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

International telephone connections and circuits – General definitions

The use of the decibel and of relative levels in speechband telecommunications

ITU-T Recommendation G.100.1

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ITU-T Recommendation G.100.1

The use of the decibel and of relative levels in speechband telecommunications

Summary

This Recommendation provides the definition for different logarithmic power level measurement units in current use in telecommunication systems. This Recommendation also provides the relationship amongst those units and usage examples. The text herein is a merger of information that has been previously dispersed over several publications and supersedes the contents of ITU-T Rec. B.12, of Annex A/G.100, of Annex A/G.101, of Annex B/Q.551, and of 3.8/G.101.

Source

ITU-T Recommendation G.100.1 was prepared by ITU-T Study Group 12 (2001-2004) and approved under the WTSA Resolution 1 procedure on 29 November 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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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

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ITU-T Recommendation G.100.1

The use of the decibel and of relative levels in speechband telecommunications

1 Introduction

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".

Relative levels has been a very useful term in transmission planning for the last 40 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.

Guidance on the use of decibels in the field of sound transmission and radio frequencies can be found in ITU-R Recommendation V.574-4 (05/00) [5].

Notations for expressing the reference of a level can be found in Part 5 of IEC Publication 60027-3 [6].

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 currently valid ITU-T Recommendations is regularly published.

[1] ITU-T Recommendation G.100 (2001), *Definitions used in Recommendations on general characteristics of international telephone connections and circuits*.

[2] ITU-T Recommendation G.121 (1993), *Loudness ratings (LRs) of national systems*.

[3] ITU-T Recommendation G.712 (2001), *Transmission performance characteristics of pulse code modulation channels*.

[4] ITU-T Recommendation O.41 (1994), *Psophometer for use on telephone-type circuits*.

[5] ITU-R Recommendation V.574-4 (2000), *Use of the decibel and the neper in telecommunications.*

[6] IEC Publication 60027-3 (2002), *Letter symbols to be used in electrical technology – Part 3: Logarithmic and related quantities, and their units.*

[7] IEC Publication 60651 (2001), *Sound level meters*.

3 Fundamentals about dB

The *bel* (symbol B) expresses *the ratio of two powers* by the decimal logarithm of this ratio. This unit is not often used, having been replaced by the *decibel* (symbol dB) which is one-tenth of a bel.

The decibel may be used to express the ratio of two *field quantities*, such as voltage, current, sound pressure, electric field, charge velocity or density, the square of which in linear systems is proportional to power. To obtain the same numerical value as a power ratio, the logarithm of the field quantity ratio is multiplied by the factor 20, assuming that the impedances are equal.

The relationship between a current or voltage ratio and that of the corresponding power ratio is impedance dependent. Use of the decibel when the impedances are not equal is not appropriate unless adequate information is given concerning the impedances involved.

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

$$\mathrm{L} = 10 \cdot \log \frac{P_1}{P_2} \quad [\mathrm{dB}] \tag{3-1}$$

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

$$\mathrm{L} = 20 \cdot \log \frac{V_1}{V_2} \quad [\mathrm{dB}] \tag{3-2}$$

Note that "power" depends on the square of "voltage", hence the coefficient 10 in equation (3-1) and the coefficient 20 in equation (3-2).

Equation 3-2 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".)

3.1 Loss and gain

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

Figure 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).

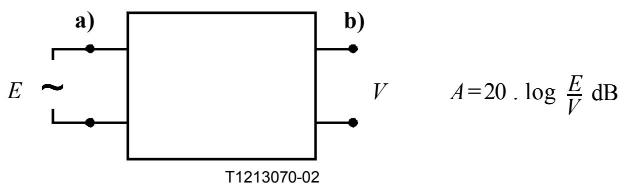


Figure 1/G.100.1 – Example of voltage loss from port a) to port b)

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:

$$A_r = 20 \cdot \log \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| \quad [\mathrm{dB}] \tag{3-3}$$

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 2 depicts the configuration.

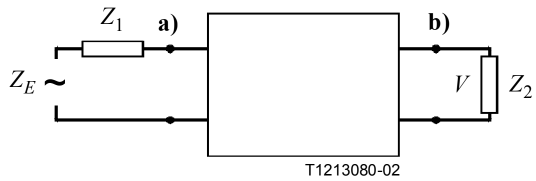


Figure 2/G.100.1 – Example for apparent power loss calculation

Note that the signal generator in Figure 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:

$$P_1 = \frac{E^2}{|Z_1|} \qquad P_2 = \frac{V^2}{|Z_2|} \tag{3-4}$$

Thus, the (apparent power) loss is:

$$A = 10 \cdot \log \frac{P_1}{P_2} \ [\mathrm{dB}] = 20 \cdot \log \sqrt{\left|\frac{Z_1}{Z_2}\right|} \cdot \frac{E}{V} \ [\mathrm{dB}] \tag{3-5}$$

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 (3-5) 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 (3-5) 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 2 is defined as having a flat frequency response when

$$20 \cdot \log \frac{E}{V} = \text{ constant} \tag{3-6}$$

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:

$$A_0 = 20 \cdot \log \frac{E(F_0)}{V(F_0)} \sqrt{\left|\frac{Z_2(F_0)}{Z_1(F_0)}\right|} \ [\mathrm{dB}] \tag{3-7}$$

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

$$A(f) = 20 \cdot \log \frac{E(f)}{V(f)} \sqrt{\left|\frac{Z_2(F_0)}{Z_1(F_0)}\right|} \quad [\text{dB}] \tag{3-8}$$

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.

3.2 The letter "p" in "dBmp" and "dBm0p"

The additional small letter "p" is derived from the French word "pondéré" for "weighted" and means that the considered value is a noise level, measured by a psophometer with a special noise weighting filter included as described in ITU-T Rec. O.41 [4].

3.3 Correction factors

Depending on the type of test instruments, auxiliary equipment and test objects, sometimes correction factors need to be used, to either adjust the correct test signal level, or to obtain the correct test result. This mainly occurs in conjunction with capacitive complex impedances.

In practice, test instruments may be used with input/output impedances only 600 Ω resistive and consequently send levels or displayed results referred to 1 mW. To provide the correct termination of test objects with complex impedances, auxiliary equipments called "impedance converter" are used. The principle of such an impedance converter is shown in Figure 3 in the application for sending and in Figure 4 for receiving.

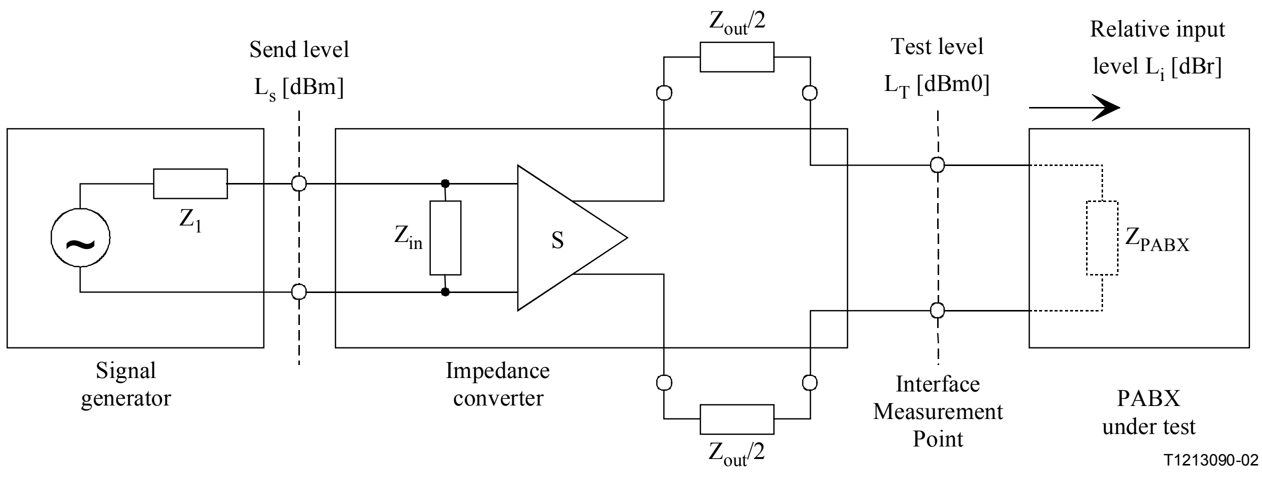


Figure 3/G.100.1 – Impedance converter in the sending path

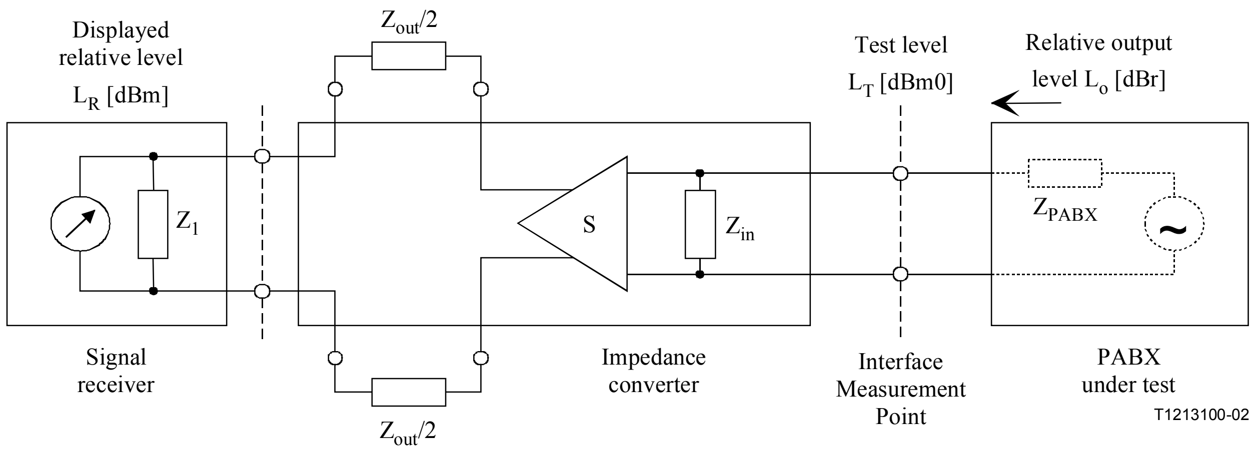


Figure 4/G.100.1 – Impedance converter in the receive path

An advantageous design is to obtain a power transfer ratio of 1 at the reference frequency 1020 Hz, if terminated with the respective nominal impedances at input and output. In this case the voltage gain "s" of the inserted amplifier is:

$$s = 6 + 10 \cdot \log \frac{Z_{out}}{Z_{in}} \ \ [\mathrm{dB}] \tag{3-9}$$

This formula is valid for send and receive part of an impedance converter. It should be noted, that if Z_{out} or Z_{in} is a complex impedance, the modulus at the reference frequency 1020 Hz has to be used.

For impedance converters in the application with different complex impedances the gain "s" is normally only adjusted to 6 dB (power transfer ratio = 1 only, if $Z_{in} = Z_{out}$) and correction values are used as follows:

3.3.1 Sending a test signal

In this application (see Figure 3) Z_{in} is exactly matched to the impedance Z_i of the signal generator (e.g. 600 Ω) and Z_{out} is the nominal value of the interface impedance ZPBX of the PBX under test.

To obtain the required test level L_T in dBm0 at the IMP, the necessary send level L_S in dBm of the signal generator can be calculated as follows:

$$L_S \ \ [\mathrm{dBm}] = L_T \ \ [\mathrm{dBm0}] + L_i \ \ [\mathrm{dBr}] + 10 \cdot \log \frac{Z_{out}}{Z_{in}} \tag{3-10}$$

Example 1: For an interface of the PBX under test with an input relative level L_i = –5 dBr and a nominal impedance ZPBX = 842 Ω (modulus at 1020 Hz for a 3-element complex impedance with 270 Ω + 750 Ω // 150 nF) a test level of L_T = –10 dBm0 shall be provided. What is the necessary send level L_S in dBm at a signal generator with 600 Ω impedance? From equation (3-10) results the following:

$$L_S = -10\,\mathrm{dBm0} + (-5\,\mathrm{dBr}) + 10 \cdot \log \frac{842}{600}$$

$$L_S = -13.53\,\mathrm{dBm}$$

3.3.2 Receiving a test signal

For receiving (see Figure 4), Z_{out} is exactly matched to the instrument impedance Z_i, and Z_{in} provides the nominal termination of the IUT with the impedance Z_{PBX}.

To obtain the correct (received) test level L_T in dBm0 at the IMP, the displayed receive level L_R in dBm at the signal receiver needs to be corrected, using the following formula:

$$L_T[\text{dBm0}] = L_R[\text{dBm}] - L_o[\text{dBr}] + 10 \cdot \log \frac{Z_{out}}{Z_{in}} \tag{3-11}$$

Example 2: Assuming the same impedances for the test instrument (600 Ω) and the PBX under test (842 Ω) as in example 1, but with an output relative level of $L_o = -7$ dBr, what is the correct received test level L_T if the signal receiver readout is $L_R = -50$ dBm? From equation (3-11) results the following:

$$L_T = -50 \text{ dBm} - (-7 \text{ dBr}) + 10 \cdot \log \frac{600}{842}$$

$$L_T = -44.47 \text{ dBm0}$$

3.4 Signal-to-noise ratio

This is either the ratio of the signal power *(P_S)* to the noise power (P_N), or the ratio of the signal voltage *(U_S)* to the noise voltage (U_N) measured at a given point with specified conditions. It is expressed in decibels:

$$R = 10 \cdot \log\left(\frac{P_S}{P_N}\right) \ [\text{dB}] \qquad \text{or} \qquad R = 20 \cdot \log\left(\frac{U_S}{U_N}\right) \ [\text{dB}] \tag{3-12}$$

The ratio of the wanted signal to the unwanted signal is expressed in the same way.

3.5 Sound pressure level

This is the logarithm, generally expressed in decibels, of the ratio of sound pressure and a reference pressure, often 20 µPa but typically referenced to 1 Pa in telephonometry. Usually the sound pressure when referenced to 20 µPa is called dB_{SPL}.

Example:

15 dB(20 µPa) or 15 dB_{SPL}

As acoustic power is linked to the square of sound pressure, this means:

20 log (p / 20 µPa) =15 $dB_{(20\ \mu Pa)}$

In the ratio (*p*/20 µPa) or (*p*/1 Pa), it is evident that both sound pressures must be expressed in the same units.

Often the sound pressure level is weighted in order to take into account the human ear sensitivity in frequency. For absolute acoustic pressure level, dB_{SPL}(A) [or dB_{SPL}(B), dB_{SPL}(C)] refers to the weighted acoustic pressure level with respect to 20 µPa, mentioning the weighting curve used (curves A, B or C, see [7]). The same weighting can be applied as well when referring the sound pressure to 1 Pa [e.g. dBPa(A)].

4 The use of a reference signal

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 clause 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:

$$V(F_0) = \sqrt{0.001 \cdot |Z_n(F_0)|} \quad [\mathrm{V}] \tag{4-1}$$

$$Z_n(F_0) \quad [\Omega]$$

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 levels 10 dB lower 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:

$$V(f) = \sqrt{0.001 \cdot |Z_n(F_0)|} \cdot 10^{\mathrm{L}/20} \quad [\mathrm{V}] \tag{4-2}$$

Note especially that the modulus of the nominal impedance in equation (4-2) is always to be taken at the reference frequency F_0. (This is in accordance with the principle previously mentioned in clause 3.)

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:

$$P = \frac{1}{F_2 - F_1} \cdot \int_{F_1}^{F_2} \frac{V^2(f)}{0.001 \cdot R} \cdot \mathrm{d}f \quad [\mathrm{mW}] \tag{4-3}$$

where:

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

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.

$$L = 10 \cdot \log P \quad [\mathrm{dBm}] \tag{4-4}$$

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 voiceband 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.

$$P = \frac{1}{F_2 - F_1} \cdot \int_{F_1}^{F_2} \frac{V^2(f)}{0.001 \cdot |Z_n(F_0)|} \cdot \mathrm{d}f \;\; [\mathrm{mVA}] \qquad (4\text{-}5)$$

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

Note that P in equation (4-5) also has the dimension of mVA or, as traditionally used in ITU-T, of mW. Therefore, the magnitude of a complex signal sometimes is stated in mW or pW on the basis of equation (4-5). 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 Rec. 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 Rec. 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:

$$P = \frac{1}{F_2 - F_1} \cdot \int_{F_1}^{F_2} \frac{V^2(f)}{0.001 \cdot |Z_n(F_0)|} \cdot 10^{W(f)/10} \cdot \mathrm{d}f \;\; [\mathrm{mVA}] \qquad (4\text{-}6)$$

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

The absolute psophometric level is designated dBmp:

$$L_p = 10 \cdot \log P_p \;\; [\mathrm{dBmp}] \qquad (4\text{-}7)$$

An instrument performing a psophometric weighting, including a certain time constant, is termed "psophometer", the performance of which is specified in ITU-T Rec. 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 Rec. 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 (3-7) 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.

5 Relations between the units "dBm","dBr" and "dBm0"

5.1 General

Transmission values for loss, gain and levels are expressed in decibels (dB) as a general principle. The basic unit "dB" is often extended with additional letters in order to distinguish between its use in different applications. The aim of this clause is to give a short description of the most common forms as used for transmission measurements at speech-band frequencies as well as an introductory explanation of certain transmission planning applications. See also clause 6 for a more complete discussion.

5.2 The unit "dB"

This basic unit is mainly used for losses, gains, return losses, etc., i.e. as a logarithmic ratio between two values which can be voltages, currents, powers, acoustic pressures, etc. If the ratio is X for voltages, currents, pressures, the dB expression is 20 log (X). If the ratio is Y for powers, the dB expression is 10 log (Y).

5.3 The unit "dBm"

This unit with the additional "m" is used as a logarithmic measure of the "magnitude" P of an actual signal. The "dBm value" of a signal is called its "absolute power level" or "absolute level".

The signal magnitude P used for signal characterization in speech-band applications has the dimension of power, i.e. expressed in mW or mVA, and has by definition the form:

$$P = \frac{1000 \cdot V^2}{|Z(f_0)|} \ [\text{mW}] \,\text{or}\, [\text{mVA}] \qquad (5\text{-}1)$$

where:

- V: the rms value in volts of the voltage across the test impedance Z which in the general case is complex and frequency dependent;
- $Z(f_0)$: the value of the test impedance in ohms at the (sinusoidal) reference frequency f_0 = 1020 Hz.

The choice of this definition is based on three conventions:

a) firstly, it is practical to characterize the signal magnitude by a unit that has the dimension of power because this has been the practice for the special case of resistive terminations;

b) secondly, electronic circuits are designed to react on voltages, i.e. the open-circuit output voltage of, for instance, an amplifier depends only upon the voltage across its input terminals, irrespective of the input and output impedances of the amplifier. Thus, the "power" absorbed by the input impedance of the amplifier has no influence on how the signal is amplified. Hence the use of a constant impedance value in the denominator instead of a frequency-dependent impedance;

c) thirdly, a sinusoidal signal with the reference frequency (1020 Hz) the numeric value of *P* shall be equal to the apparent power absorbed by *Z*, when this is complex, which is the same as the active power when *Z* is resistive.

Note that P is equal to the active power absorbed by the test impedance Z only when the latter is purely resistive and constant with frequency, for instance when $Z = 600\ \Omega$. Then P is measured in mW, otherwise in mVA. However, when Z is complex, the value of P does not represent the apparent power absorbed by the test impedance at other frequencies than the reference frequency 1020 Hz.

The definition for the so-called absolute power level L is:

$$L = 10 \cdot \log \frac{P}{P_0} \ [\text{dBm}] \tag{5-2}$$

where:

P: the power in mW to be stated;

P_0: the reference value which is $P_0 = 1$ mW.

Likewise, in the speech band the loss between two analogue points 1 and 2 is defined to be:

$$A(f) = 10 \cdot \log \frac{P_1(f)}{P_2(f)} \ [\text{dB}] = 20 \cdot \log \frac{V_1(f)}{V_2(f)} \sqrt{\frac{|Z_2(f_0)|}{|Z_1(f_0)|}} \ [\text{dB}] \tag{5-3}$$

Sometimes the unit "dBm" is used in conjunction with a voltage level, referred to a voltage of 0.775 V. The use of "dBm" in this application is only correct if the test impedance is 600 Ω resistive since 0.775 V across 600 Ω results in the reference active power of 1 mW. This fact is important to remember if capacitive complex interfaces or test impedances are used.

5.4 The unit "dBr"

This unit is used to characterize "relative levels", i.e. to express the level relations for signals between points in a signal path, with the convention that one of the points is designated as a level reference point with the relative level 0 dBr.

More specifically, a sinusoidal reference signal of 1020 Hz in the speech band is thought to pass the signal path under consideration with such an amplitude that its absolute level is 0 dBm at the 0 dBr point. The relative level in dBr at any other point in this signal path is then equal to the level (in dBm) that the reference signal has at that point. (Note that relative level designations should be used for both transmission directions.)

If the level reference point is digital, normally the reference signal is thought of as being decoded by an ideal decoder at which output terminal a power of 1 mW is produced, termination 600 Ω resistive (see also clause 5).

The relative level concept is very practical for the transmission aspects of telecommunications in several ways. It is a method for matching the power handling capacity of the transmission equipment in a connection to the levels of the actual signals in the network. Loss and gain in the network can be specified by means of relative levels. Also, relative levels can be used to characterize parameters of certain components of an equipment.

Note, however, that the application rules for relative levels depend on the context in which they are used.

It is immediately apparent that the differences of the relative levels between two points, which have the same level reference point, correspond to the loss or gain between those two points (at the reference frequency).

Moreover, relative levels are used to characterize the "power" handling capabilities of components (such as codecs) and equipment on the one hand and the expected levels of actual signals in the network on the other hand. This will be discussed in more detail in the following clauses.

The "signal path under consideration", for which a specific 0 dBr reference point is designated, can encompass:

a) a single component, such as an encoder or decoder;

b) an equipment, such as a half-channel of a digital exchange;

c) a circuit in the sense of the ITU-T definition, i.e. the fixed connection between two exchanges.

In the first two cases, the "power handling capability" is the guiding principle for the allocation of a level reference point. For the third case, the "expected absolute levels of actual signals" determines the choice of the level reference point.

The aim is of course to match the component performance to the requirement for the equipment performance which in turn should be matched to the actual range of signal levels. However, it is not always possible to achieve this exactly. For this and other reasons, the allocation of the 0 dBr reference point in the signal path may be chosen differently in the three cases above, i.e. when the component is considered alone, when it is considered as part of the equipment, and when the equipment is a part of the circuit. This means that the relative level designation for a certain point sometimes may differ in these three cases, a fact which should be remembered when discussing relative levels.

NOTE – It would be easy to surmise that there is only one level reference point in the network to which all relative levels are referred. However, this is not the case. As a matter of fact, in a complete connection, several different level reference points can be designated. These may also be different from those chosen when the parts of the transmission links are considered separately in the context of parameters for equipment or components. Thus, when stating the relative level at a point, one should make it quite clear in which context this relative level applies.

A more detailed discussion of the various applications of relative levels is given in clause 6.

Note that a so-called "level jump" may be introduced at the interconnection point between two (ITU-T) circuits. Thus, the loss or gain between two points belonging to two different circuits is not always equal to the difference in their (circuit) relative levels. Such an example is the case of the input and output relative levels of a digital exchange having no digital loss or gain pads. When the exchange is considered as an equipment, the difference between the (equipment) input and output relative levels gives the loss through the exchange because the two half-channels have the same level reference point. When the exchange is considered as a part of a connection, the two half-channels belong to two different (ITU-T) circuits which are interconnected "in the middle of" the switching matrix. The (circuit) input and output relative levels for the exchange, which are stated in the transmission plan for the connection, can differ from the specified (equipment) relative levels. This is because the (circuit) relative levels refer to two separate level reference points, each determined by estimation of expected signal levels in the two circuits (in general, however, the differences are not very large).

For the purpose of equipment parameter specification and transmission measurement, which is of interest here, the "power handling capability" is the governing factor for the choice of the 0 dBr level reference point. In this context, the digital 64 kbit/s PCM bit stream is considered as having a relative level of 0 dBr, provided that there are no digital loss or gain pads in its path. Ideal encoders and decoders connected to the bit stream are defined as having 0 dBr relative levels at their analogue ports when their clipping level for a sinusoidal signal lies at +3.14 dBm (A-law). The relative level for real encoders and decoders connected to the bit stream is determined by means of the actual clipping levels in relation to the clipping levels of the ideal codecs.

When a digital loss or gain pad is included in the digital bit stream, one has to make a choice of which side of the pad the bit stream is to be assigned to 0 dBr. In the context of equipment specification and transmission measurement, it has been found most practical to apply a convention that a digital bit stream never should be assigned a higher relative level than 0 dBr. This means that:

i) a digital pad with L dB loss has the relative levels of 0 dBr at the input and –L dBr at the output;

ii) a digital pad with G dB gain has the relative levels of –G dBr at the input and 0 dBr at the output.

Note that in the context of transmission planning, a digital bit stream sometimes may be assigned a relative level which is different from 0 dBr even if there is no digital pad in the digital path (see 6.4).

Clause 6.5.3 lists another couple of possible choices of the 0 dBr point in digital exchanges.

5.5 The unit "dBm0"

When using an additional "m" and "0" (zero) with the basic "dB", the level under consideration is expressed as the absolute level (dBm) of the same signal that would be measured at the relevant 0 dBr level reference point.

This term is used in conjunction with transmission measurements to specify test levels and test results; the term also facilitates the comparison of the power levels of different signals by referring them to a common reference point, i.e. the 0 dBr reference point. Networks are often designed to carry different types of signals (speech, modem, fax, etc.) at different levels, expressed in dBm0.

5.6 The relationship between dBm, dBr and dBm0

The relationship between relative levels at interfaces, which have the same level reference point, and the resulting transmission loss or gain "L", is given by the formula:

$$L = L_i - L_o \tag{5-4}$$

where L_i and L_o are the relative input and output levels at the interfaces.

The relation between the terms dBm, dBr and dBm0 can be expressed by the following formula:

$$\text{dBm} = \text{dBm0} + \text{dBr} \tag{5-5}$$

$$\text{dBmp} = \text{dBm0p} + \text{dBr (for weighted noise)} \tag{5-6}$$

or:

$$\text{dBm0} = \text{dBm} - \text{dBr (general)} \tag{5-7}$$

$$\text{dBm0p} = \text{dBmp} - \text{dBr (for weighted noise)} \tag{5-8}$$

Example 1: The test level for an interface with an input relative level of $L_i = -2$ dBr, is required to be –10 dBm0. To what absolute power level in dBm the signal generator should be adjusted?

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

$$= -10 + (-2) = -12 \text{ dBm}$$

Example 2: The dial-tone level at an interface with an output relative level of $L_o = -7$ dBr was measured with –19 dBm. Does this value meet the requirement given with ≤ –15 dBm0 for this type of interface?

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

$$= -19 - (-7) = -12 \text{ dBm0}$$

The result shows that the dial-tone level is outside the limit.

NOTE – Some modern test instruments are providing as well an automatic adjustment of the correct absolute test level, as the necessary correction of received levels and displaying the results in "dBm0". In those cases the above given calculation can be avoided; however, an additional adjustment (beside the test level itself) is required, to adapt the test instrument to the relative input and output levels of the test object.

5.7 The unit "dBov"

In the process of specifying speech coders and other signal processing devices, it is customary to express the codec input level specification in terms of dBs relative of the overload point of the digital system. This is a more convenient way to represent levels relative to the maximum power that can be stored in fixed- or floating-point format of a specific digital processing device.

In a generic notation, the overload point within the digital domain can be defined by the (normalised) amplitude value $x_{over} = 1.0$. The level specification for speech codecs is relative to this overload point in the digital domain. It should be noted that this overload point does **not** depend on the quantisation method used and remains identical, regardless of whether the quantisation is done e.g. with 32, 16, 13 or 8 bits. How this overload point relates to the analogue world depends on the conversion method between the analogue and digital domains, and it is beyond the scope of this Recommendation.

The power of a sampled signal $x(n)$ with a length of N samples can be defined by:

$$P = \frac{1}{N}\sum_{n=0}^{N-1} x(n)^2$$

For a digital system which has an overload point x_{over}, the maximum signal power will be $P_0 = 1.0$. In this case, the power level for a digital signal in decibels relative to the overload point (dBov, where the characters "ov" arbitrarily mean digital **ov**erload signal level) is defined by:

$$L_{ov} = 10\log_{10}\left(\frac{P}{P_0}\right) \text{ [dBov]}$$

The level of the maximum signal power P_0 is thus 0 dBov, which is chosen to be the reference level. A signal with such power level could be:

a) a sequence of maximum positive numbers ($+x_{over}$);

b) a sequence of maximum negative numbers ($-x_{over}$); or

c) a rectangular function exercising only the positive or negative maximum numbers ($\pm x_{over}$).

The level of a *tone* with a digital amplitude (peak value) of x_{over} is therefore $L = -3.01$ dBov.

5.8 Relation between overload (dBov) and maximum levels (dBm0)

While levels in digital transmission networks are expressed usually in terms of the power of tone (expressed e.g. in dBm0), the level specification for digital processing devices such as speech coders is specified in terms of dBov values. Hence, it is useful to relate these two level units. The conversion between both representations can be generically expressed as:

$$y(\text{dBm0}) = z(\text{dBov}) + C$$

There are three specific cases of interest: A-law G.711, µ-law G.711, and G.722. It should be noted that irrespective of the case, the level $T_{\max}$ of a maximum level tone would always be –3.01 dBov. For the G.711 encoding rule, a tone that exercises the maximum level has a power $T_{\max}$ of 3.14 dBm0 for A-law, and of 3.17 dBm0 for µ-law. Therefore, C above becomes 6.15 dB for A-law and 6.18 dB for µ-law. For the G.722 wideband coding algorithm, the overload point of the A/D and D/A converters should be 9 dBm0. Therefore, in that case, C becomes 12.01 dB.

The following relationships summarise the relationships described above:

$$\Lambda_A \text{ (dBm0)} = L_{ov} \text{ (dBov)} + 6.15 \text{ dB} \qquad \text{(A-law)}$$

$$\Lambda\mu \text{ (dBm0)} = L_{ov} \text{ (dBov)} + 6.18 \text{ dB} \qquad (\mu\text{-law})$$

$$\Lambda_{G.722} \text{ (dBm0)} = L_{ov} \text{ (dBov)} + 12.01 \text{ dB} \qquad \text{(G.722)}$$

6 The concept of "relative levels"

6.1 General principles

As already mentioned in clause 5, the concept of "relative level" is applied in many areas.

In transmission planning, relative levels are used to characterize "probable signal power levels" occurring in the circuits of the network.

In transmission maintenance, relative levels are used to describe loss or gain between points as well as defining levels of test signals.

For the specification and design of an equipment, relative levels are used to describe the power handling capabilities when the equipment is employed in a transmission chain.

In testing of equipment and components, relative levels are used to characterize signal parameters.

In the ideal case, the power handling capabilities of components and equipment would be accurately matched to the actual signal powers they encounter when used in the network. In practice, this is not always achievable or even desirable. For instance, in equipment design, the relative level designations for testing components do not always correspond exactly to the specified relative levels for the equipment considered as a unit.

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.

However, the distinctions between the different applications of "relative levels" have not always been clearly stated, even in ITU-T Recommendations, which sometimes has caused confusion.

Often it is clear what a relative level value refers to. However, there is a risk of misunderstanding. It is a wise precaution to make a direct statement, such as:

a) (test) relative level;

b) (equipment) relative level;

c) (circuit) relative level.

As an example of a misunderstanding, the relative levels given in a transmission plan have sometimes erroneously been taken to exactly correspond to test levels of equipment.

In the following, examples are given of "good engineering practice" with regard to relative level applications. The rules should be considered as having a certain amount of flexibility. Most difficulties seem to have occurred in conjunction with digital transmission. Therefore, digital cases are given special attention.

6.2 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.

6.3 The speech signal and the dynamic range of the voice channel

During normal, active speech periods, the variation in level between different speakers has a standard deviation of about 3 dB as recorded with a fixed distance mouth to microphone. However, when speakers are using actual telephone handsets, held according to individual preferences, the standard deviation is increased by up to 5 dB.

The performance of Frequency Division Multiplex (FDM) (carrier) equipment is governed by the total channel load. That means that the **mean** channel load capacity is of importance. According to former ITU-T Rec. G.223, this should be –15 dBm0, with speech pauses included and consideration taken of some extraneous signals. This translates into –11 dBm0 for the actual speech periods.

For PCM systems, the individual channel performance should be matched to the **dynamic range** of the speech signals. Therefore, it is of interest to study the instantaneous amplitude distribution of speech signals.

It is practical to relate the absolute amplitude V of speech signals to the root mean square (rms) – value of the speech signal (V_{eff}) during active speech periods. Investigations have shown that the statistical distribution can be simulated by the function:

$$P(X) = \frac{K}{\Gamma(L)}(KX)^{L-1}\mathrm{e}^{-KX} \tag{6-1}$$

where: $X = \dfrac{V}{V_{eff}}$

$L = \text{constant}$

$K = \sqrt{L(L+1)}$

$\Gamma = \int_0^\infty t^{x-1}\mathrm{e}^{-t}\mathrm{d}t$ (the Gamma function)

The value for L is about 0.5 for handsets with modern linear microphones (for older carbon-type microphones L is about 0.2).

The equation as shown above is to be interpreted as follows:

The probability to find a value in the interval X ± dX/2 is P(X) dX.

From the above equation the cumulative statistical distribution F(X) can be computed. This is depicted in Figure 5 with N = 20 log (V/V_{eff}) as abscissa and L = 0.2...0.6. For comparison, a similar curve is drawn for the envelope of band-limited white noise signals.

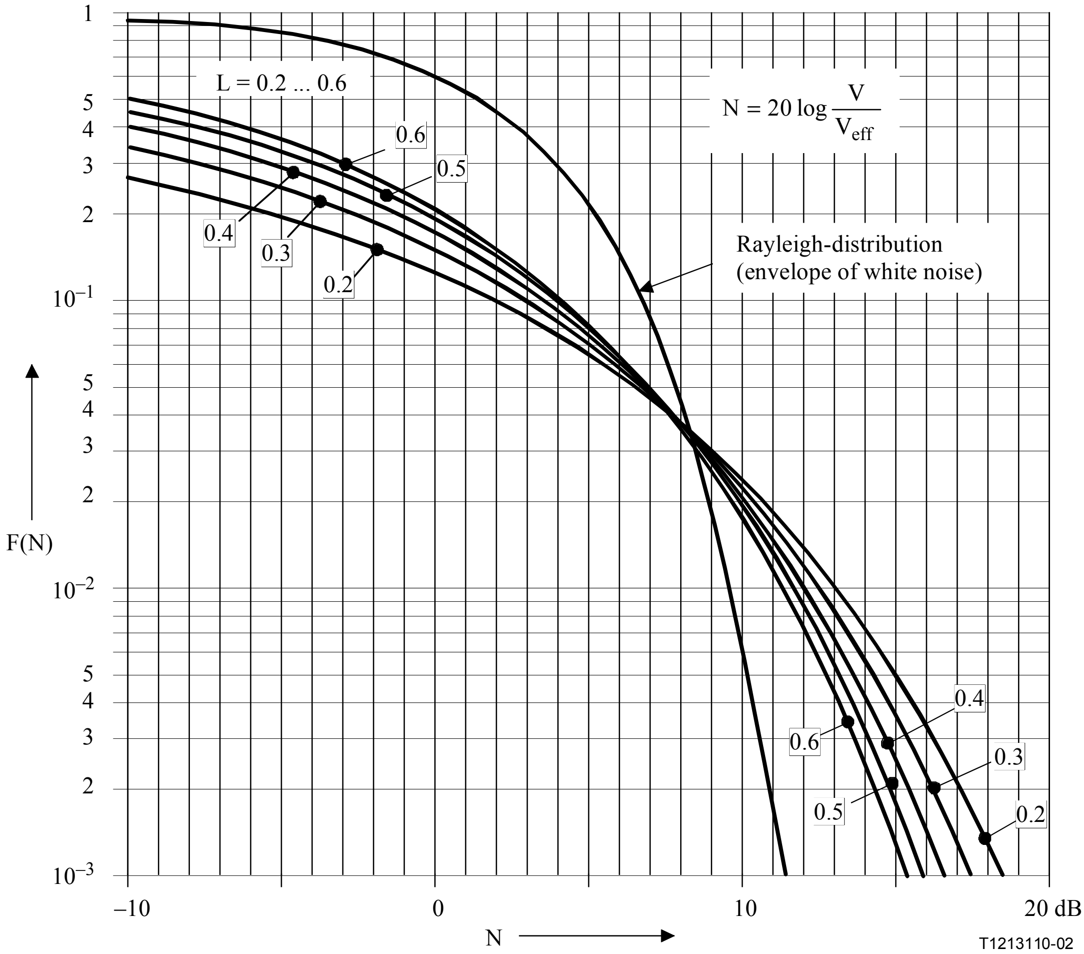


Figure 5/G.100.1 – Statistical distribution of speech signals

It is apparent from the figure that speech signals are more "peaky" than white noise. However, for those peak values that only are exceeded 1 % of the time, the difference is only about 2 dB for the most common value L = 0.5. The 1 % probability value corresponds to N = 12 dB. Subjective tests indicate that this is an acceptable lower limit for speech clipping. Measured absolute peak values of speech lie at about 18 dB above the rms value, but those peaks occur very infrequently.

The dynamic range of 64 kbit/s PCM codecs can be described in many ways. One method is to look at the limits for the signal-to-total distortion ratio as depicted in Figure I.5/G.712 [3] which is reproduced here in Figure 6. This curve applies for white noise as an input signal (method 1).

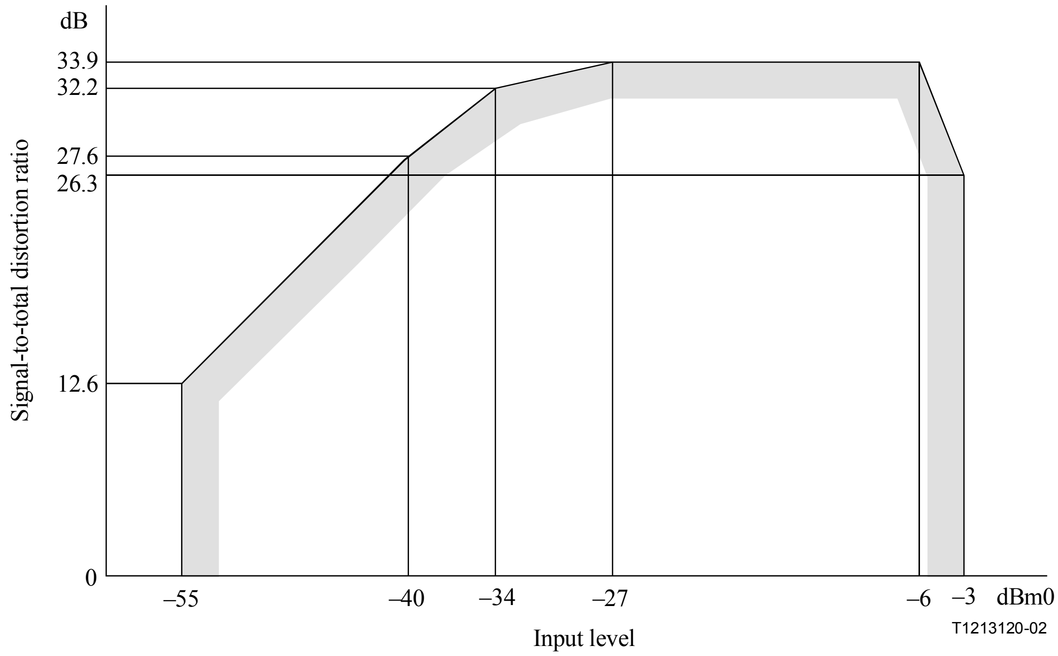


Figure 6/G.100.1 – Signal-to-total distortion ratio as a function of input level (method 1)

One can see from Figure 6 that the signal-to-total distortion ratio curve is flat from –27 dBm0 to –6 dBm0 white noise input signal. The upper limit corresponds to the level when peak clipping begins to take effect. However, the decrease in signal-to-total distortion ratio is quite moderate for –3 dBm0 input level.

The peak clipping level for sinusoidal signals is +3 dBm0, i.e. the absolute peak limit level is 6 dBm0. Thus, in the range when the peak clipping begins to take effect for white noise, the margin between the peak limit and the rms value of the noise lies between:

$$6 \text{ dB} + 3 \text{ dB} = 9 \text{ dB}$$

and

$$6 \text{ dB} + 6 \text{ dB} = 12 \text{ dB}$$

Using speech signals, these values should be increased by 2 dB, giving a desirable margin in the range of 11 dB to 14 dB. This corresponds well with the subjectively established value of 12 dB.

What actual speech levels can be expected in the network compared to the nominal speech level?

According to a recent investigations, a "reference talker" (i.e. talking a with –4.7 dBPa mean speech sound pressure at the MRP) produces during active speech at a 0 dBr point a signal level of:

$$N = -11 - SLR \text{ [dBm0]} \tag{6-2}$$

where SLR is referred to the 0 dBr point.

With equation (6-2) one can compute the margin C at the average speech level against "just noticeable" speech clipping, i.e. at 12 dB higher than the rms value. Also, using the standard deviation of 5 dB for speech levels, one can estimate the percentage P_c of talkers who talk so loudly that they are subjected to clipping. Thus:

For the nominal SLR = 7 dB: C = 12 dB, P_c = 0.8 %;

For the minimum SLR = 2 dB: C = 7 dB, P_c = 8%.

It appears that SLR > 2 gives a reasonable protection against objectionable speech clipping.

NOTE – Actual speech levels in networks are currently being studied in the ITU-T.

Thus, for normal connections, there are no problems in the matching between the dynamic ranges of the speech signal and the codecs. Moreover, it appears that reasonable margins exist in the 64 kbit/s PCM channel so that the nominal speech level can be increased by 2 or 3 dB or decreased by 6 dB from its normal value of –11 dBm0 without objectionable results. (This is confirmed by some early subjective tests performed with the help of the Modulated Noise Reference Unit (MNRU) method.)

Examples of such level shifts occur when digital loss or gain is used or when so-called level jumps have to be introduced between (ITU-T) circuits (see 6.5 and 6.6). Formally, this can be handled by assigning relative levels **differing from 0 dBr** to the digital bit stream. This is discussed in 6.4 thru 6.6.

6.4 Relative level designations for a digital path

Most often, the digital path is assigned to the relative level 0 dBr. The absolute level of a signal in a 64 kbit/s PCM path then is determined by ideal encoders and decoders as shown in Figure 7.

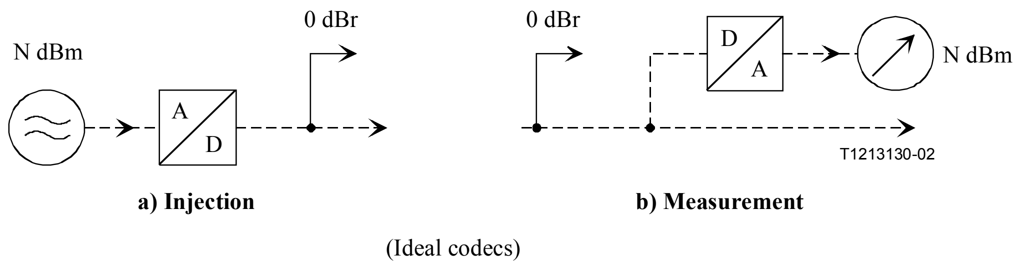
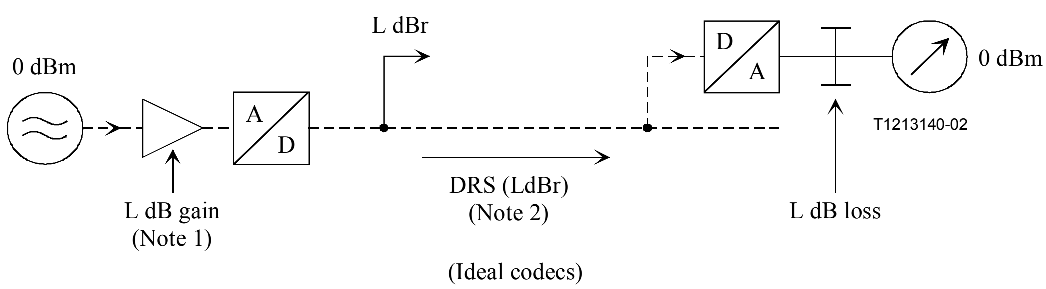


Figure 7/G.100.1 – Interpretation of absolute signal level in a digital path with relative level 0 dBr

The analogue 0 dBm0 reference signal, corresponding to N = 0 in Figure 7, has its counterpart in the standard Digital Reference Sequence (DRS).

In some exceptional cases, it is practical to assign a relative level L dBr, **differing from 0 dBr**, to the digital path. The analogue 0 dBm0 reference signal then corresponds to a different DRS which shall be termed DRS (L dBr) for clarity. The injection and detection of this is depicted in Figure 8.



NOTE 1 – Negative values of L are chosen more often. In test specifications, positive values of L shall never be used.

NOTE 2 – To create a DRS(L dBr) as shown, L < 3 dBr. If L > 0 dBr, no sinusoidal signals higher than (3-L) can be passed without clipping.

Figure 8/G.100.1 – Interpretation of a 0 dBm0 reference sequence DRS(L dBr) for a digital path with relative level L dBr

Figure 9 shows level measurement of an actual signal on a digital path with the relative level L.

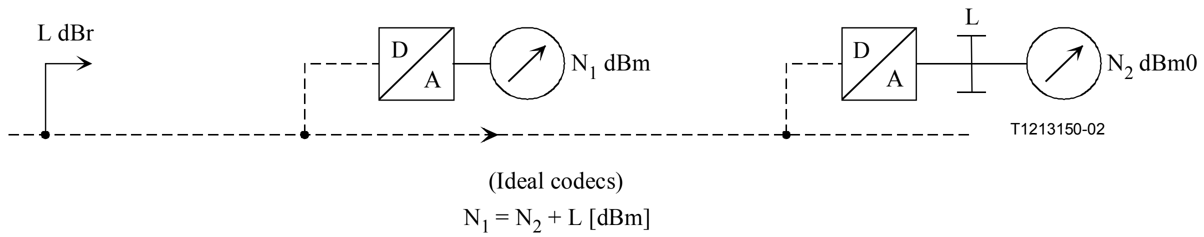


Figure 9/G.100.1 – Level measurements on a digital path with L dBr

6.5 Relative levels in equipment design, specification and testing

6.5.1 Analogue equipment

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. This corresponds to –11 dBm0 actual speech during active periods. (FDM systems with fewer than 240 channels should be designed for a higher average power per channel. For instance, a 12-channel FDM system should be able to handle –7.5 dBm0 per channel.)

Voiceband analogue equipment is in general designed with regard to relative levels so that noise and clipping do not present any problems (this implies for instance that the clipping level is higher than 3 dBm0).

6.5.2 Codecs and digital pads

For 64 kbit/s **encoders and decoders** regarded as **components** of an equipment, the digital path is taken to represent the 0 dBr level reference point (see Figure 10).

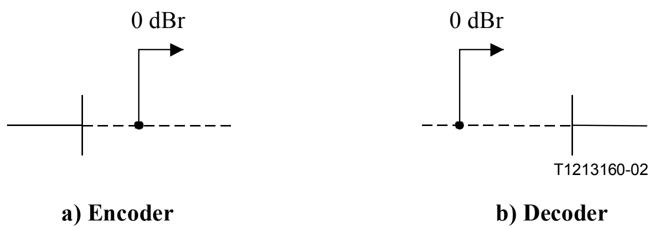


Figure 10/G.100.1 – 0 dBr level reference points for codecs

The performance specification of codecs, as described in ITU-T Rec. G.712 [3], is based on this convention and the parameters are specified with respect to 0 dBm0 values.

In general, when speech path impairments are considered, **analogue pads**, **loss** or **gain**, are to be preferred for level and loss adjustments. However, **digital pads** often allow more flexibility, especially as they can easily be controlled by software.

Experience has shown that digital pads are robust components that do not require such extensive testing as codecs do. Therefore, it has not been necessary to introduce dBm0 values in their specifications.

When codecs and digital pads are combined in an equipment, any performance testing of the equipment should be done with the pads disabled, except of course, during pure loss or gain measurements.

6.5.3 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 mentioned 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 12. 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.

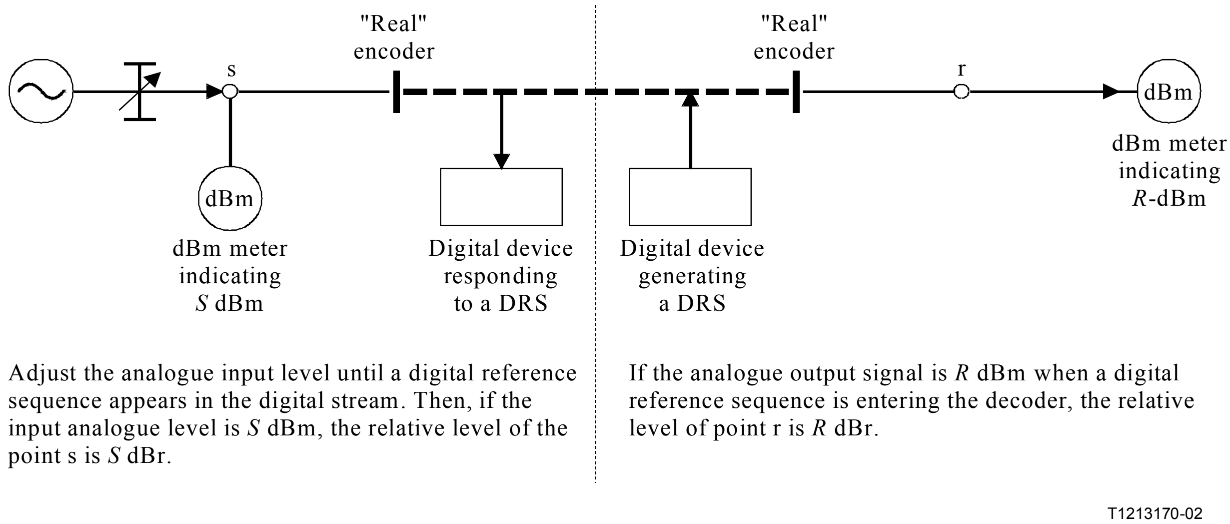
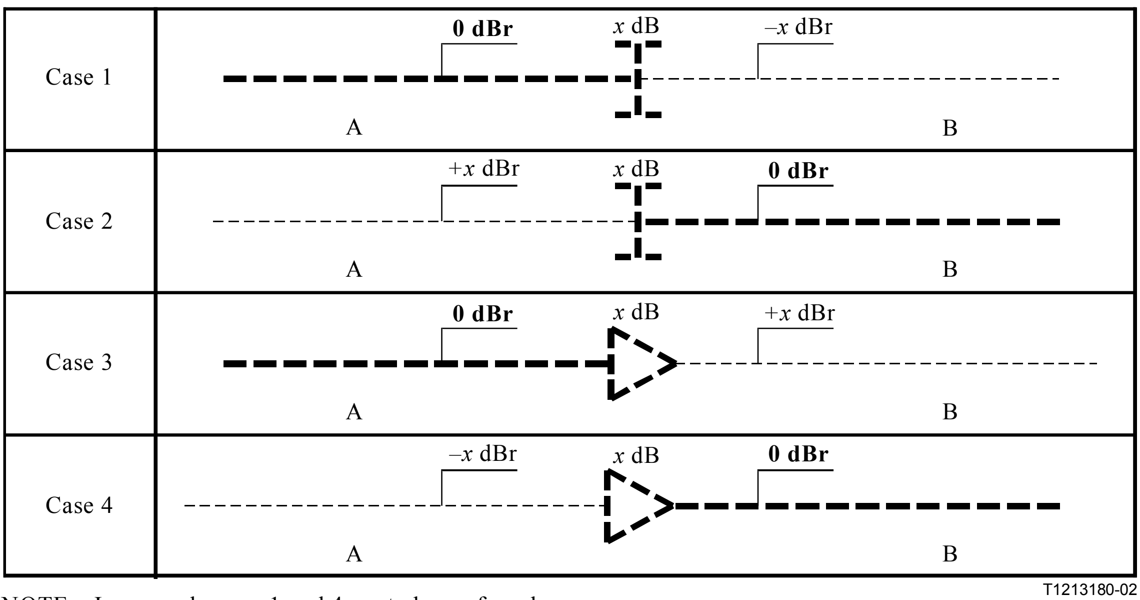


Figure 11/G.100.1 – Set-up for determination of the relative level at the input and output analogue points of a "real" codec using the digital reference sequence



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

Figure 12/G.100.1 – Relative values in a digital path

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 ITU-T Rec. G.711, then 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.

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.

6.5.4 Digital exchanges

A digital exchange is built up of half-channels interconnected by a switching matrix.

The power handling properties, which are to be used as a basis for the transmission planning of networks, are described by the relative level designations of the exchange ports. (However, these values are not necessarily the same as used in performance testing or in the transmission plan.)

For all those cases when no digital pads are used, the digital path is considered to be at 0 dBr relative level. Figure 13 shows as an example the relative level designations for a 2-wire subscriber half-channel.

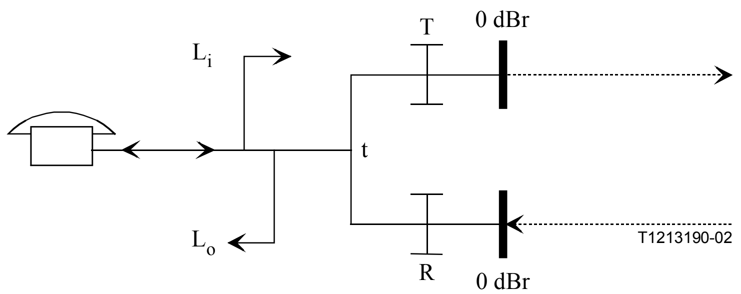


Figure 13/G.100.1 – Relative levels at a local exchange L_i = T dBr, L_o = – R dBr (it is assumed that T and R represent all those losses between t, the 2-wire point, and the digital bit streams)

When **digital pads** are used, they can either be incorporated in the switching matrix or in the half-channels.

In the first case, the relative level designations for the half-channels remain unchanged.

In the second case, in principle there are several possibilities to designate the 0 dBr reference point:

a) those digital points which are directly connected to respectively the encoder or decoder;

b) digital points near the pads, chosen in such a way that the digital relative levels never exceed 0 dBr;

c) the digital points interfacing the switching matrix.

All cases considered, method c) appears to be the most practical one when specifying data for use in transmission planning.

Figures 14 and 15 show examples of how digital loss and gain pads are introduced in the half-channel depicted in Figure 13.

The nominal losses through a half-channel can be found from the relative level designations as shown in Figures 13, 14 and 15.

The loss through the exchange is:

$$A = L_i - L_o + SL \tag{6-3}$$

where L_i, L_o are the input and output relative levels of the half-channels concerned, and SL is the loss of the switching matrix.

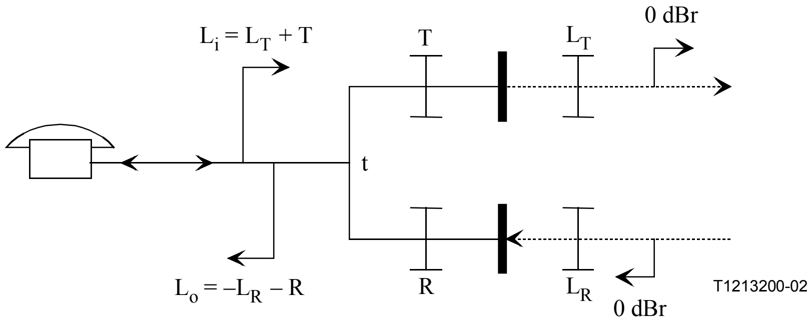


Figure 14/G.100.1 – Relative levels at a local exchange. Digital loss in the half-channel

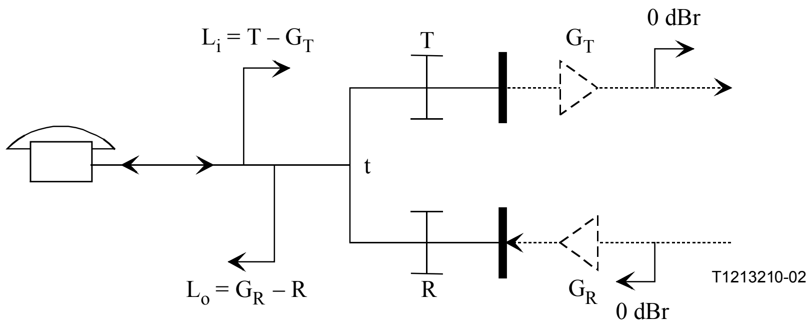


Figure 15/G.100.1 – Relative levels at a local exchange. Digital gain in the half-channel

6.6 Relative levels in transmission planning and maintenance

In transmission planning procedures the overall transmission path is divided into sections in ITU-T vocabulary termed **circuits**, each having its own 0 dBr Transmission Reference Point (TRP). Most often circuits connect switching centres. Sometimes also the subscriber line connected to a local exchange is termed circuit. Thus, a circuit is constituted by all permanently interconnected equipment. In this way maintenance personal have clearly defined segments with fixed transmission parameters to supervise.

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. This practice is common in the public networks and dates back to the times when most exchanges were analogue.

However, the transmission planner has other options to subdivide the connection into circuits, provided he clearly defines the interface. Thus, if the digital switching matrix is designed to introduce loss, the two half-channels 0 dBr points may be considered as ending of circuits with the switching matrix as a mini-circuit in between.

Exceptionally, the "transmission interface" between the 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 private links as belonging to two different circuits.

One main problem in transmission planning is to obtain a reasonable matching between expected signal levels and the power handling capabilities of the equipment used in each circuit. Sometimes also the relative levels at circuit interconnection points cannot be matched to each other so that "level jumps" have to be introduced.

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 dBrs particular to one circuit, to another set of dBrs particular for the other circuit. The loss will always be present within the circuits themselves.

However, at a speech level of –11 dBm, at a 0 dBr point, pauses excluded, expected as an average for a large number of subscribers, field measurements of actual speech levels in TRPs show a very large spread. For this reason, one resorts instead to some conventions based on general experience.

For normal telephony terminals and subscriber lines, the interconnection to the local exchange can be taken as an "anchor point" to establish a 0 dBr point (see Figure 13). Of course, the speech levels are influenced by the telephony terminal sensitivities. Nevertheless, from Annex C/G.121 [2] it can be seen that many Administrations found the optimum values to be $L_i = 0$ dBr, $L_o = -6$ dBr or –7 dBr.

Regarding how the equipment is incorporated 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 a lack of suitable level controls in certain equipment (some echo canceller designs may also need an extra margin against clipping).

An example of additional loss in an analogue 4-wire loop is shown in Figure 16 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, ITU-T recommends 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.

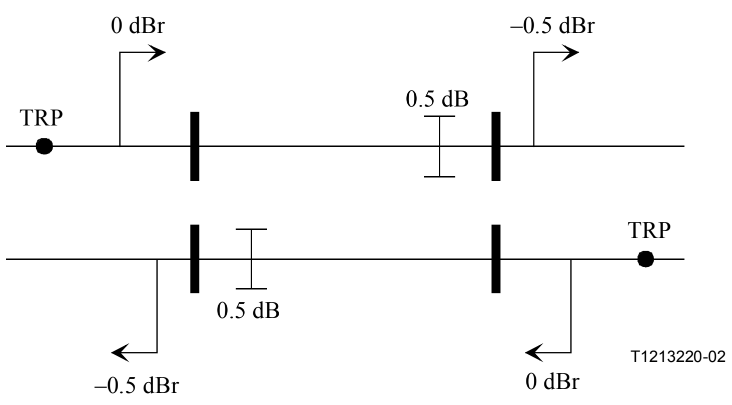


Figure 16/G.100.1 – Example of (circuit) relative levels when an analogue link is interposed in a digital chain

Two adjacent circuits each have their own 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 is 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 should be 0.5 dB for stability reasons. The relative levels at the local exchanges are determined by the properties of the telephony terminals as mentioned before. Therefore, the 0.5 dB net transmission loss corresponds to a "level-jump" of 0.5 dB at the transit exchange. A similar example of an international transit connection is given in Figure 17.

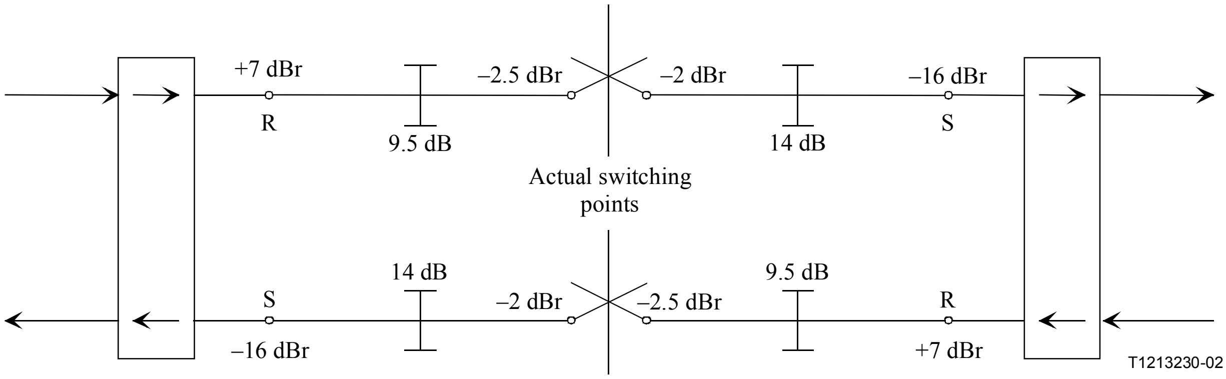


Figure 17/G.100.1 – Example showing a (simplified) representation of a transit connection in an international exchange

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 (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 should 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 consistent circuits are digital and not using digital signal processing).

With regard to digital parameters in a complete connection, the transmission planner should also consider the sum of qdus, the total amount of digital loss or gain introduced, and the sum of all level-jumps.

Appendix I

The neper

I.1 Introduction

The usage of the neper in telecommunications is no longer recommended by the ITU-T. However, for reasons of traceablity and for the sake of completeness this appendix provides the necessary information.

I.2 Definition of the neper

The *neper* (symbol Np) expresses the ratio of two field quantities such as voltage or current, the square of which is proportional to power by the natural logarithm of this ratio. The value of a power ratio in nepers is one half of the natural logarithm of the power ratio. The values in nepers of the ratio of two field quantities and of the corresponding powers are equal only if the impedances are equal.

One neper corresponds to the value of e of a field quantity ratio and to the value e^2 of a power quantity ratio.

Sub-multiples such as the decineper (dNp) are also used.

In some disciplines, nepers may be used to express the logarithm of a power ratio without the factor ½. An example is optical depth or attenuation in radiometry. Such usage is deprecated in telecommunications in order to avoid ambiguity. Under this definition, the neper would in fact be equal to 4.34 dB, instead of 8.68 dB as is traditionally the case.

I.3 Use of the decibel and of the Neper

As mentioned afore, the ITU-T has discontinued to recommend the use of the neper in telecommunications.

However, for theoretical or scientific calculations, where ratios are expressed in terms of naperian logarithms, the neper will always be used, implicitly or explicitly.

As a result of some calculations on complex quantities, a real part in nepers and an imaginary part in radians are obtained. Factors may be applied for converting to decibels or degrees.

The conversion values between the neper and the decibel are as follows:

$$1 \text{ Np} = (20 \log \text{e}) \text{ dB} \approx 8.686 \text{ dB}$$

$$1 \text{ dB} = (0.05 \ln 10) \text{ Np} \approx 0.1151 \text{ Np}$$

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