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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.