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

International telephone connections and circuits – General definitions

## Definitions used in Recommendations on general characteristics of international telephone connections and circuits

ITU-T Recommendation G.100

(Formerly CCITT Recommendation)

ITU-T G-SERIES RECOMMENDATIONS

**TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS**

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**ITU-T Recommendation G.100**

## Definitions used in Recommendations on general characteristics of international telephone connections and circuits


**Summary**

This Recommendation gives the definitions which have been found to be useful in the study of telephone connections and telephone circuits. It provides general terms and definitions related to transmission performance characteristics



**Source**

ITU-T Recommendation G.100 was revised by ITU-T Study Group 12 (2001-2004) and approved under the WTSA Resolution 1 procedure on 23 February 2001.

## FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. 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 Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

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

## NOTE

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

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## Introduction

The definitions given below have been found to be useful in the study of telephone connections and telephone circuits.

The detailed definitions appearing in ITU-T G.102 are referred to, but not reproduced.

The definitions of specialized terms which are not mentioned here can be found in

- ITU-T E.800, for availability and reliability;
- ITU-T G.117 as concerns unbalance about earth;
- Annex A/G.111 as concerns speech transmission performance;
- Recommendations of the G.16x series for echo suppressors, echo cancellers, compandors, etc.

Annex A gives an overview of some dB-related parameters as used in speech-band applications.

**ITU-T Recommendation G.100**

# Definitions used in Recommendations on general characteristics of international telephone connections and circuits

## 1 General terms

### 1.1 3.1 kHz handset telephony

A real-time two-way speech communication within the frequency range approximately from 300 to 3400 Hz using one or more telecommunication networks with suitable terminal equipment connected to the network termination points, characterized by:

- presentation of an acoustical speech signal to the mouthpiece of a traditionally shaped handset:
  - either analogue transport of said speech signal under real-time conditions through and by telecommunication networks: said networks being intended for telephony applications between network termination points;
  - or filtering of said speech signal to the frequency range approximately from 300 to 3400 Hz; transformation of said speech signal either by waveform or by non-waveform (speech analysis) encoder; transport and processing of said speech signal under real-time conditions through and by telecommunication networks: said networks being intended for telephony applications between network termination points; back transformation (speech synthesis) of said speech signal by the respective decoder;
- acoustical presentation of said speech signal in the frequency range approximately from 300 to 3400 Hz by the earpiece of a traditionally shaped handset.

### 1.2 **4-wire chain** (see ITU-T G.101)

*F: chaîne à 4 fils*

*S: cadena a cuatro hilos*

The 4-wire chain (see Figure 6/G.101) 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 (see 2.11/G.101).

### 1.3 analogue network

A network in which the access interface and all network elements are considered analogue.

### 1.4 **circuit access point** (see ITU-T G.101)

*F: point d'accès à un circuit*

*S: punto de acceso del circuito*

The circuit access points have been defined as "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 ITU-T 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 (see 2.14/G.101).

### 1.5 **circuit, telecommunication circuit** (see ITU-T G.101)

*F: circuit, circuit de télécommunication*

*S: circuito, circuito de telecomunicación*

A combination of two transmission channels permitting bidirectional transmission of signals between two points, to support a single communication.

NOTE 1 – If the telecommunication is by nature unidirectional (for example: long distance television transmission), the term "circuit" is sometimes used to designate the single channel providing the facility.

NOTE 2 – In a telecommunication network, the use of the term "circuit" is generally limited to a telecommunication circuit directly connecting two switching devices or exchanges, together with associated terminating equipment.

NOTE 3 – A telecommunication circuit may permit transmission in both directions simultaneously (duplex), or not simultaneously (simplex).

NOTE 4 – A telecommunication circuit that is used for transmission in one direction only is sometimes referred to as a unidirectional telecommunication circuit. A telecommunication circuit that is used for transmission in both directions (whether simultaneously or not) is sometimes referred to as a bidirectional telecommunication circuit.

NOTE 5 – The term circuit may be preceded by other qualifiers than telecommunication, e.g. telephone, digital, leased, etc., each defining a different application and having a different meaning.

### 1.6 **connection** (see ITU-T G.101)

*F: connexion, chaîne de connexion*

*S: conexión, cadena de conexión*

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

### 1.7 conversational quality

Quality with which a bi- or multidirectional conversation is perceived by a communication partner.

### 1.8 conversational speech quality

Speech quality as experienced in a bi- or multidirectional conversation.

### 1.9 E-model

A computational transmission rating model which is the common ITU-T transmission rating model. The algorithmic description is given in ITU-T G.107.

### 1.10 end-to-end quality

Quality related to the performance of a communication system, including all terminal equipment. For voice services it is equivalent to mouth-to-ear quality.

## 1.11 dB-related units (see ITU-T G.101)

*F: unités utilisant le dB*

*S: unidades relacionadas con el dB*

dBW: Absolute power level with respect to 1 watt, expressed in decibels;

dBm: Absolute power level with respect to 1 milliwatt, expressed in decibels;

dBu: Absolute voltage level with respect to 0.775 V, expressed in decibels;

dBrs: Relative power level expressed in decibels, referred to another point in sound-programme transmission;

dBV: Absolute power level with respect to 1 V, expressed in decibels;

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

$$V = \sqrt{10^{\frac{x}{10}} \times (1 \times 10^{-3})\text{watt} \times |Z_{1020}|} \text{ volts}$$

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.

NOTE – A discussion of the applications of other dB-related terms is given in Appendix I.

## 1.12 digital mobile system (DMS) (see ITU-T G.173)

*F: système mobile numérique (DMS)*

*S: sistema móvil digital (DMS)*

The basic configuration of a digital mobile system is shown in Figure 1/G.173. A digital mobile system consists of the mobile station, radio transmission path, base station, leased line and the mobile services switching centre up to the network connection point.

## 1.13 extension line (see ITU-T G.101)

*F: ligne téléphonique supplémentaire*

*S: línea de extensión o línea suplementaria*

Line connecting an extension either to a subscriber's main station or to a private branch exchange (IEV 722-12-12).

## 1.14 hypothetical reference connection (HRX)

*F: connexion fictive de référence*

*S: conexión ficticia de referencia (HRX)*

A hypothetical connection of defined structure, length and performance in a telecommunication network for analogue or digital (or mixed) signal transmission, to be used as a model in which studies relating to overall performance may be made, thereby allowing comparisons with standards and objectives.

**1.15 input/output** (see ITU-T G.111, ITU-T G.121, etc.)

*F: entrée/sortie*

*S: entrada/salida*

Terms used to indicate the direction of transmission at an interface of an equipment item. These terms avoid the ambiguity encountered in the use of "transmit/receive" or "send/receive".

**1.16 mixed analogue-digital channel (circuit)**

*F: voie (circuit) mixte analogique-numérique*

*S: canal (circuito) mixto analógico-digital*

A channel (circuit) comprising analogue-to-digital (digital-to-analogue) conversion. If one-type transmission channel is provided (only digital or only analogue), then analogue-to-digital (digital-to-analogue) conversion is possible only at the ends of the channel (channel translation equipment in accordance with ITU-T G.712, transmultiplexor in accordance with ITU-T G.793, ITU-T G.794). If the channel is made of separate sections of analogue and digital transmission systems, then analogue-to-digital (digital-to-analogue) conversion is possible in its separate sections (group modems are in accordance with ITU-T G.941 or ITU-T V.37, transcoders are in accordance with ITU-T G.761, group codecs are in accordance with ITU-T G.795).

**1.17 mouth-to-ear quality**

Speech quality as experienced by the user of a voice communication system. Includes the whole transmission path from the mouth of the talker to the ear of the listener.

**1.18 national system** (see ITU-T G.101)

*F: système national*

*S: sistema nacional*

The national system starting at the VICP may comprise one or more 4-wire national trunk circuits with 4-wire interconnection, as well as circuits with 2-wire connection up to the local exchange, the subscriber stations with their subscriber lines or a PBN (see Figure 5/G.101).

**1.19 normal-band telephony**

*F: téléphonie en bande normale*

*S: telefonía en banda normal*

Transmission of a signal (either speech or data) through a telephonic network with a nominal pass-band of 300-3400 Hz (see wideband telephony).

**1.20 private automatic branch exchange (PABX)**

*F: autocommutateur privé (PABX)*

*S: centralita automática privada (PABX)*

A private branch exchange consisting of an automatic telephone exchange (IEV 722-08-06).

### 1.21 private branch exchange (PBX)

*F: commutateur (téléphonique) privé (PBX)*

*S: conmutador (telefónico) privado (PBX)*

A telephone switching entity forming part of a private telephone installation that has access to the public switched telephone network (IEV 722-08-05).

### 1.22 private branch network (PBN)

*F: réseau (de télécommunication) privé*

*S: red colateral privada*

A private telecommunication network that has access to the public network.

### 1.23 private network

The term "Private Network" is used to describe a network which provides switching functions and all other features only to a single customer or to a group of customers (restricted user group) and which is not available to the general public.

In general, a private network is a terminating network and consists of several interconnected nodes (e.g. PBXs), with interconnections to other networks.

It consists of more than one element of switching equipment, connected via tie trunks or leased lines or via a Virtual Private Network (VPN). Network functionality is independent of its structure and hierarchy.

It is not limited by geographical size or to a specific national area or region and has no limitation with regard to the number of extensions and access points to other networks.

### 1.24 public switched telephone network (PSTN)

*F: réseau téléphonique public commuté (RTPC)*

*S: red telefónica pública conmutada (RTPC)*

The term "Public Switched Telephone Network" or, for short, "Public Network" is used for any network (without any relation to the legal status of the network operator) providing transmission and switching functions as well as features which are available to the general public, not restricted to a specific user group.

The PSTN provides access points to other networks or terminals only within a specific geographical area.

From the point of view of an end-to-end connection, a public network can function either as a "Transit Network" (a link between two other networks) or as a combination of "Transit and Terminating Network" in cases where the public network provides connections to terminal equipment such as telephone sets, or PBXs.

### 1.25 relative level (at a point on a circuit)

*F: niveau relatif (en un point d'un circuit)*

*S: nivel relativo (en un punto de un circuito)*

The expression $10 \log_{10} (P/P_0)$ dBr where $P$ represents the power of a test signal of 1000 Hz at the point concerned and $P_0$ the power of that signal at the *transmission reference point*.

NOTE – This quantity is independent of $P_0$; it is a composite gain (level difference).

### 1.26 relative (power) level (see ITU-T G.101)

*F: niveau relatif (de puissance)*

*S: nivel relativo (de potencia)*

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 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.

### 1.27 return loss

*F: affaiblissement d'adaptation*

*S: pérdida de retorno*

Quantity characterizing the degree of match between two impedances, $Z_1$ and $Z_2$. It is given by the expression:

$$L_R = 20\log_{10}\left|\frac{Z_1 + Z_2}{Z_1 - Z_2}\right| \text{ dB}$$

### 1.28 speech quality

Quality of spoken language as perceived when acoustically displayed. Result of a perception and assessment process, in which the assessing subject establishes a relationship between the perceived characteristics, i.e. the auditory event, and the desired or expected characteristics.

### 1.29 speech transmission quality

Speech quality related to the performance of a communication system, in general terms. Categories of speech transmission quality are defined in ITU-T G.109, based on the prediction of the E-model, i.e. in terms of ranges for the transmission rating factor $R$.

### 1.30 subscriber circuit (see ITU-T G.101)

*F: circuit d'abonné*

*S: circuito de abonado*

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/G.101. 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 (see 2.1/G.101).

NOTE – In the local exchange, the subscriber circuit usually includes "half" of the exchange and in an analogue exchange, the input and the 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.

### 1.31 telephone circuit (see ITU-T G.101)

*F: circuit téléphonique*

*S: circuito telefónico*

In transmission planning, and in the G-series Recommendations, a telephone circuit denotes a telecommunication circuit with associated equipment, directly connecting two switching devices or exchanges, in line with Note 2 to the general definition of a circuit; see definition 1.5. For simplicity, the term "circuit" is often used instead of "telephone circuit" in the G-series Recommendations (see 2.1/G.101).

NOTE 1 – Conceptually, (telephone) circuits are those parts of a connexion that remain intact and permanently associated with the switches at each end, after a connexion 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/G.101).

NOTE 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 system.

### 1.32 transmission rating model

An algorithm that calculates the effects of variations in several transmission parameters on conversational quality. The model output is one or several quality related indices that are meant to help transmission planners to ensure desired transmission performance, but are no actual customer opinion predictions.

### 1.33 transmission reference point (TRP) (see ITU-T G.101)

*F: point de référence pour la transmission*

*S: punto de referencia para la transmisión*

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 (see 2.2/G.101).

### 1.34 virtual international connecting point (VICP) (see ITU-T G.101)

*F: point de connexion international virtuel (VICP)*

*S: extremo virtual de la conexión internacional (VICP)*

The virtual international connecting points define the boundary between the national and international part of a connection; see Figure 5/G.101. The international connecting points are also used as reference points for transmission quantities recommended for the national and international part of a connection (see 2.12/G.101).

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.

### 1.35 (one-way) voice transmission quality

Speech quality related to voice signals transmitted over a communication system, experienced by a user of that system in a listening-only situation. Refers to the one-way transmission characteristics only.

### 1.36 wideband telephony

*F: téléphonie en bande élargie*

*S: telefonía de banda ancha*

Transmission of speech with a nominal pass-band wider than 300-3400 Hz, usually understood to be 100-7000 Hz (see Normal-band telephony).

## 2 Transmission performance objectives

### 2.1 performance objective

*F: objectif pour la qualité de fonctionnement*

*S: objetivo de calidad de funcionamiento*

(Defined in ITU-T G.102.)

### 2.2 design objective

*F: objectif pour les projets*

*S: objetivo de diseño*

(Defined in ITU-T G.102.)

### 2.3 commissioning objective

*F: objectif pour la mise en service*

*S: objetivo de puesta en servicio inicial*

(Defined in ITU-T G.102.)

### 2.4 limits for maintenance purposes; maintenance limits

*F: limites de maintenance*

*S: límites de mantenimiento*

(Defined in ITU-T G.102.)

## 3 Transmission impairments

### 3.1 advantage factor

A scalar number (normally positive) representing the advantage of access certain systems (e.g. mobile) have over wirebound handset telephony. Expressed in units of the transmission rating factor $R$.

### 3.2 equipment impairment factor (*Ie*)

A scalar number allocated to a network element, indicating the anticipated incremental value of impairment (decrease of the transmission rating factor $R$) resulting from the type of impairment. Expressed in units of the transmission rating factor $R$. Impairment factors are constituent parts of the overall transmission rating factor $R$ of the E-model.

### 3.3 impairment factor

A scalar number allocated to a specific type of impairment, indicating the anticipated incremental value of impairment (decrease of the transmission rating factor *R*) resulting from the type of impairment. Expressed in units of the transmission rating factor *R*. Impairment factors are constituent parts of the overall transmission rating factor *R* of the E-model.

### 3.4 group-delay distortion

*F: distorsion de temps de propagation de groupe*

*S: distorsión por retardo de grupo*

The difference between group delay at a given frequency and minimum group delay, in the frequency band of interest.

### 3.5 quantizing distortion unit (qdu) (see ITU-T G.113)

*F: unité de distorsion de quantification (qdu)*

*S: unidad de distorsión de cuantificación (qdu)*

A unit used for planning purposes, which reflects the effect of quantizing noise impairment on voice signals. One qdu is equivalent to the distortion that results from a single encoding and decoding by an average G.711 codec. The *qdu* concept is not applicable to low-bit rate codecs. Values of *qdu* associated with digital processes other then low-bit rate codecs are provided in ITU-T G.113.

### 3.6 transmission rating factor (R)

Principal output of the E-model. Scalar value which combines the effects of different transmission parameters and varies with the mouth-to-ear conversational quality.

## 4 Propagation time, echo and stability

### 4.1 balance return loss

*F: affaiblissement d'équilibrage*

*S: atenuación de equilibrado*

At a 4-wire terminating set ("hybrid"), that portion of the *semi-loop loss* which is attributable to the degree of match between the impedance, $Z_2$, connected to the 2-wire line terminals, and the balance impedance, $Z_B$. It is given approximately by the expression:

$$L_{BR} = 20\log_{10}\left|\frac{Z_2 + Z_B}{Z_2 - Z_B}\right| \text{ dB}$$

NOTE – Under most circumstances the expression given is sufficiently accurate. However, for some worst case evaluations, the exact expression must be used. The exact expression is

$$L_{BR} = 20\log_{10}\left|\frac{Z_0 + Z_B}{2Z_0} \cdot \frac{Z_2 + Z_0}{Z_2 - Z_B}\right| \text{ dB}$$

where $Z_0$ is the 2-wire input impedance. (If $Z_0 = Z_B$, the two expressions become identical.)

**4.2 circuit loudness rating (CLR)** (see ITU-T G.111)

*F: équivalent pour la sonie du circuit (CLR)*

*S: índice de sonoridad del circuito (CLR)*

The loudness loss between two electrical interfaces in a connection or circuit, each interface terminated by its nominal impedance which may be complex.

**4.3 composite loss** (see ITU-T G.101)

*F: affaiblissement composite*

*S: atenuación compuesta*

The composite loss of a quadripole inserted between two impedances $Z_E$ (of the generator) and $Z_R$ (of the load) is the expression in transmission units of the ratio $P_E/P_R$, where:

$P_E$ is the apparent power that the generator $Z_E$ would furnish to a load of impedance $Z_E$.

$P_R$ is the apparent power that the same generator furnishes via the said quadripole to the load $Z_R$.

If the number thus obtained is negative, then there is a composite gain.

**4.4 crosstalk receive loudness rating (XRLR)**

The loudness loss from a disturbing electric interface to the disturbed subscriber's ear via the crosstalk path.

**4.5 echo**

*F: écho*

*S: eco*

Unwanted signal delayed to such a degree that, for instance in telephony, it is perceived as distinct from the wanted signal (i.e. the signal directly transmitted).

NOTE 1 – Distinction is made between talker echo and listener echo.

NOTE 2 – An echo is usually considerably attenuated with respect to the wanted signal.

**4.6 echo balance return loss**

*F: affaiblissement d'équilibrage pour l'écho*

*S: atenuación de equilibrado para el eco*

*Balance return loss* averaged with $1/f$ power weighting over the telephone band, in accordance with clause 4/G.122.

**4.7 echo control device**

*F: dispositif de réduction d'écho*

*S: dispositivo de control de eco*

A voice-operated device placed in the 4-wire portion of the circuit and used for reducing the effect of echo.

NOTE – This reduction is in practice carried out either by subtracting an estimated echo from the circuit echo (i.e. cancelling it) or by introducing loss in the transmission path to suppress the echo (echo suppression).

### 4.8 echo loss ($L_{ECHO}$)

*F: affaiblissement d'écho ($L_{ÉCHO}$)*

*S: atenuación del eco ($L_{ECHO}$)*

*Semi-loop loss* averaged with $1/f$ power weighting over the telephone band, in accordance with clause 4/G.122.

NOTE 1 – In cases where a point *t* (2-wire point) exists, the echo loss is approximately equal to the sum of the transmission losses *a-t* and *t-b* and the *echo balance return loss*. (Points *a* and *b* are shown in ITU-T G.122.)

NOTE 2 – Distinction may be made between the echo loss of a given piece of equipment and that of a national system (see Note 2 to definition in 4.18.1).

### 4.9 hollowness

*F: son caverneux*

*S: cavernosidad*

Distortion in telephony caused by double reflected signals and subjectively perceived as a "hollow sound", i.e. as if the talker would speak into some hollow vessel.

NOTE – Hollowness is to be distinguished from *listener echo*.

### 4.10 listener echo; receive end echo

*F: écho à la réception*

*S: eco para el oyente; eco en la recepción*

*Echo* produced by double reflected signals and disturbing the listener, receiving voice-band data equipment, etc.

NOTE 1 – The term "received end echo" is a term preferred by some Administrations.

NOTE 2 – With small delay against the wanted signal (less than about 3 ms), listener echo may cause *hollowness* in telephony. In transmission of voice-band data signals, listener echo may cause bit errors and, in any case, reduces the margin against other disturbances.

### 4.11 listener echo loss; receive echo loss

*F: affaiblissement de l'écho à la réception*

*S: atenuación del eco para el oyente; atenuación de eco en la recepción*

Degree of attenuation of the double reflected signal with respect to the wanted signal. In terms of the absolute losses of both signals, the listener echo loss is $LE = L_2 - L_1$ (see Figure 1).

NOTE – For practical purposes, the listener echo loss is equal to the *open-loop loss* (valid if the latter exceeds 8 dB). The listener echo loss characterizes the degree of disturbance by *hollowness*, as well as the disturbing effect on voice-band data modem receivers.

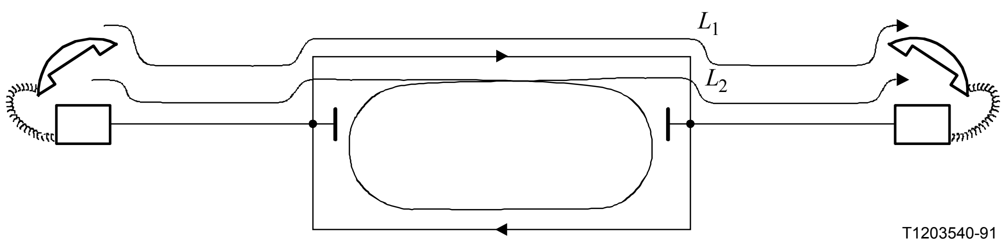



**Figure 1/G.100**

### 4.12 listener echo loudness rating (LELR)

The difference in loudness loss between the speaker's direct voice sound and its delayed echo reaching the listening subscriber's ear.

### 4.13 listener's sidetone rating (LSTR)

The loudness loss between a Hoth-type room noise source and the subscriber's (earphone) ear via the electric sidetone path (see ITU-T P.10 for a full definition).

### 4.14 loudness rating (LR)

As used in the G-series Recommendations for planning: loudness rating is an objective measure of the loudness loss, i.e. a weighted, electro-acoustic loss between certain interfaces in the telephone network. (The nature of the weighting will be dealt with later.) If the circuit between the interfaces is subdivided into sections, the sum of the individual section LRs is equal to the total LR.

How to determine and to apply LRs in the G-series Recommendations is described in A.3 and A.4. The methods are sufficiently accurate for all practical purposes. (Fundamentally, loudness ratings are based on subjective methods as described in ITU-T P.76 and ITU-T P.78. However, subjectively measured values, in general, vary too much with time and test teams to be really useful for transmission planning.)

In loudness rating contexts, the subscribers are represented from a measuring point of view by an artificial mouth and an artificial ear respectively, both being accurately specified.

### 4.15 mean one-way propagation time

*F: temps de propagation moyen dans un sens*

*S: tiempo medio de propagación en un sentido*

In a connection, the mean of the propagation times in the two directions of transmission.

NOTE – The use of this concept is explained in ITU-T G.114.

### 4.16 open-loop loss (OLL)

*F: affaiblissement en boucle ouverte (OLL)*

*S: atenuación en bucle abierto (OLL)*

In a loop formed by a 4-wire circuit (or a cascade connection of two or more 4-wire circuits) and terminated by 2-wire ends (i.e. having "4-wire terminating sets", or hybrids, at both ends), the loss measured by breaking the loop at some point, injecting a signal and measuring the loss incurred in traversing the open loop. All impedance conditions should be preserved while making the measurement. See Figure 2.

NOTE 1 – In practice the OLL is equal to the listener echo loss.

NOTE 2 – The OLL is also equal to the sum of the two *semi-loop losses* associated with a loop.

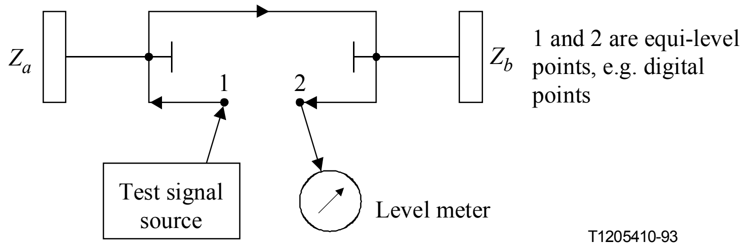



**Figure 2/G.100**

### 4.17 overall loudness rating (OLR)

The loudness loss between the speaking subscriber's mouth and the listening subscriber's ear via a connection.

### 4.18 path a-t-b (transmission loss of …); semi-loop loss

*F: affaiblissement du trajet a-t-b; affaiblissement en demi-boucle*

*S: atenuación del trayecto a-t-b; atenuación en semibucle*

The transmission loss between the points *a* and *b* of the 4-wire termination (as defined at the virtual switching points) independent of whether there exists or not a physical point *t*.

**4.18.1 semi-loop loss** (possible alternative to the definition in 4.18)

*F: affaiblissement en demi-boucle*

*S: atenuación en semibucle*

In an arrangement comprising a 4-wire circuit (or a cascade connection of several 4-wire circuits) with unwanted coupling between the go and return direction at the ends of the circuit – usually via a 4-wire terminating set, or via acoustical coupling – the loss measured between the input and output. See Figure 3.

NOTE 1 – The semi-loop loss is an important quantity in determining *echo balance return loss, echo loss, listener echo loss* (see also *open-loop loss*).

NOTE 2 – Distinction may be made between the semi-loop loss of a given piece of equipment and the semi-loop loss of a national system. The latter is measured at equi-level points in an ISC which serves as a national gateway exchange.

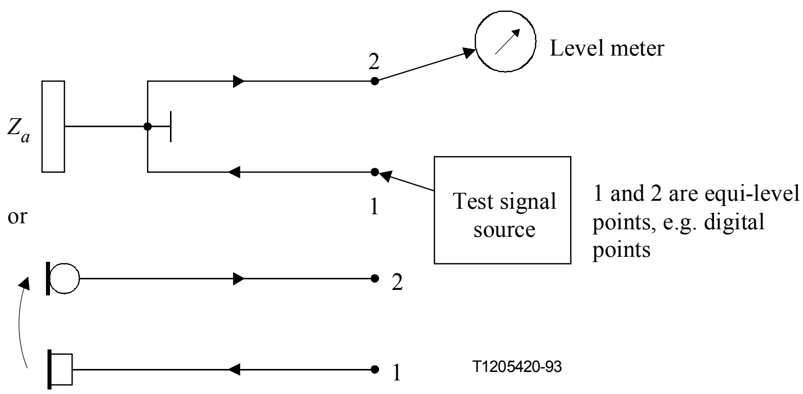



**Figure 3/G.100**

### 4.19 receive loudness rating (RLR)

The loudness loss between an electric interface in the network and the listening subscriber's ear. (The loudness loss is here defined as the weighted (dB) average of driving e.m.f. to measured sound pressure.)

### 4.20 round-trip delay (DL)

The delay in ms around the closed 4-wire loop, determined primarily by the two-way delay of the 4-wire transmission path, which is equal to listener echo path delay.

### 4.21 send loudness rating (SLR)

The loudness loss between the speaking subscriber's mouth and an electric interface in the network. (The loudness loss is here defined as the weighted (dB) average of driving sound pressure to measured voltage.)

### 4.22 sidetone masking rating, talker's sidetone (STMR)

The loudness loss between a subscriber's mouth and his (earphone) ear via the electric sidetone path (see ITU-T P.10 for a full definition).

### 4.23 singing margin (SM)

The minimum listener echo loss in dB over the frequency band involved.

### 4.24 stability loss

*F: affaiblissement pour la stabilité*

*S: atenuación para la estabilidad*

The lowest value of the semi-loop loss in the frequency band to be considered.

### 4.25 talker echo

*F: écho pour le locuteur*

*S: eco para el hablante*

Echo produced by reflection near the listener's end of a connection, and affecting the talker.

### 4.26 talker echo loudness rating (TELR); overall loudness rating of the echo path

The sum of the sending loudness rating and receiving loudness rating of the talker's national system, twice the LR of the international chain, and the echo loss (*a-b*) of the listener's national system. Points *a* and *b* are shown in ITU-T G.122 (See 4.2/G.122 and Figure 2/G.131).

### 4.27 telephone circuit loss (see ITU-T G.101)

*F: équivalent d'un circuit téléphonique*

*S: atenuación del circuito telefónico*

This is the composite loss at the reference frequency 1020 Hz between the circuit input and its output, as defined in the Note below. This will include any loss in the associated terminating equipment of the switching centres (see 2.1/G.101).

NOTE – Defined for transmission planning purposes, the input and output of a circuit are hypothetical points in an exchange where circuits are directly connected (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 ITU-T M.565; their relation to the circuit input and output are shown in Figures 1a)/M.565 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).

### 4.28 terminal coupling loss (TCL); weighted terminal coupling loss (TCLw)
(see ITU-T P.30)

*F: équivalent (pondéré) de couplage du terminal (TCL, TCLw)*

*S: atenuación por acoplamiento del terminal (TCL); ponderada TCL (TCLw)*

The (frequency-dependent) coupling loss between the receiving port and the sending port of a terminal due to:

– acoustical coupling at the user interface;

– electrical coupling due to crosstalk in the handset cord or within the electrical circuits;

– seismic coupling through the mechanical parts of the terminal.

NOTE 1 – The receiving port and the sending port of a digital voice terminal is a 0 dBr point.

NOTE 2 – The coupling at the user interface will depend on the conditions of use.

NOTE 3 – Weighted terminal coupling loss should use the weighting of ITU-T G.122.

### 4.29 test balance return loss (TBRL)

*F: affaiblissement d'équilibrage en position de mesure (TBRL)*

*S: atenuación de equilibrado en posición de medida (TBRL)*

The *balance return loss* measured against a test impedance (i.e. in this case the impedance $Z_2$ – see definition of *balance return loss* – is a specified test impedance).

NOTE – The TBRL characterizes the precision of the balance network.

### 4.30 transmission time; total transmission time (TTT) (see ITU-T G.114)

*F: temps (total) de transmission (TTT)*

*S: tiempo (total) de transmisión (TTT)*

Time between the emission of a signal and the time it is received.

NOTE 1 – (Total) transmission time for connections with digital segments includes delay due to equipment processing as well as propagation delay itself.

NOTE 2 – In the earlier version of ITU-T G.114 (*Blue Book*, 1989) the term "propagation time" was used, both for cable or satellite delay and digital equipment delay (transcoders, transmultiplexers, switches, etc.).

### 4.31 weighted listener echo path loss (WEPL)

WEPL is the weighted mean value of listener echo loss expressed by the following equation:

$$WEPL = -20\log_{10}\frac{1}{3200}\int_{200}^{3400} 10^{-\frac{EPL(f)}{20}}\,df$$

where:

EPL ($f$) magnitude of listener echo loss in dB at the frequency $f$.

This concept was originally used in North America, in the transmission rating model, which can be used to derive the subjectively equivalent effects of listener echo on voice transmission performance regardless of the frequency response of the listener echo loss in the connection.

### 4.32 weighted terminal coupling loss

*F: équivalent pondéré de couplage du terminal*

*S: atenuación ponderada por acoplamiento del terminal*

(See "terminal coupling loss".)

## 5 Equipment

### 5.1 R or T pads (in telephone extension)

*F: compléments de ligne R et T (dans un système national)*

*S: atenuadores R o T (en la prolongación telefónica)*

The R or T pad represents the transmission loss between the 0 dBr points at the digital/analogue codec and the 2-wire side of the 2-wire/4-wire terminating unit or the same in the reversed direction, respectively.

NOTE – The transmission loss introduced by the combination of the R and T pads is the subject of other ITU-T Recommendations.

## ANNEX A

## The units dB, dBm, dBmp, dBr, dBm0 and dBm0p

### A.1 Introduction

This annex will give an overview of some dB-related parameters as used in speech-band applications.

In transmission engineering, most often it would be rather impractical to characterize the magnitude of signals directly by a numerical value in volts or watts. Instead, a logarithmic measure is used, expressed in "dB", to characterize the signal magnitude in relation to some chosen reference value. Designations commonly used are "power level difference", "voltage level difference", etc., all expressed in "dB". A level difference from a standard situation is described simply as "level". Loss and gain are also measured in "dB".

Another important concept is the "reference signal", consisting of a defined analogue sinusoidal signal or its digital equivalent.

### A.2 Fundamentals about dB, loss and gain

The "dB" is a very practical unit which can be used in many different applications.

Comparing two signal powers $P_1$ mVA and $P_2$ mVA, $P_1$ is said to be at an $L$ dB higher (power) level than $P_2$, where:

$$L = 10 \cdot \log \frac{P_1}{P_2} \text{ dB} \tag{A.2-1}$$

Comparing two voltages, $V_1$ volts and $V_2$ volts, $V_1$ is said to be at an $L$ dB higher (voltage) level than $V_2$, where:

$$L = 20 \cdot \log \frac{V_1}{V_2} \quad \text{dB} \tag{A.2-2}$$

Note that "power" depends on the square of "voltage", hence the coefficient 10 in equation (A.2-1) and 20 in equation (A.2-2).

Equation (A.2-2) is also used for quantities other than volts, for instance, currents, acoustic pressure, etc. Note that the term $(V_1/V_2)$ must be a dimensionless quantity. This is automatically fulfilled when $V_1$ and $V_2$ represent two amplitudes of the same kind. Otherwise, $V_1$ and $V_2$ must each be referred to specific reference values of the proper dimension. (For instance, the send sensitivity of a telephone set is described as the relation between the input pressure in Pascal and the output voltage in volts, expressed as "dB rel. 1V/Pa".)

Of course, the unit "dB" is also used to characterize loss or gain (of power or voltage) in a system.

Figure A.2-1 shows how a voltage loss may be defined and calculated. The voltage loss is equal to the voltage level difference between ports a) and b).

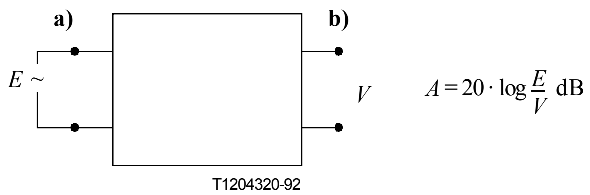



**Figure A.2-1/G.100 – Example of voltage loss from port a) to port b)**

A special case is the return loss $A_r$ which gives a measure of the mismatch between two impedances $Z_1$ and $Z_2$. ($A_r$ can be described as the voltage loss between the incident and the reflected signal at the point of mismatch.) The expression for $A_r$ is:

$$A_r = 20 \log \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| \tag{A.2-3}$$

For passive, reciprocal two-ports (like analogue, passive filters) it has been found practical to base the loss concept on the power level difference between the so-called apparent powers at the input and the output of the two-port. (It can be shown that for such types of circuits this definition of loss results in the same loss for both directions of transmission.)

Figure A.2-2 depicts the configuration.

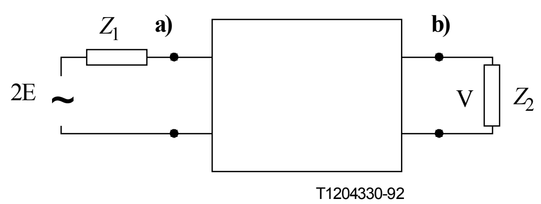



**Figure A.2-2/G.100 – Example for apparent power loss calculation**

Note that the signal generator in Figure A.2-2 produces a single-frequency tone.

The reference apparent power $P_1$ from the generator is defined to be obtained when the load is equal to the generator impedance $Z_1$. With the designation $P_2$ for the output apparent power we get:

$$P_1 = \frac{E^2}{|Z_1|} \qquad\qquad P_2 = \frac{V^2}{|Z_2|} \tag{A.2-4}$$

Thus, the (apparent power) loss is:

$$A = 10 \cdot \log \frac{P_1}{P_2} = 20 \log\left( \sqrt{\left|\frac{Z_2}{Z_1}\right|} \cdot \frac{E}{V} \right) \tag{A.2-5}$$

However, in telephone networks the transmission chain consists of cascaded units which contain amplifiers and 4-wire loops which are non-reciprocal and therefore the loss concept in equation (A.2-5) needs some modification in order to remain practical.

As long as the impedances $Z_1$ and $Z_2$ are real and constant with frequency, equation (A.2-5) is still used as a definition of loss. "Apparent power" (expressed in mVA) is in this case equal to "active power" (expressed in mW).

When one or both of the impedances are complex and varying with frequency, the transfer of "apparent power" at different frequencies is not an adequate measure of circuit performance. One of the reasons why is that the active components in the chain react on input voltage, not on apparent power.

Therefore, a circuit in accordance with Figure A.2-2 is defined as having a flat frequency response when:

$$20 \cdot \log \frac{E}{V} = \text{constant} \tag{A.2-6}$$

irrespective of how the (given) impedances $Z_1$ and $Z_2$ vary with frequency.

To retain the coupling to the power concept, the nominal loss $A_0$ is defined as the apparent power loss at a reference frequency $F_0$ = 1020 Hz as follows:

$$A_0 = 20 \cdot \log \frac{E(F_0)}{V(F_0)} \sqrt{\left|\frac{Z_2(F_0)}{Z_1(F_0)}\right|} \tag{A.2-7}$$

Thus, the frequency-dependent loss of a circuit in accordance with Figure A.2-2 is defined as:

$$A(f) = 20 \cdot \log \frac{E(f)}{V(f)} \sqrt{\left|\frac{Z_2(F_0)}{Z_1(F_0)}\right|} \tag{A.2-8}$$

The losses of cascaded units can be added to get the total overall loss of the chain provided the impedance mismatching at the interconnection points is reasonably small.

NOTE 1 – These loss definitions also apply for electro-acoustic parameters, such as telephone set sensitivities. In this case, however, for the send characteristics the input voltage in volts is divided by the output sound pressure in Pascal, and vice-versa for the receive characteristics. (Corrections are to be applied if the nominal impedance is not 600 ohms.)

NOTE 2 – The concept of apparent power at a frequency different from the reference frequency 1020 Hz is irrelevant.

NOTE 3 – The receive characteristic of a telephone set is usually rather flat with frequency within the transmitted speech band. The send characteristic often has a pronounced pre-emphasis at the high end of the frequency band.

## A.3 The use of a reference signal and dBm, dBmp

The concept of a "reference signal" sent through the network is very useful to visualize the signal transmission in general.

In the analogue parts of the network, the defined reference signal is a tone of the frequency 1020 Hz, the reference frequency $F_0$. Its magnitude is determined in such a way that it would have an apparent power value of 1 mVA at a certain level reference point. (Note that instead of mVA, ITU-T has traditionally used the designation mW.)

A level reference point may exist physically or only fictitiously. How it is located within an equipment or a circuit will be discussed in A.4.

A level reference point in the analogue part of the network has in general a complex nominal impedance $Z_n$ whose modulus $|Z_n|$ varies with frequency. Thus, in this reference point, the voltage of the reference signal is:

$$V(F_0) = \sqrt{0.001 \cdot |Z_n(F_0)|} \quad \text{volts} \qquad \text{(A.3-1)}$$

where $|Z_n(F_0)|$ is given in ohms.

NOTE – In earlier systems, the nominal impedance at an analogue level reference point was always resistive and constant with frequency. However, the modern trend is to use complex impedances in the 2-wire parts of the network.

The reference signal is said to have an absolute level of 0 dBm at the level reference point. (Note that actual test signals most often are specified at 10 dB lower levels than this reference signal.)

In a digital path the reference signal corresponds to a special case of the PCM digital reference sequence, the DRS, namely with the frequency 1020 Hz.

The unit dBm is also used to characterize the absolute level of a tone of a frequency which is different from the reference frequency $F_0$. If the absolute level of the signal is stated to be $L$ dBm at a point of nominal impedance $Z_n$, the voltage is defined to be:

$$V(f) = \sqrt{0.001 \cdot |Z_n(F_0)|} \cdot 10^{\frac{L}{20}} \quad \text{volts} \qquad \text{(A.3-2)}$$

Note especially that the modulus of the nominal impedance in equation (A.3-2) is always to be taken at the reference frequency $F_0$. (This is in accordance with the principle previously mentioned in A.2.)

How shall the magnitude of complex signals be evaluated properly (i.e. signals having a broad spectrum instead of a single tone)?

We will first discuss the case of the signal working on a resistive, constant impedance. For FDM systems, the performance is affected by the total power injected in the channels. As the FDM voice-band channel input impedances are designed to be resistive = $R$, the power is determined simply by a voltage-square-average, divided by the input resistance $R$:

$$P = \frac{1}{F_2 - F_1} \cdot \int_{F_1}^{F_2} \frac{V^2(f)}{0.001 \cdot R} \cdot df \quad \text{mW} \qquad \text{(A.3-3)}$$

where:

$V(f)$ = spectral voltage/$\sqrt{\text{Hz}}$

$R$ in ohms

$F_1$, $F_2$ in Hz, the band-limits of the signal.

The result can thus be expressed as an absolute level in dBm, i.e. in this case dB relative to an active power of 1 mW.

$$L = 10 \cdot \log P \text{ dBm} \qquad \text{(A.3-4)}$$

When the dBm-value of a voice signal acting on a constant-resistance load is calculated in this way, a fairly accurate predication can be made of many parameters, for instance, peak voltages and their statistical distribution with time.

In modern voice-band equipment, like digital exchanges, however, the signals pass interfaces with complex nominal impedances. The transfer is made on a voltage-basis as mentioned, and the active elements are sensitive to voltage, not power. The proper signal magnitude evaluation thus must also be based on voltage. To retain the principles applied for the FDM case, the signal "magnitude measure" is taken to be a voltage-square-average, but divided by the modulus of the nominal complex impedance $Z_n(F_0)$ at the reference frequency $F_0$.

$$P = \frac{1}{F_2 - F_1} \cdot \int_{F_1}^{F_2} \frac{V^2(f)}{0.001 \cdot |Z_n(F_0)|} \cdot df \qquad \text{mVA} \qquad \text{(A.3-5)}$$

The corresponding level is given by equation (A.3-4).

Note that $P$ in equation (A.3-5) also has the dimension of mVA or of mW. Therefore, the magnitude of a complex signal sometimes is stated in mW or pW on the basis of equation (A.3-5). This is quite useful for noise signals, because the pWs of uncorrelated signals can be added to get the total pW. (Note, however, that this power concept has nothing to do with apparent power.)

The magnitude of normal voice signals can be measured by means of special instruments. Formerly it was the practice to use the so-called VU-meter. Now instruments according to ITU-T P.56 are preferred. (Both these types are based on voltage-square evaluation.) From the instrument readings, such properties as long- and short-time power, peak values, etc. may be determined.

When an electric signal is transformed into acoustic pressure by the telephone receiver, the human hearing characteristics must be taken account of in order to determine the proper signal magnitude the listener perceives. For noise signals, this is done by adding a psophometric weighting $W(f)$ dB, which is specified in ITU-T O.41. (Note that the weighting includes the response of a "typical" telephone receiver, pressed hard against the listener's ear, i.e. the receiver's frequency response is quite flat within the speech band up to about 3.4 kHz, where band-limiting begins.)

The corresponding psophometric power is:

$$P = \frac{1}{F_2 - F_1} \cdot \int_{F_1}^{F_2} \frac{V^2(f)}{0.001 \cdot |Z_n(F_0)|} \cdot 10^{\frac{W(f)}{10}} \cdot df \qquad \text{mVA} \qquad \text{(A.3-6)}$$

Here, $F_1$ = 16.66 Hz, $F_2$ = 6000 Hz.

The absolute psophometric level is designated dBmp:

$$L_p = 10 \cdot \log P_p \qquad \text{(A.3-7)}$$

An instrument performing a psophometric weighting, including a certain time constant, is termed "psophometer", the performance of which is specified in ITU-T O.41.

In transmission planning it is important to know the electro-acoustic losses voice signals are subjected to when passing through the network. These losses are termed "loudness ratings" and are also measured in dB. Note, however, that it is not appropriate to determine loudness ratings as a difference in readings of speech levels (volumes), using a VU-meter, a P.56-instrument or a psophometer. The reason is that for loudness rating the signal weighting is different from the one used for speech level evaluation. For loudness ratings the weighting depends on the voice signal level and is made over an approximately logarithmic frequency scale. (See ITU-T P.79.)

For voice signals at normal levels the signal weighting is done approximately as a dB average. Send and receive loudness ratings (SLR, RLR) are measured by special instruments, specified in the P-series Recommendations. The circuit loudness rating (CLR), i.e. the loudness loss a typical circuit element like a subscriber cable introduces, is best determined by computation. Note that the nominal loss $A_0$ as defined by equation (A.2-7) turns out to be a good measure of CLR.

For weaker, voice-derived signals, the signal weighting is different. For listener's echoes it is done as a voltage average, for talker's echoes and crosstalk as a voltage-square average. (For brevity, in this context voltage-square additions are sometimes called power addition.)

Further information is found in Annex A/G.111, and in the P-series Recommendations.

## A.4 dBr, dBm0 and dBm0p

### A.4.1 General

Relative level, expressed in dBr, is a very versatile concept by which several transmission properties can be characterized in a convenient way:

– gain or loss between interfaces;

– equipment power handling capacity at an interface;

– probable speech power level at an interface.

The unit dBr is used in equipment design, equipment performance tests, transmission planning and network maintenance.

Relative level in dBr is the level a reference sinusoidal signal of 1020 Hz would have at the point in question relative to the level the reference signal would have at its unique level reference point, called a 0 dBr point.

The precise rules of application for "relative level" differ somewhat between equipment design and performance testing on one hand and transmission planning and maintenance on the other. The reason why is that in the process of designing and testing, an equipment is to be judged as a separate entity while in transmission planning and maintenance the main concern is the transmission quality of real signals through the network.

This affects the choice and allocation of level reference points and the physical limits within which a certain set of relative levels are related to the same level reference point. (The matter will be discussed in A.4.2 and A.4.3.) Thus, an equipment test point may have a certain relative level when the equipment is specified on its own but the same point may be assigned a different relative level when the equipment is considered as part of a transmission chain.

Relative levels can evidently be used to determine loss or gain between points in a signal path having the same level reference point.

The designation "dBm0" is the level in dBm an actual signal would have when passing through a 0 dBr point. For instance, if a signal is characterized as having a magnitude of $L_o$ dBm0, its absolute level at a point of relative level $X$ dBr is

$$L_a = L_o + X \text{ dBm}$$

Specifically, the voltage of a 0 dBm0 tone at any voice-band frequency at a point of $X$ dBr is:

$$V(f) = \sqrt{0.001 \cdot |Z_n(F_0)|} \cdot 10^{\frac{x}{20}} \quad \text{volts} \tag{A.4-1}$$

where:

$Z_n(F_0)$ is the nominal impedance at the point in question at the reference frequency $F_0$ = 1020 Hz.

The designation "dBm0p" applies similarly for a psophometrically weighted value.

### A.4.2 The use of dBr and dBm0 in equipment design and performance testing

Here the basis for the choice of a 0 dBr level reference point is the equipment power handling capacity. The physical limits within which this point is valid as a reference is of course the equipment interfaces to the outside.

Large-capacity FDM (carrier) systems are designed to allow in an up-modulated band a long-term average of –15 dBm0 per channel taking into account signalling carrier leaks and speech pauses. Referring to actual speech during active periods this corresponds to –11 dBm0. More detailed information is found in ITU-T G.223, including expected peak values[1]. (Note that FDM systems with fewer than 240 channels must be designed for a higher average power per channel. Thus, a 12-channel FDM system should be able to handle –7.5 dBm0 per channel.)

For equipment using 64 kbit/s PCM, the power handling capacity is directly connected with the coders and the decoders. (For the definition of the PCM digital reference sequence, the DRS (see ITU-T G.101).)

The PCM maximum coding level corresponds to clipping of a sinusoidal signal at +3.14 dBm0 for ideal coders and decoders using the A-law and at +3.17 dBm0 for the μ-law.

The procedure to determine the relative levels at real coders and decoders, when the digital path does not contain any digital processing, is illustrated in Figure A.4-1.

[1] For corresponding test levels, see ITU-T G.228.

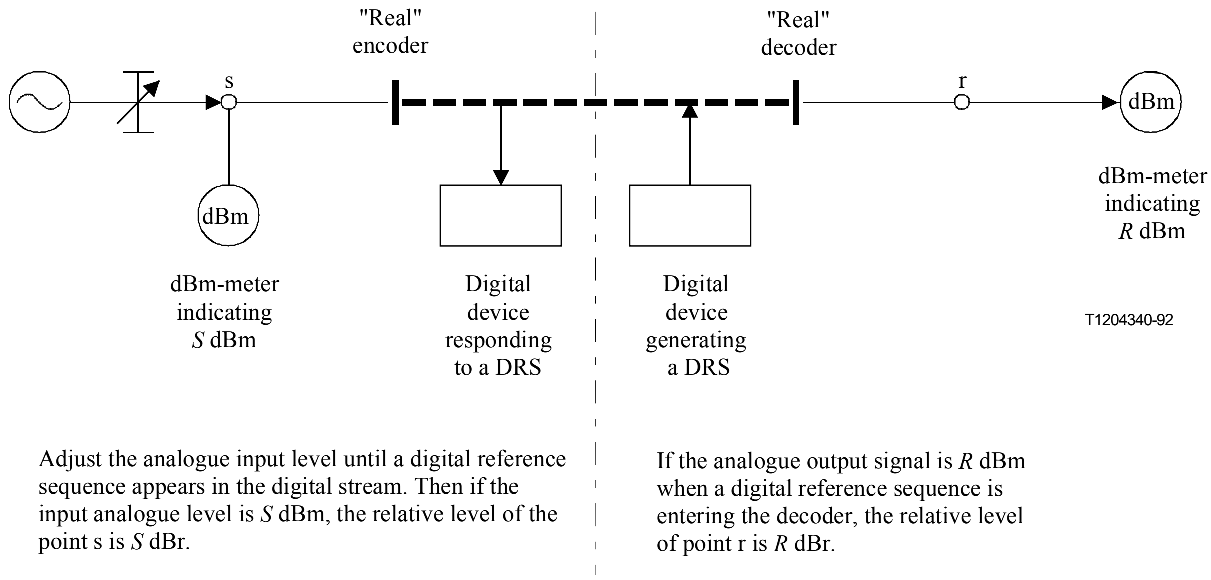



**Figure A.4-1/G.100 – Set-up for determining the relative level at the input and output analogue points of a "real" codec using the DRS at 1020 Hz (The digital path is free from digital signal processing)**

As long as the digital bit stream is not subjected to a digital signal processing, the digital path is by convention designated to be at "0 dBr" in the design and performance testing of equipment.

The signal handling performance of an equipment can then be adequately described by means of the signal level in dBm0 as a parameter. (Examples of this can be found in ITU-T G.712-ITU-T G.714 and ITU-T Q.551-ITU-T Q.553.)

If digital gain or loss is introduced, however, the conventions have to be modified. Although digital gain or loss should be used only exceptionally in a network, valid reasons to apply this form of digital signal processing could be complex interconnection cases with local exchanges, PABXs, leased lines, etc. Note also that the signal performance will be affected, which must be remembered when compliance tests are specified. (However, in most cases of compliance testing it is preferable to deactivate the digital signal processing.)

Evidently, a digital loss pad or amplifier will diminish the usable dynamic range for signals as well as increase the quantizing distortion. (For this and other reasons, the general use of digital gain or loss is depreciated. See, for instance, 3.1.4/Q.554.)

The most important parameter to be described by a dBr-designation, however, appears to be the clipping level. Therefore, the relative level conventions shown in Figure A.4-2 are chosen in most cases. In this way, a digital bit stream will never be associated with a relative level higher than 0 dBr in an equipment specification.

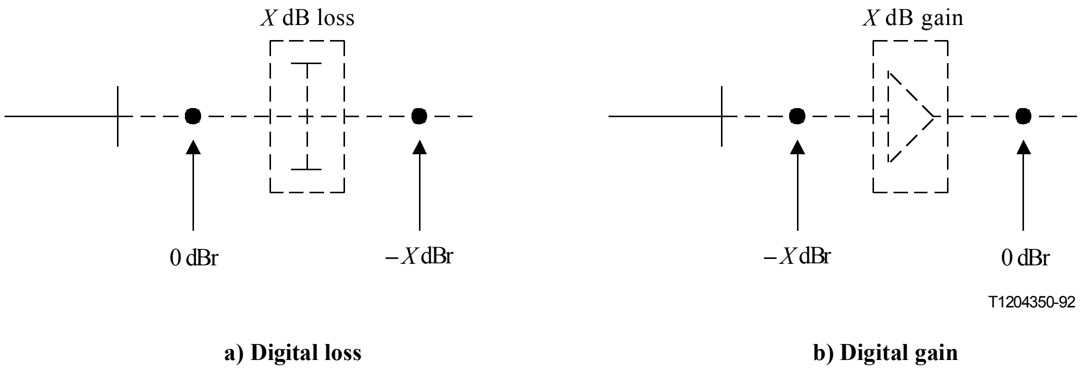


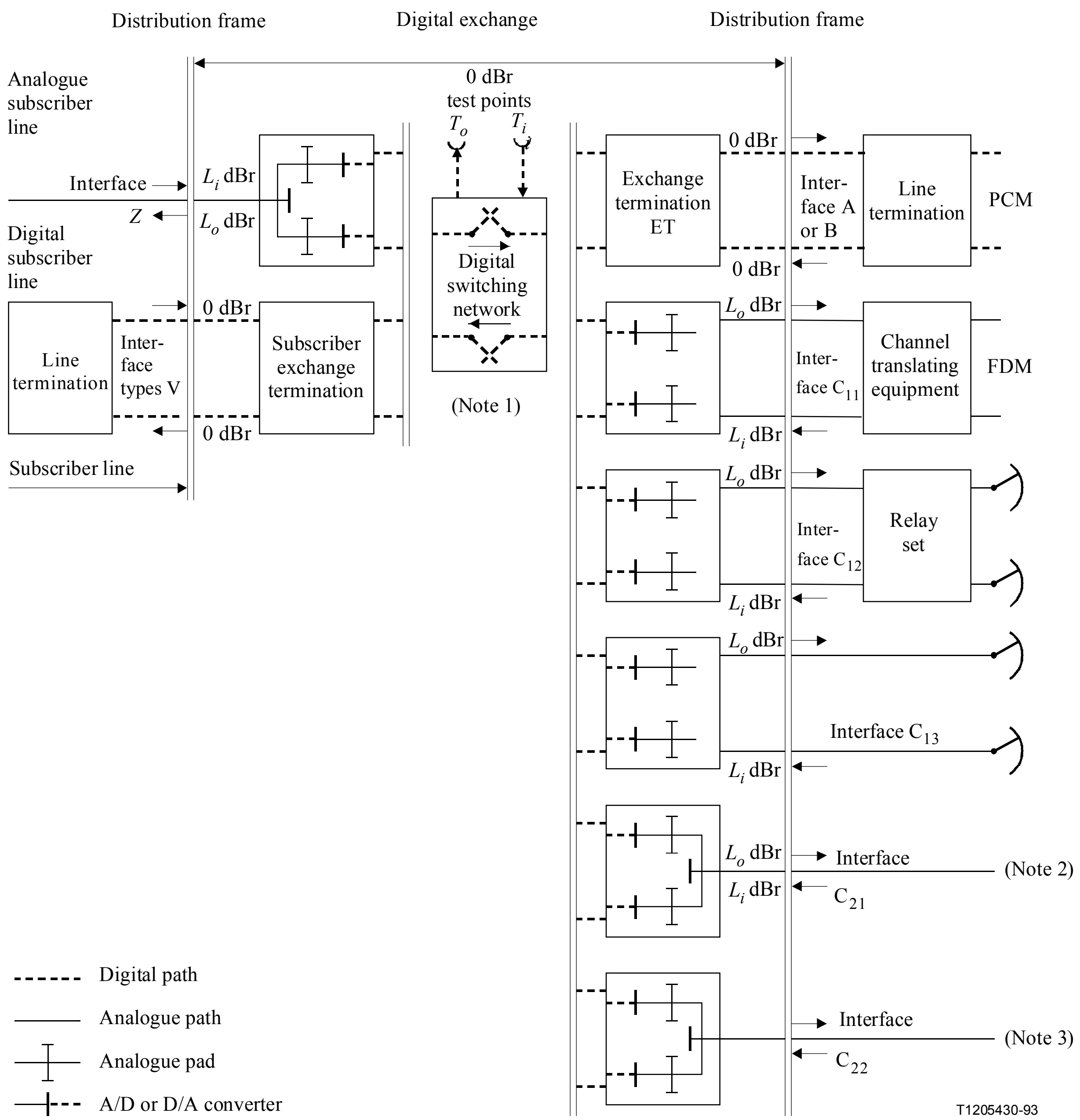

**Figure A.4-2/G.100 – Conventions for relative level of a digital bit stream when digital signal processing is applied**

ITU-T Q.551-ITU-T Q.554 apply to transmission parameters for digital exchanges in half-channel forms. In Figure A.4-3, which is a reproduction of Figure 1/Q.551, the exchange test points $T_o$ and $T_i$ for the half-channels are identified as 0 dBr level reference points. This represents, of course, the general case when specifying the exchange performance for acceptance tests. However, one must observe the following points:

NOTE 1 – If a digital loss or gain pad is located in the digital switching network as a separate unit, $T_o$ and $T_i$ may be retained as 0 dBr level reference points in the specifications for the half-channels. However, if this convention is applied, the expression for nominal loss through the exchange (see 1.2.4.1/Q.551) should be modified to include the switching loss SL, i.e. the loss of the digital pad:

$$NL = L_i - L_o + SL \qquad \text{(A.4-2)}$$

NOTE 2 – The exchange test points may be assigned relative levels different from 0 dBr in a transmission plan as discussed in 4.3. For instance, in a mixed analogue/digital circuit, $T_i$ = 0 dBr, $T_o$ = –0.5 dBr.




NOTE 1 – Digital loss pads, if required, may be located in the digital switching network or the exchange terminals (see 1.2.4.1/Q.551).
NOTE 2 – Termination of international long distance switched connections.
NOTE 3 – Termination of local or 2-wire trunk switched traffic.
NOTE 4 – The values of $L_i$ and $L_o$ for 2-wire and 4-wire interface are, in general, not equal.
NOTE 5 – This figure shows typical examples utilizing the defined interfaces.

**Figure A.4-3/G.100 – Examples of relative levels as specified for a digital exchange**

### A.4.3 The use of dBr and dBm0 in transmission planning and maintenance

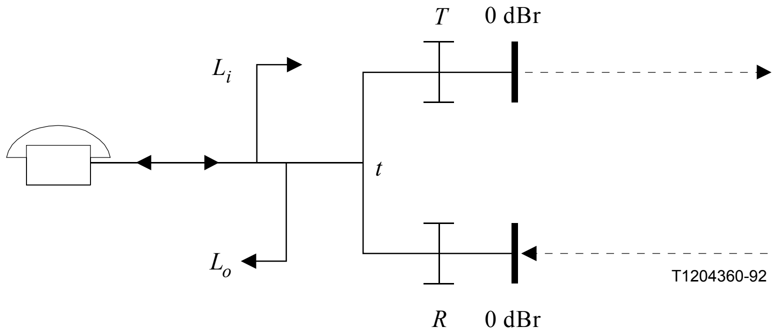
In transmission planning procedures the overall transmission path is divided into sections, termed "circuits", which connect switching centres. Sometimes also the subscriber line connected to a local exchange is termed "circuit". A circuit thus is constituted by all permanently interconnected equipment. In this way maintenance personnel have clearly defined segments with fixed transmission parameters to supervise. See the definition in ITU-T G.101.

The physical limits of a circuit are sometimes expressed as being situated at "the middle of the exchanges". In this case, the exchange terminating equipment is included in the circuit ending in the exchange test point.

Exceptionally, the "transmission interface" between two different maintenance organizations does not lie at an exchange. This may be the case when a public and a private network are interconnected. To divide the responsibilities clearly, one may designate the public and the private links as belonging to two different circuits by mutual agreement. (This could involve a level jump. See below.)

Regarding the expected speech level, in a 0 dBr point the value of –11 dBm, pauses excluded, is expected, taken as an average for a large number of subscribers as mentioned in A.4.2. However, field measurements of actual speech levels in TRPs would show a very large spread, so instead one resorts to some conventions based on general experience.

For normal telephone sets and subscriber lines, the interconnection to the local exchange can be taken as an "anchor point" to establish a 0 dBr point. See Figure A.4-4, which is a reproduction of Figure C.1/G.121. Of course, the speech levels are influenced by the telephone set sensitivities, i.e. the SLR and RLR values. Nevertheless, from Annex C/G.121 it can be seen that many Administrations have found the optimum values of the pads in Figure A.4-4 to be $T$ = 0 dB, $R$ = 6-7 dB, i.e. $L_i$ = 0 dBr, $L_o$ = –6 or –7 dBr. (Note, however, that these relative levels depend on certain national practices: if sensitivity regulation is performed in the line circuits, in the telephone sets or not at all. See, for instance, 2.2.4.3/Q.552 and 2.6/G.101.)




NOTE – It is assumed that $T$ and $R$ represent all the losses between $t$, the 2-wire point, and the digital bitstreams.

**Figure A.4-4/G.100 – Relative levels at a local exchange $L_i = T$ dBr, $L_o = R$ dBr**

Regarding how the equipment is deployed in the network, in most cases it will be possible to obtain an exact correspondence between the "equipment" and the "circuit" relative levels. Exceptions sometimes have to be allowed, for instance when for stability reasons extra loss is included in a 4-wire loop. Another reason might be lack of suitable level controls in certain equipment. (Also, some echo canceller designs may need an extra margin against clipping.)

An example of additional loss in an analogue 4-wire loop is shown in Figure A.4-5 where an analogue circuit section is interposed between digital circuit sections. To ensure that the risk of instability and "hollowness" of a connection will be insignificant, it is recommended that a 0.5 dB loss is inserted in analogue or mixed digital/analogue circuits. Thus, in the transmission plan for this circuit, part of the digital bit stream will be associated with –0.5 dBr.

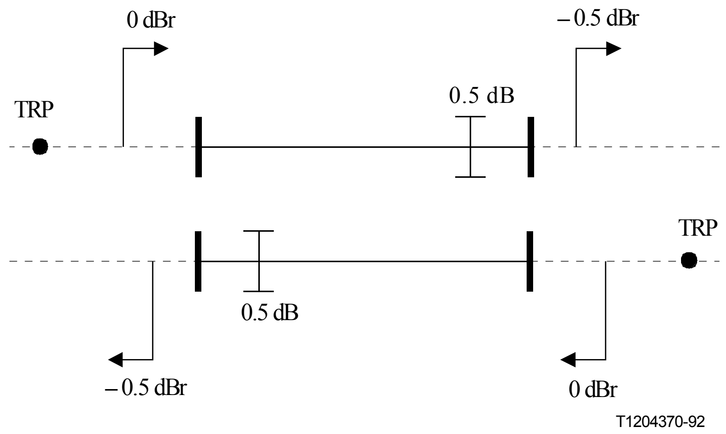



**Figure A.4-5/G.100 – Example of (circuit) relative levels when an analogue link is interposed in a digital chain**

Two adjacent circuits each have their own transmission reference points (TRPs) to which their respective relative levels are referred. Ideally, at the interface between the circuits, the two relative levels should be the same.

Occasionally, the send relative level must be set 0.5 dB lower than the receive level in order to guarantee stability, namely when analogue 4-wire transmission is used. For instance, two local exchanges are interconnected via a primary or transit centre with 4-wire analogue switching and transmission. The net loss in the transit path must be 0.5 dB for stability reasons. The relative levels at the local exchanges are determined by the properties of the telephones as mentioned before. Therefore, the 0.5 dB net transmission loss will correspond to a "level jump" of 0.5 dB at the transit exchange. For a similar example from an international transit connection see Figure A.4-6 which is a reproduction of Figure 2a)/Q.45 *bis*.

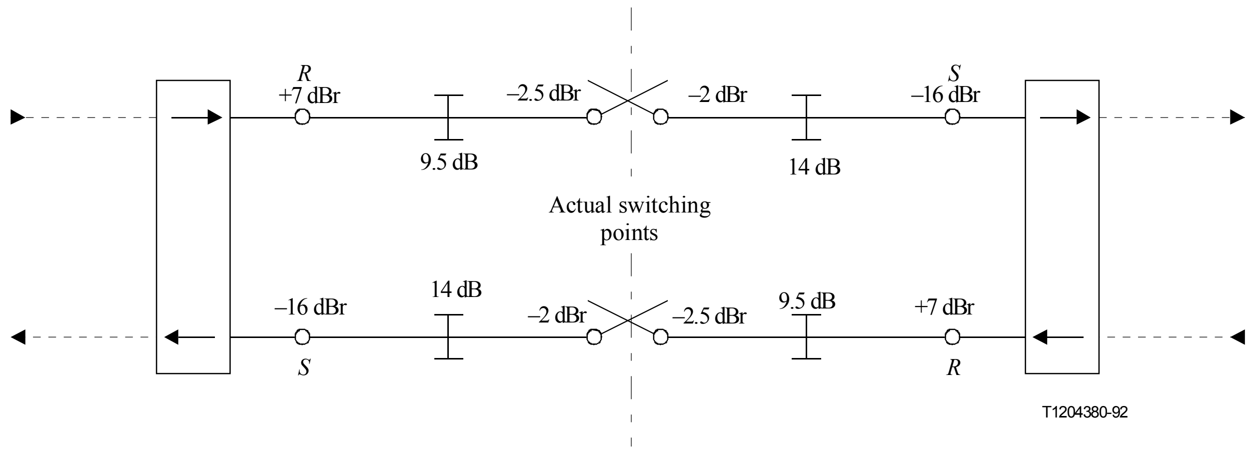



**Figure A.4-6/G.100 – Example showing a (simplified) representation of a transit connection in an international exchange, actual arrangement**

Occasionally, the transmission planner may find it convenient to assign a "level jump" at an interface between a public and a private circuit which is not associated with switching. (Note that such a level jump minimizes the dynamic range and should be as small as possible.)

Note that in general the total loss of a connection made up of several circuits must be determined by adding the losses of the individual circuits, and not by taking differences in relative levels between the input and the output of the connection ports. (The latter method is only valid when all the constituent circuits are digital and not using digital signal processing.)

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