

IEEE Standard for Measuring Electroacoustic Performance of Communication Devices

IEEE Communications Society

Developed by the
Standards Development Board

IEEE Std 269™-2019
(Revision of
IEEE Std 269-2010)

STANDARDS

IEEE Standard for Measuring Electroacoustic Performance of Communication Devices

Developed by the

**Standards Development Board
of the
IEEE Communication Society**

Approved 7 November 2019

IEEE SA Standards Board

Abstract: Practical methods for making electroacoustic laboratory measurements of analog and digital speech communications terminals and connected audio devices are contained in this standard. The methods are applicable to a wide variety of wired, wireless, cordless, and mobile communication terminals. Examples include Voice over Internet Protocol (VoIP), Wi-Fi®, and softphone devices. Tests applicable to connected audio devices such as wired, Bluetooth®, and universal serial bus (USB) handsets, headsets, and wearables are included. The standard contains objective metrics, subjective metrics, and subjective metric predictors. Application is in the frequency range from 20 Hz to 20 kHz.

Keywords: analog terminals, Bluetooth®, cordless, digital terminals, electroacoustic measurements, handsets, headsets, IEEE 269™, mean opinion score (MOS), mobile, MOS predictors, softphones, speech communication, subjective metrics, telephone, universal serial bus (USB), Voice over Internet Protocol (VoIP) terminals, voice transmission performance, wearables, Wi-Fi® terminals

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Introduction

This introduction is not part of IEEE Std 269-2019, IEEE Standard for Measuring Electroacoustic Performance of Communication Devices.

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 communication terminals, handset, and headsets. The present standard is a revision of IEEE Std 269TM-2010.

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 Secretary, IEEE Standards Board, The Institute of Electrical and Electronics Engineers, Inc., 345 East 47th Street, New York, NY 10017.

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IEEE Standard for Measuring Electroacoustic Performance of Communication Devices

1. Overview

Both objective and subjective methods are used to evaluate electroacoustic performance of communication devices. This standard contains objective measurement methods that use test instruments. The results of these objective measurements are intended to generally agree with the results of subjective testing. Many objective measurements are required to evaluate a device.

This standard also contains subjective metric predictors that are based on objective measurements. They are algorithms that use dynamic physical measurements to predict some aspect of subjective perception. Each predictor is presented as a mean opinion score (MOS). Many aspects of the performance of a communication device are combined to generate a MOS. To properly interpret a MOS, it is necessary to consider the objective measurements.

Evaluation of a device should include both objective measurements and subjective metric predictors.

Although outside the scope of this standard, there are references to other standards that contain procedures for conducting subjective tests with real human subjects.

1.1 Scope

This standard provides techniques for objective measurement of electroacoustic devices used for speech communications, including communication devices also used for multimedia applications. Application is for devices that are primarily used closely coupled to the ear. Aspects of devices with speakerphone or speaker listening features are covered by IEEE Std 1329™.¹

This standard is applicable to devices that include communication terminals that connect to a network, as well as to subsystems that connect to a terminal. The methods in this standard specifically apply to devices

¹ Information on references can be found in Clause 2.

that connect to analog, digital, Voice over Internet Protocol (VoIP), and other local area network (LAN) networks. They may also apply to devices that connect to mobile and other wireless networks.

This standard applies to terminals independently of the internal audio architecture. Examples include cordless phones, VoIP phones, and softphone clients. It also applies to subsystems such as universal serial bus (USB) audio devices, Bluetooth® headsets, and four-wire analog handsets.

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 terminals and subsystems. Guidance for the applicability of specific tests is given within this standard.

1.2 Purpose

The purpose of this standard is to provide laboratory test methods for evaluating the electroacoustic performance of communication devices.

1.3 How to use this standard

In this standard, the word “device” refers to the object to be tested (DUT). Examples include terminals and subsystems such as digital and VoIP telephones, analog telephone sets, USB headsets, analog handsets, and headsets.

Proceed with the following steps:

- a) Select the required test equipment, fixtures, and environment according to Clause 5.
- b) Set up codecs, interfaces, and test circuits according to Clause 6.
- c) Select appropriate test signals according to Clause 7.
- d) Calibrate measurement equipment and signal sources according to Clause 8.
- e) Position the device according to Clause 9.
- f) Adjust the control settings of the device according to Clause 10. Depending on the location of the control settings, they may need to be adjusted before positioning the device. It may be necessary to perform one or more preliminary measurements to verify the controls have been set properly.
- g) Arrange the test setup according to Clause 11.
- h) Perform the measurement(s) according to Clause 12.
- i) Translate measurements made at the ear simulator from drum reference point (DRP) to listener reference point (LRP) according to Clause 13. Translation shall be performed prior to any format conversion.
- j) Perform any desired data format conversion according to Clause 13.

Table 1—Index of measurements and device types

Measurement	Signal direction	Digital terminal	Digital subsystem	Analog terminal	Analog subsystem
Frequency response	Receive	12.2.2	12.2.2	12.2.2	12.2.2
	Send	12.2.3	12.2.3	12.2.3	12.2.3
	Sidetone	12.2.4		12.2.4	
	End-to-End	12.2.5		12.2.5	
Output level	Receive	12.3.2	12.3.2	12.3.2	12.3.2
	Send	12.3.3	12.3.3	12.3.3	12.3.3
	Sidetone	12.3.4		12.3.4	
	End-to-End	12.3.5		12.3.5	
Conversational gain	Receive	12.4	12.4	12.4	12.4
Tonal balance	Receive	12.5	12.5	12.5	12.5
	Send	12.5	12.5	12.5	12.5
Balance and polarity	Receive	12.6	12.6	12.6	12.6
Echo	Electrical	12.7		12.7	
Self-noise	Receive	12.8.1	12.8.1	12.8.1	12.8.1
	Send	12.8.2	12.8.2	12.8.2	12.8.2
Level-dependent dynamics	Receive	12.9.4	12.9.4	12.9.4	12.9.4
	Send	12.9.5	12.9.5	12.9.5	12.9.5
	Sidetone	12.9.5		12.9.5	
	End-to-End	12.9.5		12.9.5	
Distortion and induced artifacts	Receive	12.10.2	12.10.2	12.10.2	12.10.2
	Send	12.10.3	12.10.3	12.10.3	12.10.3
	Sidetone	12.10.4		12.10.4	
	End-to-End	12.10.5		12.10.5	
Speech quality in quiet environment	Receive	12.11	12.11	12.11	12.11
	Send	12.11	12.11	12.11	12.11
Speech quality in noisy environment	Receive	12.12	12.12	12.12	12.12
	Send	12.12	12.12	12.12	12.12
Mute leakage	Receive	12.13.1		12.13.1	
	Send	12.13.2		12.13.2	
Device delay	Receive	12.14.1	12.14.1	12.14.1	12.14.1
	Send	12.14.2	12.14.2	12.14.2	12.14.2
	Sidetone	12.14.3	12.14.3	12.14.3	12.14.3
	End-to-End	12.14.4		12.14.4	
Stability loss		12.15	12.15	12.15	12.15
Howling		12.16	12.16	12.16	12.16
Max acoustic output	Maximum	12.17.1	12.17.1	12.17.1	12.17.1
	Peak	12.17.2	12.17.2	12.17.2	12.17.2
Impedance	Receive				12.18.1
	Send				12.18.2
NOTE—The subclause numbers in this table are dynamic links to the specific subclauses in the document. ²					

Details of some procedures are found in the normative annexes referred to from within the relevant clause. Consult the informative annexes for additional information and alternative procedures.

² Notes in text, tables, and figures of a standard are given for information only and do not contain requirements needed to implement this standard.

1.4 Changes since last edition of this standard

The major changes in this revision are as follows:

- a) Restructuring of document
- b) Device classification: terminals, subsystems
- c) Applies to devices of any bandwidth between 20 Hz and 20 kHz
- d) Use of speech for most measurements except distortion
- e) Use of output level and conversational gain to supersede loudness ratings
- f) Addition of a new echo burst test method
- g) Expansion of coverage of speech quality
- h) New information about wind noise
- i) New information about Hoth and other noises
- j) New stimulus .wav files

1.5 Word usage

The word *shall* indicates mandatory requirements strictly to be followed in order to conform to the standard and from which no deviation is permitted (shall equals is required to).^{3,4}

The word *should* indicates that among several possibilities one is recommended as particularly suitable, without mentioning or excluding others; or that a certain course of action is preferred but not necessarily required (should equals is recommended that).

The word *may* is used to indicate a course of action permissible within the limits of the standard (may equals is permitted to).

The word *can* is used for statements of possibility and capability, whether material, physical, or causal (can equals is able to).

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.

AES 17-2015, AES standard method for digital audio engineering—Measurement of digital audio equipment.⁵

ANSI S1.4:1983/IEC 61672-1, American National Standard Electroacoustics— Sound Level Meters—Part 1: Specifications.^{6, 7}

³ The use of the word *must* is deprecated and cannot be used when stating mandatory requirements, *must* is used only to describe unavoidable situations.

⁴ The use of *will* is deprecated and cannot be used when stating mandatory requirements, *will* is only used in statements of fact.

⁵ AES publications are available from the Audio Engineering Society (<http://www.aes.org/>).

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

ANSI S1.11/IEC 61260-1, American National Standard Electroacoustics—Octave-band and Fractional-octave-band Filters—Part 1: Specifications.

Bellcore SR-TSV-002476-1992, CPE compatibility considerations for the voiceband data transmission interface.⁸

ETSI EG 202 396-1, Speech and multimedia Transmission Quality (STQ); Speech Quality performance in the presence of background noise; Part 1: Background noise simulation technique and background noise database.⁹

ETSI EG 202 396-3, Speech and multimedia Transmission Quality (STQ); Speech Quality performance in the presence of background noise Part 3: Background noise transmission—Objective test methods.

ETSI TS 103 106, Speech quality performance in the presence of background noise: Background noise transmission for mobile terminals-objective test methods.

ETSI TS 103 281 V1.1.1, Speech and multimedia Transmission Quality (STQ); Speech quality in the presence of background noise: Objective test methods for super-wideband and fullband terminals.

IEC 60268-4, Sound system equipment—Part 4: Microphones.

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

IEEE Std 743-1995TM (withdrawn), IEEE Standard Equipment Requirements and Measurement Techniques for Analog Transmission Parameters for telecommunications.^{10, 11, 12}

IEEE Std 802.11[®], IEEE Standard for Information technology—Telecommunications and information exchange between systems Local and metropolitan area networks—Specific requirements—Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications.

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

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

IETF RFC 3551 (2003), RTP Profile for Audio and Video Conferences with Minimal Control.¹³

ISO 3, Preferred Numbers-Series of preferred Numbers.¹⁴

ITU-T Recommendation G.100.1, The use of the decibel and of relative levels in speechband telecommunications.¹⁵

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

ITU-T Recommendation G.122, Influence of National Systems on Stability and Talker Echo in International Connections.

⁶ ANSI publications are available from the American National Standards Institute (<http://www.ansi.org/>).

⁷ IEC publications are available from the International Electrotechnical Commission (<http://www.iec.ch>) and the American National Standards Institute (<http://www.ansi.org/>).

⁸ Bellcore publications are available from iconectiv (<https://iconectiv.com/>).

⁹ ETSI publications are available from the European Telecommunications Standards Institute (<http://www.etsi.org/>).

¹⁰ The IEEE standards or products referred to in Clause 2 are trademarks owned by The Institute of Electrical and Electronics Engineers, Incorporated.

¹¹ IEEE publications are available from The Institute of Electrical and Electronics Engineers (<http://standards.ieee.org/>).

¹² IEEE Std 743-1995 has been withdrawn; however, copies can be obtained from the Institute of Electrical and Electronics Engineers (<http://standards.ieee.org/>).

¹³ IETF publications are available from the Internet Engineering Task Force (<https://www.ietf.org/>).

¹⁴ ISO publications are available from the International Organization for Standardization (<http://www.iso.org/>) and the American National Standards Institute (<http://www.ansi.org/>).

¹⁵ ITU-T publications are available from the International Telecommunications Union (<http://www.itu.int/>).

ITU-T Recommendation G.160, Voice Enhancement Devices, Appendix II.

ITU-T Recommendation G.167, Acoustic Echo Controllers.

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.

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

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

ITU-T Recommendation G.722, 7 kHz Audio-coding Within 64 kbit/s.

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.

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

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

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

ITU-T Recommendation P.50, Artificial Voices.

ITU-T Recommendation P.51, Artificial Mouths.

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.380, Electro-acoustic Measurements on Headsets.

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

ITU-T Recommendation P.570, Artificial noise-fields under laboratory conditions.

ITU-T Recommendation P.800, Methods for subjective determination of transmission quality.

ITU-T Recommendation P.800.1, Mean opinion score (MOS) terminology.

ITU-T Recommendation P.835, Subjective test methodology for evaluating speech communication systems that include noise suppression algorithm.

ITU-T Recommendation P.862, Perceptual Evaluation of Speech Quality (PESQ).

ITU-T Recommendation P.863, Perceptual objective listening quality assessment.

3. Definitions, acronyms, and abbreviations

3.1 Definitions

For the purposes of this document, the following terms and definitions apply. The *IEEE Standards Dictionary Online* should be consulted for terms not defined in this clause.¹⁶

A-weighted: A measurement made using the “A” frequency weighting specified in ANSI S1.4-1983. It approximates the frequency response of the ear at 40 dB SPL, 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 terminal conversation levels.

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

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:* **sound pressure level**.

acoustic output: The sound pressure level developed in an ear simulator or at the measuring microphone. *See:* **sound pressure level**.

active level: Level of a signal with pauses and/or silences removed. Active level (decibels) is equivalent to the level of the original signal plus $10 \times \log(100 / A)$. “A” is the nonspeech activity factor in percent.

active spectrum: Spectrum of a signal with pauses and/or silences removed. The active spectrum (decibels) is equivalent to the spectrum of the original signal plus $10 \times \log(100 / A)$. “A” is the nonspeech activity factor in percent.

active speech level (ASL): Speech level measured according to Method B of ITU-T Recommendation P.56, which accounts for pauses or silences. The method involves the use of smoothing, hangover time, and thresholds. ASL (decibels) is equivalent to the level of the entire signal, including pauses and/or silences, plus $10 \times \log(100 / A)$. “A” is the speech activity factor in percent.

active speech spectrum: Speech spectrum that accounts for pauses or silences. The active speech spectrum (dB) is equivalent to the spectrum of the entire signal, including pauses and/or silences, plus $10 \times \log(100 / A)$. “A” is the speech activity factor in percent.

activity factor (nonspeech): Percentage of time during which a nonspeech signal is present, excluding pauses or silence.

activity factor (speech): Percentage of time during which speech is present, as measured according to Method B of ITU-T Recommendation P.56.

analog subsystem: A device that connects to a two-way voice communication terminal in analog format, regardless of internal architecture. Analog handsets and headsets are examples.

NOTE—See Figure 1.

¹⁶ *IEEE Standards Dictionary Online* is available at: <http://dictionary.ieee.org>. An IEEE Account is required for access to the dictionary, and one can be created at no charge on the dictionary sign-in page.

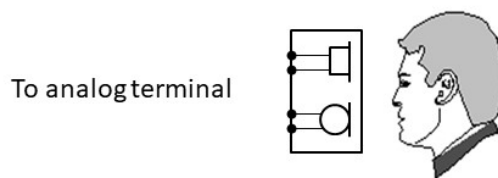


Figure 1—Analog subsystem

analog terminal: A device that connects to the network in analog format, regardless of internal architecture, which provides two-way voice communication. Wired and cordless analog telephone sets are examples.

NOTE—See Figure 2 and Figure 3.

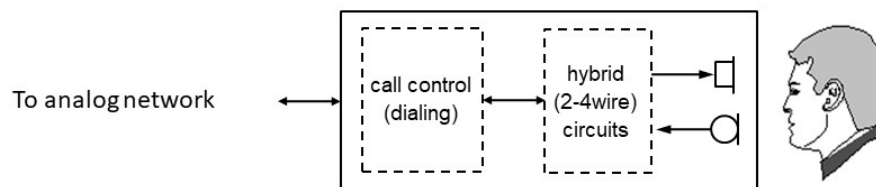


Figure 2—Analog terminal

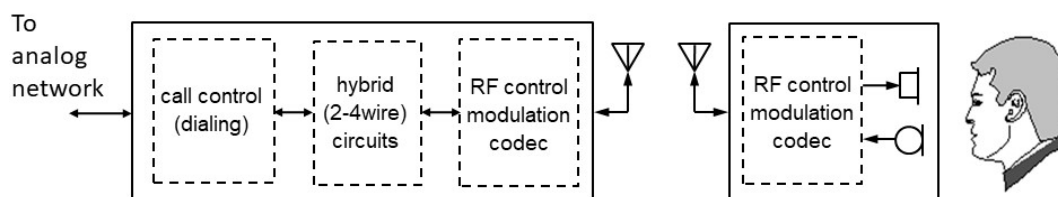


Figure 3—Analog cordless terminal

attenuation range (a_H): The difference in level in decibels 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.

attribute: An aspect or feature regarded as a characteristic or an inherent part of perceived quality, such as loudness.

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

automatic level control: A circuit or algorithm that attempts to hold the output level of the device constant regardless of changes to the input level.

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. Also known as “attack” or “rise” time.

C_{50} : In room acoustics, the ratio of energy before and after 50 ms, expressed as a difference in decibels.

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.

conditioning: The application of a signal prior to the measurement to place the device in an active and stable state. Speech is typically used as a conditioning signal.

constant percentage bandwidth (CPB) analysis: A measurement in fractional octaves, typically implemented as a digital filter bank, without time gaps, suitable for computing the spectrum of stationary or nonstationary signals. Fulfills ANSI S1.11/IEC 61260-1 and ANSI S1.4/IEC 61672-1. Sometimes called “real time analysis.”

NOTE—See Annex A for additional information.

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

conversational gain: The difference in level between a face-to-face conversation at 1 m and a conversation using a communication device.

crest factor: Absolute peak amplitude divided by root-mean-square amplitude, expressed as a ratio or in decibels.

critical distance: In a room, the distance from the source at which the level of the direct sound is equal to the level of the diffuse sound.

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

NOTE—To match common usage, we use dBA herein. You may see dB(A) in other documents.

dBFS: Decibels relative to full-scale digital saturation. 0 dBFS is the level of a square-wave signal whose amplitude is at the maximum digital value.

NOTE—See H.4.3.

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.” 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. For the L16-256 wideband codec (IETF RFC 3551), 0 dBm0 is 3.17 dB below digital full scale. *See: zero transmission level point (0 TLP).*

dBm(A): Power level in decibels, relative to a power of 1 mW (milliwatt), A-weighted. *See: 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 V AC root mean square.

dBV(A): Voltage level in decibels, relative to 1 V root mean square, measured with A-weighting.
See: A-weighted.

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

delay: The time interval between the application or detection of a signal at one point and the detection of the same signal at another point. Sometimes called “latency.”

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

diffuse field (DF): Sound field in which the time average of the mean-square sound pressure is the same everywhere and the flow of acoustic energy in all directions is equally probable.

digital subsystem: A device that connects to a two-way voice communication terminal in digital format, regardless of transmission medium or internal architecture. Universal serial bus (USB) or Bluetooth® headsets and Wi-Fi® wearables are examples.

NOTE—See Figure 4 through Figure 6.

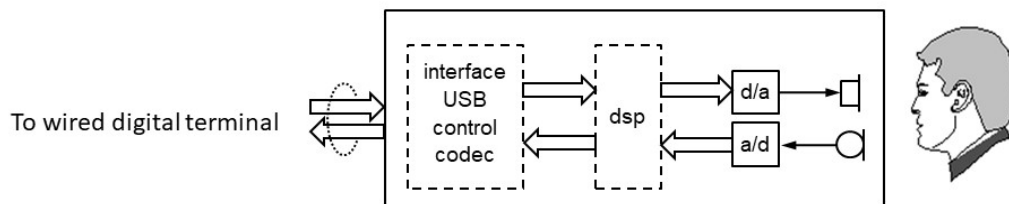


Figure 4—USB subsystem

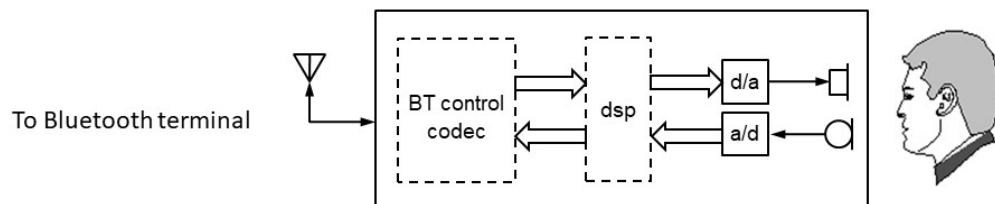


Figure 5—Bluetooth subsystem

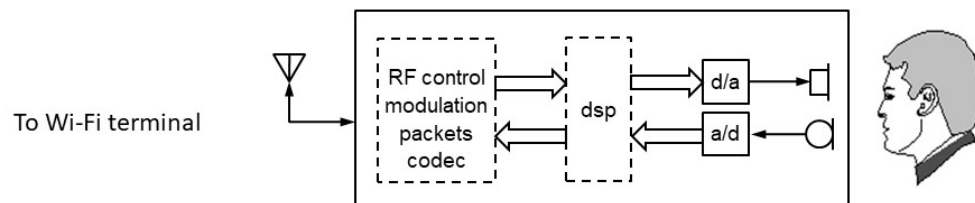


Figure 6—Wi-Fi subsystem

digital terminal: A device that connects to the network in digital format, regardless of transmission medium or internal architecture, which provides two-way voice communication. Voice over Internet Protocol (VoIP) or mobile phones, and Wi-Fi® communication devices are examples.

NOTE—See Figure 7 through Figure 9.

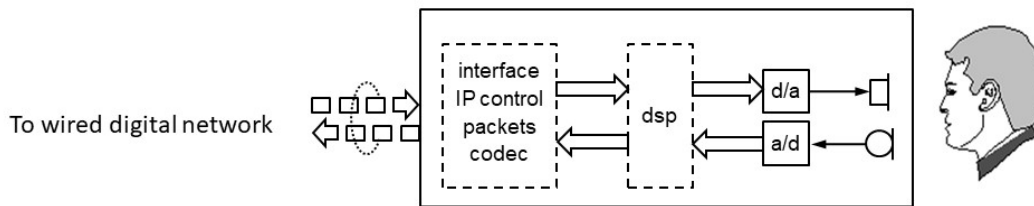


Figure 7—Wired terminal

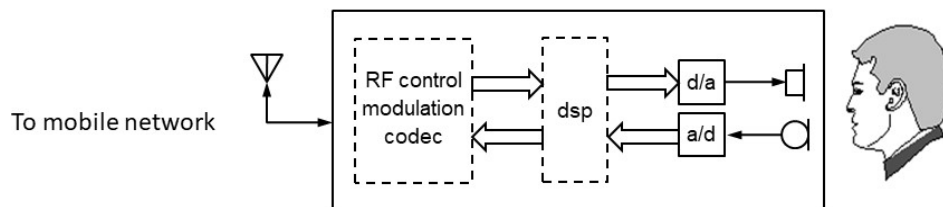


Figure 8—Mobile terminal

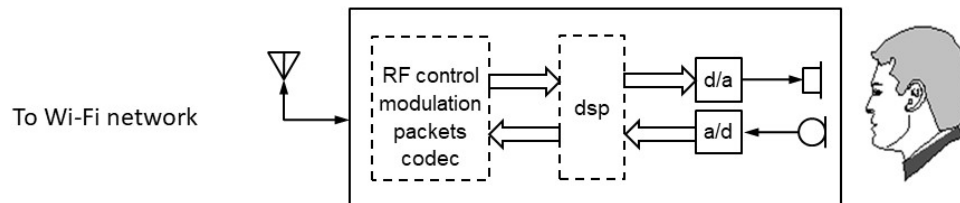


Figure 9—Wi-Fi terminal

distractor: A sound coming from a specific direction that could disturb the intended speech transmission. Examples are a nearby talker or a localized transient noise.

double-talk (DT): 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 cap reference point (ECRP): 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 drum reference point: *See:* **drum reference point (DRP).**

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

ear simulator: Device for measuring the output sound pressure of an earphone under well-defined loading conditions in a specified frequency range. It usually consists 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 eardrum.

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 to the send electrical test point, excluding any delay in the test equipment.

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

end-to-end: Speech transmission path from the mouth of the near-end talker to the ear of the far-end listener. Also called “end-to-end signal path.”

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

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

free field (FF): A sound field free of reflections. Specifically, a sound field in a homogenous, isotropic medium free from boundaries. In practice, the effects of boundaries on a free field are negligible over the region of interest.

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

fullband (FB): Bandwidth consistent with 44.1 kHz or 48 kHz sampling rate, typically 20 Hz to 20 000 Hz. Usually refers to a device or transmission channel.

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.

NOTE—See ITU-T Recommendation P.58.

howling: Audible squealing sound in a terminal, 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.

impairment: A signal or condition that may be introduced into the test environment or network that could disturb the intended speech transmission. Examples are acoustical noise, electrical noise, distractors, wind, and analog and digital network conditions.

latency: *See:* delay.

listener reference point (LRP): A generic term representing drum reference point (DRP), ear reference point (ERP), free field (FF), diffuse field (DF), natural field (NF), or any other specified point or sound field. It applies to receive, sidetone, and end-to-end speech transmission paths.

listener sidetone: The signal present at the receiver of the device under test due to sound in the local environment where the device is used.

jitter: The time variation of a clock signal from nominal. It can cause unstable delay.

loudness rating guard-ring position (LRGP): The test position a handset assumes when it is placed on an artificial test head as described in Annex A to ITU-T Recommendation P.76.

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.

method: A formal, well-documented approach for accomplishing a task.

metric: A quantitative measure of the degree to which a system, component, or process possesses a given attribute, such as signal-to-noise ratio.

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.

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.

narrowband (NB): Bandwidth consistent with 8 kHz sampling rate, typically 100 Hz to 3400 Hz or 300 Hz to 3400 Hz. Usually refers to a device or transmission channel.

natural field (NF): Sound field consisting of both free field and diffuse field components, such as found in a typical room.

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.

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

receive: Speech transmission path from the network to the ear of the terminal user.

receive electrical test point (RETP): The point in reference codec, wireless reference base station, or a battery feed circuit at which signals are applied to the terminal in the receive direction. For further discussions on wireless interfaces, see appropriate wireless standards.

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 terminals.

reference receive volume control setting: The setting of one or more receive volume controls of a terminal or subsystem that results in 6 dB of conversational gain, at nominal input level.

reference send level control setting: The setting of one or more send level controls of a terminal or subsystem that results in a nominal output level, at nominal input level.

send: Speech transmission path from the mouth of the terminal user to the network.

send electrical test point (SETP): The point in a reference codec, wireless reference base station, or battery feed circuit at which signals coming from the terminal in the send direction are accessed. For further discussions on wireless interfaces, see appropriate wireless standards.

sidetone: Speech transmission path from the microphone to the receiver of the handset or headset.
See also: **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.

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

sound pressure level: The 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.

NOTE 1—For this standard, the reference pressure is normally 1 pascal (Pa), and sound pressure levels are expressed in decibels re 1 Pa (dBPa). When a reference pressure of 20 μ Pa is used, the sound pressure level will be expressed as dB SPL. Unless otherwise indicated, root-mean-square 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 μ Pa.

NOTE 2—0 dB Pa = 94 dB SPL, 0 dB SPL = 20 μ Pa, 1 Pa = 1 N/m². A-weighted sound pressure level in decibels (dB SPL, A-weighted) is often abbreviated as dBA. (See ANSI S1.4-1983.)

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 terminal 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.

superwideband (SWB): Bandwidth consistent with 32 kHz sampling rate, typically 50 Hz to 14 000 Hz. Usually refers to a device or transmission channel.

switching time (Ts): Time taken to switch from one transmission direction to the other in alternating single talk conversation.

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

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

terminal coupling loss (TCL): The loss in the echo path from the receive electrical test point to the send electrical test point. It can be weighted in time, frequency, or both. Terminal coupling loss is also known as "echo return loss" (ERL).

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

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

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, weighted in the frequency domain per ITU-T Recommendations G.122 and G.167.

wideband (WB): Bandwidth consistent with 16 kHz sampling rate, typically 100 Hz to 6800 Hz. Usually refers to a device or transmission channel.

Wi-Fi®: A wireless local area network (LAN) as defined in IEEE Std 802.11®. It allows computers, smartphones, or other devices to connect to the Internet or communicate with one another wirelessly within a particular area.

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

3.2 Acronyms and abbreviations

AGC	automatic gain control
ANTHD	amplitude normalized total harmonic distortion
ASL	active speech level
CPB	constant percentage bandwidth
DF	diffuse field
DFTP	diffuse field test point
DRP	drum reference point
DRTTP	derived recommended test position
DSP	digital signal processor
DUT	device under test
ECRP	ear cap reference point
EPD	echo path delay
ERP	ear reference point
ERUP	estimated real use position
FF	free field
FFT	fast Fourier transform
HATS	head and torso simulator
LAN	local area network
LRGP	loudness rating guard-ring position
LRP	listener reference point
MECRP	manufacturer-defined earcap reference point
MLS	maximum length sequence
MOS	mean opinion score
MRP	mouth reference point
PCM	pulse-code modulation
PN	pulsed noise
RETP	receive electrical test point
rms	root mean square
RTP	recommended test position
RWP	recommended wearing position
SETP	send electrical test point
SDNR	signal-to-distortion-and-noise ratio
SFI	single frequency interference
S/N	signal/noise
SNR	signal-to-noise ratio
STP	standard test position
TDS	time delay spectrometry
THD	total harmonic distortion
USB	universal serial bus
VoIP	Voice over Internet Protocol
WB	wideband

4. Symbols

4.1 Letter symbols

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

$G_{LRP}(f)$	rms spectrum at listener reference point, in pascals, or in dBPa or dBSPL
$G_{MRP}(f)$	rms spectrum at mouth reference point, in pascals, or in dBPa or dBSPL
$G_{RETP}(f)$	rms spectrum at receive electrical test point, in volts, digital full scale, or in dBV, dBm0, or dBFS
$G_{SETP}(f)$	rms spectrum at send electrical test point, in volts, digital full scale, or in dBV, dBm0, or dBFS

The letter “H” is used for frequency response:

$H(f)$	frequency response, in dB
$H'(f)$	response at the new preferred ISO R10 frequency
$H_E(f)$	echo frequency response, in dBV/V or dBFS/FS
$H_{LS}(f)$	listener sidetone frequency response, in dBPa/Pa or dBSPL/SPL
$H_{ETE}(f)$	end-to-end frequency response, in dBPa/Pa or dBSPL/SPL
$H_R(f)$	receive frequency response, in dBPa/V or dBSPL/FS
$H_S(f)$	send frequency response, in dBV/Pa or dB FS/SPL
$H_{TS}(f)$	talker sidetone frequency response, in dB Pa/Pa or dBSPL/SPL

The letter “L” is used for rms levels, with the bandwidth to be specified:

L_{EB}	echo burst level, in dBV or dBFS
L_{EBMAX}	maximum echo burst level, in dBV or dBFS
L_{EQ}	level equivalent to the total sound energy in a defined time period, in dBSPL or dBV
L_{ET}	total echo level, in dBV or dBFS
L_{LPE}	log power echo level, in dBV or dBFS
L_{LRP}	level at listener reference point, in dBPa or dBSPL
L_{MID}	level of stimulus to be determined for analog subsystem tests
L_{MRP}	level at mouth reference point, in dBPa or dBSPL
L_{RETP}	level at receive electrical test point, in dBV, dBm0, or dBFS
L_{ROOM}	level of background noise in measurement environment, in dBSPL
L_{SETP}	level at send electrical test point, in dBV, dBm0, or dBFS

Other letter symbols:

A_{EB}	echo burst activity factor, in percentage
T_a	averaging time, in seconds
T_{EB}	echo burst time, in seconds
$TCLW$	frequency weighted terminal coupling loss, in dB
ΔL_{EB}	echo burst level difference, in dB

4.2 Drawing symbols

The symbols presented in Figure 10 are used throughout the standard.

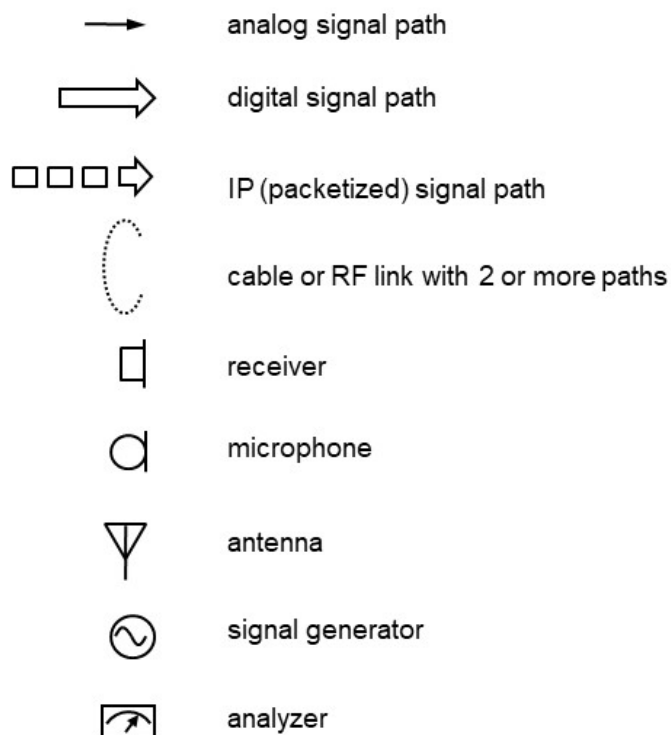


Figure 10—Drawing symbols

5. Test equipment, fixtures, and environment

The test equipment, fixtures, and environment generally required to test all the devices covered by this standard are 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. The test fixtures are selected based on the type of device under test (i.e., handsets and headsets). The test environment is determined by the type of measurement to be performed.

Calibration minimizes the effects of bias errors in the measurement. All equipment should be calibrated in accordance with the recommendations of the manufacturer before performing the system calibration procedures in Clause 8.

Sources of random error in the measurement will remain. These will affect the overall measurement uncertainty. See Annex B.

5.1 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.

5.2 Ear simulators

The fundamental purpose of ear simulators is to test a receiver under conditions that most closely approximate actual use by real persons. Either the Type 3.3 ear simulator (5.2.1) or a fullband ear simulator (5.2.2) shall be used for all measurements and devices, unless the applicable performance specification requires or allows an alternative. A fullband ear simulator is not the same as the Type 3.3, so the measurement results may not be the same.

Both the Type 3.3 and fullband ear simulators have a soft pinna that deforms when the receiver is pressed against it. There will be some acoustic leakage since the ear simulator is almost never completely blocked. Changes in force or position of the receiver against the ear simulator will cause the acoustic leak to vary. Changes in the leak will generally introduce variations in the frequency response, especially at the lower frequency range of the receiver, just as it does on a human ear. The relationship between position and force will vary depending on the shape of the receiver. Note that the correlation between measurements on ear simulators and on real ears is dependent on the device under test.

Translations from DRP to LRP should be provided by the manufacturer, and these should be used if available. Otherwise, the translations in 13.1 should be used.

The type of ear simulator used shall be stated.

Modifications to the ear simulator shall not be made.

5.2.1 Type 3.3 ear simulator

The Type 3.3 ear simulator shall comply with the specifications given in ITU-T Recommendation P.57.

The Type 3.3 ear simulator comprises a Type 2 simulator with an anatomical pinna. It simulates the characteristics of the median human ear. It is intended to give results comparable to the typical listening experience for a wide variety of devices.

The Type 3.3 ear simulator is recommended for narrowband and wideband applications. It may also be used for superwideband and fullband applications.

5.2.2 Fullband ear simulator

NOTE—As of this publication, there is no comprehensive standard for a fullband ear simulator. Standardization work by ITU-T in this area is ongoing. In the absence of an applicable standard, this subclause is provisional.

A fullband ear simulator includes an anatomical pinna. It simulates the characteristics of the human ear across the entire range of hearing. It should yield results comparable to the typical listening experience for a wide variety of devices from 20 Hz to 20 kHz.

A fullband ear simulator is recommended for all applications from 20 Hz to 20 kHz. It is preferred for superwideband and fullband devices.

If a fullband ear simulator is used for any tests, the manufacturer and model shall be stated.

5.2.3 Alternative ear simulators

If any other alternative ear simulator is used, the requirements of Annex C shall be met. Once the choice of ear simulator is made, it shall be used for all measurements on a particular device. The choice of ear simulator and positioning method shall be clearly stated in all test reports.

5.3 Mouth simulators

The purpose of mouth simulators is to test terminals and subsystems with microphones under conditions that most closely approximate typical use by real persons. The mouth simulator is generally installed in a HATS. It shall comply with the specifications given in ITU-T Recommendation P.58, unless the applicable performance specification requires or allows an alternative. See Annex C for more information about alternative mouth simulators.

ITU-T Recommendation P.58 defines the sound field in front of the lip plane of the mouth simulator but not behind it. However, formal studies (“Behind the Lip Plane Sound Field Study” [B4]¹⁷) have shown that at least one implementation of the mouth has a sound field distribution that closely approximates the sound field behind the lip plane of a real human head. The investigated region extends from behind the lip plane to the ear simulator and is equal to or greater than 5 mm above the surface of the HATS cheek. Therefore, HATS is suitable for testing terminals and subsystems with microphones that are positioned in front of the lip plane, as well as between the lip plane and the ear.

5.4 Test fixture

The purpose of the test fixture is to position a terminal or subsystem so that it may be tested under conditions that closely approximate typical use by real persons.

The preferred test fixture is a HATS that complies with ITU-T Recommendation P.58 and ITU-T Recommendation P.64, Annex E, unless the applicable performance specification requires or allows an alternative. The HATS configuration, neck position (if adjustable), the use of absorptive clothing, and any other relevant parameters shall be reported.

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

5.5 Test room

Unless otherwise specified, all measurements shall be performed in a free field (reflection free) room. (This requirement may be relaxed if it is shown that the accuracy of the measurement is not impaired or the behavior of the device under test is not changed.) In addition, a mixed field room is required for measurements in which loudspeakers are used to introduce noises as impairments. All test rooms shall have a low background noise level. The device under test shall be isolated from reflections and mechanical disturbances that could cause significant error. Minimize reflections caused by the test fixture.

¹⁷ The numbers in brackets correspond to those of the bibliography in Annex Q.

5.5.1 Test room noise level

The background noise requirements in this clause apply to any test room. (These requirements may be relaxed if it is shown that the accuracy of the measurement is not impaired or the behavior of the device under test is not changed.)

The background noise level in the test room shall not exceed 29 dB, A-weighted.

Background noise shall be measured from 20 Hz to 20 kHz. A microphone with a noise floor not exceeding 20 dB, A-weighted, is sufficient to show that the room noise is 29 dB or less. However, to measure levels with an error less than 0.5 dB, the microphone shall have a noise floor at least 10 dB less than the level to be measured.

It is also advisable to measure the room noise spectrum in 1/12th octave or finer resolution to identify strong single frequency noises, such as those from fans or other equipment. It may be possible to minimize some of this noise by switching off equipment.

If impairment noises are to be introduced as part of any test, the following requirements also apply:

- a) The unweighted room noise level should be at least 10 dB less than the unweighted level of any introduced noises.
- b) The room noise level in any 1/3rd octave band included in the test should be at least 10 dB less than the noise level in the same 1/3rd octave band of any introduced noises.

5.5.2 Free field test room

The free field test room shall be sufficiently free of reflections. There shall be no large objects within 1 m of the device under test. Deviation from free field characteristics due to the influence of reflections, including reflections from tripods or other hardware used for positioning, shall not exceed ± 1.5 dB below 800 Hz. Errors above 800 Hz shall not exceed ± 1.0 dB.

5.5.3 Mixed field test room

A mixed field test room uses several loudspeakers to introduce calibrated sounds arriving at the test point from many directions.

The test point is called the DFTP. The DFTP should be near, but not exactly at, the geometric center of the room. There should be no obstacles, including the loudspeakers, within 0.5 m of the DFTP.

A room that has both reflecting and absorbing surfaces is preferred. Some diffusing surfaces may also be used. At least eight speakers and uncorrelated sources, and possibly a subwoofer, are usually required to meet the acoustical specifications listed here.

The bandwidth of the test signals played from the loudspeakers should include at least the 1/3rd octave bands from 100 Hz to 10 000 Hz, or the bandwidth of the device under test, whichever is greater.

An anechoic room may be used as an alternative, but the results will be different.

NOTE—Every loudspeaker installed in an anechoic room also adds reflections, so these should be minimized and taken into account for all measurements to be performed in the room.

Regardless of the type of room, loudspeaker configuration, or signal processing used, the requirements for spectrum uniformity, directionality, and clarity shall be fulfilled (5.5.3.2 to 5.5.3.4). Documentation of the test room should include both a physical description and the measured acoustic parameters.

5.5.3.1 Loudspeaker considerations

The minimum distance from speaker to test point should be 1 m.

Loudspeaker equalization is generally needed to fulfil the requirements stated in this subclause. The recommended procedure is to equalize each speaker separately and then apply any necessary small correction when all speakers are active. Equalization in 1/12th octaves is recommended. (Equalization in smaller bandwidths usually introduces audible artifacts.) Although the requirements are based on 1/3rd octave measurements, 1/12th octave equalization generally results in better combination when all speakers are active.

Coaxial loudspeakers are preferred. The equalized loudspeakers should be able to produce the required bandwidth and level, for the spectrum of any signal to be used, without significant distortion. Equalization can add 10 dB or more to the signal at some frequencies, so the output level capability of the speakers should take that into account.

When loudspeakers are placed near room boundaries, strong reflections can be created. They can be difficult to equalize at the DFTP and impossible to equalize for many points simultaneously. One solution is to use loudspeakers that fit snugly against the room boundaries.

Uncorrelated signals are needed. When the signal is any type of stationary random noise, such as pink noise, independent noise generators can produce any number of completely uncorrelated signals. Uncorrelated signals can also be implemented by using several independent noise recordings. An example is babble noise, composed of several independent speech signals.

When the signal is nonstationary noise, such as many real recorded sounds, small but different delays can be inserted into each signal path. The individual delays may be adjusted as needed for the best results, but no delay should exceed 50 ms. The delays have the effect of partly decorrelating the signals at the expense of introducing some audible artifacts. There may also be smearing of the temporal features of the signal that reduce clarity.

Any other method of decorrelating signals may also be used, provided minimal audible artifacts are introduced and the requirements below are met.

5.5.3.2 Test for spectrum uniformity

Spectrum uniformity shall be measured using a 6.25 mm (1/4 in) pressure microphone pointed at the ceiling. The individual factory-calibrated frequency response of the microphone, if available, shall be accounted for. The tests shall be performed at the DFTP, with no HATS or other unnecessary objects present. The multiple loudspeaker configuration(s) to be used for testing shall be used as the sound source. All equalization and signal processing shall be active.

The spectrum at the DFTP shall meet its target within ± 0.5 dB in each 1/3rd octave band for all relevant spectra and levels. Equalization may be used to meet this requirement. The spectrum at DFTP, $G_{DFTP}(f)$, 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 $G_{DFTP}(f)$ by more than ± 3 dB in each 1/3rd octave band.

5.5.3.3 Test for directionality

Directionality of sounds arriving at the DFTP shall be measured in 1/3rd octave bands with a cardioid microphone that has a free field rejection of at least 10 dB front to back in each relevant frequency band from 250 Hz to 10 kHz. Neither super-cardioid nor hyper-cardioid microphones shall be used. A microphone with a diaphragm diameter of 18.75 mm (3/4 in) or less is recommended since this size generally has more uniform directional characteristics across a wide frequency range.

The sound field shall be considered to approximate random incidence (no directionality) if the variation is within the tolerances in Table 2. The tests are performed at the DFTP, with no HATS or other objects present. The multiple loudspeaker configuration(s) to be used for testing shall be used as the sound source. All equalization and signal processing shall be active.

The microphone is rotated about the DFTP through 360° in each of three perpendicular planes. Measurements shall be made in 15° 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 each band.

Table 2—Variation from random incidence allowable versus microphone rejection

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

5.5.3.4 Test for clarity

The clarity (either C_{50} or C_{80}) shall be measured in octave bands using a 6.25 mm (1/4 in) pressure microphone pointed at the ceiling. The individual factory-calibrated frequency response of the microphone, if available, shall be taken into account. The tests are performed at the DFTP, with no HATS or other objects present. The multiple loudspeaker configuration(s) to be used for testing shall be used as the sound source. All equalization and signal processing shall be active.

Note that in most room acoustics standards, clarity is defined based on a measurement using one omnidirectional sound source because the room is the object under test. In a mixed field room used for testing with multiple loudspeakers, the object under test is the sound field generated by all the loudspeakers in the room. This is a different situation because the number of loudspeakers and their directivity strongly influence clarity.

NOTE—Reverberation time is sometimes used to specify mixed field rooms. The reverberation time may be of interest, but it only applies to the room itself. Clarity is more relevant because it accounts for the room, the loudspeakers, and the signal processing.

Clarity is defined by Equation (1):

$$C = 10 \times \log \left(\frac{\int_0^{t_s} p^2(t) dt}{\int_{t_s}^{\infty} p^2(t) dt} \right) \quad (1)$$

where

$p^2(t)$ is the sound pressure as a function of time (pressure impulse response)

t_s is 50 ms for C_{50} and 80 ms for C_{80} . It is the time separating the direct sounds and early reflections from the reverberant sound, sometimes called “split time.”

The average of C_{50} or C_{80} over the 500 Hz, 1000 Hz, and 2000 Hz octave bands shall be greater than zero.

5.5.4 Test configurations defined in other standards

ITU-T Recommendation P.570 covers several other standard methods for creating sound fields in laboratory conditions. It also includes general suggestions about adapting the methods to particular applications as well as references to databases containing recorded background noise signals.

NOTE—As an international standard, ITU-T Recommendation P.570 is suggested as a successor to ETSI ES 202 396-1 and ETSI TS 103 224 [B6], which cover some of the same topics.

5.5.5 Reference corner

The reference corner is one specialized 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 11. 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.

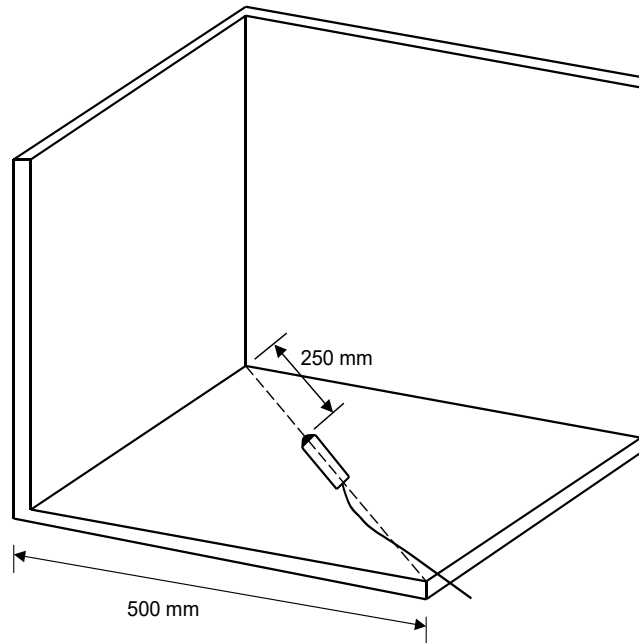


Figure 11—Reference corner for echo, howling, and stability tests

6. Codecs, interfaces, and test circuits

A terminal connects to a network and provides two-way voice communication. VoIP telephones and analog telephones are examples. A subsystem connects to a terminal. Analog headsets and USB headsets are examples. Subclause 6.1 through 6.7 define the means for interfacing both terminals and subsystems.

Digital terminals and subsystems require interfaces for applying and measuring test signals as well as for communication protocols. Digital interfaces shall incorporate a hardware or virtual reference codec.

Analog terminals require feed circuits, loop simulators, and defined termination for applying and measuring test signals. Analog subsystems use test circuits for biasing the device and testing.

Details of codecs, interfaces, and test circuits shall be stated with the measurement results.

6.1 Reference codecs

Reference codecs are used for testing digital terminals and subsystems.

6.1.1 Narrowband codecs

ITU-T Recommendation G.711 is generally the specified reference codec for testing the narrowband mode. This standard defines μ -law for encoding voice frequency signals in North America, as well as A-law for use in many other countries. For other coding schemes, an appropriate codec should be used, and tests performed over the corresponding bandwidth.

In addition, narrowband reference codec characteristics, such as attenuation versus frequency distortion and idle channel noise should meet or exceed characteristics specified in ITU-T Recommendation G.712. The quantizing distortion should approach the theoretical limits specified in Annex A of ITU-T Recommendation O.133.

6.1.2 Wideband codecs

Several wideband codecs are being used, including ITU-T Recommendation G.722, and low bit rate vocoders, such as ITU-T Recommendations G.722.1, G.723.1, and G.729. However, the L16-256 codec is generally the specified reference codec for testing the wideband mode. It is 16-bit, 16 kHz linear PCM coding, resulting in 256 kbit/s. For other coding schemes, an appropriate codec should be used, and tests should be performed over the corresponding bandwidth.

6.1.3 Other reference codecs

The test methods in this standard are also applicable to digital devices using codecs other than the reference codecs specified in 6.1.2. If a different codec is used, the means to interface to and from this codec should be provided by the device manufacturer or relevant standard. Furthermore, the 0 dBm0 level relative to full scale shall be defined. (Example: For a ITU-T G.722 codec, 0 dBm0 is defined as 9 dB below full scale. However, in practice, it is often implemented as 3.17 dBm0 below full scale. See ITU-T Recommendation G.100.1.) If the 0dBm0 level is not provided, then 3.17 dB below full scale may be assumed. In most computer-based applications for speech communication, use 3.17 dB below full scale.

The 0 dBm0 level has a significant effect on measurements. The level being used relative to digital full scale shall be stated.

6.1.4 Conversion relationships

For most codecs $-20 \text{ dBm}_0 = -26.17 \text{ dBFS}$. For ITU-T Recommendation G.722 and a few others, that relationship is different. Throughout this standard, the $-20 \text{ dBm}_0 = -26.17 \text{ dBFS}$ relationship is assumed. Please see the relevant codec specification. For a hardware implementation of a digital-to-analog (d/a) converter, a digital test sequence (DTS) representing the PCM equivalent of an analog sinusoidal signal whose rms value is 3.17 dB below the maximum full load capacity of the codec shall generate 0 dBm in a 600Ω load. This conversion is used in ITU-T Recommendation G.711 and L16-256.

For a hardware implementation of an analog-to-digital (a/d) converter, a 0 dBm signal from a 600Ω source shall give the DTS representing the PCM equivalent of an analog sinusoidal signal whose rms value is 3.17 dB below the maximum full-load capacity of the codec. Here again, this conversion is used as in ITU-T Recommendation G.711 and L16-256.

See H.4 for more details on conversions between analog and digital.

6.2 Interfaces to digital terminals

6.2.1 General

Digital interfaces include a codec and other hardware and software for connection and control. In effect, we measure the terminal plus the codec. For example, if the stimulus is wideband, and the terminal is operating in narrowband mode, the roll-off due to the codec will appear in the measurement.

The interfaces in this subclause apply to terminals of any bandwidth as long as an appropriate codec is used. If the terminal supports multiple codecs, the terminal should be tested with each codec.

The digital terminal shall be connected to a test interface such as those shown in 6.2.2 through 6.2.4. Other test interfaces may be used for specific applications. All signal processing (DSP) features in the interface, such as echo cancellation, automatic gain control, noise reduction, equalization, or voice activity detection, shall be disabled. (See “dsp OFF” in Figure 12 through Figure 16.)

6.2.2 Hardware interfaces

If analog test equipment is used, the digital terminal under test shall be connected to the reference codec through an interface such as shown in Figure 12 and Figure 13. The interface shall provide all the signaling and supervisory sequences necessary for the terminal to work in all test modes. The interface shall also be capable of converting a digital stream to or from the terminal under test to a format compatible with the reference codec.

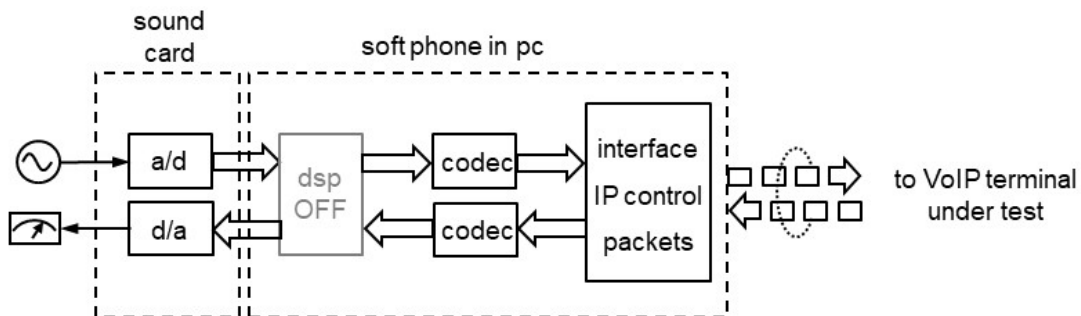


Figure 12—Hardware interface to VoIP softphone in PC

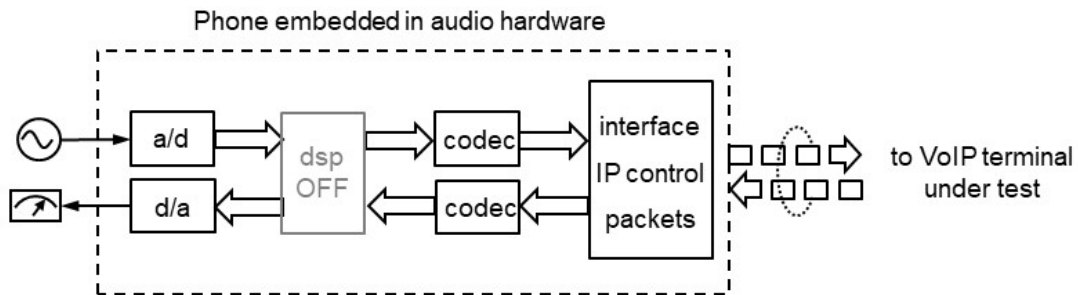


Figure 13—Hardware interface to VoIP phone embedded in audio hardware

6.2.3 Virtual interfaces

An interface can be implemented completely in software, as shown in Figure 14. The measurements are done entirely in the digital domain, since a reference codec is included in the softphone. Digital signals shall be referenced to the analog equivalent as defined in 6.1.4.

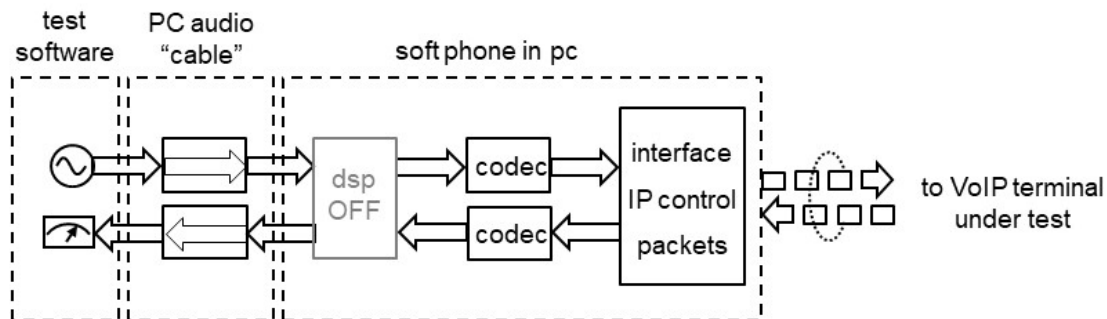


Figure 14—Virtual interface to VoIP softphone in PC

NOTE—A PC audio cable is software that allows a digital audio connection from one audio device to another.

6.2.4 Wireless interfaces

For wireless terminals, a radio link is also included, as shown in Figure 15 and Figure 16.

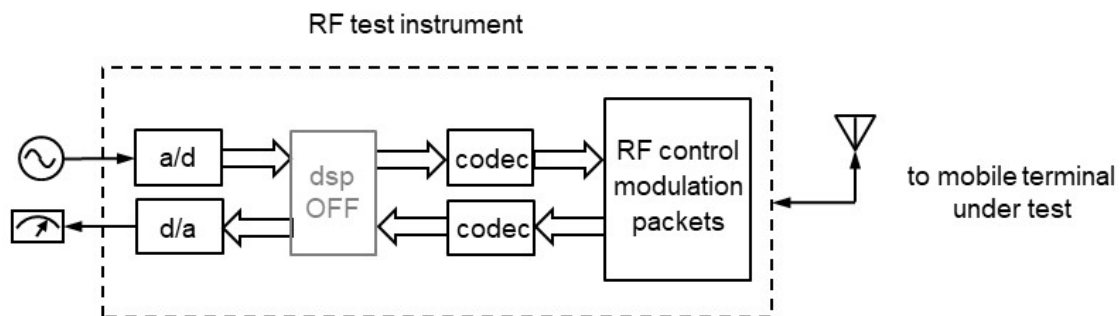


Figure 15—Hardware interface to radio-frequency (RF) test instrument

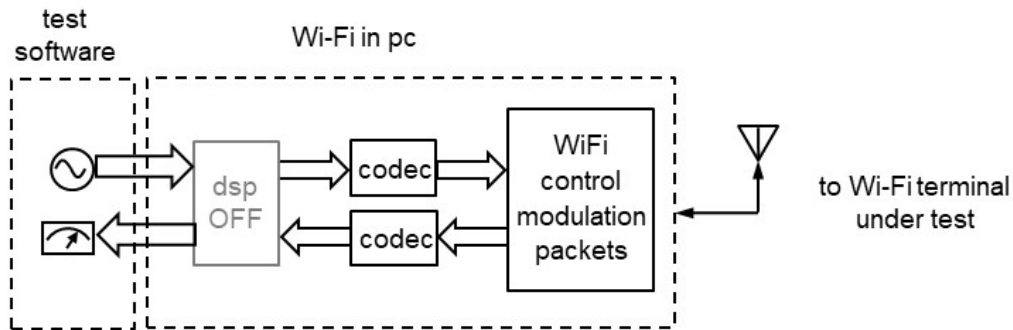


Figure 16—Virtual interface to Wi-Fi in PC

6.3 Interfaces to digital subsystems

6.3.1 General

The interfaces in this subclause apply to digital subsystems of any bandwidth as long as an appropriate codec is used. If the subsystem supports multiple codecs, the subsystem should be tested with each codec.

The digital subsystem shall be connected to a test interface such as those shown in 6.3.2 and 6.3.3. Other test interfaces may be used for specific applications.

All signal processing (DSP) features in the interface shall be disabled. (See “dsp OFF” in Figure 17 through Figure 20.)

6.3.2 Universal serial bus interface

The digital subsystem under test shall be connected using a direct digital interface as shown in Figure 17. The test software and PC interface shall provide all the signaling and supervisory sequences necessary for the subsystem to work in all test modes. In this case, a reference codec is included in the PC audio system. Digital signals shall be referenced to the analog equivalent as defined in 6.1.4.

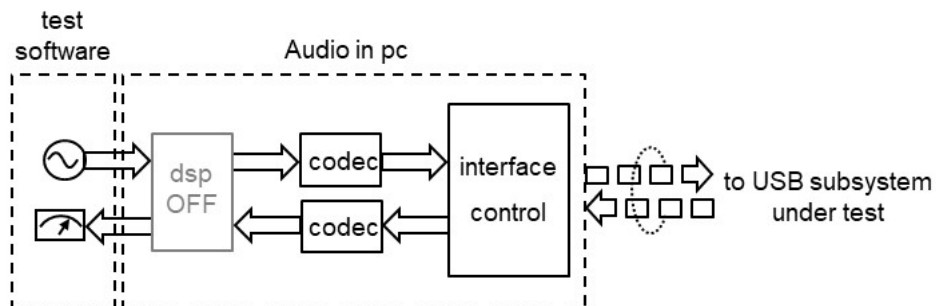


Figure 17—PC interface to USB subsystem

6.3.3 Wireless interfaces

For wireless subsystems, a radio link replaces the direct connection. The interface can use components in the PC, as shown in Figure 18 and Figure 19, or a standalone interface instrument, as shown in Figure 20.

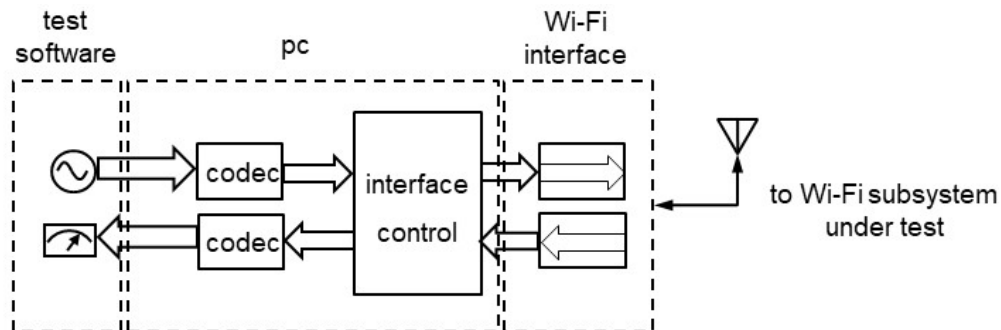


Figure 18—PC interface to Wi-Fi subsystem

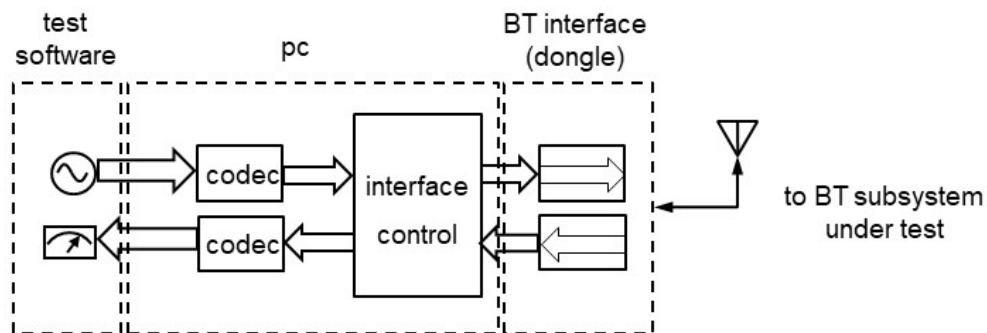


Figure 19—PC dongle interface to Bluetooth (BT) subsystem

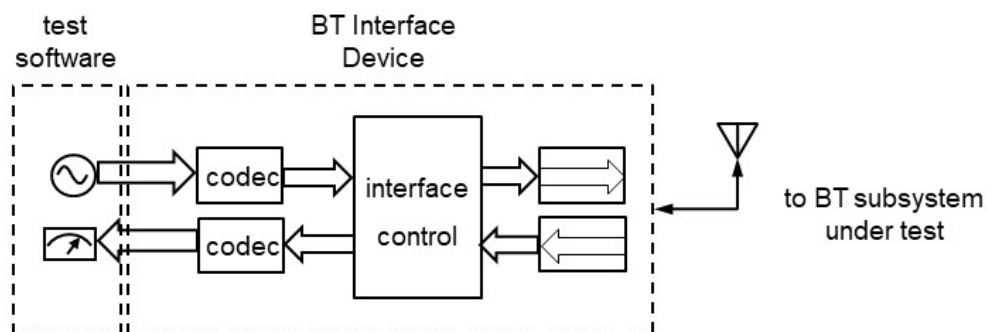


Figure 20—Hardware interface to BT subsystem

6.4 Digital impairments

The performance of a device can be influenced by transmission impairments of many kinds. There are many commercial units available to induce specific impairments which the terminal might encounter.

All tests shall be conducted without impairments. Receive and send speech quality tests (12.11) should be conducted with relevant impairments introduced and compared to results without impairments. It should be determined experimentally which other tests should be repeated with any specific impairment introduced.

6.4.1 Network delay

Network delay is the amount of time that elapses between the time a packet is transmitted and the time it is received. One-way network delays such as 50 ms, 150 ms, and 300 ms should be introduced.

6.4.2 Packet loss

Packet loss is simply the disappearance of a packet that was transmitted. Packet loss of at least 1% should be introduced.

6.4.3 Jitter

Jitter is a variation in network or device delay. Jitter of at least 30 ms should be introduced (IETF RFC 3550).

6.5 Test circuits for analog terminals

6.5.1 General

The analog terminal shall be connected to the test interface shown in Figure 21. The interface is made up of a send-to-receive switch, source and load impedances, dc feed circuit, and loop simulator. These components affect the performance of the analog terminal, so the values used should follow the relevant performance specification or match the conditions expected in use.

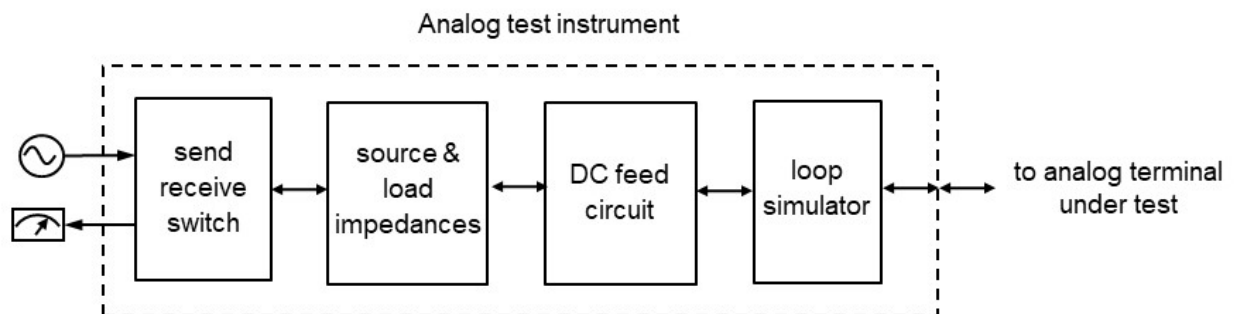


Figure 21—Hardware interface to analog terminal

6.5.2 Source and load impedances

The source and load impedances shall comply with the relevant performance specification. Otherwise, use $900\ \Omega$ for terminals connected to a central office network and $600\ \Omega$ for terminals connected to a private branch exchange (PBX) network.

6.5.3 DC feed circuits

A general-purpose dc feed circuit is shown in Figure 22. If available, parameters should be obtained from the applicable performance specification. If not, the following values should be used:

- $C \geq 100\ \mu\text{F}$.
- $L \geq 5\ \text{H}$ (each).
- $R = 400\ \Omega$, including resistance of inductors.
- $V = 50\ \text{V}$.
- A = ammeter used to measure current drawn by the terminal 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 the interface hardware. The insertion of a high-quality 1:1 audio transformer can usually prevent this. If a transformer is used, it shall be included during calibration and when determining the loss of the feed circuit.

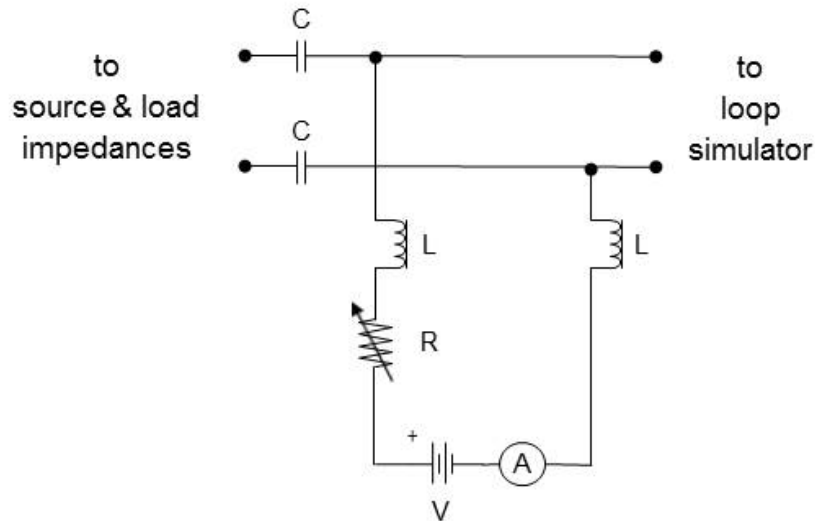


Figure 22—DC feed circuit for two-wire analog terminal

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 22 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 equipment with a $900\ \Omega$ source impedance directly to a $900\ \Omega$ 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 as follows in Equation (2).

$$\text{Feed circuit loss} = 20 \log \frac{\text{Voltage across } 900 \Omega \text{ resistor}}{\text{Voltage across feed bridge}} \quad (2)$$

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 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.

6.5.4 Loop simulator

An analog terminal should be tested with various lengths of cable or simulated cable. The test loop is inserted between the dc feed circuit and the terminal under test. The component values shall comply with the relevant performance specification. Otherwise, use the loop lengths for testing North American terminals: 0 km, 2.7 km, and 4.6 km (0 kft, 9 kft, and 15 kft) of 26 AWG nonloaded cable. The recommended loop simulator circuit is shown in Figure 23. Components for various lengths are shown in Table 3.

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

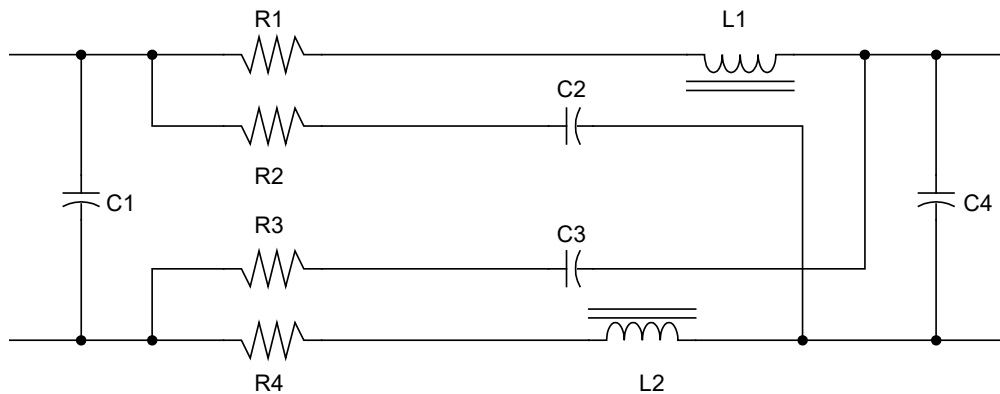


Figure 23—Loop simulator circuit

Table 3—Component values for 26 AWG cable

Component	0.305 km (1 kft)	0.914 km (3 kft)	1.83 km (6 kft)
R1, R4	41.7 Ω	124 Ω	249 Ω
R2, R3	109 Ω	174 Ω	312 Ω
C1, C4	3.77 nF	0.0113 μ F	0.0226 μ F
C2, C3	4.02 nF	0.0122 μ F	0.0255 μ F
L1, L2	96.0 μ H	0.336 mH	0.983 mH
NOTE 1—All values are $\pm 1\%$ tolerance.			
NOTE 2—2.7 km (9 kft) and 4.6 km (15 kft) can be made up of cascaded sections above.			

6.5.5 Parallel sets

The terminal should be tested with a parallel terminal 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, output level, and distortion, each measured with standard loop lengths.

The parallel terminal set simulator circuit shall have the VI curve as shown in Figure 24, ± 0.3 V over the current range of 0 mA 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 25.

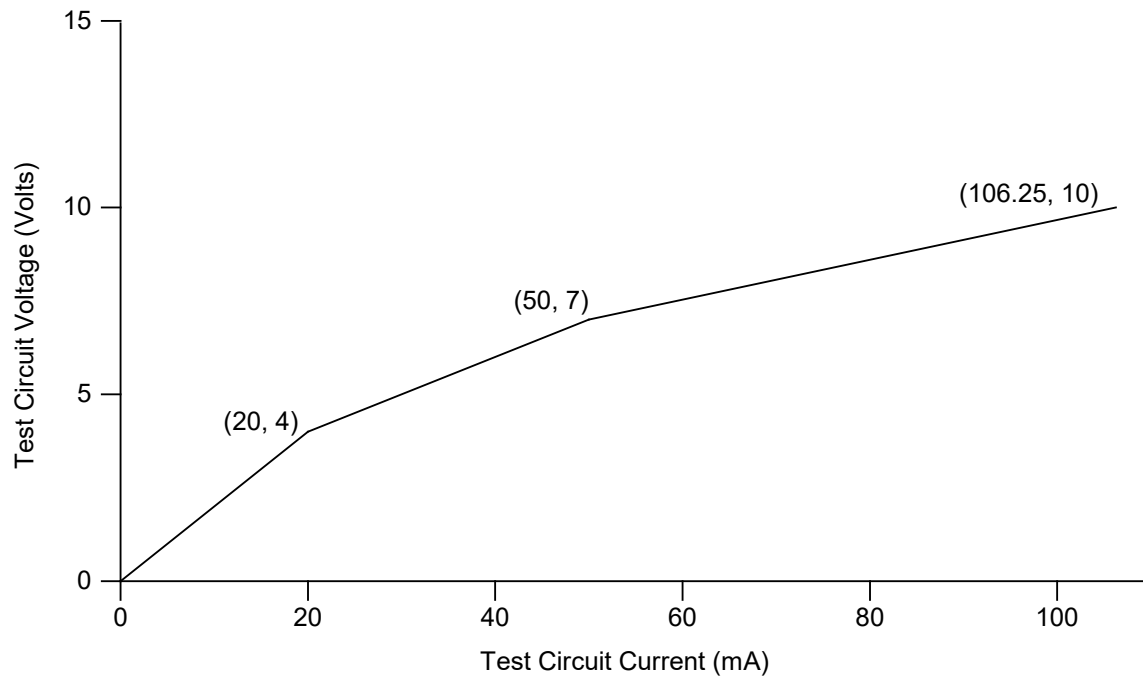


Figure 24—Parallel set test circuit VI curve

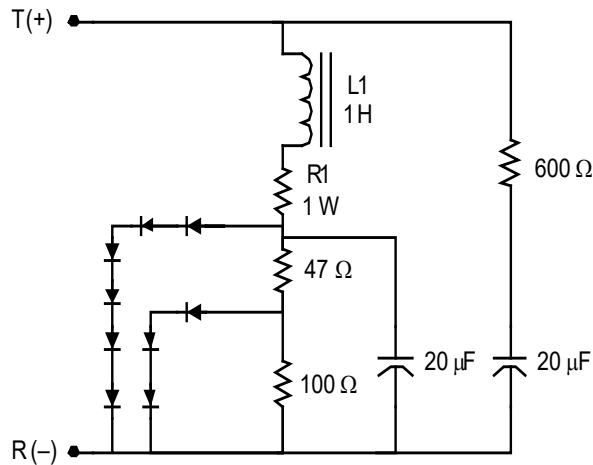


Figure 25—Parallel terminal set simulator

Other circuit features are as follows:

- Circuit is polarity sensitive.
- Use IN4004, or similar, for diode strings.
- L1 may be > 1 H.
- Resistor values shall be $\pm 1\%$.
- R1 + L1 coil resistance should be $53\ \Omega$.
- The performance of this circuit is dependent on the characteristics of the diodes used and temperature. The performance of the circuit shall be verified, and component values may need to be adjusted to meet the requirements.

6.5.6 Cordless range

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

6.6 Test circuits for analog subsystems

6.6.1 General

The analog handset or headset subsystem should be connected to the test circuit(s) described below. Other test circuits may be used for specific applications. Measurement setup and conditions should be stated.

6.6.2 Microphone test circuits

The test circuits are terminated into a load greater than $100\ \text{k}\Omega$. This termination is the SETP for measuring send output signals. This same termination is also the RETP for applying receive input signals. Other terminating loads may be substituted as defined by applicable performance specifications.

Figure 26 through Figure 28 show various microphones tested as part of analog subsystems or as standalone components.

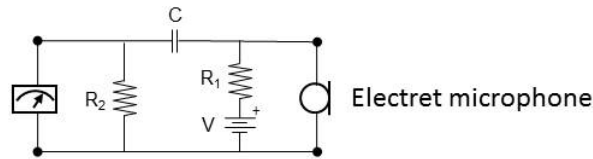


Figure 26—Electret microphone test circuit

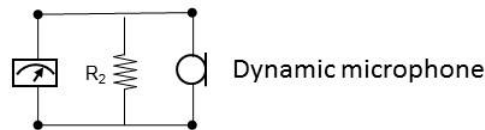


Figure 27—Dynamic microphone test circuit

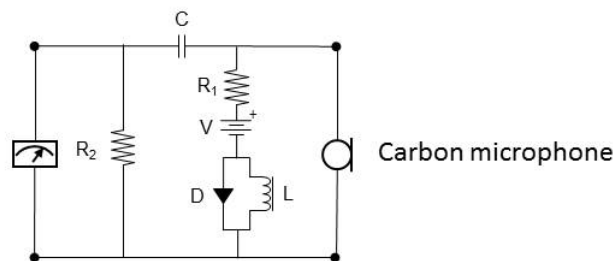


Figure 28—Carbon microphone test circuit

In the microphone test circuits of Figure 26 through Figure 28, the values for voltage V , capacitance C , resistances $R1$ and $R2$, inductance L , and diode D should simulate the range of operating parameters of the headset or handset interface. These values are intended to provide support for both dc and ac characteristics.

The effective load impedance provided by these test circuits shall be equal to the range of operating impedances of the headset or handset interface. For electret microphones, the effective load impedance $Z_L = (R1 \times R2) / (R1 + R2)$. This assumes C is large enough so that its impedance is small compared to $R1$ and $R2$ at the lowest frequency tested. In many cases, $R2$ is infinite, so the effective load impedance $Z_L = R1$.

6.6.3 Receiver test circuits

The effective source resistance $R3$ in the receiver test circuit of Figure 29 shall be equal to the specified nominal impedance of the receiver under test. $R3$ should take into account both the output impedance of the signal generator and any other added impedances. In case of a receiver system that needs to be powered, it should be connected to its intended host or suitable simulation.

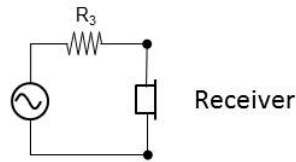


Figure 29—Receiver test circuit

6.7 Analog impairments

Terminal performance can be influenced by various conditions in the network to which a terminal is connected. Noise is the impairment of most interest.

6.7.1 Network noise

Network noise can affect nonlinear processes within an analog terminal. Receive and send frequency response (12.2.2 and 12.2.3) should be measured with network noises introduced. Network noise shall be approximated using pink noise with a bandwidth corresponding to the network (narrowband, wideband, etc.), inserted at the RETP, at levels from -90 dBm0(A) to -50 dBm0(A).

7. Test signals

The test signals should verify that the terminal, handset, or headset subsystem is characterized in realistic, stable, and well-defined states. They should verify that the unit remains stable during the measurement period, yet provide a suitable signal for the specific measurement. This is especially the case for devices with nonlinear processes such as compression or voice-activated switching (VOX) circuitry.

Unless otherwise stated, levels are specified as long-term rms averages during at least one complete period of the active part of the signal.

7.1 Speech test signals

The real speech signals provided with this standard are recommended for general use. They are described in Table 4 and are available in 269-2019_downloads.zip provided at the following URL:

<https://standards.ieee.org/downloads>

The signals extend up to and slightly beyond 20 kHz, so they are usable for test applications of any bandwidth.

For some applications, the male and female signals have a crest factor that is too great to allow playback at the “Loud” level (e.g., -10 dBm0) without clipping the device and/or the network. The compressed signals can be used at the “Loud” level because they have a reduced crest factor. The compressed signals were made by applying amplitude compression to the male and female wav files. The compressed signals without pauses are a legacy from the 2010 publication and should not be used unless a performance specification requires otherwise.

NOTE—The spectra of the compressed and uncompressed signals are not identical.

The spectrum of the acoustic or electrical test signal should match the spectrum of the speech in the original file. When used as electrical test signals, no further equalization is usually required. When used

with a mouth simulator to produce acoustic test signals, equalization is typically needed. Band limiting of the test signal may be required by the measurement or performance standard.

The files are provided at a 48 kHz sampling rate, which should be used whenever practical. The 44.1 kHz files may be used as an alternative.

Table 4—IEEE 269 speech test signals

File name	Length (s)	Crest factor (dB)
IEEE_269_2019_Male_mono_48kHz.wav	12.0	18.8
IEEE_269_2019_Male_compressed_mono_48kHz.wav	12.0	13.7
IEEE_269_2019_Male_compressed_no pauses_mono_48kHz.wav	10.0	14.5
IEEE_269_2019_Female_mono_48kHz.wav	10.0	19.3
IEEE_269_2019_Female_compressed_mono_48kHz.wav	10.0	13.9
IEEE_269_2019_Female_compressed_no pauses_mono_48kHz.wav	8.5	13.8
IEEE_269_2019_Male_mono_44.1kHz.wav	12.0	18.8
IEEE_269_2019_Male_compressed_mono_44.1kHz.wav	12.0	13.7
IEEE_269_2019_Male_compressed_no pauses_mono_44.1kHz.wav	10.0	14.7
IEEE_269_2019_Female_mono_44.1kHz.wav	10.0	19.4
IEEE_269_2019_Female_compressed_mono_44.1kHz.wav	10.0	13.9
IEEE_269_2019_Female_compressed_no pauses_mono_44.1kHz.wav	8.5	13.7

7.1.1 IEEE 269 male speech

IEEE_269-2019_Male_mono_48_kHz.wav is the recommended test signal unless specified otherwise in the relevant measurement clause. This signal consists of four sentences spoken by four male talkers. Male speech is chosen as the default because it covers most of the spectrum included in female speech, and as much as one octave lower.

IEEE_269-2010_Male_compressed_mono_48_kHz.wav is a compressed version of this signal. It is recommended for measurements at digital levels from -15 dBm0 to -10 dBm0 or at analog levels above -14 dBV, or at any other levels where clipping of the codec or any part of the test system is likely to occur.

The compressed version of this signal without pauses is a legacy from the 2010 publication and should generally not be used unless a performance specification requires otherwise.

The 44.1 kHz versions of these files may be used as an alternative.

7.1.2 IEEE 269 female speech

IEEE_269-2019_Female_mono_48_kHz.wav is provided as an alternative. It should be used to determine whether the measured results are similar for both male and female speech. This signal consists of four sentences spoken by three female talkers.

IEEE_269-2010_Female_compressed_mono_48_kHz.wav is a compressed version of this signal. It is recommended for measurements at digital levels from -15 dBm0 to -10 dBm0 or at analog levels above -14 dBV, or at any other levels where clipping of the codec or any part of the test system is likely to occur.

The compressed version of this signal without pauses is a legacy from the 2010 publication and should generally not be used unless a performance specification requires otherwise.

The 44.1 kHz files may be used as an alternative.