

269™

IEEE Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets, and Headsets

IEEE Communications Society

Sponsored by the
Transmission, Access and Optical Systems Committee



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IEEE Communications Society

Approved 8 November 2002

IEEE-SA Standards Board

Abstract: Practical methods for making laboratory measurements of electroacoustic characteristics of analog and digital telephones, handsets, and headsets are provided in this revised standard. The methods may also be applicable to a wide variety of other communications equipment, including cordless, wireless, and mobile communications devices. Measurement results may be used to evaluate these devices on a standardized basis. Application is in the frequency range 100–8500 Hz.

Keywords: analog telephones, digital telephones, electroacoustic measurements on telephones, handsets, headsets, telephony, voice transmission performance

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Introduction

(This introduction is not a part of IEEE Std 269-2002, IEEE Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets, and Headsets.)

This standard has been prepared in response to a widely expressed need by the telecommunications industry for a standard, comprehensive method for testing the transmission performance of telephone sets, handset, and headsets. The present standard is a revision of IEEE Std 269TM-1992. This revision adds coverage for a wide range of ear simulators and test signals, and incorporates and updates the contents of IEEE Std 1206TM-1994.

This revision, begun in 1999, was prepared by the Subcommittee on Telephone Instrument Testing of the Transmission, Access and Optical Systems Committee of the IEEE Communications Society.

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IEEE Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets, and Headsets

1. Overview

Objective or subjective methods can be used to measure telephone transmission performance. This standard discusses objective procedures utilizing mouth simulators, ear simulators, laboratory microphones, and test instruments to characterize transmission performance. Subjective procedures are particularly applicable for rating overall communication connections involving the real voice and real ear of human subjects. Telephones, handsets, and headsets can be evaluated by purely objective methods provided the results generally agree with the desirable performance characteristics of subjective testing.

The relationships that are established between subjective and objective measurements will vary with the physical constants of the telephone design, such as the size and shape of the handset or headset, the sound leakage between the receiver and the ear of the user, and the signal processing in the speech path. Therefore, the correlation between subjective and objective measurements should be established separately for each telephone, headset, or handset design before measurements obtained using the techniques covered herein can be interpreted to reflect performance under conditions of actual use.

Execution of this standard provides a means of determining the operational characteristics of a telephone over the range of conditions encountered during normal operation.

1.1 Scope

This standard provides the techniques for objective measurement of electroacoustic characteristics of analog and digital telephones, handsets, and headsets. Application is in the frequency range 100–8500 Hz.

Although not specifically within the scope of this standard, the methods described are generally applicable to a wide variety of other communications equipment, including cordless, wireless, and mobile communications devices.

Telephones with handsfree or loudspeaking features are covered by IEEE Std 1329TM-1999.

Due to the various characteristics of these devices and the environments in which they operate, not all of the test procedures in this standard are applicable to all types of telephones, handsets, or headsets. Application of the test procedures to atypical telephones should be determined on an individual basis.

1.2 Purpose

The purpose of this standard is to provide practical methods for making laboratory measurements of the transmission characteristics of analog and digital telephones, handsets, and headsets so that their performance may be evaluated on a standardized basis.

1.3 Contents of standard

This is a brief summary of the clauses contained in the standard. The primary measurement procedures appear in Clause 7 through Clause 9 of the document.

Clause 2, Clause 3, and Clause 4 provide references, definitions, abbreviations, acronyms, and symbols that will be useful in executing the tests of this standard. These clauses provide a background in the terminology used for telephone, handset, and headset testing.

Clause 5 specifies the test equipment, test environment, and acoustic impairments. The test equipment portion includes ear and mouth simulators, test fixtures, and measurement microphones, as well as procedures for positioning the telephone handset or headset for testing. The test environment includes both the acoustical and physical characteristics of the test space. Impairments include the acoustic conditions.

Clause 6 describes the calibration procedures needed to ensure that the equipment is in a known state. Calibration of the acoustic transducers and electrical interfaces is explained.

Clauses 7, 8, and 9 contain the transmission test procedures, such as send and receive, for analog telephones, digital telephones, and analog four-wire handsets and headsets, respectively.

The annexes contain additional information or details of procedures referred to from within the relevant clause. Normative annexes contain information that is considered to be an official part of the standard. Informative annexes contain information that may be useful, or of general interest, but is not part of the standard.

2. References

This standard shall be used in conjunction with the following publications. When the following publications are superseded by an approved revision, the revision shall apply, but the impact on results should be determined.

ANSI S1.1-1994 (Reaff 1999), American National Standard Acoustical Terminology.¹

ANSI S1.4-1983 (Reaff 2001), American National Standard Specification for Sound Level Meters.

¹ANSI publications are available from the Sales Department, American National Standards Institute, 25 West 43rd Street, 4th Floor, New York, NY 10036, USA (<http://www.asi.org/>).

ANSI S1.6-1984 (Reaff 2001), American National Standard Preferred Frequencies, Frequency Levels, and Band Numbers for Acoustical Measurements.

ANSI S1.11-1986 (Reaff 1998), American National Standard Specifications for Octave Band and Fractional-Octave-Band Analog and Digital Filters.

ANSI S1.12-1967 (Reaff 1997), American National Standard Specifications for Laboratory Standard Microphones.

ANSI/TIA/EIA-810-A-2000, Transmission Requirements for Narrowband Voice over IP and Voice over PCM Digital Wireline Telephones.

ASTM D2240-2002, Standard Test Method For Rubber Property—Durometer Hardness.

IEC 61000-4-5-2001, Electromagnetic Compatibility (EMC)—Part 4: Testing and Measurement Techniques—Section 5: Surge Immunity Test.

IEEE Std 661TM-1979 (Reaff 1998), IEEE Standard Method for Determining Objective Loudness Ratings of Telephone Connections.^{2,3}

IEEE Std 743TM-1995, IEEE Standard Equipment Requirements and Measurement Techniques for Analog Transmission Parameters for Telecommunications.

IEEE Std 1329-1999, IEEE Standard Method for Measuring Transmission Performance of Handsfree Telephone Sets.⁴

ISO 3:1973, Preferred Numbers—Series of Preferred Numbers.⁵

ITU-T Recommendation G.122-1993, Transmission Systems and Media—General Characteristics of National Systems Forming Part of International Connections—Influence of National Systems on Stability and Talker Echo in International Connections.⁶

ITU-T Recommendation G.711-1988, Pulse Code Modulation (PCM) of Voice Frequencies.

ITU-T Recommendation G.714-1988, Separate Performance Characteristics for the Encoding and Decoding Sides of PCM Channels Applicable to 4-Wire Voice-Frequency Interfaces.

ITU-T Recommendation G.722-1988, General Aspects of Digital Transmission Systems—Terminal Equipments—7 kHz Audio-coding Within 64 kbit/s.

ITU-T Recommendation G.723-1996, Speech Coders: Dual Rate Speech Coder for Multimedia Communications Transmitting at 5.3 and 6.3 kbit/s.

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⁶ITU-T publications are available from the International Telecommunications Union, Place des Nations, CH-1211, Geneva 20, Switzerland/Suisse (<http://www.itu.int/>).

ITU-T Recommendation G.726-1990, 40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM).

ITU-T Recommendation G.729-1996, Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP).

ITU-T Recommendation O.41-1994, Psophometer for Use on Telephone-Type Circuits.

ITU-T Recommendation O.133-1993, Equipment for Measuring the Performance of PCM Encoders and Decoders.

ITU-T Recommendation P.50-1998, Appendix 1, Test Signals.

ITU-T Recommendation P.50-1999, Artificial Voices.

ITU-T Recommendation P.51-1996, Artificial Mouths.

ITU-T Recommendation P.56-1993, Objective Measurement of Active Speech Level.

ITU-T Recommendation P.57-2002, Artificial Ears.

ITU-T Recommendation P.58-1996, Head and Torso Simulator for Telephonometry.

ITU-T Recommendation P.59-1993, Artificial Conversational Speech.

ITU-T Recommendation P.64-1999, Determination of Sensitivity/Frequency Characteristics of Local Telephone Systems.

ITU-T Recommendation P.79-1999, Calculation of Loudness Ratings for Telephone Sets.

ITU-T Recommendation P.360-1998, Efficiency of Devices for Preventing the Occurrence of Excessive Acoustic Pressure by Telephone Receivers.

ITU-T Recommendation P.501-2000, Test Signals for Use in Telephonometry.

ITU-T Recommendation P.862-2001, Perceptual Evaluation of Speech Quality (PESQ), an Objective Method for End-to-end Speech Quality Assessment of Narrowband Telephone Networks and Speech Codecs.

3. Definitions

These definitions apply specifically to measurements of the transmission performance of telephone sets, handsets, and headsets and may not be applicable to other disciplines. For definitions not covered in this clause, see IEEE 100TM [B2]⁷ and ANSI S1.1-1994.

3.1 A-weighted: A measurement made using the A frequency weighting specified in ANSI S1.4-1983. A-weighted sound pressure level is expressed as dBA, and the reference level is always 20 μ Pa (micropascals).

⁷The numbers in brackets correspond to those of the bibliography in Annex V.

3.2 acoustic echo path: The acoustic coupling from the handset or headset receiver to the handset or headset microphone.

3.3 acoustic input: The free-field sound pressure level developed by a mouth simulator at the mouth reference point. *See also:* **sound pressure level**.

3.4 acoustic output: The sound pressure level developed in an ear simulator. *See:* **sound pressure level**.

3.5 analog telephone set: A telephone set in which the two-way voice communication interface to the network is in an analog format.

3.6 boom microphone position (BMP): The default point at which to place a boom microphone for testing on a mouth simulator. It is specified as measurement point #21 in ITU-T Recommendation P.58-1996. With respect to the intersection of the mouth reference axis with the lip plane, it is located 6 mm back towards the mouth, 42 mm to the right (or left), and 9 mm downward.

3.7 codec: A combination of an analog-to-digital encoder and a digital-to-analog decoder operating in opposite directions of transmission within the same equipment.

3.8 dBA: Sound pressure level in decibels, relative to 20 μ Pa (micropascals), A-weighted. *See:* **A-weighted**.

3.9 dBm: Power level in decibels, relative to a power of 1 mW (milliwatt).

3.10 dBm0: Power level in dBm, relative to a reference point called the zero transmission level point, or 0 TLP. (See 3.52.) A signal level of X dBm at the 0 TLP is designated X dBm0. In a codec, the 0 TLP is specified in relationship to the full-scale digital level or saturation. However, digital saturation is generally not 0 dBm0. For μ -law codecs 0 dBm0 is 3.17 dB below digital full scale. For A-law codecs 0 dBm0 is 3.14 dB below digital full scale.

3.11 dBm(A): Power level in decibels, relative to a power of 1 mW (milliwatt), A-weighted. *See:* **A-weighted**.

3.12 dBmp: Power level in decibels, relative to a power of 1 mW (milliwatt), measured with psophometric weighting defined in ITU-T Recommendation O.41-1994.

3.13 dBPa: Sound pressure level in decibels, relative to a sound pressure of 1 Pa (pascal).

3.14 dB SPL: Sound pressure level in decibels, relative to a sound pressure of 20 μ Pa (micropascals).

3.15 dBV: Voltage level in decibels, relative to 1 V (volt) rms.

3.16 dBV(A): Voltage level in decibels, relative to 1 V (volt) rms, measured with A-weighting. *See:* **A-weighted**.

3.17 dBV(p): Voltage level in decibels, relative to 1 V (volt) rms, measured with psophometric weighting defined in ITU-T Recommendation O.41-1994.

3.18 digital telephone set: A telephone set in which the two-way voice communication interface to the network is in a digital format.

3.19 ear-cap reference point: The intersection of the external ear-cap reference plane with a normal axis through the effective acoustic center of the sound outlet ports. Generally, the acoustic center of the sound outlet ports is at the center of their distribution.

3.20 ear-reference point: A virtual point for geometric and acoustic reference located outside the entrance to the ear canal. The exact location is specified for each type of ear simulator.

3.21 feed circuit: An electrical circuit for supplying dc power to a telephone set and an ac path between the telephone set and a terminating circuit.

3.22 four-wire transmission: A transmission method, circuit, or system which provides separate paths (one pair each) for signals in the send and receive directions.

3.23 frequency response: Electrical, acoustic, or electroacoustic sensitivity (output/input), or gain, as a function of frequency.

3.24 head and torso simulator (HATS) for telephony: A manikin extending downward from the top of the head to the waist, designed to simulate the sound pick-up characteristics and acoustic diffraction produced by a median human adult and to reproduce the acoustic field generated by the human mouth. See ITU-T Recommendation P.58-1996.⁸

3.25 howling: Audible squealing sound in a telephone, handset, or headset. Acoustic feedback, or oscillation, typically caused by too much acoustic coupling between the receiver and the microphone.

3.26 ideal codec: A codec that has theoretically optimum characteristics.

3.27 listener sidetone: The signal present at the receiver due to sound in the environment where the telephone is used.

3.28 loudness rating guard-ring position (LRGP): The test position a handset assumes when it is placed on an artificial test head as described in Annex A to ITU-T Recommendation P.64-1999, Annex C.

3.29 microphone: An electroacoustic transducer that converts sound to an electrical signal.

3.30 mouth reference point (MRP): A point on the axis of the mouth simulator, 25 mm in front of the center of the external plane.

3.31 overall: The direction of speech transmission from the mouth of one person to the ear of another person. Also called end-to-end.

3.32 overall loudness rating (OLR): A single-number value which corresponds to the perceived loudness loss of an overall connection, as specified in ITU-T Recommendation P.79-1999.

3.33 pinna: The flexible part of the outer ear at the side of the head.

3.34 receive: The direction of speech transmission from the network to the ear of the telephone user.

3.35 receiver: An electroacoustic transducer that converts an electrical signal to sound and delivers it directly to the ear, sealed or unsealed.

3.36 reference codec: A codec that approaches the performance of an ideal codec and has superior, well-defined characteristics used for testing digital telephone sets.

3.37 receive loudness rating (RLR): A single-number value that corresponds to the perceived loudness loss of a receive connection, as specified in ITU-T Recommendation P.79-1999.

3.38 reference receive volume control setting: The receive volume control setting of a telephone that results in the receive loudness rating (RLR) closest to the specified target value.

3.39 reference send gain control setting: The send gain control setting of a telephone that results in the send loudness rating (SLR) closest to the specified target value.

3.40 send: The direction of speech transmission from the mouth of the telephone user to the network.

3.41 send loudness rating (SLR): A single-number value that corresponds to the perceived loudness loss of a send connection, as specified in ITU-T Recommendation P.79-1999.

3.42 sidetone: The direction of speech transmission from the microphone to the receiver of the handset or headset. There are two types of sidetone to be considered: listener sidetone and talker sidetone.

3.43 single frequency interference (SFI): An audible impairment that can be perceived as a tone relative to the overall noise level.

3.44 sidetone masking rating (STMR): A single-number value that corresponds to the perceived loudness loss of the talker sidetone connection, as specified in ITU-T Recommendation P.79-1999.

⁸Information on references can be found in Clause 2.

3.45 sound pressure level: The sound pressure level, in decibels, of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure of the sound to the reference pressure. For this standard, the reference pressure is normally 1 pascal (Pa), and sound pressure levels are expressed in dB re 1 Pa (dBPa). When a reference pressure of 20 μ Pa is used, the sound pressure level will be expressed as dBSPL. Unless otherwise indicated, rms values of pressure are used. Most telephony acoustic measurements are referenced to 1 Pa = 1 N/m² (newton per square meter). However, measurements such as receive noise and room noise are generally referenced to 20 μ Pa (micropascals).

NOTE—0 dBPa $\triangleq$ 94 dBSPL, 0 dBSPL $\triangleq$ 20 μ Pa, 1 Pa $\triangleq$ 1 N/m². A-weighted sound pressure level in dB (dBSPL, A-weighted) is often abbreviated as dBA (see ANSI S1.4-1983).

3.46 speaker (also loudspeaker): An electroacoustic transducer that converts an electrical signal to sound and delivers it to the ear from a distance of several centimeters or greater.

3.47 spectrum: A distribution of amplitude (or phase, or some other quantity) as a function of frequency. It is often expressed in bands. Bands may be of constant percentage width, such as 1/3 or 1/12th octave bands ($\sim$ 23% and $\sim$ 6% of the center frequency, respectively). Bands may also be of fixed width, regardless of center frequency (e.g., 50 Hz). Instead of bands, a spectrum may also be expressed as spectrum density, which is equivalent to 1 Hz bands.

3.48 talker sidetone: The direction of speech transmission from mouth to ear of the telephone user.

3.49 telephone set: A device that, when connected to a telephone network, allows two-way voice communication.

3.50 test head: A fixture containing a mouth simulator and an ear simulator located in a specified relationship with each other. *See: loudness rating guard-ring position.*

3.51 two-wire transmission: A transmission method, circuit, or system that provides common paths (one pair) for signals in the send and receive directions.

3.52 zero transmission level point (0 TLP): An arbitrarily established point relative to which transmission levels at all other points are specified.

4. Abbreviations, acronyms, and symbols

4.1 Abbreviations and acronyms

AGC	automatic gain control
DFTP	diffuse field test point
DRP	drum reference point
DSP	digital signal processor
DTX	discontinuous speech transmission
ECRP	ear-cap reference point
EPD	echo path delay
ERP	ear reference point
FFT	fast Fourier transform
HATS	head and torso simulator
LRGP	loudness rating guard-ring position
MRP	mouth reference point
OLR	overall loudness rating
RETP	receive electrical test point
RLR	receive loudness rating
SETP	send electrical test point

SDN	signal-to-distortion-and-noise ratio
SFI	single frequency interference
SLR	send loudness rating
SNR	signal-to-noise ratio
STMR	sidetone masking rating
TDS	time delay spectrometry
THD	total harmonic distortion
VAD	voice activity detector

4.2 Symbols

The letter “*G*” is used for spectra. This corresponds to common usage, especially in two-channel FFT analysis literature. The analysis bandwidth shall be specified:

$G_{\text{DFTP}}(f)$ = rms spectrum at diffuse field test point, in dBPa
 $G_{\text{ERP}}(f)$ = rms spectrum at ear reference point, in dBPa
 $G_{\text{MRP}}(f)$ = rms spectrum at mouth reference point, in dBPa
 $G_{(\text{MRP})(\text{ERP})}(f)$ = cross-spectrum between MRP and ERP, in dB (Pa/Pa)
 $G_{(\text{MRP})(\text{SETP})}(f)$ = cross-spectrum between MRP and SETP, in dB (V/Pa)
 $G_{\text{RETP}}(f)$ = rms spectrum at receive electrical test point, in dBV
 $G_{(\text{RETP})(\text{ERP})}(f)$ = cross-spectrum between RETP and ERP, in dB (Pa/V)
 $G_{(\text{RETP})(\text{SETP})}(f)$ = cross-spectrum between RETP and SETP, in dB (V/V)
 $G_{\text{SETP}}(f)$ = rms spectrum at send electrical test point, in dBV
 $G_{\text{SETP}(S+N)}(f)$ = rms spectrum at SETP with both the mouth simulator and noise sources active, in dBV
 $G_{\text{SETP}(N)}(f)$ = rms spectrum at SETP with only the noise source active, in dBV

The letter “*H*” is used for frequency response:

$H(f)$ = frequency response, in dB
 $H'(f)$ = response at the new preferred ISO R10 frequency
 $H_{\text{EP}}(f)$ = echo path frequency response, in dB (V/V)
 $H_{\text{LS}}(f)$ = listener sidetone frequency response, in dB (Pa/Pa)
 $H_{\text{O}}(f)$ = overall frequency response, in dB (Pa/Pa)
 $H_{\text{R}}(f)$ = receive frequency response, in dB (Pa/V)
 $H_{\text{S}}(f)$ = send frequency response, in dB (V/Pa)
 $H_{\text{SD}}(f)$ = diffuse field send frequency response, in dB (V/Pa)
 $H_{\text{TS}}(f)$ = talker sidetone frequency response, in dB (Pa/Pa)

The letter “*L*” is used for rms levels measured over a wide band, with the bandwidth to be specified. This corresponds to common usage in sound level measurements, as specified in ANSI S1.1-1994:

L_{ERP} = level at ear reference point, in dBPa
 L_{MID} = level of stimulus to be determined for headset tests (see 9.3.2)
 L_{MRP} = level at mouth reference point, in dBPa
 L_{RETP} = level at receive electrical test point, in dBV or dBm
 L_{ROOM} = level of background noise in measurement environment, in dB SPL
 L_{SETP} = level at send electrical test point, in dBV or dBm

The letter “*S*” is used for specially calculated sensitivities:

S_{DE} = translation from HATS drum reference point to ear reference point

The letter “ T ” is used for time measurements:

T = length of time window in simulated free-field techniques, in sec

T_C = convergence time of acoustic echo cancellers (AEC) algorithm, in sec

“TCL” is used for terminal coupling loss:

$ATCL_T$ = active temporally weighted terminal coupling loss, in dB

$ALTCL_T$ = active long-term temporally weighted terminal coupling loss, in dB

$LTCL_T$ = long-term temporally weighted terminal coupling loss, in dB

TCL = terminal coupling loss, in dB

TCL_T = temporally weighted terminal coupling loss, in dB

TCL_W = frequency weighted terminal coupling loss, in dB

Other symbols:

$SendSNR$ = send signal-to-noise ratio, in dB

$SendSNR_W$ = weighted send signal-to-noise ratio, in dB

5. Test equipment and setup

Test equipment generally required to test all the devices covered by this standard is covered in this clause. The specific test equipment required to produce test signals and analyze the resulting output is determined by the test signal and analysis method chosen. Test circuits, interfaces, and impairments for analog and digital telephones, as well as four-wire devices such as handsets and headsets, are described in Clause 7, Clause 8, and Clause 9, respectively.

All equipment should be calibrated in accordance with the recommendations of the manufacturer before performing the system calibration procedures in Clause 6.

5.1 Ear simulators

The fundamental purpose of ear simulators is to test a receiver under conditions that most closely approximate actual use by real persons. The recommendations that follow are based upon the manner in which the receivers are intended to be used. Modifications to an ear simulator or test procedure shall not be made. For example, flexible sealing material, such as putty, shall not be used.

5.1.1 Selection

An ear simulator with a flexible pinna shall be used for all measurements, unless the applicable performance specification requires or allows an alternative. See Annex B. The Type 3.3 ear simulator is recommended for all devices. The Type 3.4 ear simulator is recommended for all devices except supra-concha and supra-aural headsets.

The ear simulators shall comply with the specifications given in ITU-T Recommendation P.57-2002.

Type 3.3 and Type 3.4 ear simulators both simulate the acoustical and mechanical characteristics of real ears. They are likely to give results comparable to the typical listening experience of real persons for the widest possible variety of handsets or headsets and applications, including nontraditional designer handsets and headsets. Both types simulate typical leakage and how it changes with position

and/or applied force. There are, however, some differences between the two types, as well as the positioning devices available for use with them. For more information, see Annex A.

(Type 3.3 was formerly called the *soft HATS pinna*. It has a hardness of 55, ± 10 degrees Shore-OO, as measured according to ASTM D2240-2002. It is an anatomically shaped pinna which is structurally identical to the pinna formerly described as Type 3.3 in ITU-T Recommendation P.57-2002.)

The same ear simulator shall be used for all measurements on a particular device. The choice of ear simulator and positioning method shall be clearly stated in all test reports.

5.1.2 Headset measurements made on ear simulators compared to real ears

Type 3.3 and Type 3.4 ear simulators both simulate the acoustical and mechanical characteristics of real ears. They are likely to give results comparable to the typical listening experience of real persons for the widest possible variety of headsets. However, the correlation between measurements on ear simulators and on real ears is better for some headset types than others. For headsets that are in close proximity to, or in the ear canal, the correlation is not as good as for most other types.

For insert headsets, both Type 3.3 and Type 3.4 ear simulators are likely to provide a greater seal than on many human subjects, resulting in an overestimation of output at low frequencies. Nonetheless, both ear simulators are recommended for this application.

5.1.3 Translation from DRP to ERP

Type 3.3 and Type 3.4 ear simulators both measure at the eardrum reference point (DRP). Measurements collected at the DRP shall be translated to the ERP. This is done because receive and sidetone specifications are referenced to the ERP. It also permits comparison of measurements made on the various type ear simulators.

For all measurements, the translation from DRP to ERP may be fulfilled by using a filter as specified in Annex C. A filter shall be used for measurements of peak acoustic pressure, and is recommended for measurements of distortion.

For measurements made with any kind of spectrum analysis, the translation from DRP to ERP may be performed by using one of the tables in Annex C. Measurement examples include frequency response, noise, linearity, and distortion. Tables may also be used for frequency response measurements made with sine waves, if only the fundamental or total response is included.

For measurements of distortion using a sine or narrowband stimulus, a translation table may be constructed based on one of the tables in Annex C. Separate tables are required for each harmonic or difference-frequency distortion product, taking into account the frequency offset between the stimulus frequency and the frequency of the distortion product.

5.2 Mouth simulators

The fundamental purpose of mouth simulators is to test a microphone under conditions that most closely approximate actual use by real persons. The mouth simulator shall comply with the specifications given in ITU-T Recommendation P.58-1996, unless the applicable performance specification requires or allows an alternative. See Annex B. This mouth is generally installed in a HATS.

ITU-T Recommendation P.58-1996 does not define a sound field behind the lip plane of the mouth simulator. However, experience has shown that at least one implementation of the mouth has a sound

field distribution which closely approximates the sound field behind the lip plane of a real human head, up to at least 4 kHz. The investigated region extends from behind the lip plane to the base of the rubber ear and equal to or greater than 5 mm above the surface of the HATS cheek. This makes HATS suitable for testing headsets, cordless and cellular phones, handsfree phones, and traditional corded handsets. The sound field approximation may extend in frequency range as well as to other regions around HATS, but these have not yet been verified.

5.3 Test fixtures

The fundamental purpose of test fixtures is to test a device equipped with a handset or headset under conditions that most closely approximate actual use by real persons.

5.3.1 Selection

The test fixture shall be a HATS, which complies with ITU-T Recommendation P.58-1996. When using the Type 3.3 ear simulator, the HATS shall also comply with ITU-T Recommendation P.64-1999, Annex E. When using the Type 3.4 ear simulator, the HATS shall also comply with ITU-T Recommendation P.64-1999, Annex D.

The LRGP position was specified in previous editions of this standard. Send frequency response measurements made on ordinary telephones from 300 to 3400 Hz are expected to give practically identical results, whether obtained with LRGP or the HATS position. Systematic differences of about 1–2 dB in send frequency response measurements on pressure gradient microphones have to be expected from the upwards tilted speaking direction of about 19 degrees using the LRGP position. See ITU-T Recommendation P.64-1999, Annex F. (Similar effects might be observed in sidetone and overall measurements.)

For information about an alternative test fixture, and the ear simulators and mouth simulator with which it can be used, see Annex B.

5.3.2 Handset positioning

The handset receiver shall be nominally placed in the HATS position as specified by Annex D or Annex E of ITU-T Recommendation P.64-1999. To do this, the ear-cap reference point (ECRP) shall lie on the axis of motion of the positioning device. This axis is defined by a line that passes through the ERP of the left and right ears. The ECRP may be inside of or outside of ERP depending on the applied force and the shape of the receiver.

For the purposes of this standard, the ECRP is the intersection of the external ear-cap reference plane with a normal axis through the effective acoustic center of the sound outlet ports. Generally, the acoustic center of the sound outlet ports is at the center of their distribution.

For many handsets, the ear-cap reference plane is parallel to the reference plane of the positioning device.

For some handsets, the above positioning may not apply, and the position that best represents intended use shall be utilized.

The receiver shall contact the pinna with a force of 6 N (newtons). This is the default force for all measurements.

In general, it is desirable that receive frequency response should not change too much as application force changes. For this reason, the device should also be tested at 2 N and 10 N, which represent

minimum and maximum forces likely to be used by real persons on a long-term basis. These results are for information, but do not have to be included in the test report.

(One could experience a force of approximately 10 N by placing a weight of 1 kg on top of a lightweight receiver placed on his or her pinna, with the ear-cap reference plane horizontal.)

A manufacturer may specify a recommended test position (RTP) on either the Type 3.3 or Type 3.4 ear simulator. The RTP may specify position with respect to ERP, a specific force, or other aspects of the test position intended to simulate actual use. The force applied shall not exceed the range of 2–10 N. If the phone is tested at the RTP, the definition of the RTP, including evidence of its authorization by the device manufacturer, shall be included in the test report.

If the RTP is used, the device should also be tested at 2 N, 6 N, and 10 N on the same ear simulator. These results are for information, but do not have to be included in the test report.

For maximum acoustic output measurements, the device shall be tested at either 6 N or the RTP, and also at 13 N. The final result shall be an “upper envelope” curve consisting of the maximum output of each measurement at each frequency. See Figure 1 and Figure 2.

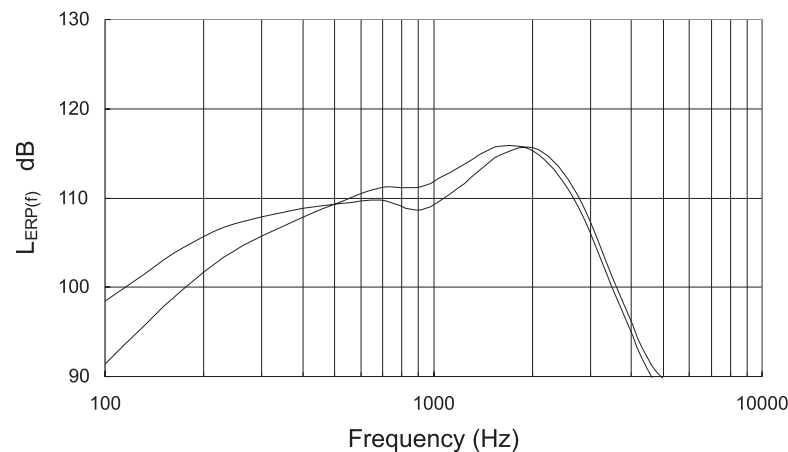


Figure 1—Maximum acoustic output, $L_{ERP}(f)$, two measurements on one handset

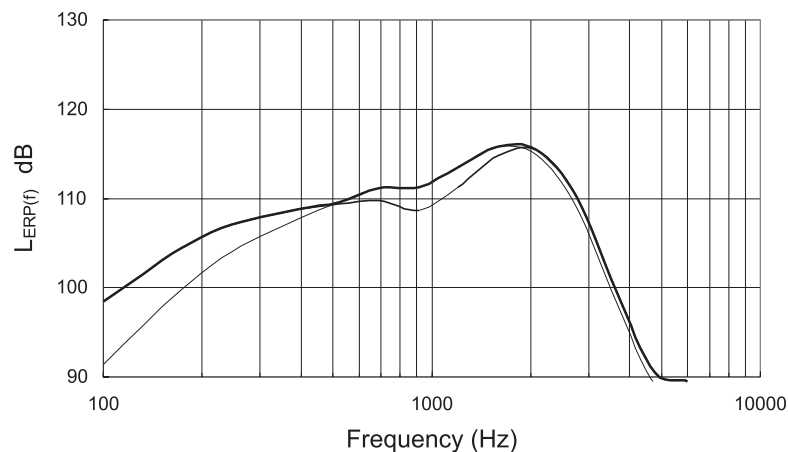


Figure 2—Maximum acoustic output, $L_{ERP}(f)$. Upper envelope (heavy line) and two individual measurements on one handset (light lines)

Except for maximum acoustic output, the same positioning shall be used for all measurements of any particular device. The positioning method shall be clearly stated in the test report.

Handsets with carbon microphones require conditioning procedures before positioning for measurement. See Annex D.

5.3.3 Headset positioning

Headsets should be tested in a position that most closely approximates real use. Natural headband pressure, or other positioning techniques normally used by a real person, shall be used for testing.

If the manufacturer specifies a recommended test position (RTP), the headset shall be tested in that position. The purpose of the RTP shall be to clarify how to position the headset in a way that corresponds to real use. The RTP shall be defined geometrically with respect to the MRP, center of the lip plane, ERP, ear entrance point (EEP), and/or the HATS reference point (HRP). See ITU-T Recommendation P.58-1996. Facial features, such as the corner of the mouth, shall not be used as reference points. If no RTP is specified, the test position can be determined by observing the actual design of the headset and by following any guidelines for positioning provided by the manufacturer.

When positioning a headset on a HATS, it is generally possible to approximate real use in an obvious way. In any case where the headset does not fit on the HATS and its ear simulator quite in the way intended for real persons, adjustments may be made so the receiver and microphone are as close as possible to positions corresponding to real use. The body of the receiver, the headband or any other non-acoustical component may be positioned as necessary.

5.3.3.1 Headset receiver positioning and averaging

When a headset is placed on the Type 3.3 or Type 3.4 ear simulator, the receiver results may vary from trial to trial due to slight variations in positioning. Relatively accurate and repeatable results can be obtained by making several measurements and averaging the results. The procedures in this clause shall be followed for receive, sidetone, and overall measurements.

A minimum of five measurements of frequency response and loudness rating shall be made on each individual unit tested. The headset shall be completely removed from the ear simulator and re-mounted for each trial. The mean and standard deviation of the loudness rating and each point of the frequency response shall be computed for this group of measurement trials.

The accuracy of the final results shall be considered acceptable if the standard deviation of the loudness rating is 1 dB or less, and if the standard deviation of the frequency response is 2 dB or less from 200–1000 Hz, and 1 dB or less from 1000–4000 Hz. If the results of the first five trials do not meet this criteria, report the results, but label them as reduced accuracy.

The reported results shall include the mean loudness rating and standard deviation, the mean frequency response and standard deviation, a description of the test position, and the number of trials. Additional measurements may be made in order to meet the mean and standard deviation criteria.

Similarly, a minimum of five measurements of distortion shall be made. A minimum of five measurements of all other parameters should also be made. The mean results shall be reported.

For maximum acoustic output measurements made using the Type 3.3 or Type 3.4 ear simulator, at least five measurements shall be made. The final result shall be an “upper envelope” curve consisting of the maximum output of each measurement at each frequency. All curves shall be reported, for a total of at least five individual measurements plus the upper envelope curve. See the example in Figure 3 and Figure 4.

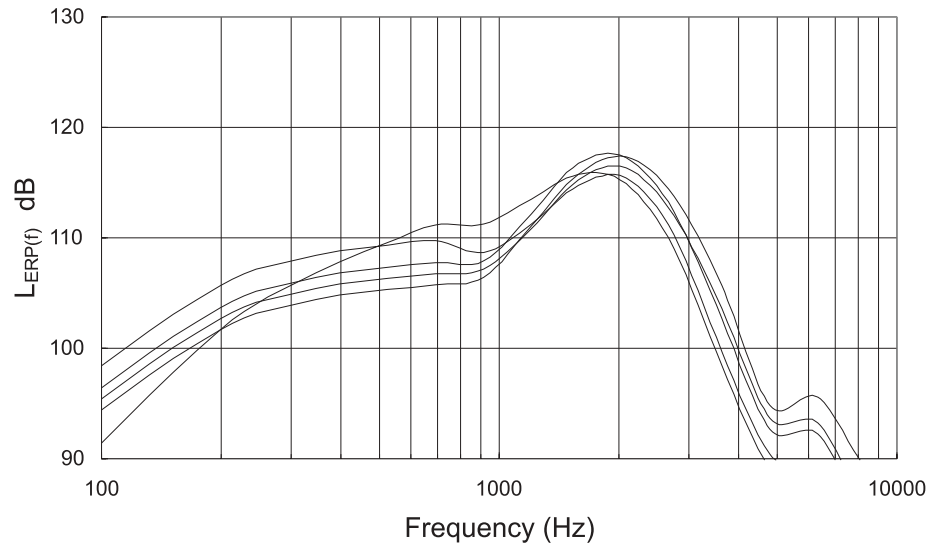


Figure 3—Maximum acoustic output, $L_{ERP}(f)$, five measurements on one receiver

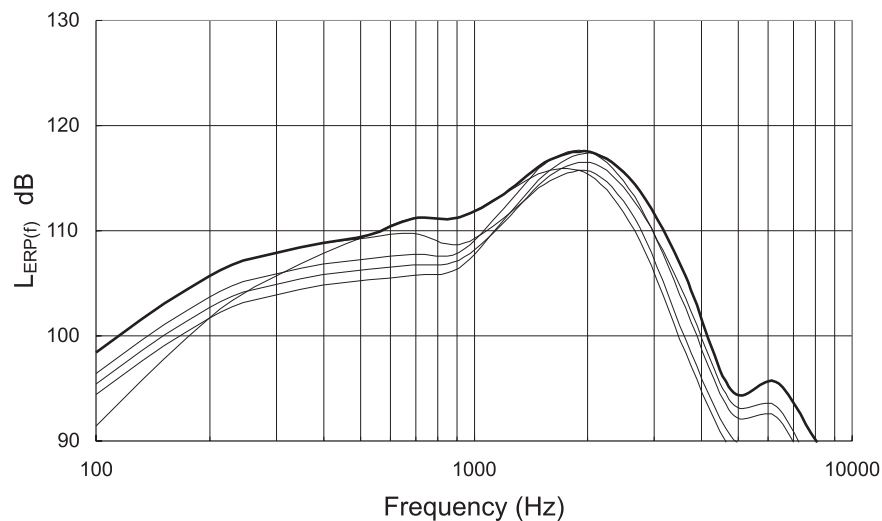


Figure 4—Maximum acoustic output, $L_{ERP}(f)$. Upper envelope (heavy line) and five individual measurements on one receiver (light lines)

For headsets with large hard-cap receivers that are similar to receivers in handsets, 5.3.2 may apply.

5.3.3.2 Headset microphone positioning

Three test positions are defined for the location of the headset microphone sound port, in order of preference

- Recommended test position (RTP).
- Boom microphone position (BMP). If an RTP is not provided, the BMP may be used if it corresponds to the intended usage of the microphone. The BMP is defined in 3.6 and in ITU-T Recommendation P.58-1996.
- Estimated real use position.

These positions may fall behind the lip plane of the mouth. See B.2 for more information.

Pressure gradient microphones (cardioid, noise canceling, etc.) are sensitive to both position and orientation. It is important that the correct orientation be used to obtain results representing actual use performance.

For microphones that are not measured at RTP or BMP, or are not on a fixed boom, state the exact geometric test position following the guidelines in 5.3.3.

5.4 Measurement microphones

The sizes and types of measurement microphones are specified in the clauses where their use is required. All microphones used to implement this standard shall comply with the relevant specifications in ANSI S1.12-1967.

5.5 Test environment

Electroacoustic measurements should be conducted in a test environment that will not affect the results beyond the intended influence of the test fixture and measurement transducers. The test environment should have a low background noise level, and the test fixtures and device under test should be isolated from reflections and mechanical disturbances that could cause significant error.

Be sure to record the test environmental conditions of temperature, humidity, and barometric pressure, in addition to the background noise. Overall A-weighted noise level and octave band sound levels are defined below.

5.5.1 Background noise level

The background noise level in the test environment shall not exceed the limits shown in Table 1. The overall level shall not exceed 29 dBA. However, these limits may be relaxed if it can be shown that the accuracy of the measurement is not impaired.

Background noise measurements shall be made using a 12.5 mm pressure microphone with a microphone system noise level not exceeding 20 dBA. The individual factory-calibrated frequency response of the microphone, if available, shall be taken into account.

Table 1—Test room noise levels

Octave band center frequency (Hz)	Octave band level (dBSPL)
63	49
125	34
250	29
500	29
1000	29
2000	29
4000	29
8000	29

5.5.2 Reflection-free conditions

The test environment should be sufficiently free of reflections. There should be no large objects within 1 m of the MRP. Small objects such as tripods that are used for positioning may be acceptable. Errors due to the influence of reflections shall not exceed ± 1.5 dB below 800 Hz. Errors above 800 Hz shall not exceed ± 1.0 dB.

5.5.3 Diffuse field conditions

A uniform diffuse sound field shall exist in a volume of radius 0.15 m. The diffuse field test point (DFTP) is at the center of this spherical volume. There shall be no obstacles, including the loudspeakers, within 0.5 m of the DFTP.

The classical method of creating a diffuse field is to construct a reverberation chamber. If one is available, it is generally the best method. (Construction and verification of a reverberation room is outside the scope of this standard.)

For the purposes of this standard, a diffuse field may be approximated by using several loudspeakers and uncorrelated noise sources. Experience has shown that four to eight speakers and uncorrelated sources in an ordinary room may be sufficient for measurements in 1/3 octaves. However, more may be required, especially if measurements are to be made in 1/12 octave resolution.

Diffuse field measurement should be made using a 6.25 mm pressure microphone, but may also be made using a 12.5 mm random pressure microphone. The individual factory-calibrated frequency response of the microphone, if available, shall be taken into account.

Diffuse field conditions shall be verified by the following two tests, performed in the same resolution and bandwidth used for measurements. The test is performed with reference to the DFTP, with no mouth simulator or other objects present.

5.5.3.1 Test for diffuse field

Diffuse field conditions at the DFTP shall be verified by measurements with a cardioid or bidirectional microphone with a free field rejection of at least 10 dB front-to-back (cardioid) or front-to-side (bidirectional) in each frequency band. The sound field shall be considered to approximate a diffuse (random-incidence) field if the variation is within the tolerance in Table 2. The microphone is rotated about the DFTP through 360 degrees in each of three perpendicular planes. Measurement shall be made in 15 degree increments or less. Since microphone rejection may not be the same in all bands, the tolerance for each band shall be determined by the microphone rejection in that particular band.

Table 2—Diffuse field variation allowable vs. microphone rejection

Microphone rejection at least	Allowable variation over 360 degrees
25 dB or greater	6 dB
20 dB	5 dB
15 dB	4 dB
10 dB	3 dB
Less than 10 dB	Microphone not suitable

5.5.3.2 Test for spectrum uniformity

$G_{\text{DFTP}}(f)$ (the spectrum at DFTP) shall be measured at the DFTP and six additional points ± 15 cm from DFTP on each of three mutually perpendicular axes passing through the DFTP. The spectrum at these six points shall not vary from $G_{\text{DFTP}}(f)$ by more than ± 3 dB in each band.

Calibration of the diffuse field shall be according to 6.7.1 and 6.7.2, except performed at the DFTP.

5.6 Acoustic impairments

In order to test a telephone, handset, or headset realistically, it may be useful to test it in environments similar to those in which it is expected to operate. Such environments can be considered acoustic impairments, which may cause the telephone to work differently than in a quiet test space. Two such impairments are described in this clause, but others may also be relevant for some applications.

5.6.1 Reference corner

The reference corner is one physical setup used for echo, howling, and stability tests. The reference corner consists of three perpendicular plane, smooth, hard surfaces 0.5 m square, as shown in Figure 5. A handset shall be placed along the diagonal from the apex of the reference corner to the outside corner, with the earcap end of the handset 250 mm from the apex. A headset is placed on the surface as if it was put down briefly by a user, with the receiver 250 mm from the apex.

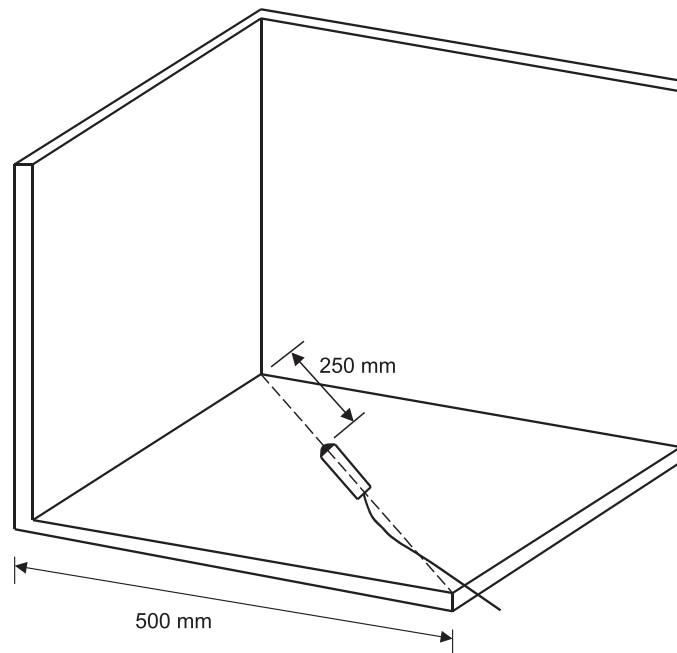


Figure 5—Reference corner for echo, howling, and stability tests

5.6.2 Hoth room noise

Hoth noise is random acoustic noise which has a spectrum designed to simulate typical ambient room noise. See Annex E for details. The noise level shall be specified in dBA.

6. Calibration

6.1 General

All equipment shall be calibrated according to the manufacturer's recommendations and in accordance with good laboratory practice. Be sure that all equipment has been powered up long enough to be stable before calibration, typically 30 min.

6.2 Electrical measurement instruments

Analyzers and level meters shall be calibrated to an accuracy of at least ± 0.5 dB.

6.3 Measurement microphones

The sensitivity of measurement microphones shall be calibrated prior to each use, to an accuracy of at least ± 0.5 dB. An acoustical calibrator with an accuracy of at least ± 0.2 dB shall be used.

The individual factory-calibrated frequency response, if available, shall be taken into account.

6.4 Ear simulator

The sensitivity of the ear simulator should be calibrated each day the system is in use. For best accuracy, it should be calibrated prior to measurements on each new device. An acoustical calibrator with an accuracy of at least ± 0.2 dB shall be used, along with the factory-supplied adapters for the ear simulator to be calibrated. Be sure to apply any correction factor required for any particular combination of calibrator, adapter, and ear simulator.

6.5 Measurement bandwidth and resolution

The calibration procedures shall be performed using the same format as will be used for measurements. Format examples are $1/N$ octave bandwidth analysis, constant bandwidth analysis, and R -series preferred frequencies. Bandwidth shall be the same as or greater than that which will be used for measurements. Resolution shall be the same as or finer than that which will be used for measurements. Amplitude accuracy shall be the same as or better than that which will be used for measurements. The actual format, bandwidth, resolution, and amplitude accuracy shall be stated. See Annex F for additional details. Also, review F.9, F.11, G.6, and G.7 for a summary of the test signal parameters, comparison of test methods and signals, and details about measurement bandwidth and resolution.

6.6 Electrical test signals

Electrical test signals should be calibrated each day the system is in use. For best accuracy, they should be calibrated prior to measurements of each new device. The analyzer or level meters used shall be calibrated first (6.2).

Electrical test signals shall be applied from a $900\ \Omega$ resistive source impedance for most analog telephones, and a $600\ \Omega$ resistive source impedance for digital telephones. These calibrations are performed across matched calibrated resistive loads. As a result, receive and echo test signals are specified under nominally loaded conditions. This is equivalent to one-half the open-circuit voltage.

Following a calibration, the resistive load is removed and the source is connected to RETP without further adjustment.

A similar calibration is required for testing handset and headset four-wire devices. See 9.2 for the source impedance requirements.

6.6.1 Electrical test spectrum

The electrical test spectrum is measured across a calibrated resistive load. For sinusoidal test signals, the spectrum shall be flat within ± 0.5 dB over the actual measurement bandwidth. Equalization may be used to meet this requirement.

For all other test signals, the electrical spectrum shall meet the target spectrum and spectrum tolerance for the type of signal used, as defined in Annex F. If no tolerance is specified in the signal definition, the default tolerance is ± 3 dB from 175 to 4500 Hz (or the 1/3 octave bands from 200 to 4000 Hz), and $+3/-5$ dB elsewhere. Equalization may be used to meet this requirement.

6.6.2 Electrical test level

The standard electrical test level, nominal L_{RETP} , is -16 dBV rms, ± 0.5 dB, for analog telephones. This test level is recommended for measurements at minimum and reference volume control settings, and -30 dBV is recommended at maximum volume. Total harmonic distortion shall be less than 1% for these test conditions.

The standard test level for digital telephones, nominal L_{RETP} , is -18.2 dBV, ± 0.5 dB, for a 600 Ω interface. This corresponds to -16 dBm0. This test level is recommended for measurements at minimum and reference volume control settings. For measurements at maximum volume control settings, L_{RETP} is -32.2 dBV. This corresponds to -30 dBm0. Total harmonic distortion of the test signal shall be less than 1% for these test conditions.

The standard test level for handsets and headsets tested as four-wire devices is determined by the procedure for setting the default receive volume control adjustment in 9.3.2.

For sinusoidal test signals, the level shall be held constant at all test frequencies.

For continuous spectrum test signals, the level shall be measured over the entire spectrum. Out-of-band signals from 40–20 000 Hz shall add no more than 0.5 dB to this level.

6.7 Mouth simulator

The mouth simulator should be calibrated each day the system is in use. For best accuracy, it should be calibrated prior to measurements on each new device. The measurement microphone used to calibrate the mouth simulator shall be calibrated first (6.3).

6.7.1 Acoustic test spectrum

The acoustic test spectrum is measured at the mouth reference point (MRP). For sinusoidal test signals, the spectrum shall be flat within ± 0.5 dB over the actual measurement bandwidth. Equalization may be used to meet this requirement.

For all other test signals, the acoustic spectrum shall meet the target spectrum and spectrum tolerance for the type of signal used, as defined in Annex F. If no tolerance is specified in the signal definition,

the default tolerance is ± 3 dB from 175 to 4500 Hz (or the 1/3 octave bands from 200 to 4000 Hz), and $+3/-5$ dB elsewhere. Equalization may be used to meet this requirement.

6.7.2 Acoustic test level

The standard acoustic test level for send, L_{MRP} , is -4.7 dBPa rms, ± 0.5 dB, at the MRP. Total harmonic distortion of the mouth simulator shall be less than 2% for this test condition.

For sinusoidal test signals, the level at MRP shall be held constant at all test frequencies.

For continuous spectrum test signals, the level shall be measured over the entire spectrum. Out-of-band signals from 40 to 20 000 Hz shall add no more than 0.5 dB to this level.

6.7.3 Mouth simulator calibration procedure

A 6.25 mm pressure or free-field microphone shall be used to calibrate the HATS mouth simulator. The microphone axis shall be oriented 90 degrees to the mouth axis with the center of the protection grid at the MRP (see Figure 6). (The HATS manufacturer generally supplies a jig for this purpose.)

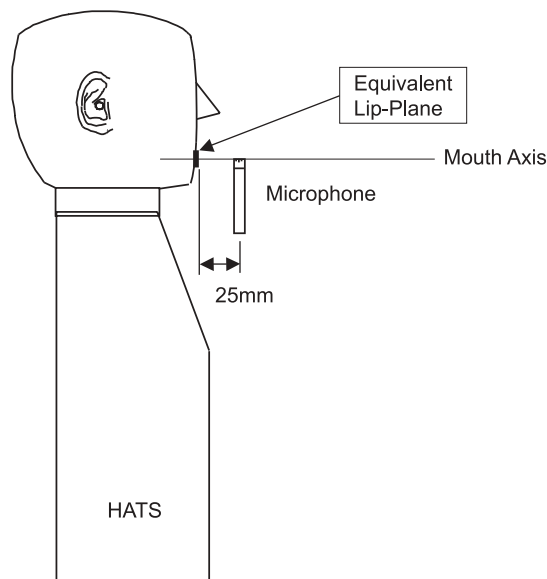


Figure 6—HATS mouth calibration

If a pressure microphone is used, the results may be used directly. The individual factory-calibrated frequency response of the microphone, if available, shall be taken into account.

If a free-field microphone is used, the free-field correction curve for 90 degrees shall be taken into account, and the individual factory-calibrated frequency response of the microphone, if available, shall be taken into account.

To calibrate the mouth, measure $G_{MRP}(f)$, the spectrum at the MRP. Adjust the mouth equalization to meet the target spectrum for the signal being used at a total sound pressure of -4.7 dBPa. This spectrum is used to calculate the send, sidetone and overall frequency responses.

NOTE—In principle, a very small ideal microphone should be used to calibrate a mouth simulator, so that the physical size of the microphone does not influence the measurement. In practice, a 6.25 mm laboratory measurement microphone with flat frequency response in a pressure field may be used to calibrate a HATS mouth simulator to the required accuracy. Some manufacturers recommend a free-field microphone instead, which typically has less sensitivity to mechanical vibration and results in a better calibration. The free-field microphone can be compensated to give the same frequency response as a pressure microphone by using free-field correction curves for the angle of sound incidence. The compensation is on the order of 1 dB at 8 kHz.

7. Test procedure for analog sets

7.1 General

Procedures are given in the following clauses for measurement of receive, send, sidetone, and overall performance characteristics of handset and headset telephones. Parameters include frequency response, noise, input–output linearity, distortion, and mute. In addition, procedures are given for measuring telephone set impedance, howling, and maximum acoustic output.

The telephone should be connected to the test circuit(s) described in 7.2. Other test circuits may be used for specific applications. Because telephone set characteristics are affected by loop impedances, terminations, loop currents, and operating levels, the measurements should be made using test loops and other conditions representative of those conditions the telephone is expected to encounter in use. Records should be kept of the measurement conditions.

The measured frequency responses shall be presented as decibels relative to one pascal per volt [dB (Pa/V)] for receive, decibels relative to one volt per pascal [dB (V/Pa)] for send, decibels relative to one pascal per pascal [dB (Pa/Pa)] for sidetone and overall, and decibels relative to one volt per volt [dB (V/V)] for echo. The stimulus level and signal type shall be reported for each test.

The calibration procedures described in Clause 6 shall be carried out before making any measurements. The acoustical test environment shall meet the specifications given in 5.5.

7.1.1 Choice of test signals and levels

In general, multiple test signals and stimulus levels should be used to ensure the telephone is characterized in realistic, stable, and well-defined states. This is especially the case for telephones with nonlinear processes such as compression or voice activated switching (VOX) circuitry, etc. See Annex F and Annex G for further information on test signals and analysis methods.

The standard test signal for all telephones consists of artificial voices defined in ITU-T Recommendation P.50-1999. See F.6.1.1 for details.

Sinusoidal test signals (F.4.1) may be used for testing telephones, handsets, or headsets, if it can be shown that they do not have adaptive, nonlinear or dynamic signal processing (e.g., compressors, AGC, voice activity detection, adaptive echo cancellers, etc.). Such evidence shall be given in the test report if sinusoidal test signals are used.

Other test signals may be used when it can be shown that they produce results consistent with actual use. They also may be necessary for some specific purposes as discussed in relevant places within this standard.

The measurements in this clause shall be performed at the standard test levels specified in 6.6.2 and 6.7.2.

7.1.2 Measurement bandwidth and resolution

The measurement shall be performed using the same format as was used for calibration. Format examples are $1/N$ octave bandwidth analysis, constant bandwidth analysis, and R-series preferred frequencies. Measurement bandwidth shall be the same as or less than that which was used for calibration. Measurement resolution shall be the same as or coarser than that which was used for calibration. The actual bandwidth used shall be stated.

In general, the test signals and analysis methods in this standard cover a frequency range from approximately 100 to 8500 Hz. The exact range depends on the analysis method, and the test signal (see G.6 and G.7)

7.1.3 Choice of ear and mouth simulators and test position

Choose the ear simulator, mouth simulator, and test position according to 5.1, 5.2, and 5.3. This equipment shall be used for all tests described in Clause 7, unless otherwise specified. The ear simulator, mouth simulator, and test position used shall be stated.

7.1.4 Tone control setting

If the telephone is equipped with a tone control, the tone control shall be set to the manufacturer's default setting. This is the default tone control adjustment that shall be used for all measurements.

If no default setting is defined by the manufacturer, the tone control shall be set so that the frequency response is as close as possible to the center of the required frequency response template. The tone control shall be set before setting the volume control. If the tone and volume controls interact, an iterative process for setting these controls may be necessary.

7.1.5 Reference receive volume control setting

All measurements shall be done at the reference receive volume control setting (3.38). A range of volume control settings may also be used where appropriate, such as minimum and maximum volume.

7.1.6 Reference send gain control setting

All measurements shall be done at the reference send volume control setting (3.39). A range of volume control settings may be used where appropriate, such as minimum and maximum volume.

7.2 Analog DC feed circuits

A general-purpose DC feed circuit is shown in Figure 7. Since the parameters of the feed circuit affect transmission performance, they should be recorded as part of the test setup. If available, parameters should be obtained from the applicable performance specification. If not, the following values should be used:

- $C \geq 50 \mu\text{F}$ (microfarads)
- $L \geq 5 \text{ H}$ (henries) (each)
- $R = 400 \Omega$ (ohms), including resistance of inductors

$V = 50$ V (volts)

A = ammeter used to measure current drawn by the telephone under test. Alternatively, the current can be fixed by a current source, regardless of the R value.

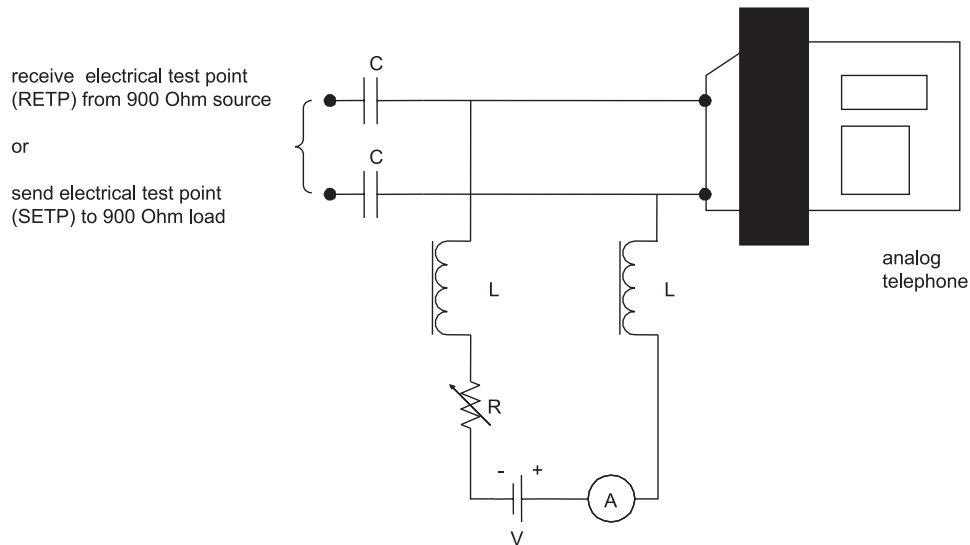


Figure 7—General purpose DC feed circuit for two-wire analog telephone

In some cases, ground loops may occur when connecting test equipment to RETP or SETP. The insertion of a high quality 1:1 audio transformer can usually prevent this. If used, this transformer shall be included during calibration and when determining the loss of the feed circuit.

The send electrical test point (SETP) is for measuring send output signals. It shall be connected to a $900\ \Omega$ load. The receive electrical test point (RETP) is for applying receive input signals. It shall be connected to a $900\ \Omega$ source. (Other terminations may be substituted as defined by applicable performance specifications.)

The loss of the feed circuit used should be measured. The loss should not be greater than 0.1 dB over the range 100–8500 Hz. The loss from 20 to 100 Hz should not exceed 1 dB. The circuit of Figure 7, using ideal components, should just meet this specification.

The following procedure may be used to determine the loss of the feed circuit:

- Connect a signal generator or similar device with a $900\ \Omega$ source impedance directly to a $900\ \Omega$ resistive termination and measure the voltage across the termination.
- Insert the feed circuit between the generator and the $900\ \Omega$ resistive termination and again measure the voltage across the termination.
- The loss of the feed circuit in decibels is

$$\text{Feed circuit loss} = 20 \log \frac{\text{Voltage across } 900\ \Omega \text{ resistor}}{\text{Voltage across feed bridge}}$$

This procedure should be followed for every value of direct current and frequency of interest. In practice, it is usually sufficient to measure a few conditions covering the range of values likely to be encountered and, if the effect of the feed circuit is relatively small and constant for the measured conditions, assume the characterization is sufficient.

The noise level of the feeding bridge should be low enough not to influence measurement results.

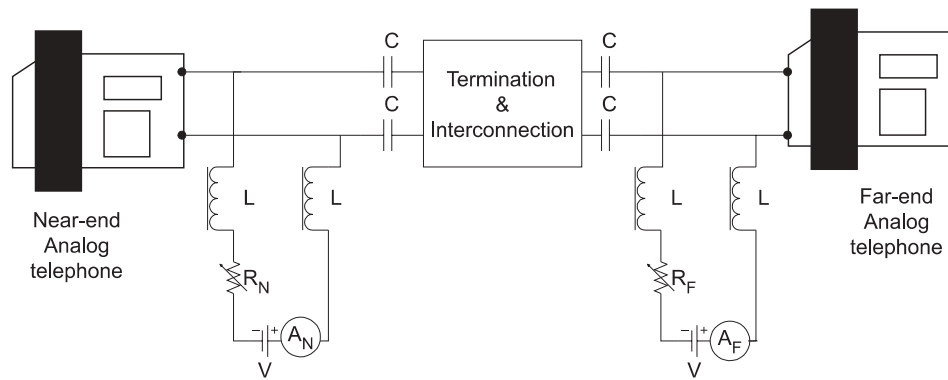


Figure 8—General purpose DC feed circuit for two analog telephones for overall measurements

Feed circuit for overall measurements using two phones is shown in Figure 8.

7.3 Analog telephone network impairments

Telephone performance can be influenced by various conditions in the network to which a telephone is connected. The specific impairments described in 7.3.1 through 7.3.6 should be investigated where applicable. Other impairments, such as ADSL signals from a high-speed modem, may be relevant for specific situations. The general method is to make a standard measurement as specified in 7.4 through 7.7, but with the impairment introduced.

7.3.1 Loop current

Loop current may be varied to determine if there are any detrimental effects. This is especially important if the telephone is powered from the line rather than from a local power supply.

7.3.2 Network noise

Network noise can affect nonlinear processes within an analog telephone. Network noise shall be approximated using white noise, with levels measured in dBmp. Noise shall be inserted at the RETP.

7.3.3 Termination impedance

Network termination impedance will affect analog telephone transmission performance. A $900\ \Omega$ termination is recommended for telephones connected to a central office network. A $600\ \Omega$ termination is recommended for telephones connected to a PBX network.

Other terminations may be used for specific applications. For example, a complex termination more typical of North American loops, may be useful for sidetone measurements.

7.3.4 Test loops

A wireline analog telephone should be tested with various lengths of cable or simulated cable. Recommended loop lengths for testing North American telephones are 0, 2.7, and 4.6 km (0, 9, and

15 kft) of 26 AWG nonloaded cable. The recommended loop simulator circuit is shown in Figure 9 and components for various lengths are shown in Table 3.

For some measurements, particularly sidetone and howling, real cable may give results more representative of actual performance compared to the loop simulator of Figure 9 and Table 3.

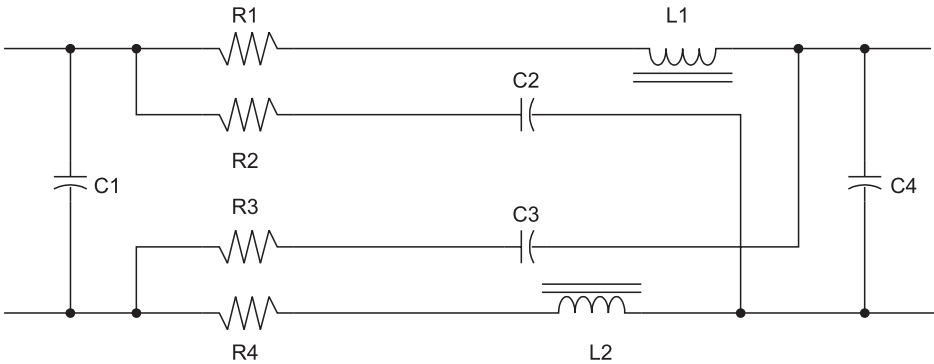


Figure 9—Loop simulator circuit

Table 3—Component values for 26 AWG cable

Component	0.305 km ^a	0.914 km ^b	1.83 km ^c
R_1, R_4	41.7 Ω	124 Ω	249 Ω
R_2, R_3	109 Ω	174 Ω	312 Ω
C_1, C_4	3.77 nF	0.0113 μ F	0.0226 μ F
C_2, C_3	4.02 nF	0.0122 μ F	0.0255 μ F
L_1, L_2	96.0 μ H	0.336 mH	0.983 mH

NOTES

- 1—All values are $\pm 1\%$
2—2.7 km and 4.6 km can be made up of cascaded sections of the above^d
^a0.305 km = 1 kft
^b0.914 km = 3 kft
^c1.83 km = 6 kft
^d2.7 km = 9 kft. 4.6 km = 15 kft

7.3.5 Parallel sets

The telephone should be tested with a parallel telephone set simulator with suitable dc and ac characteristics. In general, measurements made with the parallel set simulator should be compared to the same measurements made without the simulator. The minimum recommended measurements are send and receive frequency response, loudness ratings, and distortion, each measured with standard loop lengths.

The parallel telephone set simulator circuit shall have the VI curve as shown in Figure 10, ± 0.3 V over the current range of 0–100 mA. The return loss shall be greater than 10 dB with respect to 600 Ω from 200 to 4000 Hz. Component values may be adjusted to meet these tolerances. One possible implementation of this is shown in Figure 11.

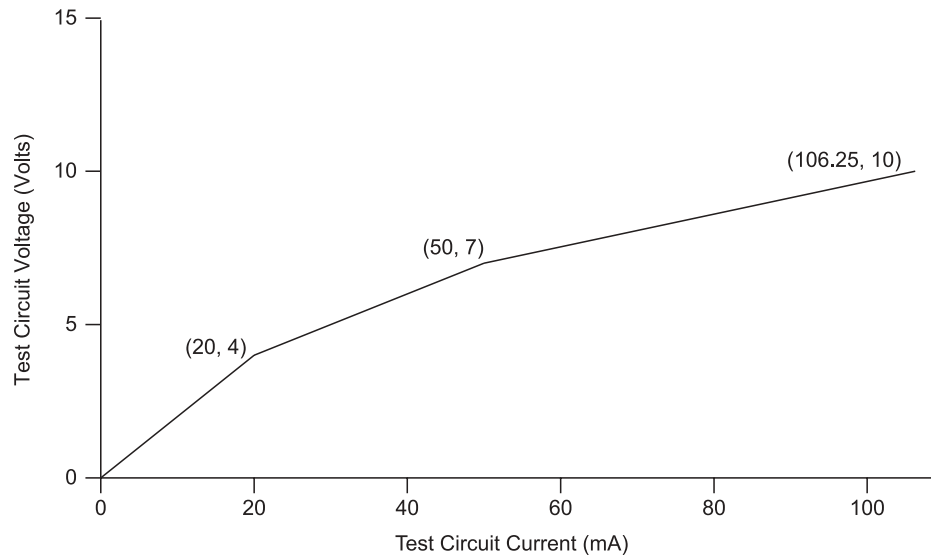


Figure 10—Parallel set test circuit VI curve

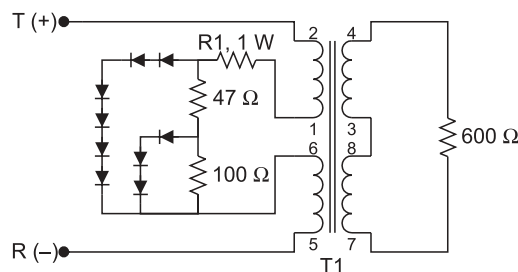


Figure 11—Parallel telephone set simulator

NOTES

- 1—Circuit is polarity sensitive.
- 2—Use IN4004, or similar, for diode strings.
- 3—T1 may be Western Electric 120C, Hammond 560G, Tellabs 4422 or equivalent (600 Ω 100 mA).
- 4—Resistor values shall be $\pm 1\%$.
- 5—R1+ coil winding should be 53 Ω.

7.3.6 Cordless range

A cordless telephone should be tested across the range of expected usage. This should include the minimum and maximum specified distance the telephone is expected to operate between the base unit and mobile unit.

7.4 Receive

7.4.1 Receive frequency response

Receive frequency response is the ratio of sound pressure measured in the ear simulator, referred to the ear reference point (ERP), to the voltage input at the receive electrical test point (RETP), which is expressed in decibels. The receive frequency response in dB, $H_R(f)$, is given by Equation (1). $H_R(f)$

may be used to calculate the receive loudness rating (RLR) according to ITU-T Recommendation P.79-1999. See Annex H.

$$H_R(f) = 20 \log \frac{G_{ERP}(f)}{G_{RETP}(f)} \quad \text{in dB (Pa/V)} \quad (1)$$

where

$G_{ERP}(f)$ is the rms spectrum at ERP,
 $G_{RETP}(f)$ is the rms spectrum at RETP.

In some cases, frequency response calculation may be performed with cross-spectrum or related techniques. Justification for such techniques shall be given in the test report. See G.1 for more information.

7.4.2 Receive noise

Receive noise is internally generated audio frequency noise present at the receiver when no stimulus is applied. The receiver shall be coupled to the ear simulator with the RETP terminated and with no signal input. The telephone's microphone should be isolated from sound input and mechanical disturbances that would cause significant error. Measure the acoustic output signal, referred to the ERP, from 100 to 8500 Hz, averaging over a minimum period of 5 s. Receive noise should be measured with the send mute feature both "on" and "off."

The overall receive noise level is measured with A-weighting in dBA. The measurement may be implemented directly using an A-weighting filter, or by using single-channel FFT with Hann windowing or real-time spectrum analysis, followed by an A-weighted power summation.

7.4.3 Receive narrowband noise

Receive narrowband noise, including single frequency interference (SFI), is an impairment that can be perceived as a tone relative to the overall weighted noise level. This test measures the weighted noise level characteristics in narrow bands of not more than 31 Hz maximum from 100 to 8500 Hz. These levels can then be compared to the receive noise (7.4.2).

The receiver shall be coupled to the ear simulator with the RETP terminated and with no signal input. Measure the A-weighted receive noise level, referred to the ERP, using a selective voltmeter or spectrum analyzer with an effective bandwidth of not more than 31 Hz, over the frequency range of 100–8500 Hz. If FFT analysis is used, then a "flat top" windowing shall be employed.

7.4.4 Receive linearity

Receive linearity is a measure of how the frequency response changes with input level.

The test consists of measuring the receive frequency response as specified in 7.4.1 and applying the procedures described in Annex I. Linearity shall be measured using the same test method and stimulus type used to measure frequency response.

If artificial voices or another wideband stimulus are used, the test shall be performed at seven levels, from –46 to –16 dBV, in 5 dB intervals, measured in 1/3 octave bands. Smaller intervals and/or a wider range of levels may also be used. The reference stimulus level is –16 dBV. These levels take into account the high crest factor of artificial voices, which approaches 23 dB.

If sine-wave signals are used, they shall be applied at the ISO R10 frequencies from 200 through 5000 Hz, at seven levels, from -36 to -5 dBV, in 5 dB intervals. (For ISO R10 preferred frequencies, see ISO 3:1973.) Smaller intervals and/or a wider range of levels may also be used. The reference stimulus level is -21 dBV.

7.4.5 Receive distortion

The preferred distortion measurement method is receive signal-to-distortion-and-noise ratio (SDN), measured using narrowband pseudo-random noise as the stimulus. See J.3 for details of the method.

Receive distortion is measured at ERP using the standard input level of -16.0 dBV. Other input levels should be tested covering a range from -30 to 0 dBV. Measurements should also be made over a range of frequencies within the telephone band, such as the ISO R10 preferred frequencies. For higher input levels above 0 dBV, verify that distortion of the test system is less than 1% THD.

For information about THD and other distortion measurement methods and test signals, and the conditions under which they may be used, see Annex J. Different distortion measurement methods are likely to give different results.

7.4.6 Receive mute leakage

Receive mute is sometimes called “DTMF mute” or “autodial mute.” Receive muting is usually automatic, but may be manually controlled, and would normally be activated by touch-tone dialing, line “hold” operation, activating the hold button, or other means. Mute leakage is the amount of signal measured at the ERP when an electrical stimulus is applied to the RETP.

To measure mute leakage, engage the mute, apply the test signal, and measure the receive noise according to 7.4.2. The test signal shall be the same as that used for receive frequency response (7.4.1), at 0 dBV. An additional measurement shall be made using the DTMF tones of the telephone set being tested. In the case of the DTMF tones of the telephone set, there will not be any control over the level. Each result is expressed in dBA, the weighted noise level which should be compared to muted receive noise measured according to 7.4.2 (with no stimulus applied).

NOTE—If a sinusoidal stimulus was used to measure receive frequency response in 7.4.1, the same sinusoidal frequency pattern shall be used for the mute measurement, but only over the range of 200–4000 Hz, at 0 dBV. The absolute level at each frequency is measured, not the frequency response. A-weighting should be applied to the result, expressed in dBA as a function of frequency. The weighting permits more relevant comparison with results obtained with artificial voices.

7.5 Send

7.5.1 Send frequency response

Send frequency response is the ratio of voltage output at the send electrical test point (SETP) to the sound pressure at the mouth reference point (MRP), which is expressed in decibels. The send frequency response in dB, $H_S(f)$, is given by Equation (2). The send frequency response, $H_S(f)$, may be used to calculate the send loudness rating (SLR) according to ITU-T Recommendation P.79-1999. See Annex H.

$$H_S(f) = 20 \log \frac{G_{SETP}(f)}{G_{MRP}(f)} \quad \text{in dB (V/Pa)} \quad (2)$$

where

$G_{\text{SETP}}(f)$ is the rms spectrum at SETP,
 $G_{\text{MRP}}(f)$ is the rms spectrum at MRP.

In some cases, frequency response calculation may be performed with cross-spectrum or related techniques. Justification for such techniques shall be given in the test report. See G.1 for more information.

7.5.2 Send noise

Send noise is internally generated audio frequency noise present at the SETP. Measure the electrical output signal at SETP, averaging over a minimum period of 5 s. The telephone's microphone should be isolated from sound input and mechanical disturbances that would cause significant error. Send noise should be measured with the mute feature both "on" and "off."

Send overall noise shall be measured and reported in units of dBmp. It shall also be measured with A-weighting (defined in ANSI S1.4-1983), reported in units of dBm(A). Measurements in dBmp and dBm(A) are generally not the same, and they may not be correlated.

Psophometric measurements are made from 100 to 6000 Hz, while A-weighted measurements are made from 100 to 8500 Hz. These measurements can be made directly using a psophometrically weighted or A-weighted noise meter with the correct terminating impedance. The measurement may also be implemented using a single-channel FFT with Hann windowing, or a real-time spectrum analysis, followed by a weighted power summation.

7.5.3 Send narrowband noise

Send narrowband noise, including single frequency interference (SFI), is an impairment that can be perceived as a tone relative to the overall weighted noise level. This test measures the weighted noise level characteristics in narrow bands of not more than 31 Hz maximum from 100 to 6500 Hz. These levels can then be compared to the send noise (7.5.2).

The handset or headset should be isolated from sound input and mechanical disturbances that would cause significant error. Measure the psophometrically weighted noise level at the SETP with a selective voltmeter or spectrum analyzer with an effective bandwidth of not more than 31 Hz, over the frequency range of 100–6500 Hz. If FFT analysis is used, then a "flat top" windowing shall be employed.

The procedure shall be repeated using A-weighting instead of psophometric weighting, and the frequency range shall be changed to 100–8500 Hz.

7.5.4 Send linearity

Send linearity is a measure of how the frequency response changes with input level.

The test consists of measuring the send frequency response as specified in 7.5.1 and applying the procedures described in Annex I. Linearity shall be measured using the same test method and stimulus type used to measure frequency response.

If artificial voices or another wideband test signal are used, the test shall be performed at seven levels from –34.7 to –4.7 dBPa, in 5 dB intervals, measured in 1/3 octave bands. Smaller intervals and/or

a wider range of levels may also be used. The reference stimulus level is -4.7 dBPa. These levels take into account the high crest factor of artificial voices, which approaches 23 dB.

If sine-wave signals are used, they shall be applied at the R10 frequencies from 200 through 5000 Hz, at seven levels, from -24.7 to $+5.3$ dBPa, in 5 dB intervals. Smaller intervals and/or a wider range of levels may also be used. The reference stimulus level is -9.7 dBPa.

7.5.5 Send distortion

The preferred distortion measurement method is send signal-to-distortion-and-noise ratio (SDN), measured using narrowband pseudo-random noise as the stimulus. See J.3 for details of the method.

Send distortion is measured at SETP using the standard input level of -4.7 dBPa. Other input levels should be tested covering a range from -30 to $+10$ dBPa. Measurements should also be made over a range of frequencies within the telephone band, such as the ISO R10 preferred frequencies. For higher input levels, verify that distortion of the test system is less than 2% THD.

For information about THD and other distortion measurement methods and test signals, and the conditions under which they may be used, see Annex J. Different distortion measurement methods are likely to give different results.

7.5.6 Send mute leakage

The mute function is for voice privacy during line hold and mute. Send muting is often manually controlled, but may be automatically controlled. Mute leakage is the amount of signal measured at the SETP when an acoustic stimulus is applied to the handset or headset microphone.

To measure mute leakage, engage the mute, apply the test signal, and measure the send noise according to 7.5.2. The test signal shall be the same as that used for send frequency response (7.5.1), at $+5$ dBPa. The result is expressed in dBmp, a weighted noise level that should be compared to muted broadband noise measured according to 7.5.2 (with no stimulus).

NOTE—If a sinusoidal stimulus was used to measure send frequency response in 7.5.1, the same sinusoidal frequency pattern shall be used for the mute measurement, but only over the range of 200–4000 Hz, at $+5$ dBPa. The absolute level at each frequency is measured, not the frequency response. Psophometric weighting should be applied to the result, expressed in dBmp as a function of frequency. The weighting permits more relevant comparison with results obtained with artificial voices.

7.5.7 Send frequency response in a diffuse field

Send frequency response in a diffuse field is a measure of how much of the noise in the room where a telephone is being used is transmitted to the network. It is the ratio of voltage output at the send electrical test point (SETP) to the sound pressure at the diffuse field test point (DFTP, see 5.5.3), which is expressed in decibels. The diffuse field send frequency response in dB, $H_{SD}(f)$, is given by Equation (3).

The diffuse field send frequency response may be sensitive to both the level and type of signal used. This measurement may be performed in 1/3 octave resolution.

During the measurement, the mouth simulator is present but not active, with the MRP is located at the DFTP. The mouth simulator is not present during calibration.

$$H_{SD}(f) = 20 \log \frac{G_{SETP}(f)}{G_{DFTP}(f)} \quad \text{in dB (V/Pa)} \quad (3)$$