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Acoustics — Application of new measurement methods in building and room acoustics

Acoustique — Application de nouvelles méthodes de mesurage dans l'acoustique des bâtiments et des salles

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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 18233 was prepared by Technical Committee ISO/TC 43, *Acoustics*, Subcommittee SC 2, *Building acoustics*.

Introduction

Stochastic signal analysis methods for the measurement of sound transmission phenomena started to be developed around 1960, but lack of available computing power excluded the use of these methods outside the best equipped research laboratories.

The development of digitizing circuitry, powerful personal computers and the use of digital signal processing components in sound measuring equipment for field use, have made the application of measuring equipment based on extended digital signal analysis readily available. Dedicated instruments, as well as specialized software used on general computers, currently apply such methods and are already widely used.

The new methods bring a number of advantages compared to the well-established classical methods, such as suppression of background noise and extended measurement range. However, there is also risk of unreliable results if certain guidelines are not followed. The new methods may demonstrate larger sensitivity to time-variations and change in the environmental conditions than the classical methods.

This International Standard is developed to give requirements and guidelines for the use of new measurement methods in building and room acoustic measurements, but can also be used in the construction of measuring equipment for the implementation of the methods.

As even an experienced user of equipment based on classical methods may be unaware of the difficulties and limitations for some applications of the new methods, the user is encouraged to develop a deeper understanding of the theoretical bases for the new methods. Instrument manufacturers are also encouraged to give further guidelines for applications and to make it an objective to design instruments that give warnings when results are not reliable.

This International Standard gives guidelines and requirements for the application of new methods for the measurement of sound insulation in buildings and building elements and for the measurement of reverberation time and related quantities. Reference is made to the standards for the classical methods regarding what to measure, the number and the selection of measurement points, and the conditions for measurements.

Acoustics — Application of new measurement methods in building and room acoustics

1 Scope

This International Standard gives guidelines and specifies requirements for the application of new methods for the measurement of the acoustic properties of buildings and building elements. Guidelines and requirements for selection of the excitation signal, signal processing and environmental control are given, together with requirements for linearity and time-invariance for the systems to be tested.

This International Standard is applicable to such measurements as airborne sound insulation between rooms and of façades, measurement of reverberation time and other acoustic parameters of rooms, measurement of sound absorption in a reverberation room, and measurement of vibration level differences and loss factor.

This International Standard specifies methods to be used as substitutes for measurement methods specified in standards covering classical methods, such as ISO 140 (all parts), ISO 3382 (all parts) and ISO 17497-1.

2 Normative references

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

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

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

3 Terms definitions and abbreviated terms

3.1 Terms and definitions

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

3.1.1
classical method
conventional method of measurement where the resulting sound pressure levels or decay rates are determined directly from the recorded responses to random noise or impulse signals

3.1.2
new method
measurement method in which various deterministic signals can be used to first obtain the impulse response of the system under test and from which the required sound pressure levels and decay rates can be obtained

NOTE The new methods may have additional, intentional features such as giving results under situations where no result is obtained by the classical method. The new methods may, for instance, be more immune to noise from other sources.

3.1.3
effective signal-to-noise ratio
signal-to-noise ratio
ten times the logarithm to the base 10 of the ratio of the mean-square value of the signal part caused by the excitation and obtained by the new method, to the mean-square value of the unwanted part of the signal obtained by the same method and caused by sources other than the excitation

NOTE 1 The effective signal-to-noise ratio is expressed in decibels.

NOTE 2 The effective signal-to-noise ratio is used as a substitute for the normal signal-to-noise ratio when establishing procedures for the new method based on a classical method.

3.1.4
peak-to-noise ratio
ten times the logarithm to the base 10 of the ratio of the squared peak value of the signal part caused by the excitation and obtained by the new method, to the mean-square value of the unwanted part of the signal obtained by the same method and caused by other sources than the excitation

NOTE The effective peak-to-noise ratio is expressed in decibels.

3.1.5
fractional-octave band
frequency range, in hertz, from lower to higher band edge frequency for a fractional-octave-band filter as specified in IEC 61260

NOTE Both full-octave- and fractional-octave-band filters are designated fractional-octave-band filters.

3.2 Abbreviated terms

MLS Maximum length sequence method

SS Swept-sine method

4 Designations

4.1 Maximum length sequence method (MLS)

An MLS method in accordance with this International Standard shall be designated as “ISO 18233–MLS”.

4.2 Swept-sine method (SS)

An SS method in accordance with this International Standard shall be designated as “ISO 18233–SS”.

5 Theory

5.1 General

The transmission of sound within a room as well as the transmission of sound between rooms may normally be considered as a close approximation to a linear and time-invariant system. The general theory applicable to such systems may therefore be used to establish the relationship between excitation and response for the sound transmission.

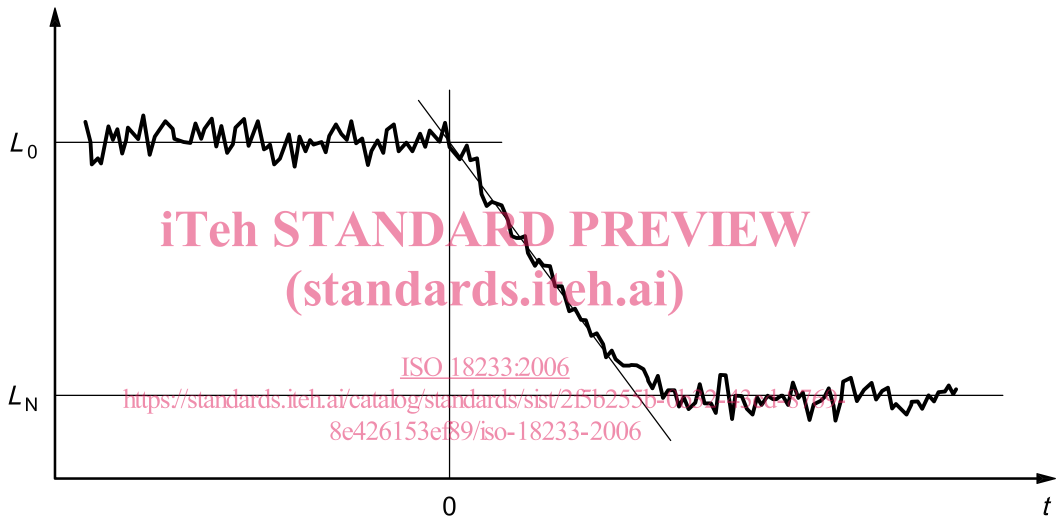
The impulse response is the basis of all measurements. The methods are applicable to the velocities measured on structures as well as to sound pressures measured in rooms.

5.2 Sound in a room

The scope of Parts 3 to 5 of ISO 140 and of Parts 9 to 12 of ISO 140 is to specify methods to measure the airborne sound insulation for building elements and the insulation between dwellings. ISO 3382 (all parts) specifies the measurement of reverberation time. In order to measure these quantities, the sound pressure level and the reverberation time in rooms by the application of noise excitation shall be measured.

For the measurement of reverberation time, the noise source is switched on for a time sufficient to obtain a steady level. The source is thereafter switched off, and the decay of the sound in the room is observed. The time for switching the noise off is set to $t = 0$ in this International Standard.

A recording of the sound pressure level versus time will, in general, contain information on the obtained stationary sound pressure level in the room as well as the reverberation time. A typical level versus time diagram is shown in Figure 1. The stationary sound pressure level before the sound source is switched off is given by the recording for $t < 0$, and information about the decay will be given for $t \geqslant 0$. The decay may be further processed to obtain the reverberation time.



Key

L_0 stationary noise level before the excitation is switched off

L_N background noise level

t time

NOTE The excitation is switched off at time $t = 0$.

Figure 1 — Typical level versus time curve

The classical methods for the measurement of airborne sound in rooms, defined in the ISO 140 and ISO 3382 series, specify a stochastic signal for the excitation. Although the room in most cases may be described as a deterministic system, statistical spread from the random excitation will lead to a certain stochastic variation in the result, which may be characterized by a standard deviation. Therefore, averaging of more measurements is normally needed to obtain results close to the stochastically expected values. Such averaging may for the classical method be combined with the spatial averaging needed to obtain a mean value for the room.

The methods described in this International Standard intend to obtain measurement values in fractional-octave bands. Requirements and guidelines are selected accordingly.

It has been shown (Reference [6]) that the expected decay in one particular observation point may be obtained without averaging, by processing the impulse response between the excitation signal (loudspeaker) and the observation point (microphone) directly. This holds for the decay curve and the stationary levels as long as the system is linear and time-invariant. The theory may be extended and applied to sound in the source room, to sound in the receiving room, and to the transmission from the source to the receiving room.

The measured response in the classical method based on noise excitation may, in theory, be described as a convolution between the excitation signal and the impulse response of the room. However, in the classical method with noise excitation, the response is recorded directly and information about the impulse response is normally not known.

According to the new methods described in this International Standard, the results may be obtained from processing of the impulse response itself.

NOTE 1 The impulse response is normally the combined impulse response of the system, consisting of amplifiers, transducers, applied filters, and the enclosure between the transmitting and the receiving points.

Several methods may be applied to obtain the impulse response or the frequency response function, which is linked to the impulse response by Fourier transformation. All such methods may be used if they are able to demonstrate reliable results within normal measurement conditions.

When a room has been excited by stationary white noise for a time sufficient to obtain stationary conditions and the noise is thereafter switched off at the time $t = 0$, the expected level at any time $t \geqslant 0$ will be [6]:

$$L(t) = 10 \lg \left[\frac{W_0}{C_{\text{ref}}} \int_t^{\infty} h^2(t)\, \mathrm{d}t \right] \text{ dB} \tag{1}$$

where

W_0 is a constant specifying the signal power per unit bandwidth of the excitation signal;

$h(t)$ is the impulse response;

C_{ref} is an arbitrary selected reference value for the level calculation.

The decay corresponds to the expected decay based on the classical method, which conventionally is approximated by a straight line.

NOTE 2 Due to the fact that the running time, t, is the lower start point for the integration, the operation in Equation (1) may be described as backwards integration. In an alternative form of the formula, the integral starts at $+\infty$ and runs backwards to the actual time. Historically, this was achieved using analog technology by playing a tape with the recorded response in the reverse direction.

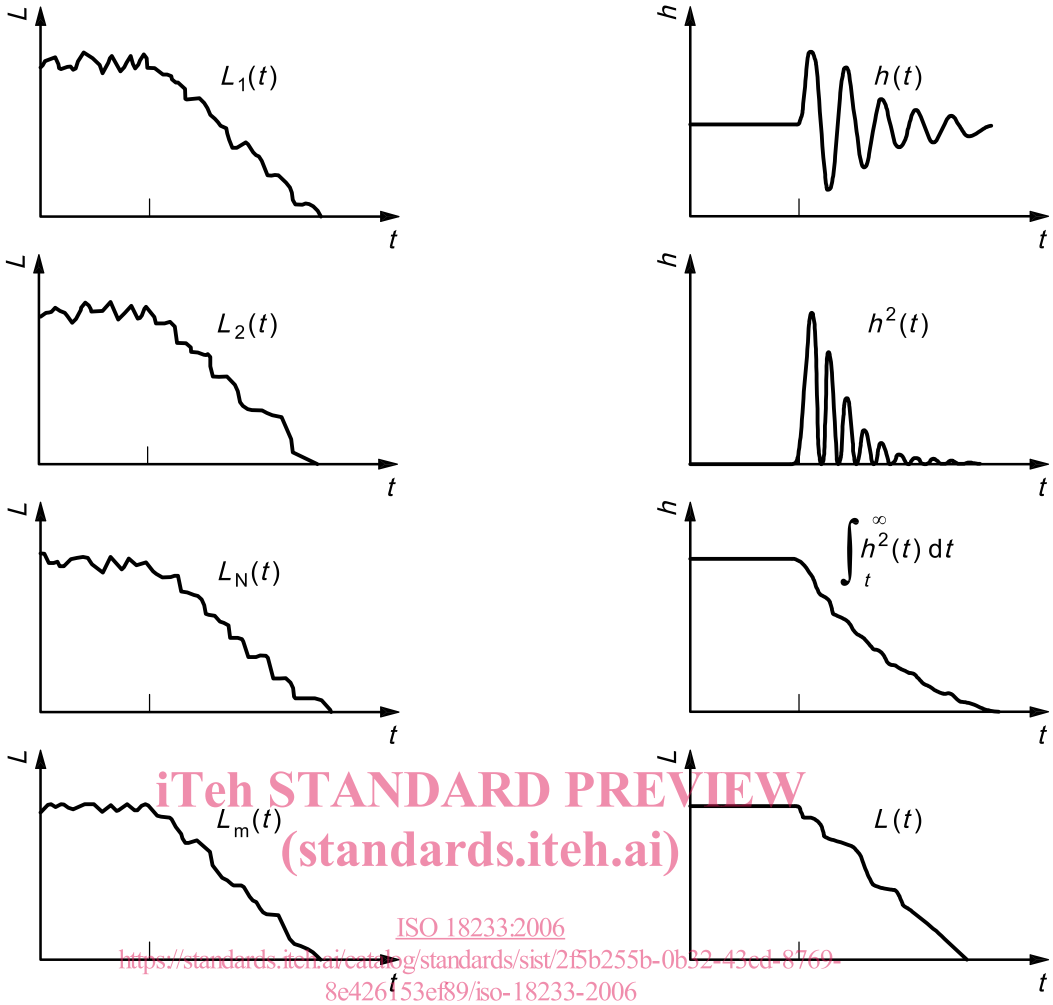
Equation (1) does not consider the extraneous noise normally accompanying a measurement.

When a fractional-octave-band filter is a part of the measured system, Equation (1) will describe the expected decay according to the classical method for the applied filter band.

Equation (1) may be used to compute the expected level at any time after the signal source was switched off. It may also be used to obtain the expected mean level before the excitation was switched off, L_0. The level may be obtained from Equation (1) by setting $t = 0$:

$$L_0 = 10 \lg \left[\frac{W_0}{C_{\text{ref}}} \int_0^{\infty} h^2(t)\, \mathrm{d}t \right] \text{ dB} \tag{2}$$

Figure 2 illustrates how the level versus time function is obtained by the classical and the new method.



a) Classical method **b) New method**

Key

L sound level

h impulse response

t running time

NOTE In the classical method, an approximation, $L_m(t)$, of the expected decay is found by averaging (ensemble) a number of individual decays, $L_1(t)$, $L_2(t)$, … $L_N(t)$, based on noise excitation. By application of the new method, the expected decay, $L(t)$, is found by processing the impulse response $h(t)$.

Figure 2 — Illustration of the difference between classical and new method

5.3 Sound transmission between two rooms

If a noise source is placed in a source room and the sound pressure level is measured at a point S, the expected level, L_1, may according to Equation (2) be obtained from the impulse response between the excitation point and the point S: $h_1(t)$.

$$L_1 = 10 \lg \left[\frac{W_0}{C_{\text{ref}}} \int_0^\infty h_1^2(t)\,\mathrm{d}t \right] \text{ dB} \qquad (3)$$

In a similar way, if the sound level is measured in an adjacent receiving room at a point R, the expected level, L_2, may be obtained from the impulse response between the excitation point and the point R: $h_2(t)$.

$$L_2 = 10 \lg \left[\frac{W_0}{C_{\text{ref}}} \int_0^{\infty} {h_2}^2(t) \mathrm{d}t \right] \quad \text{dB} \tag{4}$$

The expected sound level difference, D, between the source and the receiver room may therefore be computed as:

$$D = L_1 - L_2 = 10 \lg \left[\frac{\int_0^{\infty} h_1^2(t) \mathrm{d}t}{\int_0^{\infty} h_2^2(t) \mathrm{d}t} \right] \quad \text{dB} \tag{5}$$

The variable describing the excitation power, W_0, is eliminated in the result for the level difference as the arbitrary chosen reference C_{ref}.

NOTE The new methods specified in this International Standard can also be applied to the measurement of sound insulation of façades. In this context one of the measurement positions will be an outdoor position.

5.4 Using the frequency response function

A sinusoidal signal has a unique position in the theory of signals and linear time-invariant systems. If the transients formed when signals are switched on and off are disregarded, the response for such a system to a sinusoidal excitation will always be sinusoidal with the same frequency. The amplitude (gain) and the phase may, however, change. Information about the change of amplitude and phase between input and output as a function of frequency is called the frequency response function of the system. The frequency response function will, as the impulse response, give full information about the response to any input signal. The frequency response function may be obtained from the impulse response by Fourier transformation.

Equation (2) may be transformed by the application of Parseval's theorem:

$$W_0 \int_0^{\infty} h^2(t) \mathrm{d}t = \frac{W_0}{2\pi} \int_{-\infty}^{\infty} |H(\omega)|^2 \, \mathrm{d}\omega \tag{6}$$

where

ω is the angular frequency;

$H(\omega)$ is the frequency response function obtained by the Fourier transformation of the impulse response $h(t)$:

$$H(\omega) = \boldsymbol{F}\{h(t)\} = \int_{-\infty}^{\infty} h(t) \mathrm{e}^{-\mathrm{j}\omega t} \, \mathrm{d}t \tag{7}$$

where $\mathrm{j} = \sqrt{-1}$

NOTE In Equation (6), it is assumed that $h(t) = 0$ for $t < 0$, which will be the case for a physical, causal system.