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Acoustics — Measurement of room acoustic parameters —

Part 1:
Performance rooms

Acoustique — Mesurage des paramètres acoustiques des salles —

Partie 1: Salles de spectacles

(Revision of ISO 3382:1997)

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Foreword

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

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

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

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

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

Part 1: Performance rooms;

Part 2: Reverberation time in ordinary rooms.

The existing International Standard ISO 3382:1997, *Acoustics – Measurement of the reverberation time of rooms with reference to other acoustical parameters* has been revised and made Part 1 in a new ISO 3382 on measurement of reverberation time. Part 1 contains the technical details of the measurement technique and the information for room acoustic measurements in performance spaces, including the measurement of other room acoustic parameters. The new Part 2 does not repeat the technical details of Part 1, but it deals with the measurement of reverberation time, only, in any kind of room.

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

Part 1 of this third edition cancels and replaces the second edition (ISO 3382:1997), which has been technically revised. Annex A has been extended with information about JND (Just Noticeable Difference), recommended frequency averaging and 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.

The 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.

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 binaural measurements in auditoria. Annex C introduces the support measures which have been found useful for evaluating the acoustic conditions from the musicians' point of view.

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 room acoustic parameters —

Part 1:

Performance rooms

1 Scope

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

ISO 3741, *Acoustics — Determination of sound power levels of noise sources using sound pressure — Precision methods in reverberation room*

ISO 18233, *Acoustics — Application of new measurement methods in building and room acoustics*

IEC 60268-1, *Sound system equipment — Part 1: General*

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 International Standard, the following terms and 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 can 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. 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

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 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 the initial level

NOTE 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. The range of evaluation should always start from a level 5 dB below the initial level.

3.6
states of occupancy

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 chairs for performers, music stands and percussion instruments etc. should preferably be present

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 1 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.

NOTE 2 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 3 In theatres, a distinction should 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 is described.

NOTE 4 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 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 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). 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 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.

4.2.2.2 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 optimisation 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 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 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 $10^{0,01n}$ within ± 2 %, where n is an integer including zero.

NOTE 1 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 2 Where a tape recorder is used then in the requirements in 4.2.2.3 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 3 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 $T/30$. 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 4.2.2.2, Note 2).

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, the time scale of the display should be adjusted 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 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 should 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

Source positions should be located where the natural sound sources in the room would be typically 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 Annex A and Annex 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 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. See A.4 for more details.

No microphone position shall be too close to any source position in order to avoid 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 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.

NOTE 1 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 2 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 3 If conditions of 4.3, Notes 1 and 2 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. 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 2000 Hz for the survey method. For the engineering and precision methods the frequency range should cover at least 125 Hz to 4000 Hz in octave bands, or 100 Hz to 5000 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 it is allowed to use a short excitation or an impulse signal as an alternative to the interrupted noise signal. However, in that case the measuring accuracy is less than stated in 4.3.1.

5.2.2 Averaging of measurements

The number of microphone positions used will be determined by the accuracy 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 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 [14]). It is important that the sound power emitted by the source is 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, 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 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. 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 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 covers the frequency bands to be measured.

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 ($t \rightarrow \infty$) and proceed to the beginning of the squared impulse response. Thus the decay as a function of time is

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

where

p is the impulse response.

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

$$\int_t^{\infty} p^2(\tau)\,\mathrm{d}\tau = \int_0^{\infty} p^2(\tau)\,\mathrm{d}\tau - \int_0^{t} p^2(\tau)\,\mathrm{d}\tau \tag{3}$$

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.

$$E(t) = \int_{t+T_0}^{t} p^2(\tau)\,\mathrm{d}(-\tau) \tag{4}$$

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 displayed using a dB scale, and calculate the decay curve from

$$E(t) = \int_{t_1}^{t} p^2(\tau)\,\mathrm{d}(-\tau) + C \tag{5}$$

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

For the determination of T_{20} the evaluated range for the decay curves is from 5 dB to 25 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. The formula for the least square method is given in Annex C. 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 $T_{20} = 60/d$. For the determination of T_{30} the evaluation range is from 5 dB to 35 dB.

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.

In order to specify a reverberation time, it is essential that the decay curves 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 T_{20} or T_{30}, respectively, can be estimated from

$$\sigma(T_{20}) = 0{,}88 \times T_{20} \sqrt{\frac{1 + 1{,}90/n}{N\,B\,T_{20}}} \qquad (6)$$

$$\sigma(T_{30}) = 0{,}55 \times T_{30} \sqrt{\frac{1 + 1{,}52/n}{N\,B\,T_{30}}} \qquad (7)$$

where

n is the number of decays measured in each position;

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

B is the bandwidth, in Hz.

The equations (6) and (7) are based on certain assumptions concerning the averaging device. Further information is given in Annex A.

For an octave filter B = 0,71 $\times f_c$, and for one-third-octave filter B = 0,23 $\times f_c$, where f_c is the mid-band frequency of the filter in Hz. 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 (see [5]). For practical evaluation of the measurement uncertainty using the integrated impulse response method it can be considered 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:

$$B\,T > 16 \quad \text{and} \qquad (8)$$

$$T > 2\,T_{det} \qquad (9)$$

where

B is the filter bandwidth in Hz;

T_{det} is the reverberation time 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.

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 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 one decade. The nominal midband frequencies for octave bands according to IEC 61260 should be marked on the frequency axis.

A single figure reverberation time $T_{30,\mathrm{mid}}$ can be calculated by averaging T_{30} in the 500 Hz and 1 000 Hz octave bands ($T_{20,\mathrm{mid}}$ 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) statement that the measurements were made in conformity with this International Standard;

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;

NOTE 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: 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;

f) 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, of the source and the 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 organisation.

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, but 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 (Hz)	Just Noticeable Difference (JND)	Typical range[a]
Subjective level of sound	Sound Strength, *G*, in dB	500 to 1000	1 dB	-2 dB; +10 dB
Perceived reverberance	Early Decay Time, EDT, in s	500 to 1000	Rel. 5 %	1,0 s; 3,0 s
Perceived clarity of sound	Clarity, C_{80}, in dB	500 to 1000	1 dB	-5 dB; +5 dB
	Definition, D_{50}	500 to 1000	0,05	0,3; 0,7
	Centre Time, T_S, in ms	500 to 1000	10 ms	60 ms; 260 ms
Apparent Source Width, ASW	Early Lateral Energy Fraction, LF or LFC	125 to 1000	0,05	0,05; 0,35
Listener Envelopment, LEV	Late Lateral Sound Level, LG, in dB	125 to 1000	Not known	-14 dB; +1 dB

[a] Typical range is for 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 omni-directional 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 at a distance of 10 m from the same sound source in a free field.

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

in which

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

and

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

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} and $L_{\mathrm{pE,10}}$ 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, $L_{\mathrm{pE,10}}$ 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 ($L_{\mathrm{pE,d}}$) may be measured and then $L_{\mathrm{pE,10}}$ is obtained as follows:

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

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 according to the following equation ([1], [2]):

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

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 m²

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

$$A = 0{,}16\ V / T \quad \text{(A.6)}$$

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 omni-directional sound source as follows:

$$G = L_p - L_{p,10} \quad \text{(A.7)}$$

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

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

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:

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

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

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 should 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.

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

where

$C_{\mathrm{t_e}}$ 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 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.

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

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

$$C_{50} = 10 \times \log\left(\frac{D_{50}}{1 - D_{50}}\right) \mathrm{dB} \qquad \text{(A.12)}$$

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:

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

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 IEC 60268 [15]. This quantity is measured by using special modulated noise signals which are not covered by the impulse response methods of this standard.

A.2.4 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.

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

where

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

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]).

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

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.2.5 Late lateral energy measures

The relative level of the late arriving lateral sound energy, LG, can be measured using a calibrated omni-directional sound source, from the impulse response obtained in the auditorium from a figure-of-eight pattern microphone.

$$LG = 10 \times \log \left[\frac{\int_{0{,}080\ \mathrm{s}}^{\infty} p_L^2(t)\,\mathrm{d}t}{\int_{0}^{\infty} p_{10}^2(t)\,\mathrm{d}t} \right], \text{in dB} \tag{A.16}$$

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 omni-directional 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 energy is calculated from:

$$LG_{\mathrm{avg}} = 10 \log \left[\frac{1}{4} \sum_{i=1}^{4} 10^{\mathrm{LG}_i/10} \right], \text{in dB} \tag{A.17}$$

where

LG_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 1000 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 shall be as omni-directional as possible. Table A.2 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 during these measurements.

Table 2 — 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 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 [16] 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 (see ISO 18233). 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 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 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 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 (see [1]). 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 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 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 maximum limits in Table 2, 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.3 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 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 3 — 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 frequency averaging stated in Table A.1 should be used and the index m (for weighting) should be applied to the symbol, e.g. G_m for the strength averaged in 500 Hz to 1000 Hz octave bands, and LF_m for the Early Lateral Energy Fraction averaged in 125 Hz to 1000 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 may be useful to average the results in some sections of the hall, like the stalls, first balcony etc. Some measures like the strength G tend to vary with the distance and a graphical plot of G as a function of source-receiver distance may 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. They are described in 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 IACC

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

$$IACF_{t_1,t_2}(\tau) = \frac{\int_{t_1}^{t_2} p_\mathrm{l}(t) \cdot p_\mathrm{r}(t+\tau)\,\mathrm{d}t}{\sqrt{\int_{t_1}^{t_2} p_\mathrm{l}^2(t)\,\mathrm{d}t \int_{t_1}^{t_2} p_\mathrm{r}^2(t)\,\mathrm{d}t}} \qquad \text{(B.1)}$$

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:

$$IACC_{t_1 t_2} = \max \left| IACF_{t_1 t_2}(\tau) \right|, \quad \text{for } -1\,\mathrm{ms} < \tau < +1\,\mathrm{ms} \qquad \text{(B.2)}$$

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 [16] 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 $p_l(t)$ 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 $t_2 = 0{,}08$ s) or for the reverberant sound ($t_1 = 0{,}08$ s and t_2 = a time greater than the reverberation time of the enclosure).

The *JND* (Just Noticeable Difference) of *IACC* is assumed to be 0,075.

B.5 Measurement procedure

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

Annex C
(informative)

Stage measures derived from impulse responses

C.1 General

In concert halls and other performance spaces it is important that the acoustic conditions allow the musicians to hear each other and that there is a sufficient response from the room. For an objective evaluation of these conditions it has proven to be useful to measure on the orchestra platform with the source and microphone close together [13]. Two different parameters can be derived from the measurements, see Table C.1.

Table C.1 – Acoustic parameters measured on orchestra platforms

Subjective listener aspect	Acoustic quantity	Single number frequency averaging (Hz)	Just Noticeable Difference (JND)	Typical range
Ensemble conditions	Early Support, ST_{Early} (dB)	250 to 2000	Not known	-24 dB; -8 dB
Perceived reverberance	Late Support, ST_{Late} (dB)	250 to 2000	Not known	-24 dB; -10 dB

C.2 Definition of measures

C.2.1 Early support

The ratio in dB of the reflected energy within the first 0,1 s relative to the direct sound at a distance of 1,0 m from the acoustic centre of an omni-directional sound source.

$$ST_{\text{Early}} = 10 \times \log \left[\frac{\int_{0{,}020\ \text{s}}^{0{,}100\ \text{s}} p^2(t)\,\text{d}t}{\int_{0}^{0{,}010\ \text{s}} p^2(t)\,\text{d}t} \right], \quad \text{in dB} \tag{C.1}$$

where

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

Early support relates to ensemble, i.e. ease of hearing other members of an orchestra.

C.2.2 Late support

The ratio in dB of the reflected energy after the first 0,1 s relative to the direct sound in a distance of 1,0 m from the acoustic centre of an omni-directional sound source.

$$ST_{\text{Late}} = 10 \log \left[\frac{\int_{0,100\ \text{s}}^{1,000\ \text{s}} p^2(t)\,\mathrm{d}t}{\int_{0}^{0,010\ \text{s}} p^2(t)\,\mathrm{d}t} \right], \text{in dB} \qquad \text{(C.2)}$$

where

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

Late support relates to perceived reverberance, i.e. the response of the hall as heard by the musician.

C.2.3 Measurement positions

The height of the source and the microphone should be the same, between 1,0 m and 1,5 m above the floor. At least three different positions of the source and receiver should normally be used. Measurements should preferably be made with chairs and music stands on the orchestra platform, but the nearest chairs and music stands within a distance of 2 m from the source and microphone should be removed in order not to reflect the sound directly to the microphone. Source and receiver positions and heights should be noted with the results.

C.2.4 Statement of results

The measurements are made in octave bands. The arithmetically averaged result in the four octave bands from 250 Hz to 2 000 Hz and in the three positions should be calculated as a single number result.

The standard deviation of the result in a single position in one octave band is estimated to be 1 dB. The standard deviation of the frequency- and position averaged single number result is estimated to be 0,3 dB.

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