

# IEEE Standard Method for Measuring Transmission Performance of Handsfree Telephone Sets

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

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**Abstract:** Techniques for objective measurement of electroacoustic and voice switching characteristics of analog and digital handsfree telephones (HFTs) are provided. Due to the various characteristics of HFTs and the environments in which they operate, not all of the test procedures in this standard are applicable to all HFTs. Application of the test procedures to atypical HFTs should be determined on an individual basis.

**Keywords:** Analog handsfree telephone, digital handsfree telephone, electroacoustic transmission characteristics, handsfree telephone, voice switching characteristics, voice transmission characteristics.

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# Introduction

(This introduction is not part of IEEE Std 1329-1999, IEEE Standard Method for Measuring Transmission Performance of Handsfree Telephone Sets.)

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 handsfree telephone sets. This standard is generally in agreement with the 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: Secretary, IEEE Standards Board, The Institute of Electrical and Electronics Engineers, Inc., 445 Hoes Lane, Piscataway, NJ 08855.

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# IEEE Standard Method for Measuring Transmission Performance of Handsfree Telephone Sets

## 1. Overview

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

HFTs present a complex problem for accurate and repeatable measurements of the device characteristics. In order to meet these measurement goals, the reader should be aware of the nonlinear and time-variant characteristics of an HFT. This standard describes test signals and corresponding analysis methods, which can be chosen to ensure the HFT is in a well-defined operating state during testing.

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

### 1.1 Scope

This standard provides the techniques for objective measurement of electroacoustic and voice switching characteristics of analog and digital HFTs. Due to the various characteristics of HFTs and the environments in which they operate, not all of the test procedures in this standard are applicable to all HFTs. Application of the test procedures to atypical HFTs should be determined on an individual basis.

### 1.2 Purpose

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

### 1.3 Contents of this standard

This is a brief summary of the remaining clauses contained in this standard. The primary measurement procedures appear in Clause 8, Clause 9, Clause 10, and Clause 11 of this document. Attached annexes contain additional information or details of procedures referred to from within the relevant clause.

Clause 2, Clause 3, and Clause 4 provide references, definitions, and acronyms, which will be useful in executing the tests of this standard. These clauses provide a background in the terminology used for HFT testing.

Clause 5 details the most common analysis techniques used to make the electroacoustic and switching measurements. The clause explains the advantages and disadvantages of each technique in relation to the stimulus signals chosen for testing.

Clause 6 details the test signals used to place the HFT in a well-defined state for the measurements. This clause explains the use of these signals to maintain a stable HFT state as well as provide a suitable measurement signal.

Clause 7 details the test equipment, test environment, and impairments. The test equipment includes the interfaces and transducers. The test environment includes both the acoustical and physical characteristics of the test space. Impairments include both acoustic and network conditions.

Clause 8 details the calibration procedures needed to ensure that the equipment is in a known state. Calibration of the electrical interfaces and acoustic transducers is explained.

Clause 9 details the various send and receive transmission test procedures. This clause includes procedures for positioning the HFT for testing and generating reports of the transmission characteristics.

Clause 10 details the procedures and parameters involved in testing and classifying the voice switching characteristics of HFTs. Various environmental conditions are explained in reference to the operation of a voice switched HFT.

Clause 11 details the procedures and parameters unique to testing an HFT equipped with an acoustic echo canceller (AEC). Characteristics unique to AEC HFTs are presented. Also, environment and interface conditions specific to AECs are described.

### 1.4 How to use this standard

The following sequence is suggested when referencing the contents of this standard prior to setting up a test sequence:

- a) Choose the acoustic environment (see Clause 7). Electroacoustic measurements on HFTs generally should be made in an environment with a controlled background noise level. In addition, either the test environment or the test method should minimize or eliminate the influence of reflections. Subclause 7.3 provides guidance in the selection and setup of the test environment. Under certain circumstances, measurements may have to be made in a “real room environment.” That is, in a room having acoustic characteristics similar to those expected to exist in a room in which the HFT may be used.
- b) Choose the test signal based on HFT behavior (see Clause 6). Selection of the correct test signal(s) is critical since different HFTs can respond differently to many of the possible test signals. The choice will be a balance between one that correctly stimulates the detection circuits of the HFT, and one that is suitable for the specific measurement.

- c) Choose an analysis method based on the test signal (see Clause 5). Each analysis technique has inherent advantages and disadvantages. A particular method may be better suited for use with certain stimulus signals. In fact, certain methods rely upon the use of a synchronized or otherwise unique signal. Within a given test, different test signal types (i.e., deterministic, random, speech-like, compound) may be required.
- d) Set up the hardware, fixtures, test environment, and impairments (see Clause 7). Different equipment will be required depending upon whether analog or digital sets are being tested and which test signals and analysis methods are chosen.
- e) Calibrate the system (see Clause 8). Follow the checkout and calibration procedures provided by instrument manufacturer(s), or existing relevant standards, to verify proper operation and calibration of all equipment, as appropriate. Calibrate the test signal spectrum and level for conducting send and receive/echo measurements.
- f) Perform transmission measurements (see Clause 9). Procedures are provided for measuring parameters affecting the send and receive performance of HFTs.
- g) Perform voice switching measurements (see Clause 10 and Clause 11). Refer to Clause 10 to select the procedures and parameters to be used in testing the voice switching characteristics of HFTs. The manner in which the HFT handles the transmission of signals in both directions can be classified into different groups. This clause provides guidance in test set up, test environment, test equipment, and procedures. Clause 11 provides guidance for the testing of the unique characteristics of HFTs with AEC circuitry. The methods described will effectively evaluate AEC performance.

For each test, the test environment, position of HFT, test signal, analysis method, frequency range and resolution, and other relevant conditions shall be reported.

## 2. References

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

ANSI S1.1-1994, American National Standard Acoustical Terminology.<sup>1</sup>

ANSI S1.4-1983 (Reaff 1997), American National Standard Specification for Sound Level Meters.

ANSI S1.6-1984 (Reaff 1997), American National Standard Preferred Frequencies, Frequency Levels, and Band Numbers for Acoustical Measurements.

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

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

ANSI S3.36-1985 (Reaff 1996), American National Standard Specification for a Manikin for Simulated insitu Airborne Acoustic Measurements.

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<sup>1</sup>ANSI publications are available from the Sales Department, American National Standards Institute, 11 West 42nd Street, 13th Floor, New York, NY 10036, USA (<http://www.ansi.org/>).

IEEE Std 269-1992, IEEE Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets.<sup>2</sup>

IEEE Std 661-1979 (Reaff 1998), IEEE Standard Method for Determining Objective Loudness Ratings of Telephone Connectors.

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

ISO 3:1973, Preferred numbers - Series of preferred numbers.<sup>3</sup>

ITU-T Recommendation G.122: 1993, Influence of national systems on stability and talker echo in international connections.<sup>4</sup>

ITU-T Recommendation G.167: 1993, Acoustic echo controllers.

ITU-T Recommendation O.41: 1994, Psophometer for use on telephone-type circuits.

ITU-T Recommendation P.50: 1993, Artificial voices.

ITU-T Recommendation P.51: 1996, Artificial mouth.

ITU-T Recommendation P.56: 1993, Objective measurement of active speech level.

ITU-T Recommendation P.57: 1996, Artificial ears.

ITU-T Recommendation P.58: 1996, Head and torso simulator for telephony.

ITU-T Recommendation P.59: 1993, Artificial conversational speech.

ITU-T Recommendation P.79: 1993, Calculation of loudness ratings for telephone sets.

ITU-T Recommendation P.340: 1996, Transmission characteristics of hands-free telephones.

ITU-T Recommendation P.501: 1996, Test signals for use in telephony.

TIA/EIA-579-A-1998, Telecommunications Telephone Terminal Equipment Transmission Requirements for Digital Wireline Telephones.<sup>5</sup>

### 3. Definitions

These definitions apply specifically to measurements of the transmission performance of HFT sets, and may not be applicable to other disciplines. For definitions not covered, see ANSI S1.1-1994 and the IEEE Dictionary of Electrical and Electronic Terms.

<sup>2</sup>IEEE publications are available from the Institute of Electrical and Electronics Engineers, 445 Hoes Lane, P.O. Box 1331, Piscataway, NJ 08855-1331, USA (<http://www.standards.ieee.org/>).

<sup>3</sup>ISO publications are available from the ISO Central Secretariat, Case Postale 56, 1 rue de Varembe, CH-1211, Genève 20, Switzerland/Suisse (<http://www.iso.ch/>). ISO publications are also available in the United States from the Sales Department, American National Standards Institute, 11 West 42nd Street, 13th Floor, New York, NY 10036, USA (<http://www.ansi.org/>).

<sup>4</sup>ITU-T publications are available from the International Telecommunications Union, Place des Nations, CH-1211, Geneva 20, Switzerland/Suisse (<http://www.itu.int/>).

<sup>5</sup>TIA/EIA publications are available from Global Engineering Documents, 15 Inverness Way East, Englewood, Colorado 80112, USA (<http://www.global.ihs.com/>).

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

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

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

**3.4 attack time:** *See:* **build-up time.**

**3.5 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 send to transmit, then the larger of the two is the attenuation range.

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

**3.7 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.**

**3.8  $C_{50}$ :** The difference in dB between the first 50 ms of reverberant decay energy and the remaining decay energy from 50 ms and later. It is a member of the class of acoustic measurements commonly known as early-to-late ratios.

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

**3.10 decay time:** *See:* **hang-over time.**

**3.11 double talk (DT):** Two talkers speaking simultaneously in opposite transmission directions.

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

**3.13 ear reference point (ERP):** A virtual point for geometric reference located at the entrance to the listeners ear, traditionally used for calculating telephonometric loudness ratings.

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

**3.15 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.

**3.16 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.

**3.17 echo return loss (ERL):** *See:* **terminal coupling loss.**

**3.18 feed circuit:** An electrical circuit for supplying dc power to a handsfree telephone set and an acdc path between the handsfree telephone and a terminating circuit.

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

**3.20 full duplex:** An operating condition which 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: 1996.<sup>6</sup>

**3.21 half duplex:** An operating condition that allows communication in either send and receive directions with more than 20 dB switched loss in either direction.

NOTE—Classified as Type 3 in ITU-T Recommendation P.340: 1996.

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

**3.23 handsfree telephone (HFT):** A device for connection to a telephone network capable of two-way voice communication without close coupling to the user's mouth or ear.

**3.24 handsfree telephone test circuit:** An assembly consisting of a handsfree telephone set(s) and interface(s) as may be required to realize simulated partial telephone connections.

**3.25 hang-over 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.**

**3.26 head and torso handsfree reference point (HATSHFRP):** One of the points numbered 11–17 in ITU-T Recommendation P.58: 1996, 50 cm from the lip plane of the HATS mouth simulator. The HATSHFRP for a particular test and calibration shall be chosen so the angle most closely approximates the angle from the mouth to the HFT microphone under test.

**3.27 head and torso simulator (HATS):** A device that accurately reproduces the sound transmission and pick-up characteristics of the median head and torso of adult humans.

NOTE—See ANSI S3.36-1985 and ITU-T Recommendation P.58: 1996.

**3.28 loudspeaking telephone:** A telephone with handsfree receive but not handsfree send capability.

**3.29 mouth reference point (MRP):** The point on the reference axis of the mouth simulator, 25 mm in front of the lip plane.

**3.30 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: 1996.

**3.31 near-field test point (NFTP):** The acoustical measurement point located 1 cm from the center of the handsfree telephones (HFTs) loudspeaker along the axis of the speaker.

**3.32 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.

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

**3.34 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.

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

NOTE—Classified as Type 2 in ITU-T Recommendation P.340: 1996.

<sup>6</sup>Information on references can be found in Clause 2.

**3.36 receive:** The acoustic output of a handsfree telephone due to an electrical input.

**3.37 receive attenuation during double talk ( $A_{\text{RDT}}$ ):** Attenuation in the receive path, seen at the 50 cm test point (50TP), inserted during double talk. The send talker initiates the double talk.

**3.38 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 handsfree telephone in the receive direction.

NOTE—For further discussions on electrical interfaces, see IEEE Std 269-1992. For further discussions on wireless interfaces, see appropriate wireless standards.

**3.39 receive loudness rating directionality (RLRD):** Receive loudness rating versus angle around the handsfree telephone (HFT), normalized to the loudness rating at the 50 cm test point (50TP).

**3.40 receive speech front end clipping during double talk ( $T_{\text{RFDT}}$ ):** The length of time that speech undergoes syllabic clipping, as seen at the 50 cm test point (50TP), just after the onset of double talk. The receive talker initiates the double talk.

**3.41 recommended test position (RTP):** An acoustic test point, other than the 50 cm test point (50TP), that corresponds to the most appropriate user position for nonstandard desktop and nondesktop applications. This may be specified by the handsfree telephone (HFT) manufacturer.

**3.42 reference codec:** A well-defined analog-to-digital and digital-to-analog converter for testing digital telephones using analog test equipment.

NOTE—See Clause 5 of IEEE Std 269-1992 for details on a reference codec.

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

**3.44 release time:** *See: hang-over time.*

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

**3.46 rise time:** *See: build-up time.*

**3.47 send:** The electrical output of a handsfree telephone due to an acoustic input.

**3.48 send attenuation during double talk ( $A_{\text{SDT}}$ ):** Attenuation in the send path, seen at the send electrical test point, inserted during double talk. The receive talker initiates the double talk.

**3.49 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 handsfree telephone (HFT) in the send direction are accessed.

NOTE—For further discussion on electrical interfaces, see IEEE Std 269-1992. For further discussion on wireless interfaces, see appropriate wireless standards.

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

**3.51 send noise level:** Electrical noise at send electrical test point (SETP), measured in units of decibels relative to 1 mW (dBm), psophometrically weighted (dBmp).

NOTE—This is according to the method described in ITU-T Recommendation O.41: 1994.

**3.52 send loudness rating directionality (SLRD):** Send loudness rating versus angles around the handsfree telephone (HFT), normalized to the loudness rating at the 50 cm test point (50TP).

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

**3.54 sound pressure level:** Twenty times the logarithm to the base 10 of the ratio of the pressure of the sound to the reference pressure. For this standard, the reference pressure is normally 1 Pascal (Pa), and sound pressure levels are expressed in dB re 1 Pa (dBPa). When a reference pressure of 20  $\mu$ Pa is used, the sound pressure level will be expressed as dB SPL. Unless otherwise indicated, rms values of pressure are used. Most telephony acoustic measurements are referenced to 1 Pa. However, measurements such as receive noise and room noise are generally referenced to 20  $\mu$ Pa.

NOTE—0 dB Pa = 94 dB SPL, 0 dB SPL = 20  $\mu$ Pa, 1 Pa = 1 N/m<sup>2</sup>. An A-weighted (see ANSI S1.4-1983) sound pressure level in dB (dB SPL, A-weighted) is often abbreviated as dBA or dB(A).

**3.55 spectrum analyzer:** An instrument that measures the power of a signal in multiple frequency bands. The frequency bands may be constant bandwidth (e.g., fast Fourier transform (FFT) analyzer), or constant percentage bandwidth (e.g., real-time filter analyzer).

**3.56 switching time ( $T_S$ ):** Time taken to switch from one transmission direction to the other in alternating single talk conversation on a handsfree telephone (HFT).

**3.57 take-over time ( $T_T$ ):** Time taken to switch from one transmission direction to the other in double talk conversation on a handsfree telephone (HFT). 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.

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

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

**3.60 threshold level ( $L_{TH}$ ):** The minimum signal level necessary for removing insertion loss on a handsfree telephone (HFT).

**3.61 two-wire transmission:** A transmission scheme where the send and receive signals are carried in one pair of wires.

**3.62 weighted terminal coupling loss ( $TCL_W$ ):** Long-term, time-averaged terminal coupling loss, weighted in the frequency domain per ITU-T Recommendations G.122: 1993 and G.167: 1993.

**3.63 50 cm test point (50TP):** The acoustic test point 50 cm from the front center of the handsfree telephone (HFT) and 30 cm above the test table.

## 4. Abbreviations, acronyms, and symbols

The letter “A” is used for speech attenuations.

|            |                                                                               |
|------------|-------------------------------------------------------------------------------|
| A          | is the attenuation as a function of time (in dB)                              |
| $A_{RDT}$  | is the receive attenuation during double talk (in dB)                         |
| $A_{RADT}$ | is the receive attenuation during double talk, conversational average (in dB) |
| $A_{RTDT}$ | is the receive attenuation during double talk, as a function of time (in dB)  |
| $A_{SDT}$  | is the send attenuation during double talk (in dB)                            |
| $A_{SADT}$ | is the send attenuation during double talk, conversational average (in dB)    |

$A_{\text{STDT}}$  is the send attenuation during double talk, as a function of time (in dB)  
 $a_{\text{H}}$  is the attenuation range of the HFT (in dB)

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

$G_{\text{ERP}}(f)$  is the spectrum at Ear Reference Point (in dBPa)  
 $G_{\text{HFRP}}(f)$  is the spectrum at Handsfree Reference Point (in dBPa)  
 $G_{50\text{TP}}(f)$  is the spectrum at 50 cm Test Point (in dBPa)  
 $G_{\text{MRP}}(f)$  is the spectrum at Mouth Reference Point (in dBPa)  
 $G_{\text{RETP}}(f)$  is the spectrum at Receive Electrical Test Point (in dBV)  
 $G_{\text{SETP}}(f)$  is the spectrum at Send Electrical Test Point (in dBV)  
 $G_{(\text{MRP})(\text{SETP})}(f)$  is the cross-spectrum between MRP and SETP (in dBV/Pa)  
 $G_{(\text{RETP})(50\text{TP})}(f)$  is the cross-spectrum RETP and 50TP (in dBV/Pa)  
 $G_{(\text{RETP})(\text{SETP})}(f)$  is the cross-spectrum RETP and SETP (in dBV/V)

The letter “H” is used for frequency response.

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

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

$L_{\text{ERP}}$  is the level at ear reference point (in dBPa)  
 $L_{\text{HATSHFRP}}$  is the level at head and torso simulator handsfree reference point (in dBPa)  
 $L_{\text{HFRP}}$  is the level at Handsfree Reference Point (in dBPa)  
 $L_{50\text{TP}}$  is the level at 50 cm test point (in dBPa)  
 $L_{\text{MRP}}$  is the level at mouth reference point (in dBPa)  
 $L_{\text{RETP}}$  is the level at receive electrical test point (in dBV or dBmp)  
 $L_{\text{ROOM}}$  is the level of background noise in measurement environment (in dBPa)  
 $L_{\text{SETP}}$  is the level at send electrical test point (in dBV or dBmp)  
 $L_{\text{TH}}$  is the threshold level required to remove HFTs insertion loss (in dB)  
 $L_{\text{TH}_R}$  is the receive threshold level (in dBV)  
 $L_{\text{TH}_S}$  is the send threshold level (in dBPa)

The letter “S” is used for specially calculated sensitivities.

$S_{\text{DE}}$  is the sensitivity correction for HATS drum reference point to ear reference point  
 $\bar{S}_{\text{R}}$  is the average receive sensitivity (in dB Pa/V)  
 $\hat{S}_{\text{RO}}$  is the normalized receive sensitivity (in dB SPL/V at 1 m)  
 $\bar{S}_{\text{S}}$  is the average send sensitivity (in dB V/Pa)

The letter “T” is used for time measurements.

$T$  is the length of time window in simulated free-field techniques (in s)  
 $T_{\text{C}}$  is the convergence time of handsfree AEC algorithm (in s)  
 $T_{\text{FDT}}$  is the front-end syllabic clipping time during double talk (in s)  
 $T_{\text{SFDT}}$  is the send front-end syllabic clipping time during double talk (in s)  
 $T_{\text{RFDT}}$  is the receive front-end syllabic clipping time during double talk (in s)

$T_H$  is the hang-over time (in s)  
 $T_R$  is the build-up time (in s)  
 $T_S$  is the switching time (in s)  
 $T_T$  is the takeover time (in s)

“TCL” is used for terminal coupling loss.

TCL is the terminal coupling loss (in dB)  
 $TCL_T$  is the temporally weighted terminal coupling loss (in dB)  
 $TCL_{TDT}$  is the temporally weighted terminal coupling loss during double talk (in dB)  
 $TCL_{TST}$  is the temporally weighted terminal coupling loss during single talk (in dB)  
 $TCL_W$  is the frequency weighted terminal coupling loss (in dB)  
 $TCL_{WST}$  is the frequency weighted terminal coupling loss during single talk (in dB)

## 5. Test methods

### 5.1 General

Various analysis techniques are available for electroacoustic measurements. Each technique has inherent advantages and limitations. A particular method can be better suited for use with certain stimulus signals. Certain methods, in fact, rely upon the use of a synchronized or otherwise unique stimulus signal. This clause describes the most common techniques and their application to measurements of loudspeaking and HFTs.

### 5.2 FFT and cross-spectrum analysis

The Fourier transform is a mathematical operation that decomposes a time signal into its complex frequency components. The inverse Fourier transform reverses the process, reconstructing the time signal from its Fourier components. By applying the FFT algorithm to a sampled time signal, a spectrum can be computed. This is a parallel analysis resulting in a narrow band (constant bandwidth) frequency spectrum. Low-frequency resolution can be limited. Here, blocks of time data are analyzed. Care should be taken in the proper windowing of the data (e.g., Hanning, flat-top), overlap processing, and the number of averages to ensure an accurate analysis. The number of spectral lines and the record length determines the frequency resolution. The frequency range and time resolution are inversely related. Because the data is discrete, the highest frequency that can be measured is determined by the sampling frequency. Some degree of data processing is usually available in both the time domain and in the frequency domain. An FFT analyzer can also have a zoom capability, for increased frequency resolution across a restricted bandwidth.

#### 5.2.1 Dual-channel FFT

A dual-channel FFT analyzer performs simultaneous measurements of the HFT input and output. This type of measurement is optimized for system analysis. Most FFT analyzers calculate the frequency response from the cross spectrum and either the input or output autospectrum. In this way, different response estimators can be used to minimize noise at the system input or output. This also enables other functions such as coherence, phase, group delay, coherent power, and noncoherent power to be computed. Extensive data processing is normally available in both the time domain and in the frequency domain. It is possible to improve measurement signal-to-noise ratio (S/N) by averaging and delay compensation. Special care is needed when applying this method to HFTs that are time variant or employ nonlinear signal processing.

## 5.2.2 Single-channel FFT

Without cross-spectrum capabilities, the system input and output are measured separately. These response measurements require control of the excitation spectrum and/or a two-pass analysis. The measurement S/N due to noise at the system input or output is therefore not improved. Any post-processing features available will apply only to the directly measured spectra, not to the response function. Special care is needed when applying this method to HFTs that are time variant or employ nonlinear signal processing.

## 5.2.3 Maximum length sequence (MLS) analysis

The MLS technique (Rife and Vanderkooy [B8]<sup>7</sup>) employs a large [typically 16K ( $16 \times 1024$ )] well-defined pseudo-random pulse excitation. The length of the excitation signal is equal to the correlation length, eliminating leakage. The MLS excitation and analysis are inherently synchronized. The received response signal is cross-correlated with the MLS signal, typically using a fast Hadamard transform, to obtain the time response. An FFT is then used to obtain the frequency response. This also enables computation of coherence, phase, group delay, coherent power, and noncoherent power. Some nonlinear analysis capabilities and postprocessing are available. This method does allow improvements of measurement S/N.

## 5.3 Real-time filter analysis (RTA)

Real-time analysis is essentially a parallel filter bank, usually implemented digitally. This results in a constant percentage (logarithmic) frequency resolution. The analysis is carried out in parallel and the signal is processed continuously. The filters shall be 1/12 or 1/24 octave and comply with the ANSI S1.11-1986 standard. The statistical accuracy of real-time measurements is usually determined by the averaging time or confidence level. This type of analysis is optimized for single-port acoustical measurements (i.e., no control of the system input).

### 5.3.1 Dual-channel real-time filter analysis

Two channels enable simultaneous measurements of the system input and output and direct computation of the frequency response (output/input). This method provides limited harmonic distortion capability and some direct post-processing of the data.

### 5.3.2 Single-channel real-time filter analysis

A single-channel real-time analyzer requires separate measurements of the system input and output. Response measurements will require control of the excitation spectrum and/or a two-pass analysis. This method requires the test system to be time invariant, and is limited in measuring harmonic distortion. Some direct post-processing of the data can be available.

## 5.4 Sine-based analysis

Sinusoidal excitation provides a high measurement S/N ratio and high degree of frequency selectivity. The analysis is performed serially using either a quadrature or rms detector. This often includes a tracking filter for noise suppression and selective measurements of distortion components. The quadrature detector multiplies the response signal by a synchronized (and appropriately delayed) sine and cosine signal. This enables measurement of the complex, steady-state frequency response (i.e., magnitude and phase, real and imaginary parts). Complex averaging algorithms can be employed to improve the measurement S/N ratio. The use of an rms detector requires a separate phase meter to obtain phase information.

<sup>7</sup>The numbers in brackets correspond to those of the bibliography in Annex L.

#### 5.4.1 Discrete tone (stepped sine)

Discrete tone testing allows a measurement to be performed at precisely defined frequencies. These frequencies can be at the ANSI/ISO preferred numbers or in other user-defined formats (see ISO 3: 1973 and ANSI S1.6-1984 for preferred number series). The actual frequency interval (not resolution) used in the measurement shall be stated. In addition to frequency response measurements, intermodulation and difference frequency distortion testing are often carried out using this method. Phase and group delay information is also provided. These tests normally require an anechoic room, although tone-burst techniques can be used with gating to obtain simulated free-field results, and measurement S/N can be improved using complex averaging.

#### 5.4.2 Swept sine

This technique is similar to discrete tone testing, but instead employs a continuous linear or logarithmic sine sweep excitation. The measurement is typically slow due to sweep rate limitations. In addition to frequency response measurements, harmonic distortion is well suited to this method. This method requires an anechoic room, although tone-burst techniques can be used with gating to obtain simulated free-field results.

#### 5.4.3 Time delay spectrometry (TDS)

TDS (Heyser [B5]) utilizes a linearly swept sine excitation signal that is synchronized to the measuring instrument. With this signal, a one-to-one relationship between time and frequency is established and simulated free-field measurements can be performed. The measured response signal is multiplied with an appropriately delayed version of the excitation. This, in turn, is fed to a filter and detector. The frequency resolution of a TDS measurement is narrow band (constant bandwidth) with frequency. This is, by design, a selectable measurement parameter. Therefore, low-frequency resolution can be limited. Like other simulated free-field techniques, the effective time window determines this, i.e., the time between the arrival of the direct sound and the arrival of the first reflection. This method is also well suited for harmonic distortion, and provides phase, group delay, and time response information. TDS may be implemented using analog or digital processing. In the latter case, refinements and corrections for deterministic errors in the measurement process may also be incorporated. It is possible to improve measurement S/N through complex averaging or delay compensation. This method allows post-processing of the data. Special care is needed when applying this method to HFTs that are time variant or employ nonlinear signal processing.

### 5.5 Simulated free-field techniques

Simulated free-field techniques employ a time capture during the measurement, effectively windowing the measured response.

Time windowing enables the direct sound in a measurement to be separated from reflections, producing a simulated free-field condition. In this case, the frequency resolution is the reciprocal of the applied time window. Both gating and post-process windowing can be used on measurements in ordinary rooms.

As discussed previously, MLS and TDS are inherently simulated free-field techniques. Dual-channel FFT analysis can also be used. The time windowing may be performed as a part of the data collection or as a post-processing window operation.

The frequency resolution available in an anechoic chamber is largely determined by the measurement technique employed. Anechoic chambers are limited at low frequencies by the size of the open space available and the depth of the absorptive material on the walls, floor, and ceiling.

## 6. Test signals

### 6.1 General

The test signal should place the HFT in a well-defined reproducible state for the period of the measurement. It should ensure that the transfer function of the unit remains stable during the measurement period, and yet provide a suitable signal for the specific measurement. The choice of the signal will be a balance between one that correctly stimulates the processing algorithms in the HFT, and one that is suitable for the specific measurement.

### 6.2 Classifications

The various types of signals are divided into several groups as discussed in this clause. The classical measurement signals can be separated into deterministic signals and continuous random signals. More complex random signals include modulated random signals and speech-like signals that characterize human speech. Finally, there are compound signals composed of two sources—one for biasing the unit into a stable state, and the other which is actually used to perform the measurement.

### 6.3 Modulation types

Several types of modulation may be applied to deterministic or random signals. This is done in order to approximate the syllabic rhythm of real speech.

Test signals may be modulated in various ways to correctly stimulate an HFT, depending on the signal processing actually used in the phone. For example, a modulated noise signal is often an appropriate stimulus for a send circuit with a noise-guard feature. In the presence of a continuous signal over a few hundred milliseconds in duration, the noise-guard process reduces gain substantially. On the other hand, a continuous noise signal is often an appropriate stimulus for a receive circuit with automatic gain control (AGC).

#### 6.3.1 Square-wave modulation

Square-wave modulation is an on-off pattern. The recommended pattern is 250 ms ON and 150 ms OFF,  $\pm 10$  ms. This pattern is common in many international handsfree testing methods. It is close to the modulation rate of real speech. Other timing patterns may be used after confirming that the maximum measured response has been reached.

In some cases, a periodic pulse pattern of this type will not correctly activate the telephone circuit. In such cases, a randomly varied pulse pattern may be used. The average “on” and “off” times should approximate 250 ms and 150 ms, respectively.

With this type of modulation, all measurements are to be performed during the “on” part of the pattern. For other types of modulation, the signal is to be measured during the entire presentation time.

#### 6.3.2 Sine-wave modulation

Sine-wave modulation may be used to produce a simple and smooth speech amplitude envelope. The recommended rate is 4 Hz. Modulation depth should be at least 50%, but not so great as to introduce distortion.

### 6.3.3 Pseudo-random modulation

Pseudo-random modulation may be used to produce a relatively speech-like amplitude envelope. The modulation spectrum should cover from approximately 1–10 Hz, with the center at approximately 4 Hz. The extremes of the modulation spectrum should be rolled off gradually.

## 6.4 Deterministic signals

Deterministic (periodic) signals can always be used to measure the frequency response of linear, time invariant telephones. When modulated, they can be used to measure the response of telephones with many, but not all, speech-processing features.

### 6.4.1 Sine wave

In addition to use in measuring the frequency response of linear, time invariant telephones, sine waves are useful for measurements of harmonic and difference-frequency distortion. This signal can be modulated by square-wave, sine-wave, and pseudo-random signals.

### 6.4.2 Pseudo-random

A pseudo-random signal has a periodic structure in the time domain. In the frequency domain, almost any magnitude and phase spectrum is possible. When used with FFT types of analysis, the period of the pseudo-random signal is to be matched in length and triggered to the analysis period. When used with an MLS analyzer, the period of the MLS signal must be matched to the analysis period. This signal can be modulated by square-wave, sine-wave, and pseudo-random signals. If square-wave modulation is used, the “on” time must correspond to one or more complete periods of the pseudo-random signal.

## 6.5 Random signals

Random signals can be described by their statistical characteristics, such as the long-term power spectral density and probability density functions. These signals are not periodic, but are stationary as far as these statistical characteristics are concerned. When measuring such signals, a sufficient number of averages should be taken to obtain a given accuracy in estimating the long-term spectrum.

In practice, many practical noise generators produce pseudo-random signals, typically with a very long period. If the period of such signals is very long compared to the analysis period, and if the analysis period is not correlated to the generator period, then these signals can be considered random.

### 6.5.1 White noise

White noise has a constant spectral density per Hertz. The amplitude distribution is typically truncated Gaussian, with a crest factor of 12 dB,  $\pm 2$  dB. This signal can be modulated by square-wave, sine-wave, and pseudo-random signals.

### 6.5.2 Pink noise

Pink noise has a power spectral density that decreases 3 dB per octave. The amplitude distribution is typically truncated Gaussian, with a crest factor of 12 dB,  $\pm 2$  dB. This signal can be modulated by squarewave, sine-wave, and pseudo-random signals.

## 6.6 Speech-like signals

Speech-like signals include ITU-T Recommendation P.50: 1993 artificial voice, ITU-T Recommendation P.59: 1993 artificial conversational speech, simulated speech generator (SSG), as well as synthesized and real speech signals. When long-term averaging is used, these signals place the HFT in a well-defined reproducible state, ensure that the transfer function of the unit remains stable, and provide a suitable signal for the specific measurement.

### 6.6.1 Simulated speech

Typical parameters of simulated speech include long-term average spectrum, short-term spectrum, instantaneous amplitude distribution, speech waveform structure, and the syllabic envelope.

#### 6.6.1.1 Artificial voice

ITU-T Recommendation P.50: 1993 defines the temporal and spectral parameters for a test signal which emulates the characteristics of speech. This artificial voice is a continuous speech signal with a frequency range of 89.1–8919 Hz. Pauses may need to be inserted to emulate the on-off characteristics of conversational speech (see 6.6.1.2 for information on inserting pauses).

#### 6.6.1.2 Artificial conversational speech

Artificial conversational speech is a test signal generated by inserting pauses in the continuous artificial voice signal described by ITU-T Recommendation P.50: 1993. The on-off temporal characteristics of conversational speech are defined in ITU-T Recommendation P.59: 1993. This test signal is useful for evaluating devices that are sensitive to the on-off nature of conversational speech, in both single- and double- talk modes.

#### 6.6.1.3 Simulated speech generator (SSG)

To generate a signal approximating the amplitude distribution of speech, a main signal having a Gaussian distribution is modulated by a specially tailored modulating signal, and the resultant signal is shaped to approximate the long-term frequency spectrum of speech (see Annex A for details of this signal).

### 6.6.2 Synthesized speech

Speech-like signals may be produced using a digital processing technique rather than applying one of the signal sources described above. Conversational speech can be sampled, digitized, processed, and reproduced as synthesized speech. It may also be created from complex multiple tones that simulate the talk spurts, pauses, and activity factors associated with speech characteristics.

### 6.6.3 Real speech

Speech-like signals are not limited to signal sources or synthesized digital processing, but may also include real speech signals. This is often done by recording conversational speech, preferably in a digital format to avoid signal degradation with use. These real speech recordings are then reproduced using a playback device as the signal source.

## 6.7 Compound signals

Many test signals rely on one signal source to place the HFT in a well-defined reproducible state, ensure that the transfer function of the unit remains stable, and provide a suitable signal for the specific measurement. By applying two signal sources, one can be used specifically for “biasing” the unit into a stable, reproducible

state, while the other is the actual test signal required for measurements. These compound signals include those where the two sources are applied in sequence, and those where both sources are applied simultaneously.

Compound test signals can provide extra test flexibility and solve problems that are difficult or impossible using simple test signals. The bias signal can be a signal that, by itself, is unsuitable or very inconvenient for the actual measurement. The measurement signal can be a signal that, by itself, is unsuitable as a bias signal.

If desired, the measurement signal can be presented so as not to have a substantial effect on the action of the bias signal. This can be done by adjusting the temporal and/or level relationships between the two signals. The bias signal can be changed to put the telephone in different states with minor or even no change in the measurement signal.

### **6.7.1 Sequential presentation**

This class of test signals is characterized by the separation of the bias and analysis signals in time. The bias signal is presented until the HFT is in a stable state. Once a stable state is reached, the appropriate analysis signal is applied and a measurement is performed. The analysis should be completed while the HFT is still in its stable state. The composite source signal (CSS) is one example of this type of signal.

#### **6.7.1.1 Composite source signal**

The CSS is a compound signal using a voiced signal to simulate the voice properties, followed by a noise-like signal for measuring the transfer functions, and an inserted pause to provide amplitude modulation. The noise-like signal has either a flat or speech shaped power density spectrum. It has the advantage of short measurement periods and duplex operation where, using an uncorrelated double-talk signal, the test signals can be applied from the talking and listening directions at the same time (see ITU-T Recommendation P.501: 1996 for the definition of this signal).

### **6.7.2 Simultaneous presentation**

This class of test signals is characterized by presentation of the bias and analysis signals at the same time. Some conditioning of the HFT may be required before beginning the analysis. The bias and analysis signals must be separable by the analysis method. A synchronous analysis method is usually required. The TDS sweep with noise bursts is one example of this type of signal.

#### **6.7.2.1 TDS sweep with noise bursts**

This compound signal has two components, which are presented at the same time, but not synchronized with each other. The bias signal (ITU-T P.50 noise burst) is intended to ensure that the telephone is in a stable, well-defined operating state. The measurement signal (TDS sweep) is intended to ensure a well-defined, reproducible measurement, which is especially well adapted to simulated free-field techniques. An anechoic room is not necessary when using this signal (see Annex B for details of this signal).

#### **6.7.2.2 TDS sweep with artificial voice**

Similar to 6.7.2.1, except the bias is a continuous artificial voice signal defined in ITU-T Recommendation P.50: 1993 (see 6.6.1.1).

### 6.7.2.3 TDS sweep with real speech

Similar to 6.7.2.1, except the bias is real speech (see 6.6.3).

### 6.7.2.4 TDS sweep with random or pseudorandom noise

Similar to 6.7.2.1, except the bias is white or pink random noise (6.5). Pseudorandom noise (6.4.2) with white or pink spectrum is considered equivalent if the pseudorandom period is not correlated with the bias.

### 6.7.2.5 Pseudorandom noise with noise bursts

This compound signal has two components, which are presented at the same time, but not synchronized with each other. The bias signal (ITU-T P.50 noise burst) is intended to ensure that the telephone is in a stable, well-defined operating state. The measurement signal (pseudorandom noise) is intended to ensure a well-defined, reproducible measurement, which is especially well adapted to simulated free-field techniques. An anechoic room is not necessary when using this signal.

### 6.7.2.6 Pseudorandom noise with artificial voice

Similar to 6.7.2.5, except the bias is a continuous artificial voice signal defined in ITU-T Recommendation P.50: 1993 (see 6.6.1.1).

### 6.7.2.7 Pseudorandom noise with real speech

Similar to 6.7.2.5, except the bias is real speech (see 6.6.3).

### 6.7.2.8 Pseudorandom noise with random or pseudorandom noise

Similar to 6.7.2.5, except the bias is white or pink random noise (6.5). Pseudorandom noise (6.4.2) with white or pink spectrum is considered equivalent if the pseudorandom period is not correlated with the bias.

### 6.7.2.9 Sine wave with notched real speech

A sine wave is the measurement signal and real speech is the bias signal. A notch filter removes a band of the speech signal at the sine-wave frequency.

## 6.8 Test signal bandwidth

In general, the test signals and analysis methods in this standard cover a frequency range of 100 Hz to 10 kHz. Some signals, such as SSG (6.6.1.3), are defined only for a smaller bandwidth, and can only be used within their defined range.

The minimum bandwidth for this standard is 175 Hz to 4.5 kHz, or the 1/3 octave bands from 200 Hz to 4 kHz.

## 6.9 Signal parameter summary

Table 1 defines the bandwidth and maximum analysis interval for the various test signals identified in Clause 6. Signals may be analyzed with finer resolution if desired. These parameters shall be applied to both the calibration and test procedures.

**Table 1—Test signal parameters**

| Document reference | Test signal                           | Usable bandwidth (Hz) | Maximum analysis interval | Alternative analysis format |
|--------------------|---------------------------------------|-----------------------|---------------------------|-----------------------------|
| 6.4.1              | Sine wave                             | 100–10 000            | ISO R40 steps             | 1/12 octave steps           |
| 6.4.2              | Pseudorandom                          | 100–10 000            | 25 Hz bands               | 1/12 octave bands           |
| 6.5.1              | White noise                           | 100–10 000            | 25 Hz bands               | 1/12 octave bands           |
| 6.5.2              | Pink noise                            | 100–10 000            | 1/12 octave bands         | 25 Hz bands                 |
| 6.6.1.1            | P.50 artificial voice                 | 89.1–8 910            | 1/12 octave bands         | 25 Hz bands                 |
| 6.6.1.2            | P.59 artificial conversational speech | 89.1–8 910            | 1/12 octave bands         | 25 Hz bands                 |
| 6.6.1.3            | Simulated speech generator            | 100–5 000             | 1/12 octave bands         | 25 Hz bands                 |
| 6.6.2              | Synthesized speech                    | 100–10 000            | 1/12 octave bands         | 25 Hz bands                 |
| 6.6.3              | Real speech                           | 100–10 000            | 1/12 octave bands         | 25 Hz bands                 |
| 6.7.1.1            | Composite source signal               | 100–10 000            | 25 Hz bands               | 1/12 octave bands           |
| 6.7.2.1–6.7.2.4    | TDS sweep with bias                   | 100–10 000            | 50 Hz bands               | 1/12 octave bands           |
| 6.7.2.5–6.7.2.8    | Pseudorandom noise with bias          | 100–10 000            | 50 Hz bands               | 1/12 octave bands           |

## 6.10 Signal and test method comparative summary

Table 2 identifies the various test signals described previously. The corresponding test methods and conditions are shown for each signal. The various method classifications are described in Clause 5.

**Table 2—Signal compatibility with test method**

| Sec.<br>Ref.                                                                                                                                                                                       | Test method        | Signal type             |              |                  |            |                     |                       |             |                    |              |                                |
|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|--------------------|-------------------------|--------------|------------------|------------|---------------------|-----------------------|-------------|--------------------|--------------|--------------------------------|
|                                                                                                                                                                                                    |                    | Deterministic<br>signal |              | Random<br>signal |            | Speech-like signal  |                       |             | Compound<br>signal |              | Anechoic<br>chamber<br>needed? |
|                                                                                                                                                                                                    |                    | Sine wave               | Pseudorandom | White noise      | Pink noise | Simulated<br>speech | Synthesized<br>speech | Real speech | Sequential         | Simultaneous |                                |
|                                                                                                                                                                                                    | FFT/cross spectrum |                         |              |                  |            |                     |                       |             |                    |              |                                |
| 5.2.1                                                                                                                                                                                              | Dual-channel FFT   | Y                       | Y            | Y                | Y          | Y                   | Y                     | Y           | Y                  | Y            | Y*                             |
| 5.2.2                                                                                                                                                                                              | Single-channel FFT | Y                       | Y            | Y                | Y          | Y                   | Y                     | Y           | Y                  | Y            | Y                              |
| 5.2.3                                                                                                                                                                                              | Max. length seq.   | N                       | R            | N                | N          | N                   | N                     | N           | Y                  | Y            | Y*                             |
|                                                                                                                                                                                                    | Real-time filter   |                         |              |                  |            |                     |                       |             |                    |              |                                |
| 5.3.1                                                                                                                                                                                              | Dual-channel RTA   | Y                       | Y            | Y                | Y          | Y                   | Y                     | Y           | Y                  | N            | Y                              |
| 5.3.2                                                                                                                                                                                              | Single-channel RTA | Y                       | Y            | Y                | Y          | Y                   | Y                     | Y           | Y                  | N            | Y                              |
|                                                                                                                                                                                                    | Sine-based         |                         |              |                  |            |                     |                       |             |                    |              |                                |
| 5.4.1                                                                                                                                                                                              | Discrete tone      | R                       | N            | N                | N          | N                   | N                     | N           | N                  | Y            | Y                              |
| 5.4.2                                                                                                                                                                                              | Swept sine         | R                       | N            | N                | N          | N                   | N                     | N           | Y                  | Y            | Y                              |
| 5.4.3                                                                                                                                                                                              | Time delay spectr. | R                       | N            | N                | N          | N                   | N                     | N           | Y                  | Y            | Y*                             |
| Y—Test method is appropriate for this signal.<br>N—Should not be used.<br>R—Required signal with this test method.<br>* Anechoic chamber is required unless simulated free-field methods are used. |                    |                         |              |                  |            |                     |                       |             |                    |              |                                |

## 7. Test equipment, environment, and impairments

### 7.1 General

The test equipment, environmental concerns, and impairments recommended to evaluate HFT performance are described here (see IEEE Std 269-1992 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 HFTs, feed circuits and test loops are required. For handsfree digital or wireless telephone sets, interfaces and/or codecs are required.

### 7.2 Test equipment

#### 7.2.1 Measuring microphones

A 12.5 mm (type M) laboratory standard free-field microphone per ANSI S1.12-1967 is required for calibrating the mouth simulator and measuring receive characteristics. The sensitivity of the microphone should be constant within  $\pm 0.5$  dB, and shall be constant within  $\pm 2.0$  dB, from 100 Hz to 10 000 Hz.

A 25 mm (type L) laboratory standard free-field microphone per ANSI S1.12-1967 is required for measuring low sound pressure levels, background noise in the test environment, and receive noise from the HFT under test. The sensitivity of the microphone should be constant within  $\pm 0.5$  dB, and shall be constant within  $\pm 2.0$  dB, from 100 Hz to 5000 Hz.

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.

#### 7.2.2 Mouth simulator

The mouth simulator shall comply with the specifications given in ITU-T Recommendation P.51: 1996.

#### 7.2.3 Head and torso simulator (HATS)

A head and torso simulator for measurements of HFTs shall conform to ITU-T Recommendation P.58: 1996. The HATS shall be equipped with two ear simulators and a mouth simulator. The ear simulators shall conform to ITU-T Recommendation P.57: 1996. Electroacoustic measurements on HFTs using HATS will differ from measurements performed using a separate microphone and mouth simulator. This use of 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.

#### 7.2.4 Standard circuits for transmission and voice-switching measurements

The standard circuits for analog and digital HFT transmission and voice-switching measurements are the same as those used in IEEE Std 269-1992 for telephones. For interfacing with wireless systems, see appropriate wireless standards.

#### 7.2.5 Standard circuits for acoustic echo canceller measurements

A test hybrid is required to perform acoustic terminal coupling loss measurements on two-wire analog HFTs. The test hybrid is used to cancel or remove the HFTs hybrid reflection, making a two-wire analog HFT behave as a virtual four-wire phone. This means that 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—the use of an echo canceller with freezable coefficients, measurement post processing, or use of a freezable adaptive analog filter bank.

The requirements for the hybrid are

- High impedance bridge mode for tapping off lines without affecting impedance.

- DC blocking on the input.
- Known one way delay and round trip echo path (in/out delay). Any delay in the digital hybrid or acquisition card will impact both the echo path model delay and the synchronization of data acquisition. The fixed echo path delay inserted by the units shall be quantified.
- Noise and distortion performance 10 dB below echo detection thresholds. A full-scale signal to noise ratio (SNR) of 80 dB is preferred.
- High echo return loss of 60 dB or better is recommended. An echo return loss of 50 dB is required.
- Ability to freeze the adaptive hybrid to ensure 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 requirements in this subclause. However, one possible post-processing implementation of this technique, which does meet the requirements, is given in Annex E.

### 7.3 Test environment

Electroacoustic measurements on HFTs should be conducted in a test environment, which 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. In addition, either the test environment or the test method should minimize or eliminate the influence of reflections.

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.)

#### 7.3.1 Background noise level

The background noise level in the test environment shall not exceed the limits shown in Table 3. The overall A-weighted noise level shall not exceed 29 dBSPL.

**Table 3—Test room noise levels**

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

#### 7.3.2 Anechoic chamber

An anechoic chamber should be large enough to comfortably accommodate the test table, measurement transducers, and HFT. Free-field conditions should exist throughout the frequency range of interest. Errors due to deviations from the inverse square law or due to influence of reflections shall not exceed  $\pm 1.5$  dB below 800 Hz. Errors above 800 Hz shall not exceed  $\pm 1.0$  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.

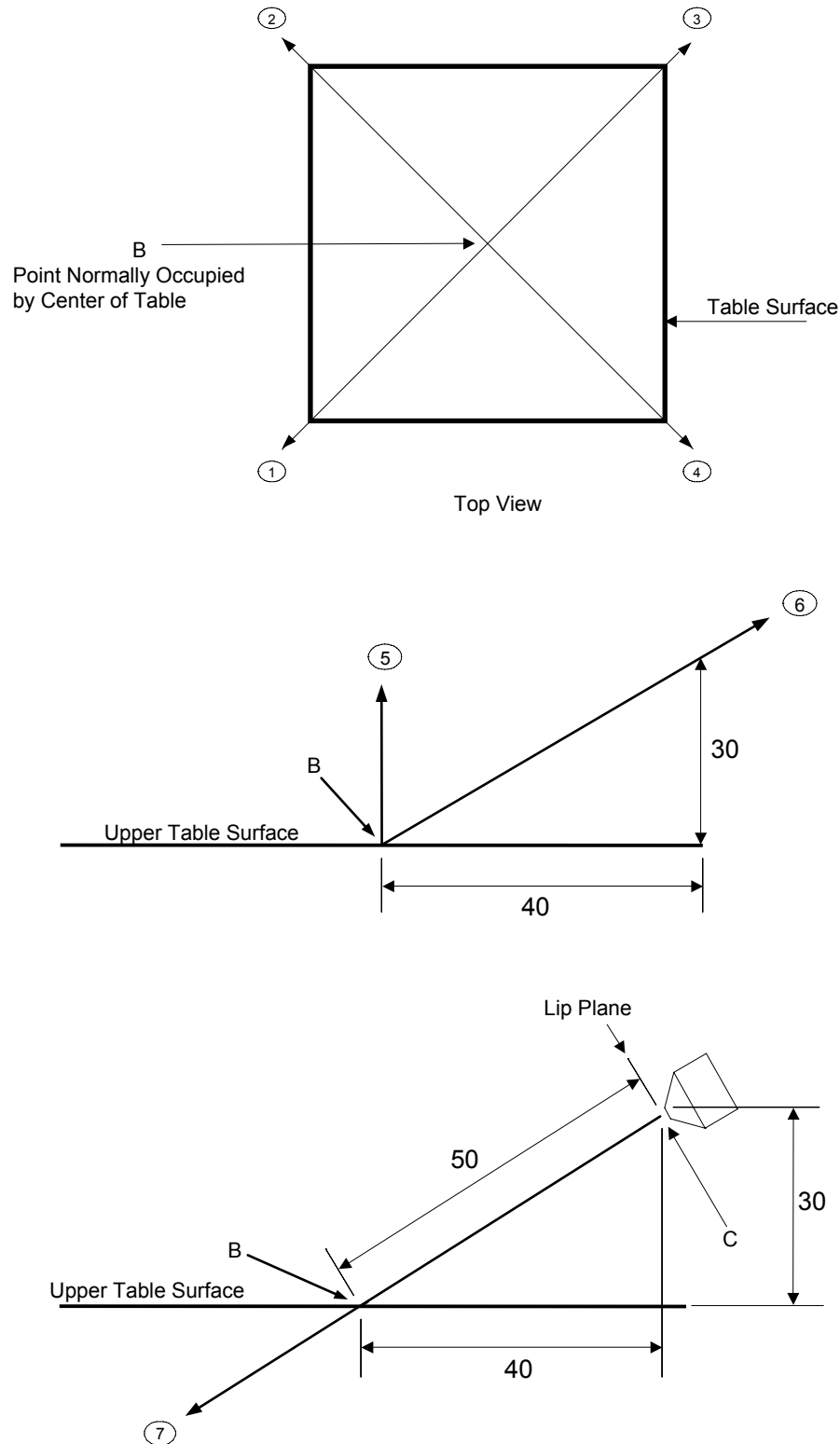


**Figure 1—Axes for determining free-field conditions**

A verification of free-field conditions can be conducted. 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.

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

The signal used for this test shall have the same frequency format as the subsequent handsfree response measurements. The tolerances specified in this subclause 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 7.3.3).

### 7.3.3 Simulated free field

Several methods exist which enable reflection-free measurements in a room with ordinary, untreated boundaries. Tone burst gating, FFT analysis of short noise bursts, TDS, MLS, and post-process windowing of an impulse response are among the possibilities which can together 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, as long as no reflections are included within the window. 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 substitutes for an anechoic room:

- a) The signal has a suitable bias effect on the HFT, so 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 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 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, and 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 ( $f_{\min}$ ) halved.

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

### 7.3.4 Test table

The test table should be a hard, bare table with a surface area of at least  $1 \text{ m}^2$ . 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 HFT 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 HFTs. It may also be used for measurements of nondesktop HFTs where appropriate.

## 7.4 Impairments

It is important to understand how an HFT 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 HFT may be evaluated. It is beyond the scope of this standard to list all impairments, but some are listed below.

### 7.4.1 Network impairments

Network impairments are those that can occur in the path of a wired or wireless telephone connection.

#### 7.4.1.1 Test loops

A wireline analog HFT should be tested with various lengths of cable or simulated cable. Recommended loop lengths for testing telephones in North America are 0 km, 2.7 km, and 4.6 km (0 ft, 9000 ft, and 15 000 ft) of 26 AWG cable.

#### 7.4.1.2 Loop current

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

#### 7.4.1.3 Termination impedance

A network termination impedance mismatch can result in the send signal being reflected back toward the telephone. This 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 effect HFT echo canceller or switching circuits.

#### 7.4.1.4 Network noise

Network noise can effect the HFT 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 an HFT. 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: 1994.

### 7.4.2 Acoustic impairments

In order to realistically test an HFT, 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 than in the anechoic test space.