INTERNATIONAL TELECOMMUNICATION UNION

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SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

International telephone connections and circuits – Transmission plan aspects of special circuits and connections using the international telephone connection network

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Transmission planning for private/public network interconnection of voice traffic

ITU-T Recommendation G.175

(Previously CCITT Recommendation)

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ITU-T RECOMMENDATION G.175

TRANSMISSION PLANNING FOR PRIVATE/PUBLIC NETWORK INTERCONNECTION OF VOICE TRAFFIC

Summary

Recommendation G.171 (1988) "Transmission plan aspects of privately operated networks" deals mainly with calls wholly within a private network. Only limited guidance is given for the interconnection of private networks with the public PSTN.

This Recommendation deals with the digital interconnection of public ISDN/PSTN and private networks. The primary application is to the overall quality of speech transmission for 3.1 kHz voiceband telephony using handsets, independent of all other types of services (e.g. facsimile and voiceband data) provided by those networks. The intention is to give guidance for transmission planning purposes, not only for a given network operator, but also for negotiations between the involved network operators.

Source

ITU-T Recommendation G.175 was prepared by ITU-T Study Group 12 (1997-2000) and was approved under the WTSC Resolution No. 1 procedure on the 18th of April 1997.

FOREWORD

ITU (International Telecommunication Union) is the United Nations Specialized Agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the ITU. The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

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

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

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

NOTE

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

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As of the date of approval of this Recommendation, the ITU had/had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

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Recommendation G.175

TRANSMISSION PLANNING FOR PRIVATE/PUBLIC NETWORK INTERCONNECTION OF VOICE TRAFFIC

(Geneva, 1997)

1 Scope

Most of the Recommendations in the G-Series are presently based on configurations where the national part of an international connection is usually terminated by a single analogue telephone set or by a digital terminal. Consequently, these Recommendations do not take into account PABXs (Private Automatic Branch Exchange) or private networks. However, modern private networks, mainly those of large size and/or using new technologies, will contribute in a specific, possibly significant amount to the overall transmission quality.

This Recommendation deals with the digital interconnection of public ISDN/PSTN and private networks. The primary application is to the overall quality of speech transmission for 3.1 kHz voiceband telephony using handsets, independent of all other types of services (e.g. facsimile and voiceband data) provided by those networks. The intention is to give guidance for transmission planning purposes, not only for a given network operator, but also for negotiations between the involved network operators.

For the purpose of this Recommendation, only call paths between the private network and other networks (private or public) including telephone sets or other speech terminals are considered. Consequently, the provision of through-connections between two interfaces to other networks, or a call path between two terminals inside the same network are not covered by this Recommendation.

NOTE – Although in principle the planning of internal- and through-connections of the Private Network is not covered here, the methods and rules described in this Recommendation may be used for those applications as well.

2 Normative references

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

- ITU-T Recommendation G.101 (1996), *The transmission plan.*
- ITU-T Recommendation G.113 (1996), *Transmission impairments.*
- ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections.*
- ITU-T Recommendation G.165 (1993), *Echo cancellers.*
- ITU-T Recommendation G.168 (1997), *Digital network echo cancellers.*
- CCITT Recommendation G.171 (1988), *Transmission plan aspects of privately operated networks.*

– CCITT Recommendation G.703 (1991), *Physical/electrical characteristics of hierarchical digital interfaces.*

– CCITT Recommendation G.711 (1988), *Pulse Code Modulation (PCM) of voice frequencies.*

– CCITT Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM).*

– CCITT Recommendation G.727 (1990), *5-, 4-, 3- and 2-bits/sample embedded Adaptive Differential Pulse Code Modulation (ADPCM).*

– CCITT Recommendation G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction.*

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

– ETSI ETS 300 580-2 (5/94), *European digital cellular telecommunications system (Phase 2); Full Rate Speech Transcoding*; (GSM 06.10).

– ETSI ETS 300 581-2 (8/95), *European digital cellular telecommunications system (Phase 2); Half Rate Speech Transcoding*; (GSM 06.20).

– ETSI ETS 300 726 (3/96), *Digital cellular telecommunications system; Enhanced Full Rate (EFR) Speech Transcoding*; (GSM 06.60).

– EIA/TIA/IS-54-B (4/92), *Cellular System Dual-Mode Mobile Station Base Station; Compatibility Standard.*

3 Abbreviations

This Recommendation uses the following abbreviations:

ATM Asynchronous Transfer Mode

DCME Digital Circuit Multiplication Equipment

ETSI European Telecommunication Standards Institute

GOB% Percentage Good or Better

ICP International Connection Point

ISDN Integrated Services Digital Network

LSTR Listener Sidetone Rating

MOS Mean Opinion Score

OLR Overall Loudness Rating

PABX Private Automatic Branch Exchange

PCM Pulse Code Modulation

POW% Percentage Poor or Worse

PSTN Public Switched Telephone Network

qdu Quantizing Distortion Unit

RLR Receiving Loudness Rating

SLR Sending Loudness Rating

STMR Sidetone Masking Rating

TELR	Talker Echo Loudness Rating
VPN	Virtual Private Network
WEPL	Weighted Echo Path Loss

4 Definitions

4.1 Private network

The term "private" is normally used in conjunction with several PABXs forming a network, mainly in an application for a restricted user group. In complement, the term "public" is usually used to describe main national or regional telecommunication networks providing services to general public.

The following list, specifying in more detail the definition of a private network, is also based on the assumption that the call path within the private network is contributing a possibly significant amount of transmission impairments to the overall transmission quality, such as loss, transmission time, number of qdus, etc.

The term "private network" is defined as follows:

1) It consists normally of more than one switching equipment (PABX), connected via private or leased lines, forming a network, independent of its structure and hierarchy. Switching equipments and leased lines can be either digital or analogue.
2) It provides switching functions and all other features only to a single customer or a group of customers, but is not accessible to everyone.
3) There is no limitation by its geographical size, it is not restricted to the national area and it is not limited on the number of extensions and access points to other networks.

A private network consists of private local exchanges providing interfaces for all types of terminal elements and for transmission elements to other private local or private transit exchanges and private transit exchanges with interfaces for transmission elements to other private transit or private local exchanges.

4.2 Public network

The term "public network" is used in this Recommendation for all networks providing their switching functions and features not only to a specific user group, but also to the general public. The word "public" is not related to the legal status of the network operator. Public networks can be restricted to only a limited size of specific features and switching functions.

Furthermore, public networks may provide access points only in a specific geographical area. From the point of view of a connection, public networks are mainly "transit networks". However, they may also be considered as a combination of "transit and terminating networks" in cases where the public network operator is also providing terminal equipments such as telephone sets, PABXs, or PABX-Features.

4.3 Network elements

All the components forming a connection can be divided into three main groups. The interconnection between private and public network is shown in the reference configurations of Figures 1 through 4. The private network is comprised of terminal elements, switching elements and transmission elements.

4.4 Types of traffic

In the case of some private networks, the "main types of traffic" via other networks (mainly public networks) may be taken into account for a possible higher amount of permitted impairments within the private network. The inclusion of the type of external traffic into planning enables the planner – wherever this is possible – to extend the limits for specific parameters (e.g. transmission time) within the private network, resulting in a more economical design of the network.

As a basic distinction for the traffic via public networks, three different types can be identified for planning purposes and with respect to the amount of transmission impairments. Referred to the switching element (local exchange) of the public network providing the access to the private network, *Local Traffic* means all connections in the local public network or in a restricted geographical area of the public network.

A second type of traffic is the *National Long Distance Call Traffic*, which designates all calls in the entire area of a country. Usually this area is identical to the area of coverage of the major public network(s) in this country.

Finally, *International Calls* must be considered to contribute in most cases with a higher amount of transmission impairments than national calls.

The distinction into these types of traffic may support negotiations between public and private network operators, not only for the partitioning of transmission impairments, but also in conjunction with other technical aspects, such as the correct insertion of echo cancellers, the use of ATM with nodes in different networks, etc.

4.5 Access to the public network

Among others, the type of access to a public network may also influence the transmission planning of the private network and may be helpful for negotiations between the network operators. "Access" in this context means not only the physical characteristics of the interfaces between public and private networks, but also the point of access with respect to the hierarchy of the public network and to additional features for private networks, provided by the public network. For large private networks, the point of access does not need to be identical to the access for single subscribers. According to the scope of this Recommendation, only digital interfaces for the access to public networks are considered.

4.5.1 Digital access at the local exchange

In most cases the access to the public network will be provided by a local exchange, or by a comparable switching element in the same hierarchy of the public network, serving the area of the respective switching element within the private network. The physical characteristics of these digital interfaces will follow standardized and commonly used frame structures and bit rates, as described in Recommendation G.703.

4.5.2 Digital access at higher hierarchies (e.g. transit exchange)

For large and complex private networks with a high number of access channels to the public network, it could be advantageous for both public and private network operators to access the public network in a higher hierarchy (e.g. a transit exchange), bypassing the local exchange. This can be done either for the entirety of all access channels, or only for those channels carrying exclusively long distance or international traffic. In both applications, physical interfaces with higher bit rates and fibre optics as the transmission media may be used.

4.5.3 Virtual Private Networks (VPN)

The meaning of the term Virtual Private Network (VPN) in this context is related to a feature where connections between two switching elements of the private network are established via switching and transmission elements of a public network on an "on-demand" basis instead of a fixed leased line. For planning purposes, such a routing should be considered as part of the private network, taking into account that the impairments of a VPN may vary for every connection in contrary to a fixed leased line.

4.6 Access to other private networks

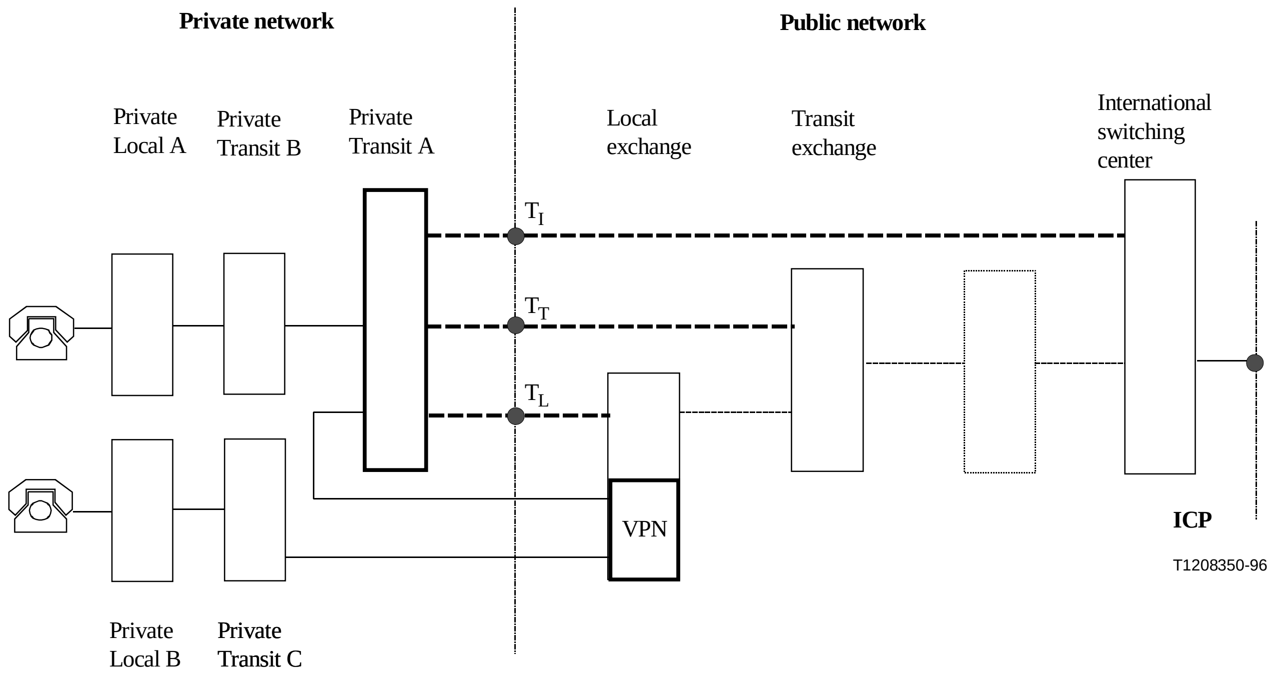
Considering the access to another private network, it is necessary to clearly identify if there is in fact a distinction between these two networks with respect to the private network definition in 4.1, or if these two networks may be considered as one network for transmission planning. The planning guidelines in this Recommendation may be also advantageous mainly in those cases, where the interconnection is only used for calls between these two networks without any routing via a public network.

While standardized interfaces and access-points will be commonly used for the access to public networks, a wider variety of physical characteristics with respect to the used frame structures, bit rates and transmission media must be considered for the interconnection between different private networks. For transmission planning, it seems important to identify if the interconnection is performed directly, or via an additional transmission element (such as a leased line, radio, or satellite link etc.), which contribute with further impairments.

5 Reference configurations

Due to the variety of hierarchies, structures, routing, and number and types of network elements in a private network, each connection investigated will result in a different reference configuration. Therefore, it is not possible to create only one basic figure for the whole task of private network planning. Figures 1 to 4 are to be considered only as examples, used mainly for definitions in this Recommendation.

The basic reference configuration for the interconnection between a public and a private network is shown in Figure 1. The private network contains transit and local exchanges with its terminals. The public network is only shown up to the international connection point of an international switching centre.



ICP International Connection Point

T_L Access point to local exchange of public network

T_T Access point to transit exchange of public network

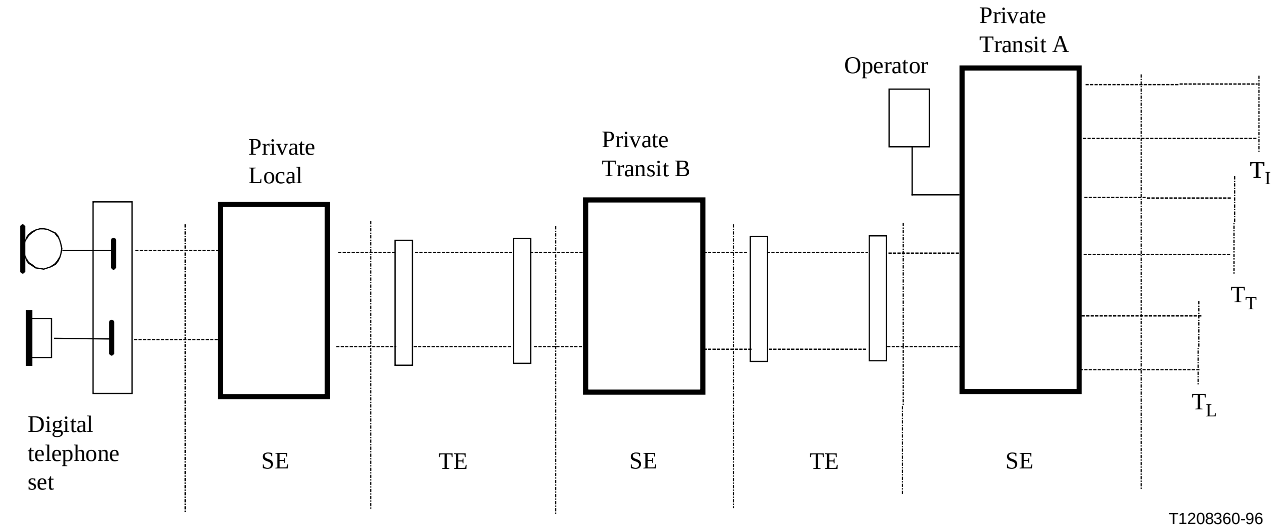
T_I Access point to international switching center

VPN Virtual Private Network

Figure 1/G.175 – Basic reference configuration for the interconnection between private and public network

It is assumed that the permitted impairments between the access points for calls within the national network are partitioned symmetrically with reference to the International Connection Point (ICP), which will be considered as a virtual center of the public network. Since calls can be terminated on both sides of private networks in the same configuration, it seems sufficient to draw Figure 1 in this simple way. From the planning point of view, the private network is also divided into local exchanges and into higher level transit exchanges, similar to the public network. The interconnection between the private and the public network is assumed in three different configurations. The access T_L represents the standard interconnection to the local exchange of the public network. Further types of access called T_T and T_I are bypassing the local exchange and entering the public network in a higher hierarchy, either in a transit exchange or directly enter the international switching centre.

Figures 2 through 4 present in more detail some examples of possible configurations within the private network, in conjunction with different types of access to the public network.



T_L Digital access point to local exchange

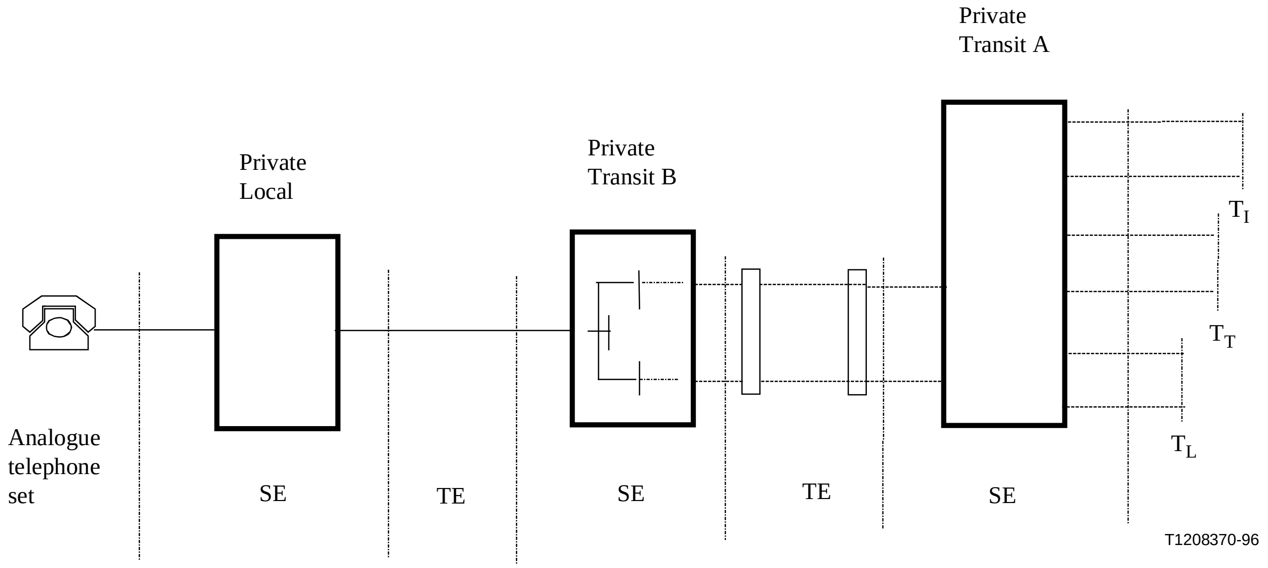
T_T Digital access point to transit exchange

T_I Digital access point to international switching center

SE Switching Element

TE Transmission Element

Figure 2/G.175 – Private network with fully digital routing



T_L Digital access point to local exchange

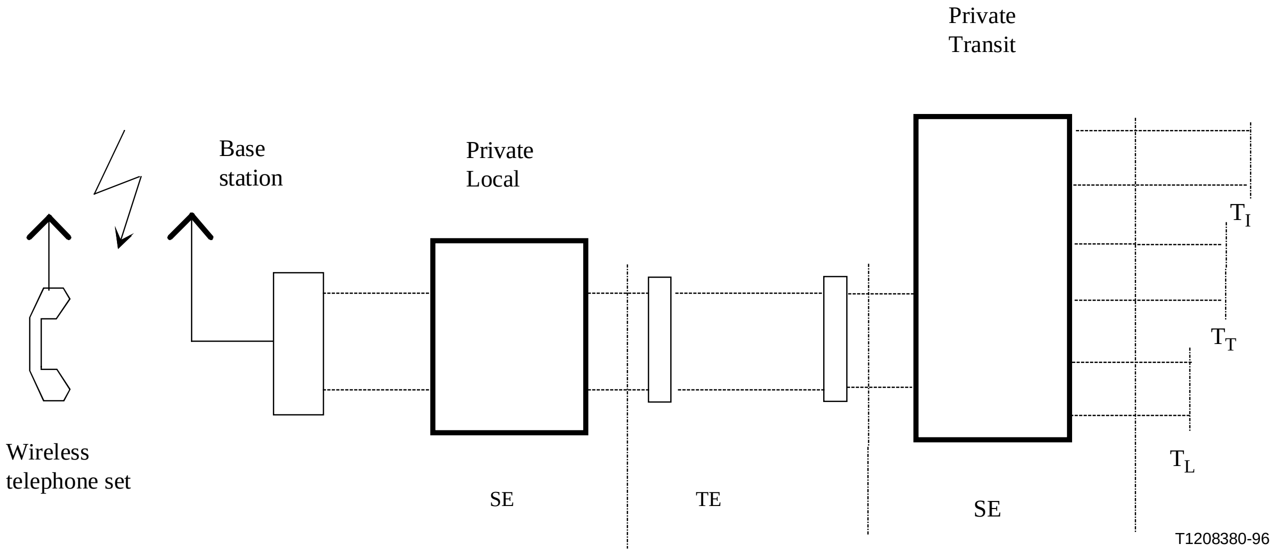
T_T Digital access point to transit exchange

T_I Digital access point to international switching center

SE Switching Element

TE Transmission Element

Figure 3/G.175 – Private network with analogue/digital routing



T_L Digital access point to local exchange
T_T Digital access point to transit exchange
T_I Digital access point to international switching center
SE Switching Element
TE Transmission Element

Figure 4/G.175 – Private network in conjunction with a digitally-connected wireless telephone

6 Basic planning principles

In general, the quality of speech transmission via telephone channels is based on a subjective judgement by the users at both ends. Therefore, transmission planning as given in Recommendation G.101 (The transmission plan) 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. For private networks, this method was in common use in the field of regulation for all calls via the public network, providing limits for the private network between the acoustical interface of the telephone set and an electrical interface to the public network. These limits were defined to guarantee a sufficient quality for all calls (national and international).

In conjunction with increasing deregulation in many countries, the responsibility for a sufficient voice transmission quality is now shifted to the private network operator. 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. This is the main task of this Recommendation.

The basic planning principle as used in this Recommendation originates from previous planning methods for private networks, which are interconnected with public networks. This interconnection is not covered by the G.100-Series of Recommendations, which still provide good guidance for end-to-end transmission planning. For all configurations subject to this Recommendation, the planning of speech transmission quality should be based on an end-to-end consideration rather than on a specification of individual objective parameter limits. End-to-end transmission quality is expressed in terms of Mean Opinion Score (MOS), Percentage Good or Better (GOB%) or Percentage Poor or Worse (POW%). The expected quality figures are calculated using Computation Models (such as the

E-Model described in Appendix I/G.101) as a planning tool and is based on the equipment impairment factor method, as described in Recommendation G.113.

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

The introduction of a quality issue for planning purposes also enables the private network operator to make the design of the network on a cost/quality relation, taking into account the specific requirements for the private network.

6.1 The equipment impairment factor method

The planning principle as recommended in clause 6 is based on a forecast of an expected quality for the investigated connection configuration. This expected quality, issued in terms of MOS, GOB% or POW%, is derived from subjective tests. In practical planning, however, it is not practical to perform subjective tests. Therefore, methods must be provided which enable the planner mathematically to combine all existing transmission impairments in the given connection into a total value of impairments. This total value must then be expressed in terms of quality using a subjectively-based algorithm. In telephone connections consisting of a variety of network elements, different transmission parameters may also simultaneously contribute to the total impairment. Therefore, the planning methods used must also incorporate combination effects.

A planning method with the approach to meet the requirements given above is available with the equipment impairment factor method in conjunction with the E-Model. This method is described in Recommendation G.113. Its basic principle consists of the allocation of a distortion value to each network element, mainly for those elements where non-waveform coders are used. This method also accounts for the impairments introduced by waveform coders and for impairments not directly related to digital processing. The distortion allocation can only be made if the results of subjective mean opinion score tests are used in conjunction with a computation model.

The equipment impairment factor method is based on the assumption that transmission impairments can be transformed into psychological factors and that these psychological factors are additive on the "Psychological Scale". If sufficient mathematical algorithms are provided by a computation model, the different transmission parameters can be transformed into "Impairment Factors *I*", resulting after addition into the "Total Impairment Value *Itot*". This method and the algorithms of the computation model also include the combination effects of those impairments which occur simultaneously in the considered connection as well as some masking effects. With that, a very useful tool is available, which provides a simplified and easy-to-handle method for practical planning purposes.

6.1.1 The total impairment value, *Itot*

The total impairment value, *Itot* can be understood as the sum of several individual impairment values,

$$Itot = Io + Iq + Idte + Idd + Ie$$

where:

- *Io* expresses the impairments which are caused by non-optimum Overall Loudness Rating,
- *Iq* represents all impairments due to quantizing distortion resulting from the PCM coding process,

Idte represents the impairments caused by talker echo,

Idd accounts for impairments introduced by long one-way transmission time in echo-free connections, like difficulties in speech conversation,

Ie represents impairments due to special equipments in a connection using low bit rate coding.

The sum of these particular impairment values can be extended by an additional value, the "Advantage Factor *A*". This value, subtracted from the total impairment value, can be used to take into account the fact that the user will tolerate some decrease in transmission quality in exchange for the "Advantage of Access", e.g. in case of mobility using wireless systems. The use of the factor *A* is expressed as the total impairment factor *Icpif*.

$$Icpif = Itot - A$$

A more detailed description of the different impairment factors can be found in Recommendation G.113 and together with the algorithms used in the E-Model in 6.2.

6.2 The E-Model

The E-Model is based on the equipment impairment factor method, following previous transmission rating models. It was developed by an ETSI ad hoc group called "Voice Transmission Quality from Mouth to Ear".

The reference connection, as shown in Figure 5, is divided in a send side and in a receive side. The model estimates the speech communication quality from mouth to ear as perceived by the user at the receive side, both as listener and talker.

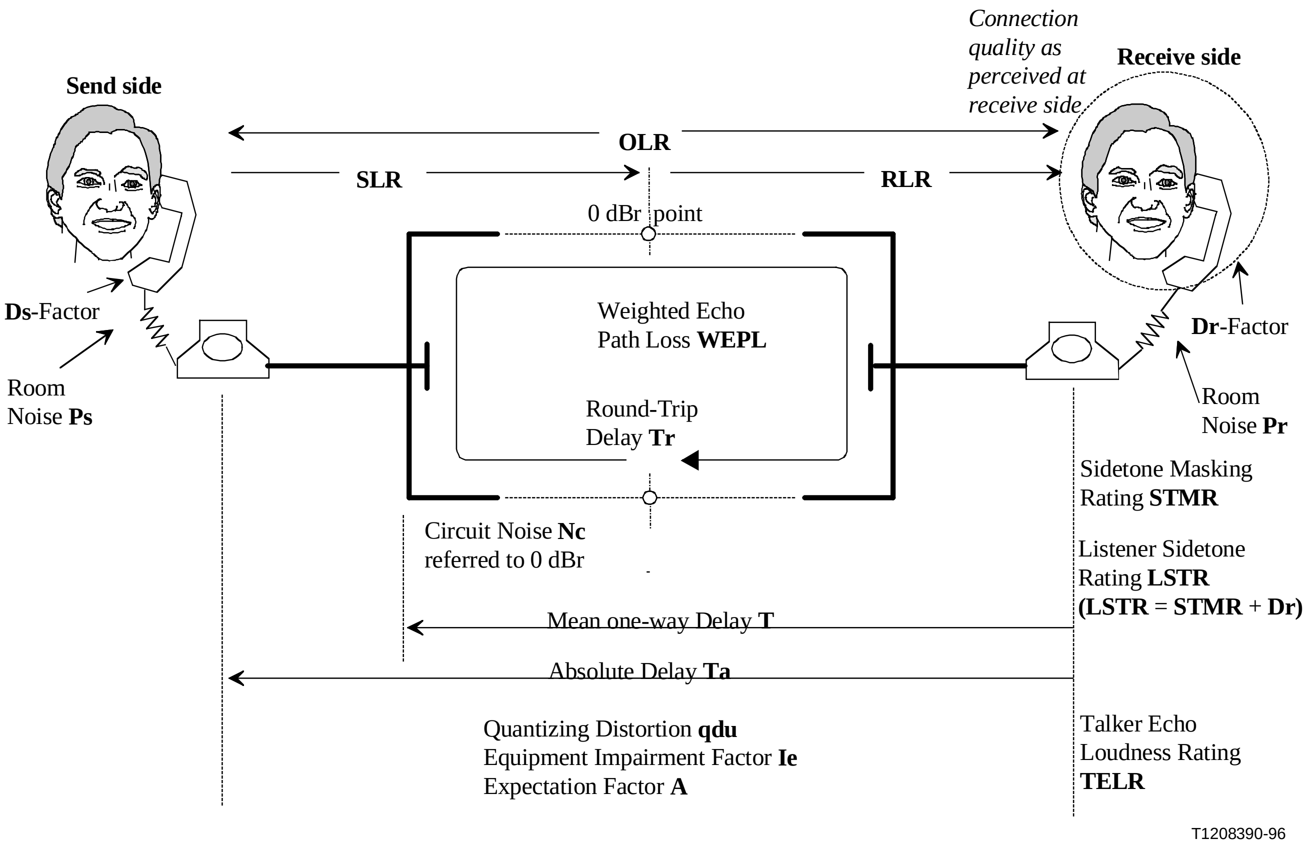


Figure 5/G.175 – Reference connection of the E-Model

The transmission parameters used as an input to the computation model are shown in Figure 5. Values for room noise and for 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 considered as values for the overall connection such as OLR (in any case, sum of SLR and RLR), number of qdu, equipment impairment factors *Ie* and advantage factor *A*, or referred only to the receive side, such as STMR, LSTR, WEPL (for calculation of Listener Echo) and TELR.

There are three different parameters associated with transmission time. The absolute delay *Ta* represents the total one-way delay between send side and receive side and is used to estimate the impairment due to too-long delay. The parameter mean one-way delay *T* represents the delay between the receive side (in talking state) and the point in a connection where a signal coupling occur as a source of echo. The round-trip delay *Tr* only represents the delay in a 4-wire loop, where the "double reflected" signal will cause impairments due to Listener Echo.

6.2.1 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 the OPINE [1] Model:

Psychological Factors on the psychological scale are additive.

The result of any calculation with the E-Model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie + A \tag{6-1}$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. The factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the equipment impairment factor *Ie* represents impairments caused by low bit rate codecs. The advantage factor *A* allows for compensation of impairment factors when there are other advantages of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following subclauses give the formulas used in the E-Model.

6.2.2 The basic signal-to-noise ratio, *Ro*

The basic signal-to-noise ratio *Ro* is defined by:

$$Ro = 15 - 1.5(SLR + No) \tag{6-2}$$

The term *No* [in dBm0p] is the power addition of different noise sources:

$$No = 10\lg\left[10^{Nc/10} + 10^{Nos/10} + 10^{Nor/10} + 10^{Nfo/10}\right] \tag{6-3}$$

Nc [in dBm0p] is the sum of all circuit noise powers, all referred to the 0 dBr point.

Nos [in dBm0p] is the equivalent circuit noise at the 0 dBr point, caused by the room noise *Ps* at the send side:

$$Nos = Ps - SLR - Ds - 100 + 0.008(Ps - OLR - Ds - 14)^2 \tag{6-4}$$

where $OLR = SLR + RLR$. In the same way the room noise *Pr* at the receive side is transferred into an equivalent circuit noise *Nor* [in dBm0p] at the 0 dBr point.

$$Nor = RLR - 121 + Pre + 0.008(Pre - 35)^2 \tag{6-5}$$

The term *Pre* [in dBm0p] is the "effective room noise" caused by the enhancement of *Pr* by the listener's sidetone path:

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

Nfo [in dBm0p] represents the "noise floor" at the receive side,

$$Nfo = Nfor + RLR \tag{6-7}$$

with *Nfor* usually set to –64 dBmp.

6.2.3 The simultaneous impairment factor, *Is*

The factor *Is* is the sum of all impairments which may occur more or less simultaneously with the voice transmission. The factor *Is* is divided into three further specific impairment factors:

$$Is = Iolr + Ist + Iq \tag{6-8}$$

Iolr represents the decrease in quality caused by too-low values of OLR and is given by:

$$Iolr = 20\left[\left\{1 + (X/8)^8\right\}^{1/8} - X/8\right] \tag{6-9}$$

where:

$$X = OLR + 0.2(64 + No - RLR) \tag{6-10}$$

The factor *Ist* represents the impairment caused by non-optimum sidetone:

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

where:

$$STMRo = -10\,\lg\left[10^{-STMR/10} + \mathrm{e}^{-T/4}\,10^{-TELR/10}\right] \tag{6-12}$$

The impairment factor *Iq* represents impairment caused by quantizing distortion:

$$Iq = 15\,\lg\left[1 + 10^{Y}\right] \tag{6-13}$$

where:

$$Y = \frac{R_o - 100}{15} + \frac{46 - G}{10} \tag{6-14}$$

and:

$$G = 1.07 + 0.258Q + 0.0602Q^2 \tag{6-15}$$

$$Q = 37 - 15\,\lg(\mathrm{qdu}) \tag{6-16}$$

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 piece of equipment, then the qdu value for that same piece of equipment must not be used.

6.2.4 The delay impairment factor, *Id*

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

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

$$Idte = \left[(Roe - Re) / 2 + \sqrt{(Roe - Re)^2 / 4 + 100} - 1 \right] \left(1 - e^{-T}\right) \tag{6-17}$$

where:

$$Roe = -1.5(No - RLR) \tag{6-18}$$

$$Re = 80 + 2.5(TERV - 14) \tag{6-19}$$

$$TERV = TELR - 40 \lg \frac{1 + T/10}{1 + T/150} + 6e^{-0.3T^2} \tag{6-20}$$

For values of $T < 1$ ms, the Talker Echo should be considered as sidetone, i.e. $Idte = 0$. The computation algorithm furthermore combines the influence of STMR to Talker Echo. Taking into account that low values of STMR may have some masking effects on the Talker Echo and for very high values of STMR the Talker Echo may become more noticeable, the terms *TERV* and *Idte* are adjusted as follows:

For STMR < 9 dB:

In equation (6-19) *TERV* is replaced by *TERVs*, where:

$$TERVs = TERV + Ist / 2 \tag{6-21}$$

For 9 dB ≤ STMR ≤ 15 dB:

the above given equations (6-17) to (6-20) apply.

For STMR > 15 dB:

Idte is replaced by *Idtes*, where:

$$Idtes = \sqrt{Idte^2 + Ist^2} \tag{6-22}$$

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

$$Idle = (Ro - Rle) / 2 + \sqrt{(Ro - Rle)^2 / 4 + 169} \tag{6-23}$$

where:

$$Rle = 10.5(WEPL + 7)(Tr + 1)^{-0.25} \tag{6-24}$$

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

For *Ta* < 100 ms:

$$Idd = 0$$

For *Ta* > 100 ms:

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

with:

$$X = \frac{\lg(Ta/100)}{\lg 2} \tag{6-26}$$

6.2.5 The equipment impairment factor, *Ie*

The values for the Equipment Impairment Factor *Ie* of elements using low bit rate codecs are not related to other input parameters. They are depending on subjective mean opinion score test results as well as on network experience. Some values listed in Table 1 are taken from Table 7/G.113.

Table 1/G.175 – Planning values for the equipment impairment factor *Ie*

Codec type	Operating rate kbit/s	Value *Ie*	Reference
ADPCM	40	2	G.726, G.727
	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[1)]	GSM 06.60, Enhanced Full Rate
CELP+	6.8	25	

[1)] Provisionally.

6.2.6 The advantage factor, *A*

Due to the specific meaning of the advantage factor *A*, there is – consequently – no relation to all other transmission parameters. Some provisional values are given in Table 2.

Table 2/G.175 – Provisional examples for the advantage factor *A*

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

It should be noted that the values in Table 2, taken from Recommendation G.113, are only provisional. The use of the factor *A* and its selected value in a specific application is up to the planner's decision. However, the values in Table 2 should be considered as absolute upper limits for *A*. With respect to subscribers of a private network, the factor *A* can be used in the same sense as for subscribers directly connected to the PSTN. Nevertheless, it is assumed that the quality expectation of subscribers in a private network is identical to that of PSTN subscribers. This means that there is no need for an additional *A*-value just for the fact that a connection is made via a private network, even if this private network provides more features than the public network.

6.2.7 Quality measures derived from the transmission rating factor, *R*

The transmission rating factor *R* can lie in the range from 0 to 100, where $R = 0$ represents an extremely bad quality and $R = 100$ represents a very high quality. The E-Model provides a statistical estimation of quality measures. The percentages for a judgement Good or Better (GOB%) or Poor or Worse (POW%) are obtained from the *R*-Factor by means of the Gaussian Error function:

$$E(x) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{x} e^{-\frac{t^2}{2}} dt \qquad (6\text{-}27)$$

The equations are:

$$GOB = 100E\left(\frac{R-60}{16}\right) \quad \% \qquad (6\text{-}28)$$

$$POW = 100E\left(\frac{45-R}{16}\right) \quad \% \qquad (6\text{-}29)$$

The Mean Opinion Score (MOS) in the scale 1-5 can be obtained from the *R*-Factor using the formulas:

For $R < 0$: $\mathrm{MOS} = 1$

For $0 < R < 100$: $\mathrm{MOS} = 1 + 0.035R + R(R-60)(100-R)\,7 \cdot 10^{-6}$ (6-30)

For $R > 100$: $\mathrm{MOS} = 4.5$

7 Planning method and limits

The introduction of the basic principles and of new planning methods were already shortly described in clause 6. Planning of private networks is performed in many cases by the operator of the private network, but it is in most applications strongly influenced by the transmission planning of the public network. Previous or still existing planning rules for interconnections with public ISDN/PSTN are applied only to the private network domain, i.e. between the acoustic interfaces and the interface to the public network. The planning values are based on a partitioning of permitted impairments, or on limits for each specific transmission parameter.

Such a partitioning, usually not taking into account the specific size, structure and complexity of a private network, will result in a very rigid handling in conjunction with stringent limits for the different transmission parameters. To provide more flexibility in this field, the planning and design of private networks should therefore be based more on individual negotiations between public and private network operators than on partitioning. Although in most cases the borders between public and private networks can be clearly identified at specific interfaces, the priority of negotiations should be more in the determination of the actual impairments in the public and private network domain. This can be supported by considering the individual configurations and requirements of a private network, such as type and point of access to the public network and the majority of connection types (international, national long distance, or local calls).

To meet these goals, a planning method is recommended and described as follows for the interconnection between public and private network. This method may also be generally applicable to multioperator networks.

7.1 Planning method

The planning method for the interconnection of private networks and the public ISDN/PSTN can be considered as a sequence of steps. The following detailed description of these different steps may be used as a guidance.

– *Configuration and requirements of the individual private network*

  In a first step, the configuration of the private network, its features with all resulting possible routings, and the type and point of access to the public network, should be considered. Furthermore, the majority of traffic via public networks should be taken into account, whenever this seems applicable, and depending on the business affairs of the private network operator.

– *Determination of reference configurations*

  As usual in network planning it is advisable to develop a reference configuration mainly for the path within the private network. This makes it easier to clearly identify all network elements and their relevant transmission impairments. It is assumed that the most critical path with respect to transmission impairments is selected as the reference configuration. However, depending on the network operator's decision, some specific configurations or routings, achieved only in exceptional cases within the network, may be accepted with lower quality, but not taken as the reference configuration.

– *Ascertainment of actual transmission impairments*

  For each of the network elements within the private network and for all main parameters, subject to planning as listed in 7.2, the actual values must be determined. It should be noticed that some elements may contribute with more than one parameter to its specific total impairment. For most of the main transmission parameters, the actual values of impairments for each element can be determined separately, followed by a simple addition of all element-related values. For some parameters such as echo and stability, however, the investigation must be performed for the relevant entire part of a connection.

  During this step it is also advisable to investigate some possible sources for further transmission impairments, such as impedance mismatching at analogue interfaces, relation between signal level and load capacity of codecs, and excessive room noise at specific locations.

  The determination of actual transmission impairments caused by public networks is subject to agreements between public and private network operators. Whenever possible, this ascertainment should include not only information about the contributing parameters and their actual values with respect to the different routing for local, national long distance and international calls, but also information about the use of echo cancellers and their performance. It must be taken into account that these values may vary in a wide range, only representing an estimate of the transmission performance of paths through the public network. Nevertheless, it is recommended that the preference should be more for a statistical than for a worst-case consideration.

  The same is valid for the opposite termination, which must be defined for the purpose of an end-to-end inspection. Most of the opposite terminations will be formed by a single analogue or digital telephone set, but PABXs or private networks must also be considered. The definition of these terminations and their transmission parameters, mainly SLR, RLR, distortions, delay and provided echo loss, should also be derived by a statistical consideration. If possible, the necessary information can be obtained from the public network operator with respect to the network configuration in the subscriber area.

The actual values with respect to all relevant transmission parameters should also be determined for all types of digital or analogue leased lines. Although these leased lines will usually be provided by public network operators, they must be considered as part of the private network from the point of view of transmission planning.

– *Planning calculation*

All actual values of the main parameters from the different elements within the private network, the public network(s) and the opposite termination, are transformed to be used in the E-Model. A detailed description on the use of this model is given in 7.4. The results can be obtained as the transmission rating factor *R* to be judged according to Table 3, or expressed as MOS, GOB% and POW%. More details about the planning calculations using the E-Model is given in 7.4.2.

– *Judgement of results*

The results of planning investigation expressed in terms of expected quality should be judged according to the limits given in 7.3. It is worth noticing that this planning method may also be used for comparing different technical solutions for a specific network element and its influence by the expected quality, e.g. different codec algorithms, use of DCME in transmission elements, deployment of ATM, etc. For the benefit of an economical design for the private network, the final decision can be made based on a ratio between costs and perceived quality.

7.2 Main parameters

Based on the assumption that in modern private networks the majority of exchanges and interconnecting (leased) lines uses digital technology and that interconnections to the public ISDN and to the PSTN are only digital, a rating of the different transmission parameters should be performed with respect to their influence on speech quality. In a digital environment, some parameters (e.g. frequency distortion, steady circuit noise, loss variation with time, etc.) have become less important. The following parameters are recommended to be included in the transmission planning.

– *Overall Loudness Rating (OLR)*

In some configurations for private networks, mainly in the lower hierarchy, small PABXs connected via 2-wire analogue sections will be in use contributing with loss. Furthermore, analogue telephone sets designed in its SLR and RLR for previous fully-analogue connections may introduce impairments caused by a non-optimum overall loudness rating.

– *Absolute delay in echo-free connections*

This parameter is mainly important in case of international calls.

– *Echo*

The investigation of impairments due to echo effects seems to be one of the most important planning aspects in a digital environment. Two different parameters, the Talker Echo Loudness Rating (TELR) and the mean one-way-delay T of the echo path must be considered. The investigation of echo should also include decisions about the use of echo cancellers.

– *Stability*

According to the assumption that private networks are interconnected digitally with the public ISDN/PSTN, any 4-wire to 2-wire conversion within the private network may terminate the international chain affecting the stability. Calculations of the stability are not covered by the E-Model, although the provision of a sufficient stability loss is strongly recommended. Values and guidance can be found in Recommendation G.122.

– *Quantization distortion*

In modern private and public networks using more and more fully digital routing, the impairment due to quantization distortion expressed in a number of qdu is decreasing. The E-Model includes this parameter, however, it should only be used for a codec pair according to Recommendation G.711. For all other coding algorithms, the equipment impairment factor *Ie*, as shown in Table 1 should be used.

– *Equipment impairment factor for complex speech processing devices*

The equipment impairment factor *Ie* which expresses the impairments caused by specific speech coding algorithms and processing devices may rise to be one of the most important impairments in modern networks. Available values to be used with the E-Model are given in Table 1.

7.3 Quality expectation and absolute upper planning limits

As described in 6.2.1 and 7.4, the results of planning calculations using the E-Model are primarily obtained in terms of a rating factor R for the considered configuration. The transmission rating factor R can lie in the range from 0 to 100 or even higher, where $R = 100$ is representing a very high transmission quality and $R = 0$ means extremely bad or unacceptable. The R-value can be transformed into a number of different quality measures such as MOS, GOB% and POW% using the Gaussian Error function as described in 6.2.7.

Basically, the judgement of the resulting quality measures is up to the planner's decision. However, it is strongly recommended to establish a specific limit which never should be exceeded, even in exceptional cases. This limit can be expressed directly in a value of the rating factor R and should never be less than 50.

In some cases, planners may not be familiar with the use of quality measures as a result from planning calculations. A provisional guidance by a verbal description of the expected quality for the different R-values is given in Table 3. This table also contains the related values of MOS, GOB% and POW%.

Table 3/G.175 – Provisional guide for the relation between *R*-value and user's satisfaction

R-value lower limit	MOS	GOB%	POW%	User's satisfaction
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Exceptional limiting case

7.4 Use of the E-Model

Initial transmission planning results using the E-Model are so promising that its use is now recommended for assessing private/public network interconnection. Formal verification of the E-Model is in progress in ITU-T Study Group 12. As with all models, there will be specific circumstances for which the results of the E-Model may not be accurate.

The basic principles and the algorithm of the E-Model are contained in clause 6. A reference configuration of the model is shown in Figure 5. When using the E-Model for planning calculations, care should be taken for a correct input of all transmission parameters. The following list of all parameters will provide the necessary guidance. As shown in Figure 5, the model distinguishes between the send side and the receive side. Both sides and most of the parameters are referred to a virtual 0 dBr point. There are 18 input transmission parameters in total, but not all of them are varied for the purpose of transmission planning. It is important to note that the model estimates the speech communication quality for both talking and listening as perceived by the user at the receive side. For transmission parameters not varied during planning calculations, default values should be set, as recommended in 7.4.3.

7.4.1 Input parameters

The following input parameters are used in the E-Model.

– *Sending Loudness Rating (SLR) and Receive Loudness Rating (RLR)*

  The values of SLR and RLR are not directly related to the values of the used telephone sets. SLR represents the loudness rating between the human mouth at the send side and the virtual 0 dBr point and RLR from the 0 dBr point to the human ear at the receive side. If network elements contributing with loss are inserted between the telephone set and the 0 dBr point and vice versa, the calculation of the total SLR and RLR must be performed separately, since the model does not allow for an input of Circuit Loudness Rating. The Overall Loudness Rating (OLR) is in any case the sum of SLR and RLR. Should a specific range of OLR be investigated, it is recommended to vary SLR and RLR simultaneously to avoid errors. If different values for SLR and RLR are existing for the telephone sets used on each end of the connection, the *R* values for both transmission directions must be calculated separately. In this application, only the SLR at the send side and the RLR at the receive side may be used as an input to the model. The remaining parameters (SLR at receive side and RLR at send side) are not used. Their influence to the speech transmission quality due to room noise and to talker echo is included via the parameters TELR, STMR and LSTR.

– *Sidetone Masking Rating (STMR) and Listener Sidetone Rating (LSTR)*

  These parameters, which are directly related to the telephone sets used, are in most cases not subject to planning and should be set to the default values. They must only be taken into account if incorrect impedance matching of analogue telephone sets, or low values for STMR and LSTR should be expected.

– *D-Factors (Ds and Dr)*

  The D-Factors *Ds* for the send side and *Dr* for the receive side are fixed values depending on the shape of the handsets of the telephone sets used. As a fixed value, they are usually not subject to planning. For the D-Factors and for the values for STMR and LSTR of a telephone set, a fixed relation is assumed:

$$LSTR = STMR + D$$

– *Talker Echo Loudness Rating (TELR)*

  The TELR, expressing the loudness rating of the echo path, is defined as the sum of SLR and RLR of the talker's telephone and the echo loss of the echo path. This value has to be calculated separately using the SLR/RLR values at the receive side with respect to the model. The echo path must be identified and calculated carefully within the reference configuration to avoid wrong inputs for the TELR.

– *Weighted Echo Path Loss (WEPL)*

In conjunction with the round-trip delay in a closed 4-wire loop, this parameter may cause impairments due to listener echo. Closed 4-wire loops may happen in a configuration if the connection includes 4-wire to 2-wire conversions. These conversions may be located in different networks and in different countries. The WEPL is defined as the sum of all losses and gains within the loop, also called the "Round-Trip Loss". In most cases listener echo can be neglected, if sufficient echo control is provided in a telephone connection.

– *Delay values (T, Ta and Tr)*

The E-Model distinguishes three different values for delay, which must be determined and used separately in the model. The mean one-way delay time *T* in ms is used to calculate the impairments due to talker echo in conjunction with TELR. It should be noted that although this is an impairment to the talker, the estimate of the impairment is referred to the receive side of the model. The mean one-way delay *T* has to be determined and calculated only for the sections in the reference configuration forming the echo path, i. e. from talker's telephone up to the identified point where signal reflections may occur, e.g. a 4-wire to 2-wire conversion. Complementarily, the absolute one-way delay *Ta* in ms is in any case the total delay via the whole connection between the two subscribers. *Ta* represents impairments due to too-long delay and must be included mainly in planning for international calls, even if perfect echo cancelling is provided. The round-trip delay *Tr* in ms will cause listener echo in conjunction with WEPL. *Tr* is defined as the total delay within the closed 4-wire loop.

– *Equipment impairment factor (Ie)*

The values for the equipment impairment factor *Ie* as an input to the E-Model are representing impairments due to low bit rate codecs in specific network elements. Provisional values based on subjective tests are given in Table 1.

– *Advantage factor (A)*

The algorithm of the E-Model also includes the Factor *A* for the calculation of the rating factor *R*; however, the inclusion and its value are subject to the planners' decision. For more information, see 6.2.6.

– *Room Noise (Ps and Pr)*

Impairments perceived at the receive side may also be caused by the room noise at the send side and the receive side, contributing to the basic signal-to-noise ratio. Values for the room noise *Ps* in dB(A) at send side and *Pr* in dB(A) at receive side are usually set to default values, but may be varied for planning purposes in case of excessive noise at a specific location. The algorithm in the E-Model transforms these values into an equivalent circuit noise referred to the 0 dBr point.

– *Circuit Noise (Nc)*

If necessary, the circuit noise *Nc* in dBm0p can be obtained by a power addition of all electric noise sources in the connection, all referred to the 0 dBr point. In most cases, sources for steady noise in a digital environment may be neglected and the input parameter may be set to its default value.

– *Noise Floor (Nfor)*

The input parameter noise floor in dBmp represents a basic noise in the equipment at the receive side. Its nominal value is set to –64 dBmp.

– *Number of Quantizing Distortion Units (qdus)*

Impairments due to quantization distortion are entered into the model as a number of qdu. It should be noted that qdus may only be used for a codec pair using a coding algorithm according to Recommendation G.711 and for distortions from digital loss or gain pads (0.7 qdu). For all other coding algorithms, the relevant equipment impairment factors *Ie* must be used.

7.4.2 Performing the calculation

If all input parameters are available, the calculation process can be described as follows:

a) Compute separately SLR and RLR and perform the calculation of the basic signal-to-noise ratio *Ro*.

b) Compute the simultaneous impairment factor *Is*.

c) Calculate separately the mean one-way delay and TELR of the echo path, the absolute one-way delay *Ta* and the round-trip delay *Tr*, and compute the delayed impairment factor *Id*.

d) Add the equipment impairment factors *Ie* for the different equipments.

e) Compute the rating factor *R* and add the *A* factor if applicable.

f) Check whether the exceptional (lower) limit of $R = 50$ is violated.

g) Compute MOS, GOB% and POW%.

In practice, computer programs will be used for the calculations providing a total run via all steps a) through g).

7.4.3 Default values

For all input parameters used in the algorithm of the E-Model, the default values are listed in Table 4. It is strongly recommended to use these default values for all parameters which are not varied during planning calculation. If all parameters are set to the default values, the calculation results in a very high quality with a rating factor of $R = 94.1$.

Table 4/G.175 – Default values for the input parameters in the E-Model

Input parameter	Abbreviation	Default value	Unit
Sending Loudness Rating	SLR	8	dB
Receiving Loudness Rating	RLR	2	dB
Sidetone Masking Rating	STMR	15	dB
Listener Sidetone Rating	LSTR	18	dB
D-Factor Send Side	Ds	3	–
D-Factor Receive Side	Dr	3	–
Talker Echo Loudness Rating	TELR	50	dB
Weighted Echo Path Loss	WEPL	80	dB
Mean One-Way Delay	*T*	0	ms
Absolute One-Way Delay	*Ta*	0	ms
Round-Trip Delay	*Tr*	0	ms
Room Noise Send Side	*Ps*	35	dB(A)
Room Noise Receive Side	*Pr*	35	dB(A)

Table 4/G.175 – Default values for the input parameters in the E-Model *(concluded)*

Input parameter	Abbreviation	Default value	Unit
Equipment Impairment Factor	*Ie*	0	–
Advantage Factor	*A*	0	–
Circuit Noise	*Nc*	–70	dBm0p
Noise Floor	*Nfor*	–64	dBmp
Quantization Distortion Unit	qdu	1	–

8 Implementation of echo cancellers

The use of speech processing devices and digital radio sections also in modern private networks will increase the delay values within the private network by an amount, which requires the implementation of echo control devices not only for national long distance calls, but also for local or internal calls. Since there seems to be less experience with the insertion of such devices in the private network domain, some guidance should be given to the planner.

Primarily it is recommended to use echo cancellers. They should meet or exceed the requirements of Recommendation G.168. Mainly in case of international calls, where echo cancellers are inserted by the public network operator, a tandeming of cancellers may occur. Practical experiences have shown that echo cancellers designed according to Recommendation G.165 will cause no major problems in a tandem configuration. It is advisable to inquire from the public network operator about the application of cancellers and their performance. Depending on the permitted echo path delay of these equipments on the point of access and on the routing within the public network, additional cancellers in the private network could possibly be avoided in some cases.

The investigation and the final decision about the insertion of echo cancellers within the private network must not only cover the echo control for the talker in the private network but also for the talker at the opposite termination in case of local or national long-distance calls via the public network.

Some terminal and transmission elements, such as cordless and mobile telephones, or speech companding devices, may provide integrated echo cancellers which must be taken into account. Those devices must be carefully controlled with respect to the relevant performance parameters, such as permitted echo path delay, residual echo loss, required echo path loss, etc.

As a basic rule, echo cancellers should be located close to the echo source within the private network for the benefit of the opposite termination. A linear echo path with a minimum echo path loss of 6 dB must be provided. For the suppression of the own talker echo (echo path via the public network and the opposite termination), information about this echo path regarding delay, echo path loss and linearity should be obtained by negotiations between public and private network operator, to allow a careful selection of the necessary devices. In most cases, these echo cancellers will be located close to the interface to the public network.

APPENDIX I

Bibliography

[1] Supplement No. 3 to the ITU-T Series-P Recommendations (1993), *Models for predicting transmission quality from objective measurements.*

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