INTERNATIONAL STANDARD

ISO 10534-2

First edition
1998-11-15

Acoustics — Determination of sound absorption coefficient and impedance in impedance tubes —

Part 2:
Transfer-function method

Acoustique — Détermination du facteur d'absorption acoustique et de l'impédance des tubes d'impédance —

Partie 2: Méthode de la fonction de transfert

Reference number
ISO 10534-2:1998(E)

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Contents

Page

© ISO 1998

All rights reserved. Unless otherwise specified, no part of this publication may be reproduced or utilized in any form or by any means, electronic or mechanical, including photocopying and microfilm, without permission in writing from the publisher.

International Organization for Standardization
Case postale 56 • CH-1211 Genève 20 • Switzerland
Internet iso@iso.ch

Printed in Switzerland

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Foreword

ISO (the International Organization for Standardization) is a worldwide federation of national standards bodies (ISO member bodies). The work of preparing International Standards is normally carried out through ISO technical committees. Each member body interested in a subject for which a technical committee has been established has the right to be represented on that committee. International organizations, governmental and non-governmental, in liaison with ISO, also take part in the work. ISO collaborates closely with the International Electrotechnical Commission (IEC) on all matters of electrotechnical standardization.

Draft International Standards adopted by the technical committees are circulated to the member bodies for voting. Publication as an International Standard requires approval by at least 75 % of the member bodies casting a vote.

International Standard ISO 10534-2 was prepared by Technical Committee ISO/TC 43, *Acoustics,* Subcommittee SC 2, *Building acoustics.*

ISO 10534 consists of the following parts, under the general title *Acoustics — Determination of sound absorption coefficient and impedance in impedance tubes*:

— *Part 1: Method using standing wave ratio*

— *Part 2: Transfer-function method*

Annnexes A to C form an integral part of this part of ISO 10534. Annexes D to G are for information only.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Acoustics — Determination of sound absorption coefficient and impedance in impedance tubes —

Part 2:
Transfer-function method

1 Scope

This test method covers the use of an impedance tube, two microphone locations and a digital frequency analysis system for the determination of the sound absorption coefficient of sound absorbers for normal sound incidence. It can also be applied for the determination of the acoustical surface impedance or surface admittance of sound absorbing materials. Since the impedance ratios of a sound absorptive material are related to its physical properties, such as airflow resistance, porosity, elasticity and density, measurements described in this test method are useful in basic research and product development.

The test method is similar to the test method specified in ISO 10534-1 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 noise 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, the normal incidence absorption and the impedance ratios of the acoustic material. The test method is intended to provide an alternative, and generally much faster, measurement technique than that of ISO 10534-1.

Compared with the measurement of the sound absorption in a reverberation room according to the method specified in ISO 354, there are some characteristic differences. The reverberation room method will (under ideal conditions) determine the sound absorption coefficient for diffuse sound incidence, and the method can be used for testing of materials with pronounced structures in the lateral and normal directions. However, the reverberation room method requires test specimens which are rather large, so it is not convenient for research and development work, where only small samples of the absorber are available. The impedance tube method is limited to parametric studies at normal incidence but 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, diffuse incidence sound absorption coefficients can be estimated from measurement results obtained by the impedance tube method. For transformation of the test results from the impedance tube method (normal incidence) to diffuse sound incidence, see annex F.

2 Definitions and symbols

For the purposes of this part of ISO 10534 the following definitions apply.

2.1
sound absorption coefficient at normal incidence
α
ratio of sound power entering the surface of the test object (without return) to the incident sound power for a plane wave at normal incidence

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

2.2
sound pressure reflection factor at normal incidence
r
complex ratio of the amplitude of the reflected wave to that of the incident wave in the reference plane for a plane wave at normal incidence

2.3
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 The reference plane is assumed to be at $x = 0$.

2.4
normal surface impedance
Z
ratio of the complex sound pressure $p(0)$ to the normal component of the complex sound particle velocity $v(0)$ at an individual frequency in the reference plane

2.5
normal surface admittance
G
inverse of the normal surface impedance Z

2.6
wave number
k_0
variable defined by

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

where

- ω is the angular frequency;
- f is the frequency;
- c_0 is the speed of sound.

NOTE In general the wave number is complex, so

$$k_0 = k_0' - jk_0''$$

where

- k_0' is the real component ($k_0' = 2\pi/\lambda_0$);
- λ_0 is the wavelength;
- k_0'' is the imaginary component which is the attenuation constant, in nepers per metre.

2.7
complex sound pressure
p
Fourier Transform of the temporal acoustic pressure

2.8
cross spectrum
S_{12}
product $p_2 \cdot p_1^*$, determined from the complex sound pressures p_1 and p_2 at two microphone positions

NOTE * means the complex conjugate.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

2.9
auto spectrum
S_{11}
product $p_1 \cdot p_1{}^*$, determined from the complex sound pressure p_1 at microphone position one

NOTE * means the complex conjugate.

2.10
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 $[(S_{12}/S_{11})(S_{22}/S_{21})]^{1/2}$

2.11
calibration factor
H_c
factor used to correct for amplitude and phase mismatches between the microphones

NOTE See 7.5.2.

3 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 (random, pseudo-random sequence, or chirp), 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 factor (see annex C), the normal-incidence absorption coefficient, and the impedance ratio of the test material.

The quantities are determined as functions of the frequency 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 width of the tube and the spacing between the microphone positions. An extended frequency range may be obtained from the combination of measurements with different widths and spacings.

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

1: two-microphone method (using two microphones in fixed locations);
2: 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 require 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 the assessment of tuned resonators and/or precision, and its requirements are described in more detail in annex B.

4 Test equipment

4.1 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, but variations involving a centre mounted microphone or probe microphone are possible.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

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.

4.2 Working frequency range

The working frequency range is

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

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 accuracy of the signal processing equipment.

f_u is chosen to avoid the occurrence of non-plane wave mode propagation.

The condition for f_u is:

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

for circular tubes with the inside diameter d in metres and f_u in hertz.

$$d < 0{,}5\,\lambda_u; \quad f_u \cdot d < 0{,}50\,c_0 \tag{3}$$

for rectangular tubes with the maximum side length d in metres; c_0 is the speed of sound in metres per second given by equation (5).

The spacing s in metres between the microphones shall be chosen so 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 accuracy of the analysis system but, as a general guide, the microphone spacing should exceed 5 % of the wavelength corresponding to the lower frequency of interest, provided that the requirements of equation (4) are satisfied. A larger spacing between the microphones enhances the accuracy of the measurements.

4.3 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.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

The loudspeaker generally will produce non-plane modes besides the plane wave. They will die out within a distance of about three tube diameters or three times the maximum 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 suggested above, but in any case no closer than one diameter or one maximum lateral dimension, as appropriate.

Test samples will also cause proximity distortions to the acoustic field and the following recommendation is given for the minimum spacing between microphone and sample, depending upon the sample type:

non-structured:	½ diameter or ½ maximum lateral dimension
semi-lateral structured:	1 diameter or 1 maximum lateral dimension
strongly asymmetrical:	2 diameters or 2 times the maximum lateral dimension

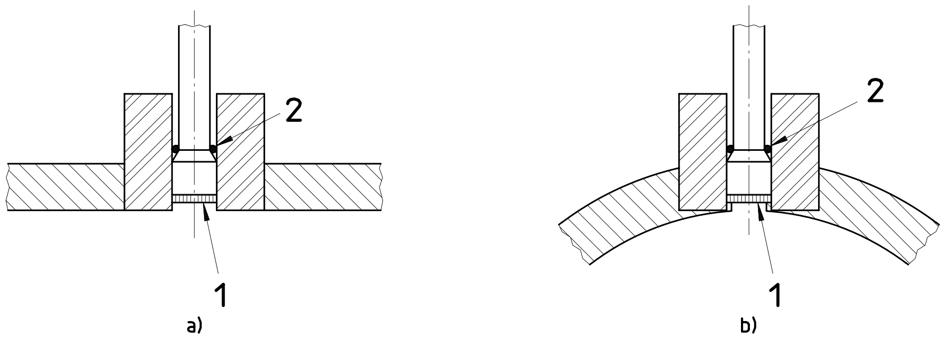
4.4 Microphones

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. In addition, it is recommended that the microphone diameters be less than 20 % of the spacing between them.

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.

4.5 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 as shown in figure 1; 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.



Key

1 Microphone

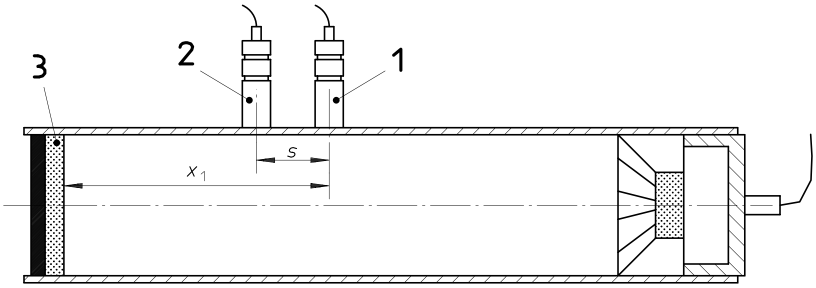
2 Sealing

Figure 1 — Examples of typical microphone mounting

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 an accuracy of ± 0,2 mm or better, and their spacing s (see figure 2) shall be recorded. Traversing microphone positions shall be known to an accuracy of ± 0,5 mm or better.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.



Key

1 Microphone A
2 Microphone B
3 Test specimen

Figure 2 — Microphone positions and distances

4.6 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.2.3.

4.7 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 as 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 (vaseline 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 (vaseline) 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 20 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.

4.8 Signal processing equipment

The signal processing system shall consist of an amplifier, and a two-channel Fast Fourier Transform (FFT) analysing system. The system is required to measure the sound pressure at two microphone locations and to calculate the transfer function H_{12} between them. A generator capable of producing the required source signal (see 4.10) compatible with the analysing system is also required.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

The dynamic range of the analyser should be greater than 65 dB. The errors in the estimated transfer function H_{12} due to nonlinearities, 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.

4.9 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.

4.10 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 and a periodic pseudo-random sequence is well suited for this method, although special signal processing will be required. The processing first involves an m-sequence correlation via the fast Hadamard transform to produce an impulse response. The frequency response is subsequently obtained by Fourier transform of the impulse response.

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

4.11 Loudspeaker termination

Resonances of the air column in the impedance tube will always arise. These should be suppressed by lining the inside of the impedance tube near the loudspeaker with at least a 200 mm length of an effective sound-absorbent material.

4.12 Thermometer and barometer

The temperature in the impedance tube 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.

5 Preliminary test and measurements

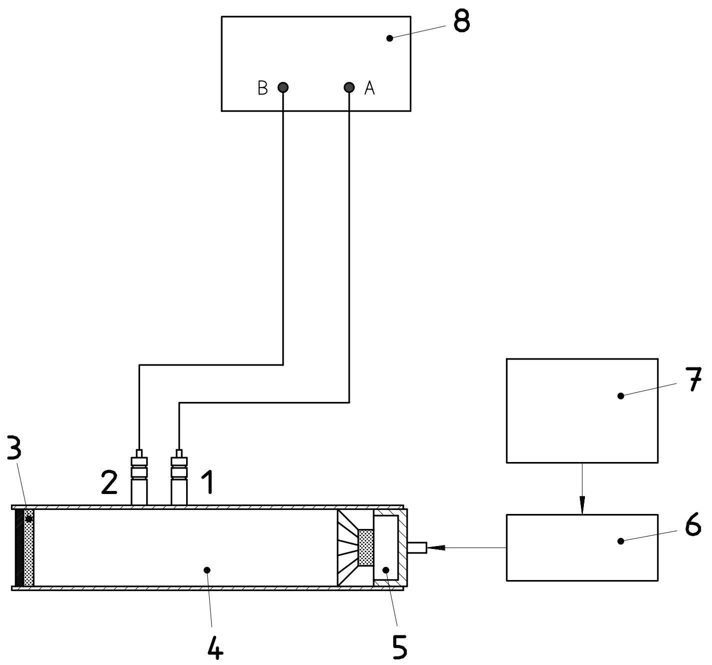
The test equipment shall be assembled, typically as shown in figure 3, and checked before use by a series of tests. These tests help to exclude error sources and secure the minimum requirements. The checks may be considered to be in two categories: prior to or following each test, and periodic calibration tests. In each case the loudspeaker should be operated for at least 10 min prior to a measurement to allow the temperature to stabilize.

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

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Periodic calibrations are performed with a rigid termination of the empty 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 B.



Key

1 Microphone A
2 Microphone B
3 Test specimen
4 Impedance tube
5 Sound source
6 Amplifier
7 Signal generator
8 Frequency analysis system

Figure 3 — Example of layout for test equipment

6 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 vaseline or plasticine is recommended. The test sample can be held more firmly, if necessary, by taping and greasing the entire edge. For example, samples such as carpet material should be firmly attached to the back plate using 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, non-vibrating wire grid with wide mesh.

If the specimen has an uneven or irregular face, surface 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 two specimens, more if the sample is not uniform, should be tested in repeated measurements using the same mounting conditions.

If the test object has a regular lateral structure (e.g. perforated cover sheets, resonator arrays, etc.), the cuts of the test samples shall be along lines of symmetry of that structure. If the dimensions of multiple structural units of the

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

test object do not fit with the cross dimensions of the impedance tube, the measurements shall be performed with several test samples with varying positions of the cuts relative to the structure. Repetition of the measurements with test samples cut from different places of the test object are also necessary with materials which are laterally inhomogeneous (such as mineral fibre products).

7 Test procedure

7.1 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 6, 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 compliance with 4.3. The reference plane location in relation to microphone 1, depicted in figure 2, shall be reported with an accuracy 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.

7.2 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 can be assessed accurately with knowledge of the tube air temperature from equation (5):

$$c_0 = 343{,}2 \sqrt{T / 293} \quad \text{m/s} \tag{5}$$

where T is the temperature, in kelvin.

The wavelength then follows from:

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

The density of the air, ρ, can be calculated from

$$\rho = \rho_0 \cdot \frac{p_a \; T_0}{p_0 \; T} \tag{7}$$

where

T is the temperature, in kelvin;

p_a is the atmospheric pressure, in kilopascals;

T_0 = 293 K;

p_0 = 101,325 kPa;

ρ_0 = 1,186 kg/m^3.

The characteristic impedance of the air is the product ρc_0.

7.3 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.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

The frequency response of the loudspeaker should idealy be equalized in the presence of an anechoic termination at the sample location to flatten out the pressure response 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.

7.4 Selection of the number of averages

Using averaging of the spectra measured at the microphone positions, errors due to noise can be cancelled out. The number of averages needed depends on the tested material and the required accuracy of the transfer function estimate. (See E.2 and E 3.)

7.5 Correction for microphone mismatch

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, preamplifier and analyser channel.

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.

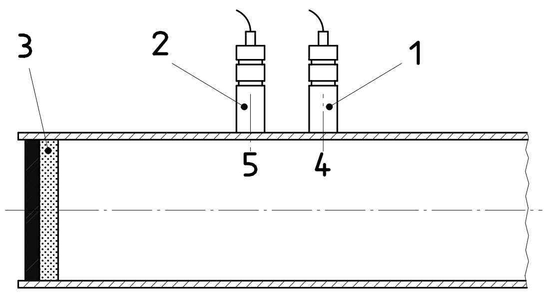
7.5.1 Measurement repeated with the microphones interchanged

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 specimen are to be tested.

Place the test specimen in the tube as described in clause 6 and measure the two transfer functions H_{12}^{I} and H_{12}^{II}, using the same mathematical expressions for both (see 7.6).

Place the microphones in configuration I (standard configuration, see figure 4) and store the transfer function H_{12}^{I}.

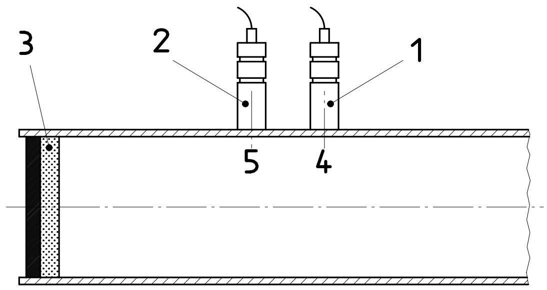
Interchange the two microphones A and B as shown in figure 5.



Key
1 Microphone A
2 Microphone B
3 Test specimen
4 Position 1
5 Position 2

Figure 4 — Standard configuration (configuration I)

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.



Key
1 Microphone B
2 Microphone A
3 Test specimen
4 Position 1
5 Position 2

Figure 5 — Configuration with microphones interchanged (configuration II)

When interchanging the microphones, ensure that microphone A in configuration II (microphones 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 equation (8):

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

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 equation (9):

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

7.5.2 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.

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 7.6).

Place the microphones in configuration I (standard configuration, see figure 4) and measure the transfer function H_{12}^{I}.

Interchange the two microphones (figure 5).

When interchanging the microphones, ensure that microphone A in configuration II (microphones 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 equation (10):

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

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

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 equation (11):

$$H_c = (H_{12}^{\mathrm{I}} \cdot H_{21}^{\mathrm{II}})^{1/2} = |H_c| e^{j\phi_c} \tag{11}$$

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

$$\hat{H}_{12} = |\hat{H}_{12}| e^{j\hat{\phi}} = \hat{H}_r + j\,\hat{H}_i \tag{12}$$

where

$\hat{H}_{12}$ is the uncorrected transfer function;

$\hat{\phi}$ is the uncorrected phase angle;

$\hat{H}_r$ is the real part of $\hat{H}_{12}$;

$\hat{H}_i$ is the imaginary part of $\hat{H}_{12}$.

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

$$H_{12} = |H_{12}| e^{j\phi} = \frac{\hat{H}_{12}}{H_c} \tag{13}$$

7.6 Determination of the transfer function between the two locations

Insert the test specimen and in accordance with the requirements of one of the two techniques described in this part of ISO 10534, measure the complex acoustic transfer function.

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

$$H_{12} = \frac{S_{12}}{S_{11}} = |H_{12}| e^{j\phi} = H_r + j\,H_i \tag{14}$$

$$H_{12} = \frac{S_{22}}{S_{21}} = |H_{12}| e^{j\phi} = H_r + j\,H_i \tag{15}$$

$$H_{12} = \left[\frac{S_{12}}{S_{11}} \cdot \frac{S_{22}}{S_{21}} \right]^{1/2} = H_r + j\,H_i \tag{16}$$

where

H_r is the real part of H_{12};

H_i is the imaginary part of H_{12}.

Equation (14) will normally be used.

Equation (15) is recommended for cases where there is noise at the input.

Equation (16) is recommended for cases involving noise at input and output.

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

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

7.7 Determination of the reflection factor

Calculate the normal incidence reflection factor (see annex D):

$$r = |r| e^{j\phi_r} = r_r + jr_i = \frac{H_{12} - H_I}{H_R - H_{12}} e^{2jk_0 x_1} \tag{17}$$

where

r_r is the real component;

r_i is the imaginary component;

x_1 is the distance between the sample and the further microphone location;

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

H_I and H_R are defined in annex D.

7.8 Determination of the sound absorption coefficient

Calculate the normal incidence sound absorption coefficient:

$$\alpha = 1 - |r|^2 = 1 - r_r^2 - r_i^2 \tag{18}$$

7.9 Determination of the specific acoustic impedance ratio

Calculate the specific acoustic impedance ratio:

$$Z/\rho c_0 = R/\rho c_0 + jX/\rho c_0 = (1+r)/(1-r) \tag{19}$$

where

R is the real component;

X is the imaginary component;

ρc_0 is the characteristic impedance.

7.10 Determination of the specific acoustic admittance ratio

Calculate the specific acoustic admittance ratio:

$$G\rho c_0 = g\rho c_0 - jb\rho c_0 = \rho c_0 / Z \tag{20}$$

where

g is the real component;

b is the imaginary component.

8 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 E.3.)

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 E) and reference plane definition.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Information concerning the reproducibility and repeatability of these test methods is not available. It is recommended that this information be obtained through interlaboratory comparison tests as soon as possible (see ISO 5725-1 for information).

9 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 part of ISO 10534; 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),
— partition walls normal to the surface in the test object;

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

g) temperature and atmospheric pressure;

h) date of the test;

i) tabular and/or graphical presentation of the test results as a function of frequency; if more than one test specimen of a sample (of the same material) was tested, the individual results or uncertainty shall be indicated, and the mean value shall be given; if different impedance tubes were applied to cover a wide frequency range, the working frequency ranges shall overlap by about one octave and test results shall be indicated separately; it is recommended to present the final, averaged and combined test results in tabular as well as graphical form;

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

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Annex A
(normative)

Preliminary measurements

A.1 Prior to or following each test

A.1.1 Microphone amplitude calibration

Prior to and following each test, the microphone amplitude shall be calibrated to an accuracy 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, will be deemed sufficient provided the microphone is known to have a linear frequency response over the working frequency range.

A.1.2 Temperature measurement

Prior to and following each test, a temperature-measuring device with a measurement accuracy 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 and reported.

A.1.4 Signal-to-noise ratio

Prior to each test, the sound pressure spectrum shall be measured at each microphone position with the noise source on and off. The noise 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 part of ISO 10534. 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 part of ISO 10534.

The attenuation constant, k_0'' is determined in accordance with A.2.1.2, and can be included in the calculations of the reflection coefficient.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

A.2.1.2 Determination of corrections for the tube attenuation

The attenuation can be described analytically through a replacement of the real wave number k_0 by a complex wave number:

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

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

The tube attenuation is best determined experimentally, but at present a reliable procedure for its determination requires knowledge of at least two pressure minima. If this is not possible, then see A.2.1.5. Under some circumstances an approximate experimental method may be employed, and for this see A.2.1.4.

A.2.1.3 Two pressure minima method

The amplitudes of the sound pressure minima $|p(x_{\text{min},n})|$ and of the sound pressure maxima $|p(x_{\text{max},n})|$ when including tube attenuation are given by:

$$\left|p(x_{\text{min},n})\right| = |p_0| \cdot \left| e^{k_0'' x_{\text{min},n}} - |r| \cdot e^{-k_0'' x_{\text{min},n}} \right| \tag{A.2}$$

$$\left|p(x_{\text{max},n})\right| = |p_0| \cdot \left| e^{k_0'' x_{\text{max},n}} + |r| \cdot e^{-k_0'' x_{\text{max},n}} \right| \tag{A.3}$$

The numbering n = 1, 2, 3 ... starts with the left most minimum to the right of the plane of reference and with the first maximum to the right of the minimum number 1.

Defining the standing wave ratio s_n of the n-th minimum and the n-th maximum (n = 1, 2, ...) by:

$$s_n = \frac{\left|p(x_{\text{max},n})\right|}{\left|p(x_{\text{min},n})\right|} = \frac{e^{k_0'' x_{\text{max},n}} + |r| \cdot e^{-k_0'' x_{\text{max},n}}}{e^{k_0'' x_{\text{min},n}} - |r| \cdot e^{-k_0'' x_{\text{min},n}}} \tag{A.4}$$

the magnitude of the reflection factor becomes:

$$|r| = \frac{s_n \cdot e^{k_0'' x_{\text{min},n}} - e^{k_0'' x_{\text{max},n}}}{s_n \cdot e^{-k_0'' x_{\text{min},n}} + e^{-k_0'' x_{\text{max},n}}} \tag{A.5}$$

Since

$$x_{\text{max},n} = x_{\text{min},n} + \lambda_0/4 \tag{A.6}$$

where λ_0 is the wavelength in metres, the magnitude of the reflection factor is in its final form:

$$|r| = e^{2k_0'' x_{\text{min},n}} \cdot \frac{s_n - e^{k_0'' \lambda_0/4}}{s_n + e^{-k_0'' \lambda_0/4}} \tag{A.7}$$

Putting

$$k_0'' x_{\text{min},n} = 4(k_0'' \lambda_0/4) \cdot (x_{\text{min},n}/\lambda_0) \tag{A.8}$$

it is recognized that the quantity $k_0''\lambda_0/4$ is needed for the correction. This quantity is determined in the empty tube with a rigid termination, $|r|$ = 1. Then:

$$\left|p(x_{\text{min},n})\right| = 2\,|p_0| \cdot \sinh(k_0'' x_{\text{min},n}) \tag{A9}$$

$$\left|p(x_{\text{max},n})\right| = 2\,|p_0| \cdot \cosh(k_0'' x_{\text{max},n}) \tag{A10}$$

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

If the pressure amplitudes are measured in the minima numbered n and (n + 1) as well as in the maximum number n between them, and if the quantity Δ_n is defined as:

$$\Delta_n = \frac{\left|p(x_{\min,n+1})\right| - \left|p(x_{\min,n})\right|}{\left|p(x_{\max,n})\right|} \tag{A.11}$$

then:

$$\Delta_n = 2 \sinh (k_0''\lambda_0/4) \tag{A.12}$$

and hence:

$$\frac{k_0''\lambda_0}{4} = \operatorname{arcsinh}\frac{\Delta_n}{2} = \ln\left(\frac{\Delta_n}{2} + \sqrt{\frac{\Delta_n^2}{4}+1}\right) \tag{A.13}$$

which is the required exponent, and for the exponential factor:

$$\mathrm{e}^{\pm k_0''\lambda_0/4} = \left(\frac{\Delta_n}{2} + \sqrt{\frac{\Delta_n^2}{4}+1}\right)^{\pm 1} \tag{A.14}$$

The attenuation constant k_0'' of the tube shall be determined according to these equations after each modification of the tube.

A.2.1.4 Approximate method

For small differences in level (less than 2 dB) between the first and the second minima, and if $x_{\min,1}/\lambda_0$ is not larger than 0,3 m, a correction for the attenuation can be derived by another, approximate, method. This method derives a corrected standing wave ratio s_0 by a straight extrapolation of the minima to the plane x = 0. In contrast to the method given in A.2.1.3, this correction by extrapolation shall be applied for each test object at each frequency.

The fictitious amplitude $|p_0|$ of the first minimum, if it were at the object surface x = 0 (see figure A.1), can be approximated by first defining the corrected standing wave ratio s_0 by

$$s_0 = \frac{\left|p(x_{\max,1})\right|}{\left|p_0\right|} \tag{A.15}$$

and then substituting for s_0 in this equation, taking s_0 from:

$$\frac{1}{s_0} = \frac{1}{s_1} + \frac{2x_{\min,1}}{\lambda_0}\left(\frac{1}{s_1} - \frac{1}{s_2}\right) \tag{A.16}$$

where s_1 and s_2 are the standing wave ratios of the first and the second minima, defined with the pressure maximum $|p(x_{\max,1})|$ between them. This corrected standing wave ratio s_0 is to be applied in the following equation:

$$|r| = \frac{s_0 - 1}{s_0 + 1} \tag{A.17}$$

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

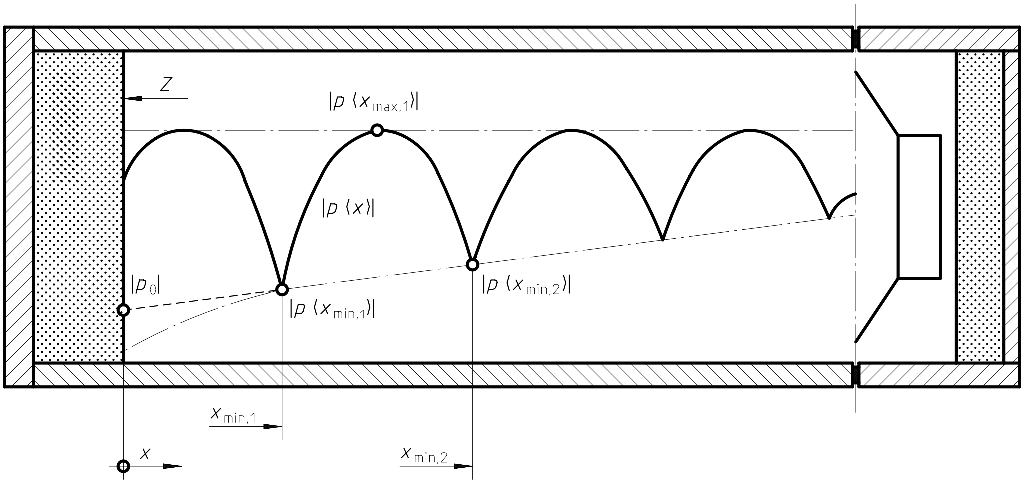


Figure A.1 —Tube attenuation correction

A.2.1.5 Estimation method

For survey measurements, and at the lower end of the working frequency range, if no two pressure minima can be explored with sufficient precision, the attenuation constant can be estimated numerically by:

$$k_0'' = 1{,}94 \times 10^{-2} \sqrt{f} / c_0 d \tag{A.18}$$

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.

If it is not certain that such additional contributions of attenuation do exist, it is recommended to determine the attenuation from equation (A.13) or (A.14) at mid and upper frequencies of the working frequency range and to extrapolate to the lower frequencies.

A.2.2 Determination of the acoustic centre of a microphone

Since the acoustic centre of a side-mounted microphone (or the pick-up opening of a probe tube, or centre-mounted microphone used as described in annex B) may be different from its geometric centre causing erroneous location of the microphone(s) and subsequent errors in acoustic property assignment, the true acoustic centre should be known.

When using probes for which measurements at two successive pressure minima at frequencies over the working frequency range are possible, the procedure outlined in A.2.3.1 shall be followed. Subsequently correction factors for each frequency of the working frequency range shall be deduced and applied to the experimental data, using, for example, a least-squares procedure. This calibration shall be undertaken within the 12-month period prior to a test or following changes to the tube, probe system or microphone.

In the case of side wall and in-tube mounted microphones, at present no acoustic centre calibration procedures have been validated. Such errors shall therefore be considered part of the uncertainties associated with the method given in this part of ISO 10534. 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.

A.2.2.1 Determination of the acoustic centre of a probe microphone

Since the acoustic centre of the pick-up opening of the probe tube or of a microphone may be different from their geometrical centres (especially with measurements of sound pressure minima), the acoustic centre of the probe

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

microphone shall be determined. This is performed at frequencies all over the working frequency range with mutual distances not larger than one-third octave and with the empty test sample holder (rigid termination).

Then the pressure minima are at distances

$$x_{\min,n} = (2n-1)\cdot\lambda_0/4 \tag{A.19}$$

where n = 1,2,...

from the rigid back plate, which is assumed to be at x = 0 (see figure A.2). Let y be the reading for the position of the geometrical centre of the probe and let y = 0 correspond to the rigid termination. If $y_{\min,1}$ and $y_{\min,2}$ are the readings for the positions of the probe microphone when positioned to the first and to the second minima, respectively, then the correction δ, by which the acoustic centre of the probe is to the left (see figure A.2) of the geometrical centre, is given by:

$$\delta = \frac{1}{2}(3y_{\min,1} - y_{\min,2}) \tag{A.20}$$

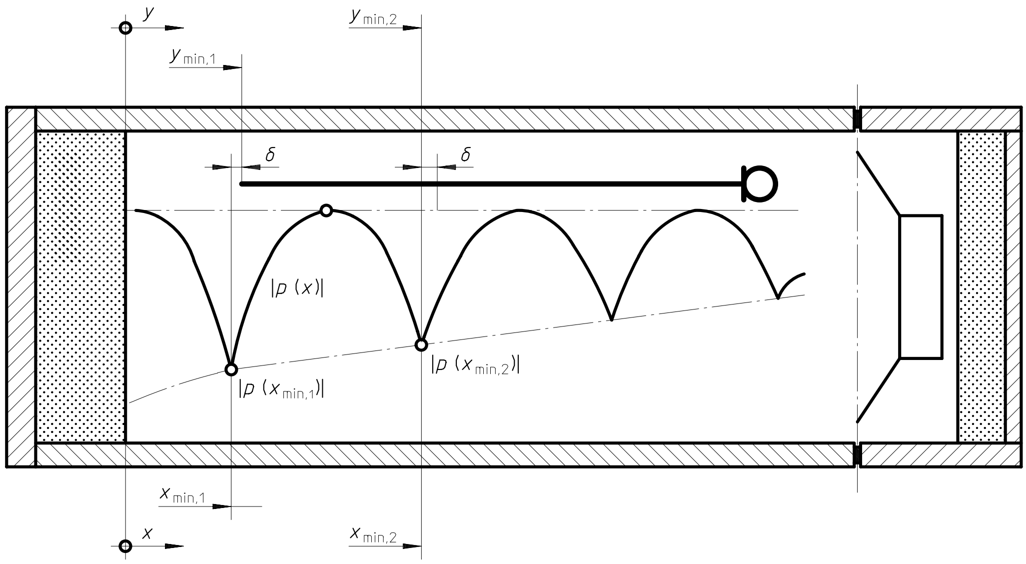


Figure A.2 — Determination of the acoustic centre of a probe microphone

This correction shall be applied to all readings of $y_{\min,n}$ for the evaluation of equations of $x_{\min,n}$:

$$x_{\min,n} = y_{\min,n} - \delta \tag{A.21}$$

NOTE δ will be negative for an acoustical centre to the right.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Annex B
(normative)

Procedure for the one-microphone technique

B.1 General

There are two different ways of applying the one-microphone technique:

1 Fixed microphone locations

2 Variable microphone locations

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 locations is sampled sequentially with a stable sound source. A deterministic signal for the sound source is recommended.

B.2 Fixed microphone locations

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

$$H_{12} = H_{x2}/H_{x1}$$

where x is the generator signal.

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

B.3 Variable microphone locations

This method uses a movable probe microphone, and the sound field at the two locations is sampled in a sequential manner. It is essential that the acoustic centre of the probe tube microphone be determined according to A.2.3.1, and the tube attenuation be included in the computation (see annex A) by replacing the real wave number k_0 by a complex wave number, $k_0' - jk_0''$.

Although the broad-band periodic pseudo-random sequence can also be used in this method, a pure-tone from a stable function generator may be preferred. To achieve high accuracy, it is necessary to choose one microphone location close to the first or second minimum (not necessarily exact), while still respecting the required minimum distance from the test specimen requirement, and the other location shall be a quarter of a wavelength apart. This choice of microphone location is required for each excitation frequency.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Annex C
(normative)

Pressure-release termination of test sample

Sometimes a pressure-release termination of a test object is wanted. This pressure-release termination can be realised by an air gap between the rear surface of the test sample and the rigid termination of the test specimen holder, the depth of which shall be exactly $\lambda_0/4$ for the test frequency. (NB: The sound velocity in the tube must be taken into account.) Therefore, the depth of the air gap shall be carefully adapted for each frequency. For this reason movable rigid plugs are sometimes used as rigid terminations of the test specimen holder. These movable plugs, however, quite often show small leaks at their edges so they are no longer acoustically rigid, and the error introduced by this cannot be taken into account quantitatively.

The reason for the use of pressure-release terminations is often the task of determination of the complex characteristic impedance Z_a and of the complex characteristic propagation constant Γ_a of homogeneous absorber materials such as mineral fibre or foam.

If the layer of this material which is used in the tests has a thickness b, and if the surface impedance of this layer is Z_r (rigid) if the termination is rigid, and if the surface impedance of the layer is Z_s (soft) if the termination is pressure release, then the characteristic impedance and propagation constant of the absorber material are given by:

$$Z_a = \sqrt{Z_r \cdot Z_s}\,; \quad \Gamma_a = \frac{1}{a} \cdot \operatorname{arctanh} \sqrt{\frac{Z_s}{Z_r}} \tag{C.1}$$

The problem of a zero-load impedance can be avoided by using another method. The depth t of the air gap shall not be exactly $\lambda_0/4$, but should be close to this value (see figure. A.2). Then the load impedance of the rear side of the absorber layer is:

$$Z_1 = -\,jZ_0 \cdot \cot k_0 t \tag{C.2}$$

and the characteristic data of the absorber material are obtained from:

$$Z_a = \left[Z_r \cdot Z_s + Z_1 \cdot (Z_s - Z_r)\right]^{1/2}; \quad \Gamma_a = \frac{1}{b} \cdot \operatorname{arctanh} \frac{Z_a}{Z_r} \tag{C.3}$$

where

Z_r is the surface impedance for rigid termination;

Z_s is the surface impedance with the air gap, as before.

The second term under the root indicates the error which is introduced into the first method by a load impedance Z_1 which is not exactly zero. It may become rather large.

In practical applications of the second method, a given depth t of the air gap shall be applied for all those frequencies for which it is approximately an odd multiple of λ_0 and for frequencies which are (about) one-third octave distant on both sides. Hence the whole frequency range can be covered with only a few settings of the depth t. This depth, the sound speed and the frequency should be determined as precisely as possible.

The thickness b of the absorber layer in such measurements of the characteristic constants of the absorber material shall not be too small (three or four tube diameters), otherwise the test sample may be excited to vibrations as a whole, and reading errors of b may become relatively large.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Annex D
(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 2.

The sound pressures of the incident wave p_I and the reflected wave p_R are, respectively:

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

and

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

where

$\hat{p}_I$ and $\hat{p}_R$ are the magnitudes of p_I and p_R at the reference plane ($x = 0$);

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

The sound pressures p_1 and p_2 in the two microphone positions are

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

and

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

The transfer function for the incident wave alone H_I is:

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

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:

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

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

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

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Transposing equation (D.7) to yield r, and using equations (D.5) and (D.6), one has:

$$r = \frac{H_{12} - H_\mathrm{I}}{H_\mathrm{R} - H_{12}} \mathrm{e}^{2jk_0 x_1} \tag{D.8}$$

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.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Annex E
(informative)

Error sources

NOTE Error sources may be considered in two main categories, bias errors and random errors.

E.1 Bias errors

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.2.3.

E.1.1 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 cross-talk 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

$$t \gg 2x_1/c_0 \quad \text{(E.1)}$$

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.

E.1.2 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 part of ISO 10534 and described in 7.5.

E.1.3 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 7.5. 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.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

E.2 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:

$$n = (1/2\,\sigma)^2 \tag{E.2}$$

where

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

σ is the standard error.

E.3 Accuracy of the transfer function

Of particular interest to this part of ISO 10534 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:

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

where

n is the number of averages;

ε is the normalized standard error;

γ^2 is the coherence function.

The coherence function is determined from:

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

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.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Annex F
(informative)

Determination of diffuse sound absorption coefficient α_{st} of locally reacting absorbers from the results of this part of ISO 10534

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 parallel to its surface) from the normalised impedance $z = z' + j \cdot z''$ which is determined according to this part of ISO 10534.

The relationship is:

$$\alpha_{st} = 8 \frac{z'}{z'^2 + z''^2} \left[1 - \frac{z'}{z'^2 + z''^2} \cdot \ln(1 + 2z' + z'^2 + z''^2) + \right.$$
$$\left. + \frac{1}{z''} \cdot \frac{z'^2 - z''^2}{z'^2 + z''^2} \cdot \arctan \frac{z''}{1 + z'} \right] \qquad \text{(F.1)}$$

where

$z = Z/\rho c_0$ the normalized impedance;

$z' = R/\rho c_0$ the real part of the normalized impedance;

$z'' = X\rho c_0$ the imaginary part of the normalized 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).

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Annex G
(informative)

Bibliography

[1] ISO 266:1975, *Acoustics — Preferred frequencies for measurements.*

[2] ISO 354:1985*, Acoustics — Measurement of sound absorption in a reverberation room.*

[3] ISO 5725-1:1994, *Accuracy (trueness and precision) of measurement methods and results — Part 1: General principles and definitions.*

[4] ISO 10534-1:1996, *Acoustics — Determination of sound absorption coefficient and impedance in impedance tubes — Part 1: Method using standing wave ratio.*

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.

ICS 17.140.01

Descriptors: acoustics, acoustic absorption, sound absorbers, tests, acoustic tests, determination, sound absorption coefficient, acoustic measurements.

Price based on 27 pages

Licensed to Universidade Federal de Santa Catarina/Arcanjo Lenzi
ANSI Store order #X309908 Downloaded: 12/9/2006 6:23:25 PM ET
Single user license only. Copying and networking prohibited.