

Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network



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Foreword

This ETSI Guide (EG) has been produced by ETSI Technical Committee Speech Processing, Transmission and Quality Aspects (STQ).

The present document represents a joint effort between TIA Subcommittee TR-41.1 (Multi-line Telecommunications Systems) and ETSI Technical Subcommittee CN-7 (Corporate Network Transmission) (see note) to develop a global private network transmission planning guide. The intent of the document is to provide the planner of multi-national and transcontinental private networks with the transmission planning principles and planning examples which will enable the planner to design a loss and level plan for the network, optimized to the desired level of voice quality between any two users of the network as well as between the network and other - e.g., public - networks.

NOTE: The main effort on the part of ETSI in the development of the present document occurred during the existence of CN-7. That subcommittee was a successor to the transmission working group in ETSI subcommittee BTC-2. Subcommittee CN-7 has since closed down and the work now continues as part of the program of ETSI Technical Committee STQ (Speech Processing, Transmission and Quality Aspects).

The text of the present document was largely generated by ETSI as an update to an earlier ETSI Technical Report, ETR 004. Concurrent with that effort, TIA TR-41.1 started a project to update the document "DPBX Loss Plan Application Guide" (TSB-32-1989). TIA TR-41.1 and ETSI CN-7 agreed in 1997 that there was a need for a common global transmission planning guide for private networks; therefore TIA TR-41.1 elected to work with the ETR 004 update project to generate the present document by adding text (network characteristics, pertinent regional parameters, planning examples, and references) which would be of value to planners in both the North American and European regions. For North American users, the document will be designated as the update of TSB-32-1989. Meanwhile, ETSI is publishing the document as an ETSI Guide, EG 201 050.

The underlying foundation for the planning guidelines in the present document is the ETSI E-Model for calculating and evaluating end-to-end connection voice quality. The E-Model is gaining universal acceptance as the tool for transmission planning. It is expected that the E-Model will undergo refinements over time; the present document may require updating accordingly.

As a component of the TSB-32-1989 update project, TIA TR-41.1 also undertook to examine a number of connection scenarios specific to North America for the illustration of loss design based on the E-Model. The rationale for this work was to provide continuity from the presentation in the initial publication of the TSB. Rather than including these scenarios in the document, it is intended to separately publish these scenarios as a companion document for demonstrating practical applications of loss design for a variety of connection scenarios.

Introduction

Private 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. In North America, this planning had to take into account the industry-prescribed loss plan for Multi-line Telecommunications Systems (e.g., for PBXs, as presented in TIA-464B); this was the premise for developing a DPBX loss plan application guide (TSB-32-1989). 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. Finally, in most cases, private networks were geographically limited to a single region or country.

The global private network transmission environment is rapidly changing. Factors which are impacting this change are deregulation of network connectivity, 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 customers' need for economical and flexible network solutions. Previous transmission planning methodology and documents, such as TSB-32-1989 or ETR 004 are no longer flexible enough to account for all these factors. For the development of this new Transmission Planning Guide, the goals can be summarized in the following:

- a) 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.
- b) Multi-national private networks are becoming common and require planning which takes into account regional differences in loss plan requirements and internetwork transmission plans.
- c) The changing situation in the field of regulation policy is a consideration in the planning process.
- d) The changing scenario in the public network operator domain impacts transmission performance.
- e) The Document should be applicable to the use of new technology within private networks, including wireless (cordless or mobile) sections, transmission of packetized voice, etc.
- f) The Document should provide sufficient planning methods and contain all the necessary information and tools which will enable the planner to design the private network plan such as to keep the resulting voice transmission quality for all end-to-end connections within expected limits.
- g) Greater flexibility in allocation of transmission parameters and provisions with more allowance for private network transmission parameters than in earlier guidelines, standards, or regulations.
- h) Private network transmission guidelines should be aligned with ongoing transmission work in various North American and international forums, e.g., ANSI committee T1, Study Group 12 of ITU-T, and with their standards, e.g., T1.508, Recommendations G.113, and G.175.
- i) The guidelines and planning examples of the Document are based on the use of the E-Model as described in ETR 250.

The goal of greater flexibility in the allocation of transmission parameters is a key feature in the present document, since the use of new technologies and the application of future new services can only be realized in an economical way, if the allowed ranges of values for the transmission parameters within a private network are no longer rigidly constrained. The guidelines and principles in this Document 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 quality for various connection scenarios.

Furthermore, the guidelines 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 about the expected speech quality (in terms of Mean Opinion Score or percentage Good-or-Better and Poor-or-Worse) for the investigated configuration.

1 Scope

The present document applies to transmission within private networks and the interconnection of private networks with other - mainly public - networks. It should be considered as a tutorial and illustration for the planning of private networks with respect to the voice transmission quality of narrowband 3,1 kHz real time telephony via handsets. Networks designed according to the present document will also provide sufficiently high speech quality for the transmission of announcements and stored speech.

The present document does not address the transmission of non-voice signals such as Fax- and Modem transmission and wholly digital data transmission.

The main application of the present document 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 this term is 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 the present document.

The document addresses only scenarios where a "private network" functions as a terminating network (one to which terminal equipment is connected). Scenarios wherein a private network provides transit connections between other networks are outside the scope of the present document, as are the following issues:

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

Notwithstanding these limitations in the scope, the principles and information described in the present document may be applied to other end-to-end connections.

For the purposes of the present document, there are no restrictions on the private network with respect to size, configuration, hierarchy, technology used, and network components. The transmission media may be cable, fiber or radio.

The discussion in the present document addresses primarily the use of digital interfaces between the private and the public network (nearly universal in Europe) but also allows for analogue private-to-public network connections. The signal transmission within the private network 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 with 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 the present document. More emphasis is placed on parameters such as echo, return loss, delay, signal processing equipment impairments, and acoustic characteristics of terminals.

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

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

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3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACELP	Algebraic-Code-Excited Linear-Prediction
ADPCM	Adaptive Differential Pulse Code Modulation
ATM	Asynchronous Transfer Mode
CLR	Circuit Loudness Rating
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 Telecommunications

DPBX	Digital PBX
EC	Echo Canceller
ECD	Echo Control Device
EFRC	Enhanced Full Rate Codec
EL	Echo Loss
ELE	Echo Loss Enhancement
ERLE	Echo Return Loss Enhancement
ERP	Ear Reference Point
ES	Echo Suppressor
GoB	Good or Better
GSM	Global System for Mobile communications
FDM	Frequency Division Multiplex
IC	Interexchange Carrier
ICP	International Connection Point
ICS	ISDN Compatible Station
INMD	In-service Non-intrusive Measurement Device
IST	Integrated Services Trunk
ITU	International Telecommunication Union
ITU-T	ITU Telecommunication Standardization Sector (former CCITT)
KTS	Key Telephone System
LD-CELP	Low-Delay Code-Excited Linear-Prediction
LEC	Local Exchange Carrier
LELR	Listeners Echo Loudness Rating
LSTR	Listener SideTone masking Rating
MOS	Mean Opinion Score
MP-MLQ	Multi-Pulse Maximum-Likelihood-Quantization
MRP	Mouth Reference Point
N.A.	North America(n)
NCP	Network Connection Points
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
SLR	Send Loudness Rating
SS	Soft Suppressor
STMR	SideTone Masking Rating
TBRL	Terminal Balance Return Loss
TCLw	Terminal Coupling Loss weighted
TDM	Time Division Multiplex Systems
TDMA	Time Division Multiple Access
TELR	Talker Echo Loudness Rating
TIA	Telecommunications Industries Association
TME	Terminate Early
UPCM	Uniform PCM
VAD	Voice Activity Detection
VNL	Via Net Loss
VPN	Virtual Private Network

VSELP	Vector Sum Excited Linear Prediction
WEPL	Weighted Echo Path Loss
WUPE	Wireless User Premises Equipment

4 Definitions

For the purposes of the present document, the following definitions apply.

4.1 Private Network

In the present document 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 consists of several interconnected nodes (i.e., PBXs), with interconnections to other (mainly public) networks. The following is a list of private network characteristics which may impact the overall transmission quality of an end-to-end connection.

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 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.

NOTE 1: As stated in the Scope, the present document addresses only the case where a private network functions as a terminating network (i.e., has terminal equipment connected directly to it).

NOTE 2: This definition is partly identical with the definition of a private network as given in the ITU-T Recommendation G.175 "Transmission planning for private / public network interconnection of voice traffic".

4.2 Public Network

The term "Public Network" is used in the present document for any network providing transmission and switching functions as well as features available to the general public, not just 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 only a limited set of features. 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).

NOTE: The Scope covers the case where public networks either:

- interconnect private networks (functions as transit network); or,
- interconnect private networks and terminal equipment (function as transit and terminating network).

4.3 Quality Aspects

Previous planning methods for private networks - or 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 voice 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 "Voice 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) used. 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'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 ITU-T Recommendations (e.g., P.800, P.82, P.830, P.84). One of the most common methods is to perform laboratory tests (e.g., "listening only tests"), where the test subjects are requested to classify the perceived quality into categories. For example, a "Quality Rating" can be defined in 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 quality of service 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. This can be done subjectively by customer interviews or via an objective method using test equipment referred to as an "In-service Non-intrusive Measurement Device" (INMD), capable of measuring several transmission parameters during the talking state of established connections. However, those methods are not applicable for the planning of (private) networks.

The main task during planning of private networks is to collect the necessary information about the various network components in the configuration investigated and their contribution of transmission impairments that impact the end-to-end connection voice quality. To assist the planner, computation models are available which, with the pertinent input data, yield a calculated value for quality measures, such as MOS, GoB and PoW. One of those tools is the "ETSI Computation Model" - in the present document, and generally referred to as the "E-Model" - which is recommended to be used for planning purposes. For more information about the E-Model, see ETR 250 and subclause 6.2 in conjunction with annex B of the present document.

4.4 Network Elements

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

4.4.1 Terminal Elements

With respect to speech transmission, terminal elements are all types of telephone sets, digital or analogue, wired, cordless, or mobile, and include 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 (the D-Factor), the frequency response in send and receive directions and the noise floor - also contribute to the end-to-end connection rating of voice quality. In case of wireless systems, additional distortions and delay may be added, depending on the coding and modulation algorithm used for the radio interface.

4.4.2 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 private networks. Connection elements may use analogue or digital switching 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. 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.

4.4.3 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), fiber-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 is in use in the existing networks. In digital transmission elements, Time Division Multiplex (TDM) systems use 64 kbit/s Pulse Code Modulation (PCM) of ITU-T Recommendation G.711, or one of the more recently introduced compression techniques based on low bit-rate codecs. Again, a major influence to the transmission quality of these systems is caused by the transmission time. Modern equipment using special coding algorithms to reduce the bit-rate of each communication channel - for example, Digital Circuit Multiplication Equipment (DCME) - contribute additional distortion and delay.

4.5 Types of Connections

For some private networks, impairment allocation may be possible by taking into account the mix of connection types. For example, 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 private network, 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 about 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 private 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 private network planning. The following figures should be considered only as examples, used mainly for definitions in the present document. One main task in planning is to identify the type of interconnection between the private network and other - mainly public - networks. Figure 1 shows a basic configuration, assuming a digital interface between the networks.

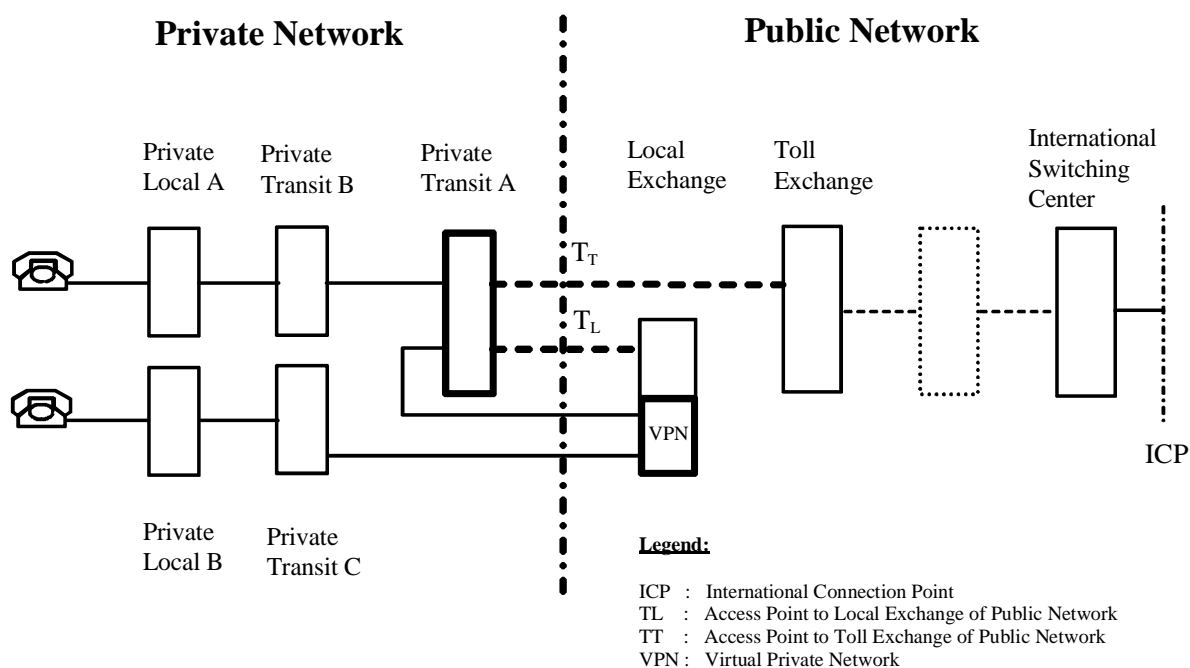


Figure 1: Basic Configuration for the Interconnection between Private and Public Network

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 the 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 to 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 likely contributes a minimum of transmission impairments to the overall connection.

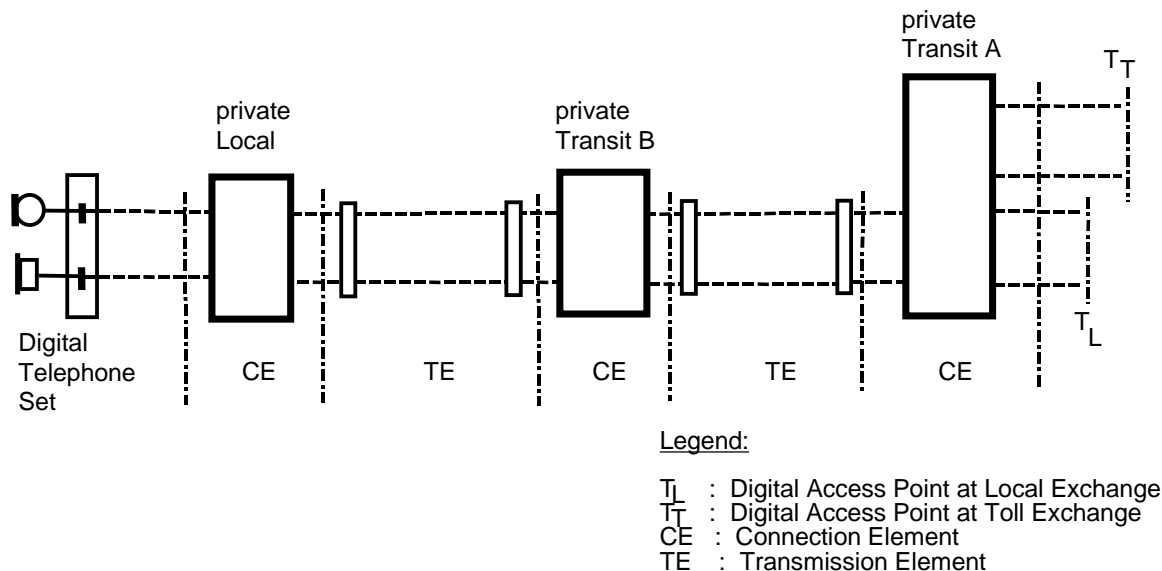


Figure 2: Standard Configuration with a fully Digital Routing within the Private Network

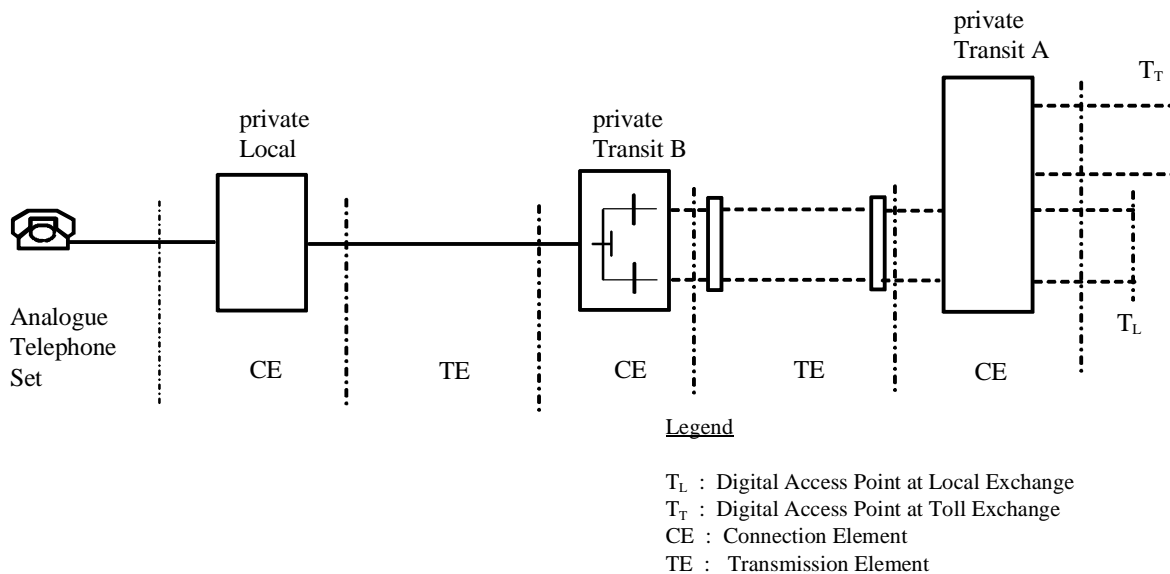


Figure 3: Private Network with 4-wire/2-wire Conversion

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. If this hybrid also forms the termination of a long national or international 4-wire connection, it could influence the stability of the connection.

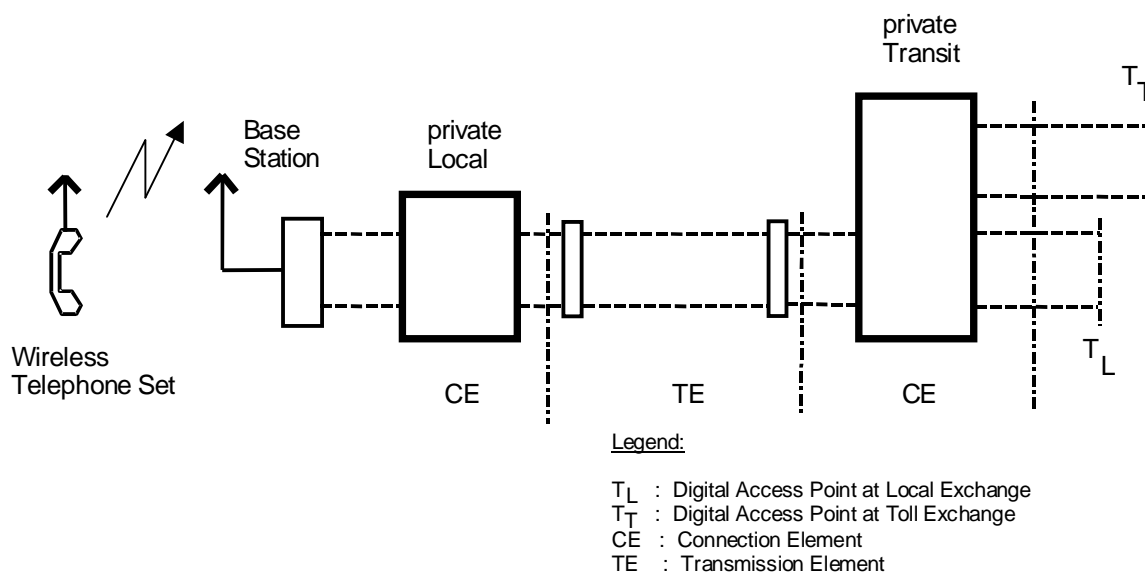


Figure 4: Digitally connected Wireless Telephone Set

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.

6 Basic Planning Principles

As stated in the introduction of the present document, the rapidly changing scenario in the field of 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 particular, with respect to the allocation of impairments between public and private networks.

In general, the users on both ends of a connection base the quality of speech transmission via telephone channels on a subjective judgement. Therefore, transmission planning is, in principle, derived from an end-to-end consideration; however, structured on the allocation of all relevant parameters between networks or, where applicable, elements within a network. Historically, this method resulted in the allocation of impairments with limits for the private network between the acoustical interface of the telephone set and an electrical interface to the public network set to be compatible with the public network transmission plan. For the simple private networks of the past, this allocation guaranteed that a sufficient quality was provided for nearly all calls (local, national, and international) even in worst case scenarios. 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 deregulation, this principle is no longer valid. The responsibility for sufficient voice transmission quality is now shifted to the planner and operator of the private network. However, planning of private networks with respect to voice transmission quality requires knowledge and experience in the field of transmission parameters and their influence on voice quality. **Therefore, it is pertinent to provide a planning method that is easy to handle, accompanied by all the necessary (tutorial) information and planning tools. This is the main intent of the present document.**

To meet the main goals of the present document, such as flexibility in planning, flexible impairment allocation, and the use of modern technology, the basic planning principle described in the present document deviates considerably from previous planning methods for private networks. For all configurations, the planning of speech transmission quality is based on an end-to-end consideration. Furthermore, the design criteria and results of planning calculations are no longer referred to in terms of units of the various impairment parameters, but to issues of perceived quality in terms of MOS, GoB or PoW.

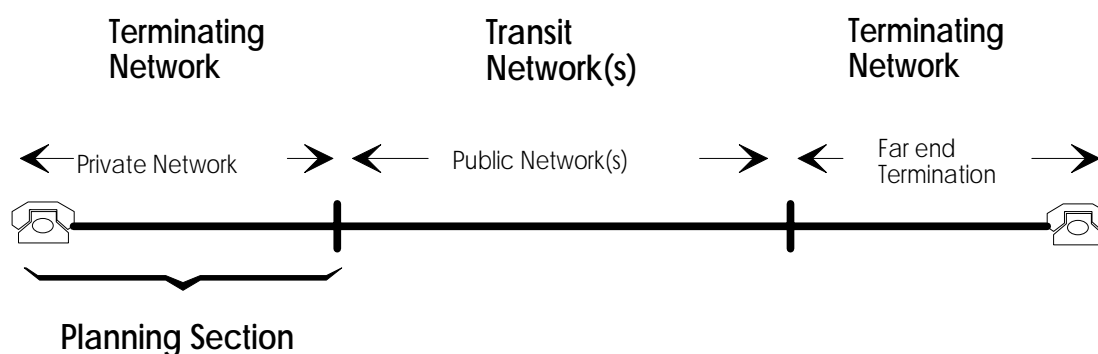


Figure 5: General Configuration for Calls via Public Networks

This principle is applied not only for internal calls within the private network, but also for calls via public networks. For calls via public networks, illustrated in figure 5, application of the "end-to-end principle" addresses parts of a connection which are not directly subject to planning by the private 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 the present document 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.

The inclusion of subjective judgements by means of quality issues requires planning rules and tools in form of a calculation algorithm simulating the subjective judgement of the users in a given configuration. Such a planning method, called the "Impairment Factor Method" is recommended and described in the revised ITU-T Recommendation G.113. The necessary planning tool, the so called E-Model, as introduced and described in ETR 250 is based on this Impairment Factor Method. For more information about this method and the E-Model, both of which are used in the present document, see the following subclauses.

6.1 The Impairment Factor Method

Voice transmission quality in telephony is influenced by several parameters such as loss, distortion, delay, echo, noise etc., each contributing to a specific amount of impairment and thereby decreasing the voice quality as perceived by users. As long as only one of these parameters impacts the quality, an upper limit or a permitted range in units of this parameter, with a relation to a "Quality Scale" based on subjective tests, can be defined. In most practical applications, several impairment parameters are present and interacting, therefore this simple way is not applicable. Thus, a planning method based on impairment contributions has been developed. This "Impairment Factor Method" is recommended by ITU-T and described in ITU-T Recommendation G.113. It is also used as the basic planning method in the present document.

The Impairment Factor Method is based on the principle, that transmission impairments can be transformed into "Psychological Factors" and that these factors are additive on a "Psychological Scale". For planning purposes, impairment values of the major transmission parameters for the different network elements, derived from subjective tests are added resulting in a Total Impairment Value as shown in figure 6.

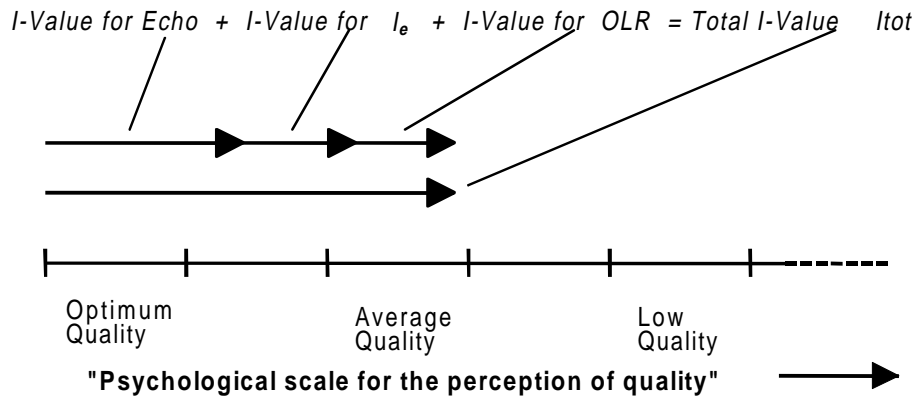


Figure 6: Example for the Addition of Impairments on a "Psychological Scale"

Impairment values are expressed in numbers, where $I = 0$ means no impairment or optimum quality. The different impairment values are described in subclause 6.2 and annex B. Using this method also for the planning of private networks, the calculation of each investigated end-to-end configuration will result in a specific I_{tot} - value. The relation between the value of I_{tot} and the perceived subjective quality level is shown in table 1a. This description of the speech communication quality is taken from ITU-T Recommendation G.113. A similar table is contained in ETR 250 but related to the "Rating Factor R" instead of Total impairment value I_{tot} .

ITU-T Recommendation G.175 contains a similar table, shown here as table 1b, also related to the Rating Factor R, but based on an assessment related to "User's Satisfaction". Both tables may be used as a guide for planning purposes, with table 1a being the preferred one.

Table 1a: Relation between Communication Quality and Total Impairment Value I_{tot} (G.113)

I_{tot} Upper limit	MOS	GoB	PoW	Speech Communication Quality
5	4,32	96,6	~0	Very Good
10	4,17	93,5	~0	Good
20	3,79	81,3	3,4	Adequate
30	3,32	60,5	11,4	Limiting Case
45	2,54	25,1	39,5	Exceptional Limiting Case
55	2,03	9,7	64,0	Customers likely to react strongly

Table 1b: Relation between Rating Factor "R" and User's Satisfaction (G.175)

R-Value Lower limit	MOS	GoB	PoW	User's satisfaction
90	4,34	97	~0	Very satisfied
80	4,03	89	~0	Satisfied
70	3,60	73	6	Some users dissatisfied
60	3,10	50	17	Many users dissatisfied
50	2,58	27	38	Exceptional Limiting Case

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, computation 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 only bibliographic status. In the present document, the E-Model is used; this model was developed by an ETSI ad hoc Group and was published in the ETR 250 "Speech Communication Quality from Mouth to Ear of 3,1 kHz handset Telephony across Networks". The E-Model is also contained in ITU-T Recommendation G.107. Detailed information about the E-Model is given in subclause 6.2, clause 9 and annex B.

- Pr Room Noise at the Receive-side;
- **A Advantage Factor**
- *) No direct input value; calculated as $OLR=SLR+RLR$
- **) 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 annex B. 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, the present document focuses mainly 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 7.8 describe those parameters which, in a mainly digital environment, should be subject to planning; subclause 7.9 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 8 to 10 dB. Connections between two digital telephone sets, i.e., designed according to TBR 8 with $SLR = 7$ dB and $RLR = 3$ dB, or to TIA 579 with $SLR = 8$ dB and $RLR = 2$ dB, and routed over a fully digital connection, will meet this optimum value for OLR (10 dB). The relation between the total impairment value I_{tot} and the OLR of a connection is shown in figure 7. This curve, calculated with the E-Model, is obtained when all other input parameters of the E-Model are set to their optimum values (default values, see subclause 9.7); i.e., OLR is the only impairment in the connection considered.

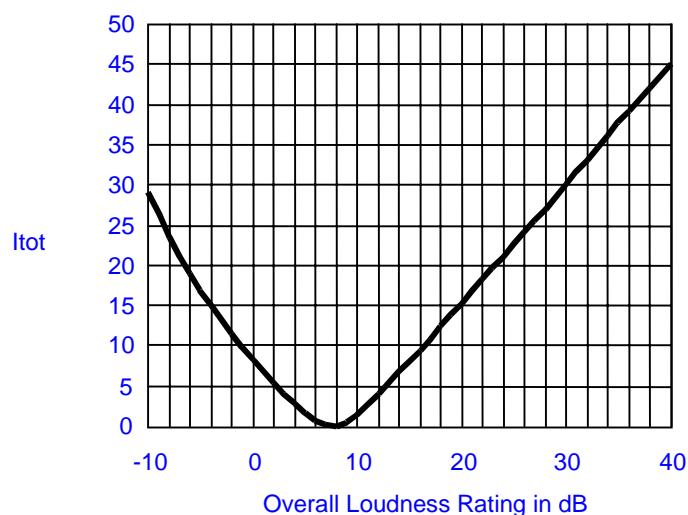


Figure 7: Relation between OLR and Total Impairment Value Itot

Comparing the curve in figure 7 with the description of the perceived quality level in table 1a, the "limiting case" for speech quality lies within a range for the OLR of -10 dB to 30 dB. 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 level of quality between "good" and "adequate" - 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 in internal connections within the private network between two analogue telephone sets, or if digital telephone sets with lower values for SLR and/or RLR than stipulated in TBR 8 or in TIA-579 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 where signal reflections together with transmission delay will cause talker echo impairments which must be taken into account. Signal reflections will mainly occur at hybrids, where the bi-directional 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; generally, one echo path is dominant. Delay contributions include not only the propagation time along the different transmission media but, for digital elements, also delays due to digital signal processing and low bit-rate codecs, e.g., in multiplexing equipment and radio sections.

The effects of echo in a conversation can cause impairments to the talker as well as to the listener. These impairments are expressed, respectively, as Talker Echo and Listener Echo. More information about these effects is given in annex A. 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.

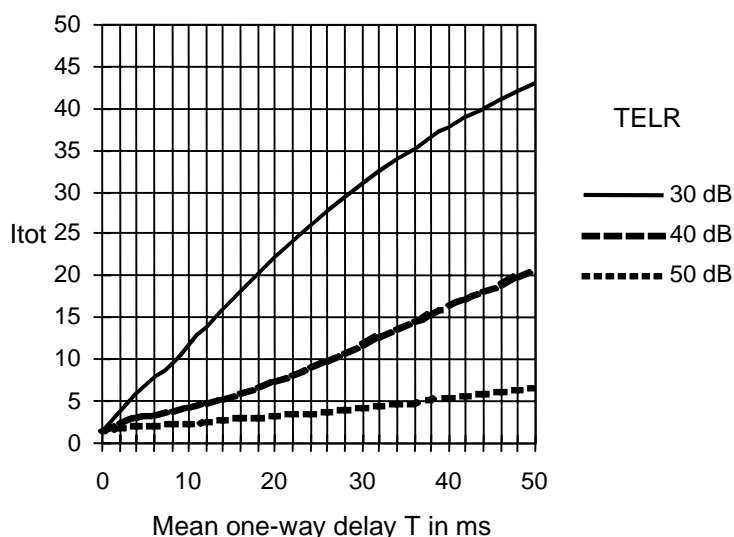


Figure 8: Total Impairment Value I_{tot} due to Talker Echo

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. EL includes losses in the bi-directional transmission path and the TBRL (see subclause C.1.3) at the 4-wire/2-wire conversion point.

$$\text{TELR} = \text{SLR} + \text{EL} + \text{RLR}$$

The relation between mean one-way delay, TELR and the resulting total impairment value I_{tot} is shown in figure 8 for three different values of TELR. The curves are calculated with all other parameters at their optimum (default) values.

Talker echo is becoming one of the more important parameters in modern mixed digital / analogue networks. To reduce the impairments caused by echo effects, the factors contributing to talker echo should be considered carefully in transmission planning. Since transmission delay in a connection usually cannot be reduced, improvements in perceived quality can only be achieved through higher values of. This can be achieved either - or both - by improvements in the 4-wire/2-wire conversion so as to increase the TBRL value, 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 2 dB). When the loss needed 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 achieve a quality level between "good" and "adequate" with an I_{tot} of approx. 15 (see table 1a), figure 8 shows that, with TELR = 30 dB, the mean one-way delay is limited to $T = 13$ ms, while with TELR = 40 dB a value of $T = 36$ 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 ms 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. If additional delay impairments, e.g., associated with low bit-rate codecs, need to be considered, this echo path delay range should be decreased accordingly.

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 (e.g., < 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 subclauses 8.2 and 9.5.

7.3 Transmission Time in Echofree Connections

In international or intercontinental connections carried via satellite links, additional impairments may arise due to very long delay, even if perfect echo cancelling is provided. This 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, T_a , 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 resulting total impairment value, I_{tot} , is shown in figure 9 for a range of $T_a = 100$ ms to 600 ms, with all other parameters at their optimum values.

With respect to a quality level between "good" and "adequate" as described in table 1a, the limit value for most connections should be in the range of 300 ms to 350 ms with an upper limit of 400 ms. This is also in accordance with ITU-T Recommendation G.114 and should be exceeded in only exceptional cases.

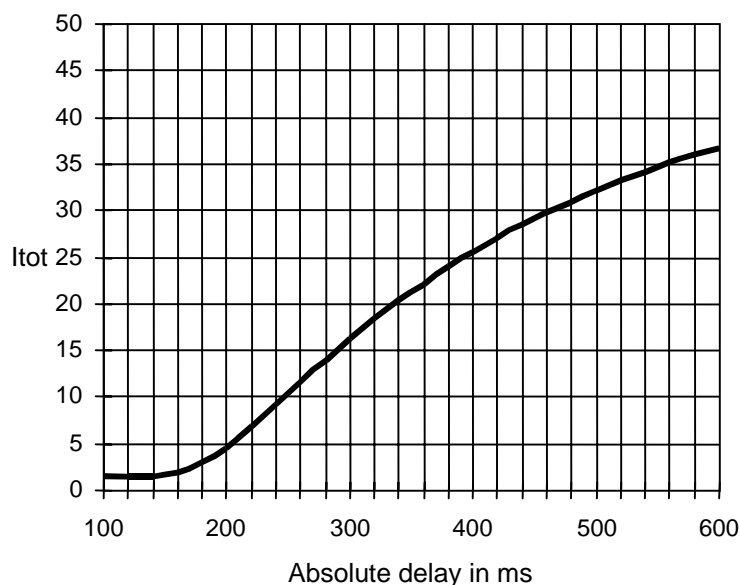


Figure 9: Relation between Absolute Delay T_a and the Total Impairment Value I_{tot}

7.4 Stability

The stability of a connection should be considered in transmission planning for any connection, where the private network contains a 4-wire loop or where a 2-wire/4-wire conversion (hybrid) within the private network 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 annex A.

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 subclause 7.9.

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 ITU-T Recommendation G.711, 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 ITU-T Recommendation G.711.

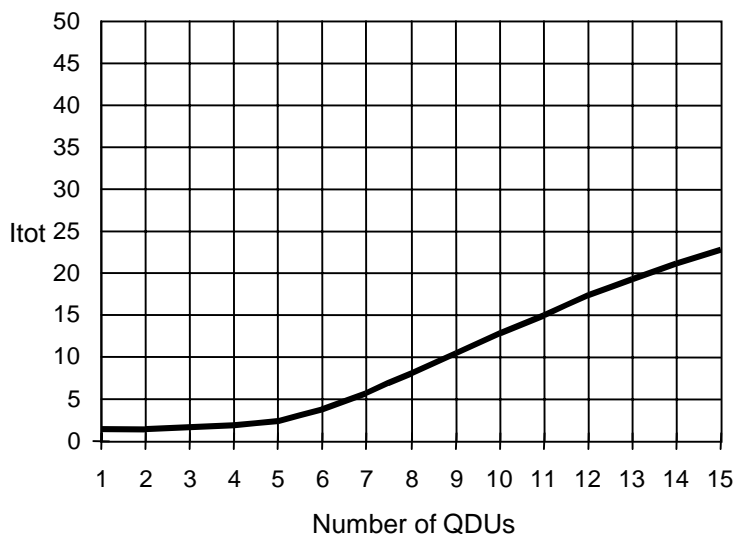


Figure 10: Relation between the Number of qdu and the Total Impairment Value Itot

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 total impairment value Itot is shown in figure 10.

The curve in figure 10, derived from the E-Model with all other parameters at their optimum (default) values, shows that the impairments are negligible for an aggregate distortion of up to 4-5 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 (Itot) 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 ITU-T Recommendations G.726, G.727 or G.728), the parameter qdu is, for transmission planning, replaced by the equipment impairment factor, Ie (see following subclause).

7.6 Equipment Impairment Factor

Modern coding laws, such as associated with low bit-rate codecs as described in, e.g., ITU-T Recommendations G.720-series or the GSM Standard, as well as ADPCM with different operating bit-rates, will contribute distortions resulting in a decrease of the perceived voice. In contrast to the quantization distortion due to the standard 8 bit 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 tests as well as from network experience. The results of these subjective mean opinion scores are transformed into a value, Ie, which follows the basic planning principle described in subclause 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 planning values for Ie for several codecs, taken from ITU-T Recommendation G.107, are listed in table 2. For information on the values referred to as provisionally in table 2 and on values for other codec types, appendix I to ITU-T Recommendation G.113 may provide guidance (appendix I / G.113 is intended to be updated regularly).

Table 2: 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	6
QCELP	IS-96a	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

7.7 Advantage Factor A

The "Advantage Factor ", A, represents an "Advantage of Access", introduced into transmission planning for the first time via the E-Model (ETR 250 and ITU-T Recommendations G.113 and G.175). 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 is expressed in units of impairments so that they can be used directly in conjunction with all other impairment values and as an input parameter to the E-Model. Provisional A values, taken from ITU-T Recommendation G.175 are listed in table 3.

Table 3: Provisional Examples for the Advantage Factor, A

Communication System Example	Maximum Value of A
Conventional (wired)	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 another degree than 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.

7.8 Limits at the Public / Private Interface with respect to Echo

Transmission planning for private networks, as described in the present document 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 the present document 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.

7.8.1 Public / Private Network Interfaces - Europe

The guidelines are derived from ETS 300 283, where limits are given for the so-called "Network Connection Point", NCP, to control echo in national calls without echo cancellers inserted. Summarizing the content of ETS 300 283, 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 subclause 4.1 of ETS 300 283). 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 a total impairment value $I_{tot} = 17$ (see figure 8), which is judged between "good" and "adequate" according to table 1a.

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 Europe, 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.

7.8.2 Public / Private Network Interfaces - North America

In North America, guidelines for the interface between a public and a private network are given in T1.508, 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 ANSI/EIA/TIA-470B). 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: 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)	$\mu = 14$	$\sigma = 3$

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 μ f), and do not include any network losses.

Digital access lines are assumed to be terminated with digital telephone sets conforming to specifications in ANSI/EIA/TIA-579 as provided in table 5, or other digital terminal equipment having equivalent characteristics.

Table 5: 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 (note)	> 45	≥ 40

NOTE: TCL_w – Weighted Terminal Coupling Loss.

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 TIA-464B.

T1.508 recommends that any new system, network or component that, by itself, adds more than 5 ms round trip delay should provide echo cancellation. Otherwise, degradation of transmission quality would occur. The rationale for the present document line 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. This 5 ms of round trip delay is derived from a widely used criterion of 4 % decrease in "Good or Better" customer opinion rating using the E-Model. (It should be observed that the North American incremental 5-ms delay limit is round-trip, whereas for Europe, the limit is one-way.)

7.9 Parameters not directly subject to Transmission Planning

As mentioned in 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.

In analogue transmission, impairment due to the *frequency response* of unloaded cable sections and other connection elements needed to be controlled. For the purposes of the present document, 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.

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., TIA-464B, ETS 300 439 or ES 201 168 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, N_c, with a value referred to the 0 dBr-point. For further information, see clause 9.

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; i.e., the parameters WEPL and Tr in the E-Model can remain at their default values for calculation.

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. For analogue telephone sets, STMR, and to some extent LSTR, depends 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 TIA-464B, ETS 300 439 or ES 201 168 and to clauses A.1 and A.4 in ETR 250.

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 influence on speech quality. The E-Model includes room noise, separately for the send and receive sides as a source for impairments. In a normal office environment as can be assumed in the business domain of private networks, the room noise can be expected in a range of 30 dB(A) 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 Synchronization

A proper synchronization design is part of the network planning strategy, because poor synchronization will affect the speech quality of the communication. Relevant documents are ETS 300 462-1/6, ISO/IEC 11573, and annex F of TIA-464B.

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 ITU-T Recommendation G.165, note 2 of subclause 3.2). 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 the present document and by the current version of the E-Model.

8 Calculation of End-to-End Parameters

As described in more detail in clause 9, 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 involving the private network 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 the present document - 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/sec frame and in an 8 bit A-law or μ -law form. In most of the network applications described in the present document, the digital interface between the private network and a public network will serve as the 0 dBr-point.

The principles of precalculation are demonstrated with the configuration shown in figure 11 (see note). 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.

NOTE: The values shown in figure 11 are for illustration only and do not necessarily reflect actual circuit or loudness values in different regions.

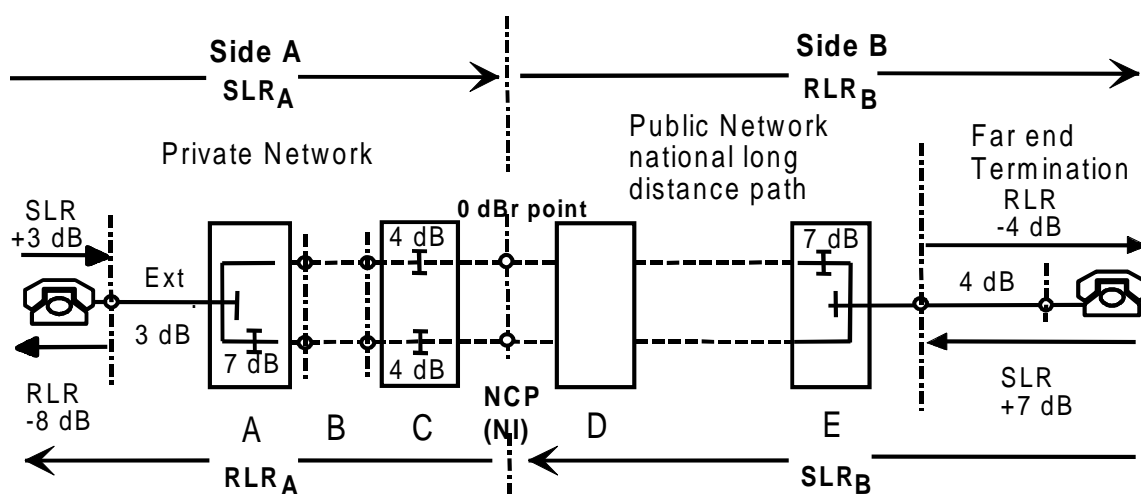


Figure 11: Reference Connection for the Calculation of SLR and RLR

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		SLR_A = 10 dB	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		SLR_B = 7 dB	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 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:

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

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 total impairment value for this direction (perceived quality of the private network customer) is calculated with $I_{tot} = 27,4$.

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 P_s is set to 65 dB(A) and P_r remains at the default value. The total impairment value in this direction is $I_{tot} = 37,2$, 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 subclause 7.8. For the example of figure 11, 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 subclause 7.8.1 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 annex A.

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 annex B. 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.

It should also be noted, 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 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 12. 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.

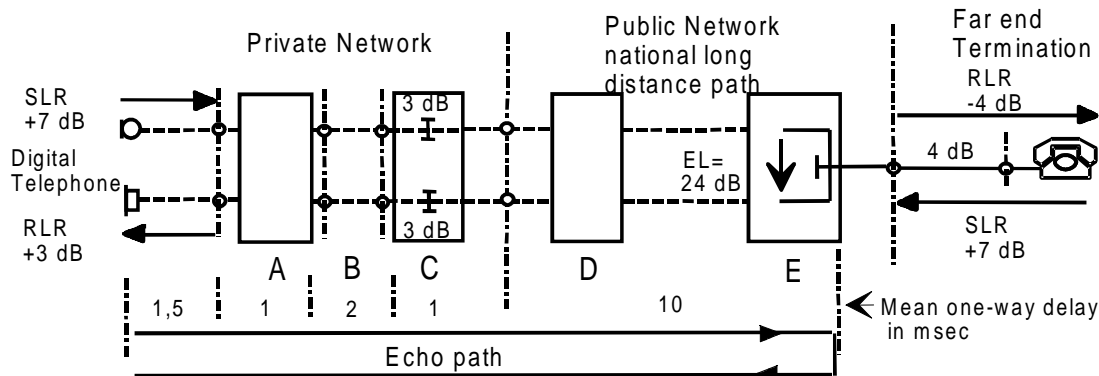


Figure 12: Reference Connection for Echo Calculations with one Echo Path

A clear identification of the echo path must be the first step of every calculation. For the example illustrated in figure 12, 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 12.

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 12 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 12 yields a total impairment value $I_{tot} = 9,9$, a value which is judged as "good" according to table 1a.

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. When possible, such configuration should be avoided in network design. The following example, derived from the configuration of figure 12, is based on the use of a cordless telephone, connected to an analogue port on PBX A. For illustration, the cordless telephone is one conforming to TBR 10 and ETS 300 175-8. 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 12). This reference connection with its two effective echo paths is illustrated in figure 13. In such a configuration the talker may be disturbed by two different echoes with different volume and different delay.

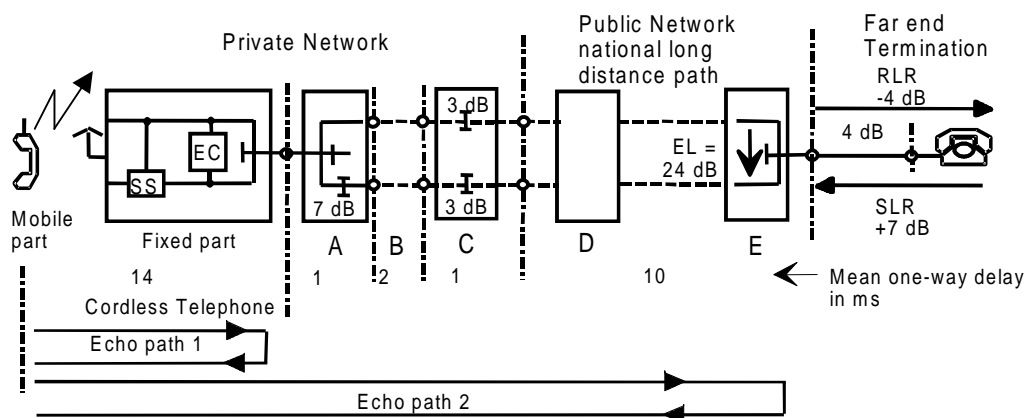


Figure 13: Reference Connection for Echo Calculations with two Echo Paths

To determine the expected overall quality, a special application of the E-Model needs to be applied here, because the E-Model does not currently accommodate and calculate two simultaneous impairments of the same type but with different impairment amount. 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 consider 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., I_e-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 subclause 9.5, together with general guidance for the application of the E-Model in planning practice. The following paragraphs describe only how to determine the parameters TELR and T for the two echo paths.

For the cordless telephone in figure 13, 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 - in this case, DECT - should be taken into account.

Cordless telephones conforming to TBR 10 and ETS 300 175-8 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 C.1, annex C). This amount of delay is likely to cause echo effects in many applications. Therefore, the DECT Standard 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 DECT cordless telephones is in annex A, subclause A.5.3.3.

When determining the TELR of this echo path 1, these "integrated" echo control devices must be taken into account. Figure 14 shows in more detail the relevant components for echo control in a DECT termination, 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.

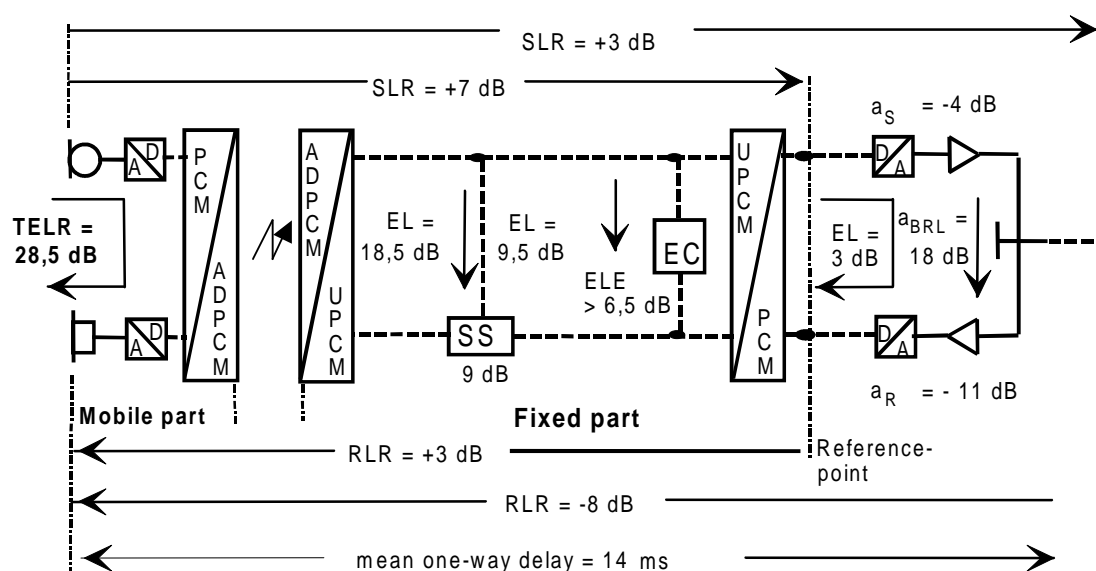


Figure 14: Details inside a Cordless Telephone with respect to Echo Control

Per TBR 10 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 14) not only provides the necessary A/D-D/A and 4-wire/2-wire conversion, but also includes, in its analogue part, an adjustment to $a_S = -4$ 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 dB 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 14 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 14 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 subclause 8.2.1 but with different values. Summing the values for delay for each network element along echo path 2, as shown in figure 13, 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 14). 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 14 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, $RLR = (-8 + 9) \text{ dB} = +1 \text{ dB}$. Thus, the sum for echo path 2, using the values as shown in figures 13 and 14, is $TELR = (3 + 3 + 24 + 3 + 7 + 1) \text{ dB} = 41 \text{ dB}$. Note that the echo canceller in the fixed part does not affect echo path 2 since the processing capability of the echo canceller with respect to the echo path delay is much less than the connection delay (generally about 4 ms).

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 subclause 7.3, a very long delay may cause impairments other than those due to echo. Connections subject to significant delay can be assumed to be designed to be "echofree" with the added assumption of 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. For the use in the E-Model, the absolute delay, T_a , in ms should be calculated as the sum of all 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 subclause 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 ITU-T Recommendation G.711 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 ITU-T Recommendation G.711 (e.g., ADPCM), the impairments due to distortion should not be expressed in terms of qdu, but rather as the equivalent Equipment Impairment Factor, I_e (see next subclause). 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.

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 subclause 7.6, modern coding laws will cause impairments due to distortions. In contrast to the standard PCM coding and decoding according to the A-law or μ -law (ITU-T Recommendation G.711), these impairments are expressed in terms of "Equipment Impairment Factor", I_e , instead of in terms of quantization distortion units (qdu). The I_e factors for different coding laws and operating rates are given in table 2. For planning practice, only the algebraic sum of all I_e 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.

9 Application of the E-Model in Planning Practice

9.1 General

The basic planning principles recommended in the present document are based on the use of the E-Model (see clause 6) for performing planning calculations which provide an evaluation 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 computation model are given in this subclause.

A detailed description of the E-Model and its associated algorithm is contained in annex B. 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 subclauses 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 contained in subclause 9.6.

The E-Model includes a variety of transmission parameters; however, not all of them are varied for planning purposes (see subclause 7.9). 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 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 15, 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 15 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.

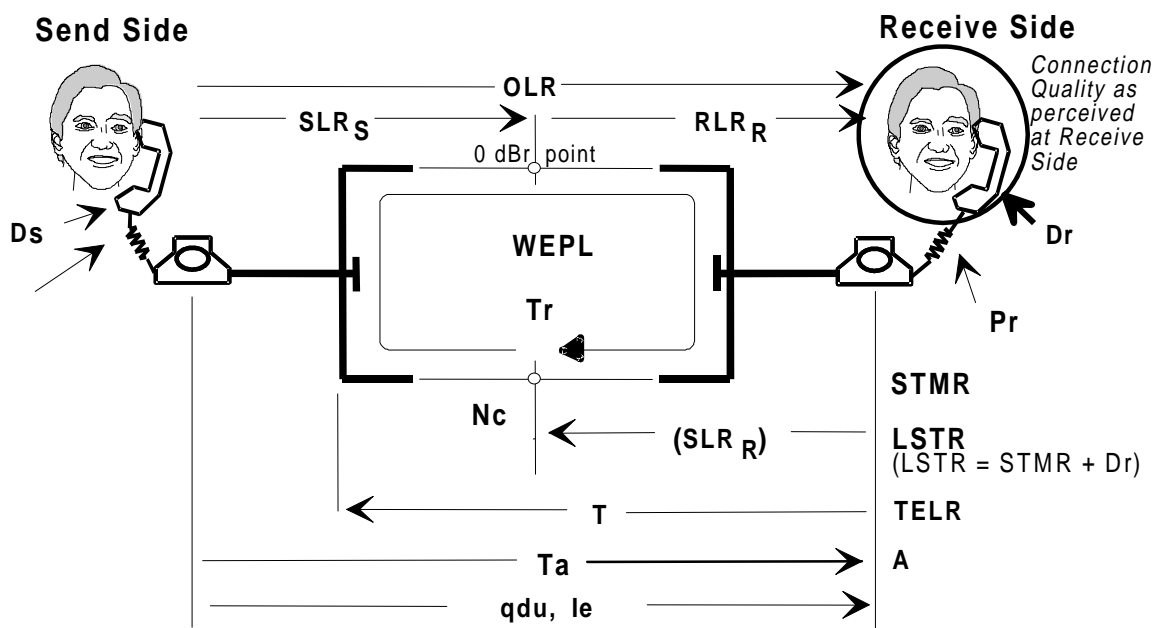


Figure 15: Basic Reference Configuration of the E-Model

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 precalculations - 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 15.

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 16, 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 center 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 D_s at the Send Side and D_r 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

$$LSTR = STMR + D$$

and thus influence the perceived quality only at the Receive Side.

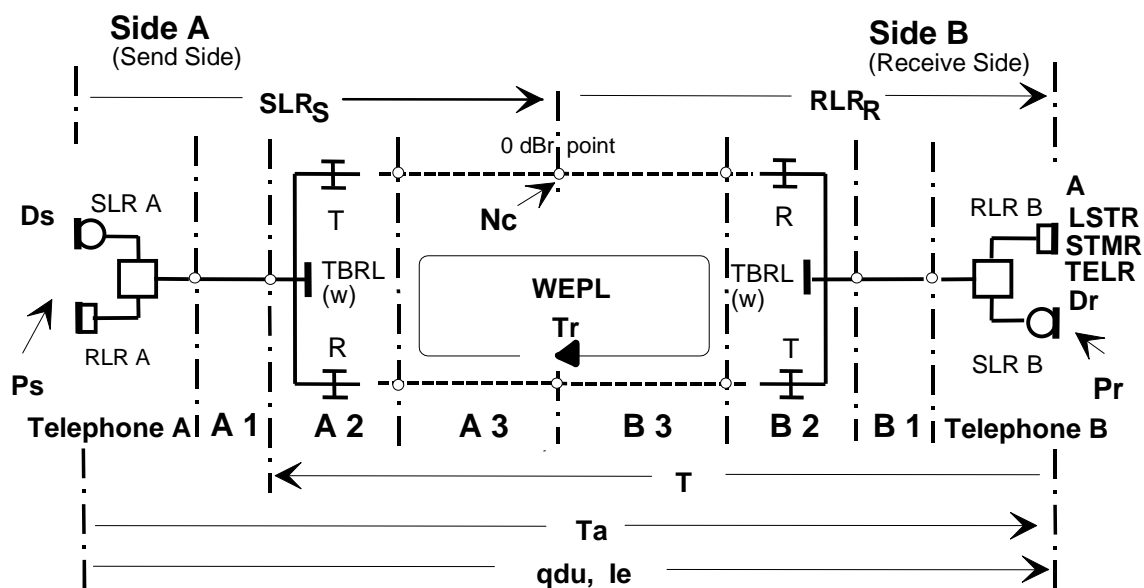


Figure 16: Working Configuration for 2-wire / 2-wire Connections.

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, P_s , at the Send Side may influence the signal to noise ratio as perceived at the Receive Side, while the room noise, P_r , 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 subclause 9.3.

Sections A1 and B1 of figure 16 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 16 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 precalculations of TELR.

Finally, sections A3 and B3 in figure 16 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 dB-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 precalculation, 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, T_r . T_r 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.

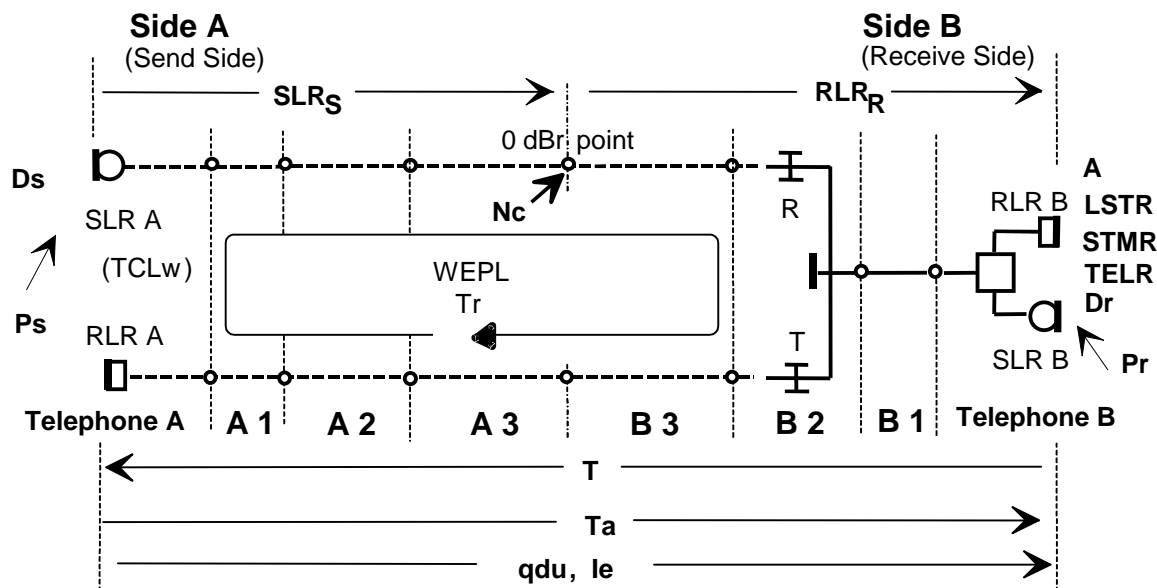


Figure 17: Working Configuration for 4-wire / 2-wire Connections

The working configuration shown in figure 17 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 and electrical coupling paths) from which a possible source for signal reflection may arise (see also annex A, subclause 5.2.3).

TCLw - which is not a direct input value to the E-Model - should be included in the precalculation of the TELR value to define impairments due to echo perceived at the Receive Side. TCLw should also be part of the precalculations 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 A.2 of figure 16. It should be noted that the mean one-way delay, T, in figure 17 includes section A1 and the telephone, in contrast to the configuration in figure 16 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 16 are also valid for this configuration.

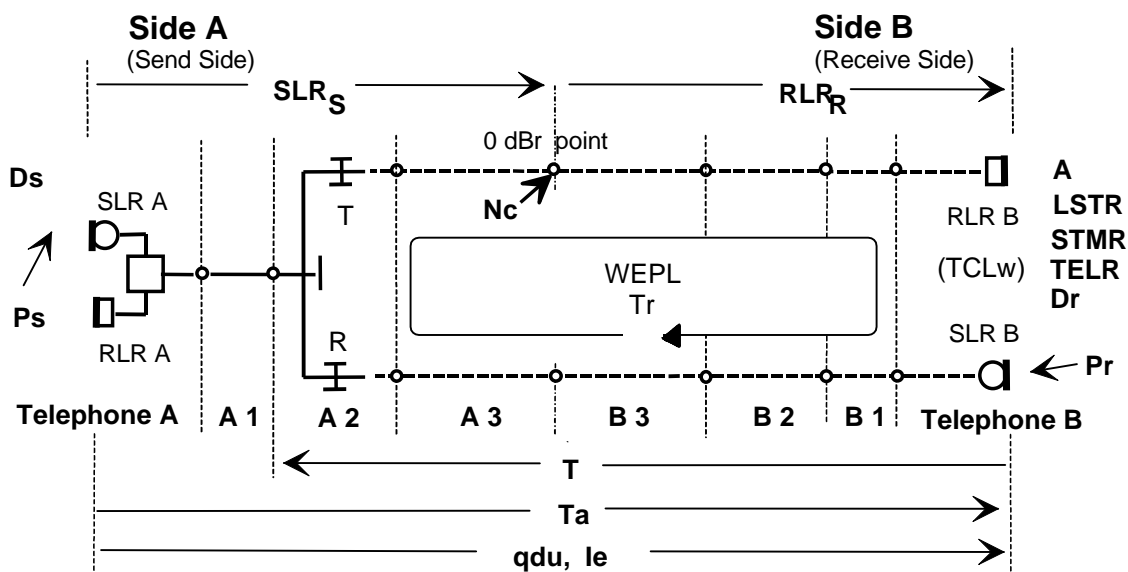


Figure 18: Working Configuration for 2-wire / 4-wire Connections

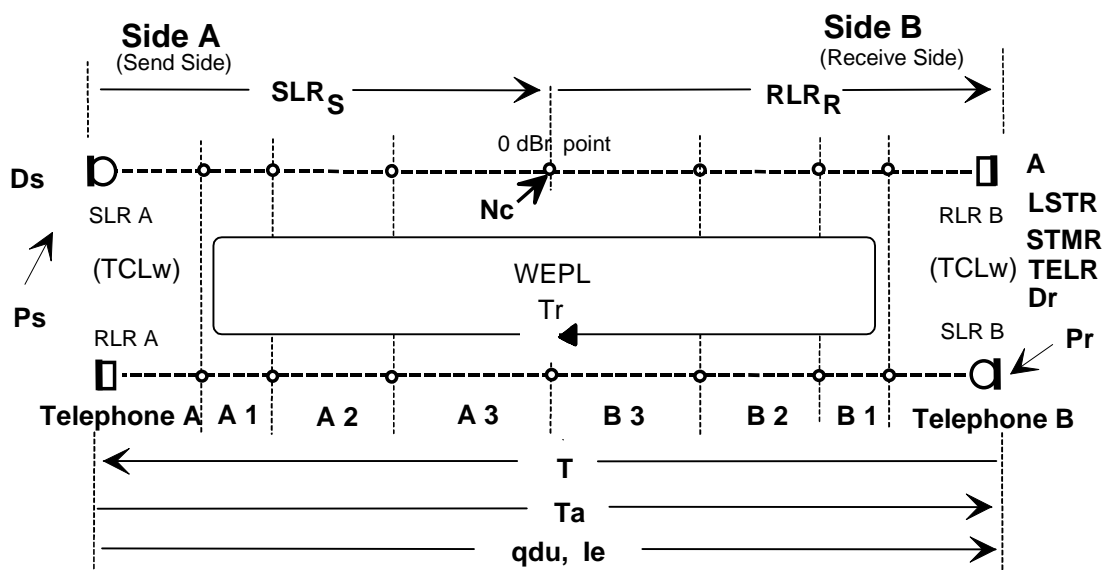


Figure 19: Working Configuration for fully digital Connections

Figure 18 illustrates the opposite working configuration with the 4-wire termination (digital telephone set) at the Receive Side. Figure 19 illustrates a fully digital connection with 4-wire termination on both ends. The application and precalculation of the input parameters in figure 18, especially for TELR, is nearly identical with the configuration in figure 16 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 19 should be included in the precalculation for WEPL. It should be noted, that WEPL can be left at its default value - i.e., can be neglected in most cases -, if digital telephone sets according to applicable European or North American standards (TBR 8, TIA-579A) with a TCLw of more than 40 dB are used and if the round trip delay, T_r , 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 dB-point. With respect to figures 16 to 19, 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 ITU-T Recommendation P.79.

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 1: For telephone sets with volume control in receive direction in the handset mode, the RLR for the default setting of the volume control and the corresponding values for STMR and LSTR, as stated by the manufacturer, and 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 subclause 8.2, the value for TELR is obtained in a precalculation according to the basic formula:

$$\text{TELR} = \text{SLR} + \text{EL} + \text{RLR}$$

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 16 through 19. 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 precalculated 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 16 and 18 and additionally in section A1 of figures 17 and 19. 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 16 and 17 is included twice within the echo path. For section A2 of figures 16 and 18, 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 17 and 19. It is absolutely necessary to use the "weighted" value for the TBRL.

The definition of the Circuit Noise parameter, N_c , can lead to wrong results if this input value to the E-Model is not specified 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 16, 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 precalculated into an equivalent value at the 0 dBr-point.

NOTE 2: Measured longitudinal interference into a cable in section A1 of the working configuration in figure 16 will result in a transversal noise level of -50 dBm0p 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 dBm0p - 3 dB = -53 dBm0p at the 0 dBr-point.

9.4 Interpretation of the Results

As described in clauses 6 and 7 in detail, the present document and the recommended planning principles are based on the Equipment Impairment Factor Method. The result of any planning investigation with respect to the perceived quality is presented with the Total Impairment Value I_{tot} as the final result. The E-Model and its algorithm however, as described in detail in annex B, does not provide this Total Impairment Value directly, but in relation to an "Overall Rating R". Therefore, a transformation from the "R-value" into an "Itot - value" should be performed before judging the result.

Assuming a value of $I_{tot} = 0$ is related to an optimum quality, this relation can be found if all input parameters of the E-Model are set to the default values as listed in table 6 (see subclause 9.7) which characterize the optimum value for each parameter. The result for the R-value in this case is $R = 94,3$. The transformation can be performed using the equation:

$$I_{tot} = 94,3 - R$$

NOTE 1: The recommended default values (see subclause 9.7) use the values SLR = 7 dB and RLR = 3 dB according to TBR 8 for digital telephone sets. The resulting sum of SLR + RLR = 10 dB deviates slightly from the optimum value of 8 dB as derived from subjective tests and which is the basis for the equations in the E-Model. Using values of SLR = 6 dB and RLR = 2 dB the optimum value would be $R = 95,8$. For the benefit of more realistic values for default setting, this small deviation can be neglected in practical planning.

NOTE 2: For use in North America, the default loudness values should be in conformance to the values in TIA-579A; i.e., SLR = 8 dB and RLR = 2 dB. Using these values of SLR and RLR, the optimum value would be $R = 94,1$.

As stated in subclause 6.2, provisions should be made that, in addition to the R-Value or the equivalent value for I_{tot} , the specific results for I_d , I_s and I_e are available after a calculation run. This is useful to recognize and to identify the different amounts each category of impairment contributes to I_{tot} when investigating solutions towards an improved quality.

9.5 Application of the E-Model for Configurations with two Echo Paths

As mentioned in subclause 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 the same type, 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 13, 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.

As a first step, the impairments due only to echo are calculated. If I_{dte_1} and I_{dte_2} are the two impairments due only to the echo of the two echo paths, these two values are calculated with the E-Model. This calculation is performed separately for each echo path, assuming that all other parameters are set to their default values (optimum situation for all other parameters) and only values for T and TELR are distinctly specified as input values to the E-Model. The two results for I_{dte_1} and I_{dte_2} are then combined to a total (sum-) value I_{dte} for both impairments using the following equation

$$I_{dte} = \sqrt{I_{dte_1}^2 + I_{dte_2}^2}$$

This value of I_{dte} now represents all impairments due to echo effects.

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 result I_{tot}' of this second calculation is then combined with the impairment value for the two echo paths I_{dte} by a simple addition.

$$I_{tot} = I_{tot}' + I_{dte}$$

For the actual calculation of this configuration with two effective echo paths, a decision about the working configuration (see subclause 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 13 will now be used to illustrate the above principles. To begin with, consider parameters which can be set to default.

As explained in subclause 7.3, the absolute one-way delay T_a causes major impairments only when the delay exceeds 150 ms (ITU-T Recommendation G.114). The algorithm of the E-Model sets the corresponding impairment to $I_{dd} = 0$ for $T_a < 100$ ms (see annex B, subclause B.3.3), i.e., for the configuration of figure 13 T_a can be left at its default value of 0 ms for the calculation.

When the configuration in figure 13 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 subclause 7.9, 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 13 result in an increase of the total impairment value I_{tot} in the range of 0,7 to 0,8 which is negligible. For the calculation, the corresponding values for round-trip delay and weighted echo path loss can be left at their default values: $T_r = 0$ and $WEPL = 110$ dB, even if the actual values deviate slightly from these values.

The input parameters T and $TELR$ for both echo paths have already been calculated in subclause 8.2.2 as $TELR = 28,5$ dB, $T = 14$ ms for echo path 1 and $TELR = 41$ dB, $T = 28$ ms for echo path 2. Performing a calculation for echo path 1 the E-Model gives a result of $R = 77,9$. The transformation into an I-Value according to subclause 9.4 gives for echo path 1:

$$Idte_1 = 94,3 - 77,9 = 16,4$$

For echo path 2 with $TELR = 41$ dB and $T = 28$ ms the result is $R = 86,2$ and

$$Idte_2 = 94,3 - 86,2 = 8,1$$

The results for $Idte_1$ and $Idte_2$ are now combined to one value using the square root relation:

$$Idte = \sqrt{Idte_1^2 + Idte_2^2} = \sqrt{16,4^2 + 8,1^2} = 18,3$$

For the final calculation including all other impairments without echo effects, one of the working configuration presented in subclause 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 16 is selected as most appropriate.

Comparing this working configuration with the actual configuration as shown in figure 13, the telephone B at the receive side in figure 16 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. Thus, the SLR can remain at its default value during calculation. 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 14) of the complete cordless system, resulting in $RLR = 2$ dB. 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 I_e set to a value of 7 according to table 2.

The calculation for the second step can now be performed with the calculated input values for $RLR = 2$ dB and $I_e = 7$, while all other parameters, including T , T_r , T_a , $TELR$ and $WEPL$ remain at their default values. Using the E-Model the result is $R = 88,1$ or:

$$I_{tot}' = 94,3 - 88,1 = 6,2$$

The total impairment value for this configuration with two echo paths is then:

$$I_{tot} = Idte + I_{tot}' = 18,3 + 6,2 = 24,5$$

The E-Model can also be used to calculate the corresponding values for the mean opinion score with the results of $MOS = 3,6$, the percentage good or better with $GoB = 73,7$ % and the percentage poor or worse with $PoW = 5,8$ %. These results can be judged as providing medium quality.

The result of $I_{tot} = 24,5$ seems to be high, considering that configurations like the one of figure 13 are in common use (the configuration is nearly identical with a cordless telephone directly connected to a public network) without complaints by the customers. The major factor contributing to the total I_{tot} value is echo path 1 with $I_{dte1} = 16,4$, 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 of value echo loss enhancement according to, for example, TBR 10 and ETS 300 175-8 is 6,5 dB. In practice, however, higher values can be expected which, for planning calculations, should be made available by the supplier. Nevertheless, this example supports the recommendation to connect, when possible, cordless telephones to the switching equipment via a digital interface, since then only echo path 2 is effective, resulting in a total impairment value in the range of only $I_{tot} = 14...16$.

9.6 Use of Computer Programs

The E-Model, the recommended main tool for all planning purposes in the present document, comprises a number of more or less complex formulas (see annex B). Thus, the planner usually relies on computer programs for performing the needed calculations. 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 precalculations, storing of often-used configurations, etc. As described in subclause 9.3, the correct handling of Loudness Rating values in conjunction with the necessary precalculations 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 precalculation run.

Programs may also provide a structure for the input of parameter values as shown in the working configurations of figures 16 through 19. In this case, precalculations for specific sections need to be performed outside the program.

Although the recommended planning principle in the present document is based on the Total Impairment Factor, I_{tot} , or the Overall Rating, R , as the outcome of calculations, the E-Model also includes the capability to calculate the corresponding values for MOS, PoW, GoB and TME (see clause 6). Thus, computer programs may provide the results of a calculation in these different presentations. The analysis of results in MOS, PoW, GoB and TME terms 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 for the Total Impairment Factor, I_{tot} , and the corresponding guidelines given in clause 7.

When planning private networks with respect to the expected perceived speech quality, connections identified as the most critical ones should be used as reference configurations and be investigated with the E-Model. This configuration will, generally, be characterized with one specific set of input parameters, leading to a single value of I_{tot} 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 form of curves or tables. Those features can be helpful to obtain an overview about 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.
- 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, I_d , the simultaneous impairment factor, I_s , and the equipment impairment factor I_e .

The correctness of calculations according to the algorithm of the E-Model should be verified; e.g., if all input parameters are set to their default values, the result should be $R = 94,3$. An additional check, using the parameter settings and the corresponding results as given in the planning examples of annex D, 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, but every parameter has influence on the 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 standard 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 I_e value.

As stated in subclause 9.1 the algorithm 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 permitted range for each of the parameters. The parameters should be viewed as related to the basic or working reference configurations as shown in figures 15 through 19.

Table 6: Default Values and permitted Ranges for the Parameters

Parameter	Abbr.	Unit	Default Value	permitted range	Note
Send Loudness Rating	SLR_S	dB	+7	0 to +18	notes 1 and 4
Receive Loudness Rating	RLR_R	dB	+3	-5 to +14	notes 1 and 4
Sidetone Masking Rating	$STMR$	dB	15	10 to 20	note 2
Listener Sidetone Rating	$LSTR$	dB	18	13 to 23	note 2
D-Value of Telephone, Send Side	D_s	-	3	-3 to +3	
D-Value of Telephone Receive Side	D_r	-	3	-3 to +3	note 2
Talker Echo Loudness Rating	TEL_R	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	T_r	ms	0	0 to 1 000	
Absolute Delay in echofree Connections	T_a	ms	0	0 to 500	
Number of Quantization Distortion Units	qdu	-	1	1 to 14	
Equipment Impairment Factor	I_e	-	0	0 to 40	
Circuit Noise referred to 0 dBr-point	N_c	dBm0p	-70	-80 to -40	
Noise Floor at the Receive Side	N_{for}	dBmp	-64	-	note 3
Room Noise at the Send Side	P_s	dB(A)	35	35 to 85	
Room Noise at the Receive Side	P_r	dB(A)	35	35 to 85	
Expectation 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.
 NOTE 4: Default values for North American use are $SLR_S = +8$ dB, $RLR_R = +2$ dB.

10 Rules for the Insertion of Echo Cancellers

10.1 Introduction

The increasing digitalization of public and private networks has resulted in connections with not only higher values of propagation delay (e.g., processing delay), but also with nearly no loss. 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 annex A.

Echo control can be achieved in three ways. One is 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. A second method is to reduce the level of the reflected signal at the talker end through deliberate insertion of loss in 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 results in excessive volume reduction for long connections; furthermore, loss insertion in digital connection adds to the distortion impairment. The third method, discussed in this clause, is the use of echo control devices.

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 operation principle of an echo canceller is described in annex A. Previously, in mostly analogue networks, echo cancellers or echo suppressors 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 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 about 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 annex C subclause C.1.9. These parameters will be the basis for investigating if 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 ITU-T Recommendations G.168 or G.165.

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 of the private network. 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 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 subclause 7.8). 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 calculation and the actual value for the impairment factor I_{dte} or I_d (if I_{idle} and I_{dd} are close to 0). These factors are part of the algorithm of the E-Model and should be available separately as output from the calculation.

North American echo canceller guidelines are based on T1.508. The present document 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 5 ms round trip 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. This 5 ms of round trip delay is derived from a widely used criterion of 4 % decrease in "Good or Better" customer opinion rating using the E-Model.

If, in a given configuration, impairments beside echo are significant then the analysis of total impairment, I_{tot} , should be the primary consideration. For values up to $I_{tot} = 15$, a sufficient good quality can be expected: i.e., the use of echo cancellers is not necessary. For higher values of I_{tot} , the specific impairment factor I_d and its relation to the total value of I_{tot} should be considered. If I_d is the main part of I_{tot} and in a range of 20 to 25, then echo cancellers should be used. As a general rule, the insertion of echo cancellers is recommended for calculated values of $I_{tot} > 15$ and if echo is the only impairment because this will likely result in a quality improvement. If other impairments, e.g., an impairment due to low bit-rate coding, contribute significantly to I_{tot} , the impairment component of I_d can be reduced to 0 by use of echo cancellers.

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 is also subject to planning.

For delays less than 5 ms within the private network, the requirements as given in subclause 7.8 will provide the necessary echo control for the far-end termination. For higher values of delay, however - e.g., when low bit-rate equipment is used within the private network as shown in figure 20 - the use of an additional echo canceller, A - in addition to the echo canceller, B for the private network talker - may be needed to control echo for the far-end talker.

In a configuration such as shown in figure 20, 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, 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.

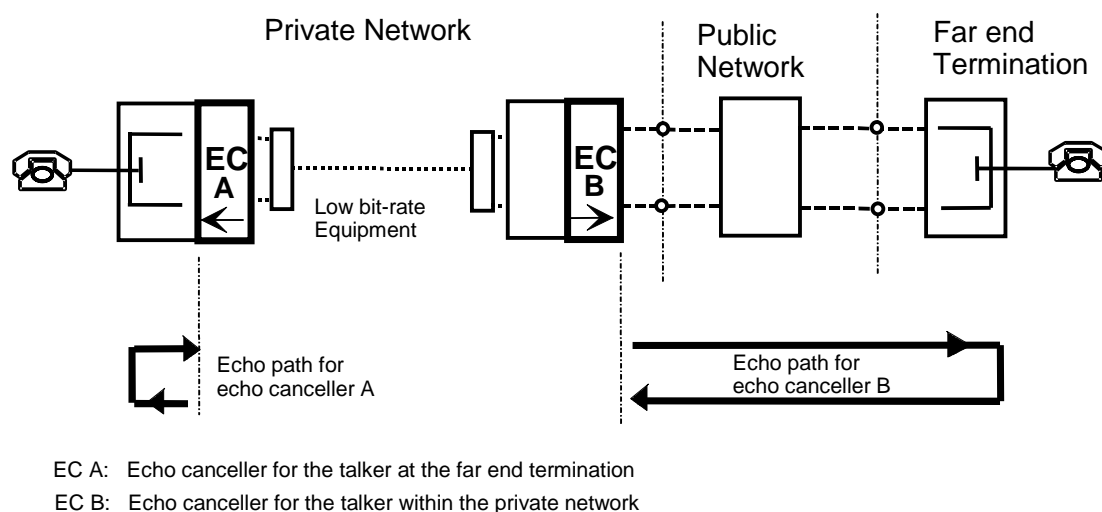


Figure 20: Application of a Pair of Echo Cancellers within the Private Network

For the selection of the appropriate echo cancellers, one of the most important properties to be investigated is the maximum echo path delay the canceller is able to compensate for (sometimes called "tail-end delay"). This value should be 6 ms to 8 ms higher than the actual total delay (twice the mean one-way delay) of the echo path. As shown in figure 20, 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, T_a , which may cause impairments resulting from excessive absolute delay.

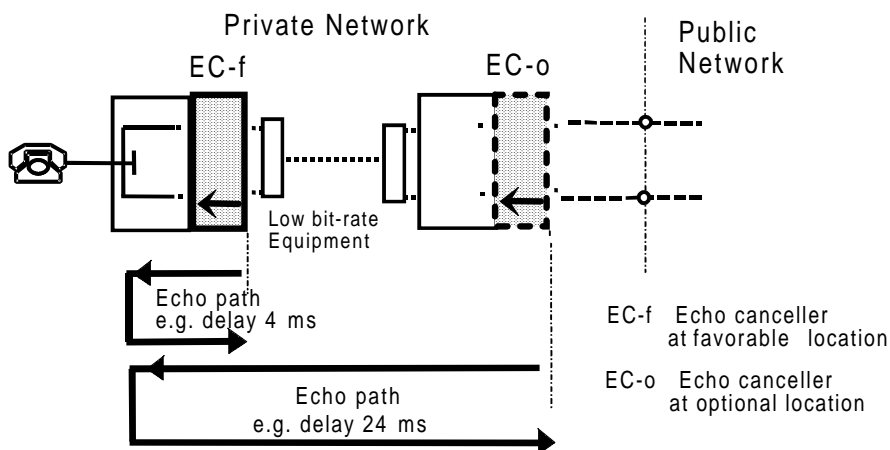
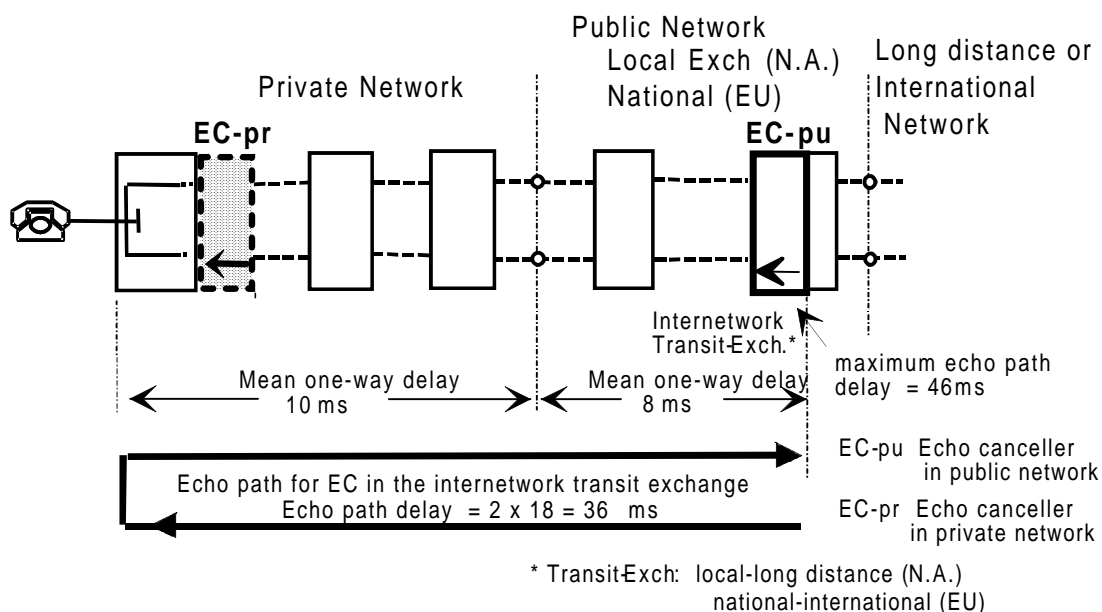


Figure 21: Options for the Location of an Echo Cancellor

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 20, associated with the hybrid, can be considered as the most favourable one. However other locations may also be considered as illustrated in figure 21 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 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 21. On the other hand, for the position EC-f in figure 21, 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 annex C subclause C.1.9 since the (non-linear) low bit-rate equipment is part of the echo path.



* Transit-Exch: local-long distance (N.A.)
national-international (EU)

Figure 22: Use of Echo Cancellers in an Internetwork or International Connection

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 22, 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.

The value of 10 ms in figure 22 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 following ITU-T Recommendations G.165 or G.168, is equivalent to an enhancement of the echo loss, and consequently also the TELR, to values in the range of 50 dB to 60 dB. When performing calculations with the E-Model for connections with inserted echo cancellers, it is recommended to leave the input parameters for TELR, WEPL, T and Tr at their default values.

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 fully new private network, but also applies in the case of an existing network being modified or amended with respects 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 by the user.
- 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, such as the specific telecommunication demands, 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 the present document, 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 C. 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 subclause 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 about the parameter values based on the type of connection and the 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 about these parameters and typical planning values, see annex C.

11.5 End-to-End Calculation with the E-Model

In this step of the planning process, the defined reference configuration(s), together with all relevant parameter values, are taken as the basis for the calculation of the expected quality for the considered configuration. As stated in subclauses 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 precalculations 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 subclause 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 19 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 subclause 9.2 suggests that the working configuration for 2-wire/4-wire connections, shown figure 18, 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 17 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 fully familiar with all the features and restrictions of the program used, to avoid 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 16. 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 16. A comparable situation is described in subclause 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 total impairment value, I_{tot} , a first concern should be for connections where the value is more than $I_{tot} = 45$ to 50. This should be considered as an absolute upper limit, which should never be exceeded, even in exceptional cases. A reference configuration which, with its corresponding parameter values, results in higher I_{tot} values shall not be realized or be used in practice. For such connections, either other solutions should be found or other equipment selected to reduce the impairments to lower values.

It is very important for the planner to fully understand the planning principle recommended in the present document.

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 subclause 4.3, quality is a subjective judgement such that assignments cannot be made to a fixed number for I_{tot} , MOS, GoB or PoW, 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 23.

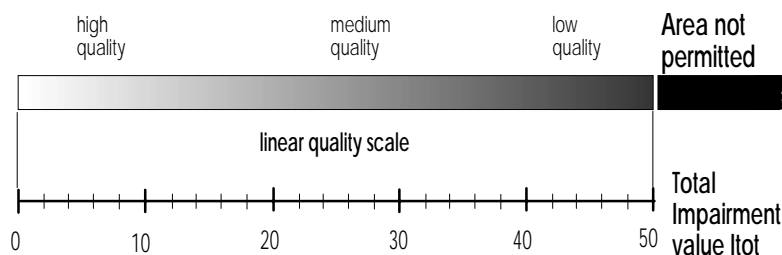


Figure 23: Judgement of a Connection on a linear Quality Scale

Although in figure 23 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 I_{tot} considered as a boundary between high and medium or between medium and low. Only the boundary between low quality and the "not permitted" or unacceptable area is fixed at the value of $I_{tot} = 50$. The linear scale of I_{tot} in the range from 0 to 50 is related to the quality scale so that a result given as a value for I_{tot} can be assigned to the linear quality scale. For further guidance in interpreting the results of planning calculations refer to the verbal quality description as given in tables 1a and 1b.

For practical planning, it is recommended that normal connections within the private network, or between the private network and the public or other networks, should result in upper values for I_{tot} in the range of 15 to 25. For exceptional configurations, values in the range from 30 to 45 are acceptable but should not be exceeded. 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 private network 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 (informative): General Guidance and Information on Transmission Parameters

A.1 General

This annex is providing additional "tutorial" information about the definition and meaning of all the relevant transmission parameters as they are necessary for the understanding of transmission planning. It is also describing specific effects such as echo or stability which are also influenced by the configuration of a voice channel, playing an important role in transmission planning. For further detailed information about all aspects of voice transmission via networks see ETR 250.

A.2 Loss Definitions

Basically a telephone connection established between two telephone sets via switching and transmission elements can be considered as a series connection of different sections as shown in figure A.1.

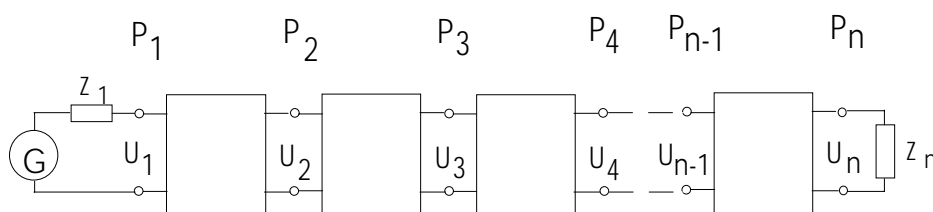


Figure A.1: Loss Contribution in a Telephone Connection

The electrical signal from a generator G - simulating the voice signal - is transmitted via the connection and received at the terminating impedance Z_n . An important attribute of a transmission channel is the "loss" of signal power, i.e. how much of the power P_1 from the signal source is still available as power P_2 at the terminating impedance. This is usually expressed as the ratio between two powers P_1 and P_2 in a configuration as illustrated in figure A.2.

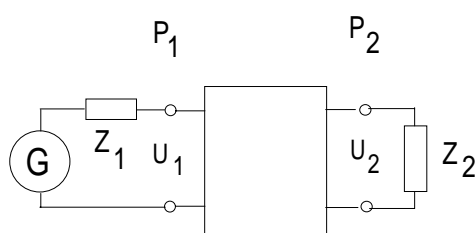


Figure A.2: Definition of Powers P_1 and P_2

For the benefit of more simple calculations the "Loss a " is defined as the logarithmic ratio of the two powers with the unit "Decibel" abbreviated "dB":

$$a = 10 \cdot \log \frac{P_1}{P_2}$$

If the impedances Z_1 and Z_2 are known, the power ratio may also be expressed using the voltages U_1 and U_2 according to the equation:

$$\frac{P_1}{P_2} = \frac{U_1^2 \cdot Z_2}{U_2^2 \cdot Z_1} = \left(\frac{U_1}{U_2} \right)^2 \cdot \frac{Z_2}{Z_1}$$

or calculated as loss in dB:

$$a = 20 \cdot \log \frac{U_1}{U_2} + 10 \cdot \log \frac{Z_2}{Z_1}$$

For the usual configuration of a telephone connection with several serial elements as shown in figure A.1, all power ratios of the different elements are multiplied to obtain the total power ratio between P_1 and P_n .

$$\frac{P_1}{P_n} = \frac{P_1}{P_2} \cdot \frac{P_2}{P_3} \cdot \dots \cdot \frac{P_{n-1}}{P_n}$$

The total loss a_n of this configuration is obtained simply by an addition of the logarithmics of the different ratios P_1 / P_2 , P_2 / P_3 ... P_{n-1} / P_n :

$$a_n = 10 \cdot \log \frac{P_1}{P_2} + 10 \cdot \log \frac{P_2}{P_3} + \dots + 10 \cdot \log \frac{P_{n-1}}{P_n}$$

If the loss values a in dB of the different elements are known, the total loss a_n of this configuration is calculated by addition:

$$a_n = a_1 + a_2 + \dots + a_{n-1}$$

It should be noted, that although in practice voltages are used and measured, the definition of the loss is referred to the ratio of two powers only. If the result of a logarithmic power ratio is negative ($P_2 > P_1$) then the transmission element is providing a gain.

Although most of the elements of a telephone channels are designed for a frequency range between 300 Hz and 3 400 Hz, the loss and gain values are given for one frequency only, usually for the reference frequency of 1 020 Hz (1 004 Hz in North American Standards). Assuming a flat frequency response for the considered transmission element, those loss and gain values can be used for planning purposes with sufficient accuracy.

A.3 Loudness Ratings

Loudness Rating is one of the important parameters in voice transmission and is mainly issued in conjunction with telephone sets. The meaning of this parameter is directly referred to the volume or "loudness" of a connection and describing the loudness in form of a weighted electro-acoustic loss. In contrary to the loss of a specific network element as described in clause A.2, the loudness rating is obtained by an objective measurement not only for one but for several frequencies in the band from 200 Hz to 4 000 Hz (usually 14 frequencies in a 1/3 octave spacing), summing up the different results together with the corresponding weighting factors to one single value. This method is defined and described in ITU-T Recommendation P.79.

In principle, the parameter loudness rating is an issue about the loudness loss between the talker's mouth (Mouth Reference Point MRP) and the listener's ear (Ear Reference Point ERP). The total loudness loss is called Overall Loudness Rating OLR. Loudness rating values however are also defined between an acoustic interface and an electrical interface or between two electrical interfaces. The following different terms are used to describe the characteristics of terminal equipment and for planning purposes:

- | | | |
|------------------------------------|------|-----------------------------------------------------------------------------------------------------------------------------------------|
| - Overall Loudness Rating, | OLR | Total loudness loss between MRP and ERP in a connection. |
| - Send Loudness Rating, | SLR | Loudness loss between MRP and an electrical interface. |
| - Receive Loudness Rating, | RLR | Loudness loss between an electrical interface and the ERP. |
| - Circuit Loudness Rating, | CLR | Loudness loss between two electrical interfaces. |
| - Sidetone Masking Rating, | STMR | Loudness loss between a talker's mouth (MRP) and his ear (ERP) via the electrical sidetone path. |
| - Listener's Sidetone Rating, | LSTR | Loudness loss of a room noise source (measured at the position of the handset microphone) and the ERP via the electrical sidetone path. |
| - Talker Echo Loudness Rating, | TELR | Loudness loss between the talker's mouth and his ear via the echo path. |
| - Listener's Echo Loudness Rating, | LELR | Difference in loudness loss between the talker's direct voice and the delayed echo both reaching the listener's ear. |

As far as telephone sets are considered, the values for their SLR, RLR, STMR and LSTR can only be obtained by measurements with an equipment according to ITU-T Recommendation P.79. STMR and LSTR are not only depending on the telephone's design but also on the terminating impedance in case of analogue telephone sets.

According to the definition of loudness rating as a "loudness loss", the values can be handled in planning practice in the same way as the loss values of other network elements. Therefore the following basic definitions can be derived.

The total loss OLR of a connection can be expressed as the sum of SLR, RLR and the sum of all n CLR values between the two telephone sets.

$$OLR = SLR + \sum_{i=1}^n CLR_i + RLR$$

If the planning is referred to a specific electrical interface, e.g. the connection point between private and public network, then SLR_{pn} and RLR_{pn} of the whole private network section can be derived from the SLR_{set} and RLR_{set} of the telephone set and the sum of m CLR values between the set and the interface

$$SLR_{pn} = SLR_{set} + \sum_{i=1}^m CLR_i$$

and

$$RLR_{pn} = RLR_{set} + \sum_{i=1}^m CLR_i$$

For the CLR of a specific network element, the loss definition as described in clause A.2 can be used with sufficient accuracy if a flat frequency response can be assumed. For unloaded cable sections the CLR can be estimated by the relation

$$CLR = K \cdot \sqrt{R \cdot C}$$

where R is the cable loop resistance in Ohm/km and C is the cable capacitance in nF/km. The factor K - depending on the terminating impedances - is 0,014 for 600 Ohm and 0,016 for capacitive complex impedances.

The parameter talker echo loudness rating TELR is defined accordingly as the sum of SLR and RLR of the telephone set and the (weighted) loss of the entire echo path, called echo loss EL.

A.4 Definitions of Levels and their Units

While the term loss is expressing the logarithmic ratio between the power at the input and at the output of a transmission element, a further important quantity in many applications is the magnitude of a signal at a specific point (interface) along the transmission path. The signal magnitude is called signal "Level" and expressed also as a logarithmic ratio between the signal at the considered point and a corresponding reference signal.

A.4.1 The Unit dB

According to the use of the logarithmic ratio between two quantities, the same unit "Decibel" or "dB" is used as for the definition of losses. The quantities can be in the form of a power, voltage, current, sound pressure etc. However if two values of powers are compared in a ratio Y, the basic definition of the level is $L = 10 \times \log(Y)$, for voltage, current and pressure ratios X the level is defined as $L = 20 \times \log(X)$. In many cases the basic unit dB is extended with one or more additional letters to distinguish between the different applications. The following subclauses are listing and describing some of these additional letters which are in common use.

A.4.2 The Unit dBm

As described in subclause A.4.1, the magnitude of a signal is compared with a reference value. In general the signal magnitude of the speech or other signals within the speechband in a telephone channel is expressed as the signal power in mW. If a power of 1 mW is used as the reference value, the definition for the level L is:

$$L = 10 \cdot \log \frac{P}{P_0}$$

where P is the signal power at the considered point and P_0 is the reference power of 1 mW. The additional letter "m" to the unit is characterizing this level as a power level with reference to 1 mW. Those levels are also called "absolute" levels. According to the mathematical definition, signal powers above 1 mW will have positive values for the level and signal powers below 1 mW will have negative values for the level.

In some applications the unit dBm is also used for a ratio of 2 voltages, where the reference voltage is 0,775 V. The use of this unit is only correct here, if the impedance at the considered point is 600 Ohm resistive, because this voltage across an impedance of 600 Ohm is resulting in a power of 1 mW.

A.4.3 The Unit dBr

In many applications it is more interesting to issue the signal power at a specific point not in its absolute value but in relation to the power of the same signal at a reference point within the network. Therefore the ratio in this case is defined between the power P_X at the considered point X and the power P_0 of the same signal as it appears at the reference point 0.

$$L = 10 \cdot \log \frac{P_X}{P_0}$$

This issue of a level value is called a "relative" level and the unit dB is extended by the letter "r". The reference point is usually designated a "0 dB - point" and defined at an important point within a connection, e.g. the interface between a private and a public network.

Also relative levels can be positive or negative. If relative levels at the input and output of a transmission element are known - both referred to the same reference point - it is immediately apparent, that the difference of these two levels is corresponding to the loss value of this equipment. Furthermore, it is common practice - mainly for test purposes - to define a relative level assuming a sinusoidal signal of 1 020 Hz (1 004 Hz in the North American Standards) with an absolute power of 1 mW (0 dBm) at the reference point. If the reference point is carrying a digital signal, the absolute power level of 0 dBm (called "digital Milliwatt") is defined as a specific 64 kbit/sec PCM code-sequence given in ITU-T Recommendation G.711 in tables 5 and 6 for A-law and μ -law.

With this definition for the absolute power at the reference point, the quantity of a relative level can also be used to characterize the "Power Handling Capability" for the maximum signal input into equipment with a nonlinear characteristic (e.g. codecs, amplifiers etc.). Relative levels at the equipment output are informing about the expected levels of actual signals.

For more information about relative levels, their definitions and applications see also ITU-T Recommendations G.100 and G.101.

A.4.4 The Unit dBm0

If the unit dB is extended by additional "m" and "0", this level issue in "dBm0" is to understand basically as an absolute power level, but referred to the 0 dBr-point. This combination is describing how a signal level at the considered point X would appear at the 0 dBr-point, independent of the relative level at the point X. The actual absolute signal level at point X is then depending of the designated relative level. If for instance the absolute level at the 0 dBr-point is - 10 dBm, the issue would be -10 dBm0. At the point X with a relative level of -4 dBr the absolute level is 4 dB lower in this case, i.e. -14 dBm.

This term is mainly used in conjunction with transmission measurements and service tests. It is also helpful to describe different types of signals (e.g. speech, data, signalling, tones etc.) and their relation with respect to signal magnitudes expressed in quantities of dBm0.

A.4.5 The Letter "p" in the Units dBmp and dBm0p

If signal levels are expressed in a "weighted" value as for noise signals, when a "psophometric weighting" is used - according to the filter curve as described in ITU-T Recommendation O.41 - the unit dBm or dBm0 is extended by the letter "p". This is derived from the French word "ponderé" for weighted.

A.4.6 Relationship between the different Levels

As mentioned already, using relative levels L_i at the input and L_o at the output of an equipment or a specific section along the connection, the difference of these two levels is corresponding to the loss a

$$a = L_i - L_o \text{ in dB}$$

Furthermore, because of their definition the relation between level values using the units dBm, dBr and dBm0 can be described as follows

- dBm = dBm0 + dBr
- dBr = dBm - dBm0
- dBm0 = dBm - dBr

The same relations are valid if all dBm and dBm0 level values are weighted values.

A.5 Delay and Echo

Amongst others, impairments with respect to speech quality may be caused by transmission delay. This delay is not only caused by the propagation time via the different transmission media, but also - mainly in modern digital networks - by additional processing delay. For the kinds of impairments due to transmission delay it is distinguished between the effect of too long transmission delay even in echofree connections, where talkers may encounter problems to follow an interactive flow of conversation and between the effect of echo. The latter effect arises in conjunction with sources for coupling between the go and return path within a telephone connection. These impairments are called "Talker Echo" and "Listener Echo", depending on whether the talker is affected by his own reflected and delayed signal, or the listener observes additional (multiple) echoes of the direct signal. The following paragraphs are containing further information, definitions and acceptable limits.

A.5.1 Influence of Delay in Telephone Connections

If the mean one way transmission time increases to several hundreds of milliseconds, problems may arise for both talkers to follow an interactive normal flow of conversation and to interrupt each other. Recent investigations regarding long pure delay, published in annex B of the ITU-T Recommendation G.114, are providing some information about the results of subjective tests.

Considering highly interactive talks combined with a specific measure of difficulty such as the ability to interrupt, the effect can be detected well below a value of 400 ms. However, the improvement of echo control devices mainly echo cancellers over the past years, has extended the pure delay values with a poor or worse rating compared with earlier investigations. Assuming the increasing use of high performance echo cancellers following ITU-T Recommendations G.165 and G.168 and a careful planning with respect to equipment causing additional delay, the following guidance about the precautions and limits for the transmission time can be given, derived mainly from the well experienced indications in ITU-T Recommendation G.114.

- a) Range 0 ms to 25 ms.

This range of transmission time can be expected for national calls within average sized countries. There are no difficulties during conversation. Usually echo is controlled by providing a sufficiently high echo loss instead of echo control devices.

- b) range 25 ms to 150 ms.

This range is acceptable for most user applications assuming the use of echo control devices.

- c) range 150 ms to 400 ms.

Within this range - in most cases including a satellite link - difficulties may arise for interruptability and normal flow of conversations, mainly in high interactive talks. High Performance Echo Cancellers according to ITU-T Recommendations G.165 and G.168 should be used and careful network planning is necessary.

- d) range above 400 ms.

Values of transmission time above 400 ms should be avoided in any case for general network planning. This value should be exceeded only in exceptional cases.

For international calls and also for some national calls, the increasing use of satellite links can be expected. Assuming a satellite link as the only international part of a connection, with a mean one-way delay of 260 ms between the two earth stations, the remaining maximum transmission time for each national section is 70 ms.

In most European countries - according to their geographical size - the maximum transmission time used by the public networks between the network connection points (NCP), will presently be in the range of 15 ms to 25 ms. The same value can normally be assumed for the section from the NCP to the International Switching Center. For terminal equipment or private networks connected to the public networks, therefore a maximum for the transmission time of 45 ms to 55 ms is available. This may allow the use of cordless systems for terminal equipment, or a limited insertion of low bit-rate-coding equipment within private networks.

However it should be noted, that in those cases the use of echo control devices is mandatory even for normal national calls. Furthermore those terminal equipment or private networks involved in an international connection, may be subject to difficulties in conversation as described above.

The given ranges of transmission time are strongly related to ITU-T Recommendations. For transmission planning according to the rules in the present document, this impairment of too long transmission time is included in the algorithm of the E-Model and the total absolute transmission time T_a is an input parameter to the E-Model. Impairments due to this effect are therefore included in the results of calculations.

A.5.2 Echo Effects in Telephone Connections

Echo generally is defined as an unwanted signal, delayed to such a degree, that it is perceived separately from the Sidetone signal. Echo is distinguished into Talker Echo, where the coupling of signals occurs near the listeners end, affecting the talker and the Listener Echo defined as an echo, produced by double reflected signals and disturbing the listener.

A.5.2.1 Talker Echo

The effect of echo is illustrated in figure A.3. The transmitted speech signal (direct signal) of the talking subscriber is delayed along the different sections of the transmission path, coupled back to the receive path at the far end and received again, affecting the talker with an unwanted signal, comparable with an echo of his own voice. Since this type of echo in the given configuration is only observed by the talker, it is called Talker Echo.

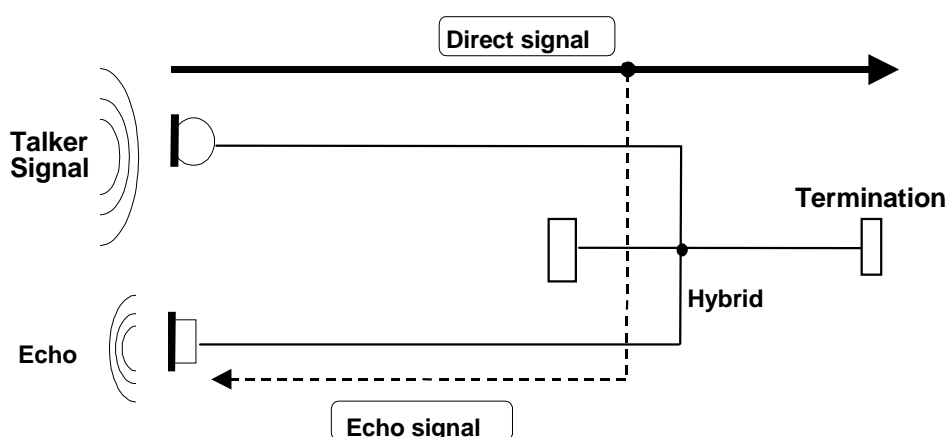


Figure A.3: Effect of Talker Echo

In context with this effect, the echo path is understood as the transmission path via the send- and receive direction of the private, national and international chain and the point where the coupling occurs. The Echo Path Delay is important mainly for consideration of echo effects. It expresses the transmission delay along the echo path in milliseconds. The term Mean One-Way Transmission Time, which is defined as half the sum of the transmission time in both transmission directions is in more common use.

Beside the mean one-way transmission time, the Talker Echo Loudness Rating TELR is the second important factor for an objectionable echo. It is defined as the sum of the sending loudness rating (SLR), the receiving loudness rating (RLR) of the talkers telephone set and the echo loss (EL) of the echo path (see also clause A.3).

A.5.2.2 Listener Echo

In case a closed 4-wire loop is contained in a telephone connection as shown in figure A.4 in addition to talker echo the so-called phenomena of listener echo will appear. The talkers signal is coupled back not only at the listeners end (hybrid B) but also again at the talkers end (hybrid A); thus the talkers signal will arrive at the listener coupled back twice in addition to the talkers direct signal delayed by a certain amount of time. The main distinction in comparison to talker echo with one single coupling back, is that the listener will perceive the annoyance which in conjunction with transmission delays of a few milliseconds only, will be objected as hollowness.

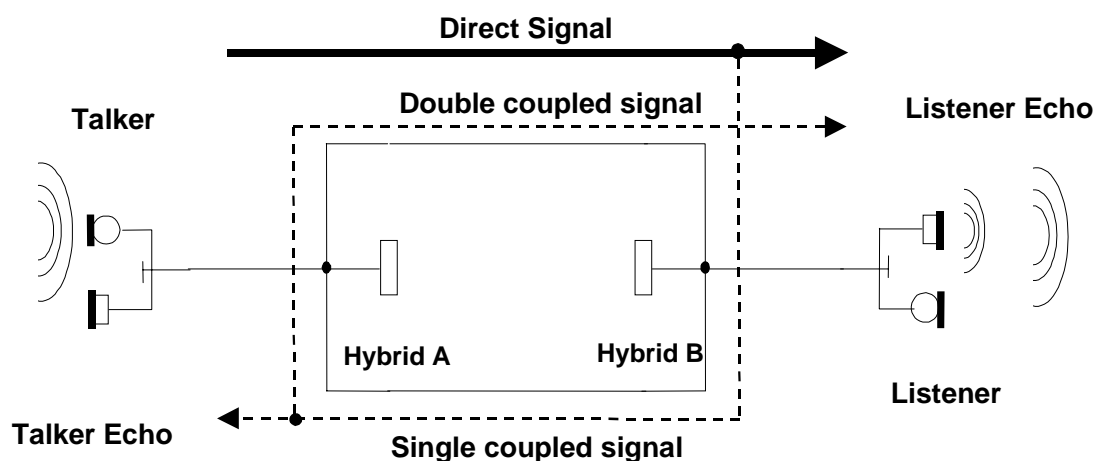


Figure A.4: Effect of Listener Echo

For Listener Echo similar terms are used. The Listener Echo Loss LE is defined as the degree of attenuation of the double reflected signal L_2 with respect to the wanted signal L_1 .

$$LE = L_2 - L_1$$

For practical purposes the LE can be set equal to the Open Loop Loss OLL, i.e. the sum of all losses and gains including the balance return losses within the 4-wire loop (see also clause A.6). If for the balance return losses weighted values are used, the term Weighted Echo Path Loss WEPL is used, one of the input parameters to the E-Model. The WEPL characterizes also the degree of disturbance by Hollowness an effect subjectively perceived as a "hollow sound".

Consequently the Listener's Echo Loudness Rating LELR is expressed by the difference in loudness loss between the speakers direct voice and its delayed echo received at the listener's ear.

A.5.2.3 Echo Sources and Echo Loss

Sources for a coupling between go- and return path which cause echo effects in a speech conversation are mainly equipment being used for conversion between 4-wire and 2-wire, so-called Hybrids. The amount of coupling within those hybrids depends mainly on the mismatch between the Balance Network Z_B and the terminating impedance Z_T at the 2-wire side (figure A.5). The degree of mismatch is expressed as Balance Return Loss a_{BRL} giving a direct impression about the magnitude of the reflected signal.

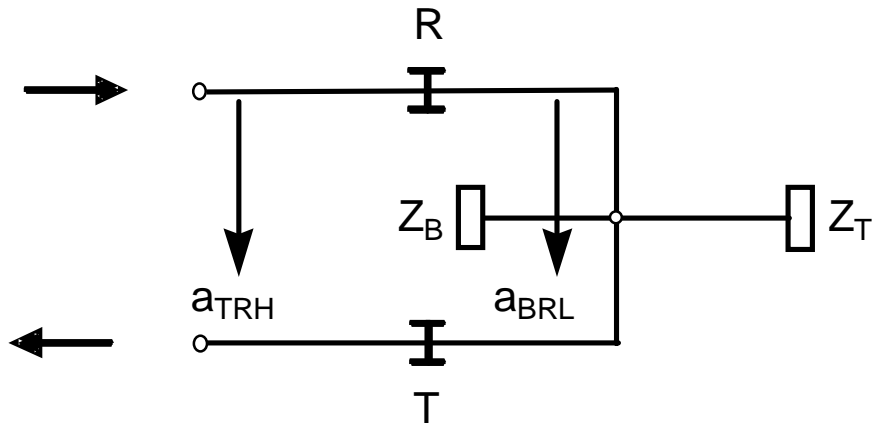


Figure A.5: Balance Return Loss and Transhybrid Loss

The definition for balance return loss is given by the equation:

$$a_{\text{BRL}} = 20 \log \left| \frac{Z_B + Z_T}{Z_B - Z_T} \right|$$

Depending on the application within a network, hybrids are also providing a specific loss in send direction (T-pad) and receive direction (R-pad) between the 2-wire and 4-wire ends, as shown in figure A.5. The resulting loss between the 4-wire input and output, including the balance return loss and the loss of the R- and T-pads, is called transhybrid loss a_{TRH} .

Signal coupling may also arise at the interconnection of two 2-wire sections, such as switching equipment interfaces, transmission systems, cable sections and terminal equipment, if a mismatch between the two impedances exists. However in practice, well planned networks are providing in most cases sufficient matching, therefore those coupling points are negligible for the effect of echo.

A further source for coupling exists within telephone sets, mainly in its acoustic path as shown in figure A.6. Those sources are of increasing importance, since digital telephone sets are providing the only echo source in fully digital connections.

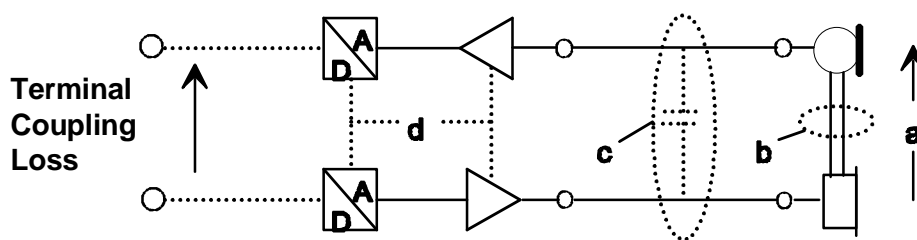


Figure A.6: Coupling in a Digital Telephone Set

For digital telephone sets several coupling paths can be identified. Primarily the acoustic path between receiver and transmitter in the handset (path a in figure A.6) should be considered. Other paths such as structure borne coupling within the handset (path b in figure A.6), capacitive coupling between the wires of the handset cord (path c in figure A.6) and coupling via the power supply for codec and amplifiers (path d in figure A.6) may have additional influence. All possible coupling paths are summarized and expressed in the term Terminal Coupling Loss TCL, referred to the digital input and output of a digital telephone set. The same principles of coupling as shown in figure A.6, can also be applied for mobile telephones or the portable part of cordless telephones. The TCL in this cases is mostly referred to a uniform PCM-interface within the fixed part, i.e. the air path is part of the TCL.

Coupling via hybrids or acoustic paths of telephone sets, is normally subject to an extensive shape in frequency response. For the effect of echo, when considering the echo behaviour of a hybrid, the transhybrid loss is weighted with a specific weighting function over the frequency range 300 Hz to 3 400 Hz. This weighted transhybrid loss is then called Echo Loss. For digital telephone sets the same weighting function is used and expressed as weighted Terminal Coupling Loss TCLw.

According to ITU-T Recommendation G.122 subclause 4.2 Echo Loss EL and TCLw are derived from the integral of the power transfer characteristic $A(f)$ additionally weighted with a negative slope of 3 dB/octave starting at 300 Hz as follows:

$$EL = 3.85 - 10 \log \left[\int_{300}^{3400} \frac{A(f)}{f} df \right] \text{dB}$$

where

$$A(f) = 10^{-\frac{L_{ab}(f)}{10}}$$

where $L_{ab}(f)$ is the loss of the echo path at frequency f . If the results are available in graphical form or as tabulated data, the echo loss may also be calculated using the trapezoidal rule. More information is given in annex B to ITU-T Recommendation G.122 clause 4.

A.5.3 Methods for Echo Control

Since propagation time and in most cases also processing delay cannot be reduced, a possible solution to reduce or suppress the effect of echo is to increase the loss of the echo path. The insertion of additional loss however is limited, since usually also the OLR of the connections is increased.

For fully digital connections terminated on both ends with digital telephone sets with sufficient high TCLw values, the TELR will result in values in the range of 45 dB to 60 dB, with negligible impairments due to echo in normal applications also in case of international connections.

For connections in mixed analogue / digital networks, lower values for TELR should be expected. The insertion of additional loss in the echo path to increase TELR is not possible in most cases, since also SLR and RLR and in total the OLR is increased, causing other impairments. Therefore in connections with not negligible values of transmission delay, specific equipment, called Echo Control Devices ECD such as Echo Suppressors ES or Echo Cancellers EC are used. The following description is referred only to the principle of Echo Suppressors and Echo Cancellers presently in use. More information are given in the ITU-T Recommendations G.164 for ES and G.165 and G.168 for EC.

Presently, Echo Control Devices are used to suppress echo in connections with a mean one-way transmission time of more than 25 ms, i.e. mainly in international connections. The basic application of ECD is shown in figure A.7. At both ends of an established connection an ECD is inserted, usually in international switching centres. The echo path in these applications is the remaining national part of the call path, including the path within a private network if present.

The main task of an ECD is to suppress or compensate signal reflections within the echo path without inserting other remarkable impairments to the speech quality. Two main basic principles are known and in common use for several years, Echo Suppressors and more modern, Echo Cancellers.

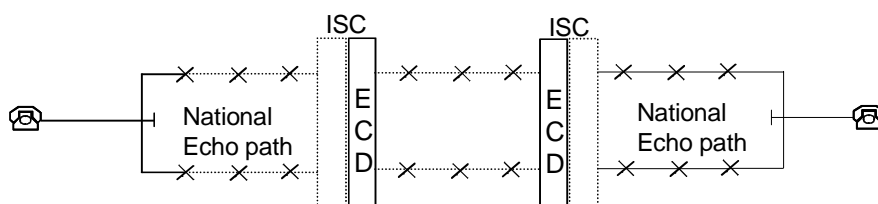


Figure A.7: Application of Echo Control Devices

A.5.3.1 Echo Suppressors

As shown in figure A.8, an ES is able to suppress the signal in the sending path (Sin to Sout) and to insert a loss in the receive path (Rin to Rout). The signal amplitude is detected from the send- and receive path, compared in a logic circuit and used to control the losses in both directions. If a signal from party A is present above a defined threshold level in the receive path, the ES changes into suppression mode, i.e. a high loss of about 50 dB is inserted in the send path, suppressing every possible signal reflection.

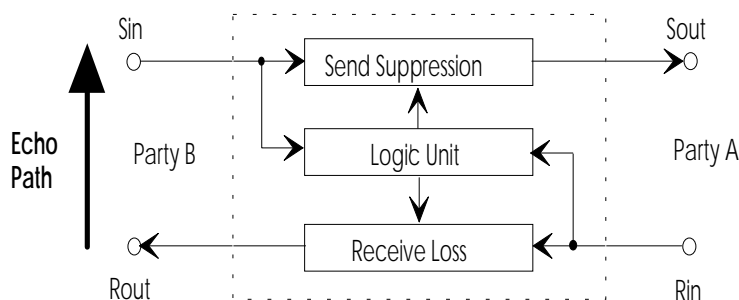


Figure A.8: Echo Suppressor

This mode of course is encountered by party A subjectively as an interruption and party B is unable to "break-in" during a conversation. For this situation, also called "double talking", a differential circuit is used to compare the speech level in send and receive path. If double talking is detected, the send suppression is removed and a loss in the range of 5 to 15 dB is inserted in the receive path. Several other technologies such as fixed or adaptive differential sensitivity are used to operate also under "break-in" conditions. Echo suppressors are applicable for echo path delays up to about 25 ms.

A.5.3.2 Echo Cancellers

The principle of an Echo Canceller EC is shown in figure A.9. 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 1 second. For single talk conditions the receive direction does not give the impression of being interrupted in contrary to ES. 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 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 Center Clipper. The task of this device is to provide a suppression of residual echo levels below a defined threshold.

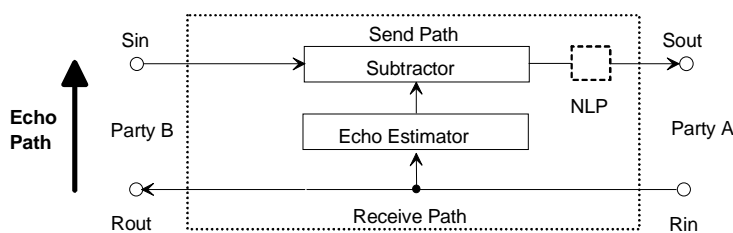


Figure A.9: Echo Canceller

A minimum echo path loss of 6 dB is a requirement common to all ECDs to achieve proper operation. As far as ESs and ECs are designed according to ITU-T Recommendations G.164, G.165 and G.168, compatibility is guaranteed also between an ES inserted at the one end and an EC at the other end.

A.5.3.3 Echo Control in specific Applications

In existing networks the insertion of echo control devices was mainly necessary in 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 amount, 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 center 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 subclauses are describing 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 TBR 10 and ETS 300 175-8 (DECT-Standard) inserting an additional mean one-way delay of about 14 ms.

A.5.3.3.1 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 A.10, which should be identified and investigated during planning.

The different echo paths in this configuration are:

- *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.

- *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.

- *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.

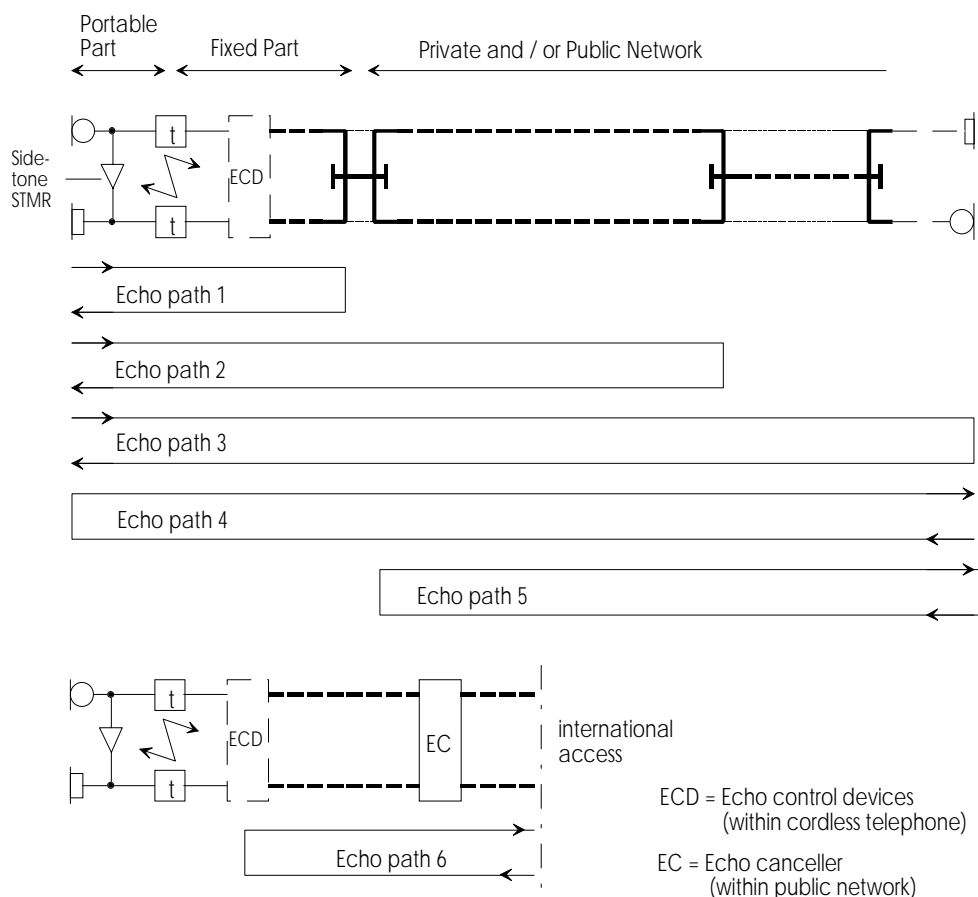


Figure A.10: Possible Echo Paths for a Cordless Telephone

- *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 is more than 34 dB if the path within the private network is fully digital.

- *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 (see also subclause 7.8).

- *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 center because of the international connection. The characteristics of these echo cancellers, presently designed according to ITU-T Recommendation G.165, 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.

A.5.3.3.2 Operation of Echo Cancellers and Soft Suppressors

The main devices in a cordless telephone according to DECT-Standard for the control of echo along the different echo paths are shown in principle in figure A.11. 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 $SLR = 7 \text{ dB}$ and $RLR = 3 \text{ dB}$ 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 $SLR = +4 \text{ dB}$, $RLR = -7 \text{ dB}$ - a gain of 3 dB in the transmit path and of 10 dB in the receive path are adjusted within the hybrid. Assuming an average terminal balance return loss of $a_{BRL} = 11,5 \text{ dB}$, the resulting transhybrid loss of $a_{TRH} = -1,5 \text{ dB}$ 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 subclause A.5.3.2, but without the NLP and with less stringent requirements compared to ITU-T Recommendation G.165 or G.168. The corresponding Standard requires only an enhancement of the transhybrid loss - called echo loss enhancement - of $> 6,5 \text{ dB}$, which is resulting in an echo loss of $-1,5 \text{ dB} + 6,5 \text{ dB} = 5 \text{ dB}$ in the example as shown in figure A.11. Furthermore, the permissible echo path delay for this EC is required with only $> 4 \text{ ms}$, since only the hybrid circuit is forming the echo path.

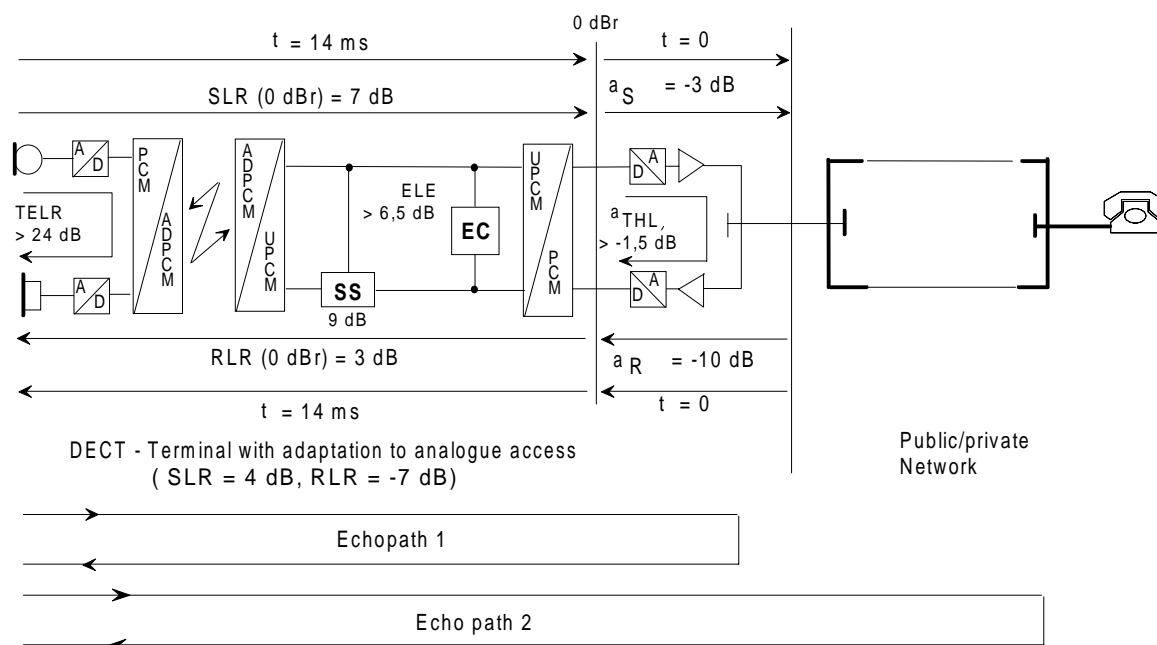


Figure A.11: Echo Control Devices in a DECT Cordless Telephone

To increase the echo loss of only 5 dB an additional device, the soft suppressor SS is inserted. This SS is in its operational mode comparable with an echo suppressor as described in subclause A.5.3.1. 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 14 dB. Together with the SLR and RLR at the 0 dBr-point the total TELR is 24 dB, which is sufficient for this echo in echo path 1.

For echo path 2, an echo loss of 15 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 of 15 dB to 24 dB, resulting in a sufficiently high value of 34 dB for the TELR.

A.5.3.3.3 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 A.10. If the connection is terminated within the private network by a hybrid the requirement for TELR is stated with 24 dB in subclause 7.8 of the present document and 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 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 A.10) 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 A.12.

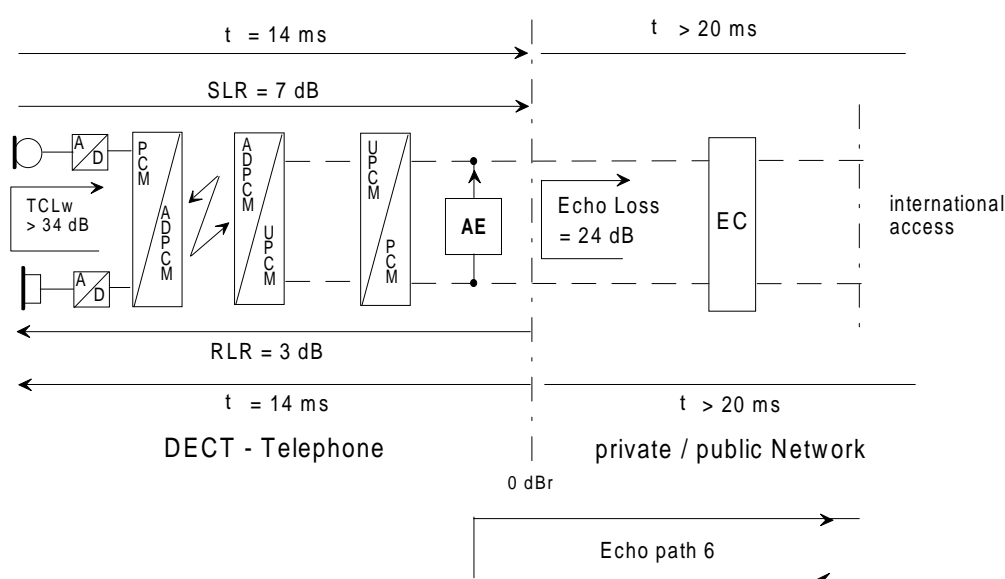


Figure A.12: TCLw and Artificial Echo Loss

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.

A.6 Stability

Two hybrids forming a closed 4-wire loop may under specific conditions enable an oscillation within the loop, the so-called "Singing". This is a separate phenomena of closed 4-wire loops beside the talker echo and listener echo mentioned before. The 4-wire loop in a mixed analogue and digital international or national connection, is providing gains "s" and losses "a" including the balance return losses a_{BRL} at the two terminating hybrids as shown in figure A.13. The point when singing starts is reached when the sum of all losses and gains at one single frequency is equal to or less than 0 dB.

The attribute of a 4-wire loop with respect to a possible singing is called its Stability. The sum of all losses and gains within the loop, responsible for the stability is called the Open Loop Loss OLL. If this OLL is close to the singing point of 0 dB, a further impairment of the speech quality may arise due to the "Near Singing Distortion". Therefore, a Stability Margin is required to avoid those effects.

The most critical point for singing is during call setup and release of a connection. In these phases, idle or short circuit termination at the 2-wire side of the hybrids may occur and the balance return loss will decrease to values close to 0 dB. But also variations of loss with time and the frequency response of transmission systems should be taken into account. Modern switching equipment do not establish the entire call path until the answer signal is received and a correct termination at the 2-wire side is provided.

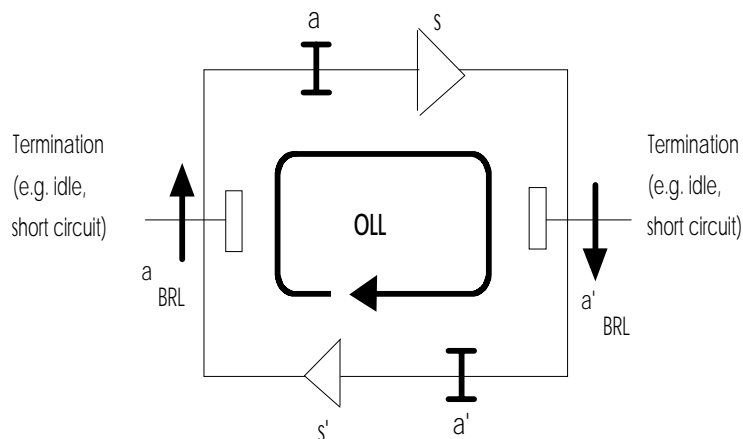


Figure A.13: Stability of a 4-wire Loop

The Stability Loss is the lowest loss (without weighting) within the considered frequency band and configuration. In contrary to talker echo and listener echo, where only frequencies in the range from 300 Hz to 3 400 Hz in conjunction with an echo weighting are considered, a wider range of 0 Hz to 4 000 Hz should be taken into account for singing, since oscillation may start at every single frequency for which the OLL has its minimum.

To avoid overloading of systems and crosstalk into other channels of FDM- and cable sections, a sufficient stability margin should be provided in any case. Guidance on values for stability loss are available in ITU-T Recommendation G.122. However, these recommended values refer to the semi-loop as the part of an international connection, formed by the public and/or private network terminating the international 4-wire chain. It is recommended, that the sum of the nominal losses shall be equal to or greater than $4 \text{ dB} + n$, where n is the number of analogue 4-wire or mixed analogue / digital circuits in the national section. Assuming, furthermore, a minimum balance return loss at the terminating hybrids of 2 dB, the recommendation results in a Stability Loss of:

$$\text{Mean value} = 6 \text{ dB} + n$$

$$\text{Standard Deviation} = \sqrt{4n} \text{ dB}$$

With increasing use of fully digital circuits, the required stability loss is 6 dB. This value is applicable not only to 4-wire/2-wire conversions located within public networks, e.g., in digital local exchanges, but also to hybrids in digital PBXs which are digitally connected to the public network and terminating an international connection.

A sufficient stability loss should not only be provided for the national or international 4-wire chain, but also for every 4-wire loop (e.g. formed by digital local exchanges or PBXs in a 2-wire environment). However, the issue of a minimum stability loss for each of the two semi-loops referred to a virtual reference point in the center of a switching element is not applicable in all cases. Due to other reasons the loss allocation is usually not symmetric. One of the two semi-loops may provide a sufficiently high loss, while the other is contributing with a gain to meet the requirements for a low value of insertion loss. The provision of a defined stability loss can only be described as a minimum OLL for the considered configuration. A suitable nominal value could be:

$$\text{OLL} \geq 4 \text{ dB}$$

assuming, that negative deviations due to frequency response in the range from 300 Hz to 3 400 Hz are limited to about 0,5 dB and loss variations with time are negligible. The loss below 300 Hz and above 3 400 Hz will in case of digital switching elements be higher than within the speech band, due to the coding / decoding process and sufficient filtering.

Annex B (informative): The E-Model

B.1 General

This overview is an extract from the detailed description as contained in the ETSI Technical Report:

"ETR 250 Speech Communication Quality from Mouth to Ear of 3,1 kHz handset telephony across Networks",
July 1996,

with an introduction to the basic methods and the used algorithm. The model was created by Mr. N. O. Johannesson, Ericsson, and discussed and agreed in a joint ETSI ad hoc group "Voice Transmission Quality Mouth to Ear".

B.2 Reference Connection and used Transmission Parameters

The E-Model as described in ITU-T Recommendation G.107 is based on the Equipment Impairment Factor Method, which is described also in the ITU-T Recommendation G.113 and is following previous Transmission Rating Models. The reference connection, shown in figure 15 of the present document, distinguishes between Send Side and Receive Side of the connection. The E-Model estimates the speech communication quality mouth to ear as perceived by the user at the receive side, both as listener and talker.

The transmission parameters used as an input to the E-Model are shown in figure 15. Values for room noise and the D-factors are handled separately in the algorithm for send side and receive side and may be of different amount. The parameters SLR, RLR and circuit noise N_c are referred to a defined 0 dBr-point. All other input parameters are either referred to the receive side only, such as STMR, LSTR, WEPL (for calculation of Listener Echo) and TELR, or are considered as values for the overall connection such as number of qdu, equipment impairment factor I_e and expectation factor A .

There are three different parameters in conjunction with transmission time. The absolute delay T_a is representing the total one way delay between send side and receive side and is used for an estimate of the impairment caused by too long delay. The parameter mean one way delay T is representing the delay between receive side (in talking state) and the point in a connection where a coupling of signals from the send path back into the receive path occurs, which then is a so-called echo source. The round trip delay T_r is representing the delay in a closed 4-wire loop only, where the "double reflected" signal may cause impairments due to Listener Echo.

B.3 Calculation of the Transmission Rating Factor R

According to the Equipment Impairment Factor Method, the fundamental principle of the E-Model is based on a concept given in the description of OPINE (see Bibliography):

"Psychological Factors on the psychological scale are additive"

In a first step the result of any calculation with the E-Model is a Transmission Rating Factor R , combining all relevant transmission parameters for the connection considered. This Rating Factor R is composed of

$$R = R_o - I_s - I_d - I_e + A \quad (1)$$

In principle, R_0 is representing the basic signal to noise ratio including noise sources such as circuit noise and room noise. The factor I_s is combining all impairments which occur more or less simultaneously with the voice signal. Factor I_d represents those impairments, which are caused due to delay and the equipment impairment factor I_e those impairments caused by low bit-rate codecs. The expectation factor A allows the combination of transmission factors and advantages of access for the user. The terms R_0 , I_s and I_d are subdivided into further specific impairment values.

The complex nature of the E-Model and its algorithm is demonstrated in the overview of figure B.1. showing the different input parameters and their influence to each other and to the final result.

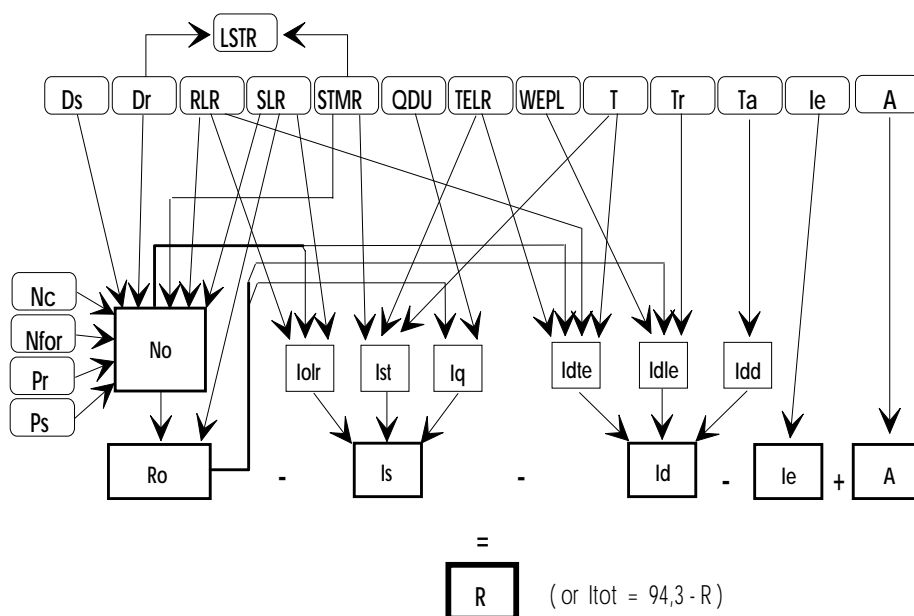


Figure B.1: Overview of the Influence of the different Input Parameters to the E-Model

The different formulas listed and described in the following subclauses, are including all these relations. To understand the principle of the E-Model and its application in planning practice, it is recommended to examine the following formulas along with this overview.

B.3.1 The Basic Signal to Noise Ratio R_0

$$R_0 = 15 - 1.5(SLR + N_0) \quad (2)$$

The term N_0 is the power addition of different noise sources, with the exception of quantization noise

$$N_0 = 10 \lg \left[10^{N_c/10} + 10^{N_{os}/10} + 10^{N_{or}/10} + 10^{N_{fo}/10} \right] \quad (3)$$

N_c (in dBm0p) is the sum of all circuit noise powers, all referred to the 0 dBr-point.

N_{os} is the equivalent circuit noise at the 0 dBr-point, caused by the room noise P_s at the send side

$$N_{os} = P_s - SLR - D_s - 100 + 0,008 \cdot (P_s - OLR - D_s - 14)^2 \quad (4)$$

where $OLR = SLR + RLR$.

N_{or} is the equivalent circuit noise at the 0 dBr-point, caused by the room noise P_r at the receive side

$$N_{or} = RLR - 121 + P_r + 0,008(P_r - 35)^2 \quad (5)$$

The term P_{re} is the "effective room noise" caused by enhancement of P_r by the listener's sidetone path

$$P_{re} = P_r + 10 \lg \left[1 + 10^{(10 - LSTR)/10} \right] \quad (6)$$

$LSTR$ is the Listener's Sidetone Rating at the receive side.

N_{fo} is representing the "noise floor" at the receive side, referred to the 0 dB_r-point

$$N_{fo} = N_{for} + RLR \quad \text{dBm0p} \quad (7)$$

with N_{for} usually set to - 64 dB_mp.

B.3.2 The Simultaneous Impairment Factor I_s

The factor I_s is the sum of three further specific impairment factors:

$$I_s = I_{olr} + I_{st} + I_q \quad (8)$$

I_{olr} represents the decrease in quality caused by too low values of OLR.

$$I_{olr} = 20 \cdot \left[\left\{ 1 + (X/8)^8 \right\}^{1/8} - X/8 \right] \quad (9)$$

where

$$X = OLR + 0,2 \cdot (64 + N_o - RLR) \quad (10)$$

and N_o is given by equation (3).

The factor I_{st} represents the impairment caused by non-optimum sidetone at the receive side

$$I_{st} = 10 \cdot \left[1 + \left\{ (STM_{Ro} - 12)/5 \right\}^6 \right]^{1/6} - 46 \cdot \left[1 + \left\{ STM_{Ro}/23 \right\}^{10} \right]^{1/10} + 36 \quad (11)$$

where

$$STM_{Ro} = -10 \lg \cdot \left[10^{-STM_{R}/10} + e^{-T/4} 10^{-TEL_{R}/10} \right] \quad (12)$$

The impairment factor I_q is representing impairment caused by quantization distortion.

$$I_q = 15 \lg \cdot \left[1 + 10^Y \right] \quad (13)$$

where

$$Y = (R_o - 100)/15 + (46 - G)/10 \quad (14)$$

and for G

$$G = 1,07 + 0,258Q + 0,0602Q^2 \quad (15)$$

$$Q = 37 - 15 \lg(qdu) \quad (16)$$

In this formula qdu means the number of qdu for the whole connection between send side and receive side.

NOTE: If an impairment factor I_e is used for a network element, then the qdu value for that same element should not be used.

B.3.3 The Delay Impairment Factor I_d

Also I_d , the impairment factor representing all impairments due to delay of voice signals is subdivided into three further factors I_{dte} , I_{dle} and I_{dd} .

$$I_d = I_{dte} + I_{dle} + I_{dd}$$

The factor **I_{dte}** gives an estimate for the impairments due to Talker Echo.

$$I_{dte} = \left[(R_{oe} - R_e) / 2 + \sqrt{(R_{oe} - R_e)^2 / 4 + 100} - 1 \right] \cdot (1 - e^{-T}) \quad (17)$$

where

$$R_{oe} = -1,5 \cdot (N_o - RLR) \quad (18)$$

$$R_e = 80 + 2,5 \cdot (TERV - 14) \quad (19)$$

$$TERV = TELR - 40 \lg \frac{1 + T / 10}{1 + T / 150} + 6e^{-0,3T^2} \quad (20)$$

For values of $T < 1,5$ ms, the talker echo should be considered as Sidetone only, i.e., in this case $I_{dte} = 0$. The computation algorithm is furthermore considering the influence of STMR to Talker Echo. Taking into account, that low values of STMR may have some effect of masking to the talker echo and that for high values of STMR the talker echo may become more noticeable, the terms TERV and I_{dte} are adjusted as follows:

For STMR ≤ 9 dB:

In equation (19) TERV is replaced by TERVs, where

$$TERVs = TERV + I_{st} / 2 \quad (21)$$

where I_{st} is representing impairments caused by too low sidetone (equation 11).

For $9 \text{ dB} < \text{STMR} \leq 15 \text{ dB}$ the equations (17) to (20) given above apply.

For STMR > 15 dB:

I_{dte} is replaced by I_{dtes} , where

$$I_{dtes} = \sqrt{I_{dte}^2 + I_{st}^2} \quad (22)$$

The factor **I_{dle}** is representing impairments due to Listener Echo. The equations are:

$$I_{dle} = (R_o - R_{le}) / 2 + \sqrt{(R_o - R_{le})^2 / 4 + 169} \quad (23)$$

where

$$R_{le} = 10,5 \cdot (WEPL + 7) \cdot (Tr + 1)^{-0,25} \quad (24)$$

and R_o is given by equation (2).

The factor **Idd** represents the impairment caused by too long absolute delay T_a , which occurs even in case of perfect echo cancelling.

For $T_a \leq 100$ ms: $I_{dd} = 0$

For $T_a > 100$ ms:

$$I_{dd} = 25 \cdot \left\{ (1 + X^6)^{1/6} - 3 \cdot (1 + [X/3]^6)^{1/6} + 2 \right\} \quad (25)$$

with

$$X = \frac{\lg(T_a/100)}{\lg 2} \quad (26)$$

B.3.4 The Equipment Impairment Factor I_e

The values of the Equipment Impairment Factors I_e of elements using low bit-rate codecs are not related to other input parameters. They are based on subjective test results as well as on network experience. Some values are already given in ITU-T Recommendations G.113, table 8, G.175 and also listed in table 2 of the present document.

B.3.5 The Expectation Factor A

Due to the specific meaning of the expectation factor A , there is - consequently - no relation to all other transmission parameters. Values are proposed in table 3 of the present document.

B.4 Quality Measures derived from the Transmission Rating Factor R

The Transmission Rating Factor R can be in the range from 0 to approximately 100, where a value of $R = 0$ is representing an extremely bad quality and R close to 100 a very high quality. The E-Model is providing a statistical estimation of quality measures. The percentages for a judgement "Good or Better (GoB)" or "Poor or Worse (PoW)" and for the reaction of an early termination (TME) of a call, are obtained from the R -Factor by means of the Gaussian Error Function

$$E(x) = \frac{1}{\sqrt{2p}} \int_{-\infty}^x e^{-t^2/2} dt \quad (27)$$

The equations are

$$GOB = 100E\left(\frac{R - 60}{16}\right)\% \quad (28)$$

$$POW = 100E\left(\frac{45 - R}{16}\right)\% \quad (29)$$

$$TME = 100E\left(\frac{36 - R}{16}\right)\% \quad (30)$$

The "Mean Opinion Score MOS" on a scale from one to five can be obtained from the R-Factor using the formulas:

For $R \leq 0$:

$$\text{MOS} = 1$$

For $0 < R < 100$:

$$\text{MOS} = 1 + 0,035R + R(R - 60)(100 - R)7 \cdot 10^{-6} \quad (31)$$

For $R \geq 100$:

$$\text{MOS} = 4.5$$

Annex C (informative): Transmission Parameters for specific Elements

This annex provides additional guidance about 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. 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 the present document, 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 private network, 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 analyze and compare this data.

C.1 Elements in Private Networks

C.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 only should be used; tolerances should not be considered. This is also valid if a volume control in receive direction is provided. In this case, only the RLR value 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.

C.1.1.1 Analogue Telephone Sets

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

Sending Loudness Rating	SLR
Receiving 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 TIA-470. 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 ITU-T Recommendation P.79. (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 not 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 subclause C.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.

C.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 TBR 8 (Europe) or with TIA-579 (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.

		<u>Europe</u>	<u>North America</u>
Sending Loudness Rating	SLR	+7 dB	+8 dB
Receiving 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 (N.A.); the desirable value is 45 dB.

North American loudness values for digital sets are being standardized in North American Standards (TIA-579 and T1.508) and reflect the long-term objectives of ITU-T Recommendation P.310, February, 1996 ("Transmission Characteristics for Telephone Band (300 Hz to 3 400 Hz) Digital Telephones"). In general, Loudness Rating values should be in accordance with the ITU-T Recommendation P.79. For the weighted Terminal Coupling Loss, TCLw, some telephones may provide values higher than those given above. If stated by the manufacturer's declaration such higher values may be used for planning purposes.

C.1.2 Wireless Telephone Sets

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 C.1, an extract is given from the DECT and the GSM Standards for all parameters relevant for planning according to this Guide. North American wireless sets use either licensed or unlicensed technology, table C.2 presents the licensed and unlicensed parameters which are relevant for voice quality planning.

For wireless telephones deviating from these standards, information about the actual values according to the list of parameters in tables C.1 or C.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 about echo cancelling devices in other sections of the network, a careful investigation of their interworking is necessary (see subclauses 8.2.2, C.1.1.9, and D.1.6 and annex A for more information, specifically related to integrated echo suppression with references to DECT.)

Table C.1: 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	TCLw (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	---	---	---
Number of qdu (note 2)	qdu	0,5	0,5	0,5	0,5
Equipment Impairment Factor	le	7	20	23	6 (note 3)
NOTE 1: A TCLw of 34 ... 46 dB is optional, artificial echo loss required.					
NOTE 2: qdu only for the A/D-D/A conversion (A-law, G.711), other processes are included in the equipment impairment factor le.					
NOTE 3: Provisionally.					

Table C.2: Planning Values for Wireless Telephones - North American Region

		Licensed: TDMA(IS641-EFRC)	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	TCLw (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 1)	qdu	0,5	0,5
Equipment Impairment Factor	le	10 (note 2)	7
NOTE 1: qdu only for the A/D-D/A conversion (μ -law, G.711), other processes are included in the equipment impairment factor le.			
NOTE 2: Provisionally.			

C.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 ES 201 168 (Europe) or TIA-464B (North America). It should be noted, that in ES 201 168, 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 ITU-T Recommendation G.711 (A-law or μ -law);
- New Coding principles, e.g., according to ITU-T 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. The following matrix illustrates the possibilities.

Table C.3: Switching Equipment Connection Possibilities

Interface to Connection Elements	2-wire Analogue	4-wire Analogue	Digital
Public Network	Common in N.A. phasing out in Europe	Unlikely, except in U.K.	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. (North America TIA-464B 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 I_e should be selected according to table 2. With the exception of inserted digital loss or gain pads, the number of q_{du} can be set to $q_{du} = 0$ 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 $q_{du} = 0,5$ can be used for analogue or digital internal switching, where one of the two interfaces is digital and finally a value of $q_{du} = 1$ is valid for internal digital switching and both interfaces analogue. In case of digital pads the number of q_{du} should be increased by 0,7 for each pad in all configurations (see note).

NOTE: It is assumed that, when loss or gain in an analogue-to-digital path is implemented via codec settings, there is no additional q_{du} increment beyond the 0,5 value. If the PBX provides A-law/ μ -law conversion, there may be additional q_{du} , 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 ETR 275, TIA-464B, and T1.508).

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. (TIA-464B specifies minimum balance requirements for various types of 2-wire interfaces.)

The echo loss (4-wire return loss in N.A.) 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 (TBRL_w or weighted hybrid balance). (See annex A 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 ETS 300 439 and in TIA-464B). 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 TBRL_w and the echo loss can be assumed in planning:

Table C.4: Average Values for TBRL_w and Echo Loss

Termination at the 2-wire side with	TBRL _w	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, TIA-464B 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).

C.1.4 Leased Lines and Tie Trunks

In private networks, leased lines and tie 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: $0,4 \text{ dB} + 0,015 \times \text{length in miles}$; maximum 2,9 dB.
- Short-haul trunks with fixed loss: 2 dB.
- Off-premises lines for terminals: 0 to 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.

C.1.5 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 16 through 18. 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 precalculation for SLR_S and RLR_R (see subclause 8.2).

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.

C.1.6 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.

C.1.7 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 ITU-T Recommendations G.721 (1988), 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 ITU-T 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 I_e . For standardized low bit-rate coders, values are given in table 2. 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 subclause C.1.1.9.

C.1.8 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.

C.1.9 Echo Cancellers

As already described in subclauses 7.2 and 8.2, the result of a planning calculation can show that the Total Impairment Value, I_{tot} , as a sum of different impairments is mainly influenced by the impairment value for echo I_d . In these situations, the decision should be made to insert echo suppressing devices. Those devices are available as either echo suppressors or echo cancellers. The basic principles of both are described in annex A. In modern networks, only echo cancellers are used due to a variety of advantages. 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 the ITU-T Recommendation G.168 (or G.165). Echo cancellers integrated in specific equipment are usually designed for this specific application and therefore not necessarily following G.165 in all data.

NOTE: The characteristics of echo cancellers according to G.165 are measured using a noise signal. If results are also available performed with other test signals, e.g., artificial voice according to ITU-T Recommendation P.50 or Composite Source Signals, these data will provide a more accurate issue about 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.

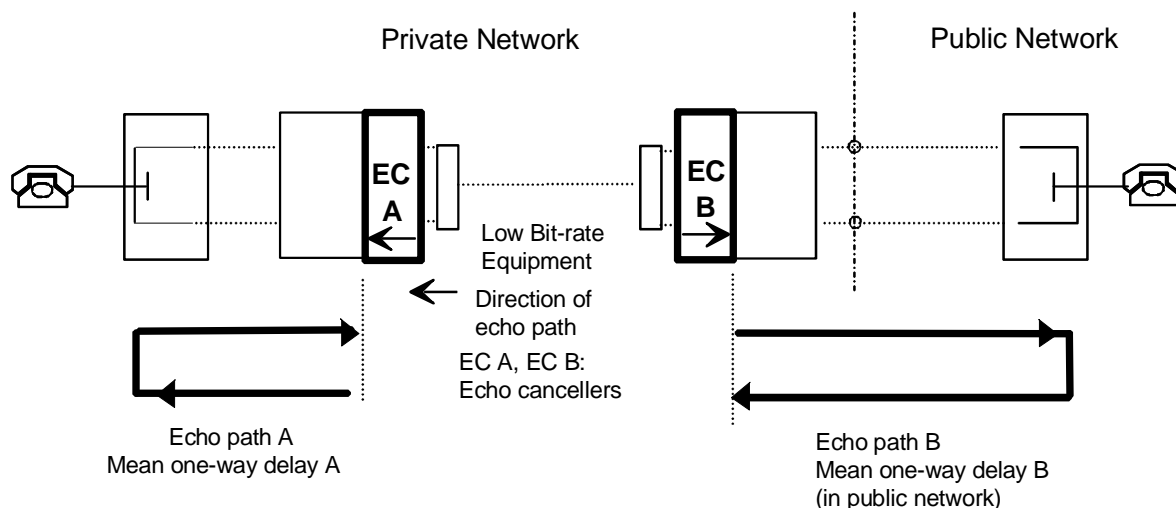


Figure C.1: Echo Cancellers in a Private Network and their Echo Paths

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 C.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 G.165, 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. This means, for the decision if a specific echo canceller is suitable in 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 ms to 8 ms higher than the value of the echo path. For echo cancellers according to G.168 or G.165, the ability to compensate can be assumed in the range of 40 ms to 60 ms. In contrary, echo cancellers integrated in specific devices may provide lower values. If for instance, the EC A and EC B in figure C.1 are integrated in a low bit-rate equipment and 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 Center 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 G.168 (G.165) 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 1 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 $SLR \geq 7$ 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 2 100 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.

C.2 Transmission Parameters of Public Networks

Along with the basic planning principles of the present document - 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 co-operation 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 C.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.

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 C. 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 an far-end termination via P.

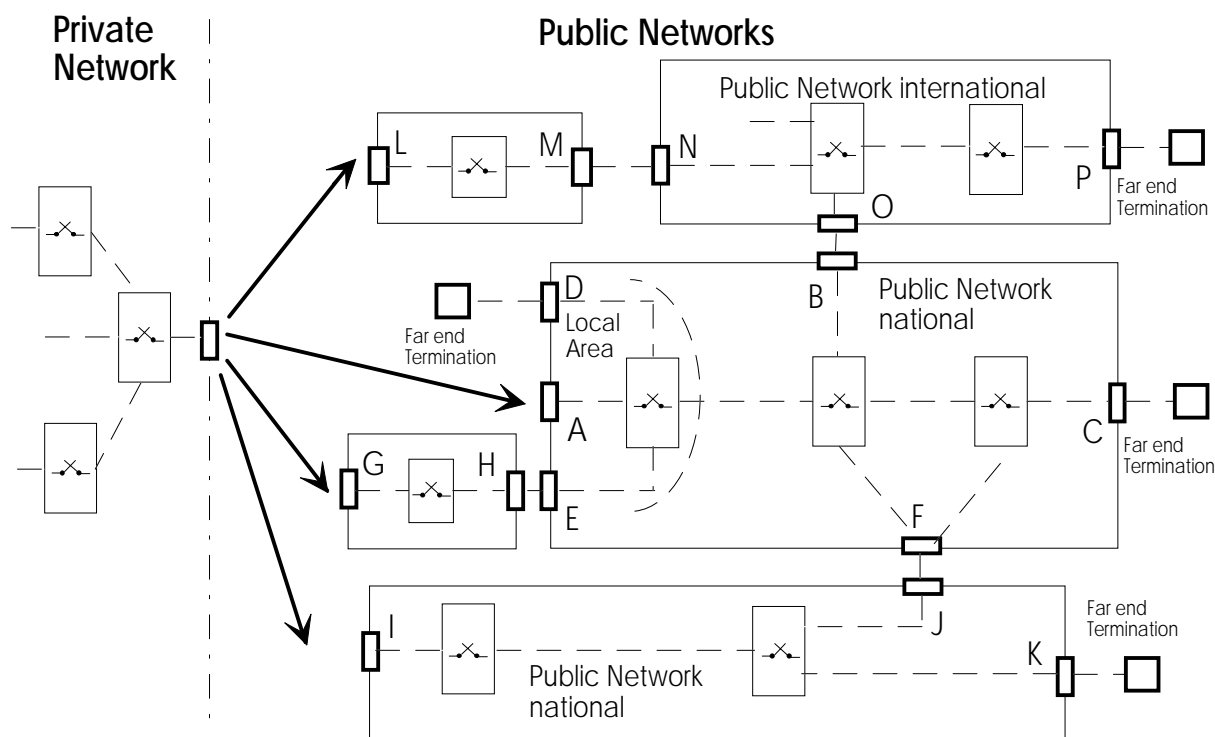


Figure C.2: Access and Routing in Public Networks

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 C.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 C.2 can be described as follows: Access point A is 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 C 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 C.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 clause C.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 values 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.

C.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.

C.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 C.2) or via satellite links (e.g. for points L and M in figure C.2).

C.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 Annex A. 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.

C.2.4 Insertion of Echo Cancellers

Information should be given from the network operator about 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 round-trip delay is likely to exceed 25 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 subclause C.1.9) it may be assumed that devices as used in public networks are, in most cases, compliant with the ITU-T Recommendation G.165 or G.168. 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.

C.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 subclause 8.4, only A/D-D/A conversions, according to ITU-T Recommendation G.711 (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 I_e should be used.

C.2.6 Equipment Impairment Values (I_e)

Information from public network operator(s) should include values for the equipment impairment value, I_e 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 TBR 10. When standardized coding laws are used, the values as listed in table 2 can be used for planning.

C.3 Transmission Parameter of the Far-end Termination

The planning principles recommended in the present document 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).

C.3.1 European Far-end Termination Scenarios

For the definition of an far-end termination including all relevant transmission parameters, three different types of termination are recommended as shown in figure C.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 C.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 q_{du} are both assumed to be 0 in this termination.

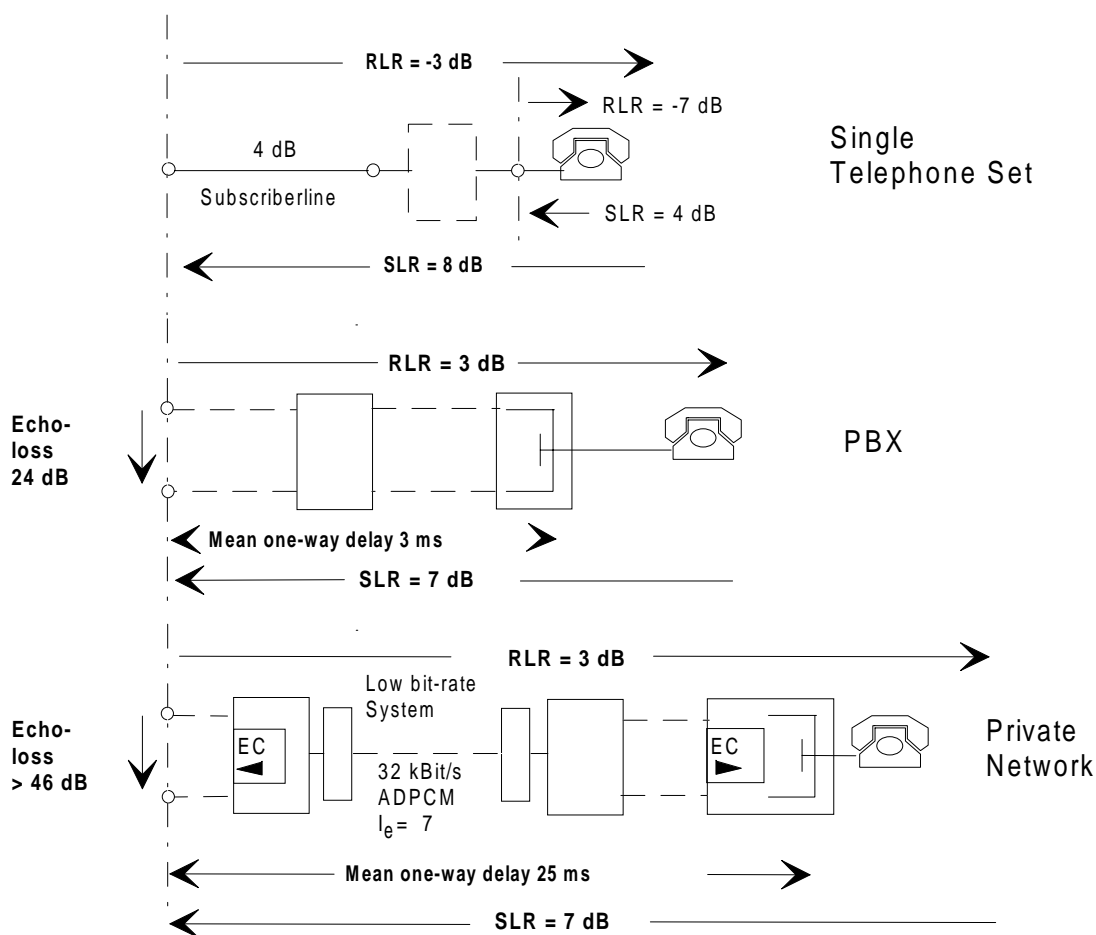


Figure C.3a: Configurations and Transmission Characteristics for the Far-end Termination - Europe -

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 ETS 300 283 for digital interfaces to public networks. The value of q_{du} 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 $I_e = 7$ 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 > 46 dB. The Loudness Ratings are the same as for the second configuration SLR = 7 dB and RLR = 3 dB (in accordance with ETS 300 283); likewise, the value of q_{du} for this far-end termination can be assumed to be 0,5.

C.3.2 North American Far-end Terminations

North American far-end terminations can best be classified by the type of access line. T1.508 defines two types of access lines, analogue and digital. These are illustrated in figure C.3b for connections to subscriber lines and connections to a PBX. Loudness values are those specified in T1.508 as characterizing access lines to the North American PSTN. For specific connections, other values may apply.

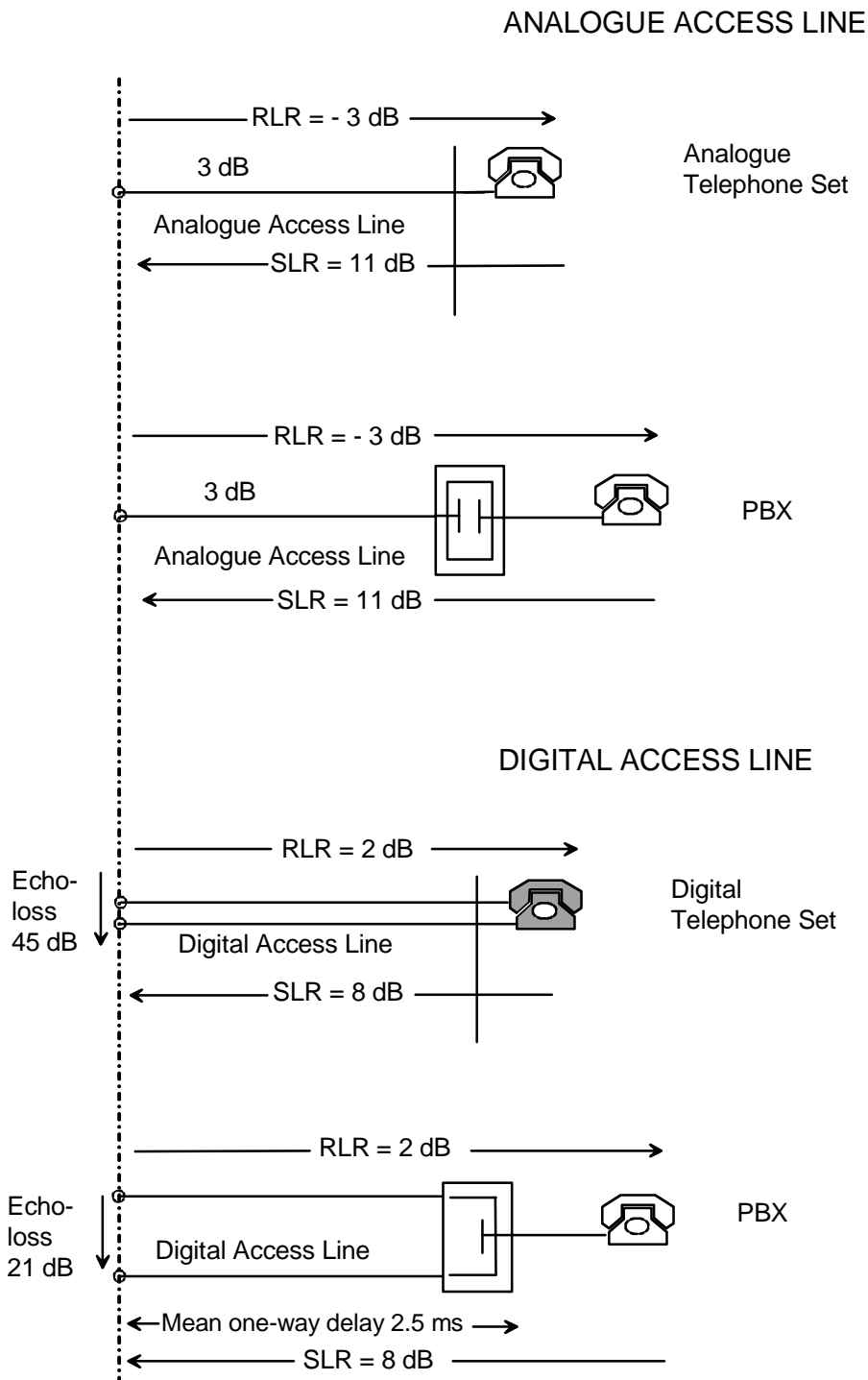


Figure C.3b: Configurations and Transmission Characteristics for the Far-end Termination - North America -

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 TIA-470, 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 0 in this termination.

The North American PBX loss plan, as specified in TIA-464B 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 a 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 T1.508 as terminating a digital (ISDN-like) telephone with nominal loudness characteristics conforming to ISDN standards. (The 1992 issue of T1.508 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 TIA-579). As shown in figure 3Cb, a digital access line terminated in a digital telephone will provide a far-end termination with the loudness and echo loss characteristics conforming to TIA-579.

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 TIA-464B 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. T1.508 guidelines for the application of echo control in networks connecting to the public network suggest a limit of 5 ms round-trip delay; for planning purposes, a 2,5 ms one-way echo delay 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 D (informative): Planning Examples

Due to a variety of differences between the European and the North American areas 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 D.1 deals with examples more related to the European situation, while clause D.2 is presented in a similar manner as clause D.1, but with specific North American details.

D.1 Planning Example for the European Scenario

The following example will demonstrate how to perform transmission planning according to the present document. This is not representing an actual private network, but the structure, routing and further requirements of the user, are assumed such, 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.

D.1.1 Description of the Network and User's Demand

The basic structure of the network is shown in figure D.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 for only 150 extensions. Both, digital and analogue telephone sets are used at all PBXs.

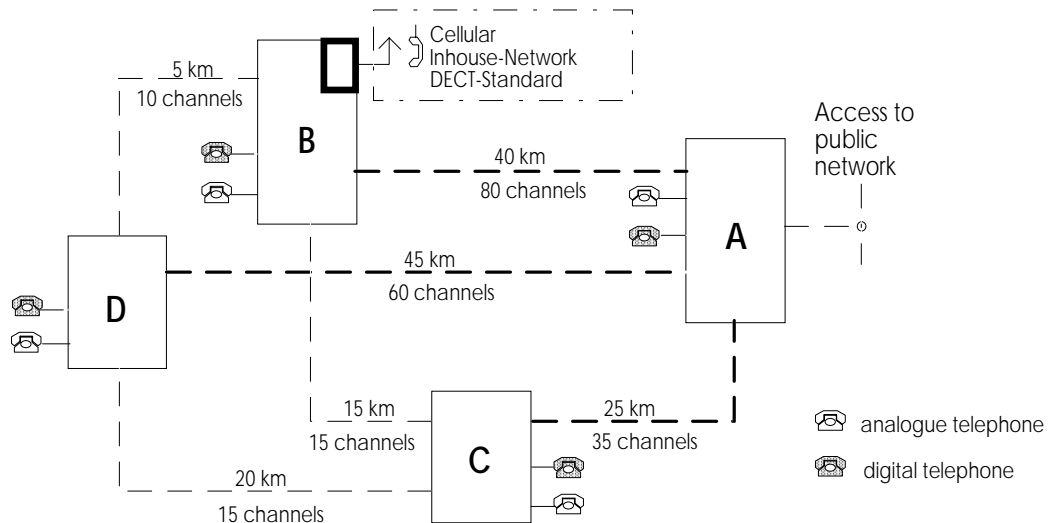


Figure D.1: Basic Structure of the Private Network

All switching elements should be interconnected via 2 Mbit/sec 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 D.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.

D.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 D.2.

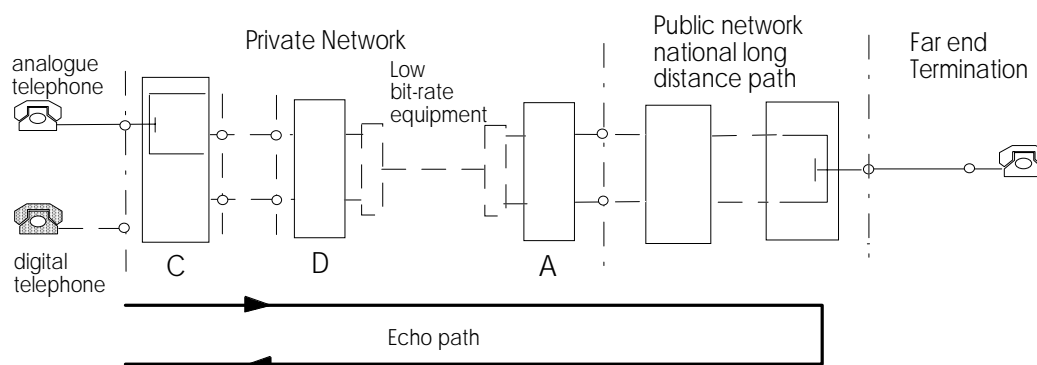


Figure D.2: Reference Configuration 1

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 D.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 D.2 and D.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.

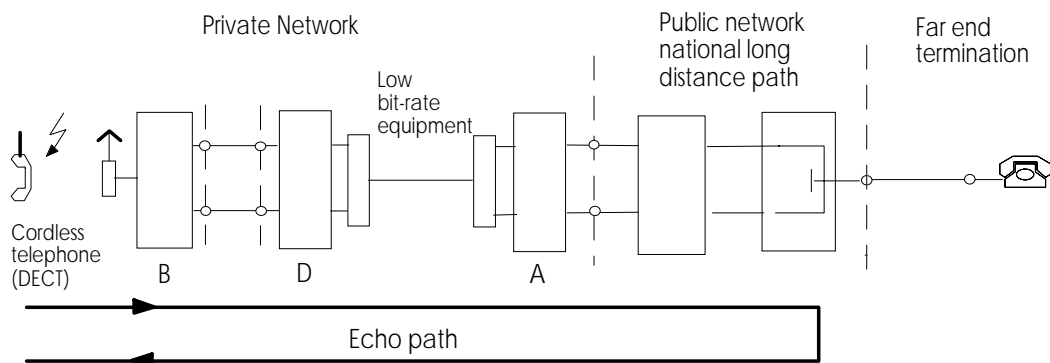


Figure D.3: Reference Configuration 2

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 D.4 is included into the planning.

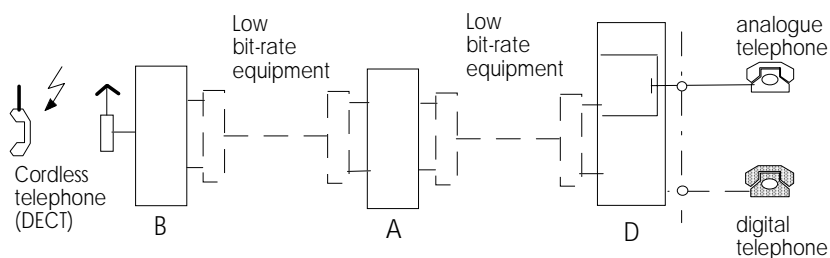


Figure D.4: Reference Configuration 3

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.

D.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 annex C subclause C.1.3 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	$Z_R = 270 \Omega + (750 \Omega \parallel 150 \text{ nF})$
Balance impedance	$Z_B = \text{optimized for termination with } Z_R$
Mean one-way delay	$\tau = \text{negligible}$

Digital telephones (according to ETSI TBR 8):

Send loudness rating	SLR = +7 dB
Receive loudness rating	RLR = +3 dB
Terminal coupling loss weighted	TCL _w = > 46 dB
Mean one-way delay	$\tau = 1,5 \text{ ms}$

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 TBR 10):

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	$\tau = 14$ 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 TBR 10 is provided.

Switching elements

All the PBXs at the different locations are of the same type with a 64 kbit/sec 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	$270 \Omega + (750 \Omega \parallel 150 \text{ nF})$
Balance impedance	$270 \Omega + (750 \Omega \parallel 150 \text{ nF})$
Echo loss (for termination with Z_R)	25 dB

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 (offhook 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/sec interfaces of the leased line and the PBX as shown in figure D.5. The equipment is providing a speech compression using either 32 kbit/sec or 24 kbit/sec ADPCM. The corresponding equipment impairment factor values can be taken from table 2 with

32 kbit/sec ADPCM equipment	$I_e = 7$
24 kbit/sec ADPCM equipment	$I_e = 25$

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

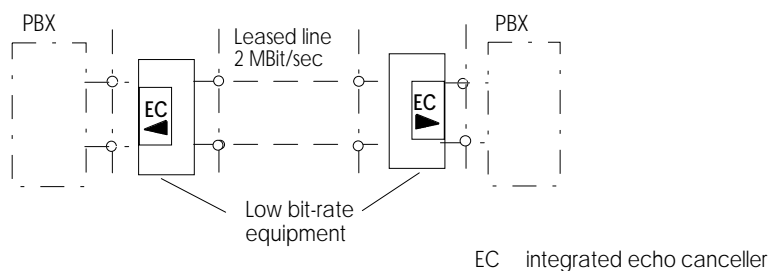


Figure D.5: Use of Low Bit-rate Equipment for a Leased Line

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 equipment relevant for planning, provided by the supplier are as follows:

Mean one-way delay	20 ms (with VAD for 32 kbit/sec or 24 kbit/sec)
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 subclauses 10.2 and annex C subclause C.1.1.9):

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 sec

It should be noted, that these echo cancellers are - with the exception of the permitted echo path delay - nearly identical with those according to ITU-T Recommendations G.165 or G.168.

D.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 D.6 through D.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. Although in the configuration in figure D.6 a closed 4-wire loop can be found between the two hybrids within the local exchange of the public network and the PBX when being connected to analogue telephones, impairments due to listener echo can be neglected because of the necessary control of the talker echo. Consequently the parameters WEPL and Tr are set to their default values. 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 absolute one-way delay T_a is well below a value of 150 ms. 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 D.6.

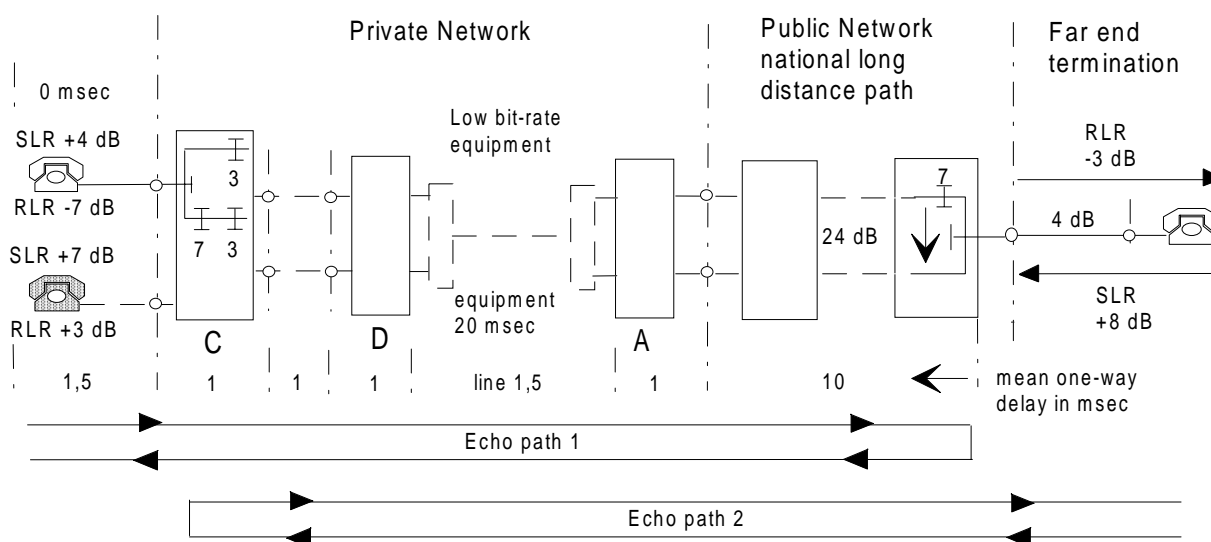


Figure D.6: Basis for Calculation of Reference Configuration 1

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 subclause 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 whether the analogue or the digital telephone of the private network is considered. For impairments due to echo path 1 - effective for both telephone sets - the telephone 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 precalculation. 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
TELRL	34 dB	34 dB

The mean one-way delay for echo path 1 is obtained by simply adding all single values along the reference configuration in figure D.6. 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 precalculation the SLR and RLR referred to the defined 0 dB_r-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, which is equal to the default value. For the SLR the path between the far-end termination (send side) and the 0 dB_r-point is relevant. The corresponding result is SLR = 8 dB.

The remaining impairment to be determined for this example is the equipment impairment factor I_e . 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/sec. According to table 2 the corresponding values are $I_e = 7$ for 32 kbit/sec and $I_e = 25$ for 24 kbit/sec. 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 (equal to the default value)

TELRL = 34 dB (for both telephones)

$T = 35,5$ ms (for analogue telephones)

$T = 37,0$ ms (for digital telephones)

$I_e = 7$ (for 32 kbit/sec ADPCM equipment)

$I_e = 25$ (for 24 kbit/sec ADPCM equipment)

$q_{du} = 1,7$ (for analogue telephones)

$q_{du} = 1$ (for digital telephones, equal to the default value)

To reduce the number of calculations the parameter T can be averaged to $T = 36$ ms and the number of q_{du} can be left at the default value, since there will be no difference in the results for values of less than $q_{du} = 4$ (see also subclause 7.5).

The result for the calculation for reference configuration 1 with the E-Model is shown below for the transmission rating R, the separate impairment values for I_s , I_d and I_e and the total impairment value I_{tot} , where this value is obtained according to the relation $I_{tot} = 94,3 - R$.

	R	I_{tot}	I_s	I_d	I_e
with 32 kbit/sec ADPCM equipment	61,2	33,1	0,3	25,0	7
with 24 kbit/sec ADPCM equipment	43,2	51,1	0,3	25,0	25

In a first analysis the results for I_{tot} are too high in both cases. Examining the separate values, mainly for I_d the sum for impairments due to echo and I_e for equipment impairments, the major impairment is caused by delay for 32 kbit/sec ADPCM equipment. Nearly the same impairment is contributing additionally when 24 kbit/sec ADPCM equipment is used. The high value for I_d can be reduced if echo control devices are used. In this case also the value for $I_{tot} = 51,1$ would be decreased, but not below 25, 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/sec 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 subclause 7.8. The required loudness rating values provided by the private network at the interface with $SLR \geq 7$ dB and $RLR \geq 3$ dB are met. Also the echo loss of ≥ 24 dB is guaranteed with 31 dB (25 dB of the hybrid in C and 2 x 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 D.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 precalculations the reference configuration of figure D.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, equal to the default value. 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 D.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
<hr/>	
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/sec ADPCM equipment is considered. The input parameters are:

SLR	=	7 dB	(equal to the default value)
RLR	=	4 dB	
TELR	=	43 dB	(for both telephones)
T	=	35,5 ms	
I_e	=	7	(for 32 kbit/sec ADPCM equipment)

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

R	Itot	Is	Id	Ie
77,4	16,9	0,3	9,0	7

With this value for I_{tot} the subscriber at the far-end termination can expect higher quality than the subscriber in the private network. However, also 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, definition of the 0 dBr-point and calculations for reference configuration 2 are nearly 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 D.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 18 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.

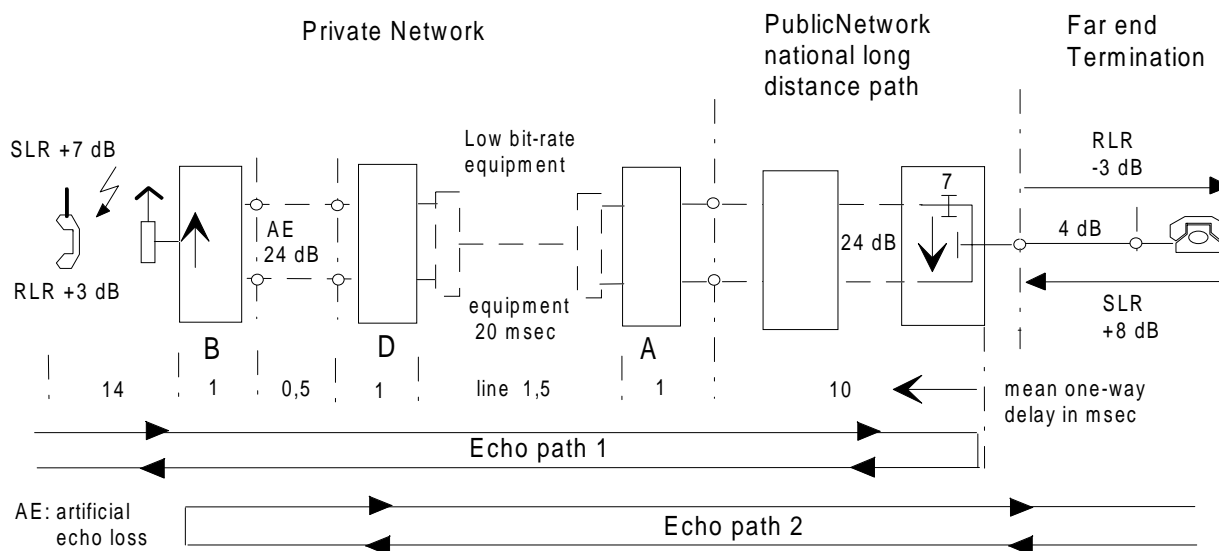


Figure D.7: Basis for Calculation of Reference Configuration 2

The precalculation 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 D.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 by a simple addition with $T = 49$ ms. 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 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 is then $RLR = 3$ 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 the equipment impairment values the 32 kbit/sec ADPCM coding is used twice resulting in a total value of $I_e = 14$.

The input values to the E-Model for calculation of reference configuration 2 are:

$$\begin{aligned} \text{SLR} &= 8 \text{ dB} \\ \text{RLR} &= 3 \text{ dB} \quad (\text{equal to the default value}) \\ \text{TELR} &= 43 \text{ dB} \\ \text{T} &= 49 \text{ ms} \\ \text{Ie} &= 14 \quad (\text{for two times 32 kbit/sec ADPCM}) \end{aligned}$$

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

R	Itot	Is	Id	Ie
65,7	28,6	0,3	13,5	14

In a first analysis also this result with $I_{\text{tot}} = 28,6$ is considered as too high, 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 D.7 should be considered and also the equipment impairments which are caused by 32 kbit/sec ADPCM equipment disturbing also the far-end termination. The far-end termination is now assigned 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 $\text{SLR} = 7 \text{ dB}$ (equal to default) and $\text{RLR} = 4 \text{ dB}$.

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 $> 46 \text{ dB}$ (this option is not indicated in figure D.7 with respect to the echo path). Now the TELR is formed by the $\text{SLR} = 8 \text{ dB}$ and $\text{RLR} = -3 \text{ dB}$ 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 $\text{TELR} = 58 \text{ dB}$. 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 D.7 and indicated by echo path 2). The whole mean one-way delay of echo path 2 is $T = 35 \text{ ms}$. The TELR can be calculated with 36 dB . For the equipment impairments the same value of $\text{Ie} = 14$ is effective as for the subscriber of the private network. The input parameters for the E-Model are

$$\begin{aligned} \text{SLR} &= 7 \text{ dB} \quad (\text{equal to the default value}) \\ \text{RLR} &= 4 \text{ dB} \\ \text{TELR} &= 36 \text{ dB} \\ \text{T} &= 35 \text{ ms} \\ \text{Ie} &= 14 \quad (\text{for two times 32 kbit/sec ADPCM}) \end{aligned}$$

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

R	Itot	Is	Id	Ie
59,1	35,2	0,3	20,3	14

This result of $I_{\text{tot}} = 35,2$ may be already considered as low quality and not be accepted for the benefit of the subscriber at the far-end termination, i.e. echo control devices should be inserted, since also here the major impairment with $\text{Id} = 20,3$ is caused by echo.

Reference Configuration 3

For the investigation of an internal connection fully 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 rerouting 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 D.8.

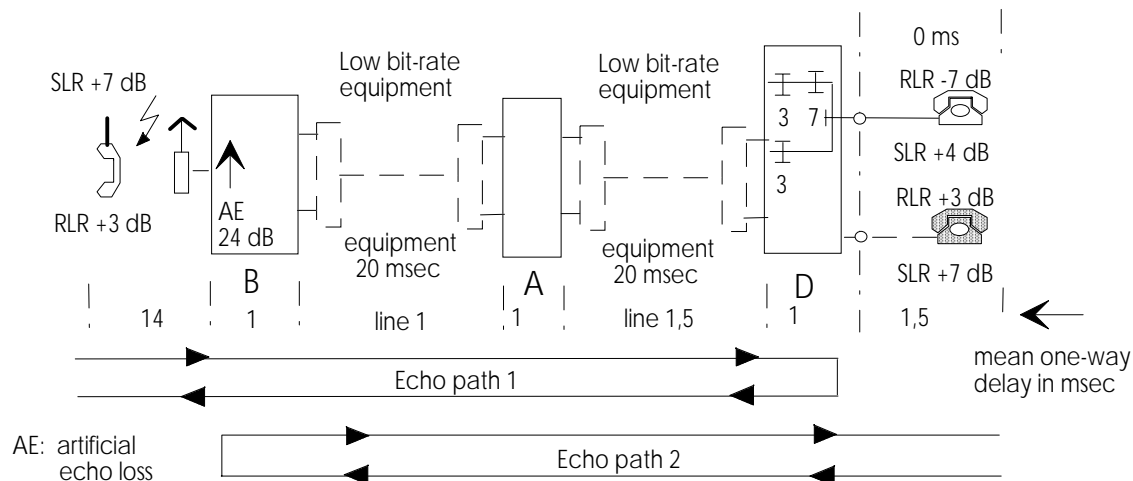


Figure D.8: Basis for Calculation of Reference Configuration 3

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 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 center 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 along echo path 1 as indicated in figure D.8 is $T = 59,5$ ms. 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 equipment impairment values a 32 kbit/sec ADPCM coding is used in three different sections cumulating to a value of $I_e = 21$. The input parameters to be set are as follows with all other parameters at default:

- SLR = 7 dB (equal to the default value)
- RLR = 3 dB (equal to the default value)
- TELR = 50 dB
- T = 59,5 ms
- $I_e = 21$ (for three times 32 kbit/sec ADPCM)

The result of the calculation is:

R	Itot	Is	Id	Ie
66,5	27,8	0,5	7,0	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 both at their default values with SLR = 7 dB and RLR = 3 dB. For the mean one-way delay along echo path 2 of figure D.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. 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. 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	(equal to the default value)
RLR	=	3 dB	(equal to the default value)
TELR	=	34 dB	
T	=	47 ms	
Ie	=	21	(for three times 32 kbit/sec ADPCM)

The result of the calculation is:

R	Itot	Is	Id	Ie
42,4	51,9	0,5	30,1	21

D.1.5 Analysis of the Results

For a better overview all results of the different calculations for the three reference configurations are repeated in table D.1 for the main impairment values Itot, 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 D.1: Summary of the Calculation Results

Reference Configuration	Private network			Opposite termination		
	Itot	Id	Ie	Itot	Id	Ie
1	33,1	25	7	16,9	9	7
2	28,6	13,5	14	35,2	20,3	14
3	27,8	7	21	51,9	30,1	21

The results for the expected quality for the different configurations is varying in a wide range from a value Itot = 16,9 - which would be judged as being between good and adequate - and Itot = 51,9 which is already outside the recommended upper limit for Itot. Most of the values are in a range from 25 to 35 which is medium quality only, or may at the higher values cause complaints. The major impairments for these reference configurations are due to echo and equipment impairment caused by the use of 32 kbit/sec ADPCM equipment. Considering the results in table D.1, the previous decision to avoid the use of 24 kbit/sec 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 high values for Itot 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 to Id = 0 if echo control devices are used. This would also reduce the total impairments by nearly the same amount as the present Id values. 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.

D.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, annex A subclause 5.3 and in annex C subclause C.1.1.9

When investigating the use of echo control devices in this private network the following questions should be raised 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 about 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 subclause D.1.3), no echo cancellers are inserted within the public network in national long distance calls. Furthermore the low bit-rate equipment are 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 rerouting, 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 $2 \times 3 \text{ ms} = 6 \text{ ms}$ 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 ITU-T Recommendation G.165 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 (fixed) 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 about the application of echo cancelling devices is illustrated in figure D.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.

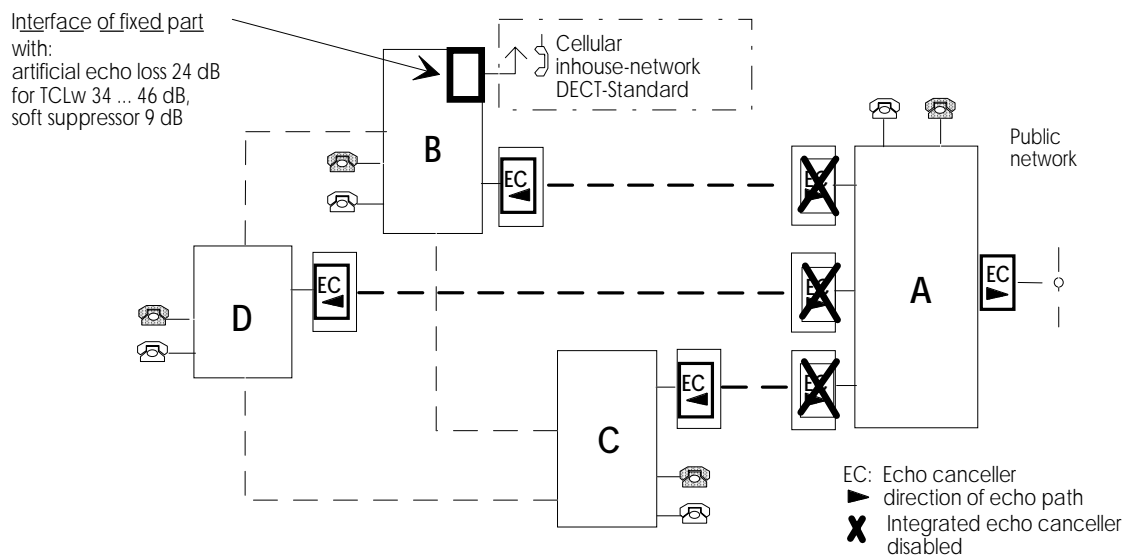


Figure D.9: Application of Echo Control Devices in the Private Network

For the cordless telephones in B, the artificial echo loss should be enabled for connections to those portable parts with $TCL_w = 34$ 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.

This investigations about 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 and the talker echo loudness rating 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 D.1 for all reference configurations, it can be assumed with sufficient accuracy, that all values for I_{tot} are reduced by the amount of the I_d values. The only impairments remaining are the equipment impairments I_e , which can be up to $I_e > 21$ for reference configuration 3. However it should be noted, that this configuration was defined as a critical connection including a rerouting with an additional low bit-rate section. Depending on other features of this private network, these rerouting will not necessarily be the "standard" routing for most of the calls.

D.2 Planning Example for the North American Scenarios

The following examples will demonstrate how to perform transmission planning according to the principles of the present document.

D.2.1 Definition of Reference Configurations

As in clause D.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 D.10. The connections are patterned after the network in figure D.1.

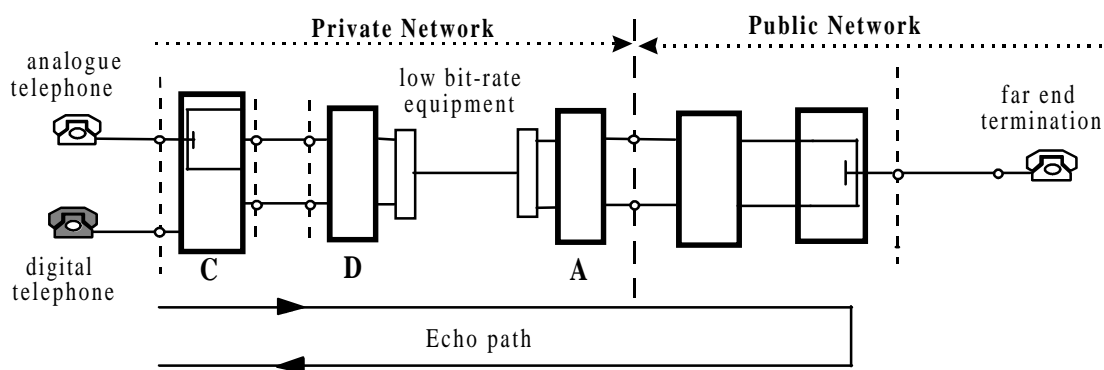


Figure D.10: Reference Configuration 1

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 D.11.

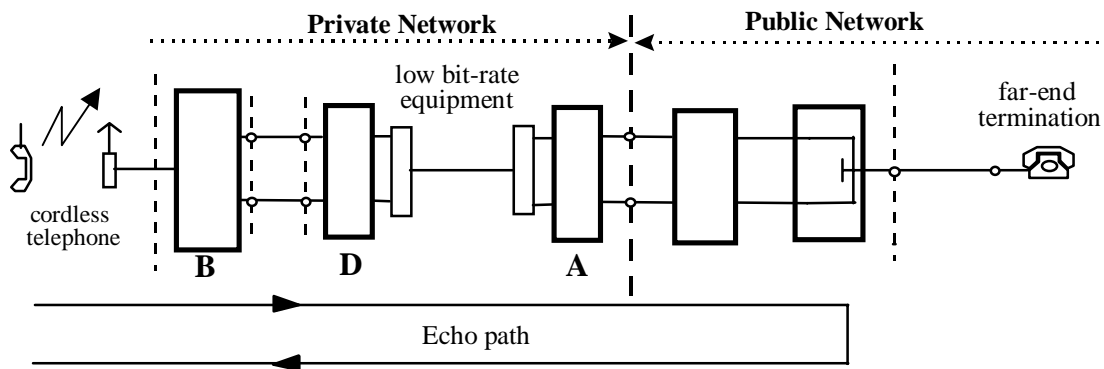


Figure D.11: Reference Configuration 2

For comparison with the above connection and consistency with the analysis in clause D.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 D.12.

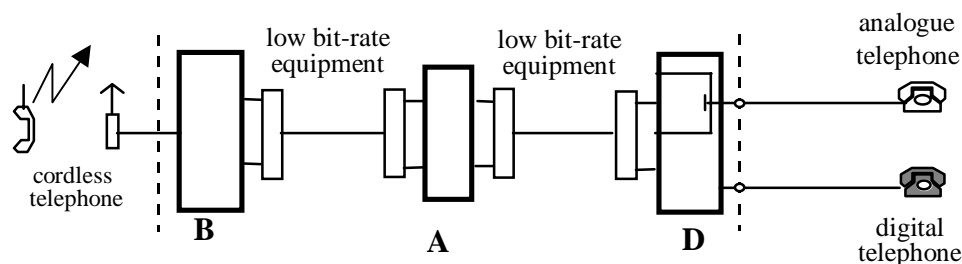


Figure D.12: Reference Configuration 3

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.

D.2.2 Determination of the Transmission Parameters

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

D.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 the present document, the calculations are made for each of the three reference configurations separately. The configurations are illustrated in the diagrams shown prior to the precalculation section.

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

D.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 clause D.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 a 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 TIA-464B for ONS-IST connection. This loss is varied in 0,5 dB steps (V) to obtain a plot of voice quality vs. 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 TIA-464B, 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.

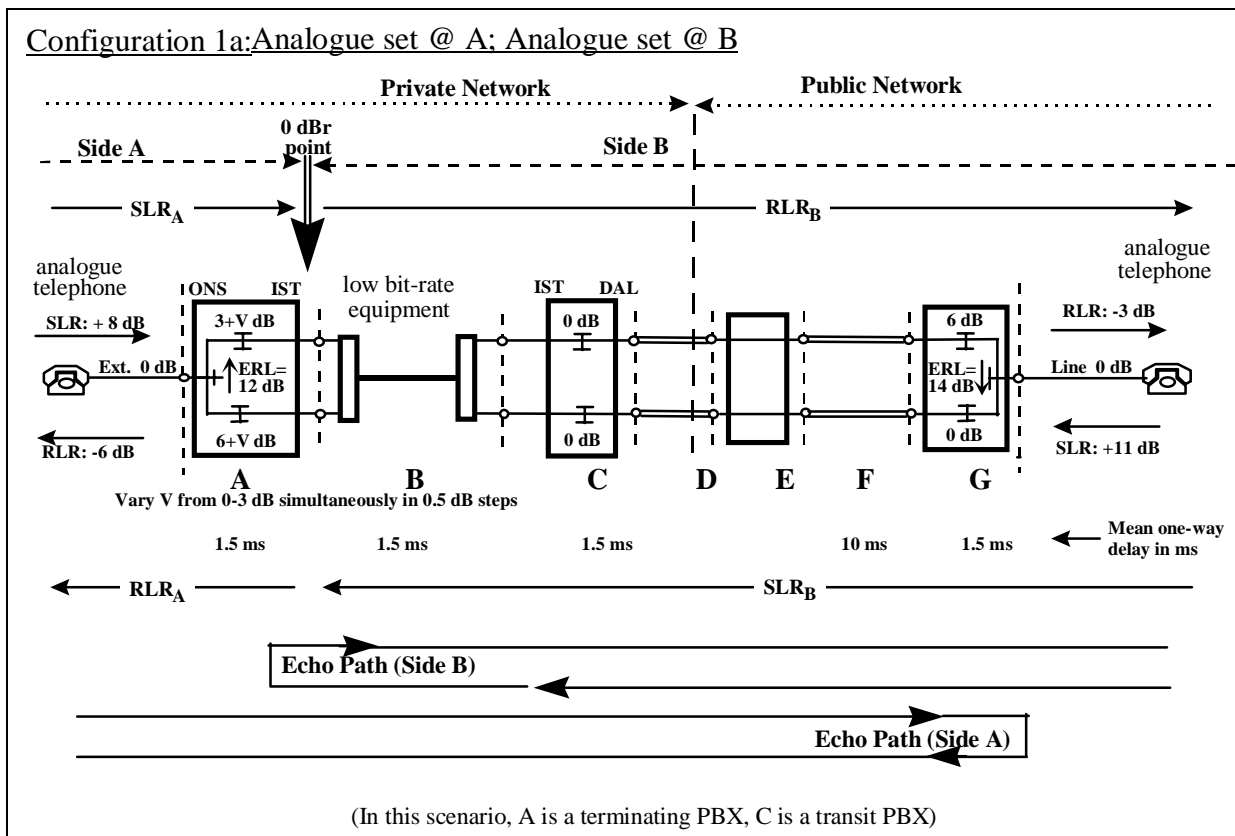


Figure D.13: Basis for Calculation for Reference Configuration 1a

D.2.4.1 Precalculations for Configuration 1a

D.2.4.1.1 Calculation of SLR, RLR

<p>Side A Telephone Set PBX A Sum at 0 dBr point</p>	<p>Set → 0 dBr point SLR 8 dB 3 + V dB SLR_A = 11+V dB</p>	<p>0 dBr point → Set RLR -6 dB 6 + V dB RLR_A = V dB</p>
<p>Side B Telephone Set DEO G DEO E Sum at 0 dBr point</p>	<p>Set → 0 dBr point SLR 11 dB 0 dB 0 dB SLR_B = 11 dB</p>	<p>0 dBr point → Set RLR -3dB 6 dB 0 dB RLR_B = 3 dB</p>
<p>Listener at Side A SLR = 11 dB RLR(base) = V dB OLR = 11 + V dB</p>	<p>Listener at Side B SLR = 11+ V dB RLR(base) = 3 dB OLR = 11 + V + 3 = 14 + V dB</p>	

D.2.4.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

$$\text{TELR} = \text{SLR}_{\text{SET A}} + 2 \times \text{Facility Loss} + \text{Return Loss (B)} + \text{sum of node losses} + \text{RLR}_{\text{SET A}}$$

$$\text{TELR}(\text{base}) = 8 + 2 \times 0 + 14 + (3 + V + 0 + 6 + 0 + 0 + 6 + V) + (-6) = 31 + 2 V \text{ dB}$$

Analysis of TELR performed at Side B:

$$\text{TELR} = \text{SLR}_{\text{SET B}} + 2 \times \text{Facility Loss} + \text{Return Loss (A)} + \text{sum of node losses} + \text{RLR}_{\text{SET B}}$$

$$\text{TELR}(\text{base}) = 11 + 2 \times 0 + 12 + (0 + 0 + 6 + V + 3 + V + 0 + 6) + (-3) = 35 + 2 V \text{ dB}$$

where as above, Facility Loss is the sum of the line losses in each direction.

D.2.4.1.3 Delay Values T, Ta and Tr

$$\begin{aligned} T(\text{ms}) &= \text{Sum of PBXs delay} + \text{Low bit-rate equipment delay} + \text{Digital End Office delay} + \text{Network delay} \\ &\quad + \text{Digital End Office delay} \end{aligned}$$

$$= 1,5 + 1,5 + 1,5 + 10 + 1,5$$

$$= 16$$

$$T_a = T$$

$$T_r = 0$$

D.2.4.1.4 Number of Quantization Distortion

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 2 digital loss pads if Side A is the listener and 2 digital pads if Side B is the listener.

$$\text{Therefore, } qdu_A = n \times 1 + m \times 0,7$$

$$= 1 \times 1 + 1 \times 0,7$$

$$= 1,7$$

$$qdu_B = n \times 1 + m \times 0,7$$

$$= 1 \times 1 + 2 \times 0,7$$

$$= 2,4$$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

D.2.4.1.5 Equipment Impairment Factor

In the above example, a low-bit rate codec was used in section B. This is the section within the private network. Typically, ADPCM 32kb/s is used. This codec does have a mild degradation to voice, and according to table 7 of ITU-T Recommendation G.113, the value of the impairment factor, I_e is 7.

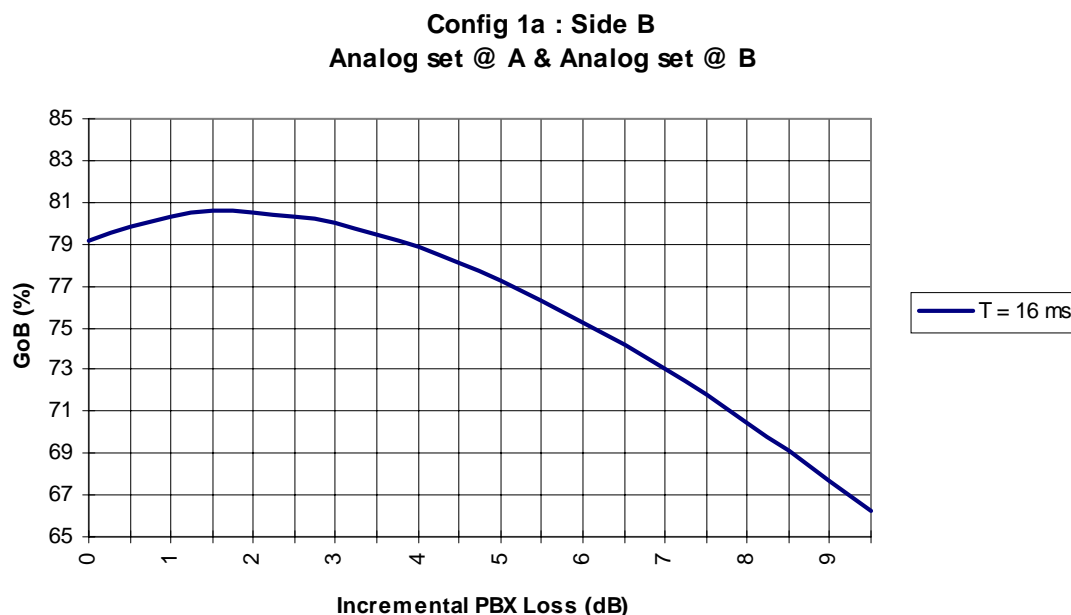
D.2.4.2 Input Parameters Summary Table

The input parameters for configuration 1a are summarized in table D.2. Note the column "Reference Side" for those that have an "*" refers to the specific input parameter for the analysis to be performed at that particular side.

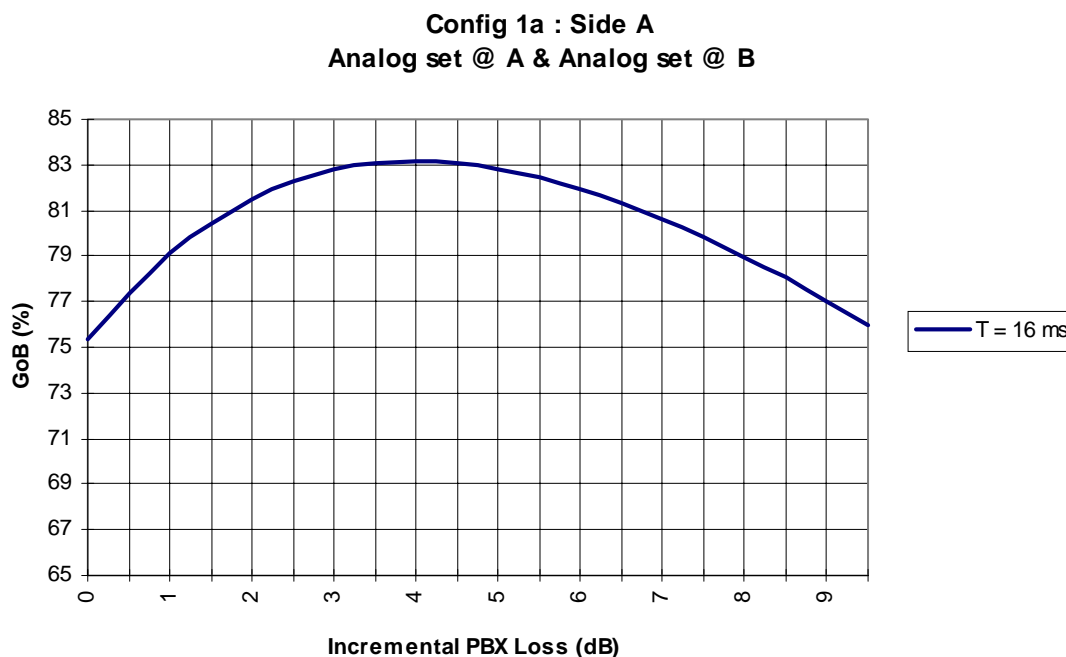
Table D.2: E-Model Input Parameters for Configuration 1a

Reference Side	Input Parameters	Values	Unit
Side A *	Send Loudness Rating (SLR)	11 + V	dB
Side B *	Send Loudness Rating (SLR)	11	dB
Side A	Room Noise (Ps)	35 (default)	dB(A)
Side A	Ds factor (Ds)	3 (default)	
Side A *	Receive Loudness Rating (RLR)	V	dB
Side B *	Receive Loudness Rating (RLR)	3	dB
Side B	Room Noise (Pr)	35 (default)	dB(A)
Side B	Ds factor (Dr)	3 (default)	
Side B	Sidetone Masking Rating (STMR)	15 (default)	dB
Side B	Listener Sidetone Rating (LSTR)	18 (default)	dB
Side A *	Talker Echo Loudness Rating	31 + 2V	dB
Side B *	Talker Echo Loudness Rating	35 + 2V	dB
Side B	Weighted Echo Path Loss	80 (default)	dB
Side A *	Circuit Noise (Nc)	-70	dBm0p
Side B *	Circuit Noise (Nc)	-70	dBm0p
Overall	Noise Floor (Nfor)	-64 (default)	dBm0p
Overall	Equipment Impairment Factor (Ie)	0	
Overall	Advantage factor (A)	0	
Overall	Quantization distortion Unit	1,7 (A), 2,4 (B)	
Overall	Round Trip delay	0 (default)	ms
Overall	Mean one-way delay	16 (independent variable)	ms
Overall	Absolute Delay	16 (independent variable)	ms

D.2.4.3 Results - Configuration 1a



The graph above shows optimal voice quality can be achieved when PBX incremental loss is in the range of 2 to 2,5 dB. The absolute optimum performance is somewhat less than for Side A (below); this is due to the somewhat greater (14 dB vs. 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.



Optimal voice quality can be achieved when incremental PBX loss is in the range of 3,5 dB to 4 dB.

D.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 clause D.1 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 a Integrated Services Trunk (IST) port. The PBX base loss is 0 dB in each direction, in accordance with the loss plan of TIA-464B for ONS-IST connection. This loss is varied in 0,5 dB steps (V) to obtain a plot of voice quality vs. 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 TIA-464B, 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.

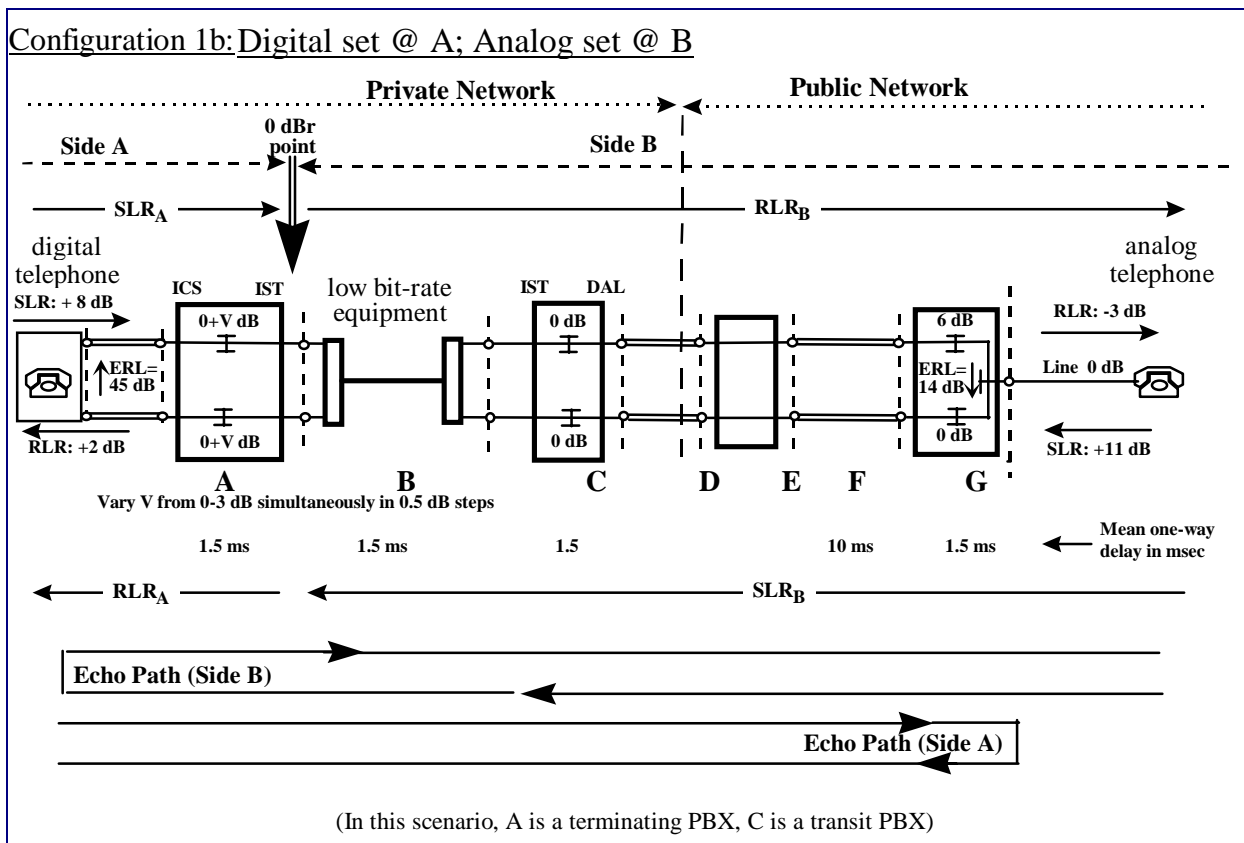


Figure D.14: Basis for Calculation of Reference Configuration 1b

D.2.5.1 Precalculations for Configuration 1b

D.2.5.1.1 Calculation of SLR, RLR

<p>Side A Telephone Set PBX A Sum at 0 dB point</p>	<p>Set → 0 dB point SLR 8 dB 0 + V dB SLR_A = 8+ V dB</p>	<p>0 dB point → Set RLR 2 dB 0 + V dB RLR_A = 2 + V dB</p>
<p>Side B Telephone Set DEO G DEO E Sum at 0 dB point</p>	<p>Set → 0 dB point SLR 11 dB 0 dB 0 dB SLR_B = 11 dB</p>	<p>0 dB point → Set RLR -3dB 6 dB 0 dB RLR_B = 3 dB</p>
<p>Listener at Side A SLR = 11 dB RLR(base) = 2 + V dB OLR = 13+ V dB</p>	<p>Listener at Side B SLR = 8+ V dB RLR(base) = 3 dB OLR = 8 + V + 3 = 11 + V dB</p>	

D.2.5.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

$$TELR = SLR_{SET A} + 2 \times \text{Facility Loss} + \text{Return Loss (B)} + \text{sum of node losses} + RLR_{SET A}$$

$$TELR(\text{base}) = 8 + 2 \times 0 + 14 + (0 + V + 0 + 6 + 0 + 0 + V) + (2) = 30 + 2 V \text{ dB}$$

Analysis of TELR performed at Side B:

$$\text{TELR} = \text{SLR}_{\text{SET B}} + 2 \times \text{Facility Loss} + \text{Return Loss (A)} + \text{sum of node losses} + \text{RLR}_{\text{SET B}}$$

$$\text{TELR}(\text{base}) = 11 + 2 \times 0 + 46 + (6 + 0 + 0 + V + 0 + V + 0 + 0) + (-3) = 60 + 2V \text{ dB}$$

where as above, Facility Loss is the sum of the line losses in each direction.

D.2.5.1.3 Delay Values T, Ta and Tr

$$\begin{aligned} T(\text{ms}) &= \text{Sum of PBXs delay} + \text{Low bit-rate equipment delay} + \text{Digital End Office delay} + \text{Network delay} \\ &\quad + \text{Digital End Office delay} \end{aligned}$$

$$= 1,5 + 1,5 + 1,5 + 10 + 1,5$$

$$= 16$$

$$T_a = T$$

$$T_r = 0$$

D.2.5.1.4 Number of Quantization Distortion

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 two digital loss pads if Side A is the listener and two digital pads if Side B is the listener.

$$\text{Therefore, } qdu_A = n \times 1 + m \times 0,7$$

$$= 1 \times 1 + 1 \times 0,7$$

$$= 1,7$$

$$qdu_B = n \times 1 + m \times 0,7$$

$$= 1 \times 1 + 2 \times 0,7$$

$$= 2,4$$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

D.2.5.1.5 Equipment Impairment Factor

In the above example, a low-bit rate codec is used in section B within the private network. This is the section within the private network. Typically, ADPCM 32kb/s is used. This codec does have a mild degradation to voice, and according to table 7 of ITU-T Recommendation G.113, the value of the impairment factor, I_e is 7.

D.2.5.2 Input Parameters Summary Table

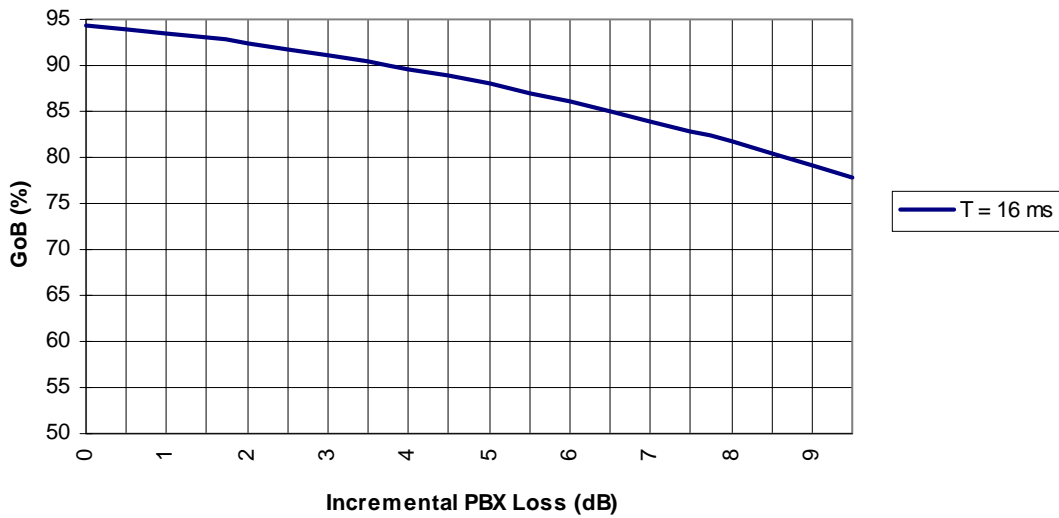
The input parameters for configuration 1b are summarized in table D.3. Note the column "Reference Side" for those that have an "*" refers to the specific input parameter for the analysis to be performed at that particular side.

Table D.3: E-Model Input Parameters for Config 1b

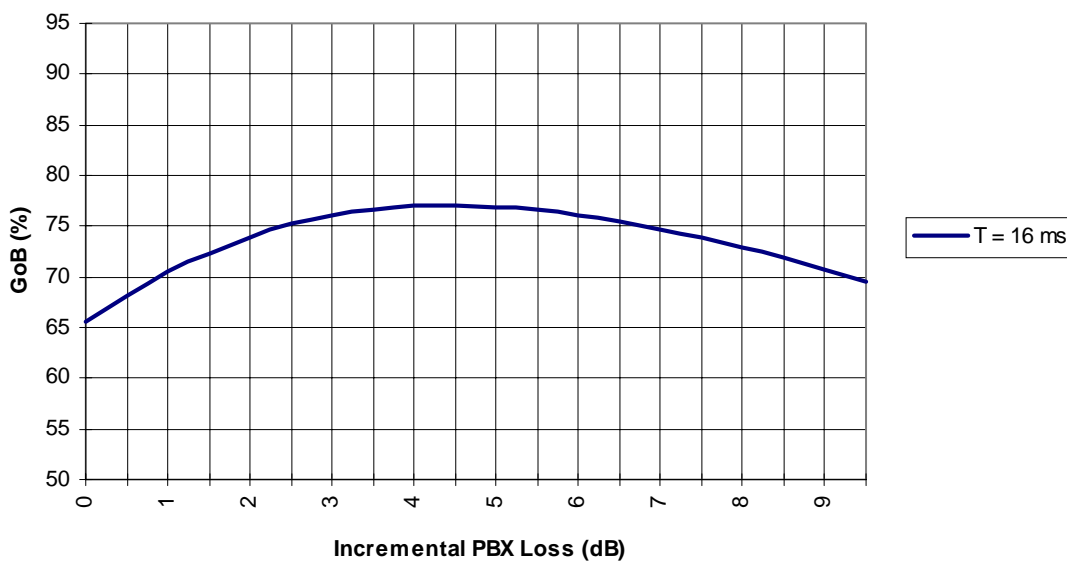
Reference Side	Input Parameters	Values	Unit
Side A *	Send Loudness Rating (SLR)	8 + V	dB
Side B *	Send Loudness Rating (SLR)	11	dB
Side A	Room Noise (Ps)	35 (default)	dB(A)
Side A	Ds factor (Ds)	3 (default)	
Side A *	Receive Loudness Rating (RLR)	2 + V	dB
Side B *	Receive Loudness Rating (RLR)	3	dB
Side B	Room Noise (Pr)	35 (default)	dB(A)
Side B	Ds factor (Dr)	3 (default)	
Side B	Sidetone Masking Rating (STMR)	15 (default)	dB
Side B	Listener Sidetone Rating (LSTR)	18 (default)	dB
Side A *	Talker Echo Loudness Rating	30 + 2V	dB
Side B *	Talker Echo Loudness Rating	60 + 2V	dB
Side B	Weighted Echo Path Loss	80 (default)	dB
Side A *	Circuit Noise (Nc)	-70	dBm0p
Side B *	Circuit Noise (Nc)	-70	dBm0p
Overall	Noise Floor (Nfor)	-64 (default)	dBm0p
Overall	Equipment Impairment Factor (Ie)	7	
Overall	Advantage factor (A)	0	
Overall	Quantization distortion Unit	1,7 (A), 2,4 (B)	
Overall	Round Trip delay	0 (default)	ms
Overall	Mean one-way delay	16 (independent variable)	ms
Overall	Absolute Delay	16 (independent variable)	ms

D.2.5.3 Results

Config 1b: Side B
Digital set @ A & Analog set @ B



Config 1b : Side A
Digital set @ A & Analog set @ B



The graphs above illustrate optimal voice quality points at respective incremental losses at the PBXs. For Side B, the optimal voice 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 dB to 4,5 dB indicates the best possible voice quality to be obtained.

D.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 clause D.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 TIA-464B 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 TIA-464B, 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.

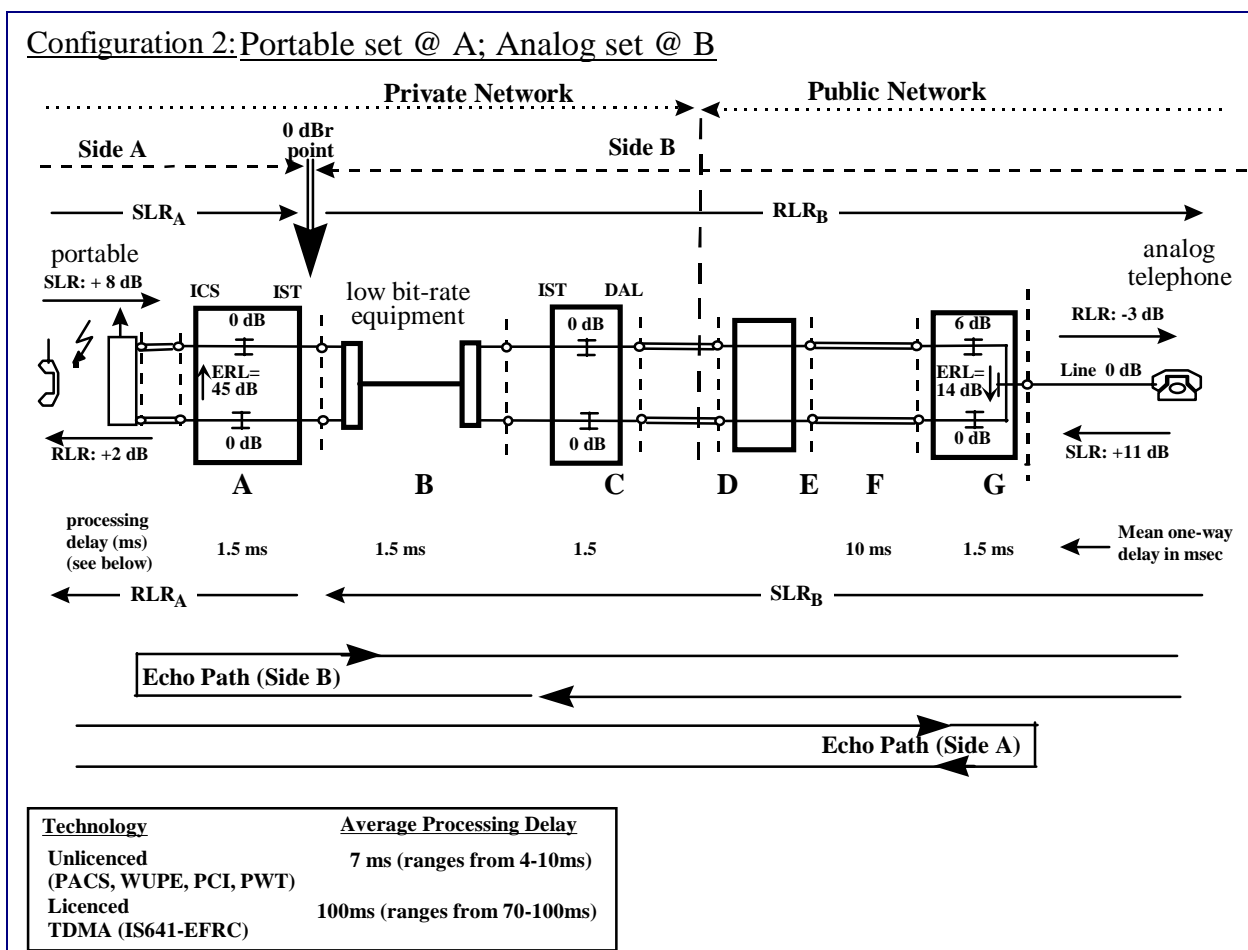


Figure D.15: Basis for Calculation of Reference Configuration 2

D.2.6.1 Precalculations for Configuration 2

D.2.6.1.1 Calculation of SLR, RLR

Side A Telephone Set PBX A Sum at 0 dBr point	Set → 0 dBr point SLR 8 dB 0 dB SLR_A = 8 dB	0 dBr point → Set RLR 2 dB 0 dB RLR_A = 2 dB
Side B Telephone Set DEO G DEO E Sum at 0 dBr point	Set → 0 dBr point SLR 11 dB 6 dB 0 dB SLR_B = 17 dB	0 dBr point → Set RLR -3dB 0 dB 0 dB RLR_B = -3 dB

Listener at Side A
 SLR = 17 dB
 RLR(base) = 2 dB
 OLR = 19 dB

Listener at Side B
 SLR = 8 dB
 RLR(base) = -3 dB
 OLR = 8 + (-3) = 5 dB

D.2.6.1.2 Talker Echo Loudness Rating (TELRL)

Analysis of TELRL performed at Side A:

$$\text{TELRL} = \text{SLR}_{\text{SET A}} + 2 \times \text{Facility Loss} + \text{Return Loss (B)} + \text{sum of node losses} + \text{RLR}_{\text{SET A}}$$

$$\text{TELRL}(\text{base}) = 8 + 2 \times 0 + 14 + (0 + 0 + 0 + 6 + 0 + 0) + (2) = 30 \text{ dB}$$

Analysis of TELRL performed at Side B:

$$\text{TELRL} = \text{SLR}_{\text{SET B}} + 2 \times \text{Facility Loss} + \text{Return Loss (A)} + \text{sum of node losses} + \text{RLR}_{\text{SET B}}$$

$$\text{TELRL}(\text{base}) = 11 + 2 \times 0 + 46 + (6 + 0 + 0 + 0 + 0 + 0) + (-3) = 60 \text{ dB}$$

where as above, Facility Loss is the sum of the line losses in each direction.

D.2.6.1.3 Delay Values T, Ta and Tr

$$T(\text{ms}) = \text{Access Delay of wireless technology} + \text{PBX delay} + \text{Low bit-rate equipment delay} + \text{Digital End Office delay} \\ + \text{Network delay} + \text{Digital End Office delay}$$

$$= 7 \text{ (unlicensed) or } 100 \text{ (licensed)} + 1,5 + 1,5 + 1,5 + 10 + 1,5$$

$$= 23 \text{ or } 116$$

$$T_a = T$$

$$T_r = 0$$

D.2.6.1.4 Number of Quantization Distortion

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 1 digital loss pad if Side A is the listener and 0 digital pads if Side B is the listener.

$$\begin{aligned} \text{Therefore, } qdu_A &= n \times 1 + m \times 0,7 \\ &= 1 \times 1 + 1 \times 0,7 \\ &= 1,7 \end{aligned}$$

$$\begin{aligned} qdu_B &= n \times 1 + m \times 0,7 \\ &= 1 \times 1 + 0 \times 0,7 \\ &= 1 \end{aligned}$$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

D.2.6.1.5 Equipment Impairment Factor

The codec used for wireless technology does have an impact on voice quality. For the two types of technology analyzed, values in table 7 of ITU-T Recommendation G.113, and a value of 10 for IS641-EFRC were used. The total impairment involved is the sum of all the impairment factors involved:

$$\begin{aligned} \text{Total impairment factor} &= \text{Impairment due to codec used in wireless technology} + \text{network low-bit rate equipment} \\ &= 7 \text{ (unlicensed) or } 10 \text{ (licensed)} + 7 \\ &= 14 \text{ or } 17 \end{aligned}$$

D.2.6.2 Input Parameters Summary Table

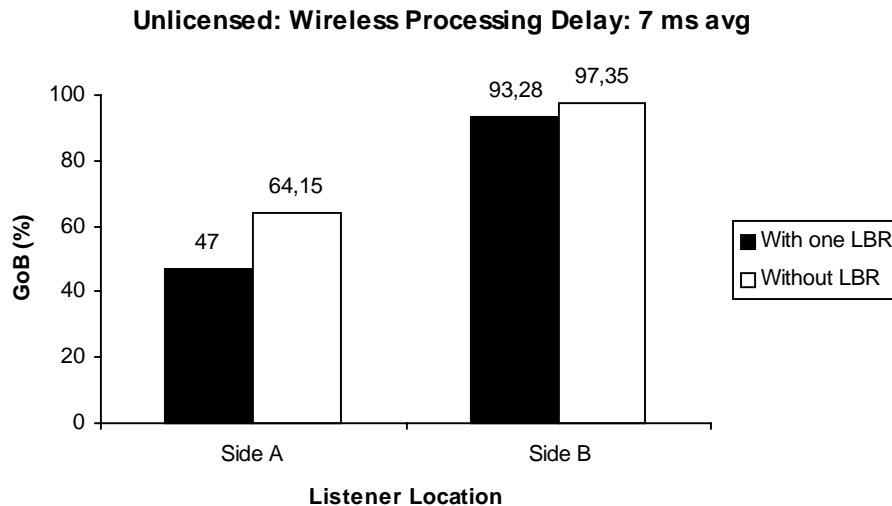
The input parameters for configuration 2 are summarized in table D.4. Note the column "Reference Side" for those that have an "*" refers to the specific input parameter for the analysis to be performed at that particular side.

Table D.4: E-Model Input Parameters for Config 2

Reference Side	Input Parameters	Values	Unit
Side A *	Send Loudness Rating (SLR)	17	dB
Side B *	Send Loudness Rating (SLR)	8	dB
Side A	Room Noise (Ps)	35 (default)	dB(A)
Side A	Ds factor (Ds)	3 (default)	
Side A *	Receive Loudness Rating (RLR)	2	dB
Side B *	Receive Loudness Rating (RLR)	-3	dB
Side B	Room Noise (Pr)	35 (default)	dB(A)
Side B	Ds factor (Dr)	3 (default)	
Side B	Sidetone Masking Rating (STMR)	15 (default)	dB
Side B	Listener Sidetone Rating (LSTR)	18 (default)	dB
Side A *	Talker Echo Loudness Rating	30	dB
Side B *	Talker Echo Loudness Rating	59	dB
Side B	Weighted Echo Path Loss	80 (default)	dB
Side A *	Circuit Noise (Nc)	-70	dBm0p
Side B *	Circuit Noise (Nc)	-70	dBm0p
Overall	Noise Floor (Nfor)	-64 (default)	dBm0p
Overall	Equipment Impairment Factor (Ie)	14 (unlicensed), 17 (licensed)	
Overall	Advantage factor (A)	5 (unlicensed), 10 (licensed)	
Overall	Quantization distortion Unit	1,7 (A), 1 (B)	
Overall	Round Trip delay	0 (default)	ms
Overall	Mean one-way delay	13 (unlicensed) 106 (licensed)	ms
Overall	Absolute Delay	13 (unlicensed) 106 (licensed)	ms

D.2.6.3 Results

D.2.6.3.1 Unlicensed: Wireless Processing Delay: 7 ms average (WUPE, PCI, PWT)



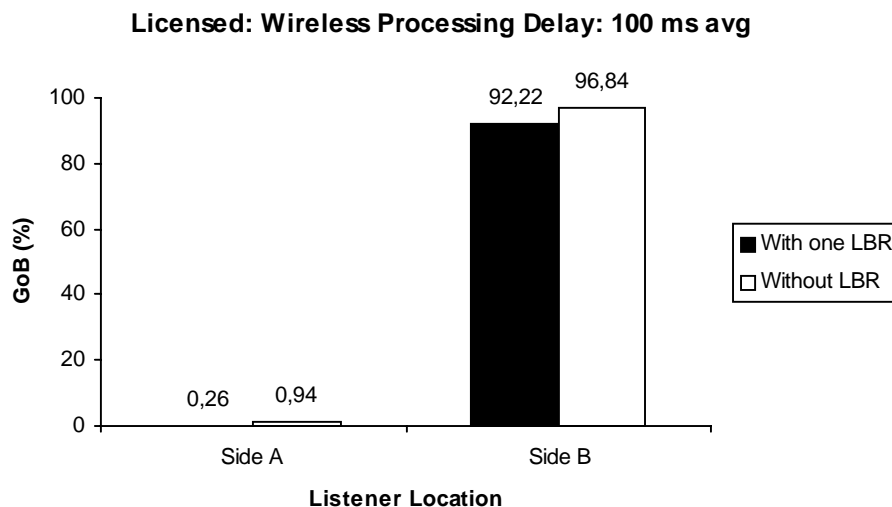
NOTE: LBR denotes Low bit-rate equipment.

Figure D.15a: Unlicensed: Wireless Processing Delay: 7ms avg

The graph above shows voice 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 voice 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 voice quality.

It can also be observed that the low bit-rate equipment degrades voice quality which can make a difference, especially when echo cancellation is not handled effectively (Larger differences at Side A compared with Side B).

D.2.6.3.2 Licensed: Wireless Processing Delay: 100 ms average (TDMA-IS641-EFRC)



NOTE: LBR denotes Low bit-rate equipment.

Figure D.15b: Licensed: Wireless Processing Delay: 100 ms avg

The same observation is shown for licensed technology. The only difference is that the absolute number in terms of performance is much lower than the unlicensed wireless technology. The reason is that licensed wireless technology has much longer delay (100 ms versus 7 ms average), and the impairment factor of the associated low-bit rate codec is larger.

D.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 clause D.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 TIA-464B 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 TIA-464B, the loss for this PBX connection is 0 dB in both direction. 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.

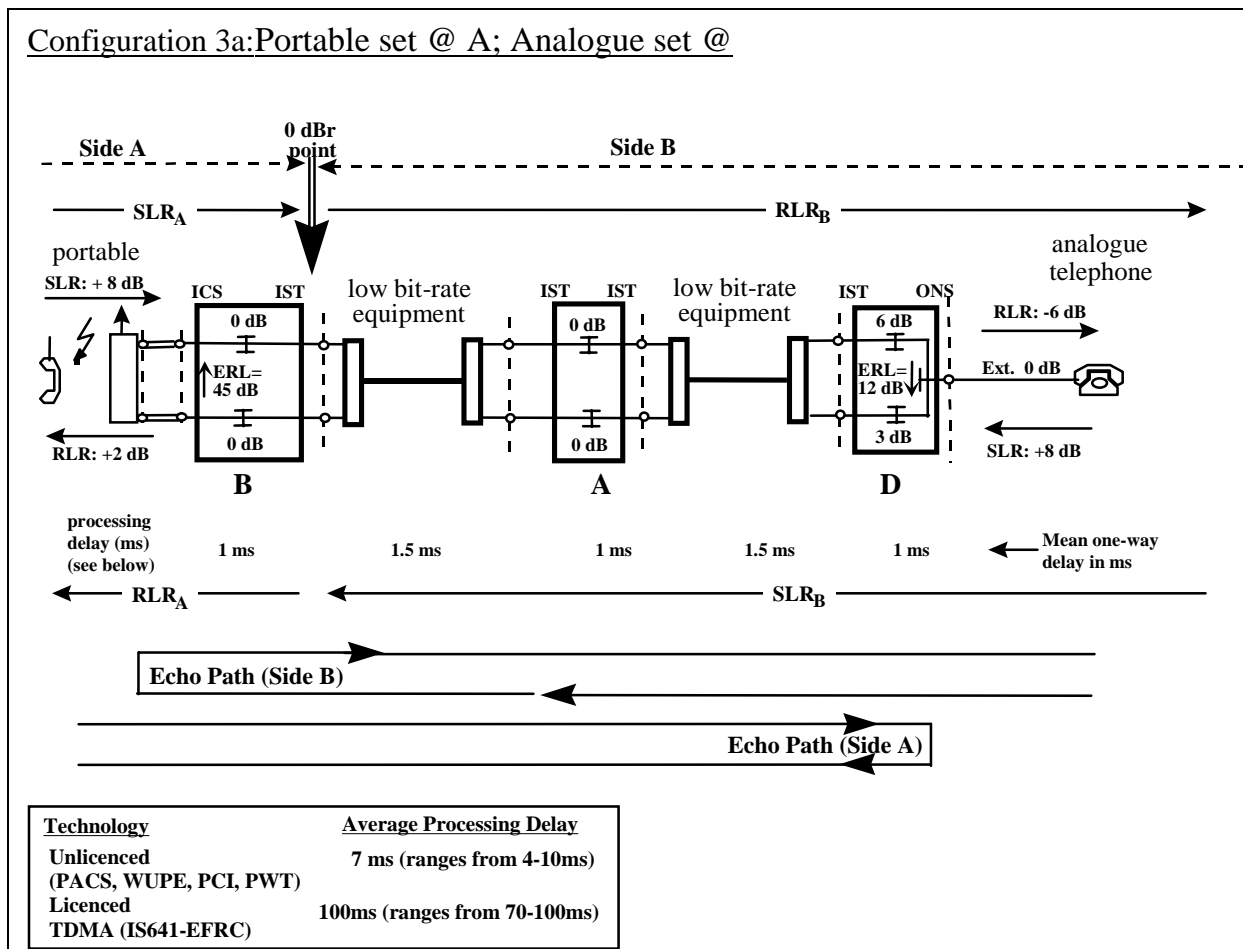


Figure D.16: Basis for Calculation of Reference Configuration 3a

D.2.7.1 Precalculations for Configuration 3a

D.2.7.1.1 Calculation of SLR, RLR

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 -6dB
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

Listener at Side A
 SLR = 11 dB
 RLR(base) = 2 dB
 OLR = 13 dB

Listener at Side B
 SLR = 8 dB
 RLR(base) = 0 dB
 OLR = 8 + 0 = 8 dB

D.2.7.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

$$\text{TELR} = \text{SLR}_{\text{SET A}} + 2 \times \text{Facility Loss} + \text{Return Loss (B)} + \text{sum of node losses} + \text{RLR}_{\text{SET A}}$$

$$\text{TELR}(\text{base}) = 8 + 2 \times 0 + 12 + (0 + 0 + 6 + 3 + 0 + 0) + (2) = 31 \text{ dB}$$

Analysis of TELR performed at Side B:

$$\text{TELR} = \text{SLR}_{\text{SET B}} + 2 \times \text{Facility Loss} + \text{Return Loss (A)} + \text{sum of node losses} + \text{RLR}_{\text{SET B}}$$

$$\text{TELR}(\text{base}) = 8 + 2 \times 0 + 45 + (3 + 0 + 0 + 0 + 0 + 6) + (-6) = 56 \text{ dB}$$

where as above, Facility Loss is the sum of the line losses in each direction.

D.2.7.1.3 Delay Values T, Ta and Tr

$$T(\text{ms}) = \text{Access Delay of wireless technology} + \Sigma \text{PBX delay / End Office Delay} + \Sigma \text{Low bit-rate equipment delay}$$

$$= 7 \text{ (unlicensed) or } 100 \text{ (licensed)} + 3 \times 1 + 2 \times 1,5$$

$$= 13 \text{ or } 106$$

$$T_a = T$$

$$T_r = 0$$

D.2.7.1.4 Number of Quantization Distortion

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 2 digital loss pads for either side analysis.

$$\text{Therefore, } qdu = n \times 1 + m \times 0,7$$

$$= 1 \times 1 + 1 \times 0,7$$

$$= 1,7$$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

D.2.7.1.5 Equipment Impairment Factor

The codec used for wireless technology does have an impact on voice quality. For the two types of technology analyzed, we used values as quoted in ITU-T Recommendation G.113 of table 7; and 10 (see note) for IS641-EFRC. The total impairment involved is the sum of all the impairment factors involved:

$$\text{Total impairment factor} = \text{Impairment due to codec used in wireless technology} + \text{network low-bit rate equipment}$$

$$= 7 \text{ (unlicensed) or } 10 \text{ (licensed)} + 7 \text{ (1 low-bit rate equipment)}$$

$$\text{or } 14 \text{ (2 low-bit rate equipment)}$$

$$= 14 \text{ or } 21 \text{ (licensed)}$$

$$= 17 \text{ or } 24 \text{ (unlicensed)}$$

NOTE: Provisionally.

D.2.7.2 Input Parameters Summary Table

The input parameters for Configuration 3a (analogue set @ B) are summarized in table D.5. Note the column "Reference Side" for those that have an "*" refers to the specific input parameter for the analysis to be performed at that particular side.

Table D.5: E-Model Input Parameters for Config 3a (analogue set @B)

Reference Side	Input Parameters	Values	Unit
Side A *	Send Loudness Rating (SLR)	11	dB
Side B *	Send Loudness Rating (SLR)	8	dB
Side A	Room Noise (Ps)	35 (default)	dB(A)
Side A	Ds factor (Ds)	3 (default)	
Side A *	Receive Loudness Rating (RLR)	2	dB
Side B *	Receive Loudness Rating (RLR)	0	dB
Side B	Room Noise (Pr)	35 (default)	dB(A)
Side B	Ds factor (Dr)	3 (default)	
Side B	Sidetone Masking Rating (STMR)	15 (default)	dB
Side B	Listener Sidetone Rating (LSTR)	18 (default)	dB
Side A *	Talker Echo Loudness Rating	31	dB
Side B *	Talker Echo Loudness Rating	56	dB
Side B	Weighted Echo Path Loss	80 (default)	dB
Side A *	Circuit Noise (Nc)	-70	dBm0p
Side B *	Circuit Noise (Nc)	-70	dBm0p
Overall	Noise Floor (Nfor)	-64 (default)	dBm0p
Overall	Equipment Impairment Factor (Ie)	14/21 (unlicensed), 17/24 (licensed)	
Overall	Advantage factor (A)	0	
Overall	Quantization distortion Unit	1,7	
Overall	Round Trip delay	0 (default)	ms
Overall	Mean one-way delay	13 (unlicensed) 106 (licensed)	ms
Overall	Absolute Delay	13 (unlicensed) 106 (licensed)	ms

D.2.7.3 Results

D.2.7.3.1 Unlicensed: Wireless Processing Delay: 7 ms average (WUPE, PCI, PWT)

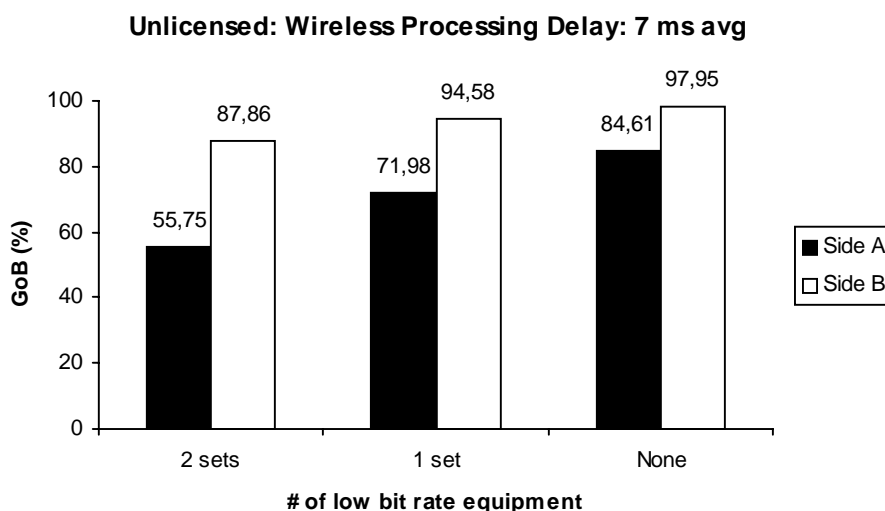


Figure D.16a: Unlicensed: Wireless Processing Delay: 7 ms avg

D.2.7.3.2 Licensed: Wireless Processing Delay: 100 ms average (TDMA-IS641-EFRC)

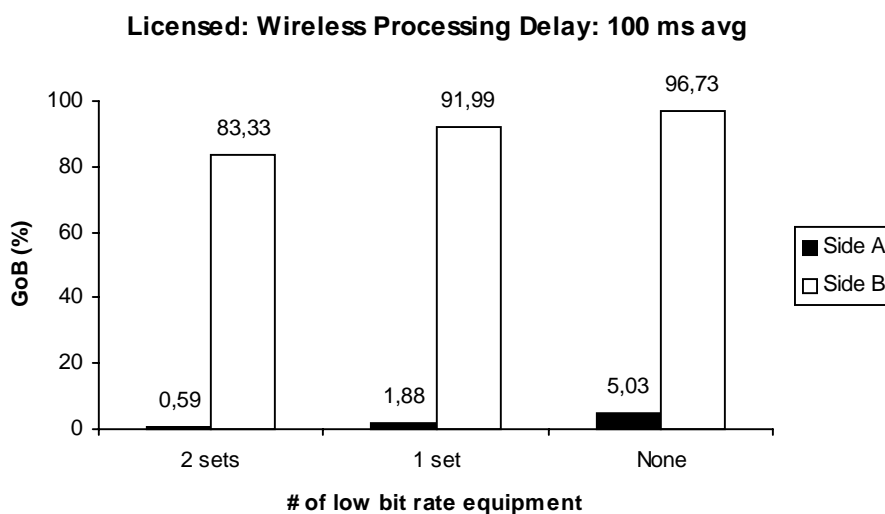


Figure D.16b: Licensed: Wireless Processing Delay: 100 ms avg

The results indicate voice 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 voice quality to the opposite party B.

Again, the more low bit-rate equipment is used, the poorer the resulting voice quality. This observation is the same for both licensed and unlicensed technology.

D.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 clause D.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 TIA-464B 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 TIA-464B, the loss for this PBX connection is 0 dB in both direction. 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.

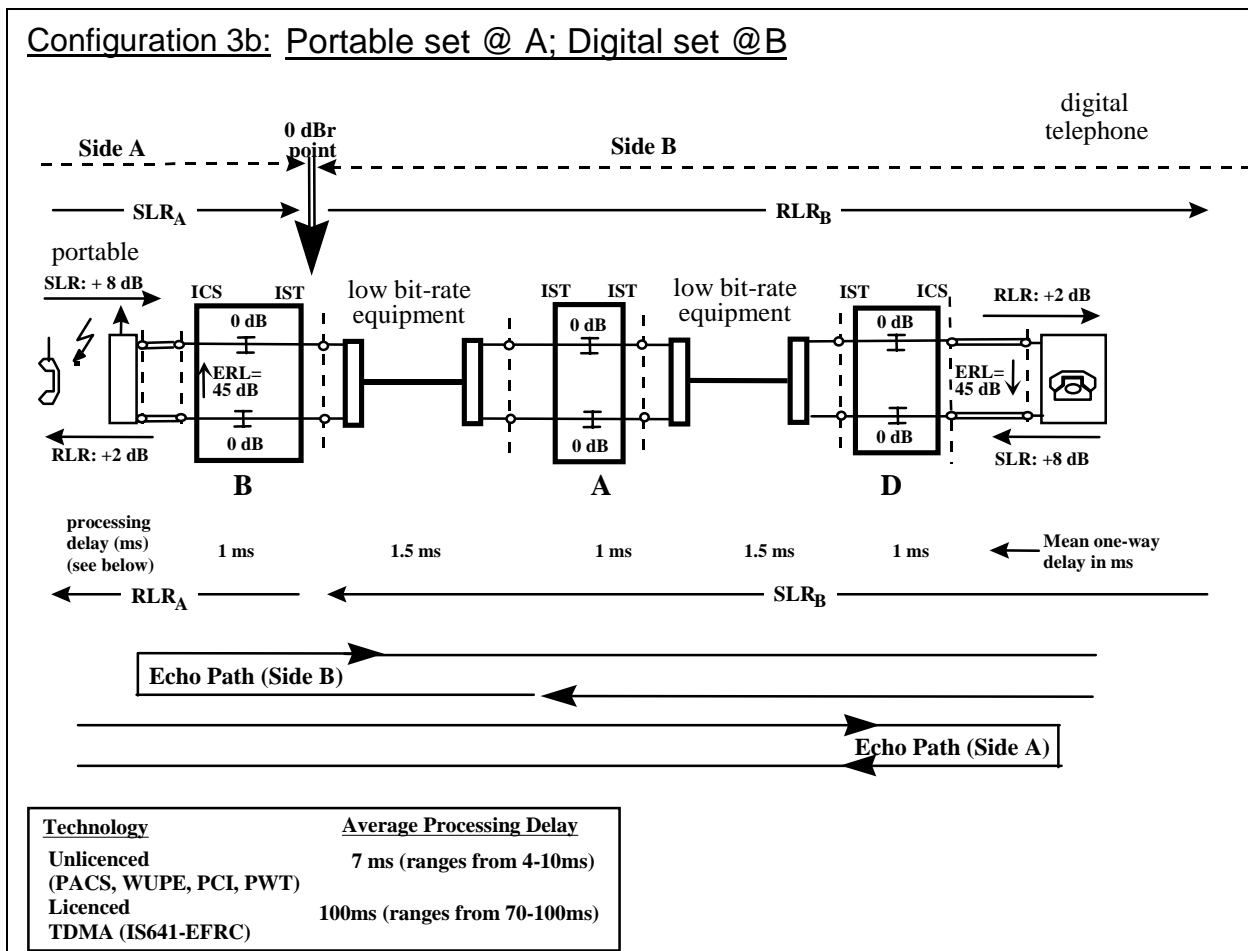


Figure D.17: Basis for Calculation of Reference Configuration 3b

D.2.8.1 Precalculations for Configuration 3b

D.2.8.1.1 Calculation of SLR, RLR

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

Listener at Side A
 SLR = 8 dB
 RLR(base) = 2 dB
 OLR = 10 dB

Listener at Side B
 SLR = 8 dB
 RLR(base) = 2 dB
 OLR = 8 + 2 = 10 dB

D.2.8.1.2 Talker Echo Loudness Rating (TELR)

Analysis of TELR performed at Side A:

$$\text{TELR} = \text{SLR}_{\text{SET A}} + 2 \times \text{Facility Loss} + \text{Return Loss (B)} + \text{sum of node losses} + \text{RLR}_{\text{SET A}}$$

$$\text{TELR}(\text{base}) = 8 + 2 \times 0 + 46 + (0 + 0 + 0 + 0 + 0 + 0) + (2) = 56 \text{ dB}$$

Analysis of TELR performed at Side B:

$$\text{TELR} = \text{SLR}_{\text{SET B}} + 2 \times \text{Facility Loss} + \text{Return Loss (A)} + \text{sum of node losses} + \text{RLR}_{\text{SET B}}$$

$$\text{TELR}(\text{base}) = 8 + 2 \times 0 + 45 + (0 + 0 + 0 + 0 + 0 + 0) + (2) = 55 \text{ dB}$$

where as above, Facility Loss is the sum of the line losses in each direction.

D.2.8.1.3 Delay Values T, Ta and Tr

T(ms) = Access Delay of wireless technology + PBX delay + Low bit-rate equipment delay + Digital End Office delay + Digital End Office delay

$$= 7 \text{ (unlicensed) or } 100 \text{ (licensed) } + 1 + 1,5 + 1 + 1,5 + 1$$

$$= 13 \text{ or } 106$$

$$\text{Ta} = \text{T}$$

$$\text{Tr} = 0$$

D.2.8.1.4 Number of Quantization Distortion

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.

$$\text{Therefore, qdu} = n \times 1 + m \times 0,7$$

$$= 1 \times 1 + 1 \times 0,7$$

$$= 1,7$$

where n = number of A/D-D/A conversion pairs

m = number of digital pads

D.2.8.1.5 Equipment Impairment Factor

The codec used for wireless technology does have an impact on voice quality. For the two types of technology analyzed, values in table 7 of ITU-T Recommendation G.113, and a value of 10 for IS641-EFRC were used. The total impairment involved is the sum of all the impairment factors involved:

Total impairment factor = Impairment due to codec used in wireless technology + network low-bit rate equipment

$$= 7 \text{ (unlicensed) or } 10 \text{ (licensed) } + 7 \text{ (1 low-bit rate equipment)}$$

$$\text{or } 14 \text{ (2 low-bit rate equipment)}$$

$$= 14 \text{ or } 21 \text{ (licensed)}$$

$$= 17 \text{ or } 24 \text{ (unlicensed)}$$

D.2.8.2 Input Parameters Summary Table

The input parameters for Configuration 3b (digital set @ B) are summarized in table D.6. Note the column "Reference Side" for those that have an "*" refers to the specific input parameter for the analysis to be performed at that particular side.

Table D.6: E-Model Input Parameters for Config 3b (digital set @ B)

Reference Side	Input Parameters	Values	Unit
Side A *	Send Loudness Rating (SLR)	8	dB
Side B *	Send Loudness Rating (SLR)	8	dB
Side A	Room Noise (Ps)	35 (default)	dB(A)
Side A	Ds factor (Ds)	3 (default)	
Side A *	Receive Loudness Rating (RLR)	2	dB
Side B *	Receive Loudness Rating (RLR)	2	dB
Side B	Room Noise (Pr)	35 (default)	dB(A)
Side B	Ds factor (Dr)	3 (default)	
Side B	Sidetone Masking Rating (STMR)	15 (default)	dB
Side B	Listener Sidetone Rating (LSTR)	18 (default)	dB
Side A *	Talker Echo Loudness Rating	56	dB
Side B *	Talker Echo Loudness Rating	55	dB
Side B	Weighted Echo Path Loss	80 (default)	dB
Side A *	Circuit Noise (Nc)	-70	dBm0p
Side B *	Circuit Noise (Nc)	-70	dBm0p
Overall	Noise Floor (Nfor)	-64 (default)	dBm0p
Overall	Equipment Impairment Factor (Ie)	14/21 (unlicensed) 17/24 (licensed)	
Overall	Advantage factor (A)	5 (unlicensed), 10 (licensed)	
Overall	Quantization distortion Unit	1,7	
Overall	Round Trip delay	0 (default)	ms
Overall	Mean one-way delay	13 (unlicensed) 106 (licensed)	ms
Overall	Absolute Delay	13 (unlicensed) 106 (licensed)	ms

D.2.8.3 Results

D.2.8.3.1 Unlicensed: Wireless Processing Delay: 7 ms average (WUPE, PCI, PWT)

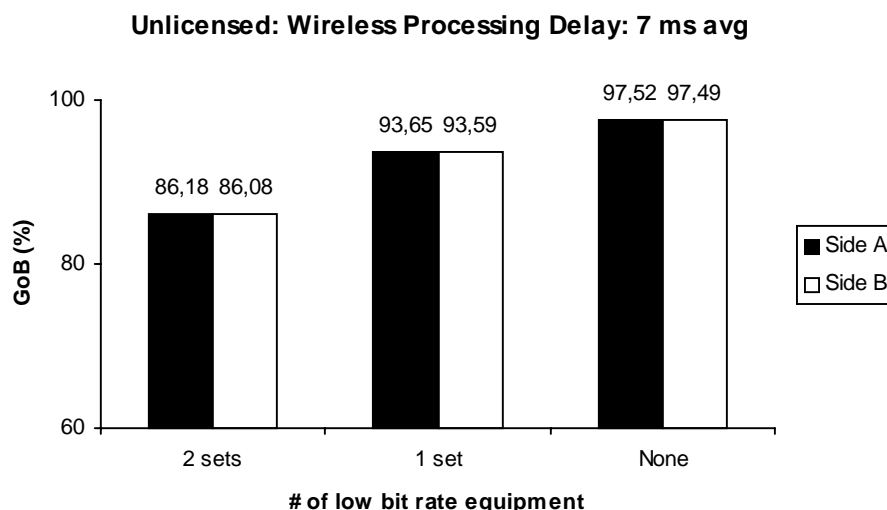


Figure D.17a: Unlicensed: Wireless Processing Delay: 7 ms avg

D.2.8.3.2 Licensed: Wireless Processing Delay: 100 ms average (TDMA-IS641-EFRC)

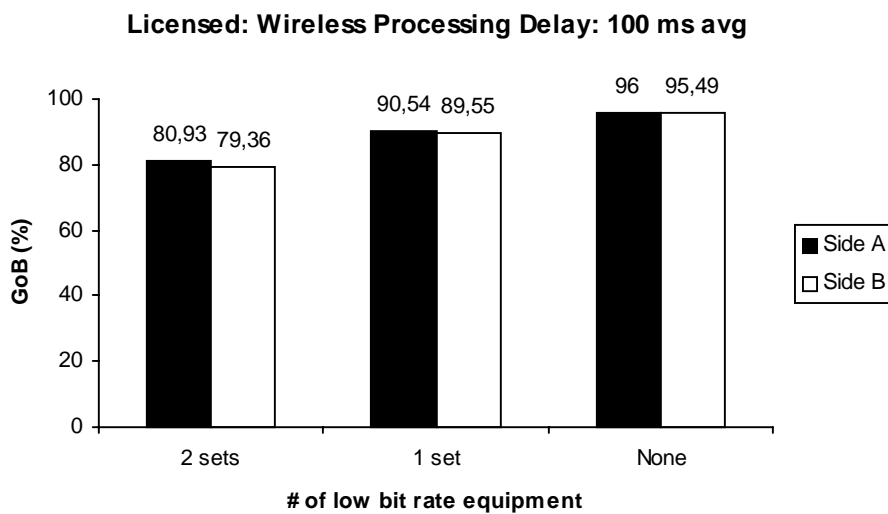


Figure D.17b: Licensed: Wireless Processing Delay: 100 ms avg

Voice quality perceived at either side A or side B is similar as the value of ERL at either end is the same value (45 dB), resulting in voice quality performance in the satisfied region. The same observation is obtained 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.

Bibliography

The following material, though not specifically referenced in the body of the present document (or not publicly available), gives supporting information.

ITU-T Recommendation G.111 (1993): "Loudness Ratings (LRs) in an international connection".

ITU-T Recommendation G.121 (1993): "Loudness Ratings (LRs) of national systems".

ITU-T Recommendation G.126 (1993): "Listener echo in telephone networks".

ITU-T Recommendation G.131 (1996): "Control of talker echo".

OPINE: Supplement III (1993) to the Series P Recommendation: Models for predicting transmission quality from objective measurements.

History

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