New Measurement Methods in Building Acoustics

Ole-Herman Bjor

Norsonic As, P.O. Box 24, N-3421 Lierskogen, Norway
ohbjor@norsonic.com

Summary

ISO TC43/SC2 is currently working with a new standard covering new measurement methods for the measurement of acoustic properties of buildings and building elements. Guidelines and requirements for selection of excitation signal, signal processing and environmental control are given together with requirements for linearity and time-invariance for the systems to be tested.

The main applications of the new measurement method are for the measurement of sound insulation in buildings and building elements and for the measurement of reverberation time and related quantities. It also covers measurement of vibration level differences and loss factor

The development of digitising circuitry, powerful 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 analysis readily available. The new methods bring a number of advantages compared to the well-established classical methods, such as suppressions of background noise and extended measurement range. However, there is also risk of unreliable results if the guidelines are not followed.

The final draft for the ISO-standard is expected to be circulated for voting among the member bodies within a few months. The paper gives a brief description of the methods covered by the standard.

Introduction

Historically, sinusoidal signals were used for measurement of the most common building acoustic properties like airborne sound insulation and reverberation time. However, as the obtained result may change considerably by even a small change in the frequency due to the narrow resonances or modes of the room, it was soon realised that band limited noise, with a bandwidth of one- or one-third octave, was more convenient. By this method the average value for a certain frequency range was obtained directly. The wanted property for the noise was its spectral distribution, the unwanted was the stochastically distribution of the result due to the randomness of the excitation.

In most cases, the system to be measured can be regarded as deterministic, linear and time invariant. This allows general signal theory to be applied for the measurement. 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 most equipped research laboratories. The recent development of digitising 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 specialised software used on general computers, are currently applying such methods and already widely used.

ISO, the International Standardisation Organisation, TC 43, has established a work group with the task to develop requirements and guidelines for the use of new measurement methods in building acoustic measurements. A draft for the standard was circulated during the spring 2005 to member bodies for comments and voting and was in general well received.

The proposed standard gives guidelines and requirements for application of new measurement methods for the measurement of acoustic properties of buildings and building elements. Guidelines and requirements for selection of excitation signal, signal processing and environmental control are given together with requirements for linearity and time-invariance for the systems to be tested. The standard is applicable to such measurement as airborne sound insulation between rooms and of façades, measurement of reverberation time and other acoustic parameters of rooms, measurement of sound absorption in reverberation rooms and for the measurement of vibration level differences and loss factor. Since the measurement of the impulse response of the room is a crucial part of the method, the method may also be used for other purposes such as auralisation.

The impulse response

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

The new methods is characterized by using various deterministic signals to first obtain the impulse response of the system under test from which the required levels and decay times can be obtained by digital signal processing. The basis is the fact that a linear, time invariant transmission system is fully described by its impulse response. The methods are applicable to sound pressures measured in rooms as well as to the velocities measured on structures.

General theory

Sound in a room

The scope of the International Standards ISO 140, Part 3 – 5 and Part 9 - 12 is to specify methods to measure the airborne sound insulation for building elements and the insulation between dwellings. In order to measure these quantities, the level and the reverberation time in rooms by the application of noise excitation has to 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. Without loss of generality, the time for switching the noise off may for this description be set to $t = 0$.

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

The classical methods for the measurement of airborne sound in rooms, defined in the ISO 140-series of International Standards 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.

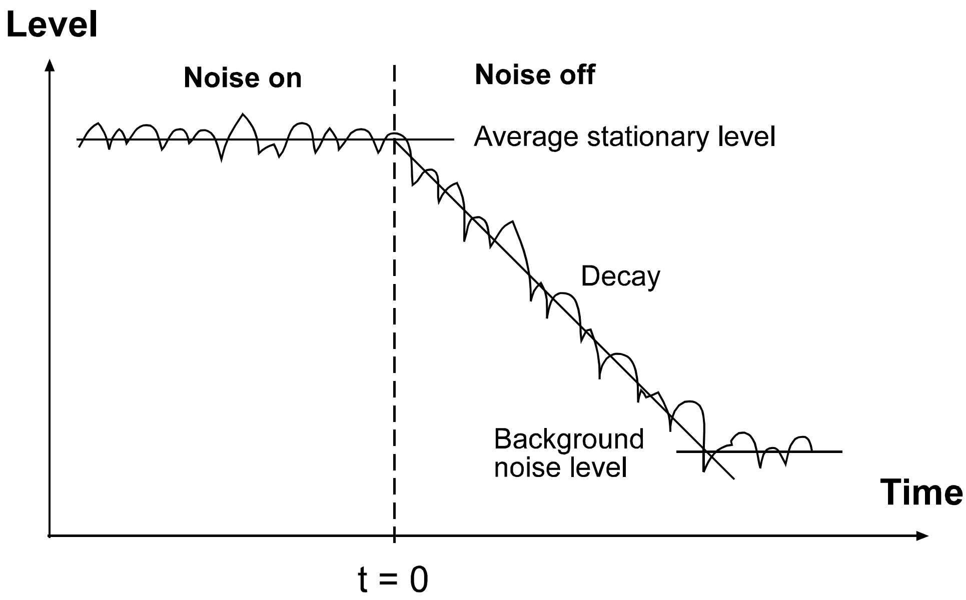

Figure 1 — *Typical level versus time curve*

M.R. Schroeder (See Bibliography [1]) has shown 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 receiver room.

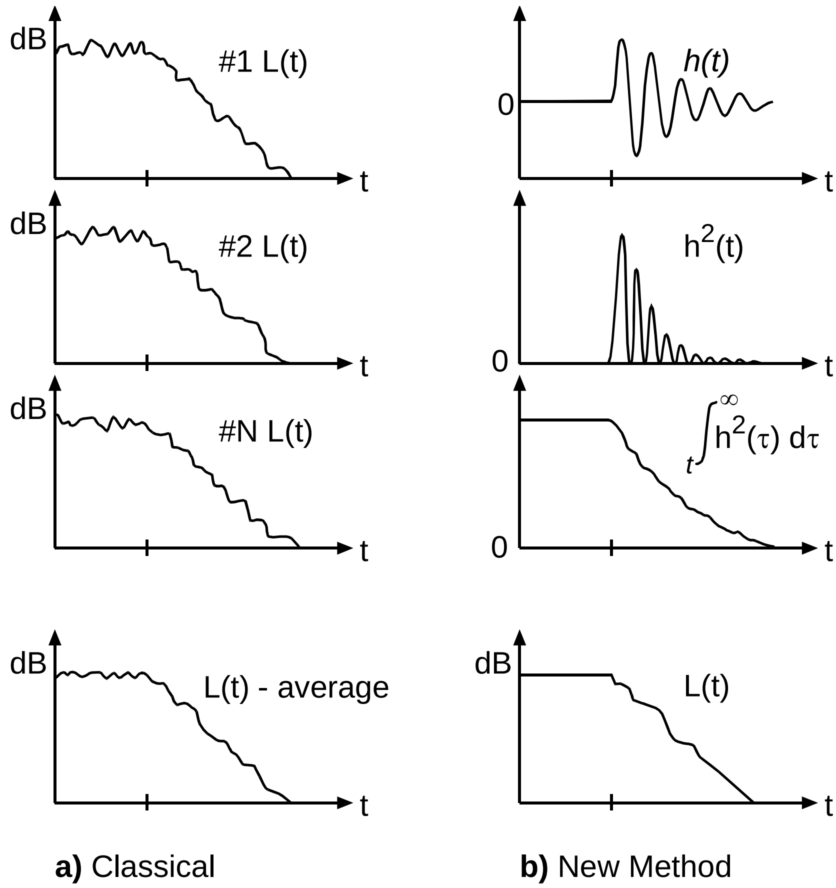

Figure 2. *Illustration of the difference between classical and new method. a) In the classical method the expected decay is found by averaging (ensemble) a number of individual decays based on noise excitation. b) By application of the new method, the expected decay is found by processing the impulse response*

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[1] is normally not known.

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.

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 $t \geq 0$ will be (See Bibliography [1]):

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

[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.

where W_0 is a constant specifying the signal power per unit bandwidth of the excitation signal, $h(t)$ is the impulse response and C_{ref} is an arbitrarily selected reference value for the level calculation.[2]

Equation (1) may be used to compute the expected level at any time after the signal source was switched off and thus give information about the reverberation. The equation may also be used to obtain the expected mean level just before the excitation was switched off, L_0. The level may be obtained from equation (1) by setting $t = 0$.

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

Sound transmission between two rooms

If a noise source is placed in a source room and the sound 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*: $h_1(t)$.
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*: $h_2(t)$.
The expected sound level difference between the source and the receiver room D, may therefore be computed as:

$$D = L_1 - L_2 = 10 \times \lg\left[\frac{W_0}{C_{\text{ref}}}\int_0^{\infty} {h_1}^2(\tau)\,\mathrm{d}\tau\right] - 10 \times \lg\left[\frac{W_0}{C_{\text{ref}}}\int_0^{\infty} {h_2}^2(\tau)\,\mathrm{d}\tau\right] = 10 \times \lg\left[\frac{\int_0^{\infty} h_1^2(\tau)\,\mathrm{d}\tau}{\int_0^{\infty} h_2^2(\tau)\,\mathrm{d}\tau}\right] \quad \text{dB} \tag{3}$$

Measurement of the impulse response

General

The impulse response for a room will typically be a signal with a large number of periods. The envelope of the signal will be irregular but typically have a fast attack-time and an exponential decay.

The impulse response may be measured as the response of the room to a very short acoustic pulse. However, it will in most cases where sources other than a loudspeaker are used, be difficult to have sufficient control of the spectral content and the directional characteristics of the excitation. Therefore, the impulse response is in most practical cases obtained by digital signal processing. The room is excited by a known signal for a certain time and the impulse response is calculated from the response to the excitation. The excitation signal is often distributed over a longer period of time to

[2] Due to the fact that the running time, t, is the lower start point for the integration, the operation of the equation may be described as backward integration. In an alternative form of equation (1), the integral starts at +∞ and runs backward to the actual time. Historically, this was achieved using analogue technology by playing a tape with the recorded response in the reversed direction.

increase the total radiated energy. This procedure will enhance the achievable dynamic range and reduce the influence of extraneous noise.

Several methods for the measurement of the impulse response are described in the literature. Most of the methods involve deconvolution techniques to obtain the impulse response. For measurements of the impulse response, movement of the source or the microphones are not acceptable as it will violate the requirement to time invariance. The impulse response of a room is formed by a complex interaction of sound waves reflected between the floor, ceiling and walls of the room. Between the reflections, the air in the room influences the transmission. Movement of the air or change in the speed of sound (temperature) may also violate the requirement for time-invariance.

Maximum Length Sequence Method

The Maximum Length Sequences (MLS) and the Hadamard transformation are often used to obtain the impulse response. A Maximum Length Sequence is a binary sequence. When used for excitation, the binary values are output at a fixed rate, which normally is equal to the sampling frequency for the recorded response. Although the sequence is deterministic, it sounds like white noise and each of the binary values appears in a random-like manner.

The MLS is characterized by an order N given by a whole number. The length of the sequence is $l_1 = 2^N - 1$. The autocorrelation of the sequence will almost be a periodic delta-pulse when the sequence is replayed periodically. The signal will thus be an approximation to a repeated replay of a recorded white noise signal. The measured impulse response will also be periodic; this means that the tail of the response outside the record will be folded back to the beginning of the record [2].

When using a periodic Maximum Length Sequence excitation signals, the calculation process for obtaining the impulse response may be done very efficiently as a Fast Hadamard Transform (FHT).

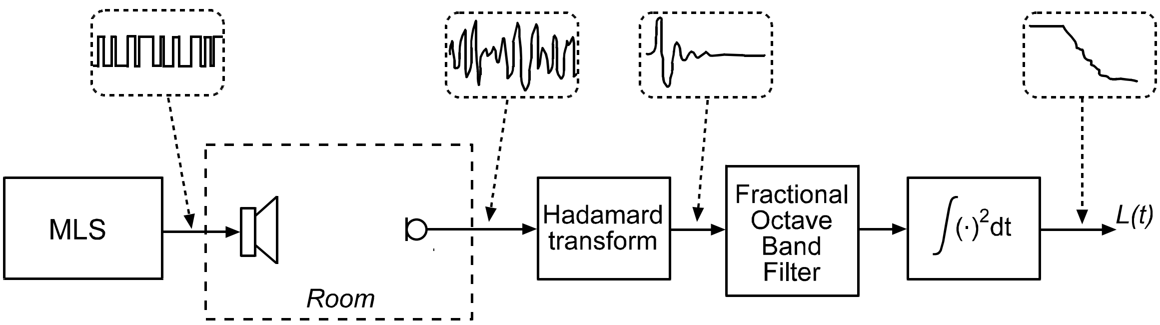

Figure 3 *Maximum Length Sequence Method*

The FHT consists of combining different samples in the recorded response by additions and subtractions. Included in the method is the addition of one extra sample in the record so the length of the output sequence is a power of two value.

The output from the Hadamard transform will be the impulse response for the measured system. The impulse response has to be further processed to obtain the fractional-octave-band filtered response. See Figure 3.

Swept-Sine Method[3]

Generally speaking, any kind of excitation signal can be used to determine the impulse response and respective frequency response function of any linear and time-invariant system, provided that it contains enough energy at every frequency of interest. The impulse response can be obtained from the response to the excitation by deconvolution, or the frequency response function can be obtained by dividing the output spectrum of the system under test by the spectrum of the input. The last implies Fourier-transformation of the input- and output-signal in order to perform the division in the spectral domain.

Using sinusoidal sweeps as the excitation signals offers a couple of crucial advantages compared to the MLS-method. The obtainable advantages include reduced sensitivity to environmental stability and elimination of the deterioration of the effective signal-to-noise ratio due to harmonic distortion. As all harmonic distortion may be deleted from the results, the sinusoidal excitation signal can be fed with substantially more power than MLS-signals. At quiet sites, sweep measurements can provide signal-to-noise ratios in excess of 100 dB.

Measurements with sweeps are less vulnerable to the deleterious effects of time variance. In outdoor measurements, these frequently occur due to air movement. Under windy weather conditions, sweeps are sometimes the only viable option when measuring impulse responses over long distances.

A single sweep, from the lower to the higher frequency, is in general recommended.

Sweep duration

In contrast to measurements with periodic excitation signals, there are no special requirements relating the sweep duration to the expected reverberation time that have to be considered. Anything from short chirps to sweeps many times longer than the reverberation time may be used. However, the acquisition time for recording the sweep response has to be longer than the sweep itself to collect the reverberated parts of the response.

In room and building acoustics, the reverberation time is normally longest for the lower frequencies. When very long sweeps (many seconds) are being used, the final gap only has to accommodate the reverberation at the highest frequencies, which generally is quite short. This holds because all the lower frequency components arrive while the excitation signal is still sweeping upwards.

Increasing the sweep duration brings more acoustic energy into the room to be measured and thus increases the effective signal-to-noise ratio. Generally a prolonged sweep should be preferred over averaging as it reduces the vulnerability to time variance and eases the separation of the distortion products.

Sweep Generation

The spectral contents of the sweep may be modified by the change of the amplitude as well as the instantaneous sweep speed. In most cases, it is advantageous to keep the amplitude at a constant value and let the sweep speed be changed with the frequency. The sweep is started at or below the lowest band-edge frequency of the lowest fractional-octave-band to be measured and continues upwards to at least the upper band-edge frequency of the highest fractional-octave-band to be

[3] A number of different names have been used in the literature to describe the swept-sine method. This includes “chirp”, “sinusoidal sweep” and “time-stretched pulse”

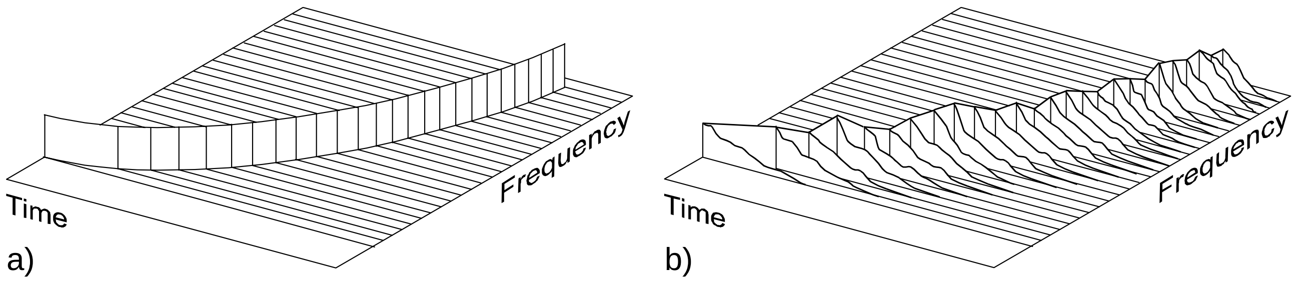

Figure 4 *Time frequency diagram for an exponential sweep. Part a) shows the excitation and b) the response.*

measured. An extension of the sweep with a quiet period is normally required and belongs to the excitation signal.

A linear sweep with constant amplitude corresponds to equal energy per hertz and is normally designated a white spectrum. If the frequency increases exponentially with the time, the time to sweep each octave is constant. The energy per fractional-octave-band will therefore be constant and the sweep mimics a pink spectrum. An exponential sweep (often also called logarithmic) is the normal excitation signal corresponding to pink noise in the referred classical methods.

Recording the response

The response to the sweep excitation has to be recorded from the start of the sweep to a time where the sound delayed by the reverberation is received. The time needed depends on the sweep speed, the frequency range to be covered and the reverberation of the room(s).

Figure 4 shows a time–frequency plot of an exponential sweep excitation and the corresponding response. Note that the received frequency components are delayed due to the reverberation.

Deconvolution

The process to obtain the impulse response of the room with the swept-sine method is illustrated on Figure 5. The complex frequency response function may be obtained by direct deconvolution or by spectral division between the spectrum of the response and the spectrum of the excitation.

When the excitation is a sweep from lower to higher frequencies, the response to harmonic components will appear before the main excitation at the same frequency. After the linear

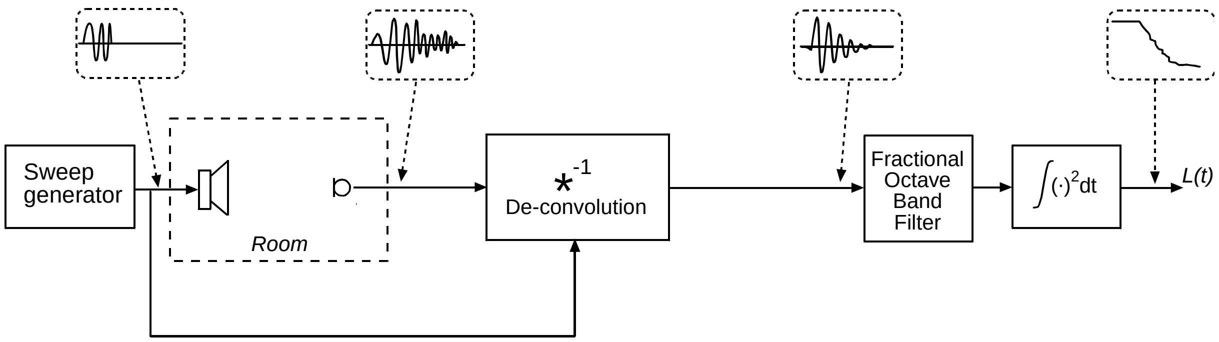

Figure 5 *The-swept sine method*

deconvolution, the responses to harmonic components in the excitation will appear at negative time and may easily be removed. See Bibliography [3] for further information.

Conclusion

Application of new measurement methods in building acoustics brings a lot of crucial advantages compared to classical methods such as suppression of background noise and extended measurement range. By application of the swept-sine method, even effects due to unlinearities, like those from a loudspeaker, may be removed. 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 changes in the environmental conditions than the classical methods.

References

[1] Schroeder, M.R., "New Method of Measuring Reverberation Time". J. Acoust. Soc. Am., vol. 37 (1965) pp. 409 – 412.

[2] Borish, J. and Angell, J.B., "An Efficient Algorithm for Measuring the Impulse Response Using Pseudorandom Noise". J. Audio Eng. Soc., vol. 31 (1983) pp. 478 – 488.

[3] Müller, S. and Massarini, P.: “Transfer-Function Measurement with Sweeps”. J. Audio Eng. Soc., vol. 49 (2001) pp. 443-471.

[4] ISO DIS 18233: “Acoustics - Application of new measurement methods in building acoustics”, ISO TC43/SC2

Fig. 6 *Example of an instrument applying the new method for building acoustic measurement. Norsonic Sound analyser Nor121 may be equipped with the swept-sine option.*