Designation: E2611 – 17

Standard Test Method for Normal Incidence Determination of Porous Material Acoustical Properties Based on the Transfer Matrix Method[1]

This standard is issued under the fixed designation E2611; 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 a tube, four microphones, and a digital frequency analysis system for the measurement of normal incident transmission loss and other important acoustic properties of materials by determination of the acoustic transfer matrix.

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

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

1.4 *This international standard was developed in accordance with internationally recognized principles on standardization established in the Decision on Principles for the Development of International Standards, Guides and Recommendations issued by the World Trade Organization Technical Barriers to Trade (TBT) Committee.*

2. Referenced Documents

2.1 *ASTM Standards:*[2]

C634 Terminology Relating to Building and Environmental Acoustics

E90 Test Method for Laboratory Measurement of Airborne Sound Transmission Loss of Building Partitions and Elements

E1050 Test Method for Impedance and Absorption of Acoustical Materials Using a Tube, Two Microphones and a Digital Frequency Analysis System

2.2 *ISO Standards:*

ISO 140-3 Acoustics—Measurement of Sound Insulation in Buildings and of Building Elements—Part 3: Laboratory Measurement of Airborne Sound Insulation of Building Elements[3]

3. Terminology

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

3.1.1 *reference plane*—an arbitrary section, perpendicular to the longitudinal axis of the tube that is used for the origin of linear dimensions. Often it is the upstream (closest to the sound source) face of the specimen but, when specimen surfaces are irregular, it may be any convenient plane near the specimen.

3.1.2 *sound transmission coefficient, τ*—(dimensionless) of a material in a specified frequency band, the fraction of airborne sound power incident on a material that is transmitted by the material and radiated on the other side.

$$\tau = \frac{W_t}{W_i}$$

where:

W_t and W_i = the transmitted and incident sound power.

3.1.3 *sound transmission loss, TL*—of a material in a specified frequency band, ten times the common logarithm of the reciprocal of the sound transmission coefficient. The quantity so obtained is expressed in decibels.

$$TL = 10\log_{10}\left(\frac{W_i}{W_t}\right) = 10\log_{10}\left(\frac{1}{\tau}\right)$$

3.1.3.1 *Discussion*—In this standard the symbol TL_n will be applied to sound which impinges at an angle normal to the test specimen, as opposed to an arbitrary or random angle of incidence.

3.2 *Symbols:*

c = speed of sound, m/s.

ρ = density of air, kg/m^3.

f = frequency, hertz, (Hz).

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[1] This test method is under the jurisdiction of ASTM Committee E33 on Building and Environmental Acoustics and is the direct responsibility of Subcommittee E33.01 on Sound Absorption.

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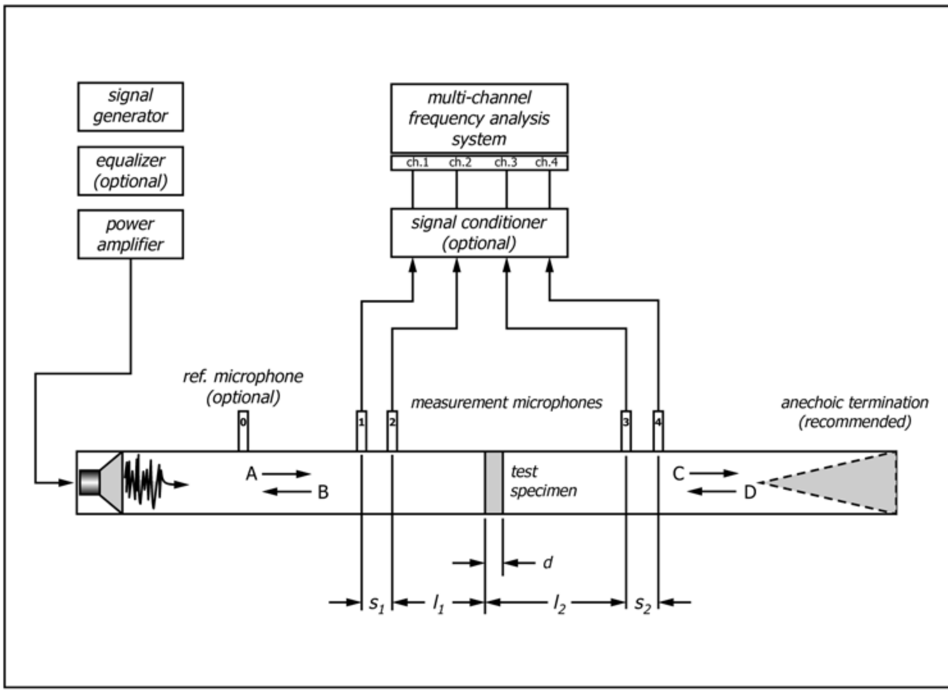
[2] For referenced ASTM standards, visit the ASTM website, www.astm.org, or contact ASTM Customer Service at service@astm.org. For *Annual Book of ASTM Standards* volume information, refer to the standard's Document Summary page on the ASTM website.

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



NOTE 1—A, B, C, and D are the forward and backward components of the standing wave field. 1, 2, 3, and 4 are the measurement locations; 0 is an optional reference location. Distances are measured from the specimen reference plane.

FIG. 1 Schematic Drawing of Measurement Setup

G_{11}, G_{22}, etc. = auto power spectra (autospectrum) of the acoustic pressure signal at microphone locations 1, 2, and so on.

G_{21}, G_{32}, etc. = cross power spectrum (cross spectrum) of the acoustic pressure signals at location 2 relative to location 1, 3 relative to 1, and so on. In general, a complex value.

$\bar{H}_{21}$, $\bar{H}_{31}$, etc. = measured transfer function of the acoustic pressure signals at location 2 relative to location 1, 3 relative to 1, and so on. In general, a complex value. Note that H_{11} is purely real and equal to 1.

H^I, H^{II} = calibration transfer functions for the microphones in the standard and switched configurations, respectively. See 8.4.

H^c = complex microphone calibration factor accounting for microphone response mismatch.

H_{21}, H_{31}, etc. = transfer function of two microphone signals corrected for microphone response mismatch. In general, a complex value.

NOTE 1—In this context, the term "transfer function" refers to the complex ratio of the Fourier transform of two signals. The term "frequency response function" arises from more general linear system theory (**1**).[4] This test method shall retain the use of the former term. Users should be aware that modern FFT analyzers might employ the latter terminology.

$j = \sqrt{-1}$

$k = 2\pi f/c$; wave number in air, m^{-1}.

NOTE 2—In general the wave number is complex where $k' = k^r - jk^i$. k^r is the real component, $2\pi\, f/c$, and k^i is the imaginary component of the wave number, also referred to as the attenuation constant, nepers/m. This accounts for the effects of viscous and thermal dissipation in the oscillatory, thermoviscous boundary layer that forms on the inner surface of the duct, (**2**). The wave number k' of the propagating wave interior to the material being tested is generally different from that in air, and may be calculated in certain cases from the acoustic transfer matrix.

d = thickness of the specimen in meters; see Fig. 1.

11, 12 = distance in meters from the reference plane (test sample front face) to the center of the nearest microphone on the upstream and downstream side of the specimen; see Fig. 1.

$s1$, $s2$ = center-to-center spacing in meters between microphone pairs on the upstream and downstream side of the specimen; see Fig. 1.

R = complex acoustic reflection coefficient.

α = normal incidence sound absorption coefficient.

TL_n = normal incidence transmission loss.

k' = complex wavenumber of propagation in the material, m^{-1}.

Z = characteristic impedance of propagation in the material, rayls.

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

Xc = calibration.

XI, XII = calibration quantities measured with microphones placed in the standard and switched configurations, respectively.

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

$|X|$ = magnitude of a complex quantity.

φ = phase of a complex quantity in radians.

Xi = imaginary part of a complex quantity.

[4] The boldface numbers in parentheses refer to the list of references at the end of this standard.

Xr = real part of a complex quantity.

3.4 *Summary of Complex Arithmetic*—The quantities in this standard, especially the transfer function spectra, are complex-valued in general. The following may be useful in evaluating the defining equations:

$$e^{j\omega} = \cos(\omega) + j\sin(\omega)$$

$$(A+jB) \times (C+jD) = (AC+BD) + j(AD+BD)$$

$$1/(A+jB) = A/(A^2+B^2) - jB/(A^2+B^2)$$

4. Summary of Test Method

4.1 This test method is similar to Test Method E1050 in that it also uses a tube with a sound source connected to one end and the test sample mounted in the tube. For transmission loss, four microphones, at two locations on each side of the sample, are mounted so the diaphragms are flush with the inside surface of the tube perimeter. Plane waves are generated in the tube using a broadband signal from a noise source. The resulting standing wave pattern is decomposed into forward- and backward-traveling components by measuring sound pressure simultaneously at the four locations and examining their relative amplitude and phase. The acoustic transfer matrix is calculated from the pressure and particle velocity, or equivalently the acoustic impedance, of the traveling waves on either side of the specimen. The transmission loss, as well as several other important acoustic properties of the material, including the normal incidence sound absorption coefficient, is extracted from the transfer matrix.

5. Significance and Use

5.1 There are several purposes of this test:

5.1.1 For transmission loss: *(a)* to characterize the sound insulation characteristics of materials in a less expensive and less time consuming approach than Test Method E90 and ISO 140-3 ("reverberant room methods"), *(b)* to allow small samples tested when larger samples are impossible to construct or to transport, *(c)* to allow a rapid technique that does not require an experienced professional to run.

5.1.2 For transfer matrix: *(a)* to determine additional acoustic properties of the material; *(b)* to allow calculation of acoustic properties of built-up or composite materials by the combination of their individual transfer matrices.

5.2 There are significant differences between this method and that of the more traditional reverberant room method. Specifically, in this approach the sound impinges on the specimen in a perpendicular direction ("normal incidence") only, compared to the random incidence of traditional methods. Additionally, revereration room methods specify certain minimum sizes for test specimens which may not be practical for all materials. At present the correlation, if any, between the two methods is not known. Even though this method may not replicate the reverberant room methods for measuring the transmission loss of materials, it can provide comparison data for small specimens, something that cannot be done in the reverberant room method. Normal incidence transmission loss may also be 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 or portable electronic device.

5.3 Transmission loss is not only a property of a material, but is also strongly dependent on boundary conditions inherent in the method and details of the way the material is mounted. This must be considered in the interpretation of the results obtained by this test method.

5.4 The quantities are measured as a function of frequency with a resolution determined by the sampling rate, transform size, and other parameters 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 microphone spacings.

5.5 The application of materials into acoustical system elements will probably not be similar to this test method and therefore results obtained by this method may not correlate with performance in-situ.

6. Apparatus

6.1 The apparatus is a set of two tubes of equal internal area that can be connected to either end of a test sample holder. The number of sets of tubes depends on the frequency range to be tested. A wider frequency range may require multiple measurements on a set of several tubes. At one end of one tube is a loudspeaker sound source. Microphone ports are mounted at two locations along the wall of each tube. A two- or four-channel digital frequency analysis system, or a computer that can effectively do the same calculations, 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 and shall have a constant cross-sectional dimension from end-to-end. The tube shall be straight and its inside surface shall be smooth, nonporous, and free of dust, in order to maintain low sound attenuation. The tube construction shall be sufficiently massive so sound transmission through the tube wall is negligible compared with transmission though the sample. See Note 3. Compliant feet or mounts must be used to attenuate extraneous vibration entering the tube structure from the work surface.

NOTE 3—The tube can be constructed from materials including metal, plastic, concrete, 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:

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

where:

f = operating frequency, Hz,
f_l = lower working frequency of the tube, Hz, and
f_u = upper working frequency of the tube, Hz.

6.2.3 The lower frequency limit f_l is determined by the spacing of the microphones and the accuracy of the analysis system. The microphone spacing shall be greater than one percent of the wavelength corresponding to the lower frequency of interest.

6.2.4 The upper frequency limit f_u depends on the diameter of the tube, the microphone spacing, and the speed of sound.

6.2.4.1 *Diameter*—In order to maintain plane wave propagation, the upper frequency limit (**3**) is defined as follows:

$$f_u < \frac{Kc}{d} \quad \text{or} \quad d < \frac{Kc}{f_u} \tag{2}$$

where:

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

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

6.2.6 Conduct the plane wave measurements within these frequency limits established by Eq 1 in order to avoid cross-modes that occur at higher frequencies, when the acoustical wave length approaches the sectional dimension of the tube.

6.2.7 *Length*—The tube should be sufficiently long for plane waves to be 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 non-plane waves along with desired plane waves. The non-plane 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 spacing, or to employ separate tubes. The overall tube length also must be chosen to satisfy the requirements of 6.5.3 and 6.5.5.

6.2.8 *Tube Termination*—The termination of the tube is arbitrary in principle, but experience has found that the most useful termination is at least weakly anechoic, causing minimal reflection of the sound wave back down the tube. A convenient way of providing this is to install a wedge or pyramidal shaped section of some sound absorbing material such as glass fiber, about 30 cm long, in the open end of the tube. As the two-load method requires a second measurement with a different tube termimation, the wedge should be easily removable so that an open or closed termination may be provided.

6.2.9 *Tube Venting*—Some tube designs cause large temporary pressure variations to be generated during installation or removal of the test specimen. This may induce microphone diaphragm deflection. By including a pressure relief opening of some type, the potential for damage to a microphone diaphragm due to excessive deflection may be reduced. One way to accomplish this is by drilling a small vent, 1 to 2 mm in diameter, 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 tube or may be a separate, detachable extension of the tube.

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 to the interior shape and dimensions of the main part of the tube. The connecting joint must be finished carefully and the use of a sealant, such as petroleum jelly or silicone grease, is recommended.

6.3.3 *Integral Holder*—If the sample holder is in an integral part of the 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.

6.3.4 *Circular Holder*—For circular tubes, it is recommended to make the specimen accessible from both the front and back of the sample holder. It is then possible to check the position and flatness of the front and back surface of the specimen. 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.

6.4 *Sound Source:*

6.4.1 *Kind and Placement*—The sound source 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 (Fig. 2).

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 tube. If a direct radiator loudspeaker is

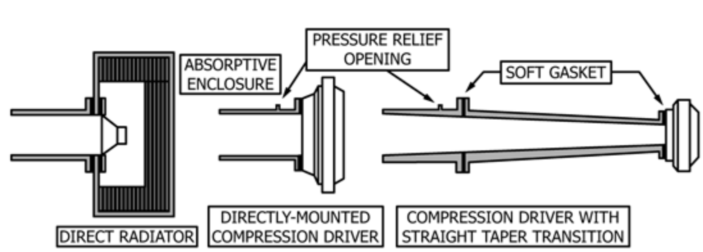


FIG. 2 Sound Source Configuration

utilized, it shall be contained in a sound-isolating enclosure in order to avoid airborne flanking transmission to the microphones (Fig. 2).

6.4.3 *Termination*—Resonances of the air column in the 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 tube near the loudspeaker or inside the sound transition. Alternatively, the locations described above may be lightly filled 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 then be required to shape the sound spectra measured 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*—Nominally identical microphones shall be mounted according to 6.5.6. 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 of 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 (e.g., on the transmitted side of high loss samples), larger diameters may be selected for use with large tubes working at low frequencies.

6.5.2 *Microphone Venting*—Some microphones may be 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) and side-vented. A microphone pair of either design may be used. Microphones must be sealed carefully when installed in the normal sound transmission 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 tube. Very low frequency accuracy is improved when the static pressure equalization vent is isolated from the sound field within the sound transmission tube (**4**). 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 (Fig. 3).

6.5.2.2 *Side-Vented Microphones*—The 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 (Fig. 3).

6.5.2.3 *Non-Vented Microphones*—Microphones with non-vented diaphragms may be used if appropriate.

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 wavelength of interest (**5**).

$$s << \frac{c}{2f_u} \tag{3}$$

where:

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

6.5.4 The maximum microphone spacing s must be no larger than 80 % of $c/2f_u$.

6.5.5 *Location*—The minimum distance between the sound source and the closest microphone must follow the requirements of 6.2.7. 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 ratio, the minimum spacing between the specimen and microphone can be modified as follows.

6.5.5.1 *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.

6.5.5.2 *Nonhomogenous Surface*—The closest microphone should be at least one tube diameter, or the largest section dimension in the case of rectangular tube, to help suppress the influence of higher-order modes induced by the rough surface of the specimen.

6.5.5.3 *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.

6.5.6 *Mounting*—Both microphone diaphragms must be flush with the interior surface of the tube using port openings through the side of the tube. If the microphones are switched, care must be taken when the microphones are removed from their port so that 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. 4. The lip should be kept small and identical for both microphone ports.

TABLE 1 Recommended Frequency Limits for Microphones

Nominal Diameter (in.)	Diaphragm Diameter (mm)	Maximum Frequency (Hz)
1	22.70	3000
½	12.20	5600
¼	5.95	11500

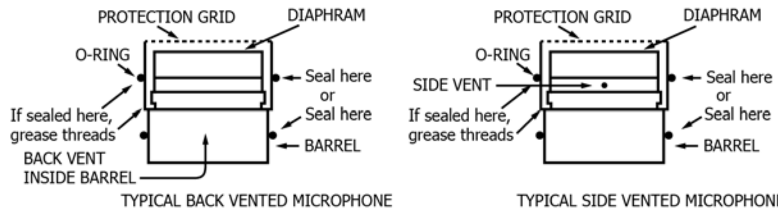


FIG. 3 Microphone Venting and Sealing

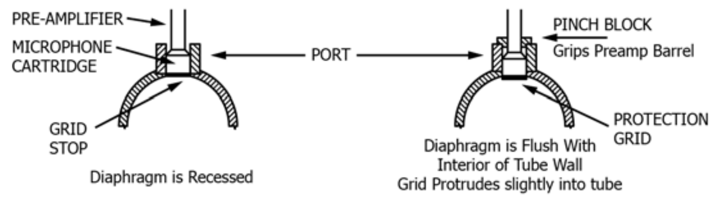


FIG. 4 Typical Microphone Mounting and Alternate Mounting

6.5.6.1 *Alternate Mounting*—In order to avoid the small recess caused by mounting the microphones according to 6.5.6, an alternative pinch block mounting technique may used as shown in Fig. 4. This method has the advantage of positioning the microphone diaphragm flush with the inside of the normal sound transmission 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.

6.5.6.2 *Microphone Acoustic Center*—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 slightly different from 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 8.4.5, 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.

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 may also be used if they 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 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 all microphone locations so 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 one, two, or preferably four similar microphones, a similar number of analog signal conditioners (optional) and a multi-channel Fast Fourier Transform (FFT) analyzer, or equivalent. The signal from each microphone system is connected to an individual channel of the analyzer. See Figs. 5 and 6.

6.7.2 *Computing Device*—A computer or other digital processsor, either separate from or part of the digital frequency analysis system, is necessary in order to calculate TL_n and other material properties. A complete set of mathematical expressions is given in 8.5.4.

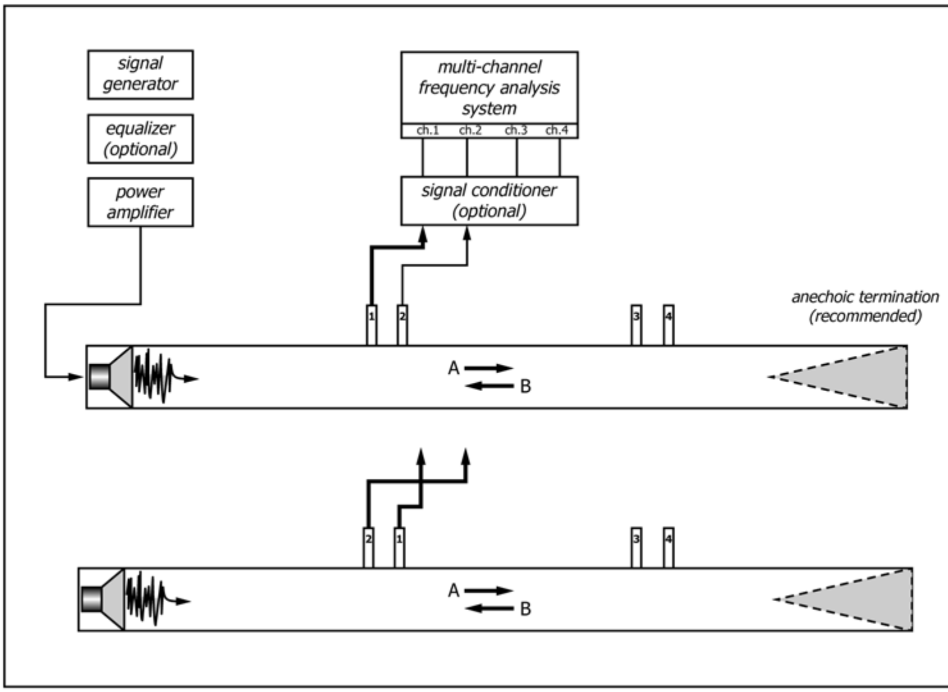
6.7.3 *Temperature Sensor*—A thermometer or other ambient temperature sensing device shall be installed so the air temperature is known to within ±1°C.

6.7.4 *Barometric Pressure Indicator*—A barometer or other equivalent indicating device shall be located in the vicinity of the sound transmission 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 sound transmission tube. The relative humidity shall be measured to within a tolerance of 5 %.

7. Test Specimen

7.1 *Mounting*—Each specimen must have the same shape and area as the tube cross-section The mounting conditions will strongly affect the measured transmission loss. The specimen may be rigidly mounted or clamped to the wall of the tube, freely suspended with a dense flexible seal, or some other method of mounting. Care must be taken to mount multiple samples in a consistant manner, and to report details of the mounting method sufficient to reproduce the mount. Any



NOTE 1—I: direct transfer function; II: switched transfer function.

FIG. 5 Apparatus and Instrumentation for Microphone Calibration in the Longitudinal Direction

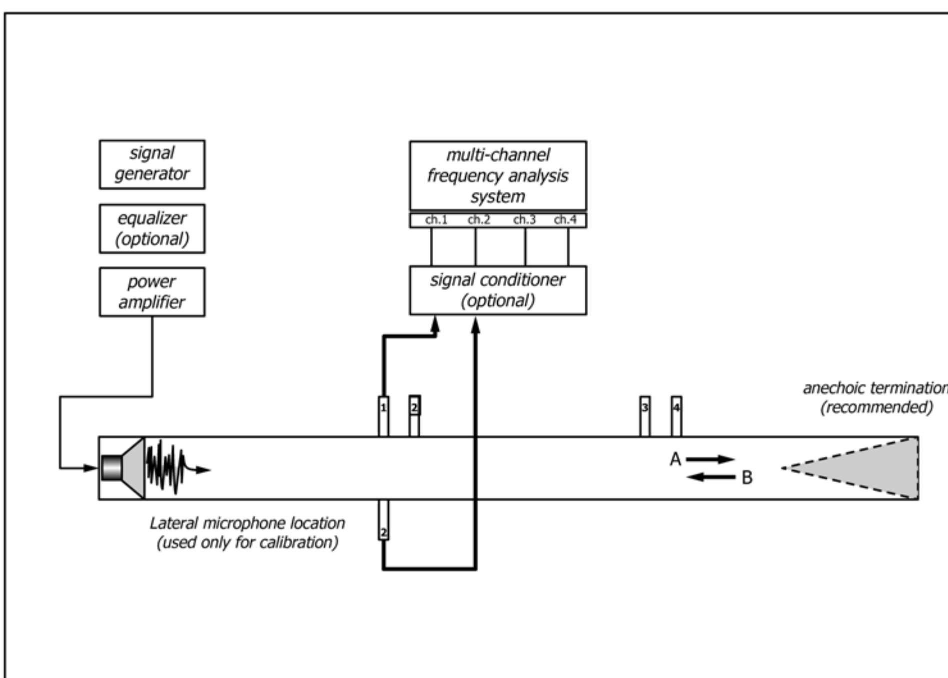


FIG. 6 Apparatus and Instrumentation for Microphone Calibration in the Same Transverse Plane

flexible mounting material must be previously shown to have a *TL* greater than the specimen material. A small opening around the edge will have a dramatic impact on the transmission loss calculations. Any peripheral cracks or gaps must be sealed with petroleum jelly, modeling clay, or putty. It is desirable to have the specimen possess a relatively flat surface for reasons stated in 6.5.5.

7.2 *Alignment*—The front surface of test specimens shall be mounted normal to the tube axis unless the surface specifically is designed otherwise.

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

7.4 *Number*—A minimum of three specimens should be cut from the sample and tested with the same mounting conditions. When the sample has a cross-section 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 *Procedure*—The determination of the transfer matrix requires a measurement of the complex sound pressure (amplitude and relative phase) at four locations, two on either side of the specimen. This is accomplished in practice by measuring the transfer function H between a reference and the four locations. There are a variety of methods for acquiring the four transfer functions, summarized in Table 2. The procedure requires as a minimum a two-channel analyzer and a single microphone. More channels and more microphones will speed the procedure but will generally require correction for amplitude and phase mis-match between the microphones, as described in 8.4.5. If there are fewer than 4 signal channels employed, the transfer function measurements are made sequentially as one or more microphones are moved from location to location. If microphones are moved, the unused locations must have their tube penetrations blocked from communication with the outside air. Four or five channels allow the transfer function measurements to be made simultaneously.

8.1.1 *Reference*—The reference for the transfer functions may be one of the four measurement locations, a separate measurement location (location 0) removed from the original four, or the electrical signal provided to the source microphone.

8.1.2 *Other Parameters*—In addition to the transfer functions, other parameters required for calculation of the transfer matrix are the ambient speed of sound and air density, the separation distance between the microphones, the location of the microphone locations relative to the reference plane, and the thickness of the specimen. These parameters are illustrated in Fig. 1.

8.2 *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:

$$c = 20.047\sqrt{273.15+T} \tag{4}$$

where:

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

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

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

where:

ρ = air density, kg/m^3,
P = atmospheric pressure, kPa, and
T = room temperature, ° C.

8.4 *Calibration:*

8.4.1 The procedure described here provides a means of correcting the measured transfer function data for mismatch in both the amplitude and phase responses of the two sets of measurement channels. Two approaches are suggested, one using the two microphone penetrations used for the measurement, and one using two penetrations located on the same plane. See Figs. 5 and 6.

8.4.2 *Signal-to-Noise Ratio*—With the representative test specimen in place, measure the sound pressure level spectrum at each microphone with the sound source on and off to ensure 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).

TABLE 2 Measurement Configurations and Procedures

Number of Channels	Number of Microphones	Transfer Function Reference	Transfer Functions Measured	Correction	Procedure
2	1	source signal	H_{1s}, H_{2s}, H_{3s} , H_{4s}	none	single microphone moves to locations 1–4
2	2	microphone1 at location 0	H_{10}, H_{20}, H_{30} , H_{40}	none	microphone 2 moves to locations 1–4
2	2	microphone 1 at location 1	H_{11}=1, H_{21}, H_{31}, H_{41}	none	microphone 2 moves to locations 2–4
4	4	microphone 1 at location 1	H_{11}=1, H_{21}, H_{31}, H_{41}	H_{21}^c, H_{31}^c, H_{41}^c	microphones 1–4 fixed in locations 1–4
5	4	source signal	H_{1s}, H_{2s}, H_{3s} , H_{4s}	H_{1s}^c, H_{2s}^c, H_{3s}^c, H_{4s}^c	microphones 1–4 fixed in locations 1–4
5	5	microphone 5 at location 0	H_{10}, H_{20}, H_{30} , H_{40}	H_{10}^c, H_{20}^c, H_{30}^c, H_{40}^c	microphones 1–4 fixed in locations 1–4

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 this test method, the Hanning window is recommended for measurement of transfer function (**6**). 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 (**7**).

8.4.5 *Amplitude and Phase Corrections*—Since the transfer function is a complex ratio of the acoustic pressure responses, any variation in the amplitude or phase responses of the transfer function pairs will affect the accuracy of the transfer function measurement. The following sequence of measurements and computations provides a means for correcting the measured transfer function between pairs of measurement channels. The general procedure is to measure the transfer function in the normal manner, and then physically switch the location of the microphones and measure again. This allows the calculation of a correction transfer function which accounts for the variation between microphones without including the phase difference due to propagation delay between them.

8.4.5.1 The termination of the tube should be at least weakly anechoic for maximum accuracy of this calibration procedure. See the termination recommendations of 6.2.8. Amplitude and phase corrections are determined using any pair of microphone locations. During the calibration procedure the unused locations must have their tube penetrations blocked from communication with the outside air, either by leaving the unused microphones in place or by some other means.

8.4.5.2 *For Microphone Correction Using Penetrations Along the Tube Axis:*

Step 1—Measure the following two transfer functions using the same computational algorithms for each microphone pair. Place the microphones in the direct configuration of Fig. 5 and measure.

$$H^I_{n,ref} = \left|H^I\right| e^{j\varphi I} = H^I_r + jH^I_i \tag{6}$$

Interchange the reference and measurement microphones to assume the switched configuration as shown in Fig. 5 and measure.

$$H^{II}_{n,ref} = \left|H^{II}\right| e^{j\varphi II} = H^{II}_r + jH^{II}_i \tag{7}$$

Care should be taken when interchanging the microphones to ensure that microphone 1 in the switched configuration occupies the precise location that microphone 2 occupied in the standard configuration, and vice versa (see 6.5.6).

Step 2—The following equations are valid for the case where the digital frequency analysis system always uses channel 1 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 correction transfer function H^c representing the amplitude and phase mismatches $|H^c|$ and φ^c for each transfer pair using the following equation:

$$H^c_{n,ref} = \left(H^I \cdot H^{II}\right)^{1/2} = \left|H^c\right| e^{j\varphi^c} \tag{8}$$

8.4.5.3 *For Microphone Correction Using Penetrations in the Same Transverse Plane*—If provision is made to place a microphone pair at the same axial location in the tube, propagation delay is zero and a single measurement of the transfer function expresses the amplitude and phase difference.

Step 1—Measure the following transfer function using the same computational algorithms for each microphone pair. Place the microphones in the configuration shown in Fig. 6 and measure.

$$H_{n,ref} = \left|H_{n,ref}\right| e^{j\varphi_{n,ref}} = H^r_{n,ref} + jH^i_{n,ref} \tag{9}$$

Step 2—The correction transfer function H^c for each transfer function pair, representing the amplitude and phase mismatches $|H^c|$ and φ^c, equals the transverse transfer function:

$$H^c_{n,ref} = \left|H_c\right| e^{j\varphi_c} = H_r + jH_i = H_{n,ref} \tag{10}$$

8.4.5.4 *For Microphone Corrections Using a Separate Signal Reference*—If the electrical signal to the loudspeaker is used as a reference, the channel pairs cannot be switched. In this case, the microphone amplitude and phase differences are normalized to one of the microphones so that the differences are the same for all transfer function pairs.

Step 1—Measure the following two transfer functions using the same computational algorithms for each microphone pair. Place the microphones in the direct configuration of Fig. 7 and measure.

$$H^I_{l,sig} = \left|H^I\right| e^{j\varphi I} = H^I_r + jH^I_i \tag{11}$$

Place the second microphone in the location used by microphone 1 to assume the switched configuration as shown in Fig. 7 and measure.

$$H^{II}_{n,sig} = \left|H^{II}\right| e^{j\varphi II} = H^{II}_r + jH^{II}_i \tag{12}$$

Care should be taken when interchanging the microphones to ensure that the second microphone in the switched configuration occupies the precise location that microphone 1 occupied in the standard configuration (see 6.5.6).

Step 2—Compute the correction transfer function H^c representing the amplitude and phase mismatches $|H^c|$ and φ^c for each transfer pair using the following equation:

$$H^c_{n,s} = \left(\frac{H^{II}}{H^I}\right) = \left|H^c\right| e^{j\varphi^c} \tag{13}$$

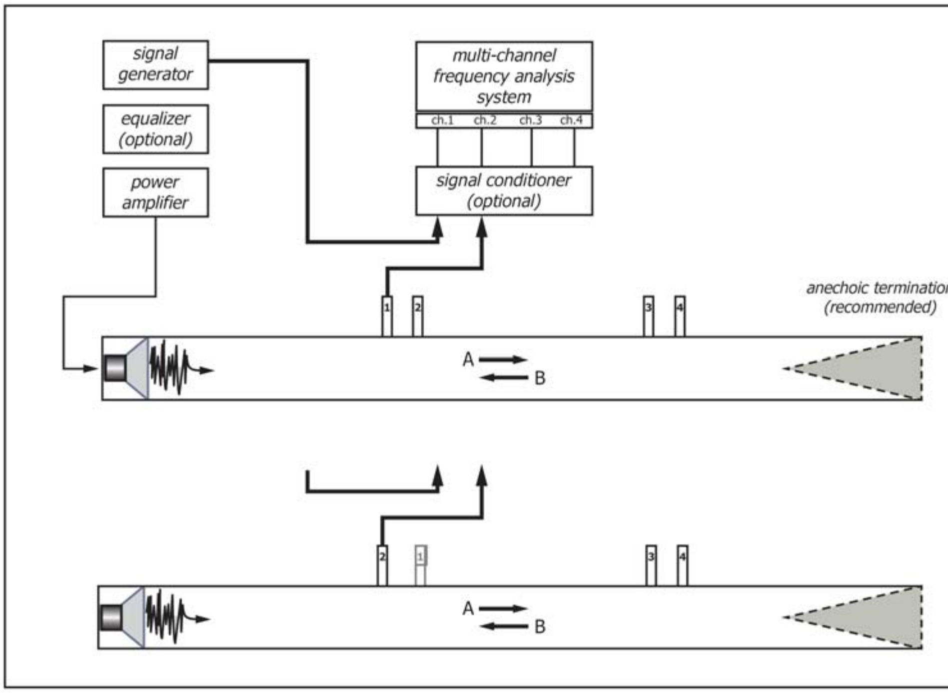
8.4.6 During the course of measurement, correct each measured transfer function by dividing it by the appropriate correction transfer function.

$$H = \frac{\bar{H}}{H^c} \tag{14}$$

8.5 *Measurements:*

8.5.1 *Transfer Function*—Insert the test specimen and measure the complex acoustic transfer functions between the reference and the remaining microphones:

$$H_{n,ref} = \frac{G_{n,ref}}{G_{ref,ref}} = \left|H\right| e^{j\varphi} = H_r + H_i \tag{15}$$



NOTE 1—I: direct transfer function; II: switched transfer function.

FIG. 7 Apparatus and Instrumentation for Microphone Calibration Using a Loudspeaker Signal as Reference

8.5.2 Eq 15 determines the transfer function by taking the ratio of the cross power spectrum to the reference autospectrum. 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 15 reduces the effects of noise. Since all FFT analysis systems do not define the cross spectrum consistently, adherence to the definitions implied in Eq 15 must be strictly observed.

8.5.3 *Mismatch Correction*—Using the method described in 8.4, correct each $\bar{H}$ for mismatch arising from the microphone amplitude and phase responses, by dividing it by its corresponding correction transfer function.

8.5.4 *Transfer Matrix*—Calculate the Normal Incidence Transfer Matrix as follows:

8.5.4.1 *Two-Load Method:*

(1) A single transfer matrix measurement involves two basic measurements with the two different terminations. The transfer matrix relates the acoustic pressure and particle velocity on front and back surface of the specimen. The tube must be configured with two different terminations (see 6.2.8), represented by indices *a* and *b*, in order to obtain four linear equations that can be used to solve for the four unknown matrix elements.

transfer matrix, termination "a" (16)

$$\begin{bmatrix} p_a \\ u_a \end{bmatrix}_{x=0} = \begin{bmatrix} T_{11} & T_{12} \\ T_{21} & T_{22} \end{bmatrix} \begin{bmatrix} p_a \\ u_a \end{bmatrix}_{x=d}$$

transfer matrix, termination "b"

$$\begin{bmatrix} p_b \\ u_b \end{bmatrix}_{x=0} = \begin{bmatrix} T_{11} & T_{12} \\ T_{21} & T_{22} \end{bmatrix} \begin{bmatrix} p_b \\ u_b \end{bmatrix}_{x=d}$$

(2) Here, Eq 16 with index "a" could represent an "anechoic" or otherwise minimally reflecting termination, and Eq 16 with index "b" could represent a blocked or open termination, reflecting a portion of incident wave.

(3) For each load case, decompose the acoustic wave field inside the tube into forward and backwards traveling waves on either side of the specimen, indicated in Fig. 1 using Eq 17-20.

$$A = j\frac{H_{1,ref}e^{-jkl_1} - H_{2,ref}e^{-jk(l_1+s_1)}}{2\sin ks_1} \tag{17}$$

$$B = j\frac{H_{2,ref}e^{+jk(l_1+s_1)} - H_{1,ref}e^{+jkl_1}}{2\sin ks_1} \tag{18}$$

$$C = j\frac{H_{3,ref}e^{+jk(l_2+s_2)} - H_{4,ref}e^{+jkl_2}}{2\sin ks_2} \tag{19}$$

$$D = j\frac{H_{4,ref}e^{-jkl_2} - H_{3,ref}e^{-jk(l_2+s_2)}}{2\sin ks_2} \tag{20}$$

(4) For each load case, determine the acoustic pressure and particle velocity on both faces of the specimen (at x = 0 and at x = d):

$$p_0 = A+B \qquad p_d = Ce^{-jkd}+De^{+jkd} \tag{21}$$

$$u_0 = (A-B)/\rho c \qquad u_d = (Ce^{-jkd}-De^{+jkd})/\rho c$$

(5) From the pressure and particle velocity values in each load case, calculate the transfer matrix for the specimen:

$$T = \begin{bmatrix} \frac{p_{0a}u_{db}-p_{0b}u_{da}}{p_{da}u_{db}-p_{db}u_{da}} & \frac{p_{0b}p_{da}-p_{0a}p_{db}}{p_{da}u_{db}-p_{db}u_{da}} \\ \frac{u_{0a}u_{db}-u_{0b}u_{da}}{p_{da}u_{db}-p_{db}u_{da}} & \frac{p_{da}u_{0b}-p_{db}u_{0a}}{p_{da}u_{db}-p_{db}u_{da}} \end{bmatrix} \tag{22}$$

8.5.4.2 *One-Load Method:*

(1) For specimens which are geometrically symmetric (presenting the same physical properties to the sound field on either side), the procedure of 8.5.4.1 may be simplified by recognizing that reciprocity places two constraints on the transfer matrix.

$$T_{11} = T_{22} \quad \text{and} \quad T_{11}T_{22} - T_{12}T_{21} = 1 \tag{23}$$

(2) This allows the elements of the matrix to be determined by a measurement of the microphone transfer function with a single termination case, preferably the anechoic case.

$$T = \begin{bmatrix} \frac{p_d u_d + p_0 u_0}{p_0 u_d + p_d u_0} & \frac{p_0{}^2 - p_d{}^2}{p_0 u_d + p_d u_0} \\ \frac{u_0{}^2 - u_d{}^2}{p_0 u_d + p_d u_0} & \frac{p_d u_d + p_0 u_0}{p_0 u_d + p_d u_0} \end{bmatrix} \tag{24}$$

8.5.5 *Calculate Material Properties:*

8.5.5.1 *Transmission Coefficient (anechoic-backed):*

$$t = \frac{2e^{jkd}}{T_{11}+(T_{12}/\rho c)+\rho c T_{21}+T_{22}} \tag{25}$$

8.5.5.2 *Normal Incidence Transmission Loss:*

$$TL_n = 20\log_{10}\left|\frac{1}{t}\right| \tag{26}$$

8.5.5.3 *Reflection Coefficient (hard-backed):*

$$R = \frac{T_{11} - \rho\, cT_{21}}{T_{11}+\rho c T_{21}} \tag{27}$$

8.5.5.4 *Absorption Coefficient (hard-backed):*

$$\alpha = 1 - |R|^2 \tag{28}$$

8.5.5.5 *Propagation Wavenumber in Material:*

$$k' = \frac{1}{d}\cos^{-1}T_{11} \tag{29}$$

8.5.5.6 *Characteristic Impedance in Material:*

$$z = \sqrt{T_{12}/T_{21}} \tag{30}$$

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 in the frequency domain, 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. Alternatively, the averaging time required to achieve a desired error level using a linear detector is given by Ref (**8**):

$$T \approx \frac{G^2}{2Be^2} \tag{31}$$

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 normal sound transmission tube system, that is:

$$t >> \frac{2(l+s)}{c} \tag{32}$$

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.

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.

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

10.1.6 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.

10.1.7 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.

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

10.2.1 The atmospheric pressure at the time of test.

10.2.2 The relative humidity at the time of test.

10.2.3 A tabular listing of the Transmission Coefficient in Material as defined in 8.5.5.1 as a function of frequency (to two significant figures). The designated reference plane must be identified clearly.

10.2.4 A tabular listing of Reflection Coefficient (hard-backed) as defined in 8.5.5.3 as a function of frequency (to two significant figures). The designated reference plane must be identified clearly.

10.2.5 A tabular listing of Propagation Wavenumber in Material as defined in 8.5.5.5 as a function of frequency (to two significant figures). The designated reference plane must be identified clearly.

10.2.6 A tabular listing of Characteristic Impedance in Material as defined in 8.5.5.6 as a function of frequency (to two significant figures). The designated reference plane must be identified clearly.

11. Precision and Bias for Transmission Loss Measurements

11.1 Imprecision in this test method arises from sources other than the measurement procedure. Some materials are not 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, and the mounting condition reproduced as closely as possible between tests.

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 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 There is no true value for this measure of transmission loss, however the results of a round robin, to be conducted, will compare the results to traditional methods. See Test Method E90 and ISO 140-3 to determine a measure of bias.

11.5 The precision will be established by the results of a round robin program, which has not yet been conducted.

11.5.1 The intra- and inter-laboratory precision of this test method, expressed in terms of the intra-laboratory 95 % Repeatability Interval I(r) and the inter-laboratory 95 % Reproducibility Interval I(R), is not yet available. These statistics will be based on the results of a round-robin test program.

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

11.5.2.1 *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.

11.5.2.2 *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 4-microphone; impedance tube; normal incidence; transfer function; transfer matrix; transmission loss

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 it is in compliance with the mandatory parts of this test method in those parts that contain the words shall or must.

A1.2.3 The laboratory shall show the construction and geometry of the tube and specimen holder as described in 6.2.

A1.2.4 The laboratory shall show calculations verifying the tube diameter in accordance with 6.2.4 and 6.2.5.

A1.2.5 The laboratory shall show the sound source and that its frequency response is in according with 6.4.

A1.2.6 The laboratory shall report the type of test signal used (see 6.6.1).

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 amplitude and phase response correction procedure used (see 8.4.5).

A1.2.9 The laboratory shall show sample calculations or the 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 periodic tests for quality assurance. It shall be so constructed or formed that it will not deteriorate quickly with use. Its acoustic properties should remain stable during at least ten years of use. As measured by this test method, the sound transmission loss of the reference specimen shall be at least 5 dB for frequencies greater than 250 Hz. The measured acoustic properties and their standard deviations shall be analyzed by the control chart method described in MNL 7 (**9**). 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 is 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 8.4.5.2 (Step 2) which describes a procedure to compute the correction transfer function H^c. All other considerations and computations remain the same as those described in the main body of this test method.

X1.2.1.1 *Step 1*—Use the procedure of 8.4.5.2 (Step 1) to determine the direct and switched transfer functions H^I and H^{II}.

X1.2.1.2 *Step 2*—Compute the calibration factor H^c representing the amplitude and phase mismatches $|H^c|$ and φ^c, using the following equation:

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

where:

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

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

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

X1.2.1.3 *Step 3*—Continue this procedure at 8.4.6.

REFERENCES

(**1**) Harris, C. M., editor, *Shock and Vibration Handbook*, McGraw-Hill, Third Edition, 1988, Chapter 21, pp. 4–8.

(**2**) Song, B. H., and Bolton, J. S., "A Transfer-Matrix Approach for Estimating the Characteristic Impedance and Wave Numbers of Limp and Rigid Porous Materials," *J. Acoust. Soc. Am.*, 107, 2000, pp. 1131–1152.

(**3**) Rayleigh, J. W. S., *The Theory of Sound*, Dover Publications, Inc., New York, NY, Vol 2, 1896, p. 161.

(**4**) *Condenser Microphones Data Handbook*, Bruel and Kjaer, Revision September 1982, p. 56.

(**5**) Chung, J. Y., and Blaser, D. A., "Transfer Function Method of Measuring In-Duct Acoustic Properties I. Theory and II. Experiment," *Journal of the Acoustical Society of America*, 68(3), 1980, pp. 907–921.

(**6**) Gade, S., and Herlufsen, H., "Use of Weighting Functions in DFT/FFT Analysis (Part 1)," *Bruel and Kjaer Technical Review*, No. 3, 1987.

(**7**) Application Note 245-1, "Signal Averaging ...," Hewlett Packard, P.O. Box 58004, Santa Clara, CA, 95052-8004.

(**8**) Beranek, L. L., *Noise and Vibration Control*, McGraw-Hill, New York, NY, 1971, pp. 116–117.

(**9**) ASTM Manual Series: *MNL 7, Manual on Presentation of Data and Control Chart Analysis*, Sixth Edition, p. 54.

BIBILOGRAPHY

(1) C384 Test Method for impedance and Absorption of Acoustical Materials by Impedance Tube Method[2]

(2) C423 Test Method for Sound Absorption and Sound Absorption Coefficients by the Reverberation Room Method[2]

(3) ISO 10534-1 Acoustics—Determination of Sound Absorption Coefficient and Impedance or Admittance—Part 1: Impedance Tube Method[3]

(4) ISO 10534–2 Acoustics—Determination of Sound Absorption Coefficient and Impedance in Impedance Tubes—Part 2: Transfer-Function Method[3]

(5) London, A., "The Determination of Reverberant Sound Absorption Coefficients from Acoustic Impedance Measurements," *Journal of the Acoustical Society of America*, Vol 22 (2), March 1950, pp. 263–269.

(6) Mechel, F. P., "Design Charts for Sound Absorber Layers," *Journal of the Acoustical Society of America*, Vol 83(3), March 1988, pp. 1002–1013.

(7) Højbjerg, K., "A New Two-Microphone Impedance Tube with Improved Microphone Design for Materials Testing," SAE 911091, Proceedings of the 1991 Noise and Vibration Conference, pp. 457–459.

(8) Beranek, L. L., *Acoustical Measurements*, Published for the Acoustical Society of America by the American Institute of Physics, Woodbury, NY, 1988 Revised Edition, pp. 72–73.

(9) Pierce, A. D., *Acoustics—An Introduction to Its Physical Principles and Applications*, published for the Acoustical Society of America by the American Institute of Physics, Woodbury, NY, 1989, p. 351.

(10) Munjal, M. L., and Doige, A. G., "Theory of a Two Source-Location Method for Direct Experimental Evaluation of the Four-Pole Parameters of an Aeroacoustic Element," *J. Sound Vib*., 141, 1990, pp. 323–333.

(11) Yoo, T., Bolton, J. S., and Alexander, J. H., "Prediction of Random Incidence Transmission Loss Based on Normal Incidence Four-Microphone Measurements," *Proc. InterNoise*, 2005.

(12) Song, B. H., and Bolton, J. S., and Kang, Y. J., "Effect of Circumferential Edge Constraint on the Acoustical Properties of Glass Fiber Materials," *J. Acoust. Soc. Am*., 110, 2001, pp. 2902–2916.

(13) Song, B. H., and Bolton, J. S., "Investigation of the Vibrational Modes of Edge-Constrained Fibrous Smaples Placed in a Standing Wave Tube," *J. Acoust. Soc. Am*., 113, 2003, pp. 1833–1849.

(14) Song, B. H., and Bolton, J. S., "Enhancement of the Barrier Performance of Porous Linings by Using Internal Constraints," *Noise Control Engineering J*., 51, 2003, pp. 16–35.