



IEEE Standard Method for Measuring Transmission Performance of Speakerphones

IEEE Communications Society

Sponsored by the
Transmission, Access, and Optical Systems Committee

1329TM

IEEE
3 Park Avenue
New York, NY 10016-5997, USA

15 October 2010

IEEE Std 1329TM-2010
(Revision of
IEEE Std 1329-1999)

IEEE Standard Method for Measuring Transmission Performance of Speakerphones

Sponsor

**Transmission, Access, and Optical Systems Committee
of the
IEEE Communications Society**

Approved 17 June 2010

IEEE-SA Standards Board

Approved 4 January 2011

American National Standards Institute

Abstract: This standard provides techniques for objective measurement of electroacoustic and voice-switching characteristics of speakerphones that connect directly or indirectly to an analog or digital telephone network. Due to the various characteristics of speakerphones and the environments in which they operate, not all of the test procedures in this standard are applicable to all speakerphones. Application of the test procedures to atypical speakerphones should be determined on an individual basis.

Keywords: full duplex, half duplex, handsfree, speakerphone, voice switching

The Institute of Electrical and Electronics Engineers, Inc.
3 Park Avenue, New York, NY 10016-5997, USA

Copyright © 2010 by the Institute of Electrical and Electronics Engineers, Inc.
All rights reserved. Published 15 October 2010. Printed in the United States of America.

IEEE is a registered trademark in the U.S. Patent & Trademark Office, owned by the Institute of Electrical and Electronics Engineers, Incorporated.

PDF: ISBN 978-0-7381-6353-6 STD96083
Print: ISBN 978-0-7381-6354-3 STDPD96083

IEEE prohibits discrimination, harassment and bullying. For more information, visit <http://www.ieee.org/web/aboutus/whatis/policies/p9-26.html>. No part of this publication may be reproduced in any form, in an electronic retrieval system or otherwise, without the prior written permission of the publisher.

IEEE Standards documents are developed within the IEEE Societies and the Standards Coordinating Committees of the IEEE Standards Association (IEEE-SA) Standards Board. The IEEE develops its standards through a consensus development process, approved by the American National Standards Institute, which brings together volunteers representing varied viewpoints and interests to achieve the final product. Volunteers are not necessarily members of the Institute and serve without compensation. While the IEEE administers the process and establishes rules to promote fairness in the consensus development process, the IEEE does not independently evaluate, test, or verify the accuracy of any of the information or the soundness of any judgments contained in its standards.

Use of an IEEE Standard is wholly voluntary. The IEEE disclaims liability for any personal injury, property or other damage, of any nature whatsoever, whether special, indirect, consequential, or compensatory, directly or indirectly resulting from the publication, use of, or reliance upon this, or any other IEEE Standard document.

The IEEE does not warrant or represent the accuracy or content of the material contained herein, and expressly disclaims any express or implied warranty, including any implied warranty of merchantability or fitness for a specific purpose, or that the use of the material contained herein is free from patent infringement. IEEE Standards documents are supplied “**AS IS**.”

The existence of an IEEE Standard does not imply that there are no other ways to produce, test, measure, purchase, market, or provide other goods and services related to the scope of the IEEE Standard. Furthermore, the viewpoint expressed at the time a standard is approved and issued is subject to change brought about through developments in the state of the art and comments received from users of the standard. Every IEEE Standard is subjected to review at least every five years for revision or reaffirmation, or every ten years for stabilization. When a document is more than five years old and has not been reaffirmed, or more than ten years old and has not been stabilized, it is reasonable to conclude that its contents, although still of some value, do not wholly reflect the present state of the art. Users are cautioned to check to determine that they have the latest edition of any IEEE Standard.

In publishing and making this document available, the IEEE is not suggesting or rendering professional or other services for, or on behalf of, any person or entity. Nor is the IEEE undertaking to perform any duty owed by any other person or entity to another. Any person utilizing this, and any other IEEE Standards document, should rely upon his or her independent judgment in the exercise of reasonable care in any given circumstances or, as appropriate, seek the advice of a competent professional in determining the appropriateness of a given IEEE standard.

Interpretations: Occasionally questions may arise regarding the meaning of portions of standards as they relate to specific applications. When the need for interpretations is brought to the attention of IEEE, the Institute will initiate action to prepare appropriate responses. Since IEEE Standards represent a consensus of concerned interests, it is important to ensure that any interpretation has also received the concurrence of a balance of interests. For this reason, IEEE and the members of its societies and Standards Coordinating Committees are not able to provide an instant response to interpretation requests except in those cases where the matter has previously received formal consideration. A statement, written or oral, that is not processed in accordance with the IEEE-SA Standards Board Operations Manual shall not be considered the official position of IEEE or any of its committees and shall not be considered to be, nor be relied upon as, a formal interpretation of the IEEE. At lectures, symposia, seminars, or educational courses, an individual presenting information on IEEE standards shall make it clear that his or her views should be considered the personal views of that individual rather than the formal position, explanation, or interpretation of the IEEE.

Comments for revision of IEEE Standards are welcome from any interested party, regardless of membership affiliation with IEEE. Suggestions for changes in documents should be in the form of a proposed change of text, together with appropriate supporting comments. Recommendations to change the status of a stabilized standard should include a rationale as to why a revision or withdrawal is required. Comments and recommendations on standards, and requests for interpretations should be addressed to:

Secretary, IEEE-SA Standards Board
445 Hoes Lane
Piscataway, NJ 08854
USA

Authorization to photocopy portions of any individual standard for internal or personal use is granted by The Institute of Electrical and Electronics Engineers, Inc., provided that the appropriate fee is paid to Copyright Clearance Center. To arrange for payment of licensing fee, please contact Copyright Clearance Center, Customer Service, 222 Rosewood Drive, Danvers, MA 01923 USA; +1 978 750 8400. Permission to photocopy portions of any individual standard for educational classroom use can also be obtained through the Copyright Clearance Center.

Introduction

This introduction is not part of IEEE Std 1329-2010, IEEE Standard Method for Measuring Transmission Performance of Speakerphones.

This standard has been developed in response to a widely expressed need by the telecommunications industry for a standard, comprehensive method for testing the transmission performance of speakerphone telephone sets. This standard is generally in agreement with the International Telecommunication Union—Telecommunication Standardization Sector (ITU-T) test arrangements and calibration procedures.

The IEEE will maintain this standard current with the state of the technology. Comments on this standard and suggestions for the additional material that should be included are invited. Comments should be addressed to the Secretary, IEEE Standards Board, The Institute of Electrical and Electronics Engineers, Inc., 445 Hoes Lane, Piscataway, NJ 08854.

This standard began in 1992 and was prepared by the Subcommittee on Telephone Instrument Testing of the Transmission, Access, and Optical Systems Committee of the IEEE Communications Society (formerly the IEEE Communications Technology Group). It was originally published in 1999 and has been updated in this 2010 edition to reflect current telecommunications technology.

Notice to users

Laws and regulations

Users of these documents should consult all applicable laws and regulations. Compliance with the provisions of this standard does not imply compliance to any applicable regulatory requirements. Implementers of the standard are responsible for observing or referring to the applicable regulatory requirements. IEEE does not, by the publication of its standards, intend to urge action that is not in compliance with applicable laws, and these documents may not be construed as doing so.

Copyrights

This document is copyrighted by the IEEE. It is made available for a wide variety of both public and private uses. These include both use, by reference, in laws and regulations, and use in private self-regulation, standardization, and the promotion of engineering practices and methods. By making this document available for use and adoption by public authorities and private users, the IEEE does not waive any rights in copyright to this document.

Updating of IEEE documents

Users of IEEE standards should be aware that these documents may be superseded at any time by the issuance of new editions or may be amended from time to time through the issuance of amendments, corrigenda, or errata. An official IEEE document at any point in time consists of the current edition of the document together with any amendments, corrigenda, or errata then in effect. In order to determine whether a given document is the current edition and whether it has been amended through the issuance of amendments, corrigenda, or errata, visit the IEEE Standards Association web site at <http://ieeexplore.ieee.org/xpl/standards.jsp>, or contact the IEEE at the address listed previously.

For more information about the IEEE Standards Association or the IEEE standards development process, visit the IEEE-SA web site at <http://standards.ieee.org>.

Errata

Errata, if any, for this and all other standards can be accessed at the following URL: <http://standards.ieee.org/reading/ieee/updates/errata/index.html>. Users are encouraged to check this URL for errata periodically.

Interpretations

Current interpretations can be accessed at the following URL: <http://standards.ieee.org/reading/ieee/interp/index.html>.

Patents

Attention is called to the possibility that implementation of this standard method may require use of subject matter covered by patent rights. By publication of this standard method, no position is taken with respect to the existence or validity of any patent rights in connection therewith. The IEEE shall not be responsible for identifying patents or patent applications for which a license may be required to implement an IEEE standard or for conducting inquiries into the legal validity or scope of those patents that are brought to its attention.

Participants

At the time this standard method was submitted to the IEEE-SA Standards Board for approval, the Telephone Instrument Testing Working Group had the following membership:

John R. Bareham, *Chair*
Glenn Hess, *Vice-chair*

Al Baum
Cliff Chamney
Miguel Dearaujo
Dan Foley
Steve Graham

Fred Kruger
Ron D. Magnuson
Scott Pennock
Tony Scott

Hong Sodoma
Christopher J. Struck
Stephen R. Whitesell
Allen Woo
Bob Young

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

John R. Bareham
Hugh Barrass
Al Baum
H. Stephen Berger
Cliff Chamney
Keith Chow
Miguel De Araujo
Carlo Donati

Sourav Dutta
Steve Graham
Randall Groves
Glenn Hess
Werner Hoelzl
Akio Iso
Piotr Karocki
Fred Kruger

G. Luri
Ron D. Magnuson
Gary Michel
Michael Newman
Gary Nissen
Scott Pennock
Venkatesha Prasad
Bartien Sayogo

Jeffrey Schenck
Tony Scott
Peretz Shekalim

Gil Shultz
Christopher J. Struck
Mitsutoshi Sugawara

Stephen R. Whitesell
Allen Woo
Oren Yuen

When the IEEE-SA Standards Board approved this method on 17 June 2010, it had the following membership:

Robert M. Grow, *Chair*
Richard H. Hulett, *Vice Chair*
Steve M. Mills, *Past Chair*
Judith Gorman, *Secretary*

Karen Bartleson
Victor Berman
Ted Burse
Clint Chaplin
Andy Drozd
Alexander Gelman
Jim Hughes

Young Kyun Kim
Joseph L. Koepfinger*
John Kulick
David J. Law
Hung Ling
Oleg Logvinov
Ted Olsen

Ronald C. Petersen
Thomas Prevost
Jon Walter Rosdahl
Sam Sciacca
Mike Seavey
Curtis Siller
Don Wright

*Member Emeritus

Also included are the following nonvoting IEEE-SA Standards Board liaisons:

Satish Aggarwal, *NRC Representative*
Richard DeBlasio, *DOE Representative*
Michael Janezic, *NIST Representative*

Catherine Berger
IEEE Standards Program Manager, Document Development

Matthew Ceglia
IEEE Standards Program Manager, Technical Program Development

Contents

1. Overview	1
1.1 General	1
1.2 Scope	1
1.3 Purpose	2
1.4 Contents of standard	2
2. Normative references	2
3. Definitions	3
4. Abbreviations, acronyms, and symbols	8
5. Test equipment, environment, and impairments	11
5.1 General	11
5.2 Test equipment	11
5.3 Test environment	12
5.4 Impairments	16
6. Test calibration	17
6.1 Electrical test signal calibration	17
7. Transmission measurements	18
7.1 General	18
7.2 Choice of measurement transducers	20
7.3 Positioning of speakerphone and test transducers	21
7.4 Test position guidelines	22
7.5 Receive	27
7.6 Send	34
7.7 Echo path response	39
8. Voice-switching time (half-duplex) measurements	42
8.1 General	42
8.2 Classification	43
8.3 Switching parameters	43
8.4 Test conditions	44
8.5 Test parameters	45
9. Acoustic echo control (full- or half-duplex) measurements	52
9.1 General	52
9.2 Test signals	53
9.3 Test conditions	53
9.4 Round-trip EPD	53
9.5 TCL—single-talk	54
9.6 Convergence time (T_C)	54
9.7 Temporally weighted terminal coupling loss during double-talk (TCL_{TDT})	55
9.8 Receive speech attenuation during double-talk (A_{RDT})	55
9.9 Send speech attenuation during double-talk (A_{SDT})	56
9.10 Receive speech front-end clipping time during double-talk (T_{RFDT})	57

9.11 Send speech front-end clipping time during double-talk (T_{SFDT}).....	57
Annex A (normative) Recommended test bed.....	58
A.1 Example implemented test hybrid	58
A.2 Example adaptive echo canceller.....	58
Annex B (normative) Detailed test methodology for TCL_T	64
B.1 General.....	64
B.2 Temporally weighted terminal coupling loss (TCL_T)	64
B.3 Modeling echo audibility	65
B.4 Expressing TCL results.....	66
Annex C (normative) TCL_T test algorithm.....	68
C.1 General.....	68
C.2 TCL_T algorithm.....	69
Annex D (normative) Double-talk testing	72
D.1 General.....	72
D.2 Cancellor training prior to double-talk.....	72
D.3 Double-talk source signals.....	74
D.4 Double-talk attenuation testing	76
D.5 Double-talk front-end syllabic clipping testing	78
D.6 Double-talk terminal coupling testing	79
D.7 Double-talk measurement filters.....	80
Annex E (informative) Microphones.....	82
Annex F (normative) Power and amplitude summations	83
F.1 Power summation of a spectrum	83
F.2 Adding spectra on a “power” basis.....	83
F.3 Averaging spectra on a “power” basis.....	84
F.4 Adding spectra on an “amplitude” basis.....	85
Annex G (normative) Preferred listening levels.....	87
G.1 Objectives	87
G.2 Apparatus.....	87
G.3 Procedure.....	87
G.4 Comments about results.....	88
G.5 Acceptance of speech levels for three levels of room noise	88
Annex H (normative) Multiple send and receive frequency response test positions.....	90
Annex I (informative) Bibliography.....	91

IEEE Standard Method for Measuring Transmission Performance of Speakerphones

IMPORTANT NOTICE: This standard is not intended to ensure safety, security, health, or environmental protection. Implementers of the standard are responsible for determining appropriate safety, security, environmental, and health practices or regulatory requirements.

This IEEE document is made available for use subject to important notices and legal disclaimers. These notices and disclaimers appear in all publications containing this document and may be found under the heading “Important Notice” or “Important Notices and Disclaimers Concerning IEEE Documents.” They can also be obtained on request from IEEE or viewed at <http://standards.ieee.org/IPR/disclaimers.html>.

1. Overview

1.1 General

Objective or subjective methods can be used to measure speakerphone transmission performance. This standard discusses objective procedures utilizing a sound source, laboratory microphone, and test instruments to characterize transmission performance. Subjective methods are particularly applicable for rating overall communication connections involving the real voice and real ear of human subjects. Speakerphones can be evaluated by purely objective methods provided the methods agree with the desirable performance characteristics of subjective testing.

It is a complex problem to obtain accurate and repeatable speakerphone test results. To meet these measurement goals, the reader should be aware of the nonlinear and time-variant characteristics of speakerphones. This standard describes test signals and corresponding analysis methods that can be chosen to place the speakerphone in a well-defined operating state during testing.

Execution of this standard provides a means of determining the operational characteristics of a speakerphone in conditions encountered during normal operation.

1.2 Scope

This standard provides techniques for objective measurement of electroacoustic and voice-switching characteristics of speakerphones that connect directly or indirectly to an analog or digital telephone network. Due to the various characteristics of speakerphones and the environments in which they operate,

not all of the test procedures in this standard are applicable to all speakerphones. Application of the test procedures to atypical speakerphones should be determined on an individual basis.

1.3 Purpose

The purpose of this standard is to provide practical methods for making laboratory measurements of the transmission and voice-switching characteristics of speakerphones so that their performance may be evaluated on a standardized basis.

1.4 Contents of standard

This subclause offers a summary of the clauses contained in the standard. The primary measurement procedures appear in Clause 6 through Clause 9. Annexes contain additional information or details of procedures referred to by relevant clauses.

Clause 2, Clause 3, and Clause 4 provide references, definitions, and abbreviations useful in executing the tests of this standard. These clauses provide background in the terminology used for speakerphone testing.

Clause 5 details test equipment, test environment, and impairments. Test equipment includes interfaces and transducers. A test environment includes both the acoustical and physical characteristics of the test space. Impairments include both acoustic and network conditions.

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

Clause 7 details send, receive, and other transmission test procedures. This clause includes procedures for positioning the speakerphone for testing.

Clause 8 details procedures and parameters involved in testing and classifying the voice-switching characteristics of speakerphones. Various environmental conditions are explained in reference to the operation of a voice-switched speakerphone.

Clause 9 details the procedures and parameters unique to testing a speakerphone equipped with an acoustic echo canceller (AEC). Characteristics unique to AEC speakerphones are presented. Also, environment and interface conditions specific to AECs are described.

2. Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments or corrigenda) applies.

ANSI S1.1, American National Standard Acoustical Terminology.¹

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

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

IEC 61094-4, Measurement microphones – Part 4: Specifications for working standard microphones.²

IEEE Std 269™, IEEE Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets, and Headsets.³

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 parts of international connections, 1993.⁴

ITU-T Recommendation O.41, Psophometer for use on telephone-type circuits.

ITU-T Recommendation P.50, Artificial voices.

ITU-T Recommendation P.51, Artificial mouth.

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 telephony.

ITU-T Recommendation P.59, Artificial conversational speech.

ITU-T Recommendation P.79, Calculation of loudness ratings for telephone sets.

ITU-T Recommendation P.340, Transmission characteristics and speech quality parameters of hands-free terminals.

3. 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.⁵

acoustic echo canceller (AEC): A circuit or algorithm designed to eliminate acoustic echoes and prevent howling due to acoustic feedback from loudspeaker to microphone.

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

acoustic output: The sound pressure level developed at the measuring microphone. *See also:* **sound pressure level**.

active speech level: Long-term average speech level that does not include pauses or silences. It is the time integral of the instantaneous power over the active time, divided by active time, in decibels relative to the appropriate reference.

² IEC publications are available from the Sales Department of the International Electrotechnical Commission, Case Postale 131, 3, rue de Varembe, CH-1211, Genève 20, Switzerland/Suisse (<http://www.iec.ch/>). IEC publications are also available in the United States from the Sales Department, American National Standards Institute, 25 West 43rd Street, 4th Floor, New York, NY 10036, USA.

³ IEEE publications are available from the Institute of Electrical and Electronics Engineers, 445 Hoes Lane, Piscataway, NJ 08854, USA (<http://standards.ieee.org/>).

⁴ ITU-T (formerly CCITT) publications are available from the ITU-T General Secretariat, International Telecommunications Union, Sales Section, Place des Nations, CH-1211, Genève 20, Switzerland/Suisse (<http://www.itu.int/ITU-T/>). ITU-T publications are also available in the United States from National Technical Information Service, Department of Commerce, 5285 Port Royal Road, Springfield, VA 22161 USA (<http://>).

⁵ *The IEEE Standards Dictionary: Glossary of Terms & Definitions* is available at <http://shop.ieee.org/>.

NOTE—See ITU-T Recommendation P.56⁶ for details.⁷

active time: Time during which the speech in question is present, excluding pauses or noises below a defined threshold.

alternate test position (ATP): A test position that reflects alternative ways of using the speakerphone.

attenuation range (a_H): The difference in level in dB between maximum inserted switched loss and the full removal of that switched loss. If the send to receive is not the same as receive to send, then the larger of the two is the attenuation range.

automatic gain control (AGC): A circuit or algorithm that varies gain as a function of the input signal amplitude.

A-weighted: A measurement made using the “A” frequency weighting. A-weighting approximates the frequency response of the ear at 40 dB SPL; therefore, 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.

NOTE—The “A” frequency weighting is specified in ANSI S1.4

binaural: Listening with two ears.

build-up time (T_R): Time from the input signal going above the threshold level until the time at which the output level reaches 3 dB below the complete removal of the insertion loss. *Syn:* **attack time; rise time.**

C_{50} : The ratio of energy before and after 50 ms, expressed as a difference in dB.

convergence time (T_C): The time required for echo to reach within 3 dB of maximum echo return loss (ERL), or 25 dB loss, whichever occurs first.

dB(A): Sound pressure in decibels relative to 20 μ Pa, A-weighted. The proper term is dB SPL(A).

NOTE—To match common usage, dB(A) in this standard. Readers may see dBA in other documents.

derived recommended test position (DRTP): A test position that reflects intended use, derived from the user’s guide.

double-talk: Two talkers speaking simultaneously in opposite transmission directions.

drum reference point (DRP): A point located at the end of the ear canal, corresponding to the eardrum position.

ear reference point (ERP): 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.

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

NOTE—See ITU-T Recommendation P.57.

⁶ Information on references can be found in Clause 2.

⁷ Notes in text, tables, and figures are given for information only and do not contain requirements needed to implement this standard.

early decay time (EDT): Reverberation time based on the first 10 dB of sound decay in a room.

echo path delay (EPD): The total delay of the echo path from the receive electrical test point (RETP) to the send electrical test point (SETP), excluding any delay in the test equipment.

echo path response: The output at the send electrical test point (SETP) due to an input at the receive electrical test point (RETP). It is a measure of acoustic, vibration, and electrical coupling from the receive circuit to the send circuit.

estimated real use position (ERUP): A test position that reflects intended use, defined by the test laboratory.

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

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

full duplex: An operating condition that allows simultaneous communication in both send and receive directions with 3 dB or less switched loss in either direction.

NOTE—Classified as Type 1 in ITU-T Recommendation P.340.

group audio terminal (GAT): A speakerphone designed to be used by several users at the same time.

half duplex: An operating condition that allows communication in either send or receive directions with more than 20 dB switched loss in the opposite direction.

NOTE—Classified as Type 3 in ITU-T Recommendation P.340.

handsfree reference point (HFRP): The calibration point on the reference axis of the mouth simulator, 50 cm in front of the lip plane.

hangover time (T_H): Time from the input signal going below the threshold level until 3 dB of switched loss is inserted in the output signal. *Syn:* **decay time**; **release time**.

head and torso simulator (HATS): 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.

NOTE— See ITU-T Recommendation P.58.

loudspeaking telephone: A telephone with speakerphone receive function, but not speakerphone send capability.

monaural: Listening with one ear.

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.

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

NOTE—See ITU-T Recommendation P.51.

near-field test point (NFTP): The acoustical measurement point located approximately 1 cm from the center of the speakerphone's loudspeaker along the axis of the speaker.

noise discriminator/noise guard/noise monitor: A circuit or algorithm intended to discriminate between speech and noise. It can affect switching, transmission, and/or noise performance.

nonlinear devices: Nonlinear processing including automatic gain control (AGC) circuits, noise guards, etc.

open listening: A mode of telephone communication in which a telephone handset is used in the normal position for send. The incoming signal is received simultaneously by the handset and loudspeaker.

partial duplex: An operating condition that allows simultaneous communication in both send and receive directions with 3 dB to 20 dB switched loss in either direction.

NOTE—Classified as Type 2 in ITU-T Recommendation P.340.

receive attenuation during double-talk (A_{RDT}): Attenuation in the receive path, seen at the test position, inserted during double-talk. The send talker initiates the double-talk.

receive electrical test point (RETP): The point in a battery feed circuit, reference codec, or wireless reference base station at which signals are applied to the speakerphone in the receive direction.

NOTE—For further discussions on electrical interfaces, see IEEE Std 269. For further discussions on wireless interfaces, see appropriate wireless standards.

receive loudness rating directionality (RLRD): Receive loudness rating versus angle around the speakerphone, normalized to the loudness rating at the test position.

receive speech front-end clipping during double-talk (T_{RFDt}): The length of time that speech undergoes syllabic clipping, as seen at the (test position), just after the onset of double-talk. The receive talker initiates the double-talk.

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

recommended test position (RTP): A test position that reflects intended use, specified by the manufacturer.

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

NOTE—See IEEE Std 269 for details on a reference codec.

reference volume control setting: The volume control position resulting in a specified nominal receive loudness rating.

reverberation time (T_{60}): The time it takes for sound in a room to decay 60 dB from its initial, steady-state value.

send attenuation during double-talk (A_{SDt}): Attenuation in the send path, seen at the send electrical test point (SETP), inserted during double-talk. The receive talker initiates the double-talk.

send electrical test point (SETP): The point in a battery feed circuit, reference codec, or wireless reference base station at which signals coming from the speakerphone in the send direction are accessed.

NOTE—For further discussions on electrical interfaces, see IEEE Std 269. For further discussions on wireless interfaces, see the appropriate wireless standards.

send front-end syllabic clipping during double-talk (T_{SFDT}): The length of time that speech undergoes syllabic clipping, as seen at the send electrical test point (SETP), just after the onset of double-talk. The send talker initiates the double-talk.

send loudness rating directionality (SLRD): The send loudness rating versus angle around the speakerphone, normalized to the loudness rating at the test position.

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

single-talk: One talker speaking while the opposite transmission direction is silent.

sound pressure level: The level, in decibels, that 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 20 μPa , and sound pressure levels are expressed in dBSPL. Measurements such as receive noise and room noise shall be referenced to 20 μPa . Alternatively, sound levels from the mouth simulator and the speakerphone (except noise) may be expressed in dB re 1 Pa (dBP_a). Unless otherwise indicated, root-mean-square (rms) values of pressure are used. Some common relationships are 0 dBP_a = 94 dBSPL, 0 dBSPL = 20 μPa , and 1 Pa = 1 N/m².

speakerphone test circuit: An assembly consisting of one or more speakerphone(s) and interface(s) as may be required to realize simulated partial telephone connections.

speakerphone: A device for connection directly or indirectly to an analog or digital telephone network capable of two-way voice communication without close coupling to the user's mouth or ear. A loudspeaker is used as a telephone receiver. The microphone(s) may be embedded or external.

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 1/12 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.

switching time (T_s): Time taken to switch from one transmission direction to the other in alternating single-talk conversation on a speakerphone.

takeover time (T_T): Time taken to switch from one transmission direction to the other in double-talk conversation on a speakerphone. The signal in the first direction is continuously applied while the interrupting signal is applied in the opposite direction. T_T is measured from the application of the interrupting signal until the output level due to the interrupting signal reaches 3 dB below its final value.

temporally weighted terminal coupling loss (TCL_T): The terminal coupling loss (TCL), weighted in both time and frequency domains to account for subjective perception.

terminal coupling loss (TCL): The loss in the echo path from the receive electrical test point (RETP) to the send electrical test point (SETP). TCL can be weighted in time, frequency, or both. *Syn:* **echo return loss (ERL).**

test position: A general term referring to the point relative to the speakerphone where most acoustic measurements are made. It is intended to simulate the position of a real user.

NOTE—Particular test positions are specified in 7.3.

threshold level (L_{TH}): The minimum signal level necessary for removing insertion loss on a speakerphone.

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

weighted terminal coupling loss (TCL_w): Long-term, time-averaged terminal coupling loss (TCL), weighted in the frequency domain.

NOTE— See ITU-T Recommendation G.122.

4. Abbreviations, acronyms, and symbols

AEC	acoustic echo canceller
AGC	automatic gain control
ATP	alternate test position
CSS	composite source signal
DATP	derived alternate test position
DRTP	derived recommended test position
DRP	drum reference point
DTX	discontinuous speech transmission
DUT	device under test
EATP	estimated alternate test position
EDT	early decay time
EPD	echo path delay
ERL	echo return loss
ERP	ear reference point
ERUP	estimated real use position
FFT	fast Fourier transform
GAT	group audio terminal
HATS	head and torso simulator
MAF	mean audible field
MAP	mean audible pressure
MLS	maximum-length sequence
MRP	mouth reference point
NFTP	near-field test point
NLMS	normalized least mean square
NLP	nonlinear processor
RATP	recommended alternate test position
RETP	receive electrical test point
RLR	receive loudness rating
RLRD	receive loudness rating directionality

RTP	recommended test position
SAD	speech activity detector
SDNR	signal-to-distortion-and-noise ratio
SETP	send electrical test point
SFI	single-frequency interference
SLR	send loudness rating
SLRD	send loudness rating directionality
SNR	signal-to-noise ratio
SPL	sound pressure level
TCL	terminal coupling loss
TDS	time delay spectrometry
VAD	voice activity detector

The letters “A” and “a” are used for speech attenuation as follows:

- A = attenuation as a function of time (in dB)
- A_{RDT} = receive attenuation during double-talk (in dB)
- A_{RADT} = receive attenuation during double-talk, conversational average (in dB)
- A_{RTDT} = receive attenuation during double-talk, as a function of time (in dB)
- A_{SDT} = send attenuation during double-talk (in dB)
- A_{SADT} = send attenuation during double-talk, conversational average (in dB)
- A_{STDT} = send attenuation during double-talk, as a function of time (in dB)
- a_{H} = attenuation range of the speakerphone (in dB)

The letter “G” is used for spectra as shown below. This practice corresponds to common usage in spectrum analysis. The analysis bandwidth shall be specified.

- $G_{\text{TP}}(f)$ = spectrum at test position (in dBSPL or dBPa)
- $G_{\text{MRP}}(f)$ = spectrum at mouth reference point (in dBSPL or dBPa)
- $G_{\text{RETP}}(f)$ = spectrum at receive electrical test point (in dBV)
- $G_{\text{SETP}}(f)$ = spectrum at send electrical test point (in dBV)

The letter “H” is used for frequency response as follows:

- $H_{\text{S}}(f)$ = send frequency response (in dB V/Pa)
- $H_{\text{R}}(f)$ = receive frequency response (in dB Pa/V)
- $H_{\text{EP}}(f)$ = echo path frequency response (in dB V/V)
- $H'(f)$ = frequency response at ISO R10 preferred frequencies (in dB)

The letter “L” is used for levels measured over a wideband, with the bandwidth to be specified, as shown below. This practice corresponds to common usage in sound level measurements, as specified in ANSI S1.1.

- L_{ERP} = level at ear reference point (in dBSPL or dBPa)
- L_{TP} = level at the test position (in dBSPL or dBPa)
- L_{MRP} = level at mouth reference point (in dBSPL or dBPa)
- L_{RETP} = level at receive electrical test point (in dBV or dBm0)
- L_{ROOM} = level of background “impairment” noise in the measurement environment (in dB(A))
- L_{SETP} = level at send electrical test point (in dBV or dBm0)
- L_{TH} = threshold level required to remove a speakerphone’s insertion loss (in dBV or dBm0)
- L_{TH_R} = receive threshold level (in dBV or dBm0)
- L_{TH_S} = send threshold level (in dBSPL or dBPa)

The letter “S” is used for specially calculated sensitivity as follows:

- S_{DE} = sensitivity correction for HATS drum reference point (DRP) to ear reference point (ERP)
- $\bar{S}_R$ = midband average receive sensitivity (in dB Pa/V)
- $\hat{S}_{RO}$ = normalized midband average receive sensitivity (in dB SPL/V at 1 m)
- $\bar{S}_S$ = midband average send sensitivity (in dB V/Pa)

The letter “T” is used for time measurement as follows:

- T = length of time window in simulated free-field techniques (in seconds)
- T_C = convergence time of speakerphone AEC algorithm (in seconds)
- T_{FDT} = front-end syllabic clipping time during double-talk (in seconds)
- T_{SFDT} = send front-end syllabic clipping time during double-talk (in seconds)
- T_{RFDT} = receive front-end syllabic clipping time during double-talk (in seconds)
- T_H = hangover time (in seconds)
- T_R = build-up time (in seconds)
- T_S = switching time (in seconds)
- T_T = takeover time (in seconds)

The abbreviation “TCL” is used for terminal coupling loss as follows:

- TCL = terminal coupling loss (in dB)
- TCL_T = temporally weighted terminal coupling loss (in dB)
- TCL_{TDT} = temporally weighted terminal coupling loss during double-talk (in dB)
- TCL_{TST} = temporally weighted terminal coupling loss during single-talk (in dB)

- TCL_W = frequency weighted terminal coupling loss (in dB)
- TCL_{WST} = frequency weighted terminal coupling loss during single-talk (in dB)

5. Test equipment, environment, and impairments

5.1 General

The test equipment, environmental concerns, and impairments recommended to evaluate speakerphone performance are described here. See IEEE Std 269 for information concerning other test equipment needed for this standard. Different equipment will be required depending on whether analog or digital sets are being tested and which test signals and analysis methods are chosen. For wireline analog speakerphones, feed circuits and test loops are required. For digital or wireless speakerphones, interfaces and/or codecs are required.

5.2 Test equipment

5.2.1 Measuring microphones

All microphones used to implement this standard shall comply with the relevant specifications in IEC 61094-4.

Either a 12.5 mm (1/2 in) or 6.25 mm (1/4 in) free-field microphone may be used whenever a free-field microphone is referred to in this document. The noise floor of the microphone, expressed as equivalent noise level in dB(A), shall be at least 10 dB below the level to be measured.

5.2.2 Mouth simulator

The mouth simulator shall comply with the specifications given in ITU-T Recommendation P.51.

5.2.3 Head and torso simulator (HATS)

A HATS for measurements of speakerphones shall conform to ITU-T Recommendation P.58. The HATS shall be equipped with two ear simulators and a mouth simulator. The ear simulators shall conform to ITU-T Recommendation P.57. Electroacoustic measurements on speakerphones using a HATS will differ from measurements performed using a separate microphone and mouth simulator. The use of a HATS shall, therefore, be reported. The HATS configuration, neck position (if variable), the use of absorptive clothing, and any other relevant parameters shall also be noted.

5.2.4 Standard circuits for transmission and voice-switching measurements

The standard circuits for analog and digital speakerphone transmission and voice-switching measurements are the same as those used in IEEE Std 269 for telephones. For interfacing to wireless systems, see appropriate wireless standards.

5.2.5 Standard circuits for AEC measurements

A test hybrid is required to perform acoustic terminal coupling loss (TCL) measurements on two-wire analog speakerphones. The test hybrid is used to cancel or remove the speakerphone's hybrid reflection and thereby make a two-wire analog speakerphone behave as a virtual four-wire phone. In other words, only the acoustic portion of the echo path will remain and thus can be measured. There are a number of ways to implement a test hybrid: with an echo canceller with freezable coefficients, measurement post-processing, or a freezable adaptive analog filter bank.

The requirements for the hybrid are as follows:

- a) High impedance bridge mode for tapping off lines without affecting impedance.
- b) DC blocking on the input.
- c) Known one-way delay and round-trip echo path (in/out delay). Any delay in the digital hybrid or acquisition card will affect both the echo path model delay and the synchronization of data acquisition. The fixed echo path delay (EPD) inserted by the units shall be quantified.
- d) Noise and distortion performance 10 dB below echo detection thresholds. A full-scale signal-to-noise ratio (SNR) of 80 dB is preferred.
- e) High echo return loss (ERL) of 50 dB. An ERL of 60 dB or better is recommended.
- f) Ability to freeze the adaptive hybrid so that the canceller does not enhance return loss.

The test hybrid is to be trained with white noise. Fulfillment of the above requirements shall be tested at a nominal level of -16 dBV at RETP with a $900\ \Omega$ load on the hybrid. Measure the long-term average level on the input and output. The difference should be 60 dB and shall be at least 50 dB.

Available commercial hybrids may not meet the above criteria. However, one possible post-processing implementation of this technique, which does meet the above specifications, is given in Annex A.

5.3 Test environment

5.3.1 General

Electroacoustic measurements on speakerphones should be conducted in a test environment that will not affect the results beyond the intended influence of the test table and the measurement transducers. The test environment should have a low background noise level.

Be sure to record for the test report the test room environmental conditions of temperature, humidity, and barometric pressure, in addition to the background noise. (Overall A-weighted noise level and 1/3 octave band sound levels are recommended.)

5.3.2 Background noise level

The background noise level in the test environment shall not exceed the limits shown in Table 1. The overall A-weighted noise level shall not exceed 29 dB(A). The noise floor of the microphone used for this measurement shall have a noise floor, expressed as equivalent noise level in dB(A), of no more than 19 dB(A). A 25 mm (1 in) microphone may be required to meet this specification.

In principle, background noise in the test environment should be measured with a low noise, random incidence microphone. In practice, use of a free-field microphone is acceptable provided six measurements are taken with the microphone aimed in each direction along three mutually perpendicular axes (left/right, front/back, up/down). The maximum of the six measurements shall be used.

Table 1—Test room noise level limits

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

5.3.3 Anechoic chamber

An anechoic chamber should be large enough to comfortably accommodate the test table, measurement transducers, and speakerphone. Free-field conditions should exist throughout the frequency range of interest. Errors due to deviations from the inverse square law or 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.

Figure 1 shows the position of transducers relative to the location of the test table, but free-field verification is done without the test table in the chamber.

Verification of free-field conditions can be conducted as follows: Using a 12.5 mm free-field microphone positioned along the seven axes shown in Figure 1, measure at distances of 315 mm, 400 mm, 500 mm, 630 mm, 800 mm, and 1000 mm from the mouth simulator. The level should decrease 6 dB for each doubling of distance. Deviations represent departure from ideal free-field conditions. The mouth shall be placed at positions equivalent to B or C, as appropriate. When placed at point B, the artificial mouth axis shall be coincident with axis 5. When placed at point C, the artificial mouth axis shall be coincident with axis 7.

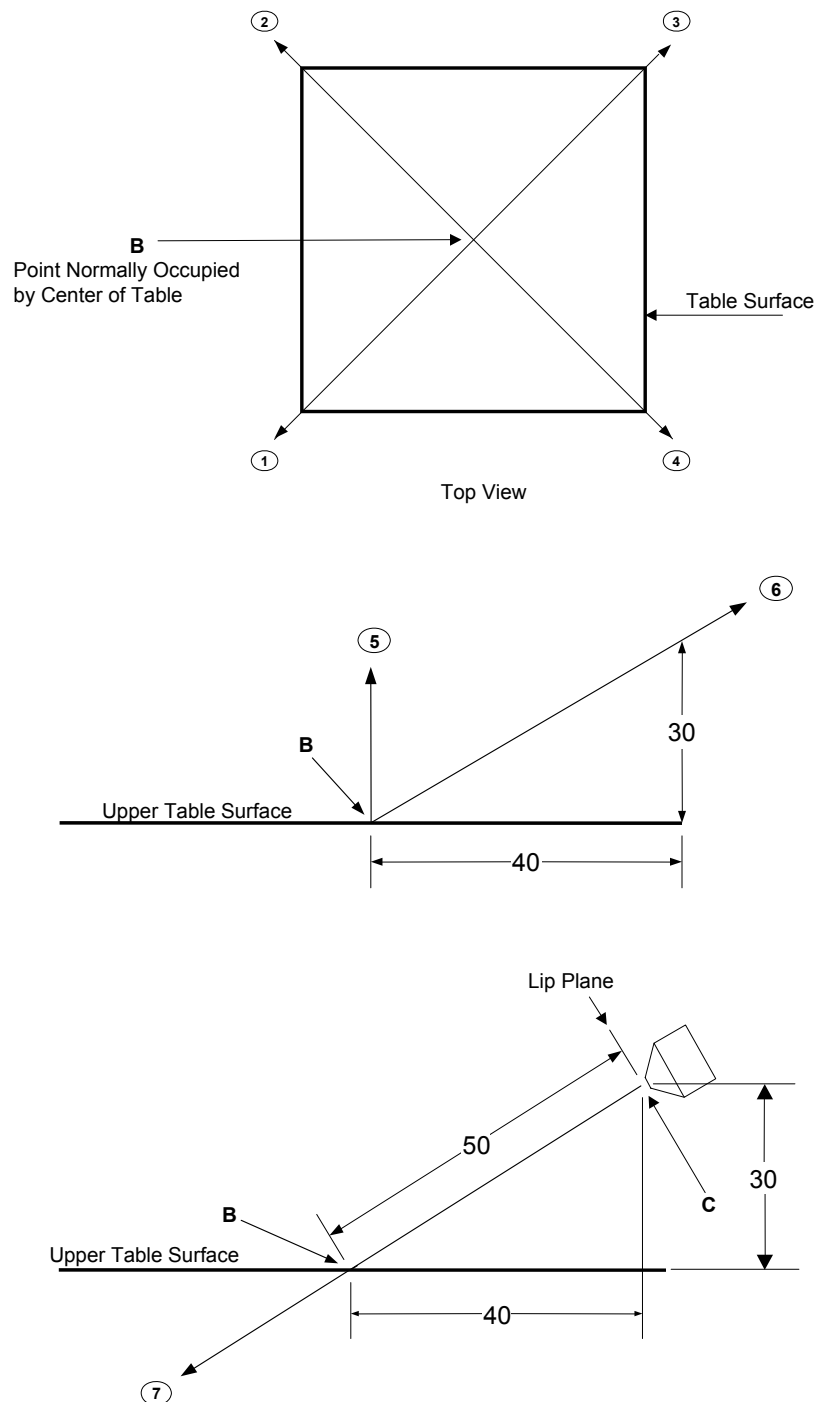


Figure 1 —Axes for determining free-field conditions

NOTE—Axes 1, 2, 3, and 4 are in the horizontal plane normally occupied by the table surface. Measurements of free-field sound pressure are made in absence of the table. Dimensions are in cm.

The signal used for this test shall have the same frequency format as the subsequent speakerphone response measurements. The tolerances specified above shall be met over the entire measurement range in that format. This verification should also be performed if a simulated free-field technique is employed (see 5.3.4).

5.3.4 Simulated free field

Several methods can enable reflection-free measurements in a room with ordinary, untreated boundaries. Tone burst gating, fast Fourier transform (FFT) analysis of short noise bursts, time delay spectrometry (TDS), maximum-length sequence (MLS), and post-process windowing of an impulse response are among the possibilities that together can be called *simulated free-field methods*.

All these methods create a time window, which is adjusted to be slightly shorter than the interval between the first arriving signal and the first reflection. Although each method creates the time window differently, the effective length of the time window is the essential parameter. The lowest frequency that can be measured ($f_{\min}$) is the inverse of the length of the time window (T). For simulated free-field results, the time window can be set as long as desired, provided no reflections are included. In general, the larger the room, the longer the time window can be.

When the following conditions are met, simulated free-field methods can be used as a substitute for an anechoic room:

- a) The signal has a suitable bias effect on the speakerphone; therefore, the unit is measured in a desired stable state. The simulated free-field signal may be the only test signal used, or it may be combined with a suitable bias signal.
- b) The effective time window (T) is long enough (> 5.8 ms) to result in a lower frequency limit ($f_{\min}$) of 175 Hz or lower. This frequency is slightly below the lowest frequency of the 200 Hz 1/3 octave band.

Example: The measurement microphone is 50 cm from the lip ring of the mouth simulator. This direct path corresponds to a signal delay of about 1.5 ms. The first reflection path from the mouth to the nearest boundary is 124 cm and another 124 cm back to the microphone, for a total of 248 cm. This distance corresponds to a delay of about 7.3 ms. The difference between the direct and first reflection path is about 5.8 ms. If the effective time window is set to this amount, the lower frequency limit ($f_{\min}$) is 175 Hz.

If the test table is positioned midway between floor and ceiling, the above example could be set up in a room of about 2.5 m ceiling height with smallest wall-to-wall dimensions slightly larger. If the smallest wall-to-wall dimensions are on the order of 5 m, with a ceiling height of 2.5 m, the test table should be located on the floor. In such a case, the time window can be approximately doubled, and the lower frequency limit ($f_{\min}$) can be halved.

The lower frequency limit ($f_{\min}$) is also the frequency resolution of the measurement. In the above example, there is effectively a data point every 175 Hz, although the curve will usually be smooth. This frequency resolution can result in some smearing of detail at the lower end of the measurement range. However, the true response of many speakerphones is already rather smooth at low frequencies; therefore, this smearing may not always be serious. A larger untreated room can be used for better resolution.

5.3.5 Test table

The test table should be a hard, bare table with a surface area of at least 1 m². No horizontal dimension of the table shall be less than 0.8 m. The table should be flat, rigid, horizontal, and thick enough to provide a sound-reflecting surface on which the speakerphone rests. The tabletop should be composed of a high-density, hard-surfaced material such as hardwood; polished marine plywood; or smooth plastic-laminated, high-density particleboard. The table shall be used for all measurements of desktop speakerphones. It may also be used for measurements of nondesktop speakerphones where appropriate.

5.4 Impairments

5.4.1 General

It is important to understand how a speakerphone performs under less-than-ideal conditions. For this reason, network and acoustic impairments may be introduced into the measurement so the effect on the performance of the speakerphone may be evaluated. It is beyond the scope of this standard to list all impairments, but some are listed below.

5.4.2 Network impairments

Network impairments occur in the path of a wired or wireless telephone connection.

5.4.2.1 Test loops

A wireline analog speakerphone 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 ft, 9000 ft, and 15 000 ft) of 26 AWG cable.

5.4.2.2 Loop current

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

5.4.2.3 Termination impedance

A network termination impedance mismatch can result in the send signal being reflected back toward the telephone. This condition is especially true with analog systems, but reflections can also occur with a digital system if some portion of an end-to-end connection is analog. This reflection can affect speakerphone echo canceller or switching circuits.

5.4.2.4 Network noise

Network noise can affect the speakerphone in various ways. Many designs sense the nominal network noise level in order to set the switching threshold. Also, network noise can affect other nonlinear processes within a speakerphone. Network noise shall be approximated as white in spectrum, and levels shall be measured in dBm, psophometrically weighted (dBmp) according to ITU-T Recommendation O.41.

5.4.3 Acoustic impairments

5.4.3.1 General

In order to realistically test a speakerphone, it may be useful to test it in environments similar to those in which it is expected to operate in ordinary use. Such environments can be considered acoustic impairments with respect to the standard anechoic test environment. They can cause switching, echo cancellation, or other signal processing to work differently from in the anechoic test space.

5.4.3.2 Nearby reflecting surfaces

The speakerphone should be evaluated with nearby reflecting surfaces. In particular, receive signals such as dial tone can cause a half-duplex telephone to become unstable and oscillate between the receive and send states. This phenomenon is commonly referred to as *self-interruption*.

5.4.3.3 Hoth room noise

Hoth noise is random acoustic noise that has a power density spectrum corresponding to that published by Hoth. This spectrum is designed to simulate typical ambient room noise. See IEEE Std 269 for details. The noise level shall be specified in dB(A).

5.4.3.4 Room reverberation

When a speakerphone is tested in realistic room environments, those environments shall be described as part of the test report. The physical description shall include major dimensions of the room, the acoustical characteristics of the major surfaces, and the location of the speakerphone. The relationship of the speakerphone to nearby surfaces shall be described in detail.

The acoustics of the room shall be described by the reverberation time (T_{60}), the early decay time (EDT), and C_{50} . Reverberation time is a standard measure, but may not be very useful in rooms that do not have a very diffuse field, such as most office and home environments. EDT may be more appropriate in such cases, as only the first 10 dB of decay is measured. C_{50} is a measure of how much of the reflected energy in the room is contained in the first 50 ms, relative to the later part after 50 ms. It is also a useful measure of the early part of a sound field. These measures shall be reported for the five octave bands from 250 Hz through 4000 Hz.

6. Test calibration

6.1 Electrical test signal calibration

Electrical test signals shall be calibrated, prior to taking measurements, according to IEEE Std 269.

6.1.1 Mouth simulator calibration

Calibration of the HATS mouth, or standard mouth simulator, shall be performed at the mouth reference point (MRP), according to IEEE Std 269. Calibration should also be performed, or at least checked, for all other test levels to be used. In particular, levels greater than 94 dB SPL (0 dB Pa) should be checked.

The mouth simulator called *standard mouth simulator* in this standard is called *alternative mouth simulator* in IEEE Std 269.

This procedure represents a change from the previous edition of this standard (IEEE Std 1329-1999). It also differs from some international standards.

The current procedure is consistent with both the method and philosophy of calibrating a mouth simulator for measuring handsets or headsets. In these cases, as well as for speakerphones, the microphone(s) in the device under test (DUT) is typically not located at the MRP. The actual level and spectrum at the device are influenced by the sound distribution around the mouth simulator and torso, just as they are for a human talker.

Different mouth simulators may give slightly different results, even if they comply with the relevant standards. The exact type and model of mouth simulator should be reported in the results.

7. Transmission measurements

7.1 General

Procedures are given below for measuring parameters affecting the send and receive performance of speakerphones. These parameters include frequency response, noise, input-output linearity, distortion, and directionality.

Because speakerphone characteristics are affected by loop impedances, terminations, loop currents, and operating levels, the measurements should be made under network and acoustic conditions representative of the conditions the speakerphone is expected to encounter in use. Records shall be kept of parameters used in all measurements. The sensitivities measured shall be presented as dBPa/V or dB V/Pa. Input levels shall be reported.

All equipment shall be calibrated in accordance with the manufacturer's recommendations prior to making measurements. The location of the mouth simulator and the characteristics of the room should be such that the level of any extraneous sound is at least 20 dB below that of the lowest signal to be measured. Proper positioning or sound treatment should be provided to minimize reflected waves and ambient noise. In general, anechoic chambers are required to perform the tests in this standard.

General test environment and impairments for all transmission measurements are described in Clause 5. A range of volume control settings should be used where appropriate, but most measurements should use the reference volume control setting.

The send stimulus for frequency response measurements should be wideband, whether the DUT is narrowband or wideband, because humans always speak in full bandwidth. Receive frequency response measurements should be wideband regardless of the stimulus bandwidth because humans always hear with full bandwidth.

Speakerphones commonly employ some form of dynamic range processing. As a result, the test methods in this standard specify several different test signal levels. The target send loudness rating (SLR) should be achievable by a test level in the range specified. The target receive loudness rating (RLR) should be achievable at some volume control setting at the specified input level.

7.1.1 Choice of test signal and level

The test signals are specified in IEEE Std 269, and they are provided with that standard.

The recommended test signal is male speech, unless specified otherwise in the relevant clause.

Female speech is an alternate. It should be used to determine whether the measured results are similar for both male and female speech.

P50-2 artificial voice may be an acceptable alternate for many applications, depending on the nature of signal processing in the DUT. The user should demonstrate that results using this signal are equivalent to those that would be obtained using male speech.

For digital test levels from -15 dBm0 to -10 dBm0, for analog test levels of -14 dBV and above, and for acoustical test levels of 99 dB SPL (+5 dBPa) and greater, the compressed version of the test signal should be used to avoid clipping.

7.1.2 Measurement bandwidth and resolution

The same bandwidth shall be used for calibration as is used for measurement. The actual bandwidth used shall be stated. Calibration and measurement shall be performed using identical measurement resolution. Measurement resolution shall be stated.

Send and receive frequency response measurements should be made in 1/12 octave bands. One-third octave band resolution is required for loudness rating calculation and may also be used for frequency response compliance testing.

7.1.3 Conversion of measured data to ISO R10 format

Measurements may be performed in various frequency formats, depending upon the analysis method employed. Response measurements of speakerphones can contain numerous peaks and dips. This conversion, therefore, should be performed using “band-averaging.” The measured points within a particular 1/3 octave band are “power averaged” according to Equation (1).

At each ISO R10 preferred frequency

$$H'(f) = 10 \log_{10} \left[\frac{1}{N} \sum_{i=1}^N 10^{\frac{H_i}{10}} \right]$$

where

$H'(f)$ is response at the new preferred ISO R10 frequency

f is preferred ISO R10 frequency

N is number of response values within the 1/3 octave band centered at f

i is index for each response value within the 1/3 octave band

H_i is measured response value (in dB)

(1)

For the lowest frequency within the band, $i = 1$. For the highest included frequency, $i = N$. The 1/3 octave passband limit frequencies can be calculated as shown in Equation (2).

$$f = 10^{(n/10) \pm 0.05}$$

where
 n is band number

(2)

Example: For the 100 Hz band, the band number = 20; for the 125 Hz band, the band number = 21; etc. IEEE Std 269 contains a list of the band numbers, 1/3 octave center frequencies, and corresponding passband limits.

For measured data at frequencies coinciding with a band-edge frequency ($i = 1$ and/or $i = N$), reduce the value by 3 dB, and use that data point in both the upper and lower frequency band calculations.

For constant percentage bandwidth measurements, there will always be the same number of points for each converted band (4 or 8, for 1/12 or 1/24 octave bands, respectively). For constant bandwidth data (e.g., FFT) on a log frequency axis, the measurement data will appear under-sampled at low frequencies and over-sampled at higher frequencies.

7.2 Choice of measurement transducers

7.2.1 General

The choice of measurement transducers should be determined by how the device will be used. There will be modest differences between the two methods described below.

7.2.2 Standard mouth and microphone

A standard mouth simulator and free-field microphone(s) shall be used for laboratory measurements of speakerphones in a one-piece housing. (*Example:* microphone and speaker in one enclosure.) The laboratory includes an anechoic chamber and may include a test table or test wall. If a test table or a test wall is used, the interaction between the speakerphone and the table or wall is considered part of the speakerphone. The standard mouth and microphone are recommended for most one-piece desktop speakerphones, handheld speakerphones, and speakerphones built into walls or similar surfaces.

7.2.3 HATS

A HATS with both right and left ear simulators may be used for all speakerphone measurements. A HATS shall be used for laboratory measurements of speakerphones in two or more housings. (*Example:* microphone and speaker in separate enclosures.) The laboratory includes an anechoic chamber and may include a test table or test wall. A HATS shall also be used for measurements in setups where there are significant reflections or complex interactions with the acoustical environment. Examples include speakerphones used in automobiles and computer workstations.

7.3 Positioning of speakerphone and test transducers

7.3.1 General

The test position should be determined by how a speakerphone will be used. Some speakerphones will be used in different ways and, therefore, should be tested in multiple positions.

The term *test position* specifically refers to the position of the test transducers relative to the speakerphone. The geometric reference point on the free-field microphone is the center of the microphone grid. The geometric reference point on the standard or the HATS mouth simulator is the intersection of the mouth axis on the lip plane, not the MRP. When the HATS mouth is at the test position, the HATS ears are located farther away from the speakerphone, just as with a real human. Some implications of the ear position relative to the mouth are discussed in (7.5.2).

There may be more than one test position. There shall be a primary test position, which reflects how the device is most likely to be used. There may be one or more alternate test positions (ATPs) that reflect alternative ways of using the device.

The primary test position and ATP include the choice of measurement transducers. Either the standard mouth and free-field microphone or a HATS will be chosen. In either case, the measurement transducers shall be used according to this standard.

All tests shall be performed in the primary test position. Send and receive frequency response, SLR, and RLR shall be tested in all ATPs. Other characteristics should be tested in ATPs when it is likely that the characteristics of the device might be significantly different in the ATP. This guidance may be specified by the manufacturer, but is otherwise left to the test laboratory to decide.

7.3.2 Primary test position

The test laboratory shall select the primary test position as follows:

- a) Recommended test position (RTP): If the manufacturer specifies an RTP, the speakerphone shall be tested in that position. The RTP shall reflect the way the speakerphone is intended by the manufacturer to be used in a real situation. The manufacturer should provide suitable drawings or pictures to illustrate the RTP.
- b) Derived recommended test position (DRTP): If the manufacturer's RTP is not provided and there are sufficient instructions in the user guide, then a test position shall be derived from these instructions.
- c) Estimated real use position (ERUP): If the manufacturer does not provide any suitable instructions, the speakerphone shall be tested in the ERUP. The test laboratory shall define an ERUP that closely approximates real use.

7.3.3 Alternate test position (ATP)

When an ATP is used, it shall follow the same principles as RTPs:

- a) Recommended alternate test position (RATP): If the manufacturer specifies a RATP, the speakerphone shall be tested in that position. The manufacturer should provide suitable drawings or pictures to illustrate the RATP.