Standard Test Method for
Impedance and Absorption of Acoustical Materials Using a Tube, Two Microphones and a Digital Frequency Analysis System[1]

This standard is issued under the fixed designation E1050; the number immediately following the designation indicates the year of original adoption or, in the case of revision, the year of last revision. A number in parentheses indicates the year of last reapproval. A superscript epsilon (ε) indicates an editorial change since the last revision or reapproval.

1. Scope

1.1 This test method covers the use of an impedance tube, two microphone locations, and a digital frequency analysis system for the determination of normal incidence sound absorption coefficients and normal specific acoustic impedance ratios of materials.

1.2 *Laboratory Accreditation*—A procedure for accrediting a laboratory for performing this test method is given in Annex A1.

1.3 The values stated in SI units are to be regarded as standard. No other units of measurement are included in this standard.

1.4 *This standard does not purport to address the safety concerns, if any, associated with its use. It is the responsibility of the use of this standard to consult and establish appropriate safety and health practices and determine the applicability of regulatory limitations prior to use.*

2. Referenced Documents

2.1 *ASTM Standards:*[2]
C384 Test Method for Impedance and Absorption of Acoustical Materials by Impedance Tube Method
C634 Terminology Relating to Building and Environmental Acoustics
E548 Guide for General Criteria Used for Evaluating Laboratory Competence (Withdrawn 2002)[3]

2.2 *ISO Standards:*
ISO 10534-1 Acoustics—Determination of Sound Absorption Coefficient and Impedance or Admittance—Part 1: Impedance Tube Method[4]
ISO 10534–2 Acoustics—Determination of Sound Absorption Coefficient and Impedance in Impedance Tubes—Part 2: Transfer-Function Method[4]

2.3 *ANSI Standards:*[4]
ANSI/ASA S1.11 Octave-Band and Fractional-Octave-Band Analog and Digital Filters

3. Terminology

3.1 *Definitions*—The acoustical terminology used in this test method is intended to be consistent with the definitions in Terminology C634.

NOTE 1—Historical literature regarding the measurement of normal incidence absorption coefficients referred to "transfer function" measurements; however, the term arises from Laplace transform theory and is not strictly rigorous when the initial conditions have a non-zero value. The term "frequency response function" arises from more general Fourier transform theory (**1**).[5] This test method shall retain the use of the former term although not technically correct. Users should be aware that modern FFT analyzers may employ the latter terminology.

3.2 *Symbols:* The following symbols are used in Section 8 (Procedure):

3.2.1 $b\rho c$—normal specific acoustics susceptance ratio.

3.2.2 c—speed of sound, m/s.

3.2.3 $g\rho c$—normal specific acoustic conductance ratio.

3.2.4 G_{11}, G_{22}—auto power spectra of the acoustic pressure signal at microphone locations 1 and 2, respectively.

3.2.5 G_{12}—cross power spectrum of the acoustic pressure signals at microphones locations 1 and 2.

3.2.6 H—transfer function of the two microphone signals corrected for microphone response mismatch.

3.2.7 $\bar{H}$—measured transfer function of the two microphone signals.

3.2.8 H^I, H^{II}—calibration transfer functions for the microphones in the standard and switched configurations, respectively.

[2] For referenced ASTM standards, visit the ASTM website, www.astm.org, or contact ASTM Customer Service at service@astm.org. For *Annual Book of ASTM Standards* volume information, refer to the standard's Document Summary page on the ASTM website.

[3] The last approved version of this historical standard is referenced on www.astm.org.

[4] Available from American National Standards Institute (ANSI), 25 W. 43rd St., 4th Floor, New York, NY 10036, http://www.ansi.org.

[5] The boldface numbers in parentheses refer to the list of references at the end of this test method.

3.2.9 $\bar{H}_c$—complex microphone calibration factor.

3.2.10 j—equals $\sqrt{-1}$.

3.2.11 k—equal $2\pi f/c$; wave number, m^{-1}.

3.2.11.1 *Discussion*—In general the wave number is complex where $k = k' - jk''$. k' is the real component, $2\pi f/c$ and k'' is the imaginary component of the wave number, also referred to as the attenuation constant, Nepers-m^{-1}.

3.2.12 l—distance from the test sample to the centre of the nearest microphone, m.

3.2.13 $r/\rho c$—normal specific acoustic resistance ratio.

3.2.14 R—complex acoustic reflection coefficient.

3.2.15 s—centre-to-center spacing between microphones, m.

3.2.16 $x/\rho c$—normal specific acoustic reactance ratio.

3.2.17 $y\rho c$—normal specific acoustic admittance ration.

3.2.18 $z/\rho c$—normal specific acoustic impedance ratio.

3.2.19 a—normal incidence sound absorption coefficient.

3.2.20 φ—phase of the complex transfer function, radians.

3.2.21 φ_R—phase of the complex acoustic reflection coefficient, radians.

3.2.22 ρ—density of air, kg/m^3.

3.3 *Subscripts, Superscripts, and Other Notation*—The following symbols, which employ the variable X for illustrative purposes, are used in Section 8:

3.3.1 X_c—calibration.

3.3.2 X_i—imaginary part of a complex quantity.

3.3.3 X_r—real part of a complex quantity.

3.3.4 X^I, X^{II}—calibration quantities measured with microphones placed in the standard and switched configurations, respectively.

3.3.5 $\bar{X}$—measured quantity prior to correction for amplitude and phase mismatch.

3.3.6 $|X|$—magnitude of a complex quantity.

4. Summary of Test Method

4.1 This test method is similar to Test Method C384 in that it also uses an impedance tube with a sound source connected to one end and the test sample mounted at the other end. The measurement techniques for the two methods are fundamentally different, however. In this test method, plane waves are generated in the tube using a broad band signal from a noise source rather than a discrete sinusoid from an oscillator. The decomposition of the stationary sound wave pattern into forward- and backward-traveling components is achieved by measuring sound pressures simultaneously at two spaced locations in the tube's side wall. Calculations of the normal-incidence absorption coefficients for the acoustical material are performed by processing an array of complex data from the measured transfer function.

4.2 The quantities are determined as functions of frequency with a resolution determined by the sampling rate of a digital frequency analysis system. The usable frequency range depends on the diameter of the tube and the spacing between the

microphone positions. An extended frequency range may be obtained by using tubes with various diameters and microphones spacings.

4.3 This test method is intended to provide a much faster measurement technique than that of Test Method C384.

5. Significance and Use

5.1 This test method can be applied to measure sound absorption coefficients of absorptive materials at normal incidence, that is, 0°. It also can be used to determine specific impedance and admittance ratios. The properties measured with this test method are useful in basic research and product development of sound absorptive materials.

5.2 Normal incidence sound absorption coefficients can be quite useful in certain situations where the material is placed within a small acoustical cavity close to a sound source, for example a closely-fitted machine enclosure.

5.3 This test method allows one to compare relative values of sound absorption when it is impractical to procure large samples for accurate random-incidence measurements in a reverberation room. Estimates of the random incidence absorption coefficients can be obtained from normal impedance data for locally-reacting materials (2).

Note 2—The classification, "locally-reacting" includes fibrous materials having high internal losses. Formulas have been developed for converting sound absorption properties from normal incidence to random incidence, for both locally-reacting and bulk-reacing materials (3).

5.4 Measurements described in this test method can be made with high precision, but these measurements may be misleading. Uncertainties of greater magnitude than those from the measurements may occur from other sources. Care should be exercised to sample nonuniform materials adequately (see 11.1).

6. Apparatus

6.1 The apparatus is a hallow cylinder, or tube, with a test sample holder at one end and a sound source at the other. Microphone ports are mounted at two or more locations along the wall of the tube. A two channel digital frequency analysis system is used for data acquisition and processing.

6.2 *Tube:*

6.2.1 *Construction*—The interior section of the tube may be circular or rectangular with a constant dimension from end-to-end. The tube shall be straight and its inside surface shall be smooth, nonporous, and free of dust to maintain low sound attenuation. The tube construction shall be massive so sound transmission through the tube wall is negligible.

Note 3—The tube can be constructed from materials including metal, plastic, cement, or wood. It may be necessary to seal the interior walls with a smooth coating in order to maintain low sound attenuation for plane waves.

6.2.2 *Working Frequency Range*—The working frequency range is:

$$f_l < f < f_u \qquad (1)$$

where:

f = operating frequency, hertz,

f_l = lower working frequency of the tube, hertz, and
f_u = upper working frequency of the tube, hertz.

6.2.2.1 The lower frequency limit depends on the spacing of the microphones and the accuracy of the analysis system. It is recommended that the microphone spacing exceed one percent of the wavelength corresponding to the lower frequency of interest.

6.2.2.2 The upper frequency limit, f_u, and the corresponding wavelength, λ_u, depends on the diameter of the tube and upon the speed of sound.

6.2.3 *Diameter*—In order to maintain plane wave propagation, the upper frequency limit (4) is defined as follows:

$$f_u < K \, c/d \quad or \quad d < K \, c/f_u \tag{2}$$

where:
f_u = upper frequency limit, hertz,
c = speed of sound in the tube, m/s,
d = diameter of the tube, m, and
K = 0.586.

6.2.3.1 For rectangular tubes, d is defined as the largest section dimension the tube and K is defined as 0.500. Extreme aspect rations greater than 2:1 or less than 1:2 should be avoided. A square cross-section is recommended.

6.2.3.2 It is best to conduct the plane wave measurements well within these frequency limits in order to avoid cross-modes that occur at higher frequencies when the acoustical wave length approaches the sectional dimension of the tube.

6.2.4 *Length*—The tube should be sufficiently long as plane waves are fully developed before reaching the microphones and test specimen. A minimum of three tube diameters must be allowed between sound source and the nearest microphone. The sound source may generate nonplane waves along with desired plane waves. The nonplane waves usually will subside at a distance equivalent to three tube diameters from the source. If measurements are conducted over a wide frequency range, it may be desirable to use a tube which provides multiple microphone spacings or to employ separate tubes. The overall tube length also must be chosen to satisfy the requirements of 6.4.3, 6.5.3, and 6.5.4.

6.2.5 *Tube Venting*—Some tube designs are such that, during during installation or removal of the test specimen, large temporary pressure variation may be generated. This may induce microphone diaphragm deflection. The potential for damage to a microphone diaphragm due to excessive deflection may be reduced including a pressure relief opening in the tube. This may be accomplished by drilling a small hole, 1 to 2 mm through the wall of the tube. It is recommended to locate the tube vent near the sound source, away from microphone locations, and to seal the vent during acoustic measurements.

6.3 *Test Specimen Holder:*

6.3.1 *General Features*—The specimen holder may either be integrated with the impedance tube or may be a separate, detachable extension of the tube. Provision must be made for mounting the specimen with its face in a known position along the tube axis and for placing a heavy backing plate behind the specimen. For some measurements it may be desirable to maintain an airspace of known dimensions between the specimen and the backing plate. One such arrangement may be to simulate a suspended ceiling tile.

6.3.2 *Detachable Holder*—As a detachable unit, the holder must make an airtight fit with the end of the tube opposite the sound source. The holder must conform with the interior shape and dimensions of the main part of the impedance tube. The connecting joint must be finished carefully and the use of a sealant, such as petroleum jelly or silicone grease, is recommended for sealing.

6.3.3 *Integral Holder*—If the sample holder is in an integral part of the impedance tube, it is recommended to make the installation section of the tube accessible for mounting of the specimen by a removable cover. The mating surfaces must be finished carefully, and the use of a sealant is recommended for sealing.

6.3.4 *Circular Holder*—For circular tubes, it is recommended to make the specimen accessible from both the front and back end of the sample holder. It is possible then to check the position and flatness of the front surface and back position. Holders may be constructed from a rigid, clear material, such as acrylic, to facilitate inspection.

6.3.5 *Rectangular Holder*—With rectangular tubes, it is recommended to install the specimen from the side, making it possible to check the fitting and the position of the specimen in the tube and to check the position and flatness of the front surface.

6.3.6 *Backing Plate*—The backing plate of the sample holder shall be rigid and shall be fixed tightly to the tube since it serves to provide a sound-reflective termination in many measurements. A metal plate having a minimum thickness of 20 mm is recommended.

6.4 *Sound Source:*

6.4.1 *Kind and Placement*—The sound sources should have a uniform power response over the frequency range of interest. It may either be coaxial with the main tube or joined to the main tube by means of a transition having a straight, tapered, or exponential section (see Fig. 1).

6.4.2 *Isolation*—The sound source and transition shall be sealed and isolated from the tube to minimize structure-borne sound excitation of the impedance tube. If a direct radiator loudspeaker is utilized, it shall be contained in a sound-isolating enclosure in order to avoid airborne flanking transmission to the microphones (see Fig. 1).

6.4.3 *Termination*—Resonances of the air column in the impedance tube may arise if the mechanical impedance of the loudspeaker membrane or diaphragm is high. In this case, it is recommended to apply a porous absorber coating or lining inside either the impedance tube near the loudspeaker or inside the sound transition. Alternatively, the locations describes above may be filled lightly with a low density absorbing material.

6.4.4 *Equalization*—When an absorptive medium is placed near the sound source as described in 6.4.3, significant sound energy will be lost at higher frequencies. An electronic equalizer may be required to shape and sound spectra measured
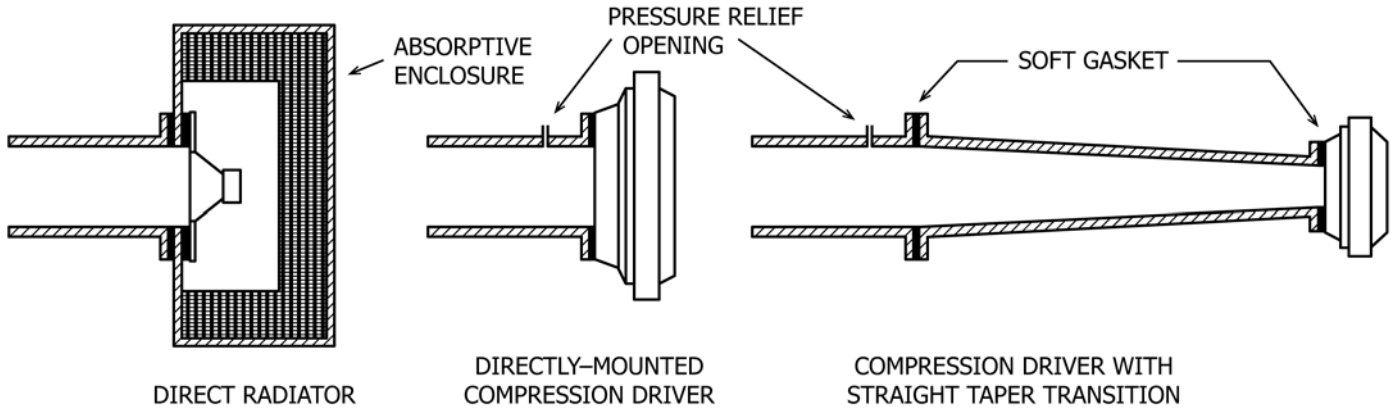
FIG. 1 Sound Source Configurations

at the microphone positions so that they are relatively flat. This will minimize the loss of signal-to-noise capability at high frequencies.

6.5 *Microphones:*

6.5.1 *Type, Diameter*—Two nominally identical microphones shall be mounted according to 6.5.4. The microphone diameter must be small in comparison with the spacing between microphone ports and also to minimize spatial averaging at higher frequencies across the diaphragm face. It is recommended that the microphone diameter be less than 20 % of the wavelength for the highest frequency of interest. Table 1 provides maximum recommended frequency limits for several typical microphone sizes used at room temperature. Where greater microphone sensitivity is required, larger diameters may be selected for use with large tubes working at low frequencies.

6.5.2 *Microphone Venting*—Microphones are designed with a vent to allow for static pressure equalization on either side of the diaphragm. In general, venting may be accomplished either to the inside or to the outside of the tube. Two alternate venting methods are available: back-vented (preferred) (**5**) and side-vented. A microphone pair of either design may be used. Microphones must be sealed carefully when installed in the impedance tube to avoid leaks, which may interfere with proper operation of the microphone vent, thus causing significant changes to the low frequency response. Blockage of a vent of an individual microphone will alter its phase response, resulting in large errors in the measurements.

6.5.2.1 *Back-Vented Microphones*—Back-vented microphones are vented out through the back of the preamplifier barrel to the outside of the the tube. Very low frequency accuracy is improved when the static pressure equalization vent is isolated from the sound field within the impedance tube

TABLE 1 Recommended Maximum Frequency Based on Microphone Diameter

Nominal Diameter (in.)	Diaphragm Diameter (mm)	Maximum Frequency (Hz)
1	22.70	3000
½	12.2	5600
¼	5.95	11 500

(**6**). Back-vented microphones are designed so that the venting path proceeds from the rear of the microphone cartridge through the preamplifier. Sealing may be accomplished either against the rear of the microphone cartridge barrel or against the protection grid. If the seal is established against the latter, the threads of the protection grid should be sealed with silicone grease to prevent leakage between the tube interior and the back vent, (see Fig. 2).

6.5.2.2 *Side-Vented Microphones*—Side-venting path proceeds from the vent opening, which is located between the protection grid threads and the diaphragm, to the front of the microphone, and therefore, vent to the inside of the tube. Sealing may be established either against the rear of microphone cartridge barrel or against the protection grid. If the seal is established against the latter, the threads of the protection should be sealed with silicone grease to prevent leakage (see Fig. 2).

6.5.3 *Spacing*—A large spacing between microphones enhances the accuracy of the measurements, however, the microphone spacing must be less than the shortest half wave length of interest (**7**).

$$s << c/2f_u \qquad (3)$$

where:

s = microphone spacing, m
c = speed of sound, m/s, and
f_u = upper frequency limit, hertz.

It is recommended that the maximum microphone spacing, *s*, be 80 % of $c/2f_u$.

6.5.4 *Location*—The minimum between the sound source and the closest microphone must follow the requirements of 6.2.4. The minimum distance between the specimen and the closest microphone depends somewhat on the surface characteristics of the specimen. In order to maintain the greatest signal-to-noise ration, the minimum spacing between the specimen and microphone can be modified as follows.

6.5.4.1 *Flat Surface*—The closest microphone can be moved to within one-half of the tube diameter, or one-half of the largest section dimension in the case of a rectangular tube.

6.5.4.2 *Nonhomogenous Surface*—The closest microphone should be at least one tube diameter, or the largest section
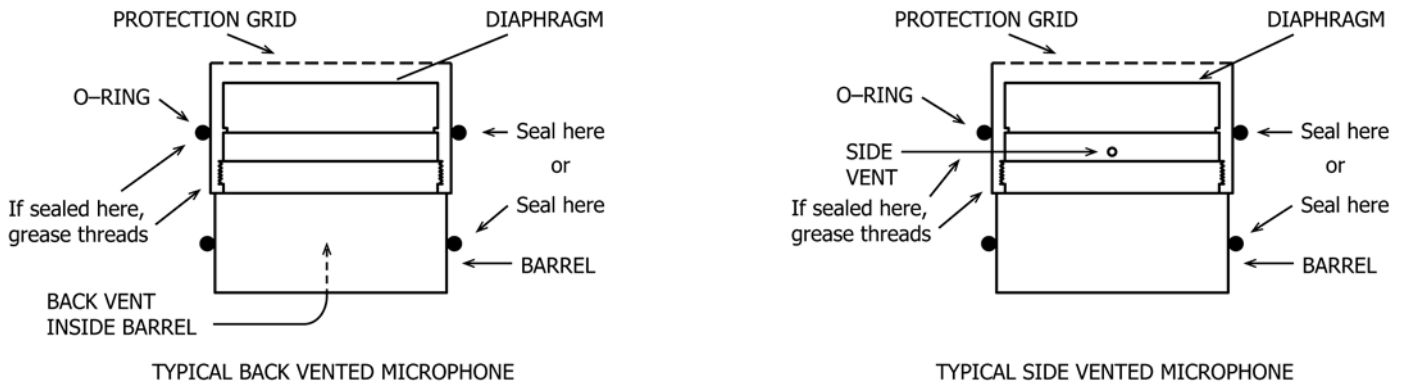
FIG. 2 Microphone Venting and Sealing

dimension in the case of rectangular tube, to help suppress the influence of higher-order modes induced by the rough surface of the specimen.

6.5.4.3 *Asymmetrical Surface*—The closest microphone should be at least two tube diameters (two times the largest section dimension in the case of a rectangular tube) to facilitate the dissipation of higher order modes generated from a rough surface. The higher order modes will decay exponentially as they propagate along the tube. If the specimen is a wedge of the kind used in anechoic rooms, the microphones must be placed far enough from the point of the wedge to measure the transfer function in the plane wave region.

6.5.5 *Mounting*—It is desirable to mount both microphone diaphragms flush with the interior surface of the tube using port openings through the side of the tube. If the microphones are switched (Section 8), care must be taken when the microphones are removed from their port so the original mounting geometry is maintained when they are replaced. A small stop may be employed to control the depth of each microphone in the port as shown in Fig. 3a. The lip should be kept small and identical for both microphone ports.

6.5.5.1 *Alternate Mounting*—In order to avoid the small recess caused by mounting the microphones according to 6.5.5, an alternative pinch block mounting technique may used as shown in Fig. 3b. This method has the advantage of positioning the microphone diaphragm flush with the inside of the impedance tube (the protection grid will protrude slightly) and the

pinch block provides lateral support for the microphone within the port. The pinch block must not interfere with venting provided by the microphone preamplifier.

6.5.5.2 *Microphone Acoustic Centre*—In order to help control measurement uncertainties in this test method, the lateral separation between the microphone axes needs to be precisely known. Unfortunately, the acoustical separation between microphones axes may be different slightly than their physical separation. This uncertainty becomes more pronounced as the ratio between microphone diameter and separation distance increases. Since there is no procedure available for determining the acoustical separation, it is recommended that the physical separation be controlled throughout the test series. If the microphones are switched to reduce phase mismatch errors, as discussed in Section 8, the physical separation should be maintained carefully with the aid of a jig, such as a rotating circular plate mounted in the sidewall of the tube. It is recommended the individual microphones be identified positively to monitor their relative positions during switching (see Fig. 4).

6.6 *Test Signal:*

6.6.1 *Signal Characteristic*—It is recommended that the test signal be random noise having a uniform spectral density across the frequency range of interest. The spectral line spacing of the test signal should be compatible with the analysis bandwidth. Alternative test signals also may be used if they
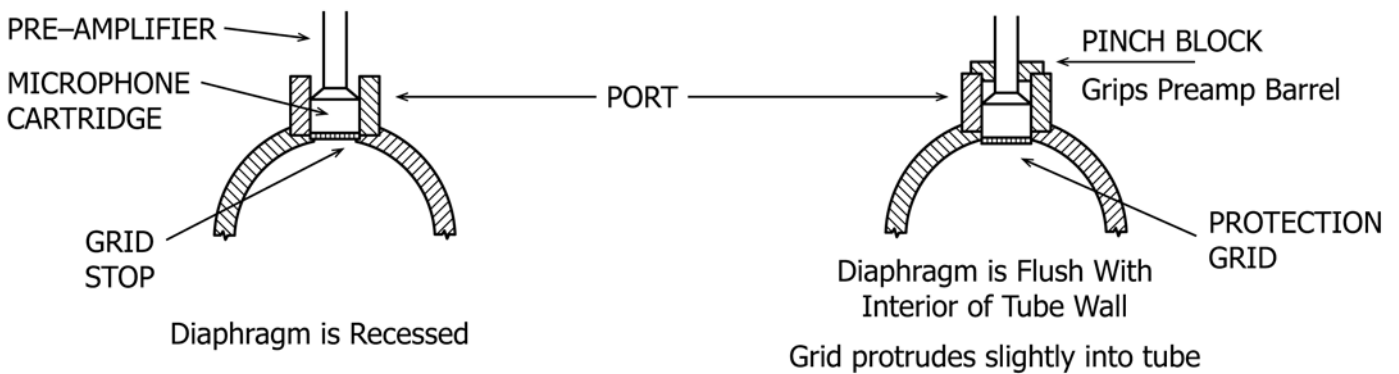

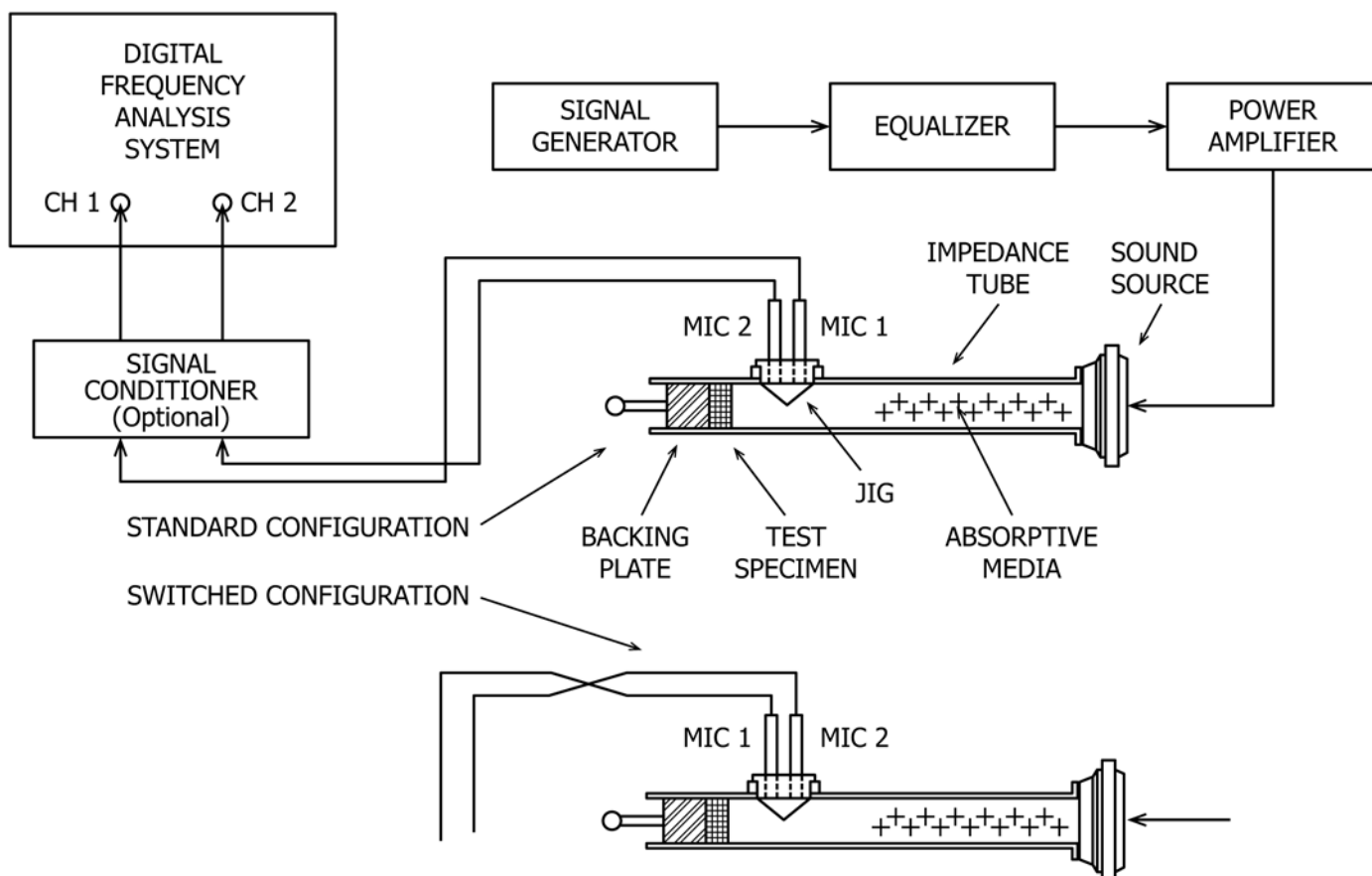FIG. 3 Typical Microphone Mounting and Alternate Mounting

FIG. 4 Apparatus and Instrumentation

have an equivalent spectral density. These alternative signals include pseudo-random noise and swept or stepped sine generation.

NOTE 4—A signal generator capable of producing a compatible test signal often is incorporated within a two-channel digital frequency analysis system. When employing alternative signals, it is recommended that each time block used in the frequency analysis be synchronized with individual repetitions of the test signal pattern.

6.6.2 *Signal-to-Noise Ratio*—The sound source shall generate sufficient signal at both microphone locations such that the measured signal in each test frequency band is at least 10 dB greater than the background noise.

6.7 *Test Measuring Equipment:*

6.7.1 *Measuring Apparatus*—The signal processing equipment shall consist of two identical microphones, two identical analog signal conditioners (optional) and a two-channel Fast Fourier Transform (FFT) analyzer, or equivalent. The signal from each microphone system is connected to an individual channel of the analyzer, (see Fig. 4).

6.7.2 *Computing Device*—A desktop calculator or a microcomputer, either separate from or part of the digital frequency analysis system, may be necessary in order to determine the acoustic absorption coefficient and normal specific impedance ratio from the measured transfer function data. One complete set of mathematical expressions is given in Section 8 (7). Alternative methods of performing such calculations are available (8).

6.7.3 *Temperature Sensor*—A thermometer or other ambient temperature sensing device shall be installed so the air temperature is known to within ± 1°C.

6.7.4 *Barometric Pressure Indicator*—A mercury barometer or other equivalent indicating device shall be located in the vicinity of the impedance tube. The atmospheric pressure shall be measured with a tolerance ±0.5 kPa.

6.7.5 *Relative Humidity Indicator*—A device capable of determining the relative humidity of the air shall be located in the vicinity of the impedance tube. The relative humidity shall be measured to within a tolerance of 5 %.

7. Test Specimen

7.1 *Shape and Size*—Each specimen must have the same shape and area as the tube cross section. It must fit snugly into the specimen holder, not so tightly that is bulges in the center, nor so loosely that there is space between its edge and the holder. It is recommended to seal any peripheral cracks or gaps with petroleum jelly or modelling clay. It is desirable to have the specimen possess a relatively flat surface for reasons stated in 6.5.4.

7.2 *Alignment*—The front surface of test specimens shall be mounted normal to the tube axis unless the surface specifically is designed otherwise. When the specimen has a very uneven back, which would introduce an unintended backing air space, for example, a specimen cut from a hollow concrete block, a minimal amount or putty-like material may be placed between

the specimen and the backing plate. This material will seal and shim the back of the specimen so it is parallel with the backing plate. Such a procedure should be used with caution as the results may be affected by the modification to the back surface of the test specimen. An unknown airspace between the rear of a specimen and the backing plate may be the ruling factor in the result.

7.3 *Containment*—With porous materials of low bulk density, it may be helpful to define the front surface by a thin, nonvibrating wire grid with wide mesh.

7.4 *Number*—A minimum of two specimens should be cut from the sample and tested with the same mounting conditions. When the sample has a surface that is not uniform, for example, a fissured acoustical tile, additional specimens should be selected in order to include representative regions of the surface. In any case, the results should be averaged.

8. Procedure

8.1 *Apparatus and Instrumentation*—A suggested arrangement of apparatus and instrumentation is shown in Fig. 4.

8.2 *Reference Plane*—Once the test specimen has been mounted in the impedance tube according to Section 7, establish the reference plane so the distance, l, to the nearest microphone may be determined. Typically the reference plane is the frontal surface of a flat test specimen. If the test specimen possesses a nonhomogenous or asymmetrical surface, the reference plane may be difficult to establish. In this case, choose one or more planes suited to the intended application including a plane corresponding to the extreme frontal edge of the material.

8.2.1 The exact location of the reference plane has no effect upon the calculated values of normal incidence sound absorption coefficients since they are only a function of the magnitude of the reflection coefficient (see Eq 23*); however, the location does affect the calculation of impedance and admittance ratios.

NOTE 5—The definition of the distance l as used in this test method is not the same as defined in Reference (7) where l is defined as the distance between the reference plane and the microphone farthest away.

8.3 *Calculations*—Measurement of the transfer function between the two microphone signals along with the microphone spacing, the distance from the reference plane to the nearest microphone, and the air temperature are required for the evaluation of the normal incidence acoustical properties of the specimen.

8.3.1 *Speed of Sound*—The speed of sound in air changes with temperature. The value of the speed of sound shall be computed from the measured temperature according to:

$$c = 20.047 \sqrt{273.15 + T} \qquad (4)$$

where:
c = speed of sound, m/s, and
T = room temperature,° C.

8.3.2 *Air Density*—The characteristic impedance of air, ρc, may be found using the following expression for the air density:

$$\rho = 1.290 \left(\frac{P}{101.325} \right) \left(\frac{273.15}{273.15 + T} \right) \qquad (5)$$

where:
ρ = air density, gm/cm³,
P = atmospheric pressure, kPa, and
T = room temperature,° C.

8.4 *Calibration:*

8.4.1 The procedure described herein provides a means of correcting the measured transfer function data for mismatch in both the amplitude and phase responses of the two measurement channels.

8.4.2 *Signal-to-Noise Ratio*—Measure the sound pressure level spectrum at each microphone with the sound source "on" and "off" to assure that the conditions of 6.6.2 are met. A highly absorptive termination must be in place for this procedure. Test data at specific frequencies where the criterion of 6.6.2 is not met, must be identified.

8.4.3 *Averaging Considerations*—The technique of ensemble averaging has the effect of reducing uncertainties due to the variance of random noise, however the ratio of signal to noise is unaltered. The number of averages needed is dependent upon the required precision of the transfer function estimate (see Section 9).

8.4.4 *Windowing*—FFT analysis is made on blocks of data as a time record of finite length. This process is a truncation of a continuous time history requiring the use of a time weighting function (window) to de-emphasize the truncated parts of the time record. A variety of windows are available and each has specific advantages depending on the type of test signal utilized. For the purposes of this test method, the Hanning Window is recommended for measurement of transfer function spectra (9). If synchronized time averaging is considered as an option (see 6.6.1), the best time weighting function is the uniform or boxcar window. Synchronizing the test signal pattern with the time blocks eliminates truncation (leakage) altogether.

8.4.4.1 If the time averaging method is selected and if the time blocks are synchronized with a repeated test signal pattern, unsynchronized "noise" will be reduced by 10/log/n decibels, where n is the number of averages (10).

8.4.5 *Amplitude and Phase Corrections*—Since the transfer is a complex ratio of the acoustic pressure responses, any mismatch in the amplitude or phase responses of the two microphone systems will affect the accuracy of the transfer function measurement. The following sequence of measurements and computations provides a means for correcting the measured transfer data in both measurement channels.

8.4.5.1 A specimen must be highly absorptive to ensure the calibration procedure of 8.4 is effective. Unfortunately, most materials exhibit low absorption at low frequencies, thus limiting the accuracy of the final results.

8.4.6 Throughout the remainder of this section a series of equations are presented that represent working expressions based on the theoretical development (7). The working expressions should be used to perform the required calculations. The working expressions are presented in terms of the real and imaginary components of the complex transfer function, thus

requiring the FFT analyzer be set up to provide spectral data in this form. These working expressions are identified with an * adjacent to the corresponding equation number.

8.4.6.1 *Step 1*—Place a highly absorptive specimen (see 8.4.5) in the tube to prevent strong acoustic reflections and to obtain the most accurate correction factor possible. Measure the following two transfer functions using the same computational algorithms for both.

8.4.6.2 Place the microphones in the standard configuration of Fig. 4, and measure as follows:

$$\bar{H}^I = \left| \bar{H}^I \right| e^{j\bar{\varphi}^I} = \bar{H}_r^I + j\,\bar{H}_i^I \qquad (*6)$$

8.4.6.3 Interchange the microphone locations to assume the switched configuration as shown in Fig. 4 and measure as follows:

$$\bar{H}^{II} = \left| \bar{H}^{II} \right| e^{j\bar{\varphi}^{II}} = \bar{H}_r^{II} + j\,\bar{H}_i^{II} \qquad (*7)$$

Care should be taken when interchanging the microphones to ensure that microphone one in the switched configuration occupies the precise location that microphone two occupied in the standard configuration, and vice versa (see 6.5.5.2).

8.4.6.4 *Step 2*—The following equations are valid for the case where the digital frequency analysis system always uses channel one as the reference channel. An alternative set of equations are presented in Appendix X1, which may be more convenient for systems where the reference analysis channel also can be switched. Compute the calibration factor $\bar{H}_c$ representing the amplitude and phase mismatches $|\bar{H}_c|$ and $\bar{\varphi}_c$, using the following equation:

$$\bar{H}_c = \left(\bar{H}^I \times \bar{H}^{II} \right)^{1/2} = \left| \bar{H}_c \right| e^{j\bar{\varphi}_c} \qquad (8)$$

where:

$$\left| \bar{H}_c \right| = \left(\left| \bar{H}^I \right| \times \left| \bar{H}^{II} \right| \right)^{1/2} = \left\{ \left[\left(\bar{H}_r^I \right)^2 + \left(\bar{H}_i^I \right)^2 \right] \times \left[\left(\bar{H}_r^{II} \right)^2 + \left(\bar{H}_i^{II} \right)^2 \right] \right\}^{1/4} \qquad (*9)$$

$$\bar{\varphi}_c = \frac{1}{2} \left(\bar{\varphi}^I + \bar{\varphi}^{II} \right) = \frac{1}{2} Tan^{-1} \left[\frac{\bar{H}_i^I \bar{H}_r^{II} + \bar{H}_r^I \bar{H}_i^{II}}{\bar{H}_r^I \bar{H}_r^{II} - \bar{H}_i^I \bar{H}_i^{II}} \right] \qquad (*10)$$

where it is assumed that the phase mismatch is between $-\pi/2$ and $\pi/2$ radians.

8.4.6.5 *Step 3*—For subsequent tests, place the microphones in the standard configuration. Insert the test specimen, and measure the transfer function as follows:

$$\bar{H} = \left| \bar{H} \right| e^{j\bar{\varphi}} = \bar{H}_r + j\,\bar{H}_i \qquad (*11)$$

8.4.6.6 Correct for mismatch in the microphone responses using the following equation:

$$H = \bar{H}/\bar{H}_c = \left| H \right| e^{j\bar{\varphi}} = H_r + j\,H_i \qquad (12)$$

where:

$$\left| H \right| = \left| \bar{H} \right| / \left| \bar{H}_c \right| \quad, \quad \varphi = \bar{\varphi} - \bar{\varphi}_c \qquad (13)$$

$$H_r = \frac{1}{\left| \bar{H}_c \right|} \left(\bar{H}_r cos\bar{\varphi}_c + \bar{H}_i sin\bar{\varphi}_c \right) \qquad (*14)$$

$$H_i = \frac{1}{\left| \bar{H}_c \right|} \left(\bar{H}_i cos\bar{\varphi}_c - \bar{H}_r sin\bar{\varphi}_c \right) \qquad (*15)$$

The values of H, $|H|$, φ, H_r and H_i are then used in the expressions of 8.5.3 to determine the reflection coefficient of the test specimen.

8.5 *Measurements:*

8.5.1 *Transfer Function*—Insert the test specimen and measure the complex acoustic transfer function, which is defined as follows: (see Section 9 to minimize estimation errors).

$$\bar{H} = \frac{G_{12}}{G_{11}} = \left| \bar{H} \right| e^{j\bar{\varphi}} = \bar{H}_r + j\,\bar{H}_i \qquad (*16)$$

8.5.1.1 Eq 16* determines the transfer function by taking the ratio of the cross power spectrum to the input auto power spectrum. The transfer function also could be calculated directly from the complex ratio of the Fourier transform of the acoustic pressure at the microphone nearest the test specimen to the Fourier transform of the acoustic pressure at the microphone nearest the sound source. For a single measurement both methods yield identical results. When averaging is employed (see 8.4.3), the method of calculation shown in Eq 16* reduces the effects of noise. Since all FFT analysis systems do not define the cross spectrum consistently, adherence to the definitions implied in Eq 16* must be strictly observed.

8.5.2 *Mismatch Correction*—Using the method described in 8.4, correct $\bar{H}$ for mismatch arising from the microphone amplitude and phase responses, and use the corrected transfer function, H, in the following computations.

8.5.3 *Complex Reflection Coefficient*—Calculate the complex reflection coefficient as follows:

$$R = \left| R \right| e^{j\varphi R} = R_r + j\,R_i = \frac{H - e^{-jks}}{e^{jks} - H} e^{j\,2k\,(l+s)} \qquad (17)$$

$$\left| R \right| = \left[\frac{1 + \left| H \right|^2 - 2\left| H \right| cos(\varphi + ks)}{1 + \left| H \right|^2 - 2\left| H \right| cos(\varphi - ks)} \right]^{1/2} \qquad (18)$$

$$\varphi_R = 2k(l+s) + Tan^{-1} \left[\frac{2\left| H \right| cos\varphi sin(ks) - sin(2ks)}{\left| H \right|^2 - 2\left| H \right| cos\varphi cos(ks) + cos(2ks)} \right] \qquad (19)$$

$$R_r = \{ 2H_r cos[k(2l+s)] - cos(2kl) - (H_r^2 + H_i^2)cos[2k(l+s)] \}/D \qquad (*20)$$

$$R_i = \{2 H_r sin[k(2\,l+s) - sin(2kl) - (H_r^2 + H_i^2) sin[2k(l+s)] \}/D \qquad (*21)$$

$$D = 1 + H_r^2 + H_i^2 - 2[H_r cos(ks) + H_i sin(ks)] \qquad (*22)$$

8.5.4 *Normal Incidence Sound Absorption Coefficient*—Calculate the normal incidence sound absorption coefficient as follows:

$$\alpha = 1 - \left| R \right|^2 = 1 - R_r^2 - R_i^2 \qquad (*23)$$

8.5.5 *Normal Specific Acoustic Impedance Ratio*—Calculate the normal specific acoustic impedance ratio as follows:

$$z/\rho c = r/\rho c + jx/\rho c = (1+R)/(1-R) \qquad (24)$$

8.5.5.1 *Normal Specific Acoustic Resistance Ratio*—Calculate the normal specific acoustic resistance ratio as follows:

$$r/\rho c = \alpha/[2(1 - R_r) - \alpha] \qquad (*25)$$

8.5.5.2 *Normal Specific Acoustic Reactance Ratio*—Calculate the normal specific acoustic reactance ratio as follows:

$$x/\rho c = 2\, R_i /\left[2\left(1 - R_r\right) - \alpha\right] \qquad (*26)$$

8.5.6 *Normal Specific Acoustic Admittance Ratio*—Calculate the normal specific acoustic admittance ratio as follows:

$$y\rho c = g\rho c - j\, b\rho c = \rho c/z \qquad (27)$$

8.5.6.1 *Normal Specific Acoustic Conductance Ratio*—Calculate the normal specific acoustic conductance ratio as follows:

$$g\rho c = \frac{r/\rho c}{(r/\rho c)^2 + (x/\rho c)^2} \qquad (*28)$$

8.5.6.2 *Normal Specific Acoustic Susceptance Ratio*—Calculate the normal specific acoustic susceptance ratio as follows:

$$b\rho c = \frac{x/\rho c}{(r/\rho c)^2 + (x/\rho c)^2} \qquad (*29)$$

8.6 *Estimation of ⅓ octave and ¹⁄₁ octave results:*

8.6.1 An estimation of ⅓ octave or ¹⁄₁ octave absorption coefficients may be calculated by:

$$\alpha_{band} = \frac{1}{n} \sum_{i=1}^{n} \alpha_{f_i} \qquad (6)$$

Where:

α_{band} = is the absorption coefficient of the desired ⅓ octave or ¹⁄₁ octave band,
n = is the number of bins in the band,
fi = is a frequency bin in the desired ⅓ octave or ¹⁄₁ octave to be estimated, and
α_{f_i} = is the absorption measured by this test method in the indicated frequency bin.

8.6.2 The upper and lower frequency limits for a ⅓ octave or ¹⁄₁ octave band are defined in ANSI/ASA S1.11.

8.6.3 The results for a given ⅓ octave or ¹⁄₁ octave band shall not be reported if that band contains less than the larger of 10 or $(f_u - f_l)$ / Δf bins where f_u and f_l are the upper and lower frequency limits of the band and Δf is the frequency spacing of the bins.

9. Sources of Error

9.1 *Estimation Errors*—Transfer function estimates are made from sample records of finite duration and frequency resolution and are susceptible to random and bias errors.

9.2 *Random Error*—Random error generally is kept low by ensemble averaging, that is, measuring several individual estimates and computing the average. Frequency smoothing, that is, averaging together the results for several frequency bands also may be employed. Typically, a product of filter bandwidth and record sample length (BT product) of 50 to 100 will keep random error sufficiently low. The number of averages performed on an FFT analyzer is essentially the same as the BT product.

9.2.1 Alternatively, the averaging time required to achieve a desired error level using a linear detector is given by (**11**):

$$T \approx \frac{G^2}{2\, B\, e^2} \qquad (30)$$

where:

T = averaging time, s,
G = confidence limit factor (= 11.91 for 95 % confidence limits),
B = filter bandwidth, hz, and
e = error, decibels.

9.3 *Bias Error*—Bias errors include errors in distance from the specimen, as well as differences between acoustic and geometric centers of microphones. Bias also can arise from uncorrected phase and amplitude mismatch in the microphones and from computational errors in post processing. These bias errors shall be considered part of the uncertainties associated with this test method.

9.3.1 *Time Aliasing*—Time aliasing arises when the duration of each record is similar to or less than the response function of the system. This type of bias error will be low, provided that the time length of each sample record is much larger than the acoustical propagation times within the impedance tube system, that is:

$$t >> \frac{2\,(l+s)}{c} \qquad (31)$$

where:

t = the sample record length, s,
l = the distance from the test sample to the nearest microphone, m,
s = microphone spacing, m, and
c = the speed of sound, m/s.

9.3.2 *Tube Attenuation*—The incident and reflected sound waves that propagate within the tube are subject to attenuation due to viscous and thermal losses. This effect causes the loci of pressure minimums to shift asymmetrically in the standing wave pattern as distance from the specimen increases (loci of maximums are minimally affected). Since the microphone positions are placed relatively close to the specimen face tube attenuation normally will not affect the results obtained from this test method.

9.3.2.1 If the distance from the face of the specimen to the nearest microphone exceeds about three tube diameters than tube attenuation must not be ignored. To account for tube attenuation, the equations involving the calculation of the reflection coefficient (Eq 17) must be modified by replacing the real variable k by the complex wave number $k' - jk''\cdot k'$ is the same as the real component of the wave number, $2\pi f/c$. The attenuation constant k'' can be found experimentally by using the technique found in C384 and requires the standing wave pattern to be explored with a traversing microphone. Alternatively, k'' may be expressed empirically by (**12**):

$$k'' = 0.02203\, \sqrt{f}/cd \qquad (32)$$

where:

k'' = attenuation constant, nepers/m,
f = frequency, hz,

c = speed of sound in the tube, m/s, and
d = diameter of the tube, m.

9.3.3 It has been shown by two studies[6] that variations in the characteristic impedance of air, ρc, in the impedance tube can affect the absorption coefficients measured by this test method. However, these effects are complex and dependent upon the impedance of the test sample and the frequency of sound. These systematic effects shall be considered part of the uncertainties of this test method.

10. Report

10.1 The report shall include the following information:

10.1.1 A statement, if true in all respects, that the test was performed in accordance with this test method. If the alternate calibration procedure in Appendix X1 or tube corrections as described in 9.3.2 are used then this shall be stated in the report. Any tube attenuation correction technique employed shall be described in detail.

10.1.2 A description of the sample adequate to identify another sample of the same material.

10.1.3 A description of the test specimen including their number, size, and method of mounting.

10.1.4 The air temperature at the time of test.

10.1.5 The relative humidity at the time of the test.

10.1.6 The barometric pressure at the time of the test.

10.1.7 The characteristic impedance (ρc) of air at the time of test using equations in 8.3.1 for c and 8.3.2 for ρ.

10.1.8 A tabular listing by frequency band of the absorption coefficients (to two significant figures).

10.1.9 If several measurements are made, include the individual results, as well as the averaged results. Results presented using a method other than arithmetic averaging, must be clearly identified.

10.2 A description of the instruments used and the details of the procedure also shall be considered part of the report. Signal processing parameters, such as the frequency resolution, the number of averages, and the windowing function also must be included.

10.3 The inclusion of following information in the test report is optional:

10.3.1 A tabular listing of the resistance and reactance ratios as a function of frequency (to two significant figures). The designated reference plane must be identified clearly.

10.3.2 A tabular listing the conductance and susceptance ratios as a function of frequency (to two significant figures).

11. Precision and Bias

11.1 Measurements described in this test method can be made with greater precision than is needed. Imprecision in this test method arises from sources other than the measurement procedure. Some materials are not very uniform so that specimens cut from the same sample differ in their properties. There is uncertainty in deciding on the location of the face of a very porous specimen. The largest causes of imprecision are related to the preparation and installation of the test specimen. The specimen must be precisely cut. The fit must not be too tight or too loose. Irregular nonreproducible airspaces behind the specimen must be prevented.

11.2 Measurements of the microphone spacing and the distance from the material surface to the center of the nearest microphone must be made to within 0.1 mm for those materials that have a well defined surface.

11.3 No quantitative statement on bias can be made at this time since there is presently no material available with known true values of performance, which can be used for determining the bias of this test method.

11.4 The within- and between-laboratory precision of this test method, expressed in terms of the within-laboratory, 95 % Repeatability Interval, $I(r)$, and the between-laboratory, 95 %, Reproducibility Interval, $I(R)$, is listed in Table 2. These statistics are based on the results of a round-robin test program involving ten laboratories.[7]

11.5 The significance of the Repeatability and Reproducibility Intervals is as follows:

11.5.1 *Repeatability Interval, I(r)*—In the same laboratory on the same material, the absolute value of the difference in two test results will be expected to exceed $I(r)$ only about 5 % of the time.

11.5.2 *Reproducibility Interval, I(R)*—In different laboratories on the same material, the absolute value of the difference in two test results will be expected to exceed $I(R)$ only about 5 % of the time.

12. Keywords

12.1 absorption; frequency response function; impedance; impedance ratio; impedance tube; normal incidence sound absorption coefficient; specific normal acoustic impedance; transfer function

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[6] Supporting data, Summary of Data and Analysis of Task Group E33.05E, Effect of Barometric pressure on Normal Incidence Sound Absorption Coefficients as Measured Using the ASTM E1050 Standard Test Method" have been filed at ASTM International Headquarters and may be obtained by requesting Research Report RR:E33-1012.

[7] Supporting data have been filed at ASTM International Headquarters and may be obtained by requesting Research Report RR:E33-1006.

TABLE 2 Within-Laboratory Repeatability, I(r), and Between-Laboratory Reproducibility, I(R)

Variable	Statistic	125	250	500	1000	2000	4000
$x/\rho c$	$I(r)$	3.4	0.4	0.3	0.1	0.1	0.5
	$I(R)$	14.7	3.3	1.5	0.4	0.2	0.8
$r/\rho c$	$I(r)$	2.4	0.5	0.2	0.3	0.2	0.2
	$I(R)$	8.1	1.7	0.6	0.3	0.3	0.5
α	$I(r)$	0.04	0.02	0.04	0.05	0.01	0.04
	$I(R)$	0.09	0.08	0.11	0.12	0.03	0.07

ANNEX

(Mandatory Information)

A1. LABORATORY ACCREDITATION

A1.1 Scope

A1.1.1 This annex describes procedures to be followed in accrediting a testing laboratory to perform tests in accordance with this test method.

A1.2 Summary of Procedures

A1.2.1 The laboratory shall allow the accrediting agency to make an on-site inspection.

A1.2.2 The laboratory shall show that is satisfies the criteria of Guide E548.

A1.2.3 The laboratory shall show that it is in compliance with the mandatory parts of this test method, that is, those parts that contain the words shall or must.

A1.2.4 The laboratory shall show the construction and geometry of the tube and specimen holder as described in 6.2 and 6.3.

A1.2.5 The laboratory shall show calculations verifying the tube diameter in accordance with 6.2.3.

A1.2.6 The laboratory shall show the sound source and that its frequency response is in according with 6.4.

A1.2.7 The laboratory shall show that the signal-to-noise ratio of the source is adequate in accordance with 6.6.2.

A1.2.8 The laboratory shall report the phase response correction procedure used (see 8.4).

A1.2.9 The laboratory shall report the type of test signal used (see 6.6.1).

A1.2.10 The laboratory shall show sample calculations or the computer program used to evaluate the equations in 8.4 and 8.5.

A1.3 Reference Tests

A1.3.1 The laboratory shall maintain a reference specimen to be used during the periodic tests for quality assurance. It shall be so constructed or formed that it will not deteriorate quickly with use. Presumably, its absorptive properties should remain stable during at least ten years of use. As measured by this test method, the sound absorption coefficients of the reference specimen shall be at least 0.20 at 250 Hz and above.

A1.3.2 For the frequency range of interest, the laboratory shall measure the sound absorption coefficients of the backing plate a minimum of once per year. Results should be compared with calculated results or with results reported in the literature (7).

A1.3.3 The sound absorption coefficients and their standard deviations shall be analyzed by the control chart method described in MNL 7 (13). The analysis shall be in accordance with the section entitled "Control-No Standard Given".

APPENDIX

(Nonmandatory Information)

X1. ALTERNATE CALIBRATION FACTOR MEASUREMENT

X1.1 Scope

X1.1.1 The information provided in this appendix is for those using a two channel FFT analyzer intended for use in conjunction with a computing device. When such a system is used, it may be more convenient to assume the complex pressure at microphone position one as the reference for both the standard and switched microphone configurations. Channel One reference for the transfer function measurement when the microphones are in the standard configuration. Channel two is the reference when the microphones are interchanged to assume the switched configuration.

X1.2 Procedure

X1.2.1 The following is intended to replace 8.4.6.4, *Step 2*, which describes a procedure to compute the calibration factor $\bar{H}$. All other considerations and computations remain the same as those described in the main body of this test method.

X1.2.2 *Step 1*—Use 8.4.6.1 – 8.4.6.3.

X1.2.3 *Step 2*—Compute the calibration factor $\bar{H}_c$ representing the amplitude and phase mismatches $|H_c|$ and $\bar{\varphi}_c$, using the following equation:

$$\bar{H}_c = \left(\bar{H}^I / \bar{H}^{II} \right)^{1/2} = \left| \bar{H}_c \right| e^{j \bar{\varphi}_c} \tag{X1.1}$$

where:

$$\left| \bar{H}_c \right| = \left(\left| \bar{H}^I \right| / \left| \bar{H}^{II} \right| \right)^{1/2} = \left\{ \left[\left(\left| \bar{H}_r^I \right| \right)^2 + \left(\bar{H}_i^I \right)^2 \right] / \left[\left(\bar{H}_r^{II} \right)^2 + \left(\bar{H}_i^{II} \right)^2 \right] \right\}^{1/4} \tag{X1.2}$$

$$\bar{\varphi} = \frac{1}{2} \left(\bar{\varphi}^I - \bar{\varphi}^{II} \right) = \frac{1}{2} Tan^{-1} \left[\frac{\bar{H}_i^I \bar{H}_r^{II} - \bar{H}_r^I \bar{H}_i^{II}}{\bar{H}_r^I \bar{H}_r^{II} + \bar{H}_i^I \bar{H}_i^{II}} \right] \tag{X1.3}$$

where it is assumed that the phase mismatch is between -π/2 and π/2 radians.

X1.2.4 *Step 3*—Continue this procedure at 8.4.6.5.

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