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International telephone connections and circuits – General definitions

Application of the E-model: A planning guide

ITU-T Recommendation G.108

(Previously CCITT Recommendation)

ITU-T G-SERIES RECOMMENDATIONS

TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

For further details, please refer to ITU-T List of Recommendations.

ITU-T RECOMMENDATION G.108

APPLICATION OF THE E-MODEL: A PLANNING GUIDE

Summary

The intent of this Recommendation is to demonstrate how the E-Model (described in Recommendation G.107 [3]) can be used in end-to-end transmission planning for a wide range of networks – local, national, multinational and transcontinental.

Source

ITU-T Recommendation G.108 was prepared by ITU-T Study Group 12 (1997-2000) and was approved under the WTSC Resolution No. 1 procedure on 30 September 1999.

Keywords

E-Model, end-to-end, interconnection, Internet, network, PBX, planning, quality, speech, terminal, transmission.

FOREWORD

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The World Telecommunication Standardization Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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Introduction

Network transmission planning has, historically, been based on scenarios reflecting the then-current (mainly analogue) technology, and the components (terminals, switches, facilities) available to the network.

Today, the global telephone network transmission environment is rapidly changing. Factors which are impacting this change are the no longer rigidly regulated network interconnections, liberalization of private network transmission parameters (particularly in Europe), increasing competition in the public networks, globalization of private networks, and the use of modern technology within private networks, driven by the customers' need for economical and flexible network solutions.

In North America, this traditional approach to planning had to take into account the industry-prescribed loss plan for Multi-line Telecommunications Systems (e.g. for PBXs, as presented in [40]; this was the premise for developing a DPBX loss plan application guide [43]. Furthermore, this planning was constrained by the transmission plan, as well as interconnecting requirements, of the North American public switched telephone network (PSTN).

Similarly, in Europe, planning had to consider not only the relevant private network parameters but also those of the main public networks in the different countries. Regulation policy in most European countries dictated rigid handling of network interconnections, in conjunction with sometimes very stringent limits, for the different transmission parameters for calls via the public network.

Similar changes may apply to other countries or regions not mentioned before.

In the ITU-T and the former CCITT, transmission planning had been based on the allocation of single parameter limits to individual sections of an international connection, thereby not considering other interconnecting networks, e.g. private networks or even the Internet.

The traditional transmission planning methodologies and documents are no longer flexible enough to account for all these new factors. For modern transmission planning, the following issues have to be considered:

a) Multinational networks (especially private networks) are becoming common and require planning which takes into account regional differences in loss plan requirements (if there are any) and inter-network transmission plans.

b) Due to the liberalization of the telecommunication markets (in Europe) there are no longer laid down ranges of values for transmission parameters by regulation.

c) The changing scenario in the public network operator domain is impacting transmission performance.

d) Planning of private networks is more complex than single PBX-configurations, therefore sufficient (tutorial) information about the planning, and calculation, methods needs to be provided.

e) The Recommendation should be applicable to the use of new technology within the networks under consideration, including wireless (cordless or mobile) sections, transmission of packetized voice, etc.

f) The Recommendation should provide sufficient planning methods and contain all the necessary information and tools which will enable the planner to design the network transmission plan such as to keep the resulting speech transmission quality for all possible end-to-end connections within expected limits.

g) Greater flexibility in allocation of transmission parameters and provisions with more allowance for network transmission parameters than in earlier guidelines, standards, or regulations is necessary.

h) The guidelines and planning examples of this Recommendation are based on the use of the E-Model as described in Recommendation G.107 [3].

All these issues mentioned above need to be taken into account. One new main aspect is that the planner has to consider that the allocation of transmission parameters has become very flexible. Due to the liberalization of the telecommunication markets there are not any longer rigid constrained allowed ranges of values for the transmission parameters within the networks under consideration. The use of new technology and the application of new services has become quite common in the different networks. So the planer has to care of various effects caused by the use of different technology in an environment of interconnected networks.

The guidelines and principles in this Recommendation reflect this flexibility, particularly in the planning examples, where it is shown how the allocation and selection of parameters is applied in optimizing the speech transmission quality for various connection scenarios.

It is a key feature of this Recommendation, to give guidance and tutorial information to the planner. The aim is to enable the planer to develop a transmission plan allowing to predict the resulting end-to-end speech transmission quality of voice telephony connection.

The guidelines and principles in this Recommendation are based on the use of the E-Model which is a useful tool to estimate the combined effects of various transmission parameters. It is recommended to use the E-Model as a transmission planning tool for end-to-end intra-network connections (connection between two terminals of the same network) and inter-network connections (connection between two terminals of different networks). By using the E-Model the planner is enabled to lay down adequate ranges of values for the different transmission parameters.

Furthermore, this Recommendation should enable the planner not only to ensure that absolute limits for various transmission parameters are not exceeded for any intra- or inter-network connection, but also to obtain an estimate of the expected speech transmission quality as perceived by the average user (in terms of E-Model Rating R, Mean Opinion Score or percentage Good-or-Better and Poor-or-Worse) for the investigated configuration.

Recommendation G.108

APPLICATION OF THE E-MODEL: A PLANNING GUIDE

(Geneva, 1999)

1 Scope

This Recommendation applies to transmission planning of speech transmission quality for end-to-end intra- and inter-network connections. It should be considered as a tutorial and illustration for the planning of networks with respect to the speech transmission quality of narrow-band 3.1 kHz real time telephony via handsets. Networks designed according to this Recommendation will also provide sufficiently high speech quality for the transmission of announcements and stored speech.

The guidelines and principles in this Recommendation are based on the use of the E-Model. Estimates of the perceived speech transmission quality calculated with the E-Model are only applicable to end-to-end connections, offering narrow-band 3.1 kHz real time telephony via handsets.

The transmission of non-voice signals such as Fax and Modem transmission and wholly digital data transmission is beyond the scope of this Recommendation.

This Recommendation does not address the following issues:

- who owns and runs the network;
- who is responsible for end-to-end transmission quality;
- to whom services are provided.

One application of this Recommendation is to medium and large private networks consisting of several interconnected MLTS. The terms "corporate network" or "enterprise network" are sometimes used to describe a large private network; in some countries these terms are used in a legal sense for a group of interconnected private networks. From the point of view of transmission planning, there is no difference between a large private network and several smaller interconnected networks. Therefore, only the term "private network" will be used in this Recommendation.

The Recommendation addresses, as examples, only scenarios where the network under consideration, e.g. a "private network" functions as a terminating network (one to which terminal equipment is connected). Scenarios wherein a network provides transit connections between other networks are not considered.

Notwithstanding these limitations, it is recommended to apply the principles described, and information given, in this Recommendation to other end-to-end connections supporting narrow-band 3.1 kHz real time telephony via handsets as well. This Recommendation can be applied to all kinds of intra- or inter-network connections. It is recommended to use these principles for the planning of end-to-end connections regardless of the number of public or private networks involved, or the specific configuration they are interconnected.

For the purposes of this Recommendation, there are no restrictions on the network with respect to size, configuration, hierarchy, technology used, nor on the components of the network. The transmission media may be cable, fibre or radio.

The discussion in this Recommendation addresses primarily the use of digital interfaces (almost universal in Europe) between the networks under consideration, but also allows for analogue network interconnections. The signal transmission within the network under consideration may be analogue or digital.

The prevalence of digital signal transmission media and digital signal handling in switching equipment impacts the relative importance of various transmission parameters to be considered in planning. For the benefit of simplification, parameters which cause only minor impairments in a digital environment, such as the frequency shape of cables, circuit noise, crosstalk, variations of loss with level or time etc., are not subject to the planning guidelines in this Recommendation. More emphasis is placed on parameters such as echo, delay, signal processing equipment impairments, and acoustic characteristics of terminals.

This Recommendation does not contain transmission requirements for specific network elements such as telephone sets, switching equipment (e.g. PBXs) or transmission equipment. It is assumed that the design of such elements conforms to applicable Recommendations, regional standards or regulations.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

[1] ITU-T Recommendation G.100 (1993), *Definitions used in Recommendations on general characteristics of international telephone connections and circuits*.

[2] ITU-T Recommendation G.101 (1996), *The transmission plan*.

[3] ITU-T Recommendation G.107 (1998), *The E-Model, a computational model for use in transmission planning*.

[4] ITU-T Recommendation G.109 (1999), *Definition of categories of speech transmission quality*.

[5] ITU-T Recommendation G.113 Appendix I (1999), T*ransmission impairments – Appendix I: Provisional planning values for the equipment impairment factor Ie*.

[6] ITU-T Recommendation G.114 (1996), *One-way transmission time*.

[7] ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections*.

[8] ITU-T Recommendation G.126 (1993), *Listener echo in telephone networks*.

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[10] ITU-T Recommendation G.164 (1988), *Echo suppressors*.

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[16] ITU-T Recommendation G.721 (1988), *32 kbit/s Adaptive differential pulse code modulation (ADPCM)*.

[17] ITU-T Recommendation G.723.1 (1996), *Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*.

[18] ITU-T Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)*.

[19] ITU-T Recommendation G.727 (1990), *5-, 4-, 3- and 2-bits/sample embedded adaptive differential pulse code modulation (ADPCM)*.

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[22] ITU-T Recommendation G.729 (1996), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction*.

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[26] ITU-T Recommendation H.323 (1998), *Packet based multimedia communications systems*.

[27] ITU-T Recommendation P.310 (1996), *Transmission characteristics for telephone band (300-3400 Hz) digital telephones*.

[28] ITU-T Recommendation P.50 (1993), *Artificial voices*.

[29] ITU-T Recommendation P.79 (1993), *Calculation of loudness ratings for telephone sets*.

[30] ITU-T Recommendation P.800 (1996), *Methods for subjective determination of transmission quality*.

[31] ITU-T Recommendation P.82 (1984), *Method for evaluation of service from the standpoint of speech transmission quality*.

[32] ITU-T Recommendation P.830 (1996), *Subjective performance assessment of telephone-band and wideband digital codecs*.

[33] ITU-T Recommendation P.84 (1993), *Subjective listening test method for evaluating digital circuit multiplication and packetized voice systems*.

[34] ITU-T Recommendation P.561 (1996), *In-service, non-intrusive measurement device – Voice service measurements*.

[35] ITU-T Recommendation P.861 (1998), *Objective quality measurement of telephone-band (300-3400 Hz) speech codecs*.

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[50] ETSI TBR 010 ed.2 (1997), *Digital Enhanced Cordless Telecommunications (DECT); General terminal attachment requirements: Telephony applications*.

[51] ETSI EG 201 377-1 V1.1.1 (1999), *Speech Processing, Transmission and Quality Aspects (STQ); Specification and measurement of speech transmission quality; Part 1: Introduction to objective comparison measurement methods for one-way speech quality across networks*.

[52] ETSI ES 201 168 V1.1.1 (1998), *Corporate telecommunication Networks (CN); Transmission characteristics of digital Private Branch eXchanges (PBXs)*.

[53] ETSI EN 300 175-8 V1.4.2 (1999), *Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech Coding and Transmission*.

[54] ETSI ETS 300 283 ed.1 (1994): *Business TeleCommunications (BTC); Planning of loudness rating and echo values for private networks digitally connected to the public network*.

[55] ETSI: EN 300 462-series (1998), *Transmission and Multiplexing (TM); Generic requirements for synchronization networks*.

[56] For the purposes of this Recommendation the following standards should be considered together as a package:

ETSI EN 300 961 V7.0.2 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Transcoding* (GSM 06.10 version 7.0.2 Release 1998).

ETSI EN 300 962 V7.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Substitution and muting of lost frames for full rate speech channels* (GSM 06.11 version 7.0.1 Release 1998).

ETSI EN 300 963 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Comfort noise aspect for full rate speech traffic channels* (GSM 06.12 version 6.0.1 Release 1997).

ETSI EN 300 964 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Discontinuous Transmission (DTX) for full rate speech traffic channels* (GSM 06.31 version 6.0.1 Release 1997).

ETSI EN 300 965 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Voice Activity Detector (VAD) for full rate speech traffic channels* (GSM 06.32 version 6.0.1 Release 1997).

[57] For the purposes of this Recommendation the following standards should be considered together as a package:

ETSI EN 300 969 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Half rate speech transcoding* (GSM 06.20 version 6.0.1 Release 1997).

ETSI EN 300 970 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Substitution and muting of lost frames for half rate speech traffic channels* (GSM 06.21 version 6.0.1 Release 1997).

ETSI EN 300 971 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Comfort noise aspects for the half rate speech traffic channels* (GSM 06.22 version 6.0.1 Release 1997).

ETSI EN 300 972 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Discontinuous Transmission (DTX) for half rate speech traffic channels* (GSM 06.41 version 6.0.1 Release 1997).

ETSI EN 300 973 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Voice Activity Detector (VAD) for half rate speech traffic channels* (GSM 06.42 version 6.0.1 Release 1997).

[58] For the purposes of this Recommendation the following standards should be considered together as a package:

ETSI EN 300 726 V7.0.2 (1999), *Digital cellular telecommunications system (Phase 2+); Enhanced Full Rate (EFR) speech transcoding* (GSM 06.60 version 7.0.2 Release 1998).

ETSI EN 300 727 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Substitution and muting of lost frames for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.61 version 6.0.1 Release 1997).

ETSI EN 300 728 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Comfort noise aspects for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.62 version 6.0.1 Release 1997).

ETSI EN 300 729 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Discontinuous Transmission (DTX) for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.81 version 6.0.1 Release 1997).

ETSI EN 300 730 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Voice Activity Detector (VAD) for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.82 version 6.0.1 Release 1997).

[59] ETSI ETR 250 ed.1 (1996), *Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3.1 kHz handset telephony across networks*.

[60] ETSI ETR 275 ed.1 (1996), *Transmission and Multiplexing (TM); Considerations on transmission delay and transmission delay values for components on connections supporting speech communications over evolving digital networks*.

[61] ISO/IEC 11573:1994, *Information technology – Telecommunications and information exchange between systems – Synchronization methods and technical requirements for Private Integrated Services Networks*.

[62] ARIB: RCR STD-27 H, Fascicle 1 (Revision H, February 2 1999), *Personal Digital Cellular Telecommunication System ARIB Standard*.

3 Abbreviations

This Recommendation uses the following abbreviations:

ACELP	Algebraic Code-Excited Linear Prediction
ADPCM	Adaptive Differential Pulse Code Modulation
ATM	Asynchronous Transfer Mode
CDMA	Code Division Multiple Access
CLR	Circuit Loudness Rating
CNG	Comfort Noise Generator
CRE	Corrected Reference Equivalent
CS-ACELP	Conjugate Structure Algebraic Code-Excited Linear Prediction
DAL	Digital Access Line
DCME	Digital Circuit Multiplication Equipment
DECT	Digital Enhanced Cordless Telecommunication
DPBX	Digital PBX
EC	Echo Canceller
ECD	Echo Control Device
EFR	Enhanced Full Rate Codec
EL	Echo Loss
ELE	Echo Loss Enhancement
ERLE	Echo Return Loss Enhancement
ERP	Ear Reference Point
ETSI	European Telecommunications Standards Institute
FDM	Frequency Division Multiplex
FEC	Forward Error Correction
GoB	Good or Better
GSM	Global System for Mobile communications
IC	Interexchange Carrier
ICP	International Connection Point
ICS	ISDN Compatible Station
INMD	In-service Non-intrusive Measurement Device
IP	Internet Protocol
ISDN	Integrated Services Digital Network
IST	Integrated Services Trunk
ITU	International Telecommunication Union

ITU-T	ITU Telecommunication Standardization Sector
IWF	InterWorking Function
KTS	Key Telephone System
LAN	Local Area Network
LD-CELP	Low-Delay Code-Excited Linear Prediction
LEC	Local Exchange Carrier
LELR	Listener Echo Loudness Rating
LSTR	Listener Sidetone Masking Rating
MLTS	Multi-line Telecommunications System
MOS	Mean Opinion Score
MRP	Mouth Reference Point
N.A.	North America(n)
NCP	Network Connection Point(s)
NLP	Non-Linear Processor
OLL	Open-Loop Loss
OLR	Overall Loudness Rating
ONS	On-Premises Station
OREM	Objective Reference Equivalent Measurement
PACS	Personal Access Communications System
PBX	Private Branch Exchange
PCI	Personal Communications Interface
PCM	Pulse Code Modulation
PoW	Poor or Worse
PSTN	Public Switched Telephone Network
PWT	Personal Wireless Telecommunications
qdu	Quantization distortion unit(s)
RLR	Receive Loudness Rating
RPE-LTP	Residual Pulse Excitation – Long Term Predictor
RTP	Real Time Protocol
SLR	Send Loudness Rating
SS	Soft Suppressor
STMR	Sidetone Masking Rating
TBRL	Terminal Balance Return Loss
TCLw	Terminal Coupling Loss (weighted)
TCP	Transmission Control Protocol
TDM	Time Division Multiplex
TDMA	Time Division Multiple Access

TELR	Talker Echo Loudness Rating
TIA	Telecommunications Industry Association
UDP	User Data Protocol
UPCM	Uniform PCM
VAD	Voice Activity Detection
VNL	Via Net Loss
VPN	Virtual Private Network
VSELP	Vector Sum Excited Linear Prediction
WAN	Wide Area Network
WEPL	Weighted Echo Path Loss
WUPE	Wireless User Premises Equipment

4 Definitions

This Recommendation defines the following terms:

4.1 Private network

In this Recommendation, the term "private network" is used to describe a network which provides features only to a restricted user group in contrast to the public network (PSTN) available to the general public. In general, a private network is a terminating network and consists of several interconnected nodes (i.e. PBXs), with interconnections to other (mainly public) networks.

A private network is characterized as follows:

1) It consists normally of more than one element of switching equipment (PBX or Key Telephone System KTS), connected via tie trunks or leased lines or via a Virtual Private Network (VPN). Network functionality is independent of its structure and hierarchy. Switching equipment and links can be either analogue or digital.
2) It provides switching functions and all other features only to a single customer or to a group of customers, and is not accessible to the general public.
3) It is not limited by geographical size or to a specific national area or region.
4) It has no limitation with regard to the number of extensions and access points to other networks.

4.2 Public network

The term "public network" is used in this Recommendation for any network providing transmission and switching functions as well as features which are available to the general public, not restricted to a specific user group. In this context, the word "public" does not imply any relation to the legal status of the network operator.

In some cases, a public network may provide a limited set of features only. In a competitive environment, a public network may be restricted to serve a limited number of customers, or restricted to specific features or functions. Generally, public networks provide access points to other networks or terminals only within a specific geographical area.

From the point of view of an end-to-end connection, a public network can function either as a "transit network" (a link between two other networks) or as a combination of "transit and terminating network" in case where the public network provides connections to terminal equipment such as telephone sets, or PBXs. In North America, Interexchange Carriers (IC) generally function as transit networks while the functions of a transit- and terminating-network are assigned to Local Exchange Carriers (LEC).

4.3 Quality aspects

Previous planning methods for terminating networks in general were usually based on limit values for the different transmission parameters between the telephone set (acoustic interface) and the interface to another, mainly public, network. This means that, only the section within the private network, as part of a full connection formed by the different network elements between a human mouth/ear and an electrical interface, was considered.

However, the perception of speech transmission quality during a telephone conversation is primarily a "subjective" judgement. The concept of "quality" may not be considered as a unique discrete quantity, but may vary, depending on the user's expectation of sufficient "speech transmission quality" for a 3.1 kHz telephony call for the terminal mode (e.g. handset) as well as the particular service (e.g. wireless). With respect to transmission planning, the planning method and the necessary calculations should be based on an end-to-end consideration between one human's mouth and another human's ear.

For the judgement of the quality in a given configuration, and the performance of "subjective tests", several methods are in use and are described in different Recommendations (e.g. Recommendations P.800 [30], P.82 [31], P.830 [32] and P.84 [33]). One of the most common methods is to perform laboratory tests (e.g. "listening only tests"), wherein the test subjects are requested to classify the perceived quality into categories. For example, a "quality rating" can be defined according to the following 5-grade scale:

Quality	Score
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

The scores are used to calculate the average value of the judgement of several test subjects for the same test configuration. The result is the so-called "Mean Opinion Score" MOS which may, theoretically, range between 1 and 5. An assessment about the speech transmission quality can also be obtained by calculating the percentage of all test persons rating the Configuration as "good or better" or as "poor or worse". For a given connection these results are expressed as "Percentage Good or Better" (%GoB) and "Percentage Poor or Worse" (%PoW).

In existing networks, public and private network providers may also use various methods to control and **monitor** the "Quality of Service" with respect to speech transmission quality. This can be done:

- subjectively by customer interviews according to Recommendation P.800 [30];
- via an objective method using test equipment referred to as an "In-service Non-intrusive Measurement Device" (INMD) according to Recommendation P.561 [34];
- by objective measurement devices according to Recommendation P.861 [35] – Note however, that a process of refinement is underway in ITU-T Study Group 12.

However, those methods as described in [51] are not applicable for the **planning** of networks.

The main task during planning of networks is the collection of the necessary information on the various network components in the configuration investigated and their contribution of transmission impairments that impact the end-to-end connection speech transmission quality. To assist the planner, computational models are available which, with the pertinent input data, yield a calculated value for the E-Model Rating R and other quality measures, such as MOS, %GoB and %PoW.

One of those tools is the **"E-Model"** which is recommended to be used for planning purposes of end-to-end 3.1 kHz handset telephony connections. For more information about the E-Model, see clause 6 in conjunction with Recommendation G.107 [3] and [59].

4.4 IP based network

IP based networks may appear with distinct realizations which can be classified as follows:

- managed and/or engineered private Enterprises Intranets (LANs);
- Service Provider domains (WANs);
- the public Internet (concatenated WANs);
- or a combination thereof.

They consist of internal routers and edge routers (e.g. between LAN and WAN) which are establishing network links between the Interworking Functions (IWFs), so-called gateways, (e.g. to the PSTN) and the IP terminals (e.g. according to Recommendation H.323 [26]).

IP based networks are relying on the Internet Protocol (IP) and provide for packet-based transport of data. Thus, a digitalized speech signal applied to the input of an Interworking Function (IWF) will be assembled to small packets. These packets then consist of an IP header which in general contains the following data:

- destination address;
- other transport related information;
- headers of other protocols for transmission, real time, etc. (e.g. TCP, UDP, RTP);
- and of a small time segment of the speech signal.

Finally, at the output of another IWF the speech segments will be decomposed to a continuous digital speech signal again. Alternatively, the packetizing as well as the decomposition of the speech signal can be provided by a terminal according to Recommendation H.323 [26] instead of the IWF.

4.5 Network elements

All components which constitute an end-to-end connection should be categorized into three main groups: terminal elements, connection elements and transmission elements.

4.5.1 Terminal elements

With respect to speech transmission, terminal elements are all types of telephone sets, digital or analogue, wired, cordless, or mobile, including the acoustical interfaces to the user's mouth and ear. These components are characterized by their Send Loudness Rating (SLR) and Receive Loudness Rating (RLR) which contribute to the Overall Loudness Rating (OLR) of a connection. Other terminal element parameters, such as the SideTone Masking Rating (STMR), the Listener SideTone Rating (LSTR), the design of the handset (D-Factor), the frequency response in send and receive directions and the noise floor, also contribute to the end-to-end connection rating of speech transmission quality. In case of wireless or IP based systems, additional distortions and delay may be added, depending on the coding and modulation algorithms used in such interfaces.

4.5.2 Connection elements

Connection elements are all types of switching equipment, such as local PBXs (for the direct connection of terminal elements) and transit PBXs in networks. Connection elements may use analogue or digital switching or packet based technology. The main impairment contributions of analogue systems are loss and noise. Digital switching systems contribute to the end-to-end delay, due to signal processing, and also to the amount of quantization distortion associated with digital pads and code conversion. Packet based routers contribute, in addition, with delay variation versus time and packet loss. Where 4-wire to 2-wire conversions take place within or between switching equipment interfaces, signal reflections contribute to impairments as a source for echo effects.

4.5.3 Transmission elements

Transmission elements are all kinds of media used as the facility between connection elements and between connection elements and terminal elements. The physical media of these elements may be metallic (copper), fibre-optics or radio. The signal form is either analogue or digital. Impairments associated with analogue signal transmission include propagation time (generally proportional to distance), loss, frequency response and noise (mainly due to longitudinal interference).

For planning purposes, impairments due to frequency response and noise can usually be neglected for short and medium line lengths.

For digital transmission elements, the main transmission impairment is caused by the propagation time via metallic, optic, and radio media. For wireless sections, additional delay is introduced, depending on the coding and modulation algorithm used. Where the transmission element includes analogue-to-digital conversion, loss and distortion are additional impairment factors.

Multiplexing is generally used to transport several channels via one single physical media. A variety of multiplexing systems are in use in the existing networks:

- Time Division Multiplex (TDM).
- Digital Circuit Multiplication Equipment (DCME).
- Packet Based Facilities:
  - connection oriented (ATM);
  - connectionless (Ethernet, IP).

In digital transmission elements, systems use either 64 kbit/s Pulse Code Modulation (PCM) of Recommendation G.711 [15], or one of the more recently introduced compression techniques based on low bit-rate codecs. Major influence to the transmission quality of these systems may be caused by additional distortions in terms of qdu, or Equipment Impairment Factor (Ie), and by additional mean one-way delay.

4.6 Types of connections

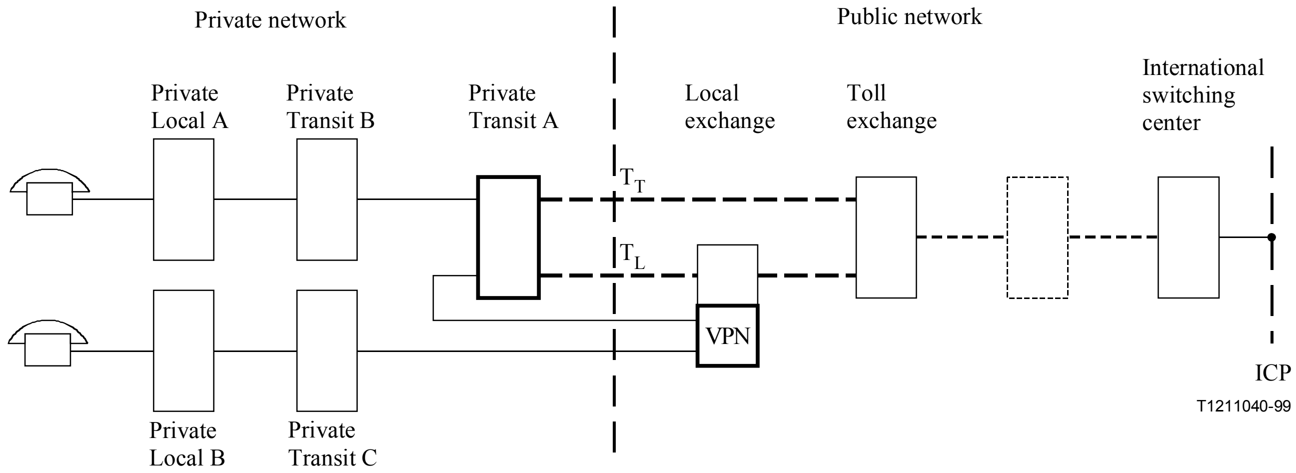
For some networks, impairment allocation may be possible by taking into account the mix of connection types. For example, private networks may provide mainly internal connections, or may route a preponderance of calls destined for a public network terminal via private network facilities to a local PSTN switch near the destination ("tail-end hop-off"). Depending on the business of the private network user, the predominance of incoming and outgoing calls may be originated or terminated only within a local calling area. It may also be feasible to segregate calls between internal connections (between two terminal elements of the same private network), and external connections via a public network, which can be further divided into "local calls" for connections in the local area only, "national long distance calls" and "international calls."

It should be noted that "predominance", in this context, is interpreted as an amount of connections, e.g. within the local area, of more than 95%. Consideration of the connection type mix into planning enables the planner, wherever this is possible, to extend the usually small allowance for specific parameters (e.g. transmission time) within the network under consideration, resulting in a more economical design of the network.

5 Reference configurations

The aim of reference configurations in transmission planning is to obtain an overview of the considered connection and to simplify the identification of all terminal-, connection- and transmission elements which contribute impairments to the end-to-end transmission performance.

Due to the variety of hierarchy, structure, routing, number and types of network elements in a network, each investigated connection will result in a different reference configuration. Therefore, it is not possible to create a single generic configuration for the whole task of network planning. The following Figures should only be considered as examples, used mainly for definitions in this Recommendation. One main task in planning is to identify the type of interconnection between the network under consideration and other networks. Figure 1 shows a basic configuration, assuming a digital interface between the networks.



ICP International Connection Point

T_L Access point to Local exchange of public network

T_T Access point to Toll exchange of public network

VPN Virtual Private Network

Figure 1/G.108 – Basic configuration for the interconnection between private and public network

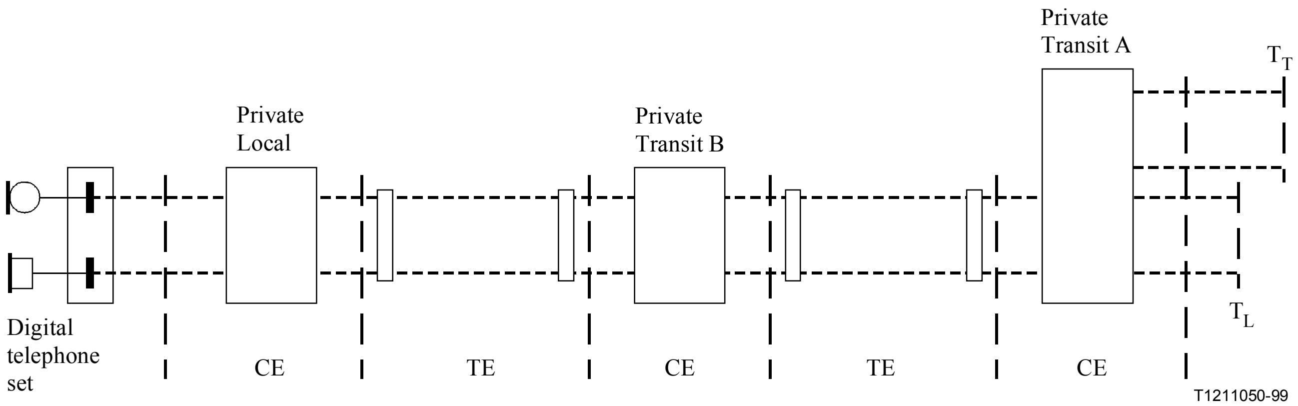
In Figure 1, which is generalized to include international scenarios, the public network, and therefore the entire connection, is shown only up to the International Connection Point, ICP, of an international switching centre. It is assumed, that the impairment allowance between the access points for calls within the national public network are allocated symmetrically with reference to the ICP, which can be considered as the virtual centre of the public network for international calls. Since calls can be terminated on both sides with private networks in the same configuration, it seems sufficient to draw the Figure in this simple way. For connections not involving an ICP, e.g. most connections within North America, the equivalent virtual centre can be assumed to be within the digital portion of the highest ranking network provider, generally an IC.

The configuration shows two different types of interfaces between public and private networks, the one called T_L, connects the private network to a local exchange (e.g. LEC), usually the lowest hierarchy and the common connection point in a public network. The other interface, called T_T, connects the private network directly to a higher hierarchy level, e.g. an IC, bypassing the local exchange. In some cases, especially for larger private networks, by-pass may permit more allocation of specific transmission parameters, e.g. delay, to the private network.

Figure 1 also illustrates the interconnection between the private Transit A and private Transit C exchanges using the feature of a Virtual Private Network VPN. For the purpose of transmission planning, this VPN, although part of the public network, should be considered as part of the private network. The same is valid for leased lines or tie trunks serving as transmission elements between different PBXs within the private network but usually provided by public network carriers.

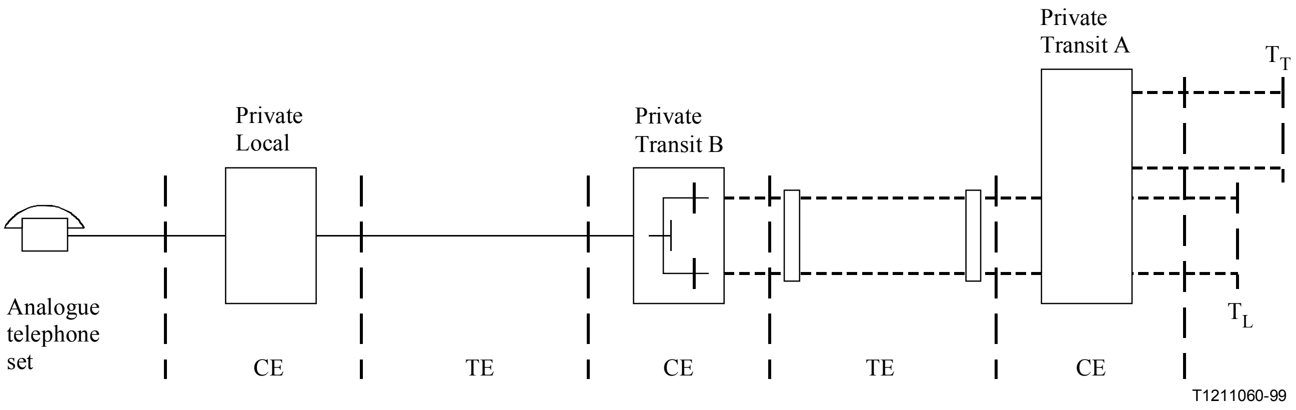
Where the private network includes leased lines or tie trunks, VPN connections, or Centrex terminals, the private network planner should obtain transmission planning information on these connections from the public network provider of these facilities or services.

Figures 2 through 4 illustrate in more detail common configurations within the private network. Figure 2 shows a fully digital connection between a digital telephone set and the digital interfaces T_L or T_T to the public network. Assuming a fully bit-transparent transmission in all elements of the private network, this configuration can be considered as the quality optimum for a connection where the private network most likely contributes a minimum of transmission impairments to the overall connection.



T_L Digital access point at Local exchange
T_T Digital access point at Toll exchange
CE Connection Element
TE Transmission Element

Figure 2/G.108 – Standard configuration with a fully digital routing within the private network



T_L Digital access point at Local exchange

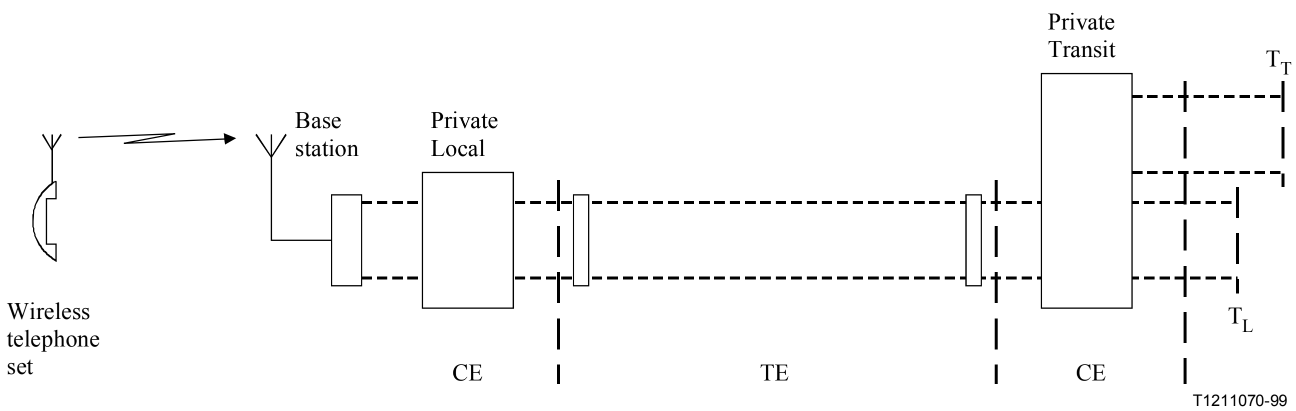
T_T Digital access point at Toll exchange

CE Connection Element

TE Transmission Element

Figure 3/G.108 – Private Network with 4-wire/2-wire Conversion

The configuration in Figure 3 assumes a 4-wire/2-wire conversion (hybrid) in the private Transit B switch and 2-wire facilities to the private terminating switch as well as to the analogue telephone set. In this case impairments due to loss in the 2-wire cable section should be expected. Furthermore, the hybrid in the Transit B switch may cause impairments with respect to being a possible source of echo for the far-end subscriber. This hybrid also forms the termination 4-wire connection with the public network and hence could influence the stability of the connection.



T_L Digital access point at Local exchange

T_T Digital access point at Toll exchange

CE Connection Element

TE Transmission Element

Figure 4/G.108 – Digitally connected wireless telephone set

When wireless telephone sets serve as terminal elements, as shown in Figure 4, impairments due to additional delay and distortion should be subject to planning. In such configurations, the use of echo control devices should also be investigated.

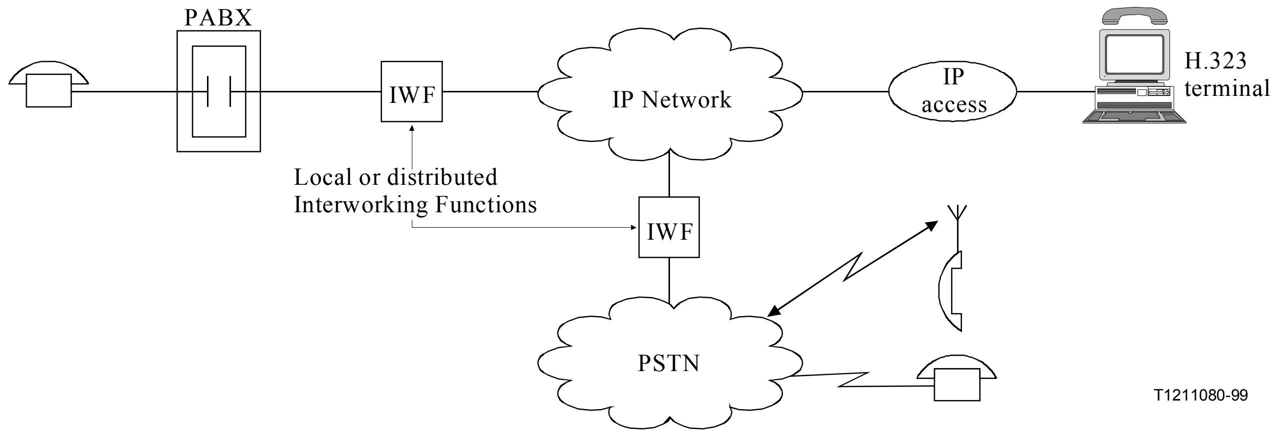


Figure 5/G.108 – Basic connection between the PSTN and an IP network with its H.323 terminal and a PBX connected via an IWF

When IP-based networks serve as transmission elements (containing connection elements), as shown in Figure 5, impairments due to transcoding or tandeming of low bit-rate codecs and packet loss as well as additional delay should be subject to planning. For such configurations, the existence, necessity and the properties of echo control devices should also be investigated.

6 Basic planning principle – the impairment factor method in conjunction with the E-Model

As indicated in the introduction to this Recommendation, the rapidly changing scenario in the field of multiple interconnected networks (e.g. private networks) with increasing size and complexity, in combination with new technologies and the constraint for more economical solutions, requires more flexibility with respect to transmission planning.

In general, the quality of speech transmission via telephone channels is based on a subjective judgement by the users at both ends. Therefore, transmission planning as given in Recommendation G.101 (The Transmission Plan) [2] is, in principle, derived from an end-to-end consideration in conjunction with a partitioning of all relevant parameters between different networks, or parts of a network. For private networks, this method was in common use in the field of regulation for all calls via the public network, providing limits for the private network between the acoustical interface of the telephone set and an electrical interface to the public network. These limits were defined to guarantee a sufficient quality for all calls (national and international). In North America, the public network providers usually mandated the allocation; in Europe, the allocation was governed by regulation.

As networks become more complex (North American standards for internetworking are modernized, and European countries are moving into liberalization) this principle is no longer valid. In conjunction with increasing liberalization in many countries, the responsibility for a sufficient speech transmission quality is now shifted to the operator of the terminating (e.g. private) network. However, planning of such networks with respect to speech transmission quality needs knowledge and experience in the field of transmission parameters and their influence to quality. Therefore, it seems necessary to provide a planning method which is easy to handle and accompanied by comprehensive tutorial information and planning tools. This is the main task of this Recommendation.

It should be noted that the preferable purpose of network planning is to control the summation of transmission impairments, caused by the different network elements in all possible configurations. It is not the task of network planning to limit the transmission impairment of a specific network element. Unless indicated otherwise, it is assumed that transmission, switching and terminal elements in general are designed to meet all relevant requirements as given in ITU-T Recommendations and in international or national standards applicable for this type of element.

Based on subjective testing users' perception is expressed in terms of MOS, %GoB or %PoW. During transmission planning, however, it is not practical to perform subjective tests. Therefore, a method must be provided which enables the planner to combine, by calculation, all existing transmission impairments in the given connection to a total value of impairment. This calculation must be performed by using an algorithm based on subjective testing. In telephone connections consisting of a variety of network elements, different transmission parameters may also simultaneously contribute to the total impairment. Therefore, the planning method used must also incorporate combination effects. For all configurations subject to this Recommendation, the planning of speech transmission quality should be based on an end-to-end consideration rather than on a specification of individual objective parameter limits.

For the calculation of the different impairment values, particularly if the combined effect from the presence of more than one parameter needs to be considered, computational models are used for planning purposes. Several such "rating-models" have been developed and were contained and described in former ITU-T and CCITT publications, which are no longer recommended for application and which today have bibliographic status only. Today, Recommendation G.107 [3] gives the algorithm for the so called E-Model as the common ITU-T Transmission Rating Model.

Transmission planning, based on the E-Model as recommended, provides a prediction of the expected quality, as perceived by the user, for an investigated connection. Based on an end-to-end assessment for each transmission parameter (including the type and number of low bit-rate codecs) impairment values are derived. This method accounts for low bit-rate coding devices as well as for impairments introduced by standard PCM coders and for impairments not directly related to digital processing. The introduction of a quality issue for planning purposes also enables the private network operator to make the design of the network on a cost versus quality relation, taking into account the specific requirements for the private network.

The basic planning principle, as used in this Recommendation, deviates from previous planning methods for network interconnection scenarios. End-to-end speech transmission quality is now expressed in terms of the E-Model Rating R, as a result of calculations with the E-Model. The E-Model Rating R can be transformed into other quality measures, which have been used in transmission planning before, such as Mean Opinion Score (MOS), Percentage Good or Better (%GoB) or Percentage Poor or Worse (%PoW), according to Annex B/G.107 [3].

Figure 6 is intended to show the relation and interdependency between Subjective MOS, Objective MOS, Predicted MOS and E-Model Rating R in detail.

For clarification:

- Subjective MOS – listening speech quality calculated by a "Subjective Test" according to Recommendation P.800 [30].
- Objective MOS – measured listening speech quality, typically a comparison rating method.
- Predicted MOS – estimated conversational quality.

The "system" box contains all the equipment (acoustic or electric input/output) which is to be tested (either subjectively or objectively).

The "subjective test" is the auditory test to provide a subjective MOS (Mean Opinion Score).

The "comparison rating method" is the objective measurement device (calibrated with the auditory test results) with "objective MOS" as the result. This result (in the case of testing a pure codec device) can additionally be transformed into the "equipment impairment factor" (Ie), for use in the E-Model. Subjective and objective MOS' are transformed into an "equipment impairment factor" (Ie), as described in Appendix I/G.113 [5].

The "E-Model" is a parameter based algorithm based on subjective test results of auditory tests done in the past with the "system" parameters (and Ie values) as inputs. The result of the E-Model calculations is "E-Model Rating R" which can be transformed to "predicted MOS".

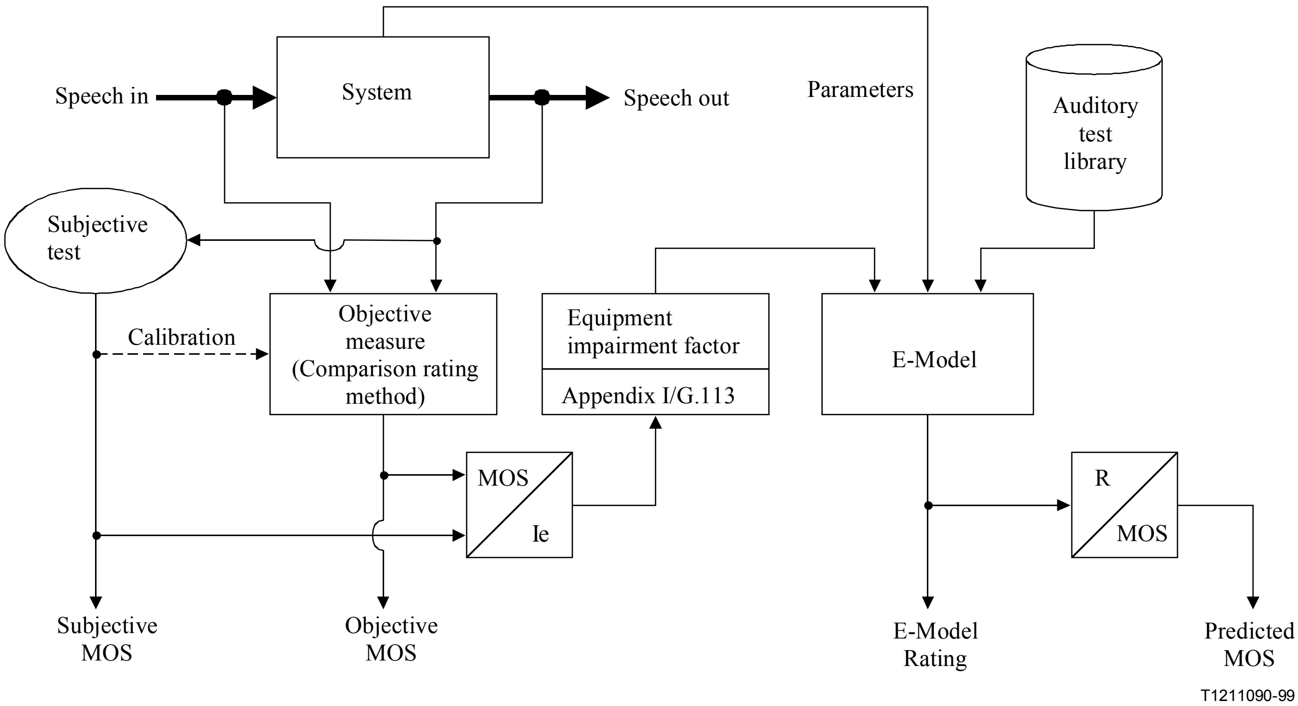


Figure 6/G.108 – The E-Model in the environment of subjective and objective testing

This basic principle is applied not only for internal calls within the network under consideration, but also for calls via public networks. For calls via public networks, illustrated in Figure 7, application of the "end-to-end principle" addresses parts of a connection which are not directly subject to planning by the network planner. Impairments caused by the public network (or tandem connections via more than one public network) and the far-end termination should be determinable and included in the transmission planning. In practice, this may cause some problems for planning, since those values may be not available in all cases, particularly for the far-end termination. In those situations, average values for different transmission parameters can be used, such values are provided in this Recommendation (see A.3) for basic terminations (single telephone set, private network). Minor problems may arise for the public network sections. Information about impairments for different routings should be available by negotiations between the public network provider and the private network planner. However, judicious application of impairment parameters in public network sections allows for a differentiation for the type of calls, national local, national long distance and international, and possibly different the allocation of impairments for the benefit of the private network design.

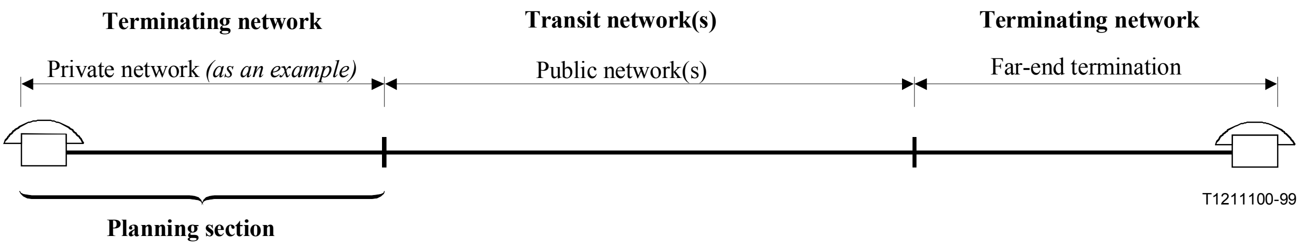


Figure 7/G.108 – General configuration for calls via public networks

The impairment factor method is based on the assumption that transmission impairments can be transformed into psychological factors and that these psychological factors are additive on the "Psychological scale" as shown in Figure 8.

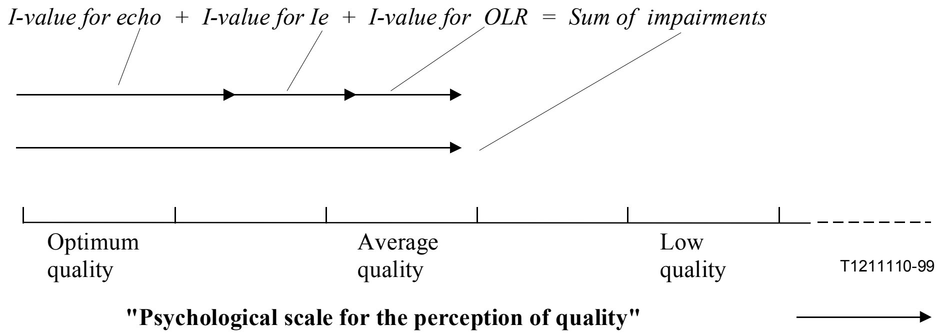


Figure 8/G.108 – Example for the addition of impairments on a "Psychological scale"

An appropriate mathematical algorithm is provided by the E-Model, with which the different transmission parameters can be transformed into different "impairment factors". This method and the algorithm of the E-Model also include the combination effects of those impairments in the considered connection which occur simultaneously as well as some masking effects. With the E-Model, a very useful tool is available, which provides a simplified and easy-to-handle method for practical planning purposes.

The final result of any calculation with the E-Model is the E-Model Rating R. The relation between the different impairment values and R is given by the equation:

$$R = Ro - Is - Id - Ie + A$$

The term Ro expresses the basic signal-to-noise ratio.

The term Is represents all impairments which occur more or less simultaneously with the voice signal, such as: too loud speech level (non-optimum OLR), non-optimum sidetone (STMR), quantization noise (qdu), etc.

The "delay impairment" factor Id sums all impairments due to delay and echo effects, and the "equipment impairment factor" (Ie), represents impairments which are caused by low bit-rate codecs used in special equipment.

The "advantage factor " A represents an "advantage of access" which certain systems may provide in comparison to conventional systems. While all other impairment factors are subtracted from the basic signal-to-noise ratio Ro, this value is added and thus compensates other impairments to a certain amount. It can be used to take into account the fact that the user will tolerate some decrease in transmission quality in exchange for the "advantage of access". Examples of such advantages are cordless and mobile systems or connections into hard-to-reach regions via multi satellite hops. The use and the amount of the advantage factor A fall into the responsibility of the individual transmission planner and are for further study.

High values of the E-Model Rating R in a range of $90 \leq R < 100$ should be interpreted as excellent quality, while a lower value of R indicates a lower quality.

It should be noted, that, in some cases, not only the final result for R is of interest, but also the specific impairment values Is, Id and Ie. Their individual contribution to the total value of impairments can be used for the determination of the major impairments in the given configuration and for possible solutions for the reduction of these impairments' severity: e.g. reducing Id by the insertion of echo cancellers.

Table 1 which has been taken from Recommendation G.109 [4] relates the E-Model Ratings R to categories of speech transmission quality and to user satisfaction.

Table 1/G.108 – Definition of categories of speech transmission quality

Range of E-Model Rating R	Speech transmission quality category	User satisfaction
$90 \le R < 100$	Best	Very satisfied
$80 \le R < 90$	High	Satisfied
$70 \le R < 80$	Medium	Some users dissatisfied
$60 \le R < 70$	Low	Many users dissatisfied
$50 \le R < 60$	Poor	Nearly all users dissatisfied
NOTE 1 – Connections with E-Model Ratings R below 50 are not recommended. NOTE 2 – Although the trend in transmission planning is to use E-Model Ratings R, equations to convert E-Model Ratings R into other metrics, e.g. MOS, %GoB, %PoW can be found in Annex B/G.107 [3].		

The E-Model and its algorithm require the following input parameters:

- **SLR** **Send Loudness Rating;**
- **RLR** **Receive Loudness Rating;**
- **OLR** **Overall Loudness Rating**[1];
- STMR Sidetone Masking Rating[2];
- LSTR Listener Sidetone Rating[2];
- Ds D-value of telephone at send-side;
- Dr D-value of telephone at receive-side[2];
- **TELR** **Talker Echo Loudness Rating**;
- WEPL Weighted Echo Path Loss;
- **T** **Mean one way delay of the echo path**;
- Tr Roundtrip delay in a closed 4-wire loop;
- **Ta** **Absolute delay in echofree connections**;
- **qdu** **Number of quantization distortion units**;
- **Ie** **Equipment impairment factor (low bit-rate codecs)**;
- Nc Circuit noise referred to the 0 dBr-point;
- Nfor Noise floor at the receive-side;
- Ps Room noise at the send-side;
- Pr Room noise at the receive-side;
- **A** **Advantage factor**

[1] No direct input value; calculated as OLR = SLR + RLR.

[2] These parameters have a fixed relation by: LSTR = STMR + Dr.

Only those parameters indicated by bold letters are usually subject to planning. The other parameters can be set to default values. For more details see clause 7.

A detailed description of the complete algorithm is given in Recommendation G.107 [3]. For information about the practical use of the E-Model for planning purposes see clause 9.

7 Parameters subject to planning and their limits

As stated in the scope, this Recommendation provides mainly examples based on connections with digital interfaces between private and public networks and with digital media for the main transmission elements within the private network. As networks and network elements migrate from analogue to digital, certain transmission parameters become of more significance with respect to the end-to-end speech quality and more important for planning purposes while others are reduced to minor influence or can even be neglected. Subclauses 7.1 through 7.9 describe those parameters which, in a mainly digital environment, should be subject to planning; subclause 7.10 deals with those parameters which, for the benefit of simplification, can be neglected or considered only in special applications.

7.1 Overall Loudness Rating

Although digital interfaces to other, mainly public, networks and digital transmission media for the main transmission elements within the private networks are the prime focus, the Overall Loudness Rating (OLR) of a connection should be considered. For economic reasons, portions of private networks may continue to rely on 2-wire analogue lines and on connection elements (PBXs) with analogue switching technology, thereby contributing impairments generally associated with the analogue environment, such as loss and noise. Likewise, within public networks, fully digital routing and termination cannot be assumed for all connections.

Basically, the OLR of a connection is calculated as the sum of the Send Loudness Rating (SLR), of the telephone set at one end, the Receive Loudness Rating (RLR), of the other end set, and the Circuit Loudness Rating (CLR), representing the sum of all analogue and digital losses between these telephone sets.

$$OLR = SLR + CLR + RLR$$

Impairments due to OLR may result from either too high or too low values of OLR. The optimum value lies in the range from 6 to 10 dB. Connections between two digital telephone sets, i.e. designed according to Recommendation P.310 [27], reference [42] with SLR = 8 dB and RLR = 2 dB, or to reference [49] with SLR = 7 dB and RLR = 3 dB, and routed over a fully digital connection, will meet this optimum value for OLR (10 dB). The relation between the E-Model Rating R and the OLR of a connection is shown in Figure 9. This graph, calculated with the E-Model, is obtained when all other input parameters of the E-Model are set to their default values (see 9.7); i.e. OLR is the only impairment in the connection considered.

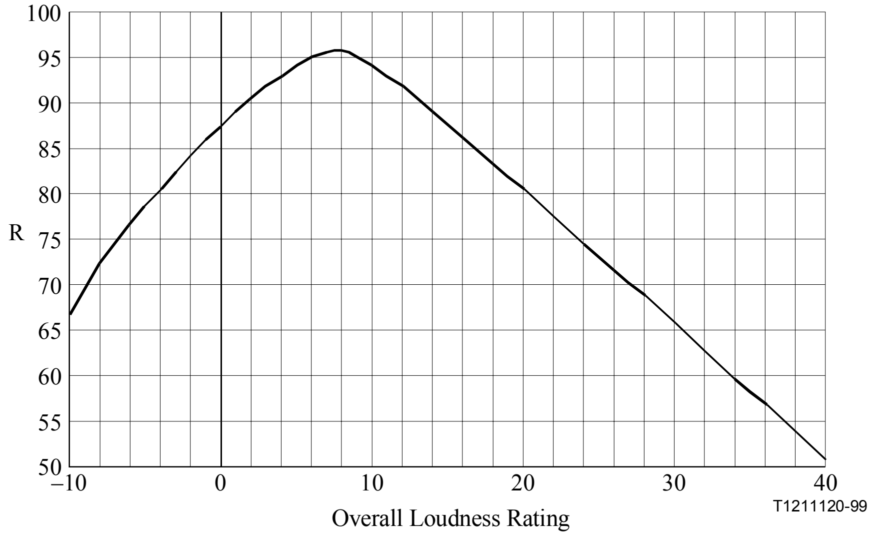


Figure 9/G.108 – Relation between OLR and E-Model Rating R

Comparing the graph in Figure 9 with the definition of categories of speech transmission quality in Table 1, a range for the OLR from –10 dB to 30 dB delivers quality between "best" and "low". However, the value 30 dB for a connection should be considered as an absolute upper limit which should never be exceeded, not even in exceptional cases. For a speech transmission quality in the "medium" category, the preferred limit for standard connections, the upper value for OLR is recommended to be in the range of 20 dB to 25 dB. Low values of OLR may exist for internal connections within a private network between two analogue telephone sets, or if digital telephone sets with lower values for SLR and/or RLR than those given in Recommendation P.310 [27] and references [42] or [49] are used. Very low values for OLR should be avoided. For OLR < 0 dB the insertion of additional loss is recommended.

7.2 Echo

As stated above, fully digital routing cannot be assumed in all configurations. Mixed analogue/digital routing, within private and public networks, generally implies the presence of 4-wire/2-wire conversions; there signal reflections, together with transmission delay, might cause impairments due to talker echo which must be taken into account. Signal reflections will mainly occur at hybrids, where the bidirectional path of the connection between the talker's telephone set and the hybrid forms the "echo path". In some connections with multiple conversion points, there are multiple echo paths.

The effects of echo in a conversation can cause impairments to the talker as well as to the listener. These impairments are expressed as Talker Echo and Listener Echo, respectively. More information about these effects is given in Recommendation G.131 [9] and Recommendation G.126 [8]. As a general rule, listener echo can be neglected if there is sufficient control of the talker echo.

The impairment due to talker echo depends on two factors, the delay and the level of the reflected speech signal received by the talker. The perceived quality decreases with increasing delay and/or increasing level of the received echo signal.

For planning purposes, the transmission delay (T), is defined as the mean one-way transmission time of the echo path. Although the total transmission time between the talker's telephone set and the hybrid and back to the talker constitutes the delay of the echo, the mean one-way transmission time is used in the E-Model calculations, assuming that, for most configurations, the transmission time is nearly equal in both transmission directions.

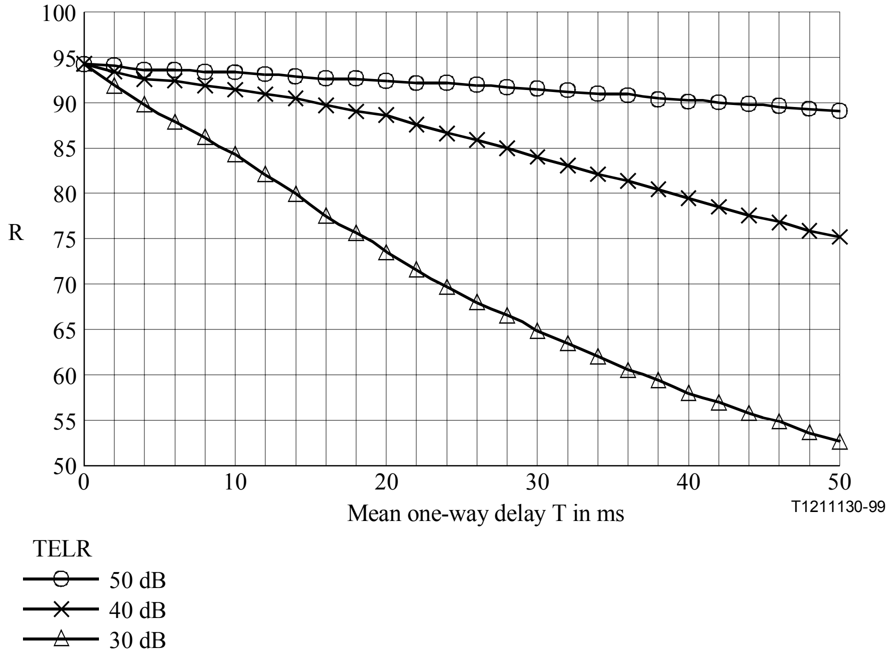


Figure 10/G.108 – E-Model Rating R due to talker echo

For planning calculations, the level of the received echo signal, as an input parameter to the E-Model, is expressed as the Talker Echo Loudness Rating (TELR). TELR is defined as the sum of SLR and RLR of the talker's telephone set and the Echo Loss EL of the echo path. The Echo Loss (EL) includes losses in the bidirectional transmission path and the TBRL (see A.1.3) at the 4-wire/2-wire conversion point.

$$TELR = SLR + EL + RLR$$

The relation between the mean one-way delay (T), and the E-Model Rating R is shown in Figure 10 for three different values of TELR. The graphs are calculated with all other parameters set to their default value.

Talker echo is the most important parameter in modern mixed digital/analogue networks.

Improvements in perceived quality with respect to talker echo can either be achieved by higher values of TELR or by lower values of mean one-way delay (T), or by a combination of both:

- The value of TELR can be increased by improvements in the 4-wire/2-wire conversion (increasing the TBRL), or by deliberately increasing the loss in the connection. However, in a given connection, increasing the TELR by adding loss is limited to avoid too large values of the OLR (note, however, that for every dB increase in the OLR, the TELR will increase by two dB). When the loss value, which would be required to achieve adequate TELR, results in excessive OLR, the use of Echo Cancellers (EC), should be considered. Information about EC application is given in clause 10.
- To decrease the value of the mean one-way delay (T) during transmission planning, it should be considered that, in modern networks, two categories of delay "sources" contribute to the end-to-end value of T. The first category comprises the traditional distance-related delay sources, e.g. analogue or digital cable sections (including switching facilities). The second category comprises modern signal processing devices as delay sources, e.g. low bit-rate codecs, where the value of the delay depends only on the type and number of the equipment inserted into a connection. Decreasing the value of T for the first category yields to possibly avoiding indirect routing, while for the second category the type and number of such signal processing devices might be changed as a result of transmission planning.

To achieve a speech transmission quality in the "high" category with a E-Model Rating R ≥ 80 (see Table 1), Figure 10 shows that, with TELR = 30 dB, the mean one-way delay is limited to T = 14 ms, while with TELR = 40 dB a value of T = 39 ms can be tolerated. In practice, values for TELR will, in most applications, be in the range of 30 dB to 40 dB. Thus, for connections with an echo path of more than 20 to 25 ms mean one-way delay, it is advisable to thoroughly investigate the impairments due to echo effects and to consider the possible use of echo cancellers.

Very low values of the mean one-way delay with T < 1.5 ms, are considered as sidetone and need not be investigated as an impairment due to echo. For low values of the Sidetone Masking Rating, STMR < 9 dB some masking of the talker echo may be observed. This effect is included in the algorithm of the E-Model.

In planning practice, it is necessary to clearly identify the echo path in a connection. As noted, more than one echo path may exist in some configurations. For planning rules in those situations, see 8.2.2 and 9.5.

7.3 Transmission time in echofree connections

As networks and network elements migrate from analogue to digital, transmission time in echofree connections becomes of more significance with respect to the end-to-end speech quality and more important for planning purposes, besides general considerations, the awareness of the specific mechanisms of codec-related processing delay needs particular attention.

7.3.1 General considerations

Additional impairments may arise due to very long delay, even if perfect echo cancelling is provided. Typically, this occurs on international or intercontinental connections carried via satellite links, or submarine cables, but also national or internal connections within the network under consideration may be affected, if modern signal processing devices, like low bit-rate codecs, are inserted. Very long delay may result in difficulties during conversation. According to subjective tests, this effect is encountered mainly for a one-way transmission time of more than 150 ms.

The relevant parameter for planning is the Absolute Delay (Ta), in ms, defined as the mean one-way delay between the two telephone sets, independent of the number of echo paths in the same connection. The E-Model Rating R, is shown in Figure 11 for a range of Ta = 100 to 600 ms, with all other parameters at their default values.

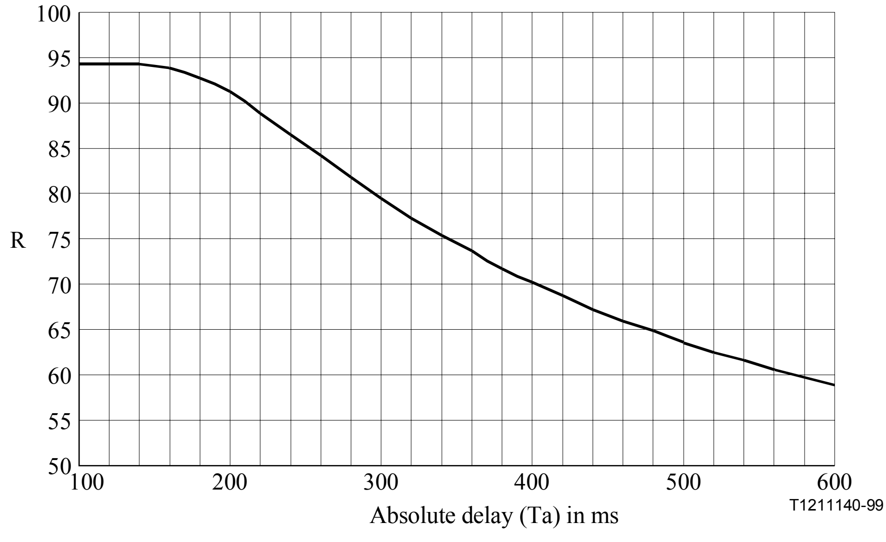


Figure 11/G.108 – Relation between absolute delay (Ta) and E-Model Rating R

With respect to a speech transmission quality in the "medium" category as described in Table 1, the limit value for most connections should be in the range of 300 to 350 ms with an upper limit of 400 ms. This is also in accordance with Recommendation G.114 [6] and should be exceeded in exceptional cases, only.

7.3.2 Considerations with respect to low bit-rate coders

Modern speech codecs operate on collections of speech samples known as frames. Each block of input speech samples is processed into a compressed frame. The coded speech frame is not generated until all speech samples in the input block have been collected by the encoder. Thus, there is a delay of one frame before processing can begin. In addition, many coders also look into the succeeding frame to improve compression efficiency. The length of this advance look is known as the look-ahead time of the coder. The time required to process an input frame is assumed to be the same as the frame length since efficient use of processor resources will be accomplished when an encoder/decoder pair (or multiple encoder/decoder pairs operating in parallel on multiple input streams) fully uses the available processing power (evenly distributed in the time domain). Thus, the delay through an encoder/decoder pair is normally assumed to be:

$$2 \times \text{frame size} + \text{look-ahead}$$

7.3.2.1 Delay in wirebound environment

If the output facility is running at the same rate as the speech codec (e.g. an 8 kbit/s facility for G.729), then an additional frame of delay is incurred when clocking the compressed frame to the facility. Thus, the maximum delay attributable to codec-related processing in conventional wirebound systems (i.e. the PSTN) is:

$$3 \times \text{frame size} + \text{look-ahead}$$

7.3.2.2 Delay in mobile and wireless environment

If the output facility is a mobile network or a cordless facility, then the frame output by the encoder will function similar to the operation in wirebound environment (see 7.3.2.1) but an additional delay is incurred for attaching the compressed frame to the airpath (assumed again that the mobile facility is running at the same rate as the speech codec). Thus, the maximum delay attributable to codec-related processing in mobile and wireless systems is:

$$3 \times \text{frame size} + \text{look-ahead} + \text{air interface framing}$$

7.3.2.3 Delay in IP environment (one frame per packet)

If the output facility is an IP network, then the frame output by the encoder will instantaneously be dropped into an IP packet. The additional delay required for IP packet assembly and presentation to the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. Thus, the minimum delay attributable to codec-related processing in IP-based systems is:

$$2 \times \text{frame size} + \text{look-ahead}$$

When the link layer is one with lower clock rate (e.g. modem connection) or one with high traffic load (e.g., congested LAN), the additional delay will increase substantially. In order to clock compressed frames at least with the same rate to the facility as the speech samples are collected at the input of the encoder, the additional delay should not exceed one frame size, Thus, the maximum delay attributable to codec-related processing in IP based systems operation in real-time is:

$$3 \times \text{frame size} + \text{look-ahead}$$

7.3.2.4 Delay in IP environment (multiple frames per packet)

If multiple voice frames are grouped together into a single IP packet, further delay is added to the speech signal. This delay will be at least the duration of one extra voice frame at the encoder for each additional voice frame added to the IP packet. Thus, the minimum delay attributable to codec-related processing in IP based systems with multiple frames per packet is:

$$(N+1) \times \text{frame size} + \text{look-ahead},$$

where N is the number of frames in each packet.

When the link layer is one with lower clock rate (e.g. modem connection) or one with high traffic load (e.g. congested LAN), additional delay will be incurred in delivering the packet to the facility. In order to clock compressed frames, at least with the same rate to the facility as the speech samples are collected at the input of the encoder, the additional delay should, in case of multiple frames per packet, not exceed the length of the frames contained in one packet. It should be noted that clocking out a packet to the IP facility cannot start before all speech frames for this packet are available. Thus, the maximum delay attributable to codec-related processing in IP based systems operating in real-time with multiple frames per packet is:

$$(2N+1) \times \text{frame size} + \text{look-ahead}$$

where N is the number of frames in each packet.

7.4 Stability

The stability of a connection should be investigated in transmission planning for any connection, where the network under consideration contains a 4-wire loop or where a 2-wire/4-wire conversion (hybrid) within the network under consideration together with a 4-wire/2-wire conversion within the public network or the far-end termination forms a 4-wire loop. For further information about stability see also Recommendation G.122 [7].

Insufficient stability may cause "singing" within the 4-wire loop, which must always be avoided. Although stability is more a problem during call setup and release (due to lower balance return loss at hybrids during these states) and usually does not impact the talking state, a circuit with singing may disturb other channels of a telecommunication network via crosstalk, mainly in analogue systems. The main parameter to control stability is the so called "Open Loop Loss (OLL)", the sum of all losses and gains in the 4-wire loop. The term stability defines the margin ("singing margin") between the actual OLL and the point where singing, usually at some frequency within the voiceband may arise. The most critical configurations during call setup are the open and short circuit conditions at the 2-wire sides of the hybrids terminating the ends of the 4-wire loop.

Since singing will not normally occur during the talking state (when proper termination at the 2-wire ends are present), stability is not a factor in the assessment of speech quality. Nevertheless, it should be considered during transmission planning. This must be done in a separate calculation, since the E-Model does not include an algorithm for calculating the stability of 4-wire loops. To avoid singing or "near-singing" (a situation close to the singing point), the stability (OLL) in every 4-wire loop should be at least 4-6 dB for every frequency in the voiceband. If a private network contains only one end of a 4-wire loop, e.g. in connections via a public network, the stability loss at the interface between private and public network should, for all possible connection scenarios, be greater than 6 dB for every frequency in the voiceband.

Connections including one or more 4-wire loops as a source for signal reflections, may, during talking state, contribute talker echo impairments due to multiple echoes as well as listener echo impairments. For information about listener echo as a subject of planning, see 7.10.3.

7.5 Quantization distortion units

Inherent to the process of PCM encoding and decoding according to the "8 bit-law" (A-law or μ-law) as described in Recommendation G.711 [15], Tables 5 and 6, is an impairment known as "quantization distortion". This impairment is perceived as a quantization noise accompanying the received voice signal (i.e. it is not perceived in a quiescent channel), thereby, reducing the signal-to-noise ratio. Quantization distortions are additive, i.e. every A/D-D/A conversion will contribute additional noise.

For transmission planning, it is common practice to express quantization noise in "quantization distortion unit " (qdu); setting a limit for the maximum number of acceptable qdu in a connection. One qdu is defined as the quantization noise arising from a complete encoding from analogue into digital (A/D) and again decoding from digital into the analogue (D/A) signal form according to Recommendation G.711 [15].

With the increasing use of digital transmission and connection elements in private and public networks, the importance of quantization noise will decrease. However, quantization distortion can be ignored in planning only if fully bit-transparent routing can be assumed. Whenever mixed digital/analogue elements are present in a connection, the resulting number of qdu needs to be subject to planning. The influence of the number of qdus in a connection to the E-Model Rating R is shown in Figure 12.

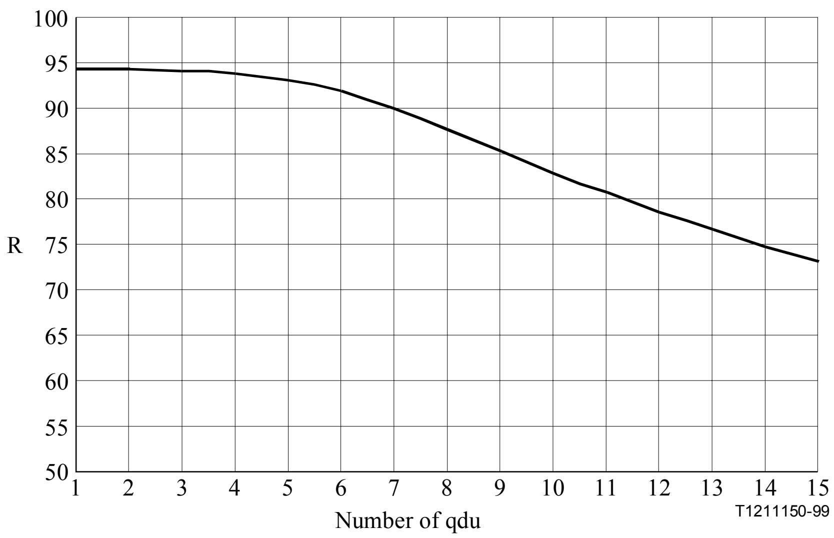


Figure 12/G.108 – Relation between the number of qdu and E-Model Rating R

The graph in Figure 12, derived from the E-Model with all other parameters at their default values, shows that the impairments are negligible for an aggregate distortion of up to 4 or 5 qdus. However, connections with more than one qdu will usually also be influenced by other impairments, such as loss and echo effects, such that the sum of all impairments influences the perceived speech quality. Because the E-Model calculations always include the number of qdu as an input parameter, it is recommended that the correct number of qdu of the connection be determined and used as an input to the model instead of the default value (1 qdu).

The parameter qdu in transmission planning applies not only to A/D-D/A conversions but also to other processes influencing the digital bit-stream. Those processes are, for example, the insertion of digital loss or gain, signal addition in conference circuits, use of digital echo cancellers, etc. For coding laws other than A-law/μ-law (e.g. according to Recommendations G.726, G.727 and G.728), the parameter qdu is, for transmission planning, replaced by the equipment impairment factor (Ie) (see 7.6).

7.6 Equipment impairment factor

Modern coding laws, like those associated with low bit-rate codecs as described in the G.720-series of Recommendations or the GSM Standards, as well as ADPCM with different operating bit-rates, will contribute distortions resulting in a decrease of the perceived speech quality. Contrary to the quantization distortion due to the standard 8 bit PCM coding (A-law or μ-law), these impairments can not readily be quantified with a number of qdu. For planning purposes, the impairments introduced by the different types of codecs are expressed by the "equipment impairment factor" (Ie). Ie values are obtained by subjective or objective tests, see Figure 6 for details. The results of these subjective or objective mean opinion scores are transformed into a value, Ie, which follows the basic planning principle described in 6.1 (addition of impairments on a linear psychological scale) and which, therefore, can be directly used as an input parameter for the E-Model.

Some provisional planning values of the equipment impairment factor Ie for several codec types are listed in Table 2a. For up to date information on those values and on values for other codec types, Appendix I/G.113 [5] may provide guidance (Appendix I/G.113 [5] is intended to be updated regularly).

Table 2a/G.108 – Provisional planning values for the equipment impairment factor (Ie)

Codec type	Reference	Operating rate kbit/s	Ie value
ADPCM	G.726, G.727	40	2
	G.721(1988), G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50
LD-CELP	G.728	16	7
		12.8	20
CS-ACELP	G.729	8	10
	G.729-A + VAD	8	11
VSELP	IS-54	8	20
ACELP	IS-641	7.4	10
QCELP	IS-96-A	8	19
RCELP	IS-127	8	6
VSELP	Japanese PDC	6.7	24
RPE-LTP	GSM 06.10, Full-rate	13	20
VSELP	GSM 06.20, Half-rate	5.6	23
ACELP	GSM 06.60, Enhanced Full Rate	12.2	5
ACELP	G.723.1	5.3	19
MP-MLQ	G.723.1	6.3	15

The speech codecs and systems mentioned in Table 2a are briefly described below.

IS-54 First generation digital TDMA cellular system in North-America utilizing Vector Sum Excited Linear Prediction (VSELP) coding at a net bit rate of 7.95 kbit/s (plus 5.05 kbit/s FEC) [45].

IS-96-A First generation digital CDMA cellular system in North-America utilizing Qualcomm Code-Excited Linear Prediction (QCELP) coding at a variable net bit rate of 8, 4 and 2 kbit/s [46].

IS-127 Second generation digital CDMA cellular system in North-America utilizing Residual Code-Excited Linear Prediction (RCELP) coding at a variable net bit rate of 8, 4 and 2 kbit/s [47].

IS-641 Second generation digital TDMA cellular system in North-America utilizing Algebraic Code-Excited Linear Prediction (ACELP) coding at a net bit rate of 7.4 kbit/s (plus 5.6 kbit/s FEC) [48].

GSM-FR First generation digital European Global System for Mobile Communication (GSM) cellular system utilizing Regular Pulse Excitation Long Term Prediction (RPE-LTP) coding at a net bit rate of 13 kbit/s (plus 9.8 kbit/s FEC) [56].

GSM-HR Half-rate version of the voice codec for the GSM system utilizing Vector Sum Excited Linear Prediction (VSELP) coding at a net bit rate of 5.6 kbit/s [57].

GSM-EFR Second generation speech codec of the digital European Global System for Mobile Communication (GSM) cellular system utilizing Algebraic Code-Excited Linear Prediction (ACELP) coding at a net bit rate of 12.2 kbit/s (plus 10.6 kbit/s FEC) [58].

PDC First generation digital Japanese Personal Digital Communication (PDC) system utilizing a Japanese version of Vector Sum Excited Linear Prediction (JVSELP) coding at a net bit rate of 6.7 kbit/s (plus 4.5 kbit/s FEC) [62].

G.723.1 ITU-T standard for speech coding in PSTN videophones utilizing Algebraic Code-Excited Linear Prediction (ACELP) coding at 5.3 kbit/s and Multipulse Maximum Likelihood Quantization (MP-MLQ) at 6.3 kbit/s [16].

G.726 ITU-T standard for speech coding at 40, 32, 24, and 16 kbit/s utilizing Adaptive Differential Pulse Code Modulation (ADPCM) [18], see also Recommendation G.721 [17] and Recommendation G.727 [19].

G.728 ITU-T standard for speech coding at 16 kbit/s utilizing Low-Delay Code-Excited Linear Prediction (LD-CELP) coding [20]. This algorithm also has 12.8 and 9.6 kbit/s bit rate extensions [21].

G.729 ITU-T standard for speech coding at 8 kbit/s utilizing Conjugate Structure Algebraic Code-Excited Linear Prediction (CS-ACELP) coding [22]. This algorithm also has 6.4 and 11.8 kbit/s bit rate extensions [23] and [24].

7.6.1 Equipment impairment factor for codecs under conditions of packet loss

Table 2a of Ie values refers to non-error conditions. For propagation errors and frame-erasures or packet loss, no definite values are available which would be valid for more than one codec or codec family. In order to help the transmission planner, examples of Ie values under conditions of packet loss are given in Table 2b and for propagation error patterns EP 1 and EP 2 in Table 2c. These values are provisional only as they were determined in single experiments.

Table 2b/G.108 – Provisional planning values for the equipment impairment factor (Ie) under conditions of packet loss, codecs G.729-A + VAD and G.723.1-A + VAD

% Packet loss	Ie for G.729-A + VAD	Ie for G.723.1-A + VAD 6.3 kbit/s
0	11	15
0.5	13	17
1	15	19
1.5	17	22
2	19	24
3	23	27
4	26	32
8	36	41
16	49	55
NOTE – No. frames per packet: – G.729-A + VAD: 2; – G.723.1-A + VAD: 1.		

Table 2c/G.108 – Provisional planning values for the equipment impairment factor (Ie) under propagation error conditions, GSM codecs

Codec type	Error pattern	Ie range
GSM-HR	EP 1	25...32
	EP 2	31...42
GSM-FR	EP 1	32...39
	EP 2	40...45
GSM-EFR	EP 1	15...22
	EP 2	26...35
NOTE 1 – The range given results from the difficulties in deriving exact impairment factor values for these conditions. NOTE 2 – EP 1 is equivalent to 10 dB C/I, EP 2 is equivalent to 7 dB C/I. C/I is the carrier to interference ratio.		

7.7 Delay variation versus time

Packetized transmission systems exhibit variable delay in packet delivery time; this is caused by the fact that different packets carrying speech samples of the same telephone conversation may be transported via distinct routes through the network: details of this effect depend strongly on the specific mechanisms for transport, queuing or prioritization, which may be implemented in such a system.

Packets which have been transported through a packet based network are collected in a buffer at the receive side. This buffer functions as the instance which re-arranges the timely order of the packets. If the delivery time of a packet exceeds the length of the receive buffer, then this packet "comes too late" with respect to the size of this buffer and will be discarded. Hence, the speech carried in this packet is lost for the decoding process. This "packet loss" impacts speech transmission quality. One approach to minimize the percentage of such lost packets is the dynamic adaptation of the length of the receive buffer.

A receive buffer with adaptive length controls its actual length via its filling level:

- if the number of packets being at a time within the buffer increases, then the length of the buffer will be increased and at the same time at the output of the buffer short pause sequences from the original signal will be dropped in order to drain the buffer with a faster rate;
- if the number of packets being at a time within the buffer decreases, then the length of the buffer will be decreased and at the same time at the output of the buffer short pause sequences will be additionally inserted into the original signal in order to drain the buffer with a slower rate.

Consequently, there will be a variable value of the end-to-end mean one-way delay between the talker's mouth and the listener's ear. It is important to clearly distinguish this effect from other delay variation discussions which refer to network internal processes, only. The impact of an end-to-end delay variation, as explained in this subclause, is strongly dependant on the length of the dropped or inserted pieces of pauses; further emphasis rests on the correct implementation of the dynamic adaptation processes, e.g. the insertion of pieces of pauses into a syllable will have more serious impact than the insertion of pieces of pauses into a pause sequence.

The parameter of end-to-end delay variation is independent of the use of Voice Activity Detection (VAD) devices (see 7.10.6).

The impact of end-to-end delay variation on speech transmission quality is for further study and not included in the algorithm of the E-Model.

7.8 Advantage factor A

The "advantage factor A" represents an "advantage of access", introduced into transmission planning for the first time via the E-Model (Recommendation G.107 [3] and [59]). This factor enables the planner to take into account the fact that customers may accept some decrease in quality for access advantage: e.g. mobility or connections into hard-to-reach regions. This value can be used directly in conjunction with all other impairment values and as an input parameter to the E-Model. Provisional A values are listed in Table 3, which has been taken from Recommendation G.107 [3].

Table 3/G.108 – Provisional examples for the advantage factor A

Communication system example	Maximum value of A
Conventional (wirebound)	0
Mobility by cellular networks in a building	5
Mobility in a geographical area or moving in a vehicle	10
Access to hard-to-reach locations, e.g. via multi-hop satellite connections	20

These values are provisional since they have not been confirmed by subjective investigations to date. Therefore, the advantage factor A should be used with care and with respect to the business interest of the private network customer, where users may judge specific advantages in telecommunication with criteria other than those used in the usual private domain. The use of the advantage factor in transmission planning of private networks and the selected values are subject to the planner's decision; however, the values in Table 3 should be considered as the maximum upper limit for A.

7.9 Limits at the public/private interface with respect to echo

Transmission planning for private networks, as described in this Recommendation is primarily directed to the performance of the private network; i.e. intended to provide acceptable speech quality for private network users. A private network, however, can also be the source for signal reflections from 2-wire terminations or network elements, connected through hybrids to a digital network interface point. In conjunction with the mean one-way delay of the public network connection, this may result in a talker echo impairment for the talker at the far (public network) end.

Although the basic planning principles of this Recommendation are based on end-to-end transmission performance, the interface between the private network and a public network needs additional control with respect to certain parameters. This is for consistency with related standards for these interfaces. Because these standards differ between Europe and North America, each is treated separately here.

7.9.1 Public/private network interfaces – Europe

The guidelines are derived from reference [54], where limits are given for the so-called "Network Connection Point" (NCP), to control echo in national calls where no echo cancellers are inserted. Summarizing the content of [54], the following limits at the NCP should be taken into account with respect to transmission planning:

Minimum Send Loudness Rating	+7 dB
Minimum Receive Loudness Rating	+3 dB
Echo loss (digital in to digital out)	> 24 dB (long term objective) > 20 dB (short term objective)
Mean one-way delay of the echo path	< 5 ms

These values are generally considered as being based on a maximum mean one-way delay of 25 ms for the whole echo path, with 15 ms for the path within the (transit) public network and 5 ms each for the paths within the (terminating) private networks (for more details see Reference [54]). The long term objective for the total TELR as the sum of SLR, RLR and echo loss EL is 34 dB. Assuming no further impairments, these values of 25 ms and TELR = 34 dB will result in an E-Model Rating R = 77.1 (see Figure 10), which falls into the "medium" category of speech transmission quality according to Table 1.

If the limit of 5 ms within the private network is exceeded and/or a TELR of 34 dB cannot be met, echo control devices, even for national calls within European countries, should be used to assure adequate echo protection for the far-end talker. For configurations wherein values of the mean one-way delay and the TELR are close to, or likely to exceed the above limits, a calculation using the E-Model is recommended.

7.9.2 Public/private network interfaces – North America

In North America, guidelines for the interface between a public and a private network are given in [44], wherein the assumed nominal characteristics for access lines between customer equipment and the public network are defined.

Analogue access lines are assumed to be terminated by equipment having nominal characteristics given in Table 4 (see [41]). An appropriate range is given to account for wide variations in telephone sets and loops. The ERL given in Table 4 is estimated as that achievable for analogue access lines in the spectrum of digital network configurations in North America.

Table 4/G.108 – Assumed analogue access characteristics

Parameter	Nominal (dB)	Range (dB)
SLR	+11 (Note 1)	+19 to +6 (Note 2)
RLR	−3 (Note 1)	−8 to +2 (Note 2)
ERL (Note 3)	μ = 14	σ = 3
NOTE 1 – SLR and RLR are specified at the loop interface at the DEO. These values are predicated on a 9 kft (2.7 km), 26-gauge (0.4 mm) cable loop and a 48-volt power supply. NOTE 2 – Variations in both telephone sets and access lines contribute to these values. NOTE 3 – ERL values shown above refer to loss across the hybrid with the access line terminated at the NI in a standard impedance (typically 600 Ω in series with 2.16 μF), and do not include any network losses.		

Digital access lines are assumed to be terminated with digital telephone sets conforming to specifications in [42] as provided in Table 5, or other digital terminal equipment having equivalent characteristics.

Table 5/G.108 – Assumed digital access characteristics for lines terminated in digital sets

Parameter	Desired (dB)	Required (dB)
SLR	8	5 to 14
RLR	2	−1.5 to 5.5
TCLw	>45	≥40

Digital access lines between the PSTN and a PBX/Private Network interface should follow the guidelines of Table 5; recognizing, however, that this is strongly dependent on the loss plan and echo loss provisions of the PBX, as given in [40].

Reference [44] recommends that any new system, network or component that, by itself, adds more than 2.5 ms mean one-way delay should provide echo cancellation. Otherwise, degradation of transmission quality would occur. The rationale for this guideline is that when a new digital technology becomes part of an end-to-end connection in the PSTN, in general, the connection may not contain an echo canceller, or the capability of the echo canceller is unknown.

7.10 Parameters not directly subject to transmission planning

As mentioned in the beginning of clause 7, certain transmission parameters can be neglected for the benefit of simplification in the calculation of speech quality. Although these parameters had previously been used in transmission planning and, in some cases, are input parameters to the E-Model, the assumption of an environment in both the private and public domains which embodies mainly digital transmission and digital connection elements will decrease the influence of these specific parameters.

7.10.1 Frequency response

In analogue transmission, impairment due to the *frequency response* of unloaded cable sections and other connection elements needed to be controlled. For the purpose of this Recommendation, it is assumed that analogue routing via unloaded cable sections within the private network occurs only in lower portions of the hierarchy and generally, only on short or medium line lengths. Furthermore, the effect of frequency slope is usually equalized by pre-emphasis of the frequency response of most analogue telephone sets. The E-Model does not cover this impairment.

7.10.2 Circuit noise

Another parameter that needs to be considered only in specific applications is *circuit noise*. Sources for circuit noise historically were associated with analogue networks using FDM systems and with certain switching systems; these have been subject to "noise-planning". However, when switching equipment (PBXs) and transmission elements within the private network are designed according to international and national standards with respect to noise (e.g. Recommendations Q.551 [36], Q.552 [37], Q.553 [38], Q.554 [39], [40] or [52] for switching systems), its influence to speech quality is negligible. Only in special cases, e.g. interference to analogue cable sections by power lines or other noise sources, should noise be part of planning calculation. For such cases, the E-Model provides for the input of the circuit noise parameter (Nc), with a value referred to the 0 dBr-point. For further information, see clause 9.

7.10.3 Listener echo

Likewise, impairments caused by *listener echo* are included in the calculation algorithm of the E-Model, with the corresponding input parameters *Weighted Echo Path Loss (WEPL)* and *round trip delay (Tr)* of a 4-wire loop, when part of the connection. The effect of listener echo generally depends on the same connection characteristics and network elements as the effect of talker echo. Therefore, if sufficient control of talker echo in a connection is provided, the listener echo can be assumed to be of minor influence.

7.10.4 Sidetone

Other parameters included in the E-Model may have important influence to speech quality but are usually not subject to planning. This is generally the case for parameters associated with various analogue and digital telephone sets. The relevant parameters for telephone sets are the *Sidetone Masking Rating (STMR)*, and the *Listener Sidetone Rating (LSTR),* in conjunction with the "D factor", a value related to the design of the handset. Analogue telephone sets, STMR, and to some extent LSTR, depend on the degree of matching between the balance impedance of the telephone circuit and the input impedance of the terminating line interface in the PBX, in conjunction with the impedance of the ports interfacing 2-wire facilities.

To simplify the task of transmission planning, and to keep these impairments as low as possible, it is recommended that telephone sets in private networks comply with relevant international or national standards concerning transmission parameters. Furthermore, it is recommended that analogue interfaces are designed to an appropriate "impedance strategy". For more information refer to Supplement 31 to G-series Recommendations [14], [40] or [52] and to A.1 and A.4 in Reference [59].

7.10.5 Room noise

One other parameter, not usually subject to transmission planning, is *room noise*. Under certain unusual environmental conditions, room noise, at both the talker and listener sides, may have a significant impact on speech quality. The E-Model includes room noise, separately for the send and receive sides as a source for impairments. The room noise, in as normal an office environment as can be assumed for the business domain of private networks, can be expected in a range of 30 to 50 dB(A). Within this range, any impairment encountered due to room noise will be minor, therefore both parameters Ps and Pr can be set to their default values in the E-Model. However, in specific applications with a significantly higher ambient noise level, such as telephone sets located in factory environments, an average value established by noise level measurements should be used as an input parameter to the E-Model.

7.10.6 Comfort noise, voice activity detection

Noise contrast occurs when background noise is interrupted due to digital speech processing, such as echo cancellation using centre clippers, and *voice activity detection* (silence removal). *Comfort noise* is noise that can be introduced to mask the negative effects of noise contrast. Recommendations on *noise contrast* limits, and comfort noise values are for future study.

For *comfort noise* insertion, some digital cellular systems (e.g. GSM) use an approach where noise parameters are extracted at the sending end and transmitted to the receiving end at a low bit rate. It is then possible to reconstruct (to a good approximation) the background noise. This approach should provide superior subjective speech transmission performance for circuits using *voice activity detection* and comfort noise insertion. The *voice activity detectors* and comfort noise generators, described in Annex B/G.729 and Annex A/G.723.1, both operate in this fashion.

The best (subjective) performance will be realized when the noise inserted at the receiving end matches, as closely as possible, the background noise at the sending end. The following comments on Comfort Noise Generators (CNGs) can be made:

- the noise used should match the background noise, both in frequency content and level;
- level of the inserted noise should match that of the background noise; appropriate level measurements and adjustments should be done using dBm0p;
- the time course of changes in the level of the inserted noise should match, as closely as possible, the level changes that occur in the background noise.

Nevertheless, it should be observed that these types of artificially generated noise, as stated previously, are of a substantially different nature than traditional noise-related parameters which are taken account of by the E-Model. Hence, noise figures provided in conjunction with CNG or VAD systems should never be used as input values for E-Model calculations in order to avoid wrong results.

7.10.7 Forward error correction

In the event that a digital sequence, which has been transmitted does not (or with errors) arrive at the destination, it is a common method for data transmission to notify the sender of this fact and to initiate a second transmission of the same sequence. For telephony speech transmission this method is not feasible, because this would result in excessive end-to-end delay and thus decrease the speech transmission performance (see further explanations in 7.7 on delay variation versus time).

One approach of replacing missing or errored speech samples is to use the last speech sample (received earlier) again. This method is very effective for waveform codecs (e.g. Recommendation G.711 [15]) where speech samples are of very short length (e.g. 125 μs) and are proportional to the actual waveform of the speech signal. However, this method fails for non-waveform codecs (e.g. Recommendation G.723.1 [17]) where one frame comprises a much longer part of the speech signal (e.g. 30 ms) and depending on the specific type of low bit-rate codec one missing frame at the receiver may cause an errored output of the decoder for a time lasting several times as long as the speech sample contained in one frame.

Hence, in conjunction with low bit-rate coding schemes (which are in most cases non-waveform codec types) *Forward Error Correction* (FEC) should be used in order to improve end-to-end speech transmission quality. This leads to increased robustness of the codec with respect to channel errors. Proper robustness is required especially for radio sections and IP based transport facilities, because there it is more likely that speech samples are lost or being discarded than in traditional wire-bound systems. FEC may appear in different realizations in various parts of a connection:

1) as internal part of the coding algorithm (e.g. Recommendation G.729 [22]);
2) as an algorithm in conjunction with the radio section (e.g. GSM);

3) in connections involving an IP based network (in the InterWorking Function (IWF) unit at the interface to the PSTN/ISDN or in the IP terminal).

The basic principle of FEC is simply that, that a frame or a packet contains, beside the distinct speech sample, additional information, which is related to earlier or later speech samples. At the receive side this additional information can be used to re-construct missing or errored speech samples.

The principle of FEC can be explained with a very plain and simplified example:

– In a packet based network, each packet may contain two consecutive speech samples and each speech sample may be contained in two consecutive packets (e.g. packet {n} contains speech samples {m} and {m+1}, packet {n+1} contains speech samples {m+1} and {m+2}, and so forth). In this case, even if every second packet will be lost or discarded, all speech samples will be available without errors at the receive side.

In practice, however, FEC algorithms are very complex, in order to cope with the real channel error behavior and the transport capacity. With respect to the first and second type of FEC, as listed afore, their properties are part of the respective Standards and their effectiveness are included in the equipment impairment factor (Ie) for such codecs under errored conditions (see Tables 2b and 2c).

For the third type of FEC no standardized method is available up to now. Thus, the effect of FEC of the third type evades practical transmission planning approaches; this is for further study.

7.11 Synchronization

Proper synchronization design is part of the network planing strategy, because synchronization impairments will affect the quality of the calls: the networks should be synchronized as defined in the series of documents [55], Annex F of [40] and ISO/IEC 11573 [61], in order to achieve the slip rate objectives defined in Recommendation G.822 [25].

Synchronization impairments result in slips, creating various degradations. This is especially important in connections where echo cancellers are deployed, because echo cancellers need, for reasons inherent to echo cancellation techniques, a time-invariant near-end echo path in order to work properly (see Note 2 of 3.2/G.165 [11]). Slips in the echo path of the echo canceller will create phase shifts, which lead to periodic divergence/reconvergence of the echo canceller. This is a new kind of impairment that is not addressed in this Recommendation and by the current version of the E-Model.

8 Calculation of end-to-end parameters

As described in clause 9 in more detail, the use of the E-Model for planning calculations requires the correct handling of this model to avoid wrong results. The E-Model is based on a basic reference configuration, separated into a send side and a receive side with a "virtual centre" referred to as a 0 dBr-point. One of the main tasks when using the E-Model is to transform the different end-to-end connections into a format which is similar to the basic reference configuration of the E-Model. In this context it is necessary to:

- define the virtual centre to be used as the 0 dBr-point;
- perform "pre-calculations" for the different parameters; and
- clearly identify existing echo paths.

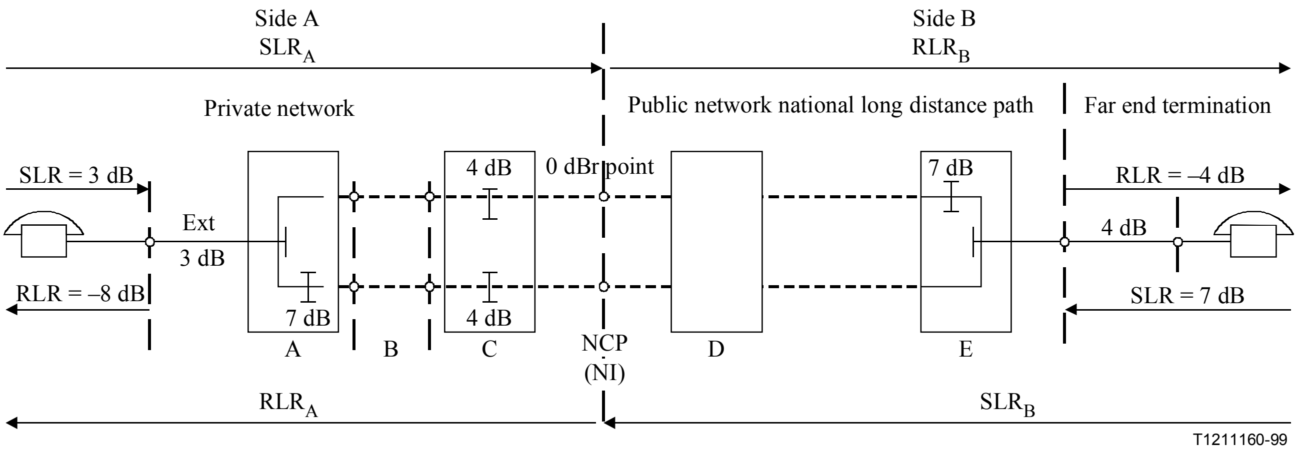
The following subclauses provide guidance for these calculations for all main parameters which are subject to planning.

8.1 Overall loudness rating

Considering the basic principle of this Recommendation, that transmission planning is based only on end-to-end performance, any impairment caused by a too high or a too low volume of a connection is consequently related only to the Overall Loudness Rating (OLR). In practice, however, it is necessary to separate the OLR at the 0 dBr-point into a Send Loudness Rating (SLR) and a Receive Loudness Rating (RLR) for both transmission directions, since the E-Model requires these input parameters separately. To obtain the correct input values SLR and RLR for the E-Model, a "pre-calculation" of these parameters is necessary. The first step, then, is to define the 0 dBr-point in the considered configuration, to be followed by the summation of all SLR/RLR values of the respective telephone sets and all distributed losses along the connection on each side of the 0 dBr-point.

For the definition of the 0 dBr-point in a given configuration, a point should be defined where the speech signal is in a standard 64 kbit/s frame and in an 8 bit PCM A-law or μ-law form. In most of the network applications described in this Recommendation, the digital interface between the private network and a public network will serve as the 0 dBr-point.

The principles of pre-calculation are demonstrated with the configuration shown in Figure 13.



NOTE – The values shown in Figure 13 are for illustration only and do not necessarily reflect actual circuit or loudness values in different regions.

Figure 13/G.108 – Reference connection for the calculation of SLR and RLR

In the private network, an analogue telephone set with SLR = 3 dB and RLR = –8 dB is connected via an analogue extension line with a loss of 3 dB to PBX A. The hybrid in the extension line circuit inserts a loss of 7 dB in the receive direction. The transmission element B between PBX A and transit PBX C is a digital facility (leased line or tie trunk) with bit-transparent routing. Transit PBX C inserts a digital loss pad of 4 dB in each direction. The path within the public network for a national long distance call provides a fully digital routing between the private network interface (NCP or NI), and the local office E, at the far end. Within this local office, the 4-wire/2-wire conversion in conjunction with an A/D-D/A conversion inserts a receive loss of 7 dB. The far-end termination is formed by a single analogue telephone set connected to the local office E via an analogue subscriber line with a loss of 4 dB. The combined loudness values for the far-end termination (SLR and RLR of the set and cable loss) are assumed to be SLR = 7 dB and RLR = –4 dB.

For this configuration, the 0 dBr-point can be defined at the Network Connection Point (NCP) or NI, the interface between private and public network. With reference to the 0 dBr-point, the configuration is divided into a Side A and a Side B (which are not automatically identical with the send side and receive side of the E-Model). The summation of SLR/RLR values and circuit losses is performed for both transmission directions so that values for SLR_A, SLR_B, RLR_A and RLR_B are obtained. The four loudness rating values can be calculated as follows:

Side A:

		Set > 0 dBr-point	**0 dBr-point > Set**
Telephone set	SLR	3 dB	–
	RLR	–	–8 dB
Extension line		3 dB	3 dB
PBX A		0 dB	7 dB
Leased line (tie trunk) B		0 dB	0 dB
Transit PBX C		4 dB	4 dB
Sum at 0 dBr-point		$\mathbf{SLR_A = 10\ dB}$	$\mathbf{RLR_A = 6\ dB}$

Side B:

		Set > 0 dBr-point	**0 dBr-point > Set**
Far-end termination	SLR	7 dB	–
	RLR	–	–4 dB
Local exchange E		0 dB	7 dB
Routing in public network		0 dB	0 dB
Sum at 0 dBr-point		$\mathbf{SLR_B = 7\ dB}$	$\mathbf{RLR_B = 3\ dB}$

It is very important to select the correct values as input values for the E-Model from these results (see clause 9). According to the basic principle of the E-Model, the expected quality is calculated as perceived at the receive side, i.e. the customer of the private network is the listener (receive side) and the customer at the far-end termination is the talker (send side). Therefore, the values for the transmission direction from far-end termination to private network should be used as input values to the model:

$$SLR_B = 7\text{ dB} \qquad RLR_A = 6\text{ dB}$$

The OLR for this transmission direction is:

$$OLR = SLR_B + RLR_A = 7\text{ dB} + 6\text{ dB} = 13\text{ dB}$$

The OLR for the opposite direction is:

$$OLR = SLR_A + RLR_B = 10\text{ dB} + 3\text{ dB} = 13\text{ dB}$$

For the configuration and the values in this example, the OLR is equal in both transmission directions. Therefore, in this case, the additional calculation for the transmission direction from private network telephone to the far-end termination is generally not necessary with respect to impairments due to OLR.

However, in the exceptional case that the telephone set in the private network is located in a noisy environment, both transmission directions should be considered and calculated separately. Assuming a measured mean room noise of 65 dB(A) at the location within the private network and the default value for the room noise of 35 dB(A) at the far-end termination the calculation procedure is as follows:

For the direction from the far-end termination to the private network (private network is the receive side) the values for SLR_B and RLR_A should be used and the parameter Pr (room noise at the receive side) is set to 65 dB(A) with Ps (room noise at the send side) remaining at the default value. For all other parameters at their default values, the E-Model Rating R for this direction (perceived quality of the private network customer) is calculated with R = 66.9.

For the direction from the private network to the far-end termination (receive side at far-end termination) the values SLR_A and RLR_B are used, the parameter Ps is set to 65 dB(A) and Pr remains at the default value. The E-Model Rating R in this direction is R = 57.1, a result which shows that the impairment of high room noise is different to the listeners at both ends.

When calculating the loudness rating values, it is advisable to also control the requirements at the NCP (NI) with respect to the SLR and RLR of the private network as described in 7.9. For the example of Figure 13, the NCP is identical with the defined 0 dBr-point, therefore the calculated values for SLR_A and RLR_A at the 0 dBr-point can be used for assessing compatibility with the NCP/NI requirements. In this example, the requirements of 7.9.1 (for the European approach) with the given minimum values of SLR = 7 dB and RLR = 3 dB are met.

8.2 Talker echo

For the impairment due to echo effects, the given configurations should be investigated with respect to sources for signal reflections, usually hybrids. For E-Model calculations of the perceived quality, two input parameters must be pre-calculated:

- mean one-way delay T in ms;
- Talker Echo Loudness Rating (TELR) in dB.

For more information about the effect of echo, see Recommendations G.126 [8] and G.131 [9].

For the investigation of echo (especially in conjunction with the E-Model), it is important to recognize that the parameter TELR is a function of the SLR and RLR of the **talker's** telephone according to the formula:

$$TELR = SLR + EL + RLR$$

where EL is the (weighted) echo loss of the echo path. Thus, the SLR and RLR values of the telephone set are included in two different input parameters of the E-Model; as part of the TELR, and as direct input of SLR and RLR for calculations of other impairments.

NOTE – The use of SLR and RLR in two separate pre-calculations is based on the principle and the algorithm of the E-Model as described in detail in Recommendation G.107 [3]. In most applications, the model will be used in conjunction with a computer program. The handling of the different input values, however, may deviate between such programs and the E-Model. Therefore, it is necessary to clearly identify what input values are required and in which form they are entered in the specific program to avoid getting wrong results. For example, if the program does automatic pre-calculations based on connection variables affecting SLR and RLR, both the loudness rating and echo effect pre-calculations are subject to changes in these variables. For further information see clause 9.

Note also, that impairments caused by talker echo are referred to the talker at the "receive" side, in accordance with the principles of the impairment factor method and the E-Model. This should be taken into account carefully when selecting parameters as input for the E-Model for each particular configuration.

In some applications, the reference connection considered may contain more than one echo path. Subclauses 8.2.1, 8.2.2, and 9.5 provide guidance for the calculation procedure for single as well as for multiple echo path cases.

8.2.1 Calculation for connections with one echo path

The following calculation example is based on the reference connection shown in Figure 14. A digital telephone set with SLR = 7 dB and RLR = 3 dB and with a mean one-way delay of 1.5 ms is connected to PBX A in the private network. PBX A is connected to the transit PBX C via a digital tie trunk or leased line B with bit-transparent routing and a mean one-way delay of 2 ms. Transit PBX C inserts a digital pad of 3 dB in each transmission direction for voice calls. PBXs A and C contribute with a mean one-way delay of 1 ms each.

The public network provides fully digital routing up to the far-end local office E where a hybrid is a source for signal reflections, and therefore part of the echo path. The far-end termination is assumed to be a single analogue telephone set. For the echo calculation, the values of this far-end termination are not relevant, since it is not part of the echo path. Based on information provided by the public network provider, the mean one-way delay within the public network for a national long distance call is assumed to be 10 ms. The hybrid in the local office E is assumed to have an average (weighted) echo loss (EL), of 24 dB, including a loss of 7 dB in the receive (towards the end termination) path.

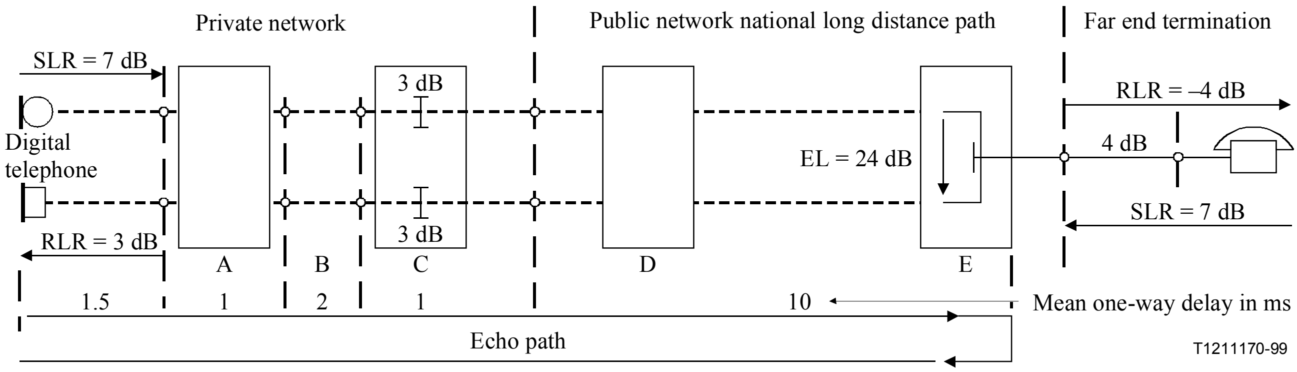


Figure 14/G.108 – Reference connection for echo calculations with one echo path

A clear identification of the echo path should be the first step of every calculation. For the example illustrated in Figure 14, the echo path consists of the digital telephone set (where the talker echo is perceived), the complete path via transmission and switching elements between that telephone set and local office E, the Echo Loss (EL) in local office E, and the return path. This echo path is identified in Figure 14.

The second step is to calculate the two relevant input parameters of the echo path for the E-Model: the mean one-way delay T, and the TELR. It is important to note that only the "one-way" delay serves as the input value for the model, even though the echo signal is delayed by twice the value of that delay (assuming the same connection path for both directions).

Once the echo path is defined, the first input value for the calculation, the mean one-way delay T is the sum of the (one-way) delay values of the different elements. For the example in Figure 14 above, this sum calculates to T = 15.5 ms. The second input value to the E-Model, the TELR, is calculated as the sum of all losses along the whole echo path (both directions) including the SLR and RLR of the (talker's) telephone set. For the example above, this value is calculated as TELR = 40 dB.

Based on an E-Model calculation using T = 15.5 ms and TELR = 40 dB as input values, the configuration of Figure 14 yields an E-Model Rating R = 85.8, a value which falls into the "high" category of speech transmission quality according to Table 1.

8.2.2 Calculation for connections with two echo paths

Due to increasing digitalization in public and private networks, configurations with more than one echo path will occur with less and less likelihood and such configuration should be avoided in network design. Nevertheless, the following gives an example of how to handle such a configuration; it is derived from the configuration of Figure 14 and is based on the use of a cordless telephone, connected to an analogue port at PBX A. The additional delay introduced by a cordless telephone termination is not negligible, hence the cordless telephone forms a separate echo path together with the end-to-end echo path via the public network (which is the same as in the configuration of Figure 14). This reference connection with its two effective echo paths is illustrated in Figure 15. In such a configuration the talker may be disturbed by two different echoes with different volume and different delay.

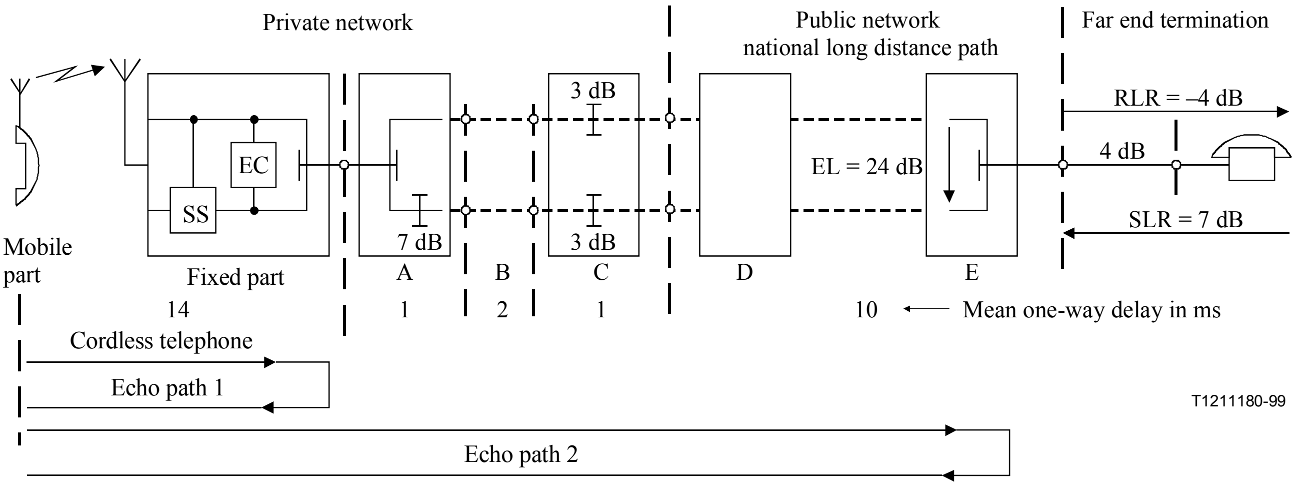


Figure 15/G.108 – Reference connection for echo calculations with two echo paths

To determine the expected overall quality, a special application of the E-Model needs to be derived here, because the E-Model does not currently accommodate and calculate two simultaneous impairments due to talker echo. The E-Model does not allow a simple addition of these two pairs of T and TELR, resulting in new input parameters. The fact, that two echo sources are present simultaneously is more complex and needs further investigation. Special psychoacoustic effects, such as masking of one echo by the other, may influence the perception of quality.

For this situation, it is generally recommended to follow a three-step approach. The first step is to consider the impairments due to echo only, calculating the two echo paths 1 and 2 separately: i.e. for each echo path, the relevant parameters of mean one-way delay, T, and TELR should be determined independently, assuming the other echo path as not existing. In the second step, the calculated impairment-values for the two echo paths should be combined with a specific formula (not provided by the E-Model). Finally, this result is combined with the result of E-Model calculations for all other impairment parameters (e.g. Ie value for ADPCM) in this configuration. This procedure and how to handle the E-Model in this special application is described in more detail in 9.5 together with general guidance for the application of the E-Model in planning practice. The following paragraphs describe how to determine the parameters TELR and T for the two echo paths.

For the cordless telephone in Figure 15, the fixed part (base station) should provide a 4-wire/2-wire conversion for the connection to an analogue 2-wire interface. That hybrid is then a source for signal reflections and forms echo path 1 together with the mobile part via the air interface. When determining the relevant parameters mean one-way delay T and TELR of this echo path, parameters specific to the cordless telephone, should be taken into account.

Cordless telephones conforming to [50] and [53] insert an additional mean one-way delay of approximately 14 ms between acoustic interface of the portable part and the network interface of the fixed part (see Table A.1). This amount of delay is likely to cause echo effects in many applications. Therefore, this configuration requires precautions to suppress the reflected signal, via the use of an Echo Canceller (EC), together with a Soft Suppressor (SS). More detailed information about echo control in specific cordless telephones is provided in Annex C.

When determining the TELR of this echo path 1, these "integrated" echo control devices must be taken into account. Figure 16 shows in more detail the relevant components for echo control in a terminal, most of which reside in the fixed part. Inside the fixed part a virtual "reference point" is defined, where the speech signal, which is transcoded into ADPCM for transmission via the air path, is again present in standard PCM format.

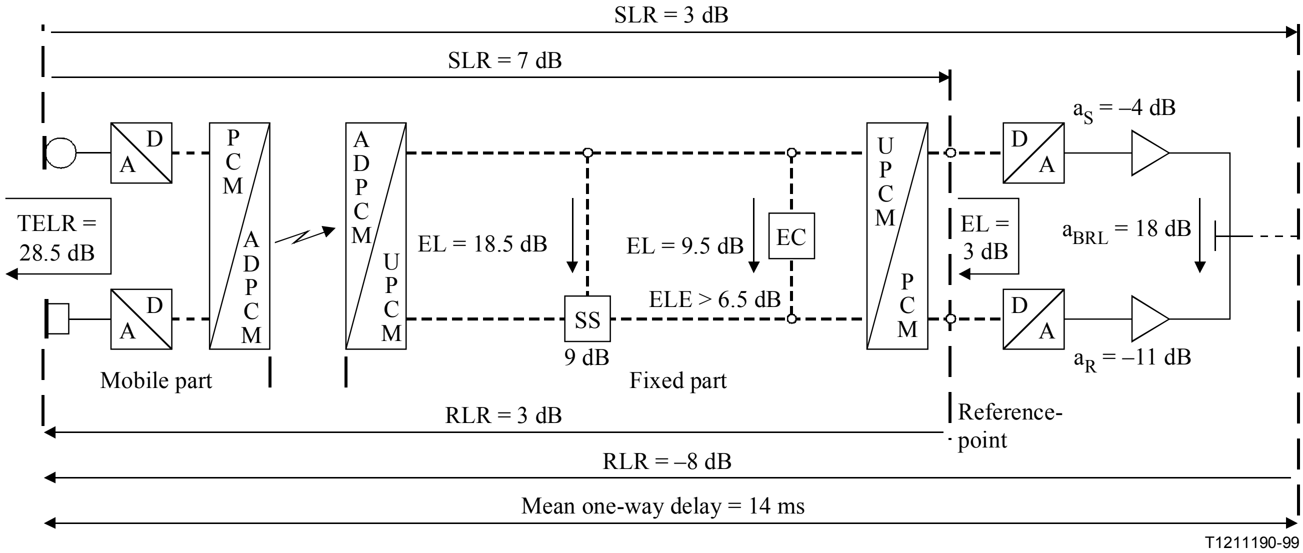


Figure 16/G.108 – Details inside a cordless telephone with respect to echo control

Per [50] the system is adjusted to SLR = 7 dB and RLR = 3 dB referred to this reference point. These loudness rating values would not meet requirements when connected to a 2-wire analogue interface. In the example of Figure 14, values of SLR = 3 dB and RLR = –8 dB are assumed to meet such requirements (other national/regional requirements may apply). The hybrid circuitry inside the fixed part (at the right side of the reference point in Figure 16) not only provides the necessary A/D-D/A and 4-wire/2-wire conversion, but also includes, in its analogue part, a level adjustment of $a_S = -4$ dB in sending direction and $a_R = -11$ dB in the receive path in order to meet the required SLR/RLR values at the 2-wire interface.

The calculation of TELR for echo path 1 should be performed in several steps. For the hybrid, an average balance return loss $a_{BRL} = 18$ dB can be assumed if correct impedance matching, using complex balance networks, is provided. For the entire hybrid circuitry, this balance return loss is reduced by the sum of the adjustments a_S and a_R, resulting in a net echo loss (EL) of 3 dB. To improve this low echo loss, an Echo Canceller (EC), is used, compensating partly for the reflected signal. This is expressed in terms of an Echo Loss Enhancement (ELE), which, in the DECT standard, requires > 6.5 dB. The echo loss is thus increased to (3 + 6.5) dB = 9.5 dB. A further improvement is available by the insertion of a Soft Suppressor (SS), in the receive path (towards the telephone) inside the fixed part. This SS inserts an additional loss of 9 dB (the DECT standard suggests 9 to 12 dB) whenever a signal is detected in the send path. The resulting value of

(9.5 + 9) dB = 18.5 dB, together with the loudness values of SLR = 7 dB and RLR = 3 dB, leads to a TELR of 28.5 dB for echo path 1. The mean one-way delay of echo path 1 is 14 ms.

NOTE – Although the values of SLR and RLR are referred to the reference point, they can also be assumed at the left Uniform PCM (UPCM) point in Figure 16 since there is no additional loss inside the UPCM path. It should also be noted, that the SS is only enabled if a signal in the send path is present, which is valid for the consideration of talker echo, but it is disabled during listening when the talker is silent. Therefore, the RLR values in Figure 16 are shown for a disabled SS.

For echo path 2, the calculation of the relevant values for mean one-way delay T and TELR is similar to the calculation in 8.2.1 but with different values. Summing up the values for delay for each network element along echo path 2, as shown in Figure 15, the mean one-way delay of echo path 2 is T = 28 ms.

The calculation of TELR in echo path 2 requires care to avoid wrong results. The send path from the microphone of the portable part via the hybrid circuitry in the fixed part can be summarized as SLR = 3 dB (Figure 16). The echo path continues via the hybrid circuitry of PBX A (0 dB), the digital pad in PBX C (3 dB), to the hybrid circuitry of local office E, with an echo loss of 24 dB.

The return path includes the digital pad in PBX C (3 dB), and the hybrid circuitry in PBX A (7 dB), and the receive path of the cordless telephone which is shown in Figure 16 as RLR = −8 dB. However, this RLR value must be adjusted by the 9 dB loss of the SS (which is enabled during talking and thus is also a factor for echo path 2), i.e. for the purpose of calculating the TELR, RLR = (−8 + 9) dB = +1 dB. Thus, the sum for echo path 2, using the values as shown in Figures 15 and 16, is TELR = (3+3+24+3+7+1) = 41 dB. Note, that the echo canceller in the fixed part does not affect echo path 2 since the processing capability of the echo canceller (which is generally about 4 ms) with respect to the echo path delay is much less than the connection delay.

The relevant values for the two echo paths are as follows:

– echo path 1 TELR = 28.5 dB T = 14 ms;

– echo path 2 TELR = 41.0 dB T = 28 ms.

8.3 Transmission time in echofree connections

As noted in 7.3, very long delay may cause impairments other than echo. Connections which exhibit a significant delay impact the perceived speech transmission quality even if they are designed "echofree" with perfect echo cancelling. For planning purposes, transmission time usually needs to be considered only for long national (e.g. in North America) or international connections or when routing via a satellite link, but also national or internal connections within the network under consideration may be affected, if modern signal processing devices like low bit-rate codecs are inserted. For the use in the E-Model, the absolute delay (Ta), in ms should be calculated as the sum of all mean one-way delay values of the different network elements throughout the entire connection. It should be noted that all elements, including the telephone sets at both ends, should be considered independent of any echo sources such as hybrids and inserted echo cancelling devices. Specific transmission elements or connection elements or even complete routings may have different values of delay in the two transmission directions. In these cases the arithmetic mean of both values should be used.

8.4 Quantization distortion units

The E-Model requires this input parameter in terms of the number of Quantization distortion units (qdu) as described in 7.5. The complete process of coding (analogue to digital conversion) and decoding (digital to analogue conversion) according to the coding laws (A-law or μ-law) as defined in Recommendation G.711 [15] is considered as one qdu. When performing the summation for the entire connection each pair of "coder" and its subsequent "decoder" should be clearly identified.

For coding laws other than those contained in Recommendation G.711 [15] (e.g. ADPCM), the impairments due to distortion should not be expressed in terms of qdu, but rather as the equipment impairment factor (Ie) (see 8.5). When elements which effect the coding (such as digital loss or gain pads, echo cancelling devices or (digital) conference circuits) are part of the connection, a standard value of 0.7 qdu for each such element should be used in the calculation.

NOTE – While some Recommendations still contain qdu value assignments for ADPCM (and some other low bit-rate codecs) the use of such qdu values for transmission planning is no longer recommended.

If the routing of a connection is fully digital, a minimum of 1 qdu should be taken into account, regardless of whether the coder/decoder is located in the digital telephone set or in the line card for the connection of an analogue telephone set. In the E-Model the default value for this parameter is already set to qdu = 1 and should not be modified to qdu = 0 in an fully digital environment.

8.5 Equipment impairment factor

As described in 7.6, modern coding laws will cause impairments due to distortions. Contrary to the standard PCM coding and decoding according to the A-law or μ-law (Recommendation G.711 [15]), these impairments are expressed in terms of "equipment impairment factor" (Ie), instead of Quantization distortion units (qdu). The Ie factors for different coding laws and operating rates are given in Table 2a. For planning practice, only the algebraic sum of all Ie values along the investigated connection should be calculated and inserted into the E-Model as an input value. However, it is very important to clearly identify, using the reference configuration, the actual location of the coder and decoder of such a low bit-rate section, since such a low bit-rate section may include several transmission elements and connection elements. A connection may also include more than one section using the same or different types of such low bit-rate coding, which must be considered for calculation.

8.6 Equipment impairment factor for codecs under conditions of packet loss

Provisional values for the equipment impairment factor for codecs under conditions of packet loss are provided in Tables 2a and 2b. Even if those values are very pre-mature they do give an estimate of the high impact this parameter has on speech transmission quality. Depending on the dynamic nature of most packet based transmission systems, today, the difficulty for the transmission planner will be that there is no network with fixed percentage of packet loss; rather the percentage of packet loss may be variable and depend on other factors like network mechanisms or network load.

Thus, the equipment impairment factor for codecs under conditions of packet loss evades practical transmission planning approaches; this is for further study.

8.7 Delay variation versus time

The impact of delay variation versus time on end-to-end speech transmission performance, as described in 7.7, depends strongly on whether the dynamic adaptation of the receive buffer length is correctly implemented in the equipment and then this parameter can than be neglected in most cases; this is for further study.

9 Application of the E-Model in planning practice

9.1 General

The basic planning principles recommended in this Recommendation are based on the use of the E-Model (see clause 6) for the performing of planning calculations which provide a prediction of the expected quality for a specified configuration. This planning methodology is a departure from earlier transmission planning guidelines which, particularly for private networks, were based on the assignment of limits for the pertinent transmission parameters. Therefore, some introduction and guidance on the application of these new principles and the use of the computational model are given in this subclause.

A detailed description of the E-Model and its associated algorithm is contained in Recommendation G.107 [3]. It should be noted that, to avoid wrong results, the use of this E-Model requires some understanding about the basic reference configuration and of the different input parameters. This is covered in detail in 9.2, 9.3 and 9.4. Subclause 9.5 covers the special configuration with two echo paths.

Because of the number and complexity of the E-Model formulas, the calculations will generally be performed by computer programs. Nevertheless, it is important for the user to be familiar with the program itself, including the handling of input parameters and the limits for the program's application. Information on this topic is provided in 9.6.

The E-Model includes a variety of transmission parameters; however, not all of them are varied for planning purposes (see 7.10). These parameters should be set to, and remain at a default value (which is not necessarily zero) during the calculation run. Furthermore, the algorithm of the E-Model is based on the results of subjective tests, in which the different parameters have been varied only within specific ranges. When using the model with input parameters outside of these ranges, the results obtained will not have been validated. Therefore, the use of such values should be avoided. Subclause 9.7 lists all default values and the valid ranges for each parameter.

9.2 Reference configurations

For the comprehension of the basic principles of the E-Model, it is useful to construct a basic reference configuration of an end-to-end telephone connection in which all transmission parameters with influence on the perceived speech quality are presented. This reference configuration, shown in Figure 17, is basically divided into a "send side" and a "receive side" with a virtual centre referred to as a "0 dBr-point". One of the most important assumptions in the model is, that the perceived quality is referred to the "receive side", i.e. the listener during a call, and to that same side with respect to impairments encountered during talking, such as sidetone, room noise and echo effects. The basic reference configuration in Figure 17 includes a "4-wire loop" to recognize the impairments of talker echo and listener echo; impairments which are only of minor influence in a fully digital (4-wire) connection.

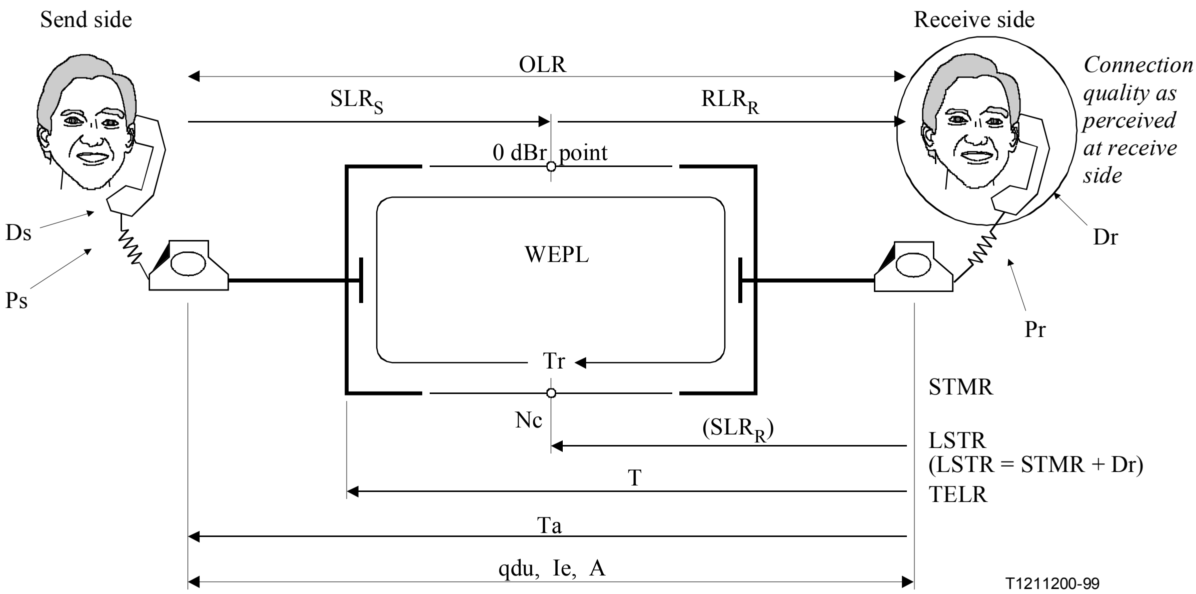


Figure 17/G.108 – Basic reference configuration of the E-Model

In the process of network planning, it is desirable to identify a "most critical" connection for investigation and planning and to designate that connection as the reference connection for that network. This reference configuration, which should contain all relevant transmission elements and connection elements, is useful for identifying and calculating the total amount of any specific parameter in the end-to-end connection. In most situations, however, this reference configuration will deviate more or less from the basic reference configuration of the E-Model. To minimize errors when entering values of different parameters into the E-Model, including results from pre-calculations, it is recommended to transform the reference configuration of the investigated connection into a form as nearly identical as possible to the basic E-Model configuration shown in Figure 17.

The following figures provide guidance for transforming actual planning configurations into E-Model-related configurations. For each of these "working" configurations which are classified as 2-wire/2-wire, 2-wire/4-wire, 4-wire/2-wire or fully 4-wire structures, the appropriate treatment for the applicable parameters is described.

In Figure 18, the same 2-wire/2-wire structure is shown as in the basic E-Model configuration, but here divided into different sections for better comparison with the actual connection. With respect to the virtual centre of the connection and related to the 0 dBr-point, the figure is divided into a "Side A" identical with the "send side" and a "Side B" identical with the "receive side". Both sides are terminated with analogue telephone sets A and B, each with its individual loudness rating values SLR A, RLR A and SLR B, RLR B.

The factors Ds at the send side and Dr at the receive side depend on the handset design and are usually not subject to planning. These factors should be set to their default values for calculation. The parameters, Sidetone Masking Rating (STMR) and Listener Sidetone Rating (LSTR), are assumed to have a fixed relation with the D-factor in the form:

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

and thus influence the perceived quality only at the receive side.

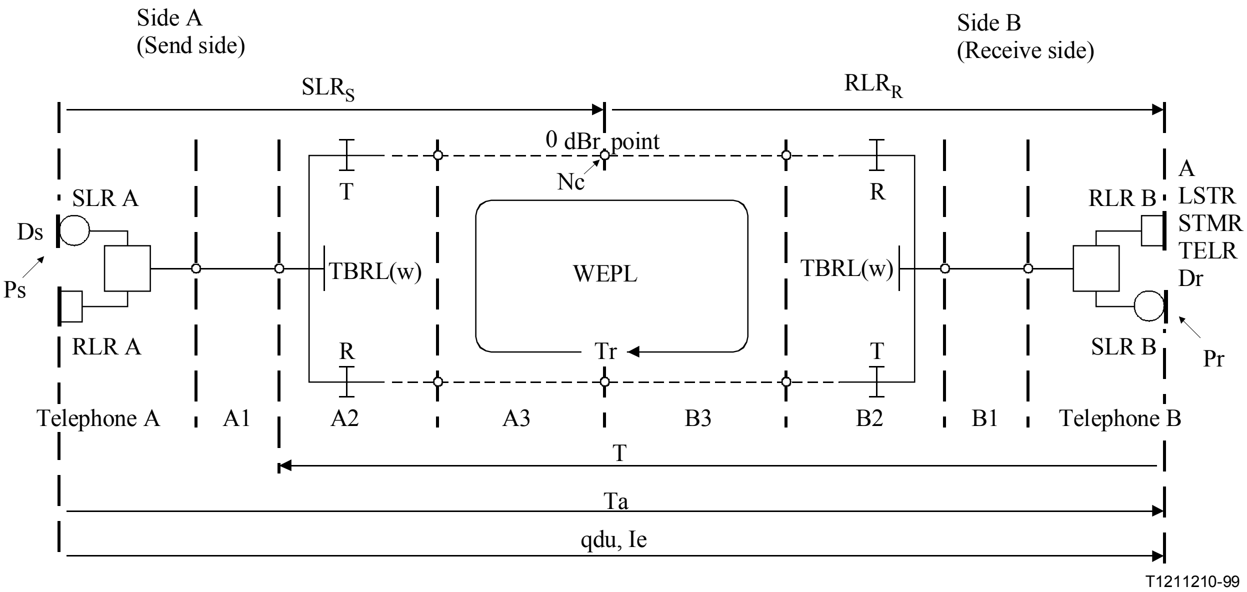


Figure 18/G.108 – Working configuration for 2-wire/2-wire connections

Parameters STMR, LSTR and D are usually not subject to planning and should be kept at their default values except for specific telephone circuit arrangements, impedance provisions, or non-standard handsets designs.

The room noise (Ps), at the Send Side may influence the signal to noise ratio as perceived at the receive side, while the room noise (Pr), at the receive side may decrease the perceived quality via the sidetone path. However, for telephone sets in normal office environments, the values for room noise can remain at their default values.

The Talker Echo Loudness Rating (TELR), as one of the input parameters to the E-Model, needs specific attention and is described in more detail in 9.3.

Sections A1 and B1 of Figure 18 should be understood as the analogue interconnection between the telephone set and the switching equipment where the 4-wire/2-wire conversion is provided. These sections may consist of transmission elements (e.g. unloaded cables) and switching elements (e.g. a PBX with an analogue switching matrix). Therefore, for A1 and B1, mainly loss values need to be calculated.

Sections A2 and B2, normally components (e.g. subscriber line card, trunk card) of switching elements, are shown in detail, since they include the 4-wire/2-wire conversion circuits (hybrids) and the loss pads (R in receiving, and T in sending, direction). In conjunction with the degree of matching (TBRL), these components have a significant influence on speech quality.

The Terminal Balance Return Loss (TBRL), shown in Figure 18 expresses this degree of impedance matching between the balance network of the hybrid and the impedance of the terminating 2-wire section. The weighted Terminal Balance Return Loss [TBRL(w)] is not a direct input parameter for the E-Model, but is required for pre-calculations of TELR.

Finally, sections A3 and B3 in Figure 18 represent the digital part of the connection between the hybrids, including the A/D- and D/A-conversion, and the point in the configuration which has been declared as the 0 dBr-point. The interface between different networks (e.g. public and private) may also be included in sections A3 and B3. These sections may contain several switching elements and transmission elements, including digital loss or gain pads, low bit-rate coding, etc. which contribute parameters such as loss, delay and distortions. In the pre-calculation, these impairment parameters should be accounted for.

The 4-wire loop composed of sections A2, A3 and B2, B3 may contribute to listener echo impairments, characterized by the parameters Weighted Echo Path Loss (WEPL), and round trip delay (Tr). Tr can be calculated as the sum of all delay values (both directions), mostly in sections A3 and B3, while WEPL is sum of all losses and gains inserted in sections A3 and B3, combined with the TBRL and the losses of the R- and T-pads associated with the hybrids in A2 and B2.

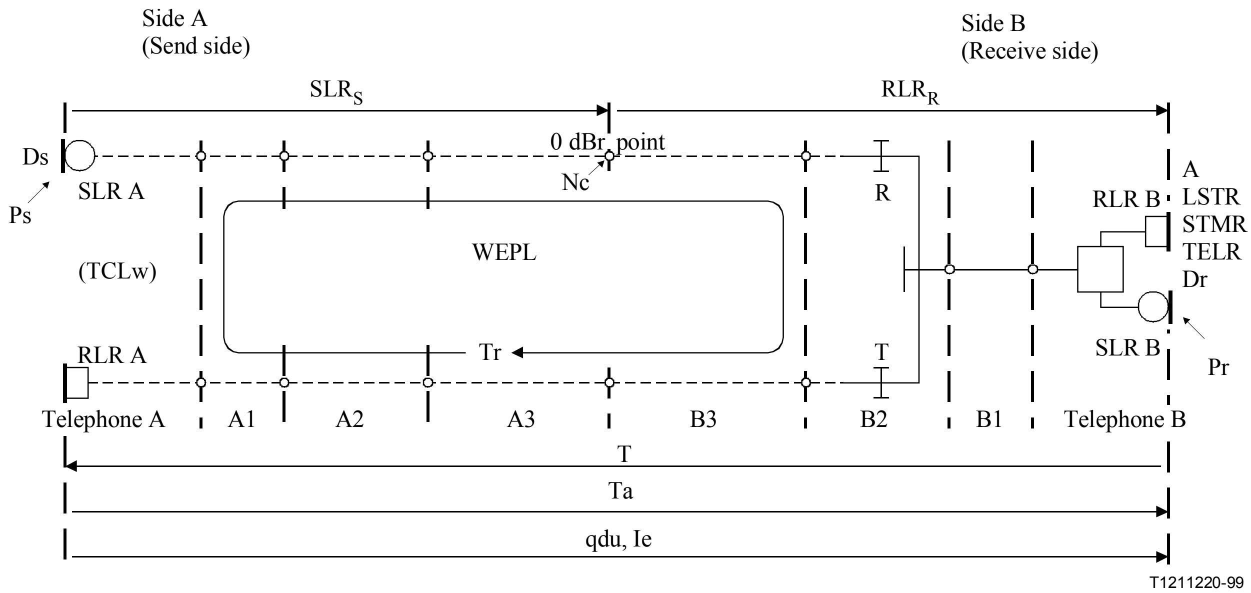


Figure 19/G.108 – Working configuration for 4-wire/2-wire connections

The working configuration shown in Figure 19 can be used for connections with a 4-wire termination (digital telephone set) at the send side and a 2-wire termination at the receive side. In this configuration, the values for SLR A and RLR A are referred to the digital telephone A. Here, an additional parameter, the weighted Terminal Coupling Loss (TCLw) (of the digital telephone set) must be considered. The TCLw characterizes the coupling between receiver and microphone (including any acoustical, electrical and structure borne coupling paths) from which a possible source for signal reflection may arise (see also Recommendation G.100 [1] and Recommendation G.122 [7]).

TCLw, which is not a direct input value to the E-Model, should be included in the pre-calculation of the TELR value to define impairments due to echo perceived at the receive side. TCLw should also be part of the pre-calculations for WEPL. With respect to talker echo and listener echo, the TCLw in this configuration replaces the echo loss (TBRL and associated R- and T-pads) of the hybrid in section A2 of Figure 18. It should be noted that the mean one-way delay, T, in Figure 19 includes section A1 and the telephone, in contrary to the configuration in Figure 18 where the echo path with respect to the talker at the receive side is terminated in section A2.

All other discussions and explanations given for the working configuration in Figure 18 are also valid for this configuration.

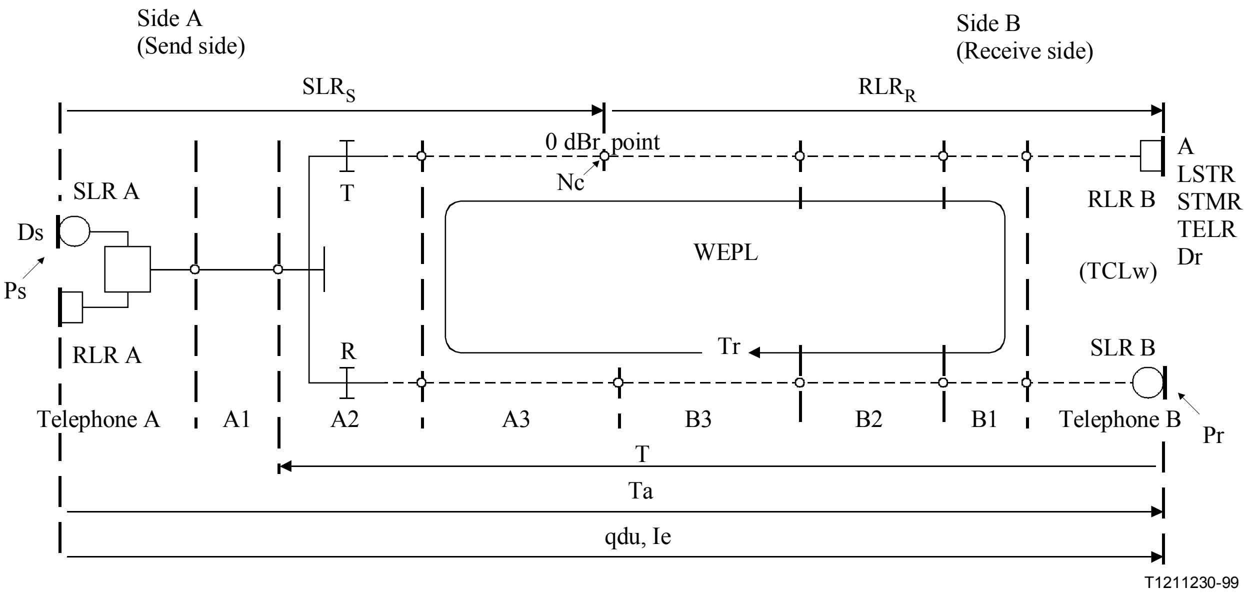


Figure 20/G.108 – Working configuration for 2-wire/4-wire connections

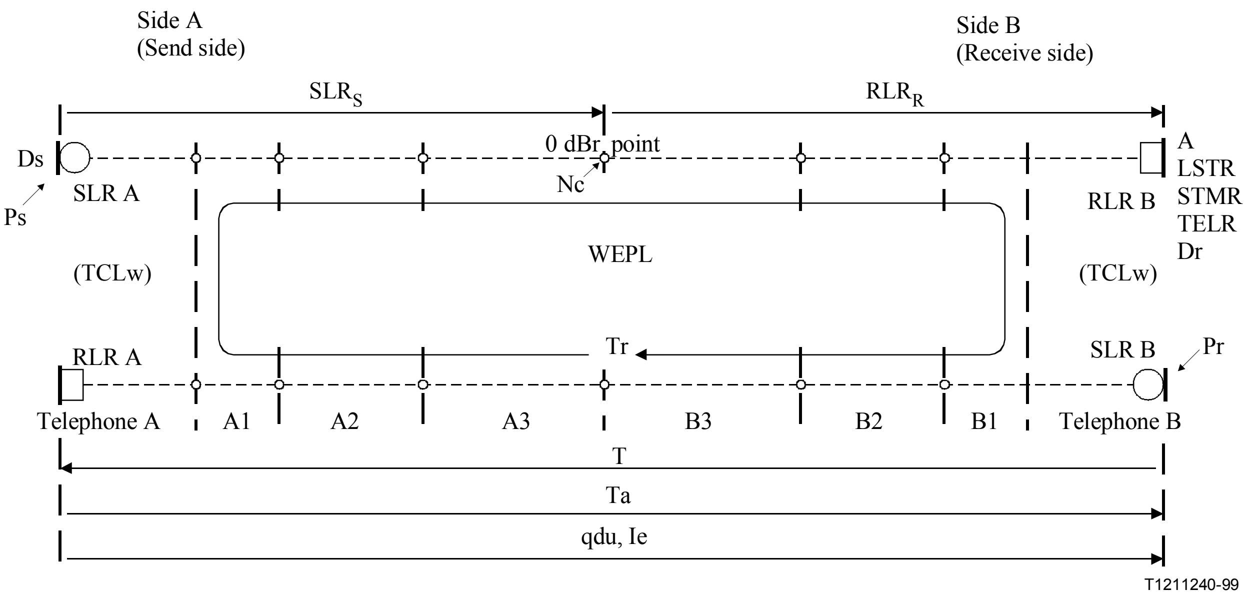


Figure 21/G.108 – Working configuration for fully digital connections

Figure 20 illustrates the opposite working configuration with the 4-wire termination (digital telephone set) at the receive side. Figure 21 illustrates a fully digital connection with 4-wire termination on both ends. The application and pre-calculation of the input parameters in Figure 20, especially for TELR, is nearly identical with the configuration in Figure 18 with the exception that, for calculation of WEPL the TCLw of the telephone B is part of the 4-wire loop. In the same way, both values for TCLw for the telephones A and B in Figure 21 should be included in the pre-calculation for WEPL. It should be noted, that WEPL will be of minor influence, if digital telephone sets with a TCLw of more than 40 dB are used and if the round trip delay (Tr), is low.

9.3 Handling of the input parameters

To obtain correct results, the specific structure of the algorithm and the handling of the different input values of the E-Model should be carefully considered. This is particularly true for Loudness Rating (LR) values. The E-Model algorithm expects, as input, the total value for SLR_S, covering the send side and a total value for RLR_R, for the receive side, both defined between the acoustical interface (microphone or receiver) and the 0 dBr-point. With respect to Figures 18 through 21, these values are calculated as follows: For the send side, the sum of SLR A of telephone A and all sending direction loss values in sections A1, A2 and A3. For the receive side, the sum of all receive direction losses of sections B3, B2, B1 and the loss values RLR B of telephone B. It is highly desirable to verify that the telephone set LR values are consistent with applicable standards via accepted measurement methods, such as Recommendation P.79 [29].

For the investigation of the transmission direction from A to B the values RLR A and SLR B are not relevant.

Besides applicable to the input parameters SLR_S and RLR_R, loudness rating values are also part of the parameters TELR, STMR and LSTR. The values for STMR and LSTR indeed depend on the SLR and RLR of the telephone set. For practical planning purposes, a fixed value for both sidetone parameters, in conjunction with the SLR and RLR, as stated by the manufacturer of the telephone sets, can be used; thus, the input parameters STMR and LSTR are set to their default values and only modified if, in case of analogue telephone sets, the likelihood of a significant impedance mismatch needs to be considered.

NOTE – For telephone sets with volume control in send and/or in receive direction in the handset mode, the SLR and RLR for the default setting of the volume control and the corresponding values for STMR and LSTR, as stated by the manufacturer should be used for planning purposes.

To include the impairments due to talker echo, the algorithm of the E-Model expects two parameters, the mean one-way delay (T), in ms along the echo path and the Talker Echo Loudness Rating (TELR), of the echo path. It is very important, to note that the talker echo is referred to the Receive side. As explained in 8.2, the value for TELR is obtained in a pre-calculation according to the basic formula:

$$TELR = SLR + EL + RLR$$

where SLR and RLR are loudness rating values of the receive side telephone set, i.e. SLR B and RLR B, with respect to Figures 18 through 21. Although RLR B is already part of the basic input value RLR_R, the formulas of the E-Model do not provide an automatic inclusion of the loudness rating values, i.e. RLR_R and TELR need to be pre-calculated separately using the same value for RLR B.

The Echo Loss (EL), in the above formula is the sum of all losses along the echo path, i.e. in sections B1, B2, B3, A3 and A2 of the working configurations in Figures 18 and 20 and additionally in section A1 of Figures 19 and 21. The losses in sections B1, B2, B3 and A3 should be identified and included for **both** transmission directions; e.g. the loss of section B1 in Figures 18 and 19 is included twice within the echo path. For section A2 of Figures 18 and 20, the R- and T-pads and the TBRL(w) of the terminating hybrids are included in EL; likewise the TCLw of the digital sets in Side A in Figures 19 and 21. It is absolutely necessary to use the "weighted" value for the TBRL.

The definition of the circuit noise parameter (Nc), can lead to wrong results if this input value to the E-Model is not included correctly. The E-Model algorithm assumes any noise sources to be defined by noise levels as they would appear at the 0 dBr-point. Usually, circuit noise is no longer a major factor in a digital environment and its specification can be neglected in most applications; i.e. the default value of −70 dBm0p can be used. However, in an analogue environment, such as in sections A1 or B1 of Figure 18, noise may arise, e.g. due to longitudinal interference into telephone cables from power lines. If such noise sources cannot be neglected, the noise level should be pre-calculated into an equivalent value at the 0 dBr-point.

Example: Measured longitudinal interference into a cable in section A1 of the working configuration in Figure 18 will result in a transversal noise level of −50 dBmp at the interface between sections A1 and A2. If the T-pad in the hybrid of section A2 has a loss of 3 dB, the noise level is reduced by this pad accordingly, resulting in a value of −50 dBmp − 3 dB = −53 dBm0p at the 0 dBr-point.

9.4 Interpretation of the results

As described in clauses 6 and 7 in detail, this Recommendation and the recommended planning principles are based on the equipment impairment factor method in conjunction with the use of the E-Model (Recommendation G.107 [3]). The result of any planning investigation with respect to the perceived quality is presented in terms of the E-Model Rating R.

As stated in clause 6, provisions should be made that, in addition to the E-Model Rating R the specific results for Id, Is and Ie are available during transmission planning. This is useful to recognize and to identify the different contributions of each category of impairment when investigating solutions towards an improved quality.

9.5 Application of the E-Model for configurations with two echo paths

As mentioned in 8.2.2, a special procedure applies when a configuration with two effective echo paths is to be calculated with the E-Model. Since the algorithm of the E-Model does not handle two impairments of talker echo, simultaneously contributing to the overall quality, this special procedure described below is necessary.

In general, the procedure is derived from the same basic principle of the E-Model, namely that impairments are additive on a psychological scale. This is also assumed for the two different echo effects with different values for the mean one-way delay, and the TELR, as given for the example in Figure 15, together with further impairments in this configuration. However, a simple addition of the two impairments, caused solely by the two different echo effects, would not be correct. Assuming that for a human listener the quality perception and its judgement is influenced more by the echo with the higher impairment value and that also some effects of masking may occur, a square root addition is recommended.

While the final result of any calculation with the E-Model will be the E-Model Rating R (see clause 6), as a partial result in the course of this calculation the impairment value Id summarizes all impairments caused by delay and echo. For the consideration of impairments due to two echo paths, a pre-calculation with Id values is necessary.

As a first step, the impairments Id_1 for echo path 1 and Id_2 for echo path 2 due only to talker echo are calculated (assuming that the computer program in use is providing the separate impairment values). This calculation is performed separately for each echo path, assuming that all other parameters are set to their default values and only values for T and TELR for the considered echo path are distinctly specified as input values to the E-Model. The two results for Id_1 and Id_2 are then combined to a total (sum-) value Id for both impairments using the following equation:

$$Id = \sqrt{Id_1^{\,2} + Id_2^{\,2}}$$

This value of Id now represents all impairments due to the talker echo effects of both echo paths.

In the second step, all other impairments effective in the configuration considered are included, setting all relevant parameters to their actual values. It is important to note that, in this second calculation, the input values for the parameters mean one-way delay (T), and TELR should be set to their default values (T = 0 ms, TELR = 65 dB). The resulting E-Model Rating R' of this second calculation is then combined with the impairment value for the two echo paths, Id as follows:

$$R = R' - Id$$

For the actual calculation of this configuration with two effective echo paths, a decision about the working configuration (see 9.2) to be used in the first step is not necessary. Since all parameters with the exception of T and TELR are set to their default values, a specific configuration is not relevant.

The configuration of Figure 15 will now be used to illustrate the above principles.

The following list reflects the input parameters T and TELR for both echo paths, which have already been calculated in 8.2.2 and the respective impairment values Id_1 and Id_2 for the two echo paths which can be calculated with the algorithm of the E-Model (see 3.4/G.107 [3]) setting all input parameters, except T and TELR, to their default values (see Table 6).

	TELR/dB	T/ms	Id
echo path 1	28.5	14	17.3
echo path 2	41	28	8.1

The results for Id_1 and Id_2 are now combined to one value using the square root relation:

$$Id = \sqrt{Id_1^2 + Id_2^2} = \sqrt{17.3^2 + 8.1^2} = 19.1$$

For the final calculation, including all other impairments without talker echo effects, one of the working configurations presented in 9.2 needs to be selected for the correct identification of the different parameters and their use as input parameters to the E-Model. For the configuration in this example, the working configuration for 2-wire/2-wire connections as shown in Figure 18 is selected as most appropriate.

Comparing this working configuration with the actual configuration as shown in Figure 15, the telephone B at the receive side in Figure 18 represents the complete cordless equipment, including the fixed part. The far-end termination is represented by telephone A. The 0 dBr-point can be defined between public and private network; i.e. at the digital interface between exchanges C and D. The loudness rating values for the entire cordless system (telephone B) and the far-end termination (telephone A) are used to define the SLR and RLR of the connection with respect to the receiving side, B.

The SLR of the send side referred to the 0 dBr-point is equal to the value SLR = 7 dB of the far-end termination, since no other gain or loss is inserted in the path between the telephone and the 0 dBr-point. The RLR for the receive path between the 0 dBr-point and the cordless equipment includes the 3 dB digital pad in exchange C, the loss of 7 dB associated with the hybrid of exchange A and the RLR of −8 dB (see Figure 16) of the complete cordless system, resulting in RLR = 2 dB. Thus, the RLR can remain at its default value during calculation. It is important to note that, for this calculation, wherein the listening conditions are judged, the Soft Suppressor (SS) should be assumed to be disabled. The other impairment in this connection is the use of 32 kbit/sec ADPCM coding within the cordless system which should be taken into account with the input parameter for Ie set to a value of 7 according to Table 2a.

The effects of listener echo and of absolute delay will be of minor influence in this configuration; nevertheless the corresponding parameters Tr, WEPL and Ta are included in the following calculation for tutorial purposes and for completeness.

When the configuration in Figure 15 is examined with respect to a possible source for impairments due to listener echo, it is seen that there is a 4-wire loop in the public/private network formed by the hybrids in the exchanges A and E. A second 4-wire loop is theoretically formed within the cordless equipment between the hybrid in the fixed part and the TCLw of the portable part.

However, as explained in 7.10, the influence of listener echo is usually negligible as long as there is sufficient control of the talker echo. Calculations for the configuration and values of Figure 15 result in values for the round-trip delay (Tr) = 28 ms and for the weighted echo path loss WEPL = 54 dB.

As explained in 7.3, the absolute one-way delay (Ta) causes major impairments only when the delay exceeds 150 ms, i.e. the value for the absolute delay (Ta) = 28 ms, for the configuration of Figure 15 will have no influence on the result.

The calculation for the second step can now be performed with the calculated input values for SLR = 7 dB, Ie = 7, Tr = 28 ms, WEPL = 54 dB and Ta = 28 ms, while all other parameters, especially T and TELR (the influence of the two pairs of T and TELR for the two echo paths has been considered in the first pre-calculation) remain at their default values. This results in an E-Model Rating R' = 87.4. The final result of this calculation, the E-Model Rating R, is then obtained by combination of the pre-calculated results R' and Id:

$$R = R' - Id = 87.4 - 19.1 = 68.3$$

This result falls into the "low" category of speech transmission quality according to Table 1. This result of low speech transmission quality seems to be surprising, considering that configurations like the one of Figure 15 are in common use (the configuration is nearly identical with a cordless telephone directly connected to a public network) without users' complaints. The major factor contributing to the E-Model Rating R is echo path 1 with $Id_1 = 17.3$, i.e. the echo path via the hybrid of the cordless telephone. For the suppression of this echo, an echo canceller can generally be assumed to be provided. The minimum value of echo loss enhancement, for example, is 6.5 dB. In practice, however, higher values can be expected which, for planning calculations, should be made available by the supplier of the cordless telephone set. Nevertheless, this example supports the recommendation to connect, whenever possible, cordless telephones to the switching equipment via a digital interface, since then only echo path 2 is effective, resulting in an E-Model Rating of R = 79.3, which is at the upper end of the "medium" category of speech transmission quality, being nearly in the "high" category.

9.6 Use of computer programs

The E-Model, the recommended main tool for all planning purposes in this Recommendation, comprises a number of more or less complex formulas (see Recommendation G.107 [3]). Thus, the planner usually relies on computer programs for performing the calculations needed. Irrespective of whether such programs are developed by the planner, or if programs available from other sources are used, it is strongly recommended that the user be fully familiar with the use of this program and the limits of its application.

Computer programs may assist the planner with a variety of features, such as the handling of input parameters, necessary pre-calculations, storing of often-used configurations, etc. As described in 9.3, the correct handling of loudness rating values in conjunction with the necessary pre-calculations is very important and can be supported by such programs. However, it is recommended to provide additional control of all input parameters after performing a pre-calculation run.

Programs may also provide a structure for the input of parameter values as shown in the working configurations of Figures 18 through 21. In this case, pre-calculations for specific sections need to be performed outside the program.

NOTE – If, for the benefit of simplification, certain parameters are not considered during transmission planning, the use of computer programs may deliver slightly different results, depending on the concept of such programs. If, for example, the parameter WEPL is not considered, because it can be neglected, a computer program with numerical input, will perform the calculations with WEPL and Tr at their default values, while a computer program with graphical input (e.g. based on the working configurations, Figures 18 through 21) will perform the calculation with the actual values (slightly deviating from default) for WEPL and Tr. This fact is important for the correct interpretation of calculation results gained by different programs.

Although the recommended planning principle in this Recommendation is based on the E-Model Rating R as the outcome of calculations, the E-Model also includes the capability to calculate the corresponding values for MOS, %PoW and %GoB (see clause 6). Thus, computer programs may provide the results of a calculation in these different presentations. The analysis of results in terms of MOS, %PoW and %GoB requires some knowledge and experience with the underlying subjective tests. Therefore, the final decision about an investigated configuration should be based on the results of calculations in terms of the E-Model Rating R and the corresponding guidelines given in clause 6.

When planning telephone networks with respect to the expected speech quality, as perceived by the user, connections identified as the most critical ones should be used as reference configuration and should be investigated with the E-Model. Such configurations will generally be characterized with one specific set of input parameters, leading to a single value of E-Model Rating R to assess the connection quality. Most of the computer programs, however, will also have a feature to vary one or more input parameters in a given range during the program run and to display the results in the form of graphs or tables. Those features can be helpful to obtain an overview of the influence of different transmission parameters on speech quality and may enable the planner to optimize certain parameters such as loss, though, for practical purposes of planning, those features are not absolutely necessary.

In general, computer programs for E-Model calculations should meet the following minimum requirements:

– Control of all actual parameter values should be possible.
– Input of parameter values outside of the permitted range should be refused (or at least indicated).
– If a program also provides the input parameter Overall Loudness Rating (OLR), as the sum of SLR_S and RLR_R and the feature to vary this OLR, the variation of SLR_S and RLR_R should be performed identically; i.e. increasing or decreasing both by the same steps, half of the OLR-steps each.
– The result should also furnish the specific values of the delay impairment factor (Id), the simultaneous impairment factor (Is), and the equipment impairment factor (Ie); it should be noted, that for the calculation of two simultaneous echo paths as described in 9.5 this is absolutely indispensable.

The correctness of calculations according to the algorithm of the E-Model should be verified; e.g. by using all default values as explained in the following subclause. An additional check, using the parameter settings and the corresponding results as given in the planning examples of Annex B, may also be performed.

9.7 Default values and parameter ranges

The E-Model is based on several transmission parameters. Not all of them are varied during the application for planning calculations, consequently each parameter influences the final result. Therefore, it is absolutely necessary to keep those parameters, which are not specifically addressed or defined in a specific configuration, at their default value. When using computer programs, it is strongly recommended to verify the parameter setting before starting a new calculation run.

The definition of the default values for the E-Model is based on a compromise between a setting that is equivalent to the optimum quality and realistic values for some parameters. Depending on the region of interest, the SLR_S and RLR_R values may deviate by 1 dB from their optimum value to be in accordance with requirements in regional standards for digital terminals. Such standards also provide the basis for the default values of the parameters STMR, LSTR and D. The number of qdu is set to 1 instead of 0 since, in a fully digital connection, a minimum of one PCM coding/decoding process is involved even if low bit-rate coding is used which itself shall only be included into the calculation as an Ie value.

As stated in 9.1, the algorithm of the E-Model is based on the results of subjective tests, varying each different parameter only within specific and realistic range. The setting of a parameter outside this range should be avoided, since the result of calculations is no longer validated. Table 6 gives all default values and the recommended range for each of the parameters. The parameters should be viewed as related to the basic or working reference configurations as shown in Figures 17 through 21.

If all input parameters of the E-Model are set to the default values as listed in Table 6 the result for the E-Model Rating R in this case is R = 94.1 (see Notes).

NOTE 1 – For the calculations shown in this Recommendation the algorithm of the E-Model has been taken from Recommendation G.107 [3] at the time of publication. In case a later revision of Recommendation G.107 [3] does show a refined version of the algorithm, the value of R = 94.1 for all input values default may slightly change. Nevertheless, this Recommendation will still provide valid guidance for tutorial purposes. For actual transmission planning tasks it should, in any case, be referred to the latest version of Recommendation G.107 [3].

NOTE 2 – The mathematically accurate value of the E-Model Rating R for all input values default, which is R = 94.154, should be used to clearly identify the correctness of any computer programs in use. For transmission planning purposes, however, it should be considered that the E-Model algorithm relies on subjective test data which, by their nature, are of limited accuracy; hence, depending on the type of actual transmission planning task, the E-Model Rating R should be expressed with no more then one decimal digit, or even as an integer.

Table 6/G.108 – Default values and recommended ranges for the parameters

Parameter	Abbr.	Unit	Default value	Recommended range	Notes
Send Loudness Rating	SLR_S	dB	+8	0 to +18	1
Receive Loudness Rating	RLR_R	dB	+2	−5 to +14	1
Sidetone Masking Rating	STMR	dB	15	10 to 20	2
Listener Sidetone Rating	LSTR	dB	18	13 to 23	2
D-value of telephone, send side	Ds	–	3	−3 to +3	
D-value of telephone receive side	Dr	–	3	−3 to +3	2
Talker Echo Loudness Rating	TELR	dB	65	5 to 65	
Weighted Echo Path Loss	WEPL	dB	110	5 to 110	
Mean one-way delay of the echo path	T	ms	0	0 to 500	
Round trip delay in a 4-wire loop	Tr	ms	0	0 to 1000	
Absolute delay in echofree connections	Ta	ms	0	0 to 500	
Number of Quantization distortion units	qdu	–	1	1 to 14	
Equipment impairment factor	Ie	–	0	0 to 40	
Circuit noise referred to 0 dBr-point	Nc	dBm0p	−70	−80 to −40	
Noise floor at the receive Side	Nfor	dBmp	−64	–	3
Room noise at the send side	Ps	dB(A)	35	35 to 85	
Room noise at the receive side	Pr	dB(A)	35	35 to 85	
Advantage factor	A	–	0	0 to 20	
NOTE 1 – Total values between microphone or receiver and 0 dBr-point. NOTE 2 – Fixed relation: LSTR = STMR + D. NOTE 3 – This value shall not be modified.					

10 Rules for the insertion of echo cancellers

10.1 Introduction

The increasing digitalization of public and private networks resulted in connections which exhibit substantially higher values of mean one-way delay (e.g. due to signal processing and low bit-rate coding), while nearly all additional loss has disappeared. The assumption of a mainly distance-related delay value is no longer valid. This increases the likelihood of impairments due to echo effects if no arrangements are made in conjunction with careful transmission planning to suppress such effects. More details about the causes and effects of echo and its control are given in Recommendations G.131 [9] and G.126 [8].

Control of talker echo caused by hybrids can be achieved in three different ways:

- Firstly, one could try to reduce or eliminate the reflected signal at hybrids. This method is limited due to the complexities of impedance matching between the hybrid balance network and the 2-wire port.
- Secondly, it is possible to reduce the level of the reflected signal at the talker end through deliberate insertion of loss into the connection. This is a practical solution on many connections; however, since the amount of loss required is proportional to the end-to-end delay, it can result in excessive volume reduction for connections with high delay. Furthermore, loss insertion in a digital connection adds quantizing distortion impairments.
- Finally, the use of echo control devices, as discussed in this clause, is applicable.

In connections where echo control cannot be achieved by mitigation of the reflection at hybrids or by loss insertion, the deployment of echo cancellers will be necessary. The basic principle of an Echo Canceller (EC) is shown in Figure 22. The received signal from party A is modified by the echo estimator which is synthesizing a replica of the echo path and subtracting this signal from the send path. Since the echo path varies for every connection mainly in loss, delay and phase, the process of converging to the new echo path should be fairly rapid, e.g. well below one second. During break-in and double talk conditions, the echo estimator attempts to adapt to this "new echo signal" and may cause a degradation of speech quality and reduction of cancellation. However, several non-standardized algorithms are used to avoid these effects.

At the output of the send path ECs may be equipped with an additional unit called Non-Linear Processor (NLP) or centre clipper. The task of this device is to provide a suppression of residual echo levels below a defined threshold.

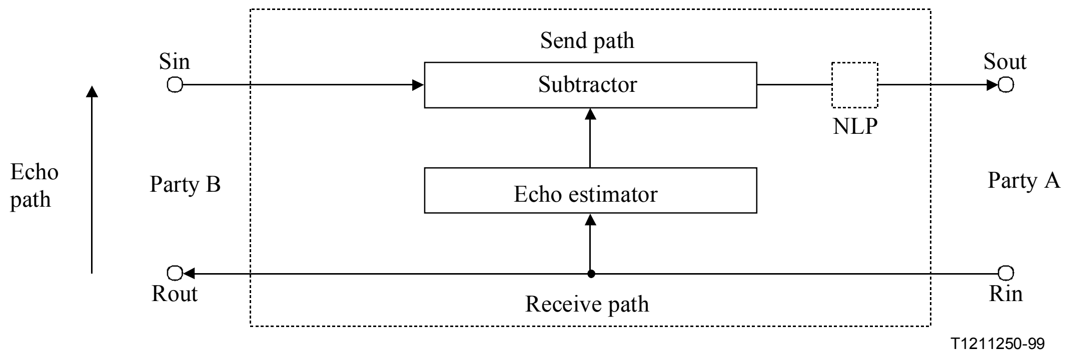


Figure 22/G.108 – Echo canceller

A minimum echo path loss of 6 dB is a requirement common to all ECs which are designed according to Recommendation G.165 [11] to achieve proper operation. As far as ECs are already designed according to Recommendation G.168 [12] this restriction is no longer necessary.

Previously, in mostly analogue networks, echo cancellers or echo suppressors according to Recommendation G.164 [10] were used mainly on long transnational or for international connections and the public network operators were responsible for their correct application. In modern networks, additional delay may require the insertion of echo cancellers also on shorter national connections and within private networks. The application of echo suppressors is no longer recommended.

The location of echo cancellers, if required, in a private network depends on various factors. Echo cancellers can either be used along with the digital interface of a specific switching equipment or transmission element, or may be provided in a pool for a flexible insertion, depending on the type of connection. Specific types of terminals with delay which is not negligible, e.g. wireless telephone terminals, are equipped with integrated echo control devices. This may also be true for systems using low bit-rate coding.

The following subclauses provide guidance to the planner of private networks on the aspects to be taken into account for the correct use of echo cancellers. It should be noted that these rules relate to additional echo cancellers when the need for such additional echo cancellers is indicated by the results of planning calculation. Integrated echo cancellers in specific equipment are, in most cases, not subject to a planning decision, though their technical characteristics should be considered for the decision on additional devices.

10.2 Characteristics of the echo cancellers

The different parameters characterizing the performance of an echo canceller are described in A.1.10. These parameters will be the basis for investigating whether the device is suitable for the designated purpose. The main parameters in this context are the maximum echo path delay to be compensated and the residual echo level. A preference should be given to those echo cancellers, which are in accordance with accepted standards, such as Recommendations G.168 [12] and G.165 [11].

10.3 Limits for the application of echo cancellers

The most important step during planning is the decision whether an echo canceller should be inserted or not. Echo cancellers may, in some situations, cause additional impairments, if they are wrongly inserted or if applied unnecessarily when sufficient echo control is already provided in other networks or network elements. The use of echo cancellers just for "safety reasons" should be avoided.

Since the amount of impairments due to echo depends on two separate factors, the amount of delay and the volume of the perceived echo, which will vary over a wide range and which are independent from each other, an absolute general rule, e.g. a limit value for the mean one-way delay above which echo cancellers are required, cannot be stated. In Europe, the use of echo cancellers is not required when the mean one-way delay in a private network is less than 5 ms for national calls via public networks with fully digital routing, and a far-end termination consisting of a single configuration (see 7.9.1). Furthermore, if this limit of 5 ms is exceeded, this should not be interpreted in such a way that echo cancellers should then be used automatically. The decision should only be based on the result of the planning with the E-Model. As explained in clause 6, the impairment value (Id), as a partial result in the course of this calculation, summarizes all impairments caused by delay and echo and should be available separately during transmission planning.

North American echo canceller guidelines are based on [44] which does not provide specific values of end-to-end delays beyond which cancellers should be deployed. However, there is a widely accepted guideline for dealing with incremental delays caused by the introduction of digital technology. The guideline is that any new system, network or component that, by itself, adds more than 2.5 ms mean one-way delay should provide echo cancellation. The rationale for this guideline is that when a new digital technology becomes part of an end-to-end connection in the PSTN, in general, the connection may not contain an echo canceller, or the capability of the echo canceller is unknown.

If, in a given configuration, impairments beside echo are significant then the analysis of the E-Model Rating R and the partial results for the impairment factors Is, Id and Ie, should be the primary consideration. For values of E-Model Rating $R \geq 80$, a sufficient good quality can be expected: i.e. the use of echo cancellers is not necessary. For lower values of E-Model Rating R, a partial result of the E-Model calculations, the impairment factor Id should be considered. If this impairment factor is in a range of $Id \geq 20$, then the insertion of echo cancellers should be further investigated because this will likely result in a quality improvement. As a general rule, the insertion of echo cancellers should be considered during transmission planning, if values of E-Model Rating $R \leq 80$ are obtained from calculation and talker echo is the main impairment.

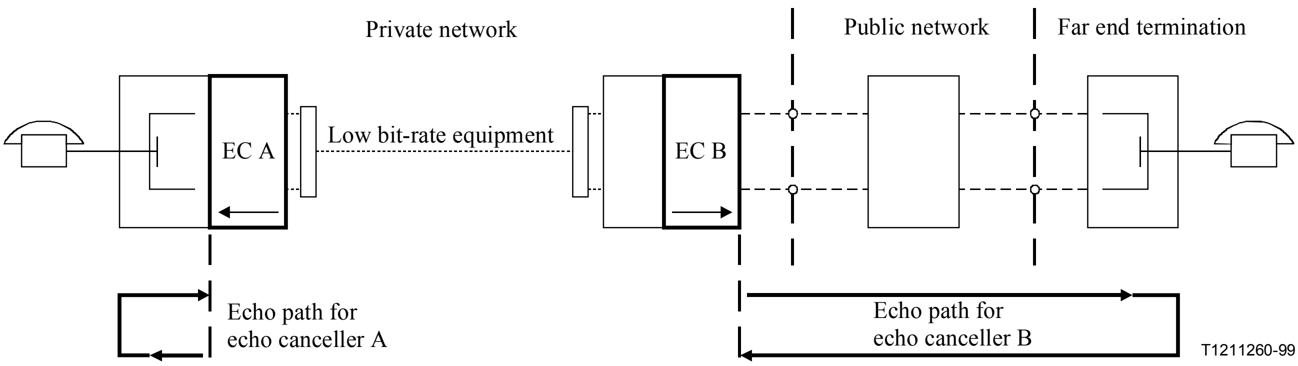
10.4 Determination of the location

If planning calculation results suggest the use of an echo canceller, the next task of planning is to investigate the appropriate location for the echo canceller as well as consideration of its properties. Due to the nature of echo effects, this investigation should be performed not only for the private network but also for the far-end talker; i.e. control of echo effects for the far end by the provision of a sufficient echo loss or echo cancelling should be subject to planning as well.

In a configuration such as shown in Figure 23, the additional delay which requires echo control for this connection is inserted by the private network. Therefore, the private network is responsible for providing the necessary echo cancellers. Furthermore, information about the routings and the types of terminal equipment, which are the factors mainly responsible for the amount of delay, are only available to the private network planner. Therefore, when additional delay will arise in specific configurations within the private network, the decision for, and correct insertion of echo cancellers must be done in the private network domain.

For delays less than 5 ms within the private network, the requirements as given in 7.9 will provide the necessary echo control for the far-end termination.

For higher values of delay, however, the use of an echo canceller A for the befit of the far-end talker besides an echo canceller B for the benefit of the private network talker may be necessary to control echo satisfactory. This may be the case when for example low bit-rate equipment is used within the network under consideration as shown in Figure 23.



EC A Echo canceller for the talker at the far end termination
EC B Echo canceller for the talker within the private network

Figure 23/G.108 – Application of a pair of echo cancellers within the private network

For the selection of the appropriate echo cancellers, one of the most important properties to be investigated, is the maximum echo path delay which the canceller is able to compensate for (sometimes called "tail-end delay"). This value should be 6-8 ms higher than the actual total delay (twice the mean one-way delay) of the echo path. As shown in Figure 23, the echo path for EC A (the arrow indicates the direction of the echo path) is formed only by the hybrid within the PBX, i.e. only a short delay to be compensated for, while for EC B, the echo path includes the entire routing via the public network with correspondingly higher values for the delay for EC B to handle. It is important to note that, for the selection of EC compensation requirements, only the delay values for the corresponding echo paths should be taken into account. The portion of delay in the section between the two ECs is not relevant for the control of echo and only becomes important for very high values as part of the total one-way delay (Ta), which may cause impairments resulting from excessive absolute delay.

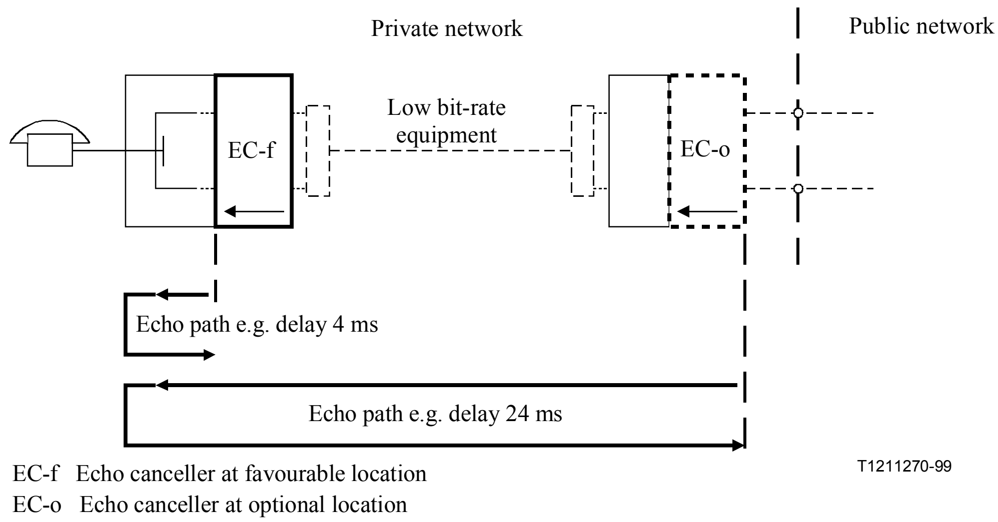
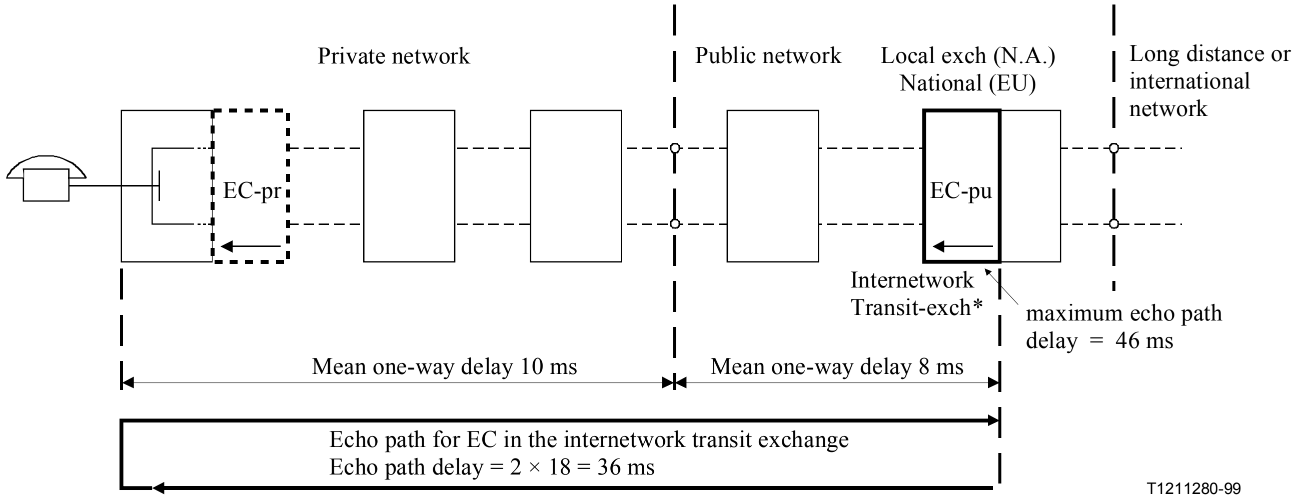


Figure 24/G.108 – Options for the location of an echo canceller

As a general rule an echo canceller should be inserted as close as possible to the echo source. Thus, the location for EC A as shown in Figure 23, associated with the hybrid, can be considered as the most favourable one. However, other locations may also be considered as illustrated in Figure 24 where the EC (for echo control benefiting the far end) is inserted at the PBX which provides access to the public network.

This solution has the advantage that the number of necessary EC devices is reduced due to a more centralized location. Also the automatic insertion of echo cancellers in only those connections which are routed to/from the public network via a low bit-rate equipment may be easier to handle when the echo control is located at a position EC-o as shown in Figure 24. On the other hand, for the position EC-f in Figure 24, echo cancellers with less stringent requirements can be used since the echo path delay is lower. Furthermore, the position EC-o does not meet the requirement for a linear echo path as described in A.1.10 since the (non-linear) low bit-rate equipment is part of the echo path.

For the planning of long-distance (e.g. via an interexchange carrier in N.A.) or international connections, the provision of echo cancellers within the public network can generally be assumed. For such configurations, it should be investigated whether the use of an additional echo canceller within the private network is really necessary. For a connection as shown in the example of Figure 25, the echo path comprises the routing via the public network (local or national) and the private network with its terminating hybrid. The mean one-way delay of the public network is assumed to be 8 ms and that of the private network 10 ms. The echo canceller EC-pu, inserted in the internetwork transit exchange of the public network, is assumed to be able to compensate for an echo path delay of 46 ms. The actual echo path delay for the configuration as shown is calculated with $2 \times (10 + 8) = 36$ ms, well below the maximum permitted echo path delay of the echo canceller EC-pu. In this situation, an additional echo canceller EC-pr within the private network is not necessary.



EC-pu Echo canceller in public network
EC-pr Echo canceller in private network
* Transit-exch: local-long distance (N.A.) national-international (EU)

Figure 25/G.108 – Use of echo cancellers in an internetwork or international connection

The value of 10 ms in Figure 25 for the mean one-way delay within the private network is low, especially for larger networks. When higher private network delay values can be expected, the use of echo canceller EC-pr should be considered, possibly even for some national or longer local carrier connections. This example illustrates, that it is advisable to ask for the relevant information about the properties of the public network, the expected average delay for long-distance and international routing, and the characteristics of the echo cancellers provided by the network.

10.5 Handling of echo cancellers in the E-Model

The proper application of echo cancellers, such as those in conformance with Recommendations G.168 [12] or G.165 [11], is equivalent to an enhancement of the echo loss. For a specific type of echo canceller, which may be inserted into the network subject to planning, there are two feasible approaches as to how this echo canceller can be considered in the E-Model calculations:

- If detailed information on the technical data of this echo canceller is available, those values and especially the value for the echo loss enhancement should be used, to calculate actual values for TELR and WEPL.
- If no detailed information on the technical data of this echo canceller (beside its conformance to Recommendation G.165 [11] or Recommendation G.168 [12]) is available, a survey of the expected quality can be achieved by leaving the input parameters for TELR, WEPL, T and Tr at their default values. Thereby the connection is assumed to be "echofree".

While the first method should be preferred, for detailed transmission planning tasks of actual networks under consideration (i.e. planning a network for a specific customer), the latter method may provide more ease in handling general overview tasks.

11 Realization of planning

11.1 General

The planning of a private network with respect to speech transmission quality addresses, in general, the investigation of connections that have been identified as critical. In most cases, the critical connection is only one specific connection, but is representative for all comparable terminal equipment: e.g. all telephone sets connected to the same exchange at the same location.

Planning is necessary in the case of establishing a totally new private network, but also applies in the case of an existing network being modified or amended with respect to major portions or components. In the latter situation, existing portions can also be subject to planning in order to investigate the expected quality between new and existing terminals and other elements, as well as the effect on quality if access and routing to the public network has been changed for the existing terminals. In general, transmission planning should be executed along the following steps:

- determination of the specific requirements and network features;
- definition of the reference configurations to be investigated;
- determination and collection of all relevant transmission parameters of:
  - elements within the private network;
  - elements within the public network(s);
  - leased lines and tie trunks;
- end-to-end calculation of the expected quality with the E-Model;
- analysis of the results.

These steps should only be considered as a recommendation. Depending on the actual planning project, they can be modified or amended. Also the proposed sequence should be viewed only as guidance. The following subclauses provide a more detailed description of each of these steps.

11.2 Determination of the specific requirements

Depending on the business of the user's company, with respect to the specific telecommunication demands, such as different locations to be interconnected, major types of connections, etc., the following characteristics of a private network are usually predetermined and can be varied only to a small degree for the benefit of speech transmission quality:

- structure and hierarchy of the network;
- routing within the network and to and from the public network(s);
- major types of connections via the public network (international, national long distance, local);
- major types of far-end termination.

Not directly related to the demands by the user, but nevertheless important for transmission planning the following network aspects should also be taken into account:

- type and point of access to the public network;
- use of Virtual Private Networks (VPN);
- type, routing and characteristics of national and international tie trunks and leased lines.

The routing algorithms, as well as routing restrictions within the private network for internal calls and for connections via the public network are of major influence to transmission planning. Detailed knowledge about the routing is necessary to identify critical connections. This should include not only the standard routing but also routing procedures used for network features (e.g. call transfer) or for alternate routing if transmission elements are busy or in failure status. If different transmission elements are used for the routing of internal calls, and for calls to and from the public networks, more economical equipment may be used in the routing paths for internal connections.

In certain private network scenarios, the nature of connections routed to public networks should be determined. As has been stated a number of times in this Recommendation, transmission planning is based on an end-to-end consideration. Consequently, the amount of impairments contributed by the public network portion is important for the planner. As a general rule, the impairments (e.g. delay) contributed by public networks are low for local calls and increasingly higher for national long distance or international connections. If, depending on the business of the user, the predominance of connections via the public network can be assigned to local routing (e.g. within the territory of the local operating company), a higher amount of impairments can then be allocated to the private network for the benefit of more economical solutions. In a competitive environment, several offers of public network access, with different amount of impairments, can be compared on the basis of expected quality and/or possible economical solutions for the private network. It should be noted, that the meaning of "predominance" in this context is a percentage in the range of 90 to 95% (not just more than 50%) for the type of connection considered.

Furthermore, depending on the customer's business, it may be possible to determine a predominance of a specific type of communication partners, e.g. whether in the residential (subscriber line) or business (another private network) domain. During planning, this will be helpful for the selection of the far-end termination as described in Annex A. If a clear definition of the far-end termination is not possible, the type "single telephone set" should be used.

Where special access (e.g. direct access to a higher hierarchy public network node), or special low impairment routing for designated calls are provided by some public network carriers, such offerings should be included in the basic determination of the reference configuration. Furthermore, public network operators generally make available tie trunks or leased lines as well as Virtual Private Network (VPN) features for the connection between the switching elements of a private network. Using the E-Model as a tool, an investigation can be made during planning on a quality/cost relation to select between various offerings of access, routing, and connection options.

11.3 Definition of the reference configurations

As stated in 11.1, the most critical connections should be identified for transmission planning. This "reference configuration" is based on the structure of the private network, in conjunction with the possible routing alternatives, information about type and point of access to public network(s), predominance of connections, and/or the type of far-end terminations (where applicable). The purpose of the reference configuration is to obtain an overview of all relevant parts of the critical connection considered. It is recommended to diagram this configuration including all relevant terminal, switching, and transmission elements that may contribute impairments. This diagram is also advantageous in the other planning steps for the determination of all parameter values, identification of echo paths and their characteristics, and for the calculation with the E-Model.

This reference configuration should be defined as an end-to-end configuration including the telephone sets of the private network and of the far-end termination. In most cases, more than one reference configuration should be taken into account, especially if the structure of the private network and the routing is complex and a clear determination of whether a path is critical or not cannot be made without calculation.

The determination of the reference configurations in large complex private networks is very important to obtaining correct planning results and usually requires a lot of experience and planning practice. When investigating the network, particular attention should be given to elements which introduce additional delay and/or equipment impairments, such as low bit-rate systems and terminals using an airpath (e.g. mobile and cordless terminals). Hybrids (4-wire/2-wire conversions) within the private or public network may form echo paths and should be carefully considered. Although connections via public networks to the far-end termination are usually the most critical connections as the basis for the reference configuration, routings fully within the private network may sometimes be found as more critical.

11.4 Determination of the transmission parameters

During this step of the planning, all relevant transmission parameters of the different elements in the reference configuration should be determined for:

– the private network;

– the public network(s);

– tie trunks and leased lines.

As a minimum the following parameters must be defined for the various elements of the reference configuration:

– loudness ratings (for telephone sets);

– loss (for switching and transmission elements);

– echo loss (for elements with a 4-wire/2-wire conversion);

– mean one-way delay (along the entire echo path);

– absolute one-way delay (between the two telephones, mainly for international calls);

– number of A/D-D/A conversions (number of qdu in all types of elements, e.g. digital pads);

– equipment impairment factor (in equipment using low bit-rate coding).

Where appropriate, the location and characteristics of existing echo cancellers in the private or public network should be determined, as well as for which routings they are applied.

As a rule, these parameter values should be available from the manufacturers or providers of each considered element. For public networks, information on the parameter values based on the type of connection and access may be obtained by negotiations between public and private network operators. This is also true for the characteristics of VPN or for tie trunks and leased lines. For additional information on these parameters and typical planning values, see Annex A.

11.5 End-to-end calculation with the E-Model

At this step of the planning process, the defined reference configuration(s), together with all relevant parameter values, are here selected as the calculation basis of the expected quality for the considered configuration. As stated in 9.2, 9.3 and 9.5, this step requires care to assure correct inputs to the E-Model. In most cases it is also necessary to perform pre-calculations for certain input parameters to the E-Model, as described in clause 8. It is strongly recommended to transform the reference configuration into one of the working configurations for the E-Model, as described in 9.2.

The selection of the appropriate working configuration depends on the reference configuration to be investigated. In case of fully digital connections, it is clear that the working configuration for fully digital connections shown in Figure 21 should be used. The selection process is somewhat more difficult for reference configurations with one or more 4-wire/2-wire conversions (hybrids) within the connection. For example, a configuration with the private network side terminated in a digital telephone set and with an echo path formed by a hybrid, either within the private network or in the public network, a review of the working configurations in 9.2 suggests that the working configuration for 2-wire/4-wire connections, shown in Figure 20, is likely to be most appropriate.

For the selection of a working configuration, it is critical to make a correct assignment of the send side and the receive side. Basically, the planning principle and the determination of the expected quality is primarily related to the user of the private network. The principle and algorithm of the

E-Model relates the perceived quality to the receive side of the working configuration. Therefore, the telephone of the private network should be assigned to the receive side of the working configuration for E-Model calculations.

However, it may also be necessary to investigate the quality for the far-end termination, particularly with respects to any echo effects. For this investigation, the assignment of send and receive side is reversed. Assuming a far-end termination terminated in a hybrid connecting an analogue telephone set (where the talker may be disturbed by echo), the working configuration of Figure 19 should be applied.

NOTE – When using computer programs, the capability to perform the calculation for both sides without changing the input parameters could be incorporated by the program. Here again it is very important that the planner is completely familiar with all the features and restrictions of the program used for avoiding wrong results.

For reference configurations with a 4-wire/2-wire conversion within the private network it is recommended to use the working configuration for 2-wire/2-wire connections of Figure 18. When transforming the reference configuration into this working configuration, it is, in some applications, necessary to consider portions of the private network (including, e.g. a PBX and/or a transmission section) as a entity (equivalent to a telephone set). In such applications, the relevant parameters should be combined such that they can be considered as the input parameters for the telephone A in Figure 18. A comparable situation is described in 9.5 in conjunction with a cordless telephone.

11.6 Analysis of the results

Once the results of planning calculations are available in terms of the E-Model Rating R, a first concern should be for connections where this value is R ≤ 50. Lower values of E-Model Rating R are not recommended, even in exceptional cases. The realization or the practical use of a reference configuration which, with its corresponding parameter values, results in lower R values is not recommended. For such connections, either other solutions should be found or other equipment should be selected in order to increase the end-to-end speech transmission quality to higher values.

It is very important for the planner to fully understand the planning principle recommended in this Recommendation.

First, that planning is based on an end-to-end consideration in contrast to previous planning practices for private networks where, either through regulation or public network-defined standards, specific limits for the various transmission parameters were specified for the private network section up to the interface to a public network.

Second, that the result is not in terms of "numbers" for the various parameters to be compared with a specific end-to-end-limit for each parameter, but in terms of a quality perception to be expected by the user when communicating via the investigated configuration. As stated in 4.3, quality is a subjective judgement such that assignments cannot be made to a fixed number for the E-Model Rating R, or to the boundaries between different ranges of the whole quality scale. Rather, the quantitative terms should be viewed as a continuum of perceived quality varying from high quality through medium values to a low quality as illustrated in Figure 26.

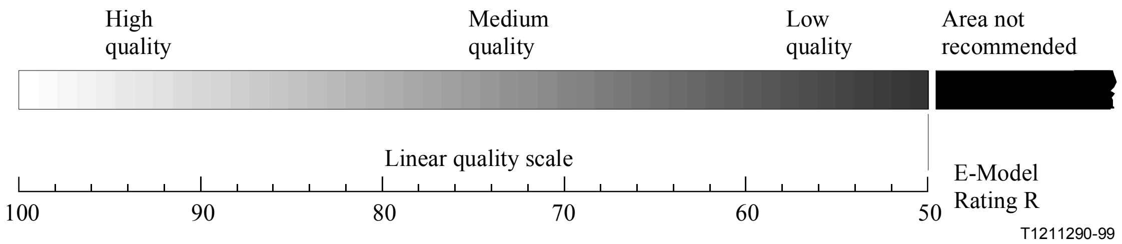


Figure 26/G.108 – Judgement of a connection on a linear quality scale

Although in Figure 26 a rough distinction is made between high, medium and low quality this should not be interpreted in a way that there is a specific value of R considered as a boundary between high and medium or between medium and low. Only the boundary between low quality and the "not recommended" or unacceptable area is fixed to a value of $R = 50$. For further guidance in interpreting the results of planning calculations refer to the definition of categories of speech transmission quality as given in Table 1 and to the text of Recommendation G.109 [4].

For practical planning, it is recommended that normal connections within the network under consideration, or between the network under consideration and the public or other networks, should result in values for the E-Model Rating of $R \geq 70$. For exceptional configurations, values for the E-Model Rating of $70 \geq R \geq 50$ are acceptable but lower values should be avoided. It is worthy of note again that, for an end-to-end analysis as performed here, the overall quality is not only influenced by the network under consideration as the subject of planning, but also by public networks. Therefore, it is not practical, in most applications, to perform the planning for a private network with the goal of "high quality" for all possible connections.

ANNEX A

Transmission parameters for specific elements

This annex provides additional guidance on the different network elements within the private network, for public networks and for the far-end termination. The nature of these parameters and their relevance to the E-Model is common to both the European and the North American regions as well as to other regions. Where network configurations and parameter values of specific network elements differ between the European and the North American regions, these differences are noted for each specific parameter.

According to the planning principles of this Recommendation, the investigation of reference configurations is based on an end-to-end consideration and analysis. This requires transmission data not only for the elements within the network under consideration, but also for other networks and for the far-end termination. As a support to the planner, guidance on specific elements within the private networks and other parts of a connection will be given in this annex. Where possible, mainly in case of commonly used elements with standardized transmission data, the parameter values are given directly and can be used for planning. For all other transmission elements or connection elements, instructions will be given as to what data needs to be collected for the purpose of planning and how to analyse and compare this data.

A.1 Elements in private and/or IP based networks

A.1.1 Wired telephone sets

In general, it is assumed, that all telephone sets used in private networks are designed according to standards applicable to the region; e.g. North American industry standards, European standards or national standards. For planning purposes, nominal values should be used, only; tolerances should not be considered. This is also valid if a volume control in receive or send direction is provided. In this case, only the RLR and SLR values for a default setting of this volume control should be used.

With respect to different impairments and relevant parameters, wired telephone sets can be classified as analogue or digital telephone sets according to their type of interface.

A.1.1.1 Analogue telephone sets

In Europe the transmission characteristics of analogue telephone sets depend mainly on traditional national loss planning. Therefore, standardized values cannot be provided in this Recommendation and should be available from the manufacturer or network operator. The following parameters are necessary for transmission planning and should be determined:

Send Loudness Rating	SLR
Receive Loudness Rating	RLR
Sidetone Masking Rating	STMR
Input impedance	Z_R
Balance impedance	Z_B
Delay (if applicable)	τ
D-factor of the handset	D

In North America transmission characteristics of analogue telephone sets generally conform to industry standards as given in [41]. This standard covers the loudness ratings and impedance values; however, delay (if applicable) and the D-factor are not covered. For sets not designed to this standard, the characteristics should be obtained from the manufacturer.

In either region, the loudness rating values SLR, RLR and STMR should be defined according to Recommendation P.79 [29]. (To avoid wrong results in conjunction with the E-Model, earlier definitions such as Corrected Reference Equivalent (CRE) values or test methods such as OREM-A or OREM-B, should no longer be used.) The input and balance impedances Z_R and $Z_{B,}$ if not specified by industry standards, should follow a modern design providing a capacitive complex impedance for optimized impedance matching between telephone set and connected equipment. [Any potential mismatch at this point may influence the STMR of the telephone set, as well as the weighted Terminal Balance Return Loss at a connected hybrid resulting in a low value for TELR (see also A.1.3).]

The D-factor of the handset should only be considered if a handset design is used which deviates from common geometry. Modern analogue telephone sets may use digital signal processing to provide additional features in some cases. The possible delay caused by this processing should then be determined.

A.1.1.2 Digital telephone sets

Beside the protocol requirements for digital telephone sets, the transmission characteristics can usually be assumed to be in conformance with Recommendation P.310 [27] or with [49] (Europe) or [42] (North America). However, deviations from these values are possible. The following parameter values may be used directly unless the telephone in use does not meet the referenced standards.

		Europe	North America
Send Loudness Rating	SLR	+7 dB	+8 dB
Receive Loudness Rating	RLR	+3 dB	+2 dB
Sidetone Masking Rating	STMR	15 dB	18 dB
Mean one-way delay	τ	1.5 ms	not specified (Note 1)
Terminal Coupling Loss (weighted)	TCLw	40 dB (Note 2)	40 dB (Note 3)
D-factor of the handset	D	3	not specified

NOTE 1 – For North American digital sets, this value is negligible when compared with other components in the connection scenario.

NOTE 2 – Normalized to SLR = 7 dB, RLR = 3 dB (Europe).

NOTE 3 – Normalized to SLR = 8 dB, RLR = 2 dB (North America); the desirable value is 45 dB.

North American loudness values for digital sets are being standardized in [42] and [44] and reflect the long-term objectives of Recommendation P.310 [27]. In general, loudness rating values should be in accordance with Recommendation P.79 [29]. For the weighted Terminal Coupling Loss (TCLw), some telephones may provide values higher than those given above. If stated by the manufacturer such higher values may be used for planning purposes.

A.1.1.3 IP terminals

Beside the functions related to the IP applications an IP terminal (H.323 [26] terminal) provides the conversion between the acoustical interface and the IP based network and vice versa.

It can be observed that practical IP terminal realizations may deviate considerably from the traditional telephone set approach; e.g. headsets/loudspeaker and tie-clamp microphones instead of a handset. Even if it operates similar to a digital telephone set regarding the acoustical interface parameters (see A.1.1.2), the various components of an IP terminal, such as personal computer, soundcard, loudspeaker, microphone and the application software may be provided by different manufacturers, the combination of those components will often be performed by the user. Therefore, the parameters SLR, RLR, TCLw STMR, LSTR and D cannot be automatically set to the default values according Table 6 and are for further study.

The digital part of an IP terminal may comprise the following components:

- encoder/decoder;
- Voice Activity Detection (VAD) devices;
- Forward Error Correction (FEC) devices;
- receive buffer;
- packet assembly/de-packetization.

The actually contained components and their relevant parameters are not standardized as a whole system and thus are for further study.

A.1.2 Wireless telephone sets (including cordless)

Wireless telephones are in common use in private networks to provide the advantage of mobility in conjunction with cellular networks. Due to the coding principles used for the airpath, these telephone sets may contribute additional delay and distortion.

For the European region it is assumed that wireless telephones comply with the appropriate European or national standards. In Table A.1 an extract is given from the DECT and the GSM standards for all parameters relevant for planning according to this Recommendation. North American wireless sets use either licensed or unlicensed technology, Table A.2 presents the licensed and unlicensed parameters which are relevant for speech transmission quality planning.

For wireless telephones deviating from these standards, information on the actual values according to the list of parameters in Tables A.1 or A.2 should be available.

These values apply to the entire configuration consisting of the mobile part and the fixed part, where the fixed part is connected digitally to the adjacent connection or transmission element.

Due to the high values of mean one-way delay wireless telephones are usually already equipped with integrated control of echo, such as echo cancellers or echo suppressors. Since this provision may also influence decisions on echo cancelling devices in other sections of the network, a careful investigation of their interworking is necessary (see 8.2.2 and A.1.10 for more information, and Annex C specifically related to integrated echo suppression with references to DECT).

Table A.1/G.108 – Planning values for wireless telephones – European region

		DECT	GSM full rate	GSM half rate	GSM enhanced full rate
Send Loudness Rating	SLR (dB)	7	7	7	7
Receive Loudness Rating	RLR (dB)	3	3	3	3
Sidetone Masking Rating	STMR (dB)	13	13	13	13
Terminal Coupling Loss weighted	TCL_w (dB)	> 46 (Note 1)	> 46	> 46	> 46
Mean one-way delay	τ (ms)	14	95	100	96
Echo loss of soft suppressor	a_{ESS} (dB)	9	–	–	–
Artificial echo loss (if required)	a_{Echo} (dB)	24 (Note 1)	–	–	–
Number of qdu (Note 2)	qdu	0.5	0.5	0.5	0.5
Equipment impairment factor	Ie	7	20	23	5

NOTE 1 – A TCLw of 34 to 46 dB is optional, artificial echo loss required.

NOTE 2 – qdu only for the A/D-D/A conversion (A-law, Recommendation G.711 [15]), other processes are included in the equipment impairment factor Ie.

Table A.2/G.108 – Planning values for wireless telephones – North American region

		Licensed: TDMA	Unlicensed: PACS WUPE, PCI, PWT
Send Loudness Rating	SLR (dB)	8	8
Receive Loudness Rating	RLR (dB)	2	2
Sidetone Masking Rating	STMR (dB)	15	15
Terminal Coupling Loss weighted	TCL_w (dB)	45	45
Mean one-way delay	τ (ms)	100	7
Echo loss of soft suppressor	a_{ESS} (dB)	–	None
Artificial echo loss (if required)	a_{Echo} (dB)	–	None
Number of qdu (Note)	qdu	0.5	0.5
Equipment impairment factor	Ie	10	7

NOTE – qdu only for the A/D-D/A conversion (μ-law, Recommendation G.711 [15]), other processes are included in the equipment impairment factor Ie.

A.1.3 Switching equipment

It is assumed that switching equipment in a private network, (e.g. PBXs), conform to pertinent regional requirements with respect to their influence on transmission quality. Appropriate requirements can be found in [52] (Europe) or [40] (North America). Note that, in [52], certain significant parameters, notably loss and one-way delay, are no longer specified but are left to manufacturer's declaration.

Basically switching equipment can be categorized according to the type of internal switching:

- analogue 2-wire or analogue 4-wire;
- Pulse Amplitude Modulation (PAM);

- Pulse Code Modulation (PCM) according to Recommendation G.711 [15] (A-law or μ-law);
- new coding principles, e.g. according to Recommendation G.728.

Furthermore, for the purpose of transmission planning, different types of interfaces to other connection elements can be considered:

- interface to connect with public networks;
- interface to connect with other switching equipment of the same private network;
- interface to connect to terminals.

For the physical layer of these interfaces, a further distinction can be made into:

- 2-wire analogue;
- 4-wire analogue;
- digital.

Not all interface/physical layer combinations need to be considered. Table A.3 illustrates the possibilities.

Table A.3/G.108 – Switching equipment connection possibilities

Interface to connection elements	2-wire Analogue	4-wire Analogue	Digital
Public network	Common in North America; phasing out in Europe	Unlikely, except in United Kingdom	Prevalent in Europe; starting in North America
Same private network	Mainly between a main PBX and a tributary	Phasing out	Common
Terminals	Usual for analogue sets	Not used	All digital terminals, including wireless

Switching equipment in a private network usually provides a "through-connection" or "port-to-port connection" between two interfaces, in other words, the switching path is "inserted" into the connection and, therefore, has the potential for contributing transmission impairments. Due to the variety of possible types of through-connections with respect to the physical layer of the interfaces, and considering the various types of internal switching, it is not possible to provide any general norm or guidance about the parameters and their degree of impairment to be considered in transmission planning. However, information should be available from the manufacturer or from general inference with reference to standards on those parameters which may contribute impairments. The most important parameters to be considered for a through-connection are:

- loss or gain between the two interfaces;
- number of qdu;
- value of equipment impairment factor;
- mean one-way delay;
- echo loss;
- input impedance of 2-wire analogue interfaces;
- balance impedance in 2-wire analogue interfaces (hybrids);
- distortion (including noise, crosstalk).

The value of loss between two interfaces is a function of the selection of the relative input- and output-level (loss adjustment) values of analogue interfaces, and the insertion of digital loss or gain pads within the switching path. (Reference [40] specifies a port-to-port loss plan between various types of interfaces but does not put requirements on the allocation of loss between interfaces.)

When new coding laws are used for the internal switching, the value for the equipment impairment factor (Ie) should be selected according to Table 2a. With the exception of inserted digital loss or gain pads, the number of qdu can be set to qdu = 0 for internal analogue switching (including PAM) with both interfaces analogue, or for internal digital switching (A-law or μ-law) with both interfaces digital. A value of qdu = 0.5 can be used for analogue or digital internal switching, where one of the two interfaces is digital and finally a value of qdu = 1 is valid for internal digital switching and both interfaces analogue. In case of digital pads the number of qdu should be increased by 0.7 for each pad in all configurations.

NOTE – It is assumed that, when loss or gain in an analogue-to-digital path is implemented via codec settings, there is no additional qdu increment beyond the 0.5 value. If the PBX provides A-law/μ-law conversion, there may be additional qdu, which must be determined.

The mean one-way delay is negligible for switching equipment using internal analogue switching and analogue interfaces. For all other types the delay depends upon the types of interfaces and the internal switching. For planning purposes an average value of T = 1 ms may be used if PCM, according to A-law or μ-law, is used for the internal switching, unless otherwise specified by the manufacturer (for guidance see Recommendations Q.551 [36], Q.552 [37], Q.553 [38], Q.554 [39], and References [60], [40] and [44]).

Whenever a switching equipment path connects a 4-wire interface (analogue or digital) to a 2-wire interface (or a 4-wire switch path connects two 2-wire interfaces), the echo loss of the terminating hybrid in the 2-wire interface(s) should be considered. The input and balance impedances of any 2-wire interface are not directly subject to planning; however, specifications should be available to determine whether adequate impedance matching at these interfaces is provided. (Reference [40] specifies minimum balance requirements for various types of 2-wire interfaces.)

The echo loss (4-wire return loss in North America) of a 4-wire/2-wire conversion is a very important value for the calculation of the Talker Echo Loudness Rating (TELR), as an input parameter to the E-Model. Together with the mean one-way delay of the echo-path, TELR contributes directly to the important impairment factor of echo performance. The echo loss of a 2-wire termination includes any loss adjustment (relative levels) of the hybrid in the 2-wire interface (in both send and receive directions) and the Terminal Balance Return Loss (TBRL) (Hybrid Balance in North America), of the interface. (In North American scenarios, the hybrid loss adjustment and PBX network loss provision, if any, are usually combined into a single value for port-to-port loss.) This TBRL (hybrid balance) is a function of the degree of matching between the balance impedance of the hybrid and the impedance of the connected terminal, transmission or connection element at the 2-wire side, and should be available as a weighted value (TBRLw or weighted hybrid balance). See Recommendation G.122 [7] for more information about this weighting algorithm.

Balance networks and input impedances of modern switching equipment provide a capacitive complex characteristic to obtain an improved match to the characteristic of unloaded cable sections. (Recommended balance networks are described in [52] and in [40].) If the 4-wire/2-wire conversion is made via an analogue 4-wire interface, the loss adjustment in this interface card should also be included. The same applies for digital pads irrespective of their location.

For European scenarios, assuming a standard loss adjustment of 0 dB (0 dBr) in sending direction and of 7 dB (–7 dBr) in receiving direction as for line cards and interfaces to other equipment and a balance network following the capacitive complex approach, the following average values for the TBRLw and the echo loss can be assumed in planning:

Table A.4/G.108 – Average values for TBRLw and echo loss (Europe)

Termination at the 2-wire side	TBRLw	Echo loss
Analogue telephone set with complex input impedance (negligible line length)	18 dB	25 dB
Analogue telephone set with non-complex input impedance, e.g. 600 ohms	7 dB	14 dB
2-wire-cable section (unloaded)	10 dB	17 dB
Other equipment with complex input impedance (negligible line length)	18 dB	25 dB

In some configurations lower values are possible. If interfaces are using adaptive balancing, the relevant information should be available from the manufacturer.

For North American switching equipment, Reference [40] provides minimum hybrid balance requirements (22 dB in the echo band) for analogue terminal interfaces (called "ONS") when measured against 600 ohms, and for all other 2-wire interfaces measured either against 600 ohms or against a complex capacitive impedance. Average hybrid balance values are not specified. North American systems do not have standard line card loss values; therefore, echo loss values cannot be given; however, the port-to-port loss of the terminating system should be included in the calculation of overall TELR of an end-to-end connection. For planning purposes in North America, assume an average hybrid balance of 12 dB to lines and 10 dB for unloaded cable sections (with no impedance matching).

A.1.4 IP gateways

The specific functions of the gateway will depend on whether the direction of transmission is from the Internet to the PSTN or vice versa. In particular, the functions in the gateway include (but are not limited to):

- Internet → PSTN
  - Packet disassembly (including "IP stack")
  - Speech decoder (including error concealment, comfort noise, silence insertion, etc.)
  - Management or regulation of delay variation
  - Echo cancellation.
- PSTN → Internet
  - Speech encoder (including silence removal, comfort noise, etc.)
  - Packet assembly (including "IP stack").

A.1.5 Leased lines and tie trunks

In private networks, leased lines and ties trunks, as provided by public network operators, are used to interconnect switching elements or to connect terminals to switching equipment. In North America, leased facilities between switching equipment are termed tie trunks; leased facilities used for connecting terminals in a location remote from their serving switching equipment to that equipment are called off-premise lines. In Europe, such facilities are called leased lines for either purpose.

With respect to their interface presentation, leased facilities can be grouped into the following basic categories:

- 2-wire analogue;
- 4-wire analogue;
- digital.

For the purpose of transmission planning, digital leased facilities are independent of their physical layer (64 kbit/s, basic rate access or primary rate access); for transmission planning, only the 64 kbit/s-channel layer is addressed. Leased facilities with analogue interfaces at both ends may also include digital sections and a closed 4-wire loop. Furthermore, in some cases analogue facilities may be available with 2-wire interface at one end and a 4-wire interface at the other end or a facility may have an analogue interface at one end and a digital interface at the other end.

Leased facilities differ not only in their type of interfaces but also in their length. Therefore, standard planning values cannot be stated here. Transmission data should be made available by the provider. The following list may be considered as a guide for the planner when asking for parameter values:

- End-to-end Loss (in both directions) for facilities with analogue interfaces, 2-wire and 4-wire.
- Relative input and output levels for facilities with analogue interfaces.
- Number of qdu for all types with the exception of facilities with fully digital routing and digital interfaces on both ends.
- Equipment impairment factor for lines using DCME, ADPCM, or other new coding laws.
- Mean one-way delay for all types.

North American analogue facilities for tie trunks generally are designed to Via Net Loss (VNL) rules (loss proportional to facility length) except for short-haul tie trunks which operate with a fixed loss. Facilities used for off-premises terminals usually employ twisted pairs (loaded or non-loaded), with Voice Frequency Repeaters (VFR) inserted, where necessary. Approximate analogue facility loss values are:

- VNL trunks: 0.4 dB + 0.0015 x length in miles; maximum 2.9 dB.
- Short-haul trunks with fixed loss: 2 dB.
- Off-premises lines for terminals: 0-4.5 dB.

It is very important that the information about these parameters as given by the providers is based on "actual" values for the specific leased facility, instead of maximum values as derived from a "worst-case" consideration. This enables the planner to avoid an unnecessary insertion of echo cancellers and, in some cases, to allocate a higher amount of impairment values to other private network elements.

A.1.6 Privately owned cable links

Beside leased facilities, privately owned cable links may be used in some private networks; mainly connecting terminals, key systems, and small PBXs to larger switching elements. Only 2-wire unloaded cable sections are considered here, contributing with loss in sections A1 or B1 of the working configurations as defined in Figures 18 through 20. For planning purposes, the loss of such a cable section can be expressed as Circuit Loudness Rating (CLR) in dB, a value which can be added directly to the SLR and RLR of the telephone sets in the pre-calculation for SLR_S and RLR_R (see 8.1).

The CLR can be calculated with the following formula:

$$CLR = 0.015\sqrt{RC} \text{ in dB/km}$$

where:

R = Cable loop resistance in Ohm per km

C = Cable capacitance in nF per km.

A.1.7 Satellite links

When satellite links are used as part of the private network, all relevant parameters, as were listed for leased facilities, should be available. The most important parameter for possible impairments is the mean one-way delay. It should be taken into account, that the total delay consists of the main delay between the two earth stations as well as a possible additional delay between the earth stations and the interface of switching equipment within the private network to which the link is connected at either or both ends. These values should be made available by the satellite operator. For the satellite links via quasi-stationary satellites in a 36 000 km orbit, a value of T = 260 ms between the earth stations can be used for planning purposes. The equivalent values for satellites in lower orbits should be provided by the operator.

A.1.8 Low bit-rate coding

For private networks, the use of low bit-rate coding can result in more economical solutions. In many cases, digital (leased) facilities used for connection elements are equipped with systems especially designed to provide a flexible "bandwidth on demand" feature, utilizing the given number of 64 kbit/s-channels of the connection in a more economical way, mainly for data transmission. For speech channels, low bit-rate coding, in conjunction with methods called Voice Activity Detection (VAD), will reduce costs in a similar way.

For transmission planning, it is absolutely necessary to identify all possible impairments which may be introduced by such systems. Beside others, the main parameters to be considered are distortions and delay. These factors depend on the type of low bit-rate encoding. In general, systems can be classified between the following principles:

- Waveform coder

  Independent of the bit-rate used, all so-called waveform coders reproduce, more or less, the original waveform at the output after decoding. These coders employ mainly the different ADPCM algorithms described in Recommendations G.721, G.726, and G.727.

- Non-waveform coder

  The basic difference in the coding process is an analysis of the speech signal at the coder input, resulting in a transmitted digital signal with reduced bit-rate which has no relation anymore to the original waveform. The decoder then performs a speech synthesis again. This category includes the RPE-LTP-coder (used in the GSM-standard) and the LD-CELP coder according to Recommendation G.728. Furthermore, there might be a variety of non-standardized coding principles, also called "proprietary coder".

- "Squelch"- oriented principles

  Reduction of the transmitted bit-rate is performed by detecting speech pauses (VAD).

The influence of those equipment and coding principles to speech quality can only be defined as a result of subjective tests, expressed in a value for the equipment impairment factor (Ie). For standardized low bit-rate coders, values are given in Table 2a. In all other cases the equivalent values and all further necessary information should be provided by the manufacturer. This applies mainly for the mean one-way delay of such a system. It should be noted that some coding principles may provide different options, with important influence to the system specific delay and that some systems are using a variable bit-rate to adapt to different traffic situations. If low bit-rate systems are used in conjunction with a digital leased line, the system-specific delay is increased by the delay of the leased line.

Some of those systems may also insert a loss to prevent other parts, e.g. integrated echo cancellers, from too high speech levels. Due to the system-specific delay some systems can already be equipped with integrated echo cancellers. The transmission data of those devices should be considered carefully during planning, mainly in conjunction with echo cancelling in other sections of the investigated connection. For further information see A.1.10.

A.1.9 Packetized voice

For the benefit of economical utilization of standard or higher order digital (leased) lines a packetized transmission will be used also in private networks, such as Asynchronous Transfer Mode (ATM) or Frame Relay. The nodes of such systems may also be located in different networks. It is necessary to clearly identify these nodes during planning and to investigate, whether more than one packetized section is included in a connection.

The packetization of the speech signal causes additional delay, depending on the cell orientation and transfer mode. Therefore, information should be available for planning purposes about this delay, expressed as a value for the mean one-way delay in ms.

A.1.10 Echo cancellers

As already described in 7.2 and 8.2, the result of a planning calculation may show that the E-Model Rating R is mainly influenced by the impairment value for echo and delay (Id). In these situations, the decision should be made to insert echo cancelling devices. In modern networks, only echo cancellers are used because they have various advantages in comparison to the formerly used echo suppressors. Therefore, this subclause only deals with the requirements and technical data of echo cancellers.

For the application of echo cancellers, several aspects should be taken into account. At first, an investigation should be performed about the correct and optimized location where cancellation should be inserted in the network. This decision can be influenced by echo control devices which are already available either within the private network, e.g. in some specific terminal elements or connection elements, or in other (public) networks. More information and rules for the insertion of echo cancellers are given in clause 10.

A second aspect which should be taken into account are the technical characteristics of echo cancellers which may vary by a high degree due to their design and application. For all echo cancellers which are not integrated in specific equipment, only devices should be used following, in all parameters Recommendations G.168 [12] or G.165 [11]. Echo cancellers integrated in specific equipment are usually designed for this specific application and, therefore, not necessarily following G.165 [11] in all data.

NOTE – The characteristics of echo cancellers according to Recommendation G.165 [11] are measured using a noise signal. If results gained with other test signals are also available, e.g. with artificial voice according to Recommendation P.50 [28] or with composite source signals, these data will provide a more accurate issue on the effectiveness.

The analysis of the technical data of an echo canceller should be made in conjunction with the designated location, mainly with the characteristics of the echo path, the part of a connection between the echo canceller and the source for signal reflections to be compensated. The routing of the echo path should be bit-transparent and the actual values for the mean one-way delay and echo loss should be determined. For local or long distance calls, where no echo control is usually applied in public networks, the use of echo cancellers can become necessary due to additional delay within the private network. In this situation, the responsibility for sufficient echo control is by the private network operator. However, in most cases both talkers will encounter echo effects, i.e. a pair of echo cancellers should be inserted within the private network if no specific arrangements with the public network operator exist.

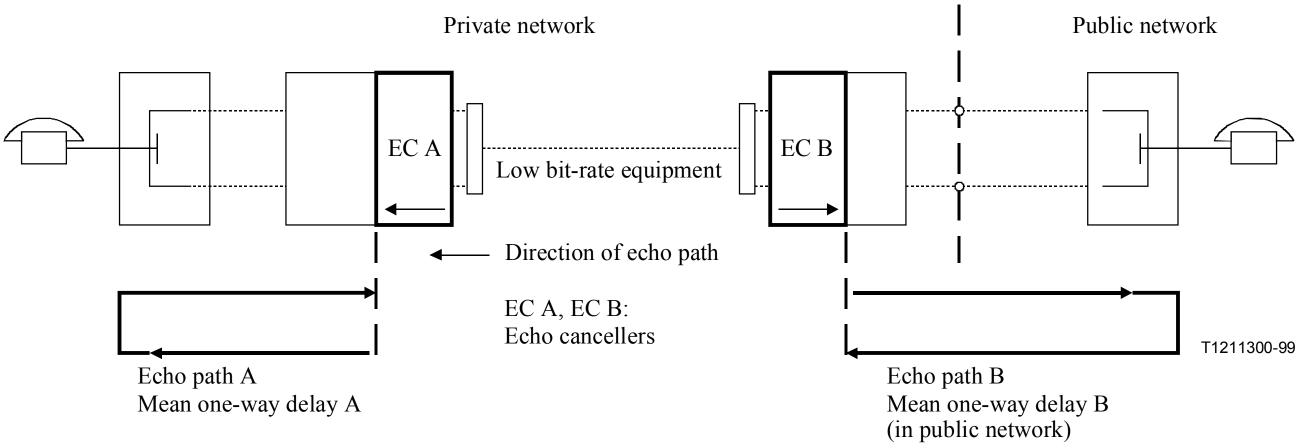


Figure A.1/G.108 – Echo cancellers in a private network and their echo paths

This configuration, with two echo cancellers EC A and EC B in a private network and the definition of both echo paths, is illustrated in Figure A.1. The device EC A with its echo path within the private network is suppressing the echo for the public network talker and vice versa.

For echo cancellers' in accordance with Recommendation G.165 [11], the echo path should provide a minimum echo loss of 6 dB for sufficient operation. This value should mainly be controlled when the echo path is terminated by a hybrid which is also used for a loss compensation (gain) of the connected 2-wire section. For some echo cancellers this required minimum value can be lower or adjustable.

The most important characteristics are the mean one-way delay of the echo path and the range of delay, also called the "tail delay", the echo canceller is able to compensate. To avoid confusion, it should be noted that tail delay of the echo path is usually expressed as mean one-way delay in transmission planning, while the corresponding data of an echo canceller are given as total tail delay in ms. Consequently, for the decision whether a specific echo canceller is suitable for the given configuration, the data of the device should be compared with twice the value of the mean one-way echo path delay. To guarantee sufficient operation, the echo canceller value should be 6-8 ms higher than the value of the echo path. For echo cancellers according to Recommendations G.168 [12] or G.165 [11], the ability to compensate can be assumed in the range of 40 to 60 ms. Contrary echo cancellers integrated in specific devices may provide lower values. If, for instance, the EC A and EC B in Figure A.1 are integrated in a low bit-rate equipment and are designed only for point-to-point connections (echo path only through a switching equipment) with low values to be compensated, the echo path delay B via the public network can be much higher than the corresponding value of EC B.

A further aspect which should be considered is the "linearity" of the echo path, which means in this case a routing consisting of only bit-transparent elements in conjunction with a standard decoding/coding at the terminating hybrid. Most echo cancellers use an adaptation and cancelling algorithm based on such a configuration. Where the echo path consists of equipment using low bit-rate coding, the correct operation of the echo canceller cannot be guaranteed.

The ability of an echo canceller to suppress echo signals is expressed as residual echo level. This is usually not a constant value; but, depends on the speech level at the input and the actual echo loss of the echo path. The value can either be given as residual echo level in dB or also as Echo Return Loss Enhancement (ERLE). Since a total compensation cannot be obtained, the residual echo level is additionally suppressed by a Non-Linear Processor (NLP), also called centre clipper. This suppression is referred to a threshold level, i.e. all residual echo below this threshold will be suppressed. This threshold level is usually expressed in dBm0-values and should be in the range of –35 to –38 dBm0. If the values are in accordance with Recommendations G.168 [12], G.165 [11] or with specific requirements (e.g. DECT standard), impairments due to echo can be neglected for the investigated connection, which means the input value for TELR to the E-Model can remain on its default value of 65 dB.

The principle of an echo canceller and its algorithm is based on an adaptation process which may take a specific time until a sufficient replica of the echo signal is obtained. This time is called the convergence time and should be as short as possible to avoid disturbing effects at the beginning of a voice sequence. Sufficient quality is given for times with less than one second.

Depending on the algorithm used, extremely high speech levels at the input of the echo canceller may cause distortions and reduce the performance of the adaptation process. The control of this level should be included in the transmission planning. In general, the speech level is sufficiently low if the sending loudness rating at the echo canceller input is SLR ≥ 7 dB.

When echo cancellers are inserted in a connection, the bit-transparency is violated. This is important for specific types of data transmission requiring transparent routing and should be taken into account accordingly. Although non-voice services via the private network such as fax and other modem applications, do not require a bit-transparent transmission path, the data handling can be disturbed in some cases. Most modems transmit a signal tone, the so called "disabling tone" with a frequency of 2100 Hz, before starting the transmission, to disable inserted echo cancellers. Depending on the given applications in the private network, the echo cancellers should provide this feature.

A.2 Transmission parameters of public networks

Along with the basic planning principles of this Recommendation, that end-to-end planning is finally executed when the actual values for all relevant transmission parameters in every section of a connection are known, the data of public networks will have a major influence to the resulting quality. If these values are available as real values with an acceptable accuracy, in most cases the unused part of the different parameters within the public network can be used by the private network to provide an economical design. This results in greater flexibility for private network planning in contrast to previous rigid regulations with a fixed apportionment between the networks.

When, during the planning process, these values have to be defined, correct values can only be obtained on the basis of cooperation or negotiations between the private and public network operators. Taking into account that this planning method is mainly applicable for large private networks with a high number of interconnecting channels, a sufficient exchange of information may be assumed.

For the investigation of the actual transmission characteristics of a public network, the main types of connections and the type of access (interconnection) should be considered. In this sense, the public network can be considered as a "transit network", providing circuit switched connections between the access point and any other far-end termination (single telephone, PBX or other private networks), or with interfaces to other public networks. Several possible scenarios for access, and routing to and between different public networks, are shown in Figure A.2. It should be noted that the connections in this figure are only examples; i.e. a variety of other configurations, depending on competition and liberalization, are possible.

NOTE – Scenarios involving IP networks are not considered in the following.

Considering first European scenarios, the several types of interconnection with public networks can be described as follows: The main national public network accessed via point A provides connections to far-end terminations either in the local area as via point D, or as a long distance call to point A. The same network usually also provides connections to international networks via point B, entering another national network in a foreign country via point O for a connection to a far-end termination via P.

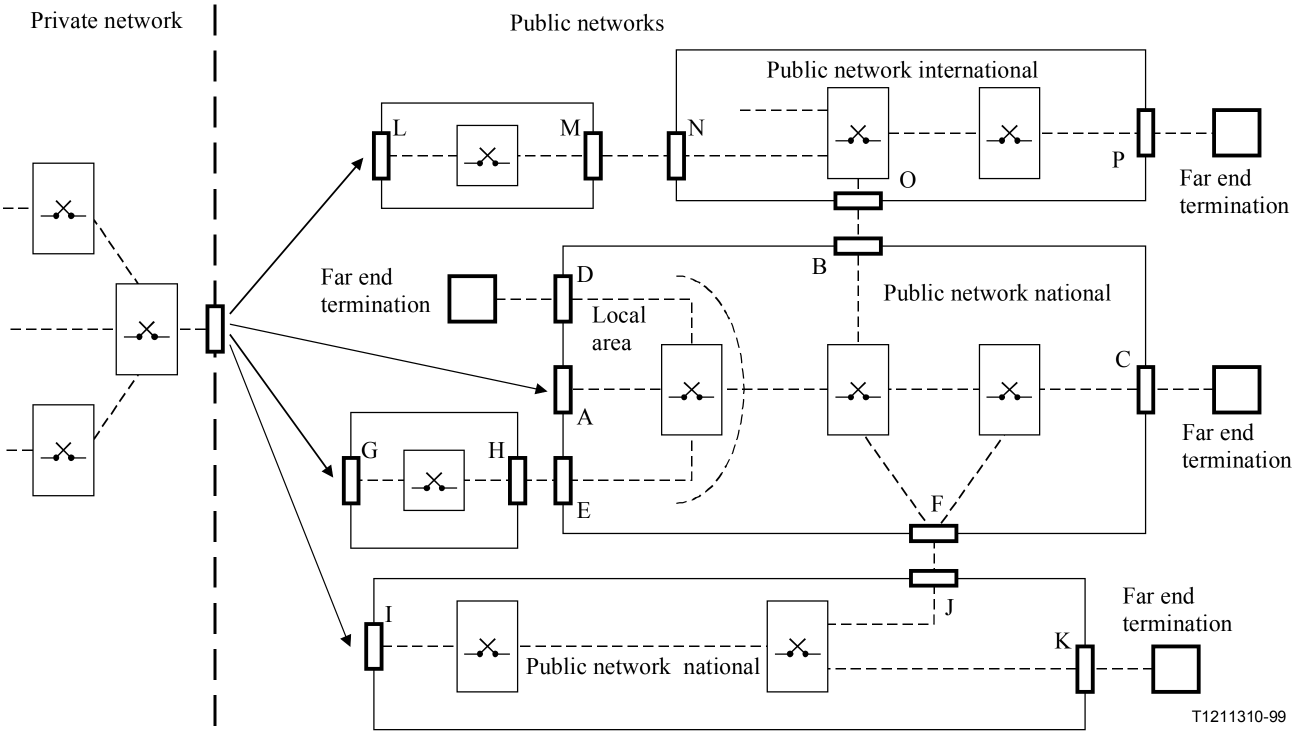


Figure A.2/G.108 – Access and routing in public networks

Beside the main national network, other public networks might be available, via access point I, with their own far-end terminations (terminals) via point K or with interconnections to the main public network via J and F. Furthermore, public networks operating only in the local area may exist, as shown in Figure A.2, via access points G and H, where H is again connecting to the main public network via E. Finally, international operating network providers can offer direct international access with the access point L, connecting the private network in a direct path with the national public network in a foreign country via the interconnections M and N.

For North American interconnection scenarios, Figure A.2 can be described as follows: Access point A is connected to the local serving office of the Local Exchange Carrier (LEC), which provides connections to another local termination (point D), to intra-LATA far terminations (point A in the same LATA), or via Interexchange Carriers (IC) to far-end terminations (point C in a different LATA) or to international carriers (via points B and O). Access could also be directly to an IC (access point I), again to terminations also with direct access to that IC (point K) or to another IC or LEC (via points J and F). Access point G illustrates connections to an alternate local service provider (either a re-seller or an IC) or connection via a wireless service. Private networks may also have direct access to international carriers (point L).

These examples, for possible interconnections of a private network with public networks, as shown in Figure A.2 for the interface points A, G, I and L, are all subject to transmission planning. The results of investigations about the expected quality for the different interconnections may, for example, be used to compare different offers for interconnection service on a cost/quality basis. Furthermore, any single private network may use more than one of these access types in parallel: e.g. national connections via A or I and international connections via access L.

For all interconnection points, A, G, I and L, between the private network and the public networks, only digital interfaces are assumed in Europe; for North American scenarios, digital, as well as analogue, interfaces must be considered. Terminations via C, D, K and P in the different networks may be either analogue or digital, depending on the network and the selected far-end termination for planning purposes (see A.3). Interconnections to international lines as via point B may be assumed to be digital or 4-wire analogue.

In large public networks, diverse routing and call handling for the same end-to-end connection is possible, resulting in a wide range for the transmission parameters (e.g. delay) rather than a single specific value for every call on that connection. It is, therefore, recommended, consistent with basic planning principles, to determine the values more on a statistical basis than on a "worst case" consideration. However, if possible, mean parameter values should be determined for the different categories of calls; i.e. such as local calls (between A and D), national long distance calls (between A and C) or international access (between A and B).

The determination of values for the different categories of calls should include all parameters which are necessary for the planning of the private network. The following list will give guidance to the planner. It is important to note that only the values between the access points (public network acting as a transit) are part of this determination, explicitly excluding the far-end terminations.

A.2.1 Loss

Loss values should be determined for both transmission directions, particularly if, within the public network, a mixed analogue/digital routing in conjunction with a 4-wire/2-wire conversion may exist. Also the insertion of digital pads and their loss values should be included.

A.2.2 Mean one-way delay

For all parts of a routing within the public network consisting of digital or 4-wire analogue sections, an average value for the mean one-way delay should be determined. In case of different values for the two transmission directions, the arithmetic mean should be used for planning. If the call routing also consists of 2-wire sections within the public network(s), any possible delay of such sections should not be included with respect to echo calculations. Specific attention should be given to the possible additional delay resulting from the use of ATM as well as for routing via radio (e.g. for points G and H in Figure A.2) or via satellite links (e.g. for points L and M in Figure A.2).

A.2.3 Echo loss

Values for an average echo loss should mainly be determined if the routing within the public network contains a 4-wire/2-wire conversion (hybrid). For such a terminating hybrid, the average echo loss should be available as weighted echo loss. For more information about the algorithm to obtain a weighted value see Recommendation G.122 [7]. If additional loss within the 4-wire part of the routing is inserted, e.g. in analogue 4-wire systems (FDM) or as digital loss or gain pads, the sum of these values of both transmission directions should be included in the information of the network operator, since they are not only part of the echo loss for the given 4-wire/2-wire configuration but also part of the final calculation of TELR.

A.2.4 Insertion of echo cancellers

Information should be given from the network operator on the insertion of echo cancellers, their location and technical data, and for which category of calls (e.g. international calls only) and routing they are inserted. This information is very important for the planner of the private network when determining whether or not echo cancellers need to be provided within the private network. In Europe echo cancellers are mainly applied on international calls; however, in some cases, mainly between adjoining European countries, echo cancellers are not required. In North America echo cancellers are used on domestic connections where the mean one-way delay is likely to exceed 12.5 ms. In either region echo cancellers may be provided if ATM or low bit-rate coding systems are in use.

From the point of view of the technical data (see also A.1.10) it may be assumed that devices as used in public networks are, in most cases, compliant with Recommendations G.165 [11] or G.168 [12]. For mean one-way delays of more than 5 ms within the private network, the maximum tail delay for which the echo canceller is able to compensate, should be stated by the provider.

A.2.5 Quantization distortion units (qdu)

The number of qdu is decreasing with the increasing digitalization of public networks. For planning purposes this value is important mainly if the private network routing or elements result in added qdu contributions. As already stated in 8.4, only A/D-D/A conversions, according to Recommendation G.711 [15] (A-law or μ-law), and digital loss or gain pads should be included in the planning by assigning a value of qdu to them. For other coding principles such as ADPCM or low bit-rate coders the factor Ie should be used.

A.2.6 Equipment impairment values (Ie)

Information from public network operator(s) should include values for the equipment impairment value (Ie) if low bit-rate coding systems are used in any portion of the connection. This is also important in case of radio sections; e.g. when using ADPCM according to the DECT standard [50]. When standardized coding laws are used, the values as listed in Table 2a can be used for planning.

A.2.7 IP transmission related parameters

Transmission in IP networks is accomplished by assembling multiple bytes into packets. These packets include headers with essential information such as source and destination for the packet, which is added at the transport and network layers. The size of the payload and of the header may vary, depending on the application and on the nature of the protocols being used. Communication between two end-points is via a "data stream" (somewhat analogous to a connection in a connection-oriented network such as the PSTN), which usually consists of multiple packets. Each of the packets from a given source may take different routes to a given destination. As a consequence, packets from a given data stream may arrive at the destination in an order that is different from the order in which they were transmitted (i.e. the packets arrive out of sequence). Protocol features such as sequence numbers or time stamps allow the packets to be reassembled in the proper order at the destination.

Proper sequencing of the packets in the data stream is the responsibility of the higher layers in the protocol stack and is not part of the IP protocol. When an application requires the arriving packets to be used in proper sequence, sufficient delay must be included to allow integration of late packets. For applications such as speech communications, where end-to-end delays as short as possible must be maintained, it may be necessary to declare "very late" packets as lost in order to achieve acceptable delay. The trade-off between long delay (which may result in higher speech transmission quality, but will increase the difficulty of conversation) and dropped packets (which will result in lower speech transmission quality, but will ease the ability to have an interactive conversation) must be given careful consideration when designing VoIP services.

A conclusion from the afore-said may be drawn by the reader of this Recommendation that the optimization processes in this field have not yet been finalized and that hitherto no standardized solutions are available. Hence, detailed explanation on the mechanisms and impacts of an IP transmission should be requested from the provider.

A.3 Transmission parameter of the far-end termination

The planning principles recommended in this Recommendation are based on end-to-end performance considerations. Thus, the results of the transmission planning should produce a reasonable estimate of the speech quality to be achieved. These principles require the inclusion of various far-end terminations. It is not possible to obtain all the relevant information regarding the complete public network connections, especially the details of the far-end terminations. Therefore, it will be necessary to use assumptions based on average values to complete the planning.

To determine the transmission characteristics of the far-end termination, it is necessary to consider two possibilities. The first is calls terminating at a single telephone (residential) and the second is calls terminating on a PBX or private network (business). If the private network planner can identify which category is predominant, this information can be used to perform a more realistic planning (predominance is considered achieved when 95% or more of the calls are in one category).

A.3.1 European far-end termination scenarios

For the definition of a far-end termination including all relevant transmission parameters, three different types of termination are recommended as shown in Figure A.3a. It should be stated, that configuration and values of these three terminations should be considered as nominal, only. National loss planning and national regulations may also result in other more realistic configurations and/or values. Where average values and additional information for a specific country or a specific far-end termination are available and deviate from those shown in Figure A.3a, such values and configurations should be preferred.

The single telephone is assumed to be connected 2-wire analogue to the public network with an average loss of the subscriber line of 4 dB. This subscriber line is considered as part of the whole termination. The configuration may also include a small PBX with analogue switching and negligible loss. The telephone set is assumed to have nominal loudness values of SLR = 4 dB and RLR = −7 dB resulting in SLR = 8 dB and RLR = 3 dB for the entire far-end termination. The LR values may deviate for standard telephone sets in a country from the values given above. The mean one-way delay and value of qdu are both assumed to be 0 in this termination.

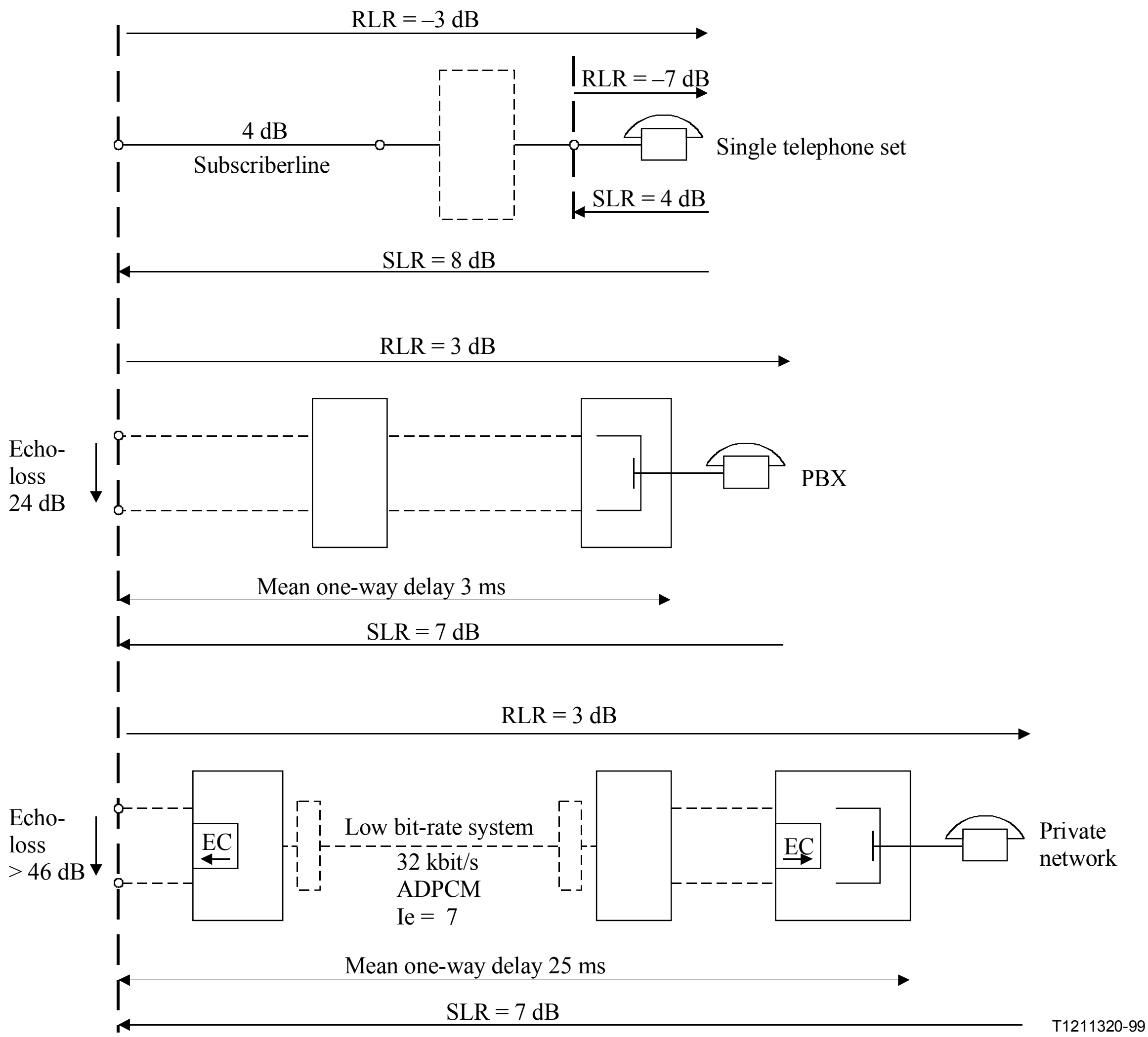


Figure A.3a/G.108 – Configurations and transmission characteristics for the far-end termination – Europe

The second termination illustrates a "standard PBX" or a small network as typical for terminations in the business environment. The interconnection with the public network and between the two PBXs is assumed to be digital, however, the telephone is connected on an analogue port via an extension line with negligible length. The required 4-wire/2-wire conversion within the PBX forms an echo path with a weighted echo loss of 24 dB. The mean one-way delay of the echo path is chosen as 3 ms. The loudness rating values are referred to the digital interface with SLR = 7 dB and RLR = 3 dB, following [54] for digital interfaces to public networks. The value of qdu is assumed to be 0.5, (one-half of an A/D-D/A conversion), with no digital loss pads being used.

The third configuration simulates a "typical" routing within a large private network. Again, the interface to the public network and the interconnections between the PBXs are assumed to be digital. Between the first and second PBX a low bit-rate system is inserted into the call path, using ADPCM with Ie = 7 in conjunction with Voice Activity Detection (VAD), thus contributing additional delay. The mean one-way delay between the public network interface and the terminating hybrid (with a 2-wire analogue extension line) is set to 25 ms, including the delay due to VAD. Because of this high value of delay, echo cancellers are enabled for both directions (echo paths). The echo loss as provided by this termination can, therefore, be assumed to be very high, and the value of TELR should be set to default according to Table 6. The loudness ratings are the same as for the second configuration SLR = 7 dB and RLR = 3 dB (in accordance with [54]); likewise, the value of qdu for this far-end termination can be assumed to be 0.5.

A.3.2 North American far-end termination scenarios

North American far-end terminations can best be classified by the type of access line. Reference [44] defines two types of access lines, analogue and digital. These are illustrated in Figure A.3b for connections to subscriber lines and connections to a PBX. Loudness values are those specified in [44] as characterizing access lines to the North American PSTN. For specific connections, other values may apply.

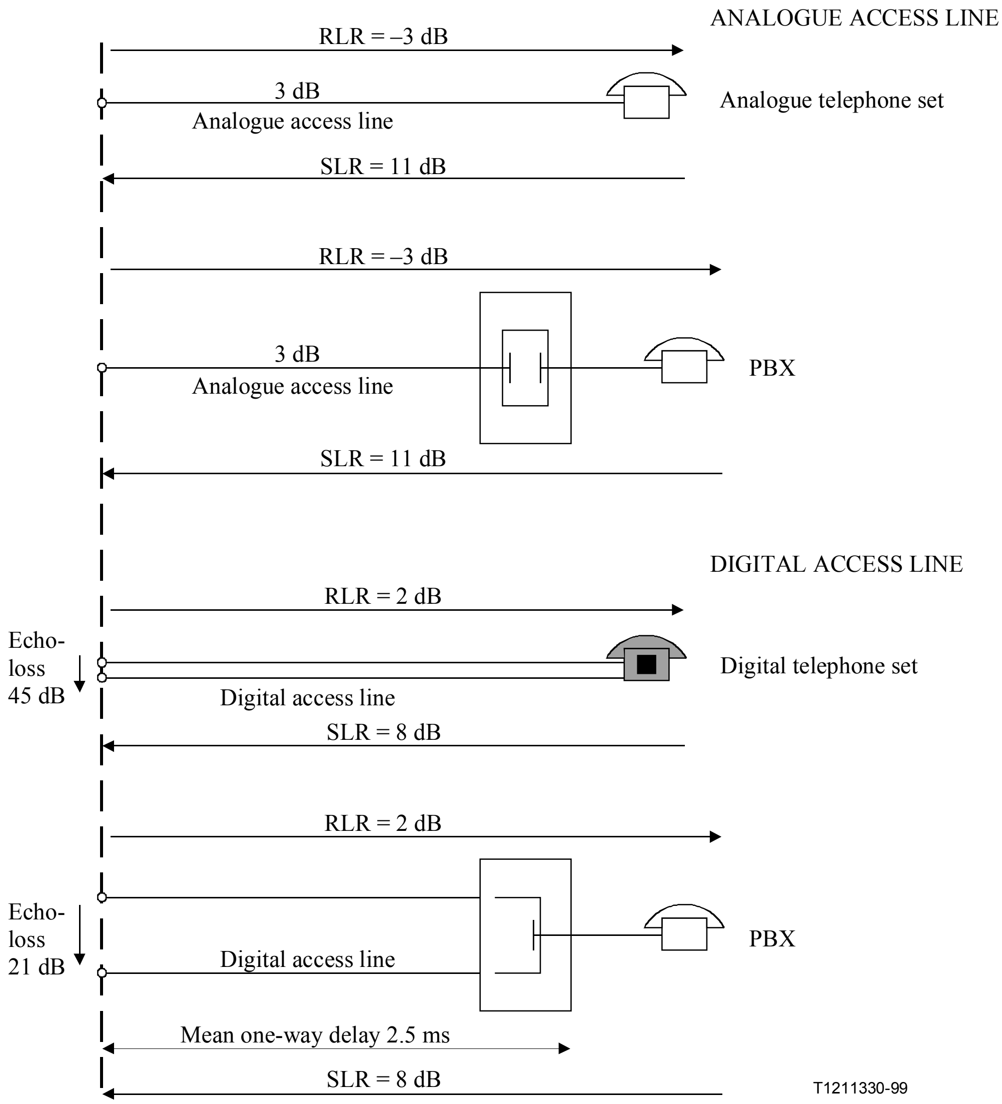


Figure A.3b/G.108 – Configurations and transmission characteristics for the far-end Termination – North America

The analogue telephone is assumed to be connected 2-wire analogue to the public network with an average loss of the subscriber line of 3 dB. The subscriber line is considered as part of the whole termination. The telephone set is assumed to conform to the requirements of [41], resulting in SLR = 11 dB and RLR = −3 dB for the entire far-end termination. The mean one-way delay and number of qdu are both assumed to be zero in this termination.

The North American PBX loss plan, as specified in [40] is designed so that an analogue telephone on a local loop behind the PBX, connected to the PSTN via an analogue access line with approximately 3 dB loss, appears with loudness values at the PSTN interface equivalent to that of an average analogue access line terminated by an analogue telephone. Thus, if the far-end termination is an analogue access line, there is no difference whether that line terminates to a telephone or to a PBX.

A digital access line was, initially, defined in [44] as terminating a digital (ISDN-like) telephone with nominal loudness characteristics conforming to ISDN standards. (The 1992 issue of [44] defines these characteristics in historic North American ISDN terms; an update, scheduled for 1998, will use long-range ITU-T terms and values for compatibility with the re-issue of [42].) As shown in Figure A.3b, a digital access line terminated in a digital telephone will provide a far-end termination with the loudness and echo loss characteristics conforming to [42].

Recently, the digital access line definition has been expanded to include terminations from any equipment which conforms to the nominal ISDN loudness levels. The PBX loss plan in [40] is designed so that a connection from an analogue telephone to a digital access line will produce loudness levels on the digital access line equivalent to those from a digital telephone. The main difference, for loss planning, is that the combination of loss insertion and the 4-wire/2-wire conversion within the PBX results in an echo path with an average echo loss of 21 dB. [44] guidelines for the application of echo control in networks connecting to the public network suggest a limit of 2.5 ms mean one-way echo delay which can be assumed for the typical connection. The value of qdu is assumed to be 0.5 (one-half of an A/D-D/A conversion), provided that the required loss insertion in the PBX does not use digital pads; otherwise, additional qdu increments must be included in the calculation.

ANNEX B

Planning examples

The planning examples contained in this annex are taken from the field of private network planning amended by considerations with respect to IP based scenarios, whereas the method demonstrated in such examples, in general, is applicable to any network Configuration.

Due to a variety of differences between the European and the North American regions, with respect to network configurations and values for the different network elements, the following planning examples are separated to provide a more realistic scenario analysis with respect to each of these areas. Therefore, clause B.1 deals with examples more related to the European situation, while clause B.2 is presented in a similar manner as clause B.1, but with specific North American details.

NOTE 1 – Planning examples for other regions have not been provided because the two different approaches to transmission planning given in this annex can easily be used as guidance for transmission planners in other regions.

NOTE 2 – For the calculation of the following examples the algorithm of the E-Model has been taken from Recommendation G.107 [3] at the time of publication. In case a later revision of Recommendation G.107 [3] does show a refined version of the algorithm, the examples in this annex will still provide valid examples for tutorial purposes. For actual transmission planning tasks it should, in any case, be referred to the latest version of Recommendation G.107 [3].

NOTE 3 – Values for the equipment impairment factor (Ie) in the following examples have been taken from Appendix I/G.113 [5] at the time of publication. In case a later revision of Appendix I/G.113 [5] does show different Ie values for a specific codec, the examples in this annex will still provide valid examples for tutorial purposes. For actual transmission planning tasks it should, in any case, be referred to the latest version of Appendix I/G.113 [5].

B.1 Planning example for the European scenario

The following example will demonstrate how to perform transmission planning according to this Recommendation. This example is not representing an actual private network, but the structure, routing and further requirements of the user are assumed in such a way, that the most critical aspects of transmission planning can be shown. The example results in impairments requiring the use of echo control devices, thus the investigations necessary for the insertion of echo cancellers can be demonstrated. Furthermore, the example follows the planning steps as proposed in clause 11.

B.1.1 Description of the network and user's demand

The basic structure of the network is shown in Figure B.1. The network is serving a medium size company, operating only in a regional area and is consisting of four PBXs with digital switching matrix. PBXs A and B are serving approximately 500 extensions each, while the PBXs at the locations C and D are smaller equipment's for only 150 extensions. Both, digital and analogue telephone sets are used at all PBXs.

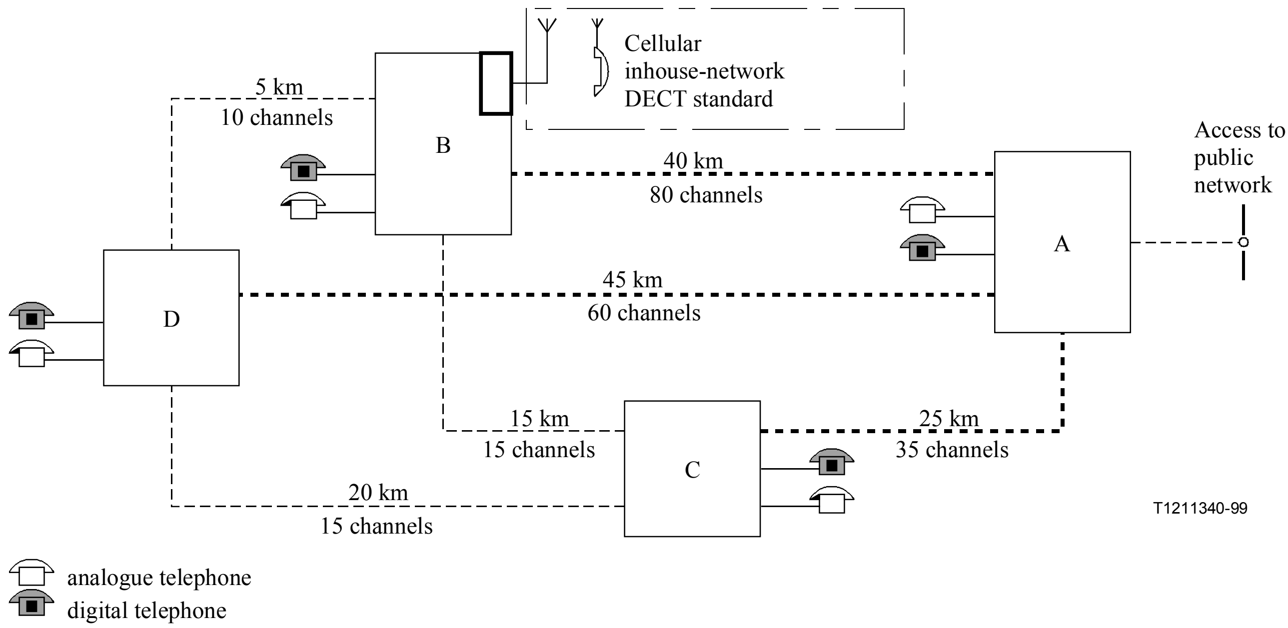


Figure B.1/G.108 – Basic structure of the private network

All switching elements should be interconnected via 2 Mbit/s digital leased lines provided by the public network operator. The required number of channels between the different locations according to the traffic load and the average distances are given in Figure B.1. Between the locations A-B, A-C and A-D a high amount of data traffic should be taken into account. For these transmission elements an economical solution using data multiplexers with integrated low bit-rate coding for speech should be investigated.

As a specific requirement by the user a cellular network, serving cordless telephones according to the DECT standard should be provided within the office building at location B. The interface between the fixed parts of this cellular network and the PBX in B is digital.

Access to and from the public network is served only by the PBX at A. The interconnection is fully digital. According to the business of this company the predominance of communication partners are in the private domain, i.e. single telephone sets, connected to the public network. Although the company is operating in a regional area only, national long distance calls cannot be excluded. International calls, however, need not be considered.

Basically there are no routing restrictions for internal connections or for connections to and from the public network. According to the "mesh" structure of the private network, rerouting in case of busy trunks via three PBXs should also be taken into account during planning. A rerouting via four PBXs (e.g. from A to B via C and D), however, is only exceptional and should not be considered.

B.1.2 Definition of reference configurations

When investigating the private network for a critical connection with respect to speech transmission quality, primarily connections via the public network should be considered. In this example the access to the public network is digital, only national long distance calls should be considered and the single telephone set with its average characteristics can be assumed as the far-end termination. The path through the public network is forming an echo path via the hybrid in the far-end local exchange.

For the most critical connection within this private network the use of low bit-rate equipment and the possibility of rerouting via three PBXs should be taken into account. Probably the cordless telephones at location B will contribute with higher impairments, mainly with echo effects, than digital or analogue wired telephones. Cordless telephones according to the DECT standard, however, are equipped with integrated echo control devices, hence, it becomes difficult to decide in advance, which telephone set will be more critical. Therefore, both reference configurations should be defined and investigated. For the first reference configuration the analogue and digital telephone sets connected to the PBX in C are selected, since a rerouting via the PBX in D to the PBX in A will (roughly estimated) contribute with more propagation time due to the length of the leased lines. The resulting reference configuration 1 is shown in Figure B.2.

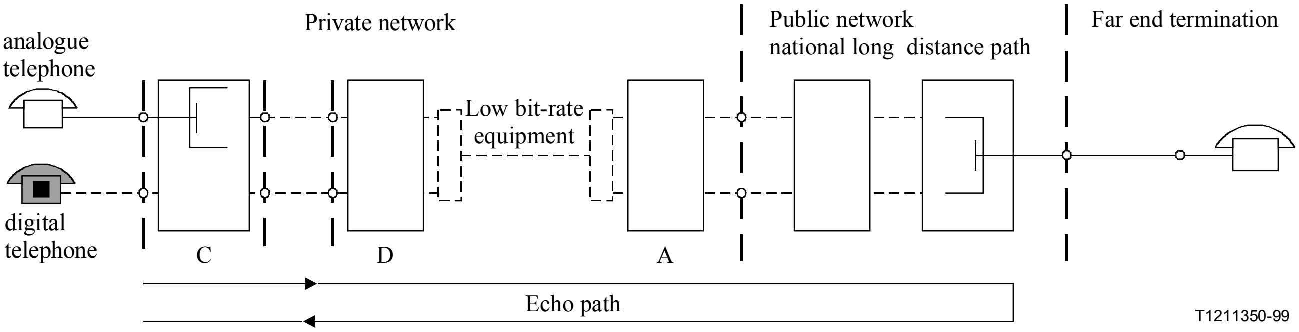


Figure B.2/G.108 – Reference configuration 1

Both types of telephone sets are included for a possible difference in impairments. This enables the planner to issue the quality estimation for all extensions in C using the same reference configuration. The configuration also contains the low bit-rate equipment between the PBXs in A and D.

The second reference configuration is based on the cordless telephones of the cellular network in location B connected via the PBXs in D and A and the public network again to the single telephone at the far-end termination. The principle of this reference configuration 2 is drawn in Figure B.3.

In both reference configurations impairments due to echo should be expected, caused not only by the low bit-rate equipment but also by the additional delay for cordless telephones. The effective echo path as shown in Figures B.2 and B.3 is comparable for both configurations. For the possible use of echo control devices both configurations should be considered together, to obtain an idea of which type of echo cancellers and which location must be selected, to serve the different terminal equipment both with the same device.

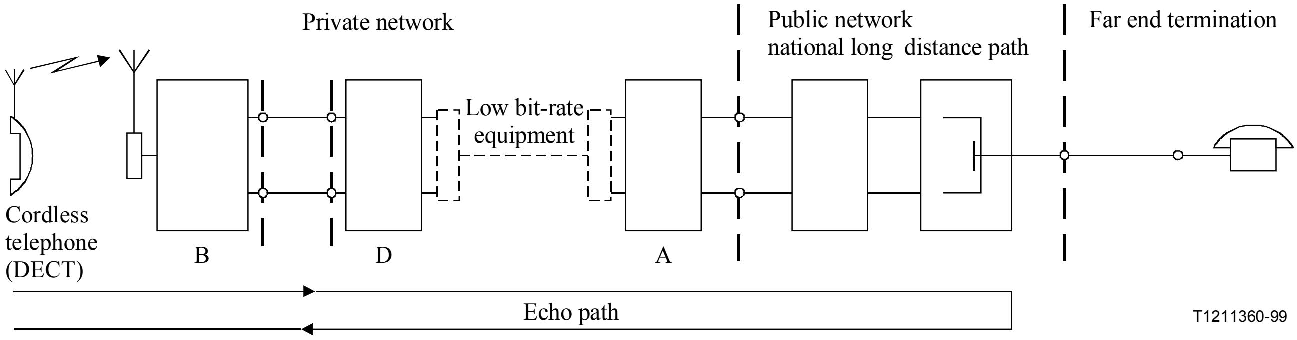


Figure B.3/G.108 – Reference configuration 2

Although, in most cases, connections via the public network are more critical than internal connections, also routings within the private network should be taken into account. For the network in this example a critical configuration may arise when the cordless telephones at location B are connected with analogue or digital telephones in D via the PBX in A. In this case two low bit-rate equipment are connected in tandem (between B-A and A-D), contributing with delay and distortions, possibly requiring echo control devices also for internal calls. Therefore, a third reference configuration as shown in Figure B.4 is included into the planning.

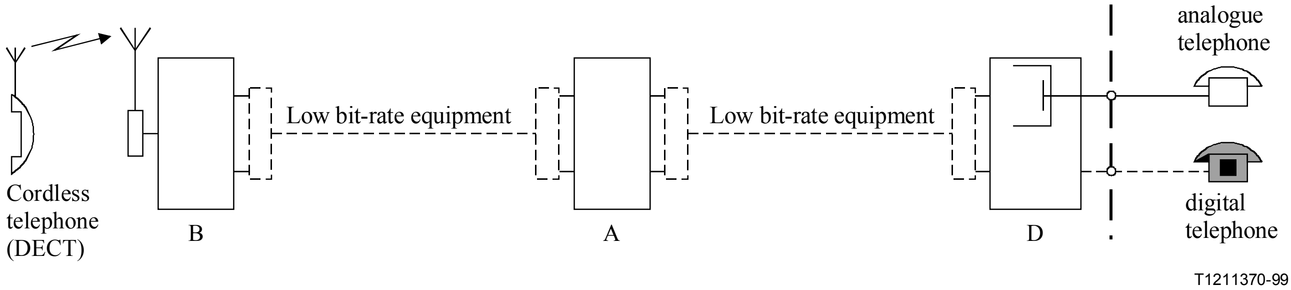


Figure B.4/G.108 – Reference configuration 3

These three reference configurations are the basis now for the determination of the relevant parameter values of the different elements and the following planning calculations. If the results for these most critical configurations, including all necessary echo cancelling devices, are in a sufficient range for the expected quality, all other connections for this private network can be assumed to have less impairments.

B.1.3 Determination of the transmission parameters

According to the three reference configurations indicating all relevant elements the values for the different transmission parameters should be determined in the next step. This information is either known, or should be provided by network operators or by the suppliers of the equipment. For the defined reference configurations values should be determined for the far-end termination, for the public network, for leased lines and for the equipment to be used within the private network.

Far-end termination

For the far-end termination the single telephone set (private domain) was selected. The corresponding values can be taken from the description in A.3.1 with SLR = 8 dB and RLR = −3 dB. These values include an average subscriber line with a loss of 4 dB. However, for planning calculations and for the assignment to a working configuration for the E-Model the entire far-end termination can be considered as a telephone set.

Public network

From the public network operator the following information is provided:

For the digital access to the public network a fully digital routing within the network up to the local exchange, serving the far-end termination, can be assumed for all local, regional and long distance calls. Depending on the location and access point of the private network, an average mean one-way delay between the access point and each terminating local exchange of 10 ms should be assumed. The hybrid at the termination is providing an average value of 24 dB for the weighted echo loss. This value is including a loss of 7 dB in the direction towards the far-end termination. There are no further losses or gains within the public network.

Leased lines

All leased lines provided by the public network operator are digital lines with a 2 Mbit/sec interface at both ends. The routing is bit-transparent in all cases. For the mean one-way delay the following actual values have been determined by the operator:

Line	A-B, A-C, C-D	each	1.0 ms
Line	A-D		1.5 ms
Line	B-C		0.8 ms
Line	B-D		0.5 ms

Terminal elements in the private network

Three types of terminals, analogue, digital and cordless telephones are used throughout the network. They are in conformance with national requirements or European TBRs. For analogue terminals only modern types with electronic circuits and capacitive complex impedances are used. The relevant parameter values for transmission planning are listed below:

Analogue telephones (For the purposes of this example the values below have been chosen):

Send Loudness Rating	SLR	= +4 dB
Receive Loudness Rating	RLR	= −7 dB
Input impedance	Z_R	= 270 Ω + (750 Ω ||150 nF)
Balance impedance	Z_B	= optimized for termination with Z_R
Mean one-way delay	τ	= negligible

Digital telephones (according to [49]):

Send Loudness Rating	SLR	= +7 dB
Receive Loudness Rating	RLR	= +3 dB
Terminal Coupling Loss weighted	TCLw	= > 46 dB
Mean one-way delay	τ	= 1.5 ms

Further parameters, in the E-Model assigned to a telephone set such as STMR, LSTR and the D-factor, can remain at their default values. For the analogue telephones this is guaranteed due to the correct impedance matching between the analogue telephone and the input impedance of the extension interfaces in the PBXs.

Cordless telephones (according to [50]):

The values are referred to the digital interface to the PBX in B, i.e. including portable and fixed part of the cordless telephone:

Send Loudness Rating	SLR	= +7 dB
Receive Loudness Rating	RLR	= +3 dB
Mean one-way delay	τ	= 14 ms
Soft suppressor (fixed part)		
additional echo loss	9 dB	
hangover time	60 ms	
Echo canceller	not applicable for digital interfaces	
Artificial echo loss	available, but disabled	
Terminal Coupling Loss weighted	TCLw = 46 dB and optional 34 dB (see Note)	

NOTE – Both types of portable parts should be assumed, however, signalling to the fixed part according to [50] is provided.

Switching elements

All the PBXs at the different locations are of the same type with a 64 kbit/s PCM switching matrix. Analogue interfaces are only available for analogue extensions. The insertion of digital loss or gain pads is possible and can be controlled depending on the types of interfaces connected. For analogue interfaces the following values are assumed:

Relative input level (A/D)	0 dBr
Relative output level (D/A)	−7 dBr
Input impedance	270 Ω + (750 Ω ‖ 150 nF)
Balance impedance	270 Ω + (750 Ω ‖ 150 nF)
Echo loss	25 dB (for termination with Z_R)

The echo loss of 25 dB includes the receive loss of 7 dB. All further characteristics are according to national or European standards. The relative input and output levels of all digital interfaces are 0 dBr, if no digital loss or gain is used. For all connections including an analogue extension interface a digital loss of 3 dB is inserted by the switching matrix in both transmission directions. For the mean one-way delay a value of 1 ms (average value) can be assumed for planning purposes for each type of connection.

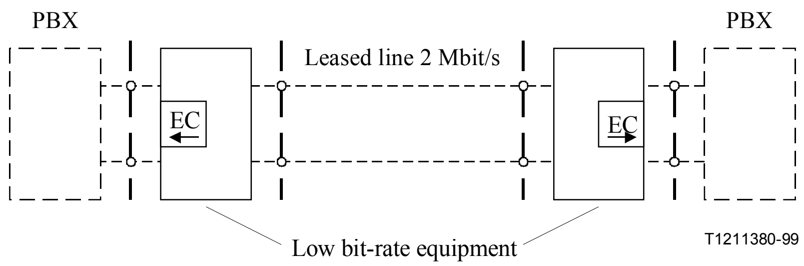
The throughconnection by the switching matrix is only performed after termination (off-hook of the telephone set). All analogue telephone sets are using DTMF signalling, i.e. idle or short circuit with respect to stability should not be considered for this network.

Low bit-rate equipment

All low bit-rate (multiplexing) equipment are assigned to the digital leased lines and installed between the 2 MBit/s interfaces of the leased line and the PBX as shown in Figure B.5. The equipment is providing a speech compression using either 32 kbit/s or 24 kbit/s ADPCM. The corresponding equipment impairment factor values can be taken from Table 2a with:

32 kbit/s ADPCM equipment	Ie = 7
24 kbit/s ADPCM equipment	Ie = 25

If two or more of this equipment is connected in tandem, where a decoding is performed for the through-connection via a PBX, the impairments should be added for each equipment in a connection.



EC Integrated echo canceller

Figure B.5/G.108 – Use of low bit-rate equipment for a leased line

For more capacity on the leased lines a Voice Activity Detection (VAD) is used in addition to the ADPCM coding. This VAD should be considered due to additional delay. Since the total delay of such an equipment is too high for most of the applications, integrated echo cancellers are provided which can be disabled or enabled manually. The characteristics for such an equipment relevant for planning, provided by the supplier are as follows:

Mean one-way delay	20 ms (with VAD for 32 or 24 kbit/s)
Loss between the two digital interfaces	0 dB
Selection of coding	fixed
Recognition of fax signals with code adaptation	yes

The characteristics of integrated echo cancellers are (see also 10.2 and A.1.9):

Minimum required echo loss (hybrid)	6 dB
Echo loss (without NLP)	25 dB
Residual echo level (with NLP)	−65 dBm0
Threshold of the NLP	−36 dBm0
Permitted echo path delay (twice the mean one-way)	15 ms
Linear echo path required	yes
Convergence time	< 1 s

It should be noted, that these echo cancellers are, with the exception of the permitted echo path delay, nearly identical with those according to Recommendations G.165 [11] and G.168 [12].

B.1.4 End-to-end calculation with the E-Model

After determining and collecting all necessary data, the calculations with the E-Model can now be executed. For this example the calculations are made for each of the three reference configurations separately. The configurations are illustrated in Figures B.6 through B.8 again, now also containing all relevant parameter values for a more clear identification of the input parameters to the E-Model. For all three configurations some parameters can already be excluded from the calculation, i.e. they will remain at their default values during calculation.

As already mentioned, the parameters related to the telephone sets such as STMR, LSTR and the D-factors are not relevant due to correct impedance matching for the analogue sets and characteristics according to the relevant standards for the digital and cordless telephones. Default values can also be used for the parameters room noise at the send and receive side, since all telephones are installed in an office environment without extensive noise. Finally, also the number of quantization distortion units can remain at the default value of qdu = 1 if the digital telephone sets at C are investigated, since in all configurations one A/D-D/A conversion is active (independent of the ADPCM coding which is handled separately). If the analogue telephone sets at C are considered, a digital loss pad (qdu = 0.7) is inserted, resulting in an input parameter of qdu = 1.7.

Reference configuration 1

The Reference configuration 1 including all necessary parameter values and indicating the echo paths to be calculated is shown in Figure B.6.

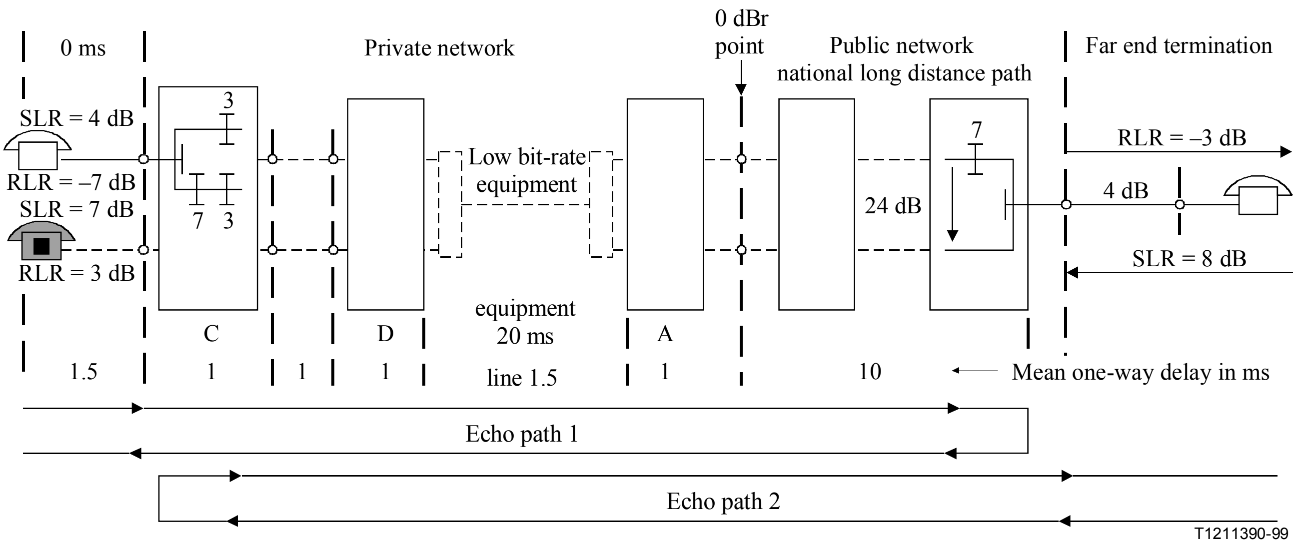


Figure B.6/G.108 – Basis for calculation of reference configuration 1

For the calculation and definition of the actual input values to the E-Model, it is the first step to select one of the working configurations (see 9.2) in conjunction with the definition of the 0 dBr-point. In this actual configuration the 0 dBr-point is defined at the interface between private and public network (access interface of PBX A).

The choice for the working configuration is depending on whether the analogue or the digital telephone set of the private network is considered. For impairments due to echo path 1, effective for both telephone sets, the telephone set of the private network should be assigned to the receive side of the working configuration. For investigation of echo path 2 the far-end termination is representing the receive side in the E-Model.

The parameters relevant for echo path 1 should be determined in a pre-calculation. It is important to note, that the total mean one-way delay of the leased line between A and D is composed of a delay of 20 ms for the low bit-rate equipment and of 1.5 ms for the leased line itself.

The TELR can be calculated as follows:

	analogue telephone	**digital telephone**
SLR of telephone	4 dB	7 dB
hybrid loss in C (transmit path)	0 dB	–
digital pad in C (transmit path)	3 dB	0 dB
echo loss in public network	24 dB	24 dB
digital pad in C (receive path)	3 dB	0 dB
hybrid loss in C (receive path)	7 dB	–
RLR of telephone	–7 dB	3 dB
TELR	34 dB	34 dB

The WEPL can be calculated as follows:

	analogue telephone	digital telephone
TCLw of the digital telephone set	–	46 dB
echo loss (analogue telephone)	25 dB	–
digital pad in C (transmit path)	3 dB	3 dB
digital pad in C (receive path)	3 dB	3 dB
echo loss in public network	24 dB	24 dB
WEPL	55 dB	76 dB

The mean one-way delay T for echo path 1 is obtained by simply adding all single values along the reference configuration in Figure B.6; and because the echo sources are at both ends of the connection, the delay parameters can be calculated as follows: Ta = T and Tr = 2 T. The result is

analogue telephone T = 35.5 ms;

digital telephone T = 37.0 ms.

The values are only slightly different in the delay between analogue and digital telephone as expected. In a further pre-calculation the SLR and RLR referred to the defined 0 dBr-point should be determined as the final input parameters. For the telephones in the private network, assigned to the receive side of the E-Model, only RLR should be available. The value is obtained with RLR = 3 dB for both telephones. For the SLR the path between the far-end termination (send side) and the 0 dBr-point is relevant. The corresponding result is SLR = 8 dB.

The remaining impairment to be determined for this example is the equipment impairment factor (Ie). The reference configuration 1 is containing the low bit-rate equipment between the PBXs in A and D. The used coding is ADPCM with a bit-rate of 32 or 24 kbit/s. According to Table 2a the corresponding values are Ie = 7 for 32 kbit/s and Ie = 25 for 24 kbit/s. These values can be used directly as input parameters. Both values should be subject to planning.

Before performing the calculation run with the E-Model the following input parameters should be set for reference configuration 1. All other input parameters should be set to their default values as given in Table 4.

SLR = 8 dB

RLR = 3 dB

TELR = 34 dB (for both telephones)

WEPL = 55 dB (for analogue telephones)

WEPL = 76 dB (for digital telephones)

T = 35.5 ms (for analogue telephones)

T = 37.0 ms (for digital telephones)

Ta = 35.5 ms (for analogue telephones)

Ta = 37.0 ms (for digital telephones)

Tr = 71 ms (for analogue telephones)

Tr = 74 ms (for digital telephones)

Ie = 7 (for 32 kbit/s ADPCM equipment)

Ie = 25 (for 24 kbit/s ADPCM equipment)

qdu = 1.7 (for analogue telephones)

qdu = 1 (for digital telephones, equal to the default value)

To reduce the number of calculations the parameters T and Ta can be averaged to T = Ta = 36 ms (consequently, Tr is set to 72 ms) and the number of qdu can be left at the default value, since there will be no difference in the results for values of less than qdu = 4 (see also 7.5).

The result for the calculation for reference configuration 1 (with analogue telephone set) with the E-Model is shown below for the E-Model Rating R, the separate impairment values for Is, Id and Ie. It should be noted, that with the digital telephone set the E-Model Rating R increases by $\Delta R = 0.5$ due to the higher value for WEPL.

	R	Is	Id	Ie
with 32 kbit/s ADPCM equipment	**60.1**	0.3	26.1	7
with 24 kbit/s ADPCM equipment	**42.1**	0.3	26.1	25

In a first analysis the results for the E-Model Rating R are too low in both cases. Examining the separate values, mainly for Id the sum for impairments due to echo and Ie for equipment impairments, the major impairment is caused by delay for 32 kbit/s ADPCM equipment. The same impairment is contributing in addition when 24 kbit/s ADPCM equipment is used. The high value for Id can be reduced if echo control devices are used. In this case also the value for R = 42.1 would be increased, but still remaining below 70, a range which should be avoided for standard connections. Standard connection means that all subscribers at the locations B, C and D would perceive a quality only in a medium range for every call to and from the public network. Therefore, as a rough estimate it can be decided to exclude the use of 24 kbit/s ADPCM.

For the benefit of the far-end termination with respect to echo effects also the requirements at the interface between public and private network should be investigated as described in 7.9. The required loudness rating values provided by the private network at the interface with SLR ≥ 7 dB and RLR ≥ 3 dB are met. Also the echo loss of ≥ 24 dB is guaranteed with 31 dB (25 dB of the hybrid in C and 2 × 3 dB digital loss in C). However, these values are restricted for networks with a mean one-way delay of less than 5 ms. For reference configuration 1, however, this value is 25.5 ms and a calculation for echo path 2 (see Figure B.6) becomes necessary.

For the execution of this calculation, only the analogue telephone set in the private network should be considered, since the TELR will in any case be lower (i.e. more critical) than with a digital telephone terminating the echo path with a TCLw of 46 dB. For the working configuration the 2-wire/2-wire connection according to Figure 16 can be used where the interface between public and private network again is used as the 0 dBr-reference point. The far-end termination should now be assigned to the receive side of the E-Model.

For pre-calculations the reference configuration of Figure B.6 can be used. The SLR (now for the analogue set in the private network) up to the reference point, is the sum of the set and the digital pad in C resulting in SLR = 7 dB. The RLR of the far-end termination is including the value of the far-end termination and the receive loss of 7 dB in the local exchange with a sum of RLR = 4 dB.

The parameters T and TELR for the echo path 2 can be added along the echo path as shown in Figure B.6. The mean one-way delay is T = 35.5 ms, identical with echo path 1. The summation for TELR is:

SLR of the far-end termination	8 dB
Hybrid loss in the local office	0 dB
Digital pad in C	3 dB
Echo loss of the hybrid in C (incl. 7 dB receive loss)	25 dB
Digital pad in C	3 dB
Hybrid loss in the local office	7 dB
RLR of the far-end termination	–3 dB
TELR	43 dB

All other input values can either be left at their default values or have the same setting as for echo path 1. For the equipment impairment value, only the 32 kbit/s ADPCM equipment is considered. The input parameters are:

SLR	=	7 dB
RLR	=	4 dB
TELR	=	43 dB
WEPL	=	55 dB
T	=	35.5 ms
Ta	=	35.5 ms
Tr	=	71 ms
Ie	=	7 (for 32 kbit/s ADPCM equipment)

The result of the calculation for the far-end termination is:

R	Is	Id	Ie
76.3	0.3	10.1	7

With this value for the E-Model Rating R of 76.3, the subscriber at the far-end termination can expect higher quality then the subscriber in the private network. However, even this value could be further improved if the corresponding echo control devices are also provided for the far-end termination.

Reference configuration 2

The procedure, the definition of the 0 dBr-point and the calculations for reference configuration 2 are similar to reference configuration 1. The only difference between these two configurations is within the private network, now considering a cordless telephone connected to the PBX in B as shown in the detailed configuration of Figure B.7.

For the quality estimate of the private network, the cordless telephone should be assigned to the receive side of the E-Model in the first investigation. In this case the working configuration for 2-wire/4-wire connections of Figure 19 can be used. Also the impairments for the far-end termination mainly with respect to echo should be considered here in a second investigation. The major impairments in this configuration can be expected due to delay and distortion caused by the use of ADPCM coding. Other parameters as described for reference configuration 1 can be left at their default values or are not relevant in this configuration.

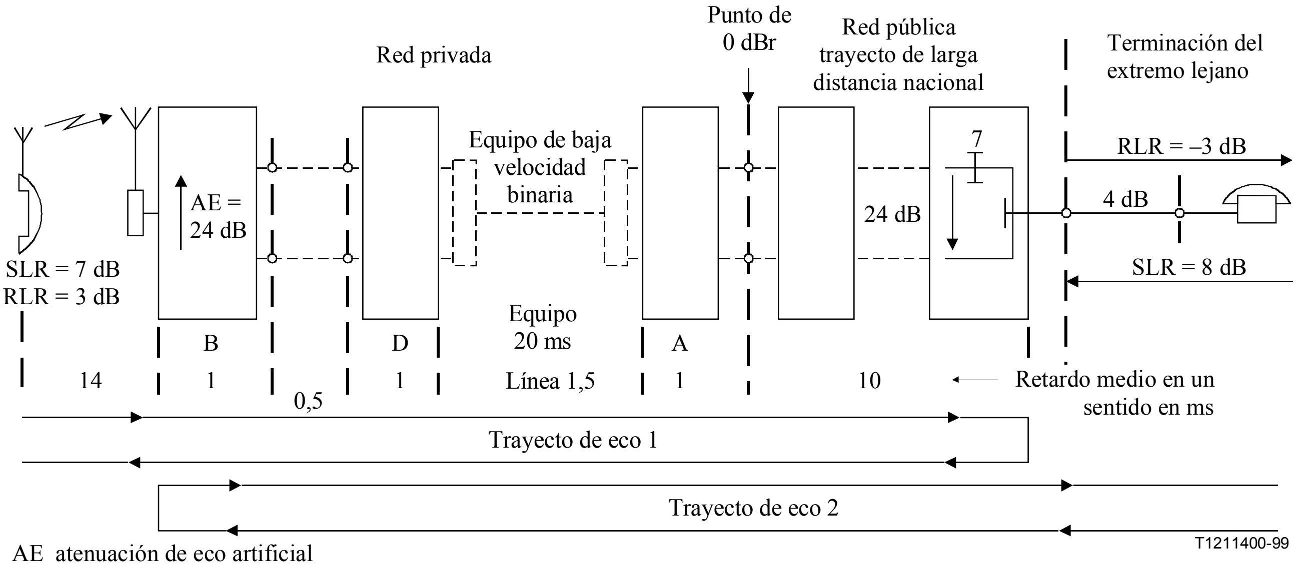


Figure B.7/G.108 – Basis for calculation of reference configuration 2

The pre-calculation for the loudness rating values is equal to configuration 1 for the investigation of the digital telephone and resulting in SLR = 8 dB (far-end termination) and RLR = 3 dB (no digital pads are enabled in PBX B). The artificial echo loss AE in B shown in Figure B.7 (if enabled, it is effective for the subscriber at the far-end termination, only) is disabled for the present.

The parameters for echo path 1 are again determined for the mean one-way delay T by a simple addition with T = 49 ms, which is equal to the absolute delay Ta. For the TELR within echo path 1 a specific characteristic of the cordless telephone, the soft suppressor (SS), should be taken into account. The SS is set to an additional loss of 9 dB and is enabled during talking. This can also be interpreted for the calculation of TELR as an increase of the RLR of the cordless telephone. The RLR to be used for the calculation of TELR then is RLR = 3 dB + 9 dB = 12 dB. Since there is no further loss neither within the private (no digital loss pads in B) nor in the public network, only the echo loss in the local exchange and the loudness ratings of the cordless telephone with SLR = 7 dB and RLR = 12 dB are contributing to the echo path loss with TELR = 43 dB.

For impairments caused by listener echo and for echo path 2 the artificial echo loss AE with 24 dB is assumed to be enabled, since this will result in a lower value for WEPL and TELR (of echo path 2) then the TCLw of the cordless telephone set. In this case, for the round-trip delay (Tr), the delay segments have to be taken into account only, which occur within the 4-wire loop, i.e. the delay of 14 ms caused by the cordless telephone set is excluded; hence $Tr = 2 \times 35$ ms $= 70$ ms. For the calculation of the WEPL the echo loss of the far-end termination with 24 dB and the AE with 24 dB have to be considered only, resulting in WEPL = 48 dB.

For the equipment impairment factor, the 32 kbit/s ADPCM coding is used twice, resulting in a total value of Ie = 14. The input values to the E-Model for calculation of reference configuration 2 are:

SLR = 8 dB

RLR = 3 dB

TELR = 43 dB

WEPL = 48 dB

T = 49 ms

Ta = 49 ms

Tr = 70 ms

Ie = 14 (for two times 32 kbit/s ADPCM)

with all other parameters set to their default values. The result of the calculation is:

R	Is	Id	Ie
64.5	0.3	15.3	14

In a first analysis this result with the E-Model Rating R = 64.5 is considered as too low as well, since all subscribers using cordless telephones would perceive only medium quality for each call to and from the public network, which seems to be unacceptable. Additional echo control devices should also be provided for this type of connections.

For the investigation of the perceived quality for the far-end termination, echo path 2 as shown in Figure B.7 should be considered and also the equipment impairments which are caused by 32 kbit/s ADPCM equipment disturbing the far-end termination too. The far-end termination is assigned now again to the receive side of the E-Model corresponding to the working Configuration for 4-wire/2-wire connections of Figure 17. The input parameters for loudness rating are SLR = 7 dB and RLR = 4 dB.

When preparing the input parameters mean one-way delay T and TELR, two different configurations at the cordless telephone with respect to echo should be taken into account. First, the portable part of the cordless system can provide a TCLw of > 46 dB (this option is not indicated in Figure B.7 with respect to the echo path). Now the TELR is formed by the SLR = 8 dB and RLR = −3 dB of the far-end termination, the 7 dB receive loss in the hybrid of the local office and the TCLw of 46 dB resulting in a value of TELR = 58 dB. This value is nearly identical with the use of an echo canceller independent of the amount of mean one-way delay in the entire echo path.

The portable part of the cordless telephone may however also provide a TCLw of only 34 dB as an option. In this situation usually the artificial echo loss AE in PBX B is enabled, terminating the connection with a fixed echo loss of 24 dB, but excluding the delay of 14 ms from the cordless system (this configuration is shown in Figure B.7 and indicated by echo path 2). The whole mean one-way delay T of echo path 2 is T = 35 ms; while the absolute delay Ta remains at its value of 49 ms. The TELR can be calculated with 36 dB, while the values of the round-trip delay Tr = 70 ms and of WEPL = 48 dB remain unchanged. For the equipment impairment factor the same value of Ie = 14 is effective as for the subscriber of the private network. The input parameters for the E-Model are:

SLR = 7 dB

RLR = 4 dB

TELR = 36 dB

WEPL = 48 dB

T = 35 ms

Ta = 49 ms

Tr = 70 ms

Ie = 14 (for two times 32 kbit/s ADPCM)

with all other parameters set to their default values. The result of the calculation is:

R	Is	Id	Ie
57.7	0.3	21.7	14

This result of E-Model Rating R = 57.7 may be already considered as low quality and should not be accepted for the benefit of the subscriber at the far-end termination, i.e. echo control devices should be inserted, since here the major impairment with Id = 21.7 is caused by echo as well.

Reference configuration 3

For the investigation of an internal connection completely within the private network, the most critical configuration is expected for a routing between a cordless telephone at PBX B to an analogue or digital wired telephone in location D with a re-routing via PBX in A and, therefore, containing two leased line sections and the airpath where all three sections are using uncorrelated ADPCM coding. The reference configuration as the basis for this planning task is shown in Figure B.8.

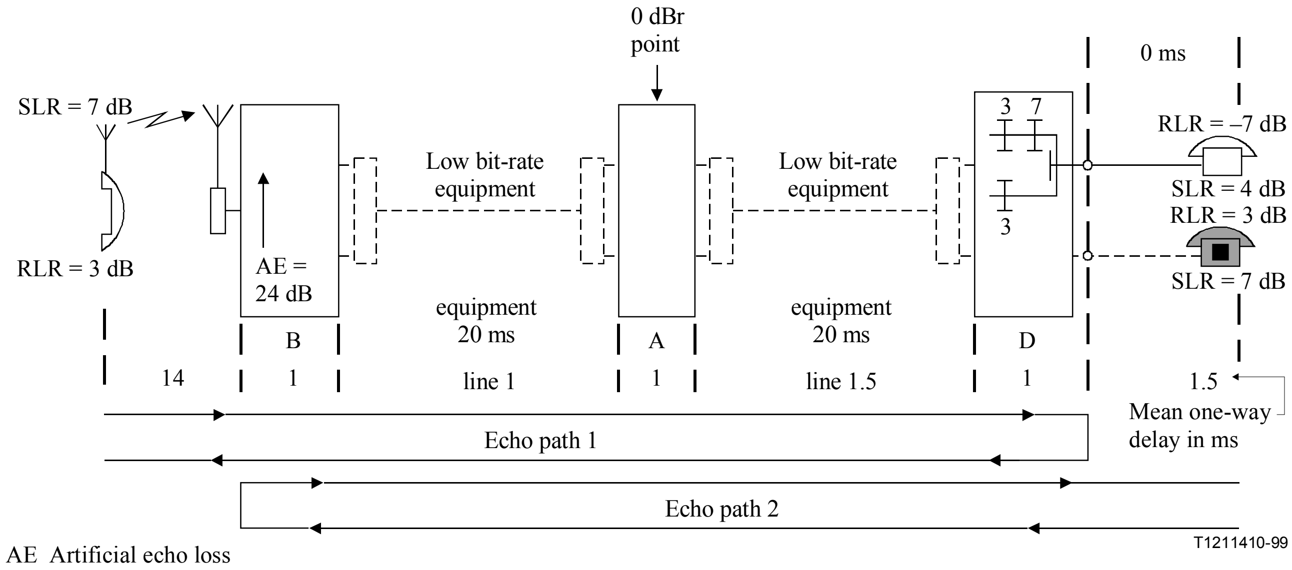


Figure B.8/G.108 – Basis for calculation of reference configuration 3

For this investigation some assumptions can be made. For the termination in B cordless telephones with a TCLw of 34 dB and consequently an enabled artificial echo loss AE is assumed, since this will in any case be more critical than a TCLw of 46 dB with respect to TELR in echo path 2. For the terminals at PBX in D analogue telephones are considered, because the provided echo loss of 25 dB for a hybrid will result in lower values for TELR in echo path 1 than a digital telephone with TCLw = 46 dB.

For this configuration it is obvious that the use of 24 kbit/sec ADPCM shall be excluded. All other assumptions with respect to the different parameters are the same as in the previous configurations. The investigations and necessary calculations should be performed for both sides of the configuration.

For the quality estimation of the cordless system, this telephone should be assigned to the receive side of the E-Model along with the working configuration of Figure 18. The 0 dBr-point in this configuration can be defined as a "virtual" reference point in the centre of PBX A, to obtain a nearly symmetrical configuration. The loudness rating values are SLR = 7 dB (including digital loss of 3 dB in PBX D) and RLR = 3 dB (without the effect of the soft suppressor). The mean one-way delay T along echo path 1 as indicated in Figure B.8 is T = 59.5 ms, which is equal to Ta. For the calculation of TELR an echo loss of 25 dB for the hybrid and the digital pads of 3 dB each in the PBX in D should be included. Furthermore, the soft suppressor in the receive path with a loss of 9 dB is contributing to the TELR. The result is TELR = 50 dB.

For impairments caused by listener echo, and for echo path 2, the artificial echo loss (AE) with 24 dB should be considered again. Then the round-trip delay Tr is calculated, as before, excluding the delay caused by the cordless telephone set and results to Tr = 91 ms. The WEPL includes the echo loss of 25 dB for the hybrid and the digital pads of 3 dB each in the PBX in D together with the artificial echo loss AE of 24 dB, resulting in WEPL = 55 dB. For the calculation of the equipment impairment factor, a 32 kbit/sec ADPCM coding is to be considered in three different sections cumulating to a value of Ie = 21. While all other parameters are at default, the input parameters to be set are as follows:

SLR = 7 dB

RLR = 3 dB

TELR = 50 dB

WEPL = 55 dB

T = 59.5 ms

Ta = 59.5 ms

Tr = 91.0 ms

Ie = 21 (for three times 32 kbit/sec ADPCM)

The result of the calculation is:

R	Is	Id	Ie
65.6	0.5	8.5	21

For the opposite direction the digital telephone at D is chosen for this investigation (1.5 ms additional delay) and is assigned to the receive side and a working configuration according to Figure 19 can be used. The loudness rating values are again SLR = 7 dB and RLR = 3 dB. For the mean one-way delay T along echo path 2 of Figure B.8, the 14 ms of the cordless system are not included due to the artificial echo loss. However, contrary to the previous calculation, additional 1.5 ms should be taken into account for the digital telephone. The total value is then calculated with T = 47 ms and the round-trip delay Tr is exactly twice this value, Tr = 94 ms. The absolute delay includes the delay caused by the cordless telephone, resulting to Ta = 61 ms.

For the corresponding TELR, only the AE and the sum of SLR and RLR of the telephone in D is contributing with a result of TELR = 34 dB. The WEPL includes the TCLw of the digital telephone set with 46 dB and the artificial echo path of the cordless telephone set with 24 dB, resulting in WEPL = 70 dB. For the impairments due to ADPCM, coding the same value of Ie = 21 as before should be applied. The input parameters to be set are as follows with all other parameters at default:

SLR = 7 dB

RLR = 3 dB

TELR = 34 dB

WEPL = 70 dB

T = 47 ms

Ta = 61 ms

Tr = 94 ms

Ie = 21 (for three times 32 kbit/sec ADPCM)

The result of the calculation is:

R	Is	Id	Ie
41.6	0.5	31.8	21

B.1.5 Analysis of the results

For a better overview all results of the different calculations for the three reference configurations are summarized in Table B.1 for the E-Model Rating R and for the main impairment values Id and Ie, and for both sides of the connections, where opposite termination means the subscriber at PBX D for the reference configuration 3.

Table B.1/G.108 – Summary of the calculation results

Reference configuration	**Private network**			**Opposite termination**		
	R	Id	Ie	**R**	Id	Ie
1	**60.1**	26.1	7	**76.3**	10.1	7
2	**64.5**	15.3	14	**57.7**	21.7	14
3	**65.6**	8.5	21	**41.6**	31.8	21

The results for the expected quality for the different configurations is varying in a wide range from a value of E-Model Rating R = 76.3, which would be judged according to Table 1 as medium quality category, to a value of R = 41.6 which is already below the recommended lower limit for R. Most of the values are in a range from 60 to 70 which is in the low quality category, only, and may at the lower values cause complaints. It should be noted that according to Table 1 in this category of low speech transmission quality many users will be dissatisfied. The major impairments for these reference configurations are due to echo and equipment impairment caused by the use of 32 kbit/s ADPCM equipment. Considering the results in Table B.1, the previous decision to avoid the use of 24 kbit/s ADPCM in the low bit-rate equipment is now confirmed again, since all Ie values would be increased and shifted into the range between Ie = 25 and Ie = 50.

The low values for the E-Model Rating R appear not only in conjunction with calls to and from the public network but also for certain calls within the private network. When examining the values, especially the separate values for Id and Ie, it can be seen, that in most cases the impairments due to echo are contributing with a high amount. As a rough estimate the values for Id can be assumed to be reduced nearly to Id = 0 if echo control devices are used. This would also improve the E-Model Rating R by nearly the amount of the current Id value. For confirmation the calculations should be executed again with inserted echo control devices.

For this private network it is, therefore, necessary to use echo cancellers. The selection of the correct echo cancellers and their location should provide echo control not only for internal calls, but also for calls to and from the public network and should be effective also for the opposite termination.

B.1.6 Application of echo cancellers

When once the decision has been made to use echo cancellers within the private network, the investigations should include an analysis of the characteristics of the echo cancellers also with respect to their type of application and location. Further information about the use of echo cancellers and all necessary characteristics, which should be taken into account, is available in detail in clause 10, and in A.1.9.

When investigating the use of echo control devices in this private network the following problems should be cleared and rules should be considered (see also clause 10):

- echo control should be provided for both talkers of a connection;
- information should be available if echo cancellers are provided within the public network and on their application and characteristics;
- echo cancellers should be located as close as possible to the echo source (e.g. hybrids);
- the permitted echo path delay of the canceller should be sufficiently higher than the actual echo path delay;
- the echo path should be linear.

According to the information given by the public network operators and the suppliers of equipment in conjunction with the determination of parameters (see B.1.3), no echo cancellers are inserted within the public network in national long distance calls. Furthermore, the low bit-rate equipment is already equipped with integrated echo cancellers, which can be enabled or disabled. In a first step these cancellers should be taken into account for a possible use.

Integrated echo cancellers are available on both ends of each of the three leased lines from A to B, C and D, located close to the leased line interfaces in the PBXs. In this case the echo path is formed only by the path through the PBX with a delay of 1 ms and the hybrid, connecting the analogue telephone set. Also in case of a re-routing, e.g. extension in D is routed via C to A, the echo path delay for the canceller in C is only increased by the additional paths through PBX C with 1 ms and the leased line between C and D with 1 ms. The actual echo path delay of 2×3 ms = 6 ms is sufficiently below the permitted delay of the integrated cancellers of 15 ms. As already stated during the determination of equipment characteristics these integrated cancellers can be assumed to follow Recommendation G.165 [11] in all other relevant values. Therefore, as a first decision all integrated echo cancellers at the ends of the leased lines in location B, C and D will be enabled. This guarantees proper echo control for all far-end talkers (including the far-end termination via the public network) when being connected with any extension in B, C or D.

The next question is, how to protect the talkers at the extensions in B, C and D themselves. For all internal connections between telephones in B, C and D, routed via A, this is guaranteed by the same equipment. For calls to and from subscribers in A and into the public network, however, there is no echo control up to now. For connections with telephones in A the integrated echo cancellers at the ends of the leased lines in A could be enabled. The leased lines between A and all other locations are carrying not only internal traffic but also traffic between the public network and all extensions in B, C and D. The echo path delay via the public network is higher than the permitted echo path delay of the integrated cancellers which is 15 ms only, therefore, these cancellers cannot be used with respect to the echo via the public network.

To solve this problem, separate echo cancellers with characteristics according to the requirements of the public network shall be used. They can be inserted in PBX A, directly at the interface to the public network. However, to avoid a tandeming, the integrated echo cancellers at A should be disabled. These decisions on the application of echo cancelling devices is illustrated in Figure B.9.

The separate echo cancellers at the public network interface are only necessary for connections to extensions in B, C and D (routed in any case via an additional delay of 20 ms) but not to A. If they remain inserted in the latter case, their impact is negligible. If possible, a more economical solution is to provide a "pool" for these cancellers, i.e. only calls between the leased line interfaces and the public network are routed via an echo canceller in the pool, while extensions in A are bypassing this pool.

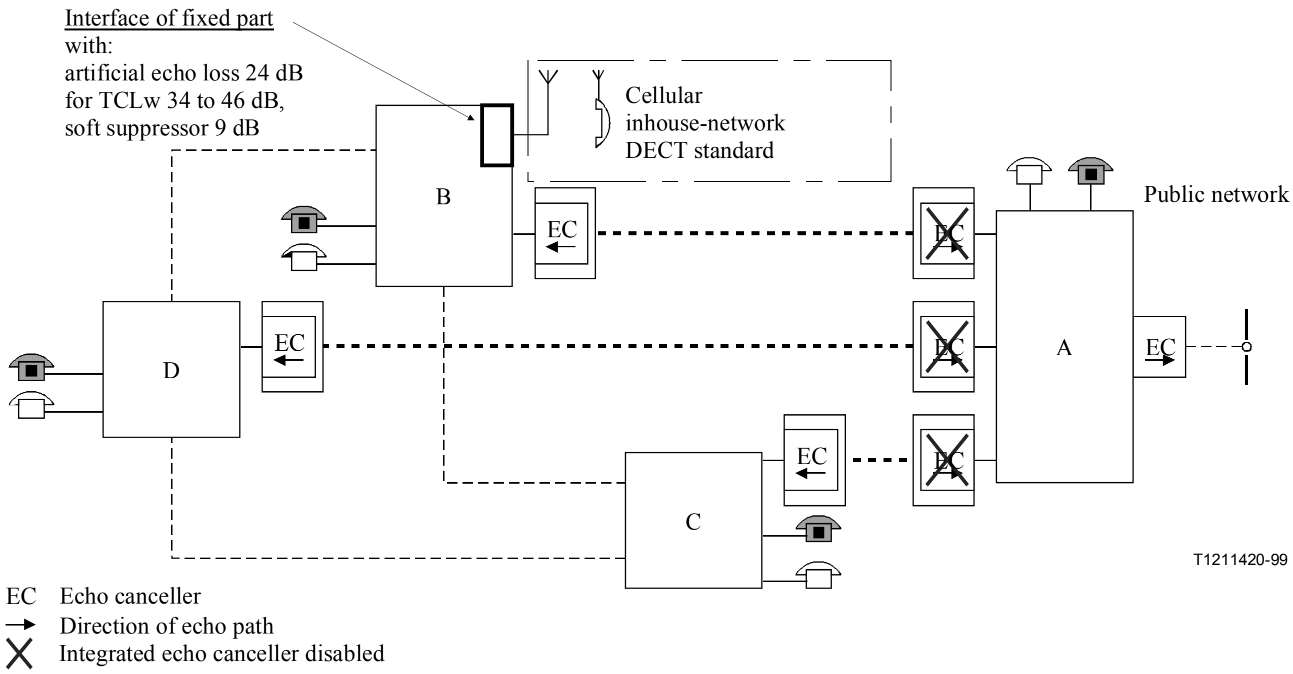


Figure B.9/G.108 – Application of echo control devices in the private network

For the cordless telephones in B, the artificial echo loss should be enabled for connections to those portable parts with TCLw = 34 dB. This is providing an "in range" operation of the integrated echo cancellers at B. The soft suppressor is not necessary in principle, since all critical echo paths with higher delay are now equipped with echo cancellers, but the echo impairments can be reduced in case of direct connections to extensions in A (echo cancellers at A disabled), and to extensions in C and D via the direct routing without echo control devices.

The only problem remaining are connections to analogue extensions in A. For talkers at B, C and D there is no echo control available due to the integrated cancellers at A being disabled. It is not possible to enable these cancellers by different reasons, as due to the need of avoiding tandeming with the separate cancellers, due to the non-linear echo path and due to exceeding the maximum permitted echo tail length for such a canceller. Therefore, it should be investigated whether the pool of echo cancellers can be used also for the internal connections to the analogue extensions in A, or the exclusive use of digital extensions in A should be taken into account.

These investigations on the correct application of echo control devices in this network should be confirmed finally by calculations for all possible types of calls within the network, to and from the public network, mainly including the cordless telephones at B. These calculations are not executed here. When performing these calculations now, the input parameters for the mean one-way delay T and for the talker echo loudness rating TELR of the echo path can be set to their default values of T = 0 and TELR = 65 dB.

With respect to the results summarized in Table B.1 for all reference configurations, it can be assumed with sufficient accuracy, that all values for the E-Model Rating R are improved by the amount of the Id values. The only impairments remaining are the equipment impairments (Ie), which can be up to Ie > 21 for reference configuration 3. However, it should be noted, that this configuration was defined as a critical connection including a re-routing with an additional low bit-rate section. Depending on other features of this private network, these re-routing will not necessarily be the "standard" routing for most of the calls.

B.2 Planning example for the North American scenario

The following examples will demonstrate how to perform transmission planning of private networks in North America according to the principles of this Recommendation.

B.2.1 Definition of reference configurations

As in clause B.1, the analysis of North American scenarios addresses connections between the private and public network, with digital access lines interconnecting the networks, far-end terminations consisting of telephones with average characteristics, and the use of low bit rate equipment in the private network (the most critical connection case). The path through the public network forms an echo path via the hybrid in the far-end local exchange. The resulting reference configuration 1 is shown in Figure B.10. The connections are patterned after the network in Figure B.1.

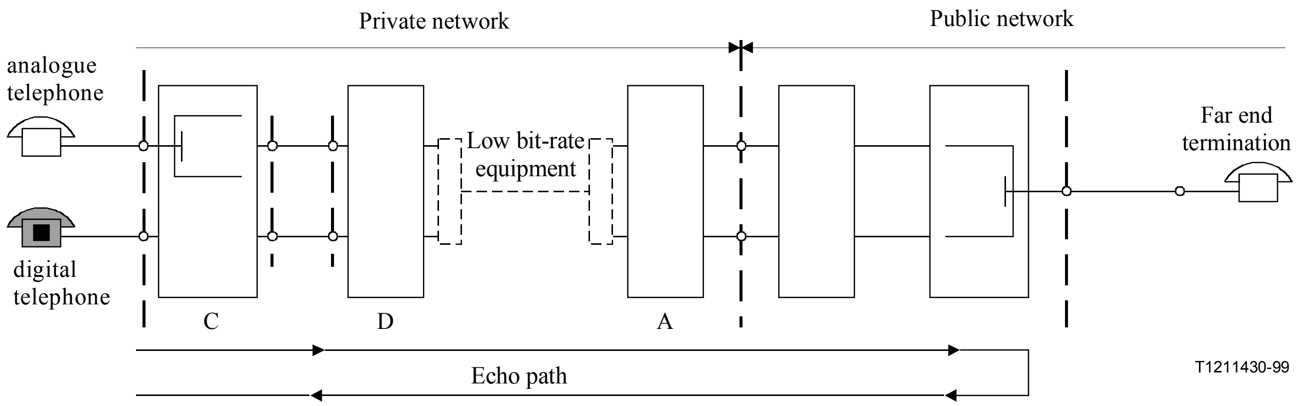


Figure B.10/G.108 – Reference configuration 1

Cordless telephones, mainly due to the added delay, affecting echo performance, can result in higher impairments on an interconnection. Thus, an analysis is made of scenarios with different types of cordless telephones, segregated by the amount of delay contribution. The reference connection is shown in Figure B.11.

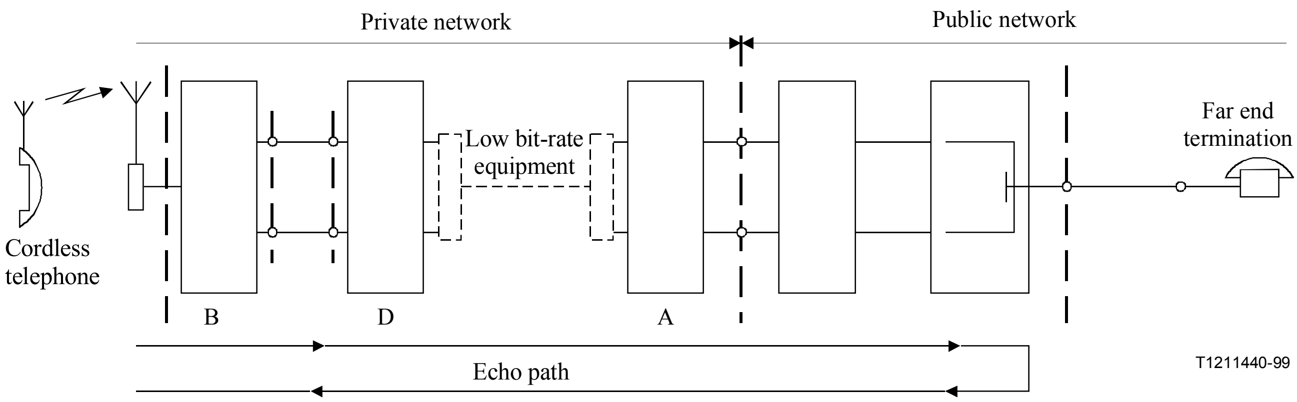


Figure B.11/G.108 – Reference configuration 2

For comparison with the above connection and consistency with the analysis in B.1, a private network scenario, involving a cordless telephone at one end and low bit-rate equipment on a tandem connection, is also examined. The reference connection is shown in Figure B.12.

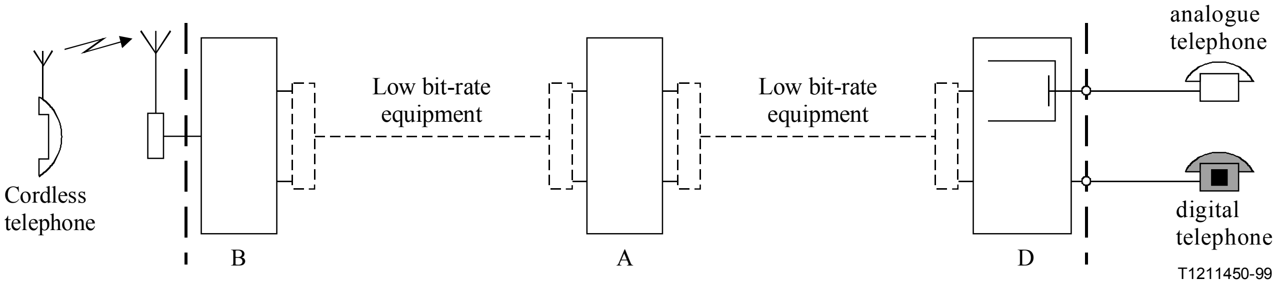


Figure B.12/G.108 – Reference configuration 3

These three reference configurations are now the basis for the determination of the relevant parameter values of the different elements and the following planning calculations. If the results for these most critical configurations, including all necessary echo cancelling devices, are in a sufficient range for the expected quality, all other connections for this private network can be assumed to have less impairments.

B.2.2 Determination of the transmission parameters

To evaluate speech transmission quality of the particular connection scenarios, the parameters to describe the situation are essential. In the examples illustrated in this Recommendation, detailed transmission calculations are illustrated, followed by a summary table.

B.2.2.1 Application of the advantage factor A

The advantage factor A as described in 7.8 has been used for the following calculations in those cases where wireless technology is included in the reference connection. An advantage factor of A = 5 has been assigned to unlicensed technology (PACS, WUPE, PCI, PWT) whereas an advantage factor of A = 10 has been assigned to licensed technology (TDMA). It should be noted, that such an assignment of a specific value of the Advantage Factor A to a specific connection under consideration is not fixed and lies completely within the responsibility of the transmission planner. Further guidance and upper limits of the advantage factor A for typical situations can be found in 7.8.

B.2.2.2 Use of digital pads

For the planning examples describing the North American scenarios it has been assumed that loss or gain adjustments in PBXs or DEOs is performed by digital pads, resulting in additional 0.7 qdu, each.

B.2.3 End-to-end calculation with the E-Model

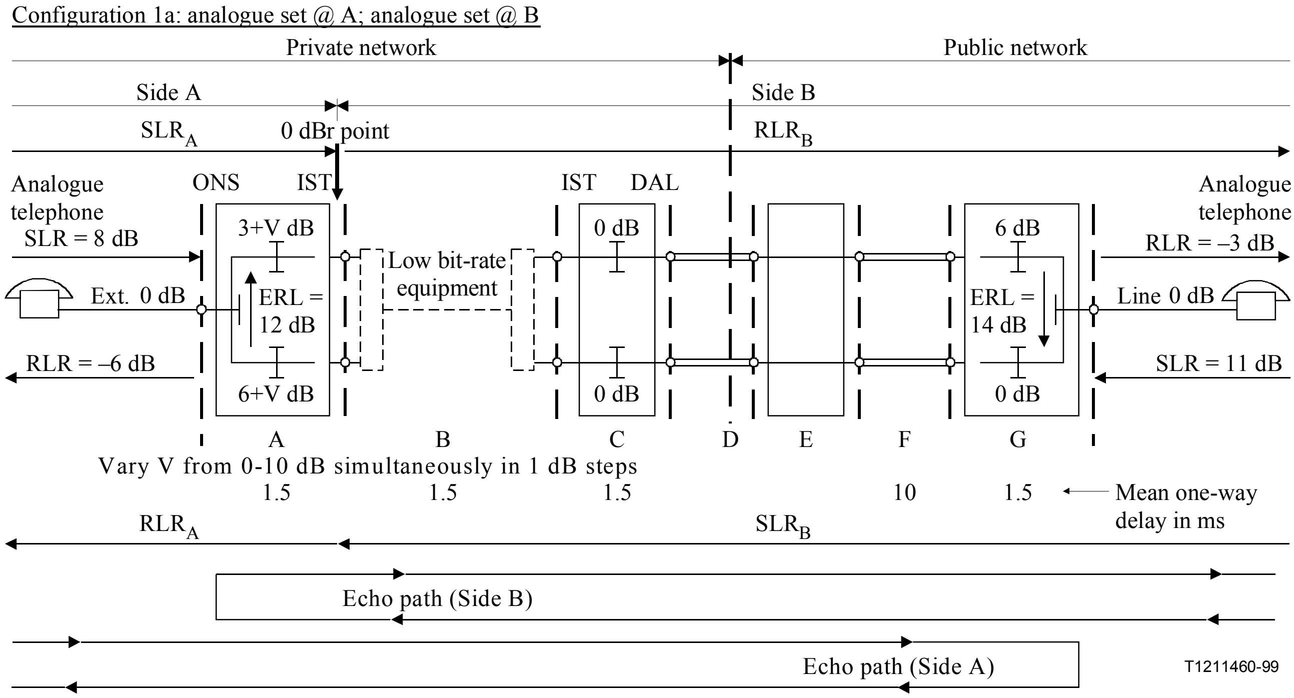
After determining and collecting all necessary data, the calculations with the E-Model can now be executed. In this Recommendation, the calculations are made for each of the three reference configurations separately. The configurations are illustrated in the diagrams shown prior to the pre-calculation section.

Parameters which are standard across North America have been used and where relevant used, as default values and are being labelled as "default" where applicable. Calculations are shown only for those parameters that change.

B.2.4 Reference configuration 1a (analogue set @ A, analogue set @ B)

This configuration represents a private network-to-public network connection with the private network termination consisting of an analogue telephone set. This connection is the North American equivalent to reference configuration 1 in B.1 with an analogue set at the private network termination.

In configuration 1a, the analogue telephone at Side A connects to digital PBX A by an ONS (ON-premises Station) port. PBX A connects to transit PBX C via an Integrated Services Trunk (IST) port. The PBX base loss is 3 dB towards the IST and 6 dB towards the ONS, in accordance with the loss plan of [40] for ONS-IST connection. This loss is varied in 1 dB steps (V) to obtain a plot of speech transmission quality versus PBX loss insertion for this connection. The digital tie trunk facility, B, uses low bit-rate encoding equipment. Transit PBX C connects the digital tie trunk (IST) to a Digital Access Line (DAL). Per the PBX loss plan in [40], the loss for this PBX connection is 0 dB in both directions. The digital access line, D, connects to local PSTN office E for connection to the public switched digital network F for termination at local digital office G. In compliance with the public network digital loss plan, office G inserts 6 dB loss in the receive (towards analogue access line of subscriber) direction.



(In this scenario, A is a terminating PBX, C is a transit PBX)

Figure B.13/G.108 – Basis for calculation for reference configuration 1a

B.2.4.1 Pre-calculations for configuration 1a

B.2.4.1.1 Calculation of SLR, RLR and OLR

Side A

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 8 dB	RLR −6 dB
PBX A	3 + V dB	6 + V dB
Sum at 0 dBr point	**SLR_A = 11 + V dB**	**RLR_A = V dB**

Side B

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 11 dB	RLR −3 dB
DEO G	0 dB	6 dB
DEO E	0 dB	0 dB
PBX C	0 dB	0 dB
Sum at 0 dBr point	**SLR_B = 11 dB**	**RLR_B = 3 dB**

OLR

Listener at Side A	**Listener at Side B**
SLR_B = 11 dB	SLR_A = 11 + V dB
RLR_A = V dB	RLR_B = 3 dB
OLR = 11 + V dB	OLR = 14 + V dB

B.2.4.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

TELR = $SLR_{SET\ A}$ + return loss (B) + sum of node losses + $RLR_{SET\ A}$

= 8 + 14 + (3 + V + 0 + 6 + 0 + 0 + 6 + V) + (−6)

= 31 + 2 V dB

Analysis of TELR performed at Side B:

TELR = $SLR_{SET\ B}$ + return loss (A) + sum of node losses + $RLR_{SET\ B}$

= 11 + 12 + (0 + 0 + 6 + V + 3 + V + 0 + 6) + (−3)

= 35 + 2 V dB

B.2.4.1.3 Weighted Echo Path Loss (WEPL)

WEPL = return loss (A) + return loss (B) + sum of node losses

= 12 + 14 + (3 + V + 0 + 6 + 0 + 0 + 6 + V)

= 41 + 2 V dB

B.2.4.1.4 Delay values T, Ta and Tr

T(ms) = sum of PBXs delay + low bit-rate equipment delay + digital end office (C) delay + network delay + digital end office (G) delay

$= 1.5 + 1.5 + 1.5 + 10 + 1.5$

$= 16$

$Ta = T$

$Tr = 2 \times T$

B.2.4.1.5 Number of quantization distortion units

This parameter represents the impairments due to quantization distortion. It is entered as the number of quantization distortion units. Note that a value of 1 is used for the A/D-D/A conversion. Digital loss pads have additional qdu value of 0.7. In this scenario, there are altogether 1 pair of A/D-D/A conversion and one digital pad if Side A is the listener and two digital pads if Side B is the listener.

Therefore,
$$qdu_A = n \times 1 + m \times 0.7 = 1 \times 1 + 1 \times 0.7 = 1.7$$

$$qdu_B = n \times 1 + m \times 0.7 = 1 \times 1 + 2 \times 0.7 = 2.4$$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

B.2.4.1.6 Equipment impairment factor

In the above example, a low-bit rate codec is used in Section B within the private network. Typically, ADPCM 32 kbit/s is used. This codec does have a mild degradation to voice, and according to Table 2, the value of the equipment impairment factor Ie is 7.

B.2.4.2 Input parameters summary tables

The input parameters for configuration 1a are summarized in Tables B.2a and B.2b; all other parameters remain at their default values according to Table 6.

Table B.2a/G.108 – E-Model input parameters for configuration 1a as perceived at side A

Input Parameters	Values	Unit
Send Loudness Rating (SLR_B)	11	dB
Receive Loudness Rating (RLR_A)	V	dB
Talker Echo Loudness Rating	31 + 2V	dB
Weighted Echo Path Loss	41 + 2V	dB
Equipment impairment factor (Ie)	7	
Quantization distortion unit	1.7	
Round trip delay	32	ms
Mean one-way delay	16	ms
Absolute delay	16	ms

Table B.2b/G.108 – E-Model input parameters for configuration 1a as perceived at side B

Input Parameters	Values	Unit
Send Loudness Rating (SLR_A)	11 + V	dB
Receive Loudness Rating (RLR_B)	3	dB
Talker Echo Loudness Rating	35 + 2V	dB
Weighted Echo Path Loss	41 + 2V	dB
Equipment impairment factor (Ie)	7	
Quantization distortion unit	2.4	
Round trip delay	32	ms
Mean one-way delay	16	ms
Absolute delay	16	ms

B.2.4.3 Results

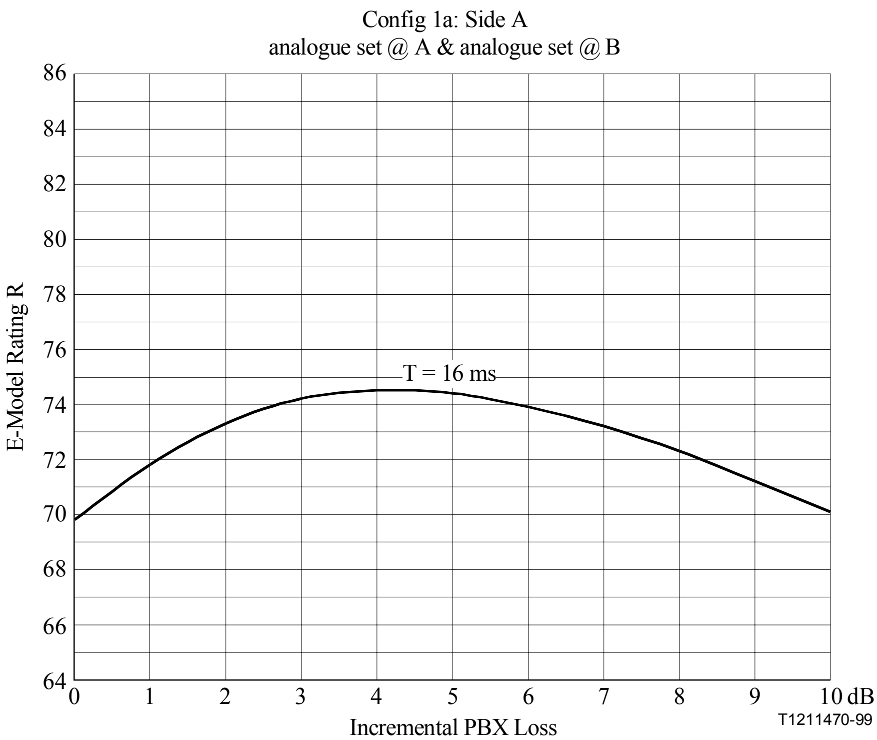


Figure B.13a/G.108 – Analogue set @ A & analogue set @ B as perceived @ side A

As Figure B.13a shows, optimal speech transmission quality for the listener at side A can be achieved when incremental PBX loss is in the range of 3 to 5 dB.

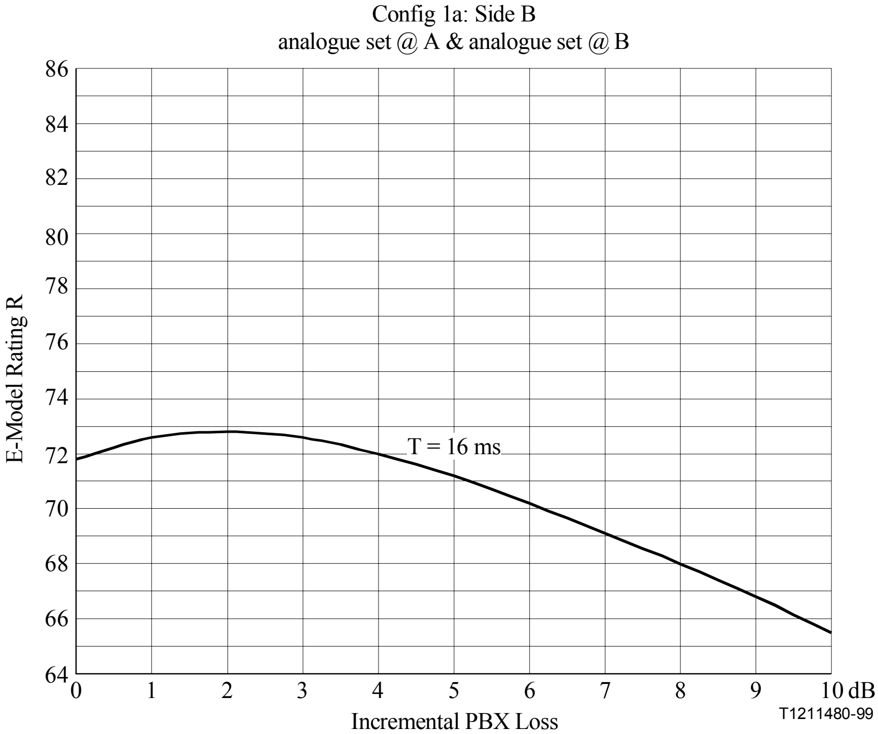


Figure B.13b/G.108 – Analogue set @ A & analogue set @ B as perceived @ side B

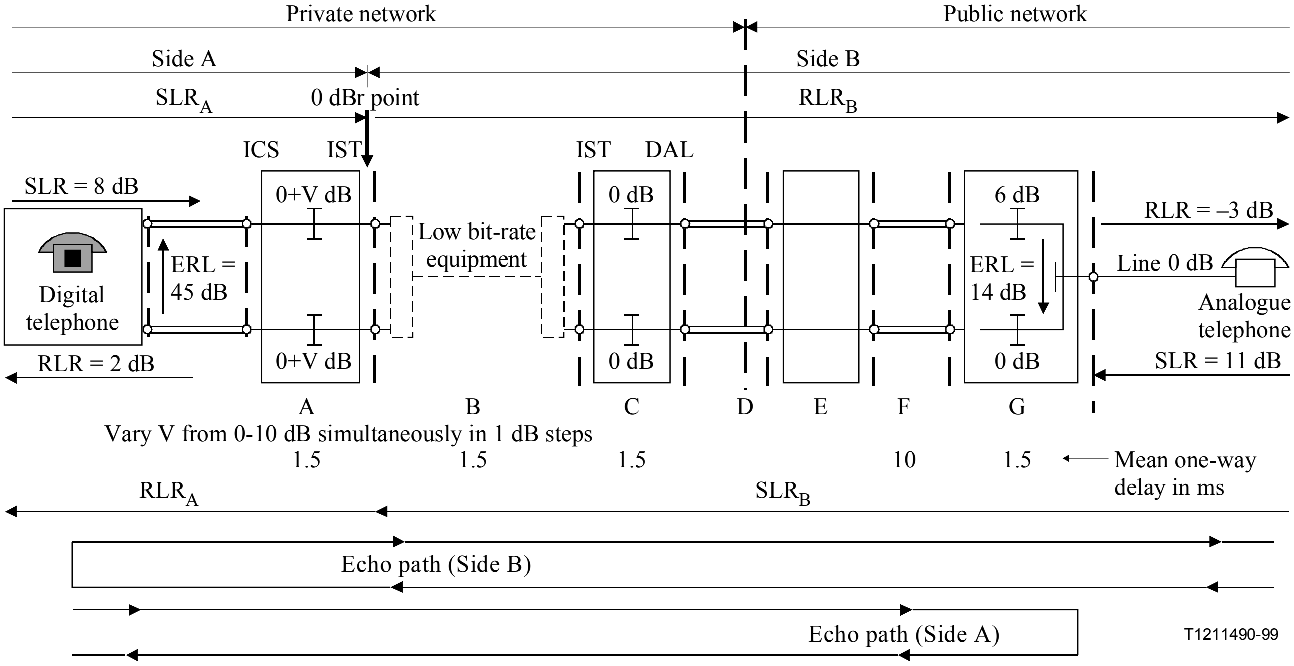
Figure B.13b shows optimal speech transmission quality can be achieved for the listener at side B when PBX incremental loss is in the range of 1 to 3 dB. The absolute optimum performance is somewhat less than for side A; this is due to the somewhat greater (14 dB versus 11 dB) acoustic end-to-end loss when B is the listener. However, the optimum occurs for a lower amount of incremental PBX loss, since the TELR is somewhat greater for side B.

B.2.5 Reference configuration 1b (digital set @ A, analogue set @ B)

This configuration represents a private network-to-public network connection with the private network termination consisting of a digital telephone set. This connection is the North American equivalent to reference configuration 1 in Section D with a digital set at the private network termination.

In configuration 1b, the digital telephone at side A connects to digital PBX A by an ICS (ISDN Compatible Station) port. PBX A connects to transit PBX C via an Integrated Services Trunk (IST) port. The PBX base loss is 0 dB in each direction, in accordance with the loss plan of [40] for ONS-IST connection. This loss is varied in 1 dB steps (V) to obtain a plot of speech transmission quality versus PBX loss insertion for this connection. The digital tie trunk facility, B, uses low bit-rate encoding equipment. Transit PBX C connects the digital tie trunk (IST) to a Digital Access Line (DAL). Per the PBX loss plan in [40], the loss for this PBX connection is 0 dB in both directions. The digital access line, D, connects to local PSTN office E for connection to the public switched digital network F for termination at local digital office G. In compliance with the public network digital loss plan, office G inserts 6 dB loss in the receive (towards analogue access line of subscriber) direction.

Configuration 1b: digital set @ A; analogue set @ B



(In this scenario, A is a terminating PBX, C is a transit PBX)

Figure B.14/G.108 – Basis for calculation of reference configuration 1b

B.2.5.1 Pre-calculations for configuration 1b

B.2.5.1.1 Calculation of SLR, RLR and OLR

Side A

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 8 dB	RLR 2 dB
PBX A	0 + V dB	0 + V dB
Sum at 0 dBr point	**SLR$_A$ = 8 + V dB**	**RLR$_A$ = 2 + V dB**

Side B

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 11 dB	RLR –3dB
DEO G	0 dB	6 dB
DEO E	0 dB	0 dB
PBX C	0 dB	0 dB
Sum at 0 dBr point	**SLR$_B$ = 11 dB**	**RLR$_B$ = 3 dB**

OLR

Listener at Side A	**Listener at Side B**
SLR$_B$ = 11 dB	SLR$_A$ = 8 + V dB
RLR$_A$ = 2 + V dB	RLR$_B$ = 3 dB
OLR = 13 + V dB	OLR = 11 + V dB

B.2.5.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

TELR $= SLR_{SET\,A}$ + return loss (B) + sum of node losses + $RLR_{SET\,A}$

$= 8 + 14 + (0 + V + 0 + 6 + 0 + 0 + 0 + V) + 2$

$= 30 + 2\ V$ dB

Analysis of TELR performed at Side B:

TELR $= SLR_{SET\,B}$ + return loss (A) + sum of node losses + $RLR_{SET\,B}$

$= 11 + 45 + (0 + 0 + 0 + V + 0 + V + 0 + 6) + (-3)$

$= 59 + 2\ V$ dB

B.2.5.1.3 Weighted Echo Path Loss (WEPL)

WEPL = return loss (A) + return loss (B) + sum of node losses

$= 45 + 14 + (0 + V + 0 + 6 + 0 + 0 + 0 + V)$

$= 65 + 2\ V$ dB

B.2.5.1.4 Delay values T, Ta and Tr

T(ms) = sum of PBXs delay + low bit-rate equipment delay + digital end office(C) delay + network delay + digital end office(G) delay

$= 1.5 + 1.5 + 1.5 + 10 + 1.5$

$= 16$

$Ta = T$

$Tr = 2 \times T$

B.2.5.1.5 Number of quantization distortion units

This parameter represents the impairments due to quantization distortion. It is entered as the number of quantization distortion units. Note that a value of 1 is used for the A/D-D/A conversion. Digital loss pads have additional qdu value of 0.7. In this scenario, there are altogether one pair of A/D-D/A conversion and one digital loss pad if side A is the listener and two digital pads if side B is the listener.

Therefore, $qdu_A = n \times 1 + m \times 0.7$

$= 1 \times 1 + 1 \times 0.7$

$= 1.7$

$qdu_B = n \times 1 + m \times 0.7$

$= 1 \times 1 + 2 \times 0.7$

$= 2.4$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

B.2.5.1.6 Equipment impairment factor

In the above example, a low-bit rate codec is used in Section B within the private network. Typically, ADPCM 32 kbit/s is used. This codec does have a mild degradation to voice, and according to Table 2, the value of the equipment impairment factor (Ie) is 7.

B.2.5.2 Input parameters summary tables

The input parameters for configuration 1b are summarized in Tables B.3a and B.3b; all other parameter remain at their default values according to Table 6.

Table B.3a/G.108 – E-Model input parameters for configuration 1b as perceived at side A

Input Parameters	Values	Unit
Send Loudness Rating (SLR_B)	11	dB
Receive Loudness Rating (RLR_A)	2 + V	dB
Talker Echo Loudness Rating	30 + 2V	dB
Weighted Echo Path Loss	65 + 2V	dB
Equipment impairment factor (Ie)	7	
Quantization distortion unit	1.7	
Round trip delay	32	ms
Mean one-way delay	16	ms
Absolute delay	16	ms

Table B.3b/G.108 – E-Model input parameters for configuration 1b as perceived at side B

Input parameters	Values	Unit
Send Loudness Rating (SLR_A)	8 + V	dB
Receive Loudness Rating (RLR_B)	3	dB
Talker Echo Loudness Rating	59 + 2V	dB
Weighted Echo Path Loss	65 + 2V	dB
Equipment impairment factor (Ie)	7	
Quantization distortion unit	2.4	
Round trip delay	32	ms
Mean one-way delay	16	ms
Absolute delay	16	ms

B.2.5.3 Results

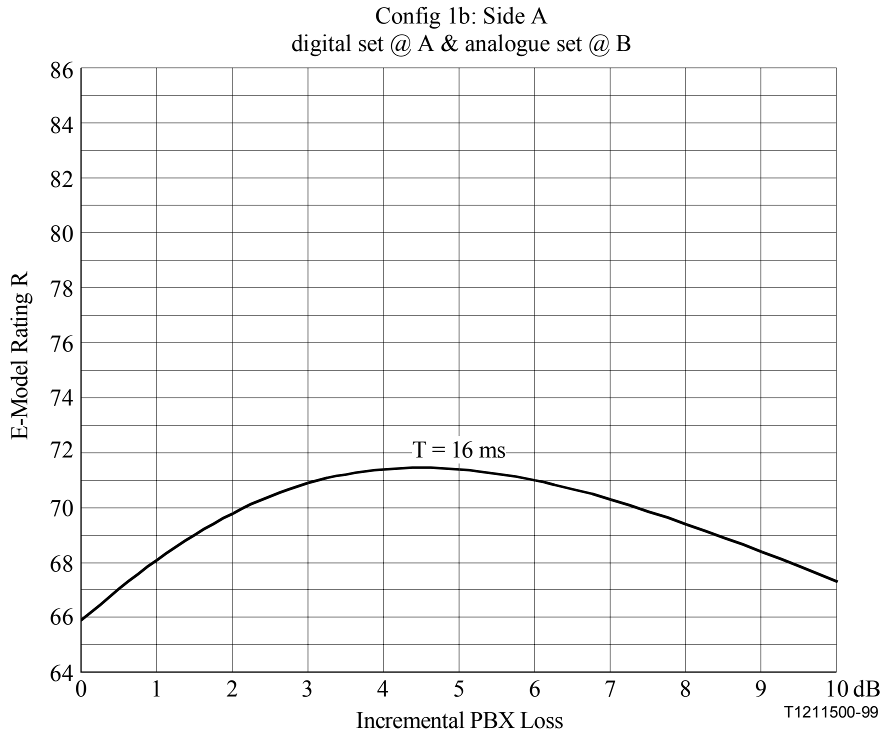


Figure B.14a/G.108 – Digital set @ A & analogue set @ B as perceived @ side A

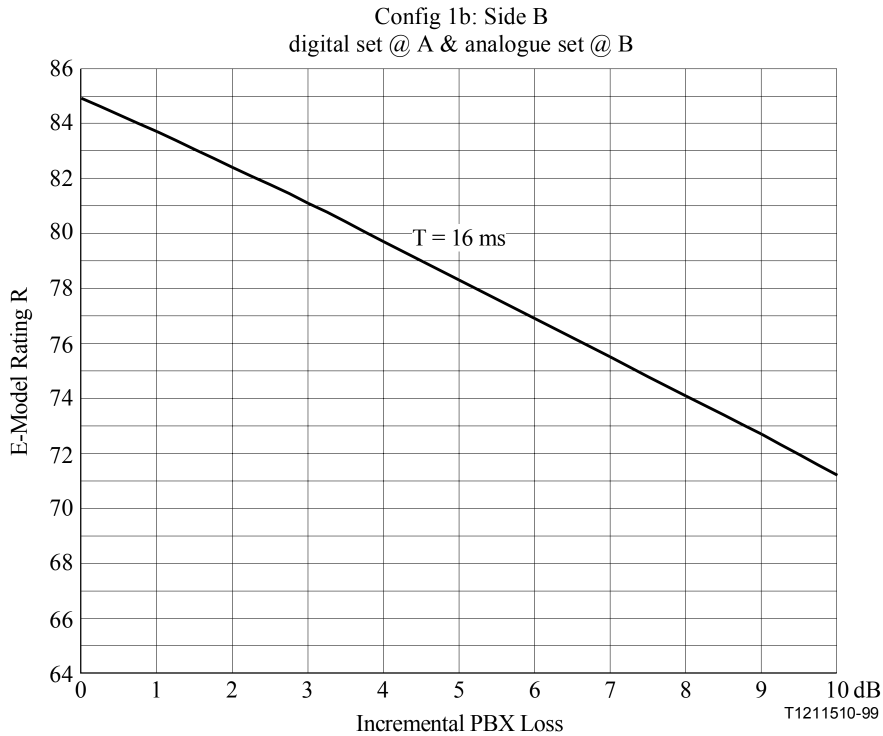


Figure B.14b/G.108 – Digital set @ A & analogue set @ B as perceived @ side B

Figures B.14a and B.14b illustrate optimal speech transmission quality points at respective incremental losses at the PBXs. For side B, the optimal speech transmission quality is located when there is no further loss in the PBX since the high echo return loss (ERL) value of 45 dB provides sufficient echo control to minimize echo effects.

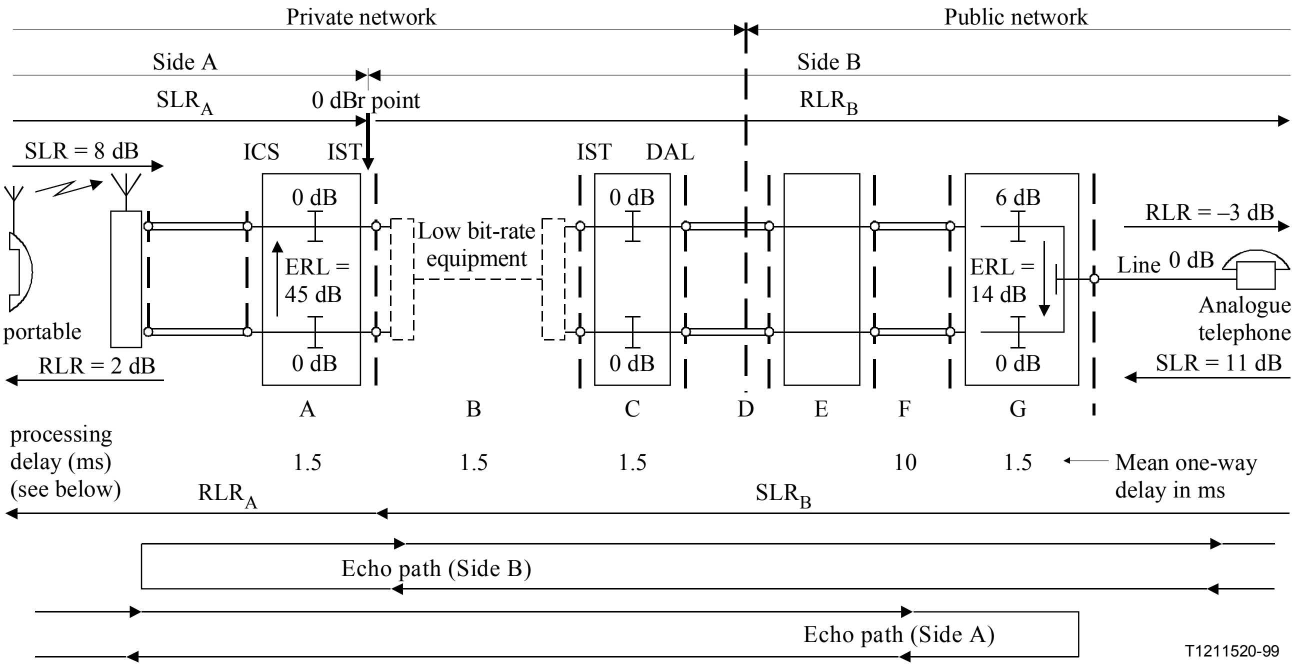
On the other hand, when listener is at side A, where the opposing side (B) has an ERL of only 14 dB, a value in the range of 4-4.5 dB indicates the best possible speech transmission quality to be obtained.

B.2.6 Reference configuration 2 (portable connection scenario)

This configuration represents a private network-to-public network connection with the private network termination consisting of a portable telephone set. This connection is the North American equivalent to reference configuration 2 in B.1.

In configuration 2, the portable (wireless set) at side A connects to digital PBX A by an ICS (ISDN Compatible Station) port. PBX A connects to transit PBX C via a Integrated Services Trunk (IST) port. The PBX loss is 0 dB in each direction, in accordance with the loss plan of [40] for ONS-IST connection. The digital tie trunk facility, B, uses low bit-rate encoding equipment. Transit PBX C connects the digital tie trunk (IST) to a Digital Access Line (DAL). Per the PBX loss plan in [40], the loss for this PBX connection is 0 dB in both direction. The digital access line, D, connects to local PSTN office E for connection to the public switched digital network F for termination at local digital office G. In compliance with the public network digital loss plan, office G inserts 6 dB loss in the receive (towards analogue access line of subscriber) direction.

Configuration 2: portable set @ A; analogue set @ B



Technology	Average processing delay
Unlicensed (PACS, WUPE, PCI, PWT)	7 ms (ranges from 4-10 ms)
Licensed TDMA	100 ms (ranges from 70-100 ms)

Figure B.15/G.108 – Basis for calculation of reference configuration 2

B.2.6.1 Pre-calculations for configuration 2

B.2.6.1.1 Calculation of SLR, RLR and OLR

Side A

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 8 dB	RLR 2 dB
PBX A	0 dB	0 dB
Sum at 0 dBr point	**SLR_A = 8 dB**	**RLR_A = 2 dB**

Side B

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 11 dB	RLR –3dB
DEO G	0 dB	6 dB
DEO E	0 dB	0 dB
PBX C	0 dB	0 dB
Sum at 0 dBr point	**SLR_B = 11 dB**	**RLR_B = 3 dB**

OLR

Listener at Side A	**Listener at Side B**
SLR_B = 11 dB	SLR_A = 8 dB
RLR_A = 2 dB	RLR_B = 3 dB
OLR = 13 dB	OLR = 11 dB

B.2.6.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

$$\begin{aligned} \text{TELR} &= \text{SLR}_{\text{SET A}} + \text{return loss (B)} + \text{sum of node losses} + \text{RLR}_{\text{SET A}} \\ &= 8 + 14 + (0 + 0 + 6 + 0 + 0 + 0) + 2 \\ &= 30 \text{ dB} \end{aligned}$$

Analysis of TELR performed at Side B:

$$\begin{aligned} \text{TELR} &= \text{SLR}_{\text{SET B}} + \text{return loss (A)} + \text{sum of node losses} + \text{RLR}_{\text{SET B}} \\ &= 11 + 45 + (0 + 0 + 0 + 0 + 0 + 6) + (-3) \\ &= 59 \text{ dB} \end{aligned}$$

B.2.6.1.3 Weighted Echo Path Loss (WEPL)

$$\begin{aligned} \text{WEPL} &= \text{return loss (A)} + \text{return loss (B)} + \text{sum of node losses} \\ &= 45 + 14 + (0 + 0 + 6 + 0 + 0 + 0) \\ &= 65 \text{ dB} \end{aligned}$$

B.2.6.1.4 Delay values T, Ta and Tr

T(ms) = Access delay of wireless technology + PBX delay + low bit-rate equipment delay + digital end office delay + network delay + digital end office delay

= 7 (unlicensed) or 100 (licensed) + 1.5 + 1.5 + 1.5 + 10 + 1.5

= 23 or 116

Ta = T

Tr = 2 × T

B.2.6.1.5 Number of quantization distortion units

This parameter represents the impairments due to quantization distortion. It is entered as the number of quantization distortion units. Note that a value of 1 is used for the A/D-D/A conversion. Digital loss pads have additional qdu value of 0.7. In this scenario, there are altogether 1 pair of A/D-D/A conversion and zero digital loss pads if side A is the listener and one digital pad if side B is the listener.

Therefore, $qdu_A = n \times 1 + m \times 0.7$

$= 1 \times 1 + 0 \times 0.7$

$= 1.0$

$qdu_B = n \times 1 + m \times 0.7$

$= 1 \times 1 + 1 \times 0.7$

$= 1.7$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

B.2.6.1.6 Equipment impairment factor

The codec used for wireless technology does have an impact on speech transmission quality. For the two types of technology analysed, values in Table 2a were used. The total equipment impairment factor (Ie) is the sum of all the equipment impairment factors involved:

Ie = Ie value for codec used in wireless technology + Ie value for network LBR equipment

= 7 (unlicensed) or 10 (licensed) + 0 (0 LBR) or 7 (1 LBR)

= 7 or 14 (unlicensed) or

= 10 or 17 (licensed)

B.2.6.2 Input parameters summary tables

The input parameters for Configuration 2 are summarized in Tables B.4a and B.4b; all other parameters remain at their default values according to Table 6.

Table B.4a/G.108 – E-Model Input Parameters for Configuration 2 as perceived at side A

Input Parameters	Values	Unit
Send Loudness Rating (SLR_B)	11	dB
Receive Loudness Rating (RLR_A)	2	dB
Talker Echo Loudness Rating	30	dB
Weighted Echo Path Loss	65	dB
Equipment impairment factor (Ie)	7 or 14 (unlicensed), 10 or 17 (licensed)	
Advantage factor (A)	5 (unlicensed), 10 (licensed)	
Quantization distortion unit	1	
Round trip delay	46 (unlicensed) 232 (licensed)	ms
Mean one-way delay	23 (unlicensed) 116 (licensed)	ms
Absolute delay	23 (unlicensed) 116 (licensed)	ms

Table B.4b/G.108 – E-Model input parameters for configuration 2 as perceived at side B

Input parameters	Values	Unit
Send Loudness Rating (SLR_A)	8	dB
Receive Loudness Rating (RLR_B)	3	dB
Talker Echo Loudness Rating	59	dB
Weighted Echo Path Loss	65	dB
Equipment impairment factor (Ie)	7 or 14 (unlicensed), 10 or 17 (licensed)	
Advantage factor (A)	5 (unlicensed), 10 (licensed)	
Quantization distortion unit	1.7	
Round trip delay	46 (unlicensed) 232 (licensed)	ms
Mean one-way delay	23 (unlicensed) 116 (licensed)	ms
Absolute delay	23 (unlicensed) 116 (licensed)	ms

B.2.6.3 Results

B.2.6.3.1 Unlicensed: Wireless processing delay: 7 ms average (WUPE, PCI, PWT)

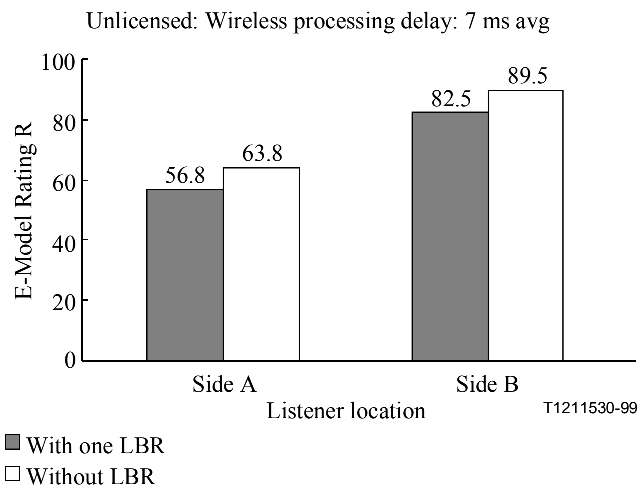


Figure B.15a/G.108 – Unlicensed: Wireless processing delay: 7 ms avg

Figure B.15a shows speech transmission quality performance when listener is at side A and side B respectively. The two bars denote the cases when there is a set of low bit-rate equipment and when there is none. It can be observed that speech transmission quality perceived at side B is better than when perceived at side A. The reason for this discrepancy is that the echo return loss value at side A is much larger than that of side B (45 dB versus 14 dB). With a larger ERL, echo cancellation is more effective and, thereby, improves the speech transmission quality.

It can also be observed that the low bit-rate equipment degrades speech transmission quality which can make a difference, especially when echo cancellation is not handled effectively (larger differences at side A compared with side B).

B.2.6.3.2 Licensed: Wireless processing delay: 100 ms average (TDMA)

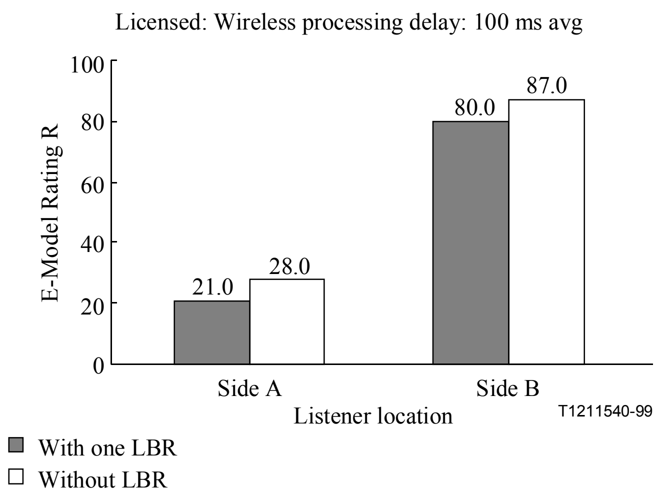


Figure B.15b/G.108 – Licensed: Wireless processing delay: 100 ms avg

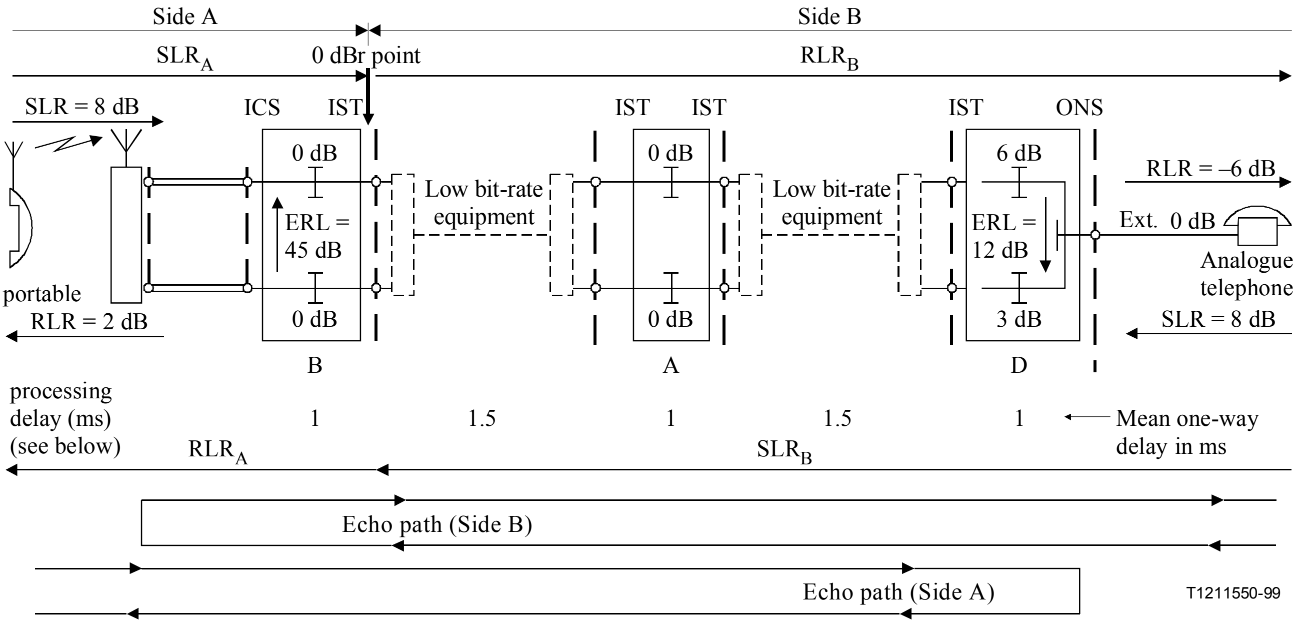
In Figure B.15b the same observation is shown for licensed technology. The only difference is that the absolute number in terms of performance is much lower than for the unlicensed wireless technology. The reason is that licensed wireless technology has much longer delay (100 ms versus 7 ms average), and the equipment impairment factor of the associated low bit rate codec is larger.

B.2.7 Reference configuration 3a (analogue set @ B)

This configuration represents a private network connection with one termination consisting of a portable telephone set and the other termination an analogue telephone set. This connection is the North American equivalent to reference configuration 3 in B.1 with an analogue telephone set at side B.

In configuration 3a, the portable (wireless set) at side A connects to digital PBX B by an ICS (ISDN Compatible Station) port. PBX A connects to transit PBX A via a Integrated Services Trunk (IST) port. The PBX loss is 0 dB in each direction, in accordance with the loss plan of [40] for ONS-IST connection. The digital tie trunk facility uses. Transit PBX A connects the digital tie trunk (IST) to another digital tie trunk (IST). Per the PBX loss plan in [40], the loss for this PBX connection is 0 dB in both directions. This digital tie trunk connects to terminating digital PBX D. As shown, both tie trunk facilities use low bit-rate encoding equipment; the analysis for this connection is made for this case, as well as for cases where only one or where neither tie trunk uses such equipment. PBX D connects to an analogue telephone via an ONS port with the losses as described in configuration 1a.

Configuration 3a: portable set @ A; analogue set @ B



Technology	Average processing delay
Unlicensed (PACS, WUPE, PCI, PWT)	7 ms (ranges from 4-10 ms)
Licensed TDMA	100 ms (ranges from 70-100 ms)

Figure B.16/G.108 – Basis for calculation of reference configuration 3a

B.2.7.1 Pre-calculations for configuration 3a

B.2.7.1.1 Calculation of SLR, RLR and OLR

Side A

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 8 dB	RLR 2 dB
PBX B	0 dB	0 dB
Sum at 0 dBr point	**SLR_A = 8 dB**	**RLR_A = 2 dB**

Side B

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 8 dB	RLR −6 dB
PBX D	3 dB	6 dB
PBX A	0 dB	0 dB
Sum at 0 dBr point	**SLR_B = 11 dB**	**RLR_B = 0 dB**

OLR

Listener at Side A	**Listener at Side B**
SLR_B = 11 dB	SLR_A = 8 dB
RLR_A = 2 dB	RLR_B = 0 dB
OLR = 13 dB	OLR = 8 dB

B.2.7.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

TELR = $SLR_{SET\ A}$ + return loss (B) + sum of node losses + $RLR_{SET\ A}$

= 8 + 12 + (0 + 0 + 6 + 3 + 0 + 0) + (2)

= 31 dB

Analysis of TELR performed at Side B:

TELR = $SLR_{SET\ B}$ + return loss (A) + sum of node losses + $RLR_{SET\ B}$

= 8 + 45 + (3 + 0 + 0 + 0 + 0 + 6) + (–6)

= 56 dB

B.2.7.1.3 Weighted Echo Path Loss (WEPL)

WEPL = return loss (A) + return loss (B) + sum of node losses

= 45 + 12 + (0 + 0 + 6 + 3 + 0 + 0)

= 66 dB

B.2.7.1.4 Delay values T, Ta and Tr

T(ms) = Access delay of wireless technology + Σ PBX delay/end office delay + Σ low bit-rate equipment delay

= 7 (unlicensed) or 100 (licensed) + $3 \times 1 + 2 \times 1.5$

= 13 or 106

Ta = T

Tr = $2 \times$ T

B.2.7.1.5 Number of quantization distortion units

This parameter represents the impairments due to quantization distortion. It is entered as the number of quantization distortion units. Note that a value of 1 is used for the A/D-D/A conversion. Digital loss pads have additional qdu value of 0.7. In this scenario, there are altogether 1 pair of A/D-D/A conversion and one digital loss pad for either side analysis.

Therefore, $\text{qdu} = n \times 1 + m \times 0.7$

$= 1 \times 1 + 1 \times 0.7$

$= 1.7$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

B.2.7.1.6 Equipment Impairment Factor

The codec used for wireless technology does have an impact on speech transmission quality. For the two types of technology analysed, values in Table 2a were used. The total equipment impairment factor (Ie) is the sum of all the equipment impairment factors involved:

Ie = Ie value for codec used in wireless technology + Ie value for network LBR equipment

= 7 (unlicensed) or 10 (licensed) + 0 (0 LBR) or 7 (1 LBR) or 14 (2 LBR)

= 7 or 14 or 21 (unlicensed) or

= 10 or 17 or 24 (licensed)

B.2.7.2 Input parameters summary tables

The input parameters for configuration 3a (analogue set @ B) are summarized in Tables B.5a and B.5b; all other parameter remain at their default values according to Table 6.

Table B.5a/G.108 – E-Model input parameters for configuration 3a (analogue set @ B) as perceived at side A

Input parameters	Values	Unit
Send Loudness Rating (SLR_B)	11	dB
Receive Loudness Rating (RLR_A)	2	dB
Talker Echo Loudness Rating	31	dB
Weighted Echo Path Loss	66	dB
Equipment impairment factor (Ie)	7 or 14 or 21 (unlicensed), 10 or 17 or 24 (licensed)	
Advantage factor (A)	5 (unlicensed) 10 (licensed)	
Quantization distortion unit	1.7	
Round trip delay	26 (unlicensed) 212 (licensed)	ms
Mean one-way delay	13 (unlicensed) 106 (licensed)	ms
Absolute delay	13 (unlicensed) 106 (licensed)	ms

Table B.5b/G.108 – E-Model input parameters for configuration 3a (analogue set @ B) as perceived at side B

Input parameters	Values	Unit
Send Loudness Rating (SLR_A)	8	dB
Receive Loudness Rating (RLR_B)	0	dB
Talker Echo Loudness Rating	56	dB
Weighted Echo Path Loss	66	dB
Equipment impairment factor (Ie)	7 or 14 or 21 (unlicensed), 10 or 17 or 24 (licensed)	
Advantage factor (A)	5 (unlicensed) 10 (licensed)	
Quantization distortion unit	1.7	
Round trip delay	26 (unlicensed) 212 (licensed)	ms
Mean one-way delay	13 (unlicensed) 106 (licensed)	ms
Absolute delay	13 (unlicensed) 106 (licensed)	ms

B.2.7.3 Results

B.2.7.3.1 Unlicensed: Wireless processing delay: 7 ms average (WUPE, PCI, PWT)

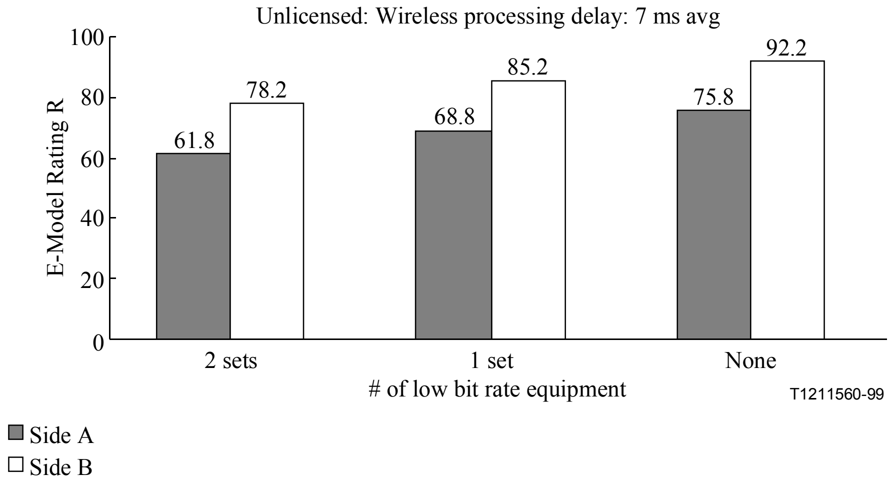


Figure B.16a/G.108 – Unlicensed: Wireless processing delay: 7 ms avg

B.2.7.3.2 Licensed: Wireless processing delay: 100 ms average (TDMA)

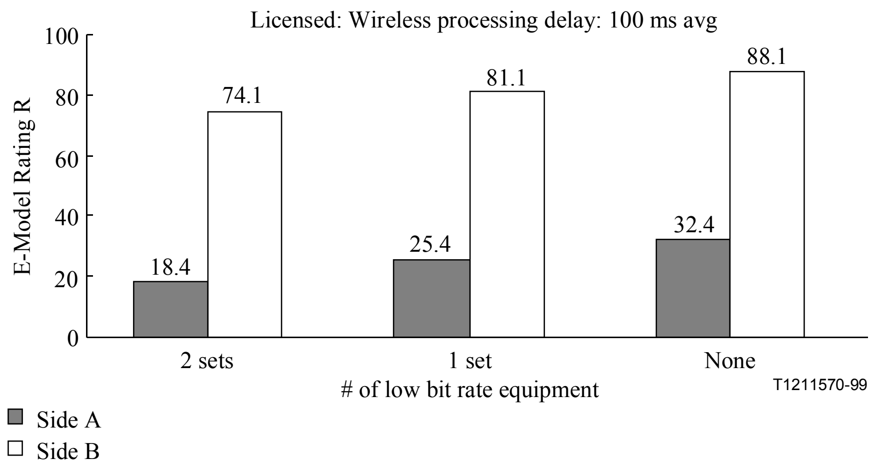


Figure B.16b/G.108 – Licensed: Wireless processing delay: 100 ms avg

The results shown in Figures B.16a and B.16b indicate speech transmission quality perceived at side B is much better than that perceived at side A. The higher value of the ERL at side A provides a better echo cancellation and hence a better speech transmission quality to the opposite party B.

Again, the more low bit-rate equipment is used, the poorer the resulting speech transmission quality. This observation is the same for both licensed and unlicensed technology.

B.2.8 Reference configuration 3b (digital set @ side B)

This configuration represents a private network connection with one termination consisting of a portable telephone set and the other termination a digital telephone set. This connection is the North American equivalent to reference configuration 3 in B.1 with a digital telephone set at side B.

In configuration 3b, the portable (wireless set) at side A connects to digital PBX B by an ICS (ISDN Compatible Station) port. PBX A connects to transit PBX A via a Integrated Services Trunk (IST) port. The PBX loss is 0 dB in each direction, in accordance with the loss plan of [40] for ONS-IST connection. Transit PBX A connects the digital tie trunk (IST) to another digital tie trunk (IST). Per the PBX loss plan in [40], the loss for this PBX connection is 0 dB in both directions. This digital tie trunk connects to terminating digital PBX D. As shown, both tie trunk facilities use low bit-rate encoding equipment; the analysis for this connection is made for this case as well as for cases where only one or where neither tie trunk uses such equipment. PBX D connects to a digital telephone via an ICS port with 0 dB loss in each direction.

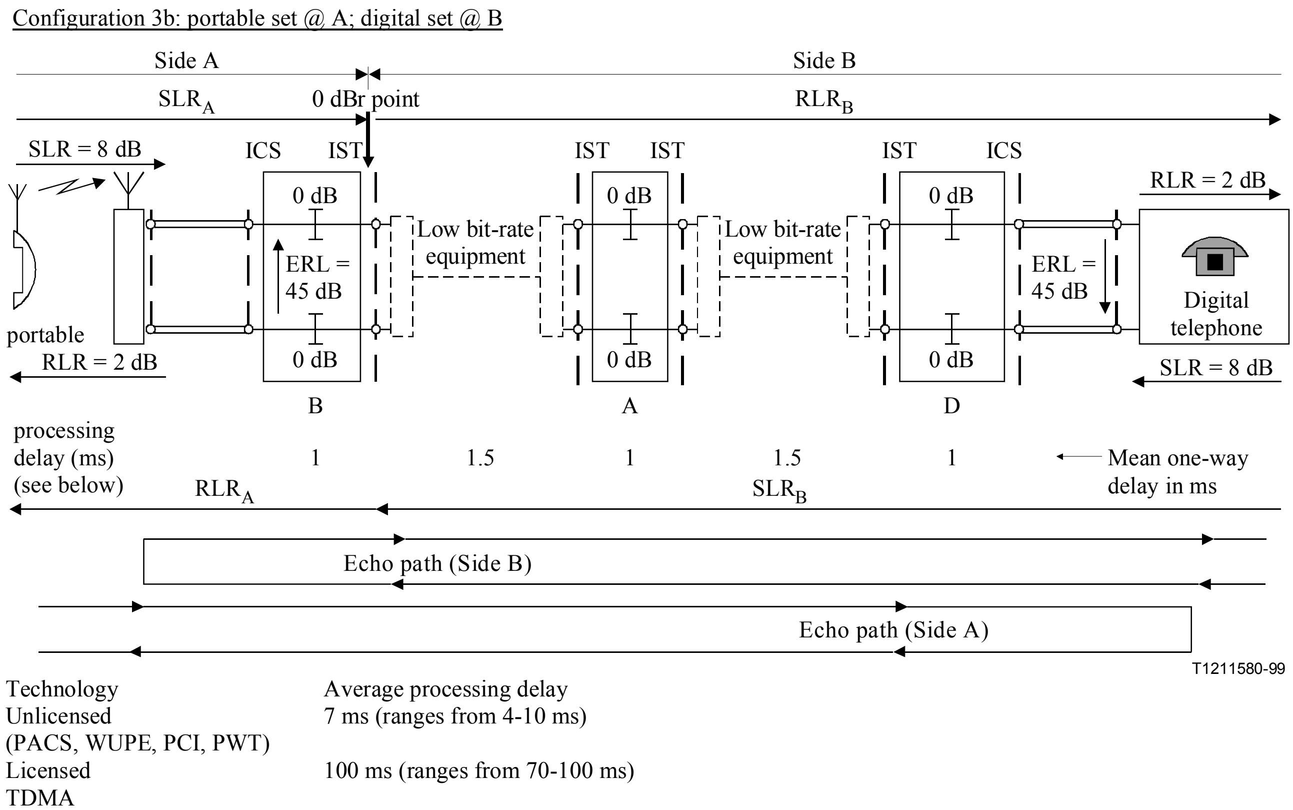


Figure B.17/G.108 – Basis for calculation of reference configuration 3b

B.2.8.1 Pre-calculations for configuration 3b

B.2.8.1.1 Calculation of SLR, RLR and OLR

Side A

	Set → 0 dBr point	0 dBr point → Set
Telephone set	SLR 8 dB	RLR 2 dB
PBX B	0 dB	0 dB
Sum at 0 dBr point	**SLR_A = 8 dB**	**RLR_A = 2 dB**

Side B

	Set → 0 dBr point	**0 dBr point → Set**
Telephone set	SLR 8 dB	RLR 2 dB
PBX D	0 dB	0 dB
PBX A	0 dB	0 dB
Sum at 0 dBr point	**SLR_B = 8 dB**	**RLR_B = 2 dB**

OLR

Listener at Side A	**Listener at Side B**
SLR_B = 8 dB	SLR_A = 8 dB
RLR_A = 2 dB	RLR_B = 2 dB
OLR = 10 dB	OLR = 10 dB

B.2.8.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

TELR = $SLR_{SET\ A}$ + return loss (B) + sum of node losses + $RLR_{SET\ A}$

= 8 + 45 + (0 + 0 + 0 + 0 + 0 + 0) + (2)

= 55 dB

Analysis of TELR performed at Side B:

TELR = $SLR_{SET\ B}$ + return loss (A) + sum of node losses + $RLR_{SET\ B}$

= 8 + 45 + (0 + 0 + 0 + 0 + 0 + 0) + (2)

= 55 dB

B.2.8.1.3 Weighted Echo Path Loss (WEPL)

WEPL = return loss (A) + return loss (B) + sum of node losses

= 45 + 45 + (0 + 0 + 0 + 0 + 0 + 0)

= 90 dB

B.2.8.1.4 Delay values T, Ta and Tr

T(ms) = Access delay of wireless technology + Σ PBX delay/end office delay + Σ low bit-rate equipment delay

= 7 (unlicensed) or 100 (licensed) + 3 × 1 + 2 × 1.5

= 13 or 106

Ta = T

Tr = 2 × T

B.2.8.1.5 Number of quantization distortion units

This parameter represents the impairments due to quantization distortion. It is entered as the number of quantization distortion units. Note that a value of 1 is used for the A/D-D/A conversion. Digital loss pads have additional qdu value of 0.7. In this scenario, there are altogether 1 pair of A/D-D/A conversion for either side analysis.

Therefore,

$$\begin{aligned} qdu &= n \times 1 + m \times 0.7 \\ &= 1 \times 1 + 1 \times 0.7 \\ &= 1.7 \end{aligned}$$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

B.2.8.1.6 Equipment impairment factor

The codec used for wireless technology does have an impact on speech transmission quality. For the two types of technology analysed, values in Table 2a were used. The total equipment impairment factor (Ie) is the sum of all the equipment impairment factors involved:

Ie = Ie value for codec used in wireless technology + Ie value for network LBR equipment

= 7 (unlicensed) or 10 (licensed) + 0 (0 LBR) or 7 (1 LBR) or 14 (2 LBR)

= 7 or 14 or 21 (unlicensed) or

= 10 or 17 or 24 (licensed).

B.2.8.2 Input parameters summary tables

The input parameters for configuration 3b (digital set @ B) are summarized in Tables B.6a and B.6b; all other parameter remain at their default values according to Table 6.

Table B.6a/G.108 – E-Model input parameters for configuration 3b (digital set @ B) as perceived at side A

Input parameters	Values	Unit
Send Loudness Rating (SLR_B)	8	dB
Receive Loudness Rating (RLR_A)	2	dB
Talker Echo Loudness Rating	55	dB
Weighted Echo Path Loss	90	dB
Equipment impairment factor (Ie)	7 or 14 or 21 (unlicensed) 10 or 17 or 24 (licensed)	
Advantage factor (A)	5 (unlicensed), 10 (licensed)	
Quantization distortion unit	1.7	
Round trip delay	26 (unlicensed) 212 (licensed)	ms
Mean one-way delay	13 (unlicensed) 106 (licensed)	ms
Absolute delay	13 (unlicensed) 106 (licensed)	ms

Table B.6b/G.108 – E-Model input parameters for configuration 3b (digital set @ B) as perceived at side B

Input parameters	Values	Unit
Send Loudness Rating (SLR_A)	8	dB
Receive Loudness Rating (RLR_B)	2	dB
Talker Echo Loudness Rating	55	dB
Weighted Echo Path Loss	90	dB
Equipment impairment factor (Ie)	7 or 14 or 21 (unlicensed) 10 or 17 or 24 (licensed)	
Advantage factor (A)	5 (unlicensed), 10 (licensed)	
Quantization distortion unit	1.7	
Round trip delay	26 (unlicensed) 212 (licensed)	ms
Mean one-way delay	13 (unlicensed) 106 (licensed)	ms
Absolute delay	13 (unlicensed) 106 (licensed)	ms

B.2.8.3 Results

B.2.8.3.1 Unlicensed: Wireless processing delay: 7 ms average (WUPE, PCI, PWT)

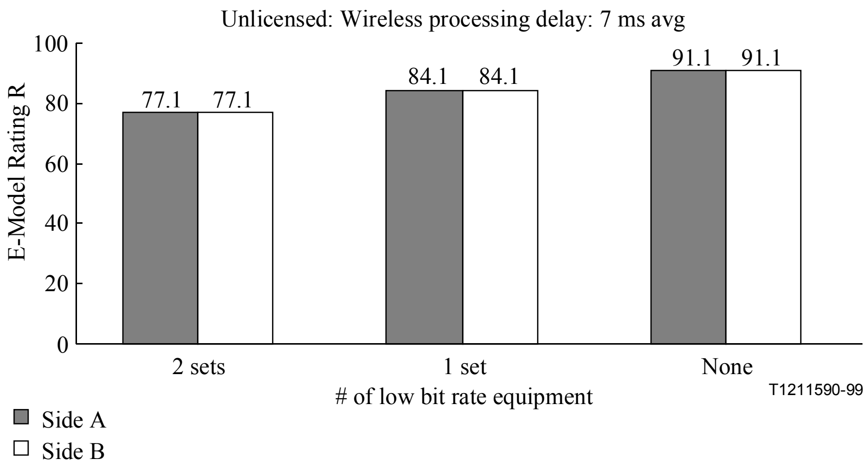


Figure B.17a/G.108 – Unlicensed: Wireless processing delay: 7 ms avg

B.2.8.3.2 Licensed: Wireless processing delay: 100 ms average (TDMA)

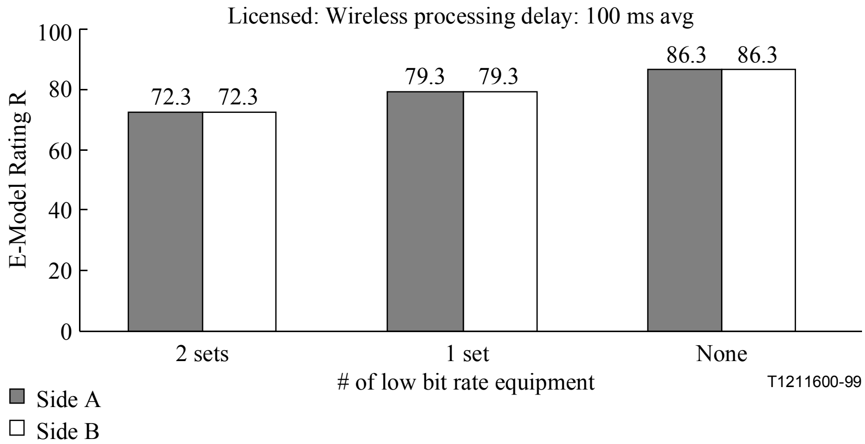


Figure B.17b/G.108 – Licensed: Wireless processing delay: 100 ms avg

As shown in Figure B.17a, speech transmission quality perceived at either side A or side B is the same as the value of ERL at either end is the same value (45 dB), resulting in speech transmission quality performance in the satisfied region. The same observation is obtained in Figure B.17b for licensed technology. The only difference is the lower value in absolute terms for licensed technology because of the longer delay and the higher impairment factor.

B.3 Planning example for IP transmission based scenarios

Table B.7 shows sample E-Model calculations for several IP telephony scenarios. An "impairment-free" connection with 50 ms one-way delay is included as a reference. Such a connection meets the delay allocation for a national segment as defined in Recommendation G.114 [6].

The only E-Model parameters that have been varied are the equipment impairment factor of the codec and the delay. All other parameters are set to the default values shown in Table 6. In particular, effects of channel errors, packet loss and delay variability are not included.

The equipment impairment factors are taken from the current Appendix I/G.113 [5].

One-way delay for a given IP telephony system is determined by the network delay and the number of encoded frames that are included in each IP packet (see 7.3.2.3). A lost packet will result in speech clipping. In order to maintain a good end-to-end speech transmission performance, the speech contained in coded frames should be less than 64 ms of speech per IP packet (see 7.5 on Temporal (Syllable) Clipping Recommendation G.177 [13]). Hence, at most two frames of G.723.1, or six frames of G.729A, are assembled into a packet. In Table B.7, the "packet delay" column is the time required for the assembly of a packet including the encoding and decoding process, and is computed as:

$$T_F(N+1)+T_L$$

where T_F is the frame size for the codec, N is the number of frames in a packet, and T_L is the look-ahead time for the codec. For G.723.1 (both bit rates), the frame size is 30 ms and the look-ahead is 7.5 ms. For G.729A, the frame size is 10 ms and the look-ahead is 5 ms.

Table B.7/G.108 – Example E-Model calculations for IP telephony

Codec	Equipment impairment factor (Ie)	# frames/ packet	Packet delay (ms)	Network delay (ms)	End-to-end mean one-way delay (ms)	E-Model Rating R
G.711 (PSTN)	0	N/A	N/A	50	50	92.7
G.723.1 (5.3)	19	1	67.5	50	117.5	72.3
G.723.1 (5.3)	19	2	97.5	50	147.5	71.6
G.723.1 (6.3)	15	1	67.5	50	117.5	76.3
G.723.1 (6.3)	15	2	97.5	50	147.5	75.6
G.729A	11	1	25	50	75	81.1
G.729A	11	2	35	50	85	80.9
G.729A	11	3	45	50	95	80.7
G.729A	11	4	55	50	105	80.5
G.729A	11	5	65	50	115	80.3
G.729A	11	6	75	50	125	80.1
NOTE – Equipment impairment factors (Ie) for Recommendations G.723.1 and G.729A are for the codec with voice activity detection turned on. See text for details on delay values.						

Table B.8 shows R-values for indicated combinations of Ie and end-to-end mean one-way delay. Note that the different shadings relate to the different categories of speech transmission quality as defined in Table 1.

Table B.8/G.108 – E-Model Ratings R for indicated combinations of Ie and end-to-end mean one-way delay

Ie=	0	5	7	10	15	19	19	20	26
	G.711	GSM-EFR	G.726 @32	G.729	G.723.1 @6.3	G.729A +VAD w/2% loss	G.723.1 @5.3	GSM-FR	G.729A +VAD w/4% loss
ms			G.728 @16				G.723.1@ 6.3+VAD w/1% loss	IS-54	
~0	94		87						
50	93		86	83		74			67
100	92	87	85	82	77	73	73	72	66
150	90	85	83	80	75	71	71	70	64
200	87	82	80	77	72	68	68	67	61
250	80	75	73	70	65	61	61	60	54
300	74	69	67	64	59	55	55	54	48
350	68	63	61	58	53	49	49	48	42
400	63	58	56	53	48	44	44	43	37
450	59	54	52	49	44	40	40	39	33

NOTE 1 – R-values in this table have been calculated using the indicated values for Ie and T (T=Ta=Tr/2) along with the default values from Table 6 for all other parameters.

NOTE 2 – Unless indicated otherwise, examples do not include packet loss or Voice Activity Detection (VAD).

NOTE 3 – Blackened cells indicate combinations of delay and codec that are impossible to realize.

ANNEX C

Echo control in specific applications

In traditional networks, the insertion of echo control devices was mainly necessary for international connections. Usually they were located in international switching centres and, therefore, in most cases, within the responsibility of public network operators. The increasing use of digital routing in conjunction with equipment using low bit-rate coding has changed these previous rules. Additional transmission delay may now be arising due to the processing delays of specific equipment such as multiplexers, voice activity detection circuits, mobile and cordless telephones, equipment which are used more in the lower hierarchy of public networks or directly within private networks.

In case calls are being routed, e.g. via multiplexing equipment or are being terminated with mobile or specific cordless telephones, the additional delay introduced into a connection may have such a high value, that echo control devices are necessary even for national calls, hence in international calls the permissible echo path delay of the echo canceller in an international switching centre could be exceeded. Therefore, in most applications those multiplexers or terminals are equipped with "integrated" echo control devices to guarantee sufficient echo control for every type of connection. However, these integrated echo control devices should be considered carefully during planning to avoid incorrect range of operation and insufficient interworking with other echo control devices.

The following clauses describe the different aspects which should be considered for those types of integrated echo control devices referring, as an example, to a cordless telephone according to [50] and [53] inserting an additional mean one-way delay of about 14 ms.

C.1 Effective echo paths

The fixed part of a cordless telephone (base station) can be connected either 2-wire analogue or digital to a switching element (e.g. PBX in a private network). If this terminal within the private network is connected to another terminal via a public network, several echo paths may be effective as shown in Figure C.1, which should be identified and investigated during planning.

The different echo paths in this configuration are described in the following subclauses:

C.1.1 Echo path 1: for talker at the cordless telephone

This echo path is only effective if the fixed part is connected 2-wire analogue via a hybrid to the PBX. This hybrid is forming an echo path for the talker at the cordless telephone with a mean one-way delay of 14 ms, a value which requires echo control.

C.1.2 Echo path 2: for talker at the cordless telephone

This echo path via the public network is effective if the far end is terminated with a 4-wire to 2-wire conversion (hybrid). For planning purposes, however, at present this should be assumed in most cases. This echo path is independent of the type of access of the fixed part (2-wire analogue or digital) and its mean one-way delay is including the delay of the public/private network and is, therefore, in any case higher than for echo path 1. Both echo paths 1 and 2 each with different mean one-way delay and different TELR values can be effective for the talker at the cordless telephone at the same time. Therefore, the cordless telephone should be equipped with echo control devices able to control both types of echo.

C.1.3 Echo path 3: for talker at the cordless telephone

This echo path via the acoustic coupling of the far-end telephone set usually is negligible compared to the impairments caused by echo paths 1 and 2. Although the mean one-way delay can be higher than for echo path 2, e.g. if a cordless telephone according to DECT standard is used on both sides, the resulting impairment will be low, since the corresponding TELR for this echo path can be expected with more than 44 dB.

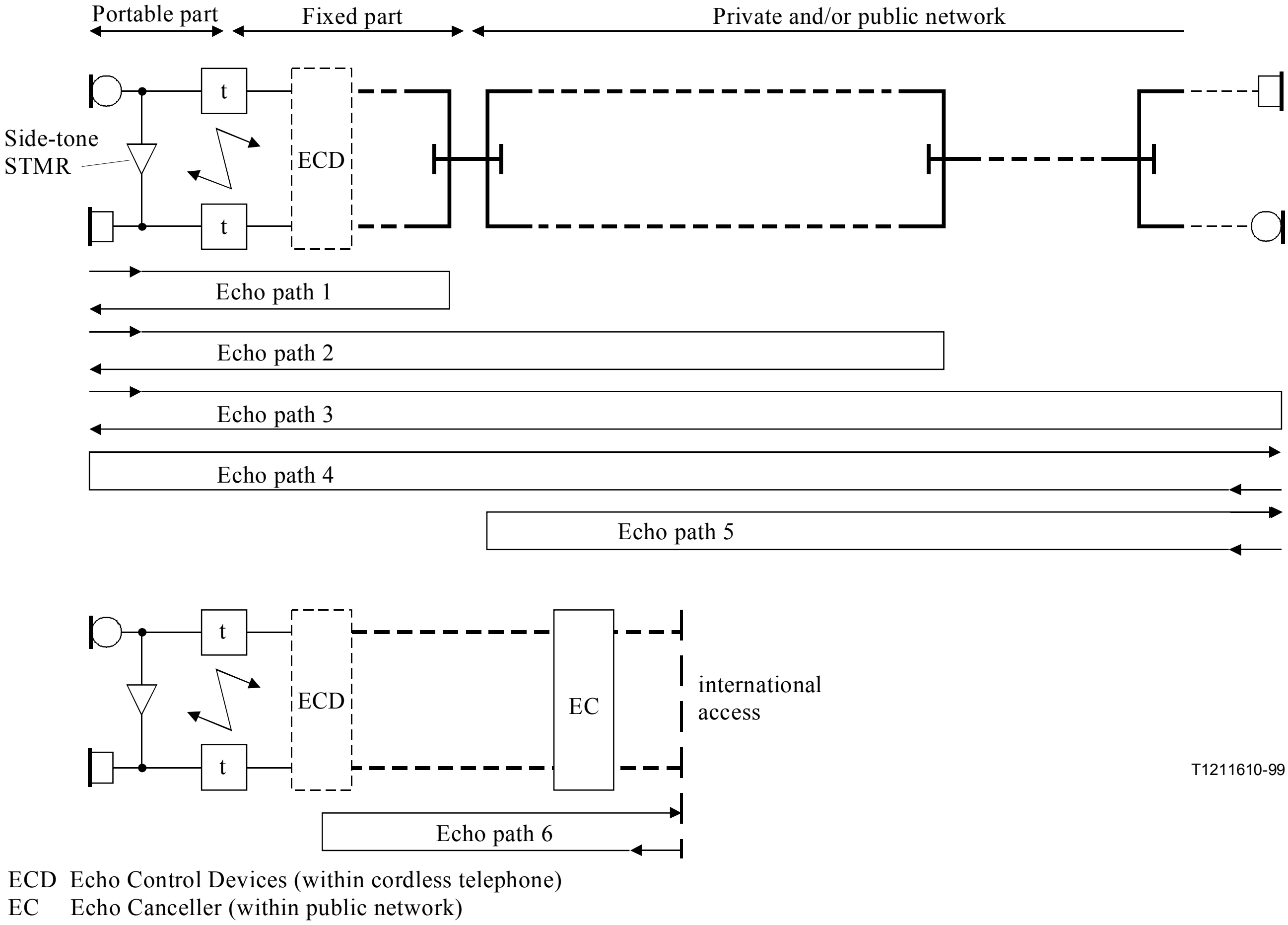


Figure C.1/G.108 – Possible echo paths for a cordless telephone

C.1.4 Echo path 4: for talker of the public network

For the echo path being effective here for the talker at the public network side, in principle the same issues are valid as for echo path 3. For planning purposes it is necessary only to investigate whether the provided TCLw referred to the fixed part amounts to more than 34 dB if the path within the private network is fully digital.

C.1.5 Echo path 5: for talker of the public network

This echo path is mentioned here only for the sake of completeness since the requirements for a sufficient echo loss referred to the interface between public and private network are in this case independent from the type of terminal equipment and its mean one-way delay. Echo path 5 will in normal configurations contribute with higher impairments than echo path 4 due to the lower value of only 24 dB for the echo loss.

C.1.6 Echo path 6: for talker of the public network in an international call

Although comparable with echo path 5, this configuration should be considered separately due to the echo canceller which is automatically enabled within the international switching centre because of the international connection. The characteristics of these echo cancellers, presently designed according to Recommendations G.165 [11] or G.168 [12], are usually not adapted to the additional delay and the high values of TCLw of a terminating cordless telephone. Therefore, precautions should be taken to guarantee a proper operation of this echo canceller.

C.2 Operation of echo cancellers and soft suppressors

The main devices in a cordless telephone according to the DECT standard for the control of echo along the different echo paths are shown in principle in Figure C.2. For the given configuration of a 2-wire analogue connection to the private/public network the two echo paths 1 and 2 should be considered. The analogue access requires not only a hybrid but also an adjustment of the SLR and RLR of the cordless terminal (including the fixed part) to national requirements. The DECT standard in general specifies values of SLR = 7 dB and RLR = 3 dB referred to an internal reference point (0 dBr-point). To meet the national requirements at the 2-wire interface of the fixed part (in this example SLR = +3 dB, RLR = −8 dB) a gain of 4 dB in the transmit path and of 11 dB in the receive path are adjusted within the hybrid. Assuming a proper impedance strategy (see C.1.3) an average terminal balance return loss of $a_{BRL} = 18$ dB can be estimated; the resulting transhybrid loss of $a_{TRH} = -3$ dB is very low and could cause remarkable impairments. To suppress this "near-end echo" of echo path 1 two devices, the echo canceller EC and the soft suppressor (SS) are inserted between the so called UPCM-interfaces (Uniform-PCM).

The principle of the echo canceller is in accordance with the description in Recommendation G.165 [11] or G.168 [12], but without the NLP and with less stringent requirements. The corresponding standard requires only an enhancement of the transhybrid loss, called echo loss enhancement, of > 6.5 dB, which is resulting in an echo loss of −3 dB + 6.5 dB = 9.5 dB in the example as shown in Figure C.2. Furthermore, the permissible echo path delay for this EC is required with only > 4 ms, since only the hybrid circuit is forming the echo path.

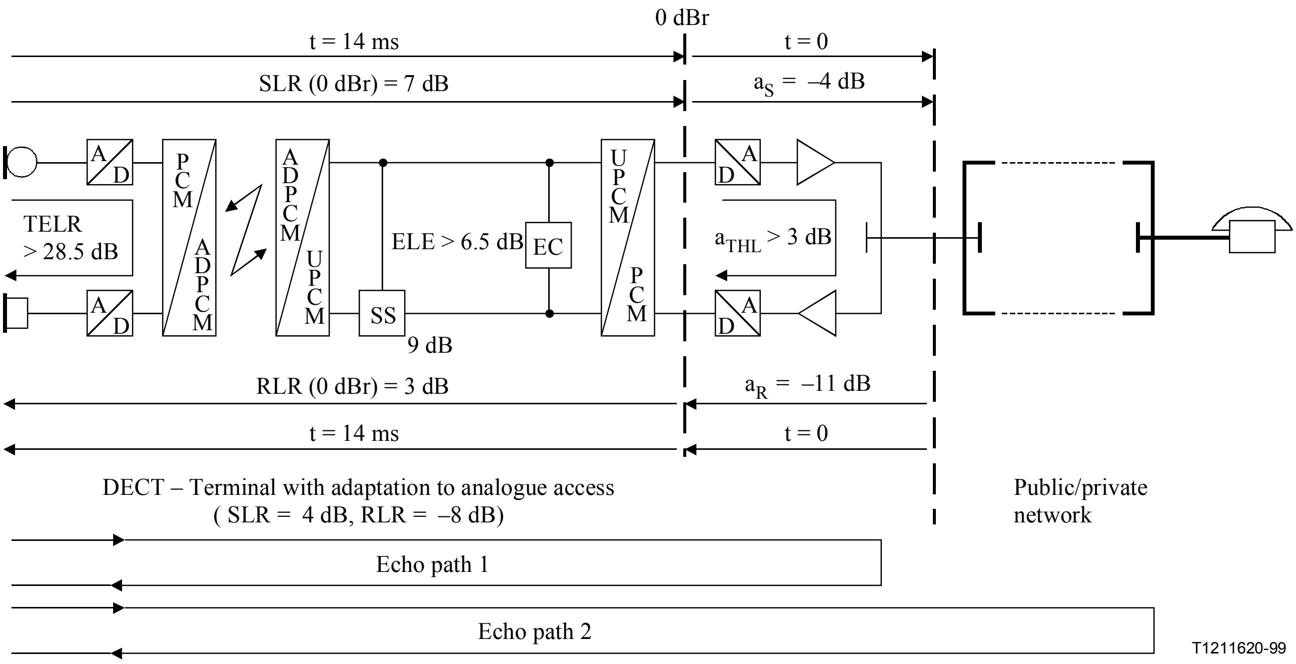


Figure C.2/G.108 – Echo control devices in a DECT cordless telephone

To increase the echo loss of only 9.5 dB an additional device, the soft suppressor (SS) is inserted. This SS is in its operational mode comparable with an echo suppressor. If the talker's voice signal is detected in the transmit path, an additional loss of 9 dB is enabled in the receive path during talking, increasing the echo loss to 18.5 dB. Together with the SLR and RLR at the 0 dBr-point the total TELR is 28.5 dB, which is sufficient for this echo in echo path 1.

For echo path 2, an echo loss of 24 dB with an additional mean one-way delay of 30 ms via the public network is assumed. The echo canceller (EC) is unable to compensate this echo path due to the low value of 4 ms for the permissible echo path delay. In this case only the soft suppressor (SS) is, during talking, increasing the echo loss from 24 dB to 33 dB, resulting in a sufficiently high value of 41 dB for the TELR.

C.3 Provision of echo control for the talker of the public network

For the far-end talker provisions should be made by the private network subject to planning for a sufficient high value of TELR. The effective echo paths for this talker are the paths 4, 5 and 6 as shown in Figure C.1. If the connection is terminated within the private network by a hybrid the value for TELR is stated with 24 dB in 7.7 and is not considered here.

For a fully digital routing within the private network and termination by a cordless telephone, the echo loss is identical with the TCLw of the portable part. For cordless telephones according to the DECT standard basically a TCLw of > 46 dB is required, however, also only 34 dB are allowed as an option. For the echo path of national calls (echo path 4 in Figure C.1) this value is providing a sufficiently high value of TELR = 44 dB (assuming SLR = 7 dB and RLR = 3 dB for the far-end termination with respect to the public/private network interface).

For international connections with an enabled echo canceller within the public network, this high value of TELR in conjunction with the additional mean one-way delay of 14 ms may cause an improper operation of this echo canceller. Therefore, if the portable part is only providing the optional TCLw of 34 dB an artificial echo loss (AE) should be inserted at the network-side of the fixed part. This configuration is shown in Figure C.3.

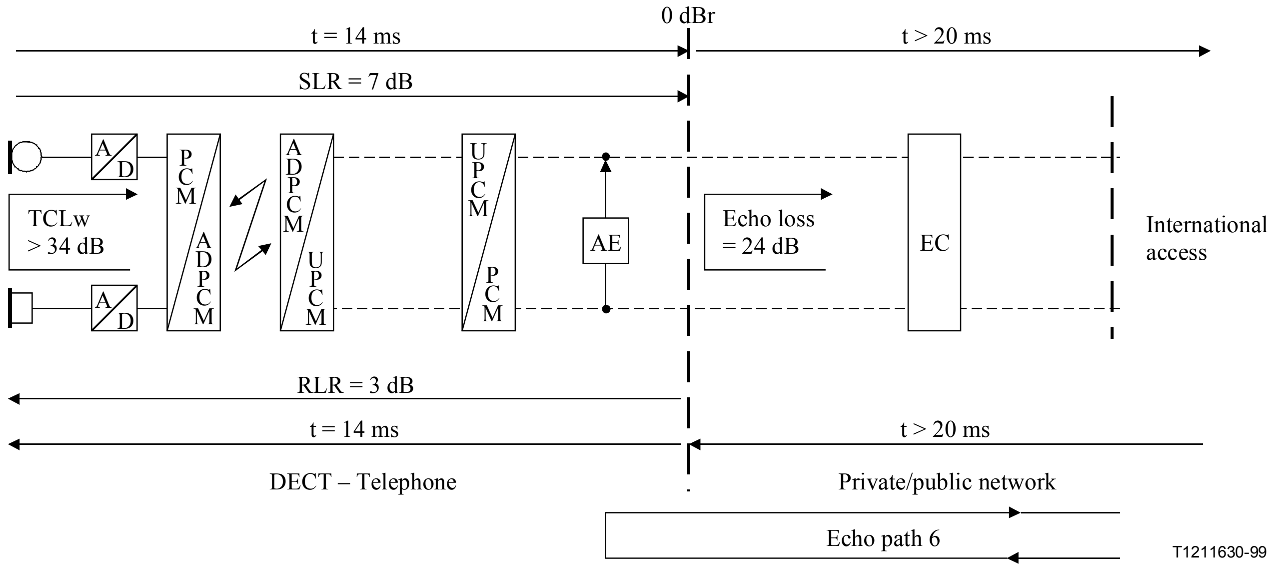


Figure C.3/G.108 – TCLw and artificial echo loss

The purpose of this artificial echo loss (AE) (a directional loss from digital in to digital out) is to provide a "virtual" echo path with an "in-range" echo loss of 24 dB and excluding the additional delay of 14 ms from the cordless telephone. Residual echo levels via the TCLw and including this additional delay are suppressed by the NLP of the echo canceller in the public network, because these residual echo levels are now well below the threshold of the NLP.

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