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ETSI

Postal address

F-06921 Sophia Antipolis Cedex - FRANCE

Office address

650 Route des Lucioles - Sophia Antipolis
Valbonne - FRANCE
Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16
Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Internet

secretariat@etsi.fr
http://www.etsi.fr
http://www.etsi.org

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Foreword

This EG (ETSI Guide) has been produced by ETSI Project Corporate telecommunication Networks (CN).

Introduction

The planning of private networks was previously - in most cases - based on existing scenarios with respect to the used (mainly analogue) technology, size and configurations, not only of the private networks, but also of the main public networks in the different countries. Regulation policy in most European countries was resulting in a rigid handling in conjunction with sometimes very stringent limits for the different transmission parameters, for calls via the public network.

There is a rapidly changing scenario in the field of liberalisation, deregulation, increasing competition in the public networks and the use of modern technology within private networks, with the need for more economical solutions. The ETR 004 [5] is no more longer flexible enough to take care of all these facts. For the development of this new ETR the goals can be summarized in the following highlights:

a) Planning of private networks will be more complex than in single PBX-configurations, therefore sufficient (tutorial) information about the planning - and calculation - methods should be provided.

b) The changing situation in the field of regulation policy should be considered.

c) The changing scenario in the public network operator domain should be included.

d) The ETR should enable the use of new technology within private networks, including wireless (cordless or mobile) sections, transmission of packetized voice, etc.

e) The ETR should provide sufficient planning methods and should contain all necessary information and tools enabling the planner to keep the resulting voice transmission quality within expected limits.

f) More "Allowance" with respect to specific transmission parameters and more flexibility compared with the recommendations in the ETR 004 [5] should be provided.

g) The activities in Study Group 12 of ITU-T, mainly the revised Recommendation G.113 [11] and the work on Question 18/12 "Interconnection of Private Networks with the public ISDN/PSTN" resulting in the new ITU-T Recommendation G.175 "Transmission planning for public/private network interconnection of voice traffic" should be taken into account.

h) The ETR should mainly be based on the ETR 250 [6] and the use of the E-Model.

One of the most important tasks in this list seems to be the provision of more "Allowance", since the use of new technologies and the application of future new services can only be realised in an economical way, if the permitted range of values within the private network for the different parameters is no more longer restricted by a rigid regulation. Therefore, the present document should provide enough flexibility in its application, also in the field of interconnection with other (public) networks and in the structure and routing within the private domain.

Furthermore, the present document should enable the planner not only to take care of absolute upper limits for the different parameters (which never should be exceeded), but also to obtain an estimate about the expected speech quality (possibly 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 their interconnections with other - mainly public - networks. It should be considered as a guidance 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 sufficient 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 for medium and large private networks consisting of more than one PBX. The term "Corporate Network" is 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 single networks interconnected to each other. Therefore only the term "Private Network" will be used in the present document.

The present document addresses only the case where a "Private Network" functions as a terminating network (one to which terminal equipment is connected). The case where a private network provides transit connections between other networks is outside the scope.

The issues of:

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

are outside the scope of the present document and the contents is independent of these issues.

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

The present document is based exclusively on the use of digital interfaces between the private and the public network, although the signal transmission within the private network may be analogue or digital. The transmission media may be cable, fiber or radio. Furthermore there are no restrictions to the private network with respect to size, configuration, hierarchy, used technology and network components.

Taking into account, that in modern private networks preferably digital signal transmission in exchanges and transmission media will be used, the selection of parameters to be considered during planning, has been slightly changed compared to previous planning methods. Therefore parameters with only minor impairments in a digital environment such as frequency shape of analogue cables, circuit noise, cross-talk, variations of loss with level or time etc., are not subject to planning for the benefit of simplification in the present document.

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, subsequent revisions do apply.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

NOTE: With regard to Recommendations from ITU-T (former CCITT), the following principle has been followed: The designation "CCITT" is used for Recommendations published before and including the Blue Books (1988). Recommendations (new or revised) after 1989 are designated "ITU-T".

[1] TBR 008: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice Attachment requirements for handset terminals".

[2] TBR 010: "Radio Equipment and Systems (RES); Digital Enhanced Cordless Telecommunications (DECT); General terminal attachment requirements; Telephony application".

[3] ETS 300 175 - 8: "Digital Enhanced Cordless Telecommunications (DECT);Common interface (CI); Part 8: Speech coding and transmission".

[4] ETS 300 283: "Business TeleCommunications (BTC); Planning of loudness rating and echo values for private networks digitally connected to the public network".

[5] ETR 004: "Business Telecommunications (BT); Overall Transmission Plan Aspects of a Private Branch Network for Voice Connections with Access to the Public Network".

[6] ETR 250: "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear of 3,1 kHz handset telephony across networks".

[7] ETR 275: "Transmission and Multiplexing (TM); Considerations on transmission delay and transmission delay values for components on connections supporting speech communications over evolving digital netsworks".

[8] ITU-T Recommendation G.100, (03/93): "Definitions used in Recommendations on general characteristics of international telephone connections and circuits".

[9] ITU-T Recommendation G.101, (03/93): "The Transmission plan".

[10] ITU-T Recommendation G.111, (03/93): "Loudness ratings (LRs) in an international connection".

[11] ITU-T Recommendation G.113, (02/96): "Transmission impairments".

[12] ITU-T Recommendation G.114, (02/96): "One -way transmission time".

[13] ITU-T Recommendation G.121, (03/93): "Loudness ratings (LRs) of national systems".

[14] ITU-T Recommendation G.122, (03/93): "Influence of national systems on stability talker echo in international connections".

[15] ITU-T Recommendation G.126, (03/93): "Listener echo in telephone networks".

[16] CCITT Recommendation G.131, Blue Book 1988, Fasc. III.1: "Stability and echo".

[17] CCITT Recommendation G.164, Blue Book 1988, Fasc. III.1: "Echo suppressors"

[18] ITU-T Recommendation G.165, (03/93): "Echo cancellers".

[19] Draft ITU-T Recommendation G.168, (05/96): "Echo cancellers".

[20] ITU-T Recommendation G.175, (05/96): "Transmission planning for private/public network interconnection of voice traffic".

[21] ITU-T Recommendation G.711, Blue Book 1988, Fasc. III.4: "Pulse code modulation (PCM) of voice frequencies".

[22] ITU-T Recommendation P.11, (03/93): "Effect of transmission impairments".

[23] ITU-T Recommendation P.79, (03/93): "Calculation of loudness ratings for telephone sets".

[24] ITU-T Recommendation P.800, (03/93): "Methods for subjective determination of transmission quality".

[25] CCITT Recommendation P.82, Blue Book 1988, Fasc. V: "Method for evaluation of service from the standpoint of speech transmission quality".

[26] ITU-T Recommendation P.830, (02/96): "Subjective performance assessment of telephone-band and wideband digital codecs".

[27] ITU-T Recommendation P.84, (03/93): "Subjective listening test method for evaluating digital circuit multiplication and packetized voice systems".

[28] ANSI Standard EIA/TIA 464B-1996: "Requirements for Private Branch Exchange (PBX) Switching Equipment".

[29] ANSI Standard EIA/TIA 470A-1987: "Telephone Instruments with Loop Signaling".

[30] ANSI Standard EIA/TIA 579-1991: "Acoustic-To-Digital and Digital-To-Acoustic Transmission Requirments for ISDN Terminals".

[31] ANSI Standard T1.508-1992: "Network Performance - Loss Plan for Evolving Digital Networks".

[32] ETS 300 462-1/6: "

[33] ISO/IEC 11573: "

3 Definitions and abbreviations

3.1 Definitions

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

3.1.1 Private Network

In the present document the term "Private Network" is used to describe a network providing its features only to a "restricted User Group" in contrary to the "General Public". It consists of several switching equipment (PBXs) forming a network, with interconnections to other (mainly public) networks. The following more detailed list is assuming, that the private network is contributing in a possibly significant amount of transmission impairments to the overall transmission quality.

The term "Private Network" is defined as follows:

1) It consists normally of more than one switching equipment (PBX or Key Telephone System KTS), connected via private or leased lines or via a Virtual Private Network (VPN), forming a network independant of its structure and hierarchy. Switching equipment and links can be either analogue or digital.

2) It provides switching functions and all other features to only a single customer or a group of customers, but is not accessible to everyone.

3) There is no limitation by its geographical size, no restriction to the national area only and no limitation for 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" [20].

3.2 Public Network

The term "Public Network" is used in the present document for all networks providing their switching functions and features not only to a specific user group, but to the "General Public". The word "Public" is not related to the legal status of the network operator. Public networks may not provide in all cases the usual features. In the field of competition they can be restricted to only a limited number of customers, or specific features and switching functions.

Furthermore, public networks may provide access points in a specific geographical area only. From the point of view of a connection, public networks are mainly "Transit Networks", but they may also be considered as a combination of "Transit- and Terminating Networks" in case where the public network operators are providing also terminal equipment such as telephone sets, PBXs (or PBX-Features).

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).

3.3 Quality aspects

Previous planning methods for private networks, or in general terminating networks, 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 the voice transmission quality during a telephone conversation is primarily a "subjective" judgement. The term "Quality" may not be considered as a unique entity, but may vary, depending on the users expectation of a sufficient "Voice Transmission Quality" for a 3,1 kHz telephony call via handset, or the used particular service. With respect to transmission planning, the planning method and the necessary calculations should be based on an end-to-end consideration between a humans mouth and ear.

For the judgement of the quality in a given configuration and the performance of "Subjective Tests" several methods are in use and described in different ITU-T Recommendations (P.800 [24], P.82 [25], P.830 [26], P.84 [27]). One of the most common methods is to perform laboratory tests (e.g. "Listening only Tests"), where the test persons are requested to "classify" the perceived quality into categories. For example this "Quality Rating" can be defined in the following 5 grade scale:

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

The numbers (Score) are used to calculate the average value of the judgement of several test persons for the same test configuration. The result is the so called "**M**ean **O**pinion **S**core **MOS"** which may vary theoretically between 1 and 5. An issue about the quality of service can also be obtained, if the percentage of all test persons judging with "Good or Better" and "Poor or Worse" are calculated. These results for a given connection are called "Percentage **G**OOD **o**r **B**ETTER **GOB**" and "Percentage **P**OOR **o**r **W**ORSE **POW**".

In existing networks, public and private network operators may also use different methods of "Field Survey" to control and monitor the "Quality of Service" with respect to speech transmission. This can be done by customer interviews or as an objective method using a test equipment called "In-service non-intrusive measurement device (INMD)", 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 different network components in the investigated configuration and their impairments contributing to the whole connection, expressed by means of quality. To assist the planner, computation models are available resulting in a calculated value for quality issues such as MOS, GOB and POW. One of those tools is the "ETSI Computation Model" - in the present document called the **"E-Model"**- which is recommended to be used for planning purposes. For more information about this E-Model, see ETR 250 [6] and subclause 6.2 in conjunction with annex B of the present document.

3.4 Network elements

All components forming a connection can be distinguished into the three main groups: terminal elements, connection elements and transmission elements.

3.4.1 Terminal elements

Terminal elements with respect to speech transmission, are all types of digital or analogue and cordless or mobile telephone sets, including the acoustical interfaces to the humans mouth and ear. These components with their Send Loudness Rating SLR and Receive Loudness Rating RLR are contributing to the Overall Loudness Rating OLR of a connection. Further impairments can be caused by the Sidetone Masking Rating STMR, the Listener Sidetone LSTR in conjunction with the D-Factor (depending on the design of the handset), the frequency response in send and receive direction and also some noise floor. In case of wireless systems, additional distortions and delay will be added according to the used coding and modulation algorithm for the radio interface.

3.4.2 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. They are using either an analogue or a digital switching matrix. Analogue systems may mainly contribute by loss. Digital exchanges will contribute mainly with transmission time due to signal processing and partly also with quantization distortion. In case of 4-wire to 2-wire conversion, they are contributing additionally with signal reflections as a source for echo effects.

3.4.3 Transmission elements

Transmission elements are all kinds of media, used as a facility between connection elements and to terminal elements. The physical media of these elements can be metallic, fiber-optic or radio. The signal form is either analogue or digital. In case of analogue signal transmission, cables are contributing by loss and by a shape in the frequency response. An additional impairment can be caused by noise due to longitudinal interference.

For planning purposes impairments due to frequency shape and noise can usually be neglected for short and medium line lengths. Using the digital signal form, the main transmission impairment is caused by the propagation time via metallic and optic media. Wireless sections may contribute with delay depending on the used coding and modulation algorithm.

Beside the transmission of only one single communication channel via the different media, also "Multiplexing" is used, to transport several channels via one single physical media. A variety of multiplexing systems is in use in the existing networks. Time Division Multiplex Systems (TDM) are using the Pulse Code Modulation (PCM), or new low bit-rate Codecs. A major influence to the transmission quality of these systems is caused by the transmission time. Modern equipment are using special coding algorithms to reduce the bit-rate of each communication channel. An example of such a device is an equipment called Digital Circuit Multiplication Equipment (DCME). Those systems are contributing with additional distortion and delay.

3.5 Types of connections

For some private networks the main "Types of connections" via other - mainly public networks - may be taken into account for a possible higher amount of permitted impairments within the private network. Depending on the business of the private network operator, the predominance of incoming and outgoing calls may be e.g. only originated or terminated in the local area. Beside internal connections between two terminal elements of the same private network, an external connection via a public network can be 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 sense means an amount of connections e.g. within the local area of more than 95 %. The inclusion of the type of external connections 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.

3.6 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
DCME	Digital Circuit Multiplication Equipment
EC	Echo Canceller
ECD	Echo Control Devices
EL	Echo Loss
ELE	Echo Loss Enhancement
ERLE	Echo Return Loss Enhancement
ERP	Ear Reference Poin
ES	Echo Suppressors
FDM	Frequency Division Multiplex
GOB%	Percentage GOOD or BETTER
ICP	International Connection Point
IFM	The Impairment Factor Method
INMD	Non-Intrusive Measurement Device
KTS	Key Telephone System
LD-CELP	Low-Delay Codebook-Excited Linear-Prediction
LELR	Listeners Echo Loudness Rating
LSTR	Listener Sidetone Rating
MOS	Mean Opinion Score
MRP	Mouth Reference Point
NCP	Network Connection Points
NLP	Non Linear Processor
OLL	Open Loop Loss
OLR	Overall Loudness Rating
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
POW%	Percentage POOR or WORSE
qdu	number of quantizing distortion units;

RLR	Receiving Loudness Rating
RPE-LTP	Residual Pulse Excitation - Long Term Predictor
SLR	Sending Loudness Rating
SS	Soft Suppressor
TBRL	Terminal Balance Return Loss
TCLw	Terminal Coupling Loss weighted
TDM	Time Division Multiplex Systems
TELR	Talker Echo Loudness Rating
TME	Terminate Early
UPCM	Uniform PCM
VAD	Voice Activity Detection
VPN	Virtual Private Network
VSELP	Vector Sum Excited Linear Prediction
WEPL	Weighted Echo Path Loss

4 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 single terminal-, connection- and transmission-elements which may contribute in a specific amount of 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 different reference configurations. Therefore it is not possible to create only one basic figure for the whole task of private network planning. The following figures should be considered as examples only, 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 digital interfaces.

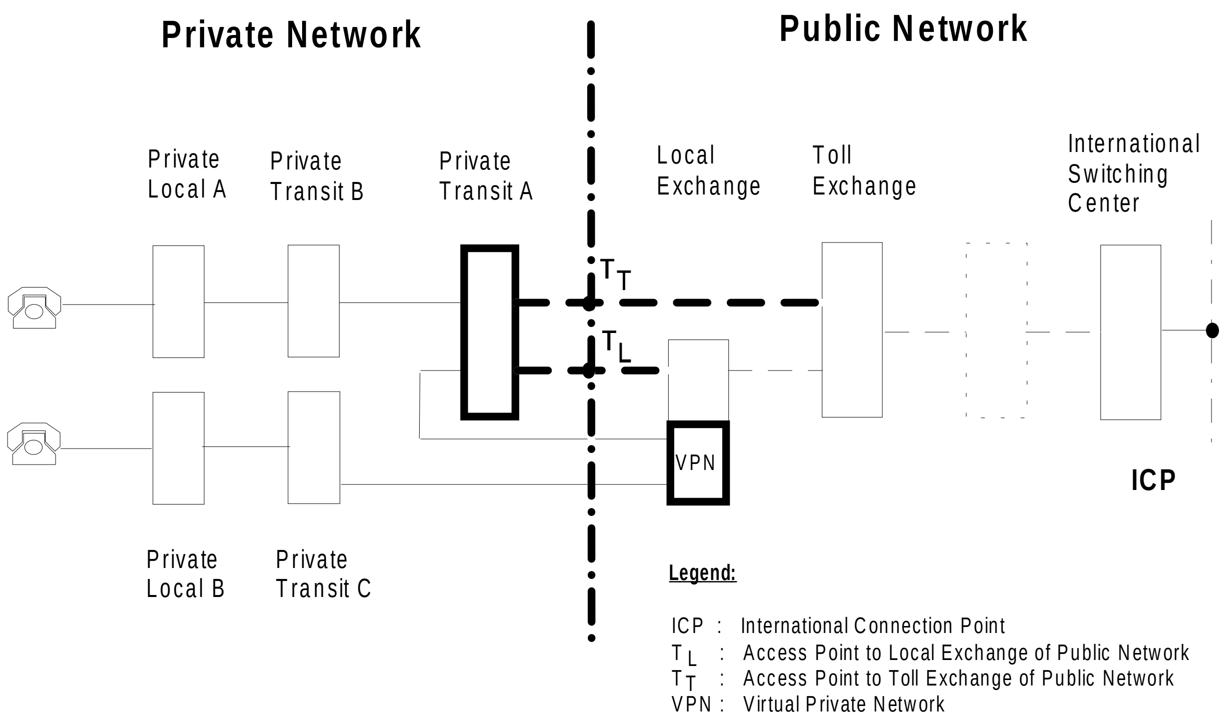

Figure 1: Basic configuration for the interconnection between private and public network

The public network and therefore the entire connection is only shown up to the International Connection Point ICP of an international switching center. It is assumed, that the permitted impairments between the access points for calls within the national public network are partitioned symmetrically with reference to the ICP, which can be considered as the virtual center of the public network. 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.

The configuration shows two different types of interfaces between public and private networks, the one called T_L is connecting the private network to a local exchange, usually the lowest hierarchy and the common connection point in a public network. The other one, called T_T is connecting the private network directly to a toll exchange, i.e. a higher hierarchy level, bypassing the local exchange. In some applications this feature could be offered by the public network operator, mainly in case of large private networks, possibly resulting in more "Allowance" with respect to specific transmission parameters.

A further possibility of a public network is shown for 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 all possible leased lines used as interconnection between the different PBXs within the private network, but provided usually by public network operators.

Where the private network includes Leased Lines, VPN connections or Centrex terminals, the private network planner should obtain planning information on these connections from the Public Network Operator providing the service.

Figures 2 - 4 are demonstrating common standard configurations only 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 is contributing with a minimum of transmission impairments.

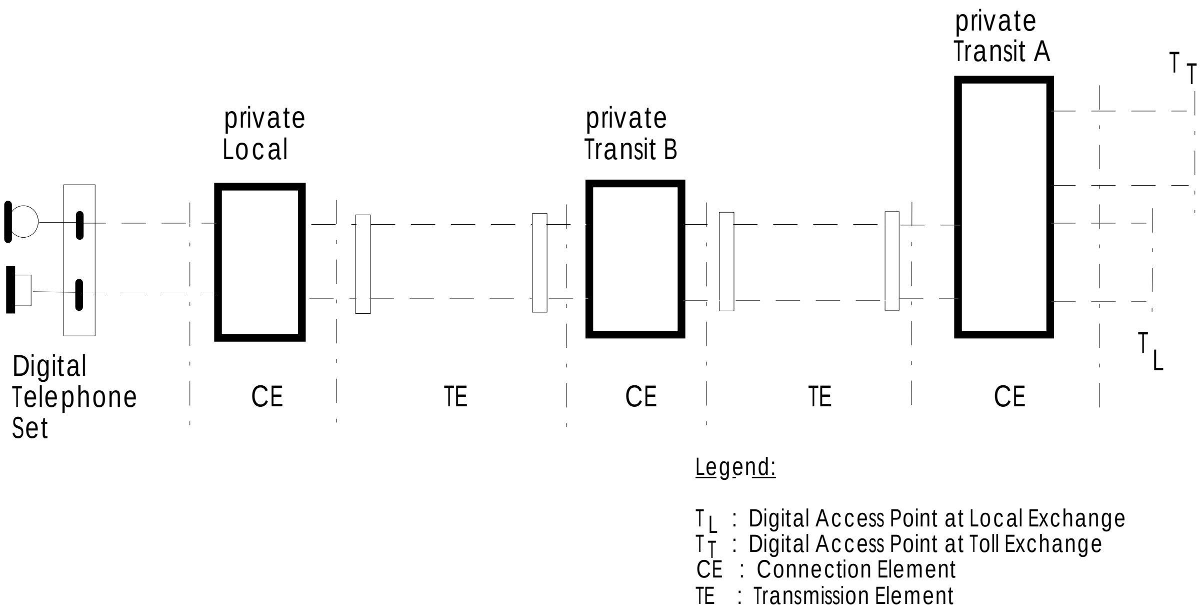

Figure 2: Standard configuration with a fully digital routing within the private network

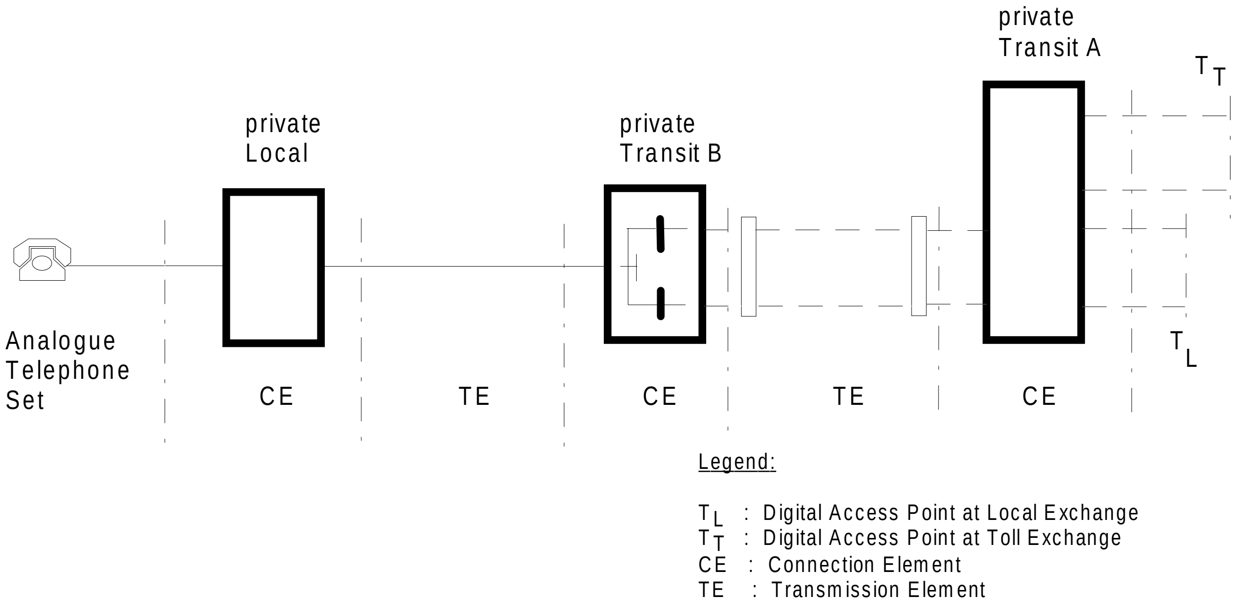

Figure 3: Private network with 4-wire/2-wire conversion

The configuration in figure 3 is assuming a 4-wire/2-wire conversion (hybrid) in the private Transit B exchange and a 2-wire cable to the private local exchange as well as to the analogue telephone set. In this case impairments due to loss via the 2-wire cable section should be expected. Furthermore the hybrid in the Transit B exchange may cause impairments as a possible source of echo for the far end subscriber. This hybrid may also form the termination of a long international connection with an influence to stability of the international 4-wire chain.

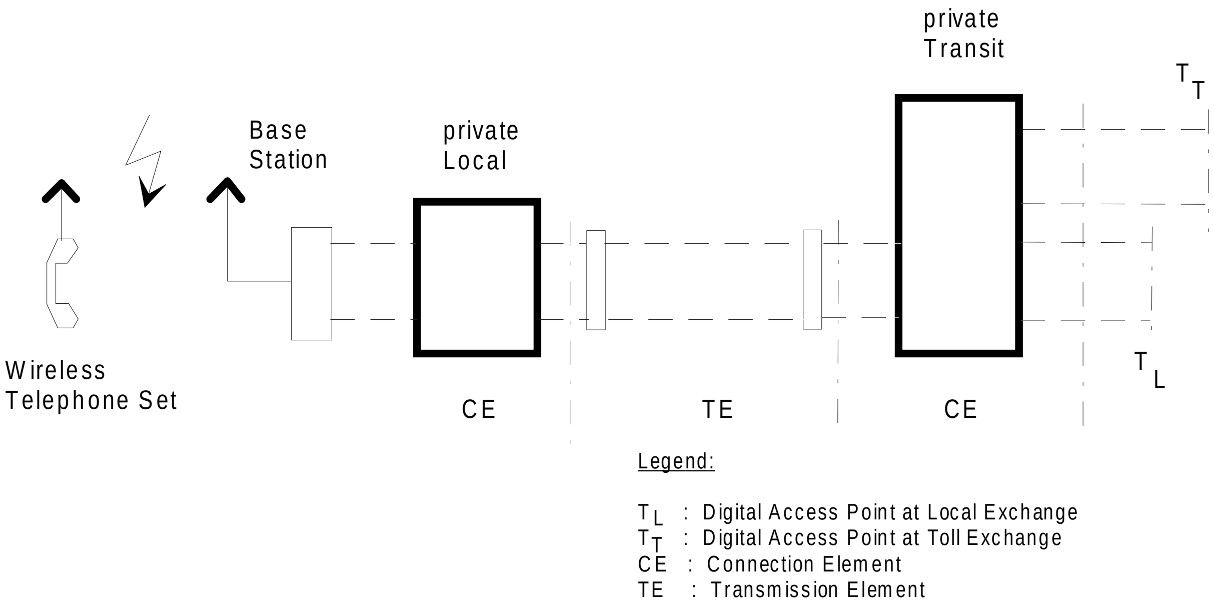

Figure 4: Digitally connected wireless telephone set

Impairments due to additional delay and distortion should be subject to planning when using wireless telephone sets as terminal elements as shown in figure 4. In those configurations also the use of echo control devices should be investigated.

5 Basic planning principles

As already 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 and more "Allowance" for the private network with respect to transmission planning. In general, the quality of speech transmission via telephone channels is based on a subjective judgement by the users on both ends. Therefore transmission planning is in principle derived from an end-to-end consideration in conjunction with a partitioning of all relevant parameters between different networks or parts of a network where applicable. This method was in common use in the field of regulation for all calls via the public network, providing limits for the private network between the acoustical interface of the telephone set and an electrical interface to the public network in such an amount, that for all calls (national and international) a sufficient quality was guaranteed even in a worst case scenario.

This principle is no longer valid in all those countries which are moving into deregulation. The responsibility for a sufficient voice transmission quality is now shifted to the operator of the private network. However, planning of private networks with respect to voice transmission quality needs knowledge and experience in the field of transmission parameters and their influence to quality. Therefore it seems necessary to provide a planning method, easy to handle and accompanied by all the necessary (tutorial) information and planning tools, the main task of the present document. To meet the main goals of this new ETR, such as flexibility, more "Allowance" and the use of modern technology, the basic planning principle as used in the present document is deviating 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 more longer referred to units of the different parameters, but to issues of perceived quality in terms of MOS, GOB or POW.

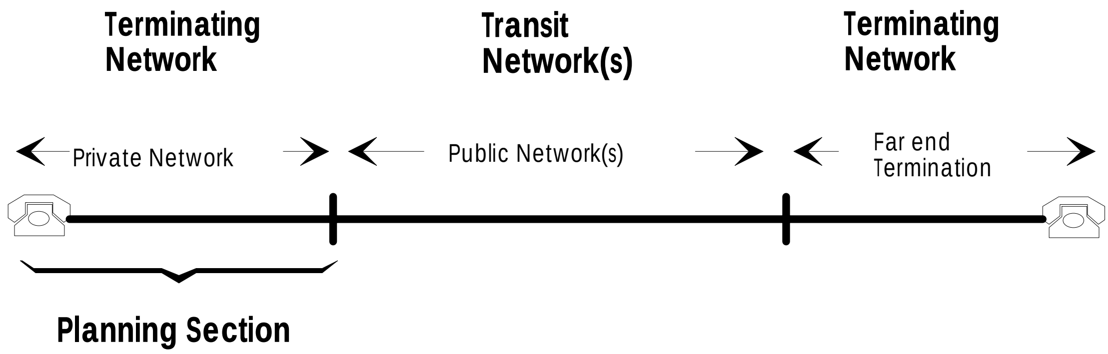

Figure 5: General configuration for calls via public networks

This principle is applied not only for internal calls within the private network, but also for calls via public networks. The "end-to-end principle" in case of calls via public networks should include parts of a connection not directly subject to planning as illustrated in figure 5, e. g. impairments caused by the public network (or tandem connections via more than one public network) and the far end termination. These impairments should be known and included in the transmission planning. This may cause some problems in planning practice, since those values may be not available in all cases, mainly for the far end termination. In those situations average values for different transmission parameters will be 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 public and private network operator. Furthermore, the inclusion of these sections allows a differentiation for the type of calls - national local, national long distance and international - assuming a different amount 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 [11]. The necessary planning tool, the so called E-Model, as introduced and described in ETR 250 [6] is based on this Impairment Factor Method. For more information about this method and the E-Model, which both are used in the present document, see the following subclauses.

5.1 The Impairment Factor Method (IFM)

Voice transmission quality in telephony is influenced by several parameters such as loss, distortions, delay, echo, noise etc., contributing to a specific amount and decreasing the quality as perceived by the users. As long as only one of these parameters is influencing the quality, simply an upper limit or permitted range in units of this parameter with a relation to a "Quality Scale" - based on subjective tests - can be defined. This simple way is not applicable if several parameters are contributing with impairments, a situation which may occur in many practical configurations. For those situations a planning method was developed, the "Impairment Factor Method" recommended by ITU-T and described in ITU-T Recommendation G.113 [11]. This method 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 derived from subjective tests and for the different network elements are added resulting in a Total Impairment Value as shown in figure 6.

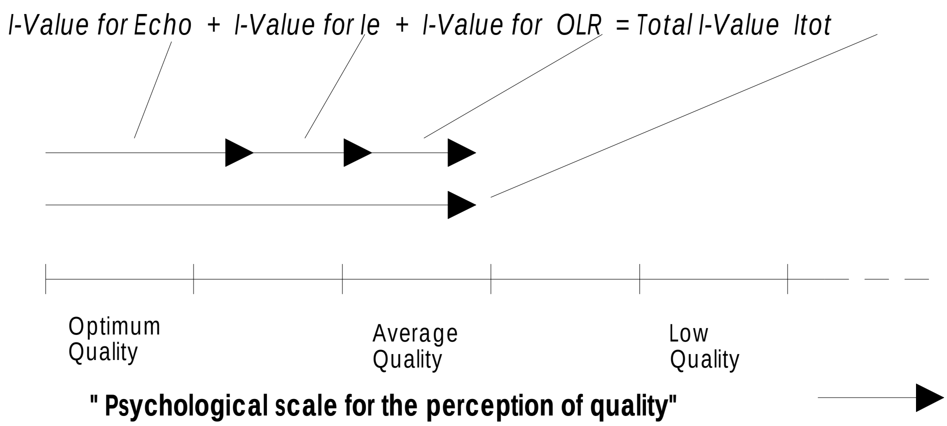

Figure 6: Example for the addition of impairments on a "Psychological Scale"

These impairment values are expressed in numbers, where I = 0 means no impairment or optimum quality. The different impairment values are described in subclause 5.2 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 Itot - value. The relation between the value of Itot and the perceived quality level, expressed by verbal description, is shown in table 1a. This description of the speech communication quality is taken from ITU-T Recommendation G.113 [11]. A similar table is contained in ETR 250 [6] but related to the "Rating Factor R" instead of Total impairment value Itot.

The ITU-T Recommendation G.175 [20] contains a similar table as shown in table 1b, also related to the Rating Factor R, but based on an issue about the "Users Satisfaction". Both tables may be used as a guidance for planning purposes, with table 1a being the preferred one.

Table 1a: Relation between communication quality and total impairment value Itot (G.113)

Itot 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 users satisfaction (G.175)

R-Value Lower limit	MOS	GOB	POW	Users 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, mainly if combination effects in the presence of more than one parameter should be considered, computation models are used for planning purposes. Several of those "Rating-Models" have been developed and are contained and described in ITU-T Recommendations. For the purpose of the present document the E-Model is used as developed by an ETSI ad hoc Group and as published in the ETR 250 "Speech Communication Quality from Mouth to Ear of 3,1 kHz handset Telephony across Networks" [6]. More information is given in subclause 5.2, clause 8 and annex B.

5.2 The E-Model

Among several rating models, the E-Model was choosen for the application in the present document since it is easy to handle, allows the incorporation of impairments if digital processing is performed by low bit-rate Codecs and it is including combination- and masking-effects of impairments occuring simultaneously in a connection. Furthermore it is providing an algorithm to transform the results into terms of MOS, GOB and POW.

NOTE: It should be noted, that this E-Model and its algorithm has not yet been fully verified and may give incorrect results in some applications. However investigations with the goal to prove the validity of this model are in progress within Study Group 12 of ITU-T in conjunction with Question 20/12 in the Study Period 1997-2000.

In general, the E-Model is based on the Impairment Factor Method, i. e. the different impairment-values are calculated and combined to an overall rating "R". This can be expressed in the following formula:

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

High values of "R" in a range of up to R = 100 should be interpreted as excellent quality and vice versa. Therefore the main impairment values Is, Id and Ie are substracted from the basic signal-to-noise ratio **Ro**. The term **Is** is representing all impairments which occur more or less simultaneously with the voice signal, such as too loud speech level (non optimum OLR), non optimum sidetone (STMR), quantization noise (qdu) etc. The "delayed impairment" factor **Id** is summarizing all impairments due to delay and echo effects and the "equipment impairment factor **Ie**" is representing all impairments which are caused by low bit-rate Codecs used in special equipment. The "Expectation Factor **A**" represents an "Advantage of Access" which certain systems may provide in comparison to conventional wired systems. Those advantages are cordless and mobile systems or connections into hard-to-reach regions via multi satellite hops. It should be noted, that in some cases not only the final result for R is of interest, but also the specific impairment values Is, Id and Ie. Their contribution to the total value can be used for the determination of the major impairments in the given configuration and of possible solutions for the reduction especially of these impairments, e.g. reducing Id by the insertion of echo cancellers.

The E-Model and its algorithms are using the following input parameters:

- SLR Sending Loudness Rating;
- RLR Receiving Loudness Rating;
- (OLR Overall Loudness Rating) *);
- STMR Sidetone Masking Rating **);
- LSTR Listener Sidetone Rating **);
- Ds D-Value of Telephone at Send-side;
- Dr D-Value of Telephone at Receive-side **);

- TELR Talker Echo Loudness Rating;
- WEPL Weighted Echo Path Loss;
- T Mean one way Delay of the Echo path;
- Tr Roundtrip Delay in a closed 4-wire loop;
- Ta Absolute Delay in echofree connections;
- qdu Number of Quantizing Distortion Units;
- Ie Equipment Impairment Factor (low bit-rate Codecs);
- Nc Circuit Noise referred to the 0 dBr-point;
- Nfor Noise Floor at the Receive-side;
- Ps Room Noise at the Send-side;
- Pr Room Noise at the Receive-side;
- A Expectation Factor:
- *) No direct input value; calculated as OLR=SLR+RLR;
- **) These parameters have a fixed relation by: LSTR=STMR+Dr.

Only parameters marked in bold letters are usually subject to planning. The other parameters can be set to default values. For more details see clause 6.

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 8.

6 Parameters subject to planning and their limits

In the present document only digital interfaces between private and public networks and digital media for the main transmission elements within the private network are assumed. The amount of influence of the different transmission parameters to the end-to-end speech quality may change in such mainly digital environment compared to previous analogue networks. Some parameters may have become more important for planning purposes while others are now of minor influence or can even be neglected. The following subclauses 6.1 to 6.8 are describing those parameters which should in any case be subject to planning and subclause 6.9 is dealing with those parameters which - for the benefit of simplification - can be neglected or considered only in special applications.

6.1 Overall Loudness Rating (OLR)

Although only digital interfaces to other - mainly public - networks and digital transmission media for the main transmission elements within the private networks are used, the Overall Loudness Rating OLR of a connection should be considered. Mainly in the lower hierarchy of private networks 2-wire analogue lines and connection elements (PBXs) with an analogue switching matrix are in use due to economical reasons and may contribute with some amount of loss. Also within public networks fully digital routing and termination cannot be assumed for all connections.

Basically the OLR of a connection can be described as the sum of the Send Loudness Rating SLR and the Receive Loudness Rating RLR of the telephone sets at both ends and the Circuit Loudness Rating CLR, 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 values of OLR or as well from too low values of OLR. The optimum value can be found in the range from 8 to 10 dB. Connections with two digital telephone sets e.g. according to ETSI TBR 8 [1] with SLR = 7 dB and RLR = 3 dB, or TIA 579 [30] with SLR = 8 dB and RLR = 2 dB, routed fully digitally, will meet this optimum value for OLR. The relation between the total impairment value Itot and the OLR of a connection is shown in figure 7. This curve, calculated with the E-Model, is obtained if all other input parameters of the E-Model are set to their optimum values (default values, see subclause 8.7), i.e. OLR is the only impairment in the considered connection.

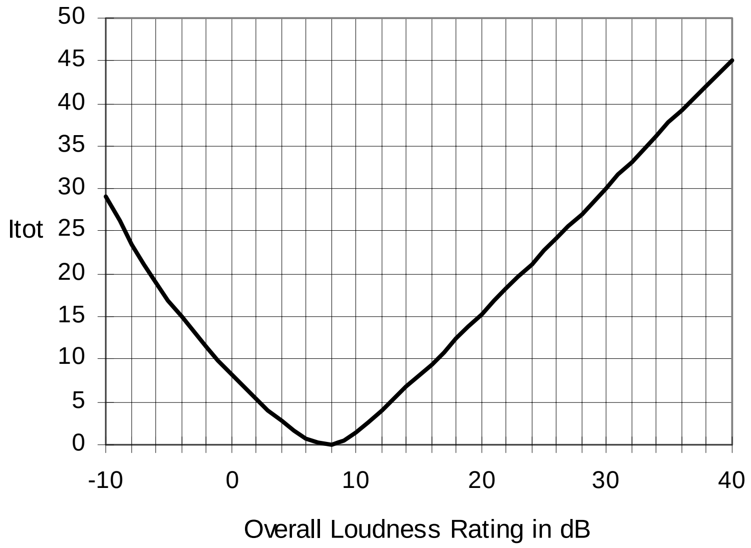

Figure 7: Relation between OLR and total impairment value Itot

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 required in ETSI TBR 8 are used. When comparing the curve in figure 7 with the description of the perceived quality level in table 1a, the "limiting case" is identical with a range for the OLR of - 10 dB to 30 dB. However, the value 30 dB for a connection should be interpreted as an absolute upper limit which should never be exceeded, also not in exceptional cases. For a quality level between "good" and "adequate" - the preferred limit for standard connections - the upper value for OLR is recommended in the range of 20 dB to 25 dB. Very low values for OLR should be avoided. For OLR < 0 dB the insertion of additional loss is recommended.

6.2 Echo

As already stated, fully digital routing cannot be assumed in all configurations, therefore mixed analogue/digital routing within private and public networks in conjunction with 4-wire/2-wire conversions should be taken into account, where signal reflections together with transmission delay will cause impairments due to talker echo. In digital networks sources for delay are not only the propagation time along the different transmission media, but also delay due to digital signal processing. Furthermore low bit-rate codecs as used e.g. in multiplexing equipment and radio sections will contribute with additional delay. Signal reflections will mainly arise at hybrids, where the whole path of the connection between the talkers telephone set and this hybrid is forming the so called Echo Path.

The effect of echo in a conversation can cause impairments to both, the talker and the listener. Consequently this is expressed as Talker Echo and Listener Echo. For more information about these effects see also annex A. As a general rule, in most configurations listener echo can be neglected if there is a sufficient control of the talker echo.

The impairment due to talker echo is depending on two factors, the delay and the volume of the received own speech signal. The perceived quality is decreasing with increasing delay and/or increasing volume of the echo signal.

For planning purposes the transmission delay T is defined as the mean one-way transmission time along the whole echo path. It should be noted, that - although the total transmission time between the talkers telephone set and the hybrid **and back** is perceived as the disturbing delay - only the mean one-way transmission time is used in the present document for planning, assuming that for standard configurations the transmission time is nearly equal in both transmission directions..

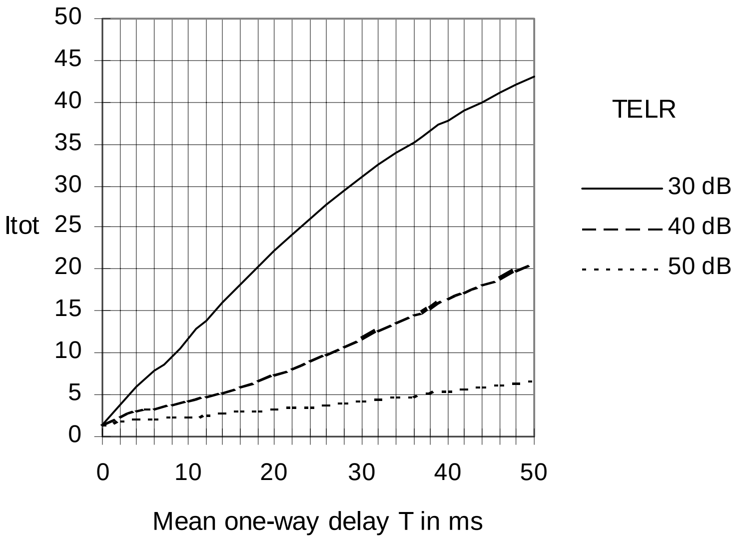

Figure 8: Total impairment value Itot due to Talker Echo

For planning calculations the volume of the received echo signal is expressed as Talker Echo Loudness Rating TELR (as an input parameter to the E-Model). This parameter is defined as the sum of SLR and RLR of the talkers telephone set and the Echo Loss EL of the echo path.

$$TELR = SLR + EL + RLR$$

The relation between mean one-way delay, TELR and the resulting total impairment value Itot 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 will be one of the most important parameters in modern mixed digital/analogue networks. To reduce the impairments caused by echo effects. they should be considered carefully in every transmission planning. Since the transmission delay in a connection usually cannot be reduced, only higher values of TELR within the echo path may improve the perceived quality. However, an increase of the TELR in a given connection is limited in most cases, to avoid an increase of the OLR. In those situations the use of Echo Cancellers EC should be taken into account. For more information about the use of EC see clause 9.

Based on a planning limit for the quality level between "good" and "adequate" (see table 1a with an Itot of approx. 15) for standard connections, figure 8 shows, that with TELR = 30 dB the mean one-way delay is limited to T = 13 msec, while with TELR = 40 dB a value of T = 36 msec can be tolerated. In practice, values for TELR will in most applications be in the range of 30 dB to 40 dB. Therefore it is advisable, in case of connections containing an echo path with more than 20 to 25 msec mean one-way delay, to investigate exactly the impairments due to echo effects and the possible necessary use of echo cancellers. This recommended range should be decreased accordingly if further impairments, e.g. due to low bit-rate codecs should be taken into account.

Very low values of the mean one-way delay with T < 1,5 msec 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. In some configurations more than one echo path may exist. For planning rules in those situations see subclauses 7.2 and 8.5.

6.3 Transmission time in echofree connections

In case of international connections mainly 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. However, according to subjective tests, this effect is encountered only for a one-way transmission time of more than 150 msec.

The relevant parameter for planning is the Absolute Delay Ta in msec, 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 Itot is shown in figure 9 for a range of Ta = 100 ... 600 msec, 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 standard connections should be in the range of 300 to 350 msec with an upper limit of 400 msec. This is also in accordance with ITU-T Recommendation G.114 [12] and should be exceeded in exceptional cases only.

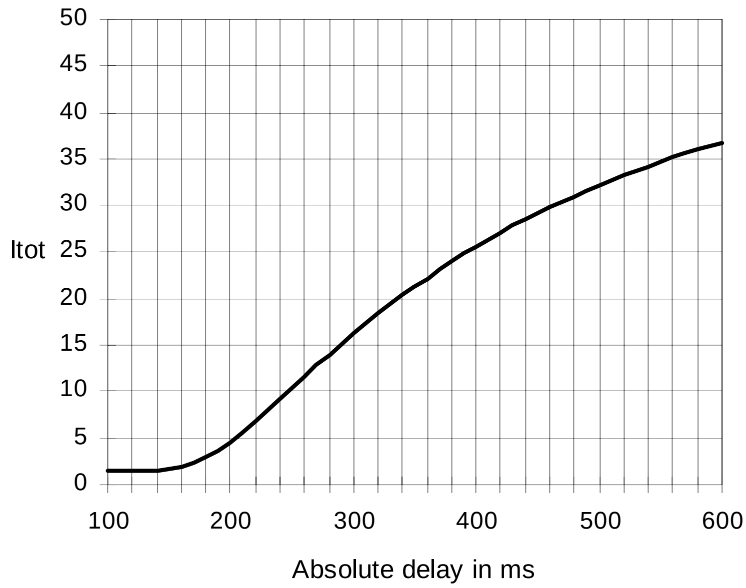

Figure 9: Relation between absolute delay Ta and the total impairment value Itot

6.4 Stability

The stability of a connection should be considered in transmission planning in all those configurations, where the private network is containing 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 is forming a 4-wire loop. For further information about stability see also annex A.

Insufficient stability may cause singing within the 4-wire loop, which should be avoided in any case. Although stability is more a problem during call setup and release, i.e. there is usually no influence to 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 is defining the margin between the actual OLL and the point where singing may arise. The most critical configurations during call setup are open and short circuit conditions at the 2-wire ends of the 4-wire loop.

Since singing will not occur during the talking state (when correct termination at the 2-wire ends are available), stability is no task of speech quality. However, it should in any case be considered during transmission planning. This should 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 in the frequency range from 0 Hz to 4 000 Hz. If a private network is forming only part of a 4-wire loop, e.g. in case of national or international connections via a public network, the stability loss at the interface between private and public network should be in all possible configurations > 6 dB in the frequency range 0 Hz to 4 000 Hz.

Connections including one or more 4-wire loops as a source for signal reflections, may - during talking state - contribute with impairments due to multiple echoes and also "Listener Echo". For information about listener echo as a subject of planning see subclause 6.9.

6.5 Quantization Distortion Units (QDUs)

The processs of PCM encoding and decoding according to the "8 bit-law" (A-law or µ-law) as described in ITU-T Recommendation G.711 [21], tables 5 and 6, will result in an additional "Quantization Distortion", perceived as a quantization noise beside the received voice signal (not perceived in a quiescent channel) reducing the signal-to-noise ratio. These distortions are additive, i. e. every A/D-D/A conversion will contribute with a new additional noise.

For transmission planning it is common practice to express this disturbing quantization noise in the unit "Quantization Distortion Unit qdu" with a limitation in form of a maximum number of permitted 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, following the 8 bit A-law (or µ-law) according to ITU-T Recommendation G.711 [21].

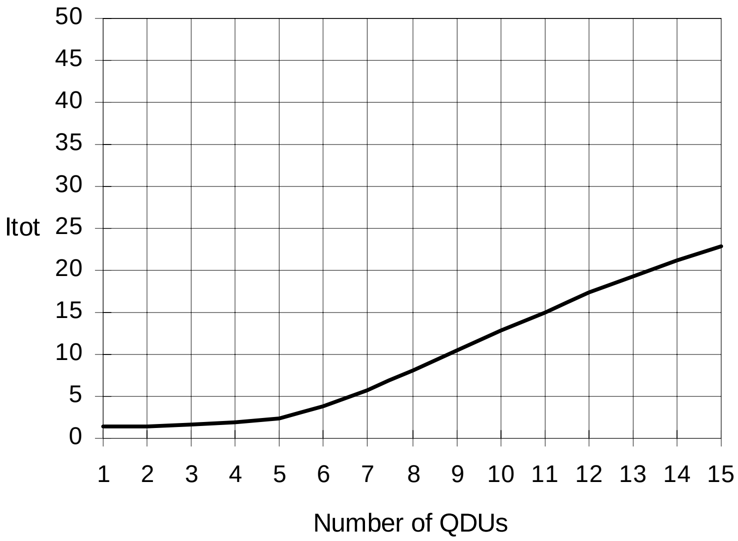

Figure 10: Relation between the number of qdus and the total impairment value Itot

As long as mixed digital/analogue connections should be taken into account for planning, the resulting number of qdus should be subject to planning. Only if fully bit-transparent routing can be assumed, the quantization noise can be neglected in planning. With the increasing use of digital transmission and connection elements in private and public networks, the importance of quantization noise will decrease. 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 a number 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, where the sum of all impairments (Itot) is at least responsible for the perceived speech quality. When using the E-Model, including the number of qdu as an input parameter, it is recommended to determine the number of qdu and to use this as an input to the model instead of the default value (1 qdu).

The parameter qdu in transmission planning is used not only for the normal A/D-D/A conversions but also for 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 other (modern) coding laws than A-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 subclause 6.6).

6.6 Equipment impairment factor

Specific modern coding laws, mainly low bit-rate codecs as described in ITU-T Recommendations e.g. G.728 or according to the GSM Standard, but also ADPCM with different operating bitrates will contribute with distortions causing a decrease of the perceived quality. In contrast to the quantization distortion due to the standard 8 bit coding (A-law or μ-law), these impairments can no more longer be described 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", a value which is obtained by subjective tests and 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 5.1 (addition of impairments on a linear psychological scale) and which therefore can directly be used as an input parameter for the E-Model. Some planning values for Ie for several codecs, taken from ITU-T Recommendation G.175 [20], are listed in table 2.

Table 2: Planning values for the equipment impairment factor Ie

Codec Type	Operating Rate kBit/s	Value Ie	Reference
	40	2	G.726, G.727
ADPCM	32	7	G.721(1988), G.726, G.727
	24	25	G.726, G.727
	16	50	G.726, G.727
LD-CELP	16	7	G.728
	12.8	20	
CS-ACELP	8	15	G.729
VSELP	8	20	IS-54-B, TIA
RPE-LTP	13	20	GSM 06.10, Full-rate
VSELP	5.6	23	GSM 06.20, Half-rate
ACELP	12.2	6 (see note)	GSM 06.60, Enhanced Full Rate
CELP+	6.8	25	
NOTE: provisionally			

6.7 Advantage Factor A

The "Advantage Factor A" is representing an "Advantage of Access" introduced into transmission planning for the first time in ETR 250 [6] and also included in the ITU-T Recommendations G.113 [11] and G.175 [20]. This factor enables the planner to take into account the fact that customers may accept some decrease in quality for the advantage of 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 also as an input parameter to the E-Model. Some provisional values taken from ITU-T Recommendation G.175 [20] 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

It should be noted, that these values are only provisional. There is no confirmation by subjective investigations up to now. Therefore the advantage factor A should be used with care, mainly in the business area of a private network, where the users may judge specific advantages in telecommunication with another degree than in the private domain. The use of the advantage factor A in transmission planning of private networks and the chosen values are in any case subject to the planners decision. However, the values in table 3 should be considered as the maximum upper limit for A.

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

Although the basic planning principle of the present document for all configurations is referred to an end-to-end consideration, the (digital) interface between the private and a public network needs additional control with respect to some parameters. This is derived from ETS 300 283 [4], where limits are given for the so called "Network Connection Point NCP" to control echo in national calls without echo cancellers inserted. Transmission planning for private networks according to the present document is primarely referred to the own customers, i.e. to provide sufficient speech quality within the considered private network. A private network however can also be the source for signal reflections, mainly in hybrids terminating the (digital) NCP. In conjunction with the mean one-way delay of the national connection this may result in a disturbing talker echo for the talker at the far end. Summarizing the content of ETS 300 283 [4] the following limits at the NCP should be also the basis in the present document:

Minimum Sending Loudness Rating	+ 7 dB
Minimum Receiving 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 msec

These values are generally considered as being based on a maximum mean one-way delay of 25 msec for the whole echo path, with 15 msec for the path within the (transit) public network and 5 msec each for the paths within the (terminating) private networks (for more details see section 4.1 of ETS 300283 [4]). The total TELR as the sum of SLR, RLR and echo loss EL is 34 dB for the long term objective. Assuming no further impairments, these values of 25 msec and TELR = 34 dB will result in a total impairment value Itot = 17 (see also figure 8), which is judged between "good" and "adequate" according to table 1a.

If the limit of 5 msec within the private network is exceeded and/or a TELR of 34 dB cannot be met, echo control, also in case of national calls, should be used for the benefit of the far end talker. For all values of the mean one-way delay and the TELR close to the limits or slightly exceeding, a calculation using the E-Model is recommended.

6.9 Parameters not directly subject to transmisison planning

As already mentioned in clause 6, some parameters can be neglected for the benefit of simplification. Although these parameters are input parameters to the E-Model or were used previously in transmission planning, the assumption of an environment in the private and public domain with mainly digital transmission and digital connection elements will decrease the influence of those specific parameters.

Those parameters are the impairment due to the frequency response of unloaded cable sections. Analogue routing via an unloaded cable section can be assumed within the private network only in the lower hierarchy and with only short or medium line lengths. Furthermore this effect of a frequency slope is usually equalized by the preemphasis in the frequency response of the used standard analogue telephone sets. The E-Model does not cover this impairment.

Another parameter which should be considered only in specific applications is the circuit noise. Sources for circuit noise were previously mainly existing in analogue networks using FDM systems and a higher share of cable sections and have been subject to "noise-planning". As long as all switching equipment (PBXs) and transmission elements within the private network are designed according to international and national Standards with respect to noise, its influence to speech quality is negligible. Only in special cases, e.g. interference to analogue cable sections by power lines or other sources, noise should be part of planning calculation. For those applications the E-Model allows the input of the parameter circuit noise Nc in a value referred to the 0 dBr-point. For further information see also clause 8.

In the same way, 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 being part of the connection. The effect of listener echo is mostly depending on the same connection characteristics and network elements as for 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 only, i.e. the parameters WEPL and Tr in the E-Model can remain at their default values during calculation.

Other parameters included in the E-Model may have important influence on speech quality, but are usually not subject to planning. This is mainly valid for parameters in conjunction with the used analogue and digital telephone sets. The relevant parameters are the "Sidetone Masking Rating STMR" and the "Listener Sidetone Rating LSTR" in conjunction with the "Factor D", a value depending on the design of the handset. Mainly STMR, but to some extend also LSTR are - for analogue telephone sets - depending 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 connecting cable.

For the task of transmission planning and for keeping these impairments as low as possible, it is important to control the used telephone sets with respect to all relevant international or national Standards concerning transmission parameters. Furthermore it is recommended to follow a sufficient "Impedance Strategy" at all analogue interfaces. For more information see also annexes A1 and A4 in ETR 250 [6].

A further parameter which is usually not subject to transmission planning is the "Room Noise". In case of unusual environmental conditions, room noise may have an important influence to speech quality, both at the talkers and listeners side. The E-Model is including room noise as a source for impairments, separated for send and receive side. For a normal office environment as it can be assumed in the business domain of private networks, the room noise can be expected in a range of 30 to 50 dB(A). Within this range only minor impairments will be encountered due to room noise, therefore both parameters Ps and Pr can be left at their default values in the E-Model. Only in specific applications, like telephone sets used in factory rooms with a significantly higher room noise, an average value resulting from measurements should be used as input parameter to the E-Model.

6.10 Synchronization

A proper synchronization design is part of the network planning strategy, because poor synchronization will affect the speech quality of the communications: the relevant documents are ETS 300 462-1/6 [32] and ISO/IEC 11573 [33].

In the case of synchronization impairments, slips will occur creating various degradations. This is especially important in situations 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 [18], note 2 of § 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 will be a new kind of impairment, which is not adressed by the present document and by the current version of the E-Model.

7 Calculation of end-to-end parameters

As described more detailed in clause 8, 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 center" referred to as a 0 dBr-point. One of the main tasks when using the E-Model is to transfer the different end-to-end connections including the private network into a form which is similar to the basic reference configuration of the E-Model. In this context it is necessary to clearly identify existing echo pathes, to define the virtual center to be used as the 0 dBr-point and to perform "pre-calculations" for the different parameters to obtain correct results for all input values to the E-Model. The following subclauses provide guidance for these calculations for all main parameters which are subject to planning.

7.1 Overall loudness rating

With respect to the basic principle of the present document that every transmission planning is based only on an end-to-end consideration, an impairment caused by a too high or a too low volume of a connection is consequently only related to the Overall Loudness Rating OLR. In practice however, it is necessary to separate the OLR into a Send Loudness Rating SLR and a Receive Loudness Rating RLR for both transmission directions, since the E-Model is expecting these input parameters separately, both with respect to the 0 dBr-point. To obtain the correct input values SLR and RLR for the E-Model, a "pre-calculation" of these parameters is necessary. The primary task is to define the 0 dBr-point in the considered configuration, followed by an addition of all SLR/RLR values of the used telephone sets and all distributed losses along the whole connection.

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 the digital interface between the private network and a public network will provide this.

The principle of such a precalculation is demonstrated with a configuration shown in figure 11. 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 the PBX A. The hybrid in the extension line circuit is inserting a loss of 7 dB in the receive direction. The transmission element B between PBX A and the transit PBX C is formed by a digital leased line with a bit-transparent routing. The transit PBX C is inserting a digital loss pad of 4 dB in each direction. The path within the public network for a national long distance call is providing a fully digital routing between the private network interface (NCP) and the Local Exchange E at the far end. Within this Local Exchange the 4-wire/2-wire conversion in conjunction with an A/D-D/A conversion is inserting a receive loss of 7 dB. The far end termination is formed by a single analogue telephone set connected to the Local Exchange E via an analogue subscriber line with a loss of 4 dB. The total 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.

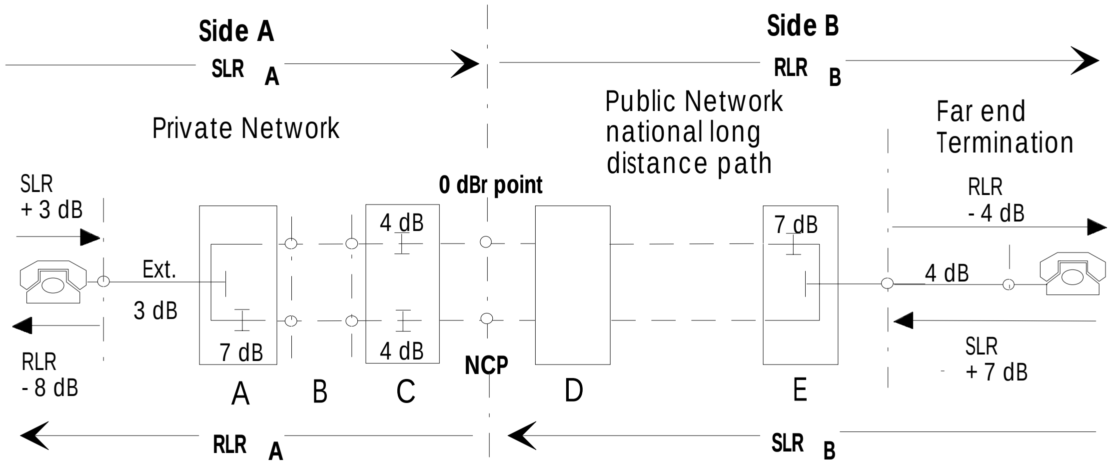

Figure 11: Reference connection for the calculation of SLR and RLR

For this configuration the 0 dBr-point can be defined at the Network Connection Point NCP, the interface between private and public network. With reference to this 0 dBr-point the figure 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 all SLR/RLR and losses should be performed for both transmission directions, so that values for SLR_A, SLR_B, RLR_A and RLR_B are available. 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 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-poin		SLR_B = 7 dB	RLR_B = 3 dB

It is very important to select from these results the correct values as input values for the E-Model (see also clause 8). 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 listener (receive side) and the customer at the far end termination is the talker (send side). Therefore the values for the transmission direction from far end termination to private network should be used as input values to the model:

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

The OLR for this transmission direction is:

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

The OLR for the opposite direction is:

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

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

For 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 is set to 65 dBA with Ps 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 to Itot = 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 Ps is set to 65 dBA and Pr remaining at the default value. The total impairment value in this direction is Itot = 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 control also the requirements at the NCP with respect to SLR and RLR of the private network as described in subclause 6.8. 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. In this example the requirements of subclause 6.8 with the given minimum values of SLR = 7 dB and RLR = 3 dB are met.

7.2 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 calculations of the perceived quality, using the E-Model, two input parameters must be available and subject to a pre-calculation:

- mean one-way delay T in msec;

- talker echo loudness rating TELR in dB.

For more information about the effect of echo see also annex A.

For the investigation of echo (especially in conjunction with the E-Model) it is important to know that the parameter TELR is including the SLR and RLR of the talkers telephone according to the formular

$$TELR = SLR + EL + RLR$$

where EL is the (weighted) echo loss of the echo path. This means that 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: This procedure, that SLR and RLR are used twice, is based on the principle and the algorithm of the E-Model as it is 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 those programs and the E-Model. Therefore it is necessary to clearly identify what input values are required and in which form they are expected in the PC program used, to avoid wrong results. For further information see also clause 8.

Furthermore it should also be noted, that impairments caused by talker echo are referred to the talker at the receive side, according to the principle of the impairment factor method and the E-Model. This should be taken into account carefully, when selecting the different parameters as input for the model.

In some applications the considered reference connection may contain more than one echo path. The following subclauses 7.2.1, 7.2.2 and subclause 8.5 are giving more guidance for the calculation procedure in both cases.

7.2.1 Calculation for connections with one echo path

The following example for the calculation is based on the reference connection shown in figure 12. A digital telephone set according to TBR 8 [1] with SLR = 7 dB and RLR = 3 dB and with a mean one-way delay of 1,5 msec is connected to the PBX A within the private network. This PBX A is connected with the transit PBX C via a digital leased line B with bit-transparent routing and a mean one-way delay of 2 msec. The transit PBX C is inserting a digital pad of 3 dB in each transmission direction for voice calls. Both PBXs A and C are contributing with a mean one-way delay of 1 msec each.

The public network is providing a fully digital routing up to the far end Local Exchange 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. According to the information provided by the public network operator, the mean one-way delay within the public network can be assumed with 10 msec for a national long distance call. The hybrid in the Local Exchange E can be assumed to have an average (weighted) echo loss EL of 24 dB, including the loss of 7 dB in the receive path of the hybrid.

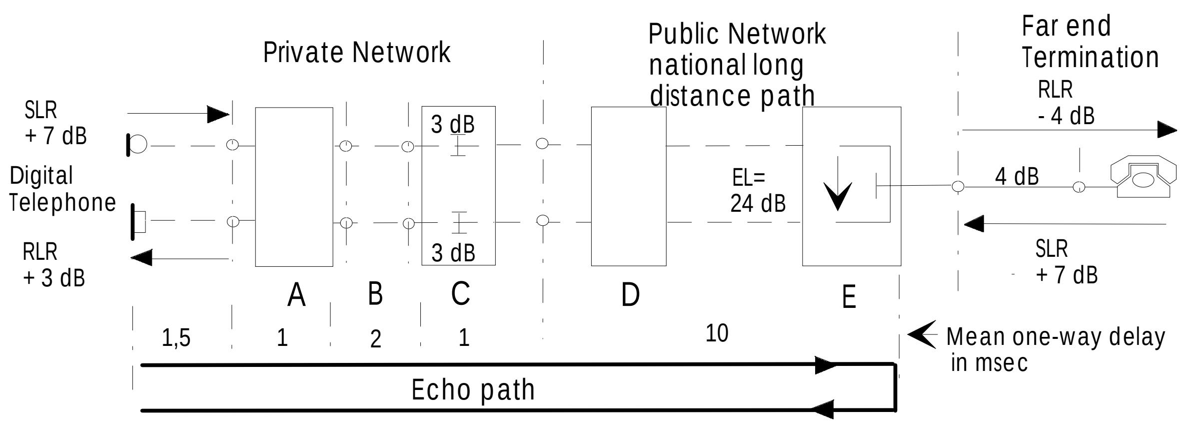

Figure 12: Reference connection for echo calculations with one echo path

A clear identification of the echo path must be the first step of every calculation. For the example assumed in figure 12 the echo path is consisting of the digital telephone set (where the talker echo is perceived), the complete path via transmission and switching elements between telephone set and Local Exchange E, the hybrid in E and backwards. This echo path is marked as part of the whole connection in figure 12.

In a second step the two relevant input parameters to the E-Model, the mean one-way delay T and the TELR of the echo path should be calculated. It is important to note that only the "one-way" delay values should be taken as an input value for the model, although the echo signal is delayed by twice the value.

If once the echo path is defined, the first input value for the calculation, the mean one-way delay T, can be found by an addition of all separate values of the different elements. For the example in figure 12 above this sum calculates to T = 15,5 msec. 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 telephone set. For the example above this value is calculated to TELR = 40 dB.

With these results of T = 15,5 msec and TELR = 40 dB used as input values, the calculation for the assumed configuration of figure 12 with the E-Model is resulting in a total impairment value Itot = 9,9, a value which is judged as "good" according to table 1a.

7.2.2 Calculation for connections with two echo paths

Due to increasing digitalization in public and private networks, configurations with more than one echo path will disappear more and more and should be avoided wherever this seems possible. The following example is based on the use of a cordless telephone according to the DECT-Standard [2, 3] which is connected analogue to a PBX within the private network. The additional delay introduced due to this DECT-Standard is not negligible, hence this cordless telephone is forming a separate echo path together with the second echo path via the public network, the same configuration as in 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 echos with different volume and delay.

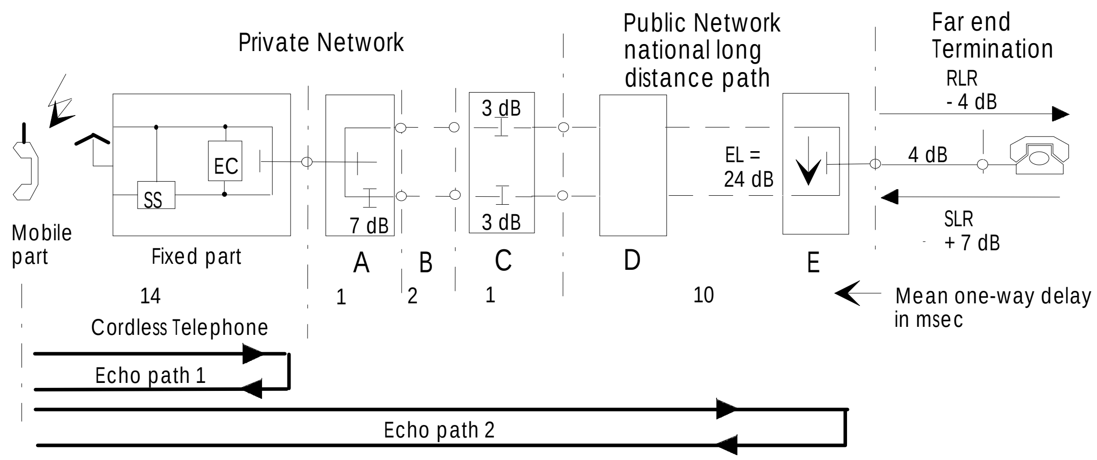

Figure 13: Reference connection for echo calculations with two echo paths

To determine the expected overall quality, a special application of the E-Model should be performed here, because the E-Model does not yet handle and calculate two simultaneous impairments of the same type but with different amount. The E-Model does not allow to simply add 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 like masking of one echo by the other may influence the perception of quality.

In this situation it is generally recommended to consider in a first step the impairments due to echo only calculating the two echo paths 1 and 2 separately, i.e. for each echo path the relevant parameters mean one-way delay T and TELR should separately be determined, assuming the other one as not existing. In a second step, the two calculated impairment-values for the two echo paths should be added separately with a specific formula, not provided by the E-Model. Finally this result is then combined with the result of E-Model calculations with all other parameters causing further impairments (e.g. Ie-value for ADPCM, loss) in this configuration. This procedure and how to handle the E-Model in this special application is described in more detail in subclause 8.5, together with the general guidance for the application of the E-Model in planning practice. The following paragraphs are describing only how to determine the parameters TELR and T for the two echo paths.

For the cordless telephone in figure 13 the fixed part (basestation) should provide a 4-wire / 2-wire conversion for the connection to an analogue 2-wire interface. The used hybrid is then a source for signal reflections and is forming 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, some specific facts in conjunction with the ETS 300 175-8 [3] should be taken into account.

Cordless telephones according to the ETS 300 175-8 [3] are inserting an additional mean one-way delay of approximately 14 msec between acoustic interface of the portable part and the network interface of the fixed part. This high value would cause echo effects in most of all applications. Therefore the Standard requires precautions to suppress the reflected signal using an echo canceller EC together with a soft suppressor SS. For more information about these cordless telephones see also annex A, subclause A 5.3.3.

These "integrated" echo equipment should be taken into account when determining the TELR of this echo path 1. Figure 14 shows in more detail all relevant parts which are mainly included in the fixed part of such a telephone. 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 available again in standard PCM format.

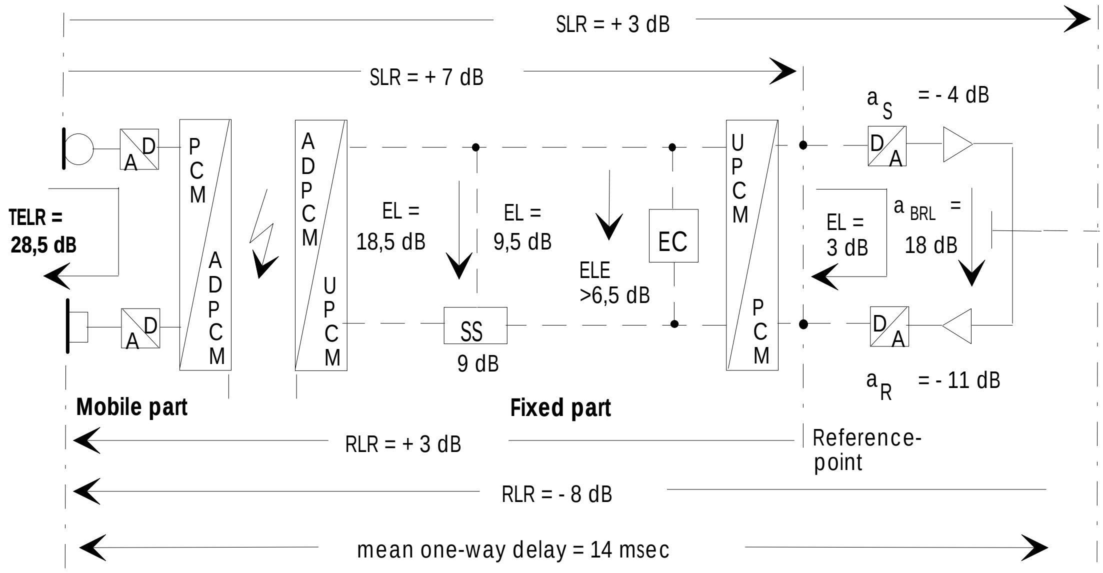

Figure 14: Details inside a cordless telephone with respect to echo control

According toTBR 10 [2] the system is adjusted to SLR = 7 dB and RLR = 3 dB referred to this reference point. These loudness rating values would not meet the requirements when connected to a 2-wire analogue interface. In the example of figure 14, a SLR = 3 dB and RLR = -8 dB are assumed (similar national requirements are possible). The hybrid inside the fixed part (at the right side of the reference point in figure 14) is not only providing the necessary A/D-D/A and 4-wire/2-wire conversion, but also in its analogue part an adjustment to a_S = - 4 dB in sending direction and a_R = - 11 dB in the receive path 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 this balance return loss is reduced by the sum of the adjustments a_S and a_R resulting in an echo loss EL of only 3 dB. To improve this low echo loss an echo canceller EC is used, compensating the reflected signal partly. This is expressed in terms of an "Echo Loss Enhancement ELE", where the DECT-Standard requires > 6,5 dB. The echo loss is increased to 3 dB + 6,5 dB = 9,5 dB. A further improvement is available by the insertion of a "Soft Suppressor SS" in the receive path inside the fixed part. This SS is inserting an additional loss of 9 dB (DECT-Standard: 9 - 12 dB) as long as a signal is detected in the send path. The resulting value of 9,5 dB + 9 dB = 18,5 dB is, together with an SLR = 7 dB and RLR = 3 dB, in total providing a TELR of 28,5 dB. The mean one.way delay of this echo path 1 is 14 msec.

NOTE: Although the values of SLR and RLR are referred to the reference point, they can be assumed also 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 encountered - which is valid for the consideration of talker echo - but not during listening. Therefore the RLR values as shown in figure 14 are given for a disabled SS.

For the echo path 2 the calculation of the relevant values for mean one-way delay T and TELR is similar to the calculation in subclause 7.2.1, but with some different values. Following all separate values for delay along echo path 2 as marked in figure 13, the sum for the mean one-way delay of echo path 2 is T = 28 msec.

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

For the return path the digital pad in PBX C (3dB) and the hybrid in PBX A (7 dB) is included, followed by the entire receive path of the cordless telephone with RLR = - 8 dB. However this value should be increased by the loss of the SS (which is enabled also for this echo path 2) with 9 dB, i.e. the value is RLR = + 1 dB for the purpose of calculating the TELR. The echo canceller in the fixed part is not supporting echo path 2 since the processing capability of the echo canceller with respect to the echo path delay is only 4 msec. The sum for echo path 2 using the values as shown in figures 13 and 14 is TELR = 41 dB.

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

- echo path 1 TELR = 28,5 dB T = 14 msec;
- echo path 2 TELR = 41,0 dB T = 28 msec.

7.3 Transmission time in echofree connections

As already described in clause 6.3, a very long delay may cause impairments other than those due to echo. Furthermore it is assumed that perfect echo cancelling is provided in those "echofree" connections. For planning purposes this parameter should usually be considered only in international connections or when a routing via a satellite link is possible. For the use in the E-Model only the absolute delay Ta in msec should be calculated. It is resulting as the sum of all **one-way delay** values of the different network elements throughout the entire connection. It should be noted, that in any case all elements including the telephone sets at both ends should be considered independant of any echo sources like hybrids and inserted echo cancelling devices. Specific transmission elements or connection elements may have different values of delay in the two transmission directions. In these cases the arithmetic mean of both values should be used.

7.4 Quantization distortion units

The E-Model is expecting this input parameter as the number of Quantization Distortion Units (qdu) as already described in subclause 6.5. However it should be noted, that only 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 [21] is considered as one qdu. When performing the summation for the entire connection each pair of "Coder" and and its subsequent "Decoder" should be clearly identified.

For other coding laws than those contained in ITU-T Recommendation G.711 [21] (e.g. ADPCM), the impairments due to distortion should not be expressed as number of qdu, but as the equivalent Equipment Impairment Factor Ie (see next subclause). For specific elements such as digital loss or gain pads, echo cancelling devices or (digital) conference circuits inserted in the connection considered, a standard value of 0,7 qdu for each of these elements should be used in the calculation.

Also if a fully digital routing is provided, a minimum of 1 qdu should be taken into account in any case, independant whether the Coder/Decoder is located either 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 a digital environment.

7.5 Equipment Impairment Factor

As already described in subclause 6.6, modern coding laws will cause impairments due to distortions. In contrary to the standard coding and decoding according to the A-law or µ-law (ITU-T Recommendation G.711[21]), these impairments are expressed with their "Equipment Impairment Factor Ie" instead of a quantization distortion unit (qdu). The factors Ie for the different coding laws and operating rates are given in table 2. For planning practice only the algebraic sum of all Ie - values along the investigated connection should be calculated and inserted into the E-Model as an input value. However it is very important to clearly identify - using the reference configuration - the actual location of the Coder and Decoder of such a low bit-rate section, since such a low bit-rate section may include several transmission elements and connection elements. A connection may also include more than one section using the same or different types of those low bit-rate coding, which should be considered for calculation.

8 Application of the E-Model in planning practice

8.1 General

The basic planning principles as recommended in the present document are based on the use of the E-Model (see also clause 5) for planning calculations, to obtain an evaluation of the expected quality for a considered configuration. This replaces the usual comparison of the different transmission parameters with given limit figures. This is therefore deviating from previous planning methods mainly for private networks. The application of this new principles and the use of a computation model need therefore some introduction and guidance on the practical use, which is contained in this subclause.

A detailed description of the E-Model and the necessary algorithm is given in annex B. It should be noted, that the use of this E-Model needs sufficient knowledge about the basic reference configuration of this E-Model and the different input parameters to avoid wrong results. This is described in detail in subclauses 8.2, 8.3, 8.4 and in 8.5 (for the special configuration with two echo paths). According to the number and complexity of the used formulas, the calculations will in most cases be performed by computer programs. Even in this case it is absolutely necessary to be familiar with the program itself, the handling of input parameters and the limits for its application. More information is contained in subclause 8.6.

The E-Model is including a variety of transmission parameters while not all of them are varied for planning purposes (see also subclause 6.9). These parameters should be set to and remain at a default value (which is not zero in all cases) during the calculation run. Furthermore, the algorithm of the model is based on the results of subjective tests, where the different parameters have been varied in a specific range only. When using the model with input parameters outside these ranges, the results obtained will not have been validated. Therefore, the use of those values should be avoided. Subclause 8.7 is listing all default values and the permitted ranges for each parameter.

8.2 Reference configurations

For the comprehension of the basic principles of the E-Model it is necessary to refer to a reference configuration, describing a telephone connection and showing nearly all transmission parameters with influence on the perceived speech quality. This reference configuration, shown in figure 15, is basically divided into a "Send Side" and a "Receive Side" with a virtual center referred to as a "0 dBr-point". It is one of the most important assumptions in this model, that the issue of perceived quality is referred only to the "Receive Side", i.e. the listener during a call, also if impairments will be encountered during talking, such as sidetone in conjunction with room noise and echo effects. The basic reference configuration in figure 15 is forming a "4-wire loop" to include also impairments such as talker echo and listener echo which are only of minor influence in a fully digital (4-wire) connection.

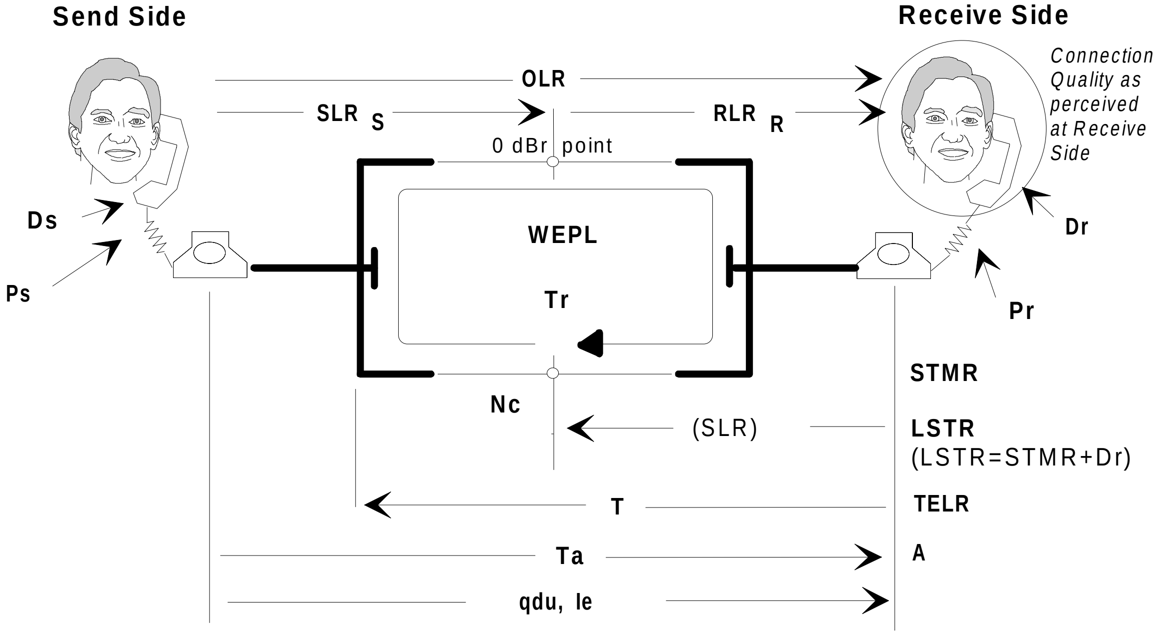

Figure 15: Basic Reference Configuration of the E-Model

Along the sequence of different steps in network planning, one step is usually resulting in a reference configuration for the investigation of a connection via the network which has been identified as the most critical one. This reference configuration shall contain all relevant transmission elements and connection elements. This is helpful to identify and calculate for a specific parameter the sum via the entire connection. In most situations however, this reference configuration will deviate more or less from the basic reference configuration of the E-Model. To avoid wrong results when entering the different parameters into the E-Model - sometimes as the result from precalculations - it is recommended to transform the reference configuration of the investigated connection into a form being nearly identical with the basic E-Model configuration as shown in figure 15.

The following figures will provide guidance when reducing the actual planning configuration into a E-Model-related configuration. Mainly for those "working" configurations which are not consisting of a closed 4-wire loop but are in a 2-wire/4-wire -, 4-wire/2-wire or fully 4-wire structure the consequences for the used parameters are described.

In figure 16 the same 2-wire/2-wire structure is shown as in the basic E-Model configuration, but divided into different sections for better comparison with the actual connection. With respect to the virtual center of the connection related to a 0 dBr-point, the figure is in general 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 with their own Loudness Rating Values SLR A, RLR A and SLR B, RLR B.

The factors Ds at the Send Side and Dr at the Receive Side, depending on the handset design are usually not subject to planning, but 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

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

and are influencing the perceived quality only at the Receive Side.

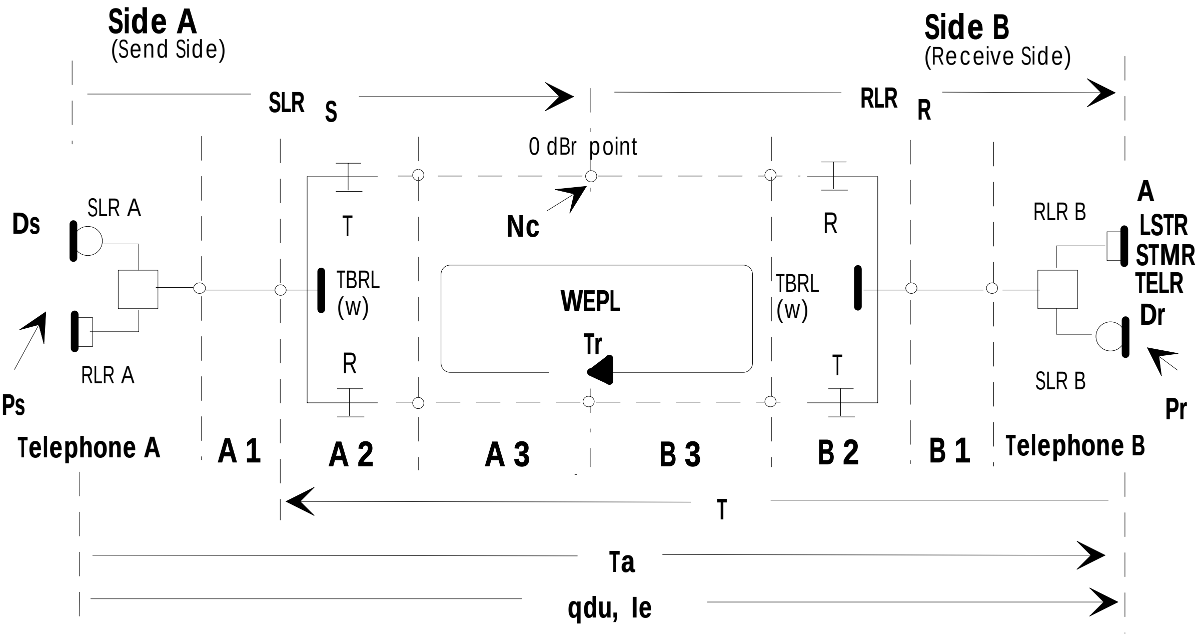

Figure 16: Working configuration for 2-wire / 2-wire connections.

The parameters STMR, LSTR and D are usually not part of the planning and should be kept at their default values with the exception of specific telephone circuits, incorrect impedance and/or for handsets deviating from common design.

The room noise Ps at the Send Side may have influence to the signal to noise ratio as perceived at the Receive Side, while the room noise Pr at the Receive Side may decrease the perceived quality via the Sidetone path. For telephone sets in normal office environment 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 care and is described in more detail in subclause 8.3.

The sections A1 and B1 of figure 16 should be understood as an 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. PBXs with analogue switching matrix). Therefore mainly loss values should be calculated for A1 and B1.

Although usually only part of a switching element (e.g. subscriber line card, trunk card), sections A2 and B2 are shown more in detail, since they are consisting of the 4-wire/2-wire conversion circuits (hybrids) and loss pads R in receiving and T in sending direction, resulting in conjunction with the degree of matching (TBRL) in an important influence on speech quality.

The Terminal Balance Return Loss TBRL, shown in the figure 16 is expressing this degree of impedance matching between the Balance network of the hybrid and the impedance of the terminating 2-wire section. This TBRL which should be available as a weighted value TBRL(w) is not directly used as an input parameter for the E-Model, but is necessary for the precalculations of the TELR.

Finally, sections A3 and B3 in figure 16 are representing the complete digital part of the connection between the hybrids - in some cases also providing the A/D- and D/A-conversion - and the point in the configuration which has been declared as the 0 dBr-point. Also the interface between different networks (e.g. public and private) can be part of A3 and B3. These sections may contain several switching elements and transmission elements, including digital loss or gain pads, low bit-rate coding etc., contributing with parameters such as loss, delay and distortions. In a precalculation all the sums of these different parameters should be defined.

The whole 4-wire loop extending via the sections A2, A3 and B2, B3 may contribute to impairments due to Listener Echo and is characterized by the parameters Weighted Echo Path Loss WEPL and the Round Trip Delay Tr. WEPL and Tr can be calculated as the sum of all delay values, mainly in sections A3 and B3 and the 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 within the hybrids in A2 and B2.

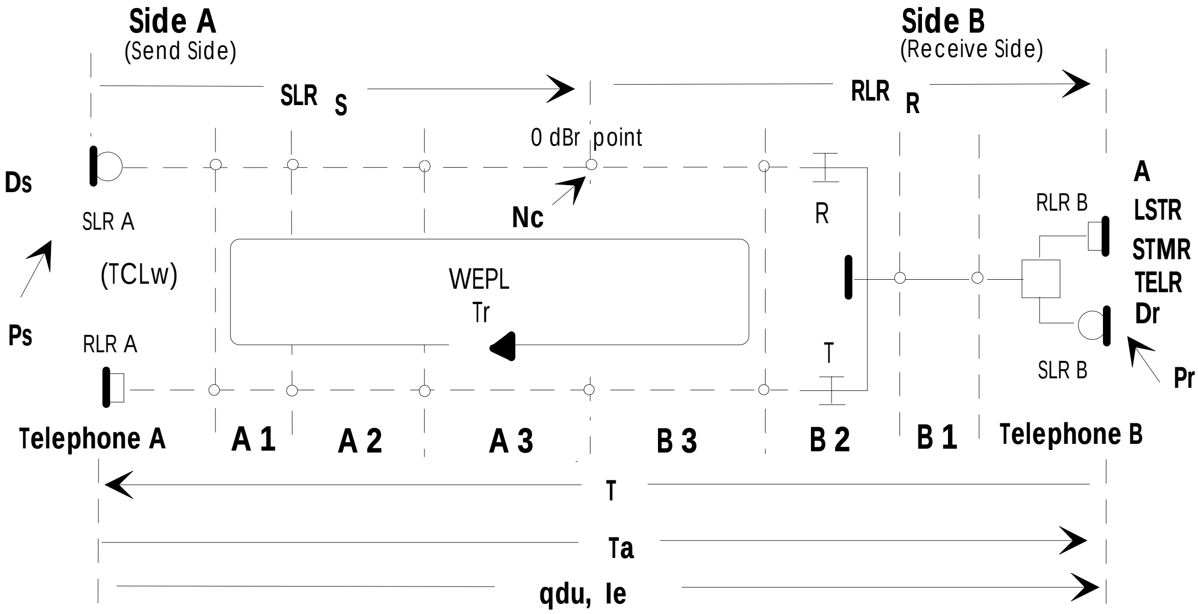

Figure 17: Working configuration for 4-wire / 2-wire connections.

The working configuration as 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. Also in this case the values for SLR A and RLR A are only referred to the digital telephone A. However in this 4-wire termination an additional parameter, the weighted Terminal Coupling Loss TCLw (of the digital telephone set) should be considered which is not used as a direct input value to the E-Model. This TCLw is characterizing the coupling between receiver and microphone (including further acoustical and electrical coupling pathes) where a possible source for signal reflection may arise (see also annex A, subclause 4.2.3).

Theoretically this TCLw should be included in the precalculation of the TELR value to include impairments due to echo perceived at the Receive Side. This TCLw should also be part of the precalculations for WEPL. With respect to Talker Echo and Listener Echo the TCLw in this configuration is replacing the Echo Loss (TBRL and 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 should include also section A1 and the telephone in contrary to the configuration in figure 16, where the echo path with respect to the talker at the Receive Side is terminated already in section A2.

All issues and explanations as given for the working configuration in figure 16 are also valid for this configuration.

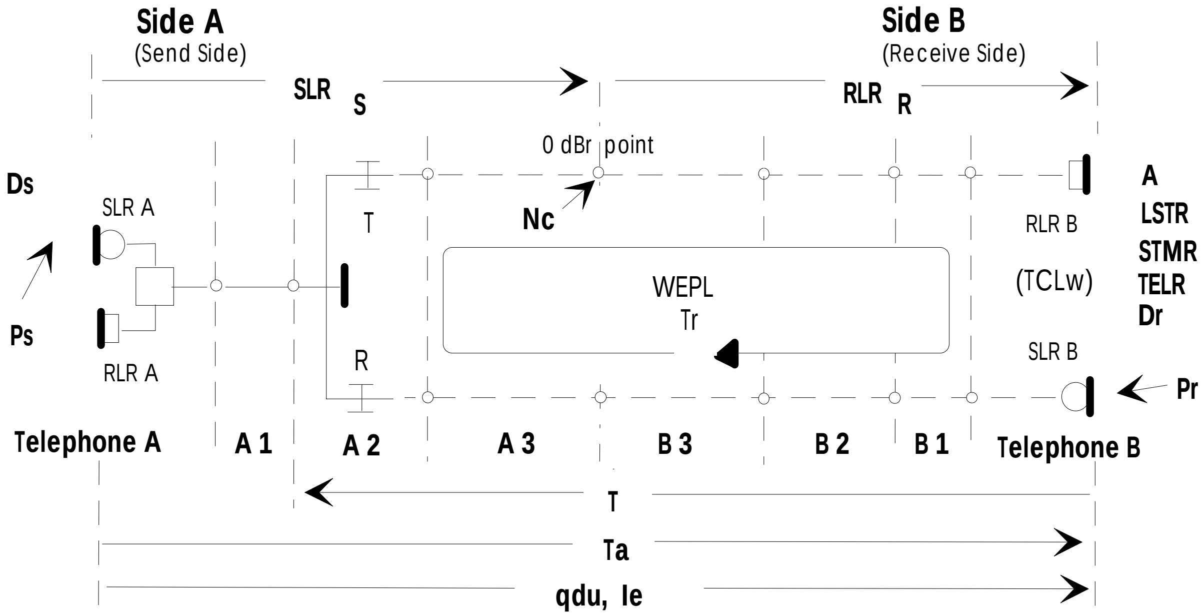

Figure 18: Working configuration for 2-wire / 4-wire connections

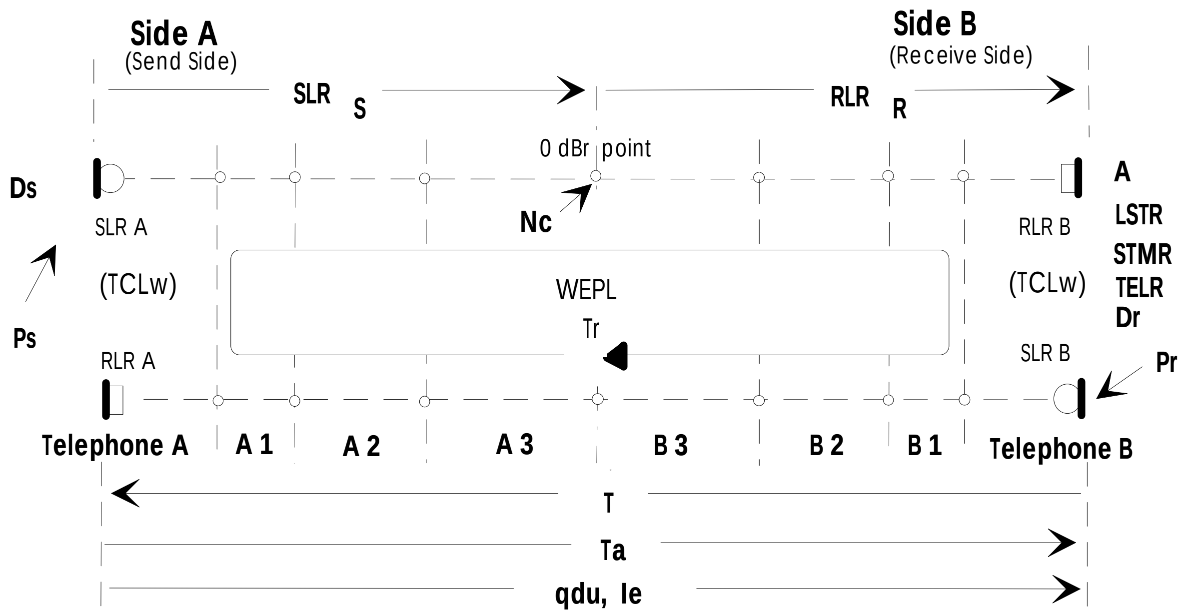

Figure 19: Working configuration for fully digital connections

Figure 18 is illustrating the opposite working configuration with the 4-wire termination (digital telephone set) at the Receive Side and figure 19 for a fully digital connection with 4-wire termination on both ends. The application and precalculation of the input parameters in figure 18, mainly 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 European Standards (TBR 8) with a TCLw of more than 40 dB are used and if the Round Trip Delay Tr is low.

8.3 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 considered carefully. This is mainly true for Loudness Rating (LR) values. Basically for the telephone sets only LR-values should be used which have been measured according to the method as described in ITU-T Recommendation P.79 [23]. The principles used in the main algorithm of the E-Model are expecting a total value for SLR_S covering the whole Send Side and a total value for RLR_R for the whole Receive Side both between the acoustical interface (microphone or receiver) and the 0 dBr-point. With respect to figures 16 to 19 these values are usually calculated as the sum of the SLR A of telephone A and all loss values in sending direction in the sections A1, A2 and A3. For the Receive Side the equivalent RLR is obtained as the sum of all losses in receive direction of sections B3, B2, B1 and the RLR B of telephone B. For the investigation of the transmission direction from A to B the values RLR A and SLR B are not relevant.

Beside 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 are indeed depending on the SLR and RLR of a telephone set. However for practical planning purposes usually a fixed value for both parameters in conjunction with the SLR and RLR is used and stated by the manufacturer of the telephone sets. The input parameters STMR and LSTR should remain at their default values and only be modified if in case of analogue telephone sets an impedance mismatch should be considered.

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

To include the impairments due to talker echo the algorithm of the E-Model is expecting two parameters, the mean one-way delay T in msec 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 only referred to the Receive Side. As already explained in subclause 7.2, the value for TELR should be obtained in a precalculation according to the basic formula:

$$TELR = SLR + EL + RLR$$

where SLR and RLR are again Loudness Ratings, but in this case only the values of the telephone set e.g. SLR B and RLR B with respect to figures 16 through 19. Although the RLR B is already part of the basic input value RLR_R , the formulas of the E-Model are not providing an automatic inclusion of the Loudness Rating values, i.e. RLR_R and TELR should be calculated separately using the same value for RLR B again.

The Echo Loss EL in the formula given above is the sum of all losses along the echo path e.g. in the working configurations the sections B1, B2, B3, A3 and A2 in figures 16 and 18 and additionally A1 in figures 17 and 19. The losses in sections B1, B2, B3 and A3 should be identified and included for both transmission directions, i.e. the loss of section B1 in figures 16 and 17 is included twice within the echo path. For the terminating hybrids in section A2 of figures 16 and 18, R- and T-pads and the TBRL(w), or TCLw in figures 17 and 19, are contributing to the TELR. It is absolutely necessary to use only a "weighted" value for the TBRL.

A further parameter, the Circuit Noise Nc can cause wrong results if this input value is not calculated correctly. The E-Model is expecting those noice sources with "noise levels" as they would appear at the 0 dBr-point. Usually circuit noise is no more longer a major factor in a digital environment and can be neglected in most applications, i.e. the default value of -70 dBm0p can be used. But, mainly in an analogue environment as in sections A1 or B1 of figure 16, noise may arise e.g. due to longitudinal interference into cables by power tractions. If those noice sources cannot be neglected, the noise level should be precalculated into an equivalent value at the 0 dBr-point.

NOTE: Longitudinal interference into a cable in section A1 of the working configuration in figure 16 will result in a transversal noise level of - 50 dBmp at the interface between sections A1 and A2. If the T-pad in the hybrid of section A2 has a loss of 3 dB, the noise level is reduced by this pad accordingly, resulting in a value of - 50 dBmp - 3 dB = - 53 dBm0p at the 0 dBr-point.

8.4 Interpretation of the results

The present document and the recommended planning principles are based on the Equipment Impairment Factor Method. The result of any planning investigation as an issue of the perceived quality is handled with the Total Impairment Value Itot as the final result. The E-Model and its algorithm however, as described in detail in annex B, is providing this Total Impairment Value not directly, but a result to be considered as 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 Itot = 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 4 (see subclause 8.7), characterizing 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:

$$\text{Itot} = 94{,}3 - \text{R}$$

NOTE: The recommended default values (see subclause 9.7) are using the values SLR = 7 dB and RLR = 3 dB according to ETSI TBR 8 [1] for digital telephone sets. The resulting sum of SLR + RLR = 10 dB is slightly deviating from the optimum value of 8 dB as derived from subjective tests and being 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.

As already stated in subclause 5.2, provisions should be made that not only the R-Value or the equivalent value for Itot are avilable after a calculation run, but also the specific results for Id, Is and Ie. This is useful to recognize the different amount of contribution to Itot when investigating solutions towards an improved quality.

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

A special procedure should be performed when a configuration with two effective echo paths should be calculated with the E-Model. Since the algorithm of the E-Model is not covering the handling of two impairments of the same type contributing simultaneously to the overall quality, this special procedure is necessary as described in the following paragraphs.

The procedure is derived in general from the same basic principle of the E-Model, 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, only caused by the two different echo effects, seems not to be correct. Assuming that for a human listener the quality perception and its judgement is more influenced by the echo with the higher impairment value and that also some effects of masking may occur, a square root addition is recommended.

If Idte_1 and Idte_2 are the two impairments only due to the echo of the two echo paths, these two values are calculated in a first step 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 used as input values to the E-Model. The two results for Idte_1 and Idte_2 are then combined to a total (sum-) value Idte for both impairments using the following equation

$$\text{Idte} = \sqrt{\text{Idte}_1^{\,2} + \text{Idte}_2^{\,2}}$$

This value Idte is now representing all impairments due to echo effects. In a 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 now of course be set to their default values (T = 0 msec, TELR = 65 dB). The result Itot of this second calculation is then combined with the impairment value for the two echo paths Idte by a simple addition.

$$\text{Itot} = \text{Itot} + \text{Idte}$$

For the actual calculation of this configuration with two effective echo paths, a decision about the working configuration (see subclause 8.2) 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 use of default values is also valid for the absolute one-way delay Ta. As explained in subclause 6.3, the absolute one-way delay Ta is causing major impairments due to too long delay only if this value is exceeding 150 msec (ITU-T Recommendation G.114 [12]). Also the algorithm of the E-Model is setting the corresponding impairment to Idd = 0 for Ta < 100 msec (see annex B, subclause B 3.3), i.e. for the configuration of figure 13 Ta can be left at its default value of 0 msec for the calculation.

When the configuration as assumed in figure 13 is examined with respect to a possible source for impairments due to listener echo, it can be seen that there is a first 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 already stated in subclause 6.9, usually the influence of listener echo is negligible as far as there is sufficient control of the talker echo. Calculations for the configuration and values of figure 13 are resulting only in an increase of the total impairment value Itot in the range of 0,7 - 0,8 and is therefore negligible. For the calculation the corresponding values for round-trip delay and weighted echo path loss can be left at their default values Tr = 0 and WEPL = 110 dB also if the actual values are slightly deviating.

The input parameters T and TELR for both echo paths have already been calculated in subclause 7.2.2 with TELR = 28,5 dB, T = 14 msec for echo path 1 and TELR = 41 dB, T = 28 msec 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 8.4 gives for echo path 1:

$$\text{Idte1} = 94{,}3 - 77{,}9 = 16{,}4$$

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

$$\text{Idte2} = 94{,}3 - 86{,}2 = 8{,}1$$

The results for Idte1 and Idte2 are now combined to one value using the square root relation:

$$\text{Idte} = \sqrt{\text{Idte}_1^2 + \text{Idte}_2^2} = \sqrt{16{,}4^2 + 8{,}1^2} = 18{,}3$$

For the final calculation including all other impairments without echo effects, a working configuration should be selected for the correct identification of the different parameters and their use as input parameters to the E-Model. For a configuration as in this example, the working configuration for 2-wire/2-wire connections as shown in figure 16 is recommended. 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 0 dBr-point can be defined between public and private network, i.e. at the digital interface between exchanges C and D.

With the loudness rating values for the entire cordless system (telephone B) and the far end termination (telephone A) 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 further gain or loss is inserted in the path between the telephone and the 0 dBr-point. This means 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 is including the digital pad with 3 dB in exchange C, the loss of 7 dB in the hybrid of exchange A and the RLR = - 8 dB of the complete cordless system (see figure 14), resulting in RLR = 2 dB. It is important to note, that in this calculation where the listening conditions are judged, the soft-suppressor SS should be assumed to be disabled. The only further impairment in this connection is the use of 32 kBit/sec ADPCM coding within the cordless system, which should be taken into account with the input parameter for Ie set to a value of 7 according to table 2.

The calculation for the second step can now be performed with the modified input values for RLR = 2 dB and Ie = 7, while all other parameters, mainly T, Tr, Ta, TELR and WEPL remain at their default values. Using the E-Model the result is R = 88,1 or:

$$\text{Itot} = 94{,}3 - 88{,}1 = 6{,}2$$

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

$$\text{Itot} = \text{Idte} + \text{Itot} = 18{,}3 + 6{,}2 = 24{,}5$$

The E-Model can also be used to calculate the corresponding values for the mean opinion score with MOS = 3,6, the percentage good or better with GOB = 73,7 % and the percentage poor or worse with POW = 5,8 %, a result which can be judged as an expected medium quality.

This result with Itot = 24,5 seems to be high, considering that those configurations 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 value contributing to the total value is echo path 1 with Idte1 = 16,4, i.e. the echo path via the hybrid of the cordless telephone. For the suppression of this echo, an echo canceller is assumed with an echo loss enhancement of only 6,5 dB, the minimum value according to the DECT-Standard [2, 3]. In practice however, higher values can be expected, which should be stated by the supplier for planning calculations. In any case it is recommended to connect those cordless telephones via a digital interface to the switching equipment, since then only echo path 2 is effective, resulting in a total impairment value in the range of only Itot = 14...16.

8.6 Use of computer programs

The E-Model, recommended main tool for all planning purposes in the present document, is based on a variety of partly complex formulas (see annex B). For the benefit of the planner in those cases computer programs performing the necessary calculations are used. Irrespective whether those programs are developed by the planner himself, or if programs available from other sources are used, it is in any case strongly recommended to 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, mainly for the handling of input parameters, necessary precalculations, storing of often used configurations etc. As described in subclause 8.3, the correct handling of Loudness Rating values in conjunction with the necessary precalculations is very important and can be supported by 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 for specific sections precalculations are necessary also outside the program.

Although the recommended planning principle in the present document is based on the Total Impairment Factor Itot or the Overall Rating R as the result of calculations, the E-Model is additionally providing formulas to calculate the corresponding values for MOS, POW, GOB and TME (see also clause 5). Also computer programs may provide the results of a calculation in these different issues. The analysis of results in MOS, POW, GOB and TME needs specific knowledge and experience with subjective tests, therefore the final decision about an investigated configuration should in any case be based on the result for the Total Impairment Factor Itot and the corresponding guidance as given in clause 6.

When planning private networks with respect to the expected perceived speech quality, connections identified as the most critical ones will be used as reference configurations and be investigated with the E-Model. Therefore only one specific set of input parameters with one result for Itot is describing the investigated reference configuration. Most of the computer programs however, will also have the 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, for practical purposes of planning those features are not absolutely necessary.

In general, such computer programs should meet the following minimum requirements:

- control of all actual parameter values should be possible;
- input of parameter values outside the permitted range should be refused;
- if a program is also providing the input parameter Overall Loudness Rating OLR as the sum of SLRS and RLRR and the feature to vary this OLR, the variation of SLRS and RLRR should be performed identical, i.e. increasing or decreasing both by the same steps, half of the OLR-steps each;
- the result should also include - if necessary - the specific amount of the delay impairment factor Id, the simultaneous impairment factor Is and the equipment impairment factor Ie.

The correct calculations according to the algorithm of the E-Model should be checked, e.g. whether the result is R = 94,3 if all input parameters are set to their default values. An additional check with the parameter settings and the corresponding results as given in the planning examples of annex D should be performed.

8.7 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 is influencing the result. Therefore it is absolutely necessary to keep those parameters which are not involved in a specific configuration at a default value. When using computer programs it is strongly recommended to control 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 which is equivalent to the optimum quality and realistic values for some parameters. The SLR_S and RLR_R are deviating by 1 dB from their optimum value to be in accordance with requirements in TBR 8 [1]. This reference to European Standards is also valid for the parameters STMR, LSTR and D. The number of qdu is set to 1 instead of 0, since also in a fully digital connection a minimum of one PCM coding/decoding process is involved even if low bit-rate coding is used which itself shall only be included into the calculation as an Ie value.

As already stated in subclause 8.1 the algorithm is based on the results of subjective tests, varying the different parameters only in a specific (realistic) range. The use of a parameter setting outside this range should be avoided, since the result of calculations is no more longer validated. Table 4 gives all default values and the permitted range for each of the parameters. The parameters should be seen in accordance with the basic or working reference configurations as shown in figures 15 through 19.

Table 4: Default Values and permitted Ranges for the Parameters

Parameter	Abbr.	Unit	Default Value	permitted range	Remark
Sending Loudness Rating	SLR_S	dB	+7	0 ... +18	note 1)
Receiving Loudness Rating	RLR_R	dB	+3	-5 ... +14	note 1)
Sidetone Masking Rating	STMR	dB	15	10 ... 20	note 2)
Listener Sidetone Rating	LSTR	dB	18	13 ... 23	note 2)
D-Value of Telephone, Send Side	Ds	-	3	-3 ... +3	note 2)
D-Value of Telephone Receive Side	Dr	-	3	-3 ... +3	note 2)
Talker Echo Loudness Rating	TELR	dB	65	5 ... 65	
Weighted Echo Path Loss	WEPL	dB	110	5 ... 110	
Mean one-way Delay of the Echo Path	T	msec	0	0 ... 500	
Round Trip Delay in a 4-wire Loop	Tr	msec	0	0 ... 1000	
Absolute Delay in echofree Connections	Ta	msec	0	0 ... 500	
Number of Quantization Distortion Units	qdu	-	1	1 ... 14	
Equipment Impairment Factor	I_E	-	0	0 ... 40	
Circuit Noise referred to 0 dBr-point	Nc	dBm0p	-70	-80 ... -40	
Noise Floor at the Receive Side	Nfor	dBmp	-64	-	note 3)
Room Noise at the Send Side	Ps	dB(A)	35	35 ... 85	
Room Noise at the Receive Side	Pr	dB(A)	35	35 ... 85	
Expectation Factor	A	-	0	0 ... 20	

NOTE 1: Total Values between microphone or receiver and 0 dBr-point.
NOTE 2: Fixed Relation: LSTR = STMR + D.
NOTE 3: This value may not be modified.

9 Rules for the insertion of echo cancellers

9.1 Introduction

The increasing digitalization of public and private networks is resulting not only in connections with higher values of propagation delay (e.g. processing delay), but also with nearly no loss. This can cause impairments due to echo effects if no arrangements are made in conjunction with careful transmission planning to suppress these effects. The effects of echo and its origins are described in more detail in annex A.

Usually the use of additional loss to reduce the volume of the echo signal is limited. Therefore the insertion of echo cancellers will be necessary. The basic operation principle of an echo canceller is described in annex A. Previously, in mainly analogue networks, echo cancellers or echo suppressors were used only in 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 in national connections and within private networks.

As far as echo cancellers have to be inserted within a private network their location is dependent on different factors. They can either be used along with the digital interface of a specific switching equipment or transmission element, or they may be provided in a pool for a flexible insertion depending on the type of connection. Specific types of terminals are - due to their delay which is not negligible - (e.g. cordless telephones according to DECT Standard [2]) equipped with integrated echo control devices. The same may apply for systems using low bit-rate coding.

The following subclauses are providing guidance to the planner of private networks about the aspects to be taken into account for the correct use of echo cancellers. However, these rules are only related to those echo cancellers which should be inserted additionally according to the results of planning calculation. Integrated echo cancellers in specific equipment are in most cases not subject to a planning decision, only their technical characteristics should be considered for the decision on additional devices.

9.2 Characteristics of the echo cancellers

The different parameters characterising the performance of an echo canceller are described in annex C subclause C 1.1.9. They will be the basis when investigating if the device is suitable for the designated purpose. 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 the ITU-T Recommendation G.168 [19] or G.165 [18].

9.3 Limits for the application of echo cancellers

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

Since the amount of impairments due to echo is depending on two different factors - the amount of delay and the volume of the perceived echo - which will vary in a wide range and which are independent from each other, a general rule, e.g. together with a limit value for the mean one-way delay cannot be given. For standard national calls via main public networks with fully digital routing, single configurations as far end termination and delays less than 5 msec within the private network, the use of echo cancellers is not required. Also if this limit of 5 msec is exceeded, this may 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 Idte or Id (if Idle and Idd are close to 0). These factors are part of the algorithm of the E-Model and should be available separately.

If in a given configuration additional impairments beside echo must be expected, then the Total Impairment Itot should be analyzed primarely. For values up to Itot = 15 a sufficient good quality can be expected, i.e. the use of echo cancellers is not necessary. In case of higher values the specific impairment factor Id and his relation to the total value should be considered. If Id is the main part of Itot and in a range of 20 ... 25, then echo cancellers should be used. As a general rule the insertion of echo cancellers is recommended if values of Itot > 15 are calculated and echo is the only impairment, because this will result in any case in a quality improvement. Mainly if other impairments e.g. due to low bit-rate coding are additionally contributing to Itot, the part of Id can be reduced to 0 by use of echo cancellers.

9.4 Determination of the location

If, according to the planning results, the decision is made in favour of an echo canceller, a further task of planning is to investigate the appropriate location for the echo canceller also under 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. a necessary suppression of echo effects by the provision of a sufficient echo loss or echo cancelling is also subject to planning.

For delays less than 5 msec within the private network, the requirements as given in subclause 6.8 will provide the necessary echo control for the far end termination. For higher values of delay however, e.g. when low bit-rate equipment are used within the private network as shown in figure 20, the use of an additional echo canceller A - beside the echo canceller B for the internal talker - is requested for the benefit of the far end talker.

In a configuration as shown in figure 20, the additional delay which requires echo control for this connection is only inserted by the private network. Therefore the private network is responsible to provide the necessary echo cancellers. Furthermore, the information about the routings and type of terminal equipment, mainly responsible for the amount of delay, are only available within the private network. Therefore the control about the necessary and correct insertion of echo cancellers when additional delay will arise in specific configurations, is only possible in the private network itself.

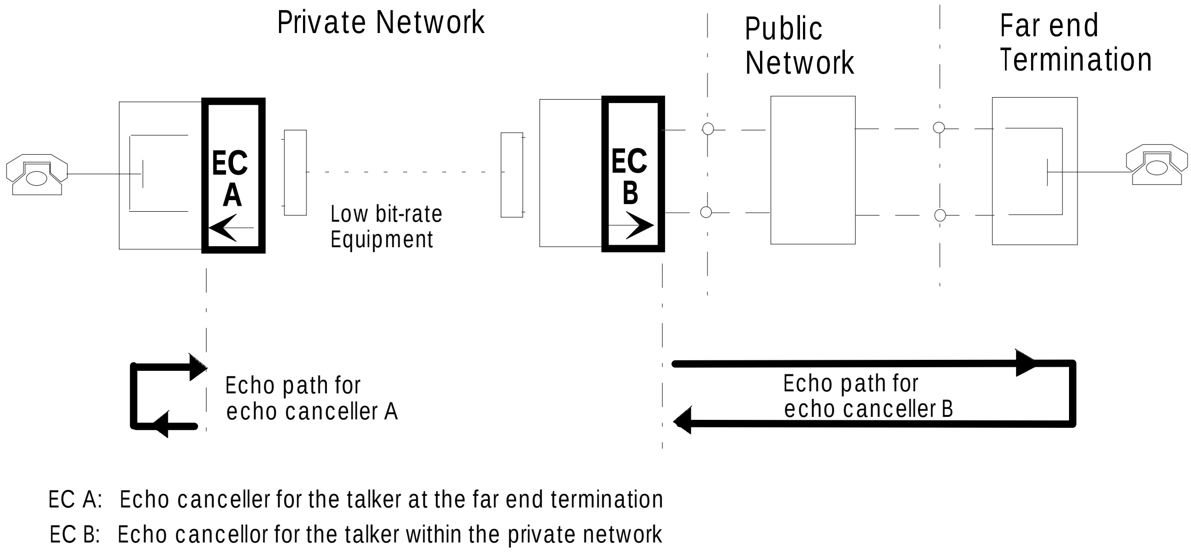

Figure 20: Application of a pair of echo cancellers within the private network

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. This value should be 6 - 8 msec 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 is indicating the direction of the echo path) is only formed by the hybrid within the PBX, i.e. only a short delay, while for EC B the echo path is including the entire routing via the public network with higher values for the delay. It is important to note, that for the selection 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 und should be considered only for very high values as part of the total one-way delay Ta, which may cause impairments resulting from too long delay.

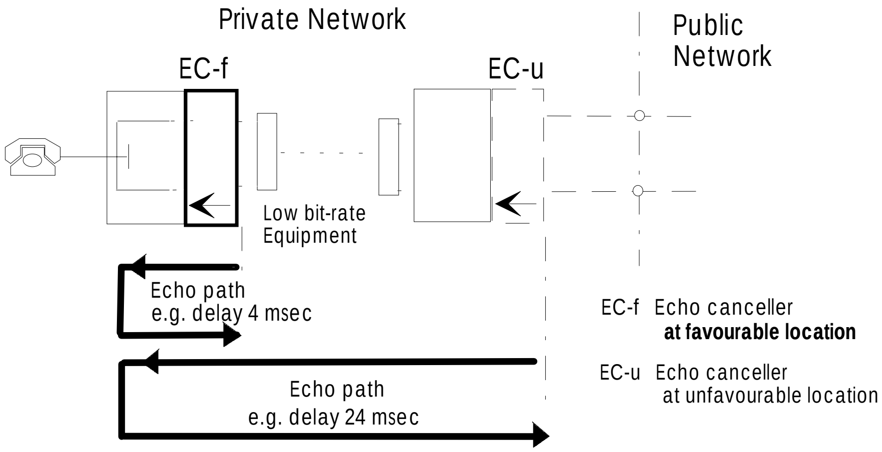

Figure 21: Most favourable location of an echo canceller

The location for EC A as shown in figure 20 can already be considered as the most favourable one. As a general rule an echo canceller should be inserted as close as possible to the echo source, in this configuration the hybrid. However other locations may also be taken into account as shown in figure 21 with EC inserted in the PBX with access to the public network.

Those solutions may have the advantage that the number of necessary devices is reduced due to this centralized location. Also the automatic insertion of echo cancellers only in those connections which are routed 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-u 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-u does not meet the requirement for a linear echo path as described in annex C subclause C 1.1.9 since the (nonlinear) low bit-rate equipment is part of the echo path.

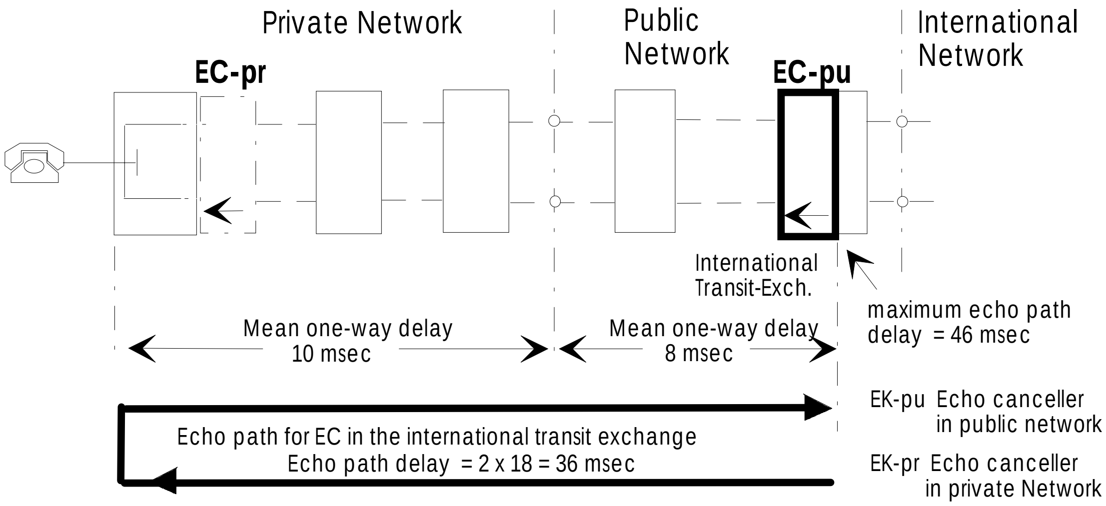

Figure 22: Use of echo cancellers in an international connection

For the planning of international connections, in most cases an automatic insertion of echo cancellers within the public network can be assumed. In these configurations it should be investigated if 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 is formed by the routing via the public network and the private network with its terminating hybrid. The mean one-way delay of the public network is assumed to be 8 msec and the private network with a value of 10 msec. The echo canceller EC-pu, inserted in the international transit exchange of the public network, is assumed to be able to compensate an echo path delay of 46 msec. The actual echo path delay is calculated with 2 x (10 + 8) = 36 msec, 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 msec in figure 22 for the mean one-way delay within the private network is an extreme case. For higher values usually the use of echo canceller EC-pr should be taken into account already for national long distance connections. This example shows, that it is advisable to ask for the relevant information about the properties of the public network, the expected average delay for international routing and the characteristics of the used echo cancellers.

9.5 Handling of echo cancellers in the E-Model

The correct use of echo cancellers, mainly those according to ITU-T Recommendations G.165 [19] or G.168 [18], is comparable with an enhancement of the echo loss and consequently also the TELR to values in the range of 50 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.

10 Realization of planning

10.1 General

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

Planning can be necessary in case of a fully new established private network, but also in case of existing networks which should be modified or amended in their major parts. In the latter situation also existing parts can be subject to planning to investigate the expected quality between new and existing terminals and if access and routing to the public network has been changed for the existing terminals. In general, transmission planning can be executed in different 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;
- end-to-end calculation of the expected quality with the E-Model;
- analysis of the results.

These steps should be considered only as a recommendation. Depending on the actual planning project they can be modified or amended. Also the proposed sequence can be seen only as a guidance. The following subclauses are providing a more detailed description of the different steps.

10.2 Determination of the specific requirements

Depending on the business of the users 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 amount 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 facts 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 leased lines.

The routing and/or 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 in case of specific features (e.g. call transfer) or 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.

For certain private networks the type of connections via public networks should be determined. According to the principles of the present document, the planning is based on an end-to-end consideration. Consequently, the amount of impairments caused by the public network is important for the planner. For most of the public networks the impairments (e.g. delay) are low in case of local calls and increasing 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 e.g. only local connections (e.g. only local operating company) a higher amount of impairments can then be permitted within the private network for the benefit of more economical solutions. In the field of competition several offers of public networks 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 conjunction is a percentage for the type of connection considered, in a range of 90 ... 95% and not just more than 50%.

Furthermore, it should be determined whether, according to the business, a predominance of communication partners with respect to a specific type e.g. in the residential or business domain only can be defined. During planning, this will be helpful for the selection of the far end termination as described in annex C subclause C 1.3 (for the European scenario). If a clear definition is not possible the type "Single Telephone Set" should be used.

As far as public network operators are providing a specific access (e.g. direct access in a higher hierarchy), or a special - low impairment - routing for a specific type of calls, this should be included in the basic determination. In most cases public network operators are also providing leased lines or the feature of a Virtual Private Network (VPN) for the connection between the different 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 different offers of leased lines or VPN.

10.3 Definition of the reference configurations

As already stated in subclause 10.1, for transmission planning the most critical connections should be identified. Based on the structure of the private network in conjunction with the possible routings and further information about type and point of access to public network(s), predominance of connection and/or type of far end terminations (where applicable) this critical connection (in this context called "Reference Configuration") should be determined now.

The purpose of such a reference configuration is to obtain an overview of all relevant parts of the critical connection considered. It is recommended to make a drawing of this configuration including all relevant terminal, switching and transmission elements which may contribute with impairments. Such a drawing 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, mainly if the structure of the private network and the routing is complex and a clear issue whether a path is critical or not cannot be obtained without calculation.

The determination of the reference configurations mainly in large complex private networks is very important to obtain correct planning results and it needs a lot of experience and planning practice. When investigating the network, attention should be drawn mainly to elements introducing additional delay and/or equipment impairments 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 considered carefully. Although critical connections as basis for the reference configurations will usually be found for connections via public networks to the far end termination, routings only within the private network may sometimes be more critical.

10.4 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);
- leased lines.

As a minimum the following parameters should be made available for the different 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);
- equipment impairment factor (in equipment using low bit-rate coding).

If appropriate, the location and characteristics of echo cancellers should be determined if they are already existing in the private or public network and for which routing they are operative. In general, these parameter values should be provided usually by the manufacturers of the considered element. For public networks, information about the parameter values based on the type of connection and the access can only be obtained by negotiations between public and private network operators. This is also valid for the characteristics of VPN or leased lines. For further information about the required parameters see also annex C subclauses C 1.1 and C 1.2 (for the European scenario).

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

For this step in 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. This step needs specific care to avoid wrong inputs to the E-Model as already described in subclauses 8.2, 8.3 and 8.5. In most cases it is necessary also to perform precalculations for different parameters as described in clause 7, to obtain the correct form and value for an input parameter to the E-Model. It is strongly recommended to transfer the reference configuration into one of the working configurations for the E-Model as proposed in subclause 8.2.

The selection of the appropriate working configuration is depending on the reference configuration to be investigated. In case of fully digital connections there is no doubt, that the working configuration for fully digital connections as shown in figure 19 should be used. More difficulties may arise for reference configurations with one or more 4-wire/2-wire conversions (hybrids) within the connection. If the private network side is terminated with a digital telephone set and a hybrid either within the private network or in the public network is forming an echo path, it is recommended to use the working configuration for 2-wire/4-wire connections of figure 18.

For the selection of a working configuration it is very important to make a correct assignment of the send side and receive side. Basically the planning principle and the determination of the expected quality is related to the user of the private network primarely. The principle and algorithm of the E-Model is relating 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 the E-Model.

For the investigation of possible echo effects for the far end termination however, the assignment of send and receive side should be changed. Assuming one of the three far end terminations which all are terminated in a hybrid connecting an analogue telephone set (where now the talker may be disturbed by echo), the working configuration of figure 17 should be applied.

NOTE: When using computer programs, the feature to perform the calculation for boths sides without changing the input parameters, could be provided by this program. Mainly in those applications however, it is very important, that the planner is fully familiar with all the features and restrictions of the used program, 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 transferring the reference configuration into this working configuration, it is in some applications necessary to consider parts of the private network (including e.g. a PBX and/or a transmission section) as a single telephone set. Therefore the relevant parameters should be transferred to be available as input parameters for the telephone A in figure 16. A comparable situation is described in subclause 8.5 in conjunction with a cordless telephone.

10.6 Analysis of the results

When the results of planning calculations are available as total impairment values Itot, a first analysis should be performed with respect to values of more than Itot = 45 ... 50. This should be considered as an absolute upper limit which should never be exceeded even in exceptional cases. If a reference configuration with its corresponding parameter values is resulting in higher values, then this configuration shall not be realized and cannot be used in practice and solutions should be found or other equipment have to be selected to reduce the impairments to lower values.

It is very important for the planner to fully understand the planning principle as recommended in the present document. First of all, the planning is based on an end-to-end consideration in contrary to previous planning practice for private networks where specific limits for the different transmission parameters were used (because of regulation) related to the private network section up to the interface to a public network.

Furthermore, the result is not providing "numbers" for the different 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 already stated in subclause 3.3, quality is a subjective judgement where no assignments can be made to a fixed number for Itot, MOS, GOB or POW, or to borders between different ranges of the whole quality scale. It should be considered as a linear scale for the perceived quality varying from high quality via medium values to a low quality as illustrated in figure 23.

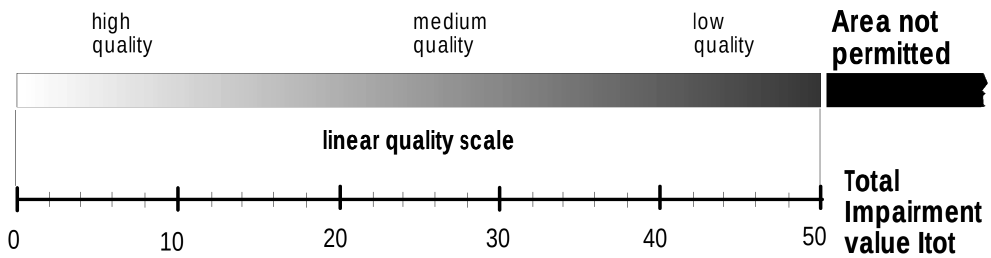

Figure 23: Judgement of a connection on a linear quality scale

Although in figure 23 a rough distinction into high, medium and low quality is made, this may not be understood in a way, that there is a specific number for Itot considered as a border between high and medium or between medium and low. Only the border between low quality and the not permitted area is related to the fixed value of Itot = 50. The linear scale of Itot in the range from 0 to 50 is related to the quality scale so that a result given as a value for Itot can be assigned to the linear quality scale. Further assistance in interpreting the results of planning is provided by the verbal quality description as given in tables 1a and 1b.

For practical planning it is recommended that standard connections within or to and from the private network should result in upper values for Itot in the range of 15 ... 25. For exceptional configurations values in the range from 30 ... 45 may not be exceeded. It is worthy of note, that for an end-to-end consideration as performed here, the overall quality is not only influenced by the private network being subject to planning, but also by public networks. Therefore, as a consequence it is not possible in most applications to perform the planning for a private network with the goal of "high quality" only.

Annex A: 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 [6].

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.

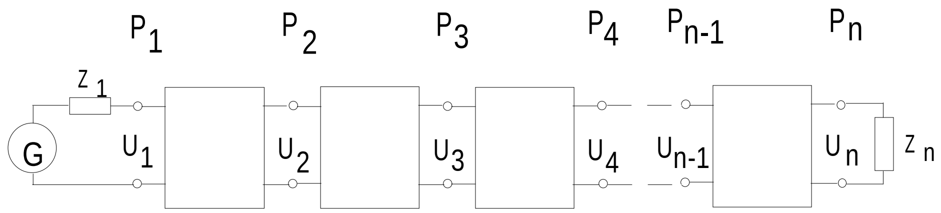

Figure A.1: Loss contribution in a telephone connection

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.

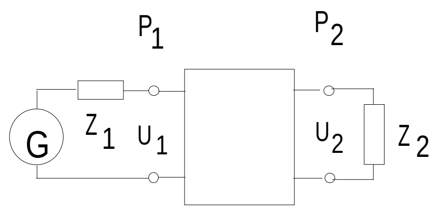

Figure A.2: Definition of powers P_1 and P_2

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":

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

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:

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

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.

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

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:

$$a_n = 10 \cdot \log \frac{P_1}{P_2} + 10 \cdot \log \frac{P_2}{P_3} + \ldots\ldots 10 \cdot \log \frac{P_{n-1}}{P_n} \quad \text{dB}$$

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:

$$a_n = a_1 + a_2 + \ldots\ldots a_{n-1} \quad \text{dB}$$

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 subclause 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 [23].

In principle, the parameter loudness rating is an issue about the loudness loss between the talkers mouth (Mouth Reference Point MRP) and the listeners 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
Sending Loudness Rating	SLR	Loudness loss between MRP and an electrical interface
Receiving 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 talkers mouth (MRP) and his ear (ERP) via the electrical sidetone path.
Listeners 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 talkers mouth and his ear via the echo path
Listeners Echo Loudness Rating	LELR	Difference in loudness loss between the talkers direct voice and the delayed echo both reaching the listeners 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 [23]. STMR and LSTR are not only depending on the telephones 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 CLRs between the set and the interface

$$SLR_{pn} = SLR_{set} + \sum_{i=1}^{m} CLR_i \quad \text{and} \quad RLR_{pn} = RLR_{set} + \sum_{i=1}^{m} CLR_i$$

For the CLR of a specific network element, the loss definition as described in section 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 faktor 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.

$$TELR = SLR_{set} + EL + RLR_{set}$$

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*log (Y), for voltage, current and pressure ratios X the level is defined as L = 20*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:

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

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 mWwill 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.

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

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 dBr - 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 1020 Hz (1004 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 [21] 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 [8] and G.101 [9].

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, independant of the relative level at the 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 Li at the input and Lo 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

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

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

$$\text{dBm} = \text{dBm0} + \text{dBr}$$

$$\text{dBr} = \text{dBm} - \text{dBm0}$$

$$\text{dBm0} = \text{dBm} - \text{dBr}$$

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) echos 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 [12], 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 msec. 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 [18] and G.168 [19] 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 [12].

a) Range 0 - 25 msec;

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 - 150 msec;

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

c) range 150 - 400 msec;

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 [18] and G.168 [19] should be used and careful network planning is necessary.

d) range above 400 msec;

Values of transmission time above 400 msec 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 msec between the two earth stations, the remaining maximum transmission time for each national section is 70 msec.

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 msec to 25 msec. 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 msec to 55 msec 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 Ta 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 from 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.

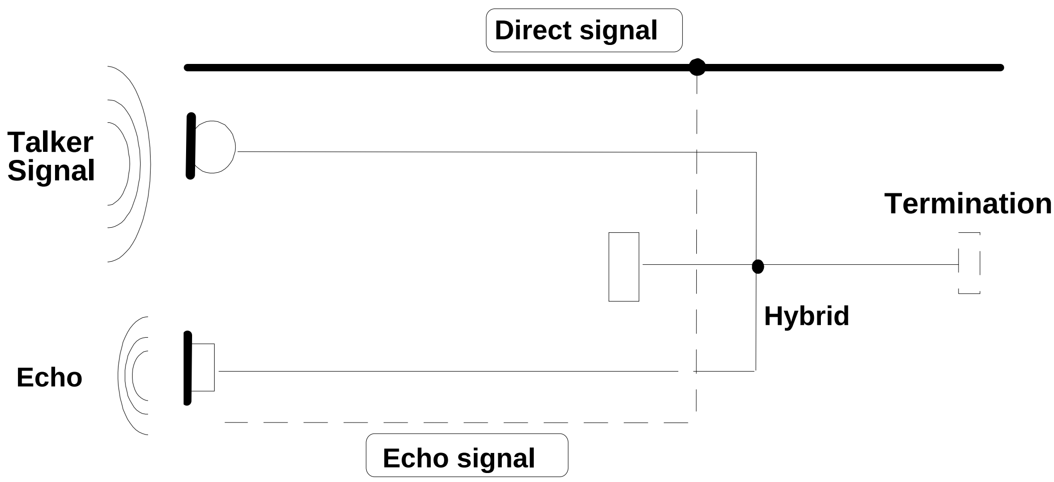

Figure A.3: Effect of Talker Echo

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 subclause 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 phenomen 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.

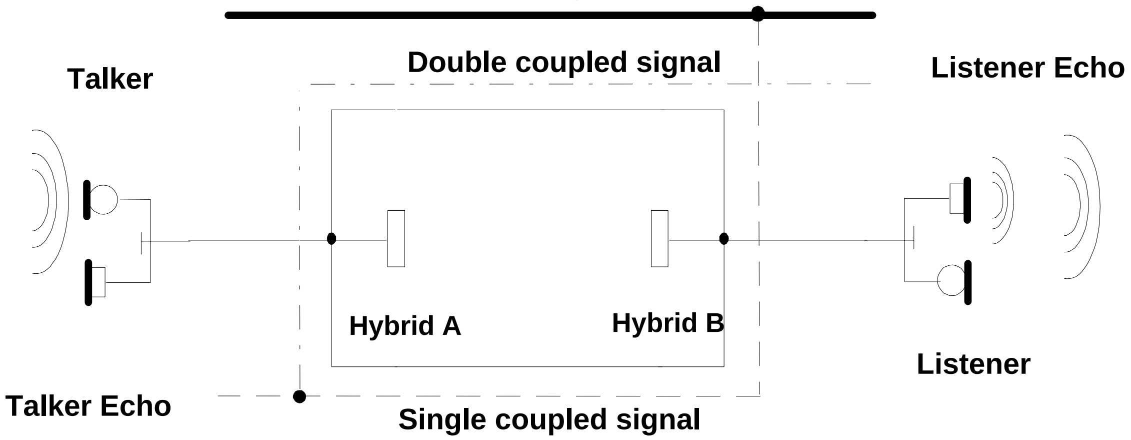

Figure A.4: Effect of Listener Echo

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 subclause 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.

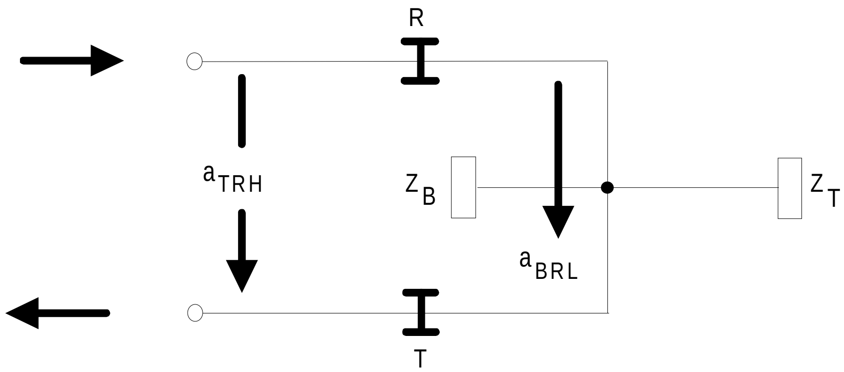

Figure A.5: Balance Return Loss and Transhybrid Loss

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

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

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.

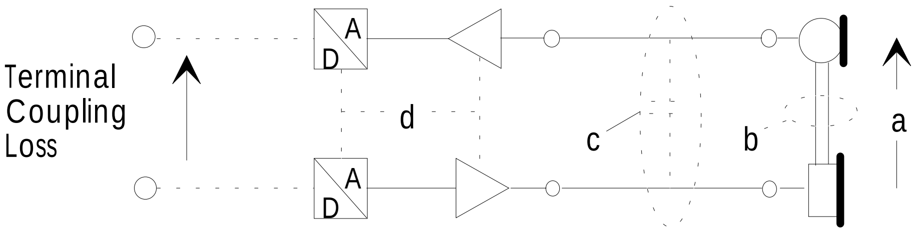

Figure A.6: Coupling in a digital telephone set

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 trans-hybrid loss is weighted with a specific weighting function over the frequency range 300 Hz to 3400 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 [14] § 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:

$$EL = 3{,}85 - 10\log\left[\int_{300}^{3400} \frac{A(f)}{f}\,df\right] dB$$

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, §4 [14].

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 transmision 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 [17] for ES and G.165 [18] and G.168 [19] for EC.

Presently, Echo Control Devices are used to suppress echo in connections with a mean one-way transmission time of more than 25 msec, 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 centers. 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.

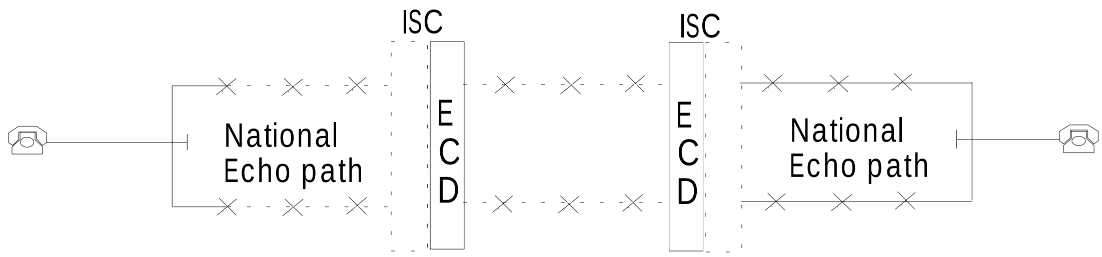

Figure A7: Application of Echo Control Devices

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.

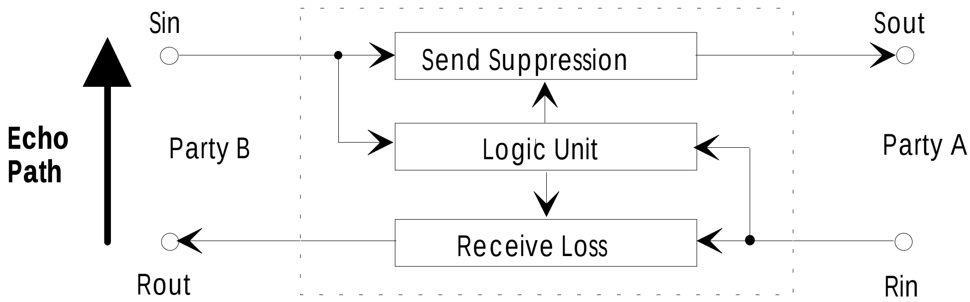

Figure A.8: Echo Suppressor

This mode of course is encountered by party A subjectively as an interuption 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 msec.

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 interupted 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.

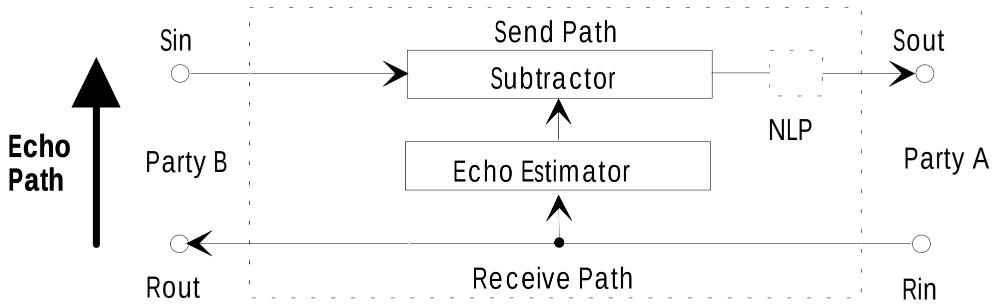

Figure A.9: Echo Canceller

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 [17], G.165 [18] and G.168 [19], 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 centers 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 [2] and ETS 300 175-8 [3] inserting an additional mean one-way delay of about 14 msec.

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 msec, 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 independant 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 to be higher than 44 dB.

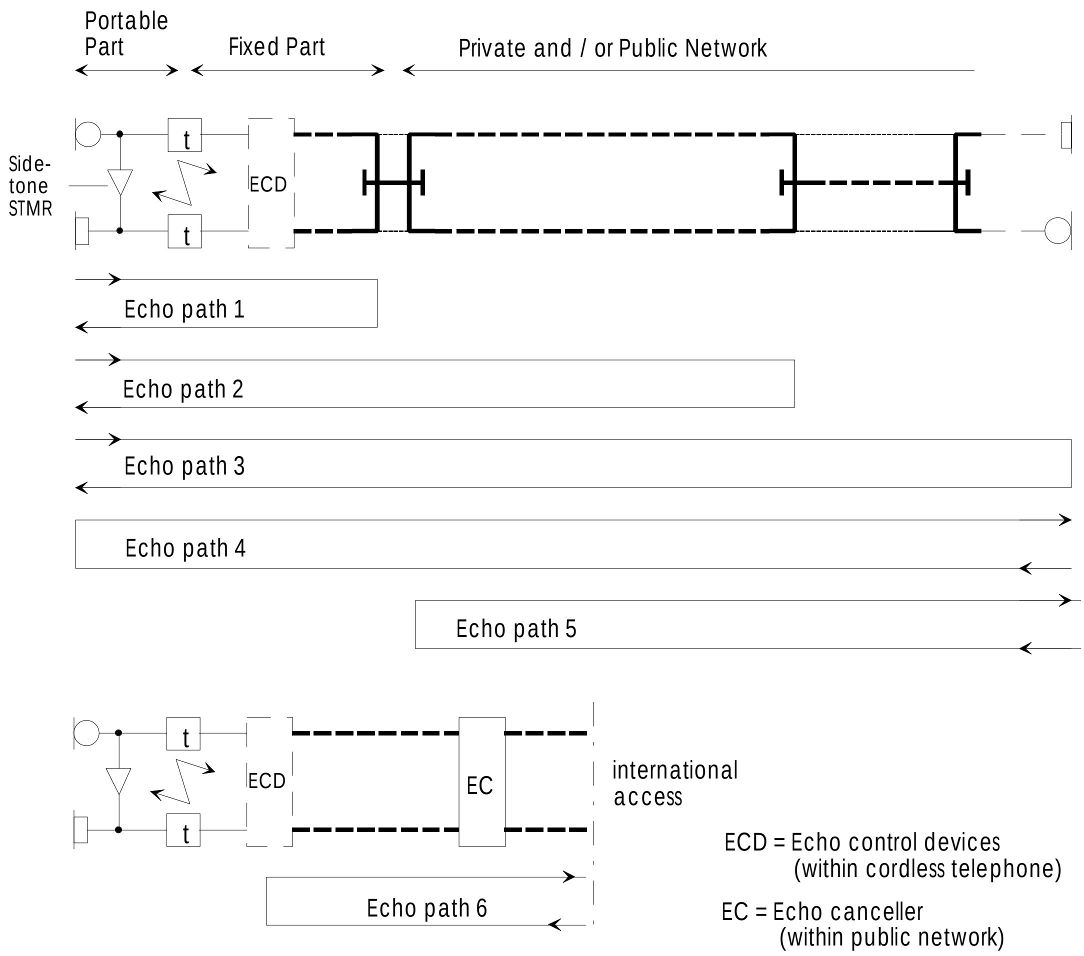

Figure A.10: Possible echo paths for a cordless telephone

- 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 independant 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 6.8).

- 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 [18], 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 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 dB and RLR = 3 dB referred to an internal reference point (0 dBr-point). To meet the national requirements at the 2-wire interface of the fixed part - in this example SLR = +4 dB, RLR = -7 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 dB, the resulting transhybrid loss of a_{TRH} = - 1,5 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 [18] or G.168 [19]. The corresponding Standard requires only an enhancement of the transhybrid loss - called echo loss enhancement - of > 6,5 dB, which is resulting in an echo loss of -1,5 dB + 6,5 dB = 5 dB in the example as shown in figure A.11. Furthermore, the permissible echo path delay for this EC is required with only > 4 msec, since only the hybrid circuit is forming the echo path.

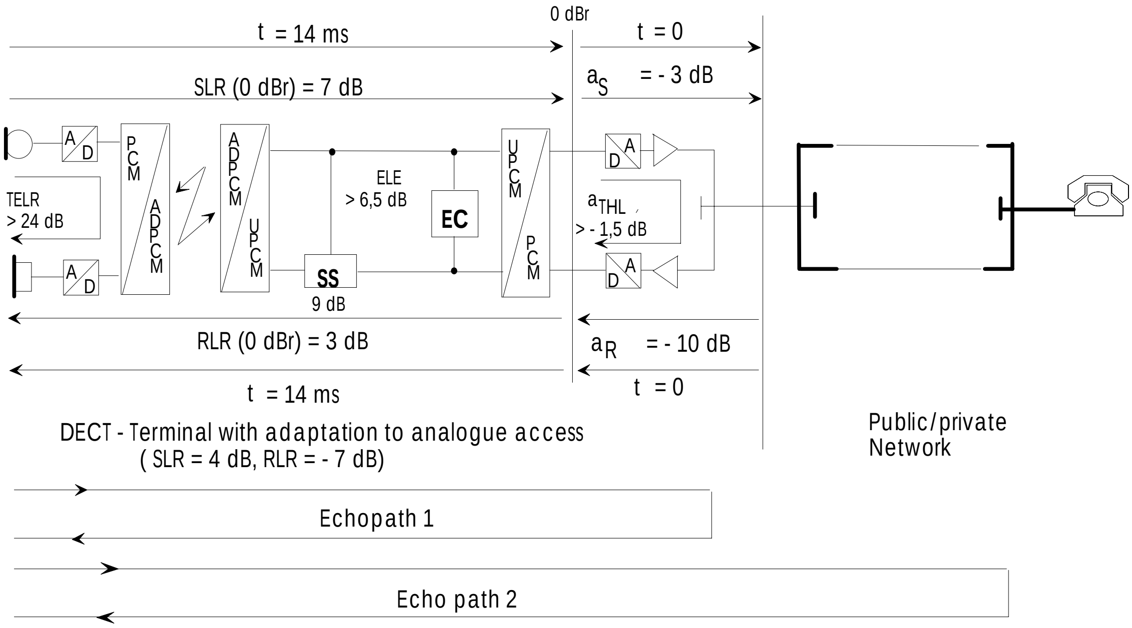

Figure A.11: Echo control devices in a DECT cordless telephone

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 talkers 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 msec via the public network is assumed. The echo canceller EC is unable to compensate this echo path due to the low value of 4 msec 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 6.8 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 msec may cause an inproper operation of this echo canceller. Therefore, if the portable part is only providing the optional TCLw of 34 dB an "Artifial Echo Loss AE" should be inserted at the network-side of the fixed part. This configuration is shown in figure A.12.

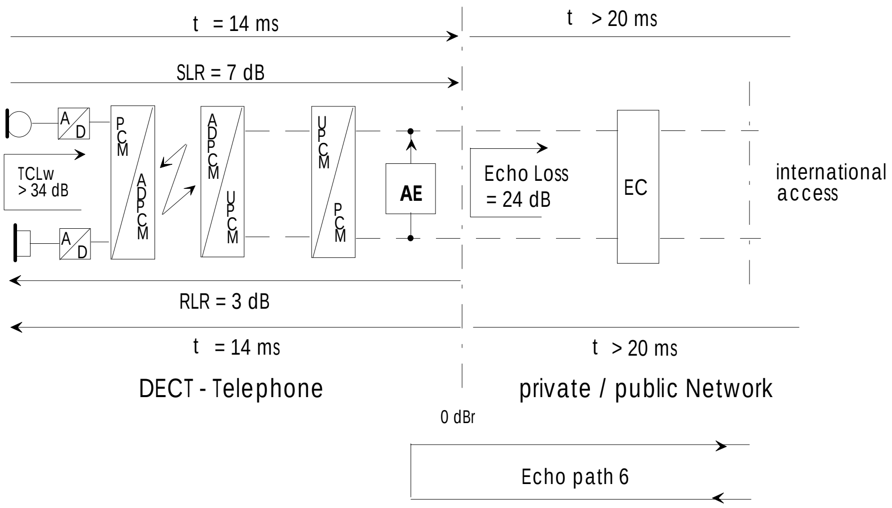

Figure A.12: TCLw and artificial echo loss

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 msec 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 phenomen 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.

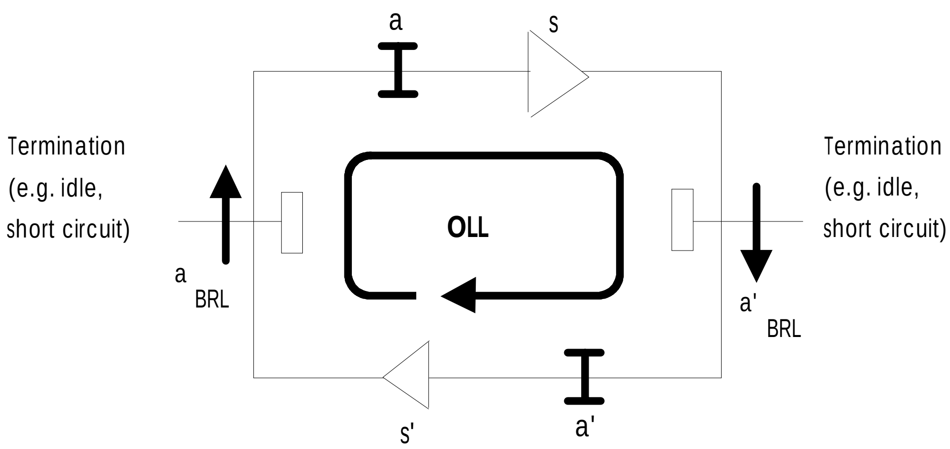

Figure A.13: Stability of a 4-wire loop

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 [14]. 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 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: 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 is based on the Equipment Impairment Factor Method, which is described also in the ITU-T Recommendation G.113 [11] 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 Nc 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 Ie and expectation factor A.

There are three different parameters in conjunction with transmission time. The absolute delay Ta 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 Tr 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 [1)]:

"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

$$R = Ro - Is - Id - Ie + A \qquad (1)$$

1) Bibliography: Supplement III (1993) to the Series P Recommendation:

- models for predicting transmission quality from objective measurements.

In principle, **Ro** is representing the basic signal to noise ratio including noise sources such as circuit noise and room noise. The factor **Is** is combining all impairments which occur more or less simultaneously with the voice signal. Factor **Id** represents those impairments, which are caused due to delay and the equipment impairment factor **Ie** 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 Ro, Is and Id 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.

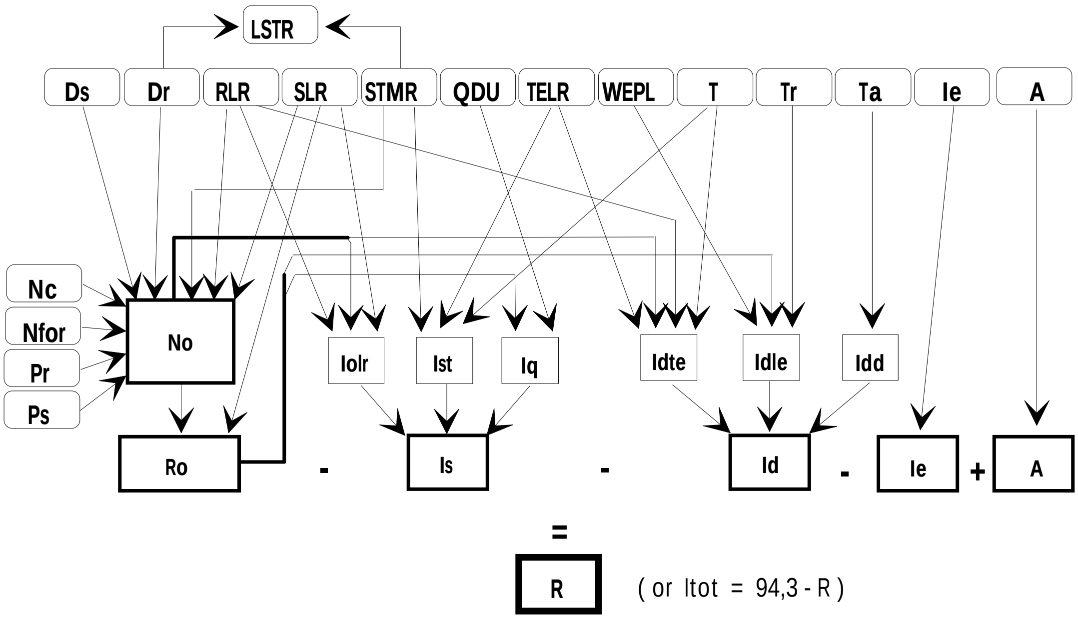

Figure B.1: Overview of the influence of the different input parameters to the E-Model

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 Ro

$$Ro = 15 - 1,5(SLR + No) \quad (2)$$

The term **No** is the power addition of different noise sources, whith the exception of quantization noise

$$No = 10\lg\left[10^{Nc/10} + 10^{Nos/10} + 10^{Nor/10} + 10^{Nfo/10}\right] \quad \text{dBm0p} \quad (3)$$

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

Nos is the equivalent circuit noise at the 0 dBr-point, caused by the room noise Ps at the send side

$$Nos = Ps - SLR - Ds - 100 + 0,008 \cdot (Ps - OLR - Ds - 14)^2 \quad \text{dBm0p} \quad (4)$$

where OLR = SLR + RLR.

Nor is the equivalent circuit noise at the 0 dBr-point, caused by the room noise Pr at the receive side

$$Nor = RLR - 121 + Pre + 0,008(Pre - 35)^2 \quad \text{dBm0p} \quad (5)$$

The term Pre is the "effective room noise" caused by enhancement of Pr by the listeners sidetone path

$$\mathrm{Pre} = \mathrm{Pr} + 10\lg\left[1 + 10^{(10-\mathrm{LSTR})/10}\right] \quad \text{dBm0p} \tag{6}$$

LSTR is the Listeners Sidetone Rating at the receive side.

Nfo is representing the "noise floor" at the receive side, referred to the 0 dBr-point

$$\mathrm{Nfo} = \mathrm{Nfor} + \mathrm{RLR} \quad \text{dBm0p} \tag{7}$$

with Nfor usually set to - 64 dBmp.

B.3.2 The simultaneous impairment factor Is

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

$$\mathrm{Is} = \mathrm{Iolr} + \mathrm{Ist} + \mathrm{Iq} \tag{8}$$

Iolr represents the decrease in quality caused by too low values of OLR.

$$\mathrm{Iolr} = 20 \cdot \left[\left\{1 + (X/8)^8\right\}^{1/8} - X/8\right] \tag{9}$$

where

$$X = \mathrm{OLR} + 0{,}2 \cdot (64 + \mathrm{No} - \mathrm{RLR}) \tag{10}$$

and No is given by equation (3).

The factor **Ist** represents the impairment caused by non-optimum sidetone at the receive side

$$\mathrm{Ist} = 10 \cdot \left[1 + \left\{(\mathrm{STMRo} - 12)/5\right\}^6\right]^{1/6} - 46 \cdot \left[1 + \left\{\mathrm{STMRo}/23\right\}^{10}\right]^{1/10} + 36 \tag{11}$$

where

$$\mathrm{STMRo} = -10\lg \cdot \left[10^{-\mathrm{STMR}/10} + e^{-T/4}\, 10^{-\mathrm{TELR}/10}\right] \tag{12}$$

The impairment factor **Iq** is representing impairment caused by quantizing distortion.

$$\mathrm{Iq} = 15\lg \cdot \left[1 + 10^{Y}\right] \tag{13}$$

where

$$Y = (\mathrm{Ro} - 100)/15 + (46 - G)/10 \tag{14}$$

and for G

$$G = 1{,}07 + 0{,}258Q + 0{,}0602Q^2 \tag{15}$$

$$Q = 37 - 15\lg(\mathrm{qdu}) \tag{16}$$

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

NOTE: If an impairment factor Ie 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 Id

Also Id, the impairment factor representing all impairments due to delay of voice signals is subdivided into three further factors Idte, Idle and Idd.

$$Id = Idte + Idle + Idd$$

The factor **Idte** gives an estimate for the impairments due to Talker Echo.

$$\mathrm{Idte} = \left[(\mathrm{Roe} - \mathrm{Re})/2 + \sqrt{(\mathrm{Roe} - \mathrm{Re})^2/4 + 100} - 1 \right] \cdot \left(1 - e^{-T}\right) \tag{17}$$

where

$$\mathrm{Roe} = -1{,}5 \cdot (\mathrm{No} - \mathrm{RLR}) \tag{18}$$

$$\mathrm{Re} = 80 + 2{,}5 \cdot (\mathrm{TERV} - 14) \tag{19}$$

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

For values of T < 1 msec, the talker echo should be considered as Sidetone only, i.e. in this case Idte = 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 Idte are adjusted as follows:

For STMR $\leq$ 9 dB:

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

$$\mathrm{TERVs} = \mathrm{TERV} + \mathrm{Ist}/2 \tag{21}$$

where Ist is representing impairments caused by too low sidetone (equation 11).

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

For STMR > 15 dB:

Idte is replaced by Idtes, where

$$\mathrm{Idtes} = \sqrt{\mathrm{Idte}^2 + \mathrm{Ist}^2} \tag{22}$$

The factor **Idle** is representing impairments due to Listener Echo. The equations are:

$$\mathrm{Idle} = (\mathrm{Ro} - \mathrm{Rle})/2 + \sqrt{(\mathrm{Ro} - \mathrm{Rle})^2/4 + 169} \tag{23}$$

where

$$\mathrm{Rle} = 10{,}5 \cdot (\mathrm{WEPL} + 7) \cdot (\mathrm{Tr} + 1)^{-0{,}25} \tag{24}$$

and Ro is given by equation (2).

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

For Ta $\leq$ 100 msec: Idd = 0

For Ta > 100 msec:

$$\mathrm{Idd} = 25 \cdot \left\{ \left(1 + X^6\right)^{1/6} - 3 \cdot \left(1 + [X/3]^6\right)^{1/6} + 2 \right\} \tag{25}$$

with

$$X = \frac{\lg(Ta/100)}{\lg 2} \qquad (26)$$

B.3.4 The equipment impairment factor Ie

The values of the Equipment Impairment Factors Ie 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 [11], Table 8, G.175 [20] 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

Error Function

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

The equations are:

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

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

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

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

For $R \leq 0$:

$$MOS = 1$$

For $0 < R < 100$:

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

For $R \geq 100$:

$$MOS = 4{,}5.$$

Annex C: Transmission parameters for specific elements

This annex is providing additional guidance about the different network elements within the private network, for public networks and for the far end termination. Since network configurations and parameter values of specific network elements may differ between the European and the North American scenario, clause C.1 of this annex is based on the European area and clause C.2 for the North American area.

C.1 Transmission parameters for the european scenario

According to the planning principles of the present document, the investigation of reference configurations is based on an end-to-end consideration in all cases. This requires data with respect to speech quality not only of the elements within the private network but also of other networks and the far end termination. As a support to the planner, guidance on some specific elements within the private networks and other parts of a connection is given in the following subclauses. If possible - mainly in case of commonly used elements with standardized transmission data - the parameter values are given and can directly be used for planning. For all other transmission elements or connection elements instructions will be given which data should be collected for the purpose of planning and how to analyse and use those data.

C.1.1 Elements in private networks

C.1.1.1 Wired telephone sets

In general it is assumed, that all telephone sets used in private networks are designed according to European or national Standards. For planning purposes the 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 should be distinguished into analogue and digital telephone sets with respect to their type of interface.

C.1.1.1.1 Analogue telephone sets

The transmission data of analogue telephone sets are mainly depending on national loss planning. Therefore standardized values cannot be provided here and should be available from the manufacturer or network operator. The following parameters are necessary for transmission planning and should be determined:

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

The Loudness Rating values SLR, RLR and STMR should be defined according to ITU-T Recommendation P.79 [23]. To avoid wrong results in conjunction with the E-Model, preceeding definitions like 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 should follow a modern design providing a "capacitive complex impedance" for a more optimized impedance matching between telephone set and connected equipment. A possible mismatch at this point may have relevant influence not only to the STMR of the telephone set, but also to the weighted Terminal Balance Return Loss within a connected hybrid resulting in a low value for TELR (see also subclause C.1.1.3).

The D-Factor of the handset should only be considered if a handset design is used deviating 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.1.1.1.2 Digital telephone sets

Beside the protocol requirements for digital telephone sets, the transmission data can usually be assumed to be identical with ETSI TBR 8 [1]. However, deviations from these values are possible. The following parameter values can be used directly or have to be determined if the telephone in use does not meet TBR 8 in all values:

Send Loudness Rating	SLR	+ 7 dB
Receive Loudness Rating	RLR	+ 3 dB
Sidetone Masking Rating	STMR	15 dB
Mean one-way Delay	τ	1,5 msec
Terminal Coupling Loss (weighted)	TCLw	40 dB
D-Factor of the handset	D	3

In any case, Loudness Rating values should be in accordance with the ITU-T Recommendation P.79 [23]. For the weighted Terminal Coupling Loss TCLw some telephones may provide values > 46 dB. According to the manufacturers declaration the higher values may be used for planning purposes.

C.1.1.2 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 will contribute with additional delay and distortions. It is assumed, that these telephones will comply with the according European or national Standards. In table C.1 an extract is given from the DECT and the GSM Standards with all parameters relevant for planning according to the present document.

These values are referred to the entire configuration consisting of mobile part and fixed part, where the fixed part is connected digitally to the adjacent connection or transmission element. For wireless telephones deviating from these Standards, information about the actual values according to the list of parameters in table C.1 should be available.

Due to the high values of mean one-way delay wireless telephones are usually already equipped with integrated parts for the control of echo, such as echo cancellers or echo suppressors. Since those equipment may also influence decisions about echo cancelling devices in other sections of the network, a careful investigation of their interworking is necessary. More information about those integrated echo suppressing devices mainly for the DECT -Standard are given in subclauses 7.2.2, C.1.1.9 and in annex A.

Table C.1: Planning values for wireless telephones

		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 [msec]	14	95	100	96
Echo Loss of Softsuppressor	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	Ie	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 Ie
Note 3: provisionally

C·1.1.3 Switching equipment

It is assumed, that switching equipment in a private network, usually called PBXs, are following European or national requirements with respect to their influence on transmission quality. Basically switching equipment can be divided according to their 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 terminals.

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

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

Switching equipment in a private network are usually providing a "throughconnection" between two interfaces, or with other words, they are "inserted" into the entire connection and therefore possibly contributing with impairments. Due to the variety of possible types of throughconnections also with respect to the physical layer of the interfaces and the types of internal switching, it is impossible to provide a general issue or guidance about the parameters and their degree of impairment to be considered in transmission planning. In any case information should be available about those parameters which may contribute with impairments. The most important parameters for a throughconnection to be considered are:

- loss or gain between the two interfaces;
- number of qdus;
- 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).

The value of loss between two interfaces is depending on the adjustment of the relative input- and output-level (loss adjustment) in case of analogue interfaces and depending whether digital loss or gain pads are inserted within the switching path.

When new coding laws are used for the internal switching, the value for the Equipment Impairment Factor Ie should be selected according to table 2. With the exception of inserted digital loss or gain pads, the number of qdu can be set to qdu = 0 for internal analogue switching (including PAM) and both interfaces analogue, or for internal digital switching (A-law or µ-law) with both interfaces digital. A value of qdu = 0,5 can be used for analogue or digital internal switching, where one of the two interfaces is digital and finally a value of qdu = 1 is valid for internal digital switching and both interfaces analogue. In case of digital pads the number of qdu should be increased by 0,7 for each pad in all configurations.

The mean one-way delay is negligible for switching equipment using internal analogue switching and analogue interfaces. For all other types the delay is depending upon the types of interfaces and the internal switching, however an average value of T = 1 msec can be used for all planning applications if PCM is used for the internal switching (for guidance see ETR 275 [7]).

Whenever a switching equipment is throughconnecting a 4-wire interface (analogue or digital) to a 2-wire interface, the echo loss of the terminating hybrid in the 2-wire interface should be considered. The input and balance impedancies of any 2-wire interface are not directly subject to planning, however these values should be available to check, whether a sufficient impedance matching at these interfaces is provided.

The echo loss 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 and responsible for a possible impairment due to delay. The echo loss of such a termination is including the loss adjustment (relative levels) of the hybrid in the 2-wire interface in send and receive direction and the Terminal Balance Return Loss TBRL. This TBRL is depending on the degree of matching between the Balance Impedance of the hybrid and the impedance of the connected terminal, transmission or connection element at the 2-wire side, and should be available as a weighted value TBRLw. (For more information about this weighting algorithm see annex A).

Balance networks and input impedancies of modern equipment are providing a capacitive complex characteristic to obtain an improved matching to the characteristic of unloaded cable sections. If the 4-wire / 2-wire conversion is made via an analogue 4-wire interface also the loss adjustment in these interface card should be included. The same applies for digital pads irrespective of their location.

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

Termination at the 2-wire side with	**TBRLw**	**Echo loss**
analogue telephone set with complex input impedance (negligible line lenght)	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 lenght)	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.

C.1.1.4 Leased lines

In private networks leased lines as provided by public network operators are mainly used to interconnect switching elements or to connect terminals to switching equipment. With respect to their interface presentation leased lines can be distinguished into the following basic categories:

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

For the purpose of transmission planning, digital leased lines are independent of their physical layer (64 kBit/s, Basic Rate Access or Primary Rate access), only the 64 kBit/s-channel should be considered. Leased lines with analogue interfaces at both ends may also include digital sections and a closed 4-wire loop. Furthermore, in some cases analogue lines may be available with a 2-wire interface at one end and a 4-wire interface at the other end.

Leased lines do not differ in their type of interfaces only but mainly in their lenght. Therefore common planning values cannot be issued here. The information about the transmission data should be made available by the provider in any case. The following list may be considered as a guidance for the planner when asking for parameter values.

- end-to-end Loss for lines with analogue interfaces, 2-wire and 4-wire. (In case of 2-wire/4-wire also for the two different transmission directions);
- relative input- and output levels for lines with analogue interfaces;
- number of qdu for all types with the exception of lines with fully digital routing and digital interfaces on both ends;
- equipment Impairment Factor for lines using DCME or other new coding laws;
- mean one-way delay for all types;

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

C.1.1.5 Privately owned cable links

Beside leased lines also privately owned cable links will be in use in some private networks. They are mainly connecting terminals and small PBXs to switching equipment. 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:

$$CLR = 0{,}015\sqrt{RC} \quad \text{in dB / km}$$

where: R = Cable loop resistance in Ohm per km;

C = Cable capacitance in nF per km.

C.1.1.6 Satellite links

When satellite links are used as part of the private network, all relevant parameters as for leased lines 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 is consisting of the main delay between the two earth stations and a possible additional delay between the earth stations and the interface of switching equipment within the private network to which the link is connected to on both ends. These values should be made available by the satellite operator. For the satellite links via quasi-stationary satellites in a 36000 km orbit, a value of T = 260 msec between the earth stations can be used for planning purposes. The equivalent values for satellites in lower orbits should be provided by the operator (for further guidance see ETR 275 [7]).

C.1.1.7 Low bit-rate coding

For private networks the use of equipment containing low bit-rate coding can result in more economical solutions. In many cases digital (leased) lines as a connection element are additionally 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 also 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 those systems. Beside others, the main parameters to be considered are distortions and delay. These factors are depending on the type of low bit-rate encoding. In general it can be distinguished between the following principles:

- Waveform Coder:
  - independant of the used bit-rate, all so called Waveform Coders are reproducing more or less the original waveform at the output after decoding. These are mainly the different ADPCM algorithms as 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 is then performing a speech synthesis again. This category is including the RPE-LTP-Coder (used in 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 Ie. 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.1.1.8 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 is causing additional delay, depending on the cell orientation and transfer mode. Therefore in any case information should be available for planning purposes about this delay, expressed as a value for the mean one-way delay in msec.

C.1.1.9 Echo cancellers

As already described in subclauses 6.2 and 7.2, the result of a planning calculation can indicate, that the Total Impairment Value Itot as a sum of different impairments is mainly influenced by the impairment value for echo Id. In this situations the decision should be made to insert echo control devices. Those devices are available as either Echo Suppressor or Echo Canceller. 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 is only dealing 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 in the network where it should be inserted. 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 9.

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 [19] (or G.165 [18]). Echo cancellers integrated in specific equipment are usually designed for this specific application and therefore not necessarely following ITU-T Recommendation G.165 [18] in all data.

NOTE: The characteristics of echo cancellers according to G.165 [18] are measured using a noise signal. If results are available also performed with other test signals (e.g. artificial voice according to ITU-T Recommendation P.50 [1993] or Composite Source Signals) these data will provide a more accurate issue.

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 national local or long distance calls, where no echo control is applied usually in public networks, the use of echo cancellers can become necessary due to additional delay only within the private network. In this situation the responsibility for sufficient echo control is born 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.

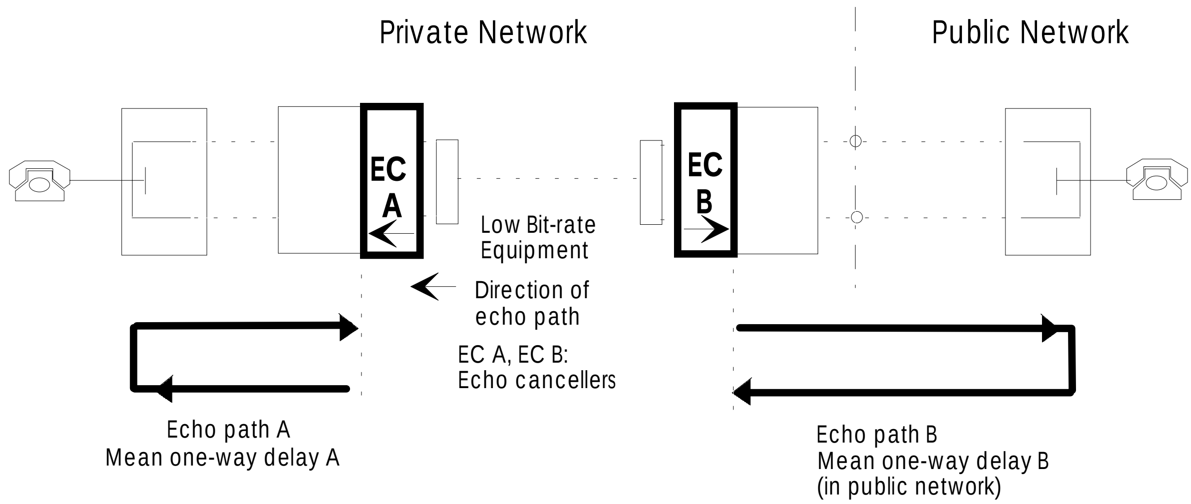

Figure C.1: Echo cancellers in a private network and their echo paths

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 talker of the public network and vice versa.

For echo cancellers in accordance with G.165 [18] the echo path should provide a **minimum echo loss** of 6 dB for sufficient operation. This value should mainly be checked 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 usually the tail delay of the echo path is expressed as mean one-way delay in transmission planning, while the corresponding data of an echo canceller is given as total tail delay in msec. 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 - 8 msec higher than the value of the echo path. For echo cancellers according to G.168 [19] or G.165 [18] the ability to compensate can be assumed in the range of 40 to 60 msec. 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 are using adaptation and cancelling algorithms based on such an assumption. 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 depending 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 (or 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 posssible 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 used algorithm, 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 Send 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 of the modems are therefore transmitting a signal tone before starting the data transmission, the so called **"Disabling Tone"** with a frequency of 2100 Hz, to disable inserted echo cancellers. Depending on the given applications in the private network, the used echo cancellers should provide this feature.

C.1.2 Transmission parameters of public networks

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

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

For the investigation of the actual values of a public network also 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 terminations (single telephones, PBXs or other private Networks), or with interfaces to other public networks. A possible scenario how access and routing with different public networks can vary in a certain country is shown in figure C.2. It should be noted, that this figure is only an example, i.e. a variety of other configurations depending on competition and liberalisation is possible.

In figure C.2 several types of interconnection with public networks are shown. The main national public network accessed via point A is providing 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 is usually also connecting 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.

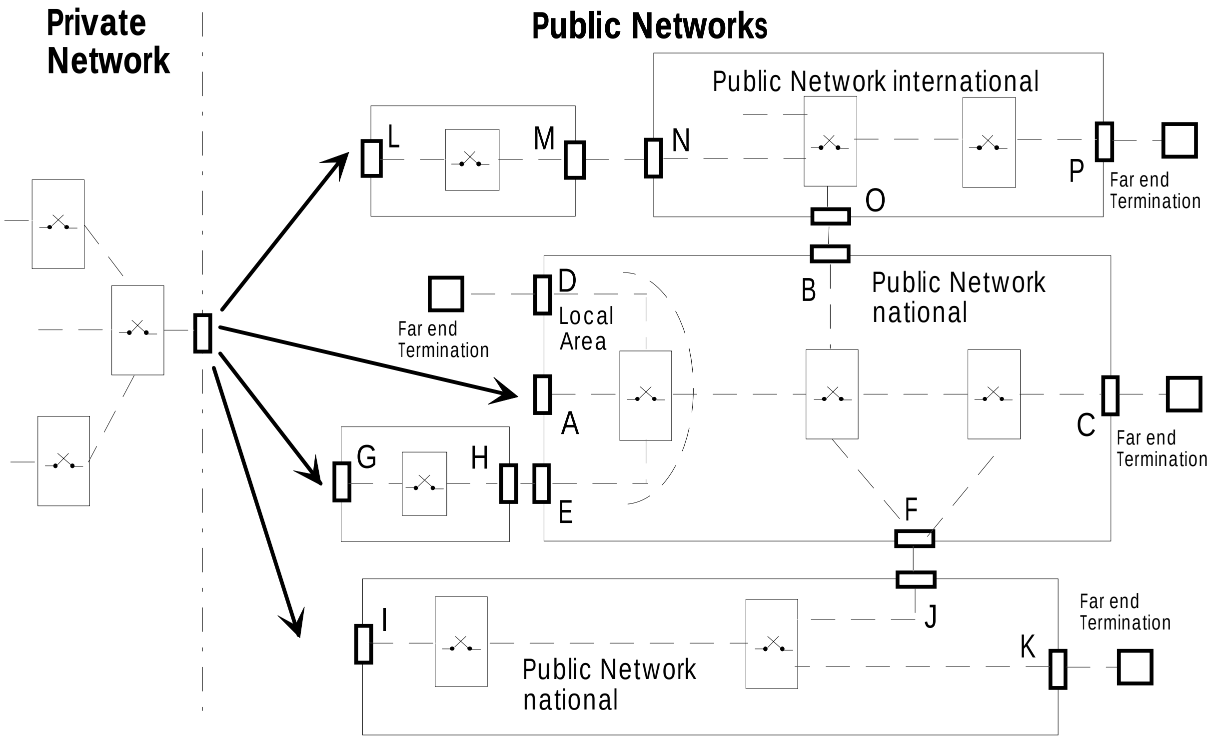

Figure C.2: Access and routing in public networks

Beside the main national network also further public networks are available as shown via the access I with own far end terminations (terminals) via point K or interconnections to the main public network via J and F. Also public networks only in the local area may exist as demonstrated in figure C.2 via the 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.

These examples for the possible interconnection of a private network with public networks as shown in figure C.2 for the interface points A, G, I and L should be subject to transmission planning, where the results of investigations about the expected quality for the different interconnections may also be used to compare the different offers on a cost/quality basis. Also more than one of these access types can be used in parallel, e.g. national connections via A or I and international connections only 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. However the terminations via C, D, K and P in the different networks can be either analogue or digital, depending on the network and the selected far end termination for planning purposes (see subclause C.1.3). Interconnections to international lines as via point B can be assumed in any case to be digital or 4-wire analogue.

In large main public networks different routings and handlings of a call will be possible, resulting in a wide range for the different transmission parameters (e.g. delay) instead of a specific value for every call. In this case it is recommended - according to the basic planning principle - to determine the values more on a statistical basis than on a "Worst Case" consideration. However, if possible own mean values should be determined for the different categories of calls 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.

Loss

Loss values should be determined for both transmission directions, mainly 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.

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 is also consisting of 2-wire sections within the public networks, a possible delay of this part should not be included with respect to echo. Specific care should be taken to the possible use of ATM with additional delay and a routing via radio (e.g. for points G and H in figure C.2) or satellite links (e.g. for points L and M in figure C.2).

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.

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 deciding whether echo cancellers are required within the private network. Echo cancellers will mainly be used for international calls. However in some cases, mainly between adjoining European countries, no echo cancellers are used. For national calls echo cancellers are possibly 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.1.1.9) it can be assumed that devices as used in public networks are in most cases in accordance with the ITU-T Recommendation G.165 [18] or G.168 [19]. For mean one-way delays of more than 5 msec within the private network, also the maximum tail delay the echo canceller is able to compensate should be known.

Quantization distortion units.

The number of qdus is decreasing with the increasing digitalisation of public networks. For planning purposes this value is only important if further qdus should be considered within the private network. As already stated in subclause 7.4 only A/D-D/A conversions according to ITU-T Recommendation G.711 [21] (A-law or µ-law) and digital loss or gain pads should be included in the planning by a value of qdu. For other coding principles such as ADPCM or low bit-rate coders the factor Ie should be used.

Equipment impairment values.

The information from the public network operators should include values for Ie if along the route low bit-rate coding systems are used. This is also important in case of radio sections e.g. when using ADPCM according to the DECT-Standard [2, 3]. As far as standardized coding laws are used, the values as listed in table 2 can be used for planning.

C.1.3 Transmission parameter of the far end termination

The planning principles recommended in this 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 collection of data.

To determine the transmission characteristics of the far end termination, it is necessary to consider two possibilities. The first are calls terminating in a single telephone (residential) and the second are 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).

For the definition of a far end termination including all relevant transmission parameters, three different types of termination are recommended as shown in figure C.3. It should be stated, that configuration and values of these three terminations may be considered as proposals only. The national loss planning and national regulations may also result in other more realistic configurations and / or values. If average values and further information for a specific country or a specific far end termination are available and deviating from the proposed terminations, the use of these 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 an analogue switching and negligible loss. The telephone set is assumed with 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 certain country from the values given above. The mean one-way delay and number of qdu are assumed to be 0 in this termination.

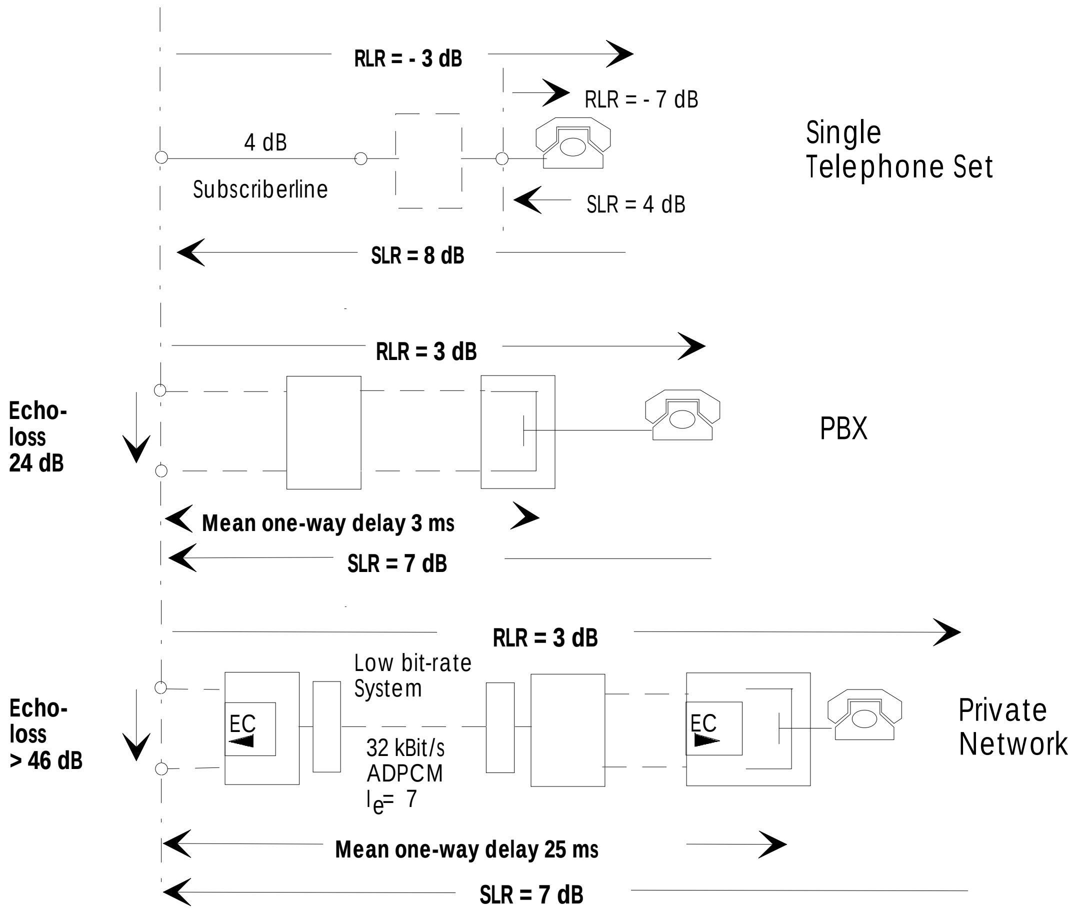

Figure C.3: Configurations and transmission characteristics for the far end termination

The second termination is representing a "Standard PBX" or a small network as usual for terminations in the business area. The interconnection with the public network and between the two PBXs is assumed to be digital, however the telephone is connected analogue via an extension line with negligible lenght. The required 4-wire/2-wire conversion within the PBX is forming the echo path with a weighted echo loss of 24 dB. The average mean one-way delay of the echo path is assumed to be 3 msec. The Loudness Rating values are referred to the digital interface with SLR = 7 dB and RLR = 3 dB, following ETS 300 283 [4] for digital interfaces to public networks. The number of qdu is assumed to be 0,5, since no digital loss pads are used.

The third configuration is simulating a "typical" routing within a large private network. Also in this case the interface to the public network and the interconnections between the PBXs are assumed to be digital. Between the first and second PBX along the call path a low bit-rate system is inserted using ADPCM with Ie = 7 in conjunction with voice activity detection (VAD) contributing with additional delay. The mean one way delay between the public network interface and the terminating hybrid (with a 2-wire analogue extension line) is assumed to be 25 msec including the influence due to VAD. With respect to this high values of delay, echo cancellers for both directions (echo paths) are enabled. The echo loss as provided by this termination can therefore be set to > 46 dB. According to ETS 300 283 [4] the Loudness Ratings are as for the second configuration SLR = 7 dB and RLR = 3 dB. The number of qdu for this far end termination can be assumed to be 0,5.

Annex D: Planning example

Due to a variety of differences between the European and the North American area with respect to network configurations and values for the different network elements, the following planning examples are separated to provide a more realistic scenario. Therefore clause D.1 is dealing with an example more related to the European situation and clause D.2 to the North American area.

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 10.

D.1.1 Description of the network and users 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.

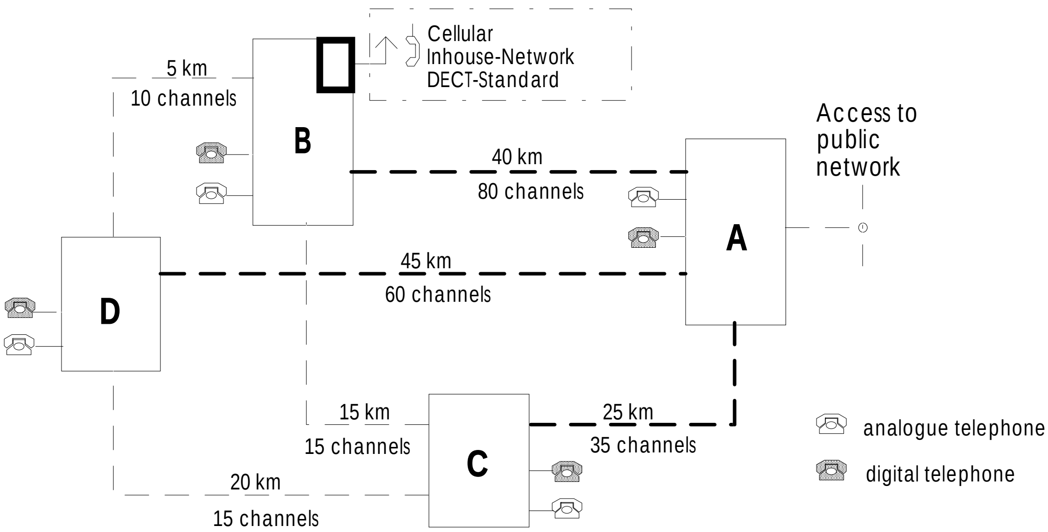

Figure D.1: Basic structure of the private network

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 lenght of the leased lines. The resulting reference configuration 1 is shown in figure D.2.

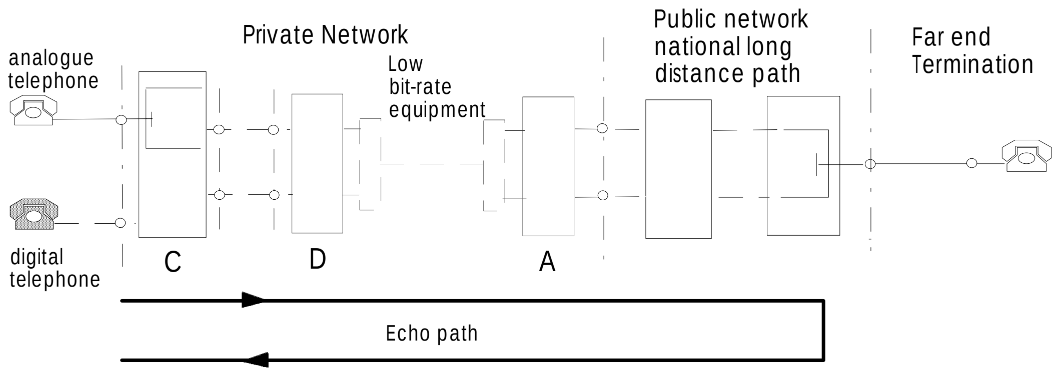

Figure D.2: Reference configuration 1

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.

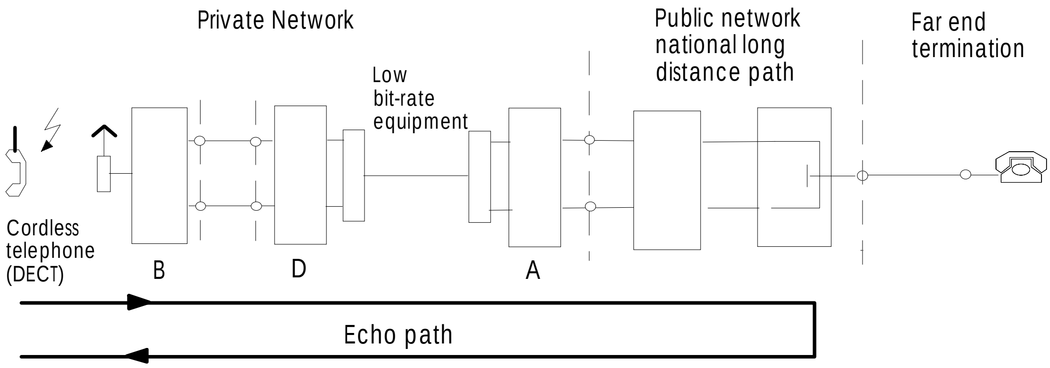

Figure D.3: Reference configuration 2

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.

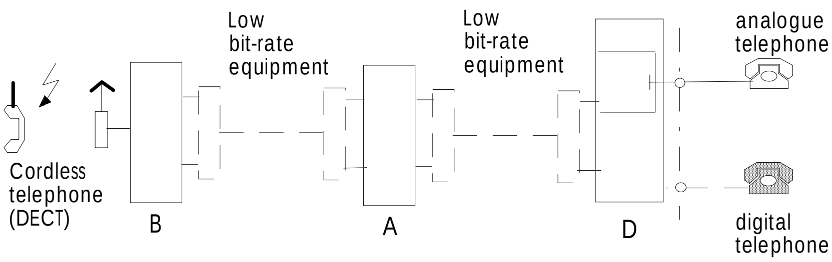

Figure D.4: Reference configuration 3

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 informations are 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 msec 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 msec
Line	A - D		1,5 msec
Line	B - C		0,8 msec
Line	B - D		0,5 msec

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 = 270Ω + (750Ω || 150 nF)
Balance impedance	Z_B = optimized for termination with Z_R
Mean one-way delay	τ = negligible

Digital telephones (according to ETSI TBR 8 [1]):

Send loudness rating	SLR = + 7 dB
Receive loudness rating	RLR = + 3 dB
Terminal coupling loss weighted	TCLw = > 46 dB
Mean one-way delay	τ = 1,5 msec

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 ETSI TBR 10 [2]):

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

Send loudness rating	SLR = + 7 dB
Receive loudness rating	RLR = + 3 dB
Mean one-way delay	τ = 14 msec
Soft suppressor (fixed part)	
additional echo loss	9 dB
hangover time	60 msec
Echo canceller	not applicable for digital interfaces
Artificial echo loss	available, but disabled
Terminal coupling loss weighted	TCLw = 46 dB and optional 34 dB 1)

1) Both types of portable parts should be assumed, however signalling to the fixed part according to TBR 10 [2] 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	270Ω + (750Ω || 150 nF)
Balance impedance	270Ω + (750Ω || 150 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 msec (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	Ie = 7
24 kBit/sec ADPCM equipment	Ie = 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.

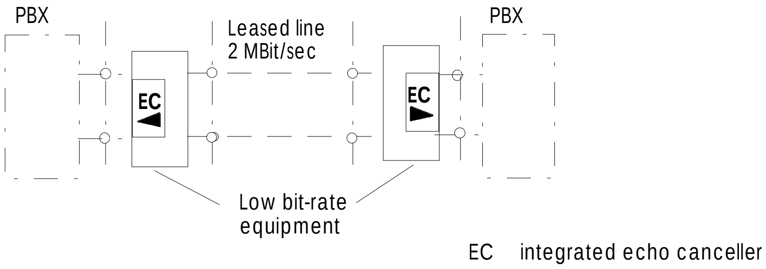

Figure D.5: Use of low bit-rate equipment for a leased line

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 (with VAD for 32 or 24 kBit/sec)	20 msec
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 11.2 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 msec
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 [18] or G.168 [19].

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 to 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 Ta is well below a value of 150 msec. 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.

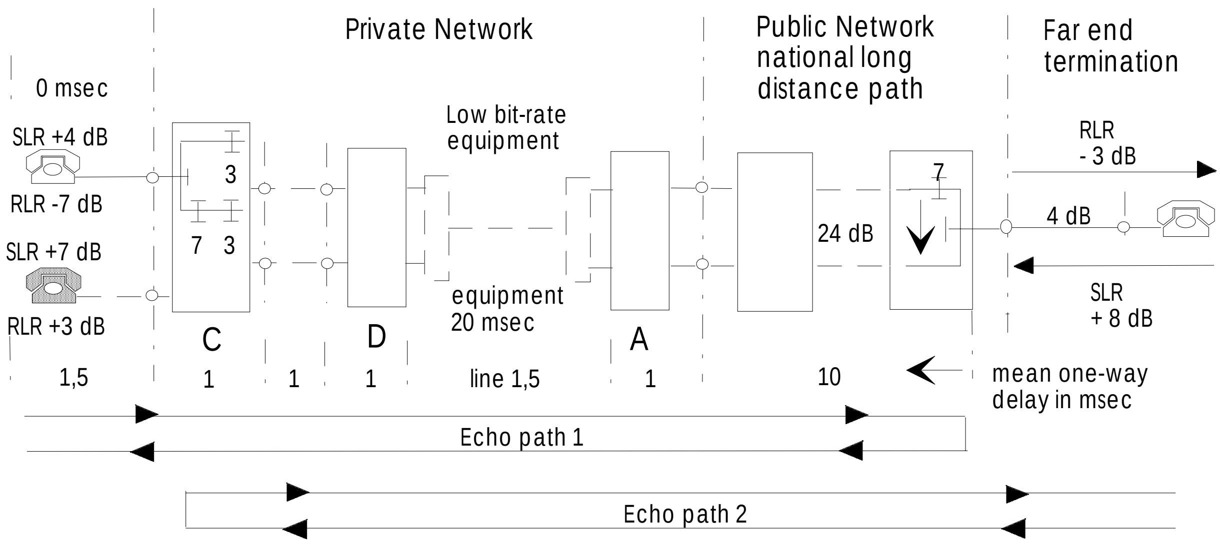

Figure D.6: Basis for calculation of reference configuration 1

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 msec for the low bit-rate equipment and of 1,5 msec for the leased line itself. The TELR can be calculated as follows:

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

The 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 msec
- digital telephone T = 37,0 msec

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 dBr-point should be determined as the final input parameters. For the telephones in the private network, assigned to the receive side of the E-Model, only RLR should be available. The value is obtained with RLR = 3 dB for both telephones, which is equal to the default value. For the SLR the path between the far end termination (send side) and the 0 dBr-point is relevant. The corresponding result is SLR = 8 dB.

The remaining impairment to be determined for this example is the equipment impairment factor Ie. The reference configuration 1 is containing the low bit-rate equipment between the PBXs in A and D. The used coding is ADPCM with a bit-rate of 32 or 24 kBit/sec. According to table 2 the corresponding values are Ie = 7 for 32 kBit/sec and Ie = 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)

TELR = 34 dB (for both telephones)

T = 35,5 msec (for analogue telephones)

T = 37,0 msec (for digital telephones)

Ie = 7 (for 32 kBit/sec ADPCM equipment)

Ie = 25 (for 24 kBit/sec ADPCM equipment)

qdu = 1,7 (for analogue telephones)

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

To reduce the number of calculations the parameter T can be averaged to T = 36 msec and the number of qdu can be left at the default value, since there will be no difference in the results for values of less than qdu = 4
(see also 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 Is, Id and Ie and the total impairment value Itot, where this value is obtained according to the relation Itot = 94,3 - R.

	R	Itot	Is	Id	Ie
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 Itot are too high in both cases. Examining the separate values, mainly for Id the sum for impairments due to echo and Ie for equipment impairments, the major impairment is caused by delay for 32 kBit/sec ADPCM equipment. Nearly the same impairment is contributing additionally when 24 kBit/sec ADPCM equipment is used. The high value for Id can be reduced if echo control devices are used. In this case also the value for Itot = 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 ≥ 7 dB and RLR ≥ 3 dB are met. Also the echo loss of ≥ 24 dB is guaranteed with 31 dB (25 dB of the hybrid in C and 2 x 3 dB digital loss in C). However these values are restricted for networks with a mean one-way delay of less than 5 msec. For reference configuration 1 however, this value is 25,5 msec 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 msec, identical with echo path 1. The summation for TELR is:

SLR of the far end termination	8 dB
Hybrid loss in the Local Exchange	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 Exchange	7 dB
RLR of the far end termination	-3 dB
TELR	43 dB

All other input values can either be left at their default values or have the same setting as for echo path 1. For the equipment impairment value only the 32 kBit/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 msec

Ie = 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 Itot the subscriber at the far end termination can expect higher quality then 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.

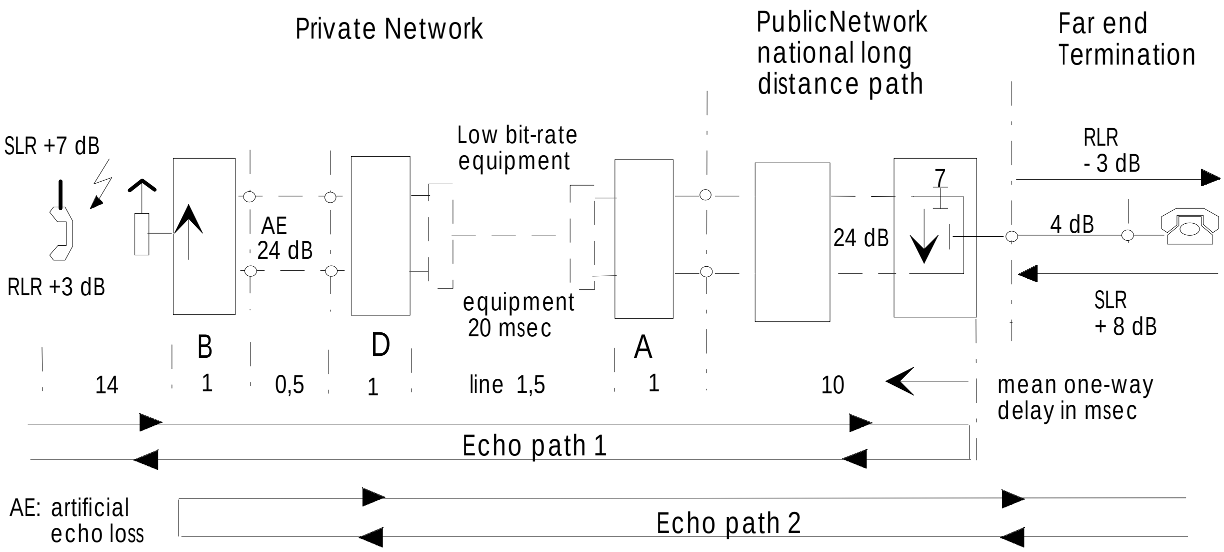

Figure D.7: Basis for calculation of reference configuration 2

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 msec. 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 Ie = 14. The input values to the E-Model for calculation of reference configuration 2 are:

SLR = 8 dB

RLR = 3 dB (equal to the default value)

TELR = 43 dB

T = 49 msec

Ie = 14 (for two times 32 kBit/sec ADPCM)

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 Itot = 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 SLR = 7 dB (equal to default) and RLR = 4 dB.

When preparing the input parameters mean one-way delay T and TELR, two different configurations at the cordless telephone with respect to echo should be taken into account. First, the portable part of the cordless system can provide a TCLw of > 46 dB (this option is not indicated in figure D.7 with respect to the echo path). In this case the TELR is formed by the SLR = 8 dB and RLR = -3 dB of the far end termination, the 7 dB receive loss in the hybrid of the Local Exchange and the TCLw of 46 dB resulting in a value of TELR = 58 dB. This value is nearly identical with the use of an echo canceller independant 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 msec from the cordless system (this configuration is shown in figure D.7 and indicated by echo path 2). The total mean one-way delay of echo path 2 is T = 35 msec. The TELR can be calculated to 36 dB. For the equipment impairments the same value of Ie = 14 is effective as for the extention of the private network. The input parameters for the E-Model are:

SLR = 7 dB (equal to the default value)

RLR = 4 dB

TELR = 36 dB

T = 35 msec

Ie = 14 (for two times 32 kBit/sec ADPCM)

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 Itot = 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 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.

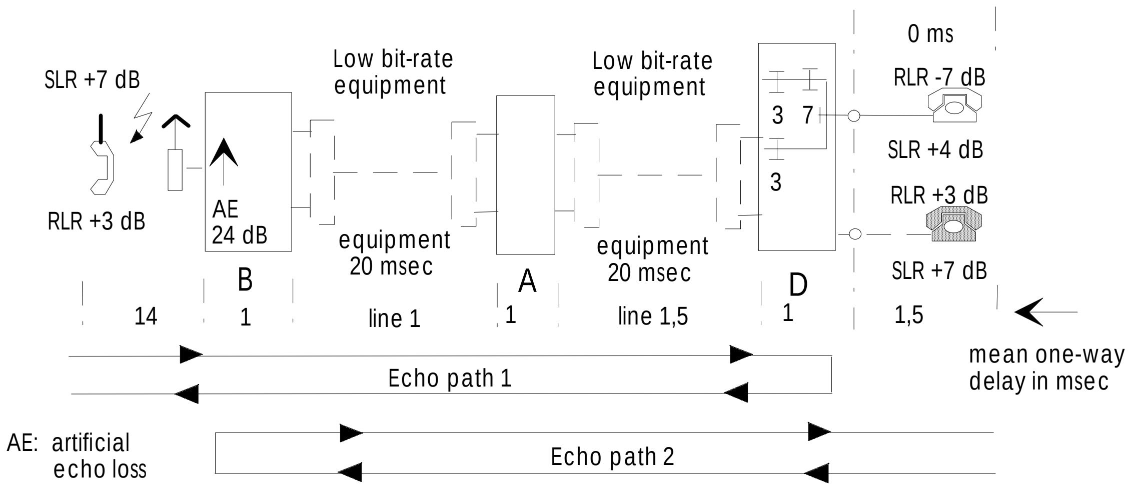

Figure D.8: Basis for calculation of reference configuration 3

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 msec. 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 Ie = 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 msec

Ie = 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 msec 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 msec of the cordless system are not included due to the artificial echo loss. However, contrary to the previous calculation, additional 1,5 msec should be taken into account for the digital telephone. The total value is then calculated with T = 47 msec. 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 msec
- 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 Config.	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 informations about the use of echo cancellers and all necessary characteristics which should be taken into account are 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 at 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 msec 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 msec and the leased line between C and D with 1 msec. The actual echo path delay of 2 x 3 msec = 6 msec 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 [18] 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 extentions 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 msec 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 msec) 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.

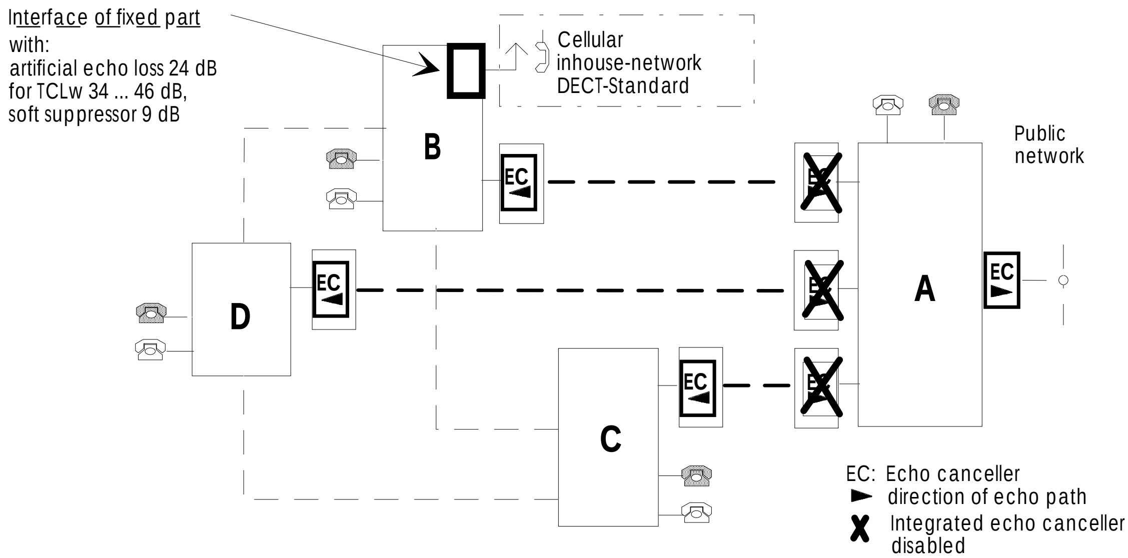

Figure D.9: Application of echo control devices in the private network

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

The only problem remaining are connections to analogue extensions in A. For talkers at B, C and D there is no echo control available due to the integrated cancellers at A being disabled. It is not possible to enable these cancellers by different reasons, as due to the need of avoiding tandeming with the separate cancellers, due to the non-linear echo path and due to exceeding the maximum permitted echo tail lenght 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 Itot are reduced by the amount of the Id values. The only impairments remaining are the equipment impairments Ie, which can be up to Ie > 21 for reference configuration 3. However it should be noted, that this configuration was defined as a critical connection including a rerouting with an additional low bit-rate section. Depending on other features of this private network, these rerouting will not necessarely be the "standard" routing for most of the calls.

History

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