S.F.

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Acoustics — Measurement of the reverberation time of rooms with reference to other acoustical parameters

Acoustique — Mesurage de la durée de réverbération des salles en référence à d'autres paramètres acoustiques

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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.

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

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

This second edition cancels and replaces the first edition (ISO 3382:1975), which has been technically revised.

Annexes A, B and C of this International Standard are for information only.

Introduction

The reverberation time of a room used to be regarded as the predominant indicator of its acoustical properties. Whilst 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 acoustical quality of rooms. This International Standard continues to specify room acoustic quality by reverberation time alone, but introduces two other levels of complexity in room acoustics measurement.

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 on but, 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 the measurements in auditoria.

Reverberation time measurements are important in the field of noise control in rooms as well as for the assessment of rooms for speech and music; this International Standard also applies to measurements in these enclosures. However, it does not apply to laboratory measurements in test facilities or reverberation rooms. Laboratory measurements require other specifications of averaging single measurements at prescribed source and microphone positions. This International Standard establishes a method for obtaining reverberation times from impulse responses and from interrupted noise. In the annexes, the concepts and details of measurement procedures for some of the newer measures are introduced, but these annexes do not constitute a part of the formal specifications of this standard. 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.

Acoustics — Measurement of the reverberation time of rooms with reference to other acoustical parameters

1 Scope

This International Standard specifies methods for the measurement of reverberation time in rooms. It is not restricted to auditoria or concert halls; it is also applicable to rooms intended for speech and music or where noise protection is a consideration. It describes the measurement procedure, the apparatus needed, the coverage required, and the method of evaluating the data and presenting the test report. Furthermore, it is intended for application of modern digital measuring techniques and for evaluation of room acoustical parameters derived from impulse responses.

2 Normative references

The following standards contain provisions which, through reference in this text, constitute provisions of this International Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this International Standard are encouraged to investigate the possibility of applying the most recent editions of the standards listed below. Members of IEC and ISO maintain registers of currently valid International Standards.

ISO 3741:1988, *Acoustics — Determination of sound power levels of noise sources — Precision methods for broad-band sources in reverberation room.*

ISO 5725-2:1994, *Accuracy (trueness and precision) of measurement methods and results — Part 2: Basic method for the determination of repeatability and reproducibility of a standard measurement method.*

IEC 268-1:1985, *Sound system equipment – Part 1: General.*

IEC 651:1979, *Sound level meters.*

IEC 1260:1995, *Electroacoustics — Octave-band filters and fractional-octave-band filters.*

ITU Recommendation P.58:1994, *Head and torso simulator for telephonometry.*

3 Definitions

For the purposes of this International Standard, the following definitions apply.

3.1 decay curve:
Decay of sound pressure level as a function of time at one point of the room after the source of sound has ceased.

NOTE 1 This decay may be either measured 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.

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 should only be used for survey purposes.

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.

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

3.4 impulse response:
Plot as a function of time of the sound pressure received in a room as a result of excitation of the room by a Dirac delta function.

NOTE 3 It is impossible in practice to create and radiate true Dirac delta functions but short transient sounds (e.g. from gunshots) may offer close enough approximations for practical measurement. An alternative measurement technique, however, is to use a period of maximum-length sequence type signal (or other deterministic, flat-spectrum signal) and transform the measured response back to an impulse response.

3.5 reverberation time, T:
Time, expressed in seconds, that would be required for the sound pressure level to decrease by 60 dB, at a rate of decay given by the linear least-squares regression of the measured decay curve from a level 5 dB below the initial level to 35 dB below.

NOTE 4 Where a decay curve is not monotonic the range to be evaluated is defined by the times at which the decay curve first reaches 5 dB and 35 dB below the initial level respectively. A value for T based on the decay rate over a smaller dynamic range (down to a minimum of 20 dB extending from 5 dB down to 25 dB down) is also allowable provided the results are appropriately labelled. In the case of ambiguity the measure for T using the decay between 5 dB and 35 dB should be called T_{30}. Using 5 dB and 25 dB, the result should be labelled T_{20} and similarly for other evaluation ranges.

3.6 States of occupancy

NOTE 5 Reverberation time measured in a room will be influenced by the number of people present and the following states of occupancy are defined for measurement purposes.

NOTE 6 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.

NOTE 7 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 these cases, measurement may be useful. If the safety curtain is up, the amount of furnishing of the stage is of importance and shall be described.

3.6.1 unoccupied state:
State of the room prepared for use and ready for speakers or performers and audience, but without these persons present; for concert halls and opera houses the presence of chairs for performers, music stands and percussion instruments etc. shall be taken into account.

3.6.2 studio state (only for rooms for speech and music):
State of the room occupied by the performers or speakers only (without audience), for example at rehearsals or during sound recordings; 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 8 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) should be noted with the results.

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 may 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.

NOTE 9 Where variable components involve active (i.e. electronic) techniques then 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 should be as close to omni-directional as possible. It shall produce a sound pressure level sufficient to provide decay curves with the required minimum dynamic range without contamination by background noise (see 3.5). Commercial domestic loudspeakers are not acceptable as an omni-directional source. 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 correlated averaging is possible. In the case of measurements which do not use a synchronous averaging (or other) technique to augment the decay range then a source level will be required which 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.

4.2.2 Microphones, recording and analysis equipment

Omni-directional 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.

4.2.2.1 Microphone and filters

The measurement equipment shall meet the requirements of a type 1 sound level meter according to IEC 651. The octave or one-third-octave filters shall conform with IEC 1260. 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.

4.2.2.2 Tape recorder

If the sound decay is initially recorded on magnetic tape, automatic gain control or other circuits for dynamic optimisation of signal-to-noise ratio shall not be used. A relatively long tape recording shall be made of each decay to enable determination of the final background level following the decay.

The tape recorder 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 within a tolerance of ± 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 $10^{0,01n}$ within ± 2 %, where n is an integer including zero.

NOTE 10 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.

NOTE 11 Where a tape recorder is used then in the requirements in 4.2.2.3 below concerning the speed of response of the apparatus for forming a record of the decay of sound pressure level with time, 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.

NOTE 12 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 [4]).

4.2.2.3 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 average time, i.e. time constant of an exponential averaging device (or appropriate equivalent) shall be less than, but as close as possible to $T/20$. Similarly, the averaging time of a linear averaging device shall be less than $T/7$. (Here T is the reverberation time being measured or, if appropriate, the effective reverberation time as described in note 11 above.)

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 13 The averaging time of an exponential averaging device is equal to 4,34 divided by the decay rate in decibels per second of the device.

NOTE 14 Commercial level recorders, in which sound pressure level is recorded graphically as a function of time, are approximately equivalent to exponential averaging devices.

NOTE 15 When an exponential averaging device is used there is little advantage in setting the averaging time very much less than $T/20$. When a linear averaging device is used there is no advantage in setting the interval between points at very much less than $T/7$. 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.

4.2.2.4 Overload indication

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

As measurements may be required for different purposes the number of measurement positions are chosen in order to achieve an appropriate coverage in the room. Microphone positions shall be at least half a wavelength apart, i.e. a minimum 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.

No microphone position shall be too close to any source position in order to avoid too strong influence from the direct sound. The minimum distance d_{min}, in metres, can be calculated from:

$$d_{\min} = 2\sqrt{\frac{V}{cT}}$$

where

V is the volume, in cubic metres;

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

T is an estimate of the expected reverberation time, in seconds.

NOTE 16 In small rooms with very short reverberation time (e.g. talks studios) it may be impossible to fulfil the above requirement. In such cases, and only for the measurement of reverberation time, it is recommended that the direct sound is eliminated by insertion of a barrier (with negligible sound absorption) between source and receiver.

Each pair of measurement positions is a combination of one source position and one microphone position. The number of positions can be chosen to yield either a low coverage or a normal coverage.

4.3.1 Low coverage (least measurement effort)

Measurements are made for assessment of the amount of room absorption for noise control purposes, including measurement of sound reduction index, or assessment of the reverberation time for sound system calculations.

Make measurements of T for two source positions which are representative of those where noise sources are located or of those used by performers and find the average of results from three or four microphone positions in areas where people normally are present or in "centre of seating" areas. If the deviations between the results from the single positions extend the tolerances set for the purpose of the measurement, use more positions.

4.3.2 Normal coverage

Measurements made for verification of building performance against a design brief.

Choose the number and location of source positions so as to include all areas likely to be occupied by performers (e.g. upon stage, risers, orchestra pits and choral seating) in addition to main stage areas. A minimum of two source positions shall be used.

A distribution of microphone positions shall be chosen which 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 may 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 amongst the surfaces which surround the room, and

b) all parts of the room volume communicate reasonably equally with each other, then 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. 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.

NOTE 17 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 may be helpful to approximate the room geometry to a rectangular parallelepiped for this assessment.)

NOTE 18 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 which is more than 10 % of the total room volume.

NOTE 19 If conditions of notes 17 and 18 are not satisfied then the room is likely to show areas with differing reverberation times, and these should be investigated and measured separately.

5 Measurement procedures

5.1 General

Two methods of measuring the reverberation time are described in this standard: the interrupted noise method and the integrated impulse response method. Both methods have the same expectation value but the latter requires more sophisticated instrumentation. If room acoustic measures other than the reverberation time are to be measured only the latter method is relevant, as these are based on the impulse response.

NOTE 20 It is preferable to measure reverberation times in octave bands from 63 Hz to 4 kHz in concert halls and rooms for speech. For measurements in rooms for other purposes measurements in one-third-octave bands from 100 Hz to 5 kHz can be applied.

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 sound source should be as omni-directional as possible.

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 an approximately pink spectrum of steady-state reverberant sound in the enclosure from 88 Hz to 5 657 Hz (i.e. a range covering the one-third-octave bands with midband frequencies from 100 Hz to 5 kHz or octave bands from 125 Hz to 4 kHz) with the reverberation time being measured simultaneously in different octave or one-third-octave bands.

The duration of excitation of the room needs to be sufficient for the sound field to have achieved a steady state before being allowed to decay, and thus it is essential for the noise to be radiated for a minimum period of $T/2$ seconds. In large volumes the duration of the excitation shall be at least a few seconds.

NOTE 21 Broadband noise excitation puts more severe requirements on the power handling capacity of the loudspeaker system to maintain the required signal-to-noise ratios.

5.2.2 Number of measurements

The number of microphone positions used will be determined by the coverage required. 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 repeatability (see 6.1.1). Therefore, a minimum of three measurements shall be made at each position and the results averaged. Then, 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 method used shall be stated in the test report. If ensemble averaging is used it is allowed to make only one measurement in each of a minimum of 18 positions instead of using six positions with three measurements at each position.

NOTE 22 In the limit of an infinite number of measurements with interrupted noise the ensemble averaged decay curve will be identical with that of a single integrated squared impulse response.

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, which can be measured in a variety of ways (e.g. using pistol shots, spark gap impulses, noise bursts, chirps or m-sequences as signals). It is not the aim of this standard 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 45 dB above the background noise 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.

Special sound signals may be used which yield the impulse response only after special processing of the recorded 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 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 (for example in order to enhance the signal to noise ratio) it is necessary to verify that the averaging process does not alter the measured impulse response.

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 an approximately pink spectrum of steady-state reverberant sound in the enclosure from 88 Hz to 5 657 Hz (i.e. a range covering the one-third-octave bands with midband frequencies from 100 Hz to 5 kHz or octave bands from 125 Hz to 4 kHz) with the reverberation time being measured simultaneously in different octave or one-third-octave bands.

5.3.3 Integration of the impulse response

Generate for each octave band the decay curve by a backward integration of the squared impulse response. In an ideal situation with no background noise the integration should 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

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

where

p is the impulse response.

This integral in reverse time is often derived by performing two integrations as follow:

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

In order to minimise the influence of the background noise on the later part of the impulse response, use one of the following two different techniques for the implementation:

a) If the level of the background noise is unknown, perform the backward integration of the squared impulse response using a sliding fixed integration time, T_0, the size of which is a compromise.

$$E(t) = \int_{t+T_0}^{t} p^2(\tau)\, \mathrm{d}(-\tau)$$

The optimum value of T_0 is 1/5 of the reverberation time. Estimate the expected reverberation time. If it turns out that the measured value of the reverberation time differs by more than 25 % from the estimated value, then change the integration time accordingly and repeat the integration. The starting time t_1 of the backward sliding integration is not critical, but it shall not be shorter than the reverberation time. The integrated background noise will appear on the decay curve as a horizontal tail, a noise floor. The level of the noise floor shall be at least 10 dB below the lower value of the evaluation range, e.g. for evaluation of T_{20} the noise floor shall be at least 35 dB below the maximum level of the integrated squared impulse response.

b) 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, and calculate the decay curve from

$$E(t) = \int_{t1}^{t} p^2(\tau)\, \mathrm{d}(-\tau) + C$$

where

$(t < t_1)$ and C is an optional correction for 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 response, which is at least 15 dB plus the dynamic range over which T is to be assessed: for instance, 45 dB below the maximum for determination of T_{30}.

6 Evaluation of decay curves

6.1 Interrupted noise method

In the case of measurements using the interrupted noise method, evaluate the decay curves over the range from 5 dB to 35 dB below the initial level for T_{30} and from 5 dB to 25 dB below for T_{20}. In this 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. The slope of the straight line gives the rate of decay in decibels per second from which the reverberation time is calculated.

NOTE 23 If it is not possible to fit a straight line to the decay curve, it is considered as a non-linear decay curve (see 6.3).

If the technique used for determining the reverberation time is based on evaluating traces plotted out 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. The method of determining the decay rate shall be stated in the report.

The lowest point on the measurement range shall be sufficiently above the background noise level. For measurements of T_{30} the noise level shall be at least 45 dB below the initial level. For measurements over a 20 dB range the noise level shall be at least 35 dB below the initial level.

6.1.1 Measurement uncertainty

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 averaging individual reverberation times have the same dependences on the number of averages. The relationship between the measurement repeatability, r, in accordance with ISO 5725-2 and the number of averages N can be estimated for T_{30} by

$$r_{30} = \frac{200}{\sqrt{B\,N\,T_{30}}}\,\%$$

and for T_{20} by

$$r_{20} = \frac{370}{\sqrt{B\,N\,T_{20}}}\,\%$$

where B is the filter bandwidth.

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 midband frequency of the filter. Octave band measurements require less averages than one-third-octave measurements. Three is the minimum number of excitations to be averaged.

6.2 Integrated impulse response method

For measurements made using the integrated impulse response method, the record shall be evaluated from between 5 dB below the total integrated level to a level at least 25 dB below the total integrated level. This gives a minimum dynamic range of 20 dB for the measurement but wherever possible a 30 dB range should be used, and in all cases the measured range shall be stated. A least-squares fit line shall be used to determine the slope for the reverberation time.

6.2.1 Measurement uncertainty

The repeatability in measurements using the integrated impulse response method is of the same order of magnitude as the comparable repeatability of an average of 10 measurements with the interrupted noise method. Normally no additional averaging is necessary to decrease the statistical measurement uncertainty. However, as described in section 5.3.3, care has to be taken to select the correct starting point for the backward integration to avoid systematic errors.

6.3 Non-linear decay curves

In cases where the decay curve is not a straight line a unique reverberation time cannot be said to exist.

Where the decay curve takes the form of two straight lines, then establish a single break point appropriate to all traces at that frequency, in terms of level relative to the initial level. Measure the slopes of the upper and lower sections of the curve and the appropriate dynamic ranges specified. The minimum acceptable dynamic range for measurement of the slope shall be 10 dB.

6.4 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:

$$B\,T > 16 \text{ and } T > 2\,T_{det}$$

where B is the filter bandwidth and T_{det} is the reverberation time of the averaging detector.

In low-coverage measurements the limits can be reduced to:

$$B\,T > 8 \text{ and } T > T_{det}$$

NOTE 24 Very short reverberation times may be analysed using the time reversal technique described in 4.2.2. In that case the lower limits for reliable results are $B\,T > 4$ and $T > T_{det}/4$.

7 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 either of the following procedures (the procedure used shall be stated in the test report):

a) Arithmetic averaging of the reverberation times. The spatial average is given by taking the mean of the individual reverberation times for all the relevant source and microphone positions. The standard deviation may be determined to provide a measure of accuracy and the spatial variance of the reverberation time.

b) Ensemble averaging of the decay curves. The individual decays are superposed with their beginnings synchronised. The discrete squared sound pressure sample values (after the exponential or linear averaging process, see 4.2.2.3) 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 clause 6).

8 Statement of results

8.1 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 graphs, 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, whilst 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 1 decade. The nominal midband frequencies for octave bands according to IEC 1260 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.

8.2 Test report

The test report shall state that the measurements were made in conformity with this International Standard. It shall include:

a) the name and place of the room tested;

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

c) the volume of the room;

NOTE 25 If the room is not completely enclosed, an explanation shall be given of how the stated volume is defined.

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

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

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

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

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

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

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

k) the type and position of sound sources employed;

l) a description of the sound signal used;

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

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

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

Annex A
(informative)
Auditorium measures derived from impulse responses

A.1 Introduction

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.

A.2 Definitions of measures

There are four groups or types of quantities. 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, but at least one quantity should be included from each of the four groups.

A.2.1 Sound strength

The sound strength G can be measured using a calibrated omni-directional sound source, as the logarithmic ratio of the sound pressure exposure (squared and integrated sound pressure) of the measured impulse response to that of the response measured at a distance of 10 m from the same sound source in a free field.

$$G = 10\ \log_{10} \frac{\int_0^\infty p^2(t)\ \mathrm{d}t}{\int_0^\infty p_{10}^2(t)\ \mathrm{d}t}\ \mathrm{dB} = L_{pE} - L_{pE,10}$$

in which

$$L_{pE} = 10 \log_{10} \left[\frac{1}{T_0} \int_0^\infty \frac{p^2(t)\ \mathrm{d}t}{p_0^2} \right] \mathrm{dB}$$

and

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

where $p(t)$ is the instantaneous sound pressure of the impulse response measured at the measurement point, $p_{10}(t)$ is that measured at a distance of 10 m in a free field, p_0 is 20 µPa and T_0 = 1 s. L_{pE} and $L_{pE,10}$ are the sound pressure exposure levels of $p(t)$ and $p_{10}(t)$, respectively.

In the above equations, t = 0 corresponds to the start of the direct sound and ∞ should correspond to a time that is equal to or greater than the point where 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 (≥ 3 m) from the source ($L_{pE,d}$) may be measured and then $L_{pE,10}$ is obtained as follows:

$$L_{pE,10} = L_{pE,d} + 20\log(d/10)\ \text{dB}$$

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 26 As an alternative method, the reference sound pressure exposure level $L_{pE,10}$ can be measured in a reverberation room according to the following equation ([1], [2]):

$$L_{pE,10} = L_{pE} + 10\log(A/S_0)\ \text{dB} - 37\ \text{dB}$$

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;

$S_0 = 1\ \text{m}^2$

A can be obtained from the reverberation time in the room according to the following equation (Sabine's formula):

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

where

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

T is the reverberation time of the room in seconds.

NOTE 27 G can alternatively be measured by using a stationary omni-directional sound source as follows:

$$G = L_p - L_{p,10}$$

where

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

$L_{p,10}$ is that 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 (≥ 3 m) from the source ($L_{p,d}$) may be measured and then $L_{p,10}$ is obtained as follows:

$$L_{p,10} = L_{p,d} + 20\log(d/10)\ \text{dB}$$

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

When using an omni-directional sound source of which sound power level is known, G can be obtained as follows:

$$G = L_p - L_W + 31\ \text{dB}$$

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

Both the early decay time, EDT, and the conventional reverberation time, T, should be measured from the slope of the octave band integrated impulse response curves. The slope of the decay curve should be determined from the slope of the best fit linear regression line to the appropriate portion of the decay curve. The EDT should be obtained from the initial 10 dB of the decay and T is normally obtained from the portion of the decay curve between -5 dB and -35 dB below the maximum initial level (or -5 dB and -25 dB, see 6.2). The decay times are to be calculated from these slopes as the time required to decay 60 dB.

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 a 80 ms early time limit depending on whether the results are intended to relate to conditions for speech or music respectively.

$$C_{t_e} = 10\log\left(\int_0^{t_e} p^2(t)\,\mathrm{d}t \Big/ \int_{t_e}^{\infty} p^2(t)\,\mathrm{d}t\right)\ \mathrm{dB}$$

where

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

t_e is the early time limit of either 50 ms or 80 ms (C_{80} is usually named "clarity").

NOTE 28 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.

$$D_{50} = \int_0^{0{,}050s} p^2(t)\,\mathrm{d}t \Big/ \int_0^{\infty} p^2(t)\,\mathrm{d}t$$

This is exactly related to C_{50} by the following relationship:

$$C_{50} = 10\log\left(\frac{D_{50}}{1-D_{50}}\right)\ \mathrm{dB}$$

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:

$$T_s = \int_0^{\infty} t\,p^2(t)\,\mathrm{d}t \Big/ \int_0^{\infty} p^2(t)\,\mathrm{d}t$$

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 29 Speech intelligibility can also be determined by measuring the speech transmission index (STI). This quantity is measured by using special modulated noise signals which are not covered by the impulse response methods of this standard.

A.2.3.1 Early lateral energy measures

The fraction of the energy, LF, arriving within the first 80 ms that arrives from lateral directions can be measured from impulse responses obtained from an omni-directional and figure-of-eight pattern microphones.

$$\mathrm{LF} = \int_{0{,}005s}^{0{,}080s} p_L^2(t)\,\mathrm{d}t \Big/ \int_{0}^{0{,}080s} p^2(t)\,\mathrm{d}t$$

where

$p_L^2(t)$ is the instantaneous sound pressure in the auditorium impulse response measured with a figure-of-eight pattern microphone.

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, approximation for obtaining lateral energy fractions, LFC, with contributions which vary as the cosine of the angle, which is thought to be subjectively more accurate, can be used (see [3]).

$$\mathrm{LFC} = \int_{0{,}005s}^{0{,}080s} \left| p_L(t) \cdot p(t) \right| \mathrm{d}t \Big/ \int_{0}^{0{,}080s} p^2(t)\,\mathrm{d}t$$

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

Interaural cross correlation measures are also thought to relate to spatial impression, envelopment and perceived source width. They are described in annex B.

A.3 Measurement procedure

A.3.1 Source

The source shall be as omni-directional as possible. Table A.1 below lists the maximum acceptable deviations from omni-directionality 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 neighbour points. The reference value shall be determined from 360° energetic average in the measurement plane. The minimum distance between source and microphone shall be 1,5 m.

Table A.1 — Maximum allowed directional deviations of the source in decibels for excitation with octave bands of pink noise and measured in a free field

Frequency, Hz	125	250	500	1 000	2 000	4 000
Maximum deviation, dB	± 1	± 1	± 1	± 3	± 5	± 6

NOTE 30 For tests relating to conditions with a human speaker, a source with a directivity approximating a human speaker should be used. Dummy heads which comply with ITU Recommendation P. 58 may be used without an explicit check of the directivity pattern.

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.

A.3.2 Microphones

An omni-directional microphone should be used to measure the impulse response for all of the measures. For LF values, a figure-of-eight pattern microphone is also required. For G values the sensitivity of the omni-directional microphone shall be calibrated. For LF values, the relative sensitivities of the omni-directional and figure-of-eight microphone in the direction of maximum sensitivity should be calibrated in a free sound field.

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 omni-directional, 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. The use of Fast Hadamard Transforms and Maximum Length Sequence (MLS) signals is one successful correlation type approach (see [8]). 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 that 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 also the filtered signal continues on after the end of the unfiltered signal. This creates particular problems for measures such as C_{80} or LF where the short early time interval portions of the signals are filtered into octave bands.

The best approach that avoids the filter delay problems is to time-window the broadband impulse response before any filtering. The start of the impulse response for the equations of subclause 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 subclause A.2 are increased to include the energy delayed by the filters.

A good approximation to the above window-before-filtering approach can be obtained using a window correction (see [1]). If the impulse signals are first filtered into octave bands, the start of the integrations for the equations of subclause 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 must start from this trigger point and continue for t_e seconds plus half the filter delay time. The late time interval should start from the point t_e 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 shall 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 is 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 characterise 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 positions would be acceptable.

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.

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

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

The source should be at positions representative of those used by performances 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 influences the parameters by more then 5 % of the value of the parameter (or 0,5 dB in the case of C_x and G), the measurement shall be repeated with the source turned in at least three steps totally. The resulting parameters related to the different angles of the source shall be arithmetically averaged.

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.

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. Lateral energy fractions in the 4 000 Hz octave band are not usually thought to be subjectively important.

Annex B
(informative)
Binaural auditorium measures derived from impulse responses

B.1 Introduction

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. They are described in annex A.)

Spatial impression may be divided into two subclasses:

subclass 1: broadening of the source, i.e. "spaciousness";

subclass 2: state of diffusion of the reverberant sound field, i.e. "envelopment".

B.2 Definition of IACC

The normalised inter-aural cross correlation function, IACF, is defined as:

$$\mathrm{IACF}_{t_1 t_2}(\tau) = \left[\int_{t_1}^{t_2} p_\mathrm{l}(t) \cdot p_\mathrm{r}(t+\tau)\,\mathrm{d}t \right] \Bigg/ \left[\int_{t_1}^{t_2} p_\mathrm{l}^2(t)\,\mathrm{d}t \int_{t_1}^{t_2} p_\mathrm{r}^2(t)\,\mathrm{d}t \right]^{1/2}$$

where

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

$p_\mathrm{r}(t)$ is that for the right ear canal.

The inter-aural cross correlation coefficients, IACC, are given by:

$$\mathrm{IACC}_{t_1 t_2} = \max \left| \mathrm{IACF}_{t_1 t_2}(\tau) \right|, \quad \text{for } -1\,\mathrm{ms} < \tau < +1\,\mathrm{ms}$$

B.3 Measurement heads

B.3.1 Dummy head

A dummy head, with pinna and ear canals, should be chosen as standard for a given set of measurements. Dummy heads which comply with ITU Recommendation P.58 may be used without verifying the geometry or the acoustical performance. Selection and usage of dummy head shall be clearly stated in the test report and the direction of the dummy head shall be described in detail.

When making measurements in an auditorium the height of the ear canals of the dummy head above the floor should be about 1,2 m.

B.3.2 Real heads

Real heads may be used in place of the standard dummy head to obtain $p_\mathrm{l}(t)$ provided K_1 < [head breadth plus two times the difference between the head length and the distance from the ear entrance point (EEP) to the occipital wall] < K_2, where K_1 and K_2 are determined from comparisons with the dummy head such that the measured IACC for the real heads chosen correlate with those of the dummy head within $r = 0,85$ or better. Selection and usage of real heads should be clearly stated in the test report and the instructions given to persons and the type of microphones used should be described in detail.

B.4 Uses of IACC

The uses of IACC have not yet been accepted uniformly. As in the case of LF and LFC, the use of IACC and its subjective relevance is still subject to discussion and research. Likewise, different approaches have been suggested regarding the choice of the time limits t_1 and t_2 and the frequency filtering of the signals (see [2]).

The most general form of IACC is defined with $t_1 = 0$ and $t_2 = \infty$ (in room acoustics a time of the order of the reverberation time) and with a wide frequency band. As in the case of monaural measurements, IACC is generally measured in octave bands ranging from 125 Hz to 4 000 Hz.

IACC can be measured to describe the dissimilarity of the signal arrival at the two ears, either for the early reflections ($t_1 = 0$ and $t_2 = 80$ ms) or for the reverberant sound ($t_1 = 80$ ms and $t_2 =$ a time greater than the reverberation time of the enclosure).

B.5 Measurement procedure

The measurement procedure should, in general, parallel that given in annex A.

Annex C
(informative)
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