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P.10/G.100

Amendment 1
(06/2019)

SERIES P: TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Vocabulary and effects of transmission parameters on customer opinion of transmission quality

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

International telephone connections and circuits – Transmission planning and the E-model

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Vocabulary for performance, quality of service and quality of experience

Amendment 1

Recommendation ITU-T P.10/G.100 (2017) – Amendment 1

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Recommendation ITU-T P.10/G.100

Vocabulary for performance, quality of service and quality of experience

Amendment 1

Summary

Recommendation ITU-T P.10/G.100 contains terms and definitions associated with network performance, quality of service and quality of experience.

Amendment 1 is coordinated with a related revision to Recommendation ITU-T P.64, *Determination of sensitivity/frequency characteristics of local telephone systems*. It includes new definitions and a bibliographic reference.

History

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Recommendation ITU-T P.10/G.100

Vocabulary for performance, quality of service and quality of experience

Amendment 1

Editorial note: This is a complete-text publication. Modifications introduced by this amendment are shown in revision marks relative to Recommendation ITU-T P.10/G.100 (2017).

1 Scope

This Recommendation contains terms and definitions associated with network performance, quality of service and quality of experience.

2 References

None.

3 Definitions

3.1 Terms defined elsewhere

None.

3.2 Terms defined in this Recommendation

The terms defined by this Recommendation are contained in clause 6.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ACR	Absolute Category Rating
ARL	Acoustic Reference Level
ASR	Automatic Speech Recognition
CCR	Comparison Category Rating
CIF	Common Intermediate Format
CMOS	Comparison Mean Opinion Score
CLR	Circuit Loudness Rating
CQE	Communication Quality Estimated
CQO	Communication Quality Objective
CQS	Communication Quality Subjective
CSS	Composite Source Signal
DCME	Digital Circuit Multiplication Equipment
DCR	Degradation Category Rating
DL	Round-Trip Delay
DMOS	Degradation Mean Opinion Score

DMS	Digital Mobile System
DRP	Eardrum Reference Point
DRS	Digital Reference Sequence
ECRP	Ear Cap Reference Point
EEP	Ear Canal Entrance Point
ERP	Ear Reference Point
HATS	Head and Torso Simulator
HFRP	Hands-Free Reference Point
HRX	Hypothetical Reference Connection
Ie	Equipment Impairment Factor
INMD	In-service Non-intrusive Measurement Device
L_E	Earphone Coupling Loss
L_{ECHO}	Echo Loss
LELR	Listener Echo Loudness Rating
LQE	Listening-only Quality Estimated
LQS	Listening-only quality subjective
LR	Loudness Rating
LRP	Level Reference Point
LS	Local System
LSTR	Listener Sidetone Rating
MNRU	Modulated Noise Reference Unit
MOS	Mean Opinion Score
MPEG	Moving Picture Expert Group
MRP	Mouth Reference Point
MTF	Modulation Transfer Function
NCP	Network Connection Point
OLL	Open-Loop Loss
OLR	Overall Loudness Rating
PABX	Private Automatic Branch Exchange
PBN	Private Branch Network
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
QDU	Quantizing Distortion Unit
QoE	Quality of Experience
QoS	Quality of Service
RLR	Receive Loudness Rating
RML	Ratio Medium/Low

SI	Spatial Perceptual Information
SIF	Source input format
SLR	Send Loudness Rating
SM	Singing Margin
STI	Speech Transmission Index
STMR	Sidetone Masking Rating
TBRL	Test Balance Return Loss
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
TELR	Talker Echo Loudness Rating
TT	Transmission Time
TTT	Total Transmission Time
TI	Temporal Perceptual Information
TTS	Text-to-Speech Synthesis
TRP	Transmission Reference Point
XRLR	Crosstalk Receive Loudness Rating
VTC/VT	Video Teleconferencing/Video Telephony Service
VICP	Virtual International Connecting Point
VPN	Virtual Private Network

5 Conventions

None.

6 Terms and definitions

This Recommendation defines the following terms in alphabetic order:

6.1 3.1 kHz handset telephony

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

- presentation of an acoustical speech signal to the mouthpiece of a traditionally shaped handset:
  - either analogue transport of the said speech signal under real-time conditions through and by telecommunication networks: the said networks being intended for telephony applications between network termination points;
  - or filtering of the said speech signal to the frequency range from approximately 300 to 3 400 Hz; transformation of the said speech signal either by waveform or by non-waveform (speech analysis) encoder; transport and processing of the said speech signal under real-time conditions through and by telecommunication networks: the said networks being intended for telephony applications between network termination points; back transformation (speech synthesis) of the said speech signal by the respective decoder;

- acoustical presentation of the said speech signal in the frequency range from approximately 300 to 3 400 Hz by the earpiece of a traditionally shaped handset.

6.2 4-wire chain

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

6.3 absolute category rating (ACR) (see [b-ITU-T P.800])

"Test method in which subjects are asked to express classification opinions by using absolute quality scales (excellent, good, ...) for their judgement."

6.4 acceptance scale (see [b-ITU-T P.85])

Opinion scale for measuring the overall quality of the message on a service point of view. Acceptance requires a Yes/No answer.

6.5 acceptance test

A contractual test to prove to the customer that the device meets certain conditions of its specification.

6.6 acoustic artificial voice

Acoustic signal at the mouth reference point (MRP) of the artificial mouth. It complies with the same time and spectral specifications as the electrical artificial voice.

6.7 acoustic coupler (in telephonometry)

A cavity of defined shape and volume used for the testing of telephone earphones or telephone transmitters in conjunction with a calibrated microphone adapted to measure the pressure developed within the cavity.

6.8 acoustic hood

A hood lined with sound-absorbing material to facilitate the use of a telephone station by reducing the ambient noise level.

6.9 acoustic reference level (ARL) (see [b-ITU-T P.310], [b-ITU-T P.311], [b-ITU-T P.341] and [b-ITU-T P.342])

The acoustic level at MRP which results in a −10 dBm0 output at the digital interface.

6.10 acoustic shock suppressor (in telephony)

A device associated with a telephone station and intended to prevent acoustic shocks, by setting an upper limit to the absolute values of the instantaneous electrical voltage that can be applied to the telephone earphone.

6.11 acoustic startle

A psychological effect caused by acoustic stimulation that may cause disturbance to some users.

6.12 acoustical telephony gain (telephonic transfer function) (see [b-ITU-T P.58])

Ratio of the pressure at the ear reference point of a listener to the pressure at the mouth reference point of a talker connected by a telephone channel.

6.13 acoustically closed earphones (nominally sealed) (see [b-ITU-T P.57])

Earphones which are intended to prevent any acoustic coupling between the external environment and the ear canal.

6.14 acoustically open earphones (nominally unsealed) (see [b-ITU-T P.57])

Earphones which intentionally provide an acoustic path between the external environment and the ear canal.

6.15 active speech level (see [b-ITU-T P.56])

A quantity, expressed in decibels relative to a stated reference, e.g., volts or pascals, formed by averaging the speech-signal's power over the active time, according to [b-ITU-T P.56], method B.

6.16 active testing

Refers to the method in which data is acquired for the measurement, i.e., that the test makes use of a dedicated channel for the measurement, or by dialling a number and making a call, or by setting-up a channel for the measurement, or by transmitting packets or frames whose purpose is dedicated to measurement.

6.17 active time

Aggregate of all intervals of time when speech is deemed to be present according to the criterion adopted by ITU-T (see [b-ITU-T P.56]) for the purpose of measurement.

6.18 activity factor

Ratio of the active time to total time elapsed during a measurement, usually expressed as a percentage.

6.19 advantage factor

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

6.20 analogue network

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

6.21 articulation index

(Definition generally used in psychoacoustics)

A measure of the intelligibility of voice signals, expressed as a percentage of speech units that are understood by the listener when heard out of context. The articulation index is based on partially empirical, partially theoretical principles to predict the speech intelligibility under known signal-to-noise conditions.

6.22 articulation scale (see [b-ITU-T P.85])

Opinion scale for measuring the impression of clarity felt by a listener. How distinguishable are the words composing the message?

6.23 artificial conversational speech (see [b-ITU-T P.59])

A signal which reproduces the on-off characteristics of human conversational speech, especially useful for characterizing speech processing systems which have speech detectors, such as speakerphones, echo control devices and digital circuit multiplication equipment (DCME).

6.24 artificial ear

A device for the calibration of earphones incorporating an acoustic coupler and a calibrated microphone for the measurement of sound pressure and having an overall acoustic impedance similar to that of the average human ear over a given frequency band.

6.25 artificial mouth

A device consisting of a loudspeaker mounted in an enclosure and having a directivity and radiation pattern similar to those of the average human mouth.

6.26 artificial mouth excitation signal

A signal applied to the artificial mouth in order to produce the acoustic artificial voice. It is obtained by equalizing the electrical artificial voice for compensating the sensitivity/frequency characteristic of the mouth.

6.27 artificial voice

A mathematically defined signal which reproduces human speech characteristics, relevant to the characterization of linear and non-linear telecommunication systems. It is intended to give a satisfactory correlation between objective measurements and tests with real speech.

6.28 automatic speech recognition (ASR)

A process or a technology which accepts a natural speech signal as an input and provides, at the output, a coded version of what has been said (word, command, expression, sentence, etc.).

6.29 automatic speech recognition (ASR) system

An implementation in hardware or software that accepts a natural speech signal as an input and provides, at the output, a coded version of what has been said (word, command, expression, sentence, etc.).

6.30 balance return loss

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

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

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

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

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

6.31 band sensation level

The difference, expressed in decibels, between the sound integrated over a frequency band and the sound pressure level in that band at the threshold of audibility, there being no other disturbing sound.

6.32 block

A group of pixel. For example, a block of 8 × 8 pixel is the smallest coding block used in MPEG-1 algorithms. There are 1320 blocks in a SIF image, 44 in the horizontal direction (352 pixel/8) and 30 in the vertical direction (240 lines/8).

6.33 block distortion

Distortion of the image characterized by the appearance of an underlying block encoding structure. Also called tiling.

6.34 blurring

A global distortion over the entire image, characterized by reduced sharpness of edges and spatial detail.

6.35 call

The establishment and use of a complete connection following a call attempt.

6.36 call attempt (by a user)

A sequence of operations made by a user of a telecommunication network trying to connect to the desired user or service.

Associated term: to call.

6.37 circuit access point

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

6.38 circuit loudness rating (CLR) (see [b-ITU-T G.111])

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

6.39 circuit, telecommunication circuit

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

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

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

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

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

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

6.40 circum-aural earphones (see [b-ITU-T P.57])

Earphones which enclose the pinna and rests on the surrounding surface of the head. Contact to the head is normally maintained by compliant cushions. Circum-aural earphones may touch, but not significantly compress the pinna.

6.41 colour errors

Distortion of all or a portion of the final image characterized by the appearance of unnatural or unexpected hues or saturation levels. These hues or saturation levels were not present in the original image.

6.42 commissioning objective

(Defined in [b-ITU-T G.102])

6.43 common intermediate format (CIF)

Common intermediate format used by [b-ITU-T H.261] coders, 352 luminance pixel × 288 lines.

6.44 communication-quality component

Communication-quality component refers to how users perceive that the system is facilitating the communication. More specifically, it refers to those quality features related with communication aspects. Examples of such aspects are conversation flow, communication effort, cognitive load and task performance.

6.45 comparison category rating (CCR) (see [b-ITU-T P.800])

Test method in which subjects are asked to express opinion judgements using comparison category scale (much better, better, slightly better, ...).

6.46 comparison mean opinion score (CMOS) (see [b-ITU-T P.800])

The mean of opinion scores as defined in clause 6.184, when the CCR method is used to evaluate the performance of a telephone transmission system.

6.47 (complete) connection

A connection between users' terminals.

6.48 complexity for an ASR system

A measure of the mean length of the sentences which are accepted by the system.

6.49 composite loss

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

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

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

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

6.50 composite source signal (CSS)

A signal composed in time by various signal elements.

6.51 connected-word mode

A string of words spoken carefully, but with no explicit pauses between them.

6.52 connection

A temporary association of transmission channels or telecommunication circuits, switching and other functional units set up to provide the means of a transfer of information between two or more points in a telecommunication network.

6.53 continuous speech understanding system

A system that can recognize continuous speech, often having phoneme-sized references, using lexical, syntactic, semantic, and pragmatic knowledge, and reacts appropriately (therefore, having interpreted the message and found the corresponding action to be taken). This term describes the final objective of an ASR research.

6.54 continuous-speech mode

A string of words spoken fluently and rapidly as in conversational speech.

6.55 conversational quality

The quality of a bi- or multidirectional conversation as perceived by a communication partner.

6.56 conversational speech quality

The speech quality as experienced in a bi- or multidirectional conversation.

6.57 crest factor

Peak-to-RMS ratio of a signal.

6.58 crosstalk receive loudness rating (XRLR)

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

6.59 daily noise exposure

Daily noise exposure is a time-weighted-average of A-weighted noise exposure for a conventional 8-hour workday.

6.60 dB-related units

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

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

dBov relative power level expressed in decibels, referred to the overload point of the digital system;

dBfs relative power level expressed in decibels, referred to the maximum possible digital level (full scale);

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

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

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

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

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

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

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

NOTE – A discussion of the applications of other dB-related terms is given in Appendix I of [b-ITU-T G.100.1].

6.61 degradation category rating (DCR) (see [b-ITU-T P.800])

A modification of the ACR test method where subjects compare the system under test with a reference system and express their opinion using a degradation scale (degradation inaudible, audible but not annoying, slightly annoying, ...).

6.62 degradation mean opinion score (DMOS) (see [b-ITU-T P.800])

The mean of opinion scores as defined in clause 6.184, when the DCR method is used to evaluate the performance of a telephone transmission system.

6.63 deletion error

An error in an ASR process in which a valid spoken word is ignored and no response is produced by the system.

6.64 DELSm (Δ_{Sm})

$Delta_{Sm}$ is defined as the difference between the sending sensitivity of a telephone set using an **artificial mouth** S_{mJ}, and that using a diffuse room noise source $S_{mJ/RN}$, such that

$$\Delta_{Sm} = S_{mJ/RN} - S_{mJ} \text{ dB}$$

(See also [b-ITU-T P.11], [b-ITU-T P.64], [b-ITU-T P.76], [b-ITU-T P.79] and the Handbook on Telephonometry.)

6.65 DELSM (Δ_{SM})

$Delta_{SM}$ is defined as the difference between the sending sensitivity of a telephone set using a **real mouth and voice**, S_{MJ}, and that using a diffuse room noise source $S_{MJ/RN}$, such that

$$\Delta_{SM} = S_{MJ/RN} - S_{MJ} \text{ dB}$$

(See also ITU-T [b-ITU-T P.11], [b-ITU-T P.64], [b-ITU-T P.76], [b-ITU-T P.79] and the Handbook on Telephonometry.)

NOTE – For most practical purposes Δ_{SM} will be closely approximated by the quantity Δ_{Sm}, which is easier to determine.

6.66 design objective

(Defined in [b-ITU-T G.102].)

6.67 digital mobile system (DMS) (see [b-ITU-T G.173])

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

6.68 digital transport

Communication using digital methods for the transmission of signals from one point to another.

6.69 diphone synthesis

Synthesis technique is based on the use of segments of speech which correspond to two consecutive sounds and cover an interval of time going from the middle part of the first sound to the middle part of the second sound.

6.70 double-ended

Refers to a type of measurement, with regard to the point(s) of the network where the signal(s) to be tested is (are) acquired. A double-ended measurement would require access to two sides of a telephony connection to, e.g., intrusively send a reference signal through a network under test while the resulting test signal will be recorded at another termination point of this network.

6.71 double talk

An operation mode, where two users are speaking simultaneously.

6.72 double talk interval

The interval during which both directions of transmission are experiencing incident speech spurts. (At the in-service non-intrusive measurement device (INMD) monitoring point, this will be different from the double talk experienced by both parties due to the delay between the termination points and the measurement equipment.)

6.73 ear canal entrance point (EEP) (see [b-ITU-T P.57])

A point located at the centre of the ear canal opening.

6.74 ear canal extension (see [b-ITU-T P.57])

Cylindrical cavity, extending the simulation of the ear canal provided by the occluded ear simulator ([b-ITU-T P.57], Type 2) out to the concha cavity.

6.74*bis* ear cap

A part of a telephone earpiece, adapted to couple to the human pinna.

6.75 ear cap reference plane

That plane formed by the contacting points of a flat surface against a telephone ear cap.

6.76 ear cap reference point (ECRP)

Point in the ear cap reference plane, used as a reference parameter.

6.76*bis* earpiece

The part of a telephone, radio receiver, or other aural device that is applied to the ear during use.

NOTE – See [b-Oxford]

6.77 ear reference point (ERP) (see [b-ITU-T P.57])

A virtual point for geometric reference located at the entrance to the listener's ear, traditionally used for calculating telephonometric loudness ratings.

6.78 ear simulator (see [b-ITU-T P.57)

Device for measuring the output sound pressure of an earphone under well-defined loading conditions in a specified frequency range. It consists essentially of a principal cavity, acoustic load networks, and a calibrated microphone. The location of the microphone is chosen so that the sound pressure at the microphone corresponds approximately to the sound pressure existing at the human ear drum.

6.79 eardrum reference point (DRP) (see [b-ITU-T P.57])

A point located at the end of the ear canal, corresponding to the eardrum position.

6.80 earphone coupling loss (L_E)

That quantity defined as the receiving sensitivity of a handset (usually as a function of frequency) when applied to an artificial ear minus the receiving sensitivity of the same handset on a human ear.

6.81 echo

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

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

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

6.82 echo (in telephony) (see [b-ITU-T P.561])

An unwanted delayed version of the directly transmitted signal, returned to the listener.

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

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

NOTE 3 – Typically, the talker is also the listener.

6.83 echo balance return loss

Balance return loss averaged with 1/f power weighting over the telephone band, in accordance with clause 4 of [b-ITU-T G.122].

6.84 echo control device

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

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

6.85 echo loss

The echo loss ([b-ITU-T G.122]) is derived from the integral of the power transfer characteristic weighted by a negative slope of 3 dB/octave starting at 300 Hz and extending to 3400 Hz. The echo loss should be calculated with the speech echo path delay removed. This echo loss figure has been found to give better agreement with subjective opinion for individual connections than an unweighted echo path loss. For a flat echo path frequency response, echo loss is equal to speech echo path loss and echo path loss.

6.86 echo loss (L_{ECHO})

Semi-loop loss averaged with 1/f power weighting over the telephone band, in accordance with clause 4 of [b-ITU-T G.122].

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

NOTE 2 – Distinction may be made between the echo loss of a given piece of equipment and that of a national system (see Note 2 to definition in clause 6.235).

6.87 echo path

The round-trip electrical path starting from the point of incident speech measurement and ending at the point where the correlated reflected speech is measured.

6.88 echo path loss

The echo path has a unique impulse response. The echo path loss is the integral of the impulse response (in the frequency domain). Echo path loss is not dependent on the speaker.

6.89 edge busyness

Distortion concentrated at or near the edge of objects, and further categorized by its temporal and spatial characteristics.

6.90 electrical artificial voice

The artificial voice produced as an electric signal, for testing transmission channels or other electric devices.

6.91 E-model

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

6.92 end-to-end quality

Quality related to the performance of a communication system, including all terminal equipment. For voice services it is equivalent to mouth-to-ear quality. For data services it is equivalent to user to user quality, or user to content server quality, or machine to machine quality where there is no user directly involved.

6.93 equipment impairment factor (Ie)

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

6.94 error blocks

A form of block distortion where one or more blocks in the image bear no resemblance to the current or previous scene and often contrast greatly with adjacent blocks.

6.95 explicit reference (source reference)

The condition used by the assessors as reference to express their opinion, when the DCR method is used. This reference is displayed first within each pair of sequences. Usually the format of the explicit reference is the format used at the input of the codecs under test (e.g., [b-ITU-R BT.601-7], CIF, QCIF, SIF, etc.).

6.96 extension line

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

6.97 fluctuation strength

(Definition generally used in psychoacoustics)

The amplitude- or frequency modulation of tones lead to different hearing events. If the envelope fluctuation is below 20 Hz the characterization for such a sound is fluctuation strength. The human ear is able to follow the fluctuation of the signal.

6.98 formant synthesis

Synthesis technique based on the use of formant and excitation parameters in which the target positions of those parameters (associated with each phonetic unit) and rules of interpolation act, are used.

6.99 fullband signal

A signal that has no significant signal components outside the range 10 Hz to 20 000 Hz. A fullband signal can be generated by applying the filter from Table 1 to an unfiltered source signal recorded without band limitation in a high quality recording environment. The application of sub-sonic and rumble filters to the source before filtering to fullband is allowed.

NOTE –The low-pass filter definition does not emulate a fullband transmission system, channel, device or codec.

Table 1 – Minimum bandwidth filter definition to derive a fullband signal from a unfiltered source signal

Frequency (Hz)	Fullband gain (dB)
10	−20 (max)
20	0 to −3
30	0
19 500	0
20 000	0 to −3
21 000	−40 (max)
24 000	−50 (max)

NOTE 1 – "−40 dB (max.)" should be interpreted as a minimum stop-band attenuation of 40 dB in the filter definition (and thus a maximum stop-band gain of 40 dB).

NOTE 2 – "−50 dB (max.)" should be interpreted as a minimum stop-band attenuation of 50 dB in the filter definition (and thus a maximum stop-band gain of −50 dB).

NOTE 3 – "−20 dB (max.)" should be interpreted as a minimum stop-band attenuation of 20 dB in the filter definition (and thus a maximum stop-band gain of −20 dB).

NOTE 4 – The values define the minimum attenuation; it can be exceeded until the technical maximum.

6.100 fullband telephony

Transmission of speech with a nominal pass-band wider than 50-14 000 Hz, usually understood to be 20-20 000 Hz (see normal-band telephony, wideband telephony and super-wideband telephony).

6.101 full-reference model

Refers to a type of algorithm; in a full-reference model the algorithm requires a reference signal, which is usually compared to a test signal at the output of a device under test.

6.102 gamma

A parameter that describes the discrimination between the grey level steps on a visual display. The relation between the screen luminance and the input signal voltage is non-linear, with the voltage raised to an exponent gamma. To compensate for this non-linearity, a correction factor that is an inverse function of gamma is generally applied in the camera. Gamma also has an impact on colour rendition.

6.103 group-audio terminal

A speakerphone set primarily designed for use by several users which will not be equipped with a handset.

6.104 group-delay distortion

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

6.105 guard-ring

Annular ring fitted, during tests, onto the transmitter housing of a telephone handset, to localize the sound source in a prescribed position relative to the microphone.

6.106 handset

A device which includes telephone receiver and transmitter which is typically coupled to the ear by hand.

6.107 handset telephone

A telephone set equipped with a handset.

6.108 hands-free reference point (HFRP) (see [b-ITU-T P.340], [b-ITU-T P.341] and [b-ITU-T P.342])

A point located on the axis of the artificial mouth, at 50 cm from the outer plane of the lip ring, where the level calibration is made, under free-field conditions. It corresponds to the measurement point 11, as defined in [b-ITU-T P.51].

6.109 hands-free terminal

A telephone set that does not require the use of hands during the communications session; examples are headset, speakerphone and group-audio terminal.

6.110 head and torso simulator (HATS) (see [b-ITU-T P.58])

Manikin extending downward from the top of the head to the waist, designed to simulate the sound pick-up characteristics and the acoustic diffraction produced by a median human adult and to reproduce the acoustic field generated by the human mouth.

6.111 headset

A device which includes telephone receiver and transmitter which is typically secured to the head or the ear of the wearer.

6.112 hollowness

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

NOTE – Hollowness is to be distinguished from listener echo.

6.113 hypothetical reference connection (HRX)

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

6.114 impairment factor

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

6.115 implicit reference

The condition used by the assessors as reference to express their opinion on the test material, when the ACR method is used. If the implicit reference is suggested by the experimenter, it must be well known to all the assessors (e.g., conventional TV systems, reality), but the condition is not explicitly presented to the subjects as a reference by the experiment designer.

6.116 input/output (see ITU-T [b-ITU-T G.111], [b-ITU-T G.121], etc.)

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

6.117 insert earphones (see [b-ITU-T P.57])

Earphones which are intended to partially or completely enter the ear canal.

6.118 insertion

A case of recognition due to either a spurious noise or an utterance which is not legitimate according to the syntax. Such a noise is either not properly rejected or a word not belonging to the recognition vocabulary is falsely accepted as an utterance from the active vocabulary.

6.119 in-service measurement

In-service measurement means that the results are obtained during regular use of a network.

6.120 interruptibility (see [b-ITU-T G.114])

The possibility for one party in a telephone conversation to interrupt the other party, as in normal conversation. Interruptibility can be affected by the use of voice-activated devices, by total transmission time, etc.

6.121 intra-concha earphones (see [b-ITU-T P.57])

Earphones which are intended to rest within the concha cavity of the ear. They have an external diameter (or maximum dimension) of less than 25 mm but are not made to enter the ear canal.

6.122 intrusive testing

According to the definitions in clause 3.10.3 of [b-ITU-T X.745], intrusive test means: "A statement made with respect to a test invocation if service/user disruption will or may occur as a result of the test". This refers to the way that data is acquired for the measurement, i.e., whether or not sending a specific predefined and known reference signal over a channel for analysis purposes is required.

NOTE 1 – In contrast to active testing, intrusive testing means that a test signal is sent over the network. The terms "active" and "intrusive" are often wrongly used as synonyms.

NOTE 2 – The combinations 'active and intrusive testing' and 'passive and non-intrusive testing' define the most common test situations.

6.123 isolated-word mode

Single words pronounced with explicit pauses between them.

6.124 jerkiness (or jerky motion)

Motion that was original smooth and continuous is perceived as a series of distinct "snapshots".

6.125 limits for maintenance purposes; maintenance limits

(Defined in [b-ITU-T G.102]).

6.126 lip plane (see [b-ITU-T P.51] and [b-ITU-T P.58])

Outer plane of the lip ring. The lip plane (of the artificial mouth or the HATS) is normally different from the plane of the mouth simulator orifice. The lip plane is vertically oriented when the HATS is in the reference position.

6.127 lip ring (see [b-ITU-T P.51] and [b-ITU-T P.58])

Circular ring of thin rigid rod, having a diameter of 25 mm and less than 2 mm thick. It shall be constructed of non-magnetic material and be solidly fixed to the artificial mouth or the HATS. The lip ring defines both the reference axis of the mouth and the mouth reference point.

6.128 lip synchronization

Operation to provide the feeling that the speaking motion of the displayed person is synchronized with that person's voice. The minimization of the relative delay between the visual display of a person speaking and the audio of the voice of the person speaking. The objective is to achieve a natural relationship between the visual image and the aural message for the viewer/listener.

6.129 listener echo loss; receive echo loss

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

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

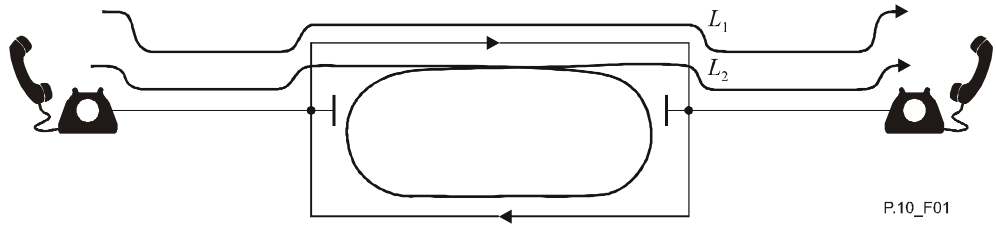

Figure 1 – Listener echo loss; receive echo loss

6.130 listener echo loudness rating (LELR)

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

6.131 listener echo; receive end echo

Echo produced by double reflected signals and disturbing the listener, receiving voiceband data equipment, etc.

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

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

6.132 listener sidetone rating (LSTR)

The loudness of a diffuse room noise source as heard at the subscriber's (earphone) ear via the electric sidetone path in the telephone instrument, compared with the loudness of the Intermediate Reference System (IRS) overall, in which the comparison is made incorporating a speech signal heard via the human sidetone path (L_{MEHS}) as a masking threshold.

6.133 listening effort scale (see [b-ITU-T P.800] and [b-ITU-T P.830])

Opinion scale for measuring the difficulty of the task performed by a person listening to a voice message, in order to understand the content of the message.

6.134 local (telephone) system (LS)

The combination of subscriber's station, subscriber's line and feeding bridge if existing.

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

6.135 local line network

All the subscribers' telephone lines and ancillary equipment provided to connect subscribers to their local switching entity.

6.136 long duration noise disturbance

A noise signal equal to or more than 500 ms duration.

6.137 loudness

(Definition generally used in psychoacoustics)

Loudness belongs to a category of intensity sensations. Loudness is that attribute of auditory sensation in terms of which sounds can be ordered on a scale extending from quiet to loud. Loudness takes into account the spectral and temporal sensitivity of the human ear. Generally masking effects in time and frequency are taken into account. The loudness level measure according to [b-Zwicker] was created to characterize the loudness sensation of tones. The loudness calculation procedure for stationary signals is defined in [b-ISO 532]. For the calculation of the loudness of time variant signal different models are known.

Specific definition used in telecommunication

In telecommunication, the generally accepted loudness measurement methodology is defined as Loudness Ratings in [b-ITU-T P.79]. The ITU-T Loudness Ratings calculations do not take into account the masking effects.

6.138 loudness rating (LR)

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

How to determine and to apply LRs in the G-series Recommendations is described in [b-ITU-T G.100.1]. The methods are sufficiently accurate for all practical purposes. (Fundamentally, loudness ratings are based on subjective methods as described in [b-ITU-T ITU-T P.76] and [b-ITU-T P.78]. However, subjectively measured values, in general, vary too much with time and test teams to be really useful for transmission planning.)

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

6.139 mean one-way propagation time

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

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

6.140 mean opinion score (MOS) (see [b-ITU-T P.800])

The mean of opinion scores.

6.141 media-signal-quality component

Media-signal-quality component refers to how users perceive that the system is providing a good signal quality. It is here considered as the combination of two components, spatial-quality component and non-spatial-quality component.

6.142 metre air path

Measured reference of sound pressure loss over a 1-metre air path. In an anechoic environment, the sound pressure attenuation of such a path is approximately 30 dB measured from the MRP.

6.143 mixed analogue-digital channel (circuit)

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

6.144 modal distance

Distance between the centre of the microphone protective grid or front sound opening on a handset, and the centre of the guard-ring.

6.145 modal gauge

Template used to check a guard-ring position on a handset relative to the receiver earcap reference plane.

6.146 modal position

Prescribed position and inclination of a handset relative to a fixed sound source.

6.147 modulated noise reference unit (MNRU) (see [b-ITU-T P.810])

A device producing a calibrated distortion which is subjectively similar to that produced by logarithmically companded PCM systems. The MNRU distortion is expressed in decibels corresponding to the ratio of a signal to the multiplicative noise.

6.148 modulation transfer function (MTF) (see [b-ITU-T P.501])

Modulation signal, derived from the envelope of a test signal. Typically, the modulation is determined in different bands. The procedure is widely used in room acoustics, mainly for determining speech intelligibility of reverberant speech signals using the STI method.

6.149 MOS-CQE

Mean opinion score – communication quality estimated

The score is calculated by a network planning model which aims at predicting the quality in a conversational application situation. Estimates of conversational quality carried out according to [b-ITU-T G.107], when transformed to mean opinion score, give results in terms of MOS-CQE.

6.150 MOS-CQO

Mean opinion score – communication quality objective

The score is calculated by means of an objective model which aims at predicting the quality for a conversational test situation. Objective measurements made using the model given in [b ITU-T P.562] give results in terms of MOS-CQO.

6.151 MOS-CQS

Mean opinion score – Communication quality subjective

The score has been collected in a laboratory test by calculating the arithmetic mean value of subjective judgments on a 5-point ACR quality scale, as it is defined in [b-ITU-T P.800]. Subjective conversation tests carried out according to [b-ITU-T P.800] give results in terms of MOS-CQS.

6.152 MOS-LQE

Mean opinion score – Listening-only quality estimated

The score is calculated by a network planning model which aims at predicting the quality in a listening-only application situation.

6.153 MOS-LQO

Mean opinion score – Listening-only quality objective

The score is calculated by means of an objective model which aims at predicting the quality for a listening-only test situation. Objective measurements made using the model given in [b-ITU-T P.862] give results in terms of MOS-LQO.

6.154 MOS-LQS

Mean opinion score – Listening-only quality subjective

The score has been collected in a laboratory test by calculating the arithmetic mean value of subjective judgments on a 5-point ACR quality scale, as it is defined in [b-ITU-T P.800]. Subjective tests carried out according to [b-ITU-T P.830] give results in terms of MOS-LQS.

6.155 mosquito noise

A form of edge business distortion sometimes associated with movement, characterized by moving artefacts around edges and/or blotchy noise patterns superimposed over the objects (resembling a mosquito flying around a person's head and shoulders).

6.156 MOS-TQE

The score is calculated by a network planning model which aims at predicting the quality in a talking-only application situation. No methods generating a MOS-TQE are currently standardized.

6.157 MOS-TQO

The score is calculated by means of an objective model which aims at predicting the quality for a talking-only test situation. Methods generating a MOS-TQO are currently under development and not yet standardized.

6.158 MOS-TQS

The score has been collected in a laboratory test by calculating the arithmetic mean value of subjective judgments on a 5-point ACR quality scale, as it is defined in [b-ITU-T P.800].

6.159 motion response degradation

The deterioration of motion video such that the video imagery has suffered a loss of spatio-temporal resolution.

6.160 motion video

Temporally varying visual imagery intended to communicate or convey movement or change.

6.161 motion-related artefacts

Distortion of motion video potentially observable by the viewer. In some instances, the distortion becomes more observable with increased motion. The distortion may appear as smearing, block distortion, jerkiness, or other impairments.

6.162 mouth reference point (MRP) (see [b-ITU-T P.51] and [b-ITU-T P.58])

Point 25 mm in front of and on the axis of the lip plane of the artificial mouth or a typical human mouth (see Figure A.1 of [b-ITU-T P.64]).

6.163 mouth-to-ear quality

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

6.164 MPEG standards

Multimedia/systems standards developed by the moving picture expert group (MPEG), a working group organized by ISO.

6.165 multimedia terminals

Terminals for multimedia services usually including video and/or audio and/or data.

6.166 narrowband signal

A signal that has no significant signal components outside the range 20 Hz to 4 000 Hz. A narrowband signal can be created by applying the filter from Table 2 to a fullband signal as defined in this Amendment, the upper cut-off frequency of 3 800 Hz is due to a realistic interpolation low-pass when re-sampling to 8 000 Hz sampling frequency.

The narrowband signal can also be derived by applying the filter in Table 2 to a wideband or super-wideband signal as defined in this Recommendation.

NOTE –The low-pass filter definition does not emulate a narrowband transmission system, channel, device or codec.

Table 2 – Minimum bandwidth filter definition to derive a narrowband signal from a fullband signal

Frequency (Hz)	Narrowband gain (dB)
20	0
3 700	0
3 800	0 to −3
4 000	−40 (max.)
NOTE 1 – "−40 dB (max.)" should be interpreted as a minimum stop-band attenuation of 40 dB in the filter definition (and thus a maximum stop-band gain of 40 dB). NOTE 2 – The value defines the minimum attenuation; it can be exceeded until the technical maximum.	

6.167 narrow-band telephony

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

6.168 national system

The national system starting at the VICP may comprise of one or more 4-wire national trunk circuits with 4-wire interconnection, as well as circuits with 2-wire connection up to the local exchange, the subscriber stations with their subscriber lines or a PBN.

6.169 network end-point

A network end-point refers to a terminal at the receiving user's side which is connected to the measurement system either electrically or acoustically.

6.170 network head-point

A network head-point refers to a terminal at the sending user's side which is connected to the measurement system either electrically or acoustically.

6.171 network mid-point

A network mid-point refers to any point in the network that is not the head point or the end-point which is connected to the measurement system either electrically or acoustically.

6.172 noise level

The electrical energy (measured in dBmp) caused by spurious signals. Spurious signals, i.e., noise, can be generated internally to the circuit or may be the result of interference from external sources.

6.173 non-intrusive testing

According to the definitions in clause 3.10.5 of Recommendation [b-ITU-T X.745], non-intrusive test means: "A statement made with respect to a test invocation if no service/user disruption will or may occur as a result of the test". This refers to the way that data is acquired for the measurement, i.e., whether or not sending a specific predefined and known reference signal over a channel for analysis purposes is required.

NOTE 1 – In contrast to passive testing, non-intrusive testing means that no specific test signal is sent over the network. The terms "passive" and "non-intrusive" are often wrongly used as synonyms.

NOTE 2 – The combinations 'active and intrusive testing' and 'passive and non-intrusive testing' define the most common test situations.

6.174 non-spatial-quality component

Non-spatial-quality component refers to how users perceive that the system is providing undistorted speech and audio signals irrespectively of their spatial attributes. More specifically, it refers to those quality features related to the non-spatial attributes of sound. Examples of such attributes are loudness, timbre, noisiness, and sharpness.

6.175 no-reference model

Refers to a type of algorithm; in a no-reference model the algorithm requires the test signal only to compute a measurement.

6.176 object persistence

Distortion where the object(s) that appeared in a previous video frame (and should no longer appear) remain(s) in current and subsequent video frames as an outline or faced image.

6.177 object retention

Distortion where a fragment of an object that appeared in a previous video frame (and should no longer appear) remains in the current and subsequent video frames.

6.178 obstacle effect; obstruction effect

The change in the acoustic field close to a human or artificial mouth as obstacles (e.g., telephone transmitter) are brought into close proximity.

6.179 occluded-ear simulator (see [b-ITU-T P.57])

Ear simulator which simulates the inner part of the ear canal, from the tip of an ear insert to the ear drum.

6.180 occlusion effect

The change in human sidetone that occurs when the ear canal is occluded, e.g., by a telephone receiver.

6.181 out-of-service measurement

Out-of-service measurements are performed when the network is out of service.

6.182 (one-way) voice transmission quality

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

6.183 open-loop loss (OLL)

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

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

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

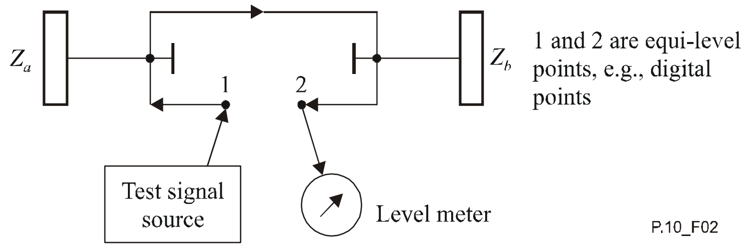

Figure 2 – Open-loop loss (OLL)

6.184 opinion score (in telephony)

The value on a predefined scale that a subject assigns to his opinion of the performance of the telephone transmission system used either for conversation or only for listening to spoken material.

6.185 optimization tests

Subjective tests that are typically carried out during either the development or the standardization of a new algorithm or system. The goal of these tests is to evaluate the performance of new tools in order to optimize the algorithms or the systems that are under study.

6.186 optimum listening level

The speech level that in a listening or conversation test corresponds to the highest opinion score on a quality scale (a rating scale going from "Excellent" to "Bad").

NOTE – It has been shown that the optimum listening level may be significantly higher than the preferred listening level. This indicates the importance of making a distinction between the optimum and preferred listening levels.

6.187 orthoreference acoustic gain for telephony (see [b-ITU-T P.58])

Ratio of the pressure at the ear reference point of the listener to the pressure at the mouth reference point of the talker under orthoreference conditions for telephony.

6.188 orthoreference condition for telephony (see [b-ITU-T P.58])

Acoustic path between a talker and a listener, facing each other at a distance of 1 metre in the free field.

6.189 orthotelephonic gain (insertion gain) (see [b-ITU-T P.58])

Ratio of the total electroacoustic gain to the orthotelephonic acoustic reference gain.

6.190 overall quality

The perceived quality of the system that is judged upon the totality of quality features that the user considers for the judgment. Overall quality is here considered as the combination of two main components, a media-signal-quality component and a communication-quality component.

6.191 overall loudness rating (OLR)

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

Refers to the method which data is acquired for the measurement, by observation only, e.g., that the test makes use of an established channel for the measurement, or by tapping a further defined point of this channel to observe the signals or packets that are normally present.

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

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

6.193 PCM digital reference sequence (DRS)

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

6.194 pel (or pixel)

A picture element that describes the brightness or colour of a discrete point in an image.

6.195 performance objective

(Defined in [b-ITU-T G.102].)

6.196 pinna simulator (see [b-ITU-T P.57])

A device which has the approximate shape of dimensions of a median adult human pinna.

6.197 pitch

(Definition generally used in psychoacoustics)

Pitch is an attribute of an auditory image that reflects listeners' impression on the location of the dominant spectral component along the frequency scale. In the case of complex harmonic tones, the pitch corresponds to a frequency close to the frequency difference between the harmonic components, i.e., the fundamental frequency.

6.198 preferred listening level

The speech level that, in a listening or conversation test, is judged as preferred on a loudness preference scale (a rating scale going from "(much) louder than preferred" to "(much) quieter than preferred").

NOTE – See "optimum listening level".

6.199 private (telephone) installation

A telephone network installed on the premises of a single individual or organization.

NOTE – By convention, private telephone installations include sets of telephone stations which are connected to one subscriber's line.

6.200 private automatic branch exchange (PABX)

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

6.201 private branch exchange (PBX)

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

6.202 private branch network (PBN)

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

6.203 private network

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

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

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

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

6.204 public switched telephone network (PSTN)

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

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

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

6.205 Q (see [b-ITU-T P.800], [b-ITU-T P.810] and [b-ITU-T P.830])

The ratio, in dB, of speech power-to-modulated noise power in the Modulated Noise Reference Unit, as described in [b-ITU-T P.810].

6.206 QCIF

Quarter CIF, 176 luminance pixel × 144 lines.

6.207 $\mathbf{Q_N}$ (see [b-ITU-T P.810] and [b-ITU-T P.830])

Q for a narrow-band Modulated Noise Reference Unit.

6.208 qualification tests

Subjective tests that are typically carried out in order to compare the performance of commercial systems or equipment. These tests must be carried out under test conditions that are as representative as possible of the real conditions of use.

6.209 quality of experience (QoE)

The degree of delight or annoyance of the user of an application or service. [b-Qualinet2013]

NOTE – Recognizing on-going research on this topic, this is a working definition which is expected to evolve for some time. (This note is not part of the definition.)

6.210 QoE influencing factors

Include the type and characteristics of the application or service, context of use, the user's expectations with respect to the application or service and their fulfilment, the user's cultural background, socio-economic issues, psychological profiles, emotional state of the user, and other factors whose number will likely expand with further research.

6.211 QoE assessment

The process of measuring or estimating the QoE for a set of users of an application or a service with a dedicated procedure, and considering the influencing factors (possibly controlled, measured, or simply collected and reported). The output of the process may be a scalar value, multi-dimensional representation of the results, and/or verbal descriptors. All assessments of QoE should be accompanied by the description of the influencing factors that are included. The assessment of QoE can be described as comprehensive when it includes many of the specific factors, for example a majority of the known factors. Therefore, a limited QoE assessment would include only one or a small number of factors.

6.212 quality of service (QoS)

The totality of characteristics of a telecommunications service that bear on its ability to satisfy stated and implied needs of the user of the service (see [b-ITU-T E.800]).

6.213 quantization noise

A "snow" or "salt and pepper" effect similar to a random noise process but not uniform over the image.

6.214 quantizing distortion unit (qdu) (see [b-ITU-T G.113])

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

6.215 Q_W (see [b-ITU-T P.810] and [b-ITU-T P.830)

Q for a wideband modulated noise reference unit.

6.216 R or T pads (in telephone extension)

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

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

6.217 ITU-R BT.601 format

ITU-R (formerly CCIR) digital video standard using interlaced formats of 720 luminance pixel × 480 lines × 30 Hz and 720 luminance pixel × 576 lines × 25 Hz.

6.218 receive loudness rating (RLR)

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

6.219 reduced-reference model

Refers to a type of algorithm; in a reduced-reference model the algorithm requires the test signal and a set of parameters derived from the reference signal to compute a measurement.

6.220 reference axis (of the mouth or the HATS)

The line perpendicular to the lip plane containing the centre of the lip ring.

6.221 reference conditions

Dummy conditions added to the test conditions in order to anchor the evaluations coming from different experiments.

6.222 reference position of HATS

The reference position of the HATS in the test space is intended to simulate a person in the upright position. The HATS is in the reference position when the following conditions are met:

– the reference point coincides with the test point;

– the HATS reference plane is horizontal.

6.223 rejection

The ability to reject spurious inputs e.g., noise, or utterances which are not parts of the active vocabulary.

– False acceptance (non-reject): a case of failure to reject input utterances which are not parts of the active vocabulary, thus resulting in the selection of a word in the vocabulary (very damaging from an ergonomic point of view).

– Wrong rejection: a case of failure to recognize a valid utterance, which is thus rejected by the system.

6.224 relative (power) level

The relative level at a point on a circuit is given by the expression $10 \log_{10} (P/P_0)$ dBr, where P represents the apparent power of a sinusoidal signal at the reference frequency 1020 Hz at the point concerned and P_0 the apparent power of that signal at the transmission reference point. This is numerically equal to the composite gain between the transmission reference point and the point concerned (or the composite loss between the point concerned and the transmission reference point), for the reference frequency 1020 Hz. For example, if a 1020 Hz signal having a level of x dBm is injected at a point in the circuit and the level measured at the transmission reference point is 0 dBm, the relative level at the point is x dBr. If y dBm is measured at another point in the circuit, the relative level at that point is y dBr.

6.225 relative level (at a point on a circuit)

The expression $10 \log_{10} (P/P_0)$ dBr where P represents the power of a test signal of 1 000 Hz at the point concerned and P_0 the power of that signal at the transmission reference point.

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

6.226 reliability of a subjective test

a) Intra-individual ("within subject") reliability refers to the agreement between a certain subject's repeated ratings of the same test condition.

b) Inter-individual ("between subjects") reliability refers to the agreement between different subject's ratings of the same test condition.

6.227 replication

Repeated presentation of the same circuit condition (with the same source material) to the same subject.

6.228 resolution

A parameter that specifies the ability to distinguish video detail in the spatial dimension or the temporal dimension.

6.229 return loss

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

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

6.230 ratio medium/low (RML)

(Specific definition used in telecommunication)

RML is the ratio of the energy in the 2/3-octave band, 1.5-kHz centred frequency, to the energy in the 2/3-octave band, 0.5-kHz centred frequency. This descriptor was defined by systematic analysis of the long-term spectra of speech recordings.

6.231 roughness

(Definition generally used in psychoacoustics)

The amplitude or frequency modulation of tones lead to different hearing events. A sound is perceived as rough if the envelope fluctuation is within the frequency range from 20 Hz to 300 Hz. The roughness perceived depends on the modulation frequency and the modulation depth.

6.232 round-trip delay (DL)

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

6.233 scene cut

Video imagery where consecutive frames are highly uncorrelated.

6.234 scene cut response

The perceived impairments associated with a scene cut. For example, a slow build-up of a video image instead of an instantaneous change of images.

6.235 semi-loop loss (possible alternative to the definition in P-1)

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

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

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

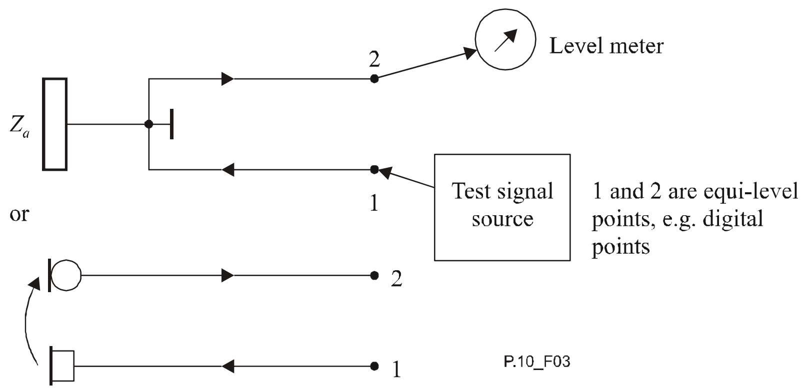

Figure 3 – Semi-loop loss

6.236 send loudness rating (SLR)

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

6.237 sharpness (also used: thinness)

(Definition generally used in psychoacoustics)

Sharpness is the centre of gravity of the spectrum and gives information on the balance between high and low frequency energy in the sound. As more the centre of gravity (of the spectral envelope) is moved to higher frequencies, as sharper a sound is perceived.

6.238 short duration noise disturbance

An instantaneous impulse noise signal of less than 500 ms duration.

6.239 sidetone balance network

An electrical network as part of a 2- to 4-wire balance point within a telephone set circuit for the purpose of controlling the telephone sidetone path loss.

6.240 sidetone masking rating (STMR)

The loudness of a telephone sidetone path compared with the loudness of the intermediate reference system (IRS) overall in which the comparison is made incorporating the speech signal heard via the human sidetone path L_{MEHS} as a masking threshold.

6.241 sidetone path

Any path, acoustic, mechanical or electrical, by which a telephone user's speech and/or room noise is heard in his own ear(s) (at ERP).

6.242 sidetone path loss

The loss of the sidetone path expressed as a loss compared with the speech at the MRP. Symbols in common use are:

L_{MEHS} for sidetone paths within a human head;

L_{MEST} for electro-acoustic sidetone paths within the telephone set;

L_{MEMS} for mechanical sidetone paths within a telephone handset;

L_{RNST} for electro-acoustic sidetone path from a diffuse room noise source to the earphone.

Each of these paths may be measured as sensitivities, in which case they become S_{MEHS}, S_{MEST}, S_{MEMS} and S_{RNST}, and experience a change of sign. Thus, for example, $S_{MEST} = -L_{MEST}$.

6.243 SIF

Source input format (SIF) used by MPEG coders, a progressive, non-interlaced format of 352 luminance pixel × 240 lines × 29.97 Hz or 352 luminance pixel × 288 lines × 25 Hz.

6.244 singing margin (SM)

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

6.245 single-ended

Refers to a type of measurement, with regard to the point(s) of the network where the signal(s) to be tested is (are) acquired. A single-ended measurement would require only access to one side of a network, e.g., at a subscriber's termination point.

6.246 single talk

An operation mode, where only one user is speaking.

6.247 smearing

A localized distortion over a sub-region of the received image, characterized by reduced sharpness of edges and spatial detail. For example, the portrayal of a fast-moving object may exhibit smearing.

6.248 spaciousness

(Definition generally used in psychoacoustics)

Spaciousness is a multidimensional perception of the auditory image that reflects a listener′s impression of the location of a sound source and of the characteristics of the space in which the sound event exists. While the perception of loudness, pitch, duration and timbre is restricted to monotic hearing, the perception of spaciousness typically arises from dichotic stimulation.

6.249 spatial application

An application needing high spatial resolution, possibly at the expense of reduced temporal resolution (or increases jerkiness). For example, spatial applications include the ability to read small characters and see fine detail in still video or motion video which contains a very limited amount of movement.

6.250 spatial edge noise

A form of edge business characterized by spatially varying distortion in close proximity to the edges of objects.

6.251 spatial perceptual information (SI)

A measure that generally indicates the amount of spatial detail of a picture. It is usually higher for more spatially complex scenes. It is not meant to be a measure of entropy nor to be associated with the information defined in communication theory. The spatial perceptual nformation (SI), is based on the Sobel filter. Each video frame (luminance plane) at time n (F_n) is first filtered with the Sobel filter ($Sobel(F_n)$). The standard deviation over the pixels (std_{space}) in each Sobel-filtered frame is then computed. This operation is repeated for each frame in the video sequence and results in a time series of spatial information of the scene. The maximum value in the time series (max_{time}) is chosen to represent the spatial information content of the scene. This process can be represented in an equation form as:

$$SI = max_{time} \{std_{space} [Sobel(F_n)]\}$$

6.252 spatial performance

A measure of the ability of a video transmission system to accurately reproduce still scenes.

6.253 spatial-quality component

Spatial-quality component refers to how users perceive that the system is providing a spatial representation of sound. More specifically, it refers to those quality features related to the spatial attributes of sound. Examples of such attributes are immersion, envelopment, localization blur, and source width.

6.254 speakerphone set

A telephone set using a loudspeaker as a telephone receiver with or without an embedded microphone as a telephone transmitter; it may be used without the handset.

6.255 speaking rate

Speaking rate may be expressed in words, syllables or phonemes per second; it takes into account speech pauses. The minimum duration to be measured must be one sentence.

6.256 speech activity factor

See activity factor.

6.257 speech database or corpus

A structured set of pre-recorded speech (phonemes, syllables, words or sentences, whether or not meaningful) from one or more talkers which may be used in either ASR system development and testing. In the latter case, it includes two distinct subsets, i.e., training and test data.

6.258 speech echo path delay

It is the period (in ms) between the detection of an incident signal at a zero reference point, on a four-wire point, to the detection of its corresponding reflected signal at the same four-wire point (in the opposite direction). (For multiple echo path reflections, the speech echo path delay should be calculated for each detection of the corresponding reflected signal.)

6.259 speech echo path loss

It is the ratio of the r.m.s. values of the incident to the reflected speech signals with the speech echo path delay removed. The speech echo path loss is highly dependent on the speaker.

6.260 speech level

A general term embracing speech volume, active speech level and any other similar quantity expressed in decibels relative to a stated reference.

6.261 speech pause interval (or quiet interval)

A period of time during which speech levels are absent due to intersyllabic and conversational pauses. (Intersyllabic pauses are the gaps inherent in the articulation process. Such gaps are short; approximately up to 350 ms, and are not noticed as such by the listener. These pauses should be considered as part of the utterance and, therefore, included in a measurement of speech level. Conversational pauses are generally longer. They will be noticed by the listener, either consciously or subconsciously, and should be excluded from the measurement of speech level since they do not contribute to the subjective loudness of the speech. When these pauses are excluded, the measurement is said to be made when the talker is "active".)

6.262 speech quality

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

6.263 speech spurt (or utterance) interval

A period of time during which speech is present due to syllabic emphasis.

6.264 speech transmission index (STI)

The index indicating the speech intelligibility especially in reverberant condition, derived from measuring the MTF.

6.265 speech transmission quality

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

6.266 speech volume or volume

A quantity which is related to speech power and is measured at a stated point in a telephone circuit by means of a specified instrument, suitable for rapid real-time control or adjustment of level by a human observer (e.g., vu meter, ARAEN volume meter, peak programme meter).

6.267 speech volume penalty

The reduction in a subscriber's talking level (usually expressed as a function of a speech sidetone rating, e.g., STMR) due to the presence of sidetone.

6.268 stability loss

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

6.269 still video

Video imagery that conveys no motion or change.

6.270 string of words

A sequence of words or expressions that are processed as a single unit in the ASR process (e.g., a telephone number).

6.271 subscriber circuit

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

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

6.272 subscriber system (in transmission planning)

A subscriber's line associated with that part of the private telephone installation connected to this line during a telephone call (see also clause 6.273).

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

6.273 subscriber's (telephone) line; subscriber loop (in telephony)

A link between a public switching entity and a telephone station or a private telephone installation or another terminal using signals compatible with the telephone network.

6.274 substitution error

An error in an ASR process in which a valid word (i.e., one in the recognition vocabulary) is incorrectly recognized as another word in the recognition vocabulary.

6.275 super-wideband signal

A signal that has no significant signal components outside the range 20 Hz to 14 000 Hz. A super-wideband signal can be generated by applying the filter from Table 3 to a fullband signal.

NOTE –The low-pass filter definition does not emulate a super-wideband transmission system, channel, device or codec.

Table 3 – Minimum bandwidth filter definition to derive a super-wideband signal from a fullband signal

Frequency (Hz)	Super-wideband gain (dB)
20	0
13 500	0
14 000	0 to −3
15 000	−40 (max)
NOTE 1 – "−40 dB (max.)" should be interpreted as a minimum stop-band attenuation of 40 dB in the filter definition (and thus a maximum stop-band gain of 40 dB). NOTE 2 – The value defines the minimum attenuation; it can be exceeded until the technical maximum.	

6.276 Super-wideband telephony

Transmission of speech with a nominal pass-band wider than 100-7 000 Hz, usually understood to be 50-14 000 Hz (see normal-band telephony and wideband telephony).

6.277 supra-aural earphones (see [b-ITU-T P.57])

Earphones which rest upon the pinna and have an external diameter (or maximum dimension) of at least 45 mm.

6.278 supra-concha earphones (see [b-ITU-T P.57])

Earphones which are intended to rest upon the ridges of the concha cavity and have an external diameter (or maximum dimension) greater than 25 mm and less than 45 mm.

6.279 talker echo

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

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

The sum of the sending loudness rating and receiving loudness rating of the talker's national system, twice the LR of the international chain, and the echo loss (a-b) of the listener's national system. Points a and b are shown in [b-ITU-T G.122] (see clause 4.2 of [b-ITU-T G.122] and Figure I.1 of [b-ITU TG.131]).

6.281 talking quality

Talking quality describes the quality of a telephone call as it is perceived by the talking party only. Talking quality will be mainly affected by the annoyance of the echo signal and effects like background noise switching and double talk.

6.282 talking resistance

A fixed resistance used for test purposes, which has a resistance equal to that of a carbon microphone at a particular current.

6.283 telephone booth

A small cabin containing a telephone station and providing a certain measure of acoustic insulation and privacy for the user.

6.284 telephone circuit

In transmission planning, and in the G-series Recommendations, a telephone circuit denotes a telecommunication circuit with associated equipment, directly connecting two switching devices or exchanges, in line with Note 2 to the general definition of a circuit; see definition in clause 6.39. For simplicity, the term "circuit" is often used instead of "telephone circuit" in the G-series Recommendations.

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

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

6.285 telephone circuit loss

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

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

6.286 telephone set; telephone instrument

An assembly of apparatus for telephony including at least a telephone transmitter, a telephone receiver and the wiring and components immediately associated with these transducers.

NOTE – A telephone set usually includes other components such as a switchhook, a built-in telephone alerter, and a dial.

6.287 telephone stall

A telephone booth without a door.

6.288 telephone station

A telephone set with associated wiring and auxiliary equipment connected to a telephone network for the purpose of telephony.

NOTE – The auxiliary equipment may include, for example, an external call-indicating device, a protector, and a local battery.

6.289 temporal application

An application needing high temporal resolution (or reduced jerkiness), possibly at the expense of reduced spatial resolution. For example, temporal applications include the ability to accurately discern moving image features such as facial expressions and lip movements.

6.290 temporal edge noise

A form of edge business characterized by time-varying sharpness (shimmering) to the edges of objects.

6.291 temporal perceptual information (TI)

A measure that generally indicates the amount of temporal changes of a video sequence. It is usually higher for high motion sequences. It is not meant to be a measure of entropy nor associated with the information defined in communication theory. The measure of temporal information (TI) is computed as the maximum over time (max_{time}) of the standard deviation over space (std_{space}) of $M_n(i,j)$ over all i and j.

$$TI = max_{time} \{std_{space} [M_n(i,j)]\}$$

where $M_n(i,j)$ is the difference between pixels at the same position in the frame, but belonging to two subsequent frames; that is:

$$M_n(i,j) = F_n(i,j) - F_{n-1} (i,j)$$

where $F_n(i,j)$ is the pixel at the ith row and jth column of nth frame in time.

6.292 temporal performance

A measure of the ability of a video transmission system to accurately reproduce motion or changing scenes.

6.293 terminal coupling loss (TCL); weighted terminal coupling loss (TCLw)
(see [b-ITU-T P.300] and [b-ITU-T P.310])

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

– acoustical coupling at the user interface;

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

– seismic coupling through the mechanical parts of the terminal.

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

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

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

6.294 test balance return loss (TBRL)

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

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

6.295 transmission time; total transmission time (TTT) (see [b-ITU-T G.114])

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

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

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

6.296 test data

Utterances used to test an ASR system, which have not been previously used in developing or modifying that system. The same set of test data may be used repeatedly to compare various systems (or subsequently as training data) but not for continuing tests of an algorithm or system development.

6.297 text-to-speech synthesis (TTS)

A TTS process generates a speech signal from text codes. It is usually composed of two parts:

– A language-dependent text processing part (the high level processing part), which generates from the character string (by reading rules, vocabulary and semantic analysis) a set of phonetic, prosodic, etc., parameters which are used by:

– An acoustical signal generating part, the synthesiser itself, which generates the audible speech.

6.298 tiling

See the definition of "block distortion".

6.299 timbre (sound colour)

(Definition generally used in psychoacoustics)

Timbre is that attribute of auditory sensation in terms of which a listener can judge to what extent two sounds, similarly presented and having the same loudness and pitch and duration, are dissimilar. Timbre depends primarily on the spectrum of the stimulus but also depends on the waveform, the sound pressure, the frequency location of the spectrum and the temporal characteristics of the stimulus.

6.300 tonality

(Definition generally used in psychoacoustics)

Tonality is the logarithm of the ratio between the arithmetical and geometrical means of the spectrum and gives information on the presence of high peaks in the spectrum.

6.301 training data

Utterances used to construct the parametric representations of speech elements which the ASR system will have to recognize. These data should not be used to test the system.

NOTE – A part of the training data is often used as development data to further improve these parametric representations.

6.302 transmission rating factor (R)

The principal output of the E-model. A scalar value which combines the effects of different transmission parameters and varies with the mouth-to-ear conversational quality.

6.303 transmission rating model

An algorithm that calculates the effects of variations in several transmission parameters on conversational quality. The model output is one or several quality-related indices that are meant to help transmission planners to ensure desired transmission performance, but are not actual customer opinion predictions.

6.304 transmission reference point (TRP)

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

6.305 transmission service channel

A transmission service channel is the one-way transmission path between two designated points (for example, analogue input, analogue output).

6.306 transparency (fidelity)

A concept describing the performance of a codec or a system in relation to an ideal transmission system without any degradation. Two types of transparency can be defined.

The first type describes how well the processed signal conforms to the input signal, or ideal signal, using a mathematical criterion. If there is no difference, the system is fully transparent. The second type describes how well the processed signal conforms to the input signal, or ideal signal, for a human observer. If no difference can be perceived under any experimental condition, the system is perceptually transparent. The term "transparent" without explicit reference to a criterion will be used for systems that are perceptually transparent.

6.307 type test

A test of one or more devices made to a certain design to show that the design meets certain specifications.

6.308 validity of a subjective test

An agreement between the mean value of ratings obtained in a test and the true value which the test purports to measure.

6.309 video

1) The visually displayed images of video teleconferencing/video telephony.
2) A signal that contains timing/synchronization information as well as luminance (intensity) and chrominance (colour) information that when displayed on an appropriate device gives a visual representation of the original image sequence.
3) Of or pertaining to the visually displayed images of video teleconferencing/video telephony.

6.310 video frame

One complete scanned image or picture from a set comprising video imagery. A video frame is usually composed of two interfaced fields.

6.311 video imagery

A sequence of video frames.

6.312 video teleconferencing/Video telephony service (VTC/VT)

The transmission of video signals capable of portraying motion and the accompanying audio signal(s) between two or more locations using bidirectional transmission facilities. Both analogue and digital transmission may be used. A typical example of this service is interactive video teleconferencing between groups or personnel located at two or more locations.

6.313 virtual international connecting point (VICP)

The virtual international connecting points define the boundary between the national and international part of a connection. The international connecting points are also used as reference points for transmission quantities recommended for the national and international part of a connection.

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

6.314 virtual source function

The change in virtual source position as a function of some other parameter, e.g., frequency, proximity of obstacles.

6.315 virtual source position

That position within a human or artificial mouth at which emitted sounds appear to have their source.

6.316 voice server

Voice servers are automatic devices having similar functions as human operators. The voice servers are connected to a speech application platform or to the telephone network and communicate with users by speech. Voice servers are usually able to handle a large number of ports. Voice servers store and/or retrieve voice messages and voice prompts. Other speech-processing technologies like recognition, understanding and synthesis of speech and general signal-processing technologies like noise processing, echo control, DTMF processing, could also be implemented in voice servers.

6.317 waveform based

Refers to the kind of signal used by the measurement modules/algorithms. In a waveform based approach the measurement algorithms employed would analyse the waveforms, as done for instance in Recommendation [b-ITU-T P.862]. Contrary to waveform based approaches, packet-information based measurement modules only work on the meta/header information of a packet stream, e.g., [b-ITU-T P.564]. Both approaches have their benefits and drawbacks and may therefore be employed in different applications according to how suitable they are.

6.318 weighted listener echo path loss (WEPL)

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

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

where:

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

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

6.319 weighted terminal coupling loss

See terminal coupling loss (clause 6.293).

6.320 wideband signal

A signal that has no significant signal components outside the range 20 Hz to 8 000 Hz. A wideband signal can be generated by applying the filter from Table 4 to a fullband signal, the upper cut-off frequency of 7 600 Hz is due to a realistic interpolation low-pass when re-sampling to 16 000 Hz sampling frequency.

The wideband signal can also be derived by applying the filter in Table 2 to a super-wideband signal as defined in this Recommendation.

NOTE –The low-pass filter definition does not emulate a wideband transmission system, channel, device or codec.

Table 4 – Minimum bandwidth filter definition to derive a wideband signal from a fullband signal

Frequency (Hz)	Wideband gain (dB)
20	0
7 400	0
7 600	0 to −3
8 000	−40 (max)
NOTE 1 – "−40 dB (max.)" should be interpreted as a minimum stop-band attenuation of 40 dB in the filter definition (and thus a maximum stop-band gain of 40 dB). NOTE 2 – The value defines the minimum attenuation; it can be exceeded until the technical maximum.	

6.321 wideband telephony

The transmission of speech with a nominal pass-band wider than 300-3400 Hz, usually understood to be 100-7 000 Hz (see narrowband telephony).

6.322 Y-ratio

The ratio between the sending and receiving efficiencies of a passive telephone set circuit.

6.323 zero sidetone line impedance (Z_{S0})

That circuit impedance which, when connected across the terminals of a telephone set, causes the sidetone to be reduced to zero.

Bibliography

[b-ITU-T E.800] Recommendation ITU-T E.800 (1988), *Definitions of terms used in international telephone operation*.

[b-ITU-T G.100.1] Recommendation ITU-T G.100.1 (2015), *The use of the decibel and of relative levels in speechband telecommunications*.

[b-ITU-T G.102] Recommendation ITU-T G.102 (1988), *Transmission performance objectives and Recommendations*.

[b-ITU-T G.107] Recommendation ITU-T G.107 (2015), *The E-model: a computational model for use in transmission planning*.

[b-ITU-T G.109] Recommendation ITU-T G.109 (1999), *Definition of categories of speech transmission quality*.

[b-ITU-T G.111] Recommendation ITU-T G.111 (1993), *Loudness ratings (LRs) in an international connection*.

[b-ITU-T G.113] Recommendation ITU-T G.113 (2007), *Transmission impairments due to speech processing*.

[b-ITU-T G.114] Recommendation ITU-T G.114 (2003), *One-way transmission time*.

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

[b-ITU-T G.122] Recommendation ITU-T G.122 (1993), *Influence of national systems on stability and talker echo in international connections*.

[b-ITU-T G.131] Recommendation ITU-T G.131 (2003), *Talker echo and its control*.

[b-ITU-T G.173] Recommendation ITU-T G.173 (1993), *Transmission planning aspects of the speech service in digital public land mobile networks*.

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

[b-ITU-T G.761] Recommendation ITU-T G.761 (1988), *General characteristics of a 60-channel transcoder equipment*.

[b-ITU-T G.793] Recommendation ITU-T G.793 (1988), *Characteristics of 60-channel transmultiplexing equipments*.

[b-ITU-T G.794] Recommendation ITU-T G.794 (1988), *Characteristics of 24-channel transmultiplexing equipments*.

[b-ITU-T G.795] Recommendation ITU-T G.795 (1988), *Characteristics of codecs for FDM assemblies*.

[b-ITU-T G.941] Recommendation ITU-T G.941 (1988), *Digital line systems provided by FDM transmission bearers*.

[b-ITU-T H.261] Recommendation ITU-T H.261 (1993), *Video codec for audiovisual services at p x 64 kbit/s*.

[b-ITU-T M.560] Recommendation ITU-T M.560 (1988), *International telephone circuits – Principles, definitions and relative transmission levels*.

[b-ITU-T M.565] Recommendation ITU-T M.565 (1988), *Access points for international telephone circuits*.

[b-ITU-T O.22] Recommendation ITU-T O.22/Q.49 (1976), *Specification for the CCITT automatic transmission measuring and signalling testing equipment ATME No.2*.

[b-ITU-T P.11] Recommendation ITU-T P.11 (1993), *Effect of transmission impairments*.

[b-ITU-T P.51] Recommendation ITU-T P.51 (1996), *Artificial mouth*.

[b-ITU-T P.56] Recommendation ITU-T P.56 (2011), *Objective measurement of active speech level*.

[b-ITU-T P.57] Recommendation ITU-T P.57 (2011), *Artificial ears*.

[b-ITU-T P.58] Recommendation ITU-T P.58 (2013), *Head and torso simulator for telephonometry*.

[b-ITU-T P.59] Recommendation ITU-T P.59 (1993), *Artificial conversational speech*.

[b-ITU-T P.64] Recommendation ITU-T P.64 (2007), *Determination of sensitivity/frequency characteristics of local telephone systems*.

[b-ITU-T P.76] Recommendation ITU-T P.76 (1988), *Determination of loudness ratings; fundamental principles*.

[b-ITU-T P.78] Recommendation ITU-T P.78 (1996), *Subjective testing method for determination of loudness ratings in accordance with Recommendation P.76*.

[b-ITU-T P.79] Recommendation ITU-T P.79 (2007), *Calculation of loudness ratings for telephone sets*.

[b-ITU-T P.85] Recommendation ITU-T P.85 (1994), *A method for subjective performance assessment of the quality of speech voice output devices*.

[b-ITU-T P.300] Recommendation ITU-T P.300 (1988), *Transmission performance of group audio terminals (GATs)*.

[b-ITU-T P.310] Recommendation ITU-T P.310 (2009), *Transmission characteristics for narrow-band digital handset and headset telephones*.

[b-ITU-T P.311] Recommendation ITU-T P.311 (2011), *Transmission characteristics for wideband digital handset and headset telephones*.

[b-ITU-T P.340] Recommendation ITU-T P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals*.

[b-ITU-T P.341] Recommendation ITU-T P.341 (2011), *Transmission characteristics for wideband digital loudspeaking and hands-free telephony terminals*.

[b-ITU-T P.342] Recommendation ITU-T P.342 (2009), *Transmission characteristics for narrow-band digital loudspeaking and hands-free telephony terminals*.

[b-ITU-T P.501] Recommendation ITU-T P.501 (2017), *Test signals for use in telephonometry*.

[b-ITU-T P.561] Recommendation ITU-T P.561 (2002), *In-service non-intrusive measurement device – Voice service measurements*.

[b-ITU-T P.562] Recommendation ITU-T P.562 (2004), *Analysis and interpretation of INMD voice-service measurements*.

[b-ITU-T P.564] Recommendation ITU-T P.564 (2007), *Conformance testing for voice over IP transmission quality assessment models*.

[b-ITU-T P.800]	Recommendation ITU-T P.800 (1996), *Methods for subjective determination of transmission quality*.
[b-ITU-T P.810]	Recommendation ITU-T P.810 (1996), *Modulated noise reference unit (MNRU).*
[b-ITU-T P.830]	Recommendation ITU-T P.830 (1996), *Subjective performance assessment of telephone-band and wideband digital codecs*.
[b-ITU-T P.862]	Recommendation ITU-T P.862 (2001), *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs*.
[b-ITU-T Q.551]	Recommendation ITU-T Q.551 (1988), *Exchange interfaces towards other exchanges*.
[b-ITU-T V.37]	Recommendation ITU-T V.37 (1988), *Synchronous data transmission at a data signalling rate higher than 72 kbit/s using 60-108 kHz group band circuits*.
[b-ITU-T X.745]	Recommendation ITU-T X.745 (1993), *Information technology – Open Systems Interconnection – Systems Management: Test management function*.
[b-ITU-R BT.601-7]	Recommendation ITU-R BT.601-7 (2011), *Studio encoding parameters of digital television for standard 4:3 and wide screen 16:9 aspect ratios*.
[b-ISO 532:1975]	ISO 532:1975, *Acoustics – Method for calculating loudness level*.
[b-Oxford]	Oxford English Dictionary (accessed 7 August 2019) https://en.oxforddictionaries.com/definition/earpiece
[b-Qualinet2013]	Qualinet *White Paper on Definitions of Quality of Experience* (2013), *Output from the fifth Qualinet meeting*, Novi Sad, March 12. http://www.qualinet.eu/index.php?option=com_content&view=article&id=45&Itemid=52
[b-Zwicker]	ZWICKER E., FASTL H. (1991), *Psychoacoustics – facts and models, ISBN 3-540-52600-5*.

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