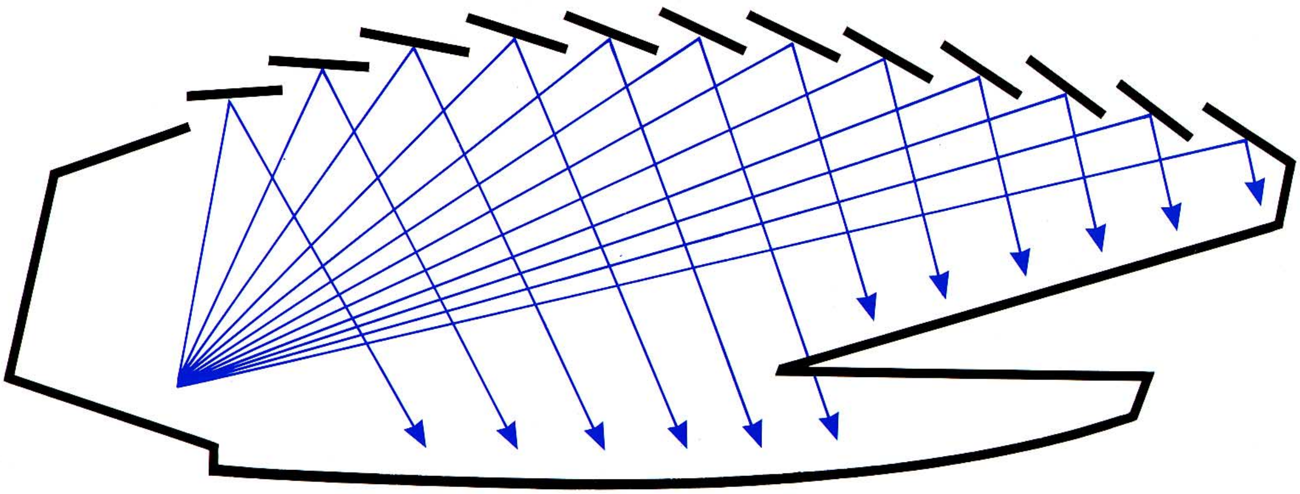
Measurements in Building Acoustics



Brüel & Kjær

This booklet answers some of the basic questions asked by the newcomer to building acoustic measurements. It gives a brief explanation of the following:

See page

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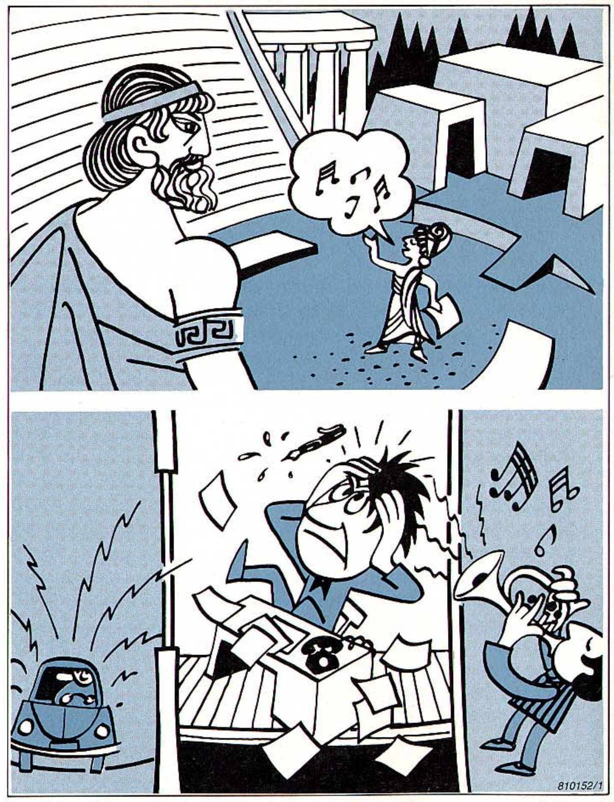
Introduction

The influence of acoustics on the design of buildings can be observed through the ages from Roman amphitheatres to the modern houses or buildings in which we spend our working hours and our leisure. The great difference, however, between life in ancient Rome and life in our crowded modern cities is the presence of noise from an ever increasing number of sources, from neighbours, traffic and industry.

Consequently, the science of building acoustics is no longer limited to the acoustic design of theatres, but has increased in scope to cover noise control and abatement in all types of buildings.



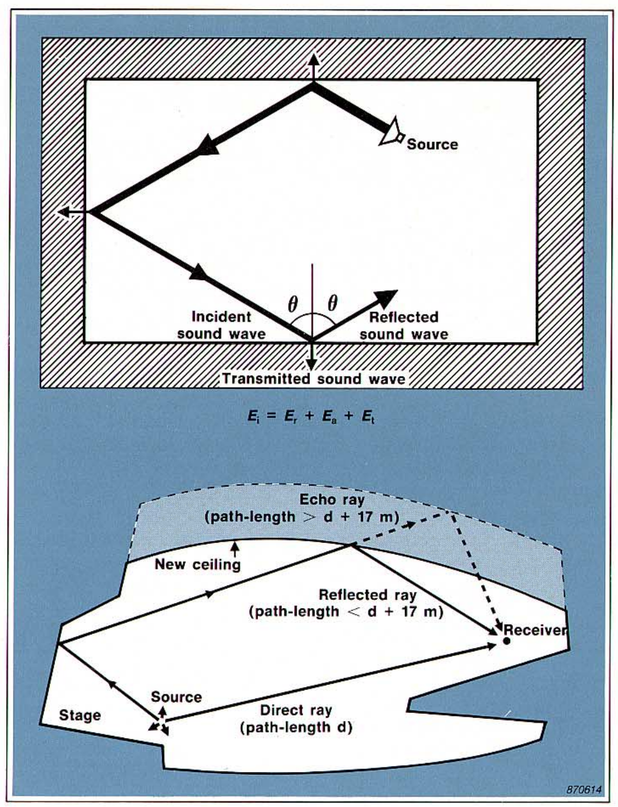
Behaviour of Sound in a Room

A knowledge of the behaviour of sound in a room is necessary if we wish to adapt the room for speech or music and if we want to attenuate external noise. Consider the effect of placing a sound source in a room. When sound energy (E_i) from the source strikes a room boundary, the reflected sound energy (E_r) contributes to the sound-field in the room, the absorbed sound (E_a) dissipates as heat, and the transmitted sound energy (E_t) propagates away through the boundary layer.

Reflection of Sound
If the wavelength of an incident sound-wave is much smaller than the dimensions of the reflecting surface, then the angle of reflection of the sound-wave equals the angle of incidence. We can use this geometrical behaviour to predict the pattern of sound rays in a room, a limitation being that only the primary and possibly the secondary reflections can be studied before the reverberant field begins to mask the ray paths.

In larger rooms such as concert halls, 'ray tracing' can identify problematic echoes, an echo being defined as a reflection which arrives more than 50 ms after the direct sound. An echo can also be thought of as a reflected ray with a path-length that is at least 17 m longer than that of the direct ray. Echo problems in large enclosures are solved by reducing the path length of the reflected ray. This can be done either by lowering the ceiling or by suspending reflectors from the ceiling.

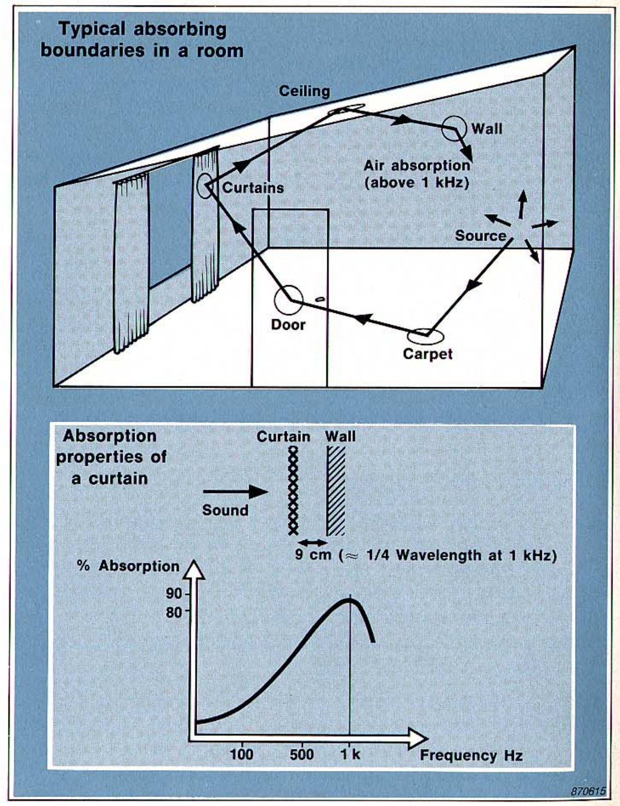
By observing the behaviour of the reflections in a room, we can control subjective properties such as *intimacy*, the quality of which depends on early arrival of reflections after the direct sound, and *diffusion* which is the evenness of the reverberant field.



Absorption of Sound
We can understand the effect of absorption by measuring, at a given position in a room, the sound pressure level caused by a steady sound power source. Instead of rising indefinitely as an increasing number of reflections arrive at the measuring position, the sound pressure level soon stabilizes. This must mean that the rate of energy input is exactly compensated by the rate at which the energy is absorbed by the different surfaces of the room. If more absorption material is put in the room, the sound pressure level is less because the energy in the reflections is reduced.

Typical absorbing surfaces in a room include carpets and curtains. These are simple porous absorbers which absorb sound energy by restricting the movement of air particles, the frictional forces causing the dissipation of energy as heat. Porous absorbers are most effective when placed at a point on the sound-wave which has maximum particle velocity. This position is a quarter wavelength away from a reflecting surface (when a wave is incident at right-angles) and is therefore frequency depedent. A carpet is an example of a porous absorber close to a reflective boundary. It absorbs best at high frequencies because the dimensions of the quarter wavelengths are then comparable with the thickness of carpet.

Other surfaces in the room absorb different frequencies to different extents, and by controlling the proportions of these absorbers it is possible to adjust the *warmth* of a room for music, or its *clarity* for speech.

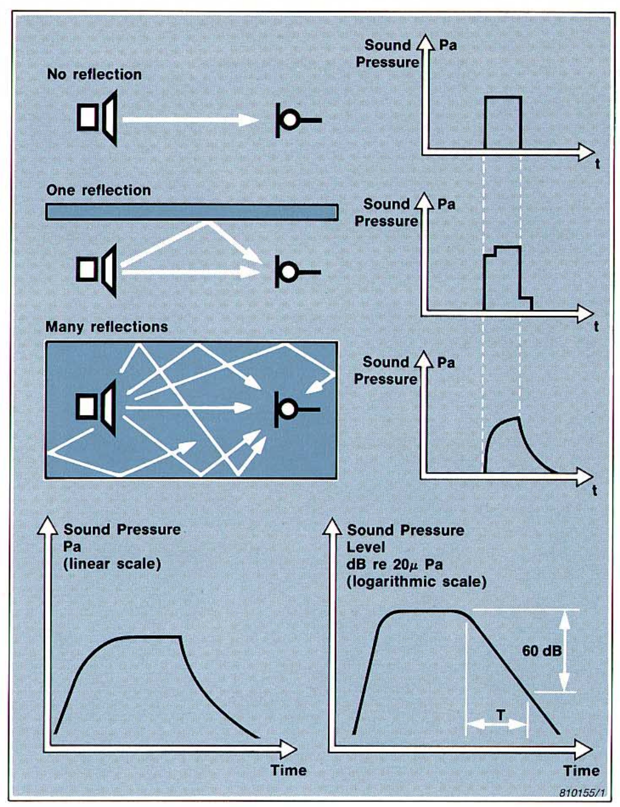


Build-up and Decay of Sound in a Room

If we position a microphone in a room and then switch on a steady sound-source, we notice that the sound pressure level does not immediately reach a steady level. This is because the first reflection and subsequent reflections take a finite time to reach the microphone.

In the resulting equilibrium state, interference between the sound-waves causes a spatial distribution of pressure maxima and minima which can be detected by moving the microphone around the room. These natural resonances or *normal room modes* are associated with the geometry of the room and the wavelengths emitted by the sound-source. Interesting consequences of these modes are that pressure doubling occurs at reflective boundaries, and that since all the room modes have antinodes at the corners of the room, they can all be "driven" by a sound-source placed there.

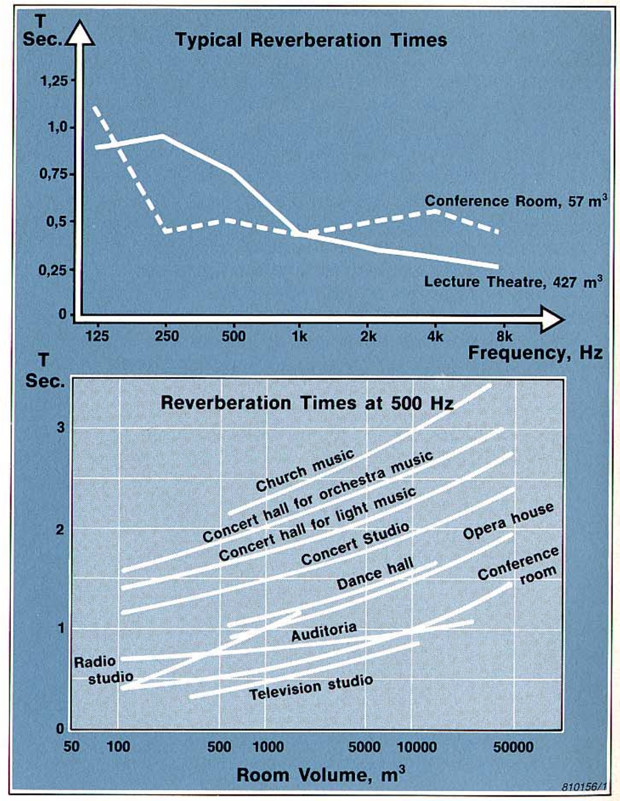
If the sound-source is now switched off, the collection of decaying room modes is called the reverberant sound-field. The rate of decay depends on the amount and positioning of absorption in the room. *Reverberation Time* is defined as the time taken for the sound pressure level in a room to decay by 60 dB. This corresponds to a decrease in sound pressure by a factor of 1000.



Importance of Reverberation Time in the Design of Rooms and Auditoria

In a room with highly reflecting surfaces, such as a bathroom, the reverberation time is relatively long, while in an anechoic chamber where all the walls, the ceiling and the floor are covered by a highly absorbent material, the reverberation time is nearly zero. The absorption of different materials varies widely with the frequency of the incident sound and the angle of incidence. It follows that the reverberation time is liable to vary with frequency. Generally, the reverberation time is longer at lower frequencies because these are usually less effectively absorbed than higher frequencies.

It is important that the reverberation time suits the intended use of the room. Too long a reverberation time renders speech less intelligible and music more cacophonous and produces higher background noise levels. A short reverberation time deadens background noise, but muffles speech and makes music sound "thin" and staccato.



Sabine's Formula for Reverberation Time

Reverberation time is related to the volume and the total absorption of a room. The relation has been empirically stated by Sabine and gives a good indication of the behaviour of most of the rooms we encounter daily. It is not suitable for a room with very absorbent boundaries such as an anechoic chamber.

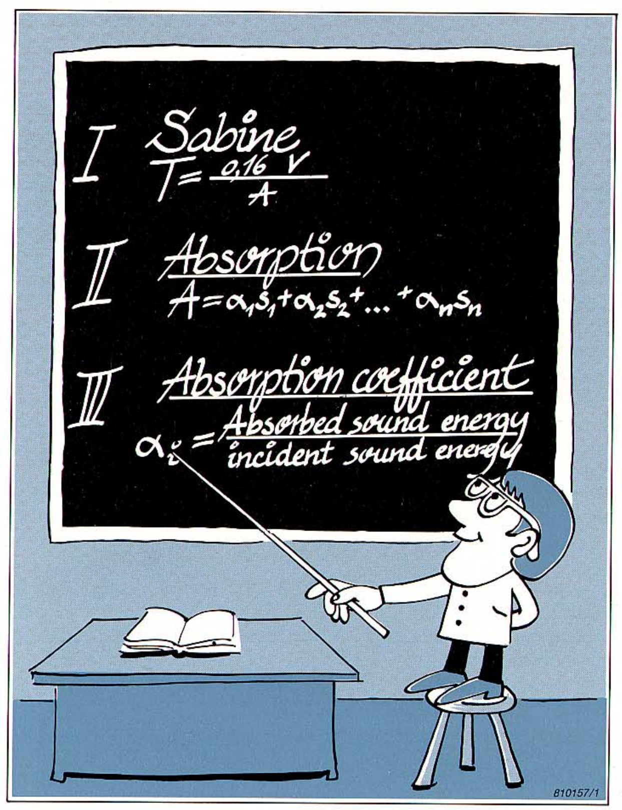
In relationship (I)

- T is the reverberation time, s
- V is the volume of the room, m^3
- A is the absorption of the room, m^2
- 0,16 is an empirical constant, s/m

The absorption of a room is obtained by summing the absorption of all the surfaces in the room, i.e. walls, ceiling, floor and all the furniture in the room. The absorption of each surface is the product of the area of the surface with its absorption coefficient, α_i, which is the ratio of the sound energy absorbed by the surface to the incident sound energy (relationship III). The absorption coefficient depends not only on the material but also on the frequency and the angle of incidence of the sound energy.

In relationship (II)

- A is the total absorption of the room
- $\alpha_1, \alpha_2, ..., \alpha_n$ are the absorption coefficients of the different surfaces of the room
- $S_1, S_2, ..., S_n$ their respective areas in m^2.



Measuring the Reverberation Time

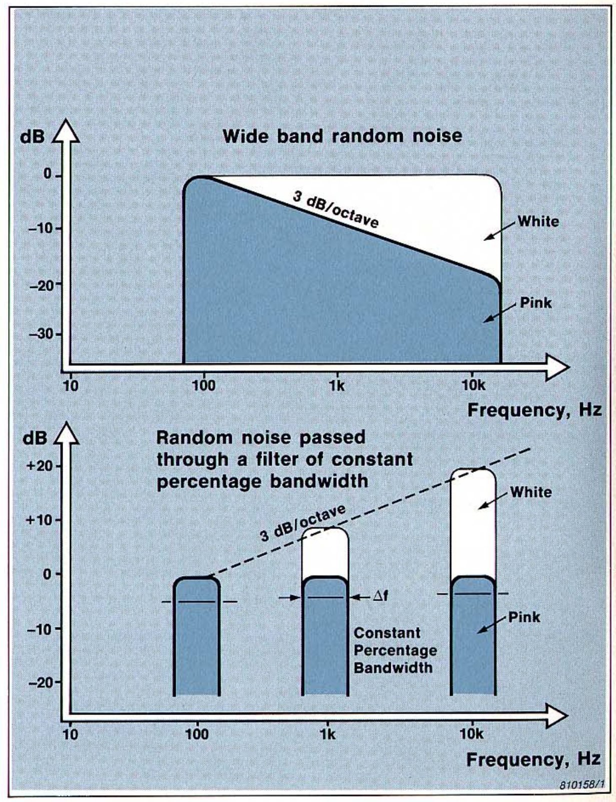
To measure the reverberation time one needs a sound-source to generate sound within the room and a receiving section to monitor the decay in sound pressure level after the sound-source ceases.

The Sound-Source
A starting pistol is a practical sound-source, but a pistol shot lacks both energy in the low frequency regions and reproducibility. A better way of excitation is to use a loudspeaker emitting noise in frequency bands. For a given power amplifier, this allows more energy to be transmitted into the room than with the starting pistol (which is important when high levels of background noise are present).

"White" noise is a wide band of random noise (i.e. a signal containing all the frequencies of the spectrum with a random amplitude distribution) with a constant level per Hertz over the entire frequency spectrum. "Pink" noise is a wide band of random noise with a level decreasing by 3 dB per octave. This attenuation is necessary to allow a constant energy to be transmitted through a filter with a bandwidth which becomes progressively wider (e.g. an oct. or 1/3 oct. filter), doubling the width for each octave.

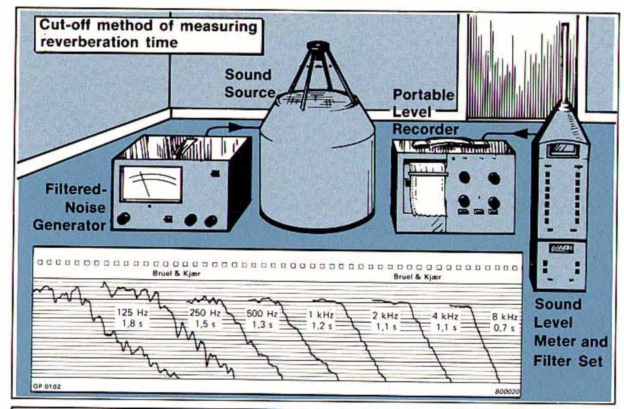
Due to the presence of background noise, it is seldom possible to measure the full 60 dB reverberation decay and one has to be content with a 40 dB, 30 dB or even 20 dB decay extrapolated to 60 dB. It is usual to specify the decay over which the reverberation time was measured, e.g. $T_r(30)$, $T_r(20)$.

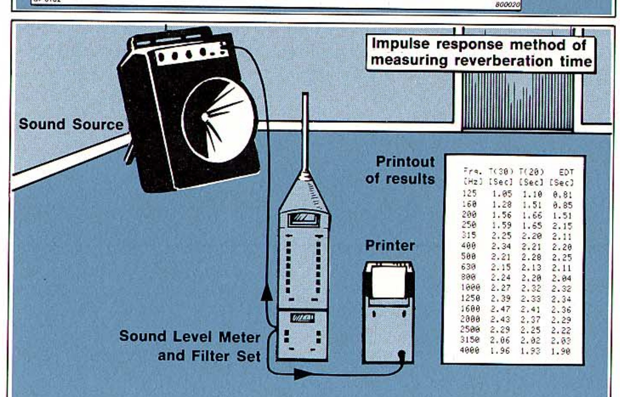
The noise can either be transmitted as a steady sound which is then cut off, or as a short pulse, the two methods having different receiving section requirements.



The Receiver
A typical receiving section may consist of a sound level meter fitted with an octave or a 1/3 octave filter set and a portable level recorder. A filter centred on the same frequency as the filter in the transmitting section reduces the influence of background noise. Since reverberation decreases in an exponential manner and is recorded on a logarithmic scale, the decay will be a straight line on the recording paper. The reverberation time result (for a given frequency band) is estimated directly from the recording. The jagged appearance of the decays at low frequencies is due to the uneven distribution of the normal room modes at these frequencies.

When the pulse method of noise transmission is used, the graphical results represent the *Impulse Response* of the room and the reverberation time cannot be obtained directly from the decay. By using the appropriate software, it is possible to calculate reverberation time results from the impulse response. An advantage of the pulse (or Schroeder) method is that accurate and reproducible results are obtained faster than with the "cut-off" method.





Using a Building Acoustics Analyzer

A Building Acoustics Analyzer is an instrument containing both the transmitting and the receiving sections. It supplies random noise in 1/3 octave bands to a power amplifier and a loudspeaker, analyzes the microphone signal through a second set of 1/3 octave band filters, and calculates the reverberation time for each frequency band.

Position of the Source and the Receiving Microphone

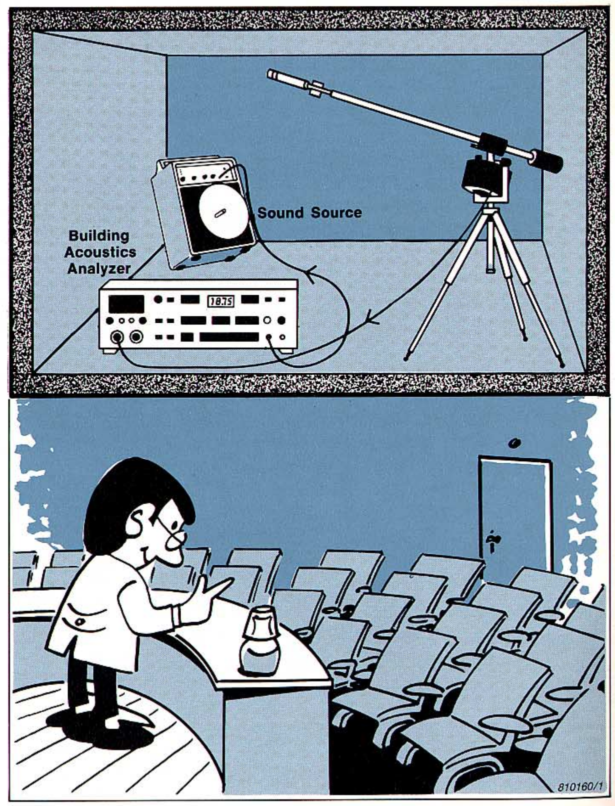
Due to room modes and echoes, the reverberation time of a room depends on the position of the source and the receiving microphone. In some cases the position of the source is obvious (e.g. the rostrum in a lecture theatre). To avoid exciting only some of the normal modes of the room, the sound-source is usually placed in a corner where every mode has a pressure maximum.

The receiving microphone should be placed at several positions in large rooms and auditoria because the reverberation time can vary from place to place. If required, the measured times should then be averaged for each frequency band by one of the following methods:

(*a*) a single microphone moved from place to place;

(*b*) several microphones scanned by a multiplexer;

(*c*) a single microphone on a rotating boom.



Measuring the Sound Absorption

The absorption coefficient of a material indicates the proportion of sound absorbed by the material relative to the total incident sound. The total absorption of a surface is given by the absorption coefficient multiplied by the area. The most usual measurement methods are:

Reverberation Chamber Method

The change in the reverberation time is measured when a $10\,m^2$ sample of absorption material is introduced into a reverberation chamber. From Sabine's Formula and the definition of absorption, α can then be found:

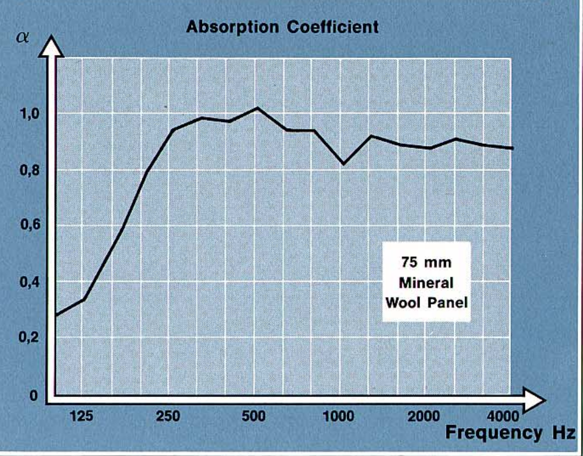
$$\alpha = \frac{0{,}16\,V}{S}\left(\frac{1}{T_s} - \frac{1}{T_e}\right)$$

where

α is the absorption coefficient of the sample
S is the area of the sample of material
V is the volume of the chamber
T_s is the reverberation time, *with* the sample
T_e is the reverberation time of the empty chamber

The measurements are performed by using an octave or 1/3 octave filter set to obtain α as function of the frequency.

Material	Frequency, Hz					
	125	250	500	1000	2000	4000
Air, per m^3	nil	nil	nil	0,001	0,003	0,007
Acoustic panelling	0,15	0,3	0,75	0,85	0,75	0,4
Plaster	0,03	0,03	0,02	0,03	0,04	0,05
Floor, concrete	0,02	0,02	0,02	0,04	0,05	0,05
Floor, wood	0,15	0,2	0,1	0,1	0,1	0,1
Floor, carpeted	0,1	0,15	0,25	0,3	0,3	0,3
Brickwall	0,05	0,04	0,02	0,04	0,05	0,05
Curtains	0,05	0,12	0,15	0,27	0,37	0,50

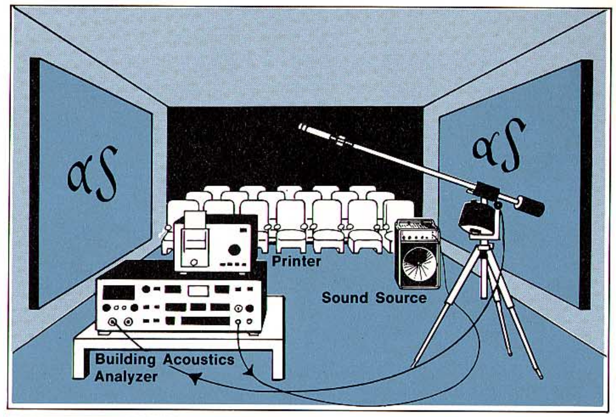


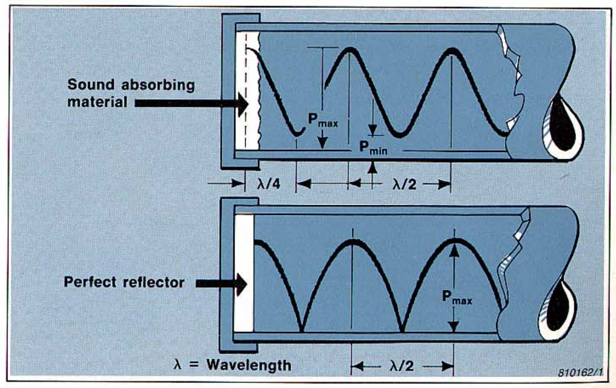
Measuring the Change of Reverberation Time "in situ"

A similar method can be used in practical situations when determining the amount of absorbent material necessary to obtain a suitable reverberation time in a room. From the absorption coefficient, α, calculated from measurement in a reverberation chamber, one calculates the area of absorbent necessary to produce a required change in reverberation time in a particular room. The absorbent material is installed, the reverberation time is measured in the actual room and, if necessary, adjusted by adding or subtracting some of the absorbent material.

Standing Wave Method

In this method a loudspeaker is used to produce standing waves in a tube terminated by the sample to be investigated. By measuring the ratio between the maximum and minimum sound pressures by means of a probe microphone moved along the axis of the tube, the absorption coefficient can be calculated. The advantage of the method is that it only requires small samples of material, gives reproducible results and yields a direct scale reading for the value of α. The disadvantages of the method are that α is obtained for normal incidence only and that the method can only be used where the sample is representative of the material.





Tone Burst Method

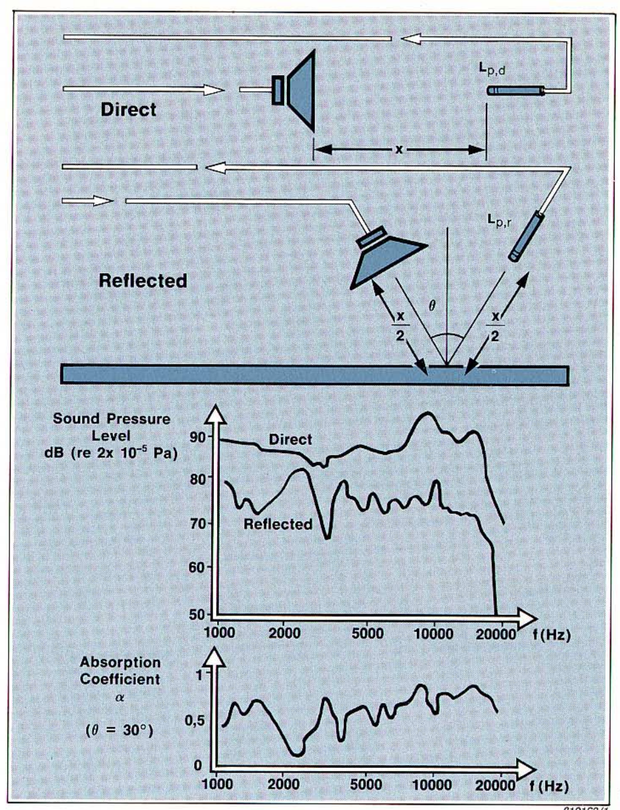
This method enables the absorption coefficient of a material to be determined for various angles of incidence of sound energy. No special reverberation room is required for this test. A short tone burst is emitted from a loudspeaker into the room at a distance x from the receiving microphone. The loudspeaker is then aimed at the test speciment at an angle of incidence, θ, such that the total path length for the reflected sound is the same as in the first case. By comparing the sound pressure level, $L_{p,r}$, of the reflected sound to the sound pressure level, $L_{p,d}$ of the direct sound, the reflection coefficient can be calculated and the absorption coefficient determined from:

$$\alpha_{\theta,f} = 1 - r_{\theta,f}$$

where $\alpha_{\theta,f}$ = the absorption coefficient

$r_{\theta,f}$ = the reflection coefficient

$$= 10^{\frac{-(L_{p,d} - L_{p,r})}{10}}$$

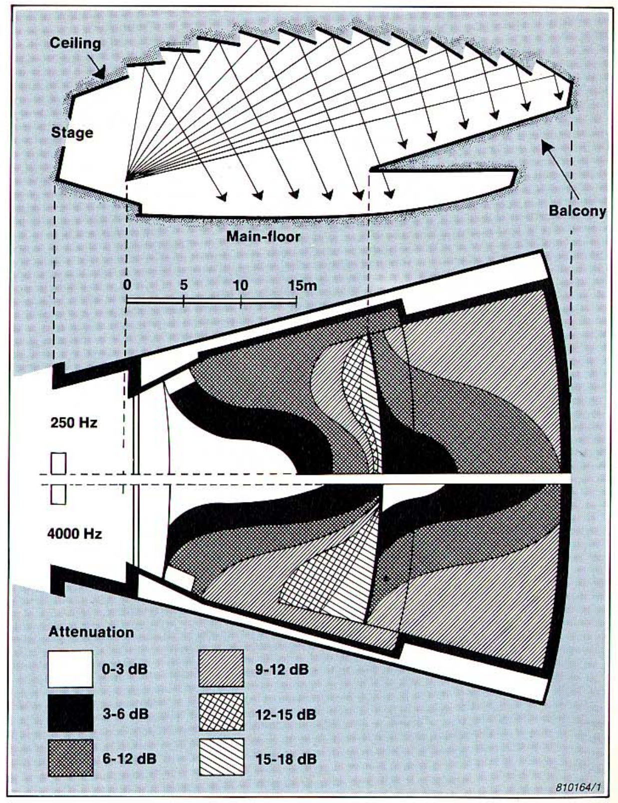


Measuring the Sound Distribution

Sound distribution measurements are especially important in theatres and concert halls or other public halls where music and speech must be heard clearly throughout the volume of the auditoria.

Measurement in Existing Room
Measurements of sound distribution in a room can be made directly by placing a source in the most probable position of the actual source (theatre stage, church pulpit, etc.) and by using a sound level meter to measure the sound pressure levels at various positions in the room. The source should be a constant sound power source radiating a wide band signal (white or pink noise).

This method can be made more informative if measurements are made at the same positions but at different frequencies. Filters (octave or third octave) can be used in the emitting section to limit the necessary power of the source and/or in the receiving section to reduce the influence of background noise.



Measurements on Models
Before the construction of a costly new theatre or auditorium, it can be economically advantageous to investigate the acoustics of the new design in a scaled-down model. Provided certain precautions are taken, model techniques can be used to investigate amongst other things, reverberation time, speech intelligibility and sound distribution.

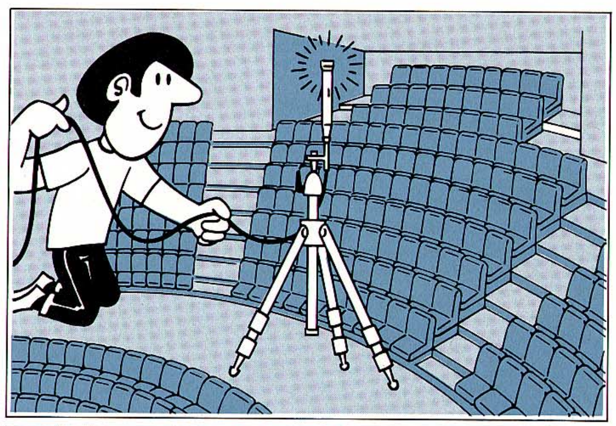
The frequency of excitation of the source should be increased by the same factor as that by which the model has been scaled down. This may be achieved in three ways:

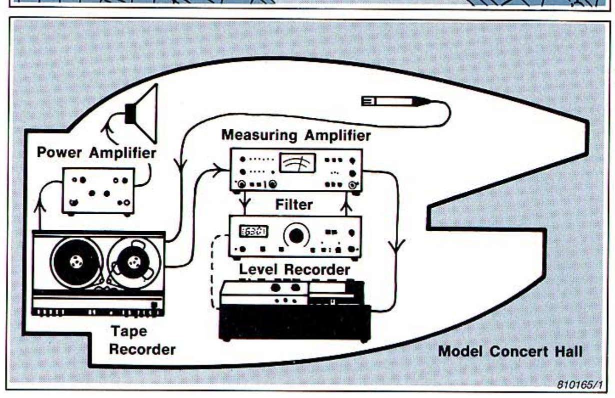
(*a*) By using a signal generator capable of producing noise at the higher frequencies required in the model;

(*b*) By recording audio range excitation noise on a tape recorder and playing back the signal in the model room at a correspondingly higher speed;

(*c*) By using a sound-source which has a frequency spectrum including relatively high frequencies e.g. an electrical spark or an ultrasonic whistle.

At these high frequencies, both the transmitting and receiving transducers should be of small dimensions to avoid disturbing the sound-field. Small condenser microphones can be driven as transmitters, the advantage being the stability of their frequency response, which can extend up to 140 kHz. The signal at the receiving position in the model is then recorded at high speed on a tape recorder. For analysis, the tape is played back at low speed, which brings the recorded signal into the audio frequency range.



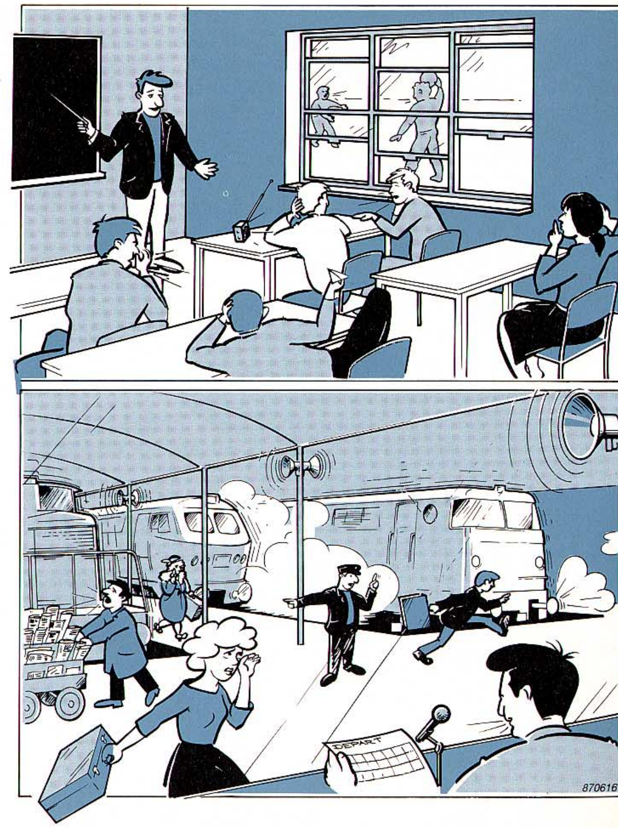


Speech Intelligibility

Speech transmitted across a room by a person or a public address system is never received at a listening position as an exact replica of the original signal. Not only is background noise added but the signal is also distorted by the reflective and reverberant properties of the room. Often a direct consequence of these distortions is a reduction in the intelligibility of speech.

To improve intelligibility, speakers usually adapt their speech to suit the room – talking slowly in a very reverberant room, or loudly either in a highly absorbent room or one with dead-spots. However, in some situations, such as when making an announcement over a public address system, speakers cannot adjust their speech. The result is often an unintelligible announcement.

By quantifying speech intelligibility and measuring it in a room, the extent to which acoustical treatment is required to solve such problems is known. Typical remedies to improve the clarity of speech include: sound reinforcement in auditoria, reduction of reverberation time in meeting rooms, prevention of echoes in large enclosures, optimisation of public address systems and attenuation of background noise.

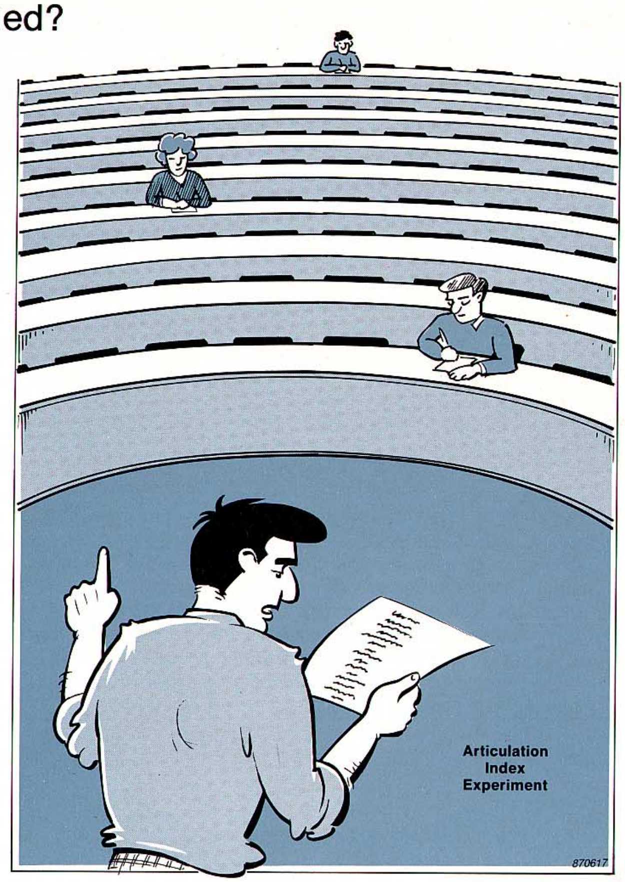


How is Speech Intelligibility Quantified?

Intelligibility is a subjective response, so it can be measured by examining the number of phonetically balanced nonsense words correctly noted down by a team of trained listeners. The results are expressed either as a percentage word score, or as an index on a scale 0 to 1. An *Articulation Index* (AI) of less than 0,3 generally suggests unintelligible speech and one over 0,7 indicates excellent intelligibility. Variabilities between different listeners will inevitably produce a large spread in the results.

Another approach is to determine the *Preferred Speech Interference Level* (PSIL) from a set of sound pressure level measurements. This involves measuring signal and noise levels over a preferred speech spectrum (the three octave bands centred on 500 Hz, 1 kHz and 2 kHz) and then adding an empirically derived correction factor to account for the effects of reverberation.

Speech Transmission Index (STI) is also a number between 0 and 1 which quantifies speech intelligibility. It is derived from a family of *Modulation Transfer Function* (MTF) curves. These describe the extent to which the original modulations in a signal are changed by a sound transmission system in the seven octave bands from 125 Hz to 8 kHz. The STI can be evaluated without speakers and listeners and also provides information about the way in which the room is distorting a signal.



Rapid Speech Transmission Index (RASTI)

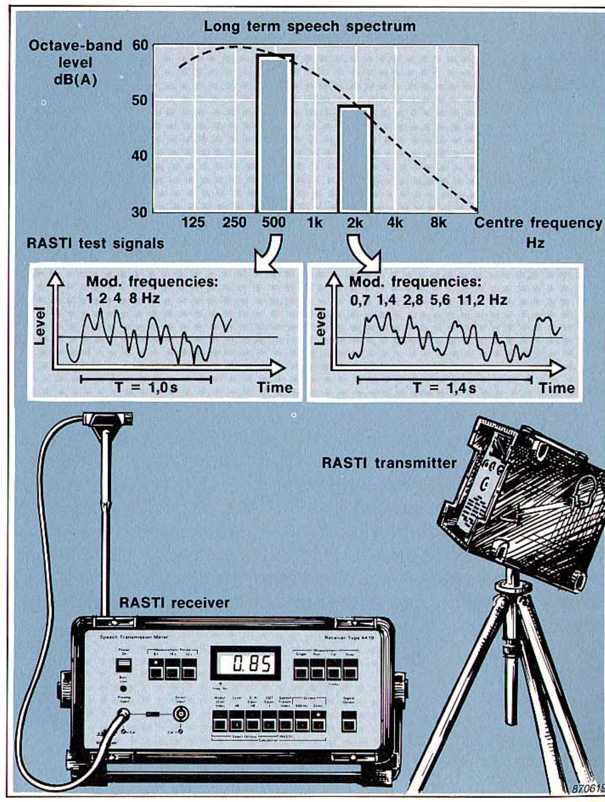
By confining the measurement of the Modulation Transfer Function to only two octave bands, the Rapid Speech Transmission Index (RASTI) can be calculated. This is much quicker than following the full STI procedure, and can easily be accomplished by using RASTI transmitting and receiving equipment.

RASTI Transmitter

A RASTI transmitter generates pink noise of levels 59 dB and 50 dB (at a distance of 1 m) in the 500 Hz and 2 kHz octave bands, respectively, to mimic the long-term speech spectrum. This noise is modulated sinusoidally by several frequencies simultaneously, representing the modulations found in normal speech. The transmitter transmits with the directional properties that would be measured 1 m from a speaker's mouth.

RASTI Receiver

An omni-directional microphone picks up the transmitted signal, which is analyzed by the RASTI receiver to detect the changes caused by the transmission medium. The receiver and transmitter are not synchronized (and are therefore independent units) because the signal is repetitive. The deviation of the received signal from the transmitted signal is recorded for each modulating frequency as a *modulation-reduction factor* (m). RASTI is calculated from the modulation reduction factors and is displayed as a number between 0 and 1.



Interpretation of RASTI Measurements

RASTI may be related to the subjective intelligibility scale shown opposite, which has been derived by comparing the phonetically balanced word score and STI methods.

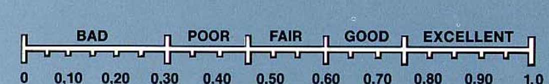
Information regarding the acoustical properties of the enclosure may also be derived from the RASTI measurements by using the *Modulation Transfer Function* (MTF). The MTF is simply a plot of modulation-reduction factor (m) against modulation frequency (M). If the MTF is flat then the source of interference is noise, if it has negative slope then the interference is reverberation. Examples of these two types are shown in the figure. A complicated MTF suggests that there is interference by a discrete echo.

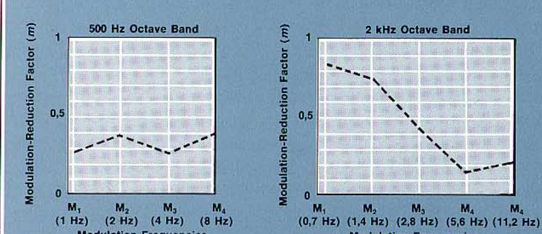
Applications of RASTI

The RASTI method identifies areas of poor speech intelligibility in a room and, because it is a quick method, the results can be displayed in the form of an iso-RASTI contour plot. Public address and sound reinforcement systems can be tested, either with the source placed at the microphone position or connected electrically to the system.

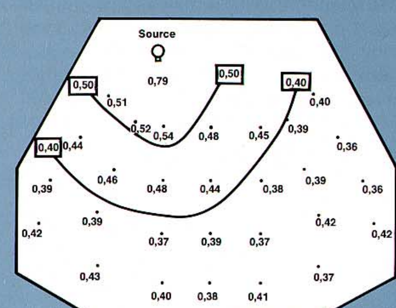
The method may also be used to assess the suitability of a room for the recording of speech, or determining the acoustical privacy of a room from adjoining rooms. In the latter case, a RASTI of less than 0,3 should be obtained if the transmitter were set up inside a room, with the receiver outside.

RASTI vs. Subjective intelligibility scale:





Modulation Transfer Function (MTF) measured in a seminar room.



RASTI contour plot for an auditorium

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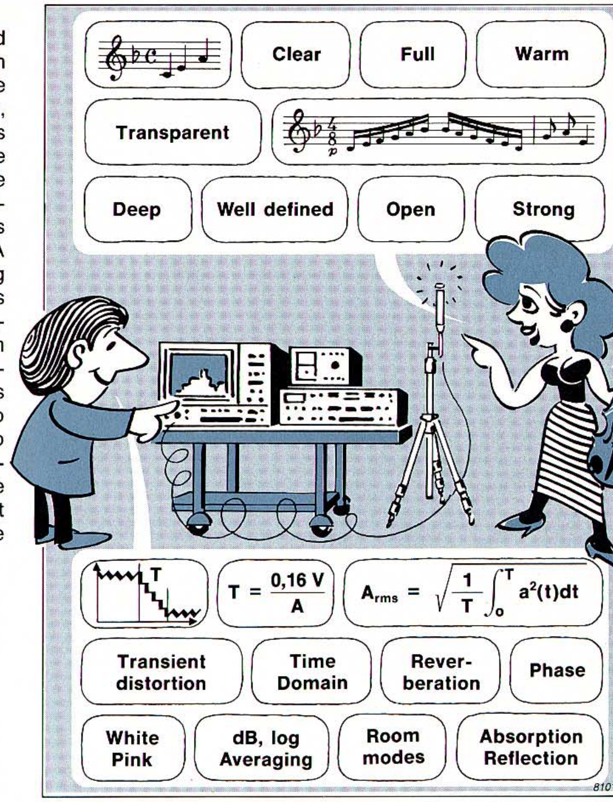
Real-Time Analysis in Room Acoustics

What are "good acoustics"?

It is generally not easy to specify what constitutes "good acoustics". Firstly, everything depends upon what the room is intended to be used for. The acoustical requirements are not the same for a concert hall, a theatre or a lecture room, and when the same hall has to be used both for concerts and theatre performances, some compromises have to be made. Secondly, it depends upon how the acoustics of the room are defined. An acoustician will talk about reverberation time, sound distribution, absorption, etc. in other words objective parameters which it is possible to measure. A musician listening to a piece of music or someone listening to a speech in the room will describe the acoustics in terms of definition, clarity of tone, warmth etc. In other words parameters which may be subjective or difficult to measure. In fact, the concept of "good acoustics" consists of a combination of most of these parameters, objective as well as subjective, considered in a "global" fashion. Therefore, to approach a more global evaluation, it may be necessary to consider several parameters simultaneously, such as amplitude, frequency and time. "Real-time analysis" allows the whole spectrum of a sound signal to be analyzed without corrupting or losing parts of the original signal. The time variations of the spectrum can therefore be studied.

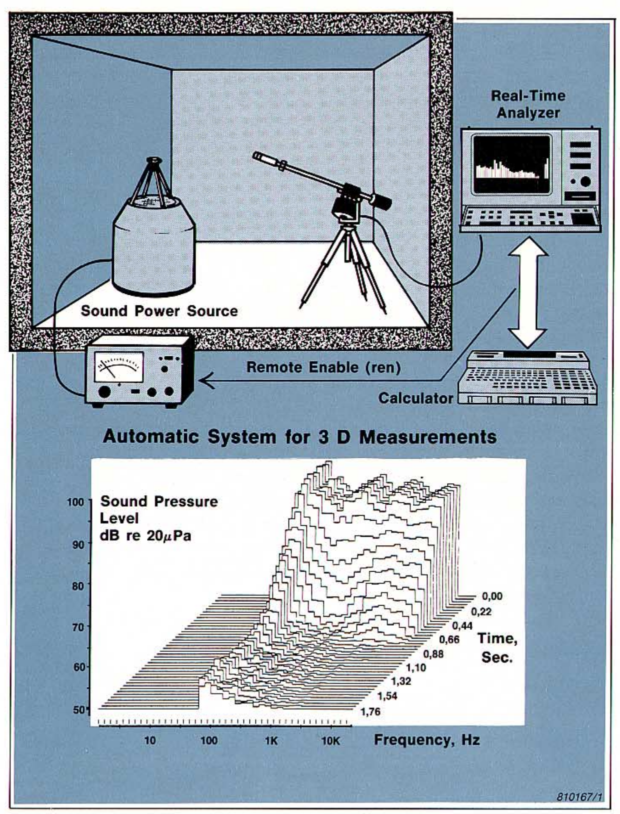


Real-Time Analysis

A real-time analyzer frequency-analyzes a sound signal and displays the results on a screen in the form of a bar graph of level against frequency band. By continuously updating the screen a fluctuating picture is obtained which closely follows the changes in level within the room. This enables "real-time" tests to be made within the room for the voice or for musical instruments so that the result can be observed immediately on the screen. For example, differences in reverberation times between lower and higher frequency bands will clearly appear on the screen as different decay rates of the columns representing the instantaneous level in the different frequency bands of the spectrum. Real-time analysis is especially useful in the detection of echoes, the positioning of reflectors, measurement of reverberation time, etc.

Reverberation Decays in Three Dimensions

The reverberation time decay curves of a sound produced in a room may be represented as a three-dimensional amplitude-frequency-time landscape by using a real-time analyzer in conjunction with a computer and a graphics plotter. If the sound-source can be started and stopped automatically by the computer, then a large number of reverberation decays can be measured and averaged to produce a final "decay curve" for each frequency band of interest.



Acoustics of Buildings: What Should be Measured?

Reverberation Time
The reverberation time should be measured in rooms or parts of the building where noise has to be reduced (e.g. flights of stairs), and in situations where sound insulation measurements are to be made (the calculation of certain insulation indices takes into account the reverberation time).

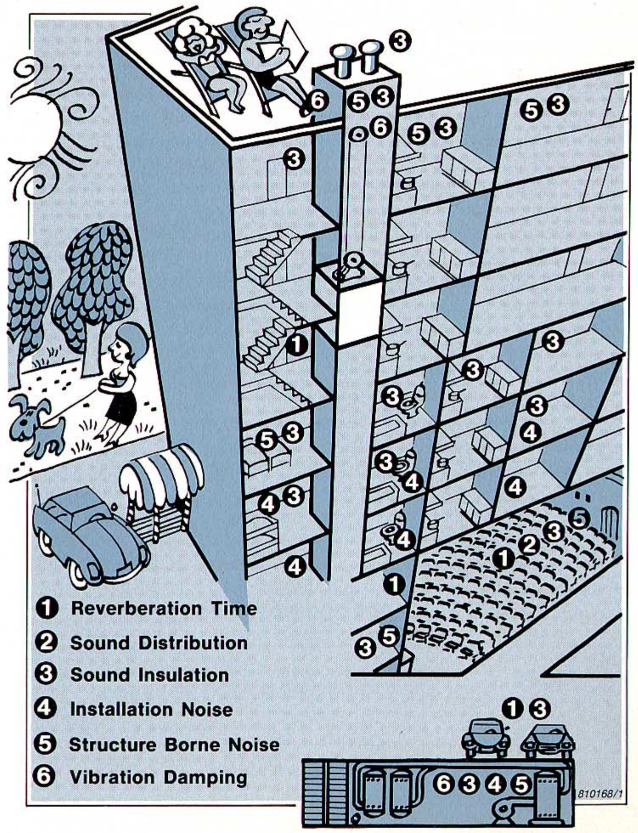
Airborne and Impact Sound Insulation
Sound energy does not remain in the room where it is produced but propagates throughout the building by any available transmission path and intrudes into other rooms as noise. Sound energy is transmitted via the air and via the structure of the building structure. In homogeneous structures of low loss factors (e.g. a solid concrete wall) sound energy is transmitted with very little attenuation. The acoustic parameters to be measured to describe the sound insulation provided by a wall or a floor are the airborne and the impact sound insulation.

installation Noise and Vibration Damping
Machinery, heating and elevator installations are often noisy. Therefore most standards of building regulations specify maximum limits of the received noise for each installation in rooms where people are living. What is required here are measurements of:

(*a*) noise and vibration at the source;

(*b*) sound and vibration transmission via the structure or via ventilation, heating system and water installations;

(*c*) the noise level in rooms affected by the installation noise.



Sound Reduction Index of a Wall

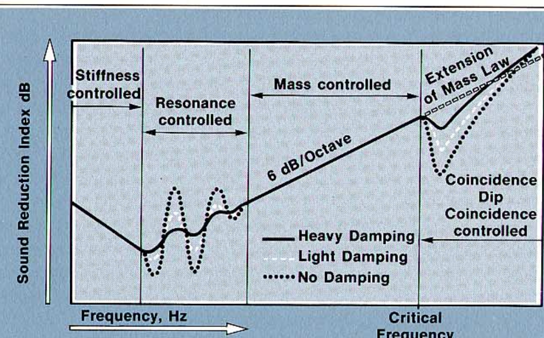
The airborne sound insulation afforded by a wall is expressed in terms of the Sound Reduction Index, R, which is the ratio in dB of the incident sound power on the wall to the sound power transmitted through the wall. The Sound Reduction Index depends on the frequency and the angle of incidence of the emitted sound.

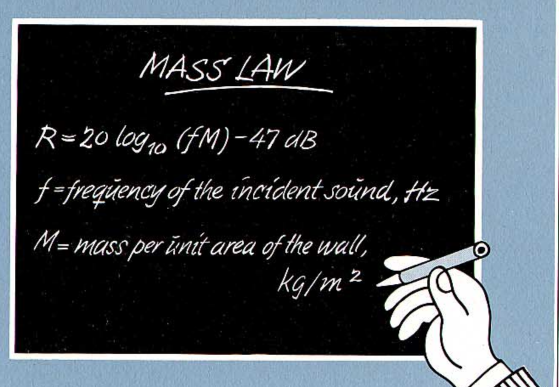
$$R = 10 \log_{10} \frac{W_i}{W_t}$$

W_i = Sound power incident on wall
W_t = Sound power transmitted through wall
R = Sound Reduction Index, dB

For a solid homogeneous wall the curve of the sound reduction index as function of frequency can be divided into several regions according to which property of the wall has most influence on the sound reduction. These properties are the stiffness, resonance, mass- and coincidence-controlled regions. The damping present in the structure affects only the profile of the curve in the resonance and the coincidence regions.

The Mass Law
In the mass controlled region, the Sound Reduction Index increases by 6 dB for each doubling in the frequency for a given mass per unit area of the wall or for each doubling of the mass per unit area (e.g. a doubling of the thickness) at a given frequency.



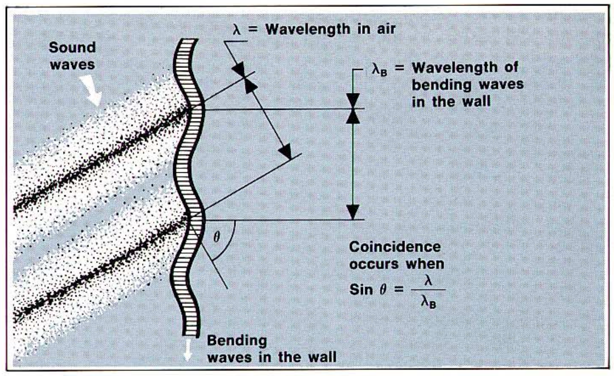


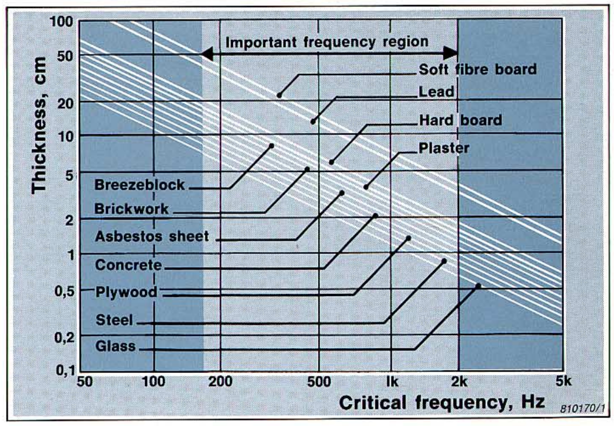
What is the Coincidence Effect?

The Coincidence Effect
The Mass Law provides a good working rule to predict the airborne sound insulation of a partition but, in practice, the application of this law is limited in the high frequency region by the coincidence effect. This effect occurs when the projected wavelength of the sound in the air is the same as the wavelength of the bending waves in the partition. For a certain frequency and a certain angle of incidence of the incident sound-waves, the bending oscillations of the partition will be amplified and the acoustic energy will be transmitted through the partition almost without attenuation. In practice, the incident sound-waves arrive from every angle of incidence to the partition, which is then almost acoustically transparent for a narrow frequency region, called the "coincidence dip".

The Critical Frequency
The lowest frequency for which the coincidence effect occurs on a certain partition is obtained when the incident sound-waves graze the partition (i.e. are parallel with it). This frequency is called the critical frequency, f_c.

The nomogram on the right may be used to determine the critical frequency in an actual situation when designing an enclosure or a dividing wall. For example, a 3 cm thick plywood partition has a critical frequency at about 500 Hz, which is unfortunately in the middle of the speech frequency region.

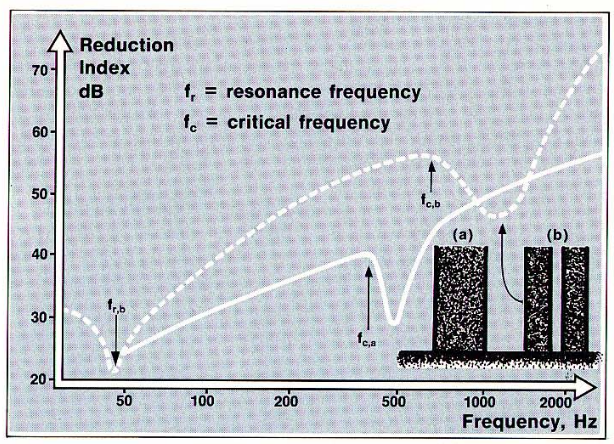


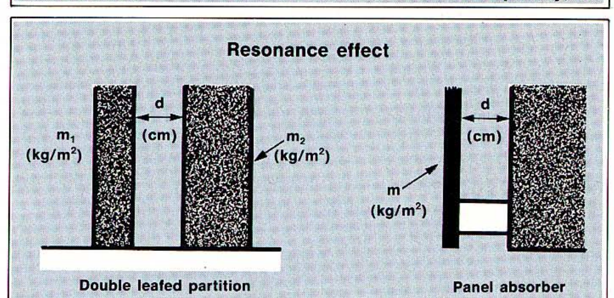


Double-Leafed Partition
One way of moving the coincidence effect to a higher frequency range without reducing the sound insulation is to use a double-leafed partition. For a double-leafed partition, the coincidence frequency is determined by the thickness of each element, while the Sound Reduction Index is even higher than that predicted by the Mass Law for a single partition of the same mass. Moreover, it is an advantage to choose two different thicknesses for both half-elements in order to avoid both coincidence effects being situated at the same frequency.

The Resonance Frequency
Generally, the sound insulation of a double-leafed partition is better than that of a single wall of the same overall mass. However, at the mass-spring-mass resonance frequency (f_r) of the partition, the sound insulation is not better — so care must be taken to keep f_r out of the frequency range of interest (i.e. below 100 Hz).

Note that the resonance effect can be used advantageously when it is desired to absorb lower-frequency sound energy in a noisy/reverberant room. A thin panel is fixed at a distance *d* from a rigid wall and the resonance frequency of the panel is chosen in that case to fall in the frequency region where the noise has to be reduced.





$$f_r = 590 \sqrt{\frac{m_1 + m_2}{m_1\, m_2} \cdot \frac{1}{d}}$$

$$f_r = 590 \sqrt{\frac{1}{md}}$$

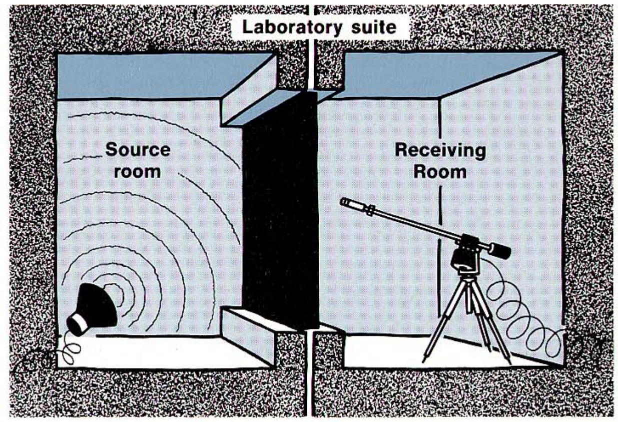
Laboratory and Field Measurements

Laboratory Measurements
Laboratory measurements are used to determine specific properties of a material or to make a complete investigation of it in order to establish acoustic data or a quality standard. They are also used to ensure that the quality of a material or a sample of building element meets international standards or local regulations.

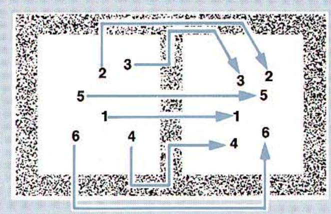
The test room suite of a laboratory is constructed very carefully to avoid any possible flanking transmission. Thus, when sound insulation tests are performed, practically all the energy in the receiving room is transmitted through the partition under test.

Field Measurements
There are so many possible transmission paths of sound in a building and so many factors influencing the acoustic quality of the construction that the only way of determining whether the building meets the legal requirements is to make measurements "in situ" in the actual building.

In most cases, a part of the sound produced in a room is transmitted indirectly via flanking elements or acoustic "leaks" into adjacent rooms. The sound insulation of building elements is therefore generally lower in situ than in the laboratory. Therefore, care should be taken when selecting building materials to include a safety factor in the calculation of the forecasted sound insulation of building constructions.



Paths of transmission between two rooms in a building



1 Direct Transmission
2–3 and 4 Flanking Transmission
5 Transmission via acoustic "leaks"
6 Transmission via adjacent Rooms

810172/1

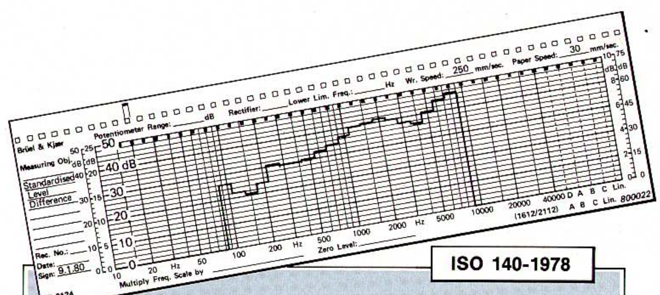
Airborne Sound Insulation

The Airborne Sound Insulation between two rooms is calculated from the difference between sound pressure levels in the source and receiving rooms, plus a factor taking into account the absorption in the receiving room. In a laboratory, the correction factor involves the area of the test specimen, *S*, and the equivalent absorption area of the receiving room, *A*, which can be determined from the volume and the reverberation time of the receiving room. In actual buildings, the correction factor depends on the way the room insulation is defined. The two most usual definitions are:

the **Standardized Level Difference**, D_{nT}, involving the reverberation time of the receiving room referred to a standard reverberation time of 0,5 s, and

the **Apparent Sound Reduction Index**, R', involving the area of the common partition, the reverberation time and volume of the receiving room.

Since the reverberation time in a furnished room is about 0,5 s, D_{nT} corresponds to the actual sound insulation experienced by people in a living-room or a bedroom. (R', on the other hand, takes into account the dimensions of the room.) For small rooms, like bathrooms, R' is the less stringent requirement of the two.



ISO 140-1978

AIRBORNE SOUND INSULATION IN LABORATORY
Sound Reduction Index of a building element

$$R = L_1 - L_2 + 10 \log \frac{S}{A} \text{ dB}$$

L_1 = Sound Pressure Level in source room, dB
L_2 = Sound Pressure Level in receiving room, dB
S = Area of the common partition, m²
A = Equivalent absorption area (receiving room), m²

AIRBORNE SOUND INSULATION IN SITU
Standardized Level Difference

$$D_{nt} = L_1 - L_2 + 10 \log \frac{T}{0{,}5} \text{ dB}$$

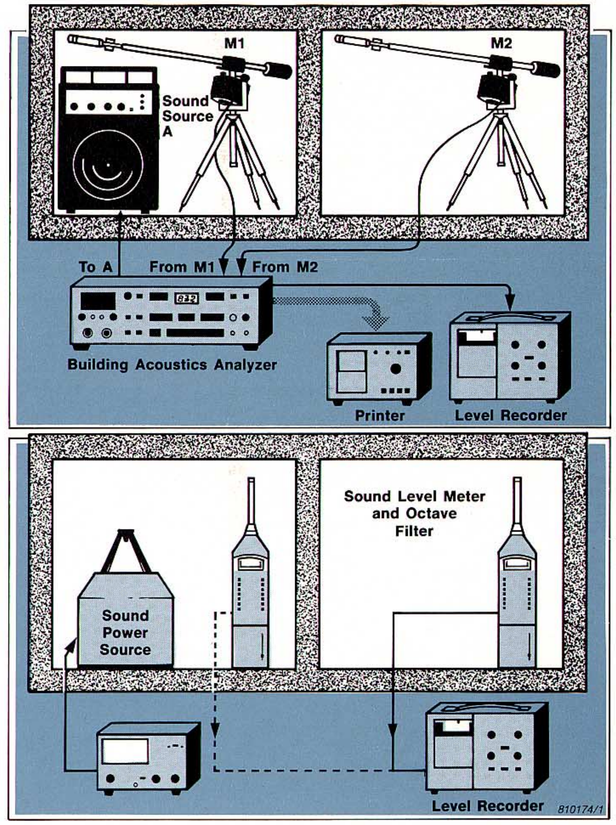
T = Reverberation Time in the receiving room, s

Apparent Sound Reduction Index
Normalized Level Difference with flanking transmission

$$R' = L_1 - L_2 + 10 \log \frac{S}{A} \text{ dB}$$

810173/1

Measuring Airborne Sound Insulation



The Transmitting Section
When measuring the sound reduction index of a building element in a laboratory, the excitation of the source room may be obtained (as for reverberation time measurements, see pp.8-9) from a broad-band signal filtered 1/3 octave bands supplied by a noise generator followed by a filter set. For "in situ" measurements, the sound-source can be a portable system generating noise in wide or narrow bands or even a noise source available on the spot such as a machine, providing that the noise emitted is stationary and broad-band without dominating frequencies. The noise levels in the source room should be high enough to allow meaningful measurements to be made.

The Receiving Section
The sound pressure levels are measured successively in the source room and the receiving room and plotted on a level recorder. A filter in the receiving section may be necessary if a broad-band noise source is used in the transmitting section or if the sound levels measured in the receiving room are not at least 6 dB higher than the background noise level. For measurements in situ, a precision sound level meter with built-in filters, or fitted with a filter set, may be used in connection with a portable level recorder. As for reverberation time measurements, it is necessary to average the sound pressure levels both spatially and temporally.

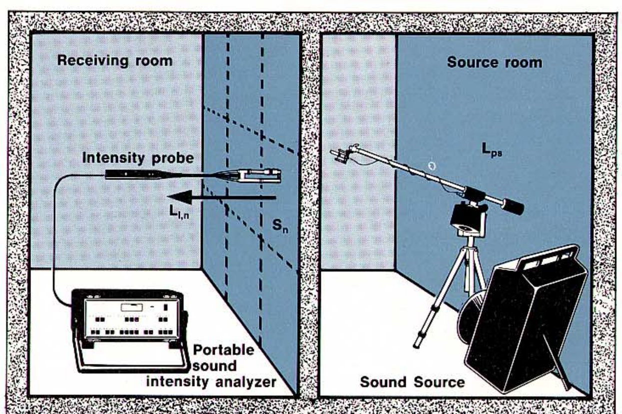
A Building Acoustics Analyzer automatically carries out the measurement sequence requiring only a microphone, a power amplifier and loudspeaker, and a printer as external equipment.

Intensity Approach
Sound intensity measurements provide an alternative approach for measuring airborne sound insulation. Intensity is a vector quantity which describes the sound energy flowing through an area. Units are W/m^2. It can be measured directly by using a two-microphone probe and an intensity analyzer.

Measurements in the source room are carried out in exactly the same way as previously. In the receiving room, a grid applied to the measurement surface defines the areas of interest. The average sound intensity flowing through each grid-segment can be measured directly by using a sound intensity analyzing system. The sound power emitted by each segment in the grid is simply the average sound intensity multiplied by the segment's area.

Since the flow of sound intensity through any surface in the room may be examined, it is possible to measure the contribution of the various flanking and leakage transmissions towards the total power in the receiving room. In this way results can be compared with those obtained by the previous method.

A significant advantage of the intensity approach is that the apparent sound reduction index of R'_n for any area on the measurement grid may be found. So if a compound partition is to be studied, for example a wall containing a window, R'_n may be found for both the wall material and the glass.



Apparent sound reduction index for surface S_n

$$R'_n = L_{ps} - 6\text{ dB} - L_{I,n} + 10 \text{ Log } \frac{S}{S_n}$$

L_{ps} = Spatially averaged sound pressure level in source room

S = Area of party wall

S_n = Area of one grid section

$L_{I,n}$ = Average intensity over surface area S_n

Apparent sound reduction index for surface S

$$R' = -10 \log \sum_{n=1}^{N} 10^{\frac{-R'_n}{10}}$$

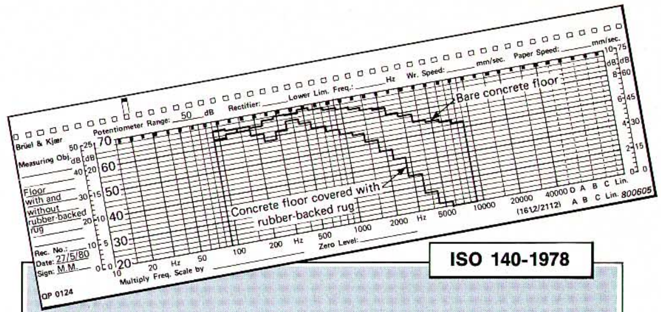
870613

Impact Sound Insulation

Impact Sound
Footsteps on floors or stairs can often be heard more clearly in other rooms than in the room where they are produced. The reason is that the building structure is set into vibration and these vibrations can be transmitted to other parts of the building almost without damping. An effective way of reducing impact noise is to attenuate the impact of the footsteps before it reaches the structure of the building by, for example, using a floating floor or laying a suitable carpet or other resilient layer on the floor.

Parameter Measured
The Impact Sound Insulation is determined from the Impact Sound Level measured in the receiving room when the source room is excited by a standard impact source. As for Airborne Sound Insulation a distinction is made between laboratory measurements and field measurements and a correction factor involving the absorption in the receiving room has to be included in the calculation of the Impact Sound Level.

The **Normalized Impact Sound Pressure Level**, L_n (or L'_n if flanking transmission is included), calls in the absorption in the receiving room, A (calculated from the volume, V, and reverberation time, T, in the receiving room by using Sabine's equation), while in the **Standardized Impact Sound Pressure Level**, L_{nT}, the reverberation time in the receiving room, T, is referred to a standard reverberation time of 0,5 s.



ISO 140-1978

IMPACT SOUND INSULATION IN LABORATORY
Normalized Impact Sound Pressure Level
(without flanking transmission)

$$L_n = L_i + 10 \log \frac{A}{10} \text{ dB}$$

L_i = Impact Sound Pressure Level (receiving room), dB
A = Equivalent Absorption area (receiving room), m^2

IMPACT SOUND INSULATION IN SITU
Standardised Impact Sound Pressure level

$$L_{nT} = L_i - 10 \log \frac{T}{0,5} \text{ dB}$$

T = Reverberation time in the receiving room, s

Normalized Impact Sound Pressure Level
(with flanking transmission)

$$L'_n = L_i + 10 \log \frac{A}{10} \text{ dB}$$

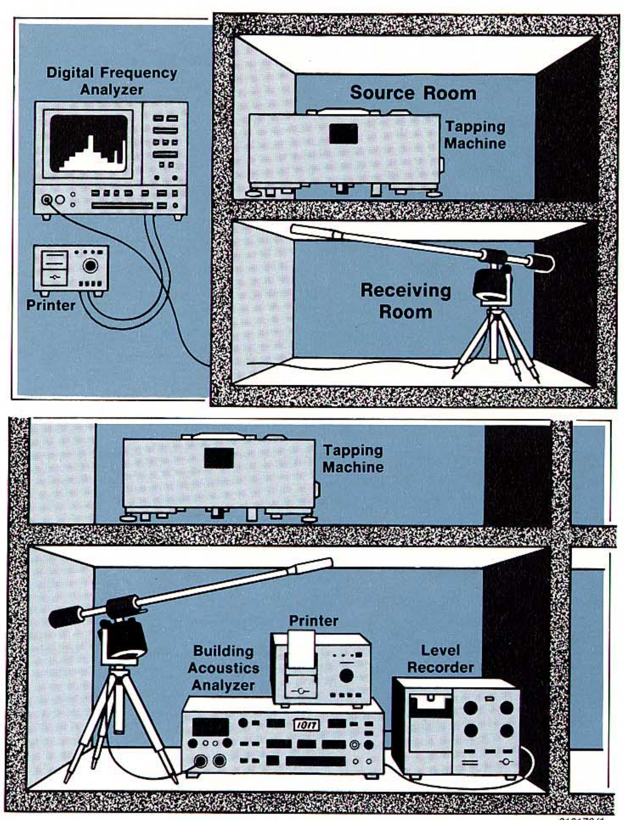
810175/1

Measuring Impact Sound Insulation

The Sound-Source
Footstep noise is simulated by a standard tapping machine containing five hammers of 0,5 kg each with a free fall of 4 cm producing 10 impacts per second. The effect on the floor is much stronger than the effect of normal footsteps, but this is necessary to obtain a suitably high sound pressure level in the receiving room. Standards specify that measurements should be carried out with several positions of the tapping machine in the source room.

The Receiving Section
Measurements in buildings assume that the sound-field is diffuse, but this is not generally the case. In practice, the sound pressure levels in the receiving room have to be averaged by measuring at several microphone positions or by using a microphone at the end of a slowly rotating boom. The received signal is filtered in octave or 1/3 octave bands. Results obtained with an 1/1 octave filter are 5 dB higher than with a 1/3 octave filter (10 log 3 = 5). The filter type should therefore always be specified on the measured curve.

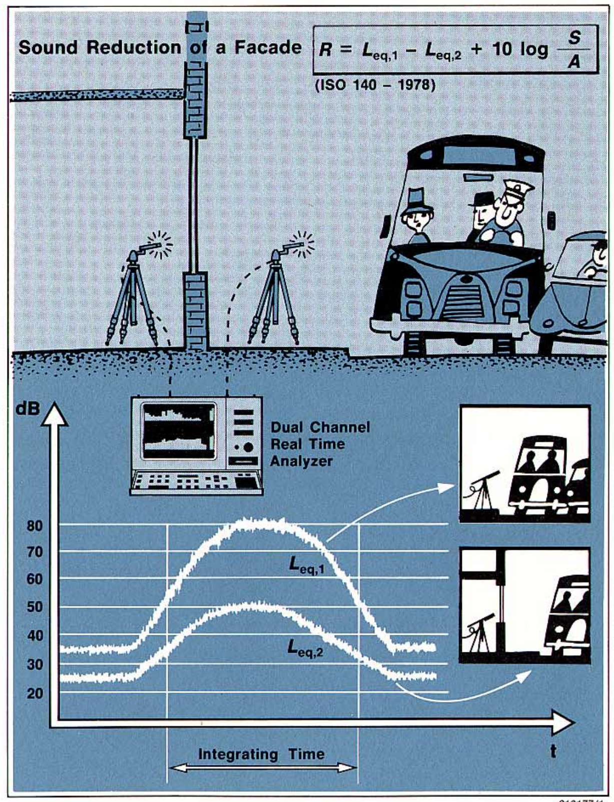
On a real-time analyzer the averaging is performed automatically. Any change in the spectrum when various resilient layers are being tested, for example, can be seen immediately. A Building Acoustics Analyzer will also perform the averaging automatically and furthermore display directly the Standardized and the Normalized Impact Sound Levels.



810176/1

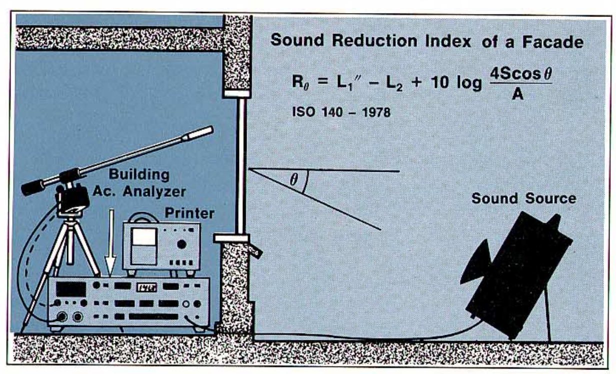
Outdoor – Indoor Noise Insulation

Sound Insulation of a Facade by Using Traffic Noise
Measuring the insulation afforded by a building against external noise must be viewed in a different light from the insulation between different parts of a building. In the latter, the sound-field is assumed to be diffuse and steady during measurements, while in the former the external sound-field is almost never diffuse or steady. The noise may arrive from various angles of incidence and often varies greatly in amplitude, e.g. traffic noise. The insulation of a facade is more a question of determining the noise level inside a building from the knowledge of the noise environment outside rather than of calculating an absolute figure from the knowledge of the reduction index of the different facade elements. The sound insulation of a facade is therefore expressed by the difference between the equivalent continuous levels in front of the facade and in the receiving room, both being measured over the same length of time. The equivalent continuous level, or L_{eq}, is the sound pressure level averaged for a relatively long measuring period on the basis of the energy. That is to say that the L_{eq} value has the same energy content as the measured sound of varying level.



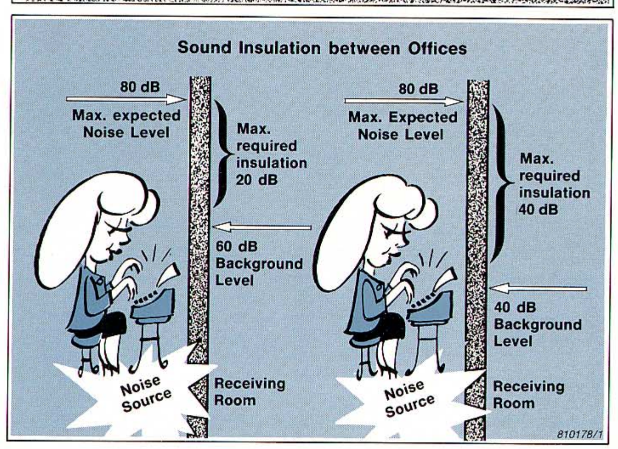
810177/1

Sound Insulation of a Facade by Using Loudspeaker Noise
In the absence of traffic noise or when the insulation of a facade or a facade element has to be investigated as function of the angle of incidence, a loudspeaker may be used as a sound-source. The loudspeaker emits a random noise filtered in 1/3 octave bands and the Sound Reduction Index, R_θ is calculated for each frequency band from the difference between the sound pressure levels with and without the test specimen. The measurements may be repeated for each value of the angle of incidence, θ, of interest.



Insulation between Offices — Influence of Background Noise

The background noise has a great influence on the requirements to the efficacy of partitions between offices. Background noise, either from external traffic or from typewriters in an office, masks the noise coming through the partitions and the insulation required is less than in the presence of a lower background noise.



810178/1

Comparing Results with Requirements

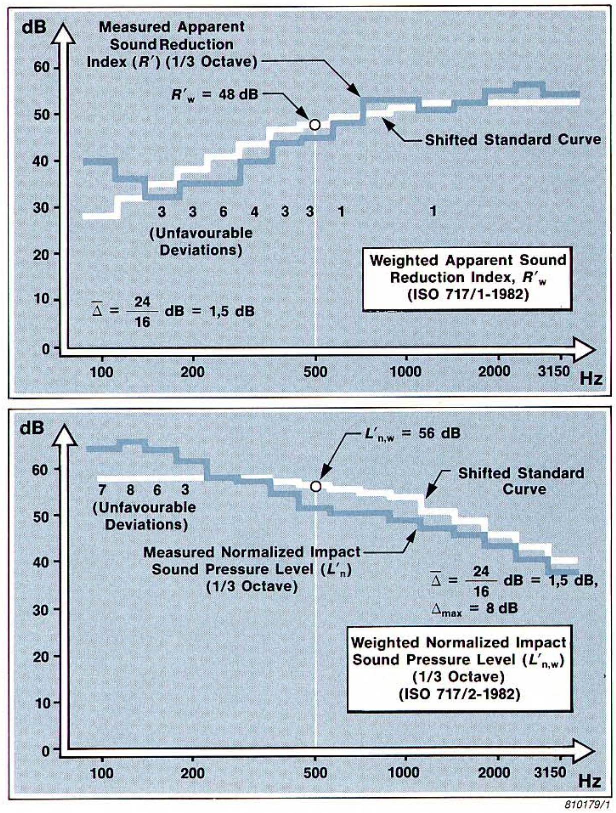
Since the sound insulation is a function of frequency, most regulations specifying the sound insulation between dwellings require an evaluation of the measurement results by comparison to reference curves covering the frequency range from 100 to 3150 Hz.

Single Figure Indices
ISO 717-1982 describes a method for obtaining single figure indices from the airborne and impact sound insulation curves measured according to ISO 140.

Weighted Apparent Sound Reduction Index, R'_w
The airborne sound insulation is characterized by an single number, R'_w, which is found by shifting in steps of 1 dB the reference curve towards the measured curve until the conditions* specified in the ISO standard are satisfied. The weighted apparent sound reduction index, R'_w, is defined as the value of the shifted reference curve at 500 Hz.

Weighted Normalized Impact Sound Pressure Level, $L'_{n,w}$
$L'_{n,w}$ is found in a similar way by shifting the reference curve towards the measured curve and is the value at 500 Hz of the shifted reference curve.

If a Building Acoustics Analyzer is used to measure the sound insulation curves, the indices R'_w and $L'_{n,w}$ can be calculated and displayed directly.



* The mean unfavourable deviation, $\bar{\Delta}$, should be as large as possible but not greater than 2 dB. The max. unfavourable deviation, Δ_{max}, must be recorded if it exceeds 8 dB at any frequency.

Vibration Measurements

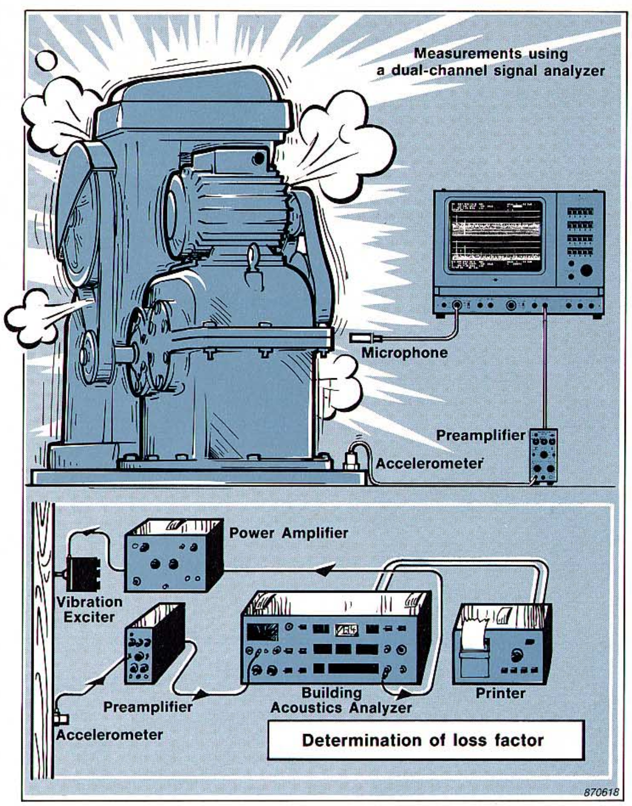
Many installations in a modern building, for example lifts and washing machines, produce both noise and vibration. Noise measurements must therefore be complemented by vibration measurements.

Vibration Isolation Measurements
These are carried out by using small mechanical transducers called accelerometers, which are attached to the vibrating structure. The accelerometer is connected to a preamplifier which may contain networks allowing the measurement of vibration velocity and displacement to be measured as well as acceleration. The output signal is analyzed by the same type of instrumentation as used for sound measurements. A frequency analysis of the vibration signal is often needed for determining the most appropriate means of damping the troublesome vibrations.

Measuring the Loss Factor of a Partition
The *Loss Factor*, η, is determined from the mechanical reverberation time of a partition which is excited by a shaker driven by white noise in 1/3 octave bands. When the partition has reached a steady level of vibration, the shaker is abruptly stopped. The reverberation time for each 1/3 octave band is determined from the decay curves recorded by an accelerometer, and the Loss Factor, η, calculated from:

$$\eta = \frac{2,2}{fT}$$

where f is the centre frequency of the 1/3 octave band and T the corresponding reverberation time.



Survey of Building Acoustic Measurements (ISO)

Measurement	Parameter to be determined	International Standard/ Recommendation	Test Environment
Reverberation time in auditoria	Reverberation time	ISO 3382-1975	Empty auditorium
			Studio- and occupied-state auditorium
Absorption coefficient	Absorption coefficient of a specimen $\alpha = \frac{0{,}16\,V}{S}\left(\frac{1}{T_s} - \frac{1}{T_e}\right)$	ISO 354-1985	Reverberation room
Airborne sound insulation of building elements	Sound Reduction Index, R $R = L_1 - L_2 + 10 \log \frac{S}{A}$	ISO 140/III-1978	Laboratory suite (specified in ISO 140/I)

T01390GB0

Source Room			Receiving Room		
Sound/ Vibration Source	**Character of Noise**	**Measurements**	**Measurements**	**Conditions of measurements**	**Observations**
Non-directional loudspeakers or pistol if T > 1,5 s below 1 kHz	Wide-band noise in oct. or 1/3 oct. bands or pistol shots. At least 40 dB above back-ground level in all freq. bands	Rev. decays in 1/3 oct. or oct. (125 Hz–4 kHz) At least 3 micro. positions with 2 records for each position (4 records for pistol shots and 6 records for music breaks)	—	—	
Non-directional loudspeakers or pistol or orche-stra (woodwind and brass instr. only)	As above or pink noise (40 dB above background level)				
Non-directional loudspeakers	Cont. freq. spectrum band-limited noise with a bandwidth of at least 1/3 octave	Rev. times at centre freq. of 1/3 octave band series 100 Hz – 5 kHz	—	—	
Loudspeaker	Steady, broad-band, may be filtered in 1/3 oct. bands	Sound Pressure Level 1/3 oct. (100 Hz – 3,15 kHz) several positions	Sound Pressure Level Rev. time	1/3 oct. (100 Hz – 3,15 kHz) several positions or moving microphone	Calculation of Weighted Sound Reduction Index: R_w (ISO 717/1-1982)

T01391GB0

Survey of Building Acoustic Measurements (ISO) — (Cont.)

Measurement	Parameter to be determined	International Standard/ Recommendation	Test Environment
Airborne sound insulation between rooms	Standardized Level Difference $D_{nT} = L_1 - L_2 + 10 \log \frac{T}{0,5}$ or Apparent Sound Reduction Index, R' $R' = L_1 - L_2 + 10 \log \frac{S}{A}$	ISO 140/IV-1978	Field measurements in buildings
Airborne sound insulation of facade elements and facades	Standardized Level Difference $D_{nT} = L_{eq,1} - L_{eq,2} + 10 \log \frac{T}{0,5}$ Sound Reduction Index $R_{tr} = L_{eq,1} - L_{eq,2} + 10 \log \frac{S}{A}$ Sound Reduction Index $R_{\theta} = L_1'' - L_2 + 10 \log \frac{4\,S \cos\theta}{A}$	ISO 140/V-1978	Field measurements

T01392GB0

Source Room			Receiving Room		
Sound/ Vibration Source	**Character of Noise**	**Measurements**	**Measurements**	**Conditions of measurements**	**Observations**
Loudspeaker	Steady, broad-band, may be filtered in 1/3 oct. bands	Sound Pressure Level oct. or 1/3 oct. (100 Hz – 3,15 kHz) several positions	Sound Pressure Level. Background level. Rev. time	Oct. (125 Hz – 2 kHz) or 1/3 oct. (100 Hz – 3,15 kHz) several positions or moving microphone	Evaluation of Weighted Apparent Sound Reduction Index: R'_w (ISO R 717/1 1982)
Traffic noise	Fluctuating	$L_{eq,1}$ at 2 m from the facade. Oct. or 1/3 oct. bands	$L_{eq,2}$ and rev. time	Oct. (125 Hz – 2 kHz) or 1/3 oct. (100 – 3,15 kHz) Several microphones or several positions	$L_{eq,1}$ and $L_{eq,2}$ measured simultaneously
Loudspeaker incidence angle $\theta = 45°$	Steady, broad-band, may be filtered in 1/3 oct. bands	Sound Pressure Level oct. or 1/3 oct.	Sound Pressure Level. Background level. Rev. time	Oct. (125 Hz – 2 kHz) or 1/3 oct. (100 Hz – 3,15 kHz) several positions or moving microphone	

T01393GB0

Survey of Building Acoustic Measurements (ISO) — (Cont.)

Measurement	Parameter to be determined	International Standard/ Recommendation	Test Environment
Impact sound insulation of floors	Normalized Impact Sound Pressure Level $L_n = L_i + 10 \log \frac{A_2}{10}$	ISO 140/VI-1978	Laboratory suite (specified in ISO 140/I)
Impact sound insulation of floors	Norm. Impact Sound Pressure Level $L'_n = L_i + 10 \log \frac{A_2}{10}$ Standard. Impact Sound Pressure Level $L_{nT} = L_i - 10 \log \frac{T_2}{0,5}$	ISO 140/VII-1978	Field measurements

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Source Room			Receiving Room		
Sound/ Vibration Source	**Character of Noise**	**Measurements**	**Measurements**	**Conditions of measurements**	**Observations**
Standard Tapping Machine	Repetitive impacts in at least 4 positions	—	Sound Pressure Level. Background level. Rev. time	Oct. (125 Hz – 2 kHz) or 1/3 oct. (100 Hz – 3,15 kHz) several positions or moving microphone	The use of oct. or 1/3 oct. shall be recorded. Evaluation of Weighted Normalized Impact Sound Pressure Level: $L_{n,w}$ (ISO 717/2 1982)
Standard Tapping Machine	Repetitive impacts in at least 4 positions	—	Sound Pressure Level. Background level. Rev. time	Oct. (125 Hz – 2 kHz) or 1/3 oct. (100 Hz – 3,15 kHz) several positions or moving microphone	As above. Evaluation of Weighted Normalized Impact Sound Pressure Level: $L'_{n,w}$ (ISO 717/2 1982)

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Survey of Building Acoustic Measurements (ISO) — (Cont.)

Measurement	Parameter to be determined	International Standard/ Recommendation	Test Environment
Reduction of impact noise by floor covering on standard floor	Reduction of Impact Sound Pressure Level $\Delta L = L_{n,0} - L_n$ $L_{n,0}$ = Norm. Impact Sound Pressure Level in the absence of floor covering	ISO 140/VIII	Laboratory suite (specified in ISO 140/I)
Flanking transmission	Radiated power, W_k, from a flanking element k, area S_k $W_k = \rho\, c\, S_k\, \overline{v_k^2}\, \sigma_k$ v_k = normal surface velocity	Airborne Sound ISO 140/III Annex A ISO 140/IV Annex B	Laboratory and field measurements
	Average Sound Pressure Level, L_k, due to a flanking element k $L_k = L_{vk} + 10 \log \frac{4 S_k}{A}$ L_{vk} = average surface velocity	Impact Sound ISO 140/VI Annex B ISO 140/VII Annex B	
Loss Factor of a partition	Total loss factor $\eta_{total} = \frac{2,2}{f\,T}$ f = 1/3 oct. centre frequency T = mechanical rev. time of the partition	ISO 140/III Annex C ISO 140/IV Annex C	Laboratory and field measurements

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Source Room			Receiving Room		
Sound/ Vibration Source	**Character of Noise**	**Measurements**	**Measurements**	**Conditions of measurements**	**Observations**
Standard Tapping Machine	Repetitive impacts in at least 3 positions on bare floor and covered floor	—	Sound Pressure Level. Background level. Rev. time	Oct. (125 Hz – 2 kHz) or 1/3 oct. (100 Hz – 3,15 kHz) several positions or moving microphone	The bandwidth used for measurements shall be stated in every graph or table
Loudspeaker or Reference Sound Source	Steady, broad-band	Incident sound power, W_1 oct. or 1/3 oct.	Normal surface velocity	Oct. or 1/3 oct. several positions on each flanking element	
Standard Tapping Machine	Repetitive impacts	—	As above Rev. Time		
Vibration Exciter	Steady vibration level White noise generator in 1/3 oct. bands	Vibration decay measured in 1/3 oct. (100 Hz – 3,15 kHz)	—	—	

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Further Reading

J. ANDERSON & T. JACOBSEN.

"RASTI Measurements in St. Paul's Cathedral, London." *Brüel & Kjær Application Note* BO 0116 – 11.

BRÜEL & KJÆR PUBLICATIONS

"Sound Intensity" *Brüel & Kjær Booklet* BR 0476 – 11.

"Reverberation Time — fast and accurate calculations with a sound level meter." *Brüel & Kjær Application Note* BO 0228 – 11.

T. R. HORRALL & T. JACOBSEN.

"RASTI Measurements: Demonstration of different applications." *Brüel & Kjær Application Note* BO 0123 – 11.

T. G. NIELSEN.

"A Powerful Combination for Building Acoustics Measurements." *Brüel & Kjær Application Note* BO 0113 – 11.

"Intensity Measurements in Building Acoustics." *Brüel & Kjær Application Note* BO 0147 – 11.

We hope this booklet has answered many of your questions and will continue to serve as a handy reference guide. If you have other questions about measurement techniques or instrumentation, please contact one of our local representatives, or write directly to:

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