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Technical Report

Speech processing, Transmission and Quality aspects (STQ); The Concept of Relative Levels

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Speech processing, Transmission and Quality aspects (STQ).

The present document is intended to provide guidance on "good engineering practice" with respect to relative levels.

1 Scope

The present document gives guidance on "good engineering practice" with respect to relative levels.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, subsequent revisions do apply.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

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

[2] ITU-T Recommendation G.101 (08/96): "The transmission plan".

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

[4] ITU-T Recommendation G.712 (11/96): "Transmission performance characteristics of pulse code modulation channels".

[5] ITU-T Recommendation O.41 (10/94): "Psophometer for use on telephone-type circuits".

3 Definition of the units dBm, dBr and dBm0

3.1 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 uses 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 4 for a more complete discussion.

3.2 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).

3.3 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 speechband applications has the dimension of power, i.e. expressed in mW or mVA, and has by definition the form:

$$P = \frac{1\,000x(V)^2}{|Z(f_0)|}[mW] \text{ or } [\mathrm{mVA}]$$

where:

V: the rms value in volts of the voltage across the test impedance Z that 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 = 1 020 Hz.

The choice of this definition is based on three conventions:

- 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;
- 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 of how the signal is amplified. Hence the use of a constant impedance value in the denominator instead of a frequency dependant impedance;
- thirdly, a sinusoidal signal with the reference frequency (1 020 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 Ω. 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 1 020 Hz.

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

$$L = 10\log\frac{P}{P_0}[dBm]$$

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:

$$\mathrm{A(f)} = 10\log\left\{\frac{\mathrm{P_1(f)}}{\mathrm{P_2(f)}}\right\} = 20\log\left\{\frac{\mathrm{V_1(f)}}{\mathrm{V_2(f)}}\sqrt{\left|\frac{\mathrm{Z_2(f_0)}}{\mathrm{Z_1(f_0)}}\right|}\right\}[dB]$$

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.

3.4 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 1 020 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 1: 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 4).

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 2: That the application rules for relative levels depend on in which context 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 subclauses.

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

- a single component, such as an encoder or decoder;
- an equipment, such as a half-channel of a digital exchange;
- 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 3: 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 4 (see also ITU-T Recommendations G.100 [1] and G.101 [2]).

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:

- a digital pad with L dB loss has the relative levels of 0 dBr at the input and -L dBr at the output;
- 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 subclause 4.3).

Subclause 4.4.3 lists another couple of possible choices of the 0 dBr point in digital exchanges.

3.5 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.

3.6 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 Recommendation O.41 [5].

3.7 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:

$$L = L_i - L_o$$

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:

dBm = dBm0 + dBr (general)

dBmp = dBm0p + dBr (for weighted noise)

or:

dBm0 = dBm - dBr (general)

dBm0p = dBmp - dBr (for weighted noise)

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?

dBm = dBm0 + dBr

= -10 + (-2) = -12 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?

dBm0 = dBm - dBr

= -19 - (-7) = -12 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.

3.8 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 1 in the application for sending and in figure 2 for receiving.

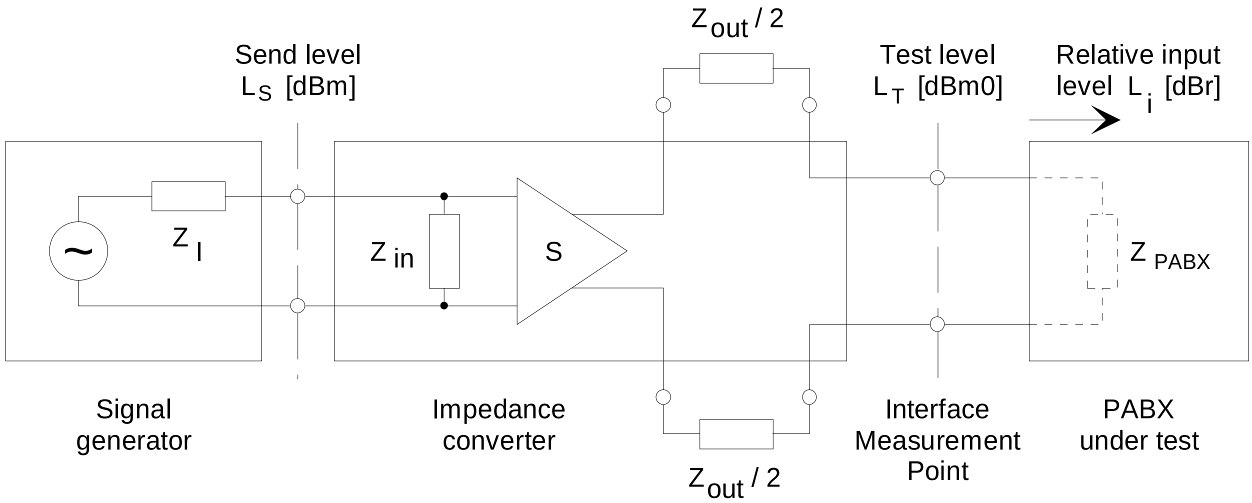


Figure 1: Impedance converter in the sending path

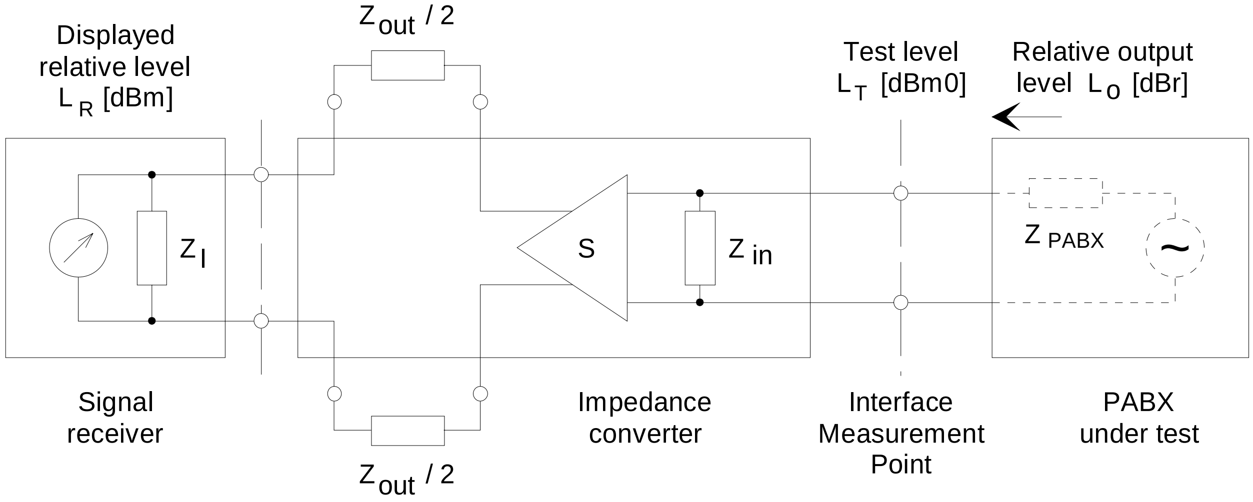


Figure 2: Impedance converter in the receive path

An advantageous design is, to obtain a power transfer ratio of 1 at the reference frequency 1 020 Hz, if terminated with the respective nominal impedances at input and output. In this case the voltage gain "s" of the inserted amplifier is:

$$s = 6 + 10\log\frac{Z_{out}}{Z_{in}}\left[dB\right]$$

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 modula at the reference frequency 1 020 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:

Sending a test signal

In this application (see figure C.0) 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 Z_{PBX} 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:

$${}_S\left[\text{dBm}\right] = L_T\left[\text{dBm0}\right] + L_i\left[\text{dBr}\right] + 10\log\frac{Z_{out}}{Z_{in}}$$

EXAMPLE 1: For an interface of the PBX under test with an input relative level L_i = -5 dBr and a nominal impedance Z_{PBX} = 842 Ω (modula at 1 020 Hz for a 3-element complex impedance with 270 Ω + 750 Ω // 150nF) 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?

$${}_S\left[\text{dBm}\right] = L_T\left[\text{dBm0}\right] + L_i\left[\text{dBr}\right] + 10\log\frac{Z_{out}}{Z_{in}}$$

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

$$\underline{\underline{L_S = -13{,}53\,dBm}}$$

Receiving a test signal

For receiving (see figure 2) 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:

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

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?

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

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

$$\underline{\underline{L_T = -\,44{,}47dBm0}}$$

4 The concept of "Relative Levels"

4.1 General principles

As already mentioned in clause 3, 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.

However, the distinctions between the different applications of "relative levels" have not always been clearly stated, not even in ITU-T documents, 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:

- (test) relative level;
- (equipment) relative level;
- (circuit) relative level.

As an example of 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.

4.2 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 an ITU-T design Recommendation, 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:

$$(X) = \frac{K}{\Gamma(L)} (K\,X)^{L-1}\, e^{-K\,X}$$

where: $X = \frac{V}{V_{eff}}$, L = a constant, $K = \sqrt{L(L+1)}, \Gamma = the \Gamma - function.$

The constant L is about 0,5 for handsets with modern linear microphones (for older, carbon types L = 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.

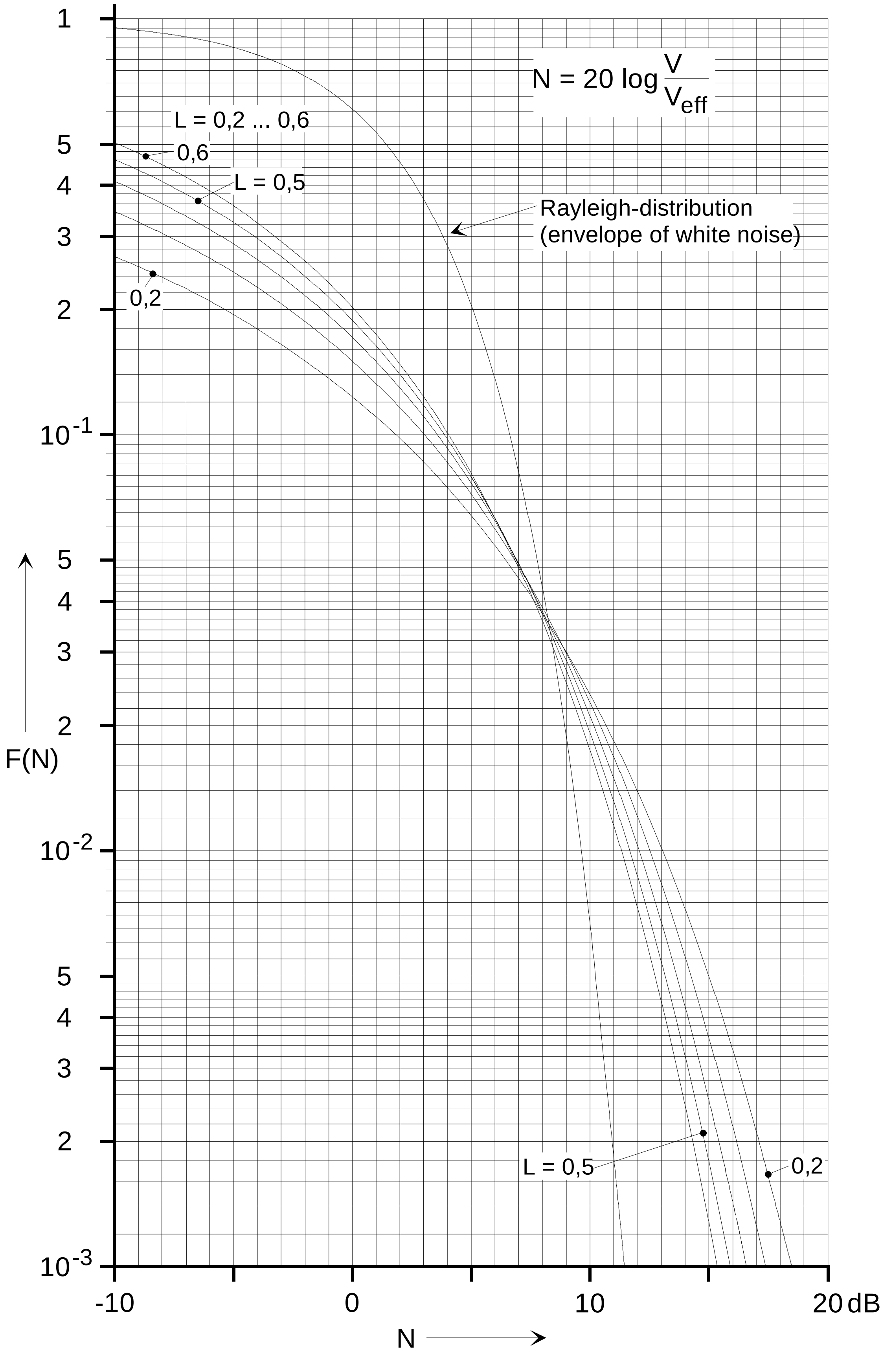


Figure 3: Statistical distribution of speech signals

From the above equation the cumulative statistical distribution F(X) can be computed. This is depicted in figure 3 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.

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 of ITU-T Recommendation G.712 [4] that is reproduced here in figure 4. This curve applies for white noise as an input signal (method 1).

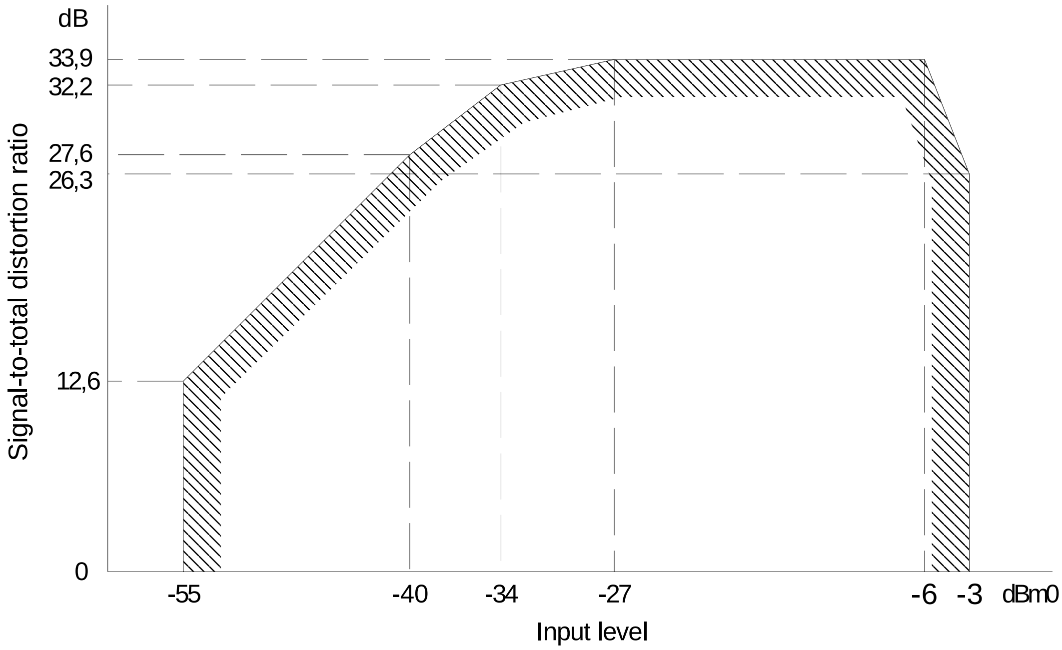


Figure 4: Signal-to-total distortion ratio as a function of input level (method 1)

One can see from the figure 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:

$$N = -11 - SLR \quad [\text{dBm0}]$$

where SLR is referred to the 0 dBr point.

By using the above formula, 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 clauses 4.4 and 4.5). Formally, this can be handled by assigning relative levels **differing from 0 dBr** to the digital bit stream. This is discussed in subclauses 4.3 to 4.5.

4.3 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 is then determined by ideal encoders and decoders as shown in figure 5.

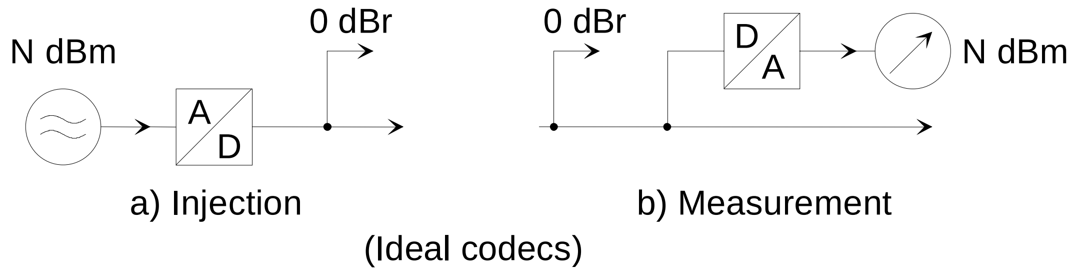
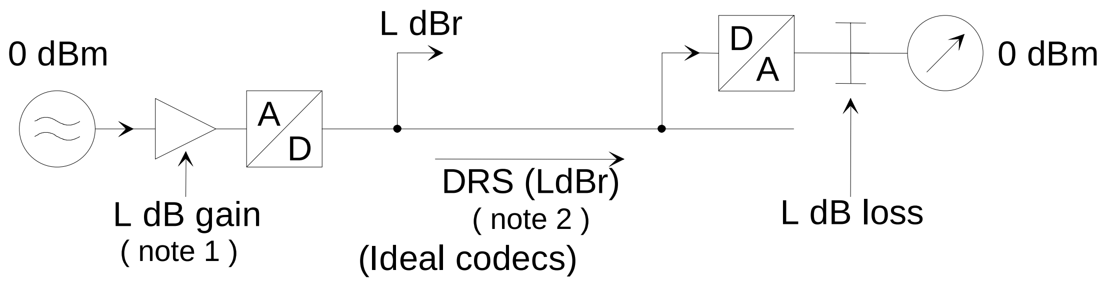


Figure 5: Interpretation of absolute signal level in a digital path with relative level 0 dBr

The analogue 0 dBm0 reference signal, corresponding to N = 0 in figure 5, 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 6.

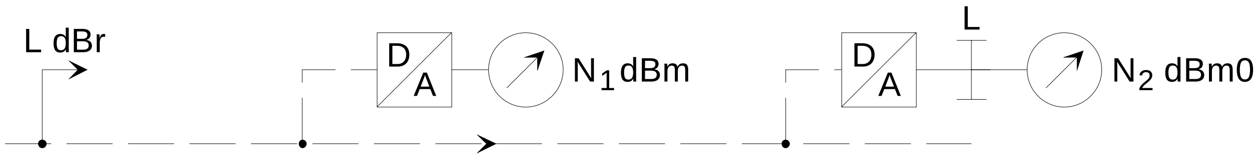


NOTE 1: Negative values of L are chosen more often. In test specifications, positive values of L shall never be used.

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 6: Interpretation of a 0 dBm0 reference sequence DRS(L dBr) for a digital path with relative level L dBr

Figure 7 shows level measurement of an actual signal on a digital path with the relative level L.



(Ideal codecs)

$$N_1 = N_2 + L \quad [\text{dBm}]$$

Figure 7: Level measurements on a digital path with L dBr

4.4 Relative levels in equipment design, specification and testing

4.4.1 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).

Voice band 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).

4.4.2 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 8).

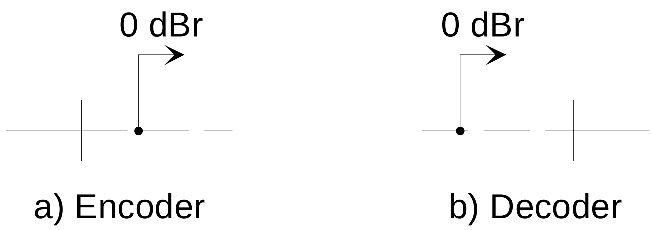


Figure 8: 0 dBr level reference points for codecs

The performance specification of codecs, as described in ITU-T Recommendation G.712 [4] and elsewhere, 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.

The amount of digital loss or gain should be kept within reasonable limits (see subclause 4.2).

4.4.3 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 9 shows as an example the relative level designations for a 2-wire subscriber half-channel.

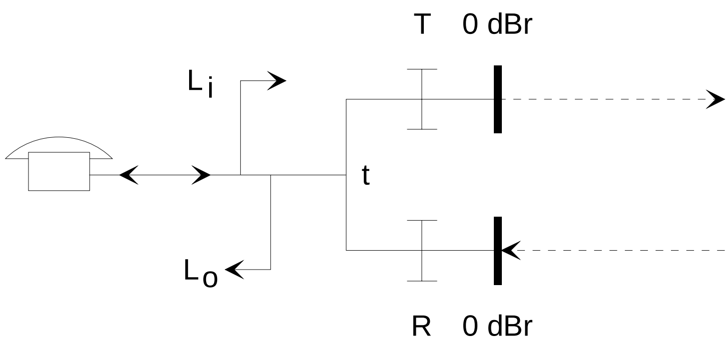


Figure 9: Relative levels at a local exchange L_i = T dBr, L_o = - R dBr (it is assumed that T and R represent all those losses between t, the 2-wire point, and the digital bit-streams)

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 10 and 11 show examples of how digital loss and gain pads are introduced in the half-channel depicted in figure 9.

The nominal losses through a half-channel can be found from the relative level designations as shown in figures 9, 10 and 11.

The loss through the exchange is:

$$A=Li-Lo+SL$$

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.

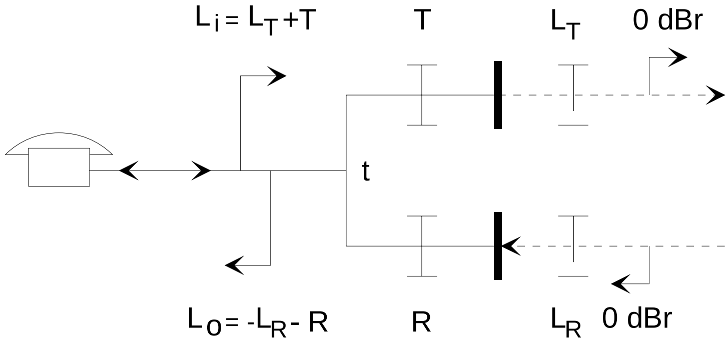


Figure 10: Relative levels at a local exchange. Digital loss in the half-channel

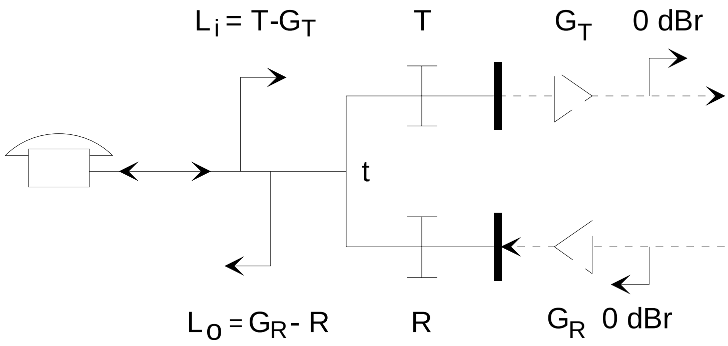


Figure 11: Relative levels at a local exchange. Digital gain in the half-channel

4.5 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 0dBr 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.

Regarding a speech level of -11 dBm, at a 0 dBr point, pauses excluded, expected as an average for a large number of subscribers, however, field measurements of actual speech levels in TRPs show a very large spread. By this instead one resorts 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 9). Of course, the speech levels are influenced by the telephony terminal sensitivities. Nevertheless, from Annex C of ITU-T Recommendation G.121 [3] 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 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 12 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.

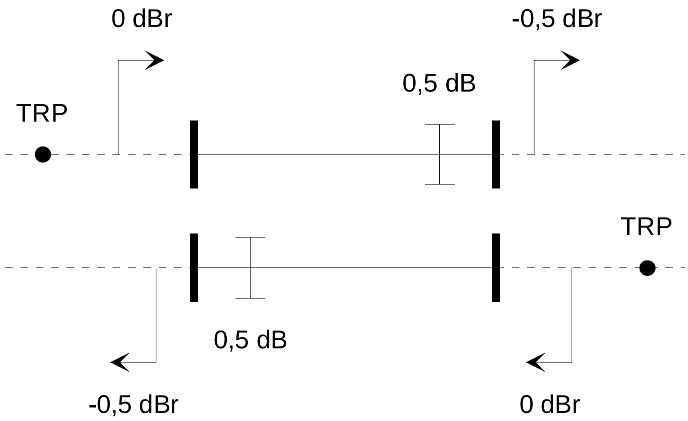


Figure 12: Example of (circuit) relative levels when an analogue link is interposed in a digital chain

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 13.

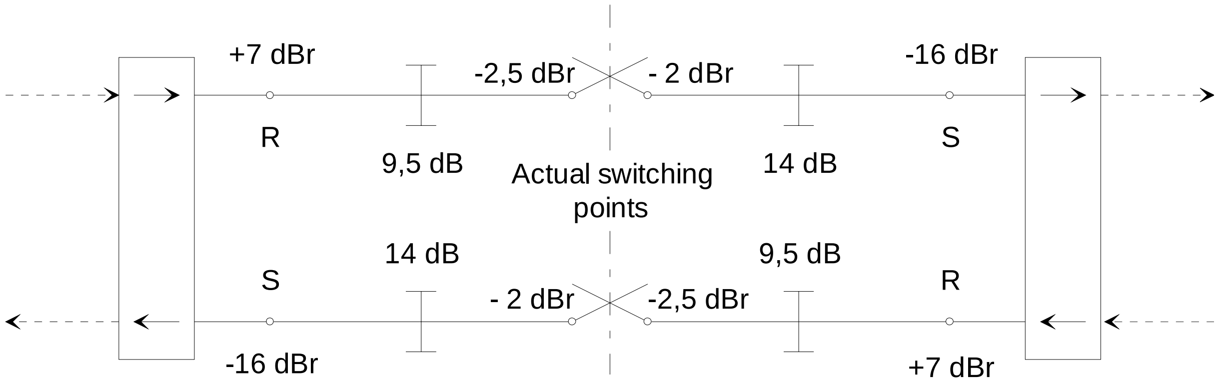


Figure 13: Example showing a (simplified) representation of a transit connection in an international exchange, actual arrangement

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 (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 dues, the total amount of digital loss or gain introduced, and the sum of all level-jumps (see subclause 4.2).

History

Document history		
V1.1.1	June 2000	Publication