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SERIES G: TRANSMISSION SYSTEMS AND MEDIA

International telephone connections and circuits – General
definitions

The transmission plan

ITU-T Recommendation G.101

(Previously “CCITT Recommendation”)

ITU-T G-SERIES RECOMMENDATIONS
TRANSMISSION SYSTEMS AND MEDIA

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FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (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 (Helsinki, March 1-12, 1993).

ITU-T Recommendation G.101, was revised by ITU-T Study Group 12 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 30th of August 1996.

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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SUMMARY

This Recommendation deals with transmission planning of modern networks. The transmission parameters and overall requirements of importance for planning are listed with reference to the relevant ITU-T Recommendations. Examples of network configurations are shown. The appropriate conventions and definitions are given. In particular, the division of responsibility between network operators in a deregulated market is reflected in the definitions of the constituent parts of connections.

THE TRANSMISSION PLAN

*(Geneva, 1964; amended at Mar del Plata, 1968, Geneva, 1972, 1976 and 1980;
Malaga-Torremolinos, 1984, Helsinki, 1993; revised in 1996)*

Scope

The aim of this Recommendation is to provide a guidance for transmission planning of modern telephone networks. This Recommendation lists the main technical (transmission) parameters of importance with reference to their relevant ITU-T Recommendations. Examples are given of network elements and network configurations, including mobile circuits and circuits using compression techniques. This Recommendation also discusses those implications for transmission planning which are due to the deregulating occurring in many telecom markets, in particular the division of responsibility between several network operators who may be involved in a connection. In this context the traditional subdivision of an international connection into “national systems” and an “international chain” is not always relevant and this Recommendation therefore has been complemented with a connection structure of “terminating networks” and “transit networks”.

1 Principles

The transmission plan established in 1964 was drawn up with the object of making use, in the international service, of the advantages offered by 4-wire switching. The main impairments included for control by transmission planning were end-to-end loss (i.e. OLR, Overall Loudness Rating), circuit noise and, by later amendments, the effect of PCM processes. Since then, the increasing use of digital switching and transmission has made 4-wire switching the rule and largely eliminated high values of OLR and/or noise as major sources of impairments in normal, wire bound connections. On the other hand, other types of impairments have now to be carefully considered in transmission planning, namely echo, total transmission time and the distortion introduced by low bit rate coders. (For mobile telephone connections, new types of noise disturbances also have to be taken into account.)

The following Recommendations give guidance about the important transmission parameters to be controlled by a modern transmission plan:

G.111 and G.121 for loudness ratings, OLR, SLR, RLR, STMR, LSTR;

G.122 for echo and stability considerations in general;

G.126 for listener echo;

G.131 for talker echo, TELR;

G.114 for total transmission time;

G.113 gives transmission impairment considerations, both taken individually and in combination.

For the transmission equipment used in connections, the appropriate ITU-T Recommendations should be applied whenever they are applicable.

Other relevant Recommendations are:

G.171 on transmission plan aspects of privately operated networks;

G.172 on conference networks;

G.173 on transmission planning aspects of the speech service in digital public land mobile networks;

G.174 on transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN.

2 References

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

- [1] ITU-T Recommendation G.100 (1993), *Definitions used in Recommendations on general characteristics of international telephone connections and circuits.*
- [2] ITU-T Recommendation G.111 (1993), *Loudness Ratings (LRs) in an international connection.*
- [3] ITU-T Recommendation G.113 (1996), *Transmission impairments.*
- [4] ITU-T Recommendation G.114 (1996), *One-way transmission time.*
- [5] ITU-T Recommendation G.121 (1993), *Loudness Ratings (LRs) of national systems.*
- [6] ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections.*
- [7] ITU-T Recommendation G.126 (1993), *Listener echo in telephone networks.*
- [8] CCITT Recommendation G.131 (1988), *Stability and echo.*
- [9] CCITT Recommendation G.171 (1988), *Transmission plan aspects of privately operated networks.*
- [10] CCITT Recommendation G.172 (1988), *Transmission plan aspects of international conference calls.*
- [11] ITU-T Recommendation G.173 (1993), *Transmission planning aspects of the speech service in digital public land mobile networks.*
- [12] ITU-T Recommendation G.174 (1994), *Transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN.*
- [13] CCITT Recommendation G.223 (1984), *Assumptions for the calculation of noise on hypothetical reference circuits for telephony.*
- [14] CCITT Recommendation G.711 (1988), *Pulse Code Modulation (PCM) of voice frequencies.*
- [15] CCITT Recommendation G.712 (1992), *Transmission performance characteristics of pulse code modulation.*
- [16] CCITT Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM).*
- [17] CCITT Recommendation G.727 (1990), *5-, 4-, 3- and 2-bits/sample embedded adaptive differential pulse code modulation (ADPCM).*
- [18] CCITT Recommendation G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction.*
- [19] ITU-T Recommendation G.763 (1994), *Digital circuit multiplication equipment using ADPCM (Recommendation G.726) and digital speech interpolation.*
- [20] CCITT Recommendation G.765 (1992), *Packet circuit multiplication equipment.*
- [21] CCITT Recommendation H.34 (1984), *Subdivision of the frequency band of a telephone-type circuit between telegraphy and other services.*
- [22] CCITT Recommendation M.120 (1988), *Access points for maintenance.*
- [23] CCITT Recommendation M.160 (1988), *Stability of transmission.*
- [24] CCITT Recommendation M.560 (1988), *International telephone circuits – Principles, definitions and relative transmission levels.*

- [25] CCITT Recommendation M.565 (1988), *Access points for international telephone circuits*.
- [26] CCITT Recommendation O.6 (1988), *1020 Hz reference test frequency*.
- [27] CCITT Recommendation O.22 (1992), *CCITT automatic transmission measuring and signalling testing equipment ATME No.2*.
- [28] ITU-T Recommendation P.10 (1993), *Vocabulary of terms on telephone transmission quality and telephone sets*.
- [29] ITU-T Recommendation P.11 (1993), *Effect of transmission impairments*.
- [30] CCITT Recommendation P.55 (1968), *Apparatus for the measurement of impulsive noise*.
- [31] ITU-T Recommendation P.56 (1993), *Objective measurement of active speech level*.
- [32] ITU-T Recommendation P.310 (1996), *Transmission characteristics for telephone-band (300-3400 Hz) digital telephones*.
- [33] CCITT Recommendation Q.45 (1984), *Transmission characteristics of an analogue international exchange*.
- [34] ITU-T Recommendation Q.551 (1994), *Transmission characteristics of digital exchanges*.
- [35] ITU-T Recommendation Q.552 (1994), *Transmission characteristics at 2-wire analogue interfaces of digital exchanges*.
- [36] ITU-T Recommendation Q.554 (1994), *Transmission characteristics at digital interfaces of digital exchanges*.
- [37] CCITT Recommendation V.2 (1980), *Power levels for data transmission over telephone lines*.
- [38] ITU-T Supplement No.3 to the P-Series Recommendations (1993), *Models for predicting transmission quality from objective measurements*.
- [39] ETSI Technical Report VTQM-E: (1994), *Transmission and multiplexing – Speech communication quality from mouth to ear of 3.1 kHz handset telephony across networks*.

3 Definitions and conventions

For the purposes of this Recommendations, the definitions in 3.1 apply.

3.1 Circuits and connections

3.1.1 telephone circuit: In transmission planning, and in the G-Series Recommendations, a telephone circuit denotes a telecommunication circuit with associated terminating equipment, directly connecting two switching devices or exchanges, in line with Note 2 to the general definition of a circuit (see 1.4/G.100). For simplicity, the term “circuit” is often used instead of “telephone circuit” in the G-Series Recommendations.

NOTES

1 Conceptually, (telephone) circuits are those parts of the connections that remain intact and permanently associated with the switches at each end, after a connection is taken down and before a new connection is established. Routine measurements of (telephone) circuits are made in a way approaching the ideal concept as closely as possible, i.e. between circuit access points which between them will include as much of the (telephone) circuit as possible (see 2.1.2./M.565).

2 In some cases, mainly in private networks, the definition of circuit is not applicable. Exchanges within a private network are normally interconnected via leased lines, specified at the interfaces of the transmission systems.

3.1.2 subscriber’s (telephone) line; subscriber’s loop (in telephony): A link between a public switching entity and a telephone station or a private telephone installation or another terminal using signals compatible with the telephone network. (See also Recommendation P.10, term 31.04.)

NOTE – In French, the term “ligne de réseau” is used only when the private telephone installation is a private branch exchange or an internal telephone system.

3.1.3 local (telephone) system; local (telephone) circuit: The combination of subscriber's station, subscriber's line and feeding bridge if present, see Figure 1. (See also Recommendation P.10, term 31.02.)

NOTES

- 1 This term is used in the context of transmission planning and performance.
- 2 In ITU-T English texts, the term "local (telephone) system" is preferred.
- 3 A local network includes the local system, the local exchanges and interconnecting circuits.

3.1.4 subscriber system (in transmission planning): A subscriber's line associated with that part of the private telephone installation connected to this line during a telephone call, see Figure 1 (see also Recommendation P.10, term 31.03).

NOTE – This term is used in the context of transmission planning and performance.

3.1.5 subscriber circuit: The circuit between the local exchange and the Network Connection Point (NCP), i.e. the interface between the public network and the subscriber's installation, see Figure 1. This interface may for instance be at the MDF of a PBX, at a socket for connecting a telephone set, etc. The location of this interface is dependent on national regulations and practice.

NOTE – In the local exchange, the subscriber circuit usually includes "half" of the exchange in an analogue exchange and in a digital exchange the input and output of the circuit usually will be a digital bit stream corresponding to the "exchange test points" defined in 1.2.1.1/Q.551.

3.1.6 telephone circuit loss: This is the composite loss at the reference frequency 1020 Hz between the circuit input and its output, as defined in Note 1 below. This will include any loss in the associated terminating equipment of the switching centres.

NOTES

1 Defined for transmission planning purposes, the input and output of a circuit are hypothetical points in an exchange where circuits are directly interconnected (see 2.3.3/M.560) and are consequently not accessible, e.g. for measurement purposes. To enable the necessary correlation to be made between planning and measured values, "circuit access points" are defined in Recommendation M.565; their relation to the circuit input and output are shown in Figures 1a and 1b/M.565 for analogue and digital exchanges, respectively. After carrying out the measurement between these points, any necessary correction is made for the effect of circuit access arrangements to allow circuit loss to be determined (see 3.1.2/O.22).

2 For digital exchanges it will be seen that the circuit input and output correspond to the "exchange test points" as defined in 1.2.1.1/Q.551. Since the levels at these points are defined in terms of the digital bit streams appearing there, neither digital access arrangements nor passage through the digital switchblock will involve any loss or gain, provided the bit sequence is not affected. On the other hand any recoding, for example such as produced by a "digital pad", will be included in the circuit loss. To allow at least the mandatory alternative of "bit-transparent" connections (i.e. retaining bit integrity, see 3.1.2/Q.554), the "pad" function must be switchable, i.e. it must be possible:

- a) to make measurements under conditions which simulate at will each real traffic condition requiring a different pad value;
- b) to check the bit error ratio (see 3.1.1/Q.554), which of course needs to be done in the absence of intentional changes in the bit stream.

3 For analogue exchanges the assumption is made that nominal switchblock losses (defined in 3.2/Q.45) are divided equally between the two circuits being interconnected in the exchange. The variance of switchblock losses contributes negligibly to the variance of circuit loss by comparison with the objective for loss variations in transmission systems (see 1.1.2/M.160).

4 The circuit access points are not to be confused with the "line access points", usually located at a distribution frame (see Recommendation M.120, last paragraph). These points are not of interest for transmission planning, but only for the maintenance services for line-up and fault localisation purposes.

5 The input and output of international circuits are defined as the Virtual International Connecting Points having defined relative levels (see 3.10.1.3). This is necessary to have a defined boundary between the national and international parts of a connection.

3.1.7 connection: A chain of circuits interconnected by switching points, between two different points in the network.

In transmission planning the loss of a connection is normally the sum of the losses of the circuits making up the connection. (The losses of the switching centres are normally included in the circuit losses.)

NOTES

- 1 A complete connection is a connection between two terminal equipment connected to the network.
- 2 When analogue or mixed analogue/digital circuits are interconnected in the exchanges, "level jumps" often have to be introduced. In a complete connection, the sum of all "level jumps" and digital losses should not exceed 6 dB in the short term and 3 dB in the long term.

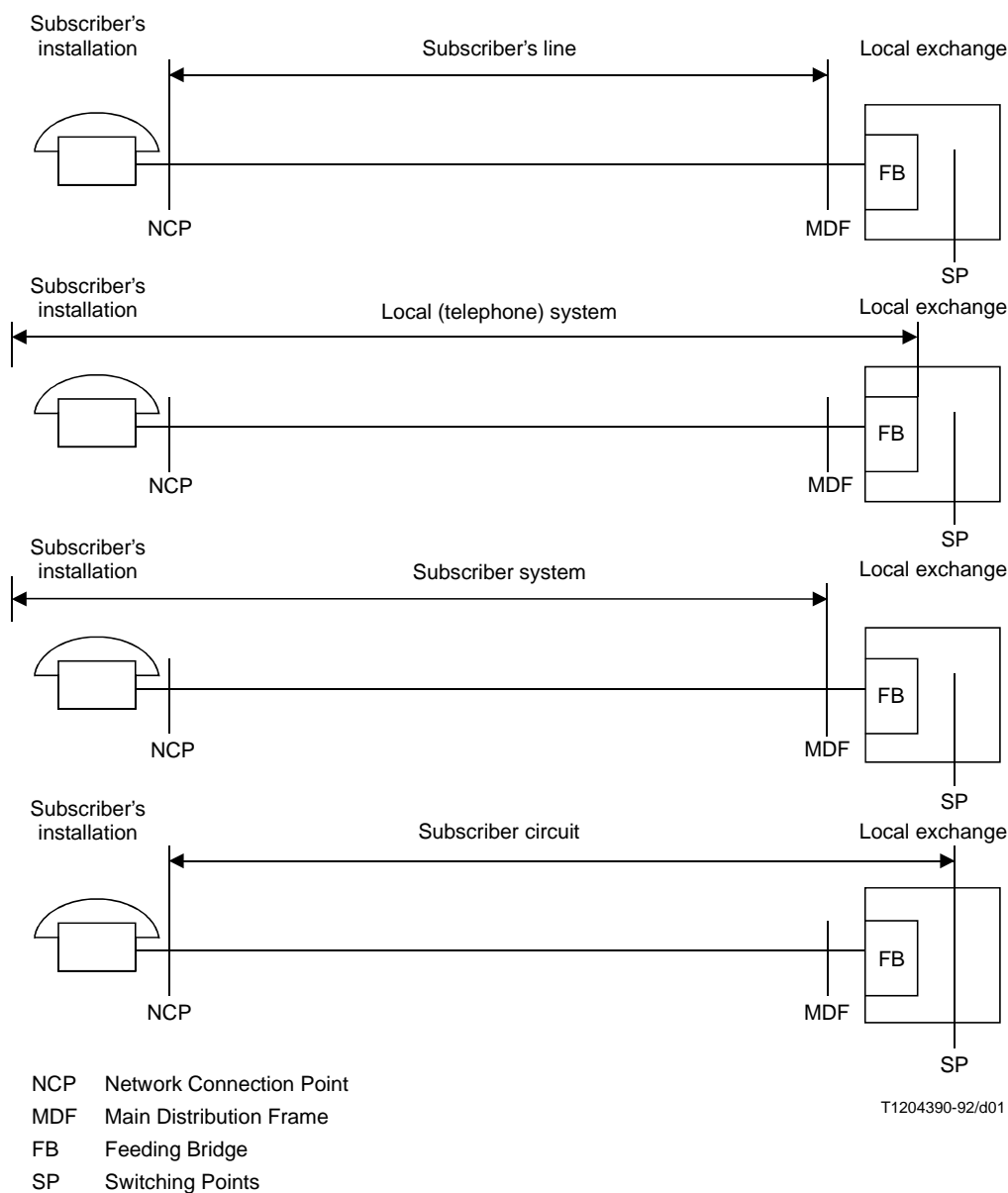


FIGURE 1/G.101

Subscriber's line, local (telephone) system, subscriber system and subscriber circuit

3.2 Transmission Reference Point (TRP)

A hypothetical point used as the zero relative level point to define the concept of relative levels. When specifying and measuring equipment, transmission systems, exchanges and PBXs, etc., the term "Level Reference Point (LRP)" is often used instead of transmission reference point

3.3 Relative (power) level

The relative level at a point on a circuit is given by the expression $10 \log_{10} (P/P_0)$ dBr, where P represents the apparent power of a sinusoidal test signal at the reference frequency 1020 Hz at the point concerned and P_0 the apparent power of that signal at the transmission reference point. This is numerically equal to the composite gain between the transmission

reference point and the point concerned (or the composite loss between the point concerned and the transmission reference point), for the reference frequency 1020 Hz. For example, if a 1020 Hz signal having a level of x dBm is injected at a point in the circuit and the level measured at the transmission reference point is 0 dBm, the relative level at the point is x dBr. If y dBm is measured at another point in the circuit, the relative level at that point is y dBr.

NOTES

1 The definition above is generally applicable to all systems, e.g. digital exchanges, transmission systems and other types of switching and transmission equipment. It should be noted that these network components often have specified relative levels at their interfaces. These relative levels may differ from the relative levels of a circuit at the same interfaces.

2 In transmission planning, each circuit will have its own specific transmission reference point.

3 The nominal reference frequency of 1020 Hz is in accordance with Recommendation O.6. For existing wholly analogue circuits, one may continue to use a reference frequency of 800 Hz.

4 The relative levels at particular points in a transmission system (e.g. input and output of distribution frames or of equipment like channel translators) are fixed by convention, either by Recommendations or by agreement between manufacturers and users.

5 In real life, the relative levels of different points in a circuit will be determined based on the fixed relative levels at the input and output of transmission systems or digital exchanges.

6 The applications of relative levels and related concepts are described in Annex A.

3.4 dBm0

At the reference frequency (1020 Hz), L dBm0 represents an absolute power level of L dBm measured at the transmission reference point (0 dBr point), and a level of $L + x$ dBm measured at a point having a relative level of x dBr.

The voltage of an 0 dBm0 tone at any voiceband frequency at a point of x dBr is given by the expression:

$$V = \sqrt{10^{x/10} \cdot 1 \text{ [W]} \cdot 10^{-3} \cdot |Z_{1020}|} \quad \text{volts}$$

where $|Z_{1020}|$ is the modulus of the nominal impedance, Z , at the point at the reference frequency 1020 Hz. Z may be resistive or complex.

3.5 Power handling capacity

FDM transmission systems (excepting the cases mentioned in Note 2) are designed for a nominal mean power during the busy hour of -15 dBm0 per channel. This is the mean with time and the mean of a large bunch of circuits. (See Recommendation G.223.) This corresponds to a mean active speech level (pauses in the speech not included) of -11 dBm0. This relation does not take account of the transmission of non-voice services. Many clauses for such services are based on the assumption of a limit of -13 dBm0 on the 1-minute mean power (see for instance 2.3 i)/V.2 for modem data; clause 1/H.34 and clause 5/H.34 for subdivided telephone-type channels, etc.). A generalised Recommendation in this sense is under study.

In PCM encoding/decoding processes one has a maximum level T_{max} of 3.14 dBm0 for A-law and 3.17 dBm0 for μ -law (see Recommendation G.711). In each channel, sinusoidal signals having levels above T_{max} will be clipped.

NOTES

1 The relative levels of the circuit should be chosen to give the best possible loading of transmission systems, i.e. by control of the mean speech level in FDM systems and the peak level in PCM systems.

2 Normal FDM systems are designed to cope with a mean active speech level of -11 dBm0, calculated on the assumption that the activity factor is 0.25. For certain systems, e.g. submarine cable systems and DCME systems, other values apply.

3.6 Relation between send loudness ratings and relative levels

The relationship between the 0 dBr point and the level of T_{max} in PCM encoding/decoding processes standardised by the ITU-T is set forth in Recommendation G.711. In particular, if the minimum nominal Send Loudness Rating (SLR) of

local systems referred to a point of 0 dBr of a PCM encoder is not less than +2 dB and the value of T_{max} of the process is set at +3 dBm0 (more accurately 3.14 dBm0 for A-law and 3.17 for μ -law), then in accordance with clause 3/G.121, the peak power of the speech will be suitably controlled.

An average talker produces, at a (network) 0 dBr point, a mean active speech level of:

$$L = -11 - SLR \quad (\text{dBm0})$$

where SLR is referred to the 0 dBr point in question and the speech level is measured with an instrument designed in accordance with Recommendation P.56.

Moreover, taking account of the variation among talkers and their handset handling, the standard deviation of the mean active speech level at a 0 dBr point is found to be about 5 or 6 dB.

3.7 Determination of relative level

Figure 2 illustrates the principle of how the relative level at the input and output analogue points of a “real” codec can be determined.

When using Figure 2 to determine the relative levels of a “real” codec with non-resistive impedances at the analogue input and output ports, the following precautions must be observed:

- i) the test frequency should be 1020 (+2, -7) Hz, see Recommendation O.6;
- ii) the power at points s and r is expressed as apparent power, i.e.

$$\text{Apparent power level} = 10 \log_{10} \left[\frac{(\text{Voltage at point})^2 \times 10^3}{(\text{Modulus of nominal impedance at 1020 Hz}) (1 \text{ W})} \right] \quad \text{dBm}$$

- iii) point r is terminated with the nominal design impedance of the decoder to avoid significant impedance mismatch errors.

NOTE – Precautions ii), iii) above are, of course, equally applicable to the case of resistive input and output impedances and would generally be observed by conventional test procedures. Standardising the reference frequency as in i) above is, however, essential for complex impedances because of the variation of nominal impedance with the test frequency.

3.8 Relative level of a point in a digital link

The relative level to be associated with a point in a digital path carrying a digital bit stream generated by a coder lined-up in accordance with the principles of 3.7 above is determined by the value of the digital loss or gain between the output of the coder and the point considered. If there is no such loss or gain the relative level at the point considered is, by convention, said to be 0 dBr.

For the application of digital loss or digital gain in telephone circuits, it is possible to discern the four basic cases pointed out in Figure 3. In the cases shown it is understood that the points denoted with 0 dBr (**bold print**) are defined by the network transmission plan. All the other relative levels in the digital path before or behind the digital pad/amplifier are derived from the aforementioned assumption.

Taking the theoretical assumption that a real signal in part A of the transmission path utilises the complete dynamic range of the PCM process according to Recommendation G.711, in part B of the transmission path:

- the dynamic range will be reduced by x dB in case 1 as well as in case 2;
- clipping effects will appear for signals with levels down to x dB below the overload limit of part A in case 3 as well as in case 4.

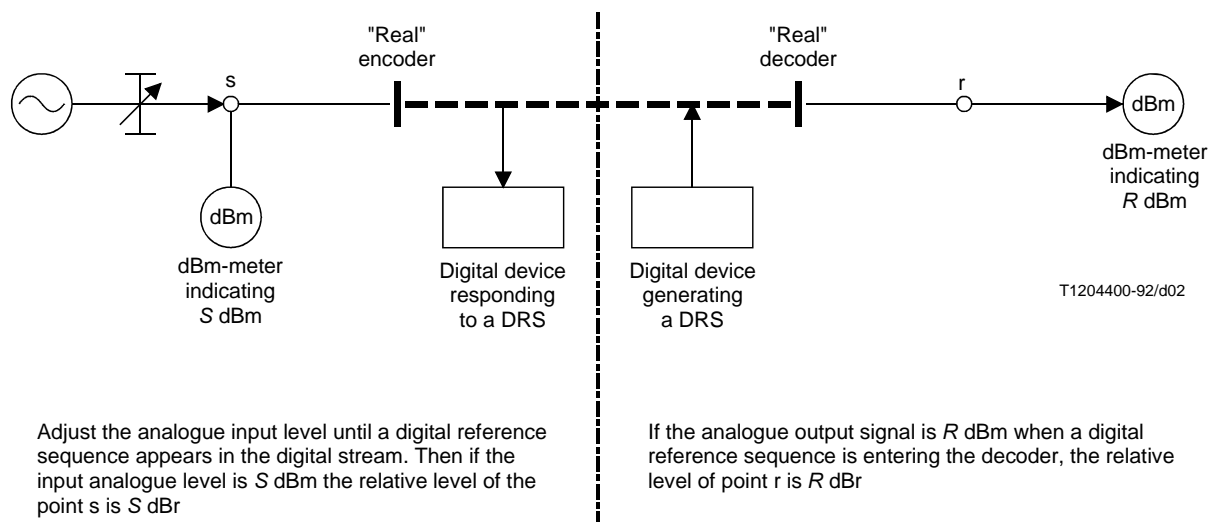


FIGURE 2/G.101

Set-up for determining the relative level at the input and output analogue points of a "real" codec using digital reference sequence

Case 1	
Case 2	
Case 3	
Case 4	

NOTE – In general, Case 1 and Case 4 are to be preferred.

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FIGURE 3/G.101

Relative values in a digital path

Therefore the occurring real signals (voice, tones, DTMF, etc.) must be observed carefully in respect to their real dynamic range in order to avoid overload. The nominal value of x of the digital gain or loss pad should be limited to a narrow range. In all four cases an additional quantizing distortion will arise.

When measuring transmission parameters (e.g. total distortion, variation of gain with input level), which usually are measured over a wide range of input levels, the input level applied to part A of the transmission path must be restricted in order to avoid improper levels at part B of the transmission path.

3.9 PCM Digital Reference Sequence (DRS)

3.9.1 Definition

A **PCM digital reference sequence** is one of the set of possible PCM code sequences that, when decoded by an ideal decoder, produces an analogue sinusoidal signal at the reference frequency (i.e. 1020 Hz) at a level of 0 dBm0.

Conversely an analogue sinusoidal signal at 0 dBm0 at the reference frequency applied to the input of an ideal coder will generate a PCM digital reference sequence.

NOTES

1 Ideal coders and decoders are assumed to show a relation between analogue and digital signals and vice versa exactly in accordance with the appropriate tables for A-law or μ -law of Recommendation G.711. "Real" coders and decoders are assumed to be such that the performance characteristics of an encoder/decoder pair between audio frequency ports will meet the requirements of Recommendation G.712 (see Recommendation P.310).

2 The digital reference sequence above is a theoretical concept used to describe the conversion between analogue and digital signals in connection with transmission planning. For practical measurements other Digital Test Sequences (DTS) are used, e.g. as described in Recommendation P.310.

3.9.2 Use of the Digital Reference Sequence (DRS)

In studying circuits and connections in mixed analogue/digital networks, use of the digital reference sequence can be helpful. For example, Figure 4 shows the various level relationships that one obtains (conceptually) on a Type 2 (as defined in Figure 14) international circuit, where one end terminates at a digital exchange and the other end at an analogue exchange. In the example of Figure 4, the analogue portion is assumed to require a loss of 0.5 dB and that provision for this loss is made by introducing a 1.0 dB pad (0.5 dB for each direction of transmission) in the receive direction at the analogue exchange. This has been deliberately chosen to illustrate the utility of the concept of a digital reference sequence.

Figure 4 gives an example where all the analogue loss is introduced in the output direction at the analogue exchange. In this case the relative levels at the various codecs can be derived from either the DRS or the transmission reference point at the input of the international circuit with no ambiguity.

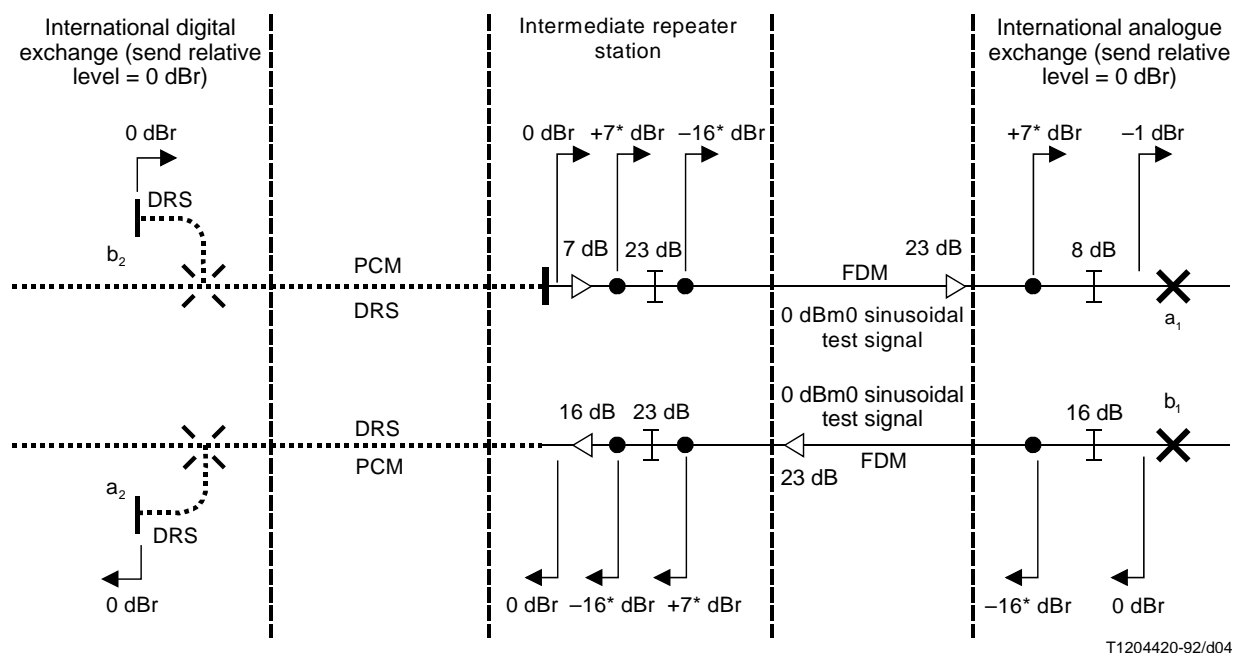
If, however, in Figure 4 the analogue circuit section is lined up so as to give an overall loss in the direction $b_1 - a_2$, care must be taken in the use of the DRS. In this case the 0 dBm0 sinusoidal reference signal and DRS may result in different levels at the point a_2 . Account should be taken of this effect when designing lining-up procedures for mixed analogue/digital circuits.

As a general principle, the relative levels on a mixed analogue/digital circuit should be referred to the transmission reference point at the input of the circuit. Where this is at a digital exchange, it will correspond to the "exchange test points" as defined in 1.2.1.1.2/Q.551.

3.10 Subdivision of telephone networks with regard to the interfaces between network operators

3.10.0 General

In the past, most often telephone users had national calls handled by their national Administration. For international calls only, more than one network operator was involved in the connections. This was reflected in the ITU-T documents by the division of a complete international connection into "national systems" and "the international chain". Subclause 3.10.1 describes this methodology, including the conventions and precautions to be observed.



DRS Digital Reference Sequence

PCM PCM channel

FDM FDM channel

* one of the set of VF relative levels cited in Recommendation G.232

—●— for the purpose of illustration
Multiplex VF input/output point

Transmission loss: $b_2 - a_1 = 1.0 \text{ dB}$

$$b_1 - a_1 = 0 \text{ dB}$$

NOTE – For the meaning of the other symbols, see legend for Figure 7.

FIGURE 4/G.101

Use of digital reference sequence in the design and line-up of a type-2 (as defined in Figure 14) international circuit

Although this situation still exists in many cases today, with the advent of deregulation many users will now have the choice between different network operators for long-distance calls and even for local calls. The tendency is toward a future with a multiplicity of interconnected operator networks. In such cases it is of course more relevant to use other designations for the constituent parts of a connection. Subclause 3.10.2 therefore introduces the terms “terminating network” and “transit networks”. (Note, however, that many of the conventions and guidelines presented in 3.10.1 also are applicable here.)

To ensure the overall (end-to-end) voice transmission quality, agreement has to be reached between the operators involved in a connection with regard to critical transmission parameters, always keeping in mind the telephone users' actual expectations and needs. For the multi-operator situations covered by 3.10.2, fewer rules of a general character can be given than for the “national network case” described in 3.10.1. However, a discussion of this is outside the scope of this Recommendation.

Note that technical details of networks are touched upon in clause 4, however, only in the form of presenting some typical examples of network components and configurations with comments on which types of transmission impairments they are likely to cause.

3.10.1 The national systems and the international chain of circuits

3.10.1.1 Definition of constituent parts

A complete international telephone connection consists of three parts, as shown in Figure 5. The division between these parts is determined by the Virtual International Connecting Points (VICPs) in the originating/terminating International Switching Centres (ISCs). These are theoretical points with specified relative levels (see 3.10.1.3 and 3.10.1.4).

The three parts of the connection are:

- Two national systems, one at each end. These may comprise of one or more 4-wire national trunk circuits with 4-wire interconnection, as well as circuits with 2-wire connection up to the local exchanges and the subscriber sets with their subscriber lines.
- An international chain made up of one or more 4-wire international circuits. These are interconnected on a 4-wire basis in the international centres which provide for transit traffic and are also connected on a 4-wire basis to national systems in the international centres.

An international 4-wire circuit is delimited by its Virtual International Connecting Points in an International Switching Centre.

NOTE – The Virtual International Connecting Points may not be the same as the points at which the circuit terminates physically in the switching equipment. These latter points are known as the circuit terminals; the exact position of these terminals is decided in each case by the Administration concerned.

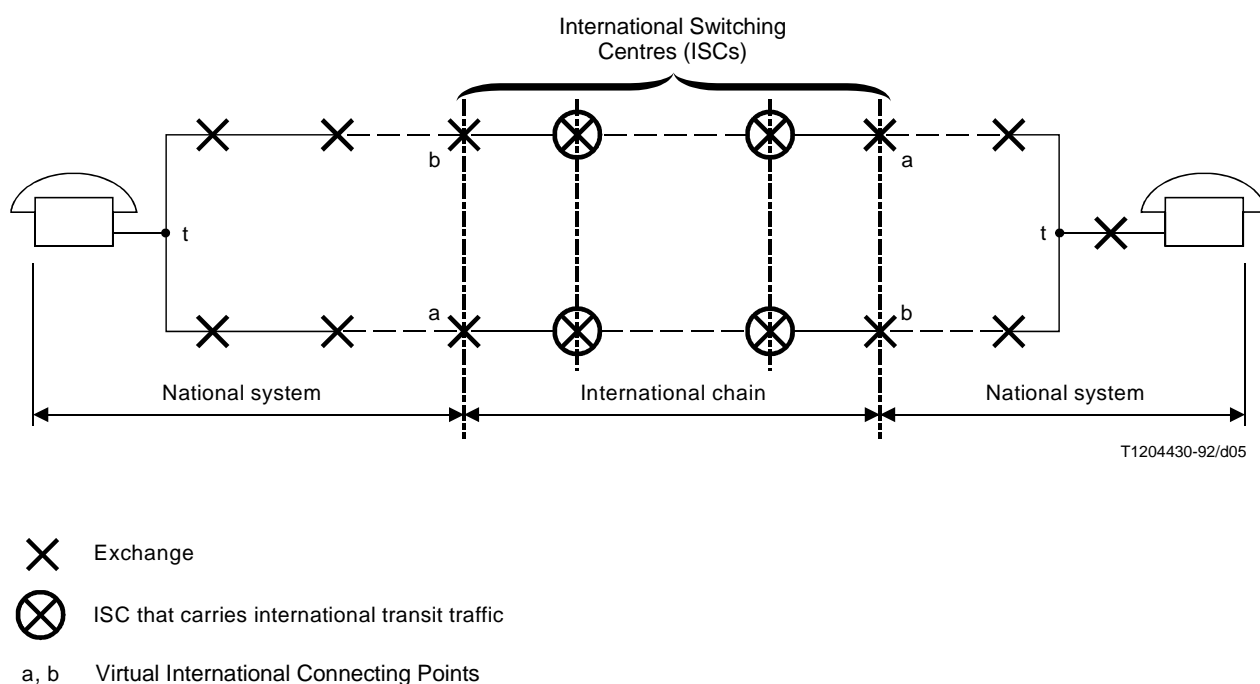


FIGURE 5/G.101

Definition of the constituent parts of an international connection

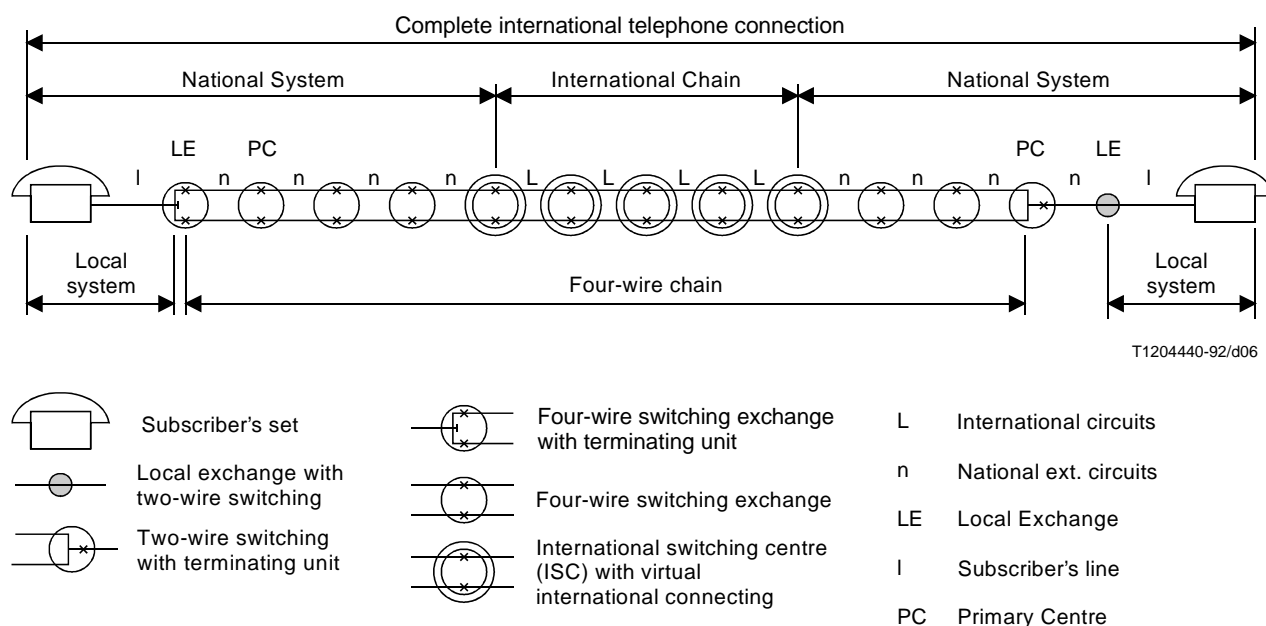
3.10.1.2 The 4-wire chain

The 4-wire chain (see Figure 6) denotes the whole unbroken chain of 4-wire national and international circuits in a complete telephone connection, including possible 4-wire circuits between the primary centre and the local exchange and on the subscriber line, e.g. ISDN access and 4-wire or digitally connected PBXs.

3.10.1.3 Virtual International Connecting Points (VICPs)

The Virtual International Connecting Points define the boundary between the national and international parts of a connection; see Figure 5. The international connecting points are also used as reference points for transmission quantities recommended for the national and international part of a connection.

NOTE – Earlier the terms “virtual switching points” and “virtual analogue switching points” were used to define the boundary between the national and international part of a connection. These points, however, were assigned other relative levels.



NOTE – The arrangements shown for the national systems are examples only. In many cases, the local exchange LE (shown as analogue in the figure) is digital.

FIGURE 6/G.101

An international connection to illustrate the nomenclature adopted

3.10.1.4 Relative levels specified in the Virtual International Connecting Points

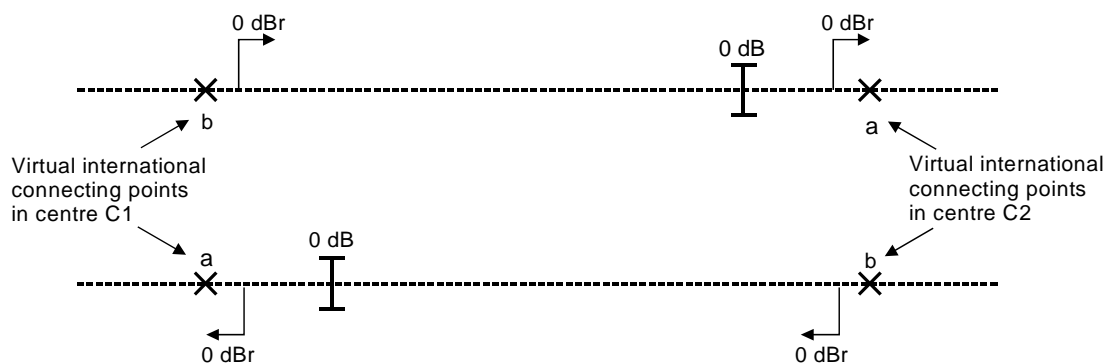
The Virtual International Connecting Points of an international 4-wire telephone circuit are by convention fixed to points in the circuit where the nominal relative levels are:

- sending: 0 dBr;
- receiving: 0 dBr for digital circuits or the very short circuits mentioned in Note 4;
–0.5 dBr for analogue and mixed analogue digital circuits.

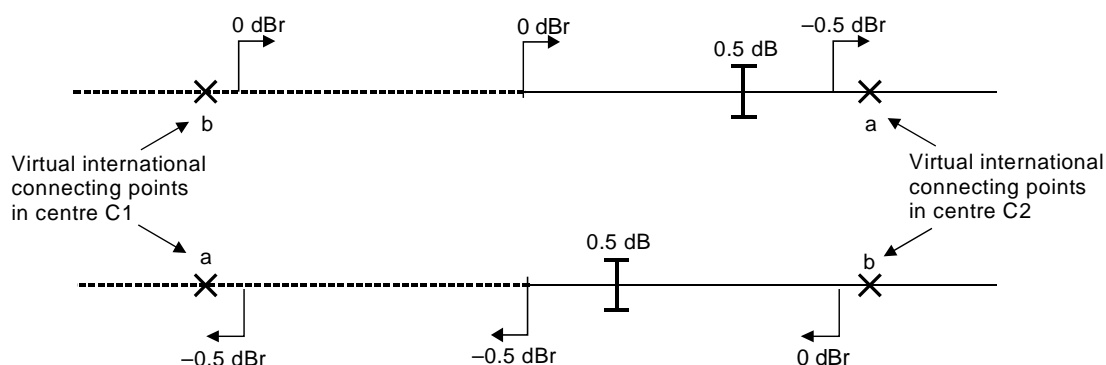
The nominal transmission loss of the international circuits is 0 dB for digital circuits and 0.5 dB for analogue and mixed analogue/digital circuits; see Figure 7.

NOTES

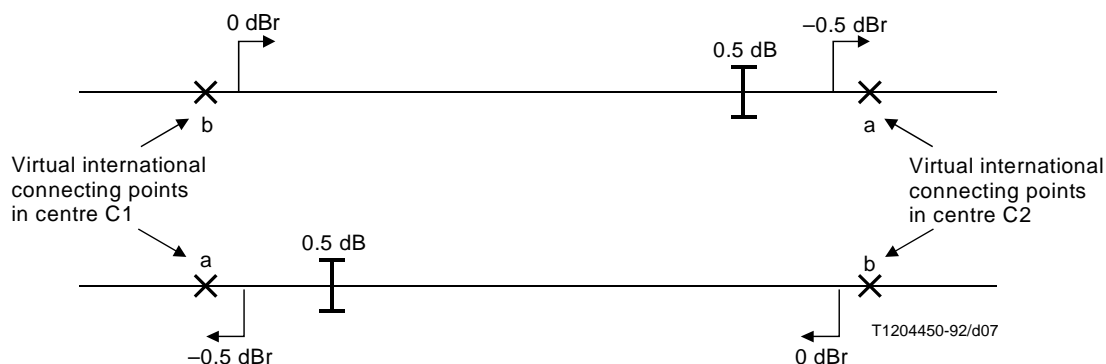
- 1 Usually a 0.5 dB loss has to be introduced in the mixed analogue/digital circuit to satisfy the stability requirements.
- 2 The “virtual analogue switching points” used earlier had the relative levels:
 - sending: –3.5 dBr;
 - receiving: –3.5 dBr for digital circuits or the very short circuits mentioned in Note 4;
–4 dBr for analogue and mixed analogue/digital circuits.
- 3 The Virtual International Connecting Points in digital exchanges referred to a digital bit stream, e.g. the exchange test points. In analogue exchanges they often will not be accessible, and will differ from the switching levels nationally used in the ISC.
- 4 If a 4-wire analogue circuit forming part of the 4-wire chain contributes negligible delay and variation of transmission loss with time, it may be operated at zero nominal transmission loss between Virtual International Connecting Points. This relaxation refers particularly to short 4-wire tie-circuits between switching centres – e.g. circuits between two international switching centres in the same city.



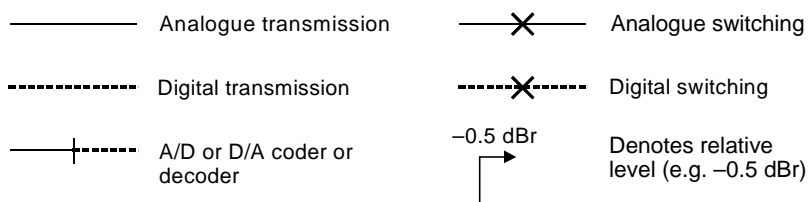
a) Definition of Virtual International Connecting Points for a digital international circuit between digital international centres



b) Definition of Virtual International Connecting Points for a mixed analogue/digital international circuit between an analogue and a digital international centre



c) Definition of Virtual International Connecting Points for an analogue international circuit between analogue international centres



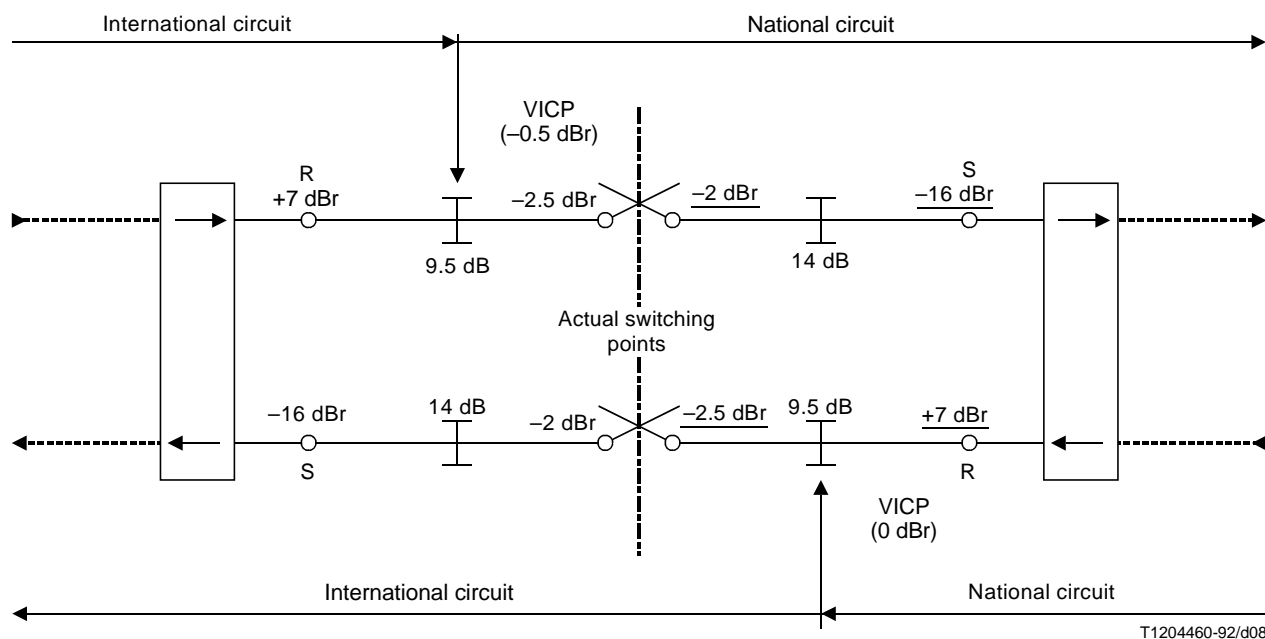
NOTE – The relative level at a point in a digital link is determined using ideal decoders as described in 3.8.

FIGURE 7/G.101
Definitions for international circuits

3.10.1.5 Circuit access point

The ITU-T has defined circuit access points as being “4-wire access points so located that as much as possible of the international circuit is included between corresponding pairs of these access points at the two centres concerned” (see Recommendation M.565). These points, and their relative level (with reference to the transmission reference point), are determined in each case by the Administration concerned. They are taken as the basic reference points of known relative level to which other transmission measurements will be related. In other words, for measurement and lining-up purposes, the relative level at the appropriate circuit access point is the relative level with respect to which other levels are adjusted.

An example showing an actual arrangement is shown in Figure 8.



NOTES

- 1 Underlined values of relative levels refer to the national circuit. Values of relative levels not underlined refer to the international circuit. In an actual switching centre the international connecting points may not physically exist. As shown in this figure, the VICP is situated inside a 9.5 dB pad.
- 2 Each of the 9.5 and 14 dB pads include half the exchange loss.
- 3 In this example, the national circuit has 0.5 dB loss, giving a 0.5 dB “level jump” in the switch at the input to the international circuit.

FIGURE 8/G.101

**Example showing a simplified representation of a transit connection
in an international switching centre**

3.10.1.6 Measurement frequency

For all international circuits 1020 Hz is the recommended frequency for single-frequency maintenance measurements. However, by agreement between the Administrations concerned, 800 Hz may be used for such measurements on wholly analogue circuits.

NOTE – The frequency should be within 1020 (+2/–7) Hz; see Recommendation O.6.

3.10.2 Multi-operator networks

For multi-operator networks (which implies deregulation) the definition of the constituent parts are “terminating networks” and “transit networks” which all are managed by separate operators. Thus this terminology reflects the *division of responsibility* between the operators involved in connections.

As the name implies, the terminals involved in a connection are to be found in the terminating networks. Each call originates and ends in a terminating network, either the same or a different one. The terminating networks can be interconnected directly or by one or more transit networks. A terminating network may contain cross connect facilities for routing calls to different switched networks.

Figure 9 shows an example of a transit network connected to a terminating network.

In modern networks it can be assumed that transit networks are all digital. At the interconnection points, the conventions, definitions and rules for digital circuits given in 3.10.1 apply.

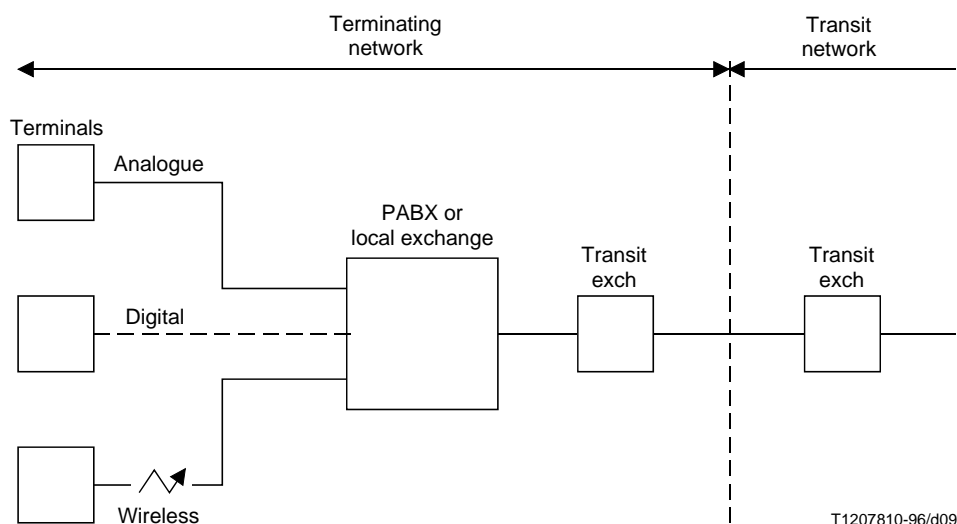


FIGURE 9/G.101

An example showing a simple “terminating network” connect to a “transit network”, the constituent parts of connections in a multi-operator market (The terminating and transit networks are managed by different network operators)

4 Typical network components and configurations

4.1 Network components

From a technical point of view, all the components forming a connection can be divided into three groups, namely Terminal Elements, Switching Elements and Transmission Elements.

Terminal Elements (for speech transmission)

All types of digital and analogue telephones, including mobile and cordless sets.

Switching Elements

All types of switching equipment.

Transmission Elements

All kinds of media, used as links between switching elements and between switching elements and terminal elements. The physical media of these elements can be metallic, fibre-optic or wireless. The signal form is either analogue or digital.

Beside the traditional multiplexing systems FDM and PCM, various other “channel compression” techniques are being increasingly used on long (and/or costly) circuits. These systems employ low bit rate codecs, sometimes with dynamic channel allocation as in DCME and VPE systems. The use of ATM systems for voice transmission is also being studied.

Echo cancellers may also be classified as a type of transmission element when they are deployed in the network. However, some terminals also contain echo cancellers.

4.2 Network configurations

The variety of possible network configurations is almost infinite. Network configurations depend on the type of connection – short, average or long national or international calls – and the type of switching and transmission elements used in the different constituent parts of the connection. Here only a few examples will be shown in order to illustrate some important cases.

Figure 10 depicts a fully analogue routing between two analogue sets. The most critical parameters are the overall loudness rating OLR and, in some instances, the noise, provided that for very long connections echo cancellers are employed. (This case was more common in the past.)

In Figure 11 a digital telephone set is connected to an analogue set via a fully digital route. In addition to the conventional PCM systems, a virtual ATM circuit is included in the digital path. (Such configurations are likely to be fairly common in the near future.) At the analogue end, reflections may occur at the hybrid in the exchange. The main impairment in those configurations is caused by talker echo effects at the digital telephone set, due to the transmission time – here increased by the ATM system – and the signal reflections at the far end hybrid. Note, however, that even when ATM circuits are not used in the digital path, there is a possibility for noticeable talker echo.

Although echo effects can be diminished by echo cancellers, a very long transmission time in itself causes impairments to the speech communication quality. A connection subjected to such a risk is shown in Figure 12, depicting a call from a mobile phone (for instance of the GSM type) via a satellite link. Although not shown in the figure, DCMEs are routinely employed on satellite circuits for voice traffic. This type of equipment also increases delay and may produce a particular kind of distortion, if not properly dimensioned.

A further example of the use of DCME systems is shown in Figure 13, where a cordless telephone is used at one end of the connection. There, the tandeming of the low rate codecs in the cordless set and the DCME decreases the transmission quality.

5 Stability control by loss allocation to circuits

This matter has been touched upon in 3.10.1.4. More detailed information is given in Recommendation G.122. Application of these rules for typical circuits is illustrated in Figure 14.

The Type 1 circuit in Figure 14 a) represents the case where digital transmission is used for the entire length of the circuit and digital switching is used at both ends. Such a circuit can generally be operated at a nominal transmission loss of 0 dB as shown because of the transmission properties exhibited by such circuits (e.g. relatively small loss variations with time).

The Type 2 circuit in Figure 14 b) represents the case where the transmission path is established on a digital transmission channel in tandem with an analogue transmission channel. Digital switching is used at the digital end and analogue switching at the analogue end.

It might be possible, in some cases, to operate Type 2 circuits with a nominal loss of 0 dB in each direction of transmission. For example, where the analogue portion could be provided with the necessary gain stability and where the attenuation distortion would permit such operation.

The Type 3 circuit in Figure 14 c) represents the case where the transmission path is established over a tandem arrangement consisting of digital/analogue/digital channels as shown. Digital switching is assumed at both ends.

The Type 4 circuit in Figure 14 d) represents the case where the transmission path is established over a tandem arrangement consisting of analogue/digital/analogue channels as shown. Analogue switching is assumed at both ends.

The Type 5 circuit in Figure 14 e) represents the case where analogue transmission is used for the entire length of the circuit and analogue switching is used at both ends.

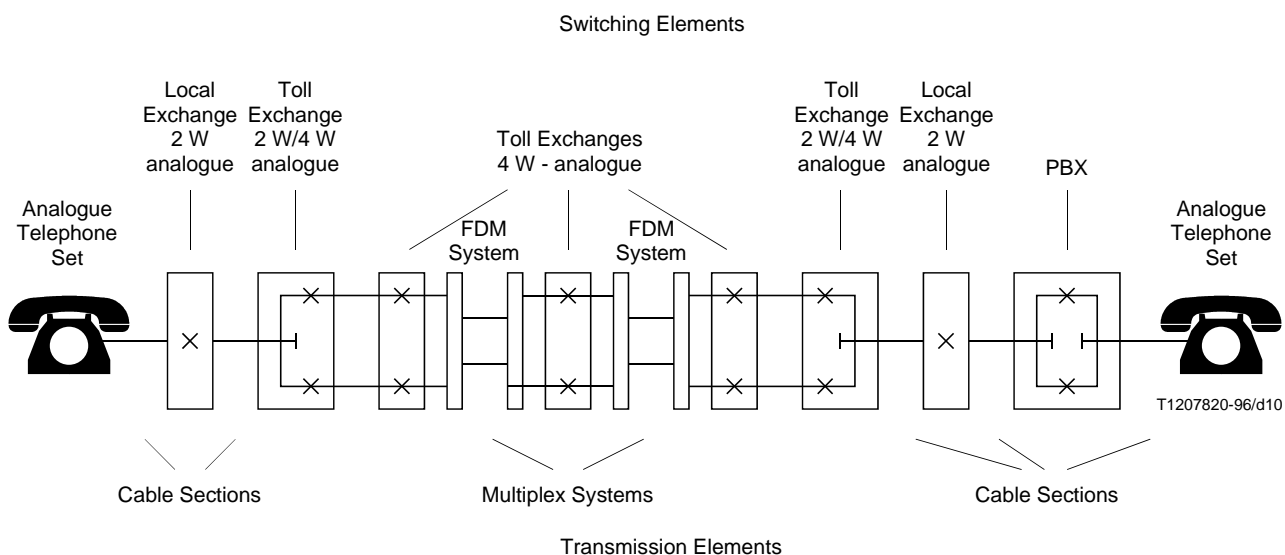


FIGURE 10/G.101
Typical configuration for a fully analogue connection

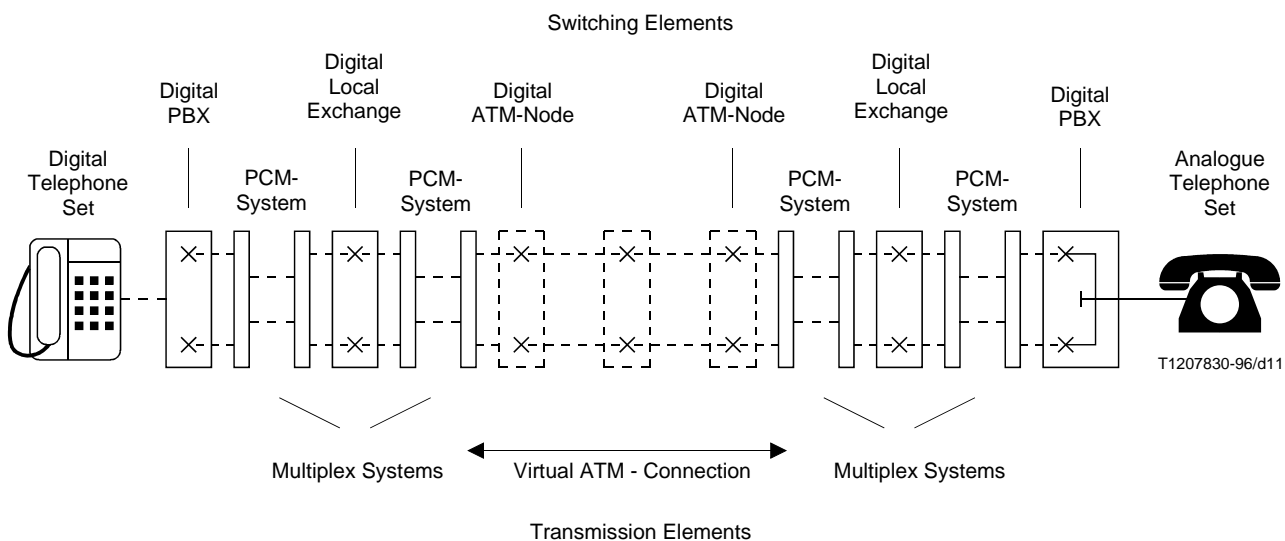


FIGURE 11/G.101
Configuration for a fully digital connection, including ATM,
between a digital and an analogue set

International circuits of this type are usually operated at a loss L , where L is nominally = 0.5 dB between Virtual International Connecting Points.

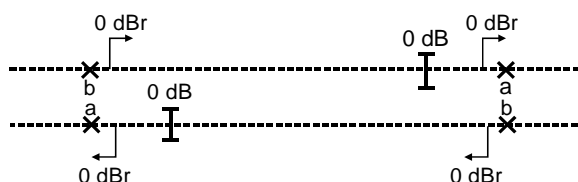
NOTE – General remarks concerning the allocation of losses in the mixed analogue/digital circuits:

In circuit types 2, 3 and 4, the pads needed to control any variability in the analogue circuit sections (arising from loss variations with time or attenuation distortion) are shown in a symmetrical fashion in both directions of transmission. However, in practice, such arrangements may require non-standard levels at the boundaries between circuit sections. Administrations are advised that should they prefer to adopt an asymmetric arrangement, e.g. by putting all the loss into the receive direction at only one end of a circuit (or circuit section); then, provided that the loss is small, e.g. a total of not more than 1 dB, there is no objection on transmission plan grounds.

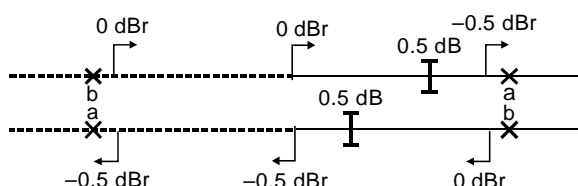
The small amount of asymmetry that results in the international portion of the connection will be acceptable, bearing in mind the small number of international circuits encountered in most actual connections.

As far as national circuits are concerned, Administrations may adopt any arrangements they wish provided that the requirements of 2.2/G.121 are complied with.

In some cases transmultiplexers may be used, in which case the circuits may not be available at audio-frequency at the point at which a pad symbol is used in the diagrams of Figure 14. Should the variability of the analogue portions merit additional loss, the precise way in which this loss can be inserted into the circuits is a matter for Administrations to decide bilaterally.

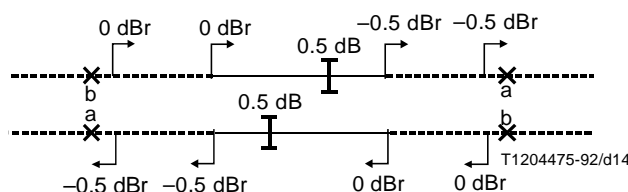


a) Type 1 circuit – All digital circuit with digital switching at both ends



NOTE – The loss is required if the analogue circuit section introduces significant amounts of attenuation distortion or variation with time.

b) Type 2 circuit – Digital/analogue circuit with digital switching at one end and analogue switching at the other end

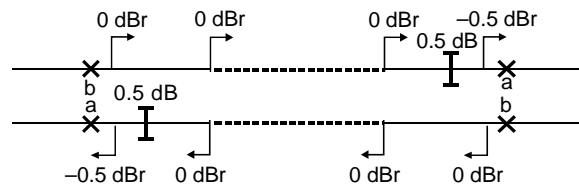


NOTE – The loss is required if the analogue circuit section introduces significant amounts of attenuation distortion or variation with time.

c) Type 3 circuit – Digital/analogue/digital circuit with digital switching at each end

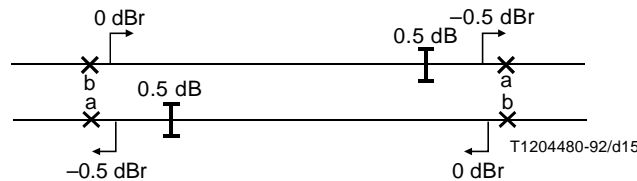
FIGURE 14/G.101 (sheet 1 of 2)

Types of international circuits

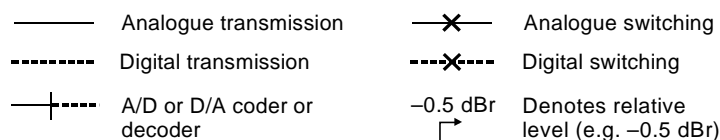


NOTE – The loss is required if the analogue circuit section introduces significant amounts of attenuation distortion or variation with time.

d) Type 4 circuit – Analogue/digital/analogue circuit with analogue switching at each end



e) Type 5 circuit – All analogue circuit with analogue switching at both ends



NOTES

- 1 The pad symbols in the circuits are not intended to imply that real attenuators are needed. They are a convention of transmission planning engineers.
- 2 The relative level at a point in a digital link is determined using ideal decoders as described in 3.8.

FIGURE 14/G.101 (sheet 2 of 2)

Types of international circuits

6 Echo control strategies

Listener echo is usually not a problem in modern networks. (Guidance can be found in Recommendation G.126.)

Rules for evaluation and control of talker echo are given in Recommendation G.131. Note that disturbance from talker echo depends not only on the mean transmission time but also on the Talker Echo Loudness Rating, (TELR). The latter is a function of the sensitivity of the talker's telephone set, the magnitude of the impedance mismatch causing signal reflection and the loss between the set and the point of reflection. One way to diminish talker echo in general is to employ a suitable impedance strategy in the 2-wire parts of the network. This is discussed in Recommendation Q.552.

Note that for echo cancellers to operate with full efficiency, the echo path should have an essentially linear amplitude characteristic.

7 Consideration of total transmission time

Guidance is given in Recommendation G.114.

8 The effect of coding and signal processing in the digital path

The impairments in the form of distortions, which are caused by low bit rate encoding and decoding or other forms of signal processing in the digital path, are described and quantified in Recommendation G.113. Note that digital processing and low bit rate coding in general increase the transmission time.

9 Channel compression techniques

Data for such equipment is given in Recommendations G.763 and G.765.

10 Evaluation of impairments, singly and in combination

This is dealt with in Recommendation G.113.

11 Transmission planning in the context of general network planning

Transmission planning is a subset of the total network planning, both when creating new networks or extending existing ones. Modern networks allow a very great flexibility in routing and “intelligent” switching features. It is important, however, that transmission aspects are not forgotten in the planning process.

For complicated networks one should keep in mind the abilities of the particular signalling that is implemented. Advanced signalling systems could, in addition to performing their normal functions, convey information about certain transmission parameters in connections. (Examples of transmission parameters of interest are accumulated delay, existence of echo cancellers in the path, terminals not needing network echo control, accumulated impairments, choice of particular routes for calls with special requirements for high quality connections, etc.)

Annex A

The concepts of relative levels, dBm0, circuits and connections, and their use in transmission planning

(This annex forms an integral part of this Recommendation)

A.1 Introduction

Relative levels has been a very useful term in transmission planning for the last 30 years and will continue to be so in the future. However, the public switched telephone networks have changed considerably in these years. Especially the introduction of digital exchanges which causes some uncertainty concerning the application of relative levels, and necessitates some changes in the traditional way of applying relative levels. Below, relative levels and associated terms have been explained and examples are shown to clarify these concepts.

A.2 Circuits and connections

The term circuit denotes the direct transmission path between two exchanges, including the associated terminating equipment in the exchanges. In transmission planning the circuit loss includes the exchange loss.

In analogue exchanges this means that “half” the exchange loss at each end of the circuit is included in the circuit loss. Therefore, the input of the circuit is in “the middle of” one exchange and the output of the circuit is “in the middle of” the other exchange. The input and output points of a circuit between analogue exchanges are not accessible points, but hypothetical points used for transmission planning.

In digital exchanges the input of the circuit will usually be a digital bit stream, e.g. at the exchange test points, and the loss in the different terminating equipment, hybrids, etc. are considered to be part of the circuit.

Circuits are linked together in the exchanges, forming connections. A connection is a chain of circuits interconnected by switching points, between different points in the switched network. A complete connection is a connection between two terminal equipment connected to the switched network.

The loss of a connection is the sum of the losses of the circuits making up the connection. (Since the loss of the exchanges is included in the circuits, the switching points have no loss. There is no loss associated with the interconnecting point between two circuits, all loss is within the circuits.)

In some cases mainly in private networks, the definition of circuit is not applicable. Exchanges within a private network are normally interconnected via leased lines, specified at the interfaces of the transmission systems.

A.3 Relative levels

Relative levels are used to describe the signalling handling capacity of transmission systems, exchanges, and other types of equipment. Relative levels are also used to describe the loss between different points within a circuit, transmission system, exchange or another type of equipment.

The relative level at a point is defined as the composite gain between a hypothetical transmission reference point (0 dBr-point) and the point (or as the composite loss from the point to the transmission reference point) at the reference frequency 1020 Hz. As a rule, the transmission reference point is not accessible, but is a purely hypothetical point used to define the concept of relative level. When specifying and measuring transmission systems, exchanges, PBXs, etc., the term “level reference point” is often used instead of transmission reference point.

In real life, the relative levels of different points in a circuit will be determined based on the fixed relative levels at the input and output of transmission systems or digital exchanges. The power handling capacity of these systems are defined, and the difficult task is to find the input relative level of the circuits that will ensure that the best possible loading of the transmission systems and exchanges are obtained.

The levels into the circuit will be determined by the SLR of the telephone sets used, the subscriber line and the loss in the circuits between the local exchange and the input of the circuit.

Traditionally, in transmission planning, each circuit has its own specific transmission reference point and the relative levels within a circuit are restricted only to that circuit and have no meaning outside that circuit. The loss between different points in a circuit may as a rule be found as the difference between the relative levels at the points. To find the loss between points in different circuits, it is necessary to know the transmission plan. (In networks where the circuits have no loss, e.g. digital networks, it is possible to have the same dBr-level at the output of a circuit as the dBr-level at the input of the interconnected circuit. In these special cases, the loss between different points in different circuits may be found directly as the difference in relative level. This means, however, that the transmission plan is known.)

The concept of relative levels is used for different applications, such as:

- 1) transmission planning;
- 2) setting up, lining up and maintenance of circuits;
- 3) specifying and measuring equipment, e.g. transmission systems, digital exchanges and PBXs.

These different applications all use the same basic concept of dBr, defined and described in this Recommendation. However, the different applications make use of the dBr in different ways, which in some cases may cause misunderstandings.

In transmission planning, the different points in the circuit are given dBr-levels to give the optimum performance of the circuit when the input levels and the performance of the different equipment being part of the circuit is taken into consideration. In some cases (especially for digital exchanges), this means that a point may have a different dBr-level when seen as part of the circuit, from what it has been assigned in specifications and test procedures. However, this should not cause problems if it is realized that this is merely because the different dBr-levels are used for different applications.

A.4 Digital pads and the designation of relative levels

When digital pads (or gain) are used within a circuit, the digital bit stream will have a change in the relative level as shown in Figure A.1. Here a 6 dB digital pad is introduced in a digital circuit between two digital exchanges. In this case the digital bit stream will have a level of -6 dBr on the right-hand side of the pad. If a Digital Reference Sequence (DRS) is applied at the right-hand side of the pad, it will give a level of 0 dBr. The use of the DRS should therefore be used with caution as described in 3.9.2.



FIGURE A.1/G.101

A.5 Level jumps

The circuits are interconnected in the exchanges. In the analogue telephone network, where the circuits have to have loss to maintain the stability, this often means that the output of one circuit having a level of A dBr is connected to the input of another circuit having a different level B dBr. This level difference is often called a “level jump”. The “level jump” is the difference in level, i.e. $B - A$ dB. The switching points have no loss, the “level jump” only shows that one goes from one set of dBr's particular to one circuit, to another set of dBr's particular for the other circuit. The loss will always be present within the circuits themselves. (See Example 1 in A.7.)

A.6 Power handling capacity

In FDM transmission systems, it is the total loading resulting from the loading of all the channels in the system that may give rise to distortion because of overloading amplifiers, etc. These systems are therefore designed for a nominal mean power during the busy hour of -15 dBm0 (32 μ W0) (see Recommendation G.223). (This is the mean with time and the mean of a large bunch of circuits.)

The mean power is assumed to consist of:

- 1) signalling and tones having a level of -20 dBm0 (10 μ W0);
- 2) a power level of -16.6 dBm0 (22 μ W0) due to:
 - speech currents, including echoes;
 - carrier leaks;
 - telegraph and phototelegraphy signals.

The contribution from carrier leaks may be as high as -26 dBm0. This gives a mean speech level of -17 dBm0. For speech, an activity factor of 0.25 is assumed. This corresponds to a mean active speech level (pauses not included) of -11 dBm0. (These relations do not account for voiceband data and facsimile transmission. This item is under study.) Note that these level values are to be used for the design of FDM systems only. Actual network speech levels are lower, see 3.6.

PCM coders have a maximum level T_{max} of 3.14 dBm0 for A-law and 3.17 dBm0 for μ -law. This means that sinusoidal signals having r.m.s.-levels exceeding T_{max} will be clipped. This limitation applies to each channel.

In transmission planning, a main task is to ensure that the speech levels entering the transmission systems will not cause overloading of the FDM systems and that the clipping of the speech signals in PCM coders is within acceptable limits. At the same time the speech levels should be as high as possible, to give an acceptable signal-to-noise ratio.

NOTE – Active speech levels may be measured using an instrument in accordance with Recommendation P.56, method B. However, it should be noted that because the instrument has a hang-over-time of 200 ms, pauses shorter than 200 ms will not be registered, resulting in an activity factor larger than 0.25. During the measurements, precautions should be taken to avoid including voiceband data and facsimile transmission.

For measuring the activity factor, instruments having a hang-over-time less than 10 ms should be used.

A.7 Examples

Example 1

Figure A.2 shows an example of a connection consisting of two circuits. The loss of circuit 1 is 1 dB and of circuit 2 is 0.5 dB. The loss of the connection will be 1.5 dB. The loss between points A and B will be -2 dB (2 dB gain), while the difference in relative levels is -3 dB.

In exchange 2, one will have a “level jump” of $-0.5 - (-1.5) = 1$ dB.

The loss between points A and B can be found as:

$$(\text{dBr-level at A}) - (\text{dBr-level at B}) + \text{“level jump”} = -5 - (-2) + 1 = -2 \text{ dB}$$

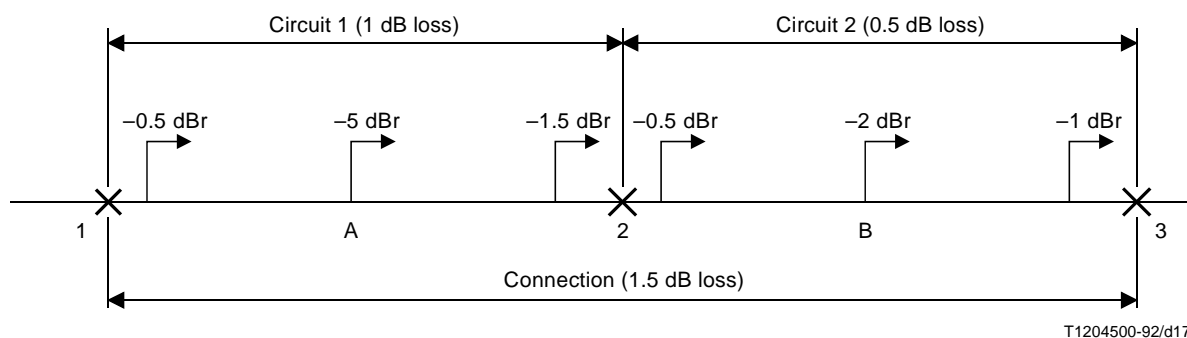


FIGURE A.2/G.101

Example 2

Figure A.3 shows an example where the circuits have no loss, and where they have the same input relative levels. In this exceptional case, the loss between the points A and B will be -3 dB, the same as the difference between the relative levels. It is possible to make a transmission plan where:

- all 4-wire circuits have 0 dB loss;
- all the circuits have the same input relative levels. In this case the whole 4-wire chain of circuits may be considered to have only one signal transmission reference point. It should be noted that this is the case only on the conditions a) and b) mentioned above. In the general case, each circuit will have its own specific transmission reference point.

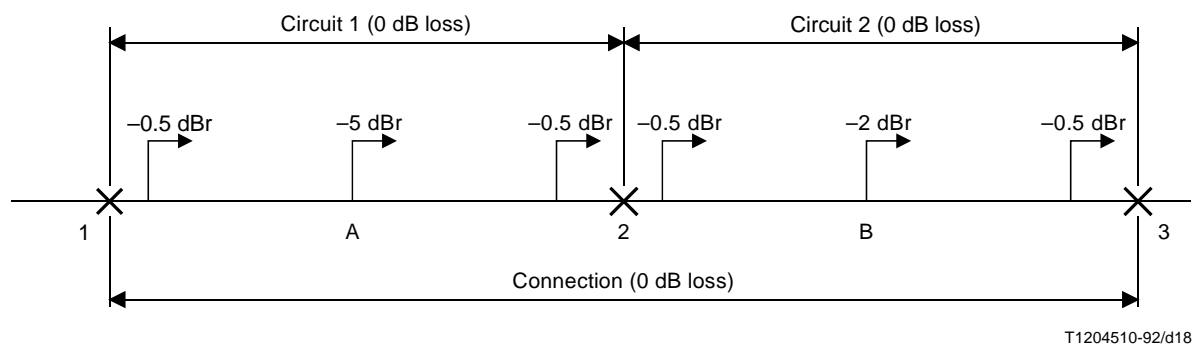


FIGURE A.3/G.101

Example 3

Figure A.4 shows an example where a circuit interconnects a digital and an analogue exchange via an analogue transmission system. For stability reasons, the circuit shall have 0.5 dB loss. The input level at the analogue exchange is 0 dBr.

The consequence of this is that on the receiving side, the digital bit stream in the digital exchange will have a level of -0.5 dBr. It should be noted that when specifying and measuring the digital exchange alone, the same digital bit stream is assigned a level of 0 dBr. This point will have another dBr-level when it is part of a circuit from what it has in specifications and equipment measurements.

NOTE – If the input relative level of the circuit had been changed to $+0.5$ dBr, this problem would have been avoided.

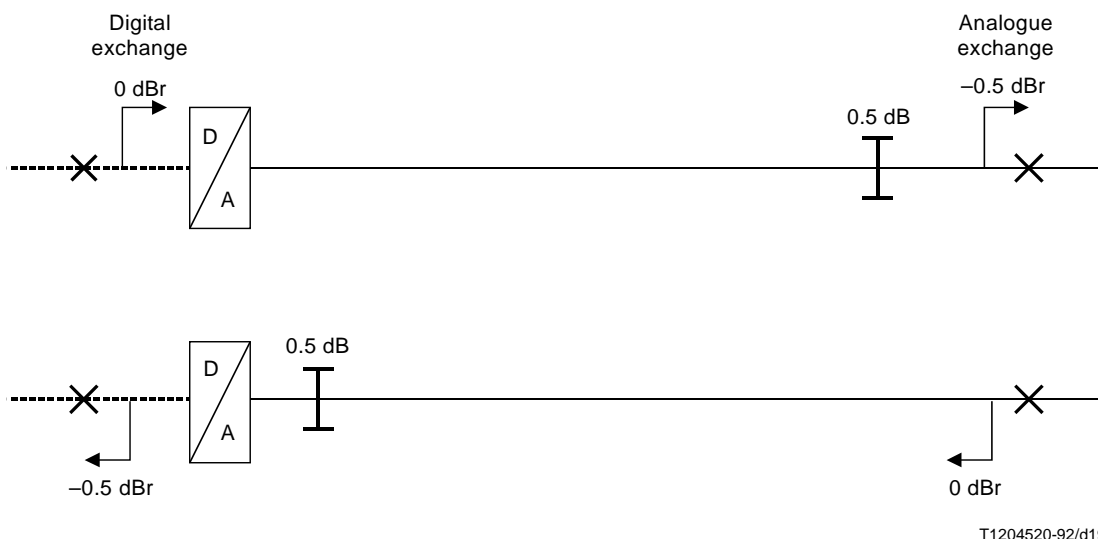


FIGURE A.4/G.101

Example 4

Figure A.5 shows an example where two digital exchanges are interconnected via an analogue and a digital transmission system. The relative levels specified for the transmission systems are as shown in the circles. In the transmission plan the mixed analogue/digital circuit shall have 0.5 dB loss. The relative levels of the circuit are as indicated by the arrows.

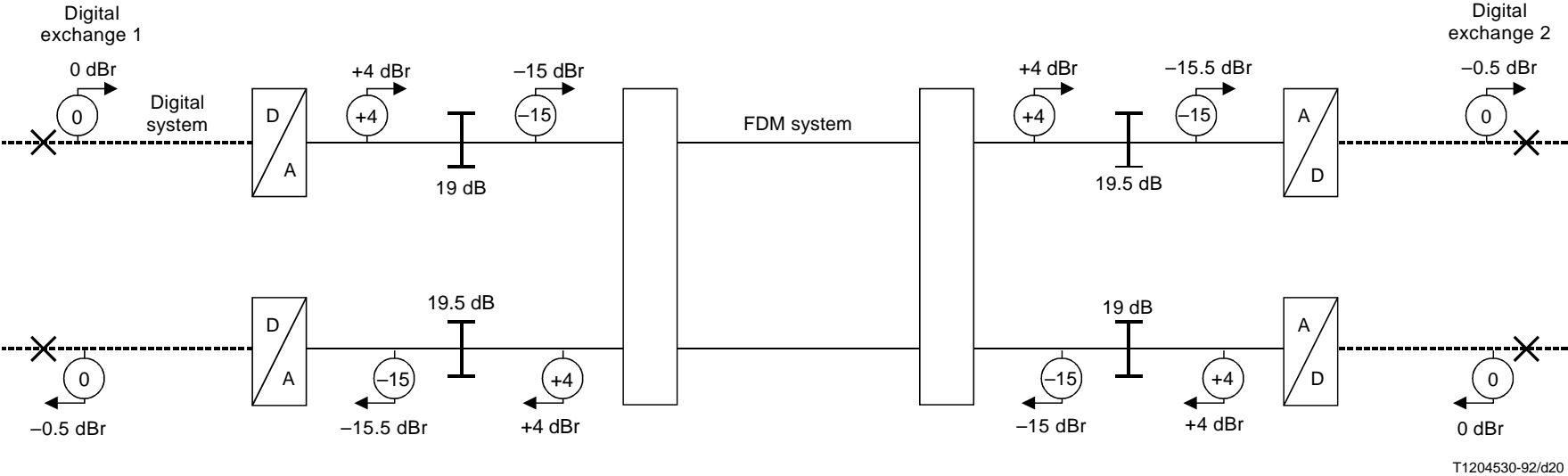


FIGURE A.5/G.101

Appendix I

A computational model for guidance in transmission planning

(This appendix does not form an integral part of this Recommendation)

I.1 Introduction

This informative appendix describes a version of a computational model published by ETSI [39] and which is here referred to as the E-model.

It should be noted that, when published, the model had not been fully verified. Until this verification has been completed, predictions resulting from its use should therefore be treated with caution.

The work continues and contributions are invited that contain information that either supports or challenges any aspect of the model and which might aid verification or improvement of the model.

Computation models such as the ETSI model are becoming increasingly useful for network operators as a help to ensure that users will be satisfied with the transmission performance whilst avoiding over-engineering of networks. Recommendation G.113 gives a guidance about impairments, including certain combination effects, based upon a simplification of the E-model. The aim of this appendix is to give more complete information about the algorithms of the model.

The complexity of modern networks requires that for transmission planning, the many transmission parameters are not only considered individually but also that their mutual combination effects are taken into account. This can be done “by expert, informed guessing”, but a more systematic approach is desirable, such as by using a plausible computation model. The output of such a model can be “nominal” estimations of user reactions, for instance in the form of percentages finding the connection “Good Or Better”, GOB, or “Poor Or Worse”, POW. An alternative, more neutral output is a rating value R, which can be associated with the speech communication quality.

In this context it is worth noticing the general fact that “the user’s perception of quality of a product or service is determined by what degree the user’s expectations are fulfilled or exceeded”. Thus, for speech communication quality, the transmission planner (at present) should consider two categories, namely:

- 1) Toll quality: Average quality of long-distance Public Switched Telephone Network connections, i.e. good intelligibility, good speaker identification, naturalness, only minor disturbing impairments.
- 2) Communications quality: Quality as found in many mobile systems, characterized by good intelligibility, speaker identity maintained, but some loss in quality when directly compared to Toll quality.

For each category, the user may find an offered system performance quite satisfactory, i.e. having a “good quality” with regard to his needs and expectations. It is only in exceptional cases that it is relevant for the user to make relative quality comparisons between the two categories.

There exist a number of computation models by which such an evaluation can be made, at least for some of the transmission parameters, see for instance Annex A/ P.11 and Supplement 3 to the P-Series Recommendations. The E-model is partly based on these models but also on some newly published subjective tests. In particular, many relevant algorithms in the Bellcore “Transmission Rating” model have been used, directly or with some modifications. However, the E-model is more complete, in particular as it can handle the impairments from low bit rate codecs and customer expectations of different transmission services. More details about the data used for the E-model is given in I.4.

I.2 The structure and basic algorithms of the E-model

The model provides “speech communication quality” measures for handset telephony in the normal speech band of nominally 300 Hz to 3400 Hz. The configuration of the connection and its relevant transmission parameters is depicted in Figure I.1. The voice communication is between Side (S) and Side (R). The model estimates the voice transmission quality mouth-to-ear as perceived by the user at Side (R), both for talking and listening conditions.

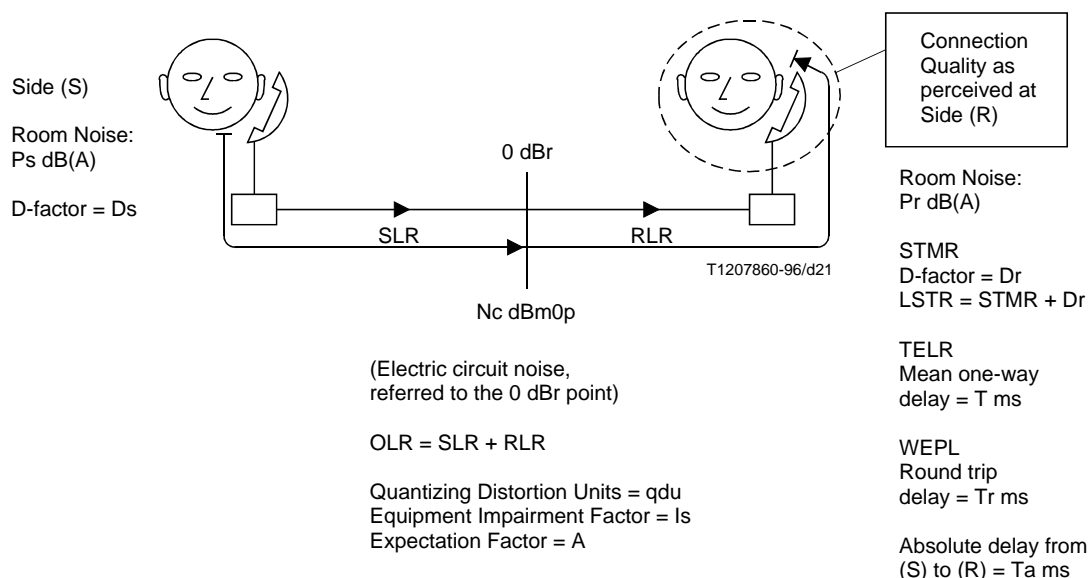


FIGURE I.1/G.101

Configuration and relevant transmission parameters of the connection

Note that the influence of any differences in the voice channel attenuation/frequency response between connections is not at present considered in the E-model. The reason is that with today's deregulation and liberalization of terminal equipment the transmission planner has no possibility to know the actual response curve for a particular user's channel, including the response of the telephone set, with sufficient accuracy to make a meaningful estimation of this influence on the speech communication quality. (It is assumed in the model that the frequency response is "within normal limits".)

Other impairments, not yet included in the E-model, are for instance poor radio channel transmission for mobile communication and the influence of impulsive noise in normal wirebound networks. (Regarding the latter, some information can be found in Recommendations P.55 and P.11.)

The fundamental principle of the E-model is based on a concept established more than twenty years ago by J. Allnatt, and used for example in the NTT model OPINE: "Psychological factors on the psychological scale are additive".

The ETSI model combines the effect of the various transmission parameters into a rating factor R , from which user reaction can be predicted, such as percentages finding the connection "good or better", "poor or worse" or even so bad that they would terminate the call early, as well as what scores would be given in a MOS experiment. The rating factor R is composed of the terms:

$$R = Ro - Is - Id - Ie + A \quad (\text{I-1})$$

Ro represents in principle the basic voice-signal-to-noise ratio.

Is, Id, Ie are so-called impairment factors:

Is represents impairments occurring simultaneously with the voice signal, like a too loud connection, loud sidetone, quantizing distortion from PCM;

Id represents delayed impairments, such as talker and listener echo as well as too long absolute delay;

Ie represents transmission impairments caused by special equipment such as certain low bit rate codecs, Digital Circuit Multiplication Equipment, etc. (This factor is a new concept.)

NOTE 1 – The individual transmission parameters can appear in more than one of the I -factors, sometimes in a masking effect.

A is termed “*Expectation Factor*”. It represents an “advantage-of-access” that certain systems have over conventional wirebound communication systems. The concept of “high quality” is intimately connected with how customers expectations are fulfilled. Thus, the overall connection transmission quality as perceived by the user is highly influenced by the ease or difficulty to establish a connection. In certain circumstances, wireless systems have an advantage in this respect over wire systems, an advantage that can compensate the subjective effect of some speech transmission defects. Examples are mobile telephony and multi-hop satellite connections to hard-to-reach regions. (Sometimes, for economical reasons, special *low-cost* long-distance connections might be considered as having a similar advantage factor.)

The expectation factor is a new concept which has not previously been used in computation models. The reason why it has been introduced is that the E-model is meant to be used for both “Toll quality” and “Communication quality” and that a fair comparison should be feasible to do with regard to a user’s actual perception of the speech communication service. (When the task is to make direct comparisons between systems belonging to respectively the “toll” and the “communication” category, the expectation factor A is omitted.)

NOTE 2 – The Bellcore “Transmission Rating” model described in Supplement 3 to the P-Series Recommendations also can be rewritten in a form equivalent to an addition of impairment factors.

NOTE 3 – The structure of additive impairment factors makes the E-model easy to update, when necessary.

The “Rating Factor” R can lie in the range from 0 (extremely bad) to 100, (very good). The R -factor is transposed into a number of different “quality measures” representing “nominal” statistical estimations of:

- a) Percentage of users finding the connection “good or better”, GOB, “poor or worse”, POW, or so bad that they terminate the call early, TME.
- b) Mean Opinion Scores, (MOS).

GOB, POW and TME are obtained from the R -factor by means of the Error function:

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

Thus:

$$\text{GOB} = 100E\left(\frac{R - 60}{16}\right) \% ; \text{POW} = 100E\left(\frac{45 - R}{16}\right) \% ; \text{TME} = 100E\left(\frac{36 - R}{16}\right) \% \quad (\text{I-3})$$

The MOS, scale 1-5, is obtained by the relations:

$$\text{For } 0 < R < 100 \quad \text{MOS} = 1 + 0.035 \cdot R + R(R - 60)(100 - R) \cdot 7 \cdot 10^{-6} \quad (\text{I-4})$$

$$\text{For } R < 0 \quad \text{MOS} = 1 \quad (\text{I-5})$$

$$\text{For } R > 100 \quad \text{MOS} = 4.5 \quad (\text{I-6})$$

NOTE 4 – Maximum value likely to be given in tests.

GOB, POW, TME and MOS as functions of R are depicted in Figures I.2 and I.3 respectively.

Regarding the terms in (I-1), the expression for R_o is:

$$R_o = 15 - 1.5 (\text{SLR} + N_o) \quad (\text{I-7})$$

SLR is referred to the 0 dBr point nearest Side (R).

No is the *total* noise in dBm0p, also referred to the 0 dBr point. No is obtained by power addition of:

- the electric circuit noise N_c dBm0p;
- the equivalent *circuit* noise N_{os} dBm0p caused by the *room* noise P_{os} dB(A), Side (S);
- the equivalent *circuit* noise N_{or} dBm0p caused by the *room* noise P_{or} dB(A), Side (R);
- the “Noise Floor” N_{fo} dBm0p caused by conditions at Side (R). [At Side (R), the noise floor is $N_{fo} = -64$ dBmp, a value established by experience.]

How to perform this power addition is described in the next subclause. The expressions for I_s and I_d are somewhat cumbersome and therefore they are also given there.

The Equipment Impairment Factor I_e applies for complex speech processing devices, in particular low bit rate codecs. Each type of codec is represented by a specific value K for its corresponding impairment. When tandeming several codecs, whether of the same or different types, the total Equipment Impairment Factor is obtained as the sum of the individual K -values, which are given in Table I.1.

The Expectation Factor A is given in Table I.2.

TABLE I.1/G.101

Codec	kbit/s	K
ADPCM (Rec. G.726, Rec. G.727)	40	2
	32	7
	24	25
	16	50
LD-CELP (Rec. G.728)	16	7
	12.8	20
VSELP (IS54; USA)	8	20
RPE-LTP (GSM)	13	20
CELP+	6.8	25

TABLE I.2/G.101

Communication systems	A
Conventional (fixed)	0
DECT and similar	5
GSM and similar	10
Multi-hop satellite to hard-to-reach places	20

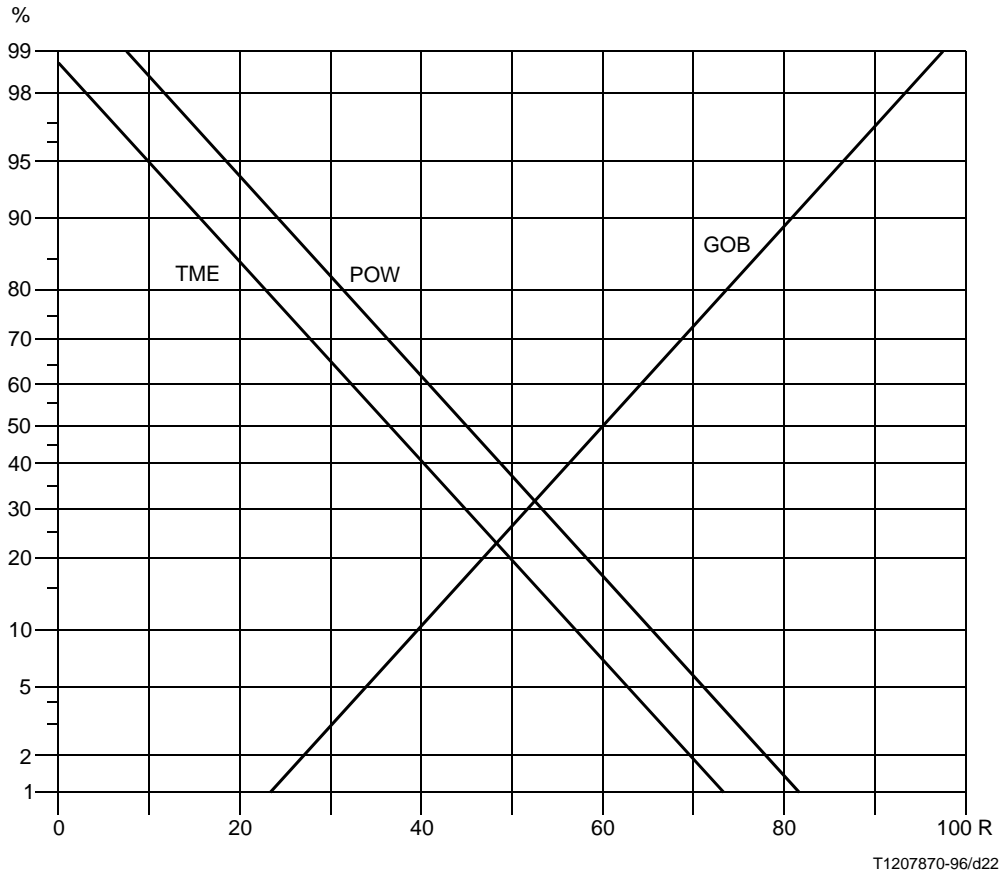


FIGURE I.2/G.101
GOB (Good Or Better), POW (Poor Or Worse) and TME (Terminate Early)
as functions of Rating factor R

I.3 Detailed description of the algorithms

I.3.1 Noise considerations

N_o is the *total* noise in dBm0p, also referred to the 0 dBr point. N_o is obtained by power addition of:

- the electric circuit noise N_c dBm0p;
- the equivalent *circuit* noise N_{os} dBm0p caused by the *room* noise P_{os} dB(A) at Side (S);
- the equivalent *circuit* noise N_{or} dBm0p caused by the *room* noise P_{or} dB(A) at Side (R);
- the “Noise Floor” N_{fo} dBm0p caused by conditions at Side (R).

The electric circuit noise N_c is obtained by power addition of the various electric noise sources in the connection, all referred to the 0 dBr point. (If a noise source of N dBmp is introduced at a point of relative level L dBr in the circuit, this corresponds to a noise level of $(N-L)$ dBm0p at the 0 dBr point.)

The equivalent circuit noise N_{os} caused by room noise P_{os} dB(A) at Side (S) is:

$$N_{os} = P_{os} - SLR - D_s - 100 + 0.008 (P_{os} - OLR - D_s - 14)^2 \quad \text{dBm0p} \quad (\text{I-8})$$

where:

$$OLR = SLR + RLR \quad (\text{I-9})$$

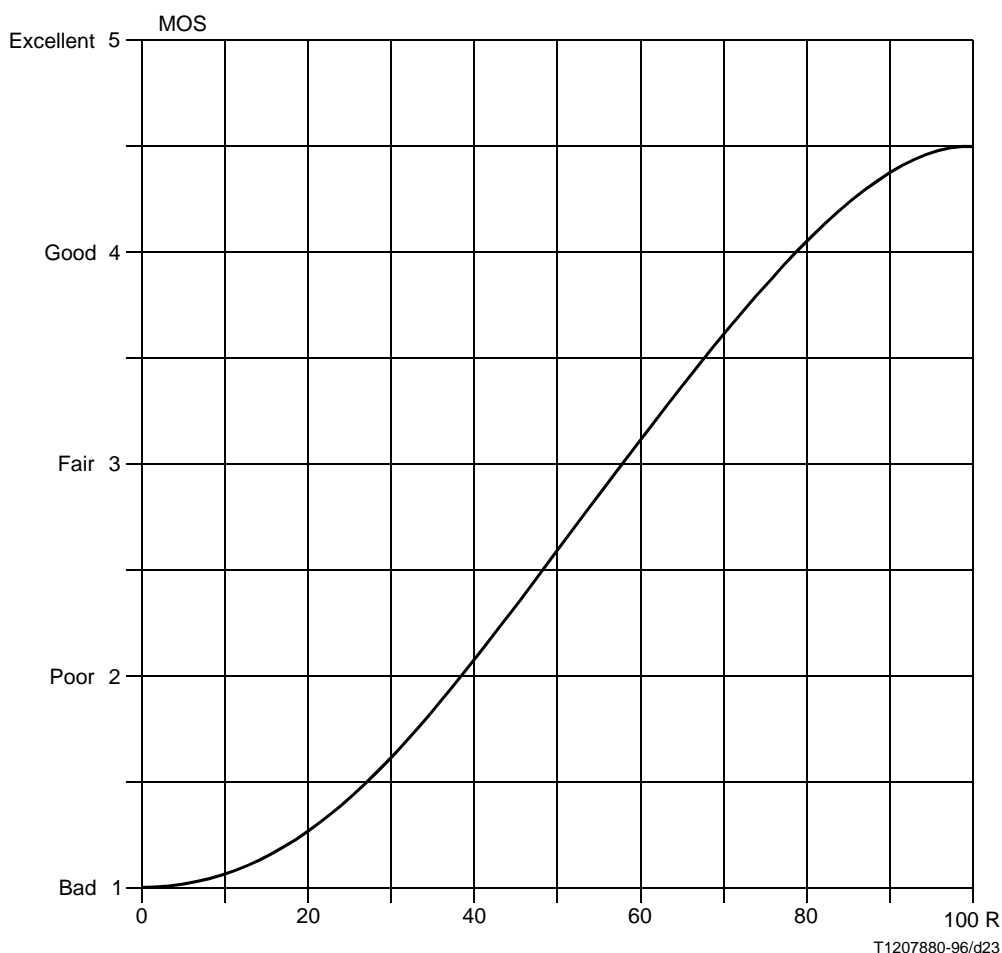


FIGURE I.3/G.101
MOS as function of Rating Factor R

[D_s is the D-factor of the handset at Side (S)].

The equivalent circuit noise N_{or} caused by room noise P_{or} dB(A) at Side (R) is:

$$N_{or} = RLR - 121 + P_{or} + 0.008 (P_{or} - 35)^2 \quad \text{dBm0p} \quad (\text{I-10})$$

where P_{or} is the *effective* room noise caused by enhancement of P_{or} by the listener's sidetone path.

$$P_{or} = P_{or} + 10 \lg \left[1 + 10^{(10 - LSTR)/10} \right] \quad \text{dBm0p} \quad (\text{I-11})$$

[$LSTR$ is the Listener's Sidetone Rating at Side (R)].

The noise floor N_{fo} dBm0p refers to a noise floor N_{for} dBmp at Side (R). Thus:

$$N_{fo} = N_{for} + RLR \quad \text{dBm0p} \quad (\text{I-12})$$

Normally, $N_{for} = -64$ dBmp

Finally, the total noise No is obtained by power addition of the noise components:

$$No = 10 \lg \left[10^{Nc/10} + 10^{Nos/10} + 10^{Nor/10} + 10^{Nfo/10} \right] \quad \text{dBmOp} \quad (\text{I-13})$$

Note that quantizing distortion noise is *not* included in this summation.

I.3.2 The simultaneous-impairment factor Is

The expression for Is is:

$$Is = Iolr + Ist + Iq \quad (\text{I-14})$$

$Iolr$ represents the decrease in quality caused by a too loud connection, i.e. when OLR is too low.

Ist represents the impairment caused by non-optimum sidetone.

Iq represents the impairment caused by quantizing distortion from PCM codecs, digital pads, etc.

The expression for $Iolr$ is:

$$Iolr = 20 \left[\left\{ 1 + (X/8)^8 \right\}^{1/8} - X/8 \right] \quad (\text{I-15})$$

where:

$$X = OLR + 0.2(64 + N_t) \quad (\text{I-16})$$

$$N_t = No - RLR \quad (\text{I-17})$$

The expression for Ist is:

$$Ist = 10 \cdot \left[1 + \left\{ (STMRO - 12)/5 \right\}^6 \right]^{1/6} - 46 \cdot \left[1 + \left\{ STMRO/23 \right\}^{10} \right]^{1/10} + 36 \quad (\text{I-18})$$

where:

$$STMRO = -10 \cdot \lg \left[10^{-STM/10} + e^{-T/4} \cdot 10^{-TEL/10} \right] \quad (\text{I-19})$$

The expression for Iq is:

$$Iq = 15 \lg \left[1 + 10^Y \right] \quad (\text{I-20})$$

where:

$$Y = (Ro - 100)/15 + (46 - G)/10 \quad (\text{I-21})$$

Ro is given by (I-7) and G by the expressions:

$$G = 1.07 + 0.258 Q + 0.0602 Q^2 \quad (\text{I-22})$$

$$Q = 37 - 15 \cdot \lg(qdu) \quad (\text{I-23})$$

qdu is the number of *quantizing units* in the connection.

I.3.3 The “delayed-impairment” factor Id

The expression for Id is:

$$Id = Idte + Idle + Idd \quad (\text{I-24})$$

$Idte$ represents the impairment caused by talker echo. Relevant parameters are the Talker Echo Loudness Rating TELR and the mean one-way delay time T ms for the echo.

$Idle$ represents the impairment caused by listener echo. Relevant parameters are the Weighted Echo Pass Loss WEPL and the round trip delay Tr ms for the echo.

Idd represents the impairment caused by too long absolute delay which occurs even with perfect echo cancelling. Relevant parameter is the one-way absolute delay Ta ms.

The expression for $Idte$ is:

$$Idte = \left[(Roe - Re) / 2 + \sqrt{(Roe - Re)^2 / 4 + 100} - 1 \right] \cdot (1 - e^{-T}) \quad (\text{I-25})$$

where:

$$Roe = -1.5 \cdot (No - RLR) \quad (\text{I-26})$$

$$Re = 80 + 2.5(TERV - 14) \quad (\text{I-27})$$

$$TERV = TELR - 40 \lg \frac{1 + T/10}{1 + T/150} + 6e^{-0.3T^2} \quad (\text{I-28})$$

Note that for $T < 1$ ms the “talker echo” should be considered as *sidetone* so that then $Idte = 0$.

Equations (I-20) to (I-22) apply when the talker sidetone is “normal”, i.e. $9 < \text{STMR} < 15$. For lower values of STMR the talker echo is partly masked by the sidetone, while for higher values of STMR the talker echo becomes more noticeable than with a normal sidetone. These phenomena are taken into consideration by adjustments of the TERV and $Idte$ respectively as follows:

For $\text{STMR} < 9$: TERV is replaced by TERVs in equation (I-27).

$$TERVs = TERV + Ist / 2 \quad (\text{I-29})$$

For $STMR > 15$: $Idte$ is replaced by $Idtes$.

$$Idtes = \sqrt{Idte^2 + Ist^2} \quad (I-30)$$

The expression for $Idle$ is:

$$Idle = (Ro - Rle) / 2 + \sqrt{(Ro - Rle)^2 / 4 + 169} \quad (I-31)$$

where Ro is given by equation (I-7) and:

$$Rle = 105(WEPL + 7)(Tr + 1)^{-0.25} \quad (I-32)$$

The expressions for Idd are:

$$\text{For } Ta < 100 \text{ ms} \quad Idd = 0 \quad (I-33)$$

For $Ta > 100 \text{ ms}$

$$Idd = 25 \left\{ \left[1 + X^6 \right]^{1/6} - 3 \left[1 + (X/3)^6 \right]^{1/6} + 2 \right\} \quad (I-34)$$

where:

$$X = \frac{\lg(Ta / 100)}{\lg 2} \quad (I-35)$$

I.4 Sources used in the E-model for evaluation of impairments and the expectation factor

In particular, the following sources for the evaluations can be mentioned.

The influence of room noise at send and receive sides, including the dependence on LSTR and D-factors: the BcTR model and published test results from the Australian and Swedish Administrations.

The influence of loudness ratings – OLR, SLR, RLR – and equivalent circuit noise: the BcTR model, however adjusted for low OLR values by considerations of results from the CATNAP model and some additional subjective tests.

The influence of the talker sidetone, STMR: interpretation of information in Supplement 11 to the P-Series Recommendations and the information that talker echo with very low delay is interpreted as a form of sidetone.

The influence of quantizing distortions for PCM systems (qdu): the CATNAP (Supplement 3 to the P-Series Recommendations) model.

The influence of talker echo, TELR: For normal values of STMR the BcTR model has been used, slightly adjusted by considerations of results from NTT investigations, and complemented for short delays by results published by France Telecom and Telia Research. For very low values of STMR results from the BcTR model have been applied, while for very high values of STMR subjective tests from Telia Research and British Telecom have been employed.

The influence of listener echo, WEPL: the BcTR model.

The influence of long absolute delays: interpretation of information given in Recommendation G.114, with a certain emphasis on how the delay influences highly interactive conversations up to about 800 ms. For still longer delays, the conversation partners are likely to realise that one has to wait for an answer.

The influence of low bit rate codecs, the equipment impairment factors: subjective tests, published in ITU-T contributions to Study Group 12 (SQEG), Study Group 15 and elsewhere.

The expectation factor A : interpretation of information from the market place that certain systems enjoy widespread public acceptance, i.e. their “speech communication quality” is deemed to be high, although their “absolute” voice transmission quality must be considered rather low in comparison with that of the conventional telephone network. The first example is mobile communication with its phenomenal growth. The second example is double-satellite hops to places that otherwise could not be reached by telephone. (The values of the expectation factor A are provisionally chosen so that they compensate half the impairments caused by the GSM (RPE-LTP) codec respectively the long absolute delay, i.e. $A = 10$ respectively 20.)

For the *subscriber opinion prediction* functions, the Bellcore Transmission Rating model and the Annex A/P.11 model have been used.

For GOB and POW the E-model represents an average of the “Long Toll”, “ITU-T” and “MH” investigations. TME corresponds to results from an old AT&T investigation.

Finally, in the ETSI model the expression for MOS as function of R has been derived indirectly, partly by comparison with relations between GOB, POW and MOS as given in Annex A/P.11 partly from experience of what the range of scores is that subjective test teams usually assign to various speech transmission channels. A direct conversion from Recommendation P.11 gives the range $1 < \text{MOS} < 5$. However, in actual subjective tests the maximum MOS is hardly ever higher than 4.5 and therefore the ETSI model has been adapted to this value by a linear transformation from the P.11 results.

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