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Acoustics — Determination of acoustic properties in impedance tubes —

Part 2: Two-microphone technique for normal sound absorption coefficient and normal surface impedance

Acoustique — Détermination des propriétés acoustiques aux tubes d'impédance —

Partie 2: Méthode à deux microphones pour le coefficient d'absorption sonore normal et l'impédance de surface normale

Reference number
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Foreword

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This document was prepared by Technical Committee ISO/TC 43 *Acoustics*, Subcommittee SC 2, *Building acoustics,* in collaboration with the European Committee for Standardization (CEN) Technical Committee CEN/TC 126, *Acoustics properties of building products and of buildings,* in accordance with the Agreement on technical cooperation between ISO and CEN (Vienna Agreement).

This second edition cancels and replaces the first edition (ISO 10534-2:1998), which has been technically revised.

The main changes are as follows:

— the introduction of the measurement procedure to estimate the characteristic properties of porous materials (characteristic impedance, wavenumber, dynamic mass density, dynamic bulk modulus) in an informative annex. The signal processing techniques have been updated since the first version of this document.

Any feedback or questions on this document should be directed to the user’s national standards body. A complete listing of these bodies can be found at www.iso.org/members.html.

Acoustics — Determination of acoustic properties in impedance tubes —

Part 2:
Two-microphone technique for normal sound absorption coefficient and normal surface impedance

1 Scope

This test method covers the use of an impedance tube, two microphone locations and a frequency analysis system for the determination of the sound absorption coefficient of sound absorbing materials for normal incidence sound incidence. It can also be applied for the determination of the acoustical surface impedance or surface admittance of sound absorbing materials. As an extension, it can also be used to assess intrinsic properties of homogeneous acoustical materials such as their characteristic impedance, characteristic wavenumber, dynamic mass density and dynamic bulk modulus.

The test method is similar to the test method specified in ISO 10534-1[1] in that it uses an impedance tube with a sound source connected to one end and the test sample mounted in the tube at the other end. However, the measurement technique is different. In this test method, plane waves are generated in a tube by a sound source, and the decomposition of the interference field is achieved by the measurement of acoustic pressures at two fixed locations using wall-mounted microphones or an in-tube traversing microphone, and subsequent calculation of the complex acoustic transfer function and quantities reported in the previous paragraph. The test method is intended to provide an alternative, and generally much faster, measurement technique than that of ISO 10534-1[1].

Normal incidence absorption coefficients coming from impedance tube measurements are not comparable with random incidence absorption coefficients measured in reverberation rooms according to ISO 354[2]. The reverberation room method will (under ideal conditions) determine the sound absorption coefficient for diffuse sound incidence. However, the reverberation room method requires test specimens which are rather large. The impedance tube method is limited to studies at normal and plane incidence and requires samples of the test object which are of the same size as the cross-section of the impedance tube. For materials that are locally reacting only, diffuse incidence sound absorption coefficients can be estimated from measurement results obtained by the impedance tube method (see Annex E).

Through the whole document, a $e^{+j\omega t}$ time convention is used.

2 Normative references

There are no normative references in this document.

3 Terms, definitions and symbols

For the purposes of this document, the following terms and definitions apply.

ISO and IEC maintain terminology databases for use in standardization at the following addresses:

— ISO Online browsing platform: available at https://www.iso.org/obp

— IEC Electropedia: available at https://www.electropedia.org/

3.1
sound absorption coefficient at normal incidence
α_n
ratio of the sound power dissipated inside the test object to the incident sound power for a plane wave at normal incidence

Note 1 to entry: “Plane wave” here describes a wave whose value, at any moment, is constant over any plane perpendicular to its direction of propagation. “Normal incidence” describes the direction of the longest axis of the impedance tube.

3.2
sound pressure reflection coefficient at normal incidence
r
complex ratio of the reflected wave sound pressure amplitude to that of the incident wave in the reference plane for a plane wave at normal incidence

3.3
reference plane
cross-section of the impedance tube for which the reflection factor r or the impedance Z or the admittance G are determined and which is usually the surface of the test object, if flat

Note 1 to entry: The reference plane is assumed to be at $x = 0$.

3.4
normal-incidence surface impedance
Z
ratio of the complex sound pressure $p(x = 0)$ to the normal component of the complex sound particle velocity $v(x = 0)$ at an individual frequency in the reference plane defined as $x = 0$

Note 1 to entry: The particle velocity vector has a positive direction pointing towards the interior of the tested object.

Note 2 to entry: Z is expressed in newton second per cubic meter (Ns/m^3)

3.5
normal-incidence surface admittance
G
inverse of the normal-incidence surface impedance Z

Note 1 to entry: G is expressed in cubic meter per newton per second ($m^3/N/s$)

3.6
wave number in air
k_0
variable, expressed in radian per metre, defined by

$$k_0 = \omega / c_0 = 2\pi f / c_0 = 2\pi / \lambda_0$$

where

- ω is the angular frequency,
- f is the frequency,
- c_0 is the speed of sound in the air,
- λ_0 is the wavelength in air.

Note 1 to entry: In general, the wave number is complex, so that $k_0 = k_0' - jk_0''$ where k_0' is the real component and k_0'' is the imaginary component (which is the attenuation constant).

Note 2 to entry: k_0' is expression in radians per metre.

3.7
material characteristic wave number
k_c
variable, expressed in radian per meter, defined by

$$k_c = \omega / c = 2\pi f / c = \omega\sqrt{\rho_{eq} / K_{eq}}$$

where

c	is the speed of sound inside the material;
ρ_{eq}	is the material dynamic mass density (defined in 3.9);
K_{eq}	is the material bulk modulus (defined in 3.10)

3.8
material characteristic impedance
Z_c
variable, expressed in Newton second per cubic metre, defined by

$$Z_c = \sqrt{\rho_{eq} K_{eq}}$$

3.9
material dynamic mass density
ρ_{eq}
variable describing the visco-inertial dissipation inside the tested material.

Note 1 to entry: The dynamic mass density can differ from the static (volume-averaged) value.

Note 2 to entry: It is expressed in kg/m^3.

3.10
material dynamic bulk modulus
K_{eq}
variable describing the thermal dissipation inside the tested material.

Note 1 to entry: The dynamic bulk modulus can differ from the static (volume-averaged) value.

Note 2 to entry: It is expressed in N/m^2 (or equivalently in pascal).

3.11
complex sound pressure
p
frequency-domain spectrum of the sound pressure time signal

3.12
cross spectrum
S_{12}
product $p_2\, p_1{}^*$, determined from the complex sound pressures p_1 and p_2 at two microphone positions

Note 1 to entry: * means the complex conjugate.

3.13
cross spectrum
S_{21}
product $p_1 \ p_2 {}^*$, determined from the complex sound pressures p_1 and p_2 at two microphone positions

Note 1 to entry: * means the complex conjugate.

3.14
auto spectrum
S_{11}
product $p_1 \ p_1 {}^*$, determined from the complex sound pressure p_1 at microphone position one

Note 1 to entry: * means the complex conjugate.

Note 2 to entry: S_{22} denotes the auto spectrum for pressure p_2 at microphone position two.

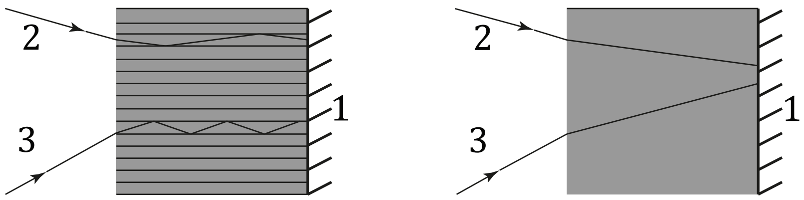
3.15
transfer function
H_{12}
transfer function from microphone position one to two, defined by the complex ratio $p_2 / p_1 = S_{12} / S_{11}$ or S_{22} / S_{21}, or $[(S_{12} / S_{11})(S_{22} / S_{21})]^{1/2}$

3.16
calibration factor
H_c
factor used to correct for amplitude and phase mismatches between the microphones

Note 1 to entry: See 8.5.3.

3.17
locally reacting material
material for which the pressure and velocity fields at a given point on the surface are independent on the behaviour at other points of the surface

Note 1 to entry: This local reaction behaviour infers specific properties for a material: its surface impedance is independent on the incidence angle of a plane wave impinging the material. Homogeneous honeycomb structures and perforated plates are examples of possible locally reacting materials (see Figure 1 a)). For a locally reacting material, its absorption coefficient depends on the angle of incidence as its reflection coefficient does as well



a) Locally reacting material sample **b) Non-locally reacting material sample**

Key

1 rigid and impervious backing
2 plane wave impinging the sample
3 plane wave impinging the sample with a different angle

Figure 1 — Propagation of plane waves inside a locally reacting material sample and comparison to a non-locally reacting material sample

3.18
bulk or extended reaction material
material for which the reaction does not occur only normal to the surface.

Note 1 to entry: The reaction in each point of the material is hence dependent on the reaction of the neighbouring points. Examples of materials experiencing bulk reactions are foams made of multiple pores and fibrous with fibres not parallel to each other's (see Figure 1 b)).

4 Principle

The test sample is mounted at one end of a straight, rigid, smooth and airtight impedance tube. Plane waves are generated in the tube by a sound source emitting a signal such as a random noise, pseudo-random sequence, or a deterministic signal such as a chirp signal, and the sound pressures are measured at two locations near to the sample. The complex acoustic transfer function of the two microphone signals is determined and used to compute the normal-incidence complex reflection coefficient (see Annex C), the normal-incidence absorption coefficient, and the normal incidence surface impedance of the test material. From two distinct measurements, the intrinsic properties of the material (characteristic wave number, characteristic impedance, dynamic mass density and dynamic bulk modulus) can be assessed assuming this material is homogeneous.

The quantities are determined as functions of the frequency (or frequency bands as detailed in ISO 266[3]) with a frequency resolution which is determined from the sampling frequency and the record length of the digital frequency analysis system used for the measurements. The usable frequency range depends on the lateral dimensions or diameter of the tube and the spacing between the microphone positions. An extended frequency range may be obtained from the combination of measurements with different lateral dimensions (or diameter) and spacings.

The measurements may be performed by employing one of two techniques:

a) two-microphone method (using two microphones in fixed locations);

b) one-microphone method (using one microphone successively in two locations).

Technique 1: requires a pre-test or in-test correction procedure to minimize the amplitude and phase difference characteristics between the microphones; however, it combines speed, high accuracy, and ease of implementation. Technique 1 is recommended for general test purposes.

Technique 2: has particular signal generation and processing requirements and may necessitate more time; however, it eliminates phase mismatch between microphones and allows the selection of optimal microphone locations for any frequency. Technique 2 is recommended for measurements with higher precision, and its requirements are described in more detail in Annex B.

5 Test equipment

5.1 Construction of the impedance tube

The apparatus is essentially a tube with a test sample holder at one end and a sound source at the other. Microphone ports are usually located at two or three locations along the wall of the tube (depending on the chosen microphone spacing).

The impedance tube shall be straight with a uniform cross-section (diameter or cross dimension within ±0,2 %) and with rigid, smooth, non-porous walls without holes or slits (except for the microphone positions) in the test section. The walls shall be heavy and thick enough so that they are not excited to vibrations by the sound signal and show no vibration resonances in the working frequency range of the tube. For metal walls, a thickness of about 5 % of the diameter is recommended for circular tubes. For rectangular tubes, the corners shall be made rigid enough to prevent distortion of the side wall plates.

It is recommended that the side wall thickness be about 10 % of the cross dimension of the tube. Tube walls made of concrete shall be sealed by a smooth adhesive finish to ensure air tightness. The same holds for tube walls made of wood; these should be reinforced and damped by an external coating of steel or lead sheets.

The shape of the cross-section of the tube is arbitrary, in principle. Circular or rectangular (if rectangular, then preferably square) cross-sections are recommended.

If rectangular tubes are composed of plates, care shall be taken that there are no air leaks (e.g. by sealing with adhesives or with a finish). Tubes should be sound and vibration isolated against external noise or vibration.

5.2 Working frequency range

The working frequency range is given by Formula (1):

$$f_l < f < f_u \tag{1}$$

where

f_l is the lower working frequency of the tube;

f is the operating frequency;

f_u is the upper working frequency of the tube.

f_l is limited by the uncertainty of the signal processing equipment and the spacing between the two microphone positions.

f_u is chosen to avoid the occurrence of non-plane wave mode propagation. The condition for f_u is given by Formula (2):

$$d < 0{,}58\ \lambda_u :\ f_u \cdot d < 0{,}58\ c_0 \tag{2}$$

for circular tubes with the inside diameter d in metres and f_u in Hertz. The same condition, given by Formula (3) is used:

$$d < 0{,}50\ \lambda_u :\ f_u \cdot d < 0{,}50\ c_0 \tag{3}$$

for rectangular tubes with the maximum side length d in metres; c_0 is the speed of sound in metres per second given by Formula (4).

The spacing s in metres between the microphones shall be chosen to avoid singularities when the distance of the two microphone positions is equal to a multiple of half the operating wavelength. The first singularity is avoided when ensuring that

$$f_u \cdot s < 0{,}45\ c_0 \tag{4}$$

The lower frequency limit is dependent on the spacing between the microphones and the uncertainty of the analysis system but, as a general guide, the microphone spacing should exceed 1,5 % of the wavelength corresponding to the lower frequency of interest, provided that the requirements of Formula (4) are satisfied. A larger spacing between the microphones enhances the accuracy of the measurements for these low frequencies but reduces the value of the upper working frequency.

Different microphone spacings can be used to cover a wider frequency range than the one allowed for a single spacing. In this case, the working frequency ranges shall overlap by about one octave (as described in ISO 266[3]). The averaging technique used to obtain the averaged and combined result should be at least mentioned.

Different impedance tubes can also be used to cover a wider frequency range than the one allowed for a single tube (see Clause 10 i).

5.3 Length of the impedance tube

The tube should be long enough to cause plane wave development between the source and the sample. Microphone measurement points shall be in the plane wave field.

The loudspeaker generally will produce non-plane waves besides the plane wave. They will die out within a distance of maximum three tube diameters or three times the lateral dimensions of rectangular tubes for frequencies below the lower cut-off frequency of the first higher mode. Thus, it is recommended that microphones be located no closer to the source than three tube diameters or three times the lateral dimensions.

Test samples will also cause proximity distortions to the acoustic field. It is recommended to have a minimum spacing between microphone and sample of ½ diameter or ½ maximum lateral dimension, but this spacing should be increased to 2 diameters or 2 times the maximum lateral dimension for non-planar materials or materials with a few small perforations (as perforated plates with a single millimetric perforation).

5.4 Microphones

Microphones of identical type shall be used in each location. When side-wall-mounted microphones are used, the diameter of the microphones shall be small compared to c_0 / f_u .

For side-wall mounting, it is recommended to use microphones of the pressure type. For in-tube microphones, it is recommended to use microphones of the free-field type.

5.5 Positions of the microphones

When side-wall-mounted microphones are used, each microphone shall be mounted with the diaphragm flush with the interior surface of the tube. A small recess is often necessary to prevent the microphone to be inserted inside the tube (see Figure 2); the recess should be kept small and be identical for both microphone mountings. The microphone grid shall be sealed tight to the microphone housing and there shall be a sealing between the microphone and the mounting hole.

a) Rectangular cross-section **b) Circular cross-section**

Key

1 microphone

2 sealing

Figure 2 — Examples of typical microphone mounting for a tube

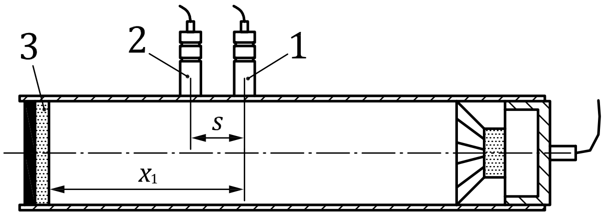
When using a single microphone in two successive wall positions, the microphone position not in use shall be sealed to avoid air leaks and to maintain a smooth surface inside the tube.

When using side-vented microphones, it is important that the pressure equalization vents are not blocked by the microphone mounting.

All fixed microphone locations shall be known to a tolerance of ±0,2 mm or better, and their spacing s (see Figure 3) shall be recorded.

Traversing microphone positions shall be known to a tolerance of ±0,5 mm or better.

Finally, it is recommended to set the microphone positions to a distance not larger than 250 mm $(x_1 < 250 \text{ mm})$ from the rigid backing of the impedance tube (i.e. the opposite end to the loudspeaker) to reduce the impact of the first acoustic resonances in the tube on the microphone measurements.



Key

1 microphone A

2 microphone B

3 test specimen

s spacing between the two microphones

x_1 distance between the surface of the test specimen and the microphone closest to the sound source

Figure 3 — Microphone positions and distances

5.6 Acoustic centre of the microphone

For the determination of the acoustic centre of a microphone, or minimizing errors associated with a difference between the acoustic and geometric centres of the microphones, see A.2.2.

5.7 Test sample holder

The test sample holder is either integrated into the impedance tube or is a separate unit, which is tightly fixed to one end of the tube during the measurement. The length of the sample holder shall be large enough to install test objects with air spaces behind them if required.

If the sample holder is a separate unit, it shall comply in its interior dimensions with the impedance tube to within ±0,2 %. The mounting of the tube shall be tight, without insertion of elastic gaskets (petroleum jelly or thread seal tape is recommended for sealing).

For rectangular tubes, it is recommended to integrate the sample holder into the impedance tube and to make the installation section of the tube accessible by a removable cover for mounting the test sample. The contact surfaces of this removable cover with the tube shall be carefully finished and the use of a sealant (like a petroleum jelly or a thread seal tape) is recommended in order to avoid small leaks.

For circular tubes, it is recommended to make the test object accessible from both the front and the back end of the sample holder. It is then possible to check the position and flatness of the front surface and the back position.

Generally, in connection with rectangular tubes, it is recommended to install the test object from the side into the tube (instead of pushing it axially into the tube). It is then possible to check the fitting and the position of the test object in the tube, to check the position and the flatness of the front surface, and to reposition the reference plane precisely in relation to the front surface. A sideways insertion also avoids compression of soft materials.

The back plate of the sample holder shall be rigid and shall be fixed tightly to the tube since it serves as a rigid termination in many measurements. A metal plate of thickness not less than 10 mm is recommended.

For some tests, a pressure-release termination of the test object by an air volume behind it is needed (this is described in Annex C).

5.8 Signal processing equipment

The signal processing system shall consist of an amplifier and an analysing system which is able to determine the transfer function H_{12} between the two microphone locations. The transfer function can be determined via a two-channel fast Fourier transform (FFT) analysing system, or via an impulse response measuring system and a subsequent Fourier transformation of the impulse responses. The impulse response measuring system can use two-channel FFT or cross-correlation. If the measurement signal is of the m-sequence type, a cross-correlation-based analysis system, which for instance uses the Fast Hadamard Transform, shall be used.

A generator capable of producing the required source signal (see 5.10) compatible with the analysing system is also required.

The dynamic range of the analyser should be greater than 65 dB. The errors in the estimated transfer function H_{12} due to non-linearities, resolution, instability and temperature sensitivity of the signal processing equipment shall be less than 0,2 dB.

Using the one-microphone technique, the analysing system shall be able to calculate the transfer function H_{12} from the generator signal and the two microphone signals measured consecutively.

5.9 Loudspeaker

A membrane loudspeaker (or a pressure chamber loudspeaker for high frequencies with a horn as a transmission element to the impedance tube) should be located at the opposite end of the tube from the test sample holder. The surface of the loudspeaker membrane shall cover at least two-thirds of the cross-sectional area of the impedance tube. The loudspeaker axis may be either coaxial with the tube, or inclined, or connected to the tube by an elbow.

The loudspeaker shall be contained in an insulating box in order to avoid airborne flanking transmission to the microphones. Elastic vibration insulation shall be applied between the impedance tube and the frame of the loudspeaker as well as to the loudspeaker box (preferably between the impedance tube and the transmission element also) in order to avoid structure-borne sound excitation of the impedance tube.

5.10 Signal generator

The signal generator shall be able to generate a stationary signal with a flat spectral density within the frequency range of interest. It may generate one or more of the following: random, pseudo-random, periodic pseudo-random, or chirp excitation, as required.

In the case of the one-microphone technique, a deterministic signal is recommended. A periodic pseudo-random sequence is also well suited for this method.

Discrete-frequency generation and display are necessary for tube calibration purposes (according to Annex A). Discrete- frequency generation and display shall have an uncertainty of less than ±2 %.

5.11 Thermometer, barometer and relative humidity

The temperature shall be measured and kept constant during a measurement with a tolerance of ±1 K. The temperature transducer shall be accurate to ±0,5 K or better.

The atmospheric pressure shall be measured with a tolerance of ±0,5 kPa.

If available, the information about the relative humidity shall be reported with a tolerance of ±2 %.

It is recommended to place the sensors for these measurements in the room where the tube lies rather than inside the tube unless the tube and the sensors are designed so that they can work properly without having a noticeable influence on the wave field inside the tube.

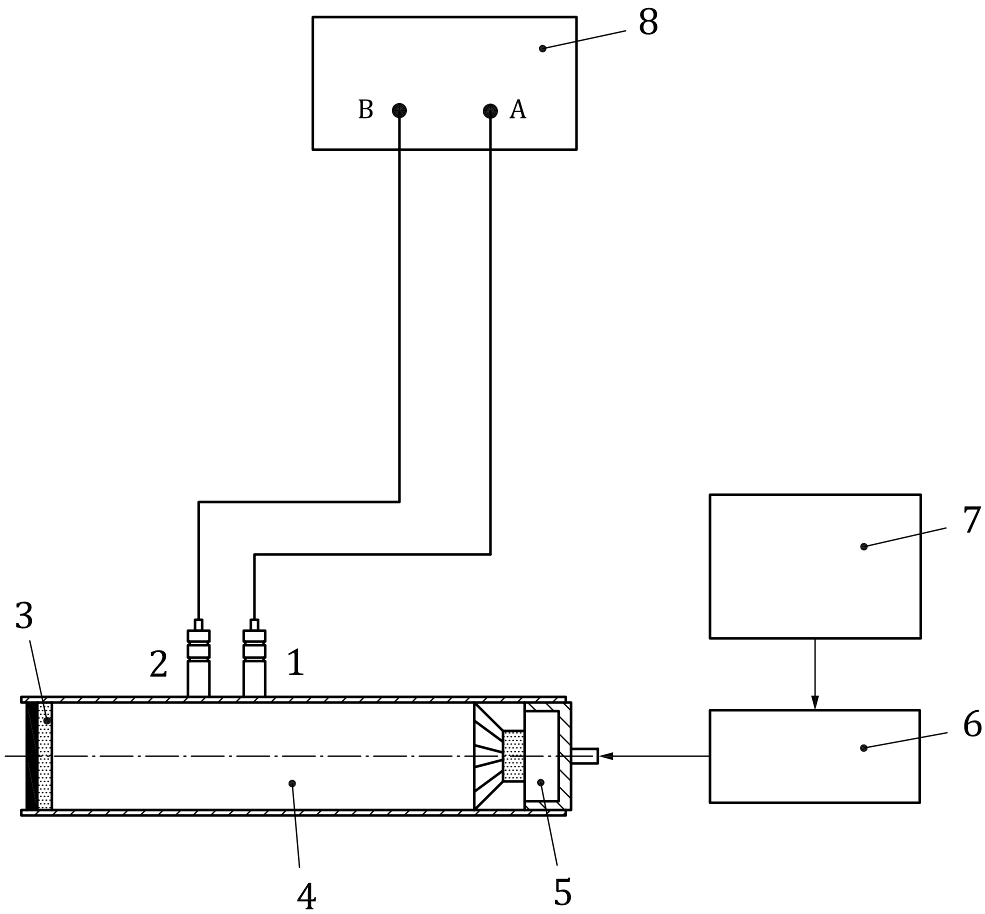
6 Preliminary test and measurements

The test equipment shall be assembled, typically as shown in Figure 4, and checked before use by a series of tests. These tests help to exclude error sources and secure the minimum requirements. The checks may be in two categories: prior to or following each test, and periodic calibration tests before each measurement session of half a day or a day maximum. In each case, the loudspeaker should be operated for at least 5 min prior to a measurement to allow the temperature to stabilize.

Checks prior to and following each test involve microphone response consistency, temperature measurement and a test of the signal-to-noise ratio.

Periodic calibrations are performed with a rigid termination of the empty impedance tube and an absorbent material (to reduce the impact of the air-column resonances inside the impedance tube). Their aim is the determination of the acoustic centre of a microphone, and/or the corrections for attenuation in the impedance tube.

These preliminary measurements are described in Annex A.



Key

1	microphone A	4	impedance tube	7	signal generator
2	microphone B	5	sound source	8	frequency analysis system
3	test specimen	6	amplifier		

Figure 4 — Example of layout for test equipment

7 Test specimen mounting

The test specimen shall fit snugly in the holder. However, it shall not be compressed unduly nor fitted so tightly that it bulges. Sealing of any crack about the edge of the sample with a petroleum jelly or a thread seal tape is recommended. Samples such as carpet material or low-density materials should be firmly attached to the back plate using a thin double-sided adhesive tape to prevent vibrational motion and unwanted air gaps.

The front surface of flat test samples shall be mounted normal to the tube axis. Their positions shall be specified with minimum tolerances: for objects with flat and smooth surfaces, to within ±0,5 mm. With porous materials of low bulk density, it may be helpful to fix and to define the surface by a thin, wire grid with wide mesh.

The vibrations of the solid phase of poroelastic material samples may significantly influence their sound absorption (usually in local frequency bands). Such a behaviour is related to the size of the sample and its boundary conditions inside the tube and is not an intrinsic characteristic of the material. Testing samples of different diameters leads to shifts in frequency of these phenomena with, consequently, difficulties in overlapping curves.

Testing high-thickness samples for a material can lead to difficulties in the installation of the samples, with compression (even not homogeneous) of the material samples. Consequently, the sample tested show absorption coefficients which are not consistent with those of the original material.

In the case of rigid and highly reflective samples (asphalts, plasters...), it is necessary to carefully seal the lateral perimeter of the samples along their entire thickness; cylindrical defects can cause lateral air cavities and consequently absorption peaks due to resonances in these cavities.

If the specimen has an uneven or irregular face, microphone locations shall be chosen to be sufficiently far away so that the measured transfer function is in the plane wave region. When the specimen has an uneven back which would introduce an unintended backing air space, a layer of putty-like material should be placed between it and the sound-reflective back plate to seal the back of the specimen and to add enough thickness to make the front surface parallel to the back plate.

A minimum of three specimens, more if the sample is not uniform, should be tested in repeated measurements using the same mounting conditions.

8 Test procedure

8.1 Specification of the reference plane

The first step in the measurement of the acoustic properties, after the mounting of the test specimen according to Clause 7, is the specification of the reference plane ($x = 0$). Typically, this coincides with the surface of the test specimen. If, however, the test specimen has a surface profile or a lateral structure, it shall be placed some distance in front of the test object.

The distance from the reference plane to the nearest microphone shall be in accordance with 5.3. The reference plane location in relation to microphone 1, depicted in Figure 3, shall be reported with a tolerance of ±0,5 mm or better.

NOTE The exact determination of the reference plane location is not required if only the absorption coefficient is measured.

If the test object exhibits a pattern which differs from the cross-section of the tube (e.g. a perforated cover sheet or an array of resonators with rectangular patterns inside a tube with a circular cross-section), the properties of the object might be severely modified. A way to quantify this mismatched pattern effect is to perform several measurements with varying positions of the cuts relative to the object structure. In any case, the information about this mismatched pattern shall be added in the report as described in Clause 10.

8.2 Determination of the sound velocity, wavelength and characteristic impedance

Before starting a measurement, the velocity of sound, c_0, in the tube shall be determined, after which the wavelengths at the frequencies of the measurements shall be calculated.

The velocity of sound, in meter per second, can be assessed accurately with knowledge of the tube air temperature from Formula (5):

$$c_0 = 343{,}2\sqrt{T_0 / 293} \tag{5}$$

where T_0 is the temperature, in Kelvin. The wavelength then follows from Formula (6):

$$\lambda_0 = c_0 / f \tag{6}$$

The density of the air, ρ_0, can be calculated from Formula (7):

$$\rho_0 = \rho_{\mathrm{ref}} \left(p_0 T_{\mathrm{ref}}\right) / \left(p_{\mathrm{ref}} T_0\right) \tag{7}$$

where

T_0 is the temperature, in kelvin;

p_0 is the atmospheric pressure, in Pa;

T_{ref} = 293 K;

p_{ref} = 101 325 Pa;

ρ_{ref} = 1,186 kg/m^3.

The characteristic impedance of the air is the product $\rho_0 c_0$.

8.3 Selection of the signal amplitude

The signal amplitude shall be selected to be at least 10 dB higher than the background noise at all frequencies of interest, as measured at the chosen microphone locations.

The frequency response of the loudspeaker should ideally be equalized in the presence of an anechoic termination (i.e. a highly absorbing termination) at the sample location to flatten out the sound pressure level measured at the microphone positions. During a test, any frequency having a response value 60 dB lower than the maximum frequency response value shall be rejected, but an equalization procedure may be performed in the presence of the test sample.

8.4 Selection of the number of averages

Using averaging techniques, errors due to noise can be reduced whatever the signal is: random, pseudo-random or deterministic. Due to the ergodicity hypothesis of the process, averaging over multiple realisations as well as averaging over a longer signal (or multiple longer signals) is also possible.

The number of averages needed depends on the tested material and the required accuracy of the transfer function estimate (see D.4).

8.5 Correction for microphone mismatch

8.5.1 General

When using the two-microphone technique, one of the following procedures for correcting the measured transfer function data for channels mismatch shall be used: repeated measurements with channels interchanged, or predetermined calibration factor. A channel consists of a microphone, a preamplifier and an analyser.

In the case of the one-microphone technique, since only one microphone is used there is no need for correction with respect to microphone mismatch in the evaluation of the transfer function.

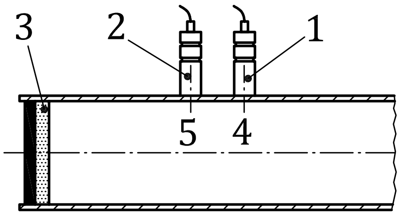
8.5.2 Measurement repeated with the channels interchanged

Correction for microphone mismatch is done by interchanging channels for every measurement on a test specimen. This procedure may be preferred when a limited number of specimens are to be tested.

Observe the following instructions:

Place the test specimen in the tube as described in Clause 7 and measure the two transfer functions H_{12}^{I} and H_{12}^{II} using the same mathematical expressions for both (see 8.6).

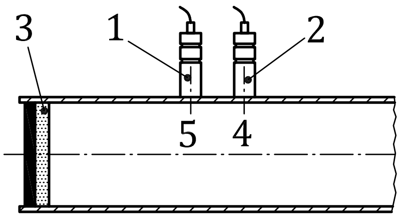
Place the microphones in configuration I (standard configuration, see Figure 5) and store the transfer function H_{12}^{I} Interchange the two channels A and B, as shown in Figure 6.



Key

1 microphone A
2 microphone B
3 test specimen
4 position 1
5 position 2

Figure 5 — Standard configuration (configuration I)



Key

1 microphone A
2 microphone B
3 test specimen
4 position 1
5 position 2

Figure 6 — Configuration with channels interchanged (configuration II)

When interchanging the microphones, ensure that microphone A in configuration II (channels interchanged) occupies the precise location that microphone B occupied in configuration I (standard configuration), and vice versa. Do not switch microphone connections to the preamplifier or signal analyser.

Measure the transfer function H_{12}^{II} and compute the transfer function using Formula (8):

$$H_{12} = \left(H_{12}^{I} \cdot H_{12}^{II}\right)^{1/2} = |H_{12}| e^{j\varphi} \tag{8}$$

If the analyser is only able to measure transfer functions in one direction (e.g from microphone A to microphone B), H_{12} can be computed using Formula (9):

$$H_{12} = \left(H_{12}^{I} / H_{21}^{II}\right)^{1/2} = |H_{12}| e^{j\varphi} \tag{9}$$

8.5.3 Predetermined calibration factor

This is a calibration procedure using a special calibration specimen and the correction is valid for all successive measurements. This procedure may be preferred as a prelude to testing a series of samples, since after calibration the microphones remain in place.

Observe the following instructions:

Place an absorptive specimen in the tube to prevent strong acoustic reflections and measure the two transfer functions H_{12}^{I} and H_{12}^{II} using the same mathematical expressions for both (see 8.6).

Place the microphones in configuration I (standard configuration, see Figure 5) and measure the transfer function H_{12}^{I}. Interchange the two channels (Figure 6).

When interchanging the microphones, ensure that microphone A in configuration II (channels interchanged) occupies the precise location that microphone B occupied in configuration I (standard configuration), and vice versa. Do not switch microphone connections to the preamplifier or signal analyser. Measure the transfer function H_{12}^{II} and compute the calibration factor H_{c} using Formula (10):

$$H_{\mathrm{c}} = \left(H_{12}^{I} / H_{21}^{II} \right)^{1/2} = \left| H_{\mathrm{c}} \right| e^{j\varphi_{\mathrm{c}}} \tag{10}$$

If the analyser is only able to measure transfer functions in one direction (e.g from microphone A to microphone B), H_{c} can be computed using Formula (11):

$$H_{\mathrm{c}} = \left(H_{12}^{I} \cdot H_{12}^{II} \right)^{1/2} = \left| H_{\mathrm{c}} \right| e^{j\varphi_{\mathrm{c}}} \tag{11}$$

For subsequent tests, place the microphones in configuration I (standard configuration). Insert the test specimen and measure the transfer function

$$\hat{H}_{12} = \left| \hat{H}_{12} \right| e^{j\hat{\varphi}} = \hat{H}_{\mathrm{r}} + j\hat{H}_{\mathrm{i}} \tag{12}$$

where

$\hat{H}_{12}$	is the uncorrected transfer function;
$\hat{\varphi}$	is the uncorrected phase angle;
$\hat{H}_{\mathrm{r}}$	is the real part of $\hat{H}_{12}$;
$\hat{H}_{\mathrm{i}}$	is the imaginary part of $\hat{H}_{12}$.

Correct for mismatch in the microphone responses using Formula (13):

$$H_{12} = \left| H_{12} \right| e^{j\varphi} = \frac{\hat{H}_{12}}{\hat{H}_{\mathrm{c}}} \tag{13}$$

8.6 Determination of the transfer function between the two locations

8.6.1 General

Insert the test specimen and in accordance with the requirements of one of the two techniques described in this document, measure the complex acoustic transfer function.

8.6.2 Cross- and autospectra-based estimate

The complex acoustic transfer function may be defined in three ways:

$$H_{12} = \frac{S_{12}}{S_{11}} = \left| H_{12} \right| e^{j\varphi} = H_{\mathrm{r}} + jH_{\mathrm{i}} \tag{14}$$

$$H_{12} = \frac{S_{22}}{S_{21}} = \left| H_{12} \right| e^{j\varphi} = H_{\mathrm{r}} + jH_{\mathrm{i}} \tag{15}$$

$$H_{12} = \left[\frac{s_{12}}{s_{11}} \cdot \frac{s_{22}}{s_{21}}\right]^{1/2} = |H_{12}| e^{j\varphi} = H_\mathrm{r} + jH_\mathrm{i} \tag{16}$$

where

H_r is the real part of H_{12};

H_i is the imaginary part of H_{12}.

Formula (14) is recommended for cases where there is noise at the output (signal 2).

Formula (15) is recommended for cases where there is noise at the input (signal 1)

Formula (16) is recommended for cases involving noise at input (signal 1) and output (signal 2).

8.6.3 Frequency-domain deconvolution

If deterministic measurement signals are used, then the transfer function can be determined as

$$H_{12} = \frac{F_2}{F_1} \tag{17}$$

where

F_1 is the fast Fourier transform of one frame of the time signal recorded by microphone 1;

F_2 is the fast Fourier transform of one frame of the time signal recorded by microphone 2.

Instead of a single frame of each of the two signals, one can use an average of a number of periodic frames of each signal.

8.6.4 Impulse-response based estimate

If deterministic measurement signals are used, then the transfer function can be determined as Formula (18):

$$H_{12} = \frac{H_{s2}}{H_{s1}} \tag{18}$$

where

H_{s1} is the transfer function from the generator to microphone 1;

H_{s2} is the transfer function from the generator to microphone 2.

The transfer functions H_{s1} and H_{s2} are, in turn, determined as a Fourier transform of the impulse responses, which can be measured with frequency-domain convolution, followed by an inverse Fast Fourier Transform, or with the cross-correlation technique, in case one uses an m-sequence measurement signal.

When the two impulse responses have been determined, it is possible to truncate the measured impulse responses at the time where the measured decaying impulse response reaches the background noise floor, and thereby reduce the effects of background noise.

An arbitrarily fine frequency resolution for H_{12} can be achieved by using zero-padding after the two impulse responses have been truncated.

For the single-microphone technique, use the procedure given in Annex B.

8.7 Determination of the reflection coefficient

Calculate the reflection coefficient using Formula (19):

$$r = |r| e^{j\phi r} = r_{\mathrm{r}} + jr_{\mathrm{c}} = \frac{H_{12} - H_{l}}{H_{R} - H_{12}} e^{2jk_0 x_1} \tag{19}$$

where

r_{r} is the real component;

x_1 is the distance between the reference plane of the sample and the further microphone location;

ϕ_r is the phase angle of the normal incidence reflection coefficient;

k_0 is the complex wave number of the waves propagating in the air contained by the impedance tube. The attenuation of the tube (not be confused with the attenuation of the material) can be described analytically through a replacement of the real wave number k_0 by a complex wave number as described in A.2.1.2.

H_{l} and H_{R} are defined in Annex D.

8.8 Determination of the sound absorption coefficient

Calculate the normal incidence sound absorption coefficient according to Formula (20):

$$\alpha_n = 1 - |r|^2 = 1 - r_{\mathrm{r}}^2 - r_{\mathrm{i}}^2 \tag{20}$$

where

r_{r} is the real component;

r_{i} is the imaginary component.

8.9 Determination of the specific acoustic impedance ratio

Calculate the specific acoustic impedance ratio according to Formula (21):

$$Z/(\rho_0 c_0) = R/(\rho_0 c_0) + jX/(\rho_0 c_0) = (1+r)/(1-r) \tag{21}$$

where

R is the real component;

X is the imaginary component;

$\rho_0 c_0$ is the characteristic impedance of air.

8.10 Determination of the specific acoustic admittance ratio

Calculate the specific acoustic admittance ratio according to Formula (22):

$$G\rho_0 c_0 = g\rho_0 c_0 - jb\rho_0 c_0 = \rho_0 c_0 / Z \tag{22}$$

where

g is the real component;

b is the imaginary component.

9 Precision

Details of the test procedure and signal processing shall be chosen so as to yield an uncertainty of 1 % or better for the amplitude and 0,6° or better for the phase of the transfer function at all reported frequencies (see D.4).

NOTE The subsequent translation of these uncertainties to the determined acoustic material properties would be speculative due to error sources other than the transfer function evaluation, particularly with respect to the material samples and placement, bias errors (see Annex D) and reference plane definition.

The accuracy of the measured quantities in this document should follow the general definitions reported in ISO 5725-1[4]. For the uncertainties related to the two-microphone method, the work by Schultz et al. 2007[5] is suggested.

Information concerning the reproducibility and repeatability of this standard is available in interlaboratory comparison tests[6][7].

10 Test report

The test report shall include the following information:

a) a statement, if true in all details, that the test was performed in accordance with this document, i.e. ISO 10534-2:2023; if not, state the deviations;

b) name and address of the testing laboratory;

c) name of the manufacturer and identification of the test specimen (tradename, if available);

d) name and address of the person or organization or person who ordered the test;

e) description of the test specimen and its acoustically relevant characteristics:

   1) structural data such as:

      — lateral dimensions and total thickness,

      — flatness of the surface or characteristic profile height, if any,

      — number, arrangement and thickness of layers, including air spaces,

      — dimensions of structural units, such as resonators, and their arrangement,

      — positions of the cuts of the test sample relative to characteristic lines of test objects with lateral structures,

      — structure, thickness and porosity of covers such as grids, and perforated metal sheet,

   2) material data such as:

      — bulk density and, if available, air flow resistivity of porous materials,

      — component materials of the test object,

   3) construction characteristics such as: connection of layers to each other (glued, or other);

f) description of the test specimen: its number, size and mounting;

g) temperature and atmospheric pressure; relative humidity if measured;

h) date of the test;

i) tabular and graphical presentation of the test results as a function of frequency:

— the results as third-octave band data shall always be reported in a tabular form (see Table 1). The calculation of the absorption coefficient for each third of octave band is obtained as the arithmetic average of the narrow band frequency values at the frequencies contained in each third of octave. For each third of octave band a minimum of six narrow band frequencies are required for the calculation.

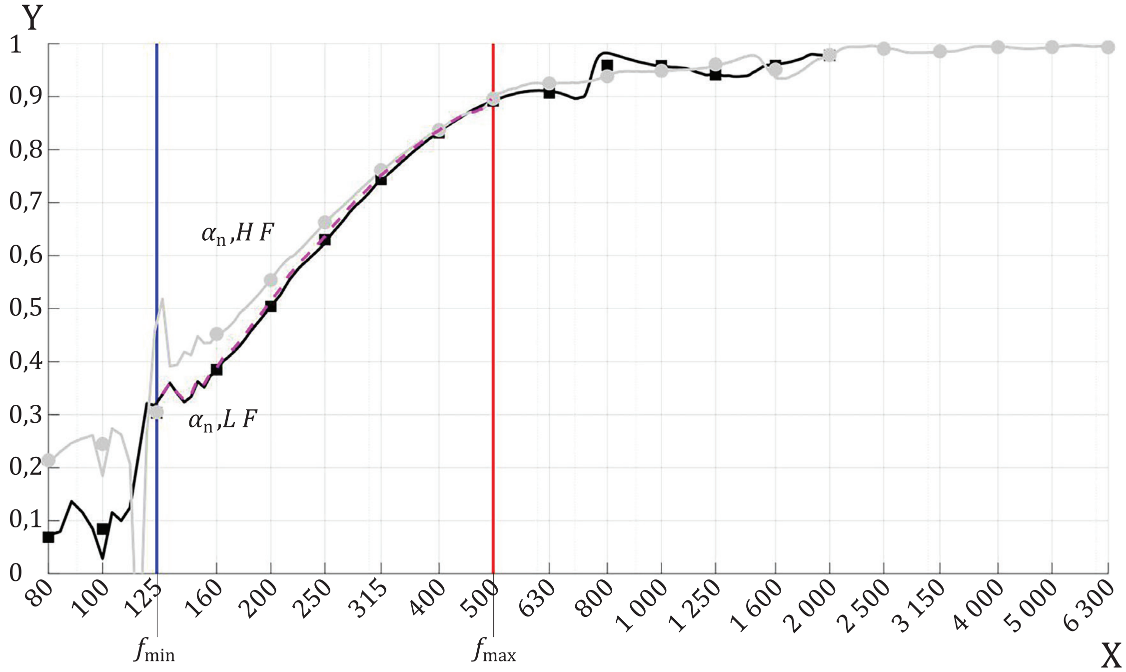
— if more than one test specimen of a sample (of the same material) was tested, the individual results or uncertainty shall be indicated, the mean value and the relative standard deviation shall be given;

— if different impedance tubes were used to cover a wider frequency range, the working frequency ranges shall overlap by a minimum of one octave (as described in ISO 266[3]) and the values, in the overlap domain, should not differ by more than 0,05 in terms of α_n without being clearly indicated or discussed. The results for each tube should be reported as well as the averaged and combined one (see example in Figure 7) to qualify and quantify the possible effects of the boundary conditions and inhomogeneity of the tested samples. The averaging technique used to obtain the averaged and combined result shall be as given in Formula (23):

$$\alpha_{\mathrm{n}}(f)=\alpha_{\mathrm{n,LF}}\frac{f_{\max}-f}{f_{\max}-f_{\min}}+\alpha_{\mathrm{n,HF}}\left(1-\frac{f_{\max}-f}{f_{\max}-f_{\min}}\right) \tag{23}$$

where $\alpha_{\mathrm{n}}(f)$ is the averaged normal incidence sound absorption coefficient at frequency f, $\alpha_{\mathrm{n,LF}}$ is the normal incidence sound absorption coefficient obtained with the tube of the larger section, $\alpha_{\mathrm{n,HF}}$ is the normal incidence sound absorption coefficient obtained with the tube of the smaller section, $f_{\min}$ is the low frequency of the overlapping domain, $f_{\max}$ is the high frequency of the overlapping domain.

j) description of the instruments used, including details about the impedance tube and the test procedure.



Key

X frequency [Hz]

Y normal incidence sound absorption coefficient

Figure 7 — Example of graphical representation of the independent and combined data for the overlapping technique described in Clause 10 i) with Formula (23)

Table 1 — Third octave frequency data of the independent measurements plotted in Figure 7

Frequency Hz	α_n **(tube Ø100 mm)**	α_n **(tube Ø29 mm)**	α_n **(combined)**
80	0,07		0,07
100	0,08		0,08
125	0,30	0,30	0,30
160	0,38	0,45	0,39
200	0,50	0,55	0,51
250	0,63	0,66	0,64
315	0,74	0,76	0,75
400	0,83	0,84	0,84
500	0,89	0,90	0,90
630	0,91	0,93	0,93
800	0,96	0,94	0,94
1 000	0,96	0,95	0,95
1 250	0,94	0,96	0,96
1 600	0,96	0,95	0,95
2 000	0,98	0,98	0,98
2 500		0,99	0,99
3 150		0,99	0,99
4 000		0,99	0,99
5 000		0,99	0,99
6 300		0,99	0,99

Annex A
(normative)

Preliminary measurements

A.1 Prior to or following each measurement session or test

A.1.1 Microphone amplitude calibration

Prior to each measurement session, it is recommended to calibrate each microphone amplitude to a tolerance of ±0,3 dB or better with respect to a stable sound source over the working frequency range. A single frequency test, for example piston phone, is deemed sufficient provided the microphone is known to have a linear frequency response over the working frequency range.

Prior to each measurement session, a check of the rigid termination should be done. This check consists in the sound absorption measurement of the empty tube (i.e. without any sample) in the frequency range of use of the tube.

The sound absorption coefficient measured for the empty tube should not exceed 0,10 in narrow frequency bands and 0,05 in third-octave bands representation. In the latter case, it is recommended to re-do the calibration, with a sample of porous material inside the tube during the calibration process, and to re-check the rigid termination.

A.1.2 Temperature measurement

Prior to and following each test, a temperature-measuring device with a measurement tolerance of ±0,5 K or better shall be used to report the air temperature.

A.1.3 Air pressure measurement

Prior to and following each test, the air pressure shall be measured with a measurement tolerance of, at least, ±0,5 kPa.

A.1.4 Relative humidity measurement

If the information is available, the relative humidity shall be measured with a tolerance of, at least, ±2 %.

A.1.5 Signal-to-noise ratio

Prior to each measurement session, the sound pressure spectrum shall be measured at each microphone position with the source sound source on and off. The source sound spectra shall be at least 10 dB higher than the background noise at all reported frequencies. Frequencies within the reported sequence which do not comply with this requirement shall be recorded in the test report.

A.2 Periodic calibration

A.2.1 Tube attenuation

A.2.1.1 Correction for tube attenuation

The incident sound wave $p_{I(x)}$ and the reflected sound wave $p_{R(x)}$ will generally be attenuated during propagation due to viscous and thermal losses. The main effect of the attenuation is a monotonic increase of the amplitudes of the sound pressure minima with increasing distance from the reflecting surface. Normally this will not influence the results obtained using the methods given in this document. However, when the distance from the surface of the test specimen to the nearest microphone is larger than three tube diameters or three times the maximum transverse dimension of a rectangular impedance tube, corrections shall be applied for this in the evaluation of the quantities which are determined according to this document.

The attenuation constant is determined in accordance with A.2.1.2 and can be included in the calculations of the reflection coefficient.

A.2.1.2 Determination of corrections for the tube attenuation

The attenuation of the tube (not be confused with the attenuation of the material) can be described analytically through a replacement of the real wave number k_0 by a complex wave number:

$$k_0 = k_0' - jk_0'';\ k_0' = 2\pi/\lambda_0 \tag{A.1}$$

where k_0'' is the attenuation, in radian per unit length.

The tube attenuation is best determined experimentally but it requires a moving microphone probe (see ISO 10534-1:1996, A.3[1]).

For fixed-microphone positions, the tube attenuation can be estimated, see A.2.1.3

A.2.1.3 Estimation method

At the lower end of the working frequency range, if two pressure minima can not be explored with sufficient precision, the attenuation constant can be estimated numerically by Formula (A.2):

$$k_0'' = 1{,}94 \times 10^{-2} \sqrt{f/(c_0 d)} \tag{A.2}$$

where d, in metres, is the diameter of circular tubes or the ratio of four times the cross-sectional area to the perimeter of rectangular tubes, and f is the frequency in hertz. This estimation, however, does not consider sources of attenuation such as porous walls and objects in the tube. Thus, it can be considered as a lower limit.

Note that the sound speed is assumed to be independent of the frequency.

A.2.2 Determination of the acoustic centre of a microphone

The acoustic centre of a side-mounted microphone may be different from its geometric centre causing erroneous location of the microphone(s) and subsequent errors in acoustic property assignment.

At present no acoustic centre calibration procedures have been validated for microphones with fixed positions as described in this document. Such errors shall therefore be considered part of the uncertainties associated with the method. It is strongly advised that orientation of a side-wall-mounted microphone with respect to the longitudinal axis of the tube be maintained throughout a test or test series. Choosing microphones with small membrane diameters in relation to their spacing will also reduce this error.

Annex B
(normative)

Procedure for the one-microphone technique

The accuracy of the one-microphone technique depends upon the way it is implemented. It is absolutely necessary to ensure that the sound field at the two fixed locations is sampled sequentially with a stable sound source. A deterministic signal for the sound source is recommended.

This method uses the same set-up as that described for the two-microphone technique (see Figure 3). However, the transfer functions between the sound pressures at the two positions is calculated from two transfer functions measured in a sequential manner:

$$H_{12} = H_{s2} / H_{s1} \quad \text{(B.1)}$$

where s is the generator signal.

The two transfer functions H_{s1} and H_{s2} can be determined with the impulse response-based method (see 8.6).

The microphone location not used shall be sealed properly, preferably by employing a dummy microphone plug.

Annex C
(informative)

Theoretical background

The measuring method is based on the fact that the sound reflection factor at normal incidence r can be determined from the measured transfer function H_{12} between two microphone positions in front of the tested material, see Figure 3.

The sound pressures of the incident wave p_I and the reflected wave p_R are, respectively given by Formula (C.1) and (C.2):

$$p_I = \hat{p}_I e^{jk_0 x} \tag{C.1}$$

and

$$p_R = \hat{p}_R e^{-jk_0 x} \tag{C.2}$$

where

p_I and p_R are the magnitudes of p_I and p_R at the reference plane ($x = 0$);

$k_0 = k_0' - jk_0''$ is a complex wave number.

The sound pressures p_1 and p_2 in the two microphone positions are respectively given by Formula (C.3) and (C.4):

$$p_1 = \hat{p}_I e^{jk_0 x_1} + \hat{p}_R e^{-jk_0 x_1} \tag{C.3}$$

and

$$p_2 = \hat{p}_I e^{jk_0 x_2} + \hat{p}_R e^{-jk_0 x_2} \tag{C.4}$$

The transfer function for the incident wave alone H_I is given by Formula (C.5):

$$H_I = \frac{p_{2I}}{p_{1I}} = e^{-jk_0 (x_1 - x_2)} = e^{-jk_0 s} \tag{C.5}$$

where $s = x_1 - x_2$ and is the separation between the two microphones.

Similarly, the transfer function for the reflected wave alone H_R is given by Formula (C.6):

$$H_R = \frac{p_{2R}}{p_{1R}} = e^{jk_0(x_1 - x_2)} = e^{jk_0 s} \tag{C.6}$$

The transfer function H_{12} for the total sound field may now be obtained by using Formula (C.3) and (C.4) and noting that $\hat{p}_R = r\hat{p}_I$ as given by Formula (C.7):

$$H_{12} = \frac{p_2}{p_1} = \frac{e^{jk_0 x_2} + re^{-jk_0 x_2}}{e^{jk_0 x_1} + re^{-jk_0 x_1}} \tag{C.7}$$

Transposing Formula (C.7) to yield *r*, and using Formulas (C.5) and (C.6), one has:

$$r = \frac{H_{12} - H_I}{H_R - H_{12}} e^{2jk_0 x_1} \tag{C.8}$$

The sound reflection factor *r* at the reference plane ($x = 0$) can now be determined from the measured transfer functions, the distance x_1 and the wave number k_0 which may include the tube attenuation constant k_0''.

NOTE It is important that the transfer function is compensated for phase and pressure amplitude mismatch of the microphones when the two-microphone technique is used.

Annex D
(informative)

Error sources

D.1 General

Errors may be considered in two main categories, bias errors and random errors.

D.2 Bias errors

D.2.1 Overview

Bias errors include potential errors either in the measurement or the analysis (after processing) such as aliasing, leakage, and picket fence errors, together with microphone mismatch and/or errors in measurement of length or distance. Frequency aliasing, leakage and picket fence errors will be minimized by employing well-known signal acquisition and processing techniques. Bias errors associated with the difference between acoustic and geometric microphone centres are discussed in A.2.2.

D.2.2 Time aliasing (non-periodic signals)

For non-periodic signals time aliasing arises when the duration of each record is similar to or less than the impulse response of the system under investigation, causing a crosstalk corruption in the signal processing.

Time aliasing may be avoided by selecting the duration of each record to be much larger than the acoustic propagation times within the impedance tube system, that is

$$t > 2x_1 / c_0 \tag{D.1}$$

where

t is the sample record length, in seconds;

x_1 is the distance from the sample to the furthest microphone, in metres;

c_0 is the sound velocity, in metres per second.

D.2.3 Phase mismatch

When using the two-microphone technique, the error of phase mismatch between microphones is unavoidable and shall be compensated for. This may be achieved by following one of the procedures required by this document and described in 8.5.

D.2.4 Amplitude mismatch

When using two microphones, a sensitivity mismatch may exist. This error is generally not important provided it is constant, and in relation to the two-microphone technique it is essentially corrected by the measurement procedure described in 8.5. To ensure consistent amplitude readings throughout a test sequence, however, a separate sound pressure level test is specified in Annex A.

When using the two-microphone technique, some advantage is gained by having the microphones calibrated so that their amplitudes will not differ by more than 0,3 dB.

D.3 Random error

Random errors arise usually from processing random noise records of finite length, but may also involve electrical noise in the instrumentation, or extraneous acoustic signals.

Random error is kept low by suitable averaging and is also minimized by employing deterministic signals. Selection of bandwidth and signal length to achieve ensemble averaging of the microphone spectra is usually effective in limiting this error for each channel.

The record length and bandwidth can be selected to yield a particular relative standard deviation for the measured r.m.s. level of a random signal. Typically, a product of frequency bandwidth and total averaging time of 50 to 100 will keep random error low.

Alternately, the number of averages required to achieve a particular standard error for measurements at a particular microphone location is given by Formula (D.2):

$$n = [1/(2\sigma)]^2 \tag{D.2}$$

where

n is the number of independent (no overlap) spectra averaged;

σ is the standard error.

D.4 Accuracy of the transfer function

Of particular interest to this document is the final accuracy of the determined transfer function. An estimate of the number of averages required to achieve a given normalized standard error for the magnitude of the transfer function estimate at a particular frequency is given by Formula (D.3):

$$n = \frac{1}{2\varepsilon^2}\left[\frac{1}{\gamma^2} - 1\right] \tag{D.3}$$

where

n is the number of averages;

ε is the normalized standard error;

γ^2 is the coherence function.

The coherence function is given by Formula (D.4):

$$\gamma^2 = |S_{12}|^2 / (S_{11} S_{22}) \tag{D.4}$$

NOTE The determination of the coherence function is subjected to bias errors associated with record length (or frequency resolution) and reverberation effect in the tube. It is expected that the coherence between microphones will be greater than 0,9 except for cases with a highly reflecting termination, the coherence will be less than 0,5 at frequencies where there is a pressure node at either one of the microphones.

Annex E
(informative)

Estimation of diffuse sound absorption coefficient α_{st} of locally reacting absorbers from the results of this document

The sound absorption coefficient α_{st} for diffuse (i.e. omnidirectional) sound incidence can be computed for absorbers of the "locally reacting" type (i.e. without sound propagation inside the absorber with a component parallel to its surface, see definition 3.17 as well as Figure 1) from the normal-incidence surface impedance, determined according to this document, normalized by the impedance of air: $z = z' + jz''$

Note that most materials are non-locally reacting ones and there is no direct link between

— the sound absorption coefficients measured with an impedance tube using this ISO 10534 series;

— and the sound absorption in diffuse sound field using e.g. ISO 354[2].

For such a non-locally reacting material, impedance tube measurements can be used to compute the intrinsic properties of the material (see Annex F) and then a characterisation can be assessed. From the characterisation, a diffuse sound field simulation can then be computed.

The relationship, for locally reacting materials and assuming infinite lateral dimensions, introduced by A. London in 1950,[5] is given by Formula (E.1):

$$\alpha_{st} = \frac{8z'}{z'^2 + z''^2}\left[1 + \frac{z'^2 - z''^2}{z''\left(z'^2 + z''^2\right)}\tan^{-1}\left(\frac{z''}{1+z'}\right) - \frac{z'}{z'^2 + z''^2}\ln\left[(1+z')^2 + z''^2\right]\right] \tag{E.1}$$

where

$z = Z/(\rho_0 c_0)$	the normal-incidence surface impedance normalized by the impedance of air: $\rho_0 c_0$;
$z' = R/(\rho_0 c_0)$	the real part of the normalized surface impedance;
$z'' = X/(\rho_0 c_0)$	the imaginary part of the normalized surface impedance.

If $z'' = 0$ then the last term in the square brackets will be $1/(1+z')$. The maximum value of α_{st} which can be obtained after this formula is 0,96.

Similar explicit analytical relationships do not exist for bulk reacting absorbers (absorbers with inside sound propagation parallel to the surface such as low-density open-cellular foams or mineral fibre absorbers).

Annex F
(informative)

Estimation of intrinsic properties

F.1 General

In this annex, two methods for the estimation of the frequency-dependent and complex intrinsic properties of a tested material are presented. These intrinsic properties are the characteristic impedance of the material Z_c, its characteristic wave number k_c, its dynamic mass density ρ_eq and its dynamic bulk modulus K_eq. From these quantities, the parameters of the material can be estimated (see e.g [9].).

These methods are sensitive to possible phase shifts during the measurements of the surface impedances. Signal analysis removing phase shifts is highly recommended while using these methods. The estimation of the intrinsic properties is also sensitive to the structural vibrations of the sample and therefore their estimates should be done outside the frequency ranges of such vibrations when they appear.

Note that, when available, the 3-microphone method or the 4-microphone method are preferred to estimate the intrinsic properties (and parameters) of materials. These methods are scheduled for standardisation in ISO 10534-3 and ISO 10534-4.

F.2 The two-cavity method

This method introduced by Utsuno et al. 1989[10] requires the measure of the acoustic surface impedance Z_0 at a reference plane of a material sample of thickness d backed by an air-gap of thickness L leading to a surface impedance behind the porous material of Z_1 (usually neglecting any flow distortion at the interface between the porous sample and the backing air-gap (see e.g [11].).

A second measurement is done, with an acoustic impedance Z'_0 at the same reference surface plane of the sample material sample, this time backed by an air-gap of thickness L' leading to a surface impedance behind the porous material of Z'_1.

Thus, two measurements of the surface impedance of the same sample of material (with thickness d) backed with two different air-gaps L and L' are needed as given by Formula (F.1) and Formula (F.2)

$$Z_\mathrm{c} = \pm\left(\frac{Z_0 Z'_0 (Z_1 - Z'_1) - Z_1 Z'_1 (Z_0 - Z'_0)}{(Z_1 - Z'_1) - (Z_0 - Z'_0)}\right)^{1/2} \tag{F.1}$$

$$k_c = \frac{1}{2jd}\ln\left(\frac{Z_0 + Z_c Z_1 - Z_c}{Z_0 - Z_c Z_1 + Z_c}\right) \tag{F.2}$$

where the sign in the formula of Z_c is selected so as to let the real part of Z_c be positive.

Note that the appropriate set of air space depths (L and L') shall be selected so as to not satisfy Formula (F.3):

$$L - L' = c_0 / (2fu) \tag{F.3}$$

to avoid the elimination of the terms $(Z_0 - Z'_0)$ in the expression of Z_c.

ρ_{eq} and K_{eq} are obtained from Z_c and k_c, as introduced in Clause 3.

F.3 The two-thickness method

This method introduced by Smith & Parrott 1983[12] requires two measurements of the surface impedance of a sample of material (with thickness d) and another one, of the same material, but of thickness $2d$. Alternatively, two samples with each a thickness d can be used to obtain the sample of thickness $2d$ ensuring no air-gap and no change in the morphology can exist at the interface between the 2 samples.

The first surface impedance writes:

$$Z_1 = -j \frac{Z_c}{\tan(k_c d)} \tag{F.4}$$

The second surface impedance writes:

$$Z_2 = -j \frac{Z_c}{\tan(k_c 2d)} \tag{F.5}$$

using the identity $\tan(2d) = 2\tan(d) / \left(1 - \tan^2 d\right)$ leads to:

$$k_c = \frac{\tan^{-1}\left[(1 - 2 Z_2 / Z_1)^{1/2}\right]}{d} \tag{F.6}$$

$$\text{and } Z_c = Z_1 (2 Z_2 / Z_1 - 1)^{1/2} \tag{F.7}$$

ρ_{eq} and K_{eq} are obtained from Z_c and k_c as introduced in Clause 3.

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