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Acoustics — Measurement of sound absorption in a reverberation room

Acoustique — Mesurage de l'absorption acoustique en salle réverbérante

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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.

International Standards are drafted in accordance with the rules given in the ISO/IEC Directives, Part 2.

The main task of technical committees is to prepare International Standards. 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.

Attention is drawn to the possibility that some of the elements of this document may be the subject of patent rights. ISO shall not be held responsible for identifying any or all such patent rights.

ISO 354 was prepared by Technical Committee ISO/TC 43, *Acoustics*, Subcommittee SC 2, *Building acoustics*.

This second edition cancels and replaces the first edition (ISO 354:1985), which has been technically revised, as follows:

— an integrated impulse response method has been introduced;

— the requirement to measure at least 36 decays has been added;

— mounting conditions according to ISO 354:1985:Amd.1:1997 and mounting conditions Type B and Type J have been introduced.

Introduction

When a sound source operates in an enclosed space, the level to which reverberant sound builds up, and the subsequent decay of reverberant sound when the source is stopped, are governed by the sound-absorbing characteristics of the boundary surfaces, the air filling the space, and objects within the space. In general, the fraction of the incident sound power absorbed at a surface depends upon the angle of incidence. In order to relate the reverberation time of an auditorium, office, workshop, etc., to the noise reduction that would be effected by an absorbing treatment, knowledge of the sound-absorbing characteristics of the surfaces, usually in the form of a suitable average over all angles of incidence, is required. Since the distribution of sound waves in typical enclosures includes a wide and largely unpredictable range of angles, a uniform distribution is taken as the basic condition for the purposes of standardization. If, in addition, the sound intensity is independent of the location within the space, the sound distribution is called a diffuse sound field, and the sounds reaching a room surface are said to be at random incidence.

The sound field in a properly designed reverberation room closely approximates a diffuse field. Hence, sound absorption measured in a reverberation room closely approximates the sound absorption that would be measured under the basic conditions assumed for standardization.

The purpose of this International Standard is to promote uniformity in the methods and conditions of measurement of sound absorption in reverberation rooms.

Acoustics — Measurement of sound absorption in a reverberation room

1 Scope

This International Standard specifies a method of measuring the sound absorption coefficient of acoustical materials used as wall or ceiling treatments, or the equivalent sound absorption area of objects, such as furniture, persons or space absorbers, in a reverberation room. It is not intended to be used for measuring the absorption characteristics of weakly damped resonators.

The results obtained can be used for comparison purposes and for design calculation with respect to room acoustics and noise control.

2 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

ISO 266, *Acoustics — Preferred frequencies*

ISO 9613-1, *Acoustics — Attenuation of sound during propagation outdoors — Part 1: Calculation of the absorption of sound by the atmosphere*

IEC 61260, *Electroacoustics — Octave-band and fractional-octave-band filters*

3 Terms and definitions

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

3.1
decay curve
graphical representation of the decay of the sound pressure level in a room as a function of time after the sound source has stopped

3.2
reverberation time
T
time, in seconds, that would be required for the sound pressure level to decrease by 60 dB after the sound source has stopped

NOTE 1 The definition of T with a decrease by 60 dB of the sound pressure level can be fulfilled by linear extrapolation of shorter evaluation ranges.

NOTE 2 This definition is based on the assumptions that, in the ideal case, there is a linear relationship between the sound pressure level and time, and that the background noise level is sufficiently low.

3.3
interrupted noise method
method of obtaining decay curves by direct recording of the decay of the sound pressure level after exciting a room with broadband or band-limited noise

3.4
integrated impulse response method
method of obtaining decay curves by reverse-time integration of the squared impulse responses

3.5
impulse response
temporal evolution of the sound pressure observed at a point in a room as a result of the emission of a Dirac impulse at another point in the room

NOTE It is impossible in practice to create and radiate true Dirac delta functions, but short transient sounds (e.g. from shots) may offer close enough approximations for practical measurements. An alternative measurement technique, however, is to use a period of maximum-length sequence type signal (MLS) or another deterministic, flat-spectrum signal and to transform the measured response back to an impulse response.

3.6
equivalent sound absorption area of a room
hypothetical area of a totally absorbing surface without diffraction effects which, if it were the only absorbing element in the room, would give the same reverberation time as the room under consideration

NOTE 1 The area is measured in square metres.

NOTE 2 For the empty reverberation room, this quantity is denoted by A_1; for the reverberation room containing the test specimen, it is denoted by A_2.

3.7
equivalent sound absorption area of the test specimen
A_T
difference between the equivalent sound absorption area of the reverberation room with and without the test specimen

NOTE The area is measured in square metres.

3.8
area of the test specimen
S
area of the floor or wall covered by the test specimen

NOTE 1 The area is measured in square metres.

NOTE 2 In the case of a test specimen surrounded by a structure (type E mounting or type J mounting), it is the area enclosed by the structure.

3.9
sound absorption coefficient
α_s
ratio of the equivalent sound absorption area of a test specimen divided by the area of the test specimen

NOTE 1 For absorbers where both sides are exposed, the sound absorption coefficient is the equivalent sound absorption area of the test specimen divided by the area of the two sides of the test specimen.

NOTE 2 The sound absorption coefficient evaluated from reverberation time measurements can have values larger than 1,0 (e.g. because of diffraction effects), and α_s is not, therefore, expressed as a percentage.

NOTE 3 The use of the subscript “s” is to avoid confusion with the sound absorption coefficient defined as the ratio of non-reflected-to-incident sound energy if a plane wave strikes a plane wall at a particular angle of incidence. That “geometric” sound absorption coefficient is always smaller than 1,0 and may therefore be expressed as a percentage.

4 Principle

The average reverberation time in the reverberation room is measured with and without the test specimen mounted. From these reverberation times, the equivalent sound absorption area of the test specimen, A_T, is calculated by using Sabine's equation (see 8.1.2.1).

In the case of a test specimen that uniformly covers a surface (a plane absorber or a specified array of test objects), the sound absorption coefficient is obtained by dividing A_T by the treated surface area S (see 3.8).

When the test specimen comprises several identical objects, the equivalent sound absorption area A_{obj} of an individual object is found by dividing A_T by the number of objects, n:

$$A_{obj} = A_T/n$$

5 Frequency range

Measurements shall be made in one-third-octave bands with the following centre frequencies, in hertz, as specified in ISO 266:

100	125	160	200	250	315
400	500	630	800	1 000	1 250
1 600	2 000	2 500	3 150	4 000	5 000

Additional measurements may be made in one-third-octave bands with centre frequencies specified by ISO 266 outside this range. Especially at low frequencies (below 100 Hz), it could be very difficult to obtain accurate measurement results due to the low modal density of the reverberation room.

6 Test arrangement

6.1 Reverberation room and diffusion of sound field

6.1.1 Volume of reverberation room

The volume of the reverberation room shall be at least 150 m^3. For new constructions, the volume is strongly recommended to be at least 200 m^3. When the volume of the room is greater than about 500 m^3, it may not be possible to measure sound absorption accurately at high frequencies because of air absorption.

6.1.2 Shape of reverberation room

The shape of the reverberation room shall be such that the following condition is fulfilled:

$$l_{max} < 1{,}9 \; V^{1/3} \tag{1}$$

where

l_{max} is the length of the longest straight line which fits within the boundary of the room (e.g. in a rectangular room it is the major diagonal), in metres;

V is the volume of the room, in cubic metres.

In order to achieve a uniform distribution of natural frequencies, especially in the low-frequency bands, no two dimensions of the room shall be in the ratio of small whole numbers.

6.1.3 Diffusion of the sound field

The decaying sound field in the room shall be sufficiently diffuse. In order to achieve satisfactory diffusion whatever the shape of the room, the use of stationary or suspended diffusers or rotating vanes is, in general, required (see Annex A).

6.1.4 Sound absorption area

The equivalent sound absorption area of the empty room, A_1, calculated according to 8.1.2.1, determined in one-third octave bands, shall not exceed the values given in Table 1.

Table 1 — Maximum equivalent sound absorption areas for room volume V = 200 m^3

Frequency, Hz	100	125	160	200	250	315	400	500	630
Equivalent sound absorption area, m^2	6,5	6,5	6,5	6,5	6,5	6,5	6,5	6,5	6,5

Frequency, Hz	800	1 000	1 250	1 600	2 000	2 500	3 150	4 000	5 000
Equivalent sound absorption area, m^2	6,5	7,0	7,5	8,0	9,5	10,5	12,0	13,0	14,0

If the volume V of the room differs from 200 m^3, the values given in Table 1 shall be multiplied by $(V/200\ \text{m}^3)^{2/3}$.

The graph of the equivalent sound absorption area of the empty room versus the frequency shall be a smooth curve and shall have no dips or peaks differing by more than 15 % from the mean of the values of both adjacent one-third-octave bands.

6.2 Test specimens

6.2.1 Plane absorbers

6.2.1.1 The test specimen shall have an area between 10 m^2 and 12 m^2. If the volume V of the room is greater than 200 m^3, the upper limit for the test specimen area shall be increased by the factor $(V/200\ \text{m}^3)^{2/3}$.

The area to be chosen depends on the room volume and on the absorption capability of the test specimen. The larger the room, the larger the test area should be. For specimens with small absorption coefficient, the upper limit area should be chosen.

6.2.1.2 The test specimen shall be of rectangular shape with a ratio of width to length of between 0,7 and 1. It should be placed so that no part of it is closer than 1 m to any edge of the boundary of the room; the distance shall be at least 0,75 m. The edges of the specimen shall preferably not be parallel to the nearest edge of the room. If necessary, heavy test specimens may be mounted vertically along the walls of the room, and directly resting on the floor. In this case, the requirement of at least 0,75 m distance need not be respected.

6.2.1.3 The test specimen shall be installed in one of the mountings specified in Annex B, unless the relevant specifications provided by the producer or the application details provided by the user require a different mounting. The measurement of the reverberation time of the empty room shall be made in the absence of the frame or the side walls of the test specimen except for the barrier around a Type J mounting.

6.2.2 Discrete sound absorbers

6.2.2.1 Rectangular unit sound absorber pads or baffles shall be installed in a Type J mounting as specified in Annex B.

6.2.2.2 Discrete objects (e.g. chairs, free-standing screens or persons) shall be installed for the test in the same manner as they are typically installed in practice. For example, chairs or free-standing screens shall rest on the floor, but they shall not be closer than 1 m to any other boundary. Space absorbers shall be mounted at least 1 m from any boundary or room diffusers and at least 1 m from any microphone. Office screens shall be mounted as individual objects.

6.2.2.3 A test specimen shall comprise a sufficient number of individual objects (in general, at least three) to provide a measurable change in the equivalent sound absorption area of the room greater than 1 m^2, but not more than 12 m^2. If the volume, V, of the room is greater than 200 m^3, these values shall be increased by the factor $(V/200\ m^3)^{2/3}$. Objects normally treated as individual objects shall be arranged randomly, spaced at least 2 m apart. If the test specimen comprises only one object, it shall be tested in at least three locations, at least 2 m apart, and the results shall be averaged.

6.3 Temperature and relative humidity

6.3.1 Changes in temperature and relative humidity during the course of a measurement can have a large effect on the measured reverberation time, especially at high frequencies and at low relative humidities. The changes are described quantitatively in ISO 9613-1.

6.3.2 Measurements should be performed in the empty room and in the room containing the test specimen under conditions of temperature and relative humidity that are almost the same so that the adjustments due to air absorption do not differ significantly. In any case, the relative humidity in the room shall be at least 30 % and max. 90 % and the temperature shall be at least 15 °C during the whole test. For all measurements, the corrections for the change in air absorption as described in 8.1.2.3 shall be applied.

Allow the test specimen to reach equilibrium with respect to temperature and relative humidity in the room before tests are carried out.

7 Measurement of reverberation time

7.1 General

7.1.1 Introduction

Two methods of measuring decay curves are described in this International Standard: the interrupted noise method and the integrated impulse response method. The decay curve measured with the interrupted noise method is the result of a statistical process, and averaging several decay curves or reverberation times measured at one microphone/loudspeaker position is mandatory in order to obtain a suitable repeatability. The integrated impulse response of a room is a deterministic function and not prone to statistical deviations, so no averaging is necessary. However, it requires more sophisticated instrumentation and data processing than the interrupted noise method.

7.1.2 Microphones and microphone positions

The directivity characteristic of the microphones used for the measurement shall be omnidirectional. The measurements shall be made with different microphone positions which are at least 1,5 m apart, 2 m from any sound source and 1 m from any room surface and the test specimen. Decay curves measured at different microphone positions shall not be combined in any way.

7.1.3 Source positions

The sound in the reverberation room shall be generated by a sound source with an omnidirectional radiation pattern. Different sound source positions which are at least 3 m apart shall be used.

7.1.4 Number of microphone and loudspeaker positions

The number of spatially independent measured decay curves shall be at least 12. Therefore the number of microphone positions times the number of sound source positions shall be at least 12. The minimum number of microphone positions shall be three, the minimum number of sound source positions shall be two. It is permissible to use more than one sound source simultaneously provided the difference in the radiated power is within a tolerance band of 3 dB for each one-third-octave band. If more than one sound source is used for excitation simultaneously, the number of spatially independent measured decay curves may be reduced to six.

7.2 Interrupted noise method

7.2.1 Excitation of the room

A loudspeaker source shall be used and the signal fed into the loudspeaker shall be derived from broad-band or band-limited noise having a continuous frequency spectrum. When using broad-band noise and a real-time analyser, the spectrum of the noise used shall be such that the differences in the resulting sound pressure levels in the room shall be less than 6 dB in adjacent one-third-octave bands. When using band-limited noise, the bandwidth shall be at least one-third octave.

The excitation signal shall be sufficiently long to produce a steady-state sound pressure level in all frequency bands of interest before it is switched off. In order to obtain steady-state conditions, the excitation time shall be at least half of the estimate of the expected reverberation time.

The level of the excitation signal before decay shall be sufficiently high that the lower decibel level of the evaluation range is at least 10 dB above the background noise level (see 7.4.1).

If a signal with a bandwidth greater than one-third octave is used, reverberation times of different length in adjacent frequency bands can influence the lower part of the decay curve. If the reverberation times in adjacent bands differ by more than a factor of 1,5, the decay curves for those bands with the shortest reverberation times shall be measured individually using one-third-octave band filtering of the sound source.

7.2.2 Averaging

As explained in 7.1.1, averaging several measurements performed at one microphone/loudspeaker position is mandatory in order to reduce the measurement uncertainty caused by statistical deviations. The number of averages shall be at least three. If the desired repeatability is to be in the same range as the repeatability produced by the integrated impulse response method, the number of averages shall be at least ten (see 8.2). Two averaging methods are possible. The first averaging method is to average the decay curves recorded at one microphone/loudspeaker position using the formula

$$L_p(t) = 10 \lg \left[\frac{1}{N} \sum_{n=1}^{N} 10^{\frac{L_{pn}(t)}{10}} \right] \quad (2)$$

where

$L_p(t)$ is the averaged sound pressure level at the time t calculated for a total number of N decays;

$L_{pn}(t)$ is the sound pressure level of the nth decay at the time t.

This method is generally referred to as “ensemble averaging”. As the second averaging method to be applied in cases where ensemble averaging is not possible, the single decay curves shall be evaluated first and the resulting reverberation times shall be averaged using arithmetic averaging. Decay curves recorded at different microphone/loudspeaker positions shall not be averaged.

NOTE In theory, for laboratory measurements, averaging the reverberation times produces similar results as ensemble averaging. When using computer-controlled devices, however, ensemble averaging should be used in any case.

The decay curve resulting from several averaged decays is normally "smoother" than a single recorded decay. This leads to a more reliable detection of the evaluation range, which is done automatically in most cases.

7.2.3 Recording system

The recording system shall be a level recorder or any other adequate system for determining the average slope of the decay curve of the corresponding reverberation time, including the necessary amplifiers and filters.

The apparatus for recording (and displaying and/or evaluating) the decay in sound pressure level may use

a) exponential averaging, with a continuous curve as output, or

b) exponential averaging, with successive discrete sample points from the continuous average as output, or

c) linear averaging, with successive discrete linear averages as output, in some cases with pauses of considerable duration between determinations of the averages.

The time constant of an exponential averaging device (or approximate equipment, see Note 2) shall be less than, but as close as possible to, $T/20$.

The averaging time of a linear averaging device shall be less than $T/12$.

For apparatus in which the decay record is formed as a succession of discrete points, the time interval between points on the record shall be less than the averaging time of the device ($\leqslant T/12$).

In all cases where the decay record must be evaluated visually, the time scale of the display should be adjusted so that the slope of the record is as close to 45° as possible.

NOTE 1 Commercial level recorders in which the sound pressure level is recorded graphically as a function of time are approximately equivalent to exponential averaging devices.

NOTE 2 When an exponential averaging device is used, there is little advantage in setting the averaging time to very much less than $T/20$. When a linear averaging device is used, there is no advantage in setting the interval between points to very much less than $T/12$. In some sequential measurement procedures, it is feasible to set the averaging time appropriately for each frequency band. In other procedures this is not feasible, and an averaging time or interval chosen as above with reference to the smallest reverberation time is strongly recommended to be used for measurements in all frequency bands.

The one-third-octave band filters included in the receiving equipment shall meet the requirements specified in IEC 61260.

7.3 Integrated impulse response method

7.3.1 Direct method

The impulse response may be measured directly by using an impulse source such as a pistol shot, balloon burst, spark gap or any other sound source that produces an impulse with sufficient bandwidth and energy to meet the requirements of 7.2.1.

NOTE Loudspeakers are usually not suited to produce broadband impulse signals with sufficient energy. It is, however, possible to generate band-filtered pulses. One practice that works well is to feed the loudspeaker system with the time-reversed impulse response of a bandpass filter, i.e. a one-third-octave band filter.

7.3.2 Indirect method

Special sound signals may be used which yield the impulse response only after special processing of the microphone signal. This can provide an improved signal-to-noise ratio. Tone sweeps or pseudo-random noise (e.g. maximum-length sequences) may be used if the requirements for the spectral characteristics of the source are fulfilled. Because of the gain in signal-to-noise ratio, the dynamic requirements of the source may be considerably lower than those set in 7.3.1. If synchronized time averaging is used (e.g. in order to enhance the signal-to-noise ratio), it is necessary to verify that the impulse response remains unchanged during the whole measurement process. The signals may be generated by devices that will consist either of external hard- and software, or devices that form an integrated part of the measuring device.

The bandwidth of the signal shall be greater than one-third octave. The spectrum should be reasonably flat within the actual one-third-octave band to be measured. Alternatively, the broadband noise spectrum may be shaped to provide an approximately pink spectrum in the range covering the one-third-octave bands with mid-band frequencies from 100 Hz to 5 kHz ,with the reverberation time being measured simultaneously in different one-third-octave bands. The test signal shall be such that the resulting decay curve for the respective frequency band meets the level requirements given in 7.2.1.

7.3.3 Recording system

The recording system shall consist of microphones and amplifiers satisfying the requirements given in 7.1.2 and 7.2.3, as well as an additional device that is capable of digitalizing the recorded signal and of performing all the necessary data processing, including the integration of the impulse response and the evaluation of the decay curve. In the case of 7.3.2, the recording system may also contain the necessary hardware and software to process the impulse response from the recorded signal and also to generate the test signal.

The impulse response shall be filtered in one-third-octave bands. The filtering may be processed before or after the digitalization of the impulse response, but in any case before performing the integration. Analog or digital filters may be used. The filters shall conform to IEC 61260.

NOTE The use of special test signals, such as maximum length sequences, not only requires more sophisticated data processing, but also a deeper knowledge of the theoretical background in order to obtain proper results. Since a detailed introduction to this technique is outside scope of this International Standard, the user should refer to appropriate literature.

7.3.4 Integration of the impulse response

The filtered impulse response shall be backward integrated. The result is theoretically equivalent to an infinite number of averaged decays obtained by the interrupted noise method. As there are several commercial systems available that offer the backward integration as an integrated feature, it will normally not be necessary for the user to program the integration. The basic procedure is the following.

Generate for each frequency band the decay curve by a backward integration of the squared impulse response. In an ideal situation with no background noise, the integration would start at the end of the impulse response ($t \rightarrow \infty$) and proceed to the beginning of the squared impulse response. Thus the decay as a function of time is

$$E(t) = \int_0^\infty p^2(\tau)\mathrm{d}\tau - \int_0^t p^2(\tau)\mathrm{d}\tau = \int_t^\infty p^2(\tau)\mathrm{d}\tau = \int_\infty^t p^2(\tau)\mathrm{d}(-\tau) \quad (3)$$

where

$E(t)$ is the backward integrated squared impulse response;

$p(\tau)$ is the sound pressure impulse response.

In order to minimize the influence of the background noise on the later part of the impulse response, use the following technique for implementation.

If the level of the background noise is known, determine the starting point of the integration t_1 as the intersection between a horizontal line through the background noise and a sloping line through a representative part of the squared impulse response decay curve. Continue the backward integration up to the beginning of the impulse response, and calculate the decay curve from

$$E(t) = \int_{t_1}^{t} p^2(\tau)\mathrm{d}(-\tau) + C \tag{4}$$

where $(t < t_1)$ and C is an optional correction for the integrated squared impulse response between t_1 and infinity. The most reliable result is obtained when C is calculated under the assumption of an exponential decay of energy with the same rate as given by the squared impulse response between t_0 and t_1, where t_0 is the time corresponding to a level 10 dB higher than the level at t_1.

If C is set to zero, the finite starting point of the integration causes a systematic underestimation of the reverberation time. For a maximum underestimation of the reverberation time of 5 %, the backward integration shall begin at a level below the maximum level of the squared impulse response, which is at least 15 dB plus the dynamic range over which T is to be assessed.

7.4 Evaluation of reverberation times based on decay curves

7.4.1 Evaluation range

The evaluation of the decay curve for each frequency band specified in Clause 5 shall start at 5 dB below the initial sound pressure level. The evaluation range shall be 20 dB. The bottom of the evaluation range shall be at least 10 dB above the overall background noise of the measuring system.

7.4.2 Evaluation method

When using a computer-controlled recording system, the calculation of a least-squares-fit line over the evaluation range is a convenient method for the determination of the reverberation time. Other algorithms may be used that provide similar results. When using a direct plot from a level recorder, a straight line shall be fitted manually as closely as possible to the decay curve. In the case of evaluation of discrete points, the number of points shall be sufficient for applying, for example, a least-squares-fit line algorithm.

8 Expression of results

8.1 Method of calculation

8.1.1 Calculation of reverberation times T_1 and T_2

The reverberation time of the room in each frequency band is expressed by the arithmetic mean of the total number of reverberation time measurements made in that frequency band.

The mean reverberation times of the room in each frequency band without and with the test specimen, T_1 and T_2 respectively, shall be calculated and expressed using at least two decimal places.

8.1.2 Calculation of A_1, A_2 and A_T

8.1.2.1 The equivalent sound absorption area of the empty reverberation room, A_1, in square metres, shall be calculated using the formula

$$A_1 = \frac{55{,}3\,V}{cT_1} - 4\,Vm_1 \tag{5}$$

where

V is the volume, in cubic metres, of the empty reverberation room;

c is the propagation speed of sound in air, in metres per second;

T_1 is the reverberation time, in seconds, of the empty reverberation room;

m_1 is the power attenuation coefficient, in reciprocal metres, calculated according to ISO 9613-1 using the climatic conditions that have been present in the empty reverberation room during the measurement. The value of m can be calculated from the attenuation coefficient, α, which is used in ISO 9613-1 according to the formula

$$m = \frac{\alpha}{10\,\lg(\mathrm{e})}$$

NOTE For temperatures in the range of 15 °C to 30 °C, c can be calculated from the formula

$$c = (331 + 0{,}6t/°\mathrm{C})\ \mathrm{m/s} \qquad (6)$$

where t is the air temperature, in degrees Celsius.

8.1.2.2 The equivalent sound absorption area of the reverberation room containing a test specimen, A_2, in square metres, shall be calculated using the formula

$$A_2 = \frac{55{,}3\,V}{c\,T_2} - 4V\,m_2 \qquad (7)$$

where

c and V have the same meanings as in 8.1.2.1;

T_2 is the reverberation time, in seconds, of the reverberation room after the test specimen has been introduced;

m_2 is the power attenuation coefficient, in reciprocal metres, calculated according to ISO 9613-1 using the climatic conditions that have been present in the empty reverberation room during the measurement. The value of m can be calculated from the attenuation coefficient, α, which is used in ISO 9613-1 according to the formula

$$m = \frac{\alpha}{10\,\lg(\mathrm{e})}$$

8.1.2.3 The equivalent sound absorption area of the test speciment, A_T, in square metres, shall be calculated using the formula

$$A_T = A_2 - A_1 = 55{,}3V\left(\frac{1}{c_2T_2} - \frac{1}{c_1T_1}\right) - 4V(m_2 - m_1) \qquad (8)$$

where

c_1 is the propagation speed of sound in air at the temperature t_1;

c_2 is the propagation speed of sound in air at the temperature t_2;

A_1, V, T_1 and m_1 have the same meanings as in 8.1.2.1;

A_2, T_2 and m_2 have the same meanings as in 8.1.2.2.

8.1.3 Calculation of α_s

The sound absorption coefficient α_s of a plane absorber or a specified array of test objects shall be calculated using the formula

$$\alpha_s = \frac{A_T}{S} \quad (9)$$

where

A_T is the equivalent sound absorption area of the test specimen, in square metres, calculated in accordance with 8.1.2.3;

S is the area, in square metres, covered by the test specimen (see 3.8).

8.1.4 Calculation of equivalent sound absorption area of discrete absorbers

For discrete absorbers, the result is generally expressed as the equivalent sound absorption area per object, which is determined by dividing A_T by the number of objects tested.

For a specified array of objects, the result is given as the sound absorption coefficient.

8.2 Precision

8.2.1 General

The overall measurement uncertainty of absorption coefficients is influenced by two effects. The first is the uncertainty of the measured reverberation times. This effect is particularly important when the interrupted noise method is used (see 8.2.2). The second factor causing uncertainty is described by reproducibility limits. It is caused by the complete measurement set-up including the reverberation room and the mounting method. Variations due to the laboratory set-up are being investigated in interlaboratory tests (see 8.2.3).

8.2.2 Repeatability of measured reverberation times

The relative standard deviation of the reverberation time T_{20}, evaluated over a 20 dB decay range, can be estimated by the following formula (see ISO/TR 140-13 for details):

$$\varepsilon_{20}(T)/T = \sqrt{\frac{2{,}42 + 3{,}59/N}{f\,T}} \quad (10)$$

$\varepsilon_{20}(T)$ is the standard deviation of the reverberation time T_{20};

T is the reverberation time measured;

f is the centre frequency of the one-third-octave band;

N is the number of decay curves evaluated.

An example of the standard deviation of measurement of T_{20} at 12 positions with 3 repetitions of decay registration at each position is illustrated in Figure 1.

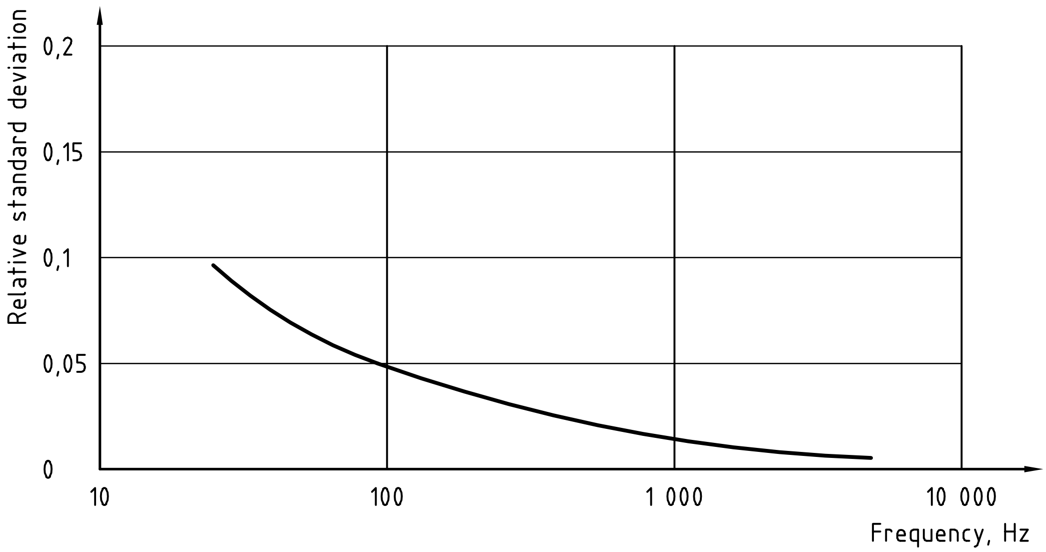


Figure 1 — Example of the standard deviation

8.2.3 Reproducibility

The reproducibility of absorption coefficient measurement is still under investigation.

8.3 Presentation of results

For all frequencies of measurement, the following results shall be reported, presented in the form of a table and as a graph:

a) for plane absorbers, the sound absorption coefficient, α_s;

b) for single objects, the equivalent sound absorption area per object, A_{obj};

c) for a specified array of objects, the sound absorption coefficient, α_s.

The equivalent sound absorption area of a test specimen shall be rounded to 0,1 m^2 and the sound absorption coefficient to 0,01.

NOTE It should be noted that the precision of the results may be less than the above decimal rounding limits might imply.

In the graphical presentation, the points of measurement shall be connected by straight lines, the x-axis giving the frequency on a logarithmic scale and the y-axis showing the equivalent sound absorption area or sound absorption coefficient on a linear scale. The ratio of the y-axis distance from $A_T = 0$ to $A_T = 10$ m^2, or from $\alpha_s = 0$ to $\alpha_s = 1$, to the x-axis distance corresponding to 5 octaves, shall be 2:3. For measuring results with $A_T \leqslant 3$ m^2, a y-axis distance from $A_T = 0$ to $A_T = 5$ m^2 can be chosen.

In addition, a single-number rating calculated according to ISO 11654 may be included. As specified in ISO 11654, octave-band values are deduced by determining the arithmetic mean of the three one-third-octave sound absorption coefficients within the octave.

9 Test report

The test report shall make reference to this International Standard and shall include the following information:

a) the name of the organization that performed the test;

b) the date of test;

c) the description of the test specimen, the test area of the test specimen, S, and its mounting and position in the reverberation room, preferably by means of drawings;

d) the shape of the reverberation room, its diffusion treatment (the number and size of diffusers) and the number of microphone and sound source positions;

e) the dimensions of the reverberation room, its volume, V, and its total surface area (walls, floor and ceiling), S_t;

f) the temperature and relative humidity during measurements of T_1 and T_2;

g) the mean reverberation times T_1 and T_2, at each frequency;

h) the test results, reported in accordance with 8.3.

Annex A
(normative)

Diffusivity of the sound field in the reverberation room

A.1 Diffusers

An acceptable diffusivity can be achieved by using fixed diffusers and/or rotating vanes. Ideally, these diffusing elements should be sheets with low sound absorption and with a mass per unit area of about 5 kg/m^2. Diffusers of different sizes, ranging from approximately 0,8 m^2 to 3 m^2 in area (for one side) are recommended. The sheets may be slightly curved and shall be oriented at random and positioned throughout the room.

If rotating vanes are used, the decay repetition frequency and frequency of rotation of the vane shall not be in the ratio of small whole numbers.

A.2 Check of diffusivity

Select a suitable test specimen, i.e. a sample 5 cm to 10 cm thick, of homogeneous, porous absorbing material which, under optimum conditions, has a sound absorption coefficient greater than 0,9 over the frequency range from 500 Hz to 4 000 Hz. (Certain glass-wools, rock-wools or polyurethane foams meet this criterion.)

Mount the test specimen in accordance with 6.2.

Perform sound absorption measurements on the test specimen as follows:

a) with no diffusers;

b) with a small number of stationary diffusers (approximately 5 m^2 in area); and

c) with increasing quantities of stationary diffusers, in steps of approximately 5 m^2 in area.

For each set of measurements, calculate the mean value of the sound absorption coefficients, in the range from 500 Hz to 5 000 Hz, and plot these values against the number (total area) of diffusers used in each case.

It will be seen that the mean sound absorption coefficient approaches a maximum and thereafter remains constant with increasing numbers (area) of diffusers.

The optimum number (area) of diffusers is that at which this constant value is attained.

If rotating vanes are used, the resulting diffusion shall be proved to be equivalent to that achieved by the procedure described above.

NOTE From experience it has been found that, in rectangular rooms, the area (both sides) of diffusers required to achieve satisfactory diffusion is approximately 15 % to 25 % of the total surface area of the room.

Annex B
(normative)

Test specimen mountings for sound absorption tests

B.1 General

The sound-absorption properties of a material depend on how that material is mounted during a test. This annex specifies several different standard mountings that shall be used during a test for sound absorption. Normally a test specimen is tested using only one of the specified mountings.

Designations for Type E and Type G mountings include a numerical suffix, for example, E-400 or G-100. The suffix is equal to a distance characteristic of the mounting in millimetres, rounded off to the nearest 5 mm.

NOTE Where applicable, the designations used for each type of mounting have been chosen to match those used in a standard that already existed when this annex was written, ASTM E 795, *Standard Practices for Mounting Test Specimens During Sound Absorption Tests*.

B.2 Type A mounting

The test specimen is mounted or placed directly against a room surface, such as the floor of the reverberation room. Adhesives or mechanical fasteners that do not leave a thin air space may be used to hold the test specimen in place during the test, if required. A complete description of the fasteners and their location or the method of surface preparation and the adhesive used to retain the specimen shall be included in the test report.

If two or more pieces of material (or separate panels) are butted together to form the test specimen, it may be necessary to cover the joints between the adjacent pieces with tape, caulking compound, or other material that is not sound absorbing. The reason for covering the joints is to prevent the side edges of the individual pieces from absorbing sound. If the joints are covered, the test report shall describe the method and material used.

The perimeter edge of the test specimen shall be sealed or covered to prevent the edges from absorbing sound. If the edges of the test specimen are exposed when the material is normally installed in an actual application, then the edges of the test specimen shall not be sealed or covered during a test. If the edges are not covered, the area of the edges shall be included in calculating the test specimen area.

The treatment of the edges of the test specimen shall be described in the test report. If the area of the edges was included in the calculation of test specimen area, this shall be noted in the test report.

The perimeter edges of the test specimen may be sealed or covered with an acoustically reflective frame. The frame shall be solid, not hollow, and shall have no air space between the test specimen and the frame and between the room surface and the frame. A frame of 1,0 mm thick steel, 12,5 mm thick gypsum board or 12,5 mm wood (minimum thicknesses) may be used. The frame shall be tightly butted to the specimen and sealed to the room surface. The exposed face of the frame shall be flush with the surface of the specimen.

If a perforated, expanded metal, or other open facing material is used over the test specimen, a complete description of this facing material shall be given in the test report.

B.3 Type B mounting

This mounting is used for products that are glued directly to a hard surface with an acoustic panel adhesive, the application of which normally leaves a thin airspace between the product and the surface to which it is adhered.

Adhere the test specimen to a gypsum board laid directly against the room surface. The thickness of the gypsum board is not critical. Apply the adhesive in accordance with the manufacturer's specification. If there are no instructions, apply four dabs of adhesive to the back of each piece of the test specimen. In order to secure the airspace, shims of 3 mm thickness of size 25 mm by 25 mm shall be located at the four corners of each piece of the test specimen. The perimeter edges of the test specimen shall be sealed or covered with an acoustically reflective frame. The frame shall be solid, not hollow, and shall have no air space between the test specimen and the frame or between the room surface and the frame. A frame of 1,0 mm steel, 12,5 mm gypsum board or 12,5 mm wood (minimum thicknesses) may be used. The frame shall be tightly butted to the test specimen and sealed to the room surface. The exposed face of the frame shall be flush with the surface of the test specimen.

B.4 Type E mounting

The test specimen is mounted with an airspace behind it. The suffix of the designation (e.g. Type E-400) shall be the distance rounded off to the nearest integral multiple of 5 mm between the exposed face of the test specimen and the room surface behind the specimen. If a Type E mounting is used, the specimen shall be tested in either an E-400, E-300 or E-200 configuration. Other air spaces may be used in addition to the 200 mm, 300 mm or 400 mm distances.

The mounting fixture shall be constructed of metal, wood or other non-porous material with a surface density of at least 20 kg m^2, and shall enclose an air space behind the sample that does not have any interior partitions unless provided as part of the sample. The joint between the fixture and the room surface shall be sealed to prevent air leaks between the enclosed space and the outside. The fixture shall cover the perimeter edges of the test specimen. The joints between the fixture and the room surface and between the fixture and the test specimen shall be sealed to prevent air leakage between the enclosed space and the outside.

The Type E mounting may be placed on the floor of the room with the test specimen pointing upwards, unless the construction of the sample will affect the sound absorption due to gravity effects.

B.5 Type G mounting

The test specimen, such as a curtain, drapery, window shade or window blind, is hung parallel to the room surface. The suffix of the mounting designation (e.g. Type G-100) shall be the distance from the face of the test specimen to the room surface. If a Type G mounting is used, the specimen shall be tested in the G-100 configuration. Other air spaces may be used in addition to the 100 mm distance.

If another distance is used, it shall be an integral multiple of 50 mm. The specimen may be tested with or without a perimeter frame, depending on how it is used in practice. If a perimeter frame is used, it shall be butted against the specimen and sealed to the room surface.

Other curtain arrangements are possible and may be tested. The test report shall describe the specific arrangement in detail.

B.6 Type I mounting

This mounting is used for spray- or trowel-applied materials, such as plaster. The material shall be applied to a suitable substrate. Care shall be taken to prevent distortion of the substrate while the applied material is curing. The test specimen shall be tested in a Type A mounting including a frame around the test specimen.

B.7 Type J mounting

This mounting shall be used for the general specification of the sound absorption per unit of rectangular unit sound absorber pads or baffles. The sound absorber pads or baffles shall be mounted with one edge resting on or touching a room surface. Optional mountings with ground clearance may be used. There shall not be an air space between the baffle edge and the room surface. The treated surface of the floor shall be between 10 m^2 and 15 m^2.

The baffles shall be arranged in two or three parallel rows. There shall not be an airspace between the single baffles in a row. The shortest distance from any baffle to a room surface other than the surface which the baffles are touching shall be at least 1 m, unless these surfaces are a part of the barrier.

The array of baffles or pads shall be surrounded by a non-absorptive barrier. One or two walls of the reverberation room may be used as part of the barrier, as shown in Figures B.1 and B.2 respectively. The part of the barrier parallel to the absorptive area of the baffles or pads shall be $d/2$ from the centreline of the nearest row of baffles or pads, where d is the distance between the parallel rows. The part of the barrier perpendicular to the rows of baffles or pads shall be flush with the ends of the baffles or pads. For the height of the barrier, the following two designs are possible.

a) Well approach:

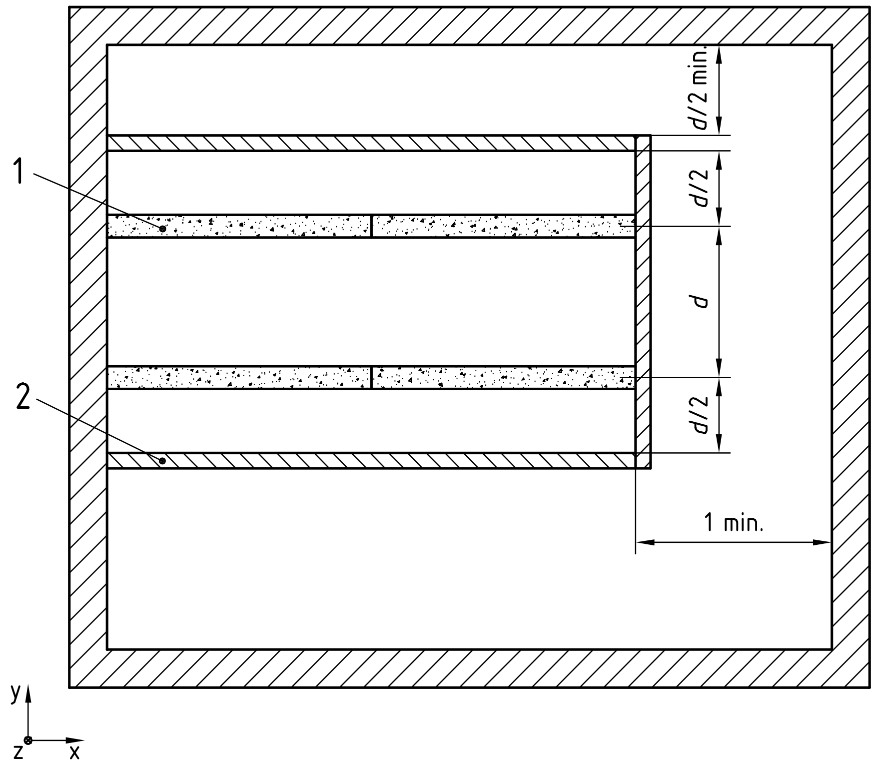
   The height of the barrier shall be the same as the height of the baffles or pads, as shown in Figure B.3.

b) Deep well approach:

   The barrier shall be 0,8 m higher than the baffles or pads, but the height of the barrier shall not exceed half of the height of the reverberation room, as shown in Figure B.4.

The barrier shall not be removed from the room for the empty room measurements.

Dimensions in metres



Key

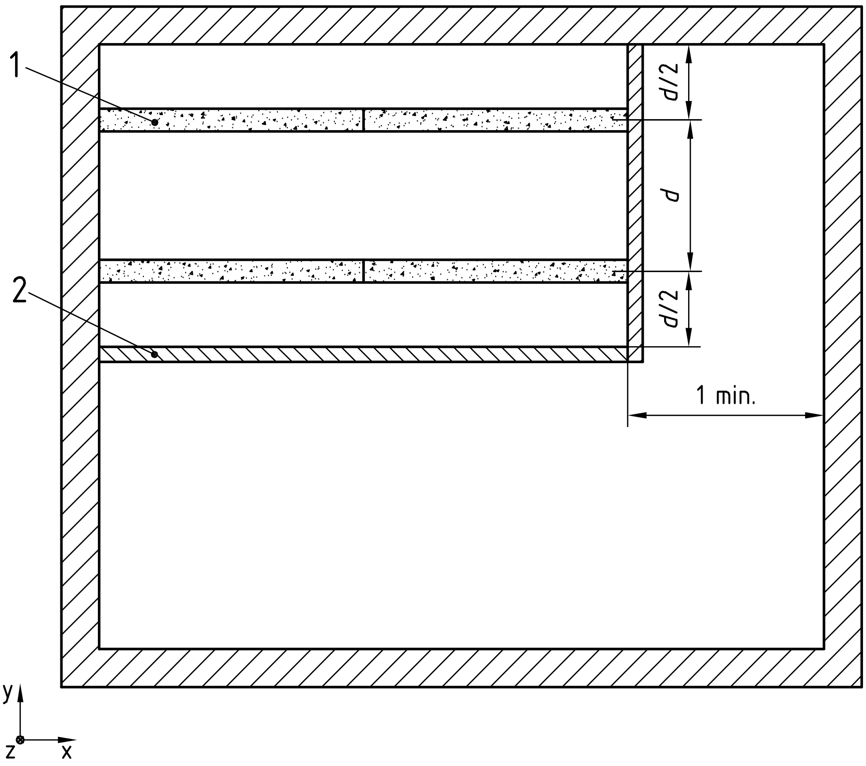
1 baffles

2 barrier

d is the distance between the parallel rows

Figure B.1 — Example of a Type J mounting using surrounding non-absorptive barrier (top view)

Dimensions in metres



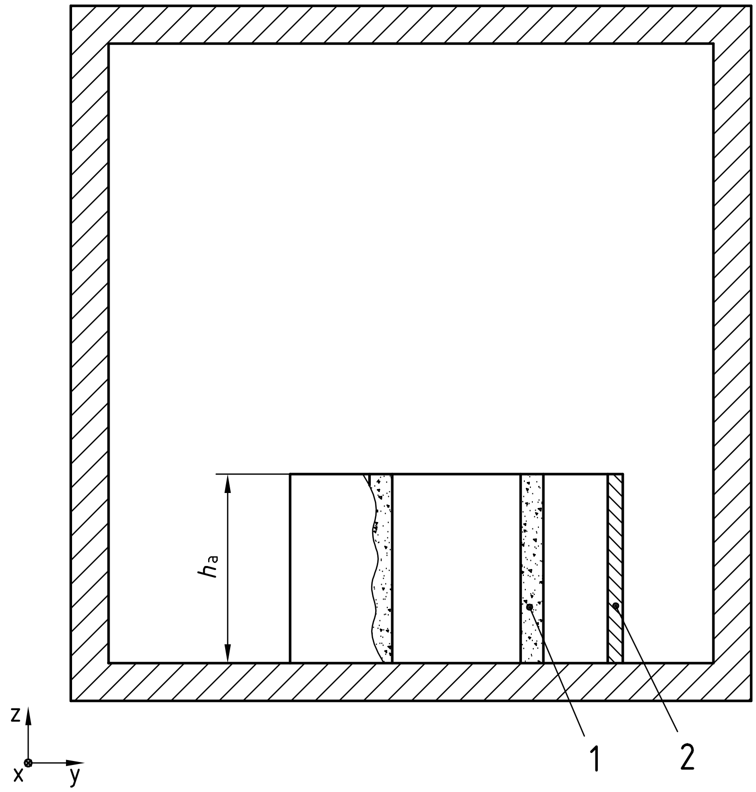
Key

1 baffles

2 barrier

d is the distance between the parallel rows

Figure B.2 — Example of a Type J mounting using surrounding non-absorptive barrier (top view)



Key

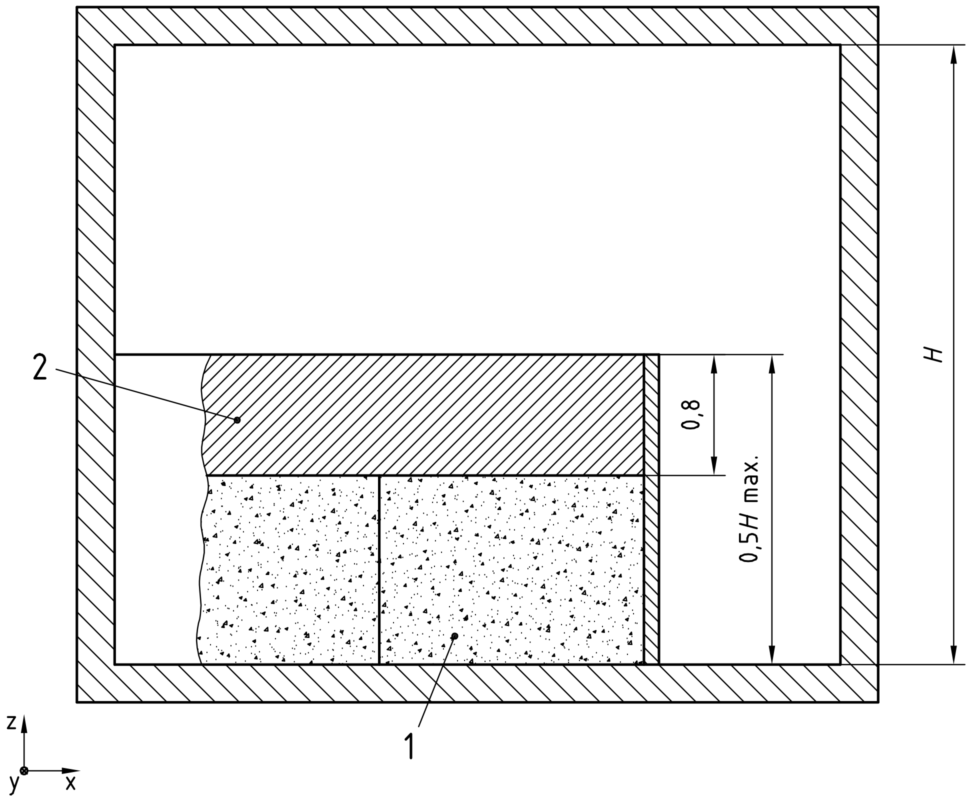
1 baffles

2 barrier

h_a is the height of the absorber

Figure B.3 — Example of a Type J mounting, “well approach”

Dimensions in metres



Key

1 baffle

2 barrier

H is the height of the reverberation room

Figure B.4 — Example of a Type J mounting, "deep well approach"

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