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Acoustics — Measurement of room acoustic parameters —

Part 1:
Performance spaces

Acoustique — Mesurage des paramètres acoustiques des salles —

Partie 1: Salles de spectacles

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Foreword

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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 3382-1 was prepared by Technical Committee ISO/TC 43, *Acoustics*, Subcommittee SC 2, *Building acoustics*.

This first edition of ISO 3382-1, together with ISO 3382-2 and ISO 3382-3, cancels and replaces ISO 3382:1997, of which it constitutes a technical revision. Annex A has been extended with information on JND (just noticeable difference), recommended frequency averaging and by the addition of a new parameter for LEV (listener envelopment). A new Annex C has been added with parameters for the acoustic conditions on the orchestra platform.

ISO 3382 consists of the following parts, under the general title *Acoustics — Measurement of room acoustic parameters*:

— *Part 1: Performance spaces*

— *Part 2: Reverberation time in ordinary rooms*

Open plan spaces are to form the subject of a future part 3.

Introduction

The reverberation time of a room was once regarded as the predominant indicator of its acoustical properties. While reverberation time continues to be regarded as a significant parameter, there is reasonable agreement that other types of measurements, such as relative sound pressure levels, early/late energy ratios, lateral energy fractions, interaural cross-correlation functions and background noise levels, are needed for a more complete evaluation of the acoustical quality of rooms.

This part of ISO 3382 establishes a method for obtaining reverberation times from impulse responses and from interrupted noise. The annexes introduce the concepts and details of measurement procedures for some of the newer measures, but these do not constitute a part of the formal specifications of this part of ISO 3382. The intention is to make it possible to compare reverberation time measurements with higher certainty and to promote the use of and consensus in measurement of the newer measures.

Annex A presents measures based on squared impulse responses: a further measure of reverberation (early decay time) and measures of relative sound levels, early/late energy fractions and lateral energy fractions in auditoria. Within these categories, there is still work to be done in determining which measures are the most suitable to standardize upon; however, since they are all derivable from impulse responses, it is appropriate to introduce the impulse response as the basis for standard measurements. Annex B introduces binaural measurements and the head and torso simulators (dummy heads) required to make binaural measurements in auditoria. Annex C introduces the support measures that have been found useful for evaluating the acoustic conditions from the musicians' point of view.

Acoustics — Measurement of room acoustic parameters —

Part 1:
Performance spaces

1 Scope

This part of ISO 3382 specifies methods for the measurement of reverberation time and other room acoustical parameters in performance spaces. It describes the measurement procedure, the apparatus needed, the coverage required, and the method of evaluating the data and presenting the test report. It is intended for the application of modern digital measuring techniques and for the evaluation of room acoustical parameters derived from impulse responses.

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.

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

IEC 61672-1, *Electroacoustics — Sound level meters — Part 1: Specifications*

3 Terms and definitions

For the purposes of this part of ISO 3382, 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

[ISO 354:2003, 3.1]

NOTE 1 It is possible to measure this decay either after the actual cut-off of a continuous sound source in the room or derived from the reverse-time integrated squared impulse response of the room (see Clause 5).

NOTE 2 The decay directly obtained after non-continuous excitation of a room (e.g. by recording a gunshot with a level recorder) is not recommended for accurate evaluation of the reverberation time. This method ought only be used for survey purposes. The decay of the impulse response in a room is in general not a simple exponential decay, and thus the slope is different from that of the integrated impulse response.

3.2
interrupted noise method
method of obtaining decay curves by direct recording of the decay of sound pressure level after exciting a room with broadband or band limited noise

[ISO 354:2003, 3.3]

3.3
integrated impulse response method
method of obtaining decay curves by reverse-time integration of the squared impulse responses

[ISO 354:2003, 3.4]

3.4
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

[ISO 354:2003, 3.5]

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

3.5
reverberation time
T
⟨room acoustic parameters⟩ duration required for the space-averaged sound energy density in an enclosure to decrease by 60 dB after the source emission has stopped

NOTE 1 The reverberation time is expressed in seconds.

NOTE 2 T can be evaluated based on a smaller dynamic range than 60 dB and extrapolated to a decay time of 60 dB. It is then labelled accordingly. Thus, if T is derived from the time at which the decay curve first reaches 5 dB and 25 dB below the initial level, it is labelled T_{20}. If decay values of 5 dB to 35 dB below the initial level are used, it is labelled T_{30}.

3.6 States of occupancy

3.6.1
unoccupied state
state of a room prepared for use and ready for speakers or for performers and audience, but without these persons being present, and in the case of concert halls and opera houses, preferably with the performers' chairs, music stands and percussion instruments, etc.

3.6.2
studio state
⟨rooms for speech and music⟩ state of a room occupied by performers or speakers only and without an audience (for example, during rehearsals or sound recordings) and with the number of performers and other persons such as technicians corresponding to the usual number

3.6.3
occupied state
state of an auditorium or theatre when 80 % to 100 % of the seats are occupied

NOTE Reverberation time measured in a room will be influenced by the number of people present and the above states of occupancy are defined for measurement purposes.

4 Measurement conditions

4.1 General

The measurements of reverberation time may be made with the room in any or all states of occupancy. Where the room has adjustable components for providing variable acoustical conditions, it can be relevant to carry out separate measurements with these components in each of their normal settings. The temperature and relative humidity of the air in the room should be measured to an accuracy of ± 1 °C and ± 5 %, respectively.

An accurate description of the state of occupancy of the room is of decisive importance in assessing the results obtained by measuring the reverberation time. Extraordinary occupancies (such as that which would be created in a concert hall by a larger than usual orchestra or the additional presence of a choir or standees) shall be noted with the results.

In theatres, a distinction shall be made between “safety curtain up” and “safety curtain down”, between “orchestra pit open” and “orchestra pit closed”, and also between “orchestra seated on the stage”, with and without concert enclosure. In all of these cases, measurement can be useful. If the safety curtain is up, the amount of furnishing of the stage is of importance and shall be described.

Where variable components involve active (i.e. electronic) techniques, the effects of these should be measured, too, but as certain types of electronic reverberation enhancement systems create non-time-stationary conditions in the room, a unique impulse response will not exist and caution should be exercised in using synchronous averaging during the course of making measurements.

4.2 Equipment

4.2.1 Sound source

The sound source shall be as close to omnidirectional as possible (see Table 1). It shall produce a sound pressure level sufficient to provide decay curves with the required minimum dynamic range, without contamination by background noise. In the case of measurements of impulse responses using pseudo-random sequences, the required sound pressure level might be quite low because a strong improvement of the signal-to-noise ratio by means of synchronous averaging is possible. In the case of measurements which do not use a synchronous averaging (or other) technique to augment the decay range, a source level will be required that gives at least 45 dB above the background level in the corresponding frequency band. If only T_{20} is to be measured, it is sufficient to create a level at least 35 dB above the background level.

Table 1 lists the maximum acceptable deviations from omnidirectionality when averaged over “gliding” 30° arcs in a free sound field. In case a turntable cannot be used, measurements per 5° should be performed, followed by “gliding” averages, each covering six neighbouring points. The reference value shall be determined from a 360° energetic average in the measurement plane. The minimum distance between source and microphone shall be 1,5 m during these measurements.

Table 1 — Maximum deviation of directivity of source in decibels for excitation with octave bands of pink noise and measured in free field

Frequency, hertz	125	250	500	1 000	2 000	4 000
Maximum deviation, decibels	± 1	± 1	± 1	± 3	± 5	± 6

4.2.2 Microphones, recording and analysis equipment

4.2.2.1 General

Omnidirectional microphones shall be used to detect the sound pressure and the output may be taken either

— directly to an amplifier, filter set and a system for displaying decay curves or analysis equipment for deriving the impulse responses, or

— to a signal recorder for later analysis.

4.2.2.2 Microphone and filters

The measurement equipment shall meet the requirements of a type 1 sound level meter according to IEC 61672-1. The octave or one-third-octave filters shall conform with IEC 61260. The microphone should be as small as possible and preferably have a maximum diaphragm diameter of 13 mm. Microphones with diameters up to 26 mm are allowed, if they are of the pressure response type or of the free field response type but supplied with a random incidence corrector yielding a flat frequency response at random incidence.

4.2.2.3 Recording device

If the sound decay is initially recorded on magnetic tape or a digital recording device, automatic gain control or other circuits for dynamic optimization of signal-to-noise ratio shall not be used. The recording time of each decay shall be sufficiently long to enable determination of the final background level following the decay; five seconds plus the expected reverberation time is recommended as a minimum.

The recording device shall have the following characteristics for the particular combination of record and playback speeds used.

a) The frequency response shall be flat over the frequency range of measurement with a smaller tolerance than ± 3 dB.

b) The dynamic range shall be sufficient to allow the required minimum decay curve range. In the case of interrupted noise decays, the recorder shall be capable of providing a signal-to-noise ratio of at least 50 dB in every frequency band concerned.

c) The ratio of the playback speed to the record speed shall be within ± 2 % of $10^{0,1 \times n}$, where n is an integer including zero.

NOTE If speed translation is used on playback, the corresponding frequency translation will then be a whole number of standard one-third-octave band spacings or, if n is a multiple of three, of octave band spacings.

Where a tape recorder is used, then in respect of the speed of response of the apparatus for forming a record of the decay of sound pressure level with time (see 4.2.2.4), T refers to the effective reverberation time of the signal being played back. This will differ from the true reverberation time of the enclosure only if the playback speed differs from the record speed.

When the decay has been recorded for replay through filters and an integrating device, it can be beneficial to time-reverse the responses during replay (see Reference [10]).

4.2.2.4 Apparatus for forming decay record of level

The apparatus for forming (and displaying and/or evaluating) the decay record shall use any of the following:

a) exponential averaging, with continuous curve as output;

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

c) linear averaging, with successive discrete linear averages as output (in some cases, with small pauses between performance of averages).

The averaging time, i.e. time constant of an exponential averaging device (or appropriate equivalent), shall be less than, but as close as possible to, $T/30$. Similarly, the averaging time of a linear averaging device shall be less than $T/12$. Here T is the reverberation time being measured or, if appropriate, the effective reverberation time as described in the penultimate paragraph of 4.2.2.3.

In apparatus where the decay record is formed as a succession of discrete points, the time interval between points on the record shall be less than 1,5 times the averaging time of the device.

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

NOTE 1 The averaging time of an exponential averaging device is equal to 4,34 dB [= 10 lg(e)] divided by the decay rate in decibels per second of the device.

NOTE 2 Commercial level recorders, in which sound pressure level is recorded graphically as a function of time, are approximately equivalent to exponential averaging devices.

NOTE 3 When an exponential averaging device is used, there is little advantage in setting the averaging time very much less than $T/30$. When a linear averaging device is used, there is no advantage in setting the interval between points at very much less than $T/12$. In some sequential measuring procedures, it is feasible to reset 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 shortest reverberation time in any band has to serve for measurements in all bands.

4.2.2.5 Overload

No overloading shall be allowed in any stage of the measuring apparatus. Where impulsive sound sources are used, peak-level indicating devices shall be used for checking against overloading.

4.3 Measurement positions

Source positions should be located where the natural sound sources in the room would typically be located. A minimum of two source positions shall be used. The height of the acoustic centre of the source should be 1,5 m above the floor.

Microphone positions should be at positions representative of positions where listeners would normally be located. For reverberation time measurements, it is important that the measurement positions sample the entire space; for the room acoustic parameters described in Annexes A and B, they should also be selected to provide information on possible systematic variations with position in the room. Microphone positions shall be at least half a wavelength apart, i.e. a distance of around 2 m for the usual frequency range. The distance from any microphone position to the nearest reflecting surface, including the floor, shall be at least a quarter of a wavelength, i.e. normally around 1 m. See A.4 for more details.

No microphone position shall be too close to any source position, in order to avoid a too-strong influence from the direct sound. In rooms for speech and music, the height of the microphones above the floor should be 1,2 m, corresponding to the ear height of average listeners in typical chairs.

A distribution of microphone positions shall be chosen that anticipates the major influences likely to cause differences in reverberation time throughout the room. Obvious examples are the differences for seating areas close to walls, underneath balconies or in spaces which are decoupled (e.g. in church transepts and chancels compared with church naves). This requires a judgement of the evenness of the "acoustical" distribution to the different seating areas, the equality of the coupling of the separate parts of the volume and the proximity to local perturbations.

For reverberation time measurement, it can be useful to assess the room against the following criteria (which in many cases will simply require a visual assessment) to determine whether single spatial averages will adequately describe the room:

a) the materials of the boundary surfaces and any suspended elements are such that, judged in terms of their absorption and diffusion properties, they are reasonably evenly distributed among the surfaces which surround the room, and

b) all parts of the room volume communicate reasonably equally with each other, in which case three or four microphone positions will be adequate — these positions being chosen to cover the seating area, in an evenly spaced array — and the results of the measurements may be averaged.

For a) above, if the ceiling, side, front and rear walls, when assessed individually, have no regions covering more than 50 % of their respective areas, with properties different from those of the remaining surfaces, then it may be considered that the distribution is acceptably even (in some spaces it can be helpful to approximate the room geometry to a rectangular parallelepiped for this assessment).

For b) above, the room volume may be considered to operate as a single space if there are no parts of the floor area which have their lines-of-sight blocked to any other part of the room that is more than 10 % of the total room volume.

If these conditions are not satisfied, then the room is likely to show areas with differing reverberation times, and these shall be investigated and measured separately.

5 Measurement procedures

5.1 General

Two methods of measuring the reverberation time are described in this part of ISO 3382: the interrupted noise method and the integrated impulse response method. Both methods have the same expectation value. The frequency range depends on the purpose of the measurements. Where there is no requirement for specific frequency bands, the frequency range should cover at least 250 Hz to 2 000 Hz for the survey method. For the engineering and precision methods, the frequency range should cover at least 125 Hz to 4 000 Hz in octave bands, or 100 Hz to 5 000 Hz in one-third octave bands.

5.2 Interrupted noise method

5.2.1 Excitation of the room

A loudspeaker source shall be used and the signal fed into the loudspeaker shall be derived from broadband random or pseudo-random electrical noise. When using a pseudo-random noise, it shall be randomly ceased, not using a repeated sequence. The source shall be able to produce a sound pressure level sufficient to ensure a decay curve starting at least 35 dB above the background noise in the corresponding frequency band. If T_{30} is to be measured, it is necessary to create a level at least 45 dB above the background level in each frequency band.

For measurements in octave bands, the bandwidth of the signal shall be greater than one octave, and for measurements in one-third-octave bands, the bandwidth of the signal shall be greater than one-third octave. The spectrum shall be reasonably flat within the actual octave band to be measured. Alternatively, the broadband noise spectrum may be shaped to provide a pink spectrum of steady-state reverberant sound in the enclosure from 88 Hz to 5 657 Hz. Thus, the frequency range covers the one-third-octave bands with mid-band frequencies from 100 Hz to 5 kHz or octave bands from 125 Hz to 4 kHz.

For the engineering and precision methods, the duration of excitation of the room needs to be sufficient for the sound field to have achieved a steady state before the source is switched off. Thus, it is essential for the noise to be radiated for at least a few seconds and not less than half the reverberation time.

For the survey method, a short excitation or an impulse signal may be used as an alternative to the interrupted noise signal. However, in that case, the measuring accuracy is less than that stated in 7.1.

5.2.2 Averaging of measurements

The number of microphone positions used will be determined by the accuracy required (see Annex A). However, in view of the randomness inherent in the source signal, it is necessary to average over a number of measurements at each position in order to achieve an acceptable measurement uncertainty (see 7.1). The averaging in each position can be made in two different ways: either

— find the individual reverberation times for all the decay curves and take the mean value, or

— make an ensemble average of the squared sound pressure decays and find the reverberation time of the resulting decay curve.

The individual decays are superposed with their beginnings synchronised. The discrete squared sound pressure sample values are summed for each time interval increment of the decays and the sequence of these sums is used as a single overall ensemble decay from which T is then evaluated (see Reference [20]). It is important that the sound power emitted by the source be kept the same for all measurements. This is the preferred method.

5.3 Integrated impulse response method

5.3.1 General

The impulse response from a source position to a receiver position in a room is a well-defined quantity that can be measured in a variety of ways (e.g. using pistol shots, spark gap impulses, noise bursts, chirps or MLSs as signals). It is not the aim of this part of ISO 3382 to exclude any other method that can yield the correct impulse response.

5.3.2 Excitation of the room

The impulse response can be measured directly using an impulse source such as a pistol shot or any other source that is not reverberant itself as long as its spectrum is broad enough to meet the requirements of 5.2.1. The impulse source shall be able to produce a peak sound pressure level sufficient to ensure a decay curve starting at least 35 dB above the background noise in the corresponding frequency band. If T_{30} is to be measured, it is necessary to create a level at least 45 dB above the background level.

Special sound signals may be used which yield the impulse response only after special processing of the recorded microphone signal (see ISO 18233). This can provide an improved signal-to-noise ratio. Sine sweeps or pseudo-random noise (e.g. MLS) may be used if the requirements for the spectrum and directional characteristics of the source are fulfilled. Because of the improvement in signal-to-noise ratio, the dynamic requirements on the source can be considerably lower than those set in the previous paragraph. If time averaging is used, it is necessary to verify that the averaging process does not alter the measured impulse response. Using these measuring techniques, the frequency filtering is often inherent in the signal analysis, and it is sufficient that the excitation signal cover the frequency bands to be measured.

5.3.3 Integration of the impulse response

Generate the decay curve for each octave band by a backward integration of the squared impulse response. In an ideal situation with no background noise, the integration ought to start at the end of the impulse response ($t \to \infty$) and proceed to the beginning of the squared impulse response. Thus, the decay as a function of time is, according to Equation (1):

$$E(t) = \int_{t}^{\infty} p^2(\tau)\,\mathrm{d}\tau = \int_{\infty}^{t} p^2(\tau)\,\mathrm{d}(-\tau) \qquad (1)$$

where

p is the sound pressure of the impulse response as a function of time;

E is the energy of the decay curve as a function of time;

t is the time.

This integral in reverse time is often derived by performing two integrations as in Equation (2):

$$\int_t^{\infty} p^2(\tau)\,\mathrm{d}\tau = \int_0^{\infty} p^2(\tau)\,\mathrm{d}\tau - \int_0^{t} p^2(\tau)\,\mathrm{d}\tau \qquad (2)$$

In order to minimize the influence of the background noise on the later part of the impulse response, the following technique may be used.

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 displayed using a decibel scale, and calculate the decay curve from Equation (3):

$$E(t) = \int_{t_1}^{t} p^2(\tau)\,\mathrm{d}(-\tau) + C \qquad (3)$$

where ($t < t_1$) and C is an optional correction for integrated squared impulse responses 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 5 %, the level of the background noise must be at least the evaluation range plus 15 dB below the maximum of the impulse response. For instance, for the determination of T_{30}, the level of the background noise must be at least 45 dB below the maximum.

6 Evaluation of decay curves

For the determination of T_{30}, the evaluated range for the decay curves is from 5 dB to 35 dB below the steady state level. For the integrated impulse response method, the steady state level is the total level of the integrated impulse response. Within the evaluation range, a least-squares fit line shall be computed for the curve or, in the case of decay curves plotted directly by level recorder, a straight line shall be fitted manually as closely as possible to the decay curve. Other algorithms that provide similar results may be used. The slope of the straight line gives the decay rate, d, in decibels per second, from which the reverberation time is calculated as $T_{30} = 60/d$. For the determination of T_{20}, the evaluation range is from 5 dB to 25 dB.

If the technique used for determining the reverberation time is based on evaluating traces plotted by a level recorder, then a visual “best fit” line may be substituted for a computed regression line, but this will not be as reliable as a regression analysis.

In order to specify a reverberation time, the decay curves shall follow approximately a straight line. If the curves are wavy or bent, this may indicate a mixture of modes with different reverberation times and thus the result may be unreliable.

7 Measurement uncertainty

7.1 Interrupted noise method

Due to the random nature of the excitation signal, the measurement uncertainty of the interrupted noise method strongly depends on the number of averages performed. Ensemble averaging and the averaging of individual reverberation times have the same dependencies on the number of averages. The standard deviation of the measurement result $\sigma(T_{20})$ or $\sigma(T_{30})$, respectively, can be estimated from Equations (4) and (5):

$$\sigma(T_{20}) = 0{,}88 T_{20} \sqrt{\frac{1 + 1{,}90/n}{N B T_{20}}} \tag{4}$$

$$\sigma(T_{30}) = 0{,}55 T_{30} \sqrt{\frac{1 + 1{,}52/n}{N B T_{30}}} \tag{5}$$

where

B is the bandwidth, in hertz;

n is the number of decays measured in each position;

N is the number of independent measurement positions (combinations of source and receiver positions);

T_{20} is the reverberation time, in seconds, based on a 20 dB evaluation range;

T_{30} is the reverberation time, in seconds, based on a 30 dB evaluation range.

Equations (4) and (5) are derived from References [21] and [22] and based on certain assumptions concerning the averaging device.

For an octave filter, B = 0,71 f_c, and for one-third-octave filter, B = 0,23 f_c, where f_c is the mid-band frequency, in hertz, of the filter. Octave-band measurements give a better measurement accuracy than one-third-octave measurements with the same number of measurement positions.

7.2 Integrated impulse response method

Theoretically, the integrated impulse response corresponds to the averaging of an infinite number of interrupted noise excitations [11]. For practical evaluation of the measurement uncertainty using the integrated impulse response method, it can be considered as being of the same order of magnitude as that using an average of n = 10 measurements in each position with the interrupted noise method. No additional averaging is necessary to increase the statistical measurement accuracy for each position.

7.3 Lower limits for reliable results caused by filter and detector

In the case of very short reverberation times, the decay curve can be influenced by the filter and the detector. Using traditional forward analysis, the lower limits for reliable results shall be according to Equations (6) and (7):

$$BT > 16 \tag{6}$$

$$T > 2T_{det} \tag{7}$$

where

B is the filter bandwidth, in hertz;

T is the measured reverberation time, in seconds;

T_{det} is the reverberation time, in seconds, of the averaging detector.

8 Spatial averaging

The results measured for the range of source and microphone positions can be combined either for separate identified areas or for the room as a whole to give spatial average values. This spatial averaging shall be achieved by arithmetic averaging of the reverberation times. The spatial average is given by taking the mean of the individual reverberation times for all the independent source and microphone positions. The standard deviation may be determined to provide a measure of accuracy and the spatial variance of the reverberation time. See also A.4.

9 Statement of results

9.1 Tables and curves

The evaluated reverberation times for each frequency of measurement shall be both plotted in the form of a graph and stated in a table.

In the case of a graph, the points shall be connected by straight lines. The abscissa shall present frequency on a logarithmic scale using a distance of 1,5 cm per octave, while the ordinate shall use either a linear time scale such that 2,5 cm corresponds to one second or a logarithmic scale with 10 cm corresponding to one decade. The nominal mid-band frequencies for octave bands according to IEC 61260 should be marked on the frequency axis.

A single figure reverberation time, $T_{30,mid}$, can be calculated by averaging T_{30} in the 500 Hz and 1 000 Hz octave bands; $T_{20,mid}$ may also be used. Alternatively, take averages over the six one-third-octave bands from 400 Hz to 1 250 Hz.

9.2 Test report

The test report shall include the following information:

a) a statement that the measurements were made in conformity with this part of ISO 3382;

b) name and place of the room tested;

c) sketch plan of the room, with an indication of the scale;

d) volume of the room — if the room is not completely enclosed, an explanation should be given of how the stated volume is defined;

e) for rooms for speech and music, the number and type of seats, e.g. whether upholstered or not, and if the information is available, the thickness and kind of upholstery, the kind of covering material (porous or non-porous, seats raised or lowered) and which parts of the seat are covered;

f) a description of the shape and material of the walls and the ceiling;

g) state or states of occupancy during measurements and the number of occupants;

h) condition of any variable equipment such as curtains, public-address system, electronic reverberation enhancement systems, etc.;

i) for theatres, whether the safety curtain or decorative curtains were up or down;

j) description, where appropriate, of the stage furnishing, including any concert enclosure, etc.;

k) temperature and relative humidity in the room during the measurement;

l) description of measuring apparatus, source and microphones, and whether tape recorders were employed;

m) description of the sound signal used;

n) coverage chosen, including details of the source and microphone positions, preferably shown on a plan, together with the heights of the sources and microphones;

o) date of measurement and name of the measuring organization.

Annex A
(informative)

Auditorium measures derived from impulse responses

A.1 General

Subjective studies of the acoustical characteristics of auditoria have shown that several quantities that can be obtained from measured impulse responses are correlated with particular subjective aspects of the acoustical character of an auditorium. While reverberation time is one fundamental description of the acoustical character of an auditorium, the addition of values of these newer quantities gives a more complete description of the acoustical conditions in the auditorium. The quantities included in this annex are limited to those that have been found to be subjectively important and that can be obtained directly from integrating impulse responses. The introduction of an audience into an auditorium can be expected to influence the reverberation time and the quantities listed below.

There are five groups or types of quantities (see Table A.1). Within each group there is often more than one measure, but values of the different quantities in each group are usually found to be strongly correlated with each other. Thus, each group contains a number of approximately equivalent measures and it is not necessary to calculate values of all of them; nevertheless, at least one quantity should be included from each of the five groups.

Table A.1 — Acoustic quantities grouped according to listener aspects

Subjective listener aspect	Acoustic quantity	Single number frequency averaging [a] Hz	Just noticeable difference (JND)	Typical range [b]
Subjective level of sound	Sound strength, G, in decibels	500 to 1 000	1 dB	−2 dB; +10 dB
Perceived reverberance	Early decay time (EDT) in seconds	500 to 1 000	Rel. 5 %	1,0 s; 3,0 s
Perceived clarity of sound	Clarity, C_{80}, in decibels	500 to 1 000	1 dB	−5 dB; +5 dB
	Definition, D_{50}	500 to 1 000	0,05	0,3; 0,7
	Centre time, T_S, in milliseconds	500 to 1 000	10 ms	60 ms; 260 ms
Apparent source width (ASW)	Early lateral energy fraction, J_{LF} or J_{LFC}	125 to 1 000	0,05	0,05; 0,35
Listener envelopment (LEV)	Late lateral sound level, L_J, in decibels	125 to 1 000	Not known	−14 dB; +1 dB

[a] The single number frequency averaging denotes the arithmetical average for the octave bands, except for L_J which shall be energy averaged [see (A.17)].

[b] Frequency-averaged values in single positions in non-occupied concert and multi-purpose halls up to 25 000 m^3.

A.2 Definitions of measures

A.2.1 Sound strength

The sound strength, G, can be measured using a calibrated omnidirectional sound source, as the logarithmic ratio of the sound energy (squared and integrated sound pressure) of the measured impulse response to that of the response measured in a free field at a distance of 10 m from the sound source, as expressed in Equations (A.1) to (A.3):

$$G = 10 \lg \frac{\int_0^\infty p^2(t)\,\mathrm{d}t}{\int_0^\infty p_{10}^2(t)\,\mathrm{d}t} = L_{pE} - L_{pE,10} \text{ dB} \tag{A.1}$$

in which

$$L_{pE} = 10 \lg \left[\frac{1}{T_0} \int_0^\infty \frac{p^2(t)\,\mathrm{d}t}{p_0^2} \right] \text{ dB} \tag{A.2}$$

and

$$L_{pE,10} = 10 \lg \left[\frac{1}{T_0} \int_0^\infty \frac{p_{10}^2(t)\,\mathrm{d}t}{p_0^2} \right] \text{ dB} \tag{A.3}$$

where

$p(t)$ is the instantaneous sound pressure of the impulse response measured at the measurement point;

$p_{10}(t)$ is the instantaneous sound pressure of the impulse response measured at a distance of 10 m in a free field,

p_0 is 20 µPa;

T_0 = 1 s

L_{pE} is the sound pressure exposure level of $p(t)$;

$L_{pE,10}$ is the sound pressure exposure level of $p_{10}(t)$.

In the above equations, $t = 0$ corresponds to the start of the direct sound, and ∞ should correspond to a time that is greater than or equal to the point at which the decay curve has decreased by 30 dB.

In the case where a large anechoic room is available, $L_{pE,10}$ can be directly measured using a source-to-receiver distance of 10 m. If this condition is not attainable, the sound pressure exposure level at a point which is d ($\geqslant$ 3 m) from the source ($L_{pE,d}$) may be measured and $L_{pE,10}$ then obtained from Equation (A.4):

$$L_{pE,10} = L_{pE,d} + 20 \lg (d/10) \text{ dB} \tag{A.4}$$

When making such a measurement in a free field, it is necessary to make the measurement at every 12,5° around the sound source and to calculate the energy-mean value of the sound pressure exposure levels in order to average the directivity of the sound source.

NOTE 1 As an alternative method, the reference sound pressure exposure level $L_{pE,10}$ can be measured in a reverberation room using Equation (A.5) [7], [8]:

$$L_{pE,10} = L_{pE} + 10 \lg (A / S_0) - 37 \text{ dB} \tag{A.5}$$

where

L_{pE} is the spatial-average sound pressure exposure level measured in the reverberation room;

A is the equivalent sound absorption area in square metres;

S_0 = 1 m^2.

A can be obtained from the reverberation time in the room using Equation (A.6) (Sabine's formula):

$$A = 0{,}16\, V/T \tag{A.6}$$

where

V is the air volume of the reverberation room in cubic metres;

T is the reverberation time of the room in seconds.

NOTE 2 G can alternatively be measured by using a stationary omnidirectional sound source using Equation (A.7):

$$G = L_p - L_{p,10} \tag{A.7}$$

where

L_p is the sound pressure level measured at each measurement point in the room under test;

$L_{p,10}$ is the sound pressure level measured at a distance of 10 m in a free field.

In the case where a large anechoic room is available, $L_{p,10}$ can be directly measured by using a source-to-receiver distance of 10 m. If this condition is not attainable, the sound pressure exposure level at a point of d ($\geqslant$ 3 m) from the source ($L_{p,d}$) can be measured and $L_{p,10}$ then obtained using Equation (A.8):

$$L_{p,10} = L_{p,d} + 20 \lg (d / 10) \text{ dB} \tag{A.8}$$

In this case, it is also necessary to average the directivity of the sound source as mentioned above.

When using an omnidirectional sound source of which the sound power level is known, the sound strength, G, can be obtained from Equation (A.9):

$$G = L_p - L_W + 31 \text{ dB} \tag{A.9}$$

where

L_p is the sound pressure level measured at every measurement point;

L_W is the sound power level of the sound source.

The sound power level of the source should be measured according to ISO 3741.

A.2.2 Early decay time measurements

The early decay time (EDT) shall be evaluated from the slope of the integrated impulse response curves (as the conventional reverberation time). The slope of the decay curve should be determined from the slope of the best-fit linear regression line of the initial 10 dB (between 0 dB and −10 dB) of the decay. The decay times should be calculated from the slope as the time required for a 60 dB decay.

Both the EDT and T should be calculated. EDT is subjectively more important and related to perceived reverberance, while T is related to the physical properties of the auditorium.

A.2.3 Balance between early- and late-arriving energy

While there are several parameters that can be used in this group, one of the simplest is an early-to-late arriving sound energy ratio. This can be calculated for either a 50 ms or an 80 ms early time limit, depending on whether the results are intended to relate to conditions for speech or music, respectively, using Equation (A.10).

$$C_{t_e} = 10 \lg \frac{\int_0^{t_e} p^2(t)\,\mathrm{d}t}{\int_{t_e}^{\infty} p^2(t)\,\mathrm{d}t}\ \mathrm{dB} \tag{A.10}$$

where

C_{t_e} is the early-to-late index;

t_e is the early time limit of either 50 ms or 80 ms (C_{80} is usually “clarity”);

$p(t)$ is the instantaneous sound pressure of the impulse response measured at the measurement point.

NOTE 1 It is also possible to measure an early to total sound energy ratio. For example, D_{50} (“definition” or “Deutlichkeit”) is sometimes used for speech conditions, as per Equation (A.11):

$$D_{50} = \frac{\int_0^{0{,}050} p^2(t)\,\mathrm{d}t}{\int_0^{\infty} p^2(t)\,\mathrm{d}t} \tag{A.11}$$

This is exactly related to C_{50} by the relationship expressed using Equation (A.12):

$$C_{50} = 10 \lg \left(\frac{D_{50}}{1 - D_{50}} \right)\ \mathrm{dB} \tag{A.12}$$

Thus it is not necessary to measure both quantities.

As a final option in this group of measures, the centre time, T_S, which is the time of the centre of gravity of the squared impulse response, can be measured, in seconds, using Equation (A.13):

$$T_S = \frac{\int_0^\infty t\, p^2(t)\, \mathrm{d}t}{\int_0^\infty p^2(t)\, \mathrm{d}t} \tag{A.13}$$

T_S avoids the discrete division of the impulse response into early and late periods.

Quantities in this group relate to perceived definition, clarity, or the balance between clarity and reverberance, as well as to speech intelligibility.

NOTE 2 Speech intelligibility can also be determined by measuring the speech transmission index (STI) (see Reference [5]). This quantity was originally measured by using special modulated noise signals which are not covered in this part of ISO 3382, but it may be derived by post-processing the impulse response as well.

A.2.4 Early lateral energy measures

The fraction of energy, J_{LF}, arriving from lateral directions within the first 80 ms can be measured from impulse responses obtained from an omnidirectional and a figure-of-eight pattern microphone using Equation (A.14):

$$J_{LF} = \frac{\int_{0,005}^{0,080} p_L^2(t)\, \mathrm{d}t}{\int_0^{0,080} p^2(t)\, \mathrm{d}t} \tag{A.14}$$

where

$p_L(t)$ is the instantaneous sound pressure in the auditorium impulse response measured with a figure-of-eight pattern microphone;

$p(t)$ is the instantaneous sound pressure of the impulse response measured at the measurement point.

It is intended that the null of the figure-of-eight pattern microphone be pointed towards an average centre-stage source position, or exactly towards individual source positions, so that this microphone responds predominantly to sound energy arriving from lateral directions and is not significantly influenced by the direct sound.

Because the directivity of the figure-of-eight microphone is essentially a cosine pattern and pressure values are squared, the resulting contribution to lateral energy for an individual reflection varies with the square of the cosine of the angle of incidence of the reflection relative to the axis of maximum sensitivity of the microphone.

As an alternative, an approximation for obtaining lateral energy fractions, J_{LFC}, with contributions which vary as the cosine of the angle, thought to be subjectively more accurate [9], can be used with Equation (A.15):

$$J_{LFC} = \frac{\int_{0,005}^{0,080} \left| p_L(t) \cdot p(t) \right| \mathrm{d}t}{\int_0^{0,080} p^2(t)\, \mathrm{d}t} \tag{A.15}$$

where

$p_L(t)$ is the instantaneous sound pressure in the auditorium impulse response measured with a figure-of-eight pattern microphone;

$p(t)$ is the instantaneous sound pressure of the impulse response measured at the measurement point.

Lateral energy fractions relate to perceived width of the sound source.

Interaural cross correlation measures are also thought to relate to spatial impression. They are described in Annex B.

A.2.5 Late lateral energy measures

The relative level, L_J, of late-arriving lateral sound energy can be measured using a calibrated omnidirectional sound source, from the impulse response obtained in the auditorium from a figure-of-eight pattern microphone, with Equation (A.16):

$$L_J = 10 \lg \left[\frac{\int_{0,080}^{\infty} p_L^2(t)\,dt}{\int_{0}^{\infty} p_{10}^2(t)\,dt} \right] \text{dB} \tag{A.16}$$

where

$p_L(t)$ is the instantaneous sound pressure in the impulse response measured with a figure-of-eight pattern microphone;

$p_{10}(t)$ is the instantaneous sound pressure in the impulse response measured with an omnidirectional microphone at a distance of 10 m in a free field.

It is intended that the null of the figure-of-eight pattern microphone be pointed towards an average centre stage source position, or exactly towards individual source positions, so that this microphone responds predominantly to sound energy arriving from lateral directions and is not significantly influenced by the direct sound.

The frequency-averaged late lateral sound energy level, $L_{J,\text{avg}}$, is calculated from Equation (A.17):

$$L_{J,\text{avg}} = 10 \lg \left[0,25 \sum_{i=1}^{4} 10^{L_{J_i}/10} \right] \text{dB} \tag{A.17}$$

where

L_{J_i} is the value in octave band i;

i is each of the four octave bands with centre frequencies 125 Hz, 250 Hz, 500 Hz and 1 000 Hz.

Late lateral sound energy relates to perceived listener envelopment or spaciousness in the auditorium.

A.3 Measurement procedure

A.3.1 Source

The source and associated equipment should be adequate to radiate a sufficient signal level in all of the octave bands for 125 Hz to 4 000 Hz, so that an adequate decay range is achieved in each octave band. The source shall be as omnidirectional as possible (see 4.2.1).

For tests relating to conditions with a human speaker, a source with a directivity approximating a human speaker may be used. Dummy heads that comply with Reference [6] may be used without an explicit check of the directivity pattern.

A.3.2 Microphones

An omnidirectional microphone should be used to measure the impulse response for all of the measures.

For J_{LF} values, a figure-of-eight pattern microphone is also required, and the relative sensitivities of the omnidirectional and figure-of-eight microphones in the direction of maximum sensitivity should be calibrated in a free sound field.

For G values, the sensitivity of the omnidirectional microphone shall be calibrated.

A.3.3 Impulse responses

Octave band impulse responses are necessary for the calculation of all quantities. These can be obtained using an impulsive source, such as a blank pistol, or from more complex techniques requiring the calculation of the impulse response from various types of signals radiated from loudspeakers. If the resulting impulse response is not exactly repeatable, then results should be the average of several repeated measurements at the same position.

Blank pistols can be modified to be closely omnidirectional, but do not produce exactly repeatable impulse responses. They can produce very high sound levels, providing results with a desirable large dynamic range, but this can lead to non-linear effects close to the gun.

Methods using a loudspeaker source are limited by the frequency and directional response of the loudspeaker. The average frequency response can, to some extent, be corrected, but variations with direction cannot be eliminated and become significantly large at higher frequencies. Using a loudspeaker to radiate various pulse signals is usually not very successful because of the limited dynamic range of the resulting impulse response, unless many pulse responses are synchronously averaged. Cross-correlation of the source signal and the received signal can provide impulse responses with good dynamic range and immunity to noise (see ISO 18233). The use of fast Hadamard transforms and MLS signals is one successful correlation-type approach [14]. Other signals with broad smooth spectrum, such as chirps and linear sweeps, can also be successfully used.

A.3.4 Time-windowing and filtering of responses

Impulse responses should be filtered into octave bands.

Filters create signal delays which can be quite significant for the narrower bandwidth, lower frequency octave bands. Thus, the start of the filtered impulse is delayed relative to the unfiltered signal, and the filtered signal also continues on after the end of the unfiltered signal. This creates particular problems for measures such as C_{80} or fraction of energy J_{LF}, where the short early time interval portions of the signals are filtered into octave bands.

The best approach for avoiding the filter delay problems is to time-window the broadband impulse response before any filtering. The start of the impulse response for the equations given in A.2 should be determined from the broadband impulse response, where the signal first rises significantly above the background but is more than 20 dB below the maximum. The early and late components of the impulse response are filtered separately, and the integration periods in the equations of A.2 are increased to include the energy delayed by the filters.

An approximation to the above window-before-filtering approach can be obtained using a window correction [7]. If the impulse signals are first filtered into octave bands, the start of the integrations for the equations of A.2 should be determined as the point where the filtered signal first rises significantly above the background but is more than 20 dB below the maximum. The early time interval, t_e, shall start from this trigger point and continue for t_e, in seconds, plus half the filter delay time. The late time interval should start from the point t_e, in seconds, plus half the filter delay time after the trigger point. In this context, the filter delay time is the time for half the energy from the filter when fed with an impulse.

Because the direct and early-arriving low frequency sound can be significantly attenuated, determining the start of the low frequency responses may not be possible. It may be necessary to determine the start time from the broadband or high frequency impulse responses and the measured delay of the filters.

A.3.5 Decay curves

The integrated impulse response technique (reverse integration) according to 5.3.3 should be used to obtain integrated octave band decay curves from which decay times are calculated. For convenience, other measures can also be calculated from these decay curves, assuming the correct time-windowing is carried out. This approach requires that the start time of each octave band response be correctly obtained from the broadband response. In other situations, forward integration can be used to separately obtain values of other quantities.

A.4 Measurement positions

The various measures are not statistical properties of the entire auditorium and will vary systematically from seat to seat. It is therefore important to include an adequate number of source and receiver positions to characterize the entire hall.

Normally, a minimum of three on-stage source positions should be used. In halls with large stages or orchestra pits, more source positions should be used. In small lecture theatres where the normal source has only one location in the room, a single source position would be acceptable.

The source should be at positions representative of those used by performers in the hall. Because most halls are symmetrical about the centre line, receiver positions can be arranged on only one side of the hall with source positions located symmetrically about the centre line. Thus there could be one central source position with other source positions at equal distances stage-right and stage-left of the centreline. A source height of 1,5 m is recommended to avoid low frequency modification of the output power of the source in the frequency range of the measurements.

If the source directivity is close to the minimum limits given in Table 1, the measurement should be repeated with the source turned in at least three steps totally. The resulting parameters related to the different angles of the source should be arithmetically averaged.

A minimum of between 6 and 10 representative microphone positions should be used, depending on the size of the hall. Table A.2 gives the minimum recommended number of receiver positions as a function of hall size. The receiver positions should be evenly distributed over all audience seating areas. Where a hall is broken up into separate areas, such as balconies and under-balcony areas, more receiver locations will be necessary.

The microphone should be placed at a height of 1,2 m above the floor at audience seat locations in order to be representative of a seated listener's ear height.

Source and receiver positions and heights should be noted with the results. Similarly, on-stage conditions such as the presence of chairs and music stands should be noted because they will produce measurable effects on the results.

Table A.2 — Minimum number of receiver positions as a function of auditorium size

Number of seats	Minimum number of microphone positions
500	6
1 000	8
2 000	10

A.5 Statement of results

In addition to the format of presentation of results specified for reverberation time, T, values can be presented in a more concise manner by determining averages for the results from pairs of octaves. Thus the 125 Hz and 250 Hz results would be arithmetically averaged to give a low frequency result; the 500 Hz and 1 000 Hz results would be averaged to give a mid-frequency result, and the 2 000 Hz and 4 000 Hz results would be averaged to give a high frequency result. However, lateral energy fractions in the 4 000 Hz octave band are not usually thought to be subjectively important.

For a single-number value of the parameters, the arithmetical average for the octave bands apply, except for L_J, which shall be energy averaged [see Equation (A.17)]. The frequency averaging stated in Table A.1 should be used and index "m" (for weighting) applied to the symbol.

EXAMPLE 1 G_{m} strength averaged in 500 Hz to 1 000 Hz octave bands

EXAMPLE 2 J_{LFm} early lateral energy fraction averaged in 125 Hz to 1 000 Hz octave bands.

The measurement results for the measures described in this annex should normally not be averaged over all microphone positions in a hall because the measures are assumed to describe local acoustical conditions. In the case of a large hall, it can be useful to average the results in some sections of the hall (stalls, first balcony, etc.) Some measures such as sound strength, G, tend to vary with the distance, and a graphical plot of G as a function of source-receiver distance can be useful.

Annex B
(informative)

Binaural auditorium measures derived from impulse responses

B.1 General

The process of hearing is binaural. Subjective studies of auditoria have shown that inter-aural cross correlation coefficients (IACC), measured with either a dummy head or a real head with average dimensions as exemplified by dummy heads, and with small microphones at the entrance to the ear canals, correlate well with the subjective quality "spatial impression" in a concert hall (early lateral energy measures are also thought to relate to spaciousness (see Annex A).

Spatial impression may be divided into two subclasses:

— Subclass 1: broadening of the source, i.e. apparent source width (ASW);

— Subclass 2: a sense of being immersed or enveloped in the sound, i.e. listener envelopment (LEV).

B.2 Definition of IAAC

The normalized inter-aural cross correlation function (IACF) is first defined using Equation (B.1):

$$\mathrm{IACF}_{t_1,t_2}(\tau) = \frac{\int_{t_1}^{t_2} p_\mathrm{l}(t) \cdot p_\mathrm{r}(t+\tau)\,\mathrm{d}t}{\sqrt{\int_{t_1}^{t_2} p_\mathrm{l}^2(t)\,dt \int_{t_1}^{t_2} p_\mathrm{r}^2(t)\,\mathrm{d}t}} \tag{B.1}$$

where

$p_\mathrm{l}(t)$ is the impulse response at the entrance to the left ear canal;

$p_\mathrm{r}(t)$ is the impulse response at the entrance to the right ear canal.

The inter-aural cross correlation coefficients, IACC, are then given by Equation (B.2):

$$\mathrm{IACC}_{t_1,t2} = \max\left|\mathrm{IACF}_{t_1,t2}\right| \quad \text{for } -1\,\mathrm{ms} < \tau < +1\,\mathrm{ms} \tag{B.2}$$