Room Acoustics and Sound Reinforcement Systems

Tadeusz Fidecki

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Introduction

Sound that propagates inside enclosures is reflected by the enclosure boundaries. The reflected waves are modified by the reflection properties of the walls and other surfaces. The size and shape of the surfaces as well as the wave frequency affect the creation and properties of the reflected wave.

The room volumes we live, work, speak and listen to music vary and the acoustic properties of that interiors strongly change over the audio frequency range. One of the important acoustical features of the rooms is the large range of the wavelength of sound in the audio frequency, from about 17 m at 20 Hz to several cm at 20 kHz.

The listening in small rooms is different from the listening in large rooms such as auditoria or concert halls. The physical analyses and listening experience indicate that the room properties depend on the frequency, particularly at low frequencies. This requires a combined approach using separate acoustics methods for the small and large rooms and for the low and high audio frequencies: wave theory, statistical and geometrical acoustic methods.

1. Wave theory approach

1.1 Eigenfrequencies and eigenmodes *)

The free waves in enclosures are standing waves that are created by the wave reflections from the enclosure walls. Figure 1.1 shows a rectangular enclosure with the dimensions L_x, L_y, L_z.

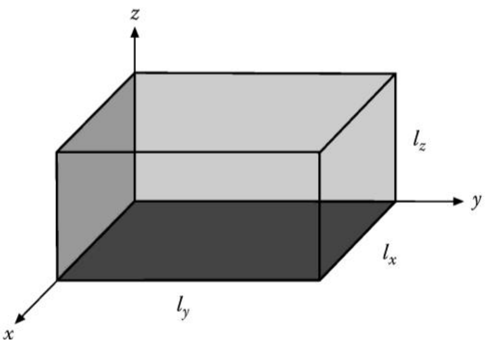


Fig. 1.1. Dimensions of rectangular room

*) "Eigenvalue" comes from the German "Eigenwert" which means proper or characteristic value. "Eigenfunction" is from "Eigenfunktion" meaning "proper or characteristic function".

The sound pressure is a function of three coordinates. The wave equation for the sound pressure of harmonic waves:

$$\frac{\partial^2 p}{\partial x^2}+\frac{\partial^2 p}{\partial y^2}+\frac{\partial^2 p}{\partial z^2}+k^2 p=0$$

In rectangular coordinate system, the eigenfunction can be separated into three functions that depend on one coordinate only.

$$p(x,y,z)=p_x(x)p_y(y)p_z(z)$$

Substituting into the wave equation results in the separation into three spatial functions:

$$\frac{1}{p_x(x)}\frac{d^2p_x(x)}{dx^2}+\frac{1}{p_y(y)}\frac{d^2p_y(y)}{dy^2}+\frac{1}{p_z(z)}\frac{d^2p_z(z)}{dz^2}=-k^2=-\left(k_x^2+k_y^2+k_z^2\right)$$

where *kx*, *ky*, *kz* are the wave numbers for wave components that propagate in one direction only. Equation can be split into three equations, each depending on one variable only [1]. For the variable *x*:

$$\frac{d^2p_x(x)}{dx^2}+k_x^2p_x(x)=0$$

Applying the boundary condition $u_x = 0$ at $x = L_x$, solution of the second-order differential equation results in:

$k_xL_x = m\pi,\ m = 0,1,2,3,\ldots$

and

$$k_m=\frac{m\pi}{L_x}\quad\text{and}\quad f_m=\frac{c}{2}\frac{m}{L_x}\quad m=0,1,2,3,\ldots$$

The wave number k_m is for a plane standing wave that propagates in the direction of *x*-axes and is therefore called axial wave. The general name of this wave is a mode and the frequencies $f_{m,n,l}$ are called characteristic frequencies or eigenfrequencies. Whenever the distance between the walls is a multiple of half wavelengths, a standing wave can exist.

The calculation may be repeated for the variables *y* and *z.* The expressions for the wave number and characteristic frequencies are:

$$k_{m,n,l}=\sqrt{\left(\frac{m\pi}{L_x}\right)^2+\left(\frac{n\pi}{L_y}\right)^2+\left(\frac{l\pi}{L_z}\right)^2}\quad\text{and}$$

$$f_{m,n,l}=\frac{c}{2}\sqrt{\left(\frac{m}{L_x}\right)^2+\left(\frac{n}{L_y}\right)^2+\left(\frac{l}{L_z}\right)^2}$$

The integers *m*, *n*, and *l* are called quantum numbers [1]. The combinations of these numbers results in various wave designations. If two of these quantum numbers are zero, the waves are called axial waves because they propagate in the direction of one of the room axes. A wave with the wave fronts perpendicular to one of the room walls is called a tangential wave (tangential to a wall). These waves have one of the wave numbers equal to zero. If none of the wave numbers is zero, the waves are called oblique waves. Their wave fronts have some general angle with the enclosure walls.

The sound pressure of an (*m*, *n*, *l*) mode:

$$p_{m,n,l}(x,y,z)=A\cos\left(\frac{m\pi}{L_x}x\right)\cos\left(\frac{n\pi}{L_y}y\right)\cos\left(\frac{l\pi}{Lz}z\right)$$

where *A* is the amplitude.

The properties of the modes are shown graphically for axial and tangential modes in following figures.

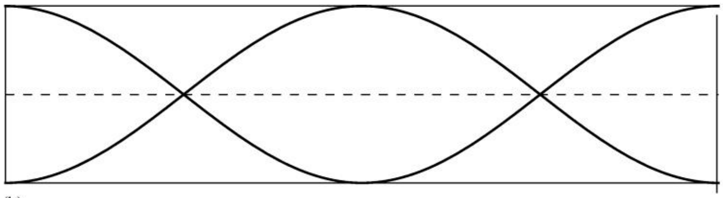
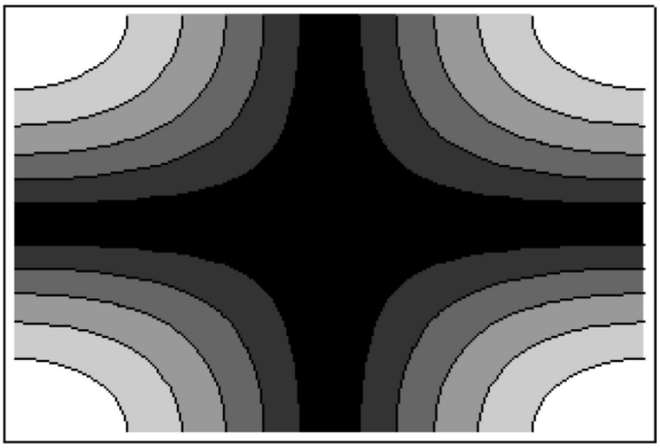
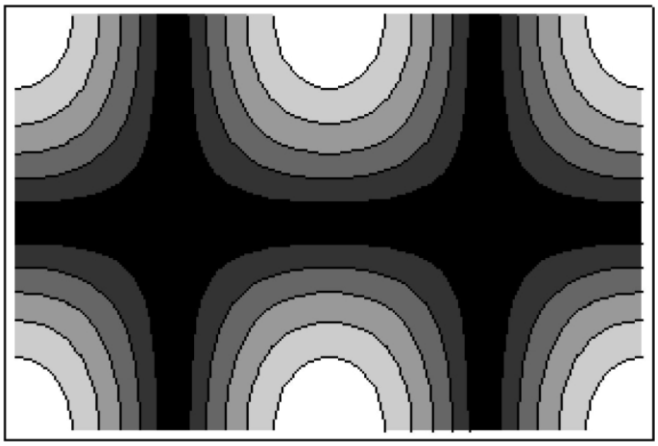


Fig 1.2. Standing wave in one dimension. A node is a point along a standing wave where the wave has minimum amplitude. Figure shows maximum and minimum pressure along the standing wave.



a) (1, 1, 0) mode



b) (2, 1, 0) mode

Fig. 1.3. (a) the pressure magnitude pattern of a (1, 1, 0) mode, (b) the pressure magnitude pattern of a (2, 1, 0) mode.

General findings for the pressure field in a room:

- all x, y, z modes: pressure maximum in corners,
- modes with one n = 0: pressure maximum at corresponding angles,
- modes with two n = 0: pressure maximum at corresponding surfaces.

The lowest eigenfrequencies calculated for an exemplary rectangular room of 4.7 x 4.1 x 3.1 m (c=340 m/s) are:

eigenfrequency [Hz]	nx	ny	nz
36.2	1	0	0
41.5	0	1	0
54.8	0	0	1
55.0	1	1	0
65.7	1	0	1
68.6	0	1	1
72.3	2	0	0

77.7	1	1	1
82.9	0	2	0
83.4	2	1	0

In real rooms, the walls may be covered with sound absorbers or sound diffusers and there is also furniture, loudspeakers and other objects. This makes calculation of the modal frequencies and spatial pressure distribution difficult. There are computational methods such as the finite element method (FEM) that can be used to predict the room modes and transfer functions.

Most objects in the rooms are usually small compared to the wavelength and the characteristic frequencies of the modes do not substantially differ from the frequencies calculated for a bare room.

The modes affect the sound transmission from the source to the listening position and the perception of the signal, both the fundamental frequencies and the overtones. The modification of the modes is generally difficult. Although most rooms have parallel walls, the sound field in rooms with nonparallel walls was studied. Fig. 1.4 shows the mode pressure distribution for a nonrectangular room calculated using the Finite Elements Method FEM. The lines in the figure represent constant sound pressure. Black areas have high sound pressure amplitude and white areas low sound pressure amplitude.

Fig. 1.4 (a) and (b) shows the two first axial modes. We see their similarity with the modes of rectangular rooms. Fig. 1.4 (c) and (f) shows higher-order modes. We note that for these modes the highest sound pressure amplitude is not necessarily found in corners.

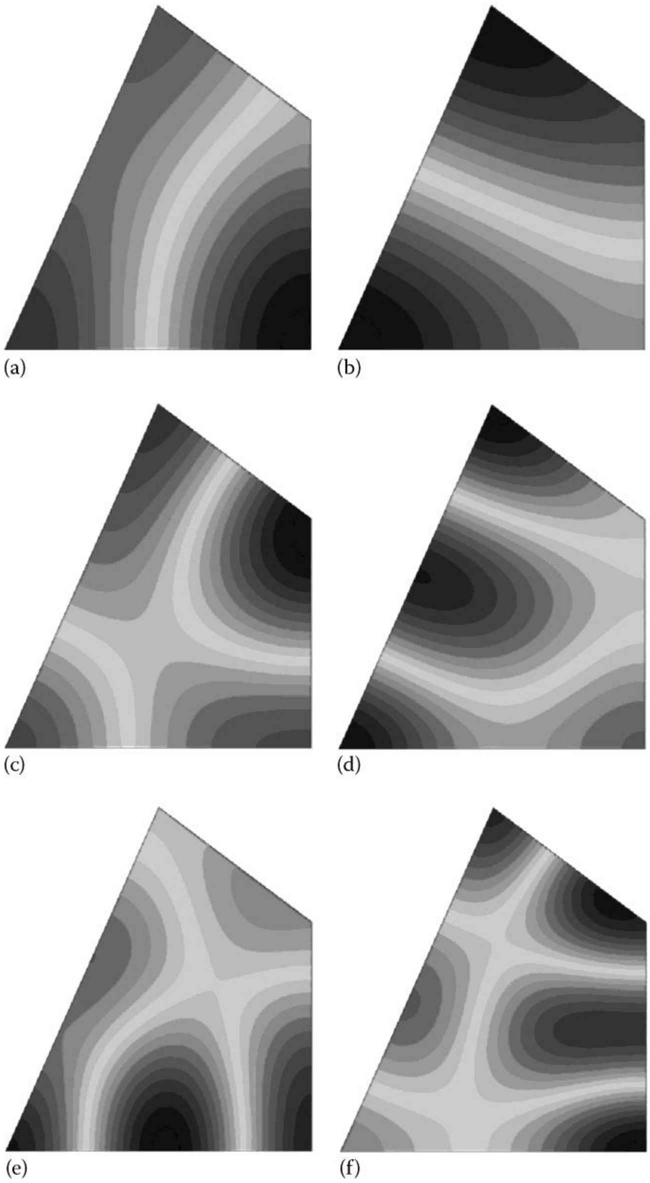


Fig. 1.4. Sound pressure magnitude distributions for waves tangential to the floor in a prism-shaped room with rigid walls, as found by the finite element method. Light regions have low sound pressure and dark regions have high pressure. Room wall lengths shown clockwise from top are 2.5 m, 3 m, 4 m, and 5.85 m. The resonance frequencies of the modes shown are (a) 52 Hz, (b) 43 Hz, (c) 77 Hz, (d) 82 Hz, (e) 99 Hz, and (f) 113 Hz [1]

1.2 Modal density

The number of modes *N* within the frequency interval 0 – *f* is the modal density.

$$N = \frac{4}{3}\frac{\pi V}{c^3} f^3 + \frac{\pi S}{4c^2} f^2 + \frac{L}{8c} f$$

where
$V = L_x L_y L_z$ is the room volume, $S = 2(L_x L_y + L_x L_z + L_y L_z)$ is the room surface, $L = 4(L_x + L_y + L_z)$ is the length of the edges.
The first term in the above equation is larger than the second and third term. At low frequencies, particularly at low volume, the number of modes is small but grows rapidly with f^3. For example, in a small room of volume 7 × 5 × 3 = 105 m3, the number of modes, between 0 and 100 Hz approximately, is *N* = 22. The first mode (axial) has the frequency 24 Hz. In a hall that has a volume of 36.000 m3, the number of modes under 100 Hz is approximately *N* = 3.800 and the frequency of the first mode is 4.8 Hz.

An important conclusion from this simple example is that the modes in small rooms are well separated in frequency and that the signal transmission is very irregular because at each modal frequency, the resonance effect elevates the sound pressure. The effects of mode separation depend on damping.

The mode separation can be followed on modal density *n,* that is, the number of modes in a unit frequency interval. It can be calculated as:

$$n = \frac{dN}{df} = \frac{4\pi V}{c^3} f^2 + \frac{\pi S}{2c^2} f + \frac{L}{8c}$$

For larger *f*, the first term dominates the other terms. With the earlier example we find out that at 100 Hz the calculated modal density is n = 0.545 modes per 1 Hz and at 50 Hz n = 0.202 modes/Hz. The frequency interval between the modes at 100 Hz is 1.83 Hz and at 50 Hz n = 4.95 Hz. When measuring the frequency response of the room, we will find the frequency intervals between the modes irregular. However, the calculated values provide good average numbers and a good help to estimate the *smoothness* of the frequency response.

1.3 Loss factor and reverberation time

When designing a room, the bandwidth of modes has to be sufficiently large. Figure 1.5 shows the energy E of a mode as a function of the frequency.

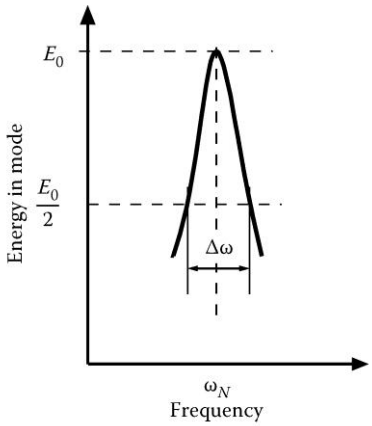


Fig. 1.5. Bandwidth of a mode.

The bandwidth of a mode is measured by the magnitude of its response in terms of $\Delta\omega$ or Δf at one-half of the energy in the mode. Because the energy of a mode is a function of pressure squared, we can calculate the bandwidth using the ratio of the one-half to the full energy to obtain the bandwidth of the mode as:

$$\Delta\omega = 2\delta_N$$

If the modes are well separated, this equation allows measurement of the modal damping from the frequency response or transfer function of the room. However, when the modes are close, the modal curves are not symmetrical and the measurements suffer from errors.

The energy in the mode is proportional to the pressure squared. If the room is fully excited by the sound and the sound source is turned off, the sound decay depends on the damping factor. The time dependence of the potential energy is given by:

$$p^2(t) = p_0^2 e^{-2\delta t}$$

where p_0^2 is the initial amplitude of the potential energy. The decay curve in the room can be measured and the reverberation time T_{60} can be determined from the slope of the decay curve. By definition, the reverberation time is the time needed for the sound energy to decrease to 10^{-6} (60 dB) of its original value. Substituting into the previous equation results in:

$$\frac{p^2(T_{60})}{p^2(0)} = 10^{-6} = e^{-2\delta T_{60}}$$

Sometimes, it is more convenient to measure the mode bandwidth Δf [Hz] than the reverberation time. The numerical evaluation leads to the relationship between the reverberation time and bandwidth that can be measured and from which the damping factor calculated:

$$\delta = \frac{6.91}{T_{60}}, \qquad T_{60} = \frac{2.2}{\Delta f}$$

Most small rooms are used for listening. Ideally, the response of the room should be frequency independent. However, due to the sound reflections from walls and objects in the room and the formation of the modes, the transmission of sound from the source to the listening position differs from point to point.

The properties of small rooms are particularly affected by modes. Following equation provides the information on the steady-state conditions and permits the analysis of sound transmission from a source point to a listening point.

$$p(r,k) \approx -jk\rho_0 c \frac{Q}{V} \sum_{N=0}^{\infty} \frac{\Psi_N(r_0)\Psi_N(r)}{\Lambda_N \left(k^2 - k_N{}^2 - j2k_N \frac{\delta_N}{c} \right)}$$

Fig. 1.6. shows schematically some modes at low frequencies and the total frequency response.

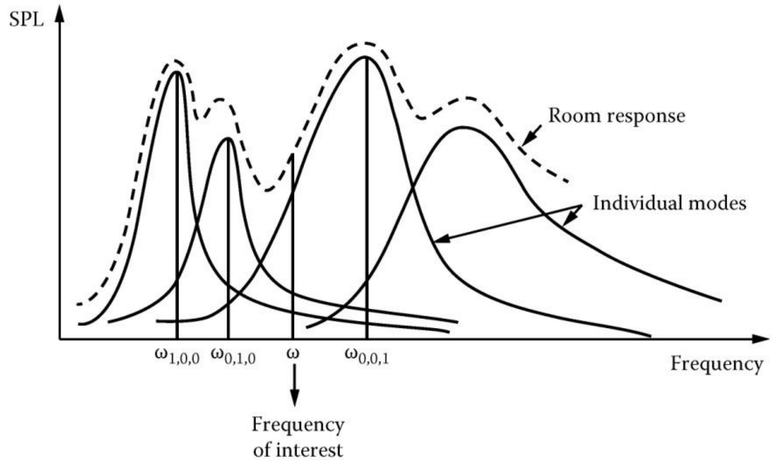


Fig. 1.6. Schematic modal contribution to room response

Due to a small damping and large frequency intervals between the modes, the room response is very irregular. Response of an individual mode varies from point to point in a room and the total response will differ in a similar way.

Fig. 1.7 shows the effects of modal damping. When the damping is low, most of the individual modes can be identified. In Fig. 1.7 b, many modes overlap because of the higher damping and only few modes are identifiable.

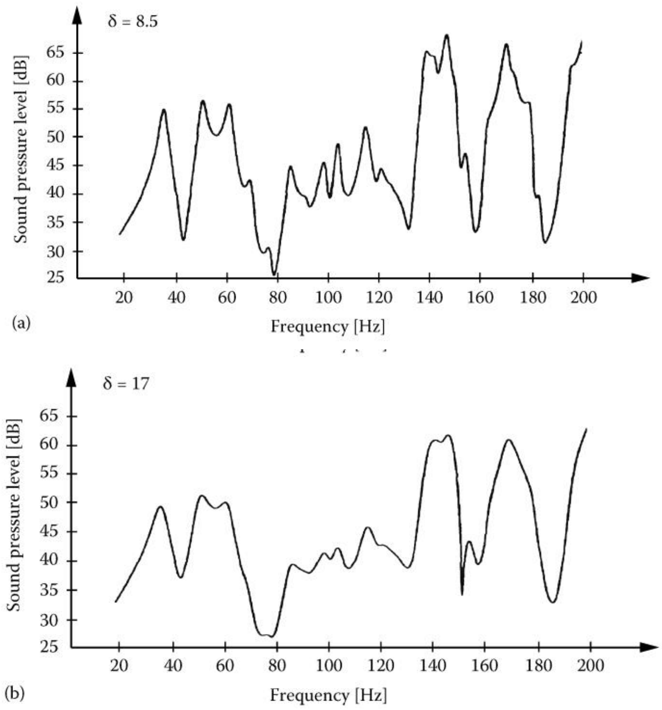


Fig. 1.7. Effect of damping on room response. (a) Little damping, (b) medium damping.

1.4 Conclusion

- in real rooms a number of Helmholtz equations, with boundary conditions defined, has to be solved in order to find the spatial distribution of the sound pressure,
- the eigenfrequencies of resonant modes are only obtainable in the low frequency range, and in rectangular or cylindrical room shapes,
- the method is useful to the preliminary analysis of small rooms,
- in case of bigger rooms and wider frequency range the statistical approach would be more practical.

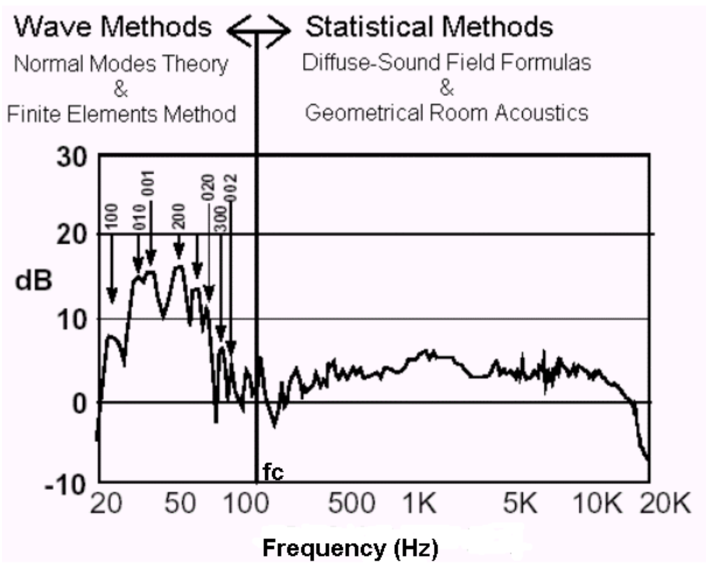


Fig. 1.8. Wave and statistical method in wide frequency range

2. Statistical method

2.1 The room response to the sound radiation

Sound field in enclosures is created by interfering modes. Because the modes exist only at certain frequencies, the frequency response at a point in the room is not smooth and reflects the resonance behavior of the modes even for a source that has a flat spectrum in a reflection-free environment. Particularly at low frequencies, the character of the sound fluctuations depends on the damping and density of modes.

At very low frequencies, only a few modes are excited and their frequencies may be identified from the frequency response curve. Because the modal density depends on the square of the frequency, the modal overlap is growing. However, the frequency response does not become smooth and is characterized by alternating maxima and minima due to wave interference. The individual modes can no longer be identified. The frequency response at different points varies.

The room response to the sound radiation can be expressed by statistical laws that were formulated by Schroeder [5] and further extended by Kuttruff [2] and others. Although there is no sharply defined frequency that separates the low- and high-frequency response parts of the room, we often use the *Schroeder* frequency as a defined division between low- and high-frequency responses of the enclosure. At this frequency, three modes overlap in one mode frequency bandwidth.

The Schroeder frequency f_s depends on modal density linked to the room volume and modal damping related to the reverberation time. It is defined in metric units as [6]:

$$f_S = 2000\sqrt{\frac{T_{60}}{V}}$$

where T_{60} is the reverberation time [s], V is the enclosure volume [m^3].

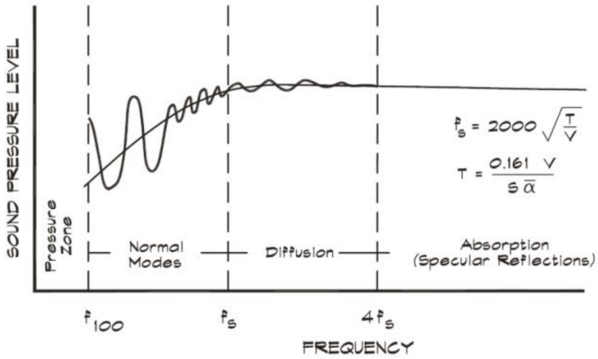


Fig. 2.1. Frequency regions in steady state room acoustic response [6]

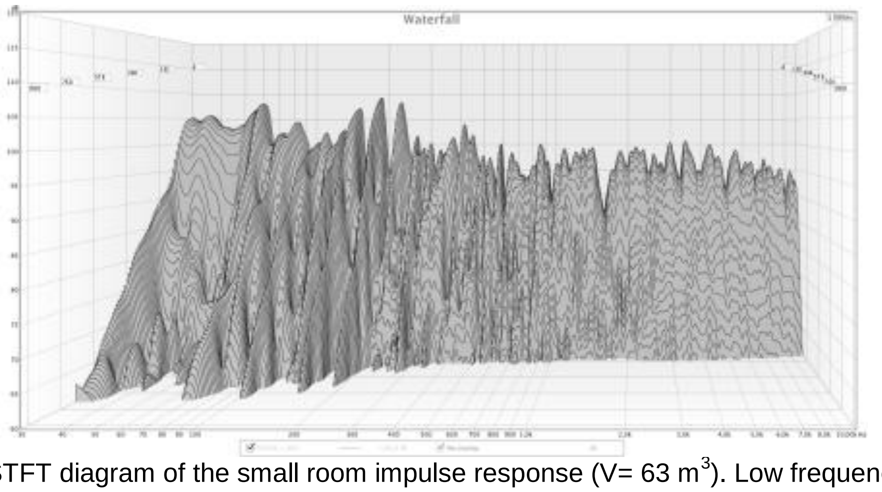


Fig. 2.2. Waterfall STFT diagram of the small room impulse response (V= 63 m^3). Low frequency resonances are seen below Schroeder frequency 200 Hz. Sound control room of the department of Sound Engineering, Fryderyk Chopin University of Music [7]

2.2 Reverberation in enclosures

The decay of sound after a sound source stops radiating is called reverberation. It is one of the most important processes in any room. It is related to the perceived quality of sound in a room.

When we listen to some sound in a room, we perceive the direct sound and successive sound reflected from the walls and other objects around us. The quality of sound perception depends on the ratio of the direct and reflected sound, type of signal, sound decay, and other physical parameters of the room.

The reverberation time is measured by the time that elapses for the sound pressure level to decrease by 60 dB from its initial steady-state level. The reverberation process is complicated and depends on the room shape, distribution of the absorbing surfaces in the room, and internal sound propagation. The derivation of reverberation time depends on various assumptions on the sound field in the room and, therefore, several different formulas for the calculation of reverberation time have been developed.

The importance of the sound decay for listening quality was first recognized by W. Sabine. Sabine established a concept of diffuse sound field. This is based on spatial uniformity of basic field descriptors such as the directional distribution of the plane waves that create the sound field, zero time-averaged sound intensity, uniform spatial energy distribution, and similar others. Such a field in strict sense cannot exist. However, the equation for the reverberation time derived by Sabine is in use due to its simplicity in many practical situations.

2.3 Growth of energy in enclosure

We assume that the walls have an absorption coefficient α. A source that radiates sound power *P* is placed somewhere inside the room. The energy increase in the room can be calculated from an energy equilibrium equation that assumes a diffuse sound field

$$P = V\frac{dw}{dt} + I_{inc}\alpha S = V\frac{dw}{dt} + \frac{wc}{4}\alpha S$$

The power *P* supplied by the source is consumed by the energy increase inside the room, expressed by the time-dependent term of the equation. At the same time, some energy is absorbed by the room walls of area S that have absorption coefficient α as expressed by the second term of the equation. The solution of this differential equation for $w = 0$ for $t = 0$ is

$$w(t) = \frac{4P}{c\alpha S}\left[1 - e^{-(c\alpha S/4V)t}\right]$$

Figure 2.3 shows the exponential time dependence of the energy density $w(t)$ increase that approaches asymptotically the value $4P/c\alpha S$.

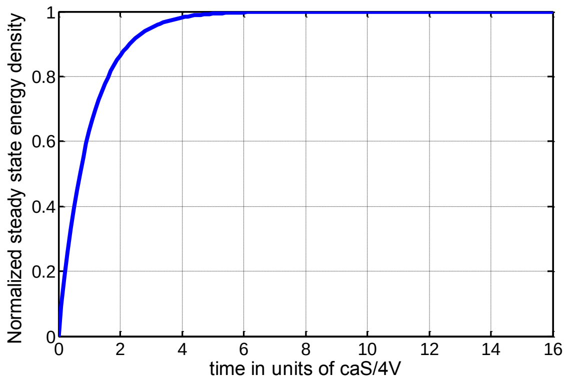


Fig. 2.3. The energy density increases asymptotically to the value where the energy supplied by the source equals the energy lost by sound absorption

2.4 Decay of energy in enclosure

The dependence of the energy density on time after the source has been turned off can be found by solving the equation for $P = 0$. We then obtain

$$w(t) = w_0 e^{-(c\alpha S/4V)t}$$

where w_0 is the initial energy density.

Figure 2.4 shows the energy density (and sound intensity) decay with time in a logarithmic scale that is more practical because of the large range of time decay values that are usually pursued. This idealized decay is represented by a smooth, straight line.

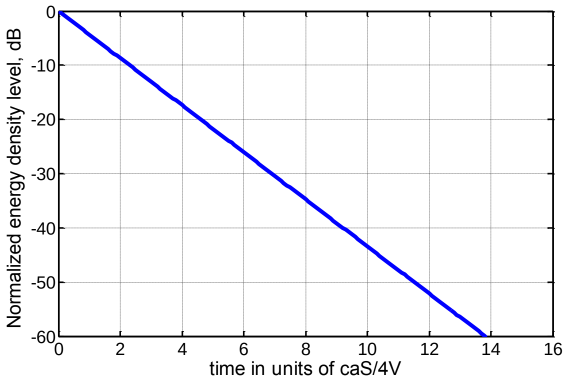


Fig. 2.4. The decay of energy using Sabine's model shown on a logarithmic scale and in decibels.

The actually measured time decay is affected, particularly in small rooms, by modes, various other resonance effects, and phenomena that make the decay curve irregular.

2.5 Reverberation time

The reverberation process is very important for the quality of listening in rooms. The quantitative definition of reverberation time is the time *T* in which the energy density of sound will decrease to one millionth of its original value or by 60 dB. Substituting for

$$w(T)/w_0 = e^{-(c\propto S/4V)T} = 10^{-6}$$

results in

$$T_{60} = \frac{24}{c\log_{10}(e)}\frac{V}{\alpha S} = 0.161\frac{V}{\alpha S}$$

where T_{60} is the decay time for a 60 dB sound pressure level decrease, *V* is the room volume, *S* is the wall surface, α is a uniform sound absorption coefficient. The constant 0.161 applies to air at 20°C and the metric system.

This formula is called Sabine's reverberation formula or equation. Although its precision is limited, it is often used because of its simplicity. The product α*S* is often called total absorption and is given in units of metric sabin [m^2 ,S]. The absorption coefficient is expected to be uniform over the wall surface *S*. However, in real rooms, the surfaces have patches that have different absorption coefficients. The total absorption is then achieved by summing the products of areas with the same absorption so that

$$\alpha S = \sum_{i=0}^{n} \alpha_i S_i$$

There is some sound attenuation when the sound propagates in air. The attenuation expressed in terms of sound intensity is given by

$$I(x) = I_0 e^{-mx}$$

where *m* is an attenuation coefficient that depends primarily on sound frequency, temperature, and humidity. The characteristics of *m* are shown in Fig. 2.5.

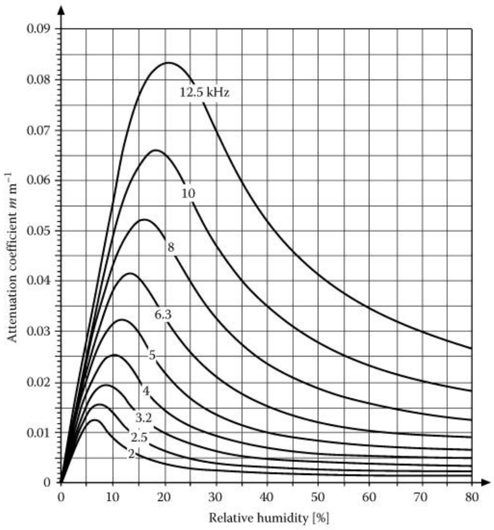


Fig. 2.5. Values of the attenuation coefficient m for air at a temperature of 20°C, for various values of relative humidity and frequency. The attenuation coefficient m depends primarily on sound frequency, temperature, and humidity. (After J. Acoust. Soc. Am., 40, 148, 1966.)

Taking into account the sound absorption in air the Sabine's formula modifies to:

$$T_{60} = 0.161\frac{V}{S\alpha + 4mV}$$

If the sound absorption by the walls is small, approaching zero in the limit, the reverberation time increases correctly to infinity. However, for large absorption where $\alpha \to 1$, as in the anechoic chamber, the reverberation time becomes finite although it should approach zero. There are other formulas that improve the reverberation time prediction.

Sabine's equation was derived under the assumption that the sound field is diffuse and that the sound power is absorbed by the walls continuously. In Eyring's approach, the sound is absorbed stepwise at each reflection and not continuously.
Using the relationships derived earlier

$$w = \frac{4P}{c\alpha S}\left[1-(1-\alpha)^{(cS/4V)t}\right] = w_0\left[1-e^{\ln(1-\alpha)(cS/4V)t}\right]$$

where:
$w_0 = (4P/c\alpha S)$ - the energy density reached after $t \to \infty$.

Examining the earlier equation for $t = 0$ and $t \to \infty$, we obtain for w the correct values 0 and w_0.

The energy density after the first reflection time τ is $w_\tau = w_0(1 - \alpha)$ and after k reflections

$$w_{k\tau} = w_0(1-\alpha)^k = w_0(1-\alpha)^{(cS/4V)t} = w_0 e^{\ln(1-\alpha)(cS/4V)t}$$

The reverberation time T_{60} can now be obtained:

$$\frac{w}{w_0} = 10^{-6} = e^{\ln(1-\alpha)(cS/4V)T_{60}}$$

and

$$T_{60} = -\frac{24\ln(10)}{c}\frac{V}{S\ln(1-\alpha)+4mV} = -0.161\frac{V}{S\ln(1-\alpha)+4mV}$$

This equation extended by the air attenuation component is called Eyring's formula. It differs from Sabine's formula by the denominator. The results are practically equal for $\alpha \leq 0.1$ and, therefore, Sabine's equation is often used for rooms with hard walls. For $\alpha > 0.1$, the magnitude estimations of the two reverberation time equations differ.

The reverberation time prediction using these two formulas may be reasonably usable for enclosures with absorption distributed over all walls and a sound field, which is reasonably diffuse.

The reverberation time can be measured in many ways, including the following *switch-off method*. A room is excited with a noise source until a stationary average sound-pressure level is reached. Then the sound source is switched off, and the sound-pressure level is recorded as a function of time, producing a curve like the one shown in Fig. 2.6. The time interval between switching-off the source and the instance where the curve has decreased by 60 dB, is taken to be the reverberation time, *T*.

Fig. 2.6. Time trace of a reverberant noise sound after being switched off [3]

The reverberation time, *T*, estimated with either *Eyring*'s or *Sabine*'s formula, is considered to be a relevant parameter of the *acoustic quality* of spaces, which includes, among other things, their suitability for specific performances. The value of reverberation time increases with the volume of the hall. Table 2.1 presents preferred values of *T* for various performance styles. Beranek [8] presented the formula for the average optimum reverberation time for a given auditorium volume based on the collected, subjective results.

The values are drawn from literature and refer to the 500–1000 Hz range. A moderate increase toward low frequencies is considered adequate since it is said to increase the listeners' sense of envelopment and warmth.

Table 2.1. Recommended values of reverberation time

Speech	Chamber music	Opera houses	Concert halls	Organ music
0.8 – 1.0 s	1.4 – 1.6 s	1.5 – 1.7 s	1.9 – 2.2 s	2.5 s and more

The plot in Fig. 2.7 shows the average optimum reverberation time *T* for a given auditorium volume *V* solved numerically from the formula [8]:

$$log_{10}V = 5.72 + log_{10}T - \frac{2.43}{\sqrt{T}}$$

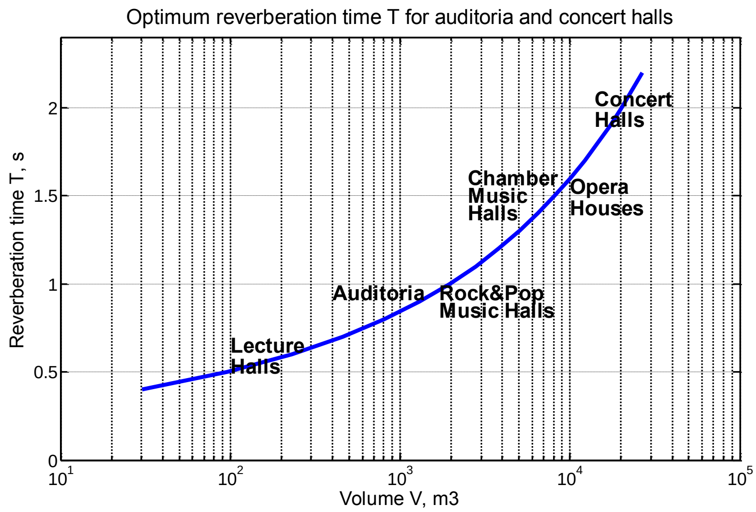


Fig. 2.7. Optimum reverberation time *T* versus auditorium volume *V*

The following guidance holds for speech. A too-short reverberation time produces high intelligibility but also increases the effort required from the speaker. If *T* is too long, auditory smearing takes place and deteriorates intelligibility. 0.8–1.0 s is a reasonable compromise for speech running at normal speed, which is roughly 50 syllables/minute.

A comment concerning the reverberation time of the halls for amplified, pop and rock music: The acoustics of these halls significantly differ from classical music interiors because of the presence of strong very low frequency components in the pop and rock music. The sound reinforcement speakers are mainly responsible for delivering the sound to audience and the influence of the hall acoustics is minimized. Reverberation time in the 125 Hz octave band is the most critical parameter for the acoustic quality of a venue for pop and rock music. The recommended value for T does not exceed 1 s for halls from 1.000–7.000 m^3. Gradual increase of the reverberation time to 2 s is suggested in larger volumes of 50.000 m^3 [9].

Reverberation time is undoubtedly an important parameter of acoustic quality but certainly not the only one. Proper guidance of early reflections and avoidance of echoes are at least as relevant.

2.6 Direct and diffuse sound field in enclosure

In the steady-state sound field generated by a wide band sound source the energy density of the sound field can be considered as consisting of two components, the direct w_d and the reverberant w_r.

The total energy density w at the observation point is:

$$w = w_d + w_r$$

Let us assume that the reverberant sound field is diffuse, i.e. has the same average energy density everywhere in space. Figure 2.8 shows schematically the sound pressure level as a function of distance from the source. Assuming a small omnidirectional source, the sound pressure will decrease with the distance. The reverberant field will be constant beyond some distance where the reverberant field energy density dominates.

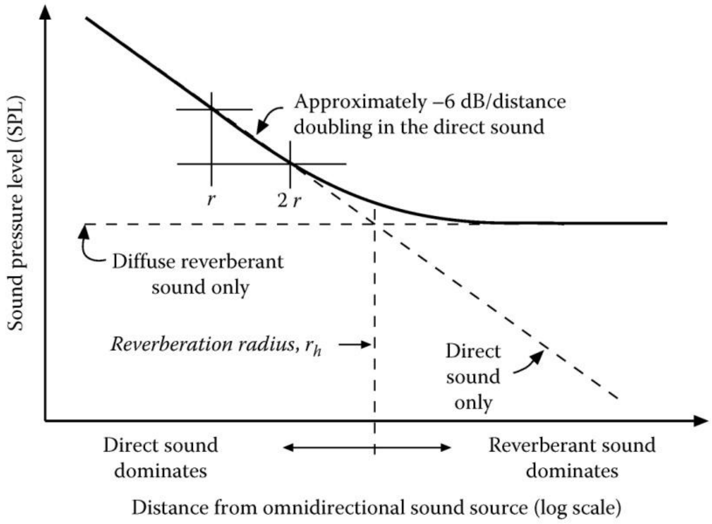


Fig. 2.8 The drawing shows the theoretical behavior of the steady-state sound pressure level as a function of distance from an omnidirectional sound source in a room with an ideally diffuse sound field. The reverberation radius r_h is where the direct and the reverberant fields are equally strong

The sound pressure squared of the direct field p_d^2 of a small source that has a directivity factor *Q*, defined as a ratio of the power radiated in certain direction to average power radiated in all direction, is at a distance *r*:

$$p_d^2 = \rho c \frac{PQ}{4\pi r^2}$$

where *P* is the radiated power. An omnidirectional source has *Q* = 1.

The sound power that creates the reverberant field is the sound power that exists in the room after the first reflection. Therefore, the original power *P* has to be multiplied by (1 − α). The energy density is of the reverberant field is

$$w_r = \frac{4P(1-\bar{\alpha})}{c\bar{\alpha}S} = \frac{4P}{cR}$$

where
α is the average absorption coefficient, *S* is the surface of the walls, *RC* is called the room constant which characterize the room properties.

$$RC = (\bar{\alpha}S) / (1-\bar{\alpha})$$

The total sound pressure squared is:

$$p^2 = \rho_0 c P\left[\frac{Q}{4\pi r^2} + \frac{4}{RC}\right]$$

We now divide sound pressure *p* and sound power *P* by the reference levels for $p_0 = 2\cdot10^{-5}$ Pa and $P_0 = 10^{-12}$ in order to obtain levels in dB.

$$L_p = L_P + 10\log_{10}\left[\frac{Q}{4\pi r^2} + \frac{4}{RC}\right]$$

(L_p – sound pressure level, L_P – sound power level)

The distance at which the energy density of the direct field and the reverberant field are equal is called the reverberation radius r_h or the critical distance r_c.

$$r_h = \sqrt{\frac{Q\cdot RC}{16\pi}} = \sqrt{\frac{Q}{16\pi}\frac{1-\bar{\alpha}}{\bar{\alpha}S}}$$

In case of omnidirectional source, Q = 1, the reverberation radius depends on the room properties only and therefore is named room radius r_0

.

$$r_0 = \sqrt{\frac{RC}{16\pi}} \cong 0.057\sqrt{\frac{V}{T_{60}}}\text{, m}$$

where
V - room volume, m^3, T_{60}, reverberation time, s.

Knowing the reverberation radius or critical distance is useful when taking sound recordings because microphone placement within this radius or distance will predominantly render direct-sound signals. Placement outside will predominantly render diffuse-sound signals, which are auditorily perceived as *spatial impression*.

3. Geometric Acoustics and Diffuse Sound Fields

The procedure for prediction of the acoustical properties of real rooms becomes very complicated, when treating sound fields inside rooms with complicated shapes like concert halls or churches. An approximate method called *geometrical acoustics* is often useful in these cases.

This method considers sound propagation in terms of so-called *sound rays*. The idea is that the wave bundle propagates along a straight line like a ray of light.

Two restrictions are important in geometrical acoustics:
1) geometrical acoustics is only applicable for mid- and high frequencies of sound where the walls and objects are large compared to wavelength,
2) geometrical acoustics is primarily useful for the study of the behavior of the initial reflections of sound in the room, typically the sound that arrives reflected at the listener within 30 ms of the direct sound.

The concept of rays is mathematically achieved by maintaining plane areas of constant phase and letting the wavelength go to zero. The wave propagation can be approximated by rays when the following condition is met. The wavelength of the sound under consideration must be small compared to the linear dimensions of boundary areas and obstacles. Diffraction is neglected in this view.

The energy density, *W”,* within a ray is equal to the energy density in a plane propagating wave. Rays are usually considered to be incoherent so that their energy densities superimpose when they meet. The sum up of the rays is

$$\sum W''_{\text{ray}} = \frac{\sum|\vec{I}|}{c} = \frac{\overline{I}_{\Sigma}}{c}$$

The assumption that the rays are incoherent is valid for most broadband signals like speech or music, assuming that the rays have traveled different distances from the source. This is not the case, however, for impinging and reflected waves close to reflecting surfaces. Incoherence can also not be assumed for narrow-band or pure-tone signals.

3.1 Mirror Sound Sources and Ray Tracing [3]

The behavior of rays at plane reflecting surfaces is particularly relevant for geometrical acoustics. Plane means here, that any unevenness of the surface is small compared to the wavelengths of the sound considered. The reflection law, $\Theta_1 = \Theta_1'$ holds, and may even be applied to slightly curved planes as long as the curvature is small compared to the wavelength – shown in Fig. 3.1.

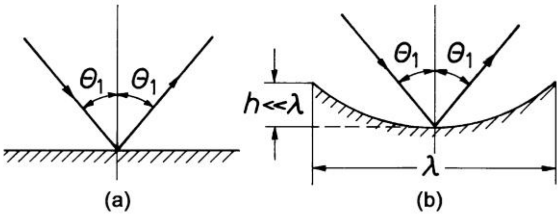


Fig 3.1 Reflection of sound rays at **(a)** planes, and **(b)** moderately curved plates

When using the concept of sound rays, relevant rules and laws from optics can directly be applied. The length of a ray is proportional to its traveling time, making it possible to not only determine the direction of sound propagation, but also the arrival times of different rays at a certain point of interest.

Reflection on plane surfaces can be depicted by *mirror sources* (*virtual sources*) – shown in Fig. 3.2 The mirror source, q_m , and the primary source, q_0 , simultaneously send out identical sound fields. The combination of these sound fields on the surface produces a reflected wave that fulfills the boundary condition for full reflection, namely, the normal component of the particle velocity, *v*, being zero.

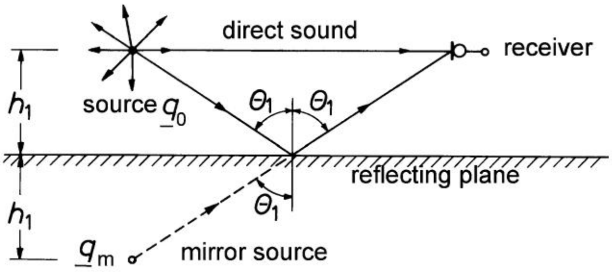


Fig. 3.2 Mirror sound sources emanating from reflection at a plane

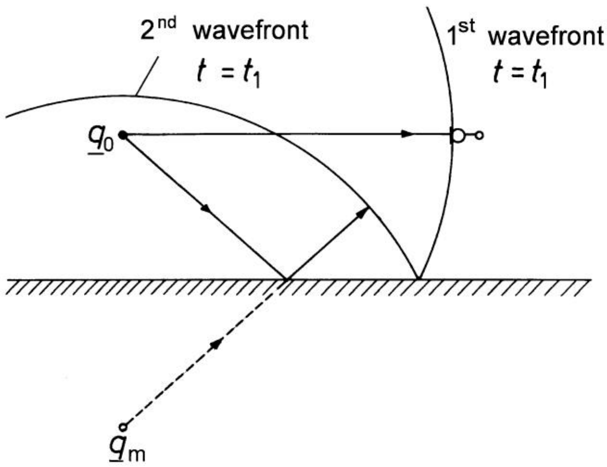


Fig. 3.3. Wave fronts of both the primary and the reflected sounds

Investigations into the relative arrival times of reflections are important, especially since reflections that arrive at the receiver with a delay may cause the perception of disturbing echoes, which should be avoided in room acoustics. The actual perceptual echo threshold is dependent on the character of the sound. It is about 50 ms for running speech, larger for music and shorter for impulses.

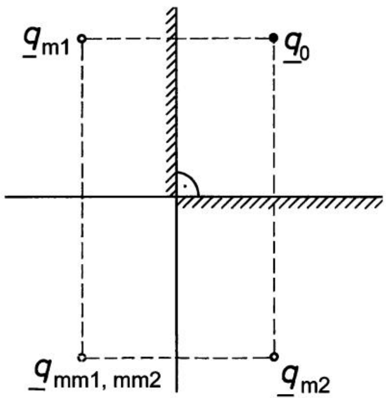


Fig. 3.4. Mirror sources at edges and in corners

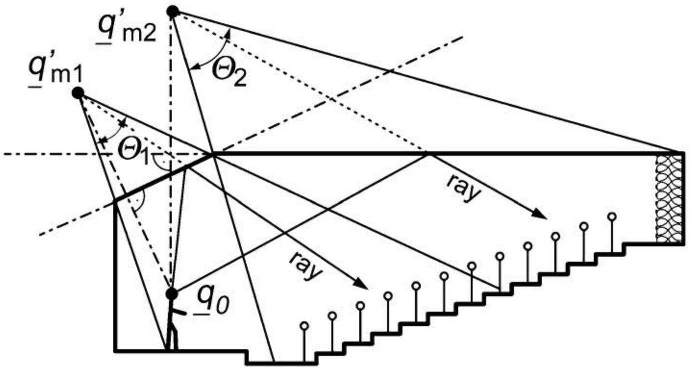


Fig. 3.5. Guiding sound via the ceiling of an auditorium

Fig. 3.5 presents an application example for geometrical acoustics. In an auditorium, the sound ray from a speaker, q_0 , is guided to the audience via ceiling reflections. q_{m1} and q_{m2} are the mirror sound sources representing the tilted and horizontal parts of the ceiling, respectively. Two sample rays are depicted for illustration. The two mirror sources illuminate the spatial sections *Θ*1 and *Θ*2. The rear wall is made absorptive to avoid audible echoes.

As to the construction of the graph please note that the mirror sources are positioned perpendicularly to the reflecting surfaces at the same distance to the surface as the original source, outside the room under consideration. The rays originating from them are restricted to the spatial sector defined by the individual reflecting surfaces concerned.

3.2 Flutter Echoes

We will consider a case involving a highly directional sound source, p_1, between two parallel walls a distance, *l*, apart. The source emits a short sound pressure impulse directed perpendicularly toward one of the walls and propagating like a ray – shown in Fig. 3.6(a). The two walls may be slightly absorbent, characterized by a degree of absorption, *α*. A microphone close to the position of the source would records a signal as schematically plotted in Fig. 3.6(b).

If the interval between the individual impulses at the receiver, $\tau = l/c$, is larger than the echo threshold, the impulses become perceptible as a series of individual echoes, called *flutter echo*. Flutter echoes should be avoided in room acoustics. This can be accomplished by slightly tilting the two walls by > 5° or by making their surfaces absorbing or scattering.

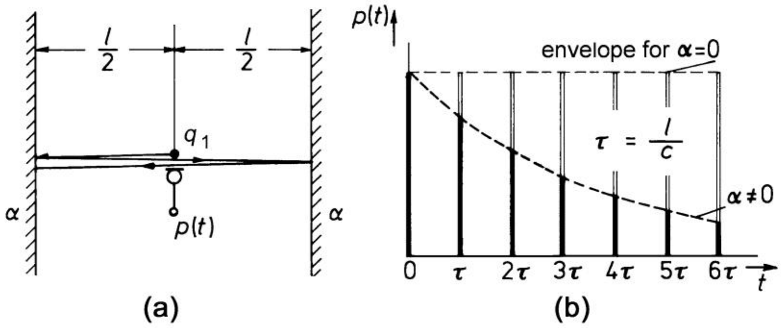


Fig. 3.6. Multiple reflections between parallel walls – the origin of flutter echoes

The envelope of the impulse series decreases exponentially for $\alpha > 0$, meaning that the ray looses a given percentage of its energy whenever a reflection takes place. This decreasing exponential function is actually an analytical description of the decreasing envelope in Fig. 3.6(b).

$$W''(t) \approx W''_0 \, \mathrm{e}^{-n'\alpha\, t}$$

3.3 Impulse Responses of Rectangular Rooms

We will now move beyond the case of two parallel walls and consider a rectangular (cuboid) room with six reflecting boundaries, namely, four walls, one floor and one ceiling. This room will illustrate an important rule in room acoustics which states that the reflection density, *n'*, increases with t^2 in many rooms.

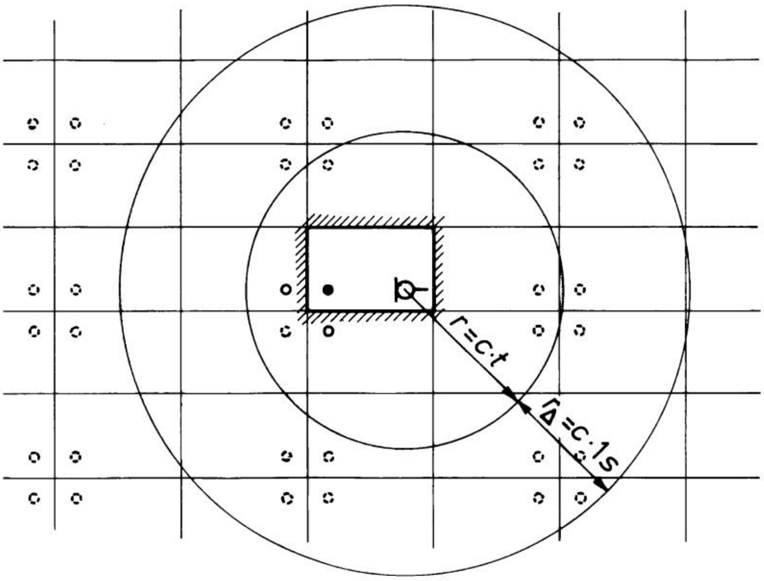


Fig. 3.7. Image sources of one sound source for a rectangular room

Fig. 3.7 illustrates this concept. The figure shows the plan of a rectangular room with a single sound source in it, along with mirrored rooms of n^{th} order with one mirror source in each of them.

Let *V* be the volume of the cuboid room. Now all of the mirror sources simultaneously transmit a sound impulse at $t = 0$. All impulses that originate from within a hollow sphere with the radius $r\Delta = c{\cdot}1\ s$, arrive at the receiver in the original room within the same interval of 1 s. The number of mirror sources in the hollow sphere is approximately the volume of the hollow sphere divided by the volume of the original cuboids, *V* , which is

$$n'(t) = \frac{4\pi\, r^2\, r_\Delta}{V \cdot 1\,\mathrm{s}} = \frac{4\pi\, c^3\, t^2}{V}$$

In other words, the density of the impulses arriving at the receiver is increasing with the square of expired time. The reflections also come from more directions over time, resulting in an ever more homogeneously distribution both over time and space.

Fig. 3.8 is a simplified illustration of what is called an *echogram*, particulary, an impulse echogram. We see the direct sound and the early, low-order reflections as discrete event. Then the echogram becomes denser and denser, so that individual impulses can no longer be discriminated. This late part of the echogram is called *reverberation tail*.

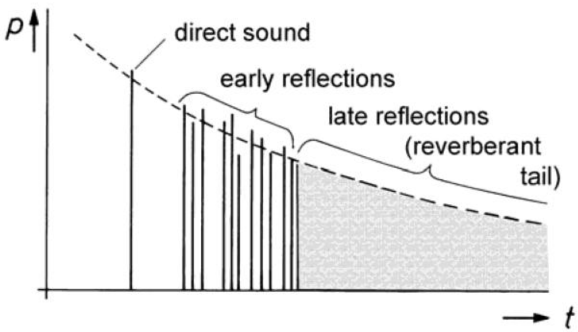


Fig. 3.8. Echogram – the time distribution of the energy arriving at receiver

3.4 Diffuse Sound Fields

From the discussion in Section 3.3, it is obvious that it is hardly possible to trace the fate of each individual sound ray, particularly in the reverberation tail. Nevertheless, it is possible to make important statements about the average fate of late reflections. Such an approach is called *statistical room acoustics*. We begin with an idealized model that adequately describes the sound field of the reverberant tail, also called the *diffuse sound field*. The model *diffuse sound field* is characterized by the following assumption, expressed in term of geometrical room acoustics.

A *diffuse sound field* is composed of many rays with the average properties of equal intensity and equal spatial distribution.

This assumes that all rays, on average, have been reflected the same number of times and have, on average, traveled the same distance. It also means that the mean free-path length between two reflections is the same for all rays. The results of statistical room acoustics are independent of room shape because only the average fate of rays is considered and described by statistical parameters.
This leads to the expression for the mean free-path length as

$$\bar{l} = \frac{c}{\overline{n}'} = \frac{4V}{A}$$

Consideration of the theory an actual reverberation plots shows that the *diffuse sound field* assumption is sufficiently valid when the following conditions are met:

- sound absorption is well distributed about the boundaries of the room,
- the shape of the room is irregular and focussing elements are particularly avoided,
- total sound absorption is small or moderate. A good check is if the ratio of room volume and equivalent absorptive area is not too small, usually > 1m.

In most ordinary rooms the diffuse sound field is not a good approximation. Each of the following conditions may indicate that the sound field is not diffuse:

- an uneven distribution of sound absorption on the surfaces, e.g. only one surface is highly absorbing,
- a lack of diffusing or sound scattering elements in the room,
- the ratio of longest to shortest room dimension is higher than three,
- the volume is very large, e.g more than 5000 m^3.

In large rooms with medium or high sound absorption ($\alpha > 0.2$) the sound pressure level will continue to decrease as a function of the distance, because the diffuse field theory is not valid in such a room.

3.5 Ray tracing and mirroring [1]

Two methods are commonly used in geometrical acoustics for prediction of impulse response: the ray tracing method (RTM) and the mirror image method (MIM). Both are based on the use of Fermat's principle that states that a wave travels by the quickest route and, in homogeneous air sound follows straight lines, thought of as rays.

When RTM is used, the sound power can be thought to move away from the source as rays or particles that carry a part of the total power radiated by the source. Usually, each ray or particle carries a fraction of the power that is inversely proportional to their number. The ray path is a straight line unless sound is reflected or scattered by planes and objects. To find the ray paths, we send out an imaginary ray or sound particle and follow it as it moves in space as time grows.

An alternative to ray tracing is mirror image analysis. Mirror images are a result of sound reflection by plane rigid surfaces. The sound that arrives at the listener may be thought to come from the mirrored copies of the source visible to the listener. When the ray hits a mirror, whether plane or curved, it will be reflected. For a mirror, the incident and reflecting rays as well as the normal to the surface are in the same plane, and the angles of incidence and reflection are equal. Once the mirrored copies of the source are found, we can draw rays from the mirror images of the source to the listener.

The number of times a reflection has been mirrored, or a ray path reflected, is called the mirror image or ray reflection order. Mirror image analysis will give the same result as analysis by ray tracing if a large number of rays are studied. Impulse response measurements in real rooms will typically show large deviations from the calculated impulse responses. Nevertheless, the RTM and MIM are important practical tools in analyzing the reflection paths of a room since both methods often give sufficient results for practical work in room acoustics in contrast to the wave theory for the same frequency range.

3.6 Sources and receivers

The following approximations and assumptions are typical for geometrical acoustics:
- sources and receivers are small,
- surfaces (of room boundaries and objects in the room) are plane and rigid and have dimensions much larger than the wavelengths of sound considered,
- sources are unaffected by their surroundings,
- source signal is assumed to be a *short pulse* or *wavelet*.

A sound source that is small compared to wavelength can be regarded as a point source but—by suitable approximation—also directional sources such as voice, musical instruments, and loudspeakers can be studied using geometrical acoustics. In the use of the MIM, the receiver is assumed to be a point, whereas in ray tracing, small spheres, polyhedra, or simply cubes are used as targets to collect the rays. In computer simulation of ray tracing and mirror imaging, both sources and receivers can be assigned complicated directivity such as that of voice, musical instruments, microphones, and hearing.

3.7 Planes

Elementary ray tracing and mirror imaging is assumes surfaces that are infinite, rigid, plane, and smooth. However, few physical surfaces can be considered as perfectly reflecting planes in the sense of geometrical acoustics. Physical surfaces are neither infinitely large, perfectly plane, nor rigid. At low frequencies, the room dimensions are too small for the room surfaces to be large compared to wavelength, whereas at high frequencies, the irregularities caused by poor workmanship and unavoidable lack of precision are so large as to make the surfaces more scattering than mirroring.

In practice, it is customary to consider as ideal mirrors such nominally plane surfaces that have dimensions larger than about three wavelengths and that have random surface irregularities smaller than 1/16 wavelength. A further requirement is rigidity, which is assumed to be present when the impedance of the surface is much larger than the sound field impedance. When the angle of incidence is larger than about 80° (*grazing incidence*), most surfaces will appear mirroring. RTM and MIM are only useful to study the major paths of sound propagation. When surfaces are curved, ray tracing is intuitively simpler to use than mirror imaging. Additionally, RTM does not require the user or software to keep track of the mirror images and find their visibility.

MIM is however more exact than RTM, but since building practice is not exact, this advantage is of little practical use. A curved surface may be subdivided into smaller surface patches and analyzed as previously or one can use the mirroring laws of optics. Subdividing large surfaces into small surfaces to take into account small irregularities will result in difficulties both in the case of ray tracing and in image analysis. Patches must be several wavelengths large to act as mirrors in either case. Keeping track of the visibility of many mirror images is cumbersome as is tracing the myriad of rays necessary to hit all small surface patches.

Because of the multitude of rays or images, it is tiresome to analyze the sound propagation in small rooms by manual ray tracing for higher orders than the first or second order of reflection. Higher orders can be calculated using computers, but both the resulting accuracy and precision will be low because of the previously mentioned inexactness of building practice that along with limited wall size and wall unevenness leads to scattering rather than specular reflection of sound. The higher the reflection order, the more destructive will their influence be on the accuracy of geometrical acoustics prediction.

3.8 Multiple reflecting planes

The impulse response at the listener from a sound source in a room is determined fundamentally by the geometry of the room in the order by which the sound is reflected by the room surfaces. The case of two opposing surfaces shown in Fig. 3.9 is fundamental. Note that the surfaces are mirrored by one another an infinite number of times although this is not indicated in the figure. The sound will bounce back and forth, reflected an infinite number of times. If the planes are not parallel, the mirror images will be located along curves instead of along a straight line.

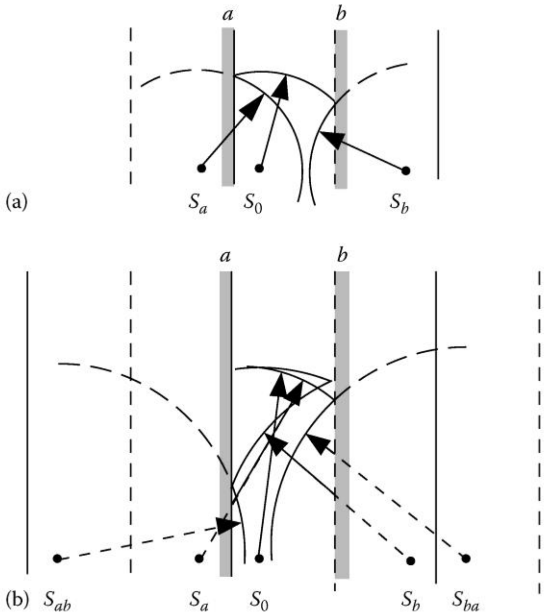


Fig. 3.9 Mirroring by two parallel surfaces a and b. (a) shows the sound source and the first-order mirror image sources and (b) shows conditions after some additional time and the additional second-order mirror images necessary to study the wave fronts. For each time instant, the radius of curvature is determined by the speed of sound and the time from the start of the sound.

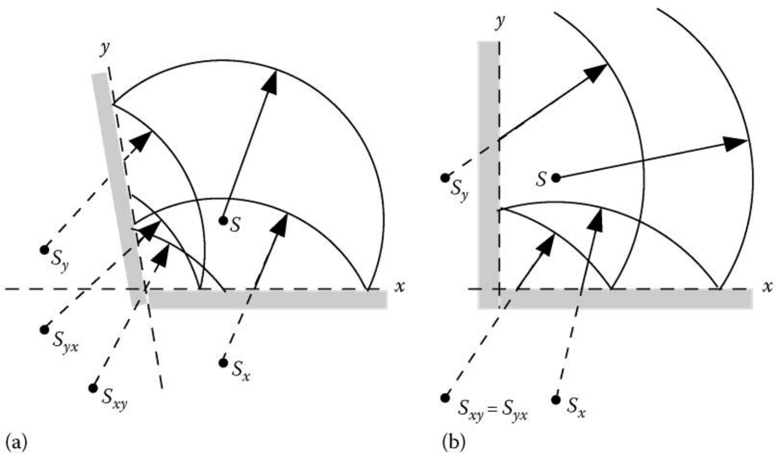


Fig. 3.10 Mirroring of sound at the corners of intersecting surfaces, showing first- and second-order image sources for two cases: (a) surfaces not at right angle and (b) surfaces at right angle.

In the case of nonparallel surfaces, the mirroring will be more complicated, exemplified in Fig. 3.10. In this case, there are four different mirror image sources since rays will be reflected twice (note that two of the image sources have identical locations in the case of the 90° corner).

Each time the wave is reflected by a wall surface, it will lose a part α of its energy, as heat or by transmission to some other acoustic system. The distances in 3D space between the mirror sources and the listener are easily calculated since here each path is the space diagonal between the image source and the receiver. Additionally, in a real room, the images will become diffused and the associated rays scattered.

If an incoming ray with intensity I_0 is specularly reflected by a surface that has a sound-absorption coefficient α, its reflected intensity I_R will be reduced:

$$I_R = I_0(1 - \alpha)$$

Many absorbers will scatter some of the incoming sound because of material inhomogenity and/or limited size. Resonant absorbers such as the Helmholtz and membrane absorbers always scatter their reflected energy. For ray tracing, there are many ways to simulate scattering of sound, such as giving the reflected rays random reflection angles or to reflect a multitude of weaker rays.

3.9 Geometrical impulse response

The geometrical impulse response (GIR) of a room is the impulse response determined by the mirror image or ray distribution of reflected sound and thus is determined by room geometry and size. An example of such a *geometrical* impulse response is shown in Fig. 3.11. The region between t_D and t_{R1} can be called the geometrical acoustics anechoic or free time.

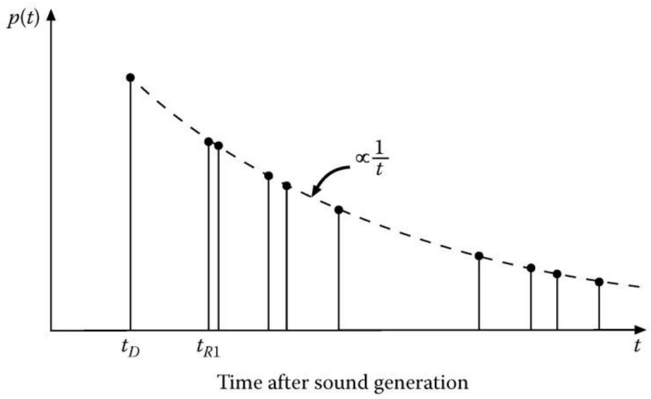


Fig 3.11 An example of the GIR of a room

Assume that source radiates a power W_S. The intensity I of sound at distance d will then be $W_S/4\pi d^2$ since the sound power is spherically distributed around a small non-directional source. If the source has been reflected N times with a mean-free-path length l_m and the reflecting surfaces have a mean sound-absorption coefficient α_d, the intensity of the arriving sound will be

$$I_N \approx \frac{W_s}{4\pi\left(N \cdot l_m\right)^2}\left(1-\alpha_d\right)^N$$

We can formulate a condition for how small the surface can be to function as an acceptable reflector of a plane wave of sound for a certain wavelength λ, under condition of source and receiver distances being very large compared to wavelength. Assuming, for example, that the source is so far away that its sound field is almost plane at the mirror, the only variable will then be the distance d between receiver and mirror. It is reasonable to assume that the surface must contain at least the inner half Fresnel zone. This approximation results in the equation

$$l_{limit} = \frac{d^2}{2\lambda}$$

where d is the smallest surface dimension, l_{limit} is the distance to the observation point.

This condition is sometimes used when one needs to design sound reflectors for auditoria. Note that a reflecting sheet must have sufficient surface mass (mass per unit area) so that it remains in place and is not moved by the sound field. An object acts as a diffuser if it is small compared to the wavelength.

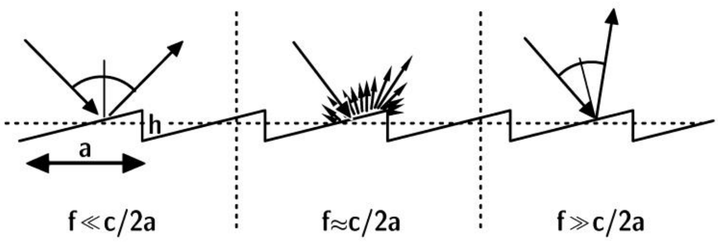


Fig. 3.12. Scattering caused by surface corrugations

3.10 Impulse response of real rooms

Because of the wave-related phenomena, real room impulse responses are not as simple as shown in Fig. 3.11. Fig. 3.13 shows a measured pressure impulse response of the room used for Fig. 3.11. The extra impulse response components are due to the presence of scattering, diffraction, complex surface impedance, loudspeaker and microphone impulse responses, etc.

The energy behavior, however, shown in Figure 3.14 can often be quite well simulated by calculating the time integral of effective value of the sound pressure in octave or better - ⅓ octave bands.

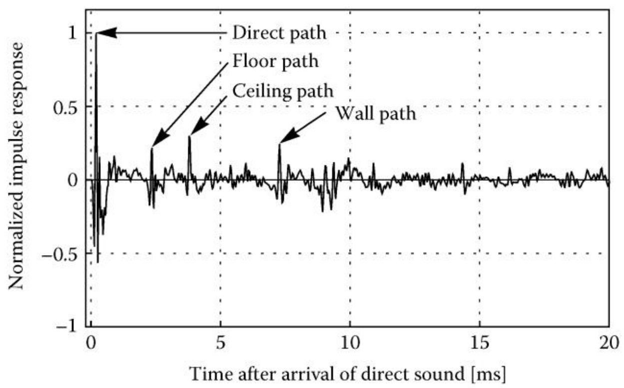


Fig. 3.13. A measured room impulse response corresponding to the GIR shown in Figure 3.11.

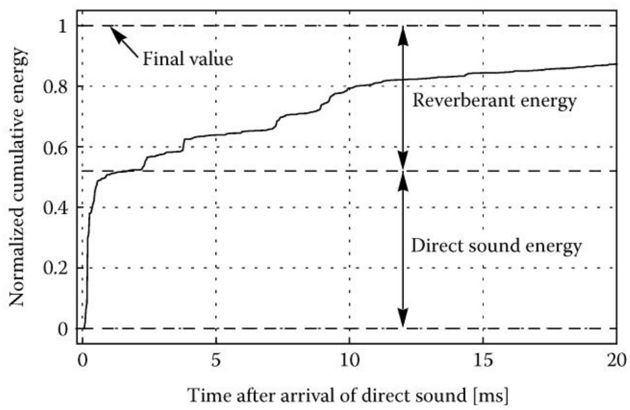


Fig. 3.14. The time integral of the effective value of the room impulse response shown in Fig. 3.13. The curve is normalized to its final value. We note that the ratio between direct D and reverberant sound R is D/R ≈ 1

3.11 Room frequency response

The measured impulse response contains all the information needed to characterize the room as a filter between sound source and receiver as long as there is no noise or nonlinear distortion. Typically, the response is measured using a dodecahedron loudspeaker and a small microphone, thus approximating a point source and point receiver.

The transfer function of a room "filter" is the Fourier transform of the room's impulse response. The frequency response of a filter or transducer is the level of its transfer function magnitude as a function of frequency. The frequency response of this room filter will depend not only on the properties of the room but also on the sound source and receiver. The sound source directivity of a loudspeaker or microphone is virtually impossible to characterize over the full audio range, but for low frequencies, loudspeakers can often be assumed omnidirectional.

The sound field in the frequency range above the Schroeder frequency is often considered *diffuse* but this is not necessarily so. For example, a shoebox-shaped room with rigid floor and ceiling but walls that are very sound absorptive. The sound field in such a room will be dominated by sound bouncing between floor and ceiling . In the high-frequency approximation of geometrical acoustics, the rays representing plane waves will be periodically reflected by these surfaces. The sound field is not diffuse and the frequency response very periodic, but one can still calculate a Schroeder frequency. Hearing can easily recognize the *flutter echo* that is a result of the short period reflected.

4. Absorption, reflection and diffusion

4.1 Absorption mechanisms

This section presents an overview of the most frequently used sound-absorbing materials and structures for the optimum adjustment of the sound field according to the room function, by means of absorbers, reflectors and diffusers.

The energy decrease of the wave is most often caused by a partial conversion of the organized molecular motion into a random motion so that heat is created. This mechanism exists in the boundary layer where the wave touches a solid surface. This is particularly the case for open-pore materials since they consist of a skeleton of tiny hard fibers with the mutually connected pores in which the surface area for the interaction with the wave is large.

The losses are proportional to the particle velocity of the incoming wave. If, for instance, a porous material is placed on the surface of a solid wall, the partially reflected wave interferes with the incoming wave and the major losses occur at locations distant a quarter wavelength from the backing wall.

Although the physics of sound conversion of energy into heat is in most absorbers the same, their mechanical structure differs. Open-pore absorbers that consist of fine fibers are (if possible) placed into regions of high particle velocity.

On the other hand, absorbers based on acoustical or mechanical resonance achieve high losses by constructing mechanism that elevate particle velocity naturally such as plates, membranes, and tubes or are using the Helmholtz acoustical mass–spring system.

In addition to these absorbing systems, sound is absorbed through various functional objects placed in rooms. People absorb sound by virtue of the textiles used for clothing that typically function as an open-pore absorber.

4.2 Fundamental quantities and their measurement

The sound absorption of a flat sound absorbing surface is characterized by its sound-absorption coefficient α defined as the ratio of the sound power absorbed W_{abs} to the sound power incident W_{inc} per unit area equal to

$$\alpha = \frac{W_{abs}}{W_{inc}}$$

The absorbing surface also reflects sound that is characterized by a sound pressure reflection factor R defined as the complex ratio of the reflected and incident sound pressures. The relationship between α and R is

$$\alpha = 1 - \left|\frac{p_{refl}}{p_{inc}}\right|^2 = 1 - |R|^2$$

Occasionally, a sound power reflection coefficient β is used where $\beta = 1 - \alpha$ that should not be confused with sound pressure reflection factor R. Both the absorption and reflection coefficients are the function of incidence of a plane wave on a unit of area and the frequency. If the absorber is large compared to wavelength and has an area S, the total absorbed energy A, usually called absorption, is equal to

$$A = S \cdot \alpha$$

Some sound absorbers cannot be characterized by the absorption per unit area but rather by their total absorption A in m^2. Such absorbers are, for example, people, chairs, the Helmholtz resonators, and other irregular objects. The sound absorption can be also expressed in terms of the impedance Z of the absorbing surface. The impedance can be considered having a real and an imaginary part:

$$z = \frac{1}{\rho_0 c} Z = \frac{1}{\rho_0 c}\frac{p}{u} = \frac{\mathrm{Re}(Z) + j\,\mathrm{Im}(Z)}{\rho_0 c}$$

where
Z is the specific acoustic impedance, z is acoustic impedance normalized to the wave impedance of the air $\rho_0 c$.

The reflection factor R is

$$R = \frac{Z - Z_0}{Z + Z_0}$$

The absorption coefficient α can be shown to be

$$\alpha = \frac{4\,\mathrm{Re}(Z) Z_0}{\left[\mathrm{Re}(Z) + Z_0\right]^2 + \mathrm{Im}(Z)^2}$$

where $Z_0 = \rho_0 c$.

These generalized notations are used for perpendicular sound incidence on the absorber. However, the absorption coefficient is angle dependent. If the plane sound wave incidents under the angle θ with the normal direction to the absorber, the normal component of the particle velocity is multiplied by cos(θ) so that the upper equations are modified and result in

$$R(\theta) = \frac{Z\cos(\theta) - Z_0}{Z\cos(\theta) + Z_0}$$

so that

$$\alpha(\theta) = \frac{4\,\mathrm{Re}\big(Z\cos(\theta)\big) Z_0}{\left[\mathrm{Re}\big(Z\cos(\theta)\big) + Z_0\right]^2 + \left[\mathrm{Im}\big(Z\cos(\theta)\big)\right]^2}$$

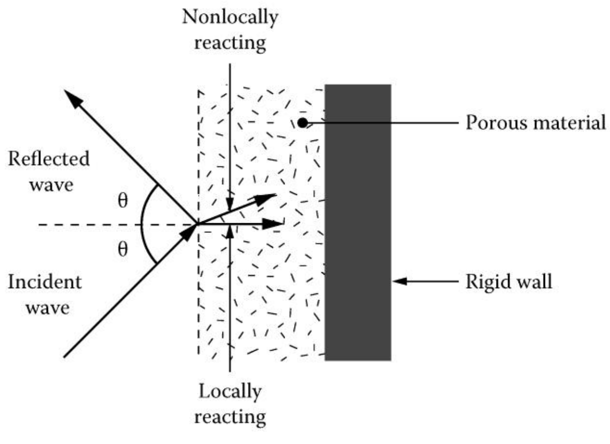


Fig 4.1. Locally and nonlocally propagating wave in a sound absorber

4.3 Measurement of absorption coefficient for normal sound incidence

When developing or comparing the performance of absorbers, the absorption coefficient is usually measured for perpendicular incidence of a plane wave. This can be done for low frequencies and homogeneous materials using a waveguide with a rectangular or circular cross section [19].

To prevent the generation of cross modes, the size *d* of the rectangular waveguide side must be smaller than one-half of the wavelength of the lowest cross mode, and for a circular waveguide, the diameter must be less than 0.63 of the wavelength. This obviously limits the use of these measurements to low frequencies although special microphone assemblies can be used to increase the frequency range of measurement. The material sample is placed at one end in the wave guide and the sound is supplied from a loudspeaker located at the other end of the tube.

As the standard ISO 9614-3 specifies in detail, the complex ratio of the incident and the reflected sound power can be determined by measuring by means of two sound pressure microphones (Fig. 4.2) [20]. The major advantage of measuring the sound intensity is that the absorption coefficient can be measured at short frequency increments using digital signal processing. The older analog technique required the measurement of sound pressure at the points of pressure maxima and minima, which are at different locations for each frequency. The pressure maxima and minima and their location with respect to the absorber sample had to be found for each frequency, and therefore, high-frequency resolution measurements were impractical.

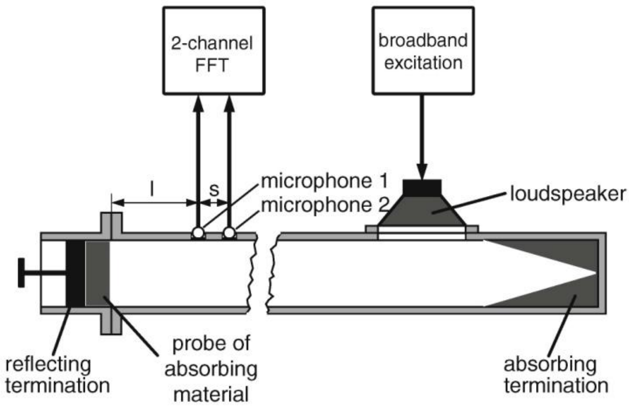


Fig. 4.2 Impedance tube of the ‘two-microphone method’ for measurement of impedances and reflection factors of material samples and other termination impedances

4.4 Measurement for random incidence

When the sound-absorbing material is installed in a room, the sound field that incidents on this material can be very complicated. Major normal sound field incidence is very rare since most sound fields in rooms are diffuse or at least the sound wave is incident at some angle. The sound incidence depends on the room shape, objects in the room, acoustical properties of the room, and location, size, and shape of the absorber, to name a few variables. These are also different for each frequency.

In order to compare the absorption coefficient of different absorbers, the measurements according to the standard ISO 354 [19] are made in a large standardized room of at least 200 m^3 volume that has bare, highly sound-reflective walls. Due to the high wall reflectivity and the density of reflections, the sound field can be considered similar to an idealized diffuse sound field. A diffuse field consists of plane waves that travel in all directions with a uniform probability distribution. Also, all waves must have the same intensity. To achieve a diffuse sound field at both low and high frequencies, a variety of reflectors are usually placed in the room interior. The sound field diffusivity can be investigated by measuring the pressure correlation function ρ, which, for a perfectly diffuse field, is given by

$$\rho(kr) = \frac{\sin(kr)}{kr}$$

where
k is the wave number, r is the distance in some selected direction.
The deviation from perfect field diffusivity is given by the deviation from the *sinc* function.

A 10 m^2 large sample of the absorber is placed in a central position of the floor. The reverberation time is measured for a bare room and for the room with the sample of the sound absorber. The absorption coefficient α_R is calculated using Sabine's equation so that

$$\alpha_R = \frac{0.161V}{S}\left[\frac{1}{T_{60,S}} - \frac{1}{T_{60,0}}\right]$$

where
V, m^3, is the volume of the chamber in metric units, S, m^2, is the area of the measured sample in metric units, $T_{60,S}$, *s*, is the reverberation time with the sample, $T_{60,0}$ is the reverberation time of the empty room.

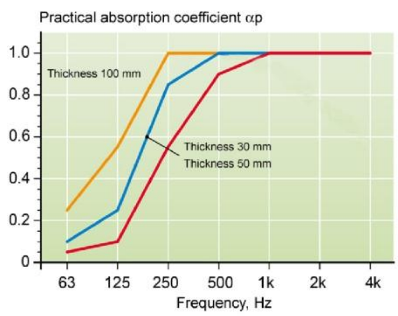
This equation was derived under the assumption made by Sabine that the sound field is perfectly diffuse and the energy density and the sound incidence on the sample are uniform. However, this is not the case in a real room. Particularly, the sound incidence on the absorbing sample is affected by diffraction on the sample boundary, sample position, frequency, and other field distortions. Different room geometries give different results. Therefore, the calculated value of the absorption coefficient for highly absorbing materials can incorrectly exceed unity. Such α_R values are usually corrected to 1 when published.

When the measured values of α_R are used for the calculation of reverberation time using Sabine's equation, we have to expect that the calculation is subject to an error that primarily depends on the position of the absorbing material. This error is particularly large for highly absorbing materials placed in the room corners as well as on the equation used to calculate the reverberation time.

4.5 Porous absorbers [1]

The porous materials most commonly used for commercial sound absorbers consist of very thin fibers that are typically compressed and glued together to form a plate of required thickness (Fig. 4.3). The internal structure of materials that are suitable for sound absorption permits the air and thus the sound to travel through the open space present in the solid skeleton. The air friction in the boundary layers on the surface of the fibers converts the acoustic energy into heat. The materials used are synthetic such as mineral, glass, and polyester wools, ceramic, metal, graphite, polypropylene, Kevlar, or natural such as cotton, compressed hay, flax, and jute.

Porous, mineral wool absorber



Absorption coefficient in octave bands

Fig 4.3. Porous absorber made of mineral wool (left). Typical frequency response of the absorption coefficient of the mineral wool layer 30, 50 and 100 mm

The choice of material depends on the desired absorption properties as a function of frequency, required thickness, mechanical properties, application, and similar criteria. As will be shown later, porous absorbers are practical to use in a variety of layer configurations, inside the cavities of resonant absorbers and as flow resistance elements. Porous absorbers are effective in absorbing medium- and high-frequency sound.

Another useful constant of the material is its characteristic wave impedance. The complex-specific acoustic wave characteristic impedance is $Z_m = R_m - jX_m$. Let us consider a very thick layer of porous material. The sound pressure and the particle velocity inside this layer are

$$p(k_m, x) = Ae^{-jk_m x} + Be^{jk_m x}$$

$$u(k_m, x) = \frac{1}{z_m}\left[Ae^{-jk_m x} - Be^{jk_m x}\right]$$

The constants A and B are the amplitudes of the pressure that travels in the positive and negative directions of the x axis. The sound pressure p_2 and the particle velocity u_2 at the back side of the material distant d from its front are

$$p_2(k_m, d) = Ae^{-jk_m d} + Be^{jk_m d}$$

$$u_2(k_m, d) = \frac{1}{z_m} Ae^{-jk_m d} - Be^{jk_m d}$$

These equations can be solved for A and B so that we can calculate p_1 and u_1 at the face of the material for $x = 0$. The impedance Z_1 is

Combination of a bulk-reacting absorber layer with (a) an air gap such that sound can only travel perpendicular to the hard wall and (b) an air gap such that sound can travel parallel to the hard wall.

If the material is backed by a rigid wall ($Z_2 = \infty$), the input impedance Z_1 is

$$Z_1 = Z_m \cot(k_m d)$$

This equation permits the calculation of the characteristic acoustic impedance of the material Z_m from measurement of the impedance Z_1 after k_m has been calculated from the empirical Equations

$$k_m = \frac{\omega}{c}\left[1 + 0.109\left(\frac{\rho_0 f}{R_1}\right)^{-0.618} - j0.16\left(\frac{\rho_0 f}{R_1}\right)^{-0.618}\right]$$

$$Z_m = \rho_0 c\left[1 + 0.070\left(\frac{\rho_0 f}{R_1}\right) - 0.632 - j0.107\left(\frac{\rho_0 f}{R_1}\right)^{-0.632}\right]$$

4.6 Porous materials in front of a rigid wall and effects of air gap

The effects of rigid wall backing mentioned in the previous section will be further analyzed because there are several alternatives. The materials can be placed either directly on a rigid wall or there might be an air gap between the absorber and the rigid wall. The latter arrangement is very frequent for ceilings where a porous tile is suspended from a supporting structure. This arrangement provides an opportunity to place engineering installations (cables, air ducts) between the actual and visual ceiling.

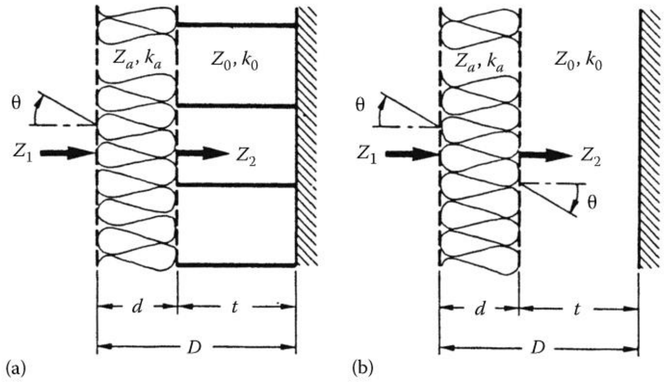


Fig 4.4. Combination of a bulk-reacting absorber layer with (a) an air gap such that sound can only travel perpendicular to the hard wall and (b) an air gap such that sound can travel parallel to the hard wall

We will now analyze the sound absorption and reflection from these installations. Let us first consider a plane sound wave that is incident normally on a wall that has a specific acoustic impedance Z_1 so that the reflection factor R and the absorption coefficients α are

$$R = \frac{Z_1 - Z_0}{Z_1 + Z_0}$$

$$\alpha = \frac{4\,\mathrm{Re}(Z_1) Z_0}{\left[\mathrm{Re}(Z_1) + Z_0\right]^2 + \mathrm{Im}(Z_1)^2}$$

4.7 Sound absorption by resonators

The sound field in enclosures consists of modes. Their density depends on the square of the frequency, and as a result, the modes are distinctly separated at the lowest frequencies. The transmission functions in the room have sharp maxima and deep minima that distort sound quality in listening. If the room is excited by a single frequency close to a mode, this frequency changes to the modal frequency when the source stops operating and only the reverberant sound still exists in the room. Sufficient wall absorption at low frequencies can help improve the modal response. Therefore, the use of low-frequency absorbers is very important.

The previous section dealing with porous absorbers revealed that they are not suitable for absorbing low-frequency sound. This is due to the absorption mechanism that is proportional to particle velocity. The maximum particle velocity of the combined incoming and reflecting wave in front of the wall is ¼ λ away from the wall. This makes it impractical since a large material thickness would be needed for absorbing low frequencies.

Low-frequency sound can be absorbed by several types of resonant systems that are excited by acoustic waves. With large amplitude at resonance, the system losses convert much acoustic energy into heat. The principal resonant systems are the Helmholtz resonators, vibrating plates and membranes, and electronically controlled absorbers. They can all operate and absorb sound at the lowest frequencies of the hearing range and have a small and practical volume.

4.8 Basic principles and properties of individual Helmholtz resonators

Figure 4.5 shows schematically a Helmholtz resonator. It is a flask that consists of a neck and closed volume. The air in the neck has a mass m:

$$m = \rho_0 l S$$

where
l is the length of the neck, S is its cross section, ρ is the air density.

The compliance C of the enclosed volume $V = Sdx$ is

$$C = \frac{dx}{dF} = \frac{V}{\rho_0 c^2 S^2}$$

where *dF* is the compression force. The resonance frequency ω_r is

$$\omega_r = \frac{1}{\sqrt{mC}} = c\sqrt{\frac{G}{V}}$$

where $G = S/l$ is called the conductivity with $l = l_0 + 0.8D$ that is corrected for the actual length l_0 of the mass in the neck.

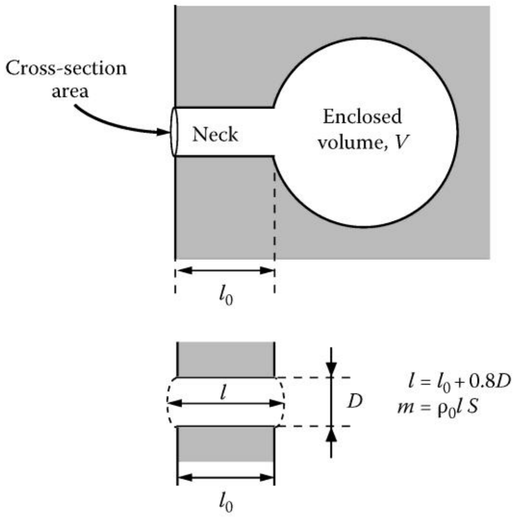


Fig. 4.5. Schematic representation of a Helmholtz resonator.

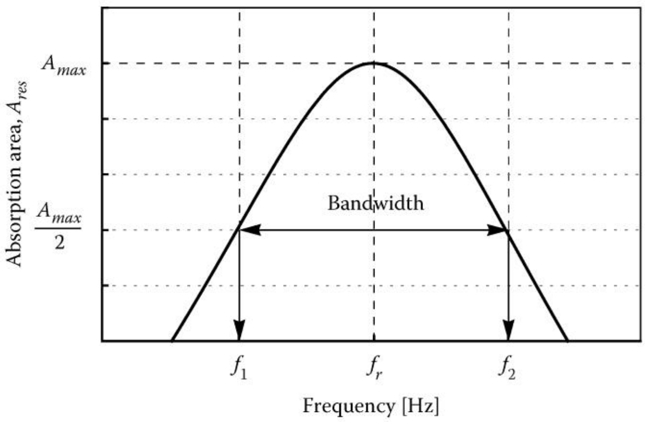


Fig 4.6 Absorption bandwidth of a Helmholtz resonator

Fig. 4.6 shows a resonance curve and the two frequencies f1 and f2 at one-half magnitude of Amax. The greater their difference, the higher is the total absorption. We define the absorption bandwidth in terms of the number of octaves O

$$O = \frac{f_1 - f_2}{f_r} = g\mu$$

$$f_r = \sqrt{f_1 f_2}$$

4.9 Perforated panels

Another type of sound-absorbing construction based on the principle of Helmholtz resonator is the resonator panel. In a resonator panel, the necks are formed by a perforated plate placed at some distance from a rigid wall, either a room wall or some sturdy plate in the case of a cassette resonator system. The damping required to *tune* the system to achieve a needed bandwidth is usually achieved by placing absorbing materials into the resonator cavity or by a flow resistance fabric sheet placed on the perforated plate.

There is also damping in each resonator neck along its boundary. Because of the size of the holes, the energy losses due to the viscous layers along the neck walls are typically insufficiently small. Because the perforated panels that have the damping material inside the cavity are still most often used, their properties will be discussed first.

Figure 4.7 shows the cross section of a resonator panel with an air space and hard wall backing. The view of the perforated panel absorber and its frequency response of absorption coefficient is shown on Fig. 4.8

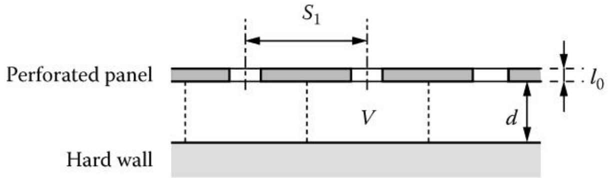
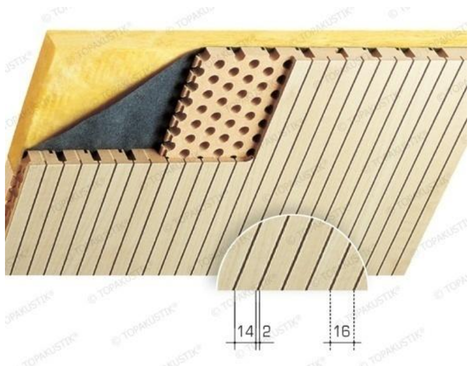
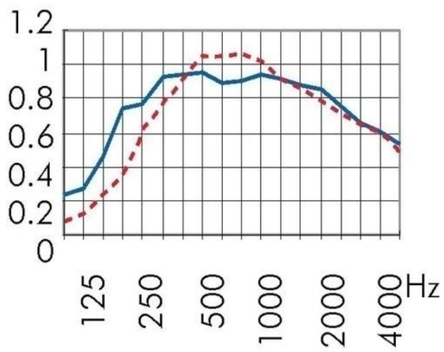


Fig.4.7. Construction parameters of perforated panel absorber



Perforated panel absorber (Topakustik 14/2 M)



Absorption coefficient

Fig 4.8. View of the perforated panel absorber (left) and absorption coefficient in 1.3 octave bands (right)

The optimal performance of both the single resonators and absorbers with perforated plates depends on their acoustical flow resistance. The conversion of acoustic energy into heat occurs on the neck of the resonator or in the hole of the perforated plate in a thin surface layer around its perimeter, and because this damping is not sufficient, usually porous layers are placed in the resonator cavity to adjust the absorption to the required frequency response.

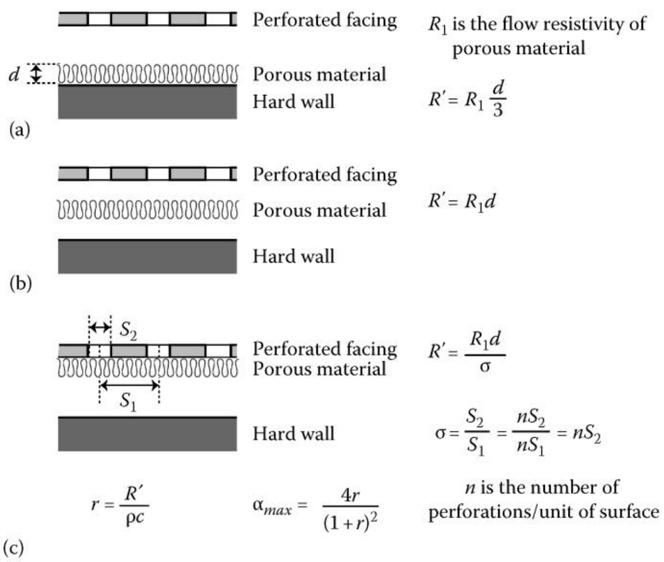
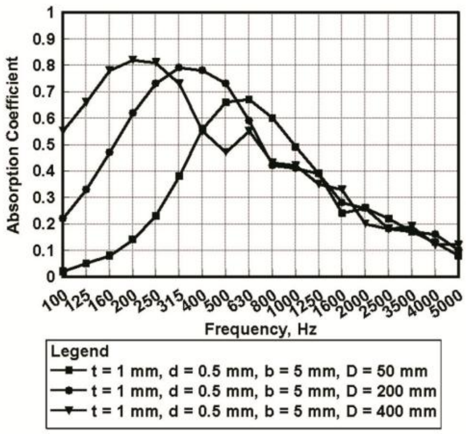


Fig.4.9. Effect of flow-resistive material placement on damping. (a) Porous material next to hard wall, (b) porous material centered in air space, (c) porous material next to perforated sheet.

4.10 Microperforated sound absorbers

With the availability of laser technology, it is feasible to manufacture thin plates and foils from materials such as glass, plastics, metals, and similar materials with holes much smaller than 1 mm. The small holes have dimensions similar to the boundary layer and provide sufficient damping so that the sheet or foil can be used as absorber without additional damping (see Fig 4.10 and 4.11).

Such microperforated absorbers can provide damping over a frequency range of several octaves. They are particularly advantageous for use in small rooms since they can absorb sound at low audio frequencies and provide damping of low frequency modes. Microperforation allows the design of translucent sound absorbers such as perforated glass plates and foils.



Microperforated polycarbonate panel absorber. Thickness 1 mm, hole diameter 0.65 mm

Absorption coefficients of MPP absorber. Three various distances from a hard base, 5, 20, 40 cm

Fig. 4.10. View od the microperforated panel (MPP) absorber (left) and 1/3 octave band absorption coefficient (right).

Also, since microperforation does not require conventional porous absorbers, it can provide sound absorption in special environments where porous absorbers would not be usable. They may provide sound absorption needed in complicated rooms with trusses, historical structures, etc.

Microperforated sheets and membranes made from rigid and particularly flexible (plastics) materials are very efficient absorbers also due to their vibrational modes. The bending modes of these materials depend on their mechanical properties as well as on their dimensions, mounting, and distance to a wall.

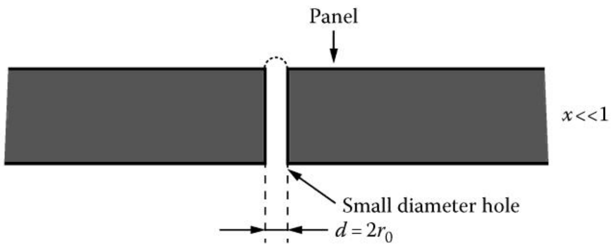


Fig. 4.11. Small perforation of a panel to achieve substantial flow resistance for conversion of acoustical wave into heat.

The perforations contribute to the vibrational damping and the vibrational modes affect the frequency characteristics and bandwidth of absorption.

4.11 Plate and membrane absorbers

As analyzed earlier, the frequencies of the low-frequency modes in small rooms are well separated unless the modes are sufficiently damped. Depending on the room wall construction, some low-frequency room damping may already exist, for example, cracks in walls can contribute to sound absorption. Walls that are heavy, such as brick or concrete walls, have high impedance and thus usually do not provide sufficient damping. However, the relatively thin plaster walls, mounted on studs and backed by porous damping (to provide sufficient transmission loss), may, because of their relatively low impedance, also offer damping of low-frequency modes. It is difficult to predetermine this damping by calculation and its determination usually requires elaborate measurements.

Plate and membrane absorbers often provide sound-absorptive resonant systems in addition to those offered by the Helmholtz resonators and resonator panels. Figure 4.12 shows the construction schematically. A plate or membrane is mounted on a suitable frame on a sufficiently rigid wall. The plate offers mass *M*, while the sealed volume formed at its back offers an air spring compliance *C*. This mass–spring resonator usually needs some additional damping to that provided by the losses in the membrane and the losses due to air pumping where the membrane is attached to the battens.

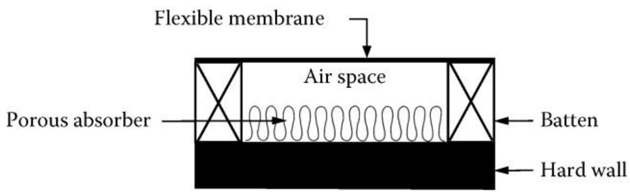


Fig. 4.12 A membrane absorber.

Such damping can be provided by adding the porous material inside the enclosed air space. The resonance frequency is

$$f_r = \frac{c}{2\pi}\sqrt{\frac{\rho_0}{md}} \simeq \frac{60}{\sqrt{md}}$$

5. Diffusion [1]

5.1 The need of diffusion

The reflection of sound from the absorption materials or room walls that we have been dealing with so far has been assumed specular. This means that if the size of the material is sufficiently large, the angle of reflection of a plane wave is the same as the angel of incidence. The sound at a listening location in a room consists of both direct sound that travels from the source and sound reflected from the walls and objects in the room. The purpose of the absorbers is to adjust the intensity and the spectrum of the sound reflection so that optimal listening conditions are achieved. However, the specular reflections, particularly in small rooms, cannot always completely satisfy this requirement. In small rooms, the time difference between the arrival of the direct and the reflected sound at the listening point is very small, usually only 10–20 ms. Their interference results in sound coloration, particularly if the reflected sound is strong. To avoid coloration early reflected sound must be 20 dB weaker than the direct sound.

Such attenuation by some absorbing material may result in excessive reduction of the overall acoustic energy in the room. This situation can be corrected by using structures that diffuse the incident sound rather than absorb it. The incident sound is reflected into many different directions. This will maintain the overall energy in the room but reduces the energy that is directly reflected to the listening point.

Sound diffusion can be achieved by a variety of objects in the room, for instance, by furniture. However, optimum placement and diffusing properties are difficult to achieve by a single reflector. Special structures have been developed to achieve optimum sound diffusion.

Most diffusers for use in room acoustics have been developed for a frequency range of about 0.5–4 kHz to be of usable size and cost in addition to working in the frequency range where hearing is the most sensitive. Substantial room correction is, however, often needed at low frequencies in small rooms due to the well-separated strong resonant modes that are often insufficiently causing the frequency response to be very nonuniform. These adverse effects are usually treated by the use of resonant absorbers. In addition to causing sound scattering or diffusion, the diffusers also have effects on sound absorption and modal density.

5.2 Basic principles of sound diffusers

The inventor of modern, numerically determined diffusers was Manfred Schroeder who in 1975 published two articles on diffuse sound reflection. He described and analyzed a line array of narrow, approximately one-quarter-wavelength deep resonators that would scatter an incident plane sound wave over an angle of 180° over a wide frequency range. Other papers on the topic were later published by Berkhout.

The principle of Schroeder's basic idea is shown schematically in Fig. 5.1 a. The two wells have the same width but different depth and, therefore, the sound that is reflected from their bottoms have different phases at their tops. The two waves interfere and the resulting wave propagates under a different angle than the incident wave arrived on the two wells. A similar idea, developed much later and shown in Figure 5.1 b, shows two narrow strips of acoustic materials that have different impedances and therefore affect the phases of the reflected waves. For certain frequencies, maxima or minima of the interfering reflected sound occur as described later in this chapter.

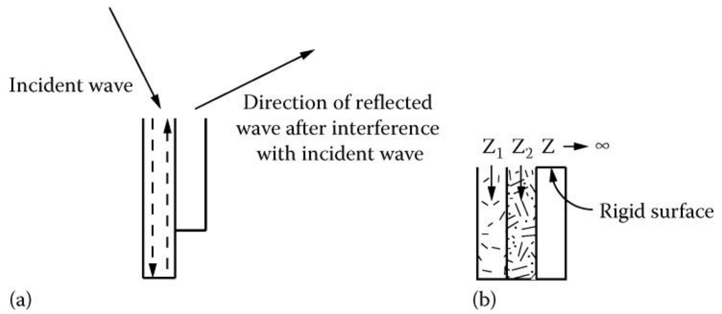


Fig.5.1. Principles of sound diffusion. (a) Two interfering tubes with phase shift. (b) Phase shift by different input impedance

An example of a Schroeder-type diffuser is shown in Fig. 5.2 and consists of *N* wells of different depths, separated by thin, hard, rigid walls.

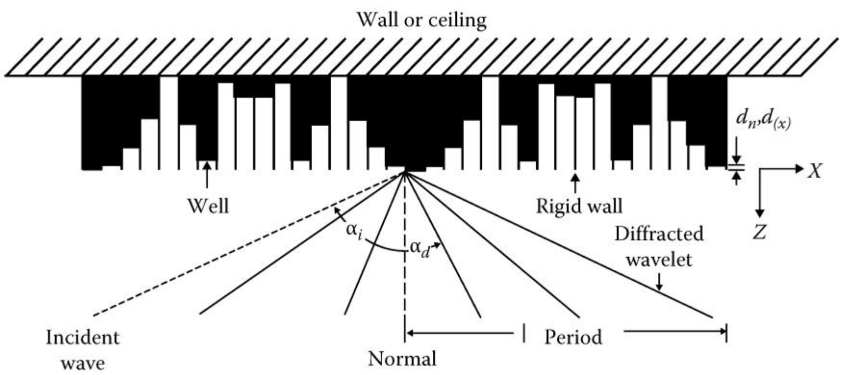


Figure 5.2 Scaled cross section of a 1D QRD described by Schroeder [5], consisting of two periods, with 17 wells per period (N = 17). The width W of each of the wells, which are separated by thin and rigid walls, is equal to 0.137λ0, where λ0 is the design wavelength, dn is the depth of the nth well, and d(x) is the depth at the lateral position x. A wave front (dashed line) making an angle of incidence αi of −65° with the surface normal and five diffracted wavelets (solid lines) occurring at αd = −54.2°, −22.4°, 2.7°, 28.5°, and 65° is shown. (From D'Antonio, P. and Konnert, J.H., J. Audio Eng. Soc., 32(4), 228, 1984.)

The width *W* of each well is smaller than $W = \lambda_m/2$ where λ_m is the wavelength of the maximum frequency for which the diffuser is designed. The wells must be sufficiently narrow to prevent the existence of cross modes at frequencies higher than f_m. The figure shows two repetitions of periods of *N* wells each. We will now calculate the required depths of the wells. Schroeder used a mathematical sequence called quadratic residue (QR). Other sequences such as pseudorandom or primitive root can also be used to determine the well depths.

The diffuser performance depends on the sequence used. Characteristic for useful sequences is that their autocorrelation has a single maximum and is close to zero elsewhere. Following will be described the properties of the diffusers called quadratic-residue diffusers (QRDs). As Schroeder has shown, the Fourier transform of the diffused sound is constant, which means that the diffused energy is uniform in direction. We can calculate the sequence of the well depths by evaluating n^2 mod N, where N is an odd prime number that is selected according to the requirements on the diffuser performance and n is an integer from zero to infinity.

A QR sequence is generated by

$$s_n = n^2 \bmod N$$

where

s_n is the sequence number of the nth well, mod N is the least nonnegative remainder, n describes the sequence of the wells.

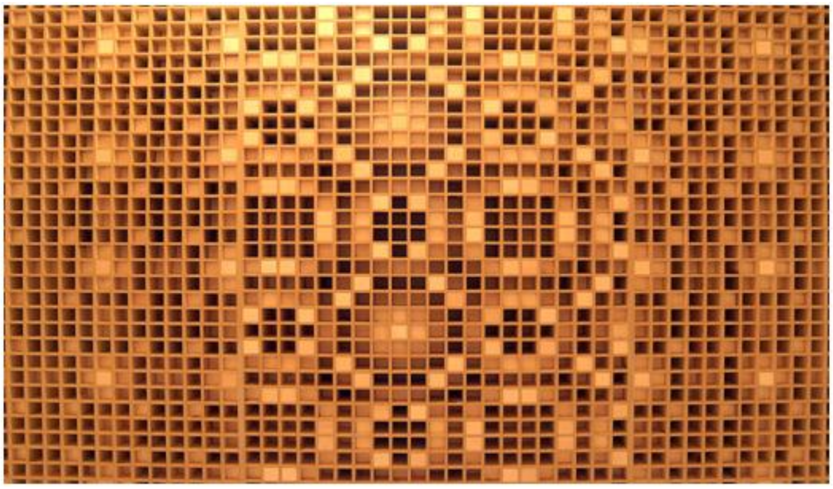


Fig. 5.3. 2D Schroeder sound diffuser made of plywood. Diffuser mounted on the ceiling of the 003 Sound Control Room at the Sound Engineering Faculty, F. Chopin University of Music

A view of the 2-dimensional Schroeder sound diffuser is shown on the Fig. 5.3. When designing the diffuser, we have to select the longest wavelength λ_0 at which sound is to be diffused. This wavelength will determine the depth of the deepest resonator. The total width of one period of the diffuser is *NW*.

As will be shown further, the selection of N, the number of the diffuser periods used, and their frequency range influence the diffuser directivity pattern and other performance qualities. Two one-period examples of sequences are, for example, for a diffuser with $N = 7$: $s_n = \{0,1,4,2,2,4,1\}$ and

for $N = 17$: $sn = \{0,1,4,9,16,25,36,49,11,28,47,15,38,10,37,13,44,24\}$.

The depth d_n of the individual wells is given by

$$d_n = n^2_{\bmod N} \frac{\lambda_0}{2N}$$

where

n is the number of the well, λ_0 is the wavelength of the lowest frequency of the diffuser, N is the number of the wells in one diffuser period.

The maximum depth of some wells can be approximately one-quarter-wavelength of the lowest design frequency. That is about 0.75 m for 100 Hz but about 15 cm for 500 Hz.

Therefore, the QRDs are practically not usable for low frequencies unless using active elements in the wells, as described later.

6. Subjective room acoustics

6.1 Subjective and objective evaluation of sound in rooms

The time structure of impulse responses of rooms starts with the short pulse of the direct sound, and includes the delay gap with negligible reflections, the portion of the early reflections and a tail of the late reflections (see Fig 6.1).

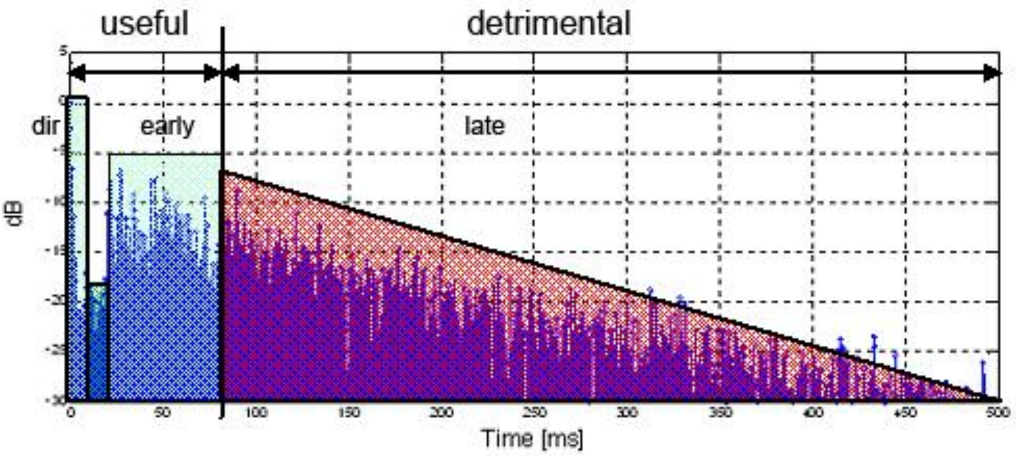


Fig. 6.1. The time structure of the squared impulse response of rooms

With the direct sound which comes first, the listener subjectively determines the direction from which the sound comes (the first wave front, precedence or Haas effect). The direct sound, initial time delay gap and the sound of several reflected components comprise the useful energy which carry a clear music and intelligible speech. The next, late portion of the delayed reflections, after 50 to 100 ms, comprise a diffuse field. The late sound provides the sensation of reverberance, spaciousness, etc., but may blur the sound transfer. There are reports that the listeners differ in their preferences to the quality of the sense of reverberation. At least three groups have been identified: those that like reverberance, those that like acoustic intimacy and those that above all prize high clarity [14].

For the objective comparisons of the interior acoustics several measures are recommended in the ISO standard 3382 [17]. Some of them make use of the division of the early and late energy of the acoustic impulse response. The temporal division between early and late for the speech communication is 50 ms and for the concert hall spaces is generally taken as 80 ms, with the early-to-late sound index known as C50 and C80 respectively.

Many attempts has been made to correlate subjective judgments of sound quality with objective measurements of acoustical attributes to establish a numerical evaluation of concert halls. Interviews with musicians, conductors, critics, and concertgoers were used to parse halls perceived quality. Several quantifiable attributes that are fairly well correlated with the subjective judgments were identified and are recommended for evaluation of the acoustical quality of concert halls.

Following is the list of the objective measures recommended for the assessment of concert halls [6], [9]:

(i) The Interaural Cross-Correlation family, IACF, IACCA ,$IACC_E$, and $IACC_L$. The Binaural Quality Index, BQI = 1-$IACC_{E3}$ is suggested as a measure for the comparative assessments of halls, (the three octave bands with center frequencies at 500, 1,000, and 2000 Hz, with integration of the early sound between 0-80 msec),

(ii) The Clarity Factor, C_{80}, dB, - the ratio of the early energy, 0-80 ms, to the late, reverberant energy 80–3000 ms,

(iii) The Lateral Energy Fraction, LF, - the ratio of the output of a figure-8 microphone with its null direction aimed at the source, to the output of a non-directional microphone,
(iv) The Strength Factor, G, dB, - a measure of the SPL at a point in a hall, with an omnidirectional source on stage, minus the SPL that would be measured at a distance of 10 m from the same sound source located in an anechoic chamber,
(v) The Bass Ratio, BR, - the ratio of the low to mid-frequency reverberation times
(vi) The Support Factor, ST1, - the difference in decibels between two measurements of sound-pressure level, made in 1 m from the source on a stage or in a pit where the orchestra members play,
(vii) The Late Lateral Energy, LG, - the ratio of the delayed output of a figure-8 microphone to the total output of a non-directional microphone, where the latter is measured at a distance of 10 m from the acoustical center of an omnidirectional source in an anechoic chamber,
(viii) The Distinctness or the Definition (Deutlichkeit) Ratio, D, %, - the ratio of the sound in the first 50 ms after arrival of the direct sound to the total sound.

The room impulse response is the basis of all measurements. The demanded acoustical quantities are obtained from digital processing of that signal.

6.2 Metrics for perceived sound and relation to physical measures [10]

The most dominant effect of hearing in rooms is the sensation of reverberation. Average value of reverberation time and its frequency dependence is the most important quantity for acoustic characterization of rooms.

There are other specific quantities important for describing the overall subjective effect in rooms. Unlike reverberation, they depend on the specific listener’s position in the room.

The reverberation in a room is the result of sound reflections at the room boundaries. The response of the room to excitation with an ideal impulse, serves well as a basis for interpretation of room acoustics. A typical structure of a room impulse response (“energy time curve”) is illustrated in Fig. 6.2. The pulse density grows with t^2 along the time axis, while the pulse energy decreases with t^{-2} and with absorption. In short time averages, the quadratic time functions cancel and the net energy loss is given by the exponential decay due to absorption.

As mentioned before, the first impulse - the direct sound - determines the perceived direction of sound incidence (precedence effect). Reflections are delayed due to the longer path of sound propagation. Even in the case of secondary sound with a higher level than the direct sound, up to 10 dB, which might happen with the sound of secondary loudspeaker, the localisation will still be determined by the first arriving component.

The reflections within an interval of early reflections contribute in a specific way to the direct sound impression. They enhance the loudness, support the intelligibility of speech, the clarity of music and the impression of the auditory source width. All reflections delayed by more than 50–80 ms build up the reverberation in more specific meaning, they create impressions of reverberance and listener envelopment.

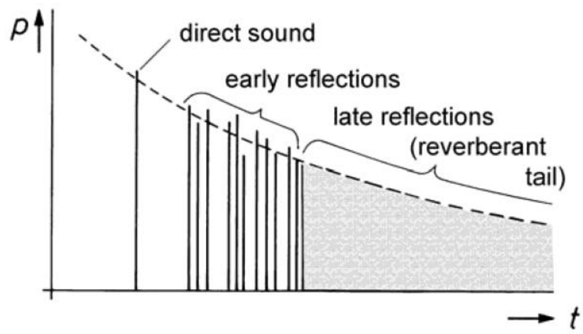


Fig. 6.2. Energy vs. time of the room impulse response

6.2.1 Reverberance

The sensation of reverberance must be seen in relation to the signal presented. Music and speech do have transient components. However, the mean syllable duration and also the mean tempo of music does not permit following the reverberation over a level drop of 60 dB. Instead, the early part of the reverberation has more significance for the perceived reverberance than the later part. For this reason, several algorithms for the evaluation of reverberation times were introduced which evaluate early parts of the energy impulse response.

Reverberation time was defined in relation to switch-off of a steady-state signal. An impulse response, in contrast, is related to a short pulse excitation. The way to transfer the impulse excitation to steady-state signals with switch-off is the integration of the impulse response.

An excitation of the room by a steady-state signal can be understood as a permanent excitation with repeated pulses. Thus, the response to steady-state excitation is the sum of all components of the impulse decay. In the stationary case, the direct sound and each of the early reflections are present permanently. The same holds for all reflections. Retardation and coincidence of all components in the end yield an integration of the impulse response. The value of the integral is the steady-state energy of the continuous excitation. The first "missing" sound after switch-off is the direct sound, while all others are still present in the total sound field, since they were radiated before switch-off time. But then, the energy from the temporal series of reflections will vanish one by one. This process is modelled by the so-called "integrated impulse response" [15]. It consists of two steps:

a) Integration of the energetic impulse response, $p^2(t)$

$$C = h^2(t<0) = N_0 \int_0^\infty p^2(\tau)\mathrm{d}\tau$$

b) Switch-off: Subtraction

$$h^2(t) = C - N_0 \int_0^t p^2(\tau)\mathrm{d}\tau = N_0 \left[\int_0^\infty p^2(\tau)\mathrm{d}\tau - \int_0^t p^2(\tau)\mathrm{d}\tau \right]$$
$$= N_0 \int_t^\infty p^2(\tau)\mathrm{d}\tau$$

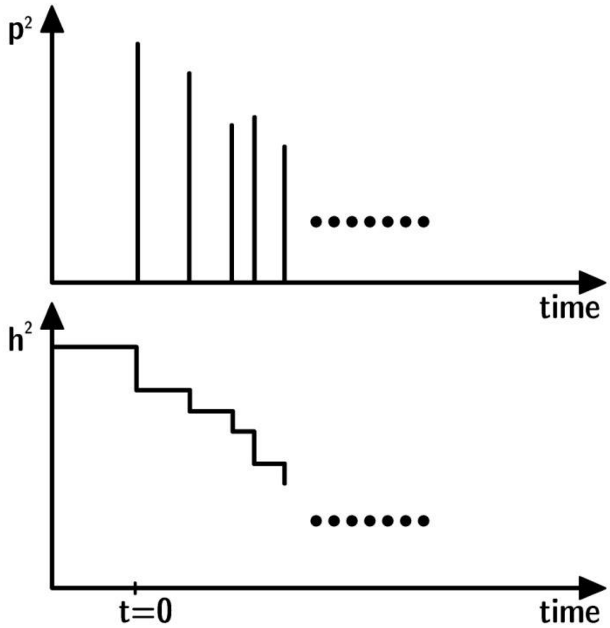


Fig. 6.3. Top: Energetic impulse response. Bottom: Integrated impulse response. Note the temporal coincidence of impulses and subtraction steps

6.2.2 Strength

As explained, the total energy for the steady state is contained in the integral of the impulse response. The main energy, however, is concentrated in the early part. For example, the energy contained in the reverberation tail later than half of the reverberation time contributes only 22% of the total energy. The total energy density expressed in decibels is hence too small by just 1 dB (which corresponds to the just audible difference), if the second half of the reverberation tail is neglected.

The total energy can be expressed independently of the sound power of the source. By choosing a reference distance in a free sound field, the parameter "strength," G, is introduced. The reference distance is 10 m. If we assume an average distance in a room of 10 m as well, G denotes the gain of the room, compared with the free field propagation of direct sound.

$$G = 10\log\frac{\int_{t_0}^{\infty} p^2(t)\mathrm{d}t}{\int_{0}^{\infty} p_{10}^2(t)\mathrm{d}t}.$$

$p(t)$ denotes the sound pressure of the room impulse response and $p_{10}(t)$ the reference sound pressure at 10 m distance with the same source in a free field. The integration limit, t_0, is defined as the arrival time of the direct sound including the direct sound.

6.2.3 Speech intelligibility and transparence

Because the early reflections support speech intelligibility, their energy sum can be used to characterize the syllable intelligibility. For simplicity, the lower integration limit (arrival time of the direct sound) is set to zero. This is the parameter "definition," D, for speech:

$$D = \frac{\int_{0}^{50\mathrm{ms}} p^2(t)\mathrm{d}t}{\int_{0}^{\infty} p^2(t)\mathrm{d}t}.$$

A similar expression is intended with the introduction of "clarity," C. It is defined as C_{80} for music and C_{50} for speech. With C_{80}, we can correlate the impression of transparence in fast pieces of music, the ability to recognize musical details behind the "curtain" of reverberation.

$$C_{80} = 10\log\frac{\int_{0}^{80\mathrm{ms}} p^2(t)\mathrm{d}t}{\int_{80\mathrm{ms}}^{\infty} p^2(t)\mathrm{d}t},$$

$$C_{50} = 10\log\frac{\int_{0}^{50\mathrm{ms}} p^2(t)\mathrm{d}t}{\int_{50\mathrm{ms}}^{\infty} p^2(t)\mathrm{d}t}.$$

$$C_{50} = 10\log\left(\frac{D}{1-D}\right).$$

The balance between the early and late parts of the impulse response can also be expressed by using the first moment of the impulse response. This way we define the "centre time," T_S:

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

Definition, clarity and centre time are interrelated and correspond to clarity of speech and music. Which parameter is best cannot be decided. Extensive listening tests and questionnaires used in the laboratory and in music halls have shown that all quantities are useful to describe this specific auditory impression of transparence related to the linguistically or musically basic elements such as syllables or notes. Another well-known approach to describe speech intelligibility is the evaluation of the modulation of the speech signal [16]. The characteristic modulation in speech signals can be affected by reverberation and by background noise, independently. The so-called “speech transmission index”, STI, expresses the degree of changes of modulation depth caused by reverberation and by noise. The basic function in this evaluation is the modulation transfer function, MTF. The MTF is a ratio of the spectra of the envelope time signal of the original and the disturbed signal. Values smaller than 1 indicate a reduction of speech intelligibility. The spectral differences can, furthermore, be discussed with regard to broadband or high-frequency components. While background noise affects the STI in all frequency bands, reverberation affects mainly the high frequencies.

6.2.4 Spatial impression

The subjective impression of spaciousness was identified as linked to lateral reflections [12]. Purely energetic integration is, thus, not sufficient to characterize this effect. Furthermore, early and late lateral reflections create two kinds of spatial impression. The early part, up to 80 ms, contributes to the auditory source width, ASW. As described above, source localization in complex sound fields is evaluated by the human hearing system such that the first arriving sound event determines the perceived direction of sound incidence, precedence effect. Early lateral reflections add some uncertainty to the localization. The source is not localized at an exact position, but related to an extended source with a characteristic width, ASW. The objective parameter which correlates well with this impression is the early lateral fraction, LF:

$$\mathrm{LF} = \frac{\int_{5\mathrm{ms}}^{80\mathrm{ms}} p_\infty^2(t)\mathrm{d}t}{\int_0^{80\mathrm{ms}} p^2(t)\mathrm{d}t} = \frac{\int_{5\mathrm{ms}}^{80\mathrm{ms}} (p\cos\vartheta)^2(t)\mathrm{d}t}{\int_0^{80\mathrm{ms}} p^2(t)\mathrm{d}t},$$

where p_∞ denotes the sound pressure impulse response obtained using a gradient (figure-of-eight) microphone oriented to the horizontal axis through the ears of the listener.

Another dimension of spatial impression is the listener envelopment, LEV. It is caused by late lateral reflections. The impression of envelopment is described by the objective parameter of late lateral strength, LG:

$$\mathrm{LG} = 10\log\frac{\int_{80\mathrm{ms}}^{\infty} p_\infty^2(t)\mathrm{d}t}{\int_0^{\infty} p_{10}^2(t)\mathrm{d}t} = \frac{\int_{80\mathrm{ms}}^{\infty} (p\cos\vartheta)^2(t)\mathrm{d}t}{\int_0^{\infty} p_{10}^2(t)\mathrm{d}t}.$$

When sound arrives at the listener’s ears from lateral directions, it will lead to a loss in correlation in the binaural pattern. Particular interaural differences occur. Hence the interaural correlation can be used as well to describe spatial measures. The interaural cross-correlation function between the sound pressure signals at the right and the left ears of a listener or a dummy head, p_r and p_l, respectively, is defined as follows:

$$IACF_{t_1,t_2}(\tau) = \frac{\int_{t_1}^{t_2} p_l(t)\cdot p_r(t+\tau)\mathrm{d}t}{\sqrt{\int_{t_1}^{t_2} p_l^{\,2}(t)\mathrm{d}t \int_{t_1}^{t_2} p_r^{\,2}(t)\mathrm{d}t}}.$$

$$\mathrm{IACC}_{t_1,t_2} = \max\left[IACF_{t_1,t_2}(\tau)\right].$$

Its maximum between –1 ms < τ < 1 ms is denoted "IACC," interaural cross-correlation coefficient.

With the choice of the integration interval in the above equation, early and late lateral reflections can be treated separately, thus, leading to an additional independent objective parameter for ASW and LEV.

6.2.5 Just noticeable differences in hearing of acoustic parameters

Parameters of hearing in rooms have been studied in laboratory tests and in real rooms. The tests have shown the extent to which human listeners can distinguish different situations. As in other psychoacoustic areas, the just noticeable differences, jnd, give important information about the precision necessary and simplification allowable in assessment of acoustic data from measurements of real objects and from computer models.

Table 6.1. Just noticeable differences of subjective room acoustical impressions [10], [17] (*) - parameters not included in the original ISO publication)

Subjective listening aspect	Acoustic quantity	Main frequency range (Hz)	jnd
Level	G (dB)	500 to 1000	1 dB
Reverberance	EDT (s)	500 to 1000	5%
Clarity	C80 (dB)	500 to 1000	1 dB
Definition	D	500 to 1000	0.05
Center Time	TS (s)	500 to 1000	10 ms
ASW	LF	125 to 1000	0.05
LEV	LG (d)	125 to 1000	1
ASW*)	Early IACC	125 to 4000	0.075
LEV*)	Late IACC	125 to 4000	0.075

6.3 Subjective preferences for room acoustical quantities

In assessing the concert hall acoustics a number of acoustical parameters are quantified and compared to the parameter values recognized as the most preferred, the best or optimal for good acoustics.

Beranek [9] selected seven parameters for symphonic concert halls, chamber music halls and opera houses. The preferred values of acoustical parameters are shown in the table 6.2. (Please see the discussion of the attributes used by Beranek at the end of this section.)

Table 6.2. Preferred values of acoustical parameters in concert halls, opera houses, and chamber music halls [9]

Conditions	**RT_{oc}, s**	**EDT_{unoc}, s**	**BQI**	**G_{mid}, dB**	**G_{125}, dB**	**ITDG, ms**	**SDI**	**C80.3, dB**	**ST1 dB**
	Ave 0.5 – 1 kHz	Ave 0.5 – 1 kHz	-	Ave 0.5 – 1 kHz	125 Hz	mid floor	visual	Ave 0.5 – 1 kHz	stage
Symphonic Repertoire, Over 1.400 Seats	1.8 - 2.1	2.2 - 2.6	0.65 - 0.71	1.5 - 5.5	3.0 - 6.0	<25	>0.8	-3.0 - 0	-14
Chamber Music, Under 700 Seats	1.6 - 1.8	1.9 - 2.3	0.70 - 0.76	9.0 - 13	9.0 - 13	<20	>0.8	-2.0 - 2.0	>-12
Opera, Over 1.200 Seats	1.4 - 1.6	1.5 - 1.9	0.60 - 0.71	-1.0 - 2.0	-1.05 - 2.3	<23	>0.5	1.0 - 3.0 –	-

In the purpose of quality ratings Beranek assigned a weighting to the importance of each acoustic attribute in order to find a single number rating of the concert halls acoustical quality. The list of weightings are given in table 6.3.

Table 6.3. Weightings of Acoustical Attributes, %

Attribute	Concert Halls	Opera Houses
Intimacy	40	40
Liveness	15	15
Warmth	15	15
Loudness of Direct Sound	10	10
Loudness of Reverberant Sound	6	6
Balance and Blend	6	10
Diffusion	4	0
Ensemble	4	4
Total	100	100

In another study, Ando [23] proposed the guidelines for evaluation of the acoustical quality of concert halls, with another rating system based on the method of subjective listener preferences. The preferred Ando values and weightings of acoustical attributes are summarized in table 6.4.

Table 6.4. Weightings of acoustical attributes, % [23]. $IACC_{E3}$ - interaural cross correlation coefficient, EDT – early decay time, s, SDI*) - surface diffusivity index, G - loudness of reverberant sound, dB, (difference of the SPL at a point in a hall and the direct field level at 10 m for an omnidirectional source on stage), ITDG – initial time delay gap, s, BR*) – bass ratio. (SDI*) and BR*) attributes added by Beranek in 1996) .

Attribute	Preferred values	Concert Halls
$IACC_{E3}$	< 0.4	25
EDT	2 – 2.3 s	25
SDI	1	15
G_{mid}	4 – 4.5 dB	15
ITDG	≤ 0.02 s	10
BR	1.1 – 1.45	10
Total	-	100

The overall assessment of the concert hall quality index is calculated as a sum of weighted preferences.

Barron [14] proposed the ranges of acceptability of the measured values of five acoustical parameters, as shown in Table 6.5.

Table 6.5. Recommended ranges of acceptability for objective measures at mid-frequencies for concert halls [14]

Measure	Acceptable range
Reverberation time (RT)	1.8 ≤ RT ≤ 2.2 s
Early decay time (EDT)	1.8 ≤ EDT ≤ 2.2 s
Early-to-late sound index (C80)	-2 ≤ C80 ≤ +2dB
Early lateral energy fraction (LF)	0.1 ≤ LF ≤ 0.35
Total relative sound level (G)	G > 0dB

Following is the discussion of the acoustical attributes as given in Table 6.2 [9].

RT_{oc} (occupate). The preferred value of RT_{oc} chosen for a hall depends on the repertoire of the performing groups that are expected to perform in it. Three possibilities are:

symphonic repertoire, chamber music, and opera. Traditional organ music is best played in a hall with a reverberation time of 3-5 sec. If a separate rehearsal space is to be provided, the optimum reverberation time should probably be lower, provided the resident conductor and orchestra agree. The argument for a lower RT, by 0.2-0.4 sec, is that the musicians can hear every note and nuance during rehearsal, which is not possible in a concert setting if the reverberation time exceeds 1.7 sec.

EDT_{unoc} (unoccupate) The preferred values given in Table 6.2 assume that the chairs are medium upholstered. With leather upholstering or with many chairs in the hall not upholstered, EDT will be longer.

BQI. BQI depends on an abundance of early lateral reflections. Its value can be crudely estimated from architectural drawings if careful consideration is made to the first 10-15 early sound reflections. The difficulty of estimation is the judgment of how much the sound wave will be attenuated and diffused on each reflection. The best method of determining BQI before construction of a hall is to make use of models. Computer models are important in the early stages of design because they can trace out the first 20 or so early reflections fairly accurately. From the model, acoustical "reflectograms" can be obtained that indicate the number and spacing of the early reflections. For greatest accuracy, a wooden model, at 1/10th or 1/20th scale, is best. A miniature loudspeaker can be moved around on the stage to represent the different sections of an orchestra and a small sphere with miniature microphones on two sides representing ears can be used to represent a listener. The "listener" can be moved about the seating areas in the model and BQI can be measured accurately. An important advantage of a "real" model is that the balcony fronts, wall diffusion, and irregularities throughout the room can be adjusted both to satisfy the architect's visual demands and to obtain uniformity of BQI over the seating areas. If the proper equipment is provided, a person using earphones can actually hear the sound as it would be heard in the completed hall. A good feature of the Binaural Quality Index is that its value is nearly the same whether the hall is occupied or unoccupied.

G_{mid} (loudness). … loudness which relates largely to G_{mid} can be too little, assuming a full orchestra is performing, in a very large hall like London's Royal Albert Hall, or too great, as in halls seating less than 1000. Gmid is related to the early decay time EDT_{unoc} divided by the cubic volume V of the hall. The EDT_{unoc} approximately equals RT_{oc} times 1.15 to 1.2 in modern halls with medium upholstering...

G_{125} .The bass response is tied to the strength of the sound at low frequencies, especially the strength of the sound in the 125-Hz octave frequency band.

ITDG. The initial-time-delay gap is usually quoted for an audience position near the center of the main floor, halfway between the stage front and the first balcony front, or the rear wall if there is no balcony. The first reflection in shoebox halls usually comes from a balcony front-otherwise from a lower side wall. In a fan-shaped hall the first reflection may come from suspended panels or the ceiling. In the best concert halls and opera houses the ITDG is less than 25 msec.

SDI (surface diffusion index). The more and varied the surface irregularities in a hall, the better. Experience in five concert halls that the author has been associated with indicates that the depth of the irregularities on the lower sidewalls can be of lesser magnitude than those on the upper walls. The irregularities on the lower walls are designed to remove acoustical "glare." Those on the upper walls primarily affect the reverberation time and need be large to cover a wide frequency range and to thoroughly homogenize the sound.

$C_{80,3}$. This term relates to clarity. If it is positive, as in opera houses, the sound is much clearer than if it is negative. Actually, $C_{80,3}$ correlates highly (inversely) with the reverberation time and is hard to control separately.

ST1. There is no question but that the degree of stage support is important. SD1 measures how well a player hears himself and other players near him. When a canopy is used to create a favorable ST1 on stage, its height should be between 7 and 13 m, adjusted according to the orchestra's preference. Depending upon what energy is reflected from other surfaces this height will make ST1 equal approximately to -12 to -15 dB. The famous Concertgebouw in Amsterdam is an unusual example. It has a high ceiling over the stage, and part of the audience sits in steep formation to either side of the orchestra. Thus, there are no reflections from stage sidewalls. The conductor and musicians have trouble hearing each other and thus must closely watch the conductor's baton. From the standpoint of the quality of the sound in the audience this makes no difference.

Notes:

1. (Beranek 2004) The values given above were measured in halls with acoustical ratings ranging from "fair" to "excellent." When using these numbers in design, observe that a large percentage of the "excellent" halls are "shoebox-shaped." Halls of different shapes could have the same measured numbers but might not sound exactly the same. In other words, there may be other less prominent attributes that contribute to the "excellence" of those halls. Further, some believe, and with some justification, that non-acoustical factors, such as the beauty of the architecture and the quality of the performances, affect the" acoustical quality" judgments of listeners.
2. (Kuttruf 2009), Assessment of concert hall acoustics, … if all designers of concert halls decided to follow the guidelines of this system, the result would be halls not only equally good but of similar acoustics. One may doubt whether this is a desirable goal of room acoustical efforts, since differences in listening environments are just as enjoyable as different architectural solutions or different musical interpretations.
3. (Kuttruf 2009) Conventional attempts to correlate the acoustical quality of a concert hall with an objective measure or a set of them have not been very satisfactory because they have concentrated on one particular aspect only or, as for instance Beranek’s elaborate rating system, relied on plausible but unproven assumptions. Since about 1970, however, researchers have tried to get a complete picture of the factors which contribute to good acoustics, including their relative significance, by employing modern psychometric methods (multidimentional scalling).

7. Measuring techniques for the evaluation of room acoustics parameters

There are two methods of measuring the reverberation time: the classical interrupted noise method and the integrated impulse response method.

In both methods the reverberation time is evaluated from the slope of a line illustrating a sound level decay after switching off the stationary sound source. The reverberation time T, s, corresponds to 60 dB decay of sound level L, dB, so it may be calculated as

$$T = 60\ (\Delta L/\Delta t)^{-1}$$

The decay curve may be plotted directly by level recorder in case of classical method or derived from squared impulse response by digital processing.

Several room acoustic measures are derivable from impulse responses: reverberation, early decay time, measures of relative sound levels, early/late energy fractions and lateral energy fractions [17].

The measured response in the classical method based on noise excitation may 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 not known.

7.1 Conventional interrupted noise methods of measurement

In case of conventional method, in order to measure the reverberation time RT in room and its frequency dependence 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. A typical level versus time diagram is shown in Fig. 7.1.

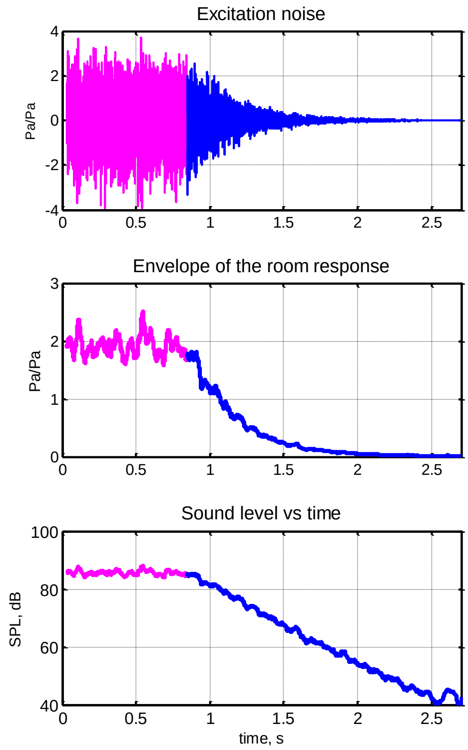


Fig. 7.1 — Typical instantaneous sound pressure sound pressure level versus time curve. Red lines - stationary noise before interruption. Blue lines – noise amplitude and level decay after switch off.

The decay may be further processed to obtain the reverberation time. The classical methods for the measurement of airborne sound in rooms specify a stochastic signal for the excitation. Statistical spread from the random excitation will lead to a certain stochastic variation in the result. 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.

7.2 New measurement method based on impulse response [18]

It has been shown [15] that the expected decay in one particular observation point may be obtained without averaging, by processing the impulse response directly between the excitation signal from a loudspeaker and the observation point at the location of microphone. We assume the system is linear and time-invariant.

According to the new methods, the results may be obtained from processing of the impulse response itself. 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 $t \geq 0$ will be:

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

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 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. Due to the fact that the running time, *t*, is the lower start point for the integration, the operation may be described as backwards integration.

Other types of measurements, such as relative sound pressure levels, early/late energy ratios, lateral energy fractions, interaural cross-correlation functions and background noise levels are needed for a more complete evaluation of the acoustical quality of rooms.

The international standard ISO 3382 [17] establishes the two methods for obtaining reverberation times, from impulse responses and from interrupted noise. The ISO recommendation presents measures based on squared impulse responses : measure of reverberation, early decay time, measures of relative sound levels, early/late energy fractions and lateral energy fractions in auditoria. They are all derivable from impulse responses. Binaural measurements are included to the ISO standard. The dummy heads are required to make these measurements in auditoria. The stage support measures are also introduced for evaluating the acoustic conditions from the musicians' point of view.

7.3 Auditorium measures derived from impulse responses

Reverberation time is the fundamental descriptor of the acoustical character of an auditorium. The addition of newer quantities gives a more complete description of the acoustical properties of the auditorium. The table 7.1 shows a list of subjective attributes and the related acoustic quantities which can be obtained directly from measured impulse responses. The quantities included are limited to those that have been found to be subjectively important.

Table 7.1. The subjective attributes and the related acoustic quantities to be obtained from the measurement of impulse responses

SUBJECTIVE ATTRIBUTES	ACOUSTIC QUANTITY	COMMENTS
Reverberance, Perceived, Reverberance, Liveness	EDT (s), T (s) ST_{Late}, dB	Early decay time EDT, Reverberation time T (occupied), Late support, ST_{Late} relates to perceived reverberance, i.e. the response of the hall as heard by the musician
Perceived clarity of sound, Definition, Center time, Speech intelligibility	D, C50 (dB), C80 (dB), T_s (s), STI	Balance between early and late-arriving energy, Clarity C80 - transparency of music in a concert hall, C50, Definition D ('deutlichkeit') - descriptors of speech intelligibility, Center time T_S, Speech transmission index, STI
Intimacy	ITDG	Initial Time Delay Gap - the time between the arrival of the direct sound at a seat and the arrival of important early reflections
Perveived Spatiousness Two subclasses: 1/ broadening of the source, (ASW); 2/ a sense of being enveloped in the sound, (LEV)).	LF JLF or JLFC, LJ, (dB) IACC	LF - lateral energy fraction. The E(arly) IACC is a measure of the ASW. L(ate} IACC is a measure of the LEV. IACC correlate with the subjective quality "spatial impression" in a concert hall.
Apparent Source Width ASW	JLF or JLFC (dB), E(arly) IACC	Early lateral energy fractions relate to perceived ASW. The E(arly) IACC is a measure of the ASW
Listener envelopment LEV	LJ, (dB), L(ate} IACC	Late lateral sound level relates to perceived LEV or spaciousness in the auditorium. L(ate} IACC is a measure of the listener envelopment LEV

Loudness of reverberant sound	G, dB	Difference of the SPL at a point in a hall and the SPL of the direct field at 10 m for an omnidirectional source on stage
Warmth	BR	Bass ratio - the ratio of the low to mid-frequency reverberation times
Ensemble conditions	ST_{Early}, (dB)	Stage support - Acoustic conditions that allow the musicians to hear each other and that there be a sufficient response from the room. Two different parameters can be derived: 1/ Early support, related to ensemble conditions (ease of hearing other members of an orchestra), 2/ Late support, related to perceived reverberance

The acoustic quantities are grouped in the ISO measurement standard into five types, according to listener aspects. Within each group (Table 7.2) there is often more than one measure, but values of the different quantities in each group are usually found to be strongly correlated with each other. Thus, each group contains a number of approximately equivalent measures and it is not necessary to calculate values of all of them; nevertheless, at least one quantity should be included from each of the five groups.

Table 7.2. Five groups of acoustic quantities for the evaluation of subjective attributes according to ISO 3382 standard

Subjective listener aspect	**Acoustic quantity**	**Single number frequency averaging** Hz	**Just noticeable difference (JND)**	**Typical range** b
Subjective level of sound	Sound strength, G, in decibels	500 to 1 000	1 dB	-2 dB; +10 dB
Perceived reverberance	Early decay time (EDT) in seconds	500 to 1 000	Rel. 5 %	1.0 s; 3.0 s
Perceived clarity of sound	Clarity, C_{80}, in decibels	500 to 1 000	1 dB	-5 dB; +5 dB
	Definition, D_{50}	500 to 1 000	0.05	0.3;0.7
	Centre time, T_S, in milliseconds	500 to 1 000	10 ms	60 ms; 260 ms
Apparent source width (ASW)	Early lateral energy fraction, J_{LF} or J_{LFC}	125 to 1 000	0.05	0.05; 0.35
Listener envelopment (LEV)	Late lateral sound level, L_J, in decibels	125 to 1 000	Not known	-14 dB; +1 dB
Ensemble conditions (c)	Early Support, ST_{Early} (dB)	250 to 2000	Not known	-24 dB; -8dB
Perceived reverberance (c)	Late Support, ST_{Late} (dB)	250 to 2000	Not known	-24dB; -10dB

a The single number frequency averaging denotes the arithmetical average for the octave bands, except for LJ which shall be energy averaged
b Frequency-averaged values in single positions in non-occupied concert and multi-purpose halls up to 25 000 m^3.
(c) Performers listener aspect, on orchestra platform.

The frequency range depends on the purpose of the measurements. For the survey method where frequency bands were not specified the frequency range should cover at least 250 Hz to 2 000 Hz. For the engineering and precision methods, the frequency range should cover at least 125 Hz to 4 000 Hz in octave bands, or 100 Hz to 5 000 Hz in one-third octave bands.

The measurement should be carried over a reasonable number of seat positions, between 10 and 20 positions for a large auditorium.

The impulse response from a source position to a receiver position in a room is a well-defined quantity that can be measured in a variety of ways, e.g. using pistol shots, spark gap impulses, noise bursts, chirps or MLSs as signals. ISO 3382-1 standard states that any other method that can yield the correct impulse response is not excluded.

7.3.1 Excitation of the room

The impulse response can be measured directly using an impulse source such as a pistol shot. Spectrum of the impulse shall be broad enough to meet the requirements. The impulse source shall be able to produce a peak sound pressure level sufficient to ensure a decay curve starting at least 35 dB above the background noise in the corresponding frequency band. If T_{30} is to be measured, it is necessary to create a level at least 45 dB above the background noise level.

In order to improve signal-to-noise ratio special sound signals may be used which yield the impulse response after processing of the recorded microphone signal. Sine sweeps or pseudo-random noise as MLS, may be used if the requirements for the spectrum and directional characteristics of the source are fulfilled. Because of the improvement in signal-to-noise ratio, the dynamic requirements on the source can be considerably lower than those for the interrupted nose method.

Using these measuring techniques, the frequency filtering is often inherent in the signal analysis, and it is sufficient that the excitation signal cover the frequency bands to be measured.

7.3.2 Integration of the impulse response

The decay curve is obtained by a backward integration of the squared impulse response for each octave band. In an ideal situation with no background noise, the integration ought to start at the end of the impulse response ($t \to \infty$) and proceed to the beginning of the squared impulse response. Thus, the time function of the decay *E(t)* may be obtained according to:

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

where

p - the sound pressure of the impulse response as a function of time,

E - the energy of the decay curve as a function of time,

t - the time.

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

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

In order to minimize the influence of the background noise on the later part of the impulse response, the following technique may be used.

If the level of the background noise is known, determine the starting point of the integration, t_1, as the intersection between a horizontal line through the background noise and a sloping line through a representative part of the squared impulse response displayed using a decibel scale, and calculate the decay curve from:

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

where ($t < t_1$) and C is an optional correction for integrated squared impulse responses between t_1 and infinity.

7.3.3 Evaluation of decay curves

For the determination of T_{30}, the evaluated range for the decay curves is from 5 dB to 35 dB below the steady state level. For the integrated impulse response method, the steady state level is the total level of the integrated impulse response. Within the evaluation range, a least-squares fit line shall be computed for the curve. The slope of the straight line gives the decay rate, d, in decibels per second, from which the reverberation time is calculated as $T_{30} = 60/d$.

For the determination of T_{20}, the evaluation range is from 5 dB to 25 dB. In order to specify a reverberation time, the decay curves shall follow approximately a straight line. If the curves are wavy or bent, this may indicate a mixture of modes with different reverberation times and thus the result may be unreliable.

7.3.4 Sound strength measurement

The sound strength, G, can be measured using a calibrated omnidirectional sound source, as the logarithmic ratio of the sound energy (squared and integrated sound pressure) of the measured impulse response to that of the response measured in a free field at a distance of 10 m from the sound source.

$$G = 10 \lg \frac{\int_0^\infty p^2(t)\, \mathrm{d}t}{\int_0^\infty p_{10}^2(t)\, \mathrm{d}t} = L_{pE} - L_{pE,10} \text{ dB}$$

In which

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

and

$$L_{pE,10} = 10 \lg \left[\frac{1}{T_0} \int_0^\infty \frac{p_{10}^2(t)\, \mathrm{d}t}{p_0^2} \right] \text{ dB}$$

where
$p(t)$ - the instantaneous sound pressure of the impulse response measured at the measurement point, $p_{10}(t)$ - the instantaneous sound pressure of the impulse response measured at a distance of 10 m in a free field, p_0 - 20 μPa; T_0 = 1 s, L_{pE} - the sound pressure exposure level of $p(t)$, $\boldsymbol{L_{pE,10}}$ - the sound pressure exposure level of $p_{10}(t)$.

In the above equations, $t = 0$ corresponds to the start of the direct sound, and ∞ should correspond to a time that is greater than or equal to the point at which the decay curve has decreased by 30 dB. In the case where a large anechoic room is available, $L_{pE,10}$ can be directly measured using a source-to receiver distance of 10 m. If this condition is not attainable, the sound pressure exposure level at a point which is $d \geq 3$ m from the source $L_{pE,d}$ may be measured and $L_{pE,10}$ then obtained from:

$$L_{pE,10} = L_{pE,d} + 20 \lg (d/10) \text{ dB}$$

When making such a measurement in a free field, it is necessary to make the measurement at every 12,5° around the sound source and to calculate the energy-mean value of the sound pressure exposure levels in order to average the directivity of the sound source.

7.3.5 Early decay time measurement

The early decay time EDT shall be evaluated from the slope of the integrated impulse response curves. The slope of the decay curve should be determined from the slope of the best-fit linear regression line of the initial 10 dB, between 0 dB and −10 dB of the decay. The decay times should be calculated from the slope as the time required for a 60 dB decay.

Both the EDT and *T* should be calculated. EDT is subjectively more important and related to perceived reverberance, while *T* is related to the physical properties of the auditorium.

7.3.6 Balance between early- and late-arriving energy

There are several parameters that can be used in this group. One of the simplest is an early-to-late arriving sound energy ratio. This can be calculated for either a 50 ms or an 80 ms early time limit, depending on whether the results are intended to relate to conditions for speech or music, respectively:

$$C_{t_e} = 10 \lg \frac{\int_0^{t_e} p^2(t)\, dt}{\int_{t_e}^{\infty} p^2(t)\, dt} \text{ dB}$$

where

C_{te} - the early-to-late index; t_e is the early time limit of either 50 ms or 80 ms (*C*80 is usually “clarity”); *p*(*t*) - the instantaneous sound pressure of the impulse response measured at the measurement point.

It is also possible to measure an early to total sound energy ratio. For example, *D*50 (“definition” or “Deutlichkeit”) is sometimes used for speech conditions, as:

$$D_{50} = \frac{\int_0^{0,050} p^2(t)\, dt}{\int_0^{\infty} p^2(t)\, dt}$$

This is related to *C*50 by the following relationship :

$$C_{50} = 10 \lg \left(\frac{D_{50}}{1 - D_{50}} \right) \text{ dB}$$

It is not necessary to measure both quantities.

As a final option in this group of measures, the centre time, T_S, which is the time of the centre of gravity of the squared impulse response, can be measured, in seconds:

$$T_S = \frac{\int_0^{\infty} t\, p^2(t)\, dt}{\int_0^{\infty} p^2(t)\, dt}$$

T_S devides the impulse response into early and late periods. Quantities in this group relate to perceived definition, clarity, or the balance between clarity and reverberance, as well as to speech intelligibility.

7.3.7 Early lateral energy measures

The fraction of energy, $\boldsymbol{J_{LF}}$, arriving from lateral directions within the first 80 ms can be measured from impulse responses obtained from an omnidirectional microphone and a figure-of-eight pattern microphone:

$$J_{LF} = \frac{\int_{0,005}^{0,080} p_L^2(t)\,dt}{\int_{0}^{0,080} p^2(t)\,dt}$$

where
$p_L(t)$ - the instantaneous sound pressure in the auditorium impulse response measured with a figure-of-eight pattern microphone; $p(t)$ - the instantaneous sound pressure of the impulse response measured at the measurement point.

The null of the figure-of-eight pattern microphone shall be pointed towards an average centre-stage source position, or exactly towards individual source positions, so that this microphone responds predominantly to sound energy arriving from lateral directions and is not significantly influenced by the direct sound. Because the directivity of the figure-of-eight microphone is essentially a cosine pattern and pressure values are squared, the resulting contribution to lateral energy for an individual reflection varies with the square of the cosine of the angle of incidence of the reflection relative to the axis of maximum sensitivity of the microphone. As an alternative, an approximation can be used for obtaining lateral energy fractions, J_{LFC}, with contributions which vary as the cosine of the angle, thought to be subjectively more accurate:

$$J_{LFC} = \frac{\int_{0,005}^{0,080} |p_L(t) \cdot p(t)|\,dt}{\int_{0}^{0,080} p^2(t)\,dt}$$

where
$p_L(t)$ - the instantaneous sound pressure in the auditorium impulse response measured with a figure-of-eight pattern microphone; $p(t)$ - the instantaneous sound pressure of the impulse response measured at the measurement point.

Lateral energy fractions relate to perceived width of the sound source. Interaural cross correlation measures are also thought to relate to spatial impression.

7.3.8 Late lateral energy measures

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

$$L_J = 10 \lg \left[\frac{\int_{0,080}^{\infty} p_L^2(t)\,dt}{\int_{0}^{\infty} p_{10}^2(t)\,dt} \right] \text{dB}$$

where
$p_L(t)$ is the instantaneous sound pressure in the impulse response measured with a figure-of-eight pattern microphone, $p_{10}(t)$ is the instantaneous sound pressure in the impulse response measured with an omnidirectional microphone at a distance of 10 m in a free field.

The null of the figure-of-eight pattern microphone shall be pointed towards an average centre stage source position, or exactly towards individual source positions, so that this microphone responds predominantly to sound energy arriving from lateral directions and is not significantly influenced by the direct sound.

The frequency-averaged late lateral sound energy level, $L_{J,\text{avg}}$, is calculated from:

$$L_{J,\text{avg}} = 10 \lg \left[0{,}25 \sum_{i=1}^{4} 10^{L_{J_i}/10} \right] \text{dB}$$

where
$L_J\, i$ - the value in octave band *i; i* - each of the four octave bands with centre frequencies 125 Hz, 250 Hz, 500 Hz and 1 000 Hz.

Late lateral sound energy relates to perceived listener envelopment or spaciousness in the auditorium.

7.3.9 Measurement procedure

Source

The source and associated equipment should be adequate to radiate a sufficient signal level in all of the octave bands for 125 Hz to 4 000 Hz, so that an adequate decay range is achieved in each octave band. The source shall be as omnidirectional as possible (see Fig. 7.2).

Fig 7.2. View of the typical omnidirectional source made of 12 loudspeakers, mounted on the surface of dodecahedron.

Microphones

An omnidirectional microphone should be used to measure the impulse response for all of the measures. For J_{LF} values, a figure-of-eight pattern microphone is also required, and the relative sensitivities of the omnidirectional and figure-of-eight microphones in the direction of maximum sensitivity should be calibrated in a free sound field.

For *G* values, the sensitivity of the omnidirectional microphone shall be calibrated.

7.3.10 Impulse responses

Octave band impulse responses are necessary for the calculation of all quantities. Methods using a loudspeaker source are limited by the frequency and directional response of the loudspeaker. The average frequency response can, to some extent, be corrected, but variations with direction cannot be eliminated and become significantly large at higher frequencies. Cross-correlation of the source signal and the received signal can provide impulse responses with good dynamic range and immunity to noise. The use of fast Hadamard transforms and MLS signals is one successful correlation-type approach [2]. Other signals with broad smooth spectrum, such as chirps and linear sweeps, can also be successfully used.

7.3.11 Binaural auditorium measures derived from impulse responses

Subjective studies of auditoria have shown that inter-aural cross correlation coefficients IACC, measured with either a dummy head or a real head with average dimensions, and with small microphones at the entrance to the ear canals, correlate well with the subjective quality “spatial impression” in a concert hall (early lateral energy measures are also thought to relate to spaciousness).

Spatial impression may be divided into two subclasses:

– Subclass 1: broadening of the source, i.e. apparent source width ASW,
– Subclass 2: a sense of being immersed or enveloped in the sound, i.e. listener envelopment LEV.

<u>Definition of IAAC</u>

The normalized inter-aural cross correlation function IACF is defined using following equation:

$$\mathrm{IACF}_{t_1,t_2}(\tau) = \frac{\int_{t_1}^{t_2} p_\mathrm{l}(t) \cdot p_\mathrm{r}(t+\tau)\,\mathrm{d}t}{\sqrt{\int_{t_1}^{t_2} p_\mathrm{l}^2(t)\,dt \int_{t_1}^{t_2} p_\mathrm{r}^2(t)\,\mathrm{d}t}}$$

where

$p_\mathrm{l}(t)$ - the impulse response at the entrance to the left ear canal, $p_\mathrm{r}(t)$ - the impulse response at the entrance to the right ear canal.

The inter-aural cross correlation coefficients, IACC, are then given by:

$$\mathrm{IACC}_{t_1,t_2} = \max\left|\mathrm{IACF}_{t_1,t_2}\right| \quad \text{for } -1\,\mathrm{ms} < \tau < +1\,\mathrm{ms}$$

<u>Dummy head</u>

A dummy head, with pinna and ear canals, should be chosen as standard for a given set of measurements.

The view of a dummy head is shown on the Fig. 7.3.

Fig. 7.3. View of the Numann KU100 dummy head

When making measurements in an auditorium, the height of the ear canals of the dummy head above the floor should be about 1.2 m.

<u>Uses of IACC</u>

The uses of IACC have not yet been accepted uniformly. As in the case of J_{LF} and J_{LFC}, the use of IACC and its subjective relevance are still subject to discussion and research. Likewise, different approaches have been suggested regarding the choice of the time limits t_1 and t_2 and the frequency filtering of the signals.

The most general form of IACC is defined with $t_1 = 0$ and $t_2 = \infty$ - in room acoustics, a time of the order of the reverberation time. As in the case of monaural measurements, IACC is generally measured in octave bands ranging from 125 Hz to 4 000 Hz.

IACC can be measured to describe the dissimilarity of the signal arrival at the two ears, either for the early reflections ($t_1 = 0$ and $t_2 = 0.08$ s) or for the reverberant sound ($t_1 = 0.08$ s and t_2 - a time greater than the reverberation time of the enclosure).

The JND (just noticeable difference) of IACC is assumed to be 0,075.

7.3.12 Stage measures derived from impulse responses

In concert halls and other performance spaces, it is important that the acoustic conditions allow the musicians to hear each other and that there be a sufficient response from the room. For an objective evaluation of these conditions, it has proven to be useful to measure on the orchestra platform with the source and microphone close together [24]. Two different parameters can be derived from the measurements (see Table 7.3).

Table 7.3. Acoustic parameters measured on orchestra platforms

Subjective listener aspect	Acoustic quantity	Single number frequency averaging Hz	JND (just noticeable difference)	Typical range
Ensemble conditions	Early support, ST_{Early}, in decibels	250 to 2 000	Not known	−24 dB; −8 dB
Perceived reverberance	Late support, ST_{Late}, in decibels	250 to 2 000	Not known	−24 dB; −10 dB

<u>Definition of measures</u>

<u>Early support</u>

This is the ratio, in decibels, of the reflected energy within the first 0.1 s relative to the direct sound (including the floor reflection), both measured at a distance of 1.0 m from the acoustic centre of an omnidirectional sound source. Other reflecting surfaces or objects should be more than 2 m from the measurement position.

$$ST_{Early} = 10\lg\left[\frac{\int_{0,020}^{0,100} p^2(t)\,dt}{\int_{0}^{0,010} p^2(t)\,dt}\right]\ \text{dB}$$

where
$p(t)$ - the instantaneous sound pressure of the impulse response, measured at the measurement point, and $t = 0$ corresponds to the arrival of the direct sound.

Early support relates to ensemble, i.e. ease of hearing other members of an orchestra. However, the influences of the direct sound, delay time and reflections from near surfaces are not included.

<u>Late support</u>

This is the ratio, in decibels, of the reflected energy after the first 0.1 s relative to the direct sound (including the floor reflection), both measured at a distance of 1.0 m from the acoustic centre of an omnidirectional sound source. Other reflecting surfaces or objects should be more than 2 m from the measurement position.

$$\mathrm{ST_{Late}} = 10 \lg \left[\frac{\int_{0,100}^{1,000} p^2(t)\,\mathrm{d}t}{\int_{0}^{0,010} p^2(t)\,\mathrm{d}t} \right] \mathrm{dB}$$

Where
$p(t)$ - the instantaneous sound pressure of the impulse response, measured at the measurement point, and $t = 0$ corresponds to the arrival of the direct sound.

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

Measurement positions

The height of the source and the microphone shall be the same: either 1.0 m or 1.5 m above the floor. At least three different positions of the source and receiver should normally be used. Measurements should preferably be made with chairs and music stands on the orchestra platform, but the nearest chairs and music stands within a distance of 2 m from the source and microphone should be removed in order not to reflect the sound directly to the microphone.

Statement of results

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

7.4 Measurement of the impulse response [18]

7.4.1 General

The room may be excited by a known signal for a certain time and the impulse response calculated from the response to the excitation. The excitation signal is distributed over a longer period of time to increase the total radiated energy.

During measurements, movement of the source or the microphones is not acceptable as it will violate the requirement for 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.

7.4.2 Excitation signal

The new methods apply deterministic excitation signals, i.e. they can be accurately reproduced, and thereby enhance the repeatability of the measurement.

7.4.2.1 Spectral requirements

The effective frequency range of the excitation signal shall at least cover the actual fractional-octave band being measured. If a broad-band measurement covering the whole audio range is being performed, the aim is to approximate the shape of the spectrum of the excitation signal, as captured at the receiver position, to that of the ambient noise prevailing there. By this, a frequency-independent signal-to-noise ratio will be obtained.

The typical sources of background noise (air-conditioning, traffic, etc.) tend to have a spectral distribution that increases with decreasing frequency. In these cases, a pink excitation signal,(with constant energy per fractional-octave band, is suitable to obtain a sufficient signal-to-noise ratio.

7.4.2.2 Repetitive excitation

If a repetitive excitation signal is used, the spectrum of the excitation will consist of narrow spectral lines where the distance between adjacent lines, Δ*f*, will be given as the inverse of the time for one repetition period

$$\Delta f = 1/T_{REP}$$

In order to ensure that all modes of the room are excited, the repetition period shall not be shorter than the reverberation time, *T*, for the room being measured.

$$T_{REP} \geq T$$

7.4.2.3 Non-repetitive excitation

A non-repetitive excitation signal may be of any suitable length. However, the excitation shall be succeeded by a period of silence in order to allow the decaying response to be properly recorded. The decay shall be recorded over a period equal to at least half of the reverberation time. For a sweep excitation from a low to a high frequency, the required length of the period of silence will normally be determined by the reverberation time for the higher frequencies.

7.4.2.4 Level and linearity

The sound power in the excitation shall be sufficiently high to obtain an effective signal-to-noise ratio. Methods involving deterministic excitation signals are generally more efficient at suppressing extraneous noise than the classical method. Enhancement of the signal-to-noise ratio by 20 dB to 30 dB compared to the classical method may be obtainable.

The use of loudspeakers typically introduces non-linear distortion in the system. Distortion violates the requirement for linearity in this method. Distortion due to the loudspeaker increases with the excitation level. Sometimes the signal-to-noise ratio may be increased by reducing the excitation level. The swept-sine method allows elimination of artefacts in the measurement result caused by harmonic distortion.

7.4.2.5 Measurement of the frequency response function

The frequency response function may be obtained from the impulse response by Fourier transformation. It may also be measured as the response to sinusoidal excitation in the required frequency range and the response recorded as amplitude and phase.

The frequency may be changed continuously, normally from below the lowest band edge frequency of the lowest fractional-octave band to be measured to above the upper band edge frequency of the upper band. A frequency sweep where the frequency increases exponentially as a function of time mimics a pink noise source in the classical method.

7.4.3 Maximum length sequence method

7.4.3.1 General

A maximum length sequence MLS is a binary sequence. When used for excitation, the binary values are output at a fixed rate, f_c, which is assumed to be 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 $l_1 = 2N - 1$. 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 record of white noise replayed with a repetition frequency

$$f_{REP} = \frac{1}{T_{REP}} = \frac{f_c}{2^N - 1}$$

For this description, it is assumed that the sequence is replayed periodically. The measured impulse response will therefore also be periodic, meaning that the tail of the response outside the record will be folded back to the beginning of the record.

When using white excitation signals, the impulse response of any linear system may in general be obtained from the cross correlation between the output and the input. When the input is a periodic maximum length sequence, using the Hadamard transformation may speed up this cross correlation process. The process is illustrated in Fig. 7.4.

The Hadamard transformation may be done very efficiently as a Fast Hadamard Transformation (FHT) and 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, l_2 , is a power of two value:

$$l_2 = l_1 + 1 = 2^N$$

The output from the Hadamard transformation will be the impulse response for the measured system. In addition to the room response, the power amplifier, loudspeaker and possibly filtering networks will be a part of the system. For most acoustic measurements, the main characteristics in the response will be determined by the room. The impulse response shall be further processed to obtain the fractional-octave-band filtered response.

Due to the periodic nature of the excitation, the measured impulse response will also be periodic. If the impulse response is longer than one period, the tail of the impulse response will add to the first part (time aliasing – circular convolution).

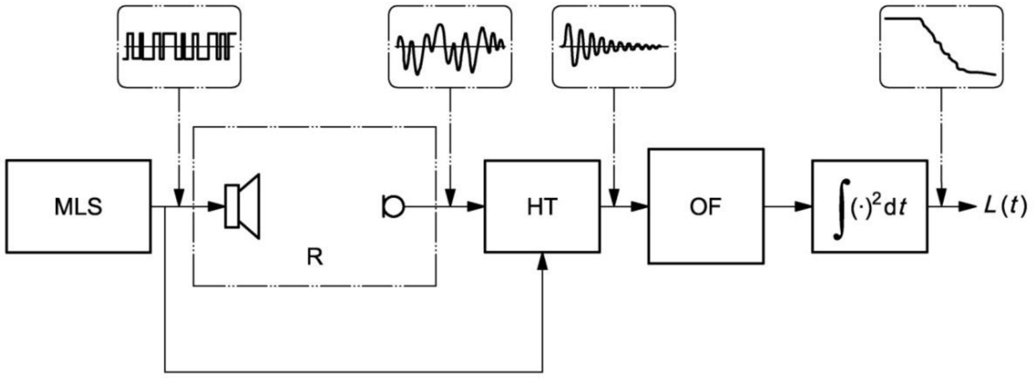


Fig. 7.4. Scheme for MLS process. MLS - generator for MLS signal, R – room, HT - Hadamard transformation, OF - fractional-octave filter.

7.4.3.2 Sequence length

The length of the sequence shall be equal to or longer than the reverberation time. The lower limit for the clock frequency, f_c, will be set by the required upper frequency range. The length of the sequence and the clock frequency will set the lower limit for the order of the MLS.

NOTE The calculation of the required order may be illustrated by the following example. The estimated reverberation time T is 1.5 s, the upper measurement frequency is 3,55 kHz (upper band edge frequency for 3.15 kHz filter band), the clock and sampling frequency is 12 kHz. The lower limit for the order is given by:

$$T_{\mathrm{REP}} = \frac{2^N - 1}{f_{\mathrm{c}}} \geqslant T$$

This corresponds to

$$N \geqslant \frac{\lg(Tf_{\mathrm{c}} + 1)}{\lg(2)} = \frac{\lg(1{,}5 \times 12\,000 + 1)}{\lg(2)} \approx 14{,}2$$

The smallest whole number satisfying this requirement is N = 15. The order may, however, be reduced to 14 if the sampling frequency is reduced closer to the limit 2 × 3.55 kHz or 7.1 kHz.

7.4.3.3 Signal-to-noise ratio

Level measurements

The signal-to-noise ratio for the classical method is the ratio obtained when the MLS-signal is used as a conventional excitation signal.

The enhancement in the effective signal-to-noise ratio in decibels, Δ, compared to the classical method will be approximately given by:

$$\Delta \approx 10\ \lg\left\{\frac{n\,T_{\mathrm{REP}}}{t_1}\right\}\ \mathrm{dB}$$

where *n* is the number of averages.

Measurement of reverberation time

The MLS sequence may be regarded as a conventional noise excitation signal and the reverberation time calculated by the classical interrupted noise method. If the signal-to-noise ratio in this measurement is used as a reference, the enhancement, Δ, in the effective signal-to-noise ratio for the MLS/Hadamard method will be approximately given by:

$$\Delta \approx 10\ \lg\left\{\frac{13{,}8 \times n\,T_{\mathrm{REP}}}{T}\right\}\ \mathrm{dB}$$

where ***T*** is the reverberation time.

7.4.3.4 Time invariance

Time-variance in the system to be measured may limit the achieved effective signal-to-noise ratio and lead to unreliable results. Besides linearity, time-invariance is a critical parameter in the application of MLS-based methods and shall always be considered.

All excitation sources, reflectors, microphones, other equipment or room boundaries shall be stationary and shall not be moved during a measurement.

7.4.4 Swept-sine method

7.4.4.1 General

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 latter 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 advantages compared to the MLS method. The obtainable advantages include reduced sensitivity to time variance (temperature and air movement) and elimination of the deterioration of the effective signal-to-noise ratio due to harmonic distortion. All harmonic distortion can be deleted from the results and 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 sweep excitation may be made once, from the lower to the higher frequency, or repeated in a periodic manner. The analysis is based on one single sweep. If convenient, the total frequency range may be divided into blocks covering only a part of the range, e.g. each fractional octave may be measured with a separate sweep. When a sweep is emitted only once, all its energy is being used for evaluating the frequency response function. This aperiodic use of the excitation signal reduces the measurement time, although a gap of silence shall follow the sweep to allow the collection of all delayed components.

7.4.4.2 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 shall be longer than the sweep itself to collect the reverberation until it has decayed under the noise floor.

In room acoustics, the reverberation time is normally longest for the lower frequencies. When very long sweeps (many seconds) are being used, the final gap only shall 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.

7.4.4.3 Sweep generation

According to the spectral requirements, a non-white excitation spectrum is preferable for the majority of measurement tasks. The spectral contents 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 measured. An extension of the sweep with a quiet period is normally required and belongs to the excitation signal.

In cases with moderate background noise, it is normally a safe practice to use sweeps with a length of two to four times the longest reverberation time and leave a silent gap equalling the expected longest reverberation time.

7.4.4.4 Sweeps with white and pink spectrum

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 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 is the normal excitation signal corresponding to pink noise in the applicable classical methods.

7.4.4.5 Synthesis of sweeps with arbitrary amplitude spectrum

The spectrum of the excitation signal may be adjusted to the requirements by changing the sweep rate. Generally, adjustment of the sweep rate is preferred compared to change of the envelope (amplitude) as it allows a constant distance to the clipping level of an amplifier to be maintained.

If the sweep rate at a particular frequency is lowered, more energy will be concentrated in this part of the spectrum. By properly controlling the sweep rate, sweeps with almost arbitrary amplitude spectrum, yet almost constant temporal amplitude envelope can be synthesized.

7.4.4.6 Recording the response

The response to the sweep excitation shall be recorded from the start of the sweep to a time where the sound delayed by the reverberation is received. The time needed for the record depends on the sweep speed, the frequency range to be covered and the reverberation of the room. Fig; 7.5 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.

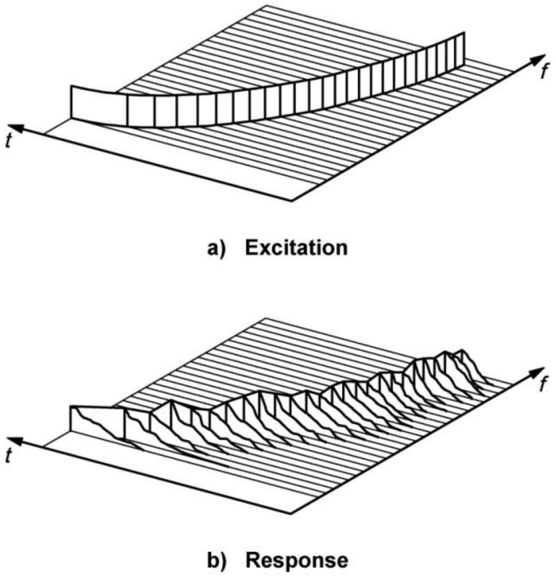


Fig. 7.5. Time-frequency diagram for exponential sweep; ***t*** - time, ***f*** – frequency.

7.4.4.7 Deconvolution

The process of deconvolution to obtain the impulse response of the room is illustrated in Fig. 7.6 and 7.7 as two alternative procedures. 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. In Fig. 7.6, the deconvolution is done by convolving the received signal with a signal that is the inverse of the excitation signal. The inverse signal is a signal with the property that an ideal impulse is created on convolving it with the excitation signal. The broadband impulse response is further processed to obtain the $L(t)$ for each fractional-octave band.

In Fig. 7.7, the transformation between the time- and frequency domain is indicated by Fast Fourier Transformation (FFT). The broadband impulse response is obtained after a transformation back to the time domain by inverse FFT. The broadband impulse response is further processed to obtain the $L(t)$ function for each fractional-octave band.

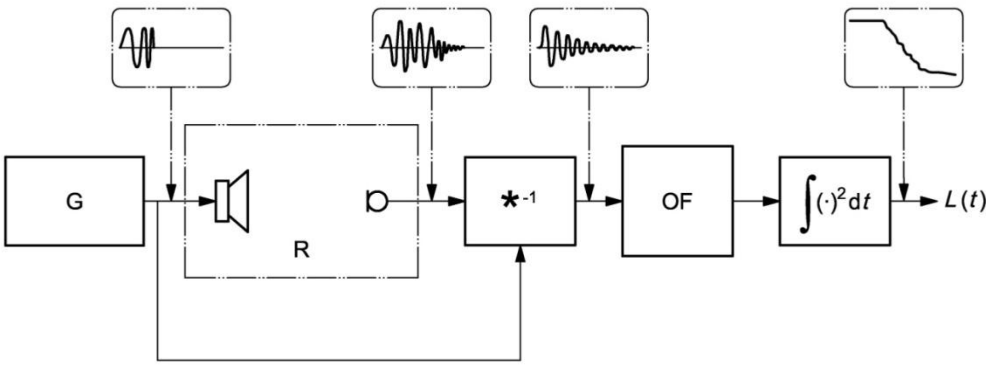


Fig. 7.6. Direct deconvolution; **G** - sweep generator, **R** - room, ***–1** - deconvolution, **OF** - fractional-octave filter.

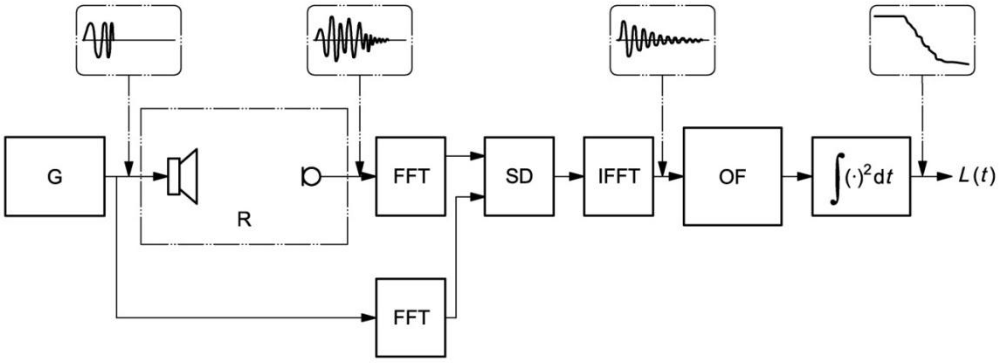


Fig. 7.7. Spectral division; **G** - sweep generator, **R** - room, **FFT** - Fast Fourier Transformation, **SD** - spectral division, **IFFT** - Inverse Fast Fourier Transformation, **OF** - fractional-octave filter.

If a Fast Fourier Transformation is used, precautions against circular convolutions shall be taken. Furthermore, the spectral division may include frequencies in the denominator with very little energy and precautions shall be taken in order not to enhance the extraneous noise accompanying the measured response at these frequencies. This will often be the case close to the boundary of the sweep range.

As the technique described here makes use of a non-periodic excitation signal, the most appropriate way to obtain the impulse response is a linear (i.e. non-circular) deconvolution. The linear deconvolution can be accomplished most simply by direct deconvolution or, if using spectral division, by extending the excitation signal and the recorded response with zeros to double their previous size (zero-padding).

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 deconvolution, the responses to harmonic components in the excitation will appear at negative time and may easily be removed.

When spectral division is applied, the excitation and the response are submitted to an FFT and the spectrum of the response is then divided by the spectrum of the excitation signal. An IFFT yields the desired impulse response in which the second half, corresponding to negative arrival times, can be disregarded. This method may also be used to remove the effects of harmonic distortion in the excitation channel.

Alternatively to the linear deconvolution, a circular deconvolution using an FFT size equal to the acquisition time may be employed. In this case however, the distortion products could smear into the decay of the impulse response. This means that the length of the excitation signal shall be chosen sufficiently longer than the decay time. The distortion products will then appear in the noise floor where they can be safely discarded by windowing without affecting the reverberant tail.

There is an important difference concerning the noise floor in the impulse responses obtained by linear and circular deconvolution. Use of a circular deconvolution results in a noise floor that is basically constant, up to the point where the first distortion products appear. The linear deconvolution, however, yields a decaying noise tail that is increasingly low-pass filtered towards its end. This stems from the fact that this last part of the deconvolution result originates from steady noise convolved with a sweep in reverse order, i.e. from high to low frequencies.

7.4.4.8 Signal-to-noise ratio

The obtained effective signal-to-noise ratio depends on a number of factors in addition to the extraneous noise level, such as the amplitude of excitation, the sweep rate and the algorithms for the signal processing. Doubling the duration of the sweep will normally lead to an increase in the effective signal-to-noise ratio by 3 dB. Although the signal-to-noise ratio may be enhanced by averaging more impulse responses, this method is, in general, not recommended as it will lead to an increased sensitivity for changes in the environmental conditions. However, most of such environmental effects are observed at levels below 30 dB from the peak level.

7.4.4.9 Time-invariance

Time-variance in the system to be measured may lead to unreliable results and limit the achieved effective signal-to-noise ratio. Most effects related to modest time-variance are seen at levels below 30 dB from the peak level of the impulse response. It will therefore seldom affect a level measurement, but may give an unreliable reverberation time. For other applications requiring a huge dynamic range, the sensitivity may be even larger.

All excitation sources, reflectors, microphones, other equipment or room boundaries shall be stationary and shall not be moved during a measurement.

8. Applied room acoustics

8.1 Design procedures in room acoustics planning

Two general groups of interiors which differ from the view point of acoustic design and acoustic treatment may be considered separately:

I. small and large rooms for speech and music presentations, production and listening, as lecture halls, theatre and music halls, etc.,
II. offices, production spaces, service facilities, etc., in which the noise control is the main interest.

The purpose of this chapter is to describe some practical issues of room acoustics planning of the rooms of the first group, namely the acoustical design of small listening rooms, rooms for music production and auditoria for performing, i.e.:

- listening rooms,
- small recording studios,
- control rooms,
- audio/video rooms,
- concert halls and opera houses,
- pop and rock interiors with amplified music.

Acoustical designing of rooms from the second group consist mainly of the protection against excessive reverberance, noises originating from room appliances, air conditioning, noise transmitted from neighboring rooms and external noise sources as street traffic noise. That kind of acoustic issues are categorized as building acoustics problems and are out of the scope of our main subject - room acoustics.

Below (table 8.1) are given allowable limits for background noise in the acoustically sensitive spaces such as recording studios, theatres and concert halls.

Table 8.1 Allowable limits for background noise in the acoustically sensitive spaces [21], [26]

Type of Room	Recommended NR Level (NR Curve)	Equivalent Sound Level, dBA
Reference listening rooms	10-15	20-25
Recording studios	15–20	25–30
Concert and recital halls	15–20	25–30
TV broadcast studios	20–25	30–35
Performing art spaces	20–25	30–45
Assembly halls	25–30	35–40
Motion picture theaters	25–30	35–40
Sport halls	45–55	55–65

The objective in any room design is to achieve a noise level appropriate for the functions of the space, not the lowest possible noise level.

8.2 The basic recommended values of objective parameters

There are a few objective sound field properties which are important for good (or poor) acoustics of a hall, namely, the duration of reverberation processes, the strength of the direct sound and the temporal and directional distribution of the early sound energy. These properties depend on:

- shape of the room,
- volume of the room,
- number of seats and their arrangement,
- materials of walls, ceiling, floor, seats, etc.

Reverberation time is the one of most recognized determinants of acoustical quality of rooms. Excessively absorptive or excessively reverberant room will deteriorate music quality. The same may happen with intelligibility of speech communication in too long reverberation time.

8.2.1 Small rooms acoustics

The suitable reverberation time value for a room may be established from normative recommendations. The acoustic parameters of listening rooms, sound control rooms and recording studios of the volume less than 300 m^3 are covered by the EBU (European Broadcasting Union) specifications [21].

The EBU document specifies the demands for direct sound, early reflections and reverberation time limits.

Two types of listening room are specifically covered by the this specifications:

- reference listening rooms used for listening tests and listening group activities for between three and seven listeners,
- high quality sound control rooms such as sound control rooms for music and drama recordings and post–production operations.

In these rooms, high quality reference type monitor loudspeakers are normally used for sound reproduction.

The walls, floor and ceiling of the listening room define the boundaries of a controlled acoustic environment, and to achieve consistent results, care must be taken in the design of the room itself and in the placement of the loudspeakers and the listeners. The quality of the listening environment is governed by the properties of the sound field produced by the loudspeakers in the listening area, at the height of the listeners' ears.

The minimum dimensions floor area should be 40 m^2 for a reference listening room; 30 m^2 for a high–quality sound control room. The size of a listening room will be determined as much by the space for installation of technical equipment, as by the acoustic aspects. In any case, the volume should not exceed 300 m^3.

To ensure a reasonably uniform distribution of the low–frequency eigentones, the proportions of the room must lie within controlled limits. The following limits for the length to height and the width to height ratios should be preserved:

$$
\begin{gathered}
1.1 * \boldsymbol{w}/\boldsymbol{h} \le \boldsymbol{l}/\boldsymbol{h} \ge 4.5 * \boldsymbol{w}/\boldsymbol{h} - 4 \\
\boldsymbol{l} < 3\boldsymbol{h} \\
\boldsymbol{w} < 3\boldsymbol{h}
\end{gathered}
$$

where:
l - length (larger dimension of floor plan, irrespective of orientation),
w - width (shorter dimension of floor plan, irrespective of orientation),
h - height.
In addition, ratios of ***l***, ***w*** and ***h*** which are within + 5% of integer values should also be avoided. All dimensions are overall, measured to the internal structural surfaces.

Following design tips are given in order to enhance the acoustic environment of listening room:

a) the room should exhibit sufficient acoustic symmetry about the listening direction. This requirement concerns not only the shape but also the absorption of the surfaces.

Surfaces in the neighbourhood of the loudspeakers are particularly important in this respect,

b) the location of sound absorbers and reflecting surfaces on the walls, ceiling and floor of the room should be chosen to avoid flutter echoes and disturbing single reflections,
c) any resonating structures in the room must be sufficiently damped (attenuation time shorter than the room reverberation time). The room must not contain any structures which cause audible rattling during loudspeaker listening,
d) the acoustic requirements and recommendations should be taken into account in regard to all surfaces and objects in the room, such as the door and window surfaces and the technical equipment. In particular, the working surface of the mixing desk can cause a disturbing early reflection,
e) for airborne and impact noise insulation, the background noise requirements should be considered.

8.2.1.1 Early reflections

Early reflections are defined in the EBU recommendation as reflections from boundary surfaces or other surfaces in the room which reach the listening area within the first 15 ms after the arrival of the direct sound. The levels of these reflections should be at least 10 dB below the level of the direct sound for all frequencies in the range 1 kHz to 8 kHz.

The amplitude and frequency responses of individual reflections may be derived from a measurement of the room impulse response, using Fourier transform methods. It is important that the effective time window and bandwidth of the measurement are appropriate. The measurement of early reflections of sounds including components in the frequency range below 500 Hz may be difficult.

The effect of early reflections may also be observable, as a comb filter effect, in the frequency response of the sound pressure level produced by the loudspeakers at the listening point. In control rooms, the top surface of the mixing desk is a potential source of strong early reflections.

8.2.1.2 Reverberation field

The reverberation field should be sufficiently diffuse over the listening area to avoid perceptible acoustic effects such as flutter echoes.

The nominal reverberation time, $\boldsymbol{T_m}$, should lie in the range:

$$0.2 < \boldsymbol{T_m} < 0.4 \text{ s}$$

(The nominal value, $\boldsymbol{T_m}$, is the average of the measured reverberation times in the 1/3–octave bands from 200 Hz to 4 kHz).

To ensure that the acoustic environment remains “natural”, the value of $\boldsymbol{T_m}$ should increase with the size of the room. The following formula is given as a guide:

$$\boldsymbol{T_m} = 0.25 * (0.001 * V)^{1/3} \text{ [s]}$$

where: $\boldsymbol{V}$ = room volume in cubic meters.

The diagram of the $\boldsymbol{T_m}$ dependence from the room volume is shown in Fig. 8.1. It may be observed that the reverberation time $\boldsymbol{T_m}$ of such small rooms does not exceed 0.4 s.

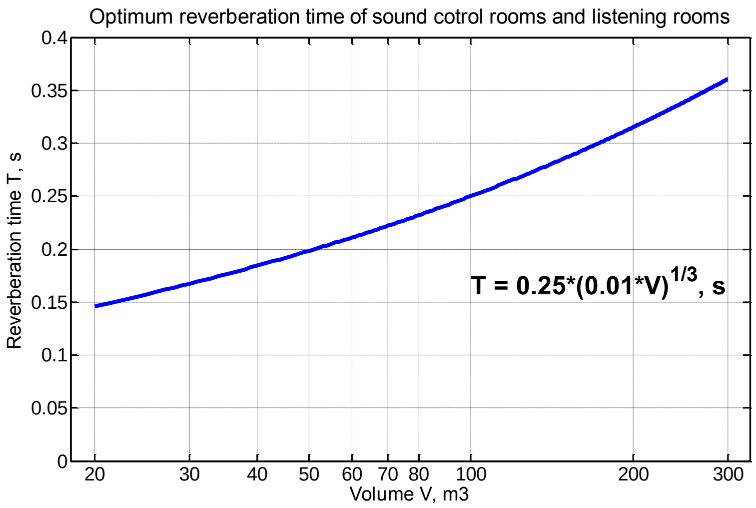


Fig. 8.1 Recommended values of the nominal reverberation time T_m for listening rooms and sound control rooms having volumes 20 m^3 – 300 m^3 [21].

The reverberation time T is frequency–dependent. Measured in 1/3–octave bands over the frequency range from 63 Hz to 8 kHz, should conform to the tolerances shown in Fig. 8.2. In addition, sudden changes in reverberation time with frequency should be avoided and the differences, $\boldsymbol{\Delta T}$, in reverberation times between adjacent 1/3–octave bands should not exceed the following limits:

$\boldsymbol{\Delta T}$ < 0.05 s	for 200Hz ≤ □*f* ≥ □8 kHz
< 25% of longer time	*f* ≤ 200 Hz.

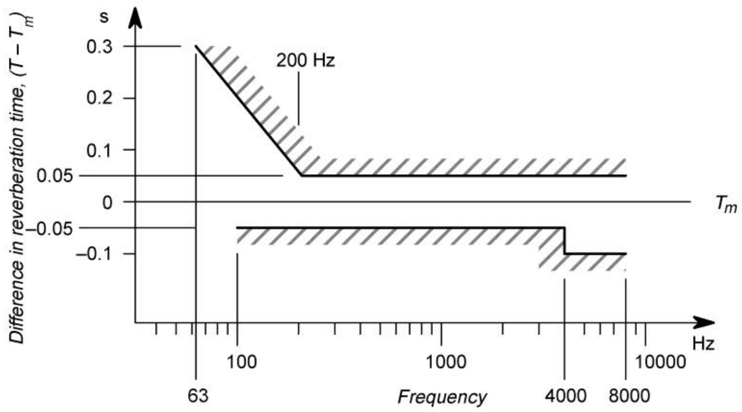


Fig. 8.2. Tolerance limits for reverberation time of listening rooms and sound control rooms

8.2.2 Medium size room acoustics

Sound recording studios, cinema, rooms for music and speech listening and production may be regarded as the medium size rooms with volumes of 100 m3 – 3000 m^3. The values of the optimum reverberation times are in this rooms in the range 0.3 s to 0.8 s [22]. Lower values of T concern smaller sizes of rooms and the highest T – rooms with volumes of 3000 m3. The diagram in Fig.8.3 gives an illustration of this relationship.

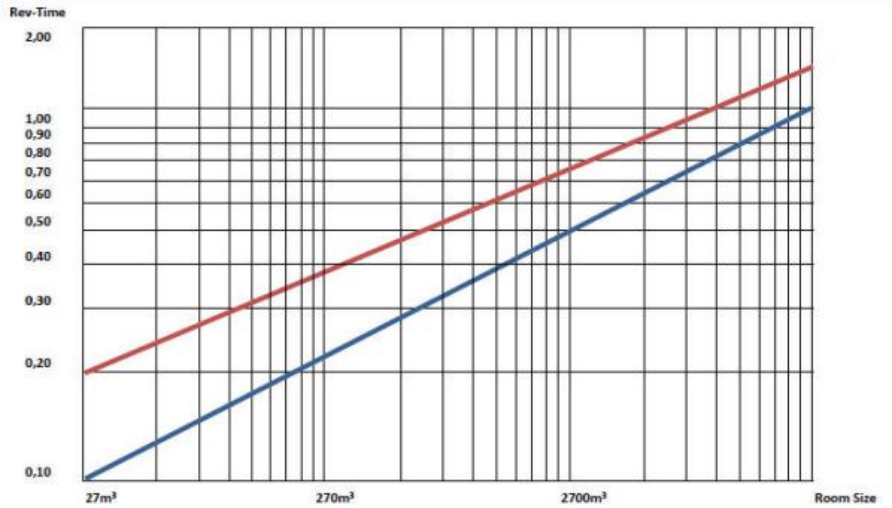


Fig. 8.3. Limits of the reverberation time versus volume of medium size rooms for music and speech listening and production (sound recording studios, cinemas, rooms for music reproduction)

8.2.3 Large performance halls

The basic acoustic criteria for the large performance halls do not differ from the smaller rooms. A large hall must have a low ambient noise level from internal and external sources. It must provide a reasonable level of acoustic gain and appropriate reverberation time. Such artifacts as echo must be avoided. A smart way of the management of the acoustic energy of music sources is required in the large halls.

The seating capacity may be a couple of hundred or several thousand. The nature of the application determines the acoustical priorities, and can also create diverse demands. Different types of music performance spaces, such as concert halls, opera houses, and chamber music halls place particular demands on the design, with different acoustical priorities. For best results, a hall must be tailored for its specific purpose.

Very generally, we may consider the design of large spaces in two aspects: spaces primarily intended for speech, and those primarily intended for music. The former emphasizes speech intelligibility while the latter requires musical sonority. Many halls designed for speech must integrate a sound playback system.

8.2.3.1 Reverberation and Echo Control

Reverberation is an important parameter that helps define the sound quality of an acoustic space. This is perhaps especially true in large halls such as concert halls, performance theaters and auditoriums. Reverberation time T_{60} must fit the intended function of a room and room volume.

Figure 8.4 shows recommended values for mean reverberation time between the octave bandwidths 500 and 1.000 Hz. Halls for speech have shorter mean reverberation times than halls designed for music performance. Also, the recommended mean reverberation time increases as a function of room volume.

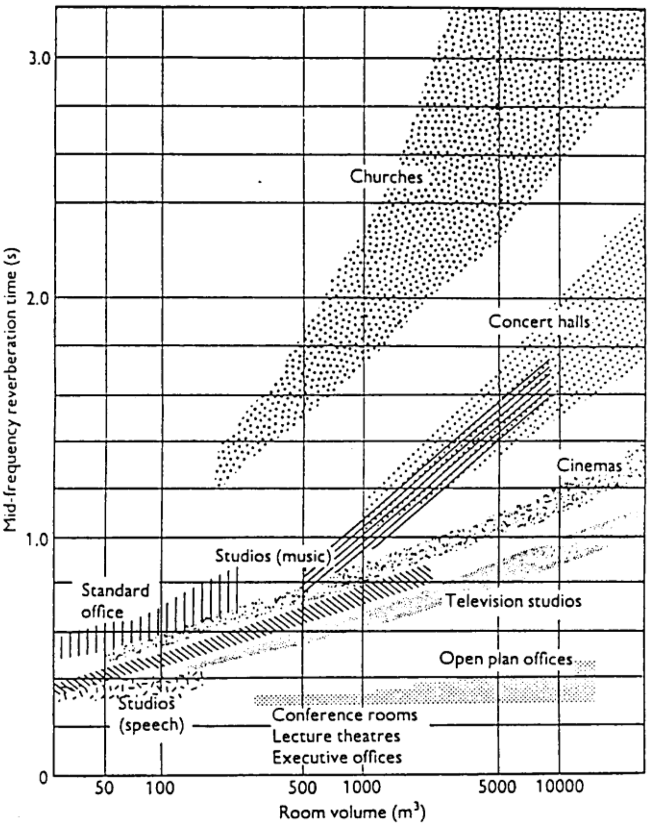


Fig. 8.4 Recommended mean values of reverberation time in octaves 500 - 1000 Hz for speech and music halls

The frequency response of the reverberant field is shown in Fig. 8.5. The tolerance ranges of reverberation time refer to the recommended mean reverberation time. The response of these tolerances for speech (Fig. 8.5 A) differs from that for music (Fig. 8.5 B). The classic music requires significantly longer reverberation times at low frequencies. This subjectively enhances warmth of the sound and is characterized as the bass ratio index BR. In case of speech, reverberation time decreases at low frequencies in order to improve speech intelligibility.

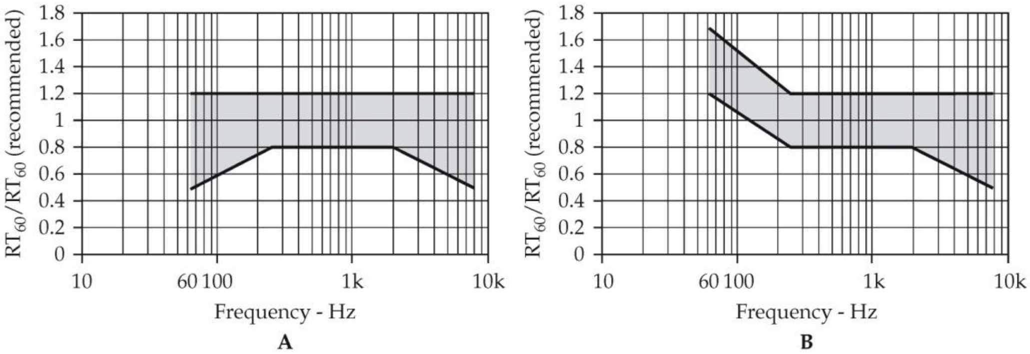


Fig. 8.5. The frequency-dependent tolerance range of the recommended reverberation time. A - Speech. B - Music. [26]

8.2.3.2 Halls Design for Speech

In halls designed for speech applications, many of the same acoustical criteria that are important for any room design will apply. Speech intelligibility is the highest design priority for any hall intended for spoken word.

Early reflections, less delayed than 35 to 50 ms, tend to be integrated with the direct sound, increase perceived loudness and assist intelligibility. Late reflections, greater than 50 ms delay, particularly degrade speech intelligibility.

In halls designed for unamplified speech, it is often necessary to limit the overall room volume. This is because a large volume requires more speech power than a small room. In a very large room, for example, 20.000 m^3 in volume, an unamplified voice cannot be satisfactorily heard with even the best acoustical design. A smaller room will require less absorption for a given reverberation time. This, in turn, allows more reflectivity which supports greater acoustical gain and a louder speech level. In a relatively large auditorium, a room volume ranging from 3 to 6 m^3 per seat is suggested. This volume index of rooms for speech is in contrary to rooms designed for music, where a relatively large volume is desirable.

A rectangular shoebox-type hall is only suitable for a relatively small speech hall. With larger seating capacities, much of the audience is placed away from the stage, at the far end of the hall. This limitation can be overcome to some extent by widening the hall.

In small speech halls, the majority of absorption is provided by the audience. Therefore, the room surfaces can be relatively reflective. In larger halls, where there is greater room volume per seat, relatively greater room absorption is needed.

A reflective front stage area provides strong early reflections that are integrated with the direct sound and enhance it (precedence effect). On the contrary, strong late reflections and reverberation, such as from rear walls, would not be integrated and may produce echoes. Absorption placed rear of the hall may prevent this defect. In some cases, when added absorption is undesirable because of decreased reverberation time, reflective diffusers can be placed on the rear wall.

To preserve speech intelligibility, although early reflections are useful, the overall reverberation time must be short, less than 1 s.

In many large halls, ceiling reflectors are used to direct sound energy from the stage to the seating area. The size of the reflectors determines the range of frequencies that are reflected; the larger the panel, the lower the cutoff frequency of the reflected sound. Both dimensions of a square reflecting panel should be at least five times the wavelength of the lowest frequency to be reflected. Panels must be solid and stiff, and securely mounted to avoid resonances. When ceilings are high, care must be taken to ensure that path-length differences between direct and reflected sound are not too great, and particularly should not exceed 20 msec. In some cases, ceiling reflectors are made absorptive, to avoid late reflections.

A raked floor allows a more direct angle of incidence which in turn allows less absorption. Generally, the slope of an auditorium floor should not be less than 8°. The floor of a lecture hall might have a 15° angle of inclination. Staggering of seats is also recommended.

8.3 Concert Hall Acoustical Design

Symphonic music, chamber music, and opera, each require very different acoustics, as well as size and room functionality. Moreover, different styles of music, such as baroque, classical, and popular have different acoustical requirements. Finally, different music cultures, such as Eastern and Western, require different design criteria.

Very generally, the problem of music hall acoustics may be considered in two parts: early sound and late reverberant sound. Early sound is considered in terms of early reverberation decay time, intimacy, clarity, and lateral spaciousness. Late reverberant sound can be considered as late reverberation decay time, warmth, loudness, and brilliance.

The ear is very sensitive to early reverberation. This is partly because in most music, later reverberation is partly masked by the following music notes. Early reverberation largely defines our subjective impression of the entire reverberation event. Unlike dense late reverberation, early reverberation comprises a relatively few primary reflections. These reflections arrive within the Haas fusion zone and are integrated with the direct sound and reinforce it. This early reverberation affect the clarity of sound. The greater the energy in the early reverberation, the better the clarity. Late reverberation affect our perception of the liveness of sound. More late reverberant energy can increase liveness or fullness. As late reverberant energy increases, the clarity relatively decreases.

In some halls, the reverberation decay has a two-part slope to achieve good clarity, as well as good liveness. The early decay has a steep slope and a short EDT, while the late decay has a shallower slope and a longer T_{60}. For a large concert hall, T_{60} in mid frequencies is about 2.0 s. When there is a single decay slope, then the reverberation time is simply measured as $\boldsymbol{T_{60}}$.

Brilliance is another metric used to quantify hall acoustics. Brilliance describes sound that has presence and clearness. Brilliance is achieved with adequate high-frequency energy from reflecting surfaces. On the other hand, a hall with good brilliance should not sound too bright or harsh. Brilliance can be estimated by comparing a high-frequency EDT to an averaged mid-frequency EDT. Some sources recommend that EDT_{2kHz} / EDT_{mid} should be at least 0.9, and EDT_{4kHz} / EDT_{mid} should be at least 0.8.

A good music hall should provide adequate acoustic gain G_{mid} at all seating positions. $G_{mid,}$ dB, is related to the early decay time EDT_{mid}, averaged at 0.5 - 1 kHz octaves, divided by the cubic volume V, m^3, of the hall.

$$G_{Mid} = 10 \log\left(\frac{EDT_{Mid}}{V}\right) + 44$$

In many concert halls with good acoustics, G_{mid} lies between 4.0 and 5.5 dB. However, in halls with different applications, G_{mid} will vary over a wider range (see table 6.2).

Room volume affects reverberation time and required absorption. In some cases room volume is specified as a volume per listener seat; for example, a concert hall may range from 10 to 12 m^3 per seat.

The early sound field is responsible for a sense of spaciousness so the listener feels enveloped by sound in the hall. A sense of spaciousness can be created by early lateral reflections from side walls, which occur within the 80 ms of the arrival of the direct sound. These reflections should arrive at the listener from either side at angles of approximately 20° to 90° relative to the front of the listener. The geometry of rectangular shoebox-type halls lends itself to lateral reflections. In fan-shaped halls, these reflections arrive more from the listener’s front and the effect of special impression is decreased. Spaciousness is also augmented by providing adequate diffusion; the decorations in many older concert halls accomplish this.

Another acoustical attributes, related to spaciousness are the apparent source width ASW and the listener envelopment LEV. The source appears wider when there is a high level of early lateral reflections, before 80 ms. Listener envelopment, LEV describes the feeling of being surrounded by and immersed in sound that fills a large space. It is improved by late-arrival lateral reflections after 80 ms.

Beranek noted [9] that highly rated concert halls had a well defined initial time-delay gap of about 20 msec. In smaller, more intimate halls, with reflective surfaces closer to the listener, ITDG is small. In larger halls, ITDG is greater for most listeners. ITDG depends on seating position - a listener sitting near a side wall, for example, would experience a smaller ITDG. An ITDG value of less than 15 ms is desirable, measured in the center of the hall. Large halls that are relatively narrow, as in a rectangular shoebox design, can have a relatively small ITDG. In some cases, a small ITDG is achieved by segmenting the audience into smaller areas and placing reflecting walls near them.

A sense of acoustic warmth is desirable in most halls dedicated to the classic music. The subjective warmth is evaluated with the bass ratio BR, which is calculated by summing the reverberation times T at 125 and 250 Hz, and dividing by the sum of T at 0.5 kHz and 1 kHz. An acoustically warm hall with longer bass reverberation times would yield a BR between 1.1 and 1.45.

8.4 Concert Hall Architectural Design

The architectural design of a concert hall requires close collaboration between the architect and the acoustician. The stage and seating area of the hall must meet very specific acoustical requirements for both performing musicians and the audience. A large concert hall must present a space where the pleasure of hearing music is never compromised.

In some halls, a balcony can be used to decrease the distance from the stage to some seating areas, and to provide good sight lines. Care must be taken to avoid acoustic shadowing in the seating areas underneath the balcony. Very generally, the balcony overhang depth should be less than twice the height of the balcony underside. The depth should not be more than the height.

The front of a balcony parapet should be designed to avoid reflections that could affect sound quality in the seating areas in the front of the hall.

Ceiling height is usually determined by the overall room volume that is required. Very generally, ceiling height should be about one-third to two-thirds of the room width; the lower ratio is used for large rooms, and the higher ratio is used for small rooms. A ceiling that is too high may result in a room volume that is too large, and may also create undesirable late reflections. To avoid potential flutter echo, a smooth ceiling should not be parallel to the floor.

In many halls, the ceiling geometry itself is designed to direct sound to the rear of the hall, or to diffuse it throughout the hall, as shown in Fig. 8.6.

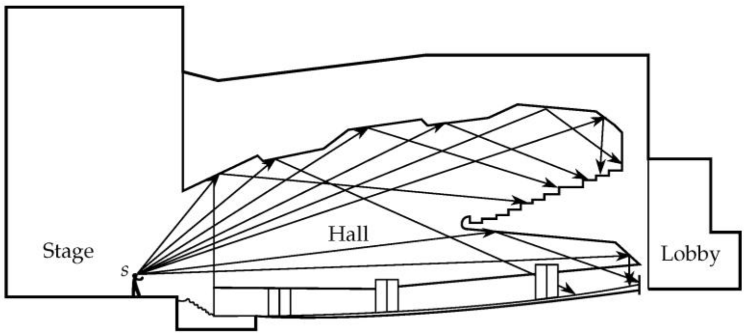


Fig. 8.6. The ceiling geometry directs reflected sound throughout the hall. Several ceiling segments may be angled to reflect sound to particular seating areas [25]

A ceiling may have several segments. For example, the segments of the ceiling near the stage reflects sound to the near rows, farther ceiling segments reflects sound to the rear rows. The rear wall must avoid large concave geometry. Side walls must avoid parallelism. This can be avoided by tilting or splaying wall surfaces. Any surface that unavoidably introduces concave geometry or an undesirable angle should be covered with absorptive material.

In large halls for music or speech a sloping floor improves sight lines, and also improves fidelity in the seating area. On a sloping floor, the listener receives more direct sound than would be available on a flat floor. In either case, the stage should be raised. A sloping floor is desirable in halls where audience sound absorption must be minimized.

The frequency response of sound passing over an audience can be affected by its angle of incidence. At low angles of incidence, a dip, 10 to 15 dB, extending for two octaves around 150 Hz occurs at the audience head position. This effect is most pronounced for direct and early arrival sound. One solution is to design a low-frequency boost in early-reverberation energy by means of strong ceiling reflections. A steep floor slope may also help to avoid that defect.

8.5 Hall Design Procedure

In practice, an acoustician may begin a hall design by carefully considering the site, and determining ambient noise levels and structure layout. Numerical design may begin by specifying a value for G_{mid}. For example, a value of 5 dB may be assumed. Furthermore, a value of T_{mid} or EDT_{mid} is assumed. The choice will depend on the type of music to be performed in the hall. For example, assume a T_{mid} of 2.0 s, which is the optimum value for classic music. Given these values, the hall volume may be calculated, as well as the number of seats and total floor area. If these results do not meet the design criteria, the value of G_{mid} may be adjusted. Using these calculated guidelines, the acoustician and architect may decide on a hall configuration, such as rectangular or fan shaped. Elements such as balconies and terraces can be included, taking into account factors such as ITDG. Absorption is added according to the assumed T and EDT values.

8.6 Case Studies

The design of a large concert hall ranks among the most complex of architectural tasks. Hall design is often unique and demonstrates creativity, and daring, on the part of the architect and acoustician. This physical uniqueness guarantees that each concert hall will have a unique sound character that is unlike any other.

The "Great Hall" (Grosser *Musikvereinssaal*) in Viena.

The *Musikvereinssaal* in Vienna due to its highly regarded acoustics is considered one of the finest concert halls in the world, along with Berlin's Konzerthaus, the Concertgebouw in Amsterdam, and Boston's Symphony Hall. None of these halls was built in the modern era with the application of acoustics science and all share a long, tall, and narrow shoebox shape.

The Musikvereinssaal was inaugurated on 6 January, 1870.

Fig 8.7. View of the famous Musikvereinssaal in Viena

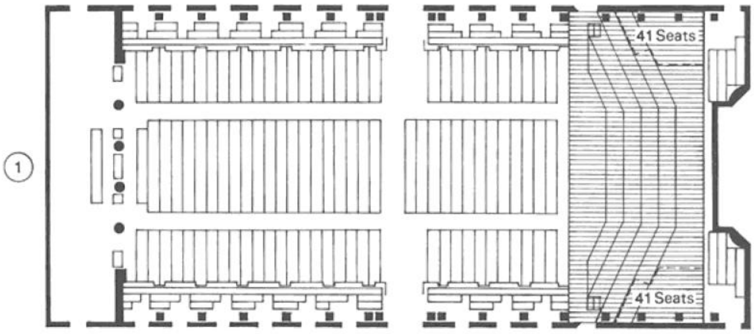
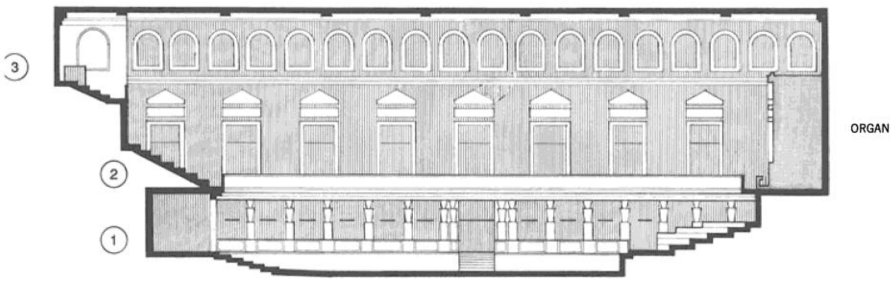


Fig 8.8. Plan of the Musikvereinssaal (from Beranek)



8.9. Section of the Musikvereinssaal

The room's rectangular shape and proportions, its boxes and sculptures allow early and numerous sound reflections. The side walls are made irregular by over forty high windows, twenty doors above the balcony, and thirty-two tall, gilded buxom female statues beneath the balcony. Everywhere are gilt, ornamentation, and statuettes. Less than 15% of the interior surfaces is made of wood. Wood is used only for the doors, for some paneling around the stage, and for trim. The other surfaces are plaster on brick or, on the ceiling and balcony fronts, plaster on wood lath.

The superior acoustics of the hall are due to its rectangular shape, its relatively small size. The Hall is about 49 m long, 19 m wide, and 18 m high. It has 1.744 seats and standing room for 300. Volume 15.000 m^3 and seats 1680. Reverberation time 2.0 s (fully occupied). Stage height: 1 m above floor level. Added absorbing material: 18.6 m2 of draperies over front railings on side loges. Seating: Wood structure on main floor and side balconies, except that tops of seat bottoms are upholstered with 10 cm of cushion covered by porous cloth; rear balcony seats, plywood.

Any hall built with these characteristics would be an excellent hall, especially for symphonic music of the Romantic and Classical periods. The Grosser Musikvereinssaal is similar acoustically to Symphony Hall in Boston. Most critical listeners agree that the clarity or definition is better than in the Amsterdam Concertgebouw. The sound in this hall is much louder than in Boston, and some feel that this is a disadvantage for a touring orchestra which may not be in the habit of restraining itself. Also, it is overly easy for the brass and percussion to dominate the strings. The string and woodwind tone are delicious and the sound is uniform throughout the hall.

Boston Symphony Hall

Symphony Hall, built in 1900, is rectangular in shape with a high, horizontal coffered ceiling and two wraparound balconies. On entering the hall, one encounters two strong architectural features: the stage with its back wall devoted to a row of gilded organ pipes, and the upper walls of the hall with their niches, in front of which stand replicas of Greek and Roman statues.

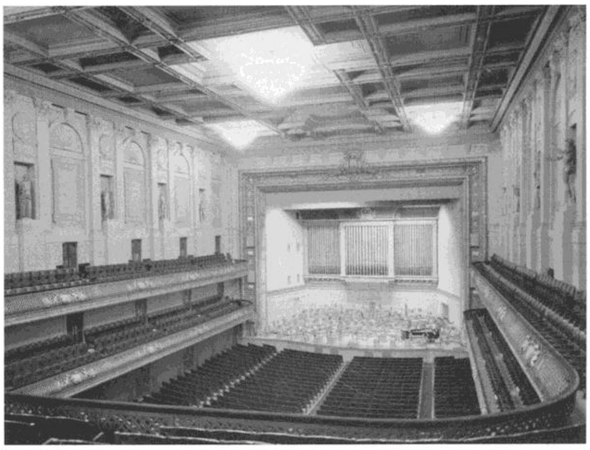
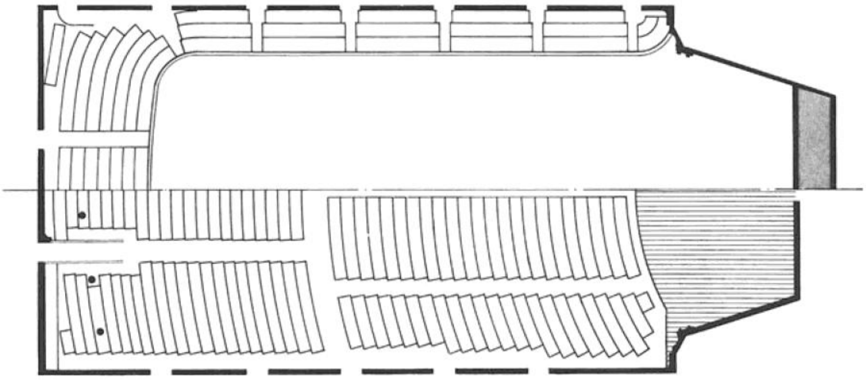


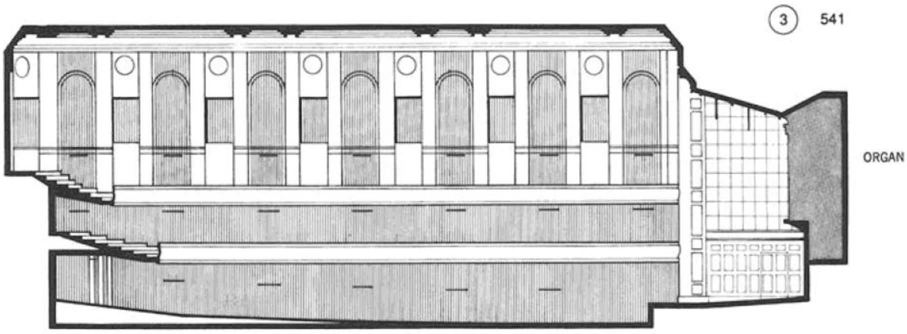
Fig 8.10. View of the Boston Symphony Hall

The combination of shades of gray and cream paint, gilded proscenium frame and balcony fronts, red-plush balcony rails, black leather seats, and red carpets would place this hall architecturally in the middle of the nineteenth century, although it was built fifty years later. In some ways it resembles the Vienna Grosser Musikvereinssaal. Nevertheless, it is different, primarily because it seats 2.625 people compared with 1.680 for the Vereinssaal.

Boston Symphony Hall volume V = 8.750 m^3, height H = 18.6 m, length L = 39 m, width W = 22.9 m, number of seats N = 2.625, V/N = 7.14 m^3, reverberation time T_{occup} = 1.8 s, ITDG = 15 ms. During May, June, and December each year, tables are installed on the main floor for "pops" concerts and the capacity is reduced to 2.369.



8.11. Plan of the Boston Symphony Hall



8.12. Section of the Boston Symphony Hall

According to [9] the sound in Symphony Hall is clear, live, warm, brilliant, and loud, without being overly loud. The hall responds immediately to an orchestra's efforts. The orchestral tone is balanced, and the ensemble is excellent.

Seven crucial design features were responsible for the immediate success of this hall. The shoebox shape, which was modeled after the old Leipzig, Gewandhaus, the proper reverberation time, determined by Sabine's formula, which set the ceiling height; the preservation of bass both by avoiding large areas of wood, which followed the decision to make the building fireproof, and by choice of seats with a minimum of upholstering; limiting its width and length to insure intimacy, providing sound diffusion by niches, statues, and ceiling coffers; and, finally, by a stage house that blends the orchestral sound beautifully and projects the music uniformly throughout the audience.

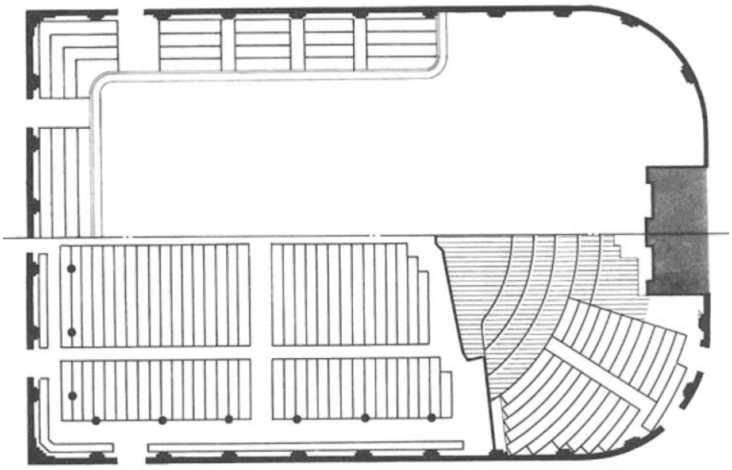
Ceiling: 1.9 cm plaster on metal screen. Walls: 30% plaster on metal lath, 50% on masonry backing, and 20% of 1.25 - 2.5 cm thick wood, including the stage walls, balcony fronts, open-pattern cast iron. Floors: base floor is flat concrete with parquet wood affixed; in winter concert season, sloping floor of 0.75-in. boards on 10 x 10 cm framing members – the airspace beneath varies from zero at the front to 1.52 m at the rear; balcony floors, wood supported above concrete. Carpets: thin, on main aisles. Stage enclosure: wood paneling about 0.5 in. thick, but from the stage floor up to a height of about 4.3 m, is about 1 in. thick. Stage floor: 1.5 in wooden planks over a large airspace with 0.75-in. flooring on top. Stage height: 1.37 m. Seating: The front and rear of the backrests and the top of the seat bottoms are leather over hair; the underseats and the arms are of solid wood.

Amsterdam Concertgebouw

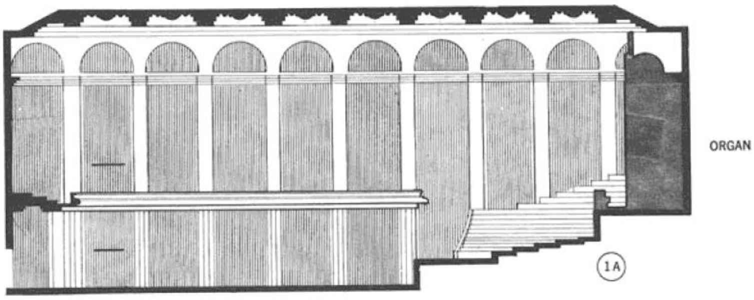
Amsterdam Concertgebouw was opened in 1888 and seats 2.037.

Fig 8.13. View of the Amsterdam Concertgebouw Hall

Several of the physical features of the Concertgebouw are different from Musikvereinssaal and Boston Symphony Hall. It is wider (27.7 m), compared, respectively, to 19.8 and 22.9 m. Height H = 17.1 m, Length L = =26.2 m. Twenty percent of the audience is seated on steep stadium steps behind the orchestra. The stage is higher than that of any hall studied, H = 153 cm. The floor of the Concertgebouw is flat and the seats are removable. The irregular walls and deeply coffered ceiling produce excellent sound diffusion. The reverberation time at mid-frequencies, fully occupied, is about 2 sec.



8.14. Plan of the Concertgebouw Hall



8.15. Section of the Concertgebouw Hall

Opinions of conductors of major orchestras and soloists obtained through the years include, "The Concertgebouw has marvelous acoustics. It is probably one of the best halls in the world." "I went into the audience and heard the Boston Symphony. It sounded excellent acoustically." "The reverberation in this hall gives great help to a violinist”. The reverberation sounds greater than that in other rectangular halls, a quality that generally pleases visiting conductors. The cello in a concerto sounds loud and luxurious. The full orchestra plays with rich tone. The sound in the balcony is better than on the main floor, probably because in those seats the articulation is somewhat better. But on the main floor, for those who love a full sound with rich bass, and the feeling of being completely surrounded in an ocean of music, this hall has no superior.

Finishing: ceiling: 3.8-cm plaster on reeds, coffered and with ornamentation; there are deep "window" recesses around the top edges. Sidewalls and rear walls: Below the balcony, plaster on brick; above the balcony, plaster on reed, which sounds dull or damped when tapped with the fingers.

Floors: 13 cm concrete, on top of which hardwood boards are nailed to 5 x 7.5-cm wooden battens; cavity filled with 4-cm layer of sand. Carpet: On main floor aisles only. Stage enclosure: None. Stage floor: Heavy wood over deep airspace. Stage height: 150 cm. Added absorptive material: 65 m2 of draperies over the front of the little room at the rear of the balcony and around the doorways. Seating: Upholstered in thick, hard-weave material.

Berlin Philharmonie

The Philharmonie in Berlin is an example of large concert hall using a vineyard configuration. In this approach, the large hall is segmented into a number of smaller multilevel audience areas separated by low walls. This contributes lateral reflections to provide intimacy within the larger space. The hall was completed in 1963, and is notable for a tent-like ceiling with reflecting clouds below it. There are 2.218 seats directly behind the orchestra and about 300 on either side. In addition, there are about 120 places on stage and spaces for 44 handicapped listeners. No listener is more than 30 m away from the stage.

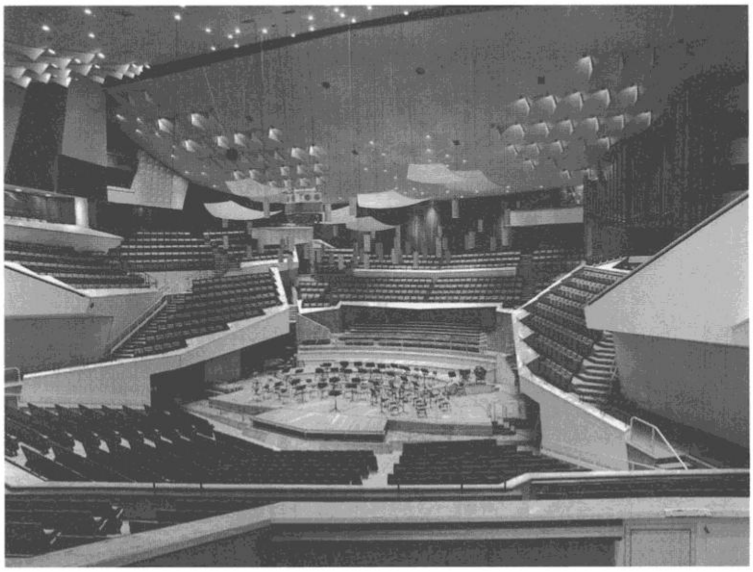
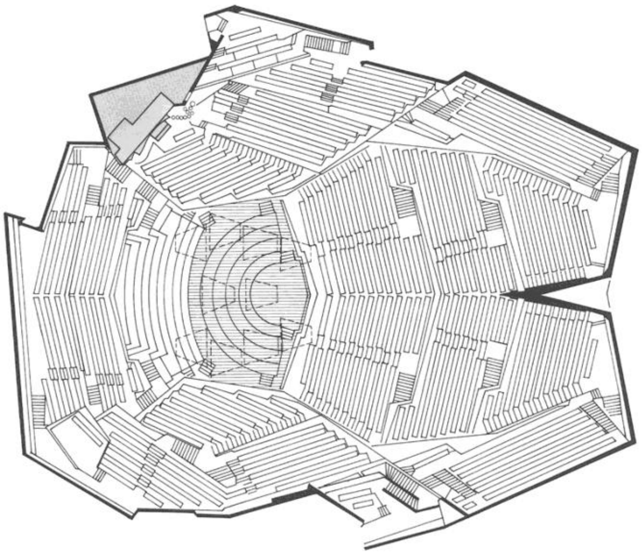
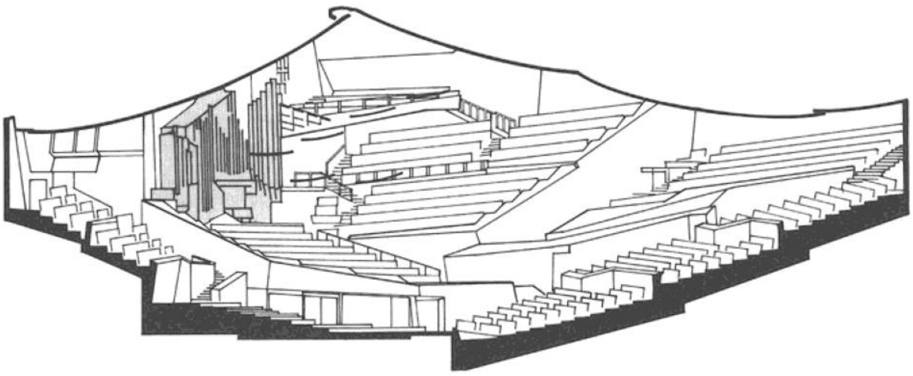


Fig. 8.16. View of the Berlin Philharmonie

Fig. 8.17 shows architectural plan of the Philharmonie, and Fig 8.18 longitudinal section. Fig. 8.19 plots reverberation time of the hall in three states: unoccupied; occupied by orchestral musicians as in a recording session; and fully occupied by orchestral musicians, chorus, and audience. In all three cases, the rise in low-frequency bass is evident.



8.17. Plan of the Berlin Philharmonie



8.18. Section of the Berlin Philharmonie

Philharmonie Hall has become one of the models of successful acoustical designs, pioneering the concept of the "vineyard" style hall. The acoustical consultant agreed on the advantage of breaking the audience into blocks, so that the first row in each block receives unimpeded direct sound. The seats in many of the blocks receive early lateral reflections from the side walls that surround them, including the wall behind. The fronts of the terraced blocks provide early reflections for both the musicians and the audience seated in the middle of the hall.

Additional early reflections are provided to the orchestra and audience by ten large suspended panels hung above the stage. The seats in the upper blocks receive early reflections from the convex, tent-shaped ceiling.

In the audience sections in front of the orchestra, the sound is beautiful, clear, balanced, and with a liveness that completely surrounds one. The principal disadvantage is that those seated to the rear, or near rear, of the stage hear a different sound: the trumpets and trombones radiate forward, and the French horns backward.

The sound from piano and singers is also troubling, a large part of the upper registers are projected forward. Fortunately, the visual impression of viewing the conductor face-on favorably shapes one's judgment of the acoustics. The mid-frequency reverberation time, fully occupied, is 1.9 sec. The bass is controllable by adjusting the 136 pyramid-shaped low-frequency resonators in the ceiling.

Finishing: Ceiling, suspended 3 cm, up to 4 cm at center, chalk-gypsum plaster on expanded metal. Suspended stage panels: 10 trapezoidal polyester panels, each 7.5 m2 in area, 50% open space between, variable in height 10-12 m above the stage. Ceiling sound-absorbing units: 136 pyramidal-shaped, combination sound-diffusing, low-frequency Helmholtz resonator type absorbing boxes. Sidewalls: Part of sidewalls are thin wood over airspace. The parapets are faced with Jurassic limestone plaster. Stage side walls: Arranged to reflect sound back to the musicians. Stage floor: Wooden floor on planks over airspace. Stage height: 76 cm. Audience floor: Oak parquet in asphalt base over precast slabs. Seating: Seat back, molded veneered plywood, the upper part bent vertical so that maximum sound reflection occurs when occupied; cushion on front of seat back does not extend to top; seat bottom is upholstered on top and the underside is covered with cloth only and is perforated; armrests are wooden.

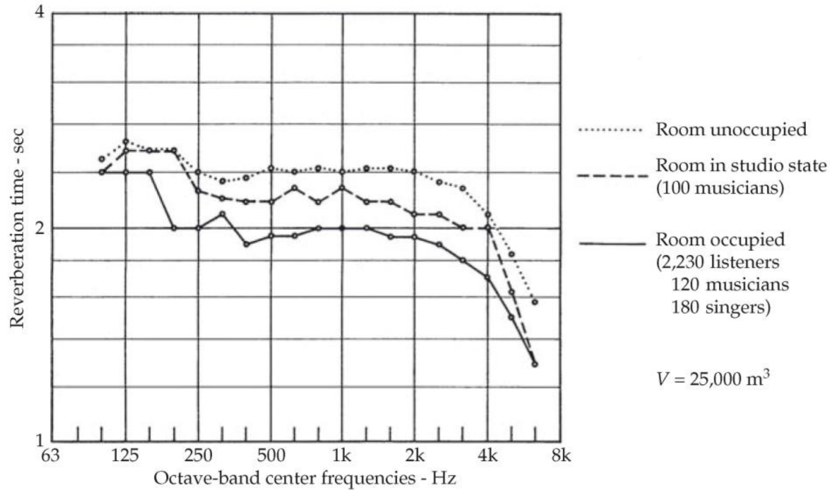


Fig. 8.19. Reverberation time of Berlin Philharmonie. The three curves illustrate how the occupancy affects the reverberation time of the hall [25]

8.7 Acoustics for Pop and Rock Music

Following analysis and design recommendations are taken from the publications of N. W. Adelman-Larsen, former rock and jazz drummer, who did the most extensive research on the acoustics in venues with amplified music. Adelman-Larsen published since 2005 several scientific papers on that topic and released in 2014 the book “Rock and Pop Venues Acoustic and Architectural Design”.

8.7.1 Acoustical demands for Pop and Rock Music

Pop and rock concerts are unique by the fact that they are depending on amplification of the sound that the band produces. A significant ingredient in the music is a highly amplified and well-articulated bass line supported by a more or less syncopated, staccato of nature, bass drum rhythm. The precise timing of both, down to few hundreds of a second, is satisfactory for the orchestra itself and to the audience.

The room’s acoustics should enable for the transmission of the clear and intelligible sound from PA microphones and loudspeakers to the audience. If the low frequency sound is not very well controlled and allowed to last too long a reverberant sound will mask the bass and higher frequency sound, significantly reducing the understanding of the content of the music.

The musicians on stage use open monitor speakers or in-ear monitoring systems to hear themselves and their colleagues. The audience listens to the music mainly through PA speakers operated by a sound engineer. These two sound systems depend on each other to some extend and on the acoustics in the two environments of stage and audience area.

In pop and rock music there is a much higher level in the 63-Hz band compared to classical music. In halls for amplified music, the reverberation time in the 63-Hz octave band may have higher value than in the 125-Hz band.

It was also found that the acoustics on stage must be similar to that of the audience area. Sound engineers especially (definitely no musicians) want the stage dead to easily handle feedback and the like.

It is pointed out that the higher value of reverberation time at low frequencies proposed here which some people find beneficial for classical music is the worst enemy of the acoustics for pop and rock music.

8.7.2 Musicians' preferences

It is important to note that musicians' taste regarding acoustics is not the same. Some instrumental groups want a more reflective hall than other groups, and there certainly is a degree of personal taste involved.

The measurements of the twenty halls for rock and pop music and interviews with musicians proved that musicians need halls not to be too acoustically dead and not too lively either. The frustrating for musicians is hearing the music reflected from the audience area loud compared to the earlier reflections from the stage surroundings including their own direct sound and that of the monitors. It gives a distancing sensation - the musician feels detached from his or her own playing and thereby disengaged from the situation.

There is a need for the early sound being enveloping for the musicians who often move around the stage and this calls for some early reflections from the stage surroundings. Without reflections from the stage area, even with a complete monitor set-up, the musician will not experience a sensation of being enveloped in his own and his colleagues' sound. With too much sound coming back from the hall he certainly will feel enveloped in sound but, too-strong late reflections will make the musician feel disengaged from his playing and will tend to affect his timing. Of course musicians experience these defects frequently and they cope with them by being somewhat conscious about the sound and navigate accordingly to get timing correct.

That does not make defects acceptable or recommended. On the contrary, if both stage and hall are too dead, small and natural timing differences between the musicians become very clear which can lead to uncertainty and for them to lose confidence. So it is seen that, according to musicians, the acoustics on stage mustn't be dead compared to those in the hall, and the acoustics in the hall must be neither too dead nor too lively.

8.7.3 Sound Engineer' preferences

Sound engineers are responsible for the sound during concerts. The sound engineer is placed in the audience area and therefore has perfect possibilities for knowing what sound impression the audience perceives. If the combination of hall and PA system does not add much reverberation, the engineer has a lot of freedom in playing with artificial effects.

It seems that sound engineers can roughly be divided into two categories: those who want the hall to give some envelopment, as is the preference of the musicians, and those who like more control over their outboard effects to be added to the mix.

Many sound engineers like the stage to be quite dead. This prevents sound from instruments and monitor speakers from being reflected to the audience area or to leak into open microphones on stage. The stage reflections entering the open microphones on stage are delayed and possibly out-of-phase with the direct signal. These reflections harm the total mix. Much in the same way regarding the audience area, a relatively dead hall will make the sound engineer capable of forming a sound experience to his taste on the PA system.

Some sound engineers find these conditions on stage as well as in the hall a little too reverberant in order to create the perfect sound they had in mind. These slightly reverberant conditions make their job a little harder if the band sounds harsh or unprofessional.

8.7.4 Recommended acoustical attributes for Pop and Rock Music venues

8.7.4.1 Recommended Reverberation Time for a given hall volume

Solid line in Fig. 8.20 shows recommended reverberation time T_{30} for an empty hall at various hall volumes. The dotted line is for the average including the 63 Hz band. The line is linear in the small interval from 1.000–7.000 m^3, but certainly cannot be extrapolated linearly to larger volumes. Applying a logarithmic scale on the *x*-axis over large volumes, recommended T_{30} would approach a straight line.

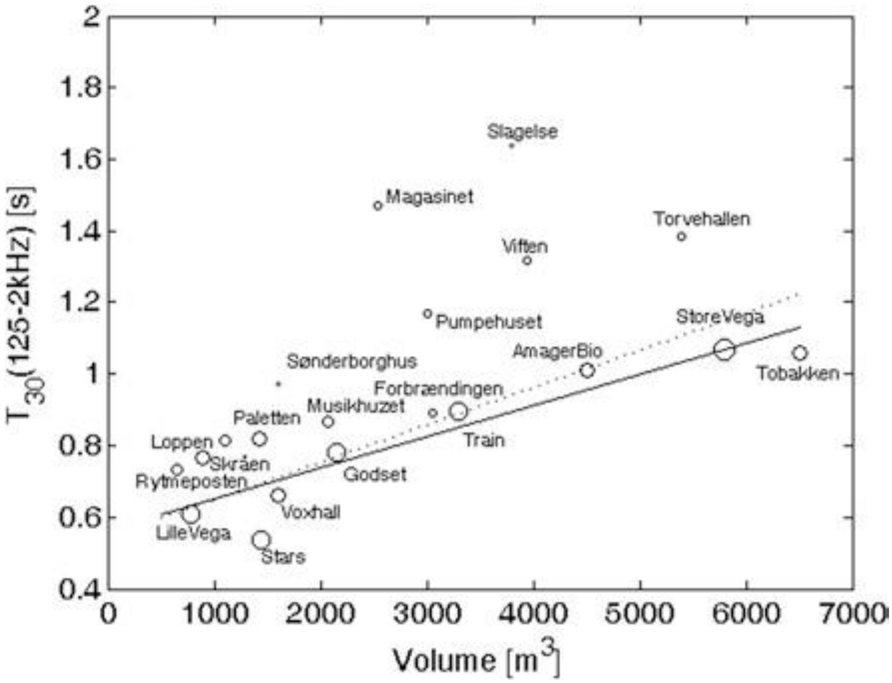


Fig. 8.20 Solid line shows recommended reverberation time T_{30} for an empty hall at various hall volumes. The dotted line is for the average including the 63 Hz band. The line is linear in the small interval from 1,000–7,000 m3. Applying a logarithmic scale on the *x*-axis over large volumes, recommended reverberation time would approach a straight line [11]

8.7.4.2 Reverberation time frequency response

Fig. 8.21 shows the recommended frequency response of the reverberation time for the rock and pop venues. Approximate factors of T_{30} in the octave bands 63 Hz – 4 kHz is scaled on the y axis. Factor 1 refers to the relevant values in Fig. 8.20.

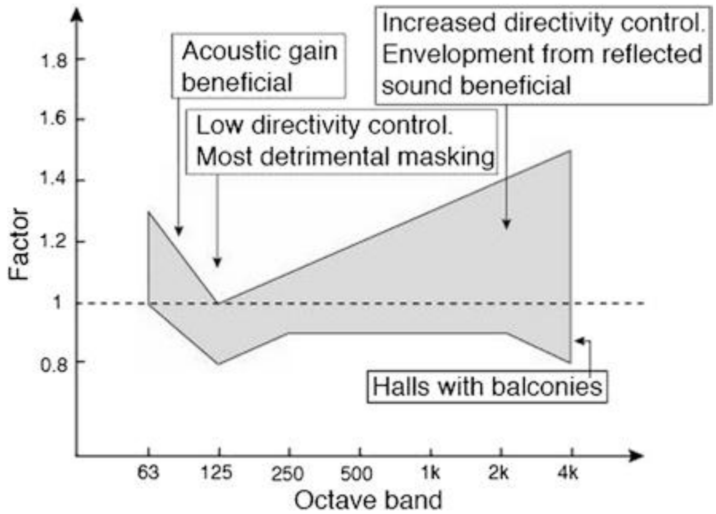


Fig. 8.21 Approximate factors of T_{30} in the octave bands 63 Hz –4 kHz. Factor 1 refers to the relevant value in Fig. 8.20

The following recommendations for empty halls of volumes between 1.000 and 7.000 m^3 has been found from the acoustical investigation of the rock and pop venues:

a) T_{30} in the 125 Hz octave band should be in accordance with Fig. 8.20. This octave band is extremely dominant in the acoustics for the rock and pop venues.
b) At higher frequencies, T_{30} can be higher according to Fig. 8.21. This is due to the high degree of absorption provided by the audience and the air, and due to higher directivity of loudspeakers at higher frequencies. It is also a fact that at amplified concerts usually artificial reverberation is added to these frequencies by the sound engineer partly to compensate for little natural hall reverberation.
c) Acceptable tolerances for the factor of T_{30} in the 63 Hz band are as follows:

50 Hz: 1.8; 63 Hz: 1.4; 80 Hz: 1.2 times the recommended value at 125 Hz. These tolerances are particularly acceptable if there is a similar increase of RT at higher frequencies that will help balance the 63 Hz band rise. The reasons why a higher value of T_{30} in this octave band is acceptable is partly that the masking effect here is less broad and that the A-weighted sound level in pop and rock music is usually somewhat lower compared to the 125 Hz band. It is also seen that a sound decay in the 63 Hz band becomes less audible to humans sooner than a decay in the 125 Hz band because the higher threshold in quiet at 63 Hz.

d) Tolerances lower than a factor of 1 from 125 Hz and up are to be used in halls with large balcony areas. It is acceptable here to place absorption material in the ceiling areas underneath the balcony whereby the RT will drop to lower levels.

8.7.4.3 Suitable Reverberation Times in Larger Halls and Arenas

Based on the knowledge that reverberation time RT in the 125 Hz octave band is the most critical parameter for the acoustic quality of a venue for pop and rock music, a graph of suitable RT over a greater span of volume stretching beyond 7.000 m^3 has been made (Fig. 8.22). This recommendation shall be regarded as the best estimate.

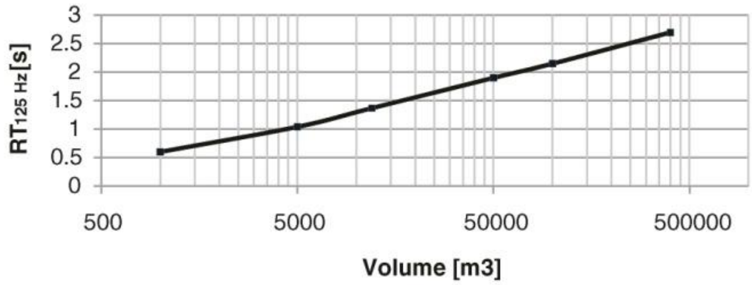


Fig. 8.22. Best estimate of recommendable values of reverberation time RT in the 125 Hz octave band for larger halls and arenas in empty halls for pop and rock music

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