ETSI TECHNICAL REPORT

ETR 004

August 1990

Source: ETSI TC/BTC

Reference: BT-2001

ICS: 33.080

Key words:

Business Telecommunications (BT); Overall transmission plan aspects of a; private branch network for voice connections with access to the public network

ETSI

European Telecommunications Standards Institute

ETSI Secretariat

Postal address: F-06921 Sophia Antipolis CEDEX - FRANCE
Office address: 650 Route des Lucioles - Sophia Antipolis - Valbonne - FRANCE
X.400: c=fr, a=atlas, p=etsi, s=secretariat - **Internet:** secretariat@etsi.fr

Tel.: +33 92 94 42 00 - Fax: +33 93 65 47 16

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Foreword

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This ETR has been produced by the Business Telecommunications (BT) Technical Committee of the European Telecommunications Standards Institute (ETSI).

Table of Contents

Overall Transmission Plan Aspects of a Private Branch Network for Voice Connections with Access to the Public Network

1 General

1.1 Scope and Introduction

This Technical Report is a Framework Document which is intended for use by Administrations, and public and private network operators. It deals with the transmission planning of Private Branch Networks (PBNs) which send or receive speech telephone calls to or from Public Switched Telephone Networks (PSTNs) or Public Integrated Services Digital Networks (ISDNs). It recognises the overall responsibility of administrations for ensuring that the quality of national communications is consistent with international quality objectives.

This Technical Report is divided into three main Sections together with two Annexes. The three Sections are:

Section 1: Scope, introduction and principal recommendation,

Section 2: General guidance and information on transmission parameters,

Section 3: Specific Recommendations.

In addition, Annex A lists the abbreviations and Annex B the terms and definitions used in this Technical Report.

This guidance and general information is given for the benefit of administrations who are drawing up national rules for private network transmission and for the benefit of private network operators who are planning their networks.

Public networks within Europe contain a mixture of analogue and digital equipment and there are significant differences in the design of the analogue networks in different countries (e.g. different loudness levels). In addition, although the public networks will eventually be very similar when they are fully digital, the conversion from analogue to digital will be carried out in different ways and at different times in different countries. During this conversion process, it may be possible to increase the impairment allowances given to private networks and the greatest possible flexibility in this respect is desirable.

For these reasons it is impracticable to establish a European Telecommunications Standard for private networks where the call paths in the public network contain analogue sections. However, the Recommendations in Section 3 for the case where the call path in the public network is fully digital may be suitable as the basis for an ETS.

The Recommendations in Section 3 relate only to calls which pass through a public switched network to an international switching centre. In the case of other calls (e.g. national calls) it may be possible for the impairment allowances for private networks to be increased.

The approach followed is to specify the performance of a private network at the point or points where it is connected to a public network. This approach may also be used for private networks that include international private leased circuits, provided that the performance at the connection point is maintained. However, the approach to more complex topologies using international private leased circuits is for further study.

The transmission planning in this Technical Report is valid, in principle, for analogue private automatic branch exchanges (PABXs), digital PABXs and PABXs with ISDN Services (ISPBXs). In this Recommendation only the term PABX is used.

The transmission characteristics of digital PABXs can be found in the following ETS's:

T/TE 10-01	Transmission Characteristics of digital PABXs
T/TE 10-02	Transmission Characteristics of 2-wire analogue Interfaces of a digital PABX
T/TE 10-03	Transmission Characteristics of 4-wire analogue Interfaces of a digital PABX
T/TE 10-04	Transmission Characteristics at digital Interfaces of a digital PABX

Section 2 of this Technical Report concerning the general transmission parameters provides information about the relevant CCITT Recommendations of the G.100 series and the P series which should be taken into account in the transmission planning of a telephone network. Unless otherwise stated, all references to CCITT Recommendations refer to CCITT's Blue Books (1989). This Section gives guidance and background information for administrations and public network operators, private service providers, manufacturers and users.

Transmission planning aspects of the speech service in the European mobile system (GSM PLMN) are covered by CEPT Recommendations GSM 03.50 and T/TM 03.12. The successful interworking of a mobile system and cordless telephones forming part of a PBN is under study.

1.2 Principal Recommendation

Administrations are recommended to introduce, within their legal and contractual arrangements for telecommunications, rules for the planning of PBNs which send or receive speech telephone calls to or from a PSTN or an ISDN. Such rules should be consistent with the Recommendations set out in Section 3 of this Technical Report, and Administrations should draw, as appropriate, upon the guidance and information in Section 2 when framing them. When an administration is framing technical rules for the public international interconnection of PBNs it should assume that the planning of PBNs in other European countries complies with rules framed in accordance with the Recommendations in Section 3.

Any new rules for the planning of private networks should apply to new networks and major modifications to existing networks which are brought into service after the rules come into force.

2 General Guidance and Information on Transmission Parameters

2.0 General

The problem of assessing transmission parameters of PBNs is dealt with in CCITT Recommendation G.171, a basic reference for the transmission planning of PBNs, mainly in the case of international 4-wire connections. However, detailed specifications applying to the other types of connection (e.g. national 2-wire, digital, etc.) and consistent with the national system requirements of individual countries are not provided by Recommendation G.171.

This Standard therefore provides detailed specifications for the transmission parameters for such connections in Europe in a single comprehensive document.

In certain cases parameter values have been found to differ from those specified by CCITT. In these cases the differences are clearly indicated and the reasons for the change are given.

The present evolution of the European PSTN towards IDN/ISDN implies a steady decrease of analogue processes and unintegrated digital processes and a contemporary shift of the public main 4-wire (long-distance) loop end towards the user.

It is recognized that the actual interconnection point of a PBN to the PSTN, i.e. the position of the main 4-wire loop terminating unit (MTU), has a major impact on the transmission parameters relevant to the connection between the two networks.

This Technical Report accepts therefore the concept that, for a given parameter, the allowances no more used in the PSTN be available, at least partially, in the PBNs under the control of the PSTN Operating Agencies. The major impact of this approach should be on the parameters such as loudness ratings, qdu and the amount of 4-wire loops allowed within a PBN.

This approach should make it easier to deal with the evolution of PBNs from analogue to digital and to introduce in PBNs new technologies, such as ADPCM, etc.

Reference connections for the connection of a PBN to the PSTN/ISDN together with examples of PBN configurations, are shown in Figure 2/1.

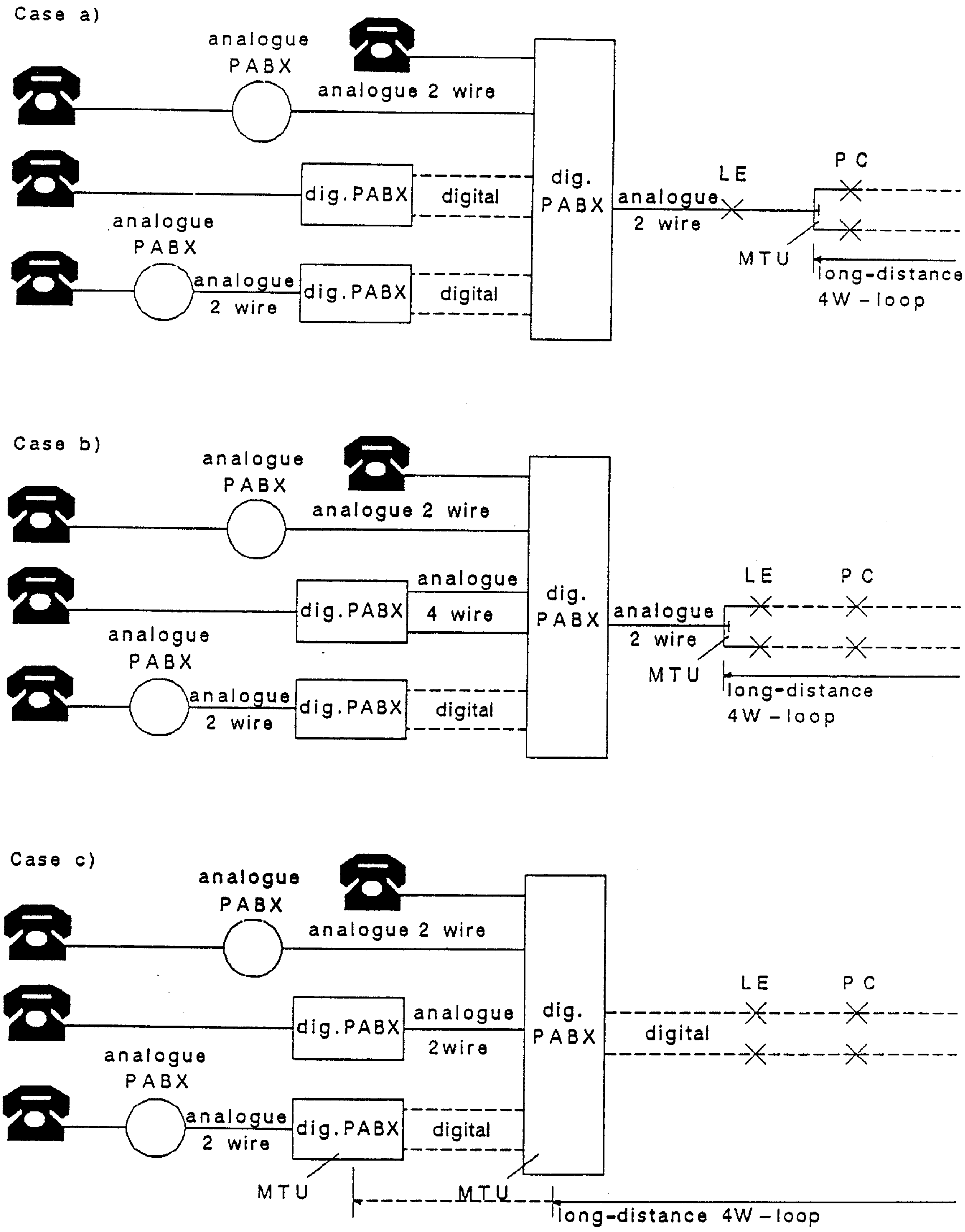


Figure 2/1 Reference Connections

2.1 Loudness Ratings

2.1.1 General Information

CCITT Recommendations G.111 and G.121 with their Annexes have been revised in such a way that the main body of the text give values in Loudness Ratings only. It is easy for Administrations to translate the Loudness Rating (LR) values into corresponding Corrected Reference Equivalent (CRE) values by means of the conversion information given in Recommendation G.111 Annex C.

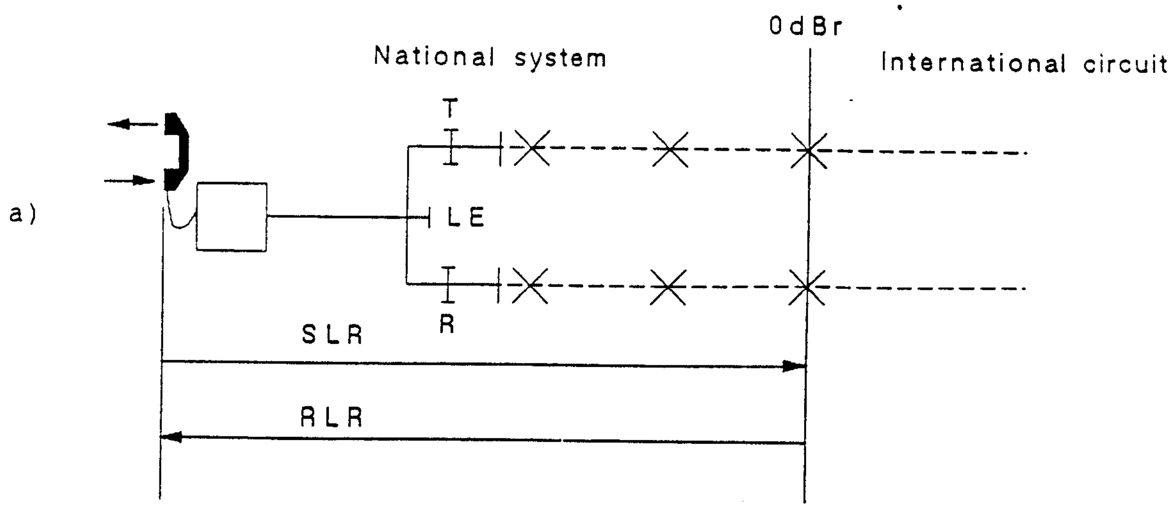
For planning purposes, LRs are defined by objective methods as described in CCITT Recommendations P.64, P.65 and P.79. The values given in terms of LR should provide an adequate loudness of speech in international telephone connections.

2.1.2 LRs in the National System

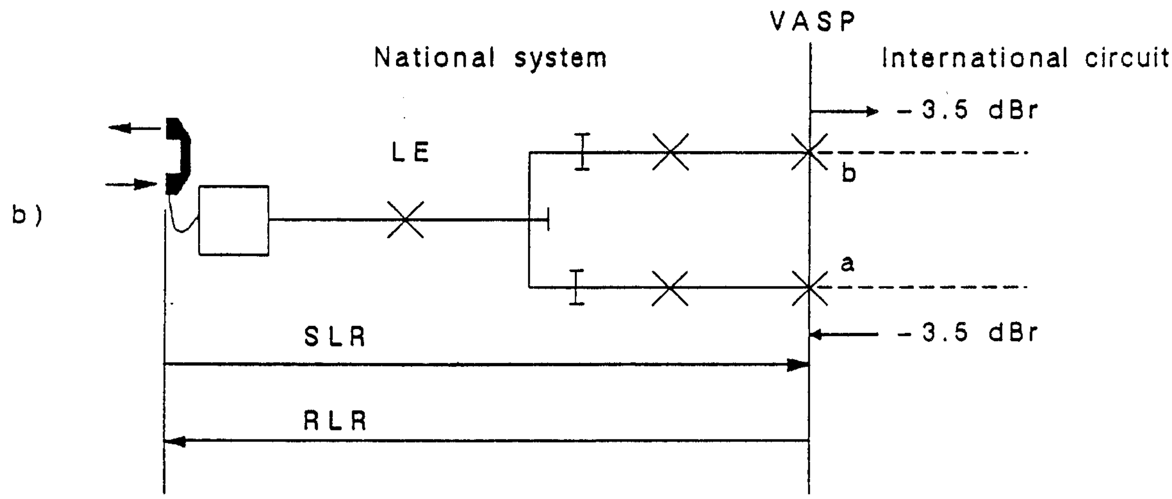
Send and Receive Loudness Ratings may, in principle, be determined at any interface in the telephone network. Values recommended by CCITT normally refer to the interface at the international switching centre.

For transmission planning purposes, the concept of the "Virtual Analogue Switching Point" (VASP) has often been used. The VASP does not physically exist but was found to be convenient when studying all-analogue and mixed analogue-digital connections. The relative levels at the VASP are -3.5 dBr in both transmission directions for a digital international circuit.

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, the SLRs and RLRs of the national system are also referred to a 0 dBr point at the international switching centre.



a) to a 0 dBr interconnection point



b) to an analogue interconnection point with -3.5 dBr

Figure 2.1/1 Definition of SLR and RLR reference points for a national system

The recommended CCITT values are given in Table 2.1/1.

referred to	SLR 0 dBr	SLR VASP	RLR 0 dBr	RLR VASP
Traffic-weighted mean value				
Long-term objective (minimum)	7	10.5	1	-3
(maximum)	9	12.5	3	-1
Short-term objective (maximum)	15	18.5	6	2
Maximum value for an average-sized country (Note 1)	16.5	20	12.5 Note 2	9
Minimum for sending	-1.5	2		

Note 1: These maximum SLR and RLR values should only be allowed in exceptional cases. When planning new networks with high traffic volumes it is preferable to use the short-term maximum values of SLR = 15 dB and RLR = 6 dB.

Note 2: This value differs from the CCITT value of 13 dB because for simplicity a relative level of 0 dBr is assumed at the receive end of the international circuit (digital transmission).

Table 2.1/1 Values of SLR, RLR cited in CCITT Recommendations G.111 and G.121

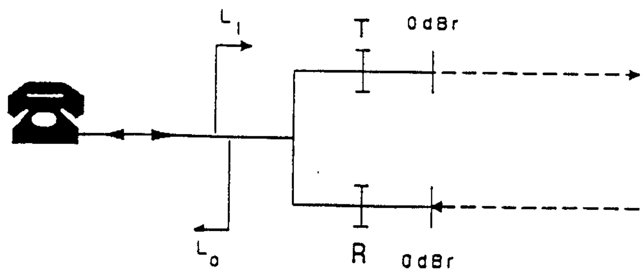
The maximum value of OLR is 29 dB and the optimum value of OLR is 10 dB.

For digital local exchanges with existing analogue 2-wire subscriber lines the typical values for R and T pads (see Figure 2.1/1) of the 2-wire interface are:

R = 7 dB
T = 0 dB

The given values for R and T pads are commonly used by many Administrations, as shown in Table C-1 of CCITT Recommendation G.121.

If these values are adopted for new digital networks then, in conjunction with adequate echo and stability balance return losses, the requirements of echo and stability will be met. The given losses for the R and T pads can be considered in terms of relative levels. The relationship is given in Figure 2.1/2



L_i = T dBr and L_o = -R dBr

Figure 2.1/2 Relationship between relative levels and R and T pads

2.1.3 Planning with LRs

For transmission planning it is convenient to evaluate the LR of the individual parts of a connection. Figure 2.1/3 depicts a speech connection between two users, consisting of several cascaded circuits.

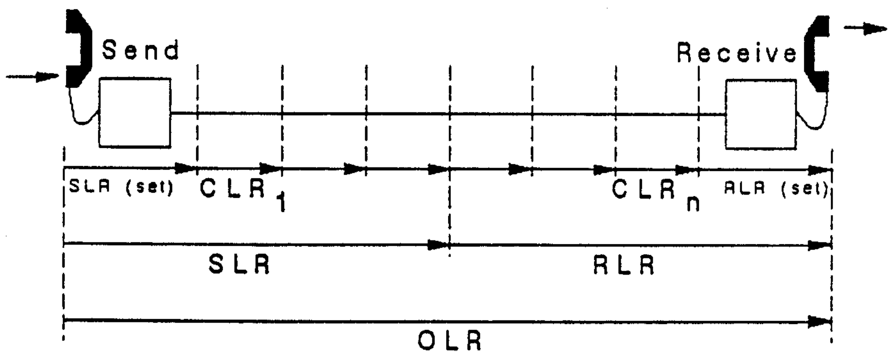


Figure 2.1/3 LRs in a normal speech connection

The Send and Receive Loudness Ratings of the telephone sets themselves are designated SLR(set) and RLR(set) respectively. The losses between the telephone sets are called Circuit Loudness Ratings (CLRs). Hence the Overall Loudness Rating (OLR) is:

$$OLR = SLR(set) + CLR_1 + \dots + CLR_n + RLR(set)$$

The CLRs are equal to the difference in relative levels between the respective interfaces, i.e. equal to the composite loss at the reference frequency 1020 Hz, using the nominal impedance appropriate to the interfaces.

Only in the unlikely case of a very unusual attenuation distortion is a somewhat more complicated computation necessary: the CLR is then the average loss over the frequency band 300 Hz to 3400 Hz on a logarithmic scale.

For unloaded subscriber cable sections, CLRs are considered to be equal to the composite loss at the frequency 1020 Hz. This mode of procedure is sufficiently accurate for planning purposes. This value can also be estimated from the cable parameters by the following expression:

$$CLR = K \sqrt{R \times C} \quad \text{in dB/km}$$

where R = cable resistance in ohm/km
C = cable capacitance in nF/km
K = a constant, which is 0.014 at 900 ohms,
0.015 at 600 ohms, and
0.016 in the case of a complex impedance cable termination

Table 2.1/2 gives the values for certain existing subscriber lines in European networks when K = 0.016.

diameter in mm	R in ohm/km	C in nF/km	CLR in dB/km
0.4	270	45.0	1.76
0.5	168	50.0	1.47
0.6	120	47.0	1.20
0.8	67	38.0	0.80

Table 2.1/2 CLR for certain existing subscriber lines in European networks

2.1.4 LRs of Telephone Sets

2.1.4.1 Analogue telephone sets

The Loudness Ratings of analogue telephone sets are determined objectively by special measuring instruments conforming to CCITT Recommendations P.64, P.65 and P.79. The measurement set-up must provide a representative current feeding bridge and may include different lengths of (artificial) unloaded subscriber lines. The parameters usually measured are SLR, RLR and STMR.

Commercial instruments following Recommendation P.79 use a measuring band of 200 Hz to 4000 Hz or even 100 Hz to 8000 Hz. This is a good deal wider than the band for which CCITT Recommendations specify an assured transmission, namely 300 Hz to 3400 Hz.

It should also be noted that the P.64, P.65 and P.79 Loudness Rating measurements are specified to be made with a terminating impedance of 600 ohms.

Therefore, to avoid confusion, measured values of the Send and Receive Loudness Rating of analogue telephone sets are designated by the index "w" (for wideband). To obtain the proper value 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 (the correction is not necessary for digital sets).

SLR(set) = SLRw + 1 dB
RLR(set) = RLRw + 1 dB

The planning values for SLR and RLR of different countries are given in Table 2.1/3:

country	SLR in dB	RLR in dB	Remarks
Austria (Note 1)	+3.5 (+0.5)	-7.5 (-10.5)	Note 3
Denmark	+2.0	-8.0	
Finland	+6.0	-4.0	
France	+5.0	-10.0	
FRG	+4.0	-8.0	
Greece	+3.0	-8.0	
Italy (Note 1)	+3.5	-7.0	
Netherlands			
Norway	-1.0	-9.0	
Spain			
Sweden (Note 1)	+4.0	-8.0	Note 2
Switzerland (Note 1)	+4.0	-7.0	
U.K.	0.0	-8.0	

Note 1: Provisional values.

Note 2: Automatic regulation (4dB lower values for long extention lines).

Note 3: The values in brackets are used for long subscriber lines.

Table 2.1/3 SLR and RLR of standard analogue telephone sets in certain countries

2.1.4.2 Digital handset telephones

CCITT Recommendation P.31 defines the transmission characteristics for digital standard telephones with handsets. For the long-term SLR and RLR, this document proposes the following nominal values:

SLR = + 8 dB;
RLR = + 2 dB with the volume control in the nominal position.

Draft ETS T/TE 04-15 (Candidate for NET 33) states the transmission characteristics of a European ISDN digital telephone. The values of SLR and RLR are:

SLR = + 7 dB;
RLR = + 3 dB

2.1.4.3 Loudspeaker telephones

For hands-free telephones using loudspeakers the SLR and RLR should be measured and computed using the methods given in CCITT Recommendation P.34 Section 6 (Note).

Note: Further work is required to develop a practical test method using, for instance, CCITT Recommendation P.50.

2.1.4.4 Headset telephones

For headset telephones the SLR and RLR should be measured and computed using the methods given in CCITT Recommendation P.38. Study is continuing in CCITT SG XII on other types of earpieces.

2.1.4.5 System-specific telephones

For system-specific telephones the SLR and RLR should be measured and computed using one of the methods given in CCITT Recommendations (P.31, P.34 Section 6 or P.38) appropriate to the type of acoustic interface, ie, handset, microphone/loudspeaker or headset. The values of SLR and RLR to be used for planning purposes are the nominal values declared by the supplier to and from a reference point declared by the supplier.

Note: The reference point will, in general, be treated within the PBX or equivalent system to which the telephone is specific.

2.1.5 Planning of a National System comprising a PBN

The network planning guidance given in this Standard is based on the use of handset telephones. However, this mode of network planning also allows loudspeaker and headset telephones to be connected.

As shown in Table 2.1/1, the following maximum LRs have been defined for national systems including PBNs connected to the PSTN by means of 2-wire analogue lines:

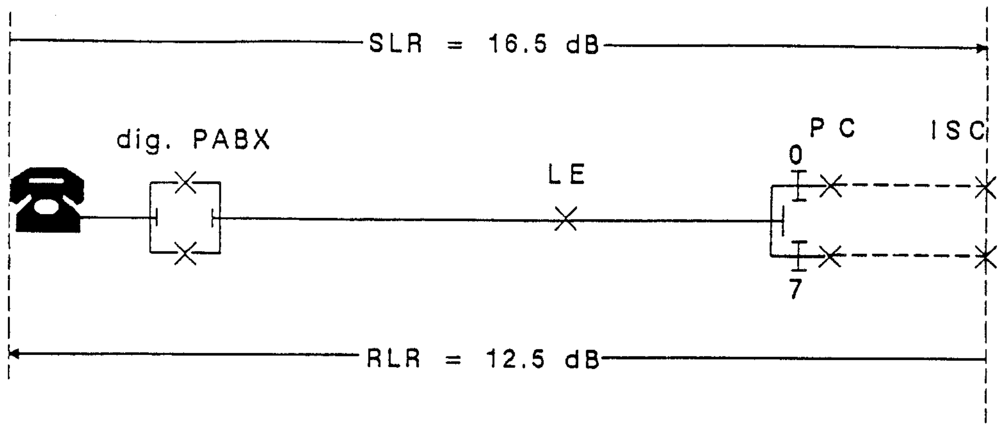
$$SLR_{max} = 16.5 \text{ dB}$$
$$RLR_{max} = 12.5 \text{ dB}$$

This results in a maximum OLR of 29 dB. However, these maximum values should only be used in exceptional cases. This is shown in Figure 2.1/4a. The division of the LRs is a national matter. However for a digital connection to the ISDN (Figure 2.1/4b) the overall value for the national system will be completly determined by the PBN values. Hence the SLR and RLR value for the PBN should meet the long term CCITT objectives as given in Table 2.1/1. This could be achieved using the configuration in Figure 2.1/4b where a digital telephone with the European values 7 dB and 3 dB is connected directly to the ISDN via a PABX.

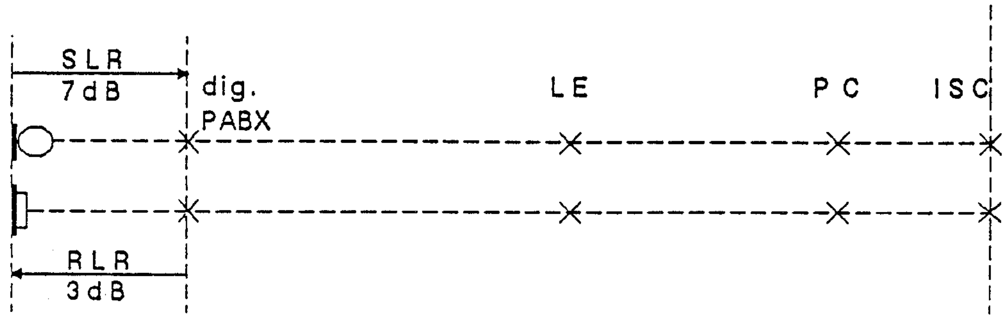
In the mixed analogue/digital PBN a digital PABX with digital telephone sets can be connected to the PSTN by means of 2-wire lines. In preparation of a future ISDN connection it is advisable that these digital telephone sets have the same LR values as defined in NET 33.

In the case of an analogue 2-wire connection to the PSTN, the objective is at least to meet the CCITT Recommendation for the maximum value (preferably the short-term maximum value) of the national system as given in Table 2.1/1. The example in Figure 2.1/4c shows how this may be achieved taking into account the minimum loss and maximum gain in the K2-interface of the digital PABX.

a)



b)



c)

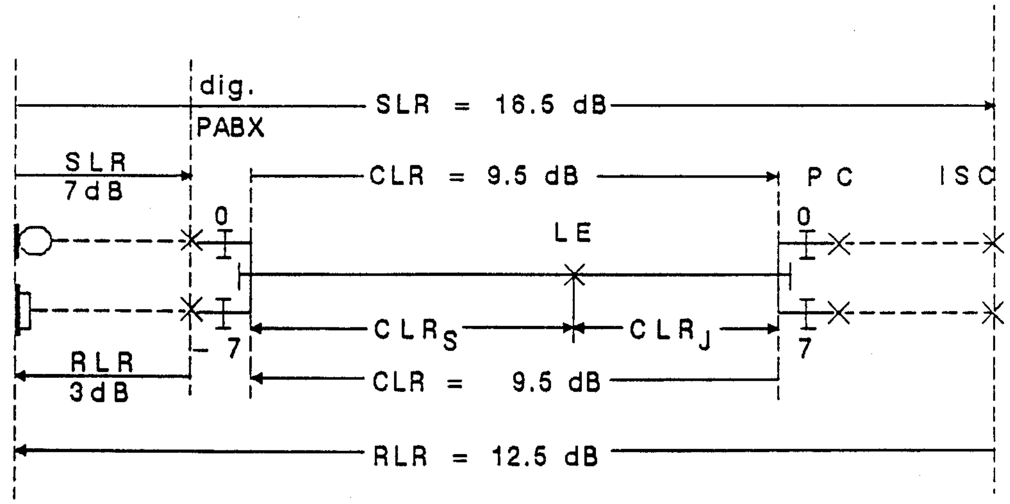


Figure 2.1/4 Examples of SLR and RLR of the national system

Figure 2.1/5 and Table 2.1/4 show planning values for the national system in a number of countries.

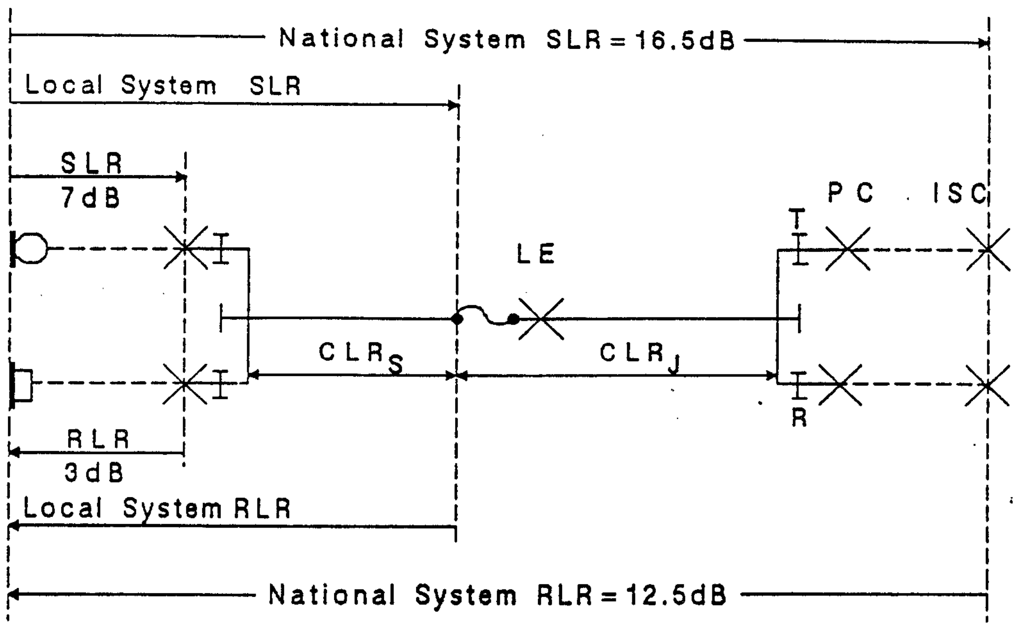


Figure 2.1/5 Configuration for which the planning values of Table 2.1/4 are applicable

Country	T in dB	R in dB	CLR_J (max) in dB	Local System (max) SLR in dB	Local System (max) RLR in dB
Austria	0.0	7.0	5.0	11.5	0.5
Denmark	0.0	6.0			
Finland	0.0	7.0	7.5	9.0	-2.0
France	0.0	7.0	3.0		
FRG	0.0	7.0	2.5	14.0	3.0
Greece	0.0	7.0	7.0	9.5	-1.5
Italy	0.0	7.0			
Netherlands					
Norway	2.0	5.0			
Spain	0.0	7.0			
Sweden (Note)	0.0	7.0			
Switzerland	0.0	7.0			
U.K.	1.0	6.0	4.5		

Note: The value of R includes 2 dB digitally implemented loss. For purely national connections the value of R is 5 dB.

Table 2.1/4 Examples of planning values of the national system as part of an international connection

In a case of an analogue telephone set the division of the SLR and RLR of the telephone set and the R and T pads are of national competence. However the objective should be to achieve the same overall value as for the digital telephone set, as shown in Figure 2.1/6.

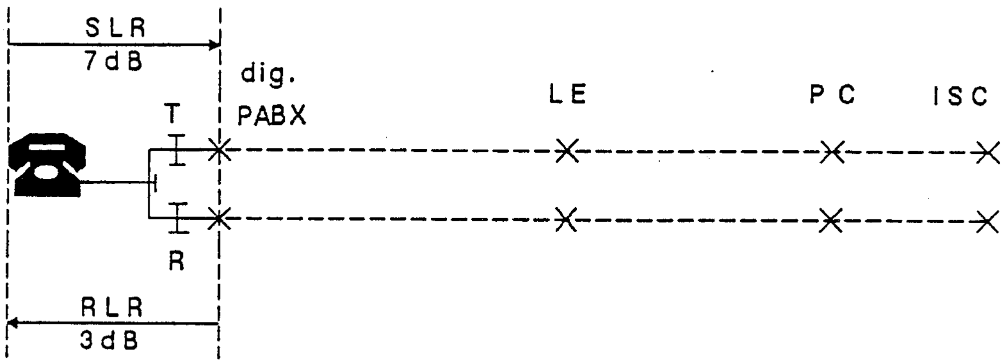


Figure 2.1/6 SLR and RLR values of an analogue telephone conected to the ISDN

2.2 Sidetone

2.2.1 General Information

Two effects of sidetone have to be taken into account:

- Sidetone Masking Rating, STMR (Talker's sidetone) and
- Listener's Sidetone Rating, LSTR

STMR is the loudness loss between a user's mouth and his ear (earphone) via the electrical sidetone path. LSTR is the loudness loss between a Hoth-type room noise source and the user's ear (earphone) via the electrical sidetone path. For analogue telephone sets both are determined by the grade of conformity between the sidetone balance impedance inside the telephone set and the line impedance. For digital telephone sets both are determined by the electrical characteristics of the artificial sidetone path between the sending and receiving direction and possible, low-delay reflections at the other end of the connection.

To reach satisfactory sidetone characteristics for the most critical case of long subscriber lines with low receiving loudness, the analogue telephone sets are usually equipped with capacitive complex sidetone balancing networks, similar to the image impedance of unloaded subscriber cables. (To provide these characteristics on short extension lines in digital PABXs as well, these PABXs should have a capacitive complex input impedance). An alternative method, sometimes used at exchanges with resistive input impedances, is to employ telephone sets with a variable sidetone balance impedance Z_{SO} controlled by the feeding current. Thus, on short extension lines Z_{SO} is made to emulate the actual resistive exchange impedance. The artificial sidetone path of digital telephone sets should have characteristics in loss and frequency response comparable with the balance return loss reached under optimal conditions in analogue telephone sets. Optimal conditions exist if the disturbing loudness is lowered to non-impairing levels yet is still high enough not to give the impression of a dead line.

This is for further study.

2.2.2 STMR

For transmission planning STMR can be calculated using the telephone set's loudness ratings SLR and RLR and the balance return loss between the mean line impedance and the sidetone balance impedance. A suitable algorithm is given in CCITT Recommendation G.111, Annex A, A.4.3.2.

For the Sidetone Masking Rating the preferred range is 7 dB to 12 dB for 2-wire telephone sets and 15 dB for digital 4-wire telephone sets (see CCITT Recommendation G.121, § 5.2).

This is for further study.

2.2.3 LSTR

For transmission planning, LSTR can be calculated from STMR, taking into account the weighted mean D of the difference between diffuse and direct sound sensitivity curves of the telephone set (see CCITT Recommendation G.111, Annex A, A.4.3.3).
It is

$$\text{LSTR} = \text{STMR} + D$$

For modern telephone sets with linear microphones, D is in the order of 1.5 to 4 dB, depending on the handset's geometric shape. Telephone sets with carbon microphones, however, typically have a higher sensitivity threshold, making them somewhat less susceptible to room noise. Their D-value is in the order of 6 to 8 dB at 60 dB(A) room noise. To guarantee a sufficient conversation quality, also under conditions of high room noise and high loss connections, the LSTR should be more than 13 dB (see CCITT Recommendation G.121, § 5.3).

This is for further study.

2.2.4 Implications for Impedance Matching in the 2-wire Parts of the Network

If sensitive telefone sets are used in order to overcome high losses in the 2-wire local network, a high degree of close impedance matching is required in order to avoid too loud sidetone signals. It is useful to consider the following relation:

$$\text{STMR} = \text{SLR(set)} + \text{RLR(set)} + \text{Am} - 1$$

(see CCITT Recommendation G.111, Annex A, Formula A.4-3)

Here SLR(set) and RLR(set) refer to the telephone set itself. Am is in essence a weighted mean of the return loss between Z_{SO}, the sidetone balance impedance of the set, and Z, the line impedance seen from the terminals of the set. The weighting may be considered to be approximately more or less flat, on a logarithmic frequency scale between 800 Hz and 3200 Hz.

As an example, suppose the set has SLR = 3.5 dB, RLR = -11.5 dB, D = 2. In order to achieve LSTR = 13 dB one must thus ensure that

$$\text{Am} = 13 - 3.5 + 11.5 + 1 = 22 \text{ dB}$$

This high value of Am means, for instance, that the range of subscriber cable gauges must be restricted.

2.3 Propagation Time

2.3.1 General Information

It is necessary in a telephone connection to limit the propagation time between the two users. A significant propagation time causes difficulties in conversation over the connection. This arises from two causes. Firstly, the signal is reflected back from the distant end, causing an echo to be returned to the talker (this is considered in item 2.4). Secondly, even if ideal echo control was achieved, the delay between a user talking and receiving a reply from the user at the distant end of the connection could cause conversational difficulty.

CCITT Recommendation G.114 gives guidance about the mean one-way propagation time in telephone connections.

2.3.2 Limits for a connection

CCITT Recommendation G.131 contains a "practical rule" Rule M), according to which some form of echo control is required to reduce the level of the echo if the one-way propagation time is about 25 ms. This rule was developed in conventional arrangements of electrical echos for two-to-four wire converters.

As a network performance specification the following limitations apply to a mean one-way propagation time in cases where echo sources exist and appropriate echo control devices, such as echo suppressors and echo cancellers, are used:

(i) 25 to 300 ms, acceptable.

(ii) 300 to 400 ms, acceptable with:
- effective control of all echos without clipping by use of good echo cancellers;
- low background noise leading to an absence of perceptible noise contrast;
- low distortion of transmitted signals;
- ideal loudness rating.

(iii) above 400 ms, unacceptable.

2.3.3 Sources of delay

CCITT Recommendation G.114 identifies various elements present in some PSTN or ISDN connections which cause delay. The following planning values applying to mean one-way propagation time should be used.

Terrestrial coaxial cable or radio relay systems; FDM and digital transmission.	4 µs/km
Optical fibre cable system; digital transmission.	5 µs/km
Geostationary satellites	260 ms

FDM channel modulator or demodulator	0.75 ms
PCM coder or decoder	0.3 ms
Echo cancellers	1 ms
Digital exchange or PABX; digital - digital.	0.45 ms
Digital exchange or PABX; analogue - analogue.	1.5 ms
Digital exchange or PABX; analogue subscriber line - digital junction.	0.975 ms
Digital exchange or PABX; digital subscriber line - digital junction.	0.825 ms
ISDN Basic Access	1 ms
ISDN Primary Access	0.2 ms
Pan-European Public Land Mobile Network	90 ms

More details on the delays of digital exchanges can be found in CCITT Recommendation Q.551 and of digital PABXs in T/TE 10-01.

For a telephone connection only one satellite hop is allowed as the limit of 400 ms would otherwise be exceeded. Planning values for the propagation time for

- Digital Speech Interpolation
- Low Rate Coding
- Cross-connectors
- Asynchronous Transfer Mode (ATM)

are under study.

2.3.4 Allocation of delay to the PBN

To avoid echo control devices in national connections including private branch networks the limit of the propagation time should be 25 ms (see Figure 2.3/1).

For an ISDN connection in an average-size country the following elements might be typical:

7 x digital exchange (digital - digital)	3.2 ms
2 x digital exchange (digital junction-digital subscriber)	1.6 ms
2 x ISDN basic access	2.0 ms
distance between the network terminations 1500 km	7.5 ms
	14.3 ms

In this particular case the propagation time of a PBN without any echo control device connected to the PSTN/ISDN should not exeed 5 ms; but in other practical cases the actual allowances for the

PBN's may vary in the range 3 ms to 7 ms, depending on the size and the network characteristics of the country.

In some larger European countries the routing of many national digital public connections may have an average propagation time above 15 ms; whereas in some smaller European countries the same parameter may not exceed 12 ms. In general the propagation delay values specified in different countries will be $(25 - t_p)/2$ ms, where t_p represents the propagation delay requirements for the public network on national calls, because identical allowances will normally be given for national and international calls and these allowances will normally be the result of the constraints on national calls. Thus the propagation delay allowances for PBNs with NCPs at the centre of a country may be greater than those for PBNs with NCPs at the periphery. During the transition from analogue to digital the propagation time may be higher.

For PBNs where the propagation time exceeds the limits calculated above echo control is required. The maximum propagation time in the PBN under such circumstances is under study.

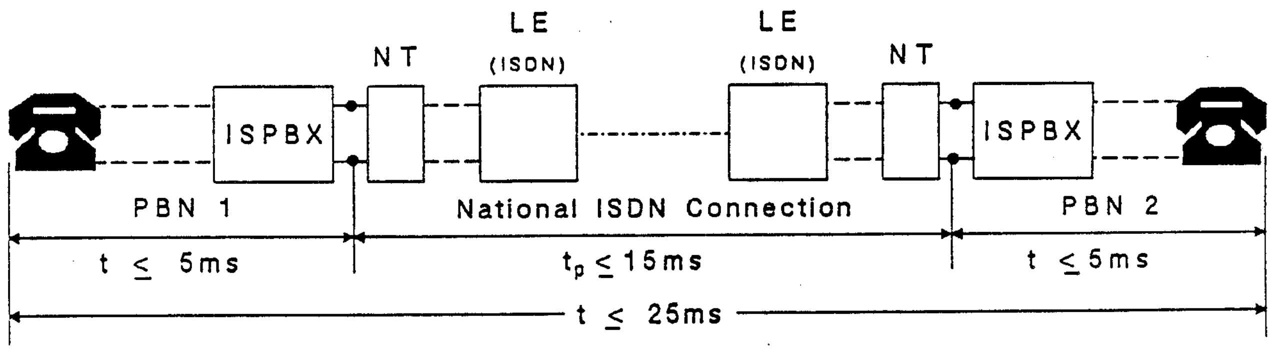


Figure 2.3/1 Example of propagation time in a national ISDN connection

2.4 Talker Echo

2.4.1 General Information

Talker echo is an undesirable phenomenon that may be observed in a telephone connection if the signal propagation time combined with echo sources are significant. There are two main sources of echo:

- electrical echo caused by coupling between the transmit and receive direction of transmission. The primary source of this form of echo is a two-to-four wire converter;

- acoustic echo caused by the acoustic path between receive and transmit transducers.

Electrical echo can be eliminated by the use of end-to-end four-wire transmission. Acoustic echo will be generated in all telephone sets with the exception of carefully designed headsets.

In general, electrical echo is characterized by a short reverberation time and low dispersion while acoustic echo is likely to have a longer reverberation time and greater dispersion. Cases involving acoustic echo may be even more complex in view of the fact that acoustic echo varies with time.

Echo planning guidelines have been provided within CCITT Recommendation G.131. Fundamentally, the planning objective is for not more than 1% of users to perceive talker echo at a level that causes them dissatisfaction with the telephone call.

To reduce the level of electrical talker echo, there are various ways of inserting echo control devices (ECDs) in connections having a one-way propagation time of more than 25 ms.

Acoustic echo control is under further study.

CCITT Recommendation Q.522 calls for ECDs to be "under the control of" an exchange. The same procedure is necessary for PABXs. For digital PABXs this will, as a rule, imply that it is most convenient to use digital ECDs and that they will actually form part of the controlling PABX. They can then be re-assigned as traffic conditions change. Furthermore, the future need of ISDNs to distinguish between different types of traffic and hence the possible need for ECDs ought to be considered.

Various ECD designs are described in the G.160 series of CCITT Recommendations. It is generally agreed that the type giving the best performance under all circumstances is the echo canceller, this is therefore the preferred type.

There has been a limited necessity to introduce echo control devices in existing analogue networks. The losses and signal propagation delays normally encountered have, in general, resulted in acceptable talker echo performance.

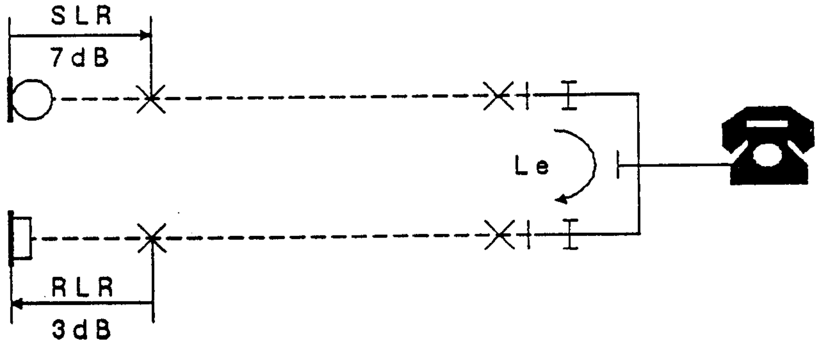
The talker echo path loss will be the addition of the losses in the transmit and receive directions of transmission and the balance return loss at the distant 2-wire to 4-wire conversion.

Most modernization programmes involve the introduction of digital PABXs to replace analogue PABXs. Although the overall loss on a connection is reduced through the use of digital equipment, one disadvantage of this is that the loss of the talker echo path is also reduced.

When the ISDN eventually extends 4-wire digital working into the PBNs, this will lead to further reductions in the overall connection loss with further increases in one-way propagation time and reductions in the talker echo path loss. Therefore, the trend towards lower network losses may create potential degradations in the echo performance within the network. Users are increasingly expecting improved performance and any new echo performance difficulties will result in an unfavourable user reaction.

2.4.2 Talker Echo Limits

The loudness of an echo path may be expressed by means of the "Talker Echo Loudness Rating (TELR)". The TELR is the addition of the SLR, RLR and the weighted Echo Loss Le (see CCITT Recommendation G.122) in the echo loop. It is normal practice to use the minimum nominal SLR and RLR values when calculating the TELR for a connection so the result will yield the worst case TELR value for the connection. Figure 2.4/1 illustrates the derivation of TELR for a typical connection.



TELR = SLR + RLR + Le

Figure 2.4/1 Derivation of TELR

The curves in Figure 2/G.131 in CCITT Recommendation G.131 indicate the minimum value of the TELR that must be introduced into the echo path if no ECD is to be fitted. The TELR is shown as a function of the mean one-way propagation time. The 1% curves applicable to fully digital connections call for the following TELR values:

TELR = 34 dB in 25 ms mean one-way propagation time

TELR = 56 dB in 400 ms mean one-way propagation time

In CCITT Blue Book, Volume V, Supplement No. 3, Part A "Transmission rating models", Annex A some more information is given about talker echo for short mean one-way propagation times. A TELR of 15 dB is an acceptable value if the echo delay is 10 ms. This echo source exists in PBNs connected to the PSTN by analogue means with a mean one-way propagation time of 5 ms in the PBN.

In the situation where the call path is wholly digital, both in the PBN and in the PSTN/ISDN the minimum nominal LR values (SLR = 7 dB and RLR = 3dB) of the telephone set, the echo loss at any 2-wire-4-wire converter in the PBN (M21 and L2-interface) should correspond to Le = 24 dB to avoid ECDs in connections with a mean one-way propagation time of up to 25 ms. For a K2-interface an echo loss of Le = 5 dB is sufficient to control the talker echo path with not more than 5 ms mean one-way propagation time in the PBN.

In the situation where the call path includes analogue or mixed analogue/digital circuits in the PBN and/or PSTN, the required Le value may be different, dependending on the number of analogue and mixed analogue/digital circuits and the loss introduced in the PSTN.

2.4.3 Electrical Echo Control

In the case of an electrical echo path ECDs are needed if the mean one-way propagation time is greater than 25 ms to reduce the level of the echo (see CCITT Recommendation G.131, Rule M). There are two types of ECDs:

- echo suppressors (CCITT Recommendation G.164) and
- echo cancellers (CCITT Recommendation G.165).

The information derived from subjective tests indicates that echo cancellers provide superior speech transmission performance compared echo suppressors. At opposite ends of a connection, different ECD types may be used since the function at one end is independent of that at the other end.

Connections should be designed to comprise the minimum practical number of ECDs.

In international connections with a high mean one-way propagation time (satellit hop) an ECD is installed in the PSTN/ISDN at the ISC (see Figure 2.4/2). In the case of echo cancellers the echo path is described by the impulse response. The response of a typical echo path shows a pure delay, 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 echo path delay does therefore not correspond to double the mean one-way propagation time. The echo path delay may vary in different national networks. It is assumed that the echo paths are basically linear and do not vary during a call.

Echo control for connections with non-linear or non-stationary conditions in the echo path (e.g. Low Rate Encoding, adaptive balancing, telephone conferencing equipment) are being studied by CCITT Study Group XII (Question 27).

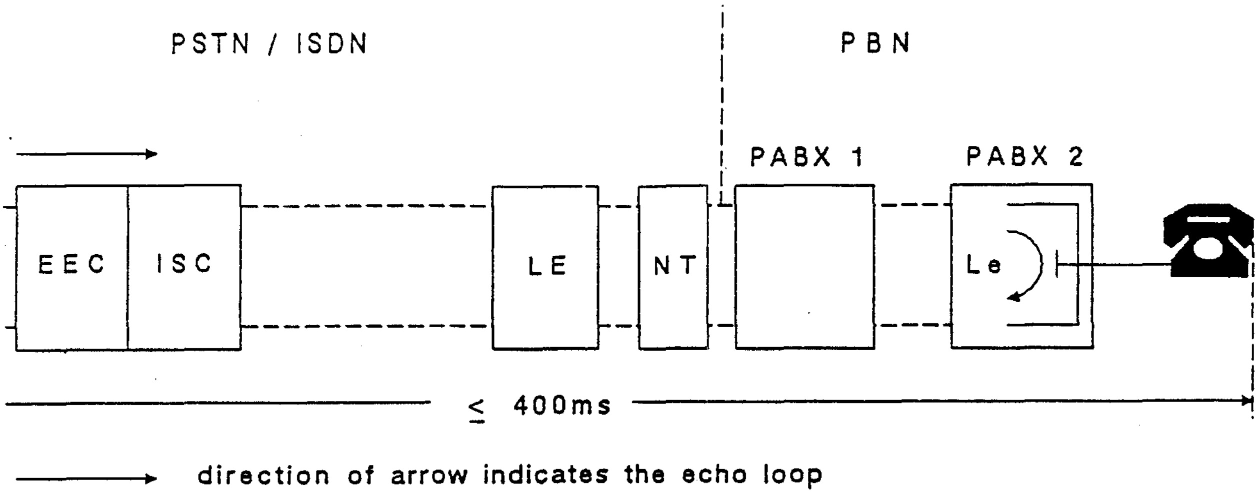


Figure 2.4/2 Electrical echo control in international connections

2.4.4 Acoustic Echo Control

The subject of acoustic echo control is under study in CCITT Study Group XII (Question 27).

At present, its main application is for handsfree telephony under critical conditions, such as video conferences. In the future, sophisticated acoustic echo control devices may also be included in loudspeaking telephones.

In the case of a 4-wire connected telephone set with an acoustic echo path any electrical echo control device in the network may be disabled. Acoustic echo control provided in such a telephone set must reach an echo loss level of 46 dB to avoid echo problems in connections with a one-way propagation time of up to 400 ms. This is shown in Figure 2.4/3 (for further study).

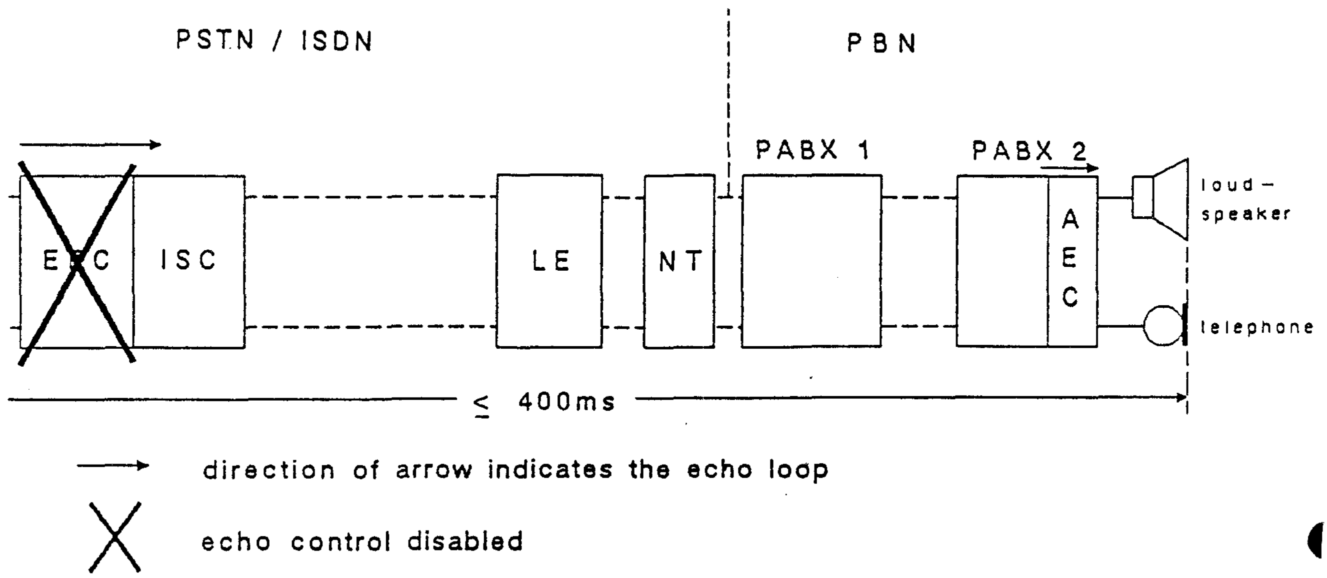


Figure 2.4/3 Acoustic echo control in international connections

If acoustic echo control is provided in a digital loudspeaker telephone using some form of echo cancellation, the cancellation algorithm should be designed to cope with the expected reverberation and dispersion. The reverberation and dispersion may vary with time. The expected dispersion values are under study.

In the case of a digital handset telephone, a careful design might make the use of echo cancellation techniques without non-linear processing possible. The implications of this possibility are under study.

In the case of a headset telephone, a careful design might eliminate the need for echo control.

2.4.5 Echo loss of PBN's

The calculation of echo loss is under study in CCITT Study Group XII, Questions 27/XII and 28/XII. In many cases the main cause of echo can be readily identified and calculations of echo loss are relatively simple.

2.5 Stability

The stability loss presented to the PSTN should meet the principles of the requirements in Sections 2 and 3 of CCITT Recommendation G.122. These requirements will be met if the attenuation between the 4-wire points of the terminating hybrid is at least 6 dB at all frequencies in the range 200 Hz to 4 kHz under the worst-case operational conditions. Short-circuit and open-loop terminating conditions at the 2-wire interface generally constitute worst case conditions (see Figure 2.5/1, case A). In some cases an inductive termination may constitute the worst case condition.

In some exchange designs it has been customary to introduce an additional loss of 3 dB to 6 dB temporarily during call set-up and call clear-down procedures. Often, in other designs the 4-wire-loop is not closed until both subscribers are off-hook.

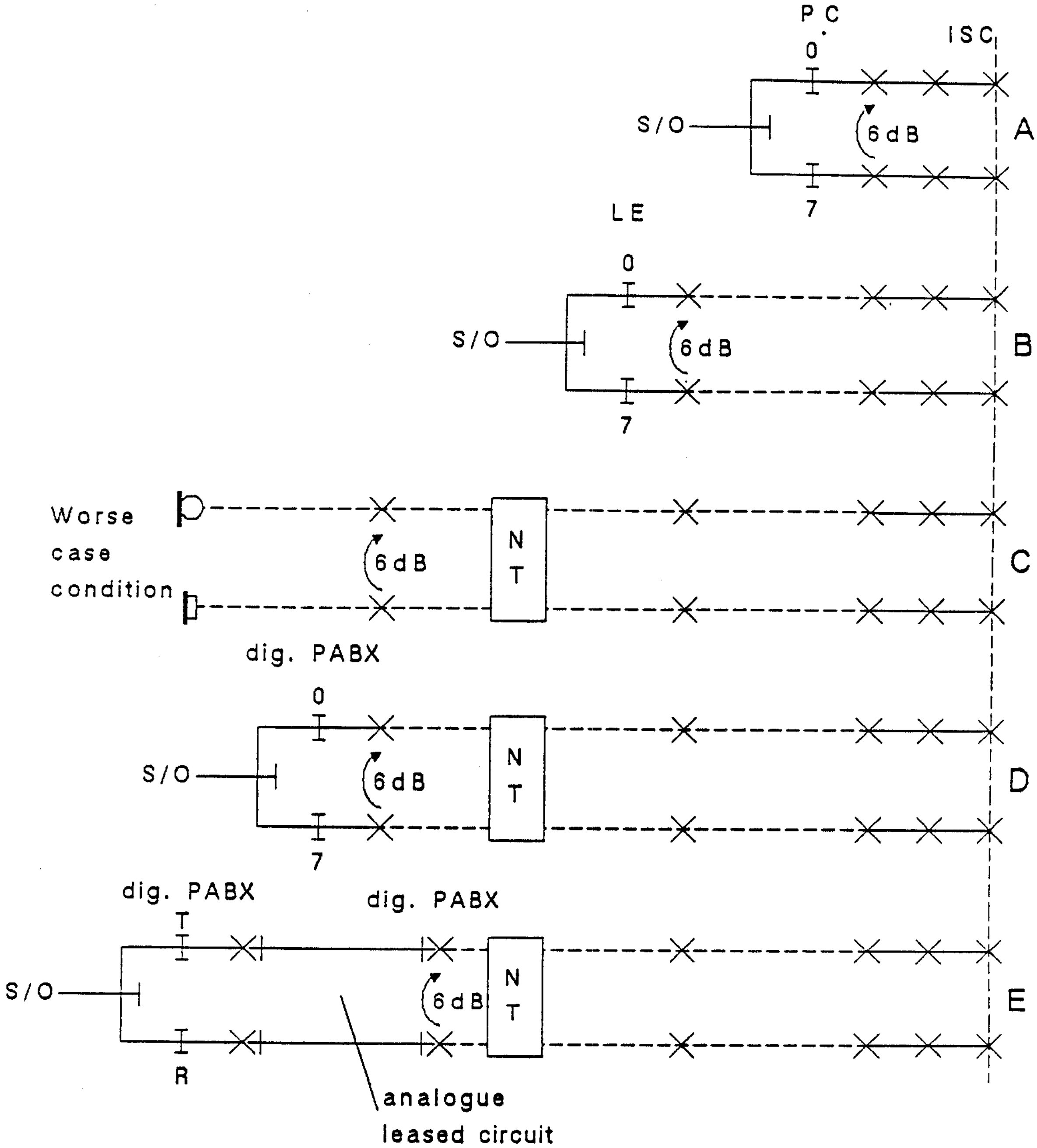
Where a digital local exchange is connected to the international chain, the half connection of the digital exchange may provide the total stability loss (see Figure 2.5/1, case B).

In a normal ISDN connection all kinds of equipments are connected 4-wire to the network. The stability requirements can be applied at the S-interface. In the case of a telephone set (see Figure 2.5/1, case C), the worst-case acoustic conditions will be as follows (with any volume control set to maximum):

- Handset telephones: the handset lying on, and the transducers facing, a hard surface.
- Loudspeaker telephones: a representative worse-case position of microphone and loudspeaker (for further study).
- Headset telephones: for further study.

Where a digital PABX is connected to the ISDN, the 2-wire analogue interfaces of the digital PABX may provide the total stability loss (see Figure 2.5/1, case D and E). For a wholly digital PBN the stability requirement can be met where the terminating interface provides for the sum of the pads R + T >= 6 dB loss.

Where the PBN call path includes 4-wire analogue transmission sections (i e, an analogue leased line as shown in Figure 2.5/1, case E) it may be necessary to increase the nominal stability loss to take account of the loss variation of the analogue sections. This can be done by increasing the values of the R and T pads or introducing additional loss in the analogue sections.



S/O - worst-case condition

Figure 2.5/1 Stability requirements for the PSTN and ISDN

2.6 2-wire-4-wire-2-wire Loops

2.6.1 General Information

In public switched telephone networks (PSTN) - also in the near future -, the subscriber line and the local exchange are often based on a 2-wire configuration whereas the trunks and higher order exchanges operate on a 4-wire basis. Thus a connection between two users will have a 4/2-wire hybrid at each end. This means that the 4-wire part of the connection and the hybrids form a 2-wire-4-wire-2-wire loop (hereinafter referred to as "4-wire loop") that may cause instability (singing).

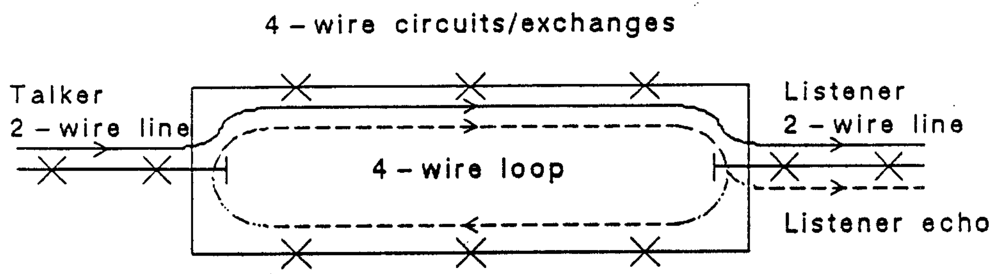


Figure 2.6/1 Connection with one 4-wire loop

The 4-wire loop also gives rise to echo. The signal is reflected twice and reaches the listener some time after the original signal. This phenomenon is called listener echo, and will be perceived as hollowness in a telephone conversation and may cause bit errors in data transmission. The degradation of the transmission will be dependent on the loss along the 4-wire loop called open loop loss (OLL).

The introduction of 4-wire digital PABXs and exchanges in a 2-wire environment leads to additional 4-wire loops, involving further reflections. This could result in increasing listener echo. If there is more than one loop in a connection, several (i.e. multiple) reflections may occur as shown in Figure 2.6/2. These reflections will add up at the receiving end.

It may be useful to carry out a computer analysis of complex circuits, involving several cascaded 4-wire loops. This would provide accurate information on Loudness Ratings, sidetone performance and echo conditions.

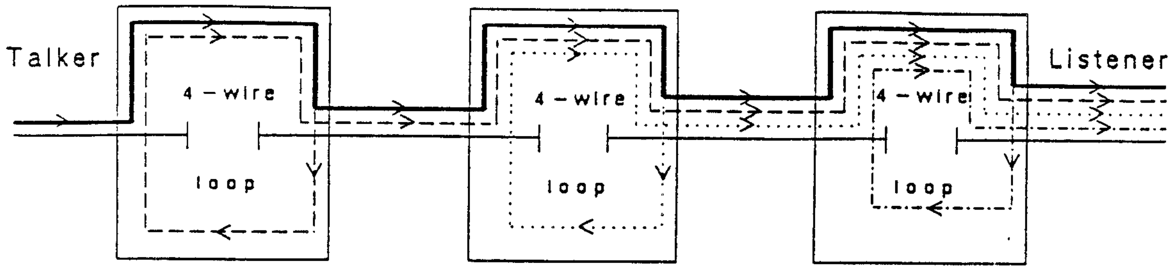


Figure 2.6/2 Example of multiple reflections in 4-wire loops (not all, possible reflections, are shown)

2.6.2 Number of 4-wire Loops

In order to control the cumulative effect of double-reflected signals, it is highly advisable to limit the number of 4-wire loops. According to G.122 in section 5.2.1 a complete connection should not contain more than five (exceptionally seven) physical 4-wire loops. To make sure that in any possible configuration this number of 4-wire loops is not exceeded, the number of 4-wire loops in a specific section of the network should be limited. At the beginning of the evolution towards a digital PSTN, within the PSTN a maximum of three 4-wire loops may be expected, in which case normally there should be only one 4-wire loop for a call path within the PBN. At the end of that evolution (or exceptionally even in the middle) two 4-wire loops could be available, if necessary, for the PBN. An extreme example of a network configuration comprising five 4-wire loops is shown in Figure 2.6/3. 4-wire loops with a very high OLL (exceeding, for example, 45 dB) need not be included in the number of loops in the connection.

A proper computer analysis will also serve as a guide as to how many 4-wire loops may be cascaded in a particular case.

The number of 4-wire loop will, in practice, be determined by the achieved values of OLL and the type of service (voice or voice band data).

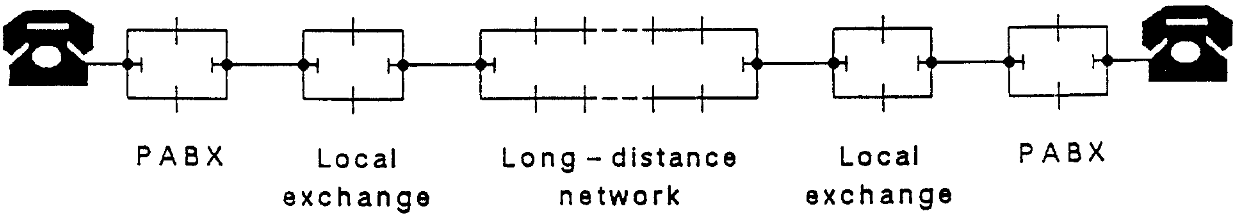


Figure 2.6/3 A worst-case configuration of five 4-wire loops

For data transmission at a bit rate of up to 2.4 kbit/s, the OLL of each 4-wire loop at any frequency in the band of 500 Hz to 2500 Hz should not be less than 25 dB. This value is based on the following formula:

$$OLL = (18 + 10 \log m)\ dB$$

where m = total number of 4-wire loops

For speech services an OLL of each 4-wire loop should be greater than (11 + 10 log m) dB. This is for further study.

2.6.3 Attenuation of 4-wire Loops

If FDM or PCM multiplex equipment or a digital exchange, or a combination of both, is inserted between 2-wire points, a 4-wire loop is formed. To secure an adequate singing margin and listener echo, some countries provide for an extra transmission loss between the 2-wire points (in contrast to normally about 1 dB in a 2-wire analogue exchange). However, the loss plan may not permit such a loss for an isolated exchange in a 2-wire environment. In such a case, a possible solution is to extend the 4-wire loop towards the analogue exchange where PCM multiplex equipment will be placed (see Figure 2.6/4). In case b) such a loss is generally permissible since it replaces the loss of the analogue 2-wire inter-line exchange circuit of case a). This type of solution is also applicable to PBNs.

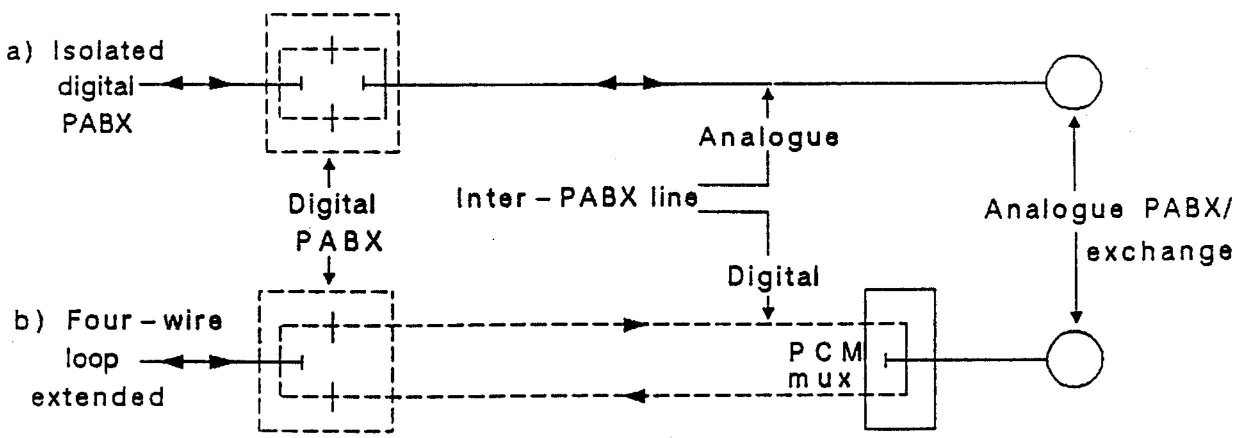


Figure 2.6/4 Digital PABX in a 2-wire environment

2.7 Quantizing Distortion

2.7.1 General Information

The incorporation of 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.

By definition, an average 8-bit codec pair (A/D and D/A conversions, A-law or u-law) which complies with CCITT Recommendation G.711 introduces 1 quantizing distortion unit (1 qdu). An average codec pair produces about 2 dB less quantizing distortion than the limits indicated in CCITT Recommendation G.712. This would correspond to a signal-to-distortion ratio of 35 dB for the sine-wave test method (see CCITT Recommendation G.113).

From the point of view of quantizing distortion, CCITT Recommendation G.113 recommends that no more than 14 units of quantizing distortion should be introduced in an international telephone connection.

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 qdu if it is equivalent to n 8-bit PCM processes in tandem.

2.7.2 Sources of Quantizing Distortion

The units of quantizing distortion tentatively assigned to a number of digital processes are given in Table 2.7/1.

Digital process	Quantizing distortion units
8-bit PCM codec pair (G.711 A- or u-law)	1
Transmultiplexer pair based on 8-bit PCM	1
32 kbit/s ADPCM (G.721)	3.5
Digital loss pad	0.7
A/u-law or u/A-law converter	0.5
A/u/A-law tandem conversion	0.5
u/A/u-law tandem conversion	0.25
PCM to ADPCM to PCM tandem conversion	2.5

Table 2.7/1 Planning values for quantizing distortion

Detailed information about planning values for quantizing distortion can be found in CCITT Recommendation G.113. The given values are only valid for speech transmission.

2.7.3 Apportionment of qdu's

The total quantizing distortion introduced by digital processes in international telephone connections should be limited to a maximum of 14 units. It is expected that the accumulated attenuation distortion and the accumulated group-delay distortion introduced by digital processes in such connections would also be kept within acceptable limits.

The number of units of transmission impairment in an international telephone connection should not exceed:

$$5 + 4 + 5 = 14 \text{ units}$$

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

In the mixed analogue/digital period it is exceptionally allowed to introduce seven units of transmission impairment in the national system. Theoretically, this could result in international connections with a total of 18 units of transmission impairment.

At the beginning of the evolution towards a digital PSTN a maximum of 3 qdu's may be expected between the LE and the·ISC, in which case 2 qdu's may be allowed within the PBN. At the end of that evolution the qdu's allowed for the PBN could be modified. This is for further study.

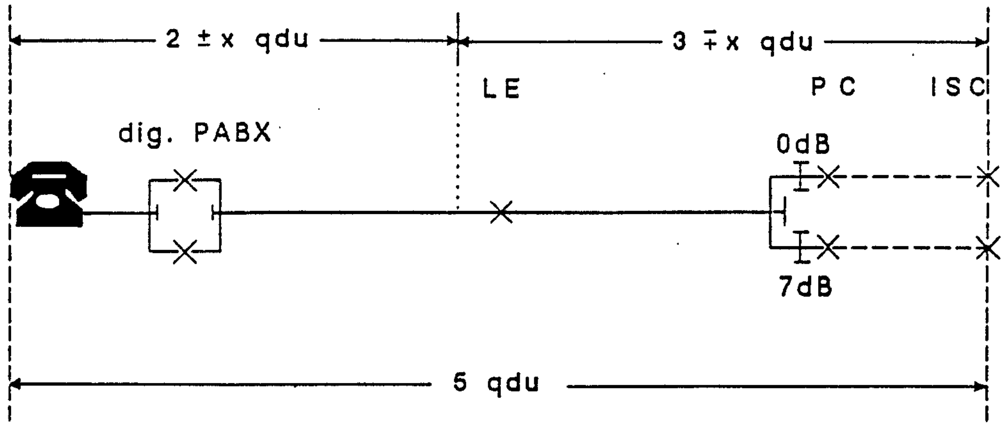


Figure 2.7/1 A possible division of the qdu in the national system

In practice this rule applies only to fixed connections. In mobile radio systems, including cordless telephones and cordless PABXs, low rate encoding techniques are frequently necessary to ensure efficient use of the radio spectrum. In these cases the qdu allowances may be exceeded. The specification of appropriate rules is for further study.

2.8 Noise

2.8.1 General Information

The recommended total weighted noise contributed by a PBN is as follows:

a. A fully digital PBN should be designed so that total noise is no worse than that from a single digital PBX, i.e. 500 pWp.

b. Where a PBN has mixed analogue/digital systems, it is recommended that the limit of total noise should be in the range of 1000 to 2000 pWp.

These figures take into account an expected improvement in the noise performance in public networks as a result of the digitalization process, and therefore they are consistent with the total noise limit set in CCITT Recommendation G.103 (10.000 pWp in a reference connection of 2500 km). Account is also taken of the principles set out in CCITT Recommendation G.171.

2.8.2 Treatment of noise within a PBN

It is rarely likely to be practicable to use normal network planning methods to determine the noise transmitted by a private network. Although it would be possible to measure the noise level transmitted to each public network interface for each relevant call path through the PBN, this is a very cumbersome procedure for private networks other than the smallest. It is therefore considered that noise should be controlled by specifying appropriate noise limits in the standards to which individual items of network apparatus must conform (e.g. for digital PABXs, the ETS TE 10-02, TE 10-03).

The PBN noise limit assumed in Fig. 1 of CCITT Rec. G.103 may prove to be too exacting when the realistic noise performance of PBNs is considered. In that case it may be appropriate to propose a change to the CCITT Study Group responsible.

2.9 Attenuation Distortion

This subject is presently under discussion in CCITT Study Group XII.

3 Specific Recommendations

3.1 General Remarks

For the planning of Private Branch Networks (PBN's) it is advisable to distinguish between the type of access to the public network. The types of access are:

- Digital access with a wholly digital call path within the ISDN/PSTN
- Digital access with a mixed analogue and digital call path within the ISDN/PSTN
- Analogue access with a wholly digital call path within the ISDN/PSTN
- Analogue access with a mixed analogue and digital call path within the PSTN.

The recommendations given in this Section apply only to calls passing through a public network from a network connection point to an international switching centre, and refer to the network connection point (NCP) unless otherwise specified.

The requirements in this Section should not automatically be applied to existing installations in retrospect. Also, one must keep in mind that details of the transmission arrangements between a public and a private network operator may be settled by separate negotiations.

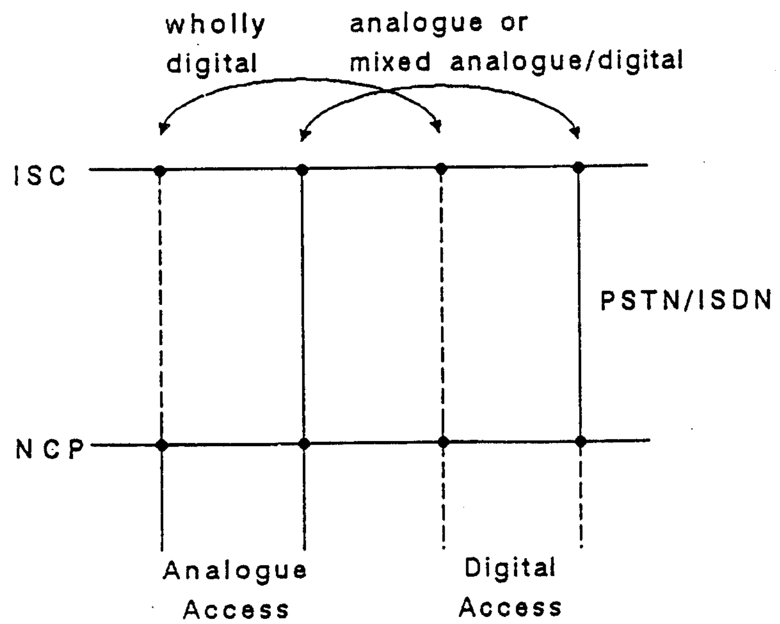


Figure 3.1/1 Possible Accesses to the Public Network

The following factors should be taken into account:

- Number of qdu
- Echo control
- Stability
- Loudness Rating.

The planning must take all types of access into account. Figure 3.1/1 illustrates the types of access under consideration.

For access to the ISDN, Section 3.2 gives planning rules which will provide basic harmonisation for the operation of PBN's in Europe. National administrations may, however, allow PBN's to have greater levels of impairments provided that the transmission performance of the national system up to the ISC is maintained. Sections 3.3 and 3.4 give information and guidance, including examples of planning rules for call paths which may include analogue sections either in the PBN or PSTN, but because of the historical differences in the design of national systems, national administrations may choose different national rules which are more appropriate for their countries.

For any particular digital interface to the PSTN or ISDN, the public network path for incoming and outgoing calls may be different, and planning aspects of incoming and outgoing calls may need to be included separately in Section 3.2 or 3.3. It should also be noted that digital ISDN interfaces conforming to NET 3 or NET 5 do not guarantee wholly digital paths from the ISDN interface to the ISC. PBN operators will need to obtain information from the relevant public network operator or to make a deliberate selection of a particular call category in order to ensure that the path from the public network interface to the ISC is wholly digital and that Section 3.2 applies.

In practice it may be very difficult for all call paths within a private network to conform to these recommendations. Either the propagation of traffic at each NCP on call paths which do conform should not exceed a small percentage of total traffic or alternatively the initial call set up or answer should take place from a point where the call path does conform to the above recommendations, with the call subsequently being extended and more impairments being added.

3.2 Digital Access with a wholly digital Call Path within the ISDN/PSTN

3.2.1 Definition and Reference Connection

Section 3.2 covers the case of a digital access at the NCP where it is known that the call path within the ISDN/PSTN up to the ISC is a wholly digital 64 kbit/s channel without code conversion (see Figure 3.2/1). The nature of the call path within the ISDN/PSTN may be known either as a result of using a facility in the signalling system, such as a particular service indicator code in ISDN, or as a result of knowledge of the structure of the public network. If the nature of the call path in the public network is not known with certainty, then the private network must be designed to satisfy Section 3.3.

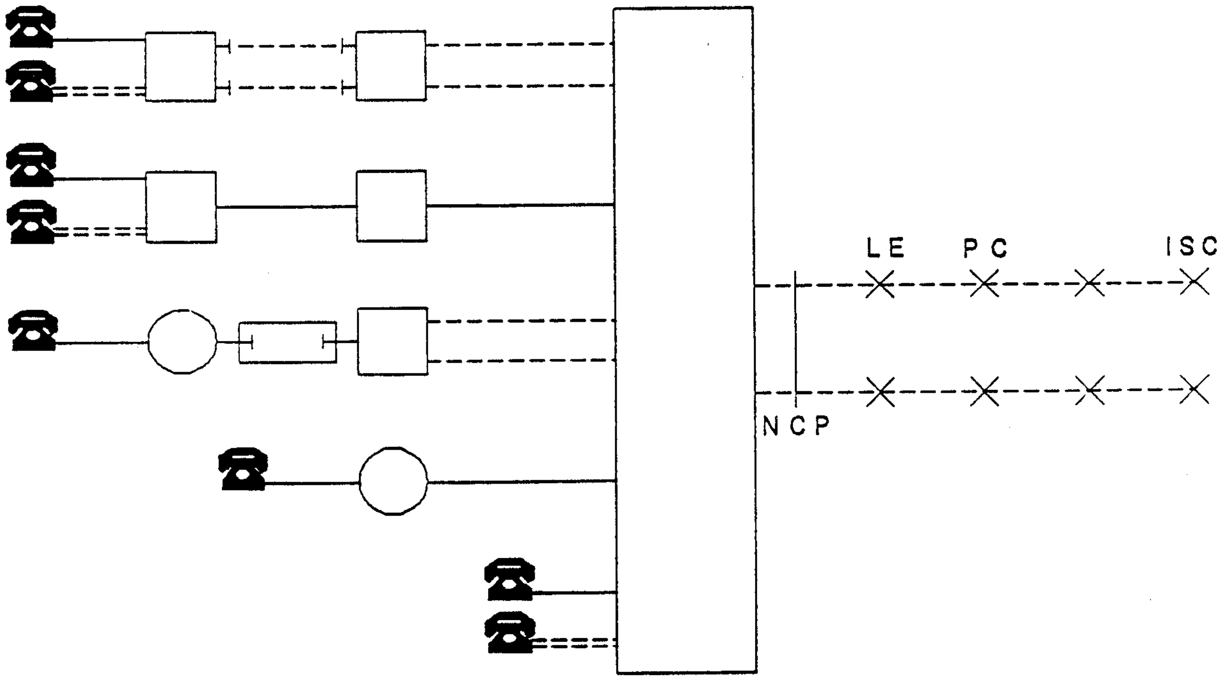


Figure 3.2/1 Reference Connection with a wholly digital call path within the ISDN/PSTN

3.2.2 Number of qdu

All possible routings within a mixed digital/analogue PBN to the PSTN/ISDN must be considered because distortion is introduced at each analogue/digital conversion. The insertion of up to 5 qdu is allowed for each call path within the PBN. Only for a small percentage of traffic it is permitted to insert up to 7 qdu. Radio systems are not covered by this rule (see Section 2.7.3).

3.2.3 Echo Control

Note 1: The following rules are based on the assumption that the one-way propagation time between the far-end talker and the NCP does not exceed 20 ms.

Note 2: For the Bearer Service Category "Circuit-mode 64 kbit/s, unrestricted" it is recognised that it is the responsibility of the customer to ensure that a compatible encoding scheme is in operation and that no network provision can be made for the control of such items as echo and loss.

3.2.3.1 Propagation Time

The private network shall be designed so that for each possible call path without echo control, the propagation time between the user apparatus and the NCP or vice versa should not exceed x ms; the standard value of x is 5 ms in average sized countries.

However, national administrations may specify other values which may vary from area to area of their country, according to the formula specified in Section 2.3.4

$$t = (25 - t_p)/2 \text{ ms}$$

In calculating the propagation time the mean delay values for the items given in Section 2.3.3 should be used.

3.2.3.2 Echo Loss of the PBN seen from the NCP

For each possible echo path, the echo loss seen from the NCP should exceed the values given in Table 3.2/2 corresponding to the number of 4-wire loops in the call path within the private network.

Two sets of figures are given in the Table below. The first set gives the long-term objective based on a TELR = 34 dB and talker SLR/RLR values of 7 dB and 3 dB respectively. In many countries it may be difficult to achieve these values in a short time and so the second set of figures gives lower values for the shorter term. However, as a result of national circumstances these values may need to be modified further in certain countries to produce a practical rule. The definition of short and long-termwill also depend on national circumstances (see also the note given in the Appendix).

Note: In countries where the SLR/RLR values at the NCP are lower than 7 dB and 3 dB respectively the TELR values may be lower than desirable and administrations may need to discuss bilaterally with each other to decide what steps may need to be taken to increase the TELR to a satisfactory level.

Number of 4-wire loops	Echo loss long-term Objective	Echo loss short-term Objective
1	24 dB	20 dB
2	26 dB	22 dB
3 or more	27 dB	23 dB

Table 3.2/2 Echo loss for each echo path

Note: Echo loss is defined in Annex B. The second paragraph of the definition gives a simple rule of its calculation.

3.2.3.3 Echo Loss wholly within the PBN

For each possible echo wholly within the PBN, the talker echo loudness rating shall exceed 15 dB. (Note: this value is lower than the requirement for the echo given above, because the total round trip delay for the whole echo path cannot exceed 10 ms).

Note: These rules for talker echo should also ensure that the listener echo is satisfactory.

In addition to the electrical echo paths of the PBN seen from the NCP, there will also be an acoustic echo path via the terminal apparatus. However the treatment of acoustic echo is for further study.

In any call path where either the propagation time of 5 ms specified above is exceeded, or where the echo path loss does not exceed the value in the Table above and cannot be increased, either an echo control device must be used or all electrical echo sources must be eliminated from the call path. (For further information about the use of echo control devices, see Section 2.4.)

3.2.4 Stability

In order to ensure stability, the loss in the shortest electrical echo path in the private network seen from the NCP must be considered at all frequencies, taking account of possible tolerances and variations in loss in any analogue equipment. For more information see Section 2.5.

3.2.5 Loudness Ratings

At the NCP the nominal SLR and RLR values for each call path within the PBN and for each type of telephone set shall lie in the following ranges:

SLR: 7 dB to 16.5 dB
RLR: 3 dB to 12.5 dB

To meet the CCITT's long-term objectives, the PBN should be designed to evolve towards nominal values close to SLR = 7 dB and RLR = 3 dB.

Taking the loudness ratings of existing analogue telephones (as specified in present national standards for telephones intended for connection to the public telephone network) into account it may be necessary to accept lower values of loudness ratings at the NCP.

3.3 Digital Access with a mixed analogue and digital Call Path within the ISDN/PSTN

3.3.1 Definition and Reference Connection

Section 3.3 covers the case of a digital access at the NCP where it is known that the call path within the ISDN/PSTN up to the ISC is a mixed analogue and digital channel (see Figure 3.3/1 and 3.3/2).

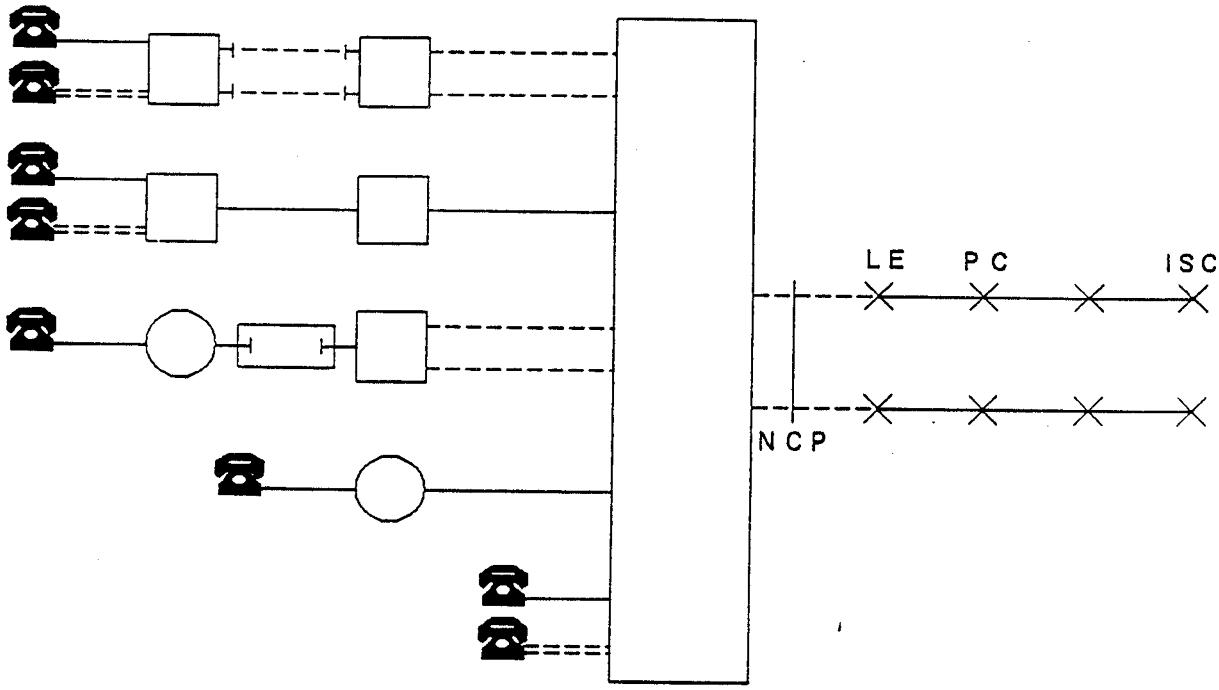


Figure 3.3/1 Reference Connection with a wholly 4-wire analogue and digital call path within the ISDN/PSTN

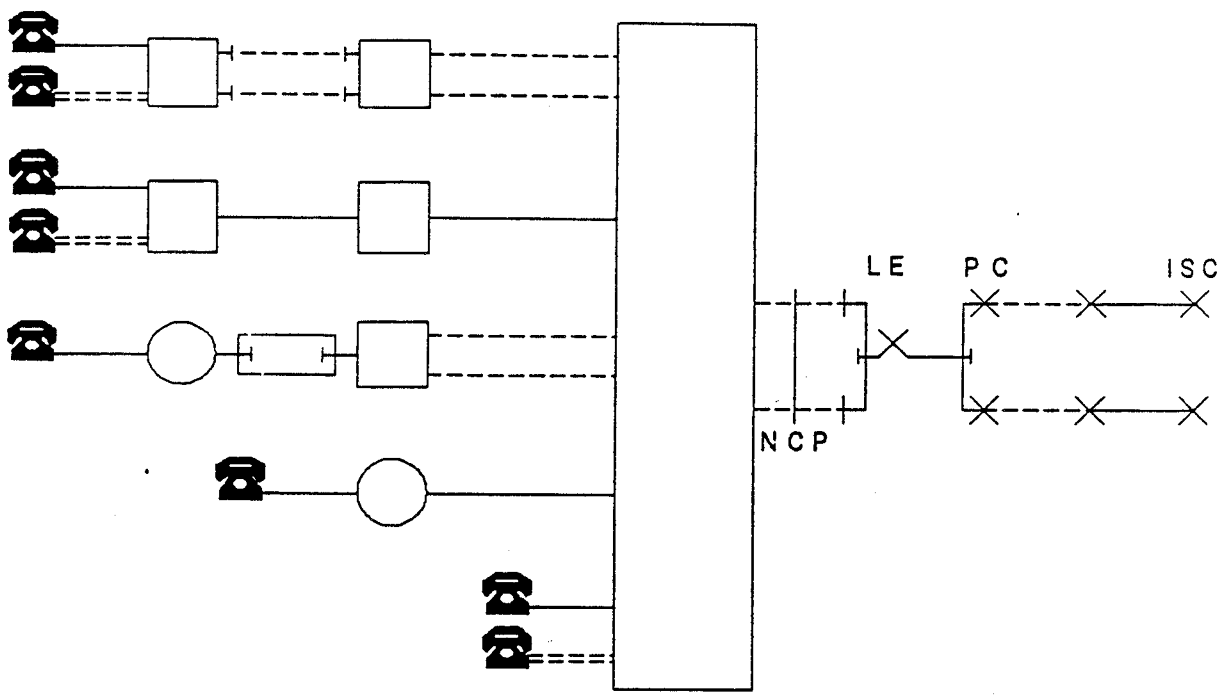


Figure 3.3/2 Reference Connection with an analogue and digital call path within the ISDN/PSTN which includes a 2-wire section

3.3.2 Number of qdu

Each call path within the PBN should insert no more than 1.5 qdu, but certain call paths which in total carry a small proportion of the total PBN traffic may insert 2.5 qdu. In addition, these qdu values may be increased in specific areas (LE's) by an amount which takes account of the extent of the digitalization of the public network. Radio systems are not covered by this rule (see Section 2.7.3).

3.3.3 Echo Control

Figures 3.3/1 and 3.3/2 show the 2 possible cases for echo. National administrations should declare whether the call path within the PSTN is fully 4-wire (Figure 3.3/1) or a combination of 4- and 2-wire (Figure 3.3/2).

3.3.3.1 Propagation Time

The propagation time limits should be in accordance with Section 3.2.3.1.

3.3.3.2 Echo Loss of the PBN seen from the NCP

The echo loss limits applicable to a fully 4-wire path within the PSTN are given in Section 3.2.3.2.

National administrations may need to alter the echo loss values to take account of the number and loss of analogue 4-wire circuits in the PSTN path.

For the case of a mixed 4-wire/2-wire path within the PSTN the limits for the echo path loss are under study.

3.3.3.3 Echo Loss wholly within the PBN or PSTN access

For each possible echo wholly within the PBN, the talker echo loudness rating shall exceed 15 dB. (Note: This value is lower than the echo requirement given above, because the total round trip delay for the whole echo path cannot exceed 10 ms).

In the case of a mixed 4-wire/2-wire call path in the PSTN access, the PSTN should present to the PBN at the NCP an echo loss due to any near-end hybrid of at least 5 dB and this value should be included in the calculation of the talker echo loudness rating.

National administrations may need to increase this value if the hybrid in the PSTN access is not close to the NCP.

3.3.4 Stability

In the case that the call path within the PSTN is fully 4-wire, in order to ensure stability the loss in the shortest electrical echo path in the private network seen from the NCP must be considered at all frequencies, taking account of possible tolerances and variations in loss in any analogue equipment. For further information see Section 2.5.

3.3.5 Loudness Ratings

At the NCP the nominal SLR and RLR values for each call path within the PBN and for each telephone set shall lie in the following ranges:

SLR:	7 dB	to	(16.5 - x) dB
RLR:	3 dB	to	(12.5 - y) dB

where x and y are the circuit loudness ratings in the send and receive directions respectively between the NCP and the ISC. The values x and y should be negotiated between the public and private network operator.

For the reference connections given in Figure 3.3/1, the values x and y are 0 dB.

Taking the loudness ratings of existing analogue telephones (as specified in present national standards for telephones intended for connection to the public telephone network) into account it may be necessary to accept lower values of loudness ratings at the NCP.

3.4 Analogue Access to the PSTN with a wholly digital Call Path within the PSTN

3.4.1 Definition and Reference Connection

Section 3.4 covers the case of an analogue 2-wire access at the NCP where it is known that the call path within the ISDN/PSTN up to the ISC is a wholly digital 64 kbit/s channel without code conversion (see Figure 3.4/1). The nature of the call path within the ISDN/PSTN may be determined from knowledge of the structure of the public network. It is assumed that there is only one analogue/digital conversion between NCP and the ISC. If the nature of the call path in the public network is not known with certainty, then the private network must be designed to satisfy Section 3.5.

This Section does not cover the case of an analogue 4-wire access which may exist in some countries and for which planning rules need to be determined nationally.

Note: The call path between the NCP and the LE may not be a wholly 2-wire path.

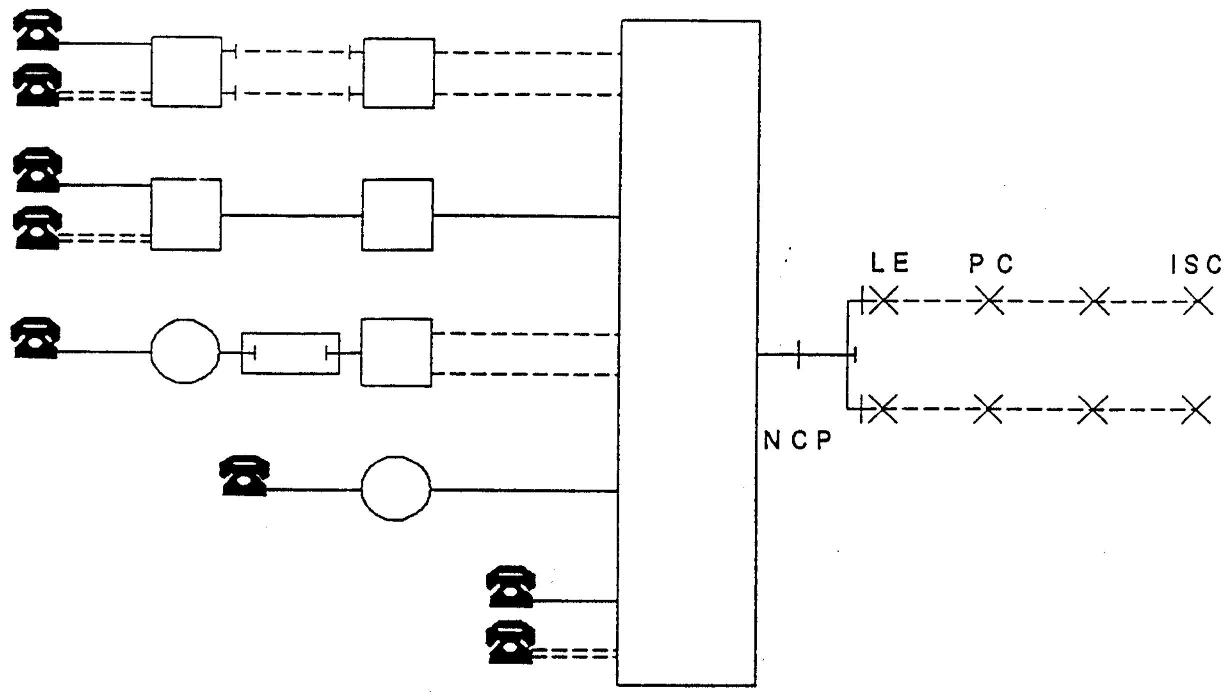


Figure 3.4/1 Reference Connection for an analogue access with a wholly digital call path within the ISDN/PSTN

3.4.2 Number of qdu

Each call path within the PBN should insert no more than 4.5 qdu, but certain call paths which in total carry a small proportion of the total PBN traffic may insert 6.5 qdu. Radio systems are not covered by this rule (see Section 2.7.3).

3.4.3 Echo Control

The requirements for the echo loss of the PBN seen from the NCP are to be determined according to national circumstances. The requirements for echo loss on echo paths wholly within the PBN are given in Section 3.3.3.3.

The propagation time limits should be in accordance with Section 3.2.3.1.

3.4.4 Stability

No requirements.
(The stability requirement for the international section of the call will be met by the public network).

3.4.5 Loudness Ratings

At the NCP the nominal SLR and RLR values for each call path within the PBN and for each type of telephone set should not exceed

SLR: (16.5 - x) dB
RLR: (12.5 - y) dB

where x and y are the circuit loudness ratings in the send and receive directions respectively between the NCP and the ISC. The exact value for x and y should be negotiated between the public and the private network operator.

3.5 Analogue Access to the PSTN with a mixed analogue and digital Call Path within the PSTN

3.5.1 Definition and Reference Connection

Section 3.5 covers the case of an analogue access at the NCP where it is known that the call path within the ISDN/PSTN up to the ISC is a mixed analogue and digital channel (see Figure 3.5/1).

Note: The call path between the NCP and the LE may not be a wholly 2-wire path.

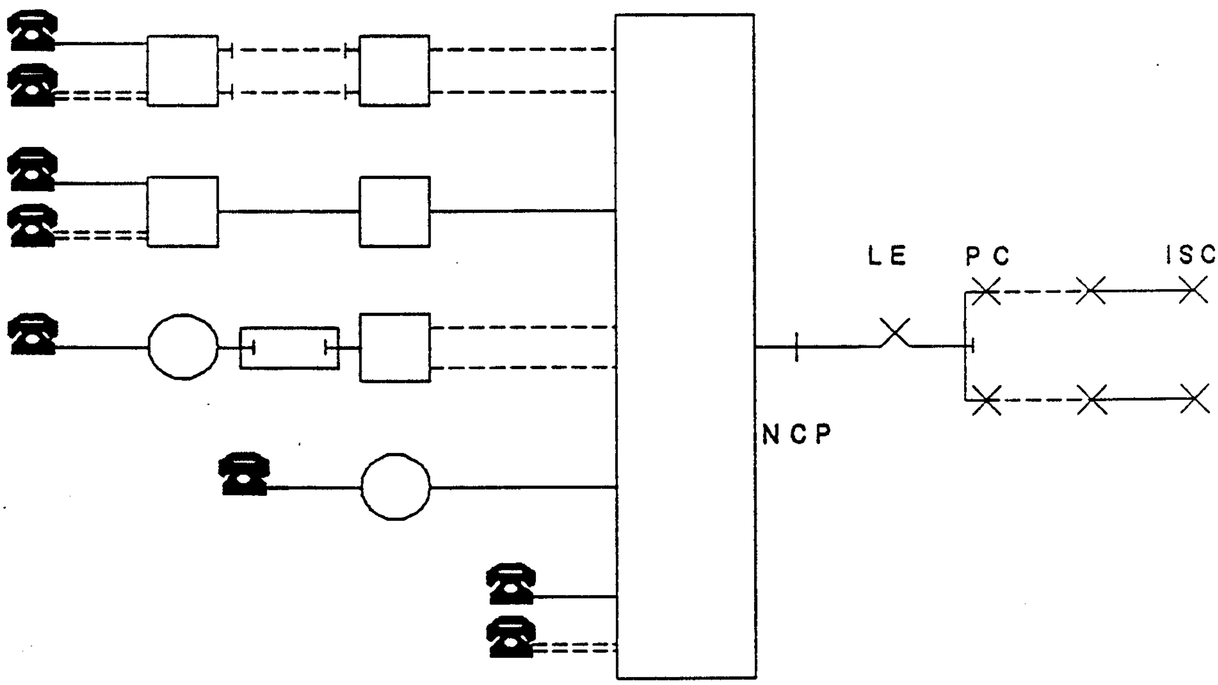


Figure 3.5/1 Reference Connection with an analogue and digital call path within the ISDN/PSTN

3.5.2 Number of qdu

Each call path within the PBN should insert no more than 2 qdu, but certain call paths which in total carry a small proportion of the total PBN traffic may insert 3 qdu.

The value may be altered by negotiation between the public and private network operators according to local circumstances. Radio systems are not covered by this rule (see Section 2.7.3).

3.5.3 Echo Control

See Section 3.4.3.

3.5.4 Stability

No requirements (see Section 3.4.4).

3.5.5 Loudness Rating

At the NCP the nominal SLR and RLR values for each call path within the PBN and for each type of telephone set should not exceed

SLR: (16.5 - x) dB
RLR: (12.5 - y) dB

where x and y are the circuit loudness ratings in the send and receive directions respectively between the NCP and the ISC. The exact value for x and y should be negotiated between the public and the private network operator.

Annex A

Abbreviations

ADPCM	Adaptive Differential PCM
AEC	Acoustic Echo Control
ATM	Asynchronous Transfer Mode
CCITT	International Telegraph and Telephone Consultative Committee
CLR	Circuit Loudness Rating
CRE	Corrected Reference Equivalent
EEC	Electric Echo Control
ECD	Echo Control Device
GSM	Special Mobile Group
ISC	International Switching Centre
ISDN	Integrated Services Digital Network
ISBPX	PABX with ISDN Services
LE	Local Exchange
LR	Loudness Rating
LS	Local System
LSTR	Listener's Sidetone Rating
MDF	Main Distribution Frame
MTU	Main Terminating Unit
NCP	Network Connection Point
NT	Network Termination
OLR	Overall Loudness Rating
OLL	Open Loop Loss
PABX	Private Automatic Branch Exchange
PBN	Private Branch Network
PC	Primary Centre
PCM	Pulse Code Modulation
PLMN	Public Land Mobile Network
PSTN	Public Switched Telephone Network
qdu	Quantizing Distortion Unit
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
STMR	Sidetone Masking Rating
TELR	Talker's Echo Loudness Rating
VASP	Virtual Analogue Switching Point

Annex B

Terms and Definitions

Attenuation/Frequency Distortion

"Attenuation/frequency distortion" or "loss distortion" is the result of imperfect amplitude/frequency response and is generally specified in addition to the relative levels of a transmission section, from which the nominal transmission loss is derived. The definition of the attenuation/frequency distortion (LD) is the difference between the actual response of voltage versus frequency U(f) and the ideal (planned) response of voltage versus frequency $U^*(f)$, referred to the corresponding difference at 1020 Hz.

$$LD = 20 \log \left| \frac{U(1020\ \text{Hz})}{U(f)} \right| - 20 \log \left| \frac{U^*(1020\ \text{Hz})}{U^*(f)} \right|$$

For practical reasons the ideal response of voltage versus frequency $U^*(f)$ is flat. Taking this into account, the equation reduces to

$$LD = 20 \log \left| \frac{U(1020\ \text{Hz})}{U(f)} \right|$$

It should be noted that the equation for LD is valid regardless of whether Z_{o1} is equal to Z_{o2} or not. However, impedance matching at input and output is assumed.

Balance Return Loss

At a 4-wire terminating set ("hybrid"), 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 terminal, and the balance impedance, Z_B. It is given approximately by the expression:

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

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

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

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

Channel, Transmission Channel

A means of unidirectional transmission of signals between two points.

Circuit, Telecommunication Circuit

A combination of two transmission channels permitting bidirectional transmission of signals between two points, to support a single communication.

Note 1: In a telecommunication network, the use of the term "circuit" is generally limited to a telecommunication circuit directly connecting two switching devices or exchanges, together with associated terminating equipment.

Note 2: A telecommunication circuit may permit transmission in both directions simultaneously (duplex), or not simultaneously (simplex).

Composite Loss

The composite loss of a quadripole inserted between two impedances Z_E (of the generator) and Z_R (of the load) is the expression in transmission units of the ratio P_E/P_R, where

P_E is the apparent power that the generator Z_E would furnish to a load of impedance Z_E.

P_R is the apparent power that the same generator furnishes via the said quadripole to the load Z_R.

If the number thus obtained is negative, then there is a composite gain.

Connection (in telecommunication)

An association of transmission channels or circuits, switching and other functional units set up to provide means for the transfer of information between terminals in a telecommunication network.

Note 1: A connection is the result of a switching operation.

Note 2: A connection which allows an end-to-end communication, e.g. conversation, may be called a "complete connection".

Note 3: A connection makes a communication possible but is not a communication.

Echo Balance Return Loss

The echo balance return loss is the balace return loss averaged with 1/f power weighting over the telephone band, in accordance with Section 4 of CCITT Recommendation G.122.

Echo Loss (L_e)

The echo loss is the semi-loop loss averaged with 1/f power weighting over the telephone band, in accordance with Section 4 of CCITT Recommendation G.122.

In the case where a 2-wire point exists, the echo loss is approximately equal to the sum of the transmission losses from 4-wire to 2-wire and 2-wire to 4-wire and the echo balance return loss.

Extension Line

The extension line is the connection between the terminal and the PABX.

Integrated Services Digital Network (ISDN)

An integrated services network that provides digital connections between user-network interfaces. An integrated services network provides or supports a range of different telecommunication services.

Inter-PABX line

The inter-PABX line connects two PABXs together in a permanent way, e.g. a leased line.

Local System (LS)

The local system consists of the telephone set, extension line, one or more PABXs, inter-PABX line and the subscriber line (see Figure B1).

Loudness Ratings

Within the context of CCITT, a Loudness Rating is an objective measure of the loudness loss, i.e. a weighted, electro-acoustic loss between certain interfaces in the telephone network.

An important attribute of the Loudness Rating is the additivity. This means that if the circuit between the interfaces is subdivided into sections, the sum of the individual section Loudness Ratings is equal to the Overall Loudness Rating.

Loudness Ratings provide a logical basis for judging both the wanted transmission from the talking to the listening subscriber as well as some unwanted phenomena such as excessive sidetone, echos and crosstalk.

Useful Loudness Ratings are:

Overall Loudness Rating OLR

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

Send Loudness Rating SLR

The loudness loss between the speaking subscriber's mouth and an electric interface in the network.

Receive Loudness Rating RLR

The loudness loss between an electric interface in the network and the listening subscriber's ear.

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.

Sidetone Masking Rating STMR (Talker's sidetone)

The loudness loss between a subscriber's mouth and his earphone via the electric sidetone path.

Listener's Sidetone Rating LSTR

The loudness loss between a Hoth-type room noise source and the subscriber's earphone via the electric sidetone path.

Talker's Echo Loudness Rating TELR

The loudness loss between a subscriber's mouth and his earphone via the delayed echo path.

Main Terminating Unit (MTU)

The main terminating unit is a 2-wire/4-wire hybrid which terminates the long-distance 4-wire loop. The MTU can be part of the PSTN or PBN if it is connected via 4-wire circuits to the PSTN/ISDN. The MTU has to fulfil the requirements for stability and echo.

National System

The national system starting at the ISC may comprise one or more 4-wire national trunk circuits with 4-wire interconnection, as well as circuits with 2-wire connection to the local exchange, the subscriber lines and the PBN (see Figure B1).

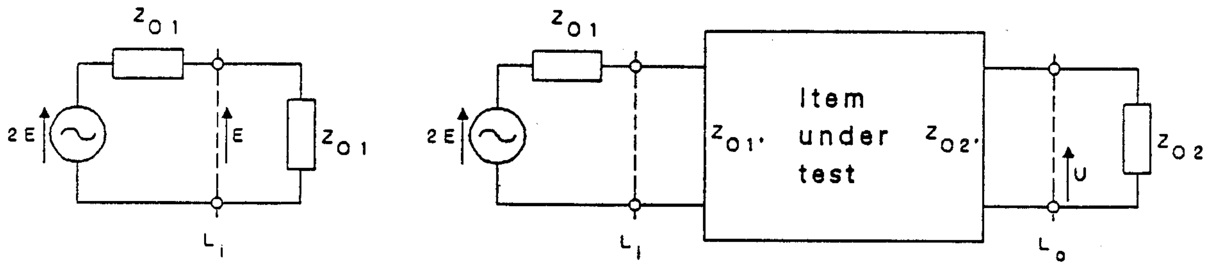
Network Connection Point (NCP)

A network connection point is a point at which two separately run networks are interconnected. In the context of this document NCP refers to the connection point between a PBN and PSTN/ISDN.

Nominal Transmission Loss

The nominal transmission loss (NL) is defined between two points as the difference between the relative levels associated with these points. If, for instance, the relative level at the input is L_i, and at the output, L_o, then the nominal transmission loss is

$$NL = L_i - L_o$$



Power reference circuit Test circuit

Figure B2 Power Reference Circuit and Test Circuit

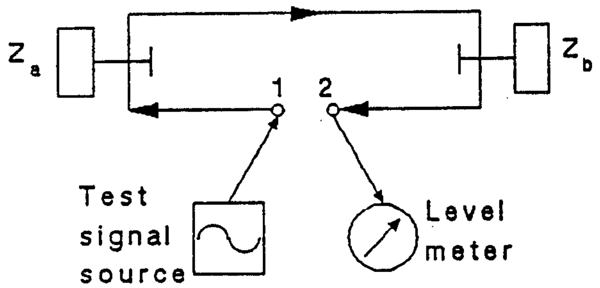
Taking into account that, according to the definition of the power reference circuit, E is frequency-independent, one obtains the nominal transmission loss

$$NL = 20 \log \left| \frac{E}{U(1020\ Hz)} \right| + 10 \log \left| \frac{Z_{o2}\ (1020\ Hz)}{Z_{o1}\ (1020\ Hz)} \right|$$

It should be noted that the NL represents the "composite loss" (ITU definition 05.20) at 1020 Hz. The composite loss is the only measure of attenuation that allows adding of the losses of "half connections", regardless of the specific impedances at the input and output ports.

Open-Loop Loss (OLL)

In a loop formed by a 4-wire circuit (or cascade connection of two or more 4-wire circuit) and terminated by 2-wire ends (i.e. having 4-wire terminating sets, or hybrids, at both ends), the loss is measured by breaking the loop at some points, injecting a signal and measuring the loss incurred in traversing the open loop. All impedance conditions should be preserved whilst making the measurement. In practice the OLL is equal to the listener echo loss. The OLL is also equal to the sum of the two semi-loop losses associated with the loop.



1 and 2 are points with equal relative levels, e.g. digital points.

Figure B3 Measurement of the OLL

Private Branch Network (PBN)

A private branch network comprises a private telecommunication network (PTN) and the connected terminal equipment for the speech service.

Propagation Time

Time lost between the transmission of a signal and the time it is received due only to delays in the transmission medium itself.

There are several terms (e.g. absolute delay, envelope delay, group delay) with nearly the same meaning.

Public Switched Telephone Network (PSTN)

The term is used to describe the ordinary telephone system including subscriber lines, local exchange and the complete system of trunks and the exchange hierarchy which makes up the network (see Figure B1).

Relative Level

The relative levels (L_r) - even at ports of complex impedances - relate to power (in general, apparent power) at a reference frequency of 1000 Hz. Accordingly, at a point of zero relative level (i.e. transmission reference point, cf. CCITT Recommendation G.101, § 5.3.1) and at an impedance Z, the reference power of 1 mW (at 1000 Hz) corresponds to the voltage:

$$U_o = \sqrt{1 \text{ mW } |Z|}$$

It follows that generally at a point of relative level L_r the voltage will be

$$U = 10^{L_r/20} \sqrt{1 \text{ mW } |Z|}$$

and that consequently the relative level L_r can be expressed as

$$L_r = 20 \log \frac{U}{\sqrt{1 \text{ mW } |Z|}} \text{ in dBr}$$

This is the basis for a coherent definition of the transmission loss, and subsequently of attenuation/frequency distortion.

Also, the expression

$$L_r = 10 \log \frac{P}{P_0} \text{ in dBr}$$

is used, 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. The quantity is independent of P_0, it is a composite loss (level difference). For further details, see CCITT Recommendation G.101, § 5.3.2

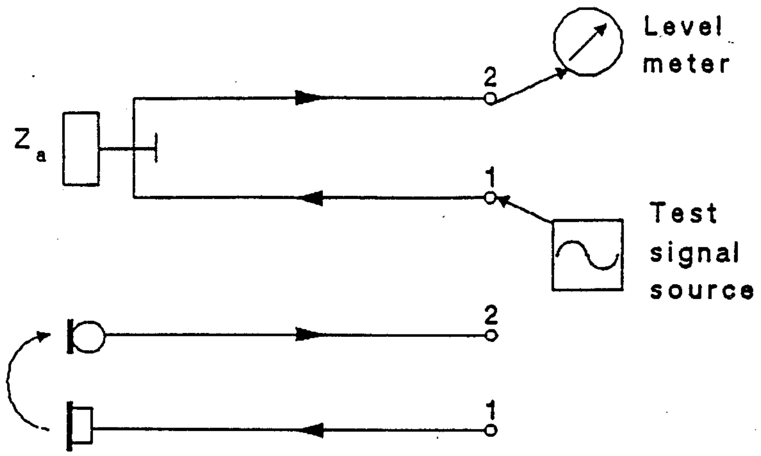
Return Loss

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

$$L_R = 20 \log \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| \text{ in dB}$$

Semi-Loop Loss

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 accoustical coupling), the semi-loop loss is the loss measured between input and output:



1 and 2 are points with equal relative levels, e.g. digital points.

Figure B4 Measurement of the semi-loop loss

The semi-loop loss is an important quantity in determining echo balance return loss, echo loss and open-loop loss. 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-relative level points in an ISC.

Stability loss

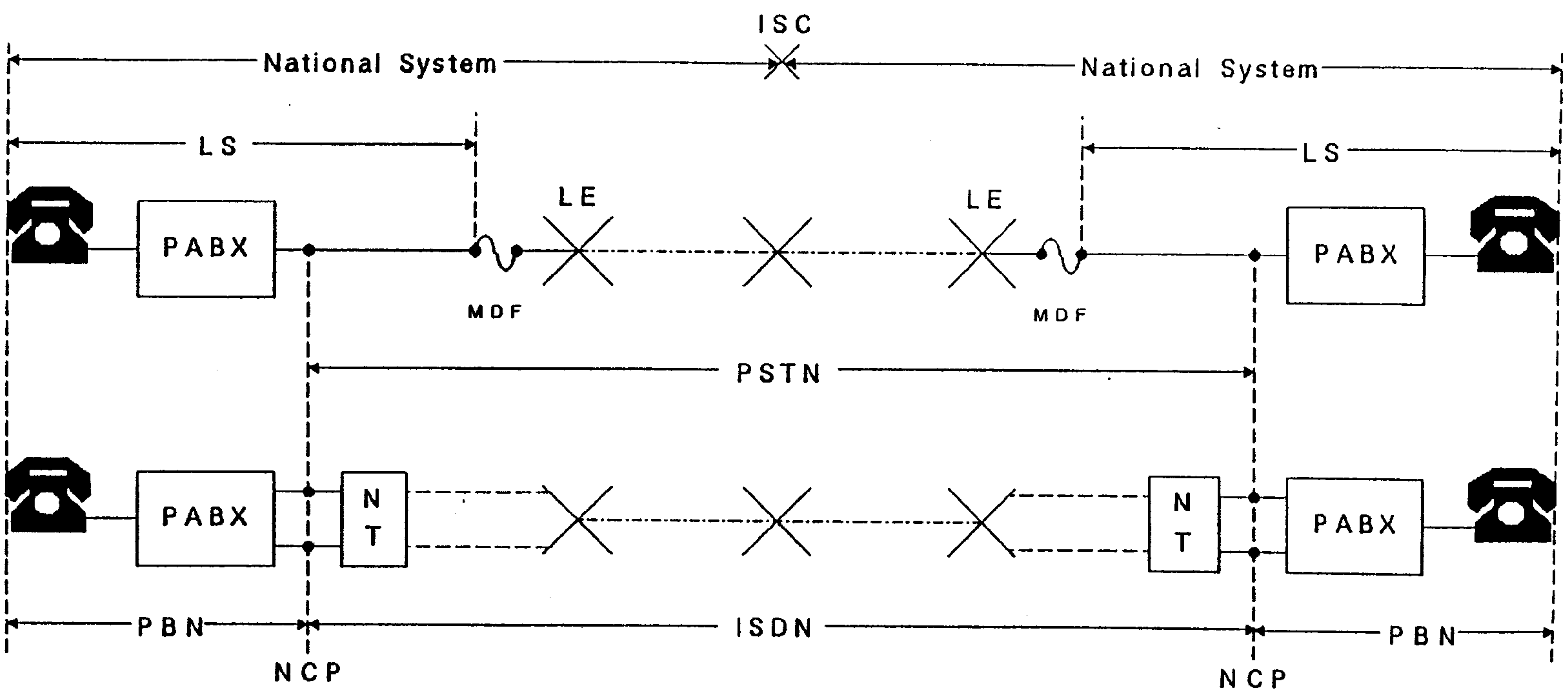
Stability loss is the lowest value of the semi-loop loss in the frequency band to be considered under worst case conditions (e.g. open- and short-circuit terminations).

Subscriber Line

The transmission link between a subscriber's terminal (telephone set, PABX etc.) and the local exchange with which it is permanently associated.

Virtual Analogue Switching Point (VASP)

The virtual analogue switching point is a theoretical analogue point with specified relative levels (-3.5 dBr/-3.5 dBr for a digital international circuit). The division between the national system and the international chain is determined by the VASP in the originating/terminating international switching centre.



ISC	International Switching Centre
ISDN	Integrated Services Digital Network
LE	Local Exchange
LS	Local System
MDF	Main Distribution Frame
NT	Network Termination
PBN	Private Branch Network
NCP	Network Connection Point

Figure B1 Definition of network-related terms

Appendix

Note regarding Deutsche Bundespost TELEKOM's long-term echo loss objectives for PBN's

From 1992, Deutsche Bundespost TELEKOM will accept only PBN's with access to its ISDN which fulfil the long-term objectives. A possible solution for a satisfactory TELR value for analogue telephone sets is to adjust the relative levels +3 dBr/-10 dBr at L2-interfaces of a PABX.

History

Document history	
August 1990	First Edition
March 1996	Converted into Adobe Acrobat Portable Document Format (PDF)