



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

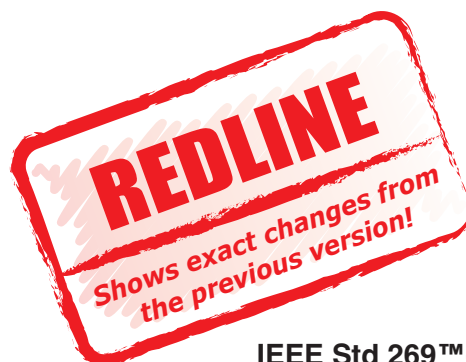
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IEEE Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets, and Headsets

Sponsor

Transmission Access & Optical Systems Committee
of the
IEEE Communications Society

Approved 25 March 2010

IEEE-SA Standards Board

Abstract: Practical methods for making laboratory measurements of electroacoustic characteristics of analog and digital telephones, handsets and headsets. 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 from 100 Hz to 8500 Hz. This edition incorporates IEEE Std 269a™-2007. Other substantive changes have been made since the 2002 edition.

Keywords: analog telephones, digital telephones, electroacoustic measurements on telephones, handsets, headsets, IEEE 269, telephony, voice transmission performance

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Introduction

This introduction is not part of IEEE Std 269-2010, 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 269-2002. 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.

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Miguel De Araujo
Daniel Foley

Frederick Kruger
Ron D. Magnuson
Anthony Scott

Christopher J. Struck
Stephen R. Whitesell
Allen Woo
Robert Young

The following members of the individual balloting committee voted on this standard. Balloters may have voted for approval, disapproval, or abstention.

John Bareham
Hugh Barrass
Al Baum
Stephen H. Berger
Clifford Chamney
Keith Chow
Miguel De Araujo
Carlo Donati
Daniel Foley
Steve Graham

Randall Groves
Glenn Hess
Werner Hoelzl
Raj Jain
Piotr Karocki
Frederick Kruger
G. Luri
Gary Michel
Michael S. Newman
Charles Ngethe

Gary Nissen
Bartien Sayogo
Jeffrey Schenck
Anthony Scott
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Michelle D. Turner
IEEE Standards Program Manager, Document Development

Matthew J. Ceglia
IEEE Standards Program Manager, Technical Program Development

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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 from 100 Hz to 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.

Digital devices that have audio characteristics of a digital telephone can be tested as a digital telephone, even if they do not have a dialpad or other similar telephone functions. USB headsets are one example of devices in this class.

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Telephones with speakerphone features are covered by IEEE Std 1329™.¹

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 6 through Clause 8 of the document.

Clause 2 – Clause 3 provides references, definitions, abbreviations, acronyms, and symbols which 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 4 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 Acoustic impairments include specific acoustic conditions [which could degrade the performance of the device under test](#).

Clause 5 describes the calibration procedures needed to ensure that the equipment is in a known state. Calibration of the acoustic transducers and electrical interfaces are explained.

[Clause 6 contains](#) the transmission test procedures for telephones with handsets or headsets [that connect to an analog telephone network](#).

¹[Information on normative references can be found in Clause 2.](#)

[Clause 7 contains the transmission test procedures for telephones with handsets or headsets that connect to a digital telephone network or digital communications interface. It also applies to handsets or headsets that connect to a digital communications interface.](#)

[Clause 8 contains the transmission test procedures for 4-wire analog headsets and analog handsets that are not connected to a telephone.](#)

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

1.4 Changes since last edition of this standard

The following major changes in this revision are:

- a) The IEEE Std 269-2002 has been integrated with IEEE Std 269a™-2007, Amendment 1.
- b) The Head And Torso Simulator (HATS) fitted with the Type 3.3 Ear Simulator is the preferred test fixture for most measurements.
- c) Most tests are performed using real speech. Specific speech signal files are provided.
- d) New receive parameters have been included covering acoustic output level, conversational gain, and tonal balance.
- e) A new pulsed noise distortion test is described.
- f) The maximum acoustic pressure test has been revised.

2. Normative references

The following referenced documents are indispensable for the application of this document (i.e., they must be understood and used, so each referenced document is cited in text and its relationship to this document is explained). For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments or corrigenda) applies. ~~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, American National Standard Acoustical Terminology.²

ANSI S1.4, American National Standard Specification for Sound Level Meters.

ANSI S1.6, American National Standard Preferred Frequencies, Frequency Levels, and Band Numbers for Acoustical Measurements.

ANSI S1.11, American National Standard Specifications for Octave Band and Fractional-octave-band Analog and Digital Filters.

IEC 61094-4, Measurement microphones – Part 4: Specifications for working standard microphones.

IEEE Std 1652™, IEEE Standard for the Application of Free Field Acoustic Reference to Telephony Measurements.

ITU-T Recommendation G.712, Transmission performance characteristics of pulse code modulation channels.

ITU-T Recommendation G.115-1996, Mean Active Speech Level for Announcements and Speech Synthesis Systems.

ITU-T Recommendation G.169, Automatic Level Control Devices.

ITU-T Recommendation G.223, Assumptions for the Calculation of Noise on Hypothetical Reference Circuits for Telephony.

Bellcore SR-TSV-002476, December, 1992.

ITU-T Recommendation G.723, Extensions of Recommendation G.721 Adaptive Differential Pulse Code Modulation to 24 and 40 kbit/s for Digital Circuit Multiplication Equipment Application.

² ANSI publications are available from the Customer Service Department, American National Standards Institute, 25 W. 43rd Street, 4th Floor, New York, NY 10036, USA (<http://www.ansi.org/>).

ANSI S1.16, American National Standard ~~Specifications for Laboratory Standard~~ [Method for Measuring the Performance of Noise Discriminating and Noise Cancelling](#) Microphones.

ANSI/TIA/EIA-579-A- 1998 , [Telecommunications Telephone Terminal Equipment 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 4-5: Testing and measurement techniques –Surge immunity test.³

IEEE Std 661™-1979 (Reaff. 1998), IEEE Standard Method for Determining Objective Loudness Ratings of Telephone Connections.^{4, 5}

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

IEEE Std 1329™, IEEE Standard Method for Measuring Transmission Performance of Handsfree Telephone Sets.

ISO 3, Preferred Numbers-Series of preferred Numbers.

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

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

[ITU-T Recommendation G.722](#), 7 kHz Audio-coding Within 64 kbit/s.

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

ITU-T Recommendation G.729, Coding of Speech at 8 kbit/s Using Conjugate-structure Algebraic-code-excited Linear-prediction (CS-ACELP).

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~~⁵ISO publications are available from the ISO Central Secretariat, Case Postale 1 rue de Varembe, CH 1211, Geneva 20, Switzerland/Suisse (<http://www.iso.ch/>). ISO publications are also available in the United States from the Sales Department, American National Standards Institute, 11 West 42nd Street, 13th Floor, New York, NY 10036, USA (<http://www.ansi.org/>).~~

~~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 O.133, Equipment for Measuring the Performance of PCM Encoders and Decoders.

ITU-T Recommendation P.50, Artificial Voices.

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

ITU-T Recommendation P.51, Artificial Mouths.

ITU-T Recommendation O.41 (1994), Psophometer for Use on Telephone-type Circuits.

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

ITU-T Recommendation P.57, Artificial Ears.

ITU-T Recommendation P.58, Head and Torso Simulator for Telephonometry. ITU-T Recommendation P.59, Artificial Conversational Speech.

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

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

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

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

~~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—~~

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

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

3. Definitions, acronyms and abbreviations, symbols

~~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]² and ANSI S1.1 1994.~~

3.1 Definitions

For the purposes of this document, the following terms and definitions apply. The IEEE Standards Dictionary: Glossary of Terms & Definitions should be referenced for terms not defined in this clause.⁶

A-weighted: A measurement made using the “A” frequency weighting specified in ANSI S1.4-1983. A-weighting approximates the frequency response of the ear at 40dBSPL, so it is frequently used for measurements at low sound levels. For completely different reasons, it is also used for many acoustic safety measurements. It is generally not appropriate for use at typical telephone conversation levels. ~~sound pressure level is expressed as dBA, and the reference level is always 20 mPa (micropascals)~~

²~~The numbers in brackets correspond to those of the bibliography in Annex V.~~

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

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

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

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.~~

⁶The IEEE Standards Dictionary: Glossary of Terms & Definitions is available at <http://shop.ieee.org/>.

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.

dB(A): Sound pressure level in decibels, relative to 20 micropascals, A-weighted ([see First definition](#)). [The proper term is dB SPL\(A\). To match common usage we use dB\(A\) herein. The term dB\(A\) may be seen in other documents.](#)

[dBFS: Decibels relative to full-scale digital saturation. 0 dBFS is the level of any signal whose maximum peak is at saturation.](#)

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

dBm0: Power level in dBm, relative to a reference point called the zero transmission level point, or 0 TLP. (Check xref). 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.

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

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

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

dB SPL: Sound pressure level in decibels, relative to a sound pressure of 20 micropascals.

dBV: Voltage level in decibels, relative to 1 volt rms.

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

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

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

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.

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.

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.

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

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

head and torso simulator (HATS) for telephonometry: A manikin [incorporating a mouth simulator and ear simulator\(s\)](#), 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).

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.

ideal codec: A codec that has theoretically optimum characteristics.

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

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

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

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

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

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.

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

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

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

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

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

reference receive volume control setting: The [setting of one or more](#) receive volume control of a telephone system which results in the receive loudness rating (RLR) closest to the specified target value. [\(Other words are often used to describe this concept, such as “nominal.”\)](#)

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

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

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

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.

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

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

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 (dBP_a). When a reference pressure of 20 µPa 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 newton per square meter). However, measurements such as receive noise and room noise are generally referenced to 20 µPa.

Note: 0 dB Pa = 94 dBSPL, 0 dBSPL = 20 micropascals, 1 Pa = 1 N/m². A-weighted sound pressure level in dB (dBSPL, A-weighted) is often abbreviated as dB(A). [see ANSI S1.4- 1983 (R 1997)]

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.

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 (~23 % and ~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.

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

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

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

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

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

3.2 Abbreviations and acronyms

AGC	automatic gain control
ANTHD	amplitude normalized total harmonic distortion
DFTP	diffuse field test point
DRP	drum reference point
DRTP	derived recommended test position
DSP	digital signal processor
DTX	discontinuous speech transmission
ECRP	ear cap reference point
EPD	echo path delay
ERP	ear reference point
ERUP	estimated real use position
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
RTR	recommended test position

SETP	send electrical test point
SDNR	signal-to-distortion-and-noise ratio
SFI	single frequency interference
SLR	send loudness rating
SNR	signal-to-noise ratio
STMR	sidetone masking rating
STP	standard test position
TDS	time delay spectrometry
THD	total harmonic distortion
VAD	voice activity detector

3.3 Symbols

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

$G_{DFTP}(f)$ = rms spectrum at Diffuse Field Test Point, in dBPa
 $G_{ERP}(f)$ = rms spectrum at Ear Reference Point, in dBPa
 $G_{MRP}(f)$ = rms spectrum at Mouth Reference Point, in dBPa
 $G_{(MRP)(ERP)}(f)$ = cross-spectrum between MRP and ERP, in dB (Pa/Pa)
 $G_{(MRP)(SETP)}(f)$ = cross-spectrum between MRP and SETP, in dB (V/Pa)
 $G_{RETP}(f)$ = rms spectrum at Receive Electrical Test Point, in dBV
 $G_{(RETP)(ERP)}(f)$ = cross-spectrum between RETP and ERP, in dB (Pa/V)
 $G_{(RETP)(SETP)}(f)$ = cross-spectrum RETP and SETP, in dB (V/V)
 $G_{SETP}(f)$ = rms spectrum at Send Electrical Test Point, in dBV
 $G_{SETP(S+N)}(f)$ = rms spectrum at SETP with both the mouth simulator and noise sources active, in dBV
 $G_{SETP(N)}(f)$ = 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
 $H_{EP}(f)$ = echo path frequency response, in dB (V/V)
 $H_{LS}(f)$ = listener sidetone frequency response, in dB (Pa/Pa)
 $H_O(f)$ = overall frequency response, in dB (Pa/Pa)
 $H_R(f)$ = receive frequency response, in dB (Pa/V)
 $H_S(f)$ = send frequency response, in dB (V/Pa)
 $H_{SD}(f)$ = diffuse field send frequency response, in dB (V/Pa)
 $H_{TS}(f)$ = 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 8.3.2)
 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
TCL = 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

4. 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 4-wire devices such as handsets and headsets, are described in Clause 6, Clause 7, and Clause 8 respectively.

All equipment should be calibrated in accordance with the recommendations of the manufacturer before performing the system calibration procedures in Clause 5.

4.1 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.~~

4.1.1 Selection

~~The preferred ear simulator with a flexible pinna is the Type 3.3. (Please see Annex A). It shall be used for all measurements and devices, unless the applicable performance specification requires or allows an alternative. If an alternative is used, the requirements of Annex B shall be met.~~

The Type 3.3 ear simulator shall comply with the specifications given in ITU-T Recommendation P.57. The Type 3.3 ~~shall have a hardness of 35 ± 6 degrees Shore-OO, as measured according to ASTM 2240. (ITU-T Recommendation P.57-2002 originally specified a hardness of 55 ± 10 degrees Shore-OO for Type 3.3) is recommended for all devices. The Type 3.4 ear simulator is recommended for all devices except supra-concha and supra-aural headsets.~~

~~Type 3.3 and 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.~~

~~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.

4.1.2 Headset Measurements made on [the Type 3.3](#) ear simulator compared to real ears

Type 3.3 ear simulator simulates the acoustical and mechanical characteristics of [the median human ear](#). It is likely to give results comparable to the typical listening experience for the widest possible variety of headsets, [handsets and similar devices](#). However, the correlation between measurements on ear simulators and on real ears is better for some ~~headset types~~ [devices](#) than others.

The correlation for headsets that are in close proximity to, or in the ear canal, is not as good as for most other types.

For insert headsets, the Type 3.3 ear simulator may provide a greater seal than on many human subjects, resulting in an overestimation of output at low frequencies. Nonetheless, the Type 3.3 ear simulator is recommended for this application.

4.1.3 Translation from DRP to ERP

Type 3.3 ear simulator measures at the eardrum reference point (DRP). [Unless otherwise specified or permitted in this standard](#), 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.

[4.1.4 Translation from DRP to free field](#)

[Measurements collected at the DRP may be translated to the free field. Some clauses in this standard require translation to the free field. The free field translation may be done along with the translation to ERP.](#)

[Please see IEEE Std 1652 for the method, translation curves, and background information.](#)

4.2 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, 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 does not define a sound field behind the lip plane of the mouth simulator.
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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, speakerphones, 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.

4.3 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.

4.3.1 Selection

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

The LRGP position was specified in previous editions of this standard. Send frequency response measurements made on ordinary telephones from 300 Hz to 3400 Hz are expected to give practically identical results, whether obtained with LRGP or the HATS position. Systematic differences of about 1 dB to 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, 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.

4.3.2 Handset positioning

The two alternatives for handset positioning are the standard test position (STP) and the manufacturer's recommended test position (RTP). STP is defined in Figure 1 and RTP is defined in 4.3.2.2. STP shall be used unless an RTP is defined by the manufacturer. Figure 1 shows coordinates as defined in ITU-T P.64, Annex E.

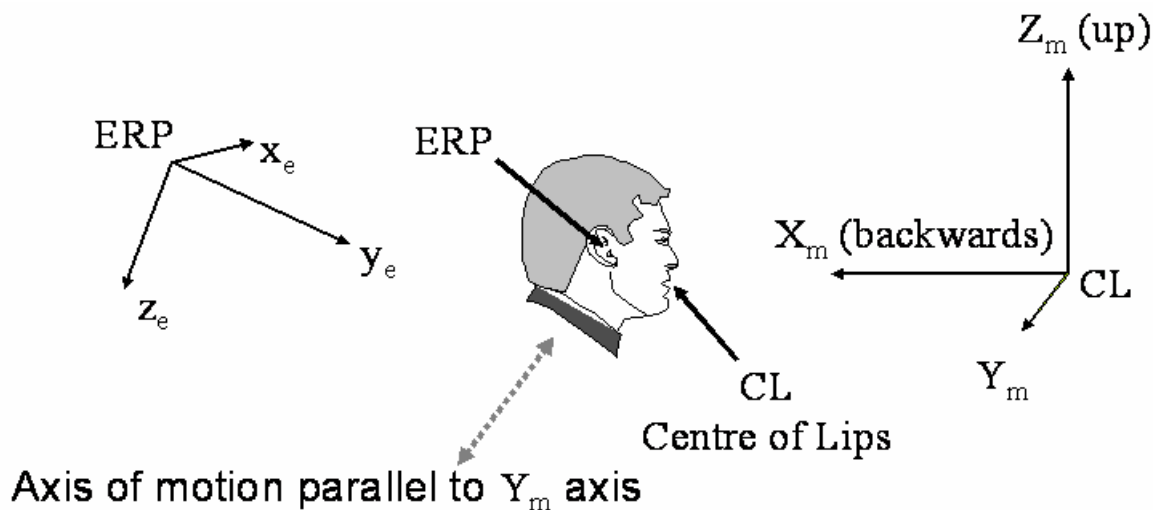


Figure 1— Coordinates as defined in ITU-T P.64, Annex

4.3.2.1 Standard test position (STP)

For STP, the handset receiver shall be nominally placed in the HATS position as specified by Annex D or E of ITU-T Recommendation P.64. 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. It is parallel to the Y_m axis (see ITU-T P.64, Annex E). The ECRP may be inside of or outside of ERP depending on the applied force and the shape of the receiver.

Unless otherwise specified by the manufacturer, the ECRP is the intersection of the external ear-cap reference plane (ECRP Plane) with a normal axis (x_e axis as defined by ITU-T P.64, Annex E) 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 ECRP 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 to know how the receive frequency response and loudness rating change ~~too much~~ as application force changes. For this reason, the device shall be tested at 2 N a high leak position and 10 N a low leak position, which represent loosely coupled and firmly coupled conditions.

For the Type 3.3 ear simulator, the high leak position is an application force of 10 N and the low leak position is an application force of 18 N. All tests shall be performed at the high leak position. The tests in Table 1 shall also be performed at the low leak position.

Table 1—Low-leak position tests

Test type	Analog	Digital	4-Wire Handsets
Receive	6.4.1	7.4.1	8.3.3
Send	6.5.1	7.5.1	8.4.2
Sidetone	6.6.1 , 6.6.2	7.6.1 , 7.6.2	
Max Acoustic Output	6.10	7.13	8.6

Maximum acoustic output (see 7.10, 7.13, and 8.6) shall be tested at both high leak and low leak position. The final result shall be the 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 of the~~ two measurements.

~~(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.)~~

4.3.2.2 Recommended Test Position (RTP)

A manufacturer may specify a recommended test position (RTP). ~~This can be described as a 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 translation, followed by a rotation, followed by an application of force.~~ The force applied shall not exceed ~~the range of 2–10 N. If the phone is tested at the RTP, the definition~~ 18 N. The definition of the RTP, including evidence of its authorization by the device manufacturer, shall be included in the test report.

RTP is defined in following steps, using coordinates as defined in ITU-T P.64, Annex E as shown in Figure 1:

- 1) Find ear cap reference point (ECRP) on the handset. Unless otherwise specified by the manufacturer, the ECRP is the intersection of the external ear-cap reference plane with a normal axis (x_e) 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 some handsets, the ear cap reference plane has to be estimated. For example, a tangent to a curved surface at the effective acoustic center.
- 2) Line up the handset ECRP at ear simulator ERP on the positioning device . The ear cap reference plane shall be identical to the reference plane of the positioning device. When the positioning device is set to ERP, then the ear cap reference plane, the reference plane the of positioning device and the ERP plane are identical
- 3) Translation: Move the handset ECRP in the ear cap reference plane relative to ERP along the y_e and/or z_e axis. If no coordinates are given, leave the ECRP centered on ERP, equivalent to (0, 0) coordinates.

The y_e axis is defined along the length of the handset with positive y_e being in a direction towards the microphone from ECRP. Positive z_e axis is in a downward direction towards the floor.

- 4) Rotation: Adjust the angle(s) of the handset positioner about the ERP of the coordinate system. If no rotation angles are given, use angles consistent with HATS position.
- 5) Application of force: Adjust pressure or distance along 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. It is parallel to the Y_m axis. If no force or position is given, use 10 N.

- 6) RTP can then be defined as the combination of the above translation, rotation and force

specifications. The manufacturer of the device under test is responsible for providing this data.



Documents

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All tests shall be performed at the RTP.

Maximum acoustic output (see 6.10, 7.13 and 8.6) shall be tested at RTP, and at RTP two more times, except the force is set to first 10 N and then to 18 N. The final result shall be the maximum of the three measurements.

~~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.~~

4.3.3 Headset positioning

For a given headset placement, the same position shall be used to test all the electroacoustic parameters.

- a) RTP: If the manufacturer specifies a recommended test position (RTP), the headset shall be tested in that position. The RTP shall reflect the way the headset is intended by the manufacturer to be used in a real situation. The manufacturer should provide suitable pictures to illustrate the RTP.
- b) DRTP: If the manufacturer's RTP is not provided, but there is a recommended wearing position (RWP) provided in the user's guide, then an RTP shall be derived from the RWP and used. It shall be called the derived recommended test position (DRTP).
- c) ERUP: If the manufacturer's RTP and RWP are not provided, then the headset shall be tested in the estimated real use position (ERUP). The test lab shall define an ERUP that closely approximates real use. Natural headband pressure, or other positioning techniques normally used by a real person, shall be used.

Additionally, for binaural headsets, the following applies:

- d) For headsets that do not have left and right ear wearing indication, it can be tested on a HATS that is equipped with one functional ear simulator. The primary receiver (where the microphone system is located) should be positioned on the functional ear simulator for all tests. The secondary receiver shall be moved into position for evaluating frequency response, RLR and distortion measurements.
- e) For headsets that are designed to be worn in only one manner or fashion, then a HATS with two functional ear simulators shall be used. The two receivers shall be tested separately. The secondary receiver need only be tested for frequency response, RLR and distortion.

The test position shall define how the receiving part of the headset is to be placed against or inside the ear simulator. After providing receiver placement instructions, the test position shall then describe the subsequent positioning and orientation of the microphone.

The exact positioning of the microphone shall be specified using geometric coordinates relative to centre of lips using three axes (See ITU-T Rec. P.64):

- f) The X_m axis coincides with the mouth reference axis and has positive direction into the mouth.

- g) The Y_m axis is horizontal, perpendicular to the X_m axis with positive direction towards the right side of the head.

h) The Z_m axis is perpendicular to the X_m axis and Y_m axis with positive direction upwards.

The closer the microphone is to the mouth, the more sensitive the results will be to any inaccuracy of the geometrical positioning. Pressure gradient microphones (cardioid, noise canceling, etc.) are especially sensitive to both position and orientation. The recommended orientation of the microphone towards the mouth shall be stated. For headsets that are fixed at the ear, and have a short microphone boom configuration, then the receiver coupling provides the main positioning element, with the boom pointed at MRP.

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. Particular care should be exercised when positioning receivers (such as earbuds) near or in the ear canal, as they are subject to more variation than other receiver types. This generally corresponds to variation in measurements of such receivers made on large groups of human subjects. The body of the receiver, the headband or any other non-acoustical component may be positioned as necessary.

Jigs may be used to improve repeatability of positioning provided that they do not cause any acoustic impairment to the measurement.

The test operator should become acquainted with the specific headset by wearing it, using it, and also by running some preliminary learning tests on the headset mounted on the HATS.

The final positioning used should be documented with pictures. Closeups of both send and receive positioning are strongly recommended.

4.3.4 Headset measurement repeatability and averaging

When a headset is placed on HATS, the 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 send, receive, sidetone and overall measurements.

A minimum of 5 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 deviations of the measurements meet the following criteria: For receive, the standard deviation of loudness ratings shall be 1 dB or less, and the standard deviation of the group of frequency response curves shall be 2 dB or less from 200 Hz to 4000 Hz. For send, the standard deviation of loudness ratings shall be 1 dB or less, and the standard deviation of the group of frequency response curves shall be 1 dB or less from 200 Hz to 4000 Hz. If the results of the first 5 trials do not meet these criteria, report the results, but label them as reduced accuracy.

For sidetone, the standard deviation will likely be wider than either send or receive. The standard deviation of the loudness ratings and frequency response measurements shall be reported.

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 5 measurements of all other parameters should also be made, and the mean should be reported.

For maximum acoustic output, at least 5 measurements shall be made. The final result shall be the maximum of all the measured results.

~~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 remounted 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. 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~~

~~a) Recommended test position (RTP);~~

~~b) 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~~

~~T Recommendation P.58 1996.
e) 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.~~

4.4 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 IEC 61094-4.

4.5 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.

4.5.1 Background noise level

The background noise level in the test environment shall not exceed the limits shown in Table 2. The overall level shall not exceed 29 dB(A). However, these limits may be relaxed if it is 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 dB(A). The individual factory-calibrated frequency response of the microphone, if available, shall be taken into account.

Table 2— 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

4.5.2 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.

4.5.3 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 multiple speakers and uncorrelated sources in an ordinary room may be sufficient for measurements in 1/3 octaves. At least 8 speakers and sources are usually required to meet the acoustical specifications below. 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.

4.5.3.1 Test for spectrum uniformity

Spectrum uniformity should be verified by measurements made using a 6.25 mm (1/4 inch) pressure microphone, but may also be made using a 12.5 mm (1/2 inch) random incidence microphone. The individual factory-calibrated frequency response of the microphone, if available, shall be taken into account.

The acoustic test spectrum is measured at the DFTP. The spectrum shall meet its target within ± 0.5 dB. Equalization may be used to meet this requirement. The typical target is actual spectrum of the noise signal to be used.

$G_{\text{DFTP}}(f)$ (the spectrum at DFTP) shall be measured at 6 additional points ± 15 cm from DFTP on each of three mutually perpendicular axes passing through the DFTP. The spectrum at these 6 points shall not vary from $G_{\text{DFTP}}(f)$ by more than ± 3 dB in each band.

4.5.3.2 Test for diffuse field

Diffuse field conditions at the DFTP shall be verified by measurements with a cardioid or bi-directional microphone with a free field rejection of at least 10 dB front-to-back (cardioid) or front-to-side (bi-directional) in each frequency band. A cardioid microphone is preferred. The sound field shall be considered to approximate a diffuse (random-incidence) field if the variation is within the tolerance in Table 3. 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 3—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

4.6 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.

4.6.1 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 2. 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.

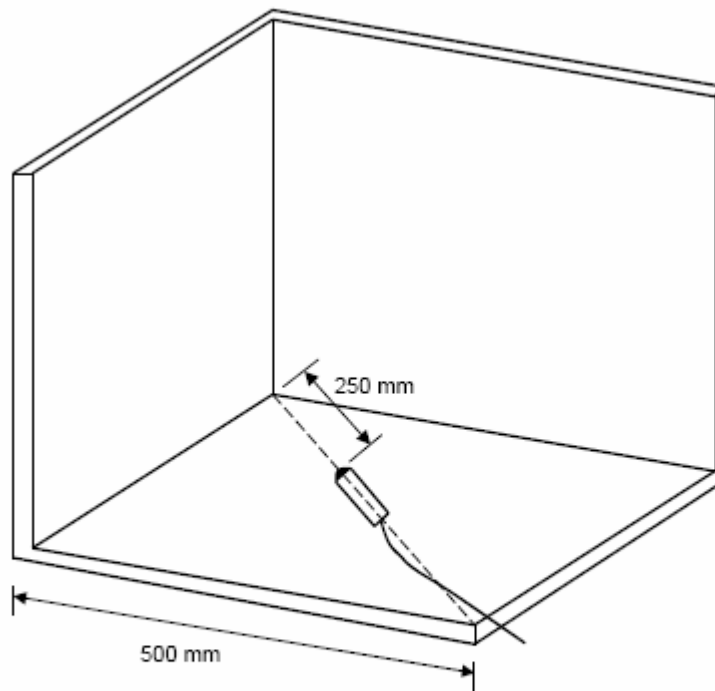


Figure 2— Reference corner for echo, howling, and stability tests

4.6.2 Hoth room noise

Hoth noise is random acoustic noise which has a spectrum designed to simulate typical ambient room noise. See Annex D for details. The noise level shall be specified in dB(A).

5. Calibration

5.1 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 minutes.

5.2 Electrical components and measurement instruments

Analyzers and level meters shall have an accuracy of ± 0.2 dB or better.

Resistors, capacitors, and small inductors shall have a tolerance of ± 1 %. Large inductors, such as those found in feeding bridges, shall be at least the specified minimum value.

5.3 Measurement microphones

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

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

5.4 Ear simulator

The sensitivity of the ear simulator shall be calibrated according to the ~~system is in use. For best accuracy, it should be calibrated prior to measurements on each new device.~~ manufacturer's directions. An acoustical calibrator with an accuracy of ~~at least~~ ± 0.2 dB or better 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.

5.5 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. The overall 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 E for additional details. Also, review E.9, E.11 and 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.

5.6 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 to calibrate the test signals shall be ~~applied from a 900 resistive source impedance for most analog telephones, and a 600 resistive source impedance for digital telephones.~~ calibrated first (see 5.2).

For analog phones, electrical test signals shall be applied from a 900 ohm resistive source impedance with a tolerance of ± 1 % or better. These calibrations are performed across matched calibrated resistive loads. As a result, electrical 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 600 Ω reference codec may be used for testing digital telephones with analog test equipment (see 7.2.2 and 7.2.3).

Alternatively, a virtual codec may be used. In this case, the digital code corresponding to 0 dBm is defined as 0.775 V. There may or may not be physical components, and their value is irrelevant. Only the voltage is relevant.

Direct digital generation of the receive signal and analysis of the send signal may be used in place of a codec.

For testing handsets and headsets as 4-wire devices, calibration is similar to that of analog phones.

See 8.2 for the source impedance requirements.

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~~A similar calibration is required for testing handset and headset four-wire devices.~~

5.6.1 Electrical test spectrum

The electrical test spectrum is measured across a calibrated resistive load. ~~For sinusoidal test signals,~~ The spectrum shall meet its target within ± 0.5 dB ~~over the actual measurement bandwidth.~~ Equalization may be used to meet this requirement. Typical targets can be the P.50 speech spectrum or the actual spectrum of the speech signal to be used. See Annex E.

In general, the test signals and analysis methods in this standard cover a frequency range from approximately 100 to 8500 Hz. This bandwidth shall be used for analog telephones.

For digital telephones, the codec will generally reduce the test signal bandwidth; however, no level compensation is required when speech or speech-like signals are used for receive measurements. For example, a narrow band codec will reduce the level of speech signals by less than 0.1 dB compared to the full bandwidth signal.

The exact frequency range also depends on the analysis method and the test signal (see G.6 and G.7).

~~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 ± 5 dB elsewhere. Equalization may be used to meet this requirement.~~

5.6.2 Electrical test level

The calibrated digital levels are -10 dBm₀, -20 dBm₀ and -35 dBm₀.

The calibrated analog levels are -14 dBV, -25 dBV, and -38 dBV closed circuit with generator load impedances of 900Ω resistive.

Refer to Annex F for details about the derivation of electrical test levels.

The standard test level for handsets and headsets tested as 4-wire devices is determined by the procedure for setting the default receive volume control adjustment in 8.3.2.

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

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

5.7 Mouth simulator

The mouth simulator ~~should be calibrated each day the system is in use. For best accuracy, it~~ shall be calibrated prior to measurements ~~on each new device~~. The measurement microphone used to calibrate the mouth simulator shall be calibrated first (see 5.3).

5.7.1 Acoustic test spectrum

The acoustic test spectrum is measured at the mouth reference point (MRP). ~~For sinusoidal test signals,~~ The spectrum shall meet its target within ± 0.5 dB ~~over the actual measurement bandwidth~~. Equalization may be used to meet this requirement. Typical targets can be the P.50 speech spectrum or the actual spectrum of the speech signal to be used. Annex E.

The bandwidth of acoustical test signals shall not be reduced when testing narrow band or any reduced bandwidth device.

The exact frequency range also depends on the analysis method and the test signal (see G.6 and G.7).

~~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 $\pm 3/5$ dB elsewhere. Equalization may be used to meet this requirement.~~

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.~~

The stimulus levels L_{MRP} are as follows:

- Loud: +5 dBPa
- Nominal: –5 dBPa
- Quiet: –15 dBPa

In this edition of the document, –5 dBPa is used as the nominal level. Previous editions of this document specified a nominal level of –4.7 dBPa. A nominal level of –4.7 dBPa is an acceptable alternative.

Please see Annex F.6 for additional information.

Before performing tests at levels above –5 dBPa, verify the linearity of the mouth simulator. Some compression of the acoustic output of the mouth is normal at high levels. It may be necessary to perform a separate calibration of the mouth at high levels.

When the mouth simulator is used for measuring distortion, the distortion produced by the mouth is included in the measured result, so it should be checked. Distortion from the mouth simulator depends on the type of test signal, the level, and the frequency or frequency range. Each case should be evaluated individually.

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

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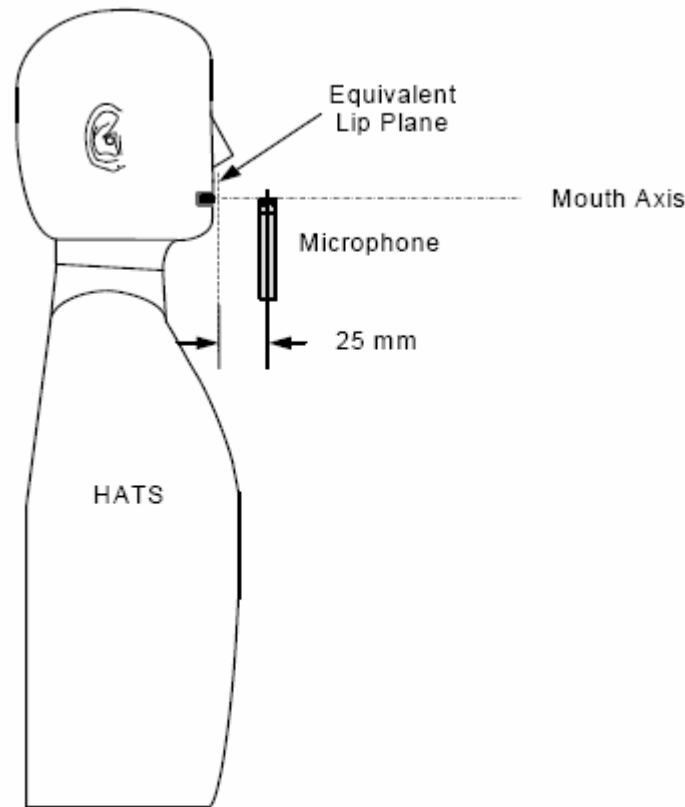


Figure 3— HATS mouth calibration

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 $G_{\text{MRP}}(f)$, 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 -5 dBPa. This spectrum is used to calculate the send, sidetone and overall frequency responses.

Calibration shall be verified for other levels and test signals.

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 (1/4 inch) 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

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

⁷ [Notes in text, tables, and figures of a standard are given for information only and do not contain requirements needed to implement this standard.](#)

6. Test procedure for analog sets

6.1 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. [New receive parameters have been included covering acoustic output level, conversational gain, and tonal balance.](#)

The telephone should be connected to the test circuit(s) described in 6.2. 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 5 shall be carried out before making any measurements. The acoustical test environment shall meet the specifications given in 4.5.

6.1.1 Choice of test signals and levels

The test signal should ensure that the telephone is characterized in realistic, stable, and well-defined states. This is especially the case for telephones with non-linear processes such as compression or voice activated switching (VOX) circuitry, etc.

[The recommended test signal is a specific recording of male speech, unless specified otherwise in the relevant clause. This speech recording is described in E.6.3.1, and a file containing the speech is an attachment to this standard.](#)

[Female speech is also provided as an alternate. It should be used to determine if the measured results are similar for both male and female speech. This speech recording is described in E.6.3.3, and a file containing the speech is an attachment to this standard.](#)

[P50-2 artificial voice may be an acceptable alternate for many applications, depending on the nature of signal processing in the device under test. The user should demonstrate that results using this signal are equivalent to those that would be obtained using male speech. P50-2 is described in E.6.1.2, and a file containing the speech is an attachment to this standard.](#)

[For electrical test levels from -14 dBV and greater, and for acoustical test levels of +5dBPa and greater, compressed test signals should be used to avoid clipping. The compressed versions of male speech, female speech, and P50-2 are found in E.6.1.3, E.6.3.2, and E.6.3.4](#)

[The standard measurement method is spectrum analysis using constant percentage band filters \(CPB\).](#)

[All other test signals are considered alternatives for special cases.](#) These alternative 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, such as distortion measurements, as discussed in relevant places within this standard. See Annex E and Annex F for further information on test signals and analysis methods.

~~See Annex F and Annex C 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.~~

6.1.2 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 and constant bandwidth analysis. The overall 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 and resolution used shall be stated.

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

6.1.3 Choice of ear and mouth simulators and test position

Choose the ear simulator, mouth simulator and test position according to 4.1, 4.2, and 4.3. This equipment shall be used for all tests described in Clause 6, unless otherwise specified. The ear simulator, mouth simulator, and test position used shall be stated.

6.1.4 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.

6.1.5 Reference receive volume control setting

For telephone systems with only one volume control, all measurements shall be done at the reference receive volume control setting (see 3.1). For analog phones, the reference receive volume control setting is determined at -25 dBV and with a 2.7 km loop.

A range of volume control settings may also be used where appropriate, such as minimum and maximum volume.

For compound systems with more than one volume control, such as a telephone plus headset system, find the reference volume control setting according to the following procedure:

a) Configure the telephone and headset system per manufacturer's specifications, or as required.

b) The test level shall be -25 dBV with a 2.7 km loop. The test signal shall be applied for at least

- c) Find a starting point on first system (telephone). Use reference volume control setting if possible, else use manufacturers power-up or return to default nominal if so equipped, else measure electrical receive signal level at handset/headset port to find mid point, else use physical midpoint, else make best possible guess.
- d) Set second system (headset system) to roughly midpoint. Measure RLR and adjust second system to obtain the reference volume control setting. If the reference setting can not be obtained, then re-adjust the first system volume control as necessary.
- e) Once the reference volume control setting is obtained, perform the remaining tests without adjusting the volume controls of either system. Record both the procedure and the settings, and ensure that they are repeatable.

A range of volume control settings may also be used where appropriate, such as minimum and maximum volume for both systems.

6.1.6 Reference send gain control setting

All measurements shall be done at the reference send volume control setting (see 3.1). A range of volume control settings may be used where appropriate, such as minimum and maximum volume. For analog phones, the reference send volume control setting is determined at -5 dBPa and with a 2.7 km loop.

6.2 Analog DC feed circuits

A general-purpose DC feed circuit is shown in Figure 4. 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:

$C \geq 100$ microfarads

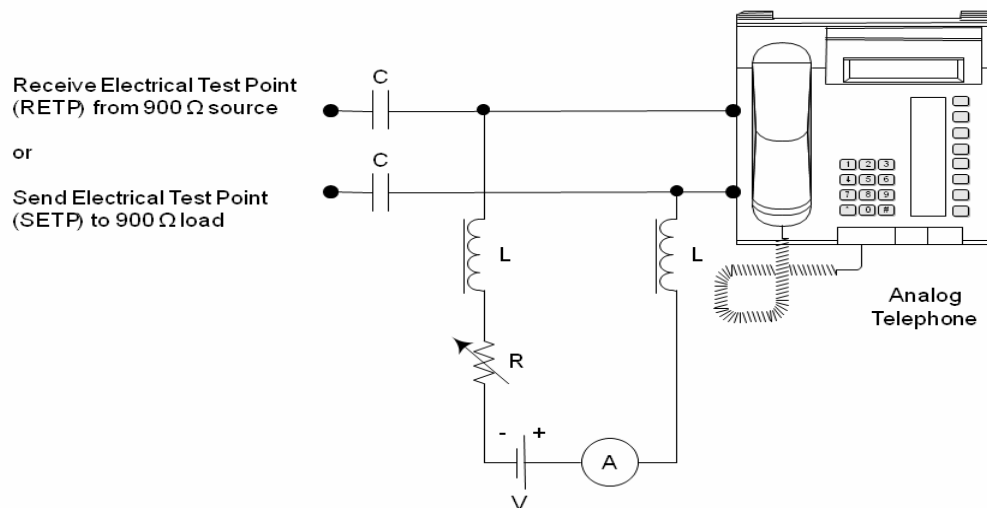
$L \geq 5$ (henries (each))

$R = 400 \Omega$, including resistance of inductors

$V = 50$ 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.

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 Ω load. The receive electrical test point (RETP) is for applying receive input signals. It shall be connected to a 900 Ω 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 of 100 Hz to 8500 Hz. The loss from 20 Hz to 100 Hz should not exceed 1 dB. The circuit of Figure 4, using ideal components, should just meet this specification.

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

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

$$\text{Feed circuit loss} = 20 \log \frac{\text{Voltage across } 900 \Omega \text{ resistor}}{\text{Voltage across feed bridge}} \quad (1)$$

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.

Feed circuit for overall measurements using two phones is shown in Figure 5.

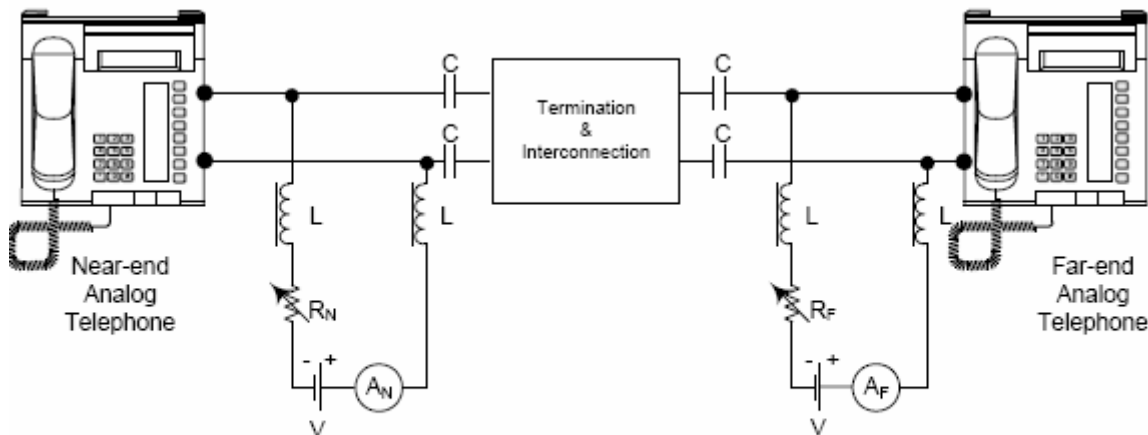


Figure 5— General purpose DC feed circuit for 2 analog telephones for overall measurements

6.3 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 6.3.1 through 6.3.6 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 6.4, 6.5, 6.6, and 6.7, but with the impairment introduced.

6.3.1 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.

6.3.2 Network noise

Network noise can affect non-linear 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.

6.3.3 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.

6.3.4 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 km, 2.7 km, and 4.6 km (0 kft, 9 kft, and 15 kft) of 26 AWG non-loaded cable. The recommended loop simulator circuit is shown in Figure 6 and components for various lengths are shown in Table 4.

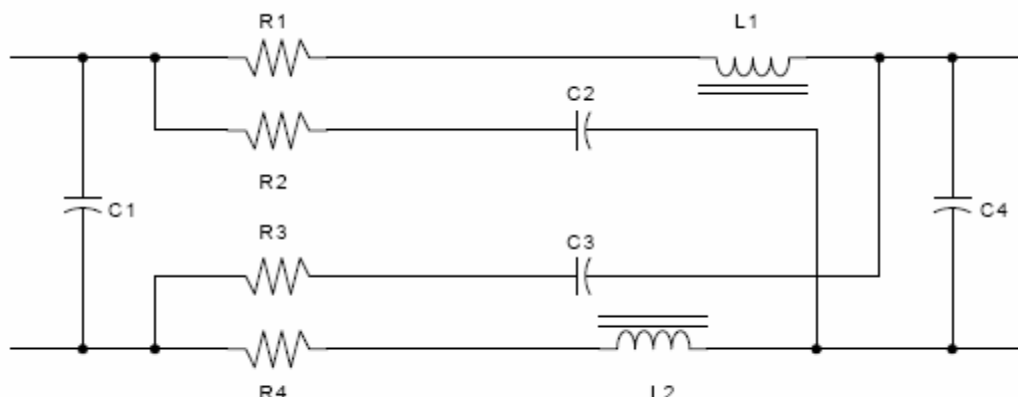


Figure 6—Loop simulator circuit

Table 4—Loop simulator circuit

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 μH	0.0226 μH
C ₂ , C ₃	4.02 nF	0.0122 μH	0.0255 μH
L ₁ , L ₂	96.0 μH	0.336 mH	0.983 mH
^a 0.305 km = 1 kft ^b 0.914 km = 3 kft ^c 1.83 km = 6kft ^d 2.7 km = 9 kft 4.6 km =15 kft			
NOTE 1— All values are ± 1 %.			
NOTE 2—2.7 km and 4.6 km can be made up of cascaded sections of section of shorter loops. ^d			

6.3.5 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 7, ± 0.3 V over the current range of 0 to 100 mA. The return loss shall be greater than 10 dB with respect to 600 Ω from 200 Hz to 4000 Hz. Component values may be adjusted to meet these tolerances. One possible implementation of this is shown in Figure 8.

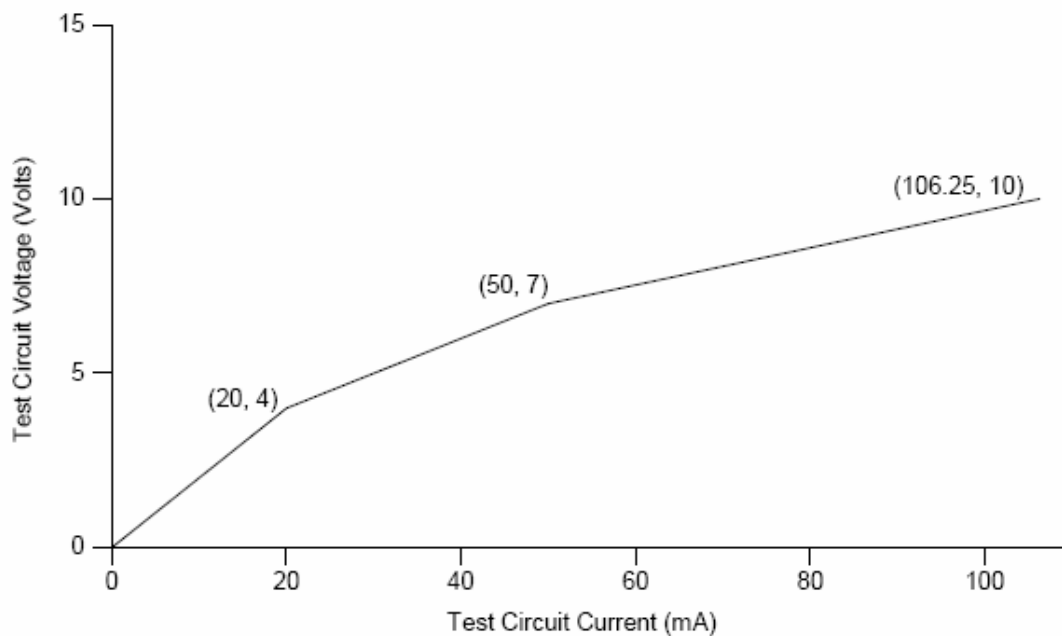
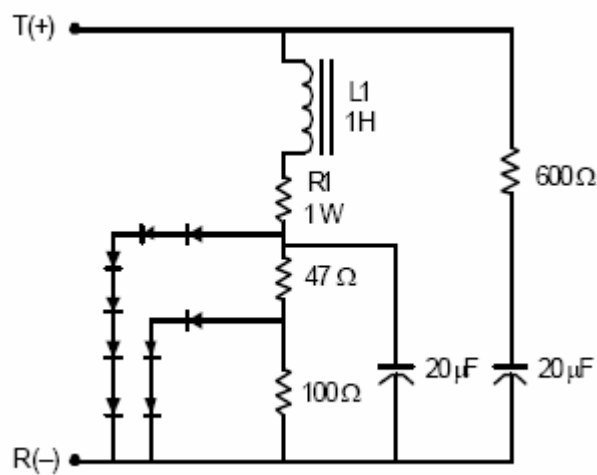


Figure 7— Parallel set test circuit VI curve



NOTE 1— Circuit is polarity sensitive.

NOTE 2— Use IN4004, or similar, for diode strings.

NOTE 3— L1 may be > 1 H.

NOTE 4— Resistor values shall be $\pm 1\%$.

NOTE 5— R1 + L1 coil resistance should be 53Ω .

NOTE 6— The performance of this circuit is dependent on the characteristics of the diodes used and temperature. Performance of the circuit must be verified and component values may need to be adjusted to meet the requirements.

Figure 8—Parallel telephone set simulator

6.3.6 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.

6.4 Receive

6.4.1 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 (2). $H_R(f)$ may be used to calculate the receive loudness rating (RLR) according to ITU-T Recommendation P.79. Please see Annex H.

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

where:

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

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

Receive frequency response shall be measured at -14 dBV with no loop, at -25 dBV with no loop, 2.7 km, and 4.6 km loops, and at -38 dBV with a 4.6 km loop. Measurements with other combinations of levels and impairments may be desirable. ~~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 C.1 for more information~~

6.4.2 Acoustic output level

Acoustic output level is a direct measure of the sound level at the ear of the listener. It should be measured over a range of line levels, and at both the reference and the maximum volume control settings.

Using the reference volume control setting, measure the acoustic output level at the loud, quiet and nominal signal levels. Repeat with the volume control set to maximum.

The test signal levels shall be -14 dBV with no loop, at -25 dBV with a 2.7 km loop, and at -38 dBV with a 4.6 km loop. (The RETP is ahead of the loop.) The test signal shall be male composite speech. It is also advisable to perform the test using the female composite speech.

To measure the acoustic output level, apply the test signal at the RETP. Measure the active total SPL, averaged over the entire length of the test signal. (Active speech level does not include pauses or silences. Please see Annex E.) The measurement shall be translated to the free field as specified in IEEE Std 1652. The measurement bandwidth shall be the same as the applied test signal, nominally 100 Hz to 8500 Hz. The level shall be unweighted.

6.4.3 Conversational gain

The conversational gain of a receiver expresses the difference in level between a face-to-face conversation at 1 meter and a conversation using a telephone. Use the method described in J.3.

Using the reference volume control setting, measure the conversational gain at the loud, quiet and nominal signal levels. Repeat with the volume control set to maximum.

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6.4.4 Tonal balance

Tonal balance is a set of three values which describe the general shape of the frequency response. Use the method described in J.4.

Measure the tonal balance at the nominal signal level, with the reference volume control setting. Repeat with the volume control set to maximum.

6.4.5 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 25 Hz to 8500 Hz, averaging over a minimum period of 5 seconds. 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 dB(A). 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 (see Annex I).

Receive noise shall be measured with 0 km, 2.7 km, and 4.6 km loops.

6.4.6 Receive narrow-band noise

Receive narrow-band 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 from 25 Hz to 8500 Hz. These levels can then be compared to the receive noise (see 6.4.5).

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 25 Hz to 8500 Hz, averaging over a minimum period of 5 s. If FFT analysis is used, then a "Flat Top" windowing shall be employed.

Receive narrow-band noise shall be measured with 0 km, 2.7 km, and 4.6 km loops.

6.4.7 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 6.4.1 and applying the procedures described in Annex K. Linearity shall be measured using the same test method and stimulus type used to measure frequency response.

The test shall be performed at 7 levels, from -45 to -15 dBV, in 5 dB intervals, measured in 1/3 octave bands. ~~If artificial voices or another wideband stimulus are used,~~ Smaller intervals and/or a wider range of levels may also be used. The reference stimulus level is -25 dBV. These levels take into account the high crest factor of ~~artificial voices, which approaches -23 dB~~ speech.

This test shall be performed with a 2.7 km loop, and may be performed at other loop lengths.

~~If sine wave signals are used, they shall be applied at the ISO R10 frequencies from 200 through 5000 Hz, at a level of -25 dBV, in 3 dB intervals. (For ISO R10, see 6.4.1.)~~

~~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.~~

6.4.8 Receive distortion

The standard distortion measurement method is pulsed noise signal-to-distortion-and-noise ratio (PN-SDNR). See L.3.

Receive distortion shall be measured at the ERP.

At a minimum, distortion shall be measured at -15 dBV at 400 Hz, 800 Hz, 1600 Hz, and 2500 Hz. At 800 Hz, it shall also be measured at levels from -40 to -15 dBV in 5 dB steps.

At a minimum, this test shall be performed with 0 km and 4.6 km loops.

Verify that distortion of the test system is sufficiently low.

See Annex L for alternative distortion measurement methods and test signals, and the conditions under which they may be used. Different distortion measurement methods are likely to give different results.

~~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 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.~~

6.4.9 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, and then apply the test signal. Measure the receive noise according to 6.4.5, but ensure that at least one complete period of the test signal is measured. The test signal shall be the same as that used for receive frequency response (see 6.4.1), at -15 dBV at 0 km loop. 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 won't be any control over the level. Each result is expressed in dB(A), the weighted noise level which should be compared to muted receive noise measured according to 6.4.(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.~~

6.5 Send

6.5.1 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 (3). The send frequency response, $H_S(f)$ may be used to calculate the send loudness rating (SLR) according to ITU-T Recommendation P.79. (see Annex H).

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

where:

$G_{SETP}(f)$ is the rms spectrum at SETP

$G_{MRP}(f)$ is the rms spectrum at MRP

Send frequency response shall be measured at +5 dBPa with no loop, at -5 dBPa with no loop, 2.7 km, and 4.6 km loops, and at -15 dBPa with a 4.6 km loop. Measurements with other combinations of levels and impairments may be desirable. 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 C.1 for more information.

6.5.2 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), 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 25 Hz to 6000 Hz, while A-weighted measurements are made from 25 Hz 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 (see Annex I).

Send noise shall be measured with 0 km, 2.7 km and 4.6 km loops.

6.5.3 Send narrow-band noise

Send narrow-band 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 25 Hz to 6000 Hz. These levels can then be compared to the send noise (see 6.5.2).

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 25 Hz to 6000 Hz, averaging over a minimum period of 5 s. 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 25 Hz to 8500 Hz.

Send narrow-band noise shall be measured with 0 km, 2.7 km, and 4.6 km loops.

6.5.4 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 6.5.1 and applying the~~

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procedures described in Annex K. 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 7 levels, from -25 to $+5$ 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 -5 dBPa. These levels take into account the high crest factor of ~~artificial voices, which approaches 23 dB~~ [speech](#).

[This test shall be performed with a 2.7 km loop, and may be performed at other loop lengths.](#)
[Before performing tests at levels above \$-5\$ dBPa, verify the linearity of the mouth simulator. Some compression of the acoustic output of the mouth is normal at high levels. It may be necessary to perform a separate calibration of the mouth at high levels.](#)

~~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~~

6.5.5 Send distortion

The standard distortion measurement method is ~~send-pulsed noise~~ signal-to-distortion-and-noise ratio (SDN), ~~measured using narrowband pseudo-random noise as the stimulus.~~ [PN-SDNR](#). See L.3.

~~Send distortion shall be 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.~~

[At a minimum, distortion shall be measured at \$+5\$ dBPa at 400 Hz, 800 Hz, 1600 Hz, and 2500 Hz. At 800 Hz, it shall also be measured at levels from \$-20\$ to \$+5\$ dBPa in 5 dB steps.](#)

[At a minimum, this test shall be performed with 0 km and 4.6 km loops.](#)

[For higher test levels, verify that distortion of the mouth simulator is sufficiently low.](#)

~~For information about THD and other-~~ [See Annex L for alternative](#) distortion measurement methods and test signals, and the conditions under which they may be used. Different distortion measurement methods are likely to give different results.

6.5.6 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, and then apply the test signal. Measure the send noise according to 6.5.2, [but ensure that at least one complete period of the test signal is measured](#). The test signal shall be the same as that used for send frequency response (see 6.5.1), at $+5$ dBPa [with 0 km loop](#).

The result is expressed in dBmp, a weighted noise level which should be compared to muted broad-band noise measured according to 6.5.2 (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, as prescribed in dBmp as a function of frequency. The weighting permits more relevant comparison with results obtained with artificial voices.~~

6.5.7 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 4.5.3), which is expressed in decibels. The diffuse field send frequency response in dB, $H_{SD}(f)$, is given by Equation (4).

The diffuse field send frequency response may be sensitive to both the level and type of signal used. The test level is typically in the range of 60 dBSPL to 80 dBSPL. This noise signal may be Hoth noise (Annex D) or any other noise signal representative of actual working conditions. This measurement may be performed in 1/3 octave resolution.

For measurement of send frequency response in diffuse field, the handset or headset is mounted on the HATS (or alternate fixture, if applicable). The MRP is positioned at the DFTP, but the mouth is not active for this measurement. The mouth simulator is not present during calibration.

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

Where:

$G_{SETP}(f)$ is the rms spectrum at SETP

$G_{DFTP}(f)$ is the rms spectrum at DFTP

~~The cross spectrum method is not recommended.~~

6.5.8 Send signal-to-noise ratio

Send signal-to-noise ratio is a measure of the desired speech transmission relative to unwanted noise in the room where the talker's phone is used. See Annex M.

6.6 Sidetone

Sidetone should be measured at minimum, reference, and maximum volume settings.

6.6.1 Talker sidetone frequency response

Talker sidetone frequency response is the ratio of the sound pressure measured in the ear simulator, referred to the Ear Reference Point (ERP), to the sound pressure at the Mouth Reference Point (MRP), which is expressed in decibels. The talker sidetone frequency response in dB, $H_{TS}(f)$, is given by Equation (5). Talker sidetone frequency response may be used to calculate the sidetone masking rating (STMR) according to ITU-T Recommendation P.79. Please see Annex H.

The STMR measured on an open-ear HATS is approximately 24 dB. This represents the effective floor of STMR measurements on actual telephones.

$$H_{TS}(f) = 20 \log \frac{G_{ERP}(f)}{G_{MRP}(f)} \text{ in } dBPa / Pa$$

Where

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

$G_{MRP}(f)$ is the rms spectrum at MRP