4 of this Recommendation) will be somewhat arbitrary.

3.2 *Operational aspects*

3.2.1 *Transfer of the switched signals*

The relevant parameters and the recommended values are listed in the lower part of Table 1/G.180 for the non-regenerative RSE and in the lower part of Table 2/G.180 for the regenerative RSE.

TABLE 2/G.180

Interface and transfer characteristics recommended for regenerative restoration switching equipment

General	Nominal bit rate and tolerance	As stated in Rec. G.703
INTERFACE	Connecting pairs Test impedance and return loss at the input ports Pulse shape and levels Tolerable input jitter	As stated in Rec. G.703
	Intrinsic output jitter	As stated in Supplement A to Table 2/G.180
TRANSFER	Jitter transfer	As stated in Supplement B to Table 2/G.180
	Error performance	99.99% error-free seconds (Note 3)
	Others (Note 1)	The paths across each switch shall maintain bit sequence independence and integrity (Note 2)

Note 1 – A delay limit will also be considered in future if benefit is expected from that.

Note 2 – Further study is necessary whether or not the digital signal should be replaced by a signal other than AIS in a restoration switching condition.

Note 3 – Evaluated under maximum loading condition and excluding any external source of interference.

3.2.2 *Response*

For RSE providing M restoration paths to N normal paths (M = 1 included) it is recommended that in response to a RSCE command the RSE should apply the incoming interface signal belonging to a given normal link to the input port of a given restoration link. The signal should not be removed from the input port of the concerned normal link, except that it may be replaced by a test signal.

For RSE providing N + M link to N accesses it is recommended that in response to a RSCE command, the SCE should apply the incoming interface signal belonging to a given access from 1 to N to a given link from 1 to N + M.

It is recommended that the time required for the above response actions, that is the "restoration transfer time", should be less than tx ms. The value for tx is under study.

Note – The characteristics necessary to specify the option of detecting in the RSE a failed path and to pass this information to the RSCE are under study.

3.2.3 *Other operational aspects*

A recognized failure of the RSCE or its disconnection from the RSE at interface X (when applicable) should either:

a) Cause the RSE to route all the signals on the N normal links. After the failure of the RSCE is cleared or the RSCE is reconnected to the RSE, normal restoration operations will resume.

b) Not alter the state of the RSE. The cross-connection pattern of the RSE should be available by interrogation from the RSCE to enable it to update, when the failure is cleared or it is reconnected to the RSE, its own record on the cross-connection pattern.

For the restoration systems of the second type (as defined in § 2 of this Recommendation) alternative b) only holds. For the systems of the first and third types both alternatives are applicable.

Note – The recommended behaviour of the RSE in case of own power failure is under study.

4 RSCE specifications

4.1 *Interfaces*

Interfaces Y, Z and Q of the RSCE (see Figure 1/G.180) are under study, including the bit rate and the tolerable bit error ratio for the Z interface.

4.2 *Operational aspects*

4.2.1 *Responses*

A switching to a restoration link should be initiated under a request coming from interfaces Y, Z, Q (and X where the faults are detected within the RSE) or on command from the local manual control.

When decided in the RSCE the allocation of a restoration link can optionally take place according to defined priority rules based on:

- defined priority for each normal link;
- request type (low or high priority request).

Otherwise the allocation should be specified by the information coming from interfaces Z, Q or local manual control.

For the restoration systems providing M restoration links (M = 1 included) on N normal links, when a successful restoration request clears, traffic should be returned to the pertinent normal link and the pertinent restoration link should be released.

It should be possible from interfaces Z, Q and under local manual control to lock in a working link (e.g. during system maintenance).

The time required for the above recommended restoration action is the sum of the "waiting time" and the "restoration procedure time". The two components should remain within the following limits:

- waiting time (under study);
- restoration procedure time (under study).

Note – Values to be recommended may be different for the three types of systems considered under § 2 and could depend on the interface over which the information is transferred. No precise proposed value exists at the moment. For a N + 1 system, one proposal indicates that the sum of the "restoration procedure time" and of the "restoration transfer time" should not exceed, in 90% of the occasions, 50 ms plus the time required for the communications.

4.2.2 *Alarm and status criteria*

Under study (see Appendix I to this Recommendation).

4.2.3 *Monitoring and self-test procedures*

Under study (see Appendix II to this Recommendation).

SUPPLEMENT A

(to Table 2/G.180)

Maximum permissible intrinsic jitter at output ports of regenerative restoration switching equipmemt

(Values for bit rates of the 1544 kbit/s digital hierarchy are under study)

For asynchronous space matrix RSE

Parameter value / Digital rate (kbit/s)	Maximum value Unit interval peak-peak	Measurement filter bandwidth: Bandpass filter having a lower cut-off frequency f_1 and an upper cut-off frequency f_4	
		f_1	f_4
2 048	0.1	20 Hz	100 kHz
8 448	0.1	20 Hz	400 kHz
34 368	0.075	100 Hz	800 kHz
139 264	0.05	200 Hz	3500 kHz

Parameter value / Digital rate (kbit/s)	Maximum value Unit interval peak-peak		Measurement filter bandwidth: Bandpass filter having a lower cut-off frequency f_1 or f_3 and an upper cut-off frequency f_4		
	B_1 ($f_1 \div f_4$ filter)	B_2 ($f_3 \div f_4$ filter)	f_1	f_3	f_4
2 048	0.25	0.05	20 Hz	18 kHz (700 Hz)	100 kHz
8 448	0.25	0.05	20 Hz	3 kHz (80 kHz)	400 kHz
34 368	0.35	0.05	100 Hz	10 kHz	800 kHz
139 264	under study	0.05	200 Hz	10 kHz	3500 kHz

Note 1 – UI Unit interval

for 2 048 kbit/s 1 UI 488 ns
for 8 448 kbit/s 1 UI 118 ns
for 34 368 kbit/s 1 UI 29.1 ns
for 139 264 kbit/s 1 UI 7.18 ns.

Note 2 – These figures shall be met for any valid signal in the absence of input jitter. The measurement shall be implemented using equipment designed in accordance with CCITT Recommendation O.171.

Note 3 – Recommendation G.823 § 2 indicates the measurement method.

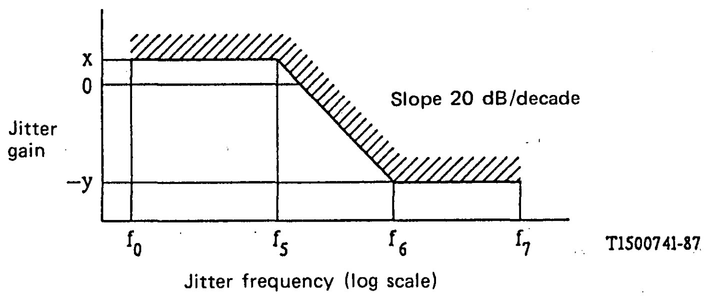
Note 4 – The frequency values in parentheses only apply to certain national interfaces.

SUPPLEMENT B

(to Table 2/G.180)

Jitter transfer characteristics recommended for regenerative restoration switching equipment

(Values for the bit rates of the 1544 kbit/s digital hierachy are under study.)



Parameter value / Digital rate (kbit/s)	x (dB) (Note 5)	$-y$ (dB)	f_0 (Hz)	f_5 (kHz)	f_6 (kHz)	f_7 (kHz)	Test signal (pseudo-random as Rec. O.151)
2 048	0.5	–8.4 (–9.5)	(Note 1)	36 (1.4)	100 (4.4)	100	$2^{15} - 1$
8 448	0.5	–9.5 (–7.5)	(Note 1)	6 (160)	19 (400)	400	$2^{15} - 1$
34 368	0.5	–9.5	(Note 1)	20	64	800	$2^{23} - 1$
139 264	0.5	–9.5	(Note 1)	20	64	3500	$2^{23} - 1$

Note 1 – The frequency f_0 should be as low as possible (e.g., 10 Hz) taking into account the limitations of measuring equipment.

Note 2 – The measuring method should be selective with a bandwidth sufficiently small referred to the relevant measuring frequency, but not wider than 40 Hz.

Note 3 – In the f_6 to f_7 frequency range the jitter gain should be less than y dB, with the exception of spurious responses, which should be suppressed below –6 dB.

Note 4 – The frequency values shown in parentheses only apply to certain national interfaces.

Note 5 – A value of 0.2 dB has been suggested as technically possible for this type of equipment. This may be useful where large numbers of RSE are employed in the network.

Parameter value / Digital rate (kbit/s)	x (dB)	$-y$ (dB)	f_0 (Hz)	f_5 (kHz)	f_6 (kHz)	f_7 (kHz)	Test signal (pseudo-random as Rec. O.151)
2 048	0.5	19.5	(Note 1)	40	400	100	$2^{15} - 1$
8 448	0.5	19.5	(Note 1)	100	1000	400	$2^{15} - 1$
34 368	0.5	19.5	(Note 1)	300	3000	800	$2^{23} - 1$
139 264	0.5	under study	(Note 1)	900	under study	3500	$2^{23} - 1$

Note 1 – The frequency f_0 should be as low as possible (e.g., 10 Hz) taking into account the limitations of measuring equipment.

Note 2 – The measuring method should be selective with a bandwidth sufficiently small referred to the relevant measuring frequency, but not wider than 40 Hz.

Note 3 – The need to tolerate spurious responses greater than y dB in the frequency range f_6 to f_7 is for further study.

APPENDIX I

(to Recommendation G.180)

Proposals for alarms and status criteria
(Both refer to a N + 1 system)

Proposed by STC PLC	Proposed by AT&T and Philips Telecommunications
Alarms It is proposed that the system should include: a) System fail. b) Protection failure. c) Manual switch in operation. d) System software self check in operation. e) Control system failure. f) System software failure. g) Communication failure. h) Stand-by channel failure. i) Power supply failure. j) Card removal.	Separate alarm criteria shall be issued at the occurrence of the following faulty conditions: a) Loss of signal at the traffic input port, transmit side. b) Loss of signal at the traffic output port, receive side. c) Automatic lock-in (see Note). d) Switch failure. e) Protection failure. f) Control system failure. g) Communication failure. h) Stand-by channel failure. i) Power supply unit failure. j) Loss of power supply. k) Switch exerciser failure. Separate status criteria shall be issued, on the occurrence of the following situations: a) Switch operated. b) Switch locked. c) Switch request pending. d) Switch in manual mode. The protective switching control equipment shall make available to the remote control and maintenance centre alarm and status information corresponding to the criteria shown above. *Note* – This system is required to automatically lock in the normal or the protection channel if an excessive number of switching operations are made in a given period.

APPENDIX II

(to Recommendation G.180)

Proposals for monitoring and self-test procedures
(Both refer to a N + 1 system)

Proposed STC PLC	*Standby Channel Monitoring* The system should include means of monitoring the standby channel continuously for proper operation. *Self-check* The system should include self-check facilities as follows: a) Communication channel. b) Background-checking of the memory, coaxial relay drive buffer and other hardware. c) Correct programme execution.
Proposed by AT&T and Philips Telecommun- cations	*Standby Channel Monitoring* The standby channel shall be monitored continuously for proper operation. *Switch Exerciser* The protective switching system shall provide a switch exerciser meeting the following requirements: The exerciser shall test the complete switch-over procedure up to but excluding the last transfer switch in the direction of transmission. The switching system shall drop the exerciser routine and serve switch requests from failed or deteriorated channels. A facility for including the last switch in the exercise routine may be provided. This feature shall have the capability of being disabled.

Recommendation G.181

CHARACTERISTICS OF 1 + 1 TYPE RESTORATION SYSTEMS FOR USE ON DIGITAL TRANSMISSION LINKS

(Melbourne, 1988)

1 General

Transmission restoration functions are often implemented in the modern telecommunication network to improve the availability and quality of services, by minimizing the effects or potential effects of a transmission failure, and to make the maintenance operations easier.

The terminology and general principles of transmission restoration are described in Recommendation M.495. The functional organization for automatic transmission restoration is described in Recommendation M.496.

2 Object of Recommendation

This Recommendation specifies the characteristics of equipment for 1 + 1 type transmission restoration systems (protection link switching) for digital transmission links, (see Recommendation G.701). The general arrangement of a system of this type is shown in Figure 1/G.181. It uses hybrid on the send side, splitting the input path into two output paths. On the receive side the two paths are supervised and are connected further by a switch automatically controlled by the received signals. The switch may additionally be operated manually or by some kind of remote procedure. The two transmission directions are handled independently.

This Recommendation refers to the equipment labelled as H (hybrid) RSE (restoration switching equipment) and RSCE (restoration switching control equipment).

This Recommendation does not cover the restoration systems fully embedded in transmission systems.

The hierarchical level at interface T is 2048 kbit/s. Other hierarchical levels are under study.

3 Equipment specifications

Equipment H and RSE (see Figure 1/G.181) may be of the regenerative or non-regenerative type.

3.1 *Interfaces*

3.1.1 *Transmission path interfaces* (T)

For H and RSE equipment of the regenerative type the interfaces shall be as specified in Recommendation G.703. The intrinsic output jitter should be not greater than 0.05 UI (measurement filter bandwidth: 20 Hz to 100 kHz).

For H and RSE equipment of the non-regenerative type the interface characteristics are under study.

3.1.2 *Control interface* (X)

The control interface X is not at present standardized by CCITT. However in the future it may be specified as a Q interface (see Recommendation G.771).

3.2 *Operational aspects*

3.2.1 *Transfer of the switched signals*

For H and RSE equipment of the regenerative type the jitter transfer gain should be not greater than 0.5 dB (the frequency limits are under study).

For H and RSE equipment of the non-regenerative type the transfer characteristics are under study.

3.2.2 *Response*

Switching between the two paths occurs only on the receive side, as indicated in Figure 1/G.181.

One of the two paths may be the path with the primary right, e.g. path II/II'. If this path fails the switch is operated to path III/III'. After restoration of path II/II' the switch will automatically be set back to this path.

If the two paths have the same right the switch will remain in the last position even after restoration of a failed path. This is the preferred method.

Note – Paths II/II' and III/III' have the same performance under normal planning conditions of transmission routes and systems. The "method of the same right" reduces the frequency of switching and resynchronization by a factor of 2.

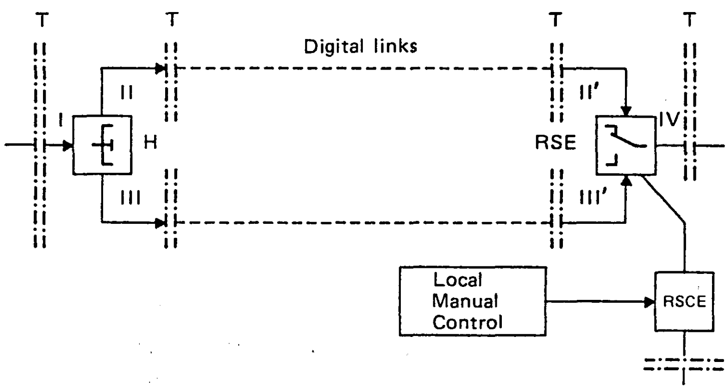
Switching to a failed path must be avoided.

The RSCE should operate the switch:

– automatically according to the criteria expressed by Tables 1/G.181 and 2/G.181, based on AIS reception and loss of incoming signal or (as an option) on transmission quality (see Note);
– manually on command from the local manual control;
– under a request coming from interface X.

Note – As an option the switching can be based on transmission quality as defined in Recommendation G.821 and by agreement between Administrations. In this case the transmitted signals need to have a standard frame structure in accordance with Recommendation G.704 which for 2048 kbit/s should also include the CRC4 option.

It is recommended that the time required for the above response actions, that is the "confirmation time" plus the "restoration transfer time" should be less than 10 ms for terrestrial routes and 500 ms for satellite routes.



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H Hybrid
RSE Restoration switching equipment
RSCE Restoration switching control equipment
T Transmission interface
X Control interface

Note 1 – The two digital links can be in the same cable or follow different routes.
Note 2 – The interface between the RSCE and the Local Manual Control is not standardized and it is not covered by this Recommendation.

FIGURE 1/G.181

1 + 1 transmission restoration system

TABLE 1/G.181

Response criteria for the hybrid H at the transmit side

Fault condition	Consequent action (signal at II and III)
No signal at I	AIS
AIS receive at I	AIS
Failure of power supply, system failure	AIS (if possible [a])

[a] The equipment may not be able to send AIS; this depends on the nature of the nature of the failure.

TABLE 2/G.181

Switching criterion for the RSE at the receive side

Condition	Consequence action	Remark	
Received signal II′ and III′	(See note)	Signal at IV	
Received signal at II′ AIS or no signal at III′	Switch to II′	Signal at IV	
Received signal at III′ AIS or no signal at II′	Switch to III′	Signal at IV	
AIS at II′ and III′	Switch to II′ or III′	Received AIS is through-connected	
No signal at II′ and III′	The switching equipment sends AIS at IV		
AIS at II′ and no signal at III′	Switch to II′	Received AIS is through-connected	
AIS at III′ and no signal at II′	Switch to III′		
Correct signal at II′ Bad quality at III′	Switch to II′	Correct signal at IV	Optional
Correct signal at III′ Bad quality at II′	Switch to III′	idem	
Bad quality at II′ AIS or no signal at III′	Switch to II′	Bad quality at IV	
Bad quality at III′ AIS or no signal at II′	Switch to III′	idem	
Bad quality at III′ Bad quality at II′	(See Note)	Bad quality at IV	
Failure of power supply system failure	The switching equipment sends AIS (if possible) at IV		

Note – Switch to paths II/II′ or III/III′ if both paths have the same right. Switch to the path with the primary right if the other method is used (see § 3.2.2).

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PART II

SUPPLEMENTS TO SECTION 1
OF THE SERIES G RECOMMENDATIONS

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Supplement No. 1

CALCULATION OF THE STABILITY OF INTERNATIONAL CONNECTIONS ESTABLISHED IN ACCORDANCE WITH THE TRANSMISSION AND SWITCHING PLAN

(Referred to in Recommendation G.131; this Supplement is to be found on page 555 of Volume III.2 of the *Green Book*, Geneva, 1973)

Supplement No. 2

TALKER ECHO ON INTERNATIONAL CONNECTIONS

(Geneva, 1964; amended at Mar del Plata, 1968 and Geneva, 1976 and 1980; Malaga-Torremolinos, 1984; and Melbourne, 1988; referred to in Recommendation G.131, § 2)

1 The curves of Figure 2/G.131 may be used to determine whether a given international connection requires an echo control device (echo suppressor or echo canceller). Alternatively they may be used to find what value of nominal overall loss shall be adopted for the 4-wire chain of a complete connection so that an echo control device is not needed. Before the curves can be used it must be decided what proportion of calls are to be allowed to exhibit an objectionable echo and Recommendation G.131 gives guidance on this matter.

The coordinates of the graph represent two of the parameters of a telephone connection that govern echo, i.e. the overall loudness rating (OLR) of the echo path and the mean one-way propagation time. By making certain assumptions (discussed below) these two parameters become the principal ones.

Each curve divides the coordinate plane into two portions and the position, relative to the curve, of the point describing the connection indicates whether an echo control device is needed, bearing in mind the percentage of calls permitted to exhibit an objectionable echo.

2 Factors governing echo

The principal factors which must be considered in order to describe whether an echo control device is needed on a particular connection are:

a) the number of echo paths;

b) the time taken by the echo currents to traverse these paths;

c) the OLRs of the echo path including the subscriber lines;

d) the tolerance to echo exhibited by subscribers.

These factors are discussed in turn in the following.

When circuits are switched together 4-wire there is only one echo path, assuming negligible go-to-return crosstalk. This is also substantially true if the circuits are switched together 2-wire and good echo return losses are achieved at these connection points (e.g. a mean value of 27 dB and a standard deviation of 3 dB). The principal echo currents are those due to the relatively poor echo balance return losses at the ends of the two extreme 4-wire circuits, where the connection is reduced to 2-wire.

The time taken to traverse the echo path is virtually dependent solely on the length of the 4-wire connection, because the main circuits of modern national and international networks are high-velocity circuits.

The OLR of the talker echo path for a symmetrical connection for planning purposes is approximately given by the sum of:

- twice the junction loudness rating (JLR) of the connection between the 2-wire point in the talker's local terminal exchange and the 2-wire side of the 4-wire/2-wire terminating set at the listener's end [1];
- the echo balance return loss at the listener's end;
- the sum of the sending LR and receiving LR of the talker's telephone and subscriber line;

In general, values of sending LR and receiving LR corresponding to low-loss subscriber lines should be used.

The echo experienced by subscribers on lines with more loss will be further attenuated. This is, therefore, a conservative assumption.

The data on tolerance to echo exhibited by subscribers given in Table 1 are furnished by the American Telephone and Telegraph Co. and are based on a series of studies completed in 1971. These tests provided information on the overall loudness rating (EARS) of the echo path for echo, just detectable, as a function of echo-path delay. In addition, ratings of quality on a five-point scale (excellent, good, fair, poor, unsatisfactory) were also obtained. The values in terms of EARS loudness ratings (then used by AT&T) were subsequently translated to values of CCITT loudness ratings by adding 1 dB. Table 1 indicates the mean echo path loss for the threshold of detectability and for ratings of unsatisfactory. These mean values are the loudness rating of the echo path for 50% detectability and 50% unsatisfactory. The standard deviation is also given.

TABLE 1

Results of echo tolerance tests

One-way propagation time (ms)	Overall loudness rating of the talker echo path			
	Threshold		Unsatisfactory	
	Mean (dB)	Standard deviation (dB)	Mean (dB)	Standard deviation (dB)
10	26	≃ 4	9	≃ 6
20	35	≃ 4	16	≃ 6
30	40	≃ 4	20	≃ 6
40	45	≃ 4	23	≃ 6
50	50	≃ 4	25	≃ 6
75	–	–	29	≃ 6
100	–	–	32	≃ 6
150	–	–	35	≃ 6
200	–	–	37	≃ 6
300	–	–	39	≃ 6

[1] According to Recommendation G.111, § A.3.3 the Junction Loudness Rating of 4-wire circuits should be taken as the 800 or 1000 Hz loss.

3 Construction of Figure 2/G.131

The mean margin against poor or unsatisfactory echo performance is given by:

$$M = 2T + B - E + \text{SLR} + \text{RLR}$$

where

T	is the mean junction loudness rating of the connection between the 2-wire point in the talker's local terminal exchange and the 2-wire side of the 4-wire/2-wire terminating set at the listener's end. The loudness rating is assumed to be the same in both directions of transmission;
B	is the mean echo balance return loss at the listener end;
E	is the mean value of loudness rating of the echo path required for an opinion rating of unsatisfactory [2];
SLR	is the sending loudness rating at the 2-wire point in the originating local exchange for short subscriber lines;
RLR	is the receiving loudness rating at the 2-wire point in the originating local exchange for short subscriber lines.

4 Fully analogue connections

The echo balance return loss is assumed to have a mean value of not less than 11 dB, with a standard deviation of 3 dB expressed as a weighted mean-power ratio (see Recommendation G.122). The mean value of the transmission loss is assumed to be uniform over this band and the standard deviation of transmission loss for each 4-wire circuit is assumed to be 1 dB for each direction of transmission. The correlation between the variations of loss of the two directions of transmission is assumed to be unity.

The standard deviation of the margin is given by:

$$m^2 = n({t^2}_1 + 2rt_1 t_2 + {t^2}_2) + b^2 + e^2$$

where

m	is the standard deviation of the margin;
t_1, t_2	are the standard deviation of the transmission loss in the two directions of transmission of one 4-wire circuit, national or international;
b	is the standard deviation of echo balance return loss;
e	is the standard deviation of the distribution of talker echo path loudness ratings required for opinion ratings of unsatisfactory;
r	is the correlation factor between t_1 and t_2;
n	is the the number of 4-wire circuits in the 4-wire chain.

Inserting $t_1 = t_2 = 1$ dB; $r = 1$; $b = 3$ dB; $e = 6$ dB gives $m^2 = (4n + 45)$.

In Recommendation G.131, § 2.3, Rules A and E refer to 1% and 10% probabilities of encountering unsatisfactory echo and for these cases nine 4-wire circuits are assumed (3 national and 3 international + 3 national). For both the 1% and 10% curves therefore $m = 9.0$ dB.

For 10% probability, the margin may fall to 1.28 times the standard deviation. The corresponding factor for the 1% curve is 2.33. Hence the corresponding values of M are:

$M = 1.28 \times 9.0$ dB $= 11.5$ dB for 10% probability
$M = 2.33 \times 9.0$ dB $= 21$ dB for 1% probability.

Putting these values into $M = 2T + B - E + \text{SLR} + \text{RLR}$ gives the following values for the mean talker echo attenuation, $2T + B + \text{SLR} + \text{RLR}$:

$2T + B + \text{SLR} + \text{RLR} = 11.5$ dB $+ E$ for 10% probability
$2T + B + \text{SLR} + \text{RLR} = 21$ dB $+ E$ for 1% probability.

[2] This corresponds to the value of overall loudness rating of the echo path at which 50% of the opinion ratings are unsatisfactory.

The values in Table 2 have been calculated (to the nearest whole decibel) using these equations. The figures in the length of connection column have been calculated assuming a velocity of propagation of 160 km/ms.

TABLE 2

Mean one-way propagation time (ms)	Length of connection (km)	Mean loudness rating of the talker echo path $2T + B + \text{SLR} + \text{RLR}$ (dB)	
		10% unsatisfactory	1% unsatisfactory
10	1 600	21	30
20	3 200	28	37
30	4 800	32	41
40	6 400	35	47
50	8 000	37	46
75	12 000	41	50
100	16 000	44	53
150	24 000	47	56
200	32 000	49	58
300	48 000	51	60

The solid last line for $n = g$ in Figure 2/G.131 has been constructed from these values and similar values calculated for other values of n (analogue circuits).

5 **Fully digital connections with analogue 2-wire subscribers lines** (conform to Figure 2/G.111)

The standard deviation of the margin is given by:

$$m^2 = n(2t^2) + b^2 + e^2$$

where

- m is the standard deviation of the margin;
- n is the number of coder/decoder pairs;
- t is the standard deviation of the transmission loss in the two directions of transmission;
- b is the standard deviation of the echo balance return loss;
- e is the standard deviation of the distribution of talker echo path loudness ratings required for opinion ratings of unsatisfactory.

The term $(2t^2)$ represents the loss variance of a coder/decoder pair, where $t = 0.2$ dB. For a fully digital connection between 2-wire analogue subscribers lines there are 2 coder/decoder pairs (i.e. one at the talker's local exchange and one at the listener's local exchange).

Inserting $n = 2$, $t = 0.2$ dB, $e = 6$ dB and assuming $b = 3$ dB gives $m^2 = 45.2$ and $m = 6.7$.

Hence the values of m are:

$m = 1.28 \times 6.7$ dB $= 8.6$ dB for 10% probability
$m = 2.33 \times 6.7$ dB $= 15.6$ dB for 1% probability.

Putting these values into $M = 2T + B - E + \text{SLR} + \text{RLR}$ gives the following values for the mean talker echo path attenuation, $2T + B + \text{SLR} + \text{RLR}$:

$2T + B + \text{SLR} + \text{RLR} = 8.6 \text{ dB} + E$ for 10% probability
$2T + B + \text{SLR} + \text{RLR} = 15.6 \text{ dB} + E$ for 1% probability.

The values in Table 3 have been calculated using these equations.

TABLE 3

Mean one-way propagation time (ms)	Mean loudness rating of the talker echo path $2T + B + \text{SLR} + \text{RLR}$	
	10% unsatisfactory (dB)	1% unsatisfactory (dB)
10	17.6	24.6
20	24.6	31.6
30	28.6	35.6
40	31.6	38.6
50	33.6	40.6
75	37.6	44.6
100	40.6	47.6
150	43.6	50.6
200	45.6	52.6
300	47.6	54.6

The dashed line in Figure 2/G.131 has been constructed from these values (fully digital connections).

6 **Fully digital connections with digital subscribers lines** (conform to Recommendation G.801)

In this case there are no 2-wire points in the connection. However, there is an acoustical feed-back path between the earpiece and mouthpiece of the telephone set. Therefore the echo balance return loss used above is now represented by the loss of this acoustical path. Representative values of this acoustical loss are under wider study. The appendix to this supplement gives some information on this question.

It may be assumed that the standard deviation of the transmission loss of the coder/decoder pair equals the value given above for digital connections with 2-wire subscriber lines. The value of the equivalent of T should be taken as zero. The quantities SLR and RLR now refer to virtual analogue 4-wire points of 0 dBr level.

If it can be assumed that the standard deviation of the acoustical echo path loss equals 3 dB and a normal distribution applies, then the values of Table 3 also apply to the digital subscriber line case and the dashed curve of Figure 2/G.131 may be used.

7 Mixed analogue/digital connections

This case is a combination of the cases given above and the appropriate variables and their values should be taken from the above information and an appropriate table can be constructed.

In general, if there are only two coder/decoder pairs in the connection, the variability of the transmission loss of the codecs may be ignored compared with the variability of the analogue circuits and the other variabilities. For such connections the solid curve given in Figure 2/G.131 for the number of analogue circuits in the connection may be used with good accuracy.

APPENDIX I

(to Supplement No. 2)

Echo loss in 4-wire telephone sets

(Contribution by Norway)

Abstract

In a 4-wire telephone set, echo may arise both by electrical crosstalk in the cord and by acoustical coupling between earpiece and mouthpiece in the handset. The echo loss for these paths has been determined for two analogue 2-wire telephone sets. This data is used to derive the echo loss of a hypothetical 4-wire set having SLR + RLR = 3 dB, and acoustical and electrical properties the same as the 2-wire telephone sets.

I.1 *Introduction*

It has been pointed out in several contributions that the choice of LRs for digital telephone sets has to be made considering aspects of loudness and echo in a complete 4-wire connection. To enable a study of the risk of objectionable echo, Study Group XVI has asked Study Group XII to present information on the subjective effect of talker echo as a function of delay, overall LR and echo path loss.

In a digital 4-wire connection, including 4-wire subscriber lines and digital telephone sets, the main echo paths are found in the telephone set itself:

- the acoustical coupling between earpiece and mouthpiece of the handset, and
- the electrical coupling in the flexible cord to the handset.

In order to assess the echo performance of a 4-wire connection, the echo loss of the digital telephone set must be estimated.

As an example of what can be expected, measurements of these echo paths have been made on two different analogue telephone sets. These results have been used to derive the echo loss between the receive and send terminals of a hypothetical 4-wire telephone set having SLR = 6 dB and RLR = −3 dB, and having the same electrical and acoustical properties as the measured sets.

I.2 *Measurements*

Figure I-1 shows a set-up for measurement of the loss between the receive and send direction in an ordinary 2-wire telephone. Two telephone sets are used to separate the two directions of transmission.

The acoustical path is measured by replacing the cord of the handset by shielded wires and the electrical path is measured by replacing the microphone by an appropriate resistor. When measuring the acoustic coupling, the handset was placed both in "free field" and held in a normal listening position.

Two different Norwegian standard telephone sets were included in the measurements. Both sets are equipped with linear microphones. EB model 67 has a "conventional" handset whereas Testafon is a "modern" set.

I.3 *Echo loss results*

In order to enable comparison of the data obtained for the two telephone sets, the measurements have been referred to a telephone set having SLR + RLR = 3 dB. The echo loss, as defined in Recommendation G.122, § 2.2, for this hypothetical telephone is shown in Table I-1.

The acoustical conditions refer to:

1) the handset held in a normal listening position, tightly against the ear ("real ear"), and
2) the handset held in "free field".

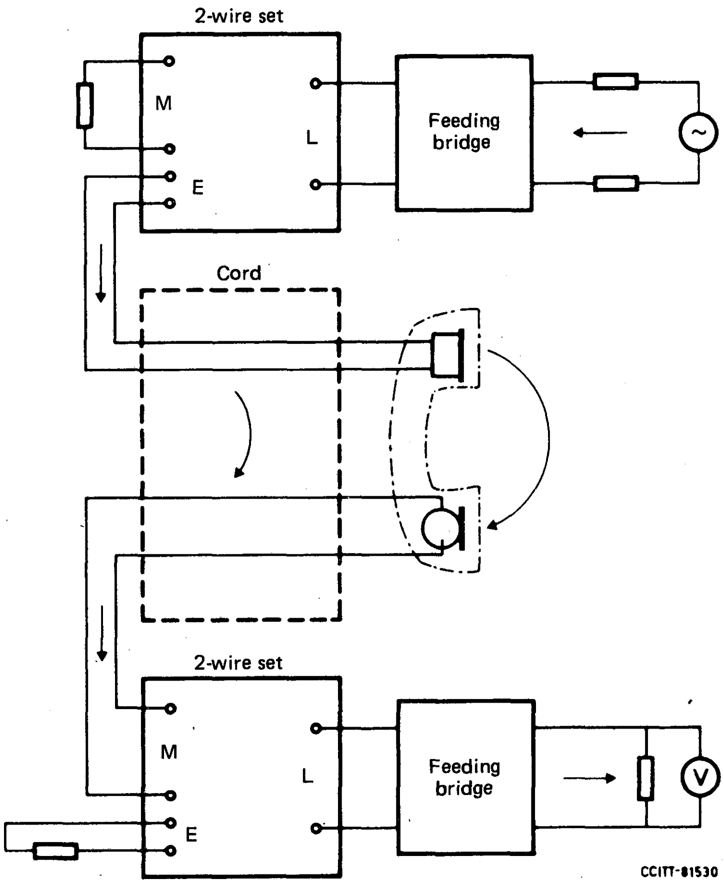

FIGURE I-1

Measurement set up

TABLE I-1

Echo loss in dB of hypothetical telephone set having SLR + RLR = 3 dB

Acoustical condition	EB model 67		Tastafon	
	Free field	Real ear	Free field	Real ear
Acoustical path	28.2	31.7	41.5	44.4
Electrical path	32.2	32.2	37.0	37.0

I.4 *Discussion*

It should be noted that high echo loss has not been design objective for either of the measured telephone sets. The results may therefore be considered as representative of what may be obtained when no special precautions are taken.

The echo loss of the acoustical path is apparently highly dependent on the physical design of the telephone handset and of the acoustical properties of the transducers. A difference of 13 dB is obtained in Table I-1 between the two sets in the test. The effect of acoustical termination of the earphone, i.e. "free field" or "real ear", is fairly small, approximately 3 dB for both sets.

Table I-1 shows that the electrical crosstalk in the flexible cord is an important echo source in both sets. For a given *SLR* and *RLR*, the level of crosstalk will depend on the partitioning of the gain between the handset (i.e. the microphone) and the telephone apparatus. Increasing the gain in the handset (by increasing the microphone sensitivity or by placing the microphone amplifier in the handset) will increase the signal level in the cord and improve the signal-to-crosstalk level. The crosstalk may also be reduced by using shielded wires. The electrical echo path may therefore be eliminated by proper design, and the acoustical component may be considered as the lower limit for the echo loss.

Supplement No. 3

EVALUATION OF ECHO CONTROL DEVICES

(Referred to in Recommendation G.161; this Supplement is to be found on page 559 of Volume III.2 of the *Green Book*, Geneva, 1973)

Supplement No. 10

APPLICATION OF RECOMMENDATION B.4 CONCERNING THE USE OF DECIBEL

(This Supplement is to be found on page 598 of Volume III.2 of the *Green Book*, Geneva, 1973)

Supplement No. 20

POSSIBLE COMBINATIONS OF BASIC TRANSMISSION IMPAIRMENTS IN HYPOTHETICAL REFERENCE CONNECTIONS

(Referred to in Recommendation G.103; this Supplement is to be found on page 319 of Fascicle III.1 of the *Red Book*, Geneva, 1985)

Supplement No. 21

THE USE OF QUANTIZING DISTORTION UNITS IN THE PLANNING OF INTERNATIONAL CONNECTIONS

(Contribution of Bell-Northern Research)

(Referred to in Recommendation G.113; this Supplement is to be found on page 326 of Fascicle III.1 of the *Red Book*, Geneva, 1985)

Supplement No. 24

CONSIDERATION CONCERNING QUANTIZING DISTORSION UNITS OF SOME DIGITAL DEVICES THAT PROCESS ENCODED SIGNALS

(Referred to in Recommendation G.113; this Supplement is to be found on page 333 of Fascicle III.1 of the *Red Book*, Geneva, 1985)

Supplement No. 25

GUIDELINES FOR PLACEMENT OF MICROPHONES AND LOUDSPEAKERS IN TELEPHONE CONFERENCE ROOM

(Referred to in Recommendation G.172; this Supplement is to be found on page 335 of Fascicle III.1 of the *Red Book*, Geneva, 1985)

Supplement No. 29

OBJECTIVE FOR THE MIXED ANALOGUE/DIGITAL CHAIN OF 4-WIRE CIRCUITS

Draft Recommendation G.136

(This Supplement is proposed for further study during the present study period with the aim to convert the supplement into a Recommendation.)

1 General

In the period of transition from a fully analogue to a fully digital network, there will be, on international and national networks, mixed type chain of 4-wire telephone circuits (see Recommendation G.101, § 4.2), some sections of which can be made with analogue or digital transmission systems.

Considering the fact that the transition period may last for a fairly prolonged time, and also considering the need for guaranteeing a certain quality of transmission on mixed chain of circuits, the CCITT recommends observance of some principles for the composition of mixed chain of circuits as set forth below and some objectives for their parameters.

The main principle in the standardization of mixed circuits lies in the retaining of the standards adopted for the FDM circuits. This would have resulted in retaining the transmission quality over the 4-wire chain formed by the international circuits and national extension circuits.

For some parameters this can be achieved, but as far as some other parameters are concerned due to analogue/digital conversions and errors in digital sections there are some considerable differences in standards and measuring methods.

Objectives for some mixed circuit parameters are contained in a number of G-, Q-, and M-series Recommendations. However, these objectives do not take due account of the addition laws for distortions based on the multitude of mixed circuit structures and specific features of the measuring methods involved.

Considering the importance of retaining the transmission quality during the transition period and attaching great importance to the standardization of mixed analogue/digital circuits the multitudinous types of which emerge while using various kinds of analogue-to-digital conversions, CCITT thinks it worth while to have a specific Recommendation on objectives for mixed analogue/digital circuits and 4-wire chains including both analogue and digital circuits.

The present Recommendation related to mixed 4-wire chain of circuits and the analogue/digital mixed connections dealt with in this Recommendation are those with analogue telephone sets at both ends.

It is based on the existing Recommendations for FDM channel equipment G.232, for PCM channel equipment G.712, for analogue switching centres Q.45, Q.45 *bis*, for digital switching centres Q.551 to Q.554, and takes account of other existing Recommendations of G- and M-series.

Later on in accordance with the study results of Question 26/XII the present Recommendation will have to be supplimented by objectives for mixed chain of circuits formed with the help of various methods of analogue-to-digital conversion such as transmultiplexers (Recommendations G.793, G.794), modems (Recommendations G.941, V.37), transcoders (Recommendation G.761), group codecs (Recommendation G.795), DCME, as well as connections with a digital telephone at one end and an analogue telephone at the other end.

2 Structure of a mixed analogue/digital voice frequence chain of 4-wire circuit

The parameters of a mixed 4-wire chain are essentially dependent on the number of analogue sections and on the number of analogue/digital conversions in the chain.

According to Recommendation G.103 the total number of 4-wire circuits in a 4-wire chain of the maximum length is 12 in exceptional cases (Table 2/G.101) so that it may be assumed that the number of circuits will not exceed 12. The worst cases in terms of distortions occur when:

- all switching centres are digital, and the circuit sections from and to the centres are set up on analogue transmission systems. The number of analogue/digital conversions is then 11, the number of analogue sections is 12;
- all switching centres are analogue, and the circuit sections from and to the centres are set up on digital systems. The number of analogue/digital conversions is 12 in this case, the number of digital sections is 12.

Such cases are very rare. More representative is considered to be a case where the number of analogue/digital conversions makes one half of the maximum number (Recommendation G.103, Annex B), that is 6, and digital islands are available. The structure of such a 4-wire chain is presented in Figure 1. The number of analogue sections is 6, the number of digital sections is also 6. Other structures of mixed 4-wire chain come into the picture when connection of the sections is realized without a switching equipment. These structures are considered in Recommendation M.562 (§ 3.2). The worst case for a circuit of 12 sections without switching centres occurs when digital and analogue sections alternate (see Figure 2), the number of analogue-digital conversions being equal to 6, the number of digital sections to 6, and the number of analogue sections also to 6.

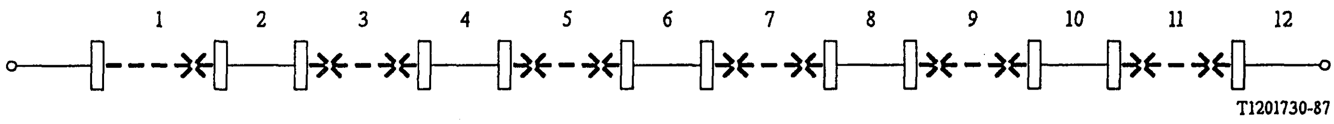

FIGURE 1

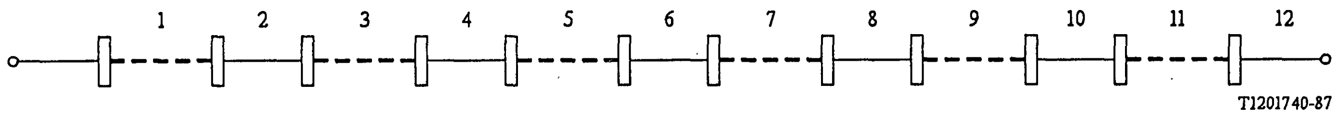

FIGURE 2

Thus, the examination of various structures of mixed analogue/digital voice-frequency chain of circuits shows that for a 4-wire chain of maximum length having 12 sections, it is advisable to establish objectives of distortions based on 6 analogue/digital conversions, 6 analogue and 6 digital sections.

Intermediate variants for combinations of analogue, digital sections and analogue-to-digital conversions will be:

11 analogue sections + 1 a/d conversion
(1 digital section) = 12

. . . .

6 analogue sections + 6 a/d conversions
(6 digital sections) = 12

It should be borne in mind that the chains may most frequently consist of less than 12 sections. The contribution of switching centres to distortion is negligible, if they do not contain analogue/digital conversions.

3 Objectives for parameters of mixed analogue/digital circuits

3.1 The nominal value of the input/output impedance of the analogue and digital sections and of a switching equipment should be 600 ohms.

3.2 Return loss of the input/output impedance referred to the nominal value of the analogue and digital sections and of a switching equipment should preferably be not less than 20 dB in the 300-3400 Hz band.

Note – For a switching centre and channel FDM equipment, the value of 15 dB is permissible in the 300-600 Hz band (see Recommendation Q.45, § 6.3 and Recommendation G.232, § 7).

3.3 *Unbalance loss in respect to earth*

The existing Recommendations for switching centres (Q.45, Q.553) and channel FDM equipment (G.712) standardize the unbalance loss in respect to the earth in different ways. There are differences in the measuring methods as well. The Recommendation for the FDM-channel equipment (G.232), does not specify this parameter. The question of standardization and methods of measuring this parameter for mixed circuits channels is under study.

Pending the establishment of unified objectives and measuring methods, Recommendation K.10 on the unbalance loss of communication equipment should be referred to in general guidelines in the case of mixed chain of 4-wire circuits.

3.4 *Nominal relative level*

The nominal relative level on the transmit side of each section (analogue and digital) is -14 (-16) dBr. The nominal relative level on the receive side of each section (analogue and digital) is $+4$ ($+7$) dBr (see Recommendations G.232, § 11, G.712, § 14, Q.45, § 3 and Q.553 § 2.2)

The nominal relative level at the virtual analogue switching point is

– sending: -3.5 dBr

– receiving: -4.0 dBr for analogue
-3.5 dBr for digital

(See Recommendation G.101, § 5.2.)

The nominal relative value in a mixed circuit is defined for a frequency which is not a subharmonic of the sampling frequency. The recommended tentative value for the frequency is 1020 Hz.

3.5 *Variations of transmission loss with time*

The standard deviation of the transmission loss should not exceed 1 dB.

The difference between mean and nominal value of the transmission loss should not exceed 0.5 dB.

Note – The indicated values are defined in Recommendation G.151, § 3 for a fully analogue circuit under the condition that the channels are part of a single group equipped with automatic regulation.

For mixed chains the stability conditions improve on the one hand because of the existence of digital sections which have a higher stability than analogue ones; but on the other hand in the mixed circuits there is no possibility of a transit automatic regulation of analogue sections, which deteriorates the overall stability. That is why the indicated values should be considered as tentative and are to be confirmed.

3.6 *Attenuation/frequency distortion*

Attenuation/frequency distortion for the whole 4-wire chain should not exceed the values given in Figure 1/G.132.

For mixed chains (without consideration of switching centre distortions) the accumulation law of attenuation/frequency distortions is expressed by the following formula:

$$\Delta a = n_1 \bar{a}_{\mathrm{FDM}} + \sum_{i=1}^{n_2} a_{i\,\mathrm{PCM}} \pm \mathrm{K}\sqrt{\sigma^2_{\mathrm{FDM}} \cdot n_1} \qquad (1)$$

with

n_1 : number of analogue sections;

n_2 : number of analogue/digital conversions;

$\bar{a}_{\mathrm{FDM}}$: average value (determined component) of attenuation/frequency distortions of the analogue sections;

σ_{FDM} : r.m.s. deviation of attenuation/frequency distortions of analogue sections;

a_{PCM} : attenuation/frequency characteristics of analogue/digital equipment;

K = 1, 2 or 3: factor defining the probability of maximum/minimum value of attenuation/frequency distortions.

"K" is usually taken as equal to 3. The justification of the choice for K = 3 depending on a given probability can be found in [1, 2].

Note 1 – Attenuation/frequency characteristics of analogue/digital equipment of the same type are similar. That is why, if in a mixed/chain of circuits analogue/digital equipment of the same type is used, in the sum formula (1)

$$\sum_{i=1}^{n_2} a_{i\,\mathrm{PCM}}$$

can be replaced by a product $n_2\ a_{\mathrm{PCM}}$.

Note 2 – The analogue-digital equipment distortion limits recommended in Recommendation G.712 (§ 1, Figure 1) and the FDM-channel equipment distortion limits recommended in Recommendation G.232 (§ 1, Figure 1) meet the limits indicated in Recommendation G.132 for mixed circuits in which the number of sections does not exceed 4.

When composing mixed chains with a greater number of sections, it is advisable to utilize modern channel equipment the attenuation/frequency distortions of which are considerably lower than those indicated in Recommendations G.232 and G.712.

Note 3 – Attenuation/frequency distortions are measured relative to the reference frequency of 1020 (1000) Hz.

Note 4 – See Recommendation Q.45 (§ 3.4 and Q.553) to take account of the switching equipment distortions.

3.7 *Group delay distortions*

Group delay distortions should not exceed the values indicated in Recommendation G.133 for the 4-wire chain.

The law of imposition of group delay distortions is expressed by the following formula:

$$\Delta\tau = n_1\ \tau_{\mathrm{FDM}} + \sum_{i=1}^{n_2} \tau_{i\,\mathrm{PCM}} \qquad (2)$$

where

n_1 the number of analogue sections,

n_2 the number of analogue/digital conversions.

Note 1 – If, in a mixed chain, analogue/digital equipment of the same type is used, then the sum

$$\sum_{i=1}^{n_2} \tau_{i\,\mathrm{PCM}}$$

is substituted by a product $n_2 \cdot \tau_{\mathrm{PCM}}$.

Note 2 – It is expected that the group delay distortion in mixed chains will be less than that of a fully analogue link for any combination of analogue and digital sections. But nevertheless the characteristics of distortion (symmetry) can change considerably. This should be taken into account when transmitting data on mixed circuits containing group delay equalizers.

Note 3 – Group delay distortions are measured with reference to a frequency situated at the lower band end of the analogue channel, i.e. 190-200 Hz.

Note 4 – Switching centre distortions are negligible and can be ignored.

3.8 *Intelligible crosstalk*

Near-end and far-end signal-to-intelligible crosstalk ratios between circuits and between send and receive directions should satisfy Recommendation G.151 (§ 4).

Note 1 – It is expected that the values indicated in Recommendation G.151, will be maintained and even better for mixed chains for any combination of analogue and digital sections, due to higher values achieved in the analogue/digital conversion equipment.

Note 2 – Measurement of the signal-to-crosstalk ratio between circuits can be performed without feeding an auxiliary signal into a channel affected by crosstalk (unlike that provided for in the note to point 11 of Recommendation G.712). This can be explained by the fact that in a mixed circuit, as a rule, and in an analogue circuit noise will be present at the input of analogue/digital converters in a mixed chain.

3.9 *Non-linear distortions*

The existing Recommendations for analogue circuits (M.1020, § 2.11), for switching equipment (Q.45, § 6.1) and Recommendation G.712 for analogue/digital equipment contain different specifications for non-linear distortions, the methods of their measurement differ too. The Recommendations for digital centres (Q.551 to Q.554) do not contain specifications for non-linear distortions.

At present it is not possible to recommend permissible values of non-linear distortions and a method for measuring mixed chains of circuits. This question needs to be studied.

3.10 *Noise (total distortions)*

The notion of noise in mixed chains of circuits due to analogue-to-digital conversions producing quantization distortions which accompany the signal has lost its initial meaning and therefore instead of the term "noise" applicable to mixed chain of circuits the term "total distortions" is used very often. This is stipulated by the fact that the measurement of quantization distortions (Recommendation Q.132) includes part of non-linear distortions and single-frequency interferences.

From this view point the total distortions in mixed chains include analogue section noise which depends on the length of the sections in case of terrestrial transmission systems and on the quantization distortion which are determined by the number and type of analogue-to-digital conversions.

The addition law of total distortions is expressed by the following formula:

$$P = 10 \log_{10} \left\{ 10^{-9} \cdot W_{FDM} + 10^{0.1 \left[S - \left((S/N) - 10 \log \eta_2 \, qdu \right) \right]} \right\} \tag{3}$$

where

– W_{FDM} noise power of analogue sections (pWp0)

– $W_{FDM} = W_o \dfrac{pWp0}{km} L \text{ km}$

(for a section provided by a satellite the terrestrial length is taken to be equal to 2500 km).

– S/N signal-to-quantization distortion ratio of one analogue-to-digital conversion.

– η_2 qdu total number of quantization distortion units of analogue-to-digital conversions.

To determine S/N and the total number of qdu's one should refer to Recommendation G.113.

– S signal level at which general distortions are measured.

To eliminate any effect of non-linear distortion the value of S should be no more than –10 dBm0.

The permissible value of P is to be determined in the studies in Study Group XII.

The value of –36 dBm0 (with S = –10 dBm0), i.e. signal-to-total distortions ratio 26 dB, can be indicated as a premilinary value.

The noise in an idle channel should comply with Recommendations G.123 and G.153, § 1.

Note 1 – Total distortions also include a component determined by errors in digital sections. It is assumed that if BER at each digital section is 10^{-6} (with the bit rate of 64 kbit/s) the respective component can be omitted.

Note 2 – The values of total distortions for various length of analogue sections and various numbers of qdu's mixed chains are available in Tables 5/M.580 and 6/M.580 of Annex A to this Recommendation.

3.11 *Single tone interference*

The level of any single tone signal should not exceed −73 dBm0 (see Recommendation G.151, § 8). The indicated value does not relate to the interfering signal at the sampling frequency.

The level of the interference at the sampling frequency should not exceed the value of $-50 + 10 \log n_2$ where n_2 is the number of analogue/digital conversions in a mixed circuit. The indicated value is tentative and needs to be confirmed by study results in Study Group XII.

3.12 *Products of unwanted modulation*

Product levels of unwanted modulation caused by power sources should not exceed −45 dB (see Recommendation G.151, § 7).

3.13 *Impulse noise*

Impulse noise is specified for analogue circuits used for data transmission (Recommendations M.1020 and M.1025) and for switching equipment (Recommendation Q.45, § 5.2 and Q.553). For voice-frequency circuits in PCM transmission systems the impulsive noise is not specified because it is supposed that it should not be there at all. In practice, it has been noticed, however, that with accumulation of errors, impulse noise can appear in a voice-frequency circuit which leads to interference in the transmisison of data signals. (Preliminary results on the effect of digital link errors on impulse noise in idle PCM voice-frequency channels is given in [4].)

The effect of impulsive noise appearing in digital sections on the overall value of interference in a mixed 4-wire chain is subject of study.

3.14 *Short-time interruptions, phase jitter, amplitude and phase hits*

These parameters strongly influence data transmission. For analogue circuits they are specified in Recommendations M.1020, M.1060 and M.910. For voice-frequency circuits set up on PCM systems, objectives are not available. It can be tentatively presumed that in mixed chains of circuits the presence of digital sections does not have a considerable effect. However, the question needs to be studied.

3.15 *Error performance*

Further study.

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[3] COM XII-19 (period 1985-1988), USSR Attenuation/frequency distortions and delay distortions of mixed audiofrequency analogue/digital circuits.

[4] COM XII-188 (period 1985-1988), USSR Interrelation between errors of a digital line and impulse noise in voice-frequency channels of the PCM System.

ANNEX A

(to draft Recommendation G.136)

TABLE 5/M.580

Signal-to-total distortion ratio for public telephone circuit maintenance using a test frequency level of − 10 dBm0

Type of circuit	Number of QDUs (Note 1)	Unit	Distance in analogue transmission (Note 3) (km)						
			< 320	321 to 640	641 to 1600	1601 to 2500	2501 to 5000	5001 to 10000	10 001 to 20 000
Analogue	0 (Note 2)	dB	45	43	41	39	36	33	30
Composite circuit	0.5	dB	35	35	34	34	33	31	29
	1	dB	33	33	32	32	31	30	28
	2	dB	30	30	30	29	29	28	27
	3	dB	28	28	28	28	28	27	26
	3.5	dB	27	27	27	27	27	26	26
	4	dB	27	27	27	27	26	26	25

Note 1 — The number of QDUs contributed by various processes are given in Table 1/G.113 [8].

Note 2 — The values are idle noise terminated with a nominal impedance of 600 Ω.

Note 3 — The section of the circuit provided by satellite (between earth stations), employing FDM techniques, contributes approximately 10 000 pWp (−50 dBm0p) of noise. Therefore, for the purpose of determining the total distortion limits for international public telephony circuits, the length of this section may be considered, from Table 4/M.580, to be equivalent to 2500 km.

TABLE 6/M.580

Signal-to-total distortion ratio for public telephone circuit maintenance using a test frequency level of −25 dBm0

Type of circuit	Number of QDUs (Note 1)	Unit	Distance in analogue transmission (Note 3) (km)						
			< 320	321 to 640	641 to 1600	1601 to 2500	2501 to 5000	5001 to 10000	10 001 to 20 000
Analogue	0 (Note 2)	dB	30	28	26	24	21	18	15
Composite circuit	0.5	dB	29	27	26	24	21	18	15
	1	dB	28	27	25	23	21	18	15
	2	dB	27	26	25	23	20	18	15
	3	dB	26	25	24	23	20	18	15
	3.5	dB	26	25	24	22	20	18	15
	4	dB	25	24	23	22	20	17	15

Note 1 – The number of QDUs contributed by various processes are given in Table 1/G.113 [8].

Note 2 – The values are idle noise terminated with a nominal impedance of 600 Ω.

Note 3 – The section of the circuit provided by satellite (between earth stations), employing FDM techniques, contributes approximately 10 000 pWp (−50 dBm0p) of noise. Therefore, for the purpose of determining the total distortion limits for international public telephony circuits, the length of this section may be considered, from Table 4/M.580, to be equivalent to 2500 km.

ANNEX B

(to draft Recommendation G.136)

SOURCE: THE URSS TELECOMMUNICATION ADMINISTRATION

TITLE: INTERRELATION BETWEEN ERRORS IN A DIGITAL CIRCUIT AND IMPULSE NOISE IN VOICE-FREQUENCY CHANNELS OF THE PCM SYSTEM

B.1 *Introduction*

Voice-frequency channels of PCM as well as FDM systems should be fit for transmitting various types of signals. It is well known that the transmission quality of discrete signals in voice-frequency channels is affected by impulse noise. At present, Recommendation G.712 has no requirements to voice-frequency PCM-channels regarding impulse noise. However, under real-life conditions in a voice-frequency PCM channel impulse noise contributes to the error-rate of digital links. The present contribution gives the investigation results of impulse noise in voice-frequency PCM-channels.

B.2 *Influence of digital circuit errors on impulse noise in an idle voice-frequency PCM channel*

Evaluation of error influence on digital links on the value of impulse noise in voice-frequency channels was conducted experimentally on a channel equipment (satisfying Recommendation G.712) of a PCM transmission system (2048 kbit/s). With the help of an error simulator errors had been introduced into one or several bits corresponding to a chosen idle voice-frequency channel of a digital link (Figure 1). In the voice-frequency channel impulse noise could be observed with the help of an oscillograph. The shape of the pulse response in the voice-frequency channel is presented in Figure B-2.

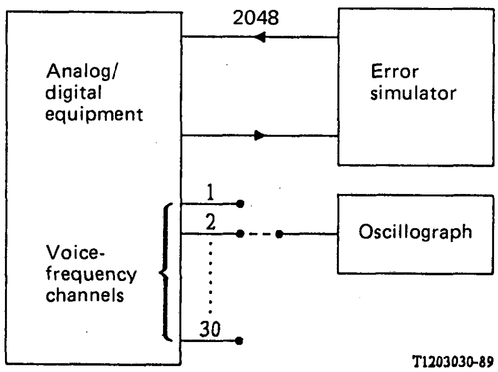

FIGURE B-1

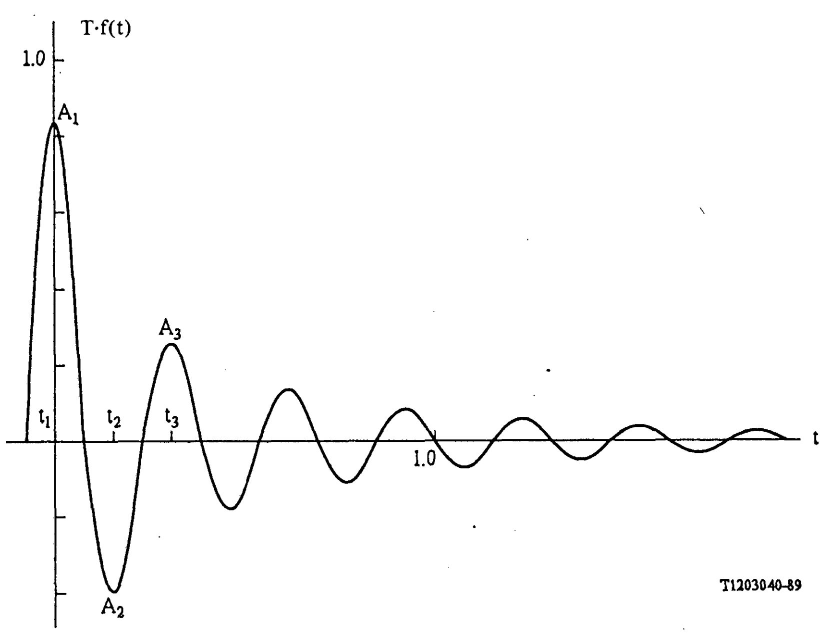

FIGURE B-2

The parameters of pulse response are given in Table 1 (the values are chosen for the point of the relative zero level at a resistance of 600 ohms). These data allow us to formulate the following conclusions:

- The pulse amplitude of the response depends on the bit number which contains the error; the errors in the more significant bits cause a greater amplitude of the response.
- With single errors the maximum value of the pulse peak A_1 (in case of an error in the second bit) is -22.1 dBm0.
- With burst-building and with an increase in the number of errored bits in the code word of the prime digital path (2048 kbit/s) the response amplitude values A_1, A_2, A_3, ... grow, but their duration, as determined by the response of the channel's low frequency receiving filter, remains unchanged. This applies to the cases where in a prime digital path, the error bursts affect the digital stream for not more than one discretization period, i.e. the number of the errors in a burst does not exceed 256. With errors in code words occurring every 125 µs the superposition of responses takes place as a result of the receiving filter reaction on the error pulses in each following discretization period.

TABLE B-1

Errored bits in a frame of primary multiplex	Amplitude of pulse response			Duration of pulse response		
	A_1	A_2	A_3	t_1	t_2	t_3
	dBm0	dBm0	dBm0	µs	µs	µs
2	−22.1	−28.2	−33.8	320	160	130
3	−34.1	−40.2	−45.8	320	160	130
2 and 3	−10.1	−16.2	−21.8	320	160	130
2 and 3 and 4 from 2 to 8,	−4.1	−10.2	−15.8	320	160	130
2 discretization periods from 2 to 8,	+4.3	−6.7	−14.8	440	180	100
3 discretization periods from 2 to 8,	+4.3	−4.9	−14.8	600	200	100
4 discretization periods from 2 to 8,	+4.3	−4.7	−14.8	680	180	120
5 discretization periods from 2 to 8,	+4.3	−6.7	−14.8	840	200	120
6 discretization periods from 2 to 8,	+3.8	−4.3	−14.8	930	200	100
7 discretization periods	+5.25	−8.7	−14.8	1100	180	140

Thus, when errors, on a 2048 kbit/s digital path grow into burst of 2 errors and more there is a certain probability that the value of the impulse noise in a PCM voice-frequency channel exceeds −21 dBm0 given in Recommendation M.1020, § 2.6.

With error bursts of 256 and more bits the above-mentioned impulse noise will always be present.

The quantitative relationship between the number of bursts, the number of errors in them within a definite time interval and the number of impulse noise interferences and the BER in a voice-frequency channel is under study at present.

TRANSMISSION PLAN ASPECTS OF LAND MOBILE TELEPHONY NETWORKS

Draft Recommendation G.173

(This Supplement is proposed for study during the present study period with the aim to convert it into a Recommendation.)

1 General

This Recommendation is primarily concerned with the special planning aspects which pertain to analogue or digital land mobile systems. Such systems, due to technical or economic factors, will prevent a full compliance with the general characteristics of international telephone connections and circuits recommended by CCITT.

The scope of this Recommendation is thus to give guidelines and advice to Administrations as to what kind of precautions, measures and minimum requirements which are needed for a successful incorporation of such networks in the national PSTN.

The performance objectives of such systems may vary between different groups of customers. For normal customers the objective should be to reach a quality as close as possible to CCITT standards. For other groups of very disciplines customers other performance objectives might be acceptable.

2 Network configurations

Under study.

Under this headline Administrations should be advised to use 4-wire transmission to avoid problems when accessing inherently 4-wire mobile links.

3 Nominal transmission loss of mobile links

Under study.

Under this headline the problems with the application of loudness ratings and the correct loading of the radio channels should be discussed.

The recommended LR values in CCITT Recommendation G.121 are not directly applicable due to the fact that the background noise level is higher in a car than what is assumed in Recommendation G.121.

What is the design objective for the speech levels from the radio path and what levels should be delivered to the network?

4 Stability

Under study.

5 Echo

Under study.

Under this headline the need for echo control devices should be discussed.

6 Noise

Under study.

(Can the European group give indications of the inherent noise performance of the codec algorithms being considered?)

7 Delay

Under study.

8 Effects of errors in digital systems

Several coding methods, such as SBC, ATC, RELP and APC-AB with transmission bit rates 16 kbit/s have been proposed to achieve spectrum utilization efficiency and quality comparable with conventional analog FM systems. However, the application of such highly efficient speech coding methods to land mobile radio can lead to a significant degradation in quality because of transmission errors.

Mobile radio links are not always error-free. Burst errors occur frequently due to multipath fading. It has been reported that the average bit error rate (BER) performance of diversity reception is 10^{-2}-10^{-4} in the 10 to 20 dB range of the average carrier to noise power ratio (CNR), and burst error length reaches 20 to 100 bits in case of 16 kbit/s digital signal transmissions. Therefore, robustness against burst error is an important characteristic for speech coding applied to mobile communication. Speech CODECs in mobile radio links should involve error control techniques so as to provide robustness in multipath fading channels. Thus, the transmission bit rate includes redundancy bits for error control.

Concerning quality evaluations, it may be better to use the average CNR as the receiving level for comparisons among analogue and digital systems. This is because it can present the receiving level as a normalized unit for both analog FM and digital systems. In quality evaluations between digital systems, the average signal energy per bit to noise power density ratio (E_b/N_o) is suitable for the presentation of the receiving level. This is because it can describe the receiving level as a normalized unit for any transmission bit rate and receiving bandwidth.

9 Quantizing distortion

Under study.

10 Effect of transmission impairments on voiceband data performance

Under study.

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