Chapter 5
Foundations of Room Acoustics

5.1 Reflection and Refraction

5.1.1 Reflection from a Flat Surface

Inasmuch as sound propagates in a straight line in a homogeneous medium, as is the case for air at rest at uniform temperature, the presentation of a sound ray, which connects the sound source and the observer in a straight line, is well suited for a number of acoustical considerations. It forms a visual foundation for a geometrical point of view of sound propagation in a room.

When a sound ray impinges on a sufficiently large flat surface, it is reflected. The well-known law of optics, that the angle of incidence is equal to the angle of reflection, goes into effect. This is a phenomenon which has its cause in the wave nature of sound or light respectively. Furthermore, the incident ray, reflected ray and a line perpendicular to the wall at the contact point lie in the same plane. This reflection process is represented schematically in Fig. 5.1a. This illustrates clearly that the angle between the incident and reflected ray depends on the incident angle, thus, by appropriate orientation of the surface, the incoming ray can be reflected into any desired direction.

When two walls are perpendicular to each other the incident sound is reflected twice. It always leaves the corner in the direction exactly opposite to the incident direction. This case is illustrated in Fig. 5.1b. From the two indicated ray paths it is noted that, in contrast to simple reflection from one plane, the direction of the reflected ray depends only on the direction of incidence of the incoming sound.

If two walls in contrast, form an obtuse angle, a single reflection results for steep incidence. For shallow incidence, double reflection results in the corner. Both possibilities are represented in Fig. 5.1c for the case that the sound initially impinges on the lower wall. The second reflection of the sound ray represented by the broken line leads to a direction which is relatively flat in relation to the left wall. When considering, that this secondary reflection can only become steeper when the incident ray on the right becomes even shallower one recognizes that there is a limiting case for double reflection when the incoming sound ray is parallel to

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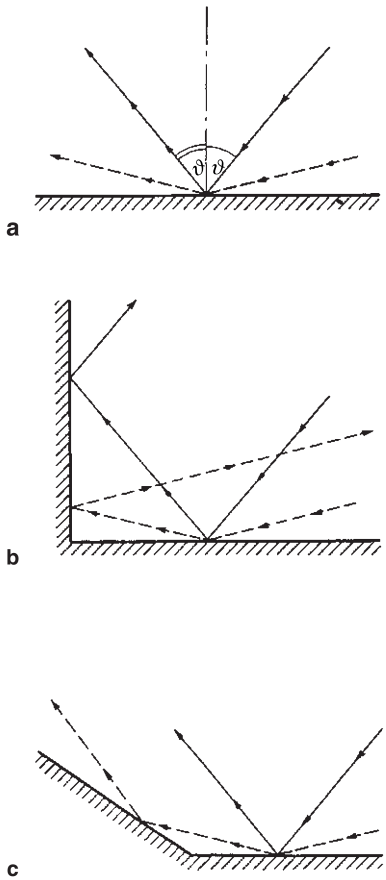


Fig. 5.1 Reflections from a flat surface

the wall. The consequence is that for an obtuse corner, in a certain region near the angular bisector, there are no secondary reflections.

5.1.2 Reflection from Curved Surfaces

When sound falls on a large wall with a curved surface, it will be reflected according to the angle of incidence of the ray, in the plane tangential to the curve at the point of contact, where again the incidence ray, the reflected ray, and the normal to the tangential plane, all lie in the same plane. Depending on the distance of the sound source from the wall and the radius of curvature, focusing or spreading effects can occur in analogy to optical curved mirrors. The most important cases are represented in Fig. 5.2:

(a) If the distance from the sound source to the wall is greater than half the radius of curvature, a focusing point beyond the center of curvature is formed. If the

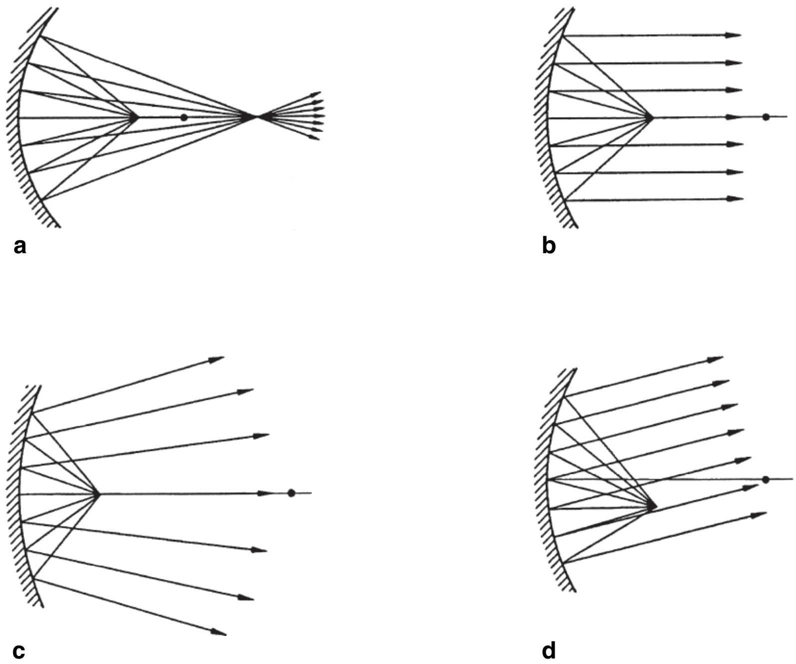


Fig. 5.2 Reflection from curved surfaces

sound source, as drawn, lies between the center of curvature and the wall, then the point of focus is located at a greater distance. Such sound energy concentration is perceived as detrimental when the focusing point falls within a region of the audience. It is, however, perceived as advantageous when the sound is to be focused onto a microphone by a ellipsoidal mirror (for example, such an arrangement has found long time use in the "Marktkirche" in Hannover, prior to the development of technically superior micro-port arrangements).

(b) When the distance of the sound source to the wall is equal to half the radius of curvature, a parallel ray bundle in the direction of the axis of the reflector is generated. A parabolic mirror is the most favorable shape to take advantage of this effect. Fig. 5.2d, however, can also be interpreted in the inverse direction: When a parallel ray bundle, i.e., sound from a far distance source falls on a hollow curve, a focal point is created at a distance corresponding to half the radius of curvature.

(c) When the source to wall distance is smaller than half the radius of curvature, the reflected sound rays spread as though they came from a single point behind the wall. Such a broadening of the sound field can occasionally be advantageous for uniform energy distribution over a certain angular region. Similarly, convex curvatures can lead to a fanning of an incoming ray bundle.

(d) When the sound source is not located on the axis of the reflector, then, for a parabolic mirror, for a source to wall distance equal to half the radius of curvature, a nearly parallel ray bundle results which, however, according to the angle of incidence at the center of the mirror is directed at an angle with respect to the axis. In similar fashion, the focusing point is shifted for a sideways repositioning of the sound source in case (a) to the opposite side, and in addition, it will lose sharpness.

In summary, one can already conclude from these examples that for the positioning for a large number of musicians in front of a concave wall, the danger exists that individual instrument groups will be reflected into different directions, or that for different locations in the hall, their intensity could stand out of the ensemble sound because of the focusing effect.

5.1.3 Influence of the Wavelength

In considering reflection processes so far, the assumption was made that the surfaces impacted by the sound were sufficiently large. Relevant dimensions must be related to the magnitude of the wavelength. The conditions for forming reflections in analogy to optics can be stated more precisely by requiring that the dimensions of sound obstacles must be at least several wavelengths. Otherwise, the sound is bent around the obstacle and hardly reflected. Thus, a wall of dimensions comparable to three wavelengths, presents only a short sound shadow and elements of the size of one wavelength practically do not disturb the sound field at all. Inasmuch as the relationship of the geometric dimensions relative to wavelength depends on frequency, the acoustic effect of reflectors or sound barriers changes with pitch. Low frequency contributions, i.e., components with long wavelengths, can still be heard strongly even when the sound source is hidden from view. High tones, however, are even reflected by small objects.

The shadowing effect of obstacles between sound source and listener can be quantitatively determined on the basis of Fig. 5.3. If a wall is located between sound source and listener, as for example the partition of the orchestra pit in the opera house, then, on the one hand, the angle with which the sound ray is bent at the edge is important. On the other hand, the effective height with reference to the straight-line connection between sound source and listener plays a role, more exactly, the relation of this height to the wavelength. It is noteworthy that already for an angle of 0°, i.e., a just possible sight connection, an attenuation of from 5 to 6 dB occurs. This is understandable when one considers that by the diffraction a part of the sound energy is bent into those regions behind the barrier which would not receive any sound energy without refraction. It is further noteworthy, that the degree of shadowing even for small angles varies strongly, that however, between 30 and 90°, there is relatively little change.

Inasmuch as in practice, frequently individual free-standing or hung reflectors are used, the question naturally becomes of interest: Above which frequencies do

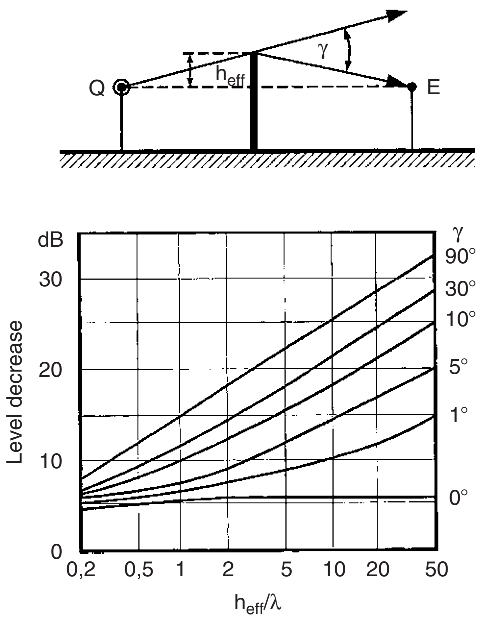


Fig. 5.3 Level decrease for diffraction of direct sound around a barrier (after Redfearn, 1940). S Sound source, R Receiver, λ wavelength of the sound

they reflect effectively? As sketched in Fig. 5.4, if the distance from the sound source to the middle of the reflector is designated as a_1, the distance from listener as a_2, the width of the reflector in the observed plane with b and the angle of incidence as θ, then wave theoretical considerations concerning the effectiveness of reflectors give a lower frequency limit as

$$f_u = \frac{2c}{(\mathrm{b}\ .\ \cos\,\theta)^2} \cdot \frac{\mathrm{a}_1\ .\ \mathrm{a}_2}{\mathrm{a}_1 + \mathrm{a}_2}$$

In this equation, c is the speed of sound in air. Furthermore, the condition must be satisfied that the distances a_1 and a_2 are larger than the reflector width b (Cremer, 1953). Below this limiting frequency, the level of reflective sound drops with 6 dB per octave (Rindel, 1992).

This formula indicates that the regions of reflector effectiveness reaches increasingly lower frequencies as:

Reflector size increases
Distance to the sound source decreases
Distance to the listener decreases
The sound falls more steeply onto the reflector

This means that seats located far back in the hall will receive fewer low frequency contributions than the front rows when all reflectors are of equal size.

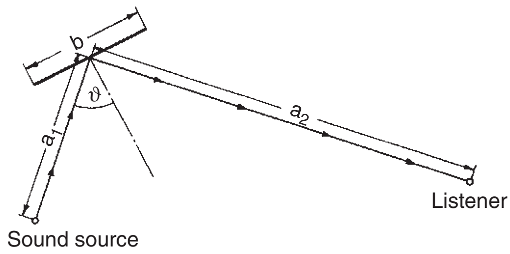


Fig. 5.4 Parameters to calculate the effectiveness region of a reflector

It also results in the requirement to increase the size of those reflectors which are oriented to reach the back of the hall.

In order to simplify the practical application of this equation, Fig. 5.5 presents graphical solutions for the most important variation regions of individual dimensions. If for example, the two distances $a_1 = 10$ m and $a_2 = 20$ m are given, then the lower diagram gives a value of 6.7 for the second term of the formula. From the upper diagram, for a reflector width of $b = 1.5$ m and perpendicular sound incidence ($\theta = 0°$), an upper frequency of about 2,000 Hz is relevant, for a sound incidence of 45°, however, approximately 4,000 Hz. In contrast, if a reflector of 2×2 m size is located at a 2 m distance behind the player, the effective region is increased to include frequencies below 300 Hz.

Aside from the size of the reflector, its mass also plays a role for its effectiveness. Plates (or foils) which are too light transmit a portion of the sound energy, or they themselves vibrate too strongly. A desirable area weight of at least 10 kg m^{-2} for reflectors of mid and high-frequencies can be used as an initial point of reference, this corresponds approximately to a 12 mm wood plate. This requirement is especially relevant for reflectors of speech and singing. If frequencies of the bass region also need to be reflected, approximately 40 kg m^{-2} are required.

The previous considerations dealing with reflection relationships were relevant for smooth, i.e., unstructured reflection surfaces. Frequently, however, wall and ceiling surfaces are structured by small protruding or receding surfaces or added profiles. Such surfaces can have a scattering effect, i.e., instead of the previously described geometric reflection of the sound waves (in one direction), the sound energy is reflected diffusely in all directions. This effect is most strongly pronounced when the depth of the structure is of the order of magnitude of one-fourth to one-half of a wavelength. The frequency region for diffuse reflection can be broadened by a differential depth arrangement of the wall structure. Staggering, according to principles of certain probabilistic sequences of whole numbers, according to elementary number theory, or so-called maximal sequences, are particularly advantageous (Schroeder, 1979).

Below the frequency regions for diffuse reflection, i.e., for larger wavelengths, the structured surface behaves like a smooth wall; i.e., geometric reflection occurs. Above the frequency regions for diffuse reflection, the individual surfaces of the structure act as geometric reflection surfaces and angular mirrors. This can certainly result in reflection directions other than those corresponding to the main orientation

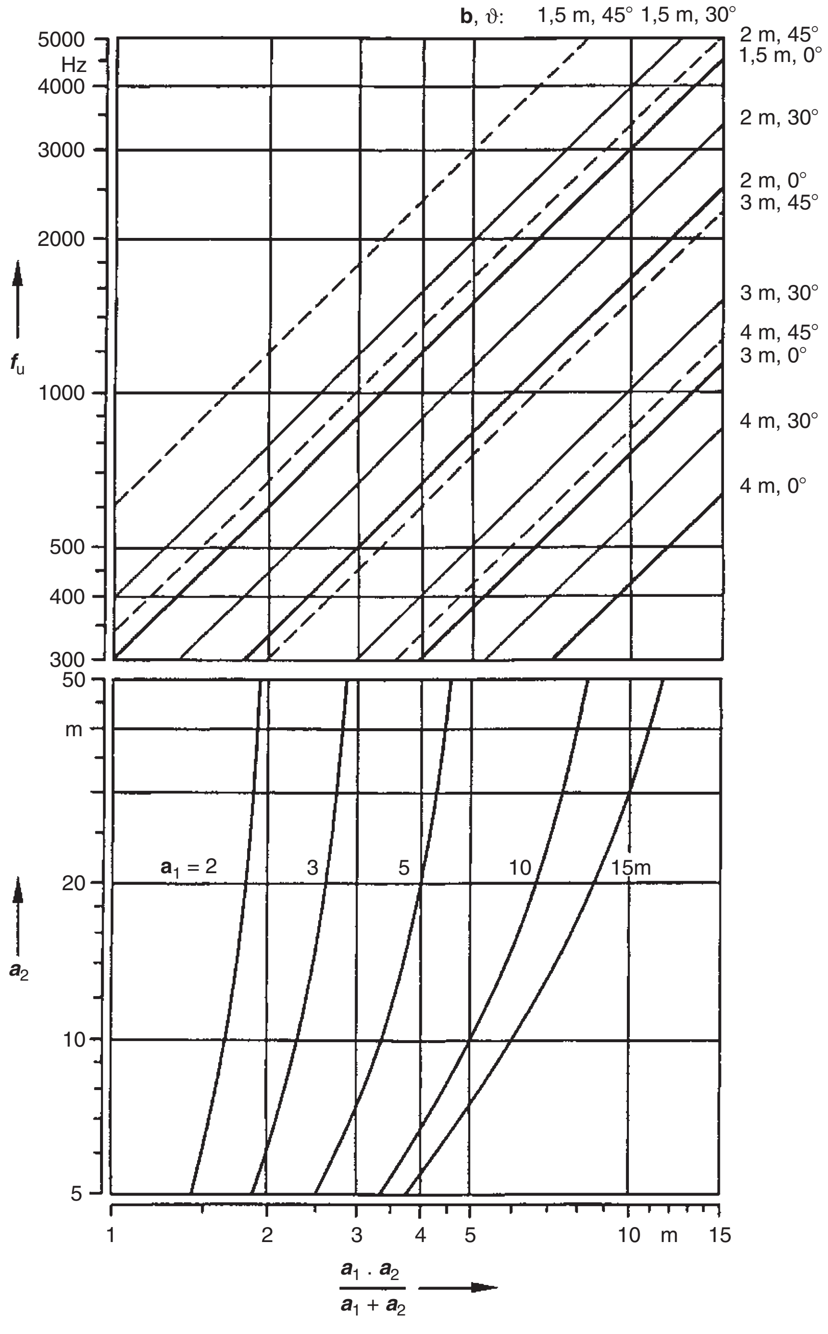


Fig. 5.5 Diagrams for the determination of the lower frequency limit of a reflector. Values for f_u, corresponding to given values for the reflector width (b) and the angle of incidence (θ), are obtained by starting from a given distance a_2 at the left edge on the bottom diagram and going to the curve of known distance a_1, and proceeding vertically upward from that point of intersection to the straight line on the upper diagram corresponding to the given values of b and θ. f_u can then be read from the left edge of the upper diagram

of the wall. Strongly structured surfaces can lead to differing tone colorations of the sound reflected in different directions because of this triple division of the entire frequency region.

For regularly stepped structures, an additional impulse sequence may be formed in certain directions when the higher frequency reflections from the individual steps arrive in regularly spaces sequences. A unique pitch is perceived (mostly in the middle register) corresponding to the frequency for which half a wavelength corresponds to depths of the individual steps.

5.2 Absorption

In the previous section, when considering reflection processes, the only questions considered dealt with directions of incident and reflected sound for different spatial circumstances. Not considered were the amplitude relationships between incident and reflected sound. As in the case of optics, previously cited for comparison reasons, where dark surfaces only reflect a small amount of light, so it is in acoustics, where the portion of reflected energy differs depending on the nature and composition of the walls.

Generally the percentage of energy absorbed during reflection, is indicated when describing material or construction characteristics. This quantity is designated as the "absorption coefficient" and is usually specified by the letter α. On the other hand the "equivalent absorption area" A refers to the sound absorption of a surface of specific size or a room (as the sum of all its surfaces). The following relationship is valid

$$A = \alpha \cdot S$$

where S represents the size of the surface with absorption coefficient α. An important property of the absorption coefficient is its frequency dependence. Sound contributions of different pitch are generally not absorbed or reflected in equal strength by the same material.

There are, for example walls, which by reason of the porous structure only absorb high frequency components, while the low components are nearly totally reflected. Such materials are therefore called high frequency absorbers. The typical frequency dependence of the absorption coefficient for such a case is schematically represented on the left of Fig. 5.6. The height of the absorption coefficient as well as the location of the frequency limit above which significant sound absorption occurs, depends on the composition and thickness of the porous layer: as this layer becomes thinner the frequency limit moves upward.

The audience in a hall can essentially be considered as a high frequency absorber. Thus the audience absorbs all sound contributions from approximately 500 Hz on upwards. In this, the surface occupied by the audience is a determining factor, while the seating density is only of subordinate significance. This means that the same number of persons affect a higher degree of absorption when distributed over a larger surface. For example, 200 persons distributed over an area of 100 m^2

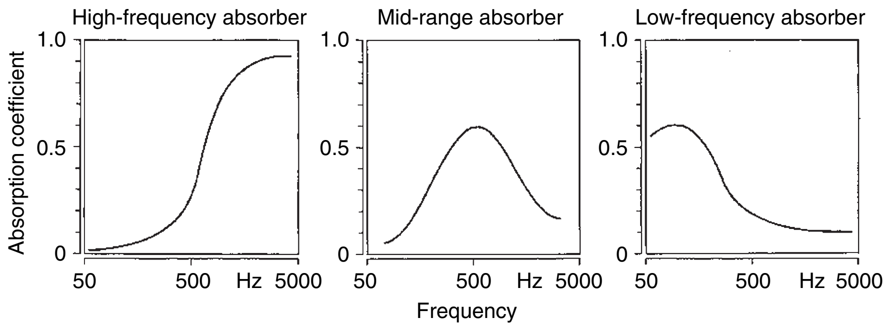


Fig. 5.6 Degree of absorption of materials for differing frequency dependence

possess an equivalent absorption area of 95 m^2 at a frequency of 1,000 Hz. However when they are distributed over 200 or even 300 m^2 their equivalent absorption area rises to above 140 or 165 m^2 respectively.

In contrast, the diagram on the right shows the typical behavior of a low frequency absorber which predominantly absorbs the low tone contributions and reflects the high ones. This effect comes about when plates can vibrate in front of a hollow space as is most often the case for wood paneling for example. As the hollow space becomes deeper and the plate becomes heavier the frequency region of maximum absorption moves toward lower frequencies. For very thin plates and hollow spaces of shallow depth, and above all for constructions where the hollow space opens into the room through slits or holes, the absorption maximum is shifted toward the region of intermediate frequencies, one therefore speaks of midfrequency absorbers. The effective frequency region can thus also be influenced by construction techniques.

There are virtually no materials with frequency independent absorption. The only ones worth mentioning are the "sound – hard" materials such as concrete, marble, or plastered stone walls, which reflect sound of all pitch regions with nearly no attenuation. In this context, however, it should be mentioned that an organ, in reference to its front surface, provides an absorption coefficient between 0.55 and 0.60 in the entire frequency range from 125 to 4,000 Hz. Deviations from this, determined by construction differences are relevant only for low frequencies (Meyer, 1976; Graner, 1988). In concert halls or radio studios it can be meaningful, in individual cases to consider covering the organ while not in use when a reflecting surface rather than an absorbing one is desirable for the orchestra sound at the relevant location.

In practice, curtains are a particularly interesting case, because they can be used without structural changes, or as a temporary proviso. As a porous material, they absorb, as do carpets, preferentially high frequency contributions. However, their absorption regions can be broadened to include lower frequencies when suspended at certain distances from the wall. A somewhat uniform absorption results above the frequency for which the wall distance amounts to one-fourth of the wavelength. From this condition the following formula can be derived

$$f_u = 8{,}500/d \quad f_u \text{ in Hz, } d \text{ in cm},$$

where f_u is the limiting frequency and d is the wall distance (Cremer, 1961). However the material can not be too light, furthermore the curtain should be hung with folds. As an example, for wall distance of 10 cm a limiting frequency of 850 Hz results, and for a distance of 25 cm this is already 340 Hz.

5.3 Reverberation

The geometric viewpoint of the sound-ray path between source and listener naturally has to be limited to the direct path, as well as to detours with only a few reflections, because otherwise the process becomes too cumbersome and visually complex. In order to include all reflection processes to the point of complete absorption, and at the same time describe, if possible, relationships at all points of the room, statistical methods become necessary. For consideration of sound processes after turning off the sound source, it becomes particularly advantageous to include all reflections: The sound reflections arrive at the listener in an increasingly dense time sequence and thus they shape the reverberation in slowly decreasing intensity.

A typical level progress is represented in Fig. 5.7 for such a case. After turning the sound source off, the level (in logarithmic dB scale) decreases approximately linearly with small variations, until it merges with the ambient noise level in the room. In that context it is irrelevant for the basic shape of the curve whether the concern is with the stopping of an electroacoustic sound source or the termination of an orchestral chord. The listener can follow this reverberation until it is submerged in the noise level. In the graphically represented example this

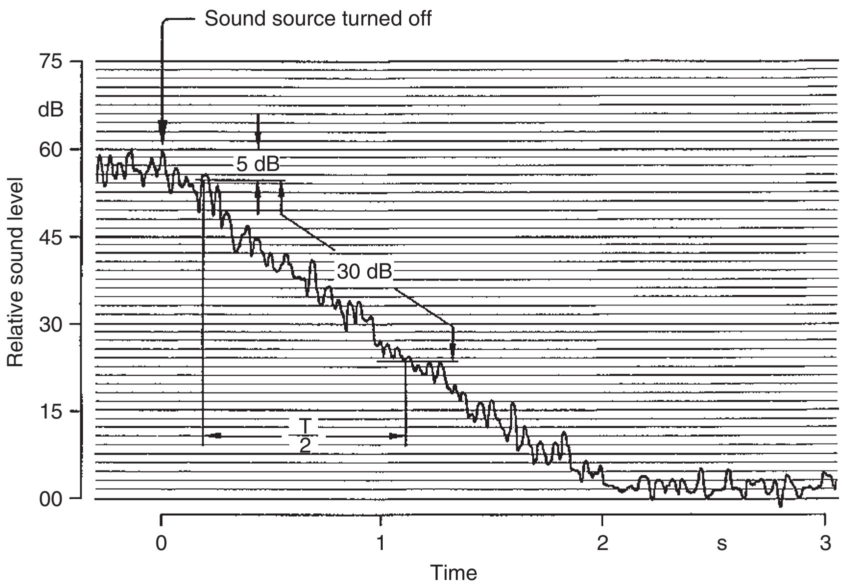


Fig. 5.7 Sound level during a reverberation process

is approximately 2 s. This time is designated as the decay time. It naturally depends on the value of the initial sound level on the one hand, and the value of noise level on the other.

It can be demonstrated that the slope of the linear level drop depends only on the characteristics of the room and not on the sound source, as long as the discontinuation is sudden. Consequently an objective quantity for the acoustic behavior of a room can be derived: the time during which the sound level drop by 60 dB in comparison to its initial value is designated as the reverberation time. This value of 60 dB corresponds approximately to the dynamic range of a large orchestra. Inasmuch as such a dynamic range is not always accessible for measurements, and furthermore, the initial point often is not uniquely recognizable, the following procedure has been specified for the determination of reverberation times: starting with a drop of 30 dB from a value of 5 dB below the steady level, the measured time is doubled. This method is indicated in Fig. 5.7. From the graphically represented level drop a reverberation time of approximately $2 \times 0.9 = 1.8$ s is obtained.

In the course of a musical performance, the temporal note sequence and the actually dynamic structure rarely provides a drop of 60 dB. Consequently, particular attention is given to the beginning portion of the decay curve. A slope of the first 10–20 dB is determined, and from that, the time for a uniform level drop of 60 dB is calculated. For an evaluation of the first 10 dB one speaks of the "Early-Decay-Time" (Jordan, 1968), for the first 15 dB the "Initial-Reverberation-Time" (Atal et al., 1965) and for 20 dB "Beginning-Reverberation-Time" (Kürer and Kurze, 1967).

The reverberation time of a room decreases as individual reflections become weaker, i.e., the stronger the walls, the floor, and the ceiling, etc. absorb sound. In contrast it becomes longer for increased time separations between individual reflection processes; this "free path length" increases with the size of the room. When the total absorption of the room is not too large, these relations are represented by the Sabine reverberation formula:

$$T = 0.163\ V/A \quad T\,\text{in}\,\text{s},\ V\,\text{in}\,\text{m}^3,\ A\ \text{in}\ \text{m}^2,$$

where T is the "Sabine" reverberation time (corresponding to the 60 dB definition), V is the volume of the room and A is its equivalent absorption area which is calculated from the sum of the A values of the individual surfaces and objects.

Inasmuch as absorption is frequency dependent, the reverberation time also shows a frequency dependence, which, depending on the nature and furnishings of the room, can exhibit rather differing characteristics. As an example for a reverberation curve, the reverberation time of the fully occupied Bayreuth Festspielhaus is represented in Fig. 5.8. It is noted that the reverberation time for low frequencies is longest and decreases for the higher registers. It should further be noted that for high frequencies, an attenuation occurs during sound spreading (dissipation loss) in addition to the absorption at the walls as described.

The frequency dependence of the reverberation time is of great importance for the auditory tonal impression, since it causes a tone color change during the decay.

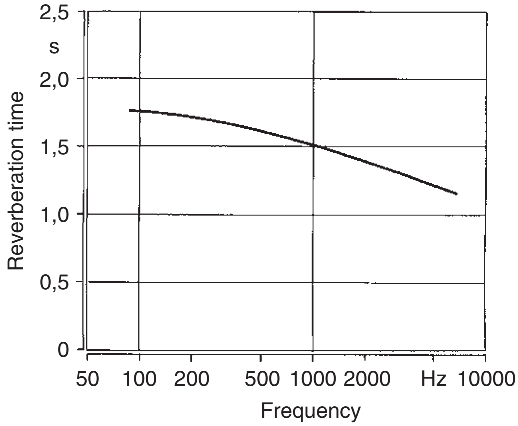


Fig. 5.8 Reverberation curve of the Bayreuth Festspielhaus under fully occupied conditions (after Beranek, 1962)

The faster the high components lose in intensity, the duller the decay process, as caused by the room. However, consideration must be given to the fact that in a certain sense the rise in the reverberation time below 125 Hz finds compensation in the characteristics of the ear. Since the "equal loudness curves" (see Fig. 1.1) are more closely spaced at those frequencies, a region of 60 phons is traversed by a level range of less than 60 dB so that a reverberation process at low frequencies appears shorter than an equally steep level drop at higher frequencies.

The early time decay in the region from 125 to 2,000 Hz has proven particularly suitable for characterizing the influence of a room on tone color (Lehnmann, 1976). This is because the dynamic short-time structure of music occupies a region of relatively narrow level difference. The Early-Decay-Time for most rooms is somewhat shorter than the Sabine reverberation time, particularly at low frequencies, when coupled room sections – e.g., above ceiling reflectors- contribute to late reverberations.

When evaluating reverberation curves of different rooms it becomes of interest to determine which smallest variations in the reverberation time are discernable for the trained ear. In a simplified approach this question can be answered as follows: for short reverberation times below 0.8 s, steps of approximately 0.02 s are noticeable, while the sensitivity above this limit amounts to approximately 3.5% of the relevant reverberation time (Seraphim, 1958). In practice, this means that values need not be specified more accurately than 0.1 s.

5.4 Direct Sound and Diffuse Field

5.4.1 The Energy Density

When a sound source radiates a long tone or a continuing sound, the direct sound and the multiplicity of sound reflections arrive at the listener at the same point and time, all of which have different run times and are attenuated in varying degrees. The combination of all of these components results in the observed sound level.

When an energy balance is established one must assume an equilibrium between the energy swallowed by the absorption surfaces and the energy provided by the source once a stationary state is reached. From this condition the so called energy density, i.e., the sound energy per unit volume, can be calculated. From this, the sound pressure level L_p of the diffuse sound field in the room is given by

$$L_p = L_w - 10\log(V/V_0) + 10\log(T/T_0) + 14\,\mathrm{dB},$$

where L_w represents the sound power level of the source V, the volume of the room (V_0 = 1 m^3) and T its reverberation time (T_0 = 1 s). The term 14 dB is the result of several constants as well as the reference point for sound power-, and sound pressure- levels. This sound pressure level of the diffuse sound field is, at least theoretically, the same everywhere within the room.

It depends on the size of the room and the power of the sound source, it is, however, also influenced by the reverberation time. Since the latter is frequency dependent, a pitch dependence results also for the energy density, and thus for the sound level of the diffuse field. This means, that because of its reverberation characteristics, the room not only influences the sound level but also the tone color. The shorter the reverberation time is in a certain frequency region, the more this region is disadvantaged in the tonal spectrum.

Still, the variation range of the reverberation time is not as large as the range in room volume which from a small chamber music room to a large cathedral encompasses several powers of 10. Consequently, practical experience has taught that it is advantageous when the reverberation time increases somewhat with increasing room size. However, this can only compensate in small measure for the decrease in energy density. On the other hand this presents an advantage since the duration of the audible reverberation is slightly shortened by the reduced sound level, so that, again the extension of the reverberation time finds partial compensation in the subjective reverberation impression. Finally, within limits, it is of course also possible to adapt the power of the sound source to the room acoustical conditions.

In order to make it possible to compare several rooms directly, the option is presented to combine the last three expressions on the right side in the formula for the sound level of the diffuse field into one quantity:

$$L_p = L_w - D_A.$$

The quantity D_A will be designated as “room damping index.” Accordingly, the room damping index (which is independent of the sound source) is the numerical difference between the sound power level of the source (also room independent) and the sound pressure level achieved by the source in the room. If the volume and reverberation time of the room are known, the room damping index can be calculated as

$$D_A = 10\log(V/V_0) - 10\log(T/T_0) - 14\,\text{dB}.$$

In practice, the room damping index indicates by how much the average sound level is changed when the same sound source is moved to different rooms. However, it should be emphasized, that in all cases this applies to a spatial average for the diffuse sound field i.e., for the steady state of the room (such as for long notes).

In order to represent the differences between energy densities in different locations in the same hall, as determined by different room reflections and directional characteristics of the sound source, the so called strength factor (also in dB) must be determined. It indicates the difference between the sound power level of the sound source and the sound pressure level as it actually occurs at the relevant location in the hall. This strength factor, however, must be measured in each situation by means of an impulse response, whereby it makes sense to include reflection with up to 80 ms delay relative to the direct sound, because it is also within this time duration that loudness impressions are formed. With some difficulty the strength factor can be calculated from several sets of data for the room (Lehmann, 1976).

5.4.2 *The Direct Sound*

As the sound radiated by the source spreads, its level naturally decreases with increasing distance. Energy conservation mandates that in an undisturbed sound field the sound pressure is inversely proportional to sound source distance, one speaks of the $1/r$ law or $1/r$ decay, where r represents the distance to the sound source. The dependence of the sound pressure level on sound source distance is given by

$$L = L_w - 20\log(r/r_0) - 11\,\text{dB} + D_i,$$

where again L_w is the sound power level of the source, r_0 the reference value of 1 m, and D_i is the directivity index of the source for the radiation direction under consideration. From this it follows for the sound level, that it decreases by 6 dB for each doubling of the distance to the source; a tenfold increase of the distance leads to a decrease by 20 dB.

Figure 5.9 represents this level change of the direct sound for a range of 28 cm to 20 m, where the reference level of 0 dB corresponds to the sound power level of the source radiating equally in all directions. In the immediate neighborhood of the sound source, this curve can deviate somewhat from the line drawn for the omnidirectional sound source, depending on its size and vibration shapes. Furthermore, dissipation losses already mentioned, effect a somewhat steeper level drop at very high frequencies. When the sound passes a strongly absorbing surface at a very shallow angle, the level drops somewhat faster than that which would correspond to a $1/r$ decay, since the sound waves are practically bent into the absorber. This case is of particular interest to the spread of sound over seating rows in an audience.

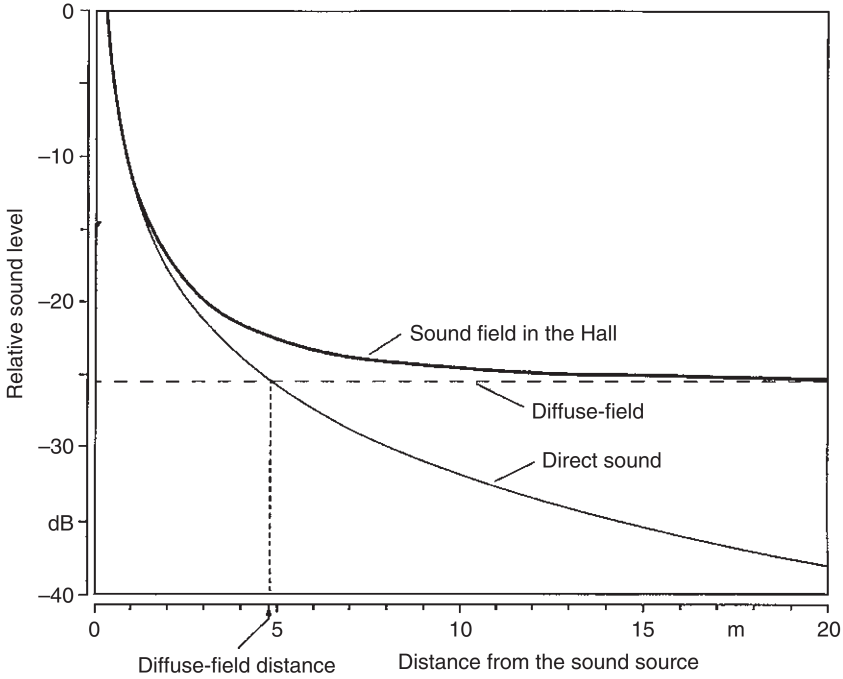
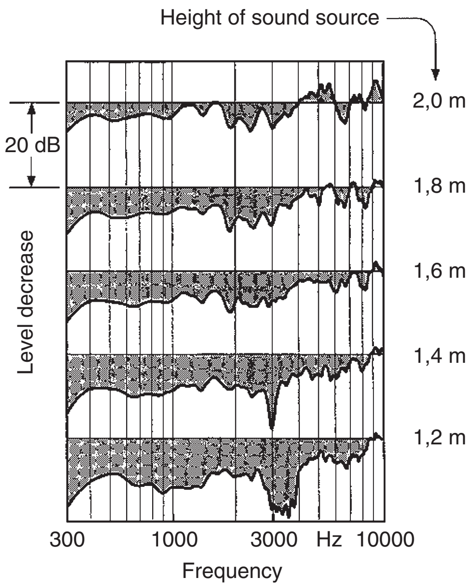


Fig. 5.9 Sound level in dependence on sound source distance

In Fig. 5.10 sound level reduction beyond the $1/r$ decay is represented for a listener position at a 13.5 m distance from the source (12th row) for different sound source heights. A more or less pronounced frequency dependence appears, which is most strongly noted for a sound source at head height. This finds its explanation in the bending of the sound around the head, especially at relatively high frequencies. This weakens the direct sound, especially for those tone contributions important for brilliance and clarity of articulation. This effect is not as strongly pronounced when audience seating is offset than is the case when heads are located exactly behind each other in the direction of sound spreading. Furthermore one notices a clear decrease of the additional damping when raising the sound source on steps. At a height of 2 m the influence of audience rows is nearly negligible: this additional damping therefore can be avoided in large measure with a higher podium and rising audience seating (Mommertz, 1993).

As the left end of the curves already indicates, the seating rows, however, effect an additional level decrease beyond the $1/r$ decay for a certain region of lower frequencies. The maximum of the resonance-like weakening lies in the region of 130 and 170 Hz. This frequency location is independent of seating row occupation and separation, it is only determined by the height of the backrests and their lower connection to the floor. Inasmuch as this additional level drop can take a value of 10–20 dB for a source distance of 20–25 m, it can affect an essential change of the direct sound of the low instruments. It is particularly noticeable in the sound

Fig. 5.10 Sound level reduction over audience seating rows which are not offset relative to each other, distance between sound source and observation point 13.5 m (after Mommertz, 1993)



impression when the components are contained only weakly in the first reflections from the ceiling or from hanging reflectors. This effect can be reduced when absorbers for the relevant frequency region are installed on the floor between rows of chairs (Schultz and Watters, 1964; Ando, 1985; Davies and Lam, 1994).

5.4.3 Diffuse-Field Distance

In the sound field present in a room, the direct sound, which decreases in intensity with distance from the source, is superimposed onto the diffuse field distance, which has the same level everywhere. Consequently, close to the sound source, the direct sound dominates, while at large distances it only makes a minor contribution to the overall sound level. This connection is represented in Fig. 5.9. In addition to the $1/r$ decay of the direct sound already explained earlier, the level of the diffuse field (for a certain volume and a certain reverberation time), as well as the total sound level resulting from the super-position of these two contributions is entered. One notes, that in consequence of the logarithmic dB scale, for the energy addition of the direct sound and statistical field, a curve results which deviates only a little from the one $1/r$ decay curve in the neighborhood of the sound source, which however, for increasing distance approaches the level of the diffuse field. Additionally, it should be noted, that in large halls the diffuse field is often not formed exactly, but rather its level drops by approximately 0.85 dB per 10 m distance from the sound source. This has the consequence that the energy density level can be by about 3 dB lower in the rear of the hall, which represents an audible loudness loss (Barron and Lee, 1988).

For the description of room acoustical quantities, it has been useful to delineate the region in which the direct sound is stronger in comparison to the region of predominantly diffuse sound contributions. The distance of this boundary from the source is referred to as the "diffuse – field distance." In the case of omnidirectionally uniform radiation by the source one also speaks of a diffuse-field radius (previously the designation "diffuse-field radius" was in general use, even for directional sound sources).The intersection of the two relevant curves in Fig. 5.9 correspond to the condition, that at this point the levels of the direct sound and the diffuse field are equal. From this, the example represented yields a diffuse field distance of 4.8 m for an omni-directional source. The following numerical-value equation applies

$$r_{\mathrm{H}} = 0.057\Gamma_{\mathrm{st}}\sqrt{(V/T)} \quad r_{\mathrm{H}}\ \mathrm{in\,m};\ V\,\mathrm{in\,m}^3;\ T\,\mathrm{in\,s},$$

where again V represents the room volume, T the reverberation time and Γ_{st} the statistical directivity factor of the sound source. When the sound source has a pronounced directional effect, the diffuse-field distance in the preferred direction becomes correspondingly large while in other directions, it becomes smaller than for a spherical radiator. It is important to note, that the diffuse-field distance is independent of the power of the sound source (Cremer, 1961).

A graphical solution of the equation for the diffuse-field distance is represented in Fig. 5.11, where it is assumed that the sound source has spherical characteristics,

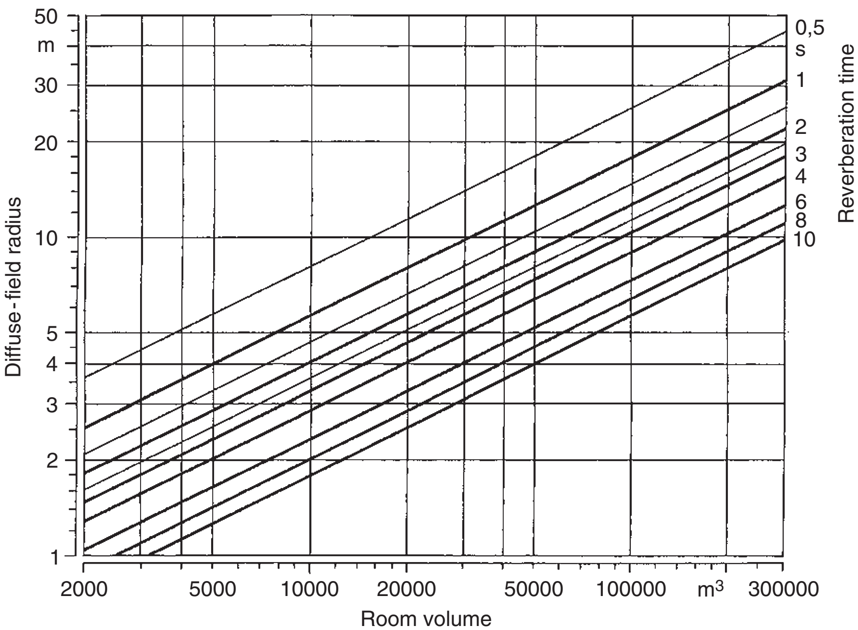


Fig. 5.11 Dependence of diffuse-field distance on room volume and reverberation time (for omnidirectional sound sources)

Table 5.1

Distance from the source	$r_H/2$	r_H	$2\ r_H$	$3\ r_H$
Relative sound level in dB:				
Direct sound	+6	0	−6	−10
Diffuse field	0	0	0	0
Superposition	7	3	1	0.4
Level increase in dB:				
By direct sound	–	3	1	0.4
By diffuse field	1	–	–	–

that is, the statistical directivity factor in all directions is equal to one. It is noticed that the diffuse-field distance increases with increasing room size and constant reverberation time, that, however for a fixed volume, it remains smaller for longer reverberation times. The advantage of a larger energy density and thus greater loudness, which accompanies the increase in reverberation time in large rooms is thus, purchased with a reduction in the diffuse-field distance and consequently, with a decrease in clarity.

In Table 5.1, some levels for the combined values of the sound field at several distances are assembled, using diffuse-field distance as a specific room characteristic for the starting point. The tripled diffuse-field distance proves to be of great importance, since at this point the direct sound lies by 10 dB below the diffuse field, which constitutes a boundary for certain locations of the sound source based on the reaction of the ear to the wave front arriving first. It can furthermore be seen that for the double diffuse-field distance, the direct sound effects an increase on the total level by only 1 dB; at half the diffuse-field distance, the influence of the diffuse field in comparison to the direct sound is correspondingly minute.

5.5 Temporal Structure of the Sound Field

The diffuse sound field can only be formed when sound waves arrive from all sides in a dense time sequence. However, this condition is not satisfied during the initial transient of a sound or noise within a room. The behavior of a room during an initial transient, therefore, does not represent a process to be treated statistically. Rather, the direct sound is followed after a short pause by the first reflection whose delayed time in comparison to the direct sound depends on the length of the detour the sound must traverse. Only after several additional reflections, which already arrive with shorter time separations, the uniform reverberation follows.

This process is represented schematically in Fig. 5.12 for a single sound source, where for the sake of clarity, only a selection of early reflections is shown in the picture of the hall. Among these are the specially important early reflections from the side walls (W_1 and W_2) as well as those formed by the gallery and the side wall barriers (W_1' and W_2'). The strength and the time of arrival of the individual

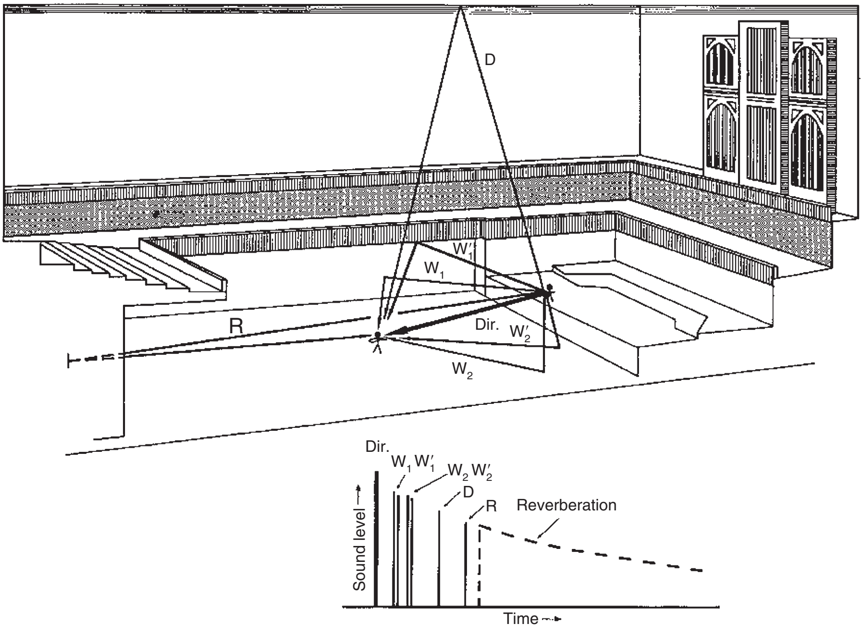


Fig. 5.12 Development of sound field in a concert hall: Sound paths (*top*), and time sequence (*bottom*) of the direct sound and the first reflections

reflections naturally depends in large measure on the shape of the room, as well as the reflection characteristics of the wall and ceiling, furthermore, both characteristics vary for different seating areas within the hall.

These room acoustical processes play an important role in the tonal impression for a musical performance. In this context, the direct sound is responsible for the clarity, in particular for rapid tone sequences, for the transparency of the tonal picture. It also transmits the tonal impression of the spatial arrangement of the performer. Furthermore, the direct sound contributions affect the sensation of the proximity of the orchestra or the stage to the listener. The increasingly delayed reflections, which in their totality shape the reverberation, particularly influence the melting of the individual voices into a complete sound. In similar fashion, they bridge the time sequence of short breaks between individual notes, whereby the melodic line obtains greater uniformity. In addition, the reverberation raises the loudness impression considerably.

Depending on delay times and intensities, the early reflections can influence the tonal impressions in different ways (Kuhl, 1965). Also the directions, from which the reflections reach the listener, play an important role, and finally, the nature of the performed music itself has an influence on the subjective perception of the first reflections (Schubert, 1969). For increasingly impulsive music, i.e., for sharp *staccato* and *pizzicato*, the ear responds more sensitively to reflections from the room. The reflection arriving first carries particular significance. For a delay time of approximately 30 ms, which corresponds to a detour of about 10 m, it affects the orchestral music by way of enhancing the direct sound without detracting from the clarity. This could merely result in minor tone color changes, which are connected to the relative phases of direct and reflected waves. However, this effect is only audible when the first reflection is dominant for the initial room sound, i.e., when further reflections are only weak or follow relatively late, as is the case above all for open air performances or in extremely dry halls. For solo passages, with a strongly pronounced rhythmic dynamic structure, this time span is reduced to approximately 20 ms. Since a long delay naturally causes an impression of a larger room, the time span between direct sound and first reflection is responsible for the degree of intimacy of the spatial sound impression (Beranek, 1962). The shorter the time span, the closer one feels acoustically to the musicians. Beyond that, an impression of the dimensions of the room is transmitted to the listener by the time sequence of the first wall and ceiling reflections, independently of their strengths.

As already mentioned elsewhere, the first reflection can even be 10 dB stronger than the direct sound without having the latter lose its direction-determining characteristic (Haas, 1951). This can occur when the sound source is blocked from the listener as is the case, for example, for instruments in the orchestra pit in relationship to the seats on the main floor in an opera house.

Concerning the minimum intensity necessary to obtain any tonal impression from the reflection, the direction of incidence at the listener plays an important role. While reflections, which reach the listener from the front or above within the time interval of the first 30 ms, depending on the nature of the music, can be by 10–20 dB weaker than the direct sound, reflections arriving from a side can still be perceived when their level is lower by an additional 10 dB (Schubert, 1969).

Under the assumption that the first reflection arrives sufficiently early, one can generally conclude that all reflections which follow the direct sound within 80 ms are useful for the precision and clarity of the musical tone picture, while later reflections and reverberations diminish the clarity. One can therefore define a "clarity factor" (C_{80}), wherein the sound energy (E_{80}) arriving within the first 80 ms is compared to the remaining energy:

$$C_{80} = 10\log(E_{80}/(E_{\infty} - E_{80}))\,\mathrm{dB},$$

where E_{∞} is the total arriving sound energy. This clarity factor should lie between -2 and +4 dB, for distant seats a value of -5 dB is still defensible. For speech – and to some extent for sharply structured music – the boundary between reflections which enhance clarity and those which reduce it lies at 50 ms. Accordingly, the "clarity factor" (C_{50}) is defined as

$$C_{50} = 10\log(E_{50}/(E_{\infty} - E_{50}))\,\mathrm{dB},$$

where E_{50} is the sound energy arriving during the first 50 ms. This clarity factor should lie above 0 dB (Reichardt et al., 1975).

Individual reflections, which arrive with a delay of more than 25 ms can effect very different sensations, depending on their directions. When coming from the median plane, i.e., from the plane standing vertically in the room, which also includes the direction of sight of the listener, they do enhance the sound, however they do not create a spatial effect. The upper limit for this lies at about 80 ms. Side reflections arriving in this time range determine the tonal room impression, as do all reflections arriving later (Barron, 1971). The boundary between side reflections and those not coming from the side is naturally somewhat fluid; The difference between the sound contributions arriving at the two ears increases as the incident direction is increasingly turned toward the side. In fact, the value of side reflections in relation to the room impression increases with the sine of the incident angle (calculated in relation to the direction of sight) (Alrutz and Gottlob, 1978). With that, the "inter-aural correlation" which determines the spatial impression decreases (Gottlob, 1973). In order to be able to measure this sound effect as "spatial impression measure", a 40° cone about the direction of sight is specified as a boundary (Lehmann, 1975; Reichardt et al., 1975). All sound energy contributions with more than 80 ms delay, as well as all sound reflections arriving from the side (i.e., outside of the 40° cone) with 25–80 ms, delay enhance the spatial sensation. All sound contributions arriving within the first 25 ms, and those arriving from the front (within the 40° cone) with 25–80 ms delay do not contribute to the spatial sensation. When designating the spatially enhancing sound energy as E_{R} and the sound energy which does not contribute to the spatial sensation as E_{NR} the spatial impression measure is given as

$$R = 10\log(E_{\mathrm{R}}/E_{\mathrm{NR}})\,\mathrm{dB}.$$

A range of 1–7 dB is desirable for concert halls. From a standpoint of measurement it is simpler to determine the relationship of the sound energy arriving from the side between 25 and 80 ms to the entire sound energy arriving between 0 and 80 ms ("lateral efficiency") (Jordan, 1979).

The spatial impression measure, as well as the "lateral efficiency", are quantities which are independent of the loudness of the sound impression for the listener. Thus they are not concerned with the sensation that music only assumes spatial characteristics with increasing loudness (Keet, 1968). This effect is represented by the concept of "spatial impression". This is to be understood as the subjective impression that the sound of an orchestra seems to expand into the space between the musicians and the side walls and possibly from the floor to the ceiling without limiting the localization of the instruments (Kuhl, 1978). The sensation that sound in a room also comes from behind occurs only rarely, e.g., when one is very close to the sound source in a large church. As a matter of principle one can assume that the acoustic space impression is limited to the hemisphere in front.

Sufficient loudness is an essential condition for an impression of space, so that the effect, when present at all, only occurs for *forte* passages in music. Precisely therein lies the value of this criterion: even in mediocre halls a *piano* passage can be made to sound well, while a convincing tonal development in *forte* succeeds only in acoustically good halls. This is because both sufficient loudness and the impression of spatial expansion of the sound are of great significance for an emotional experience.

In addition to loudness, the spatial sensation demands an energy relationship between reflections not in the median plane with a delay time of 10–80 ms on the one hand, and the direct sound on the other. The level difference between the sum of these reflection and the direct sound is also designated as the "lateral sound level". For a quantitative description of the spatial sensation, a scale is determined in such a way that an increase in the spatial sensation by one step is caused by an increase in the sound level by 5 dB, provided that a spatial effect is already sensed at the initial level. Below a certain level, no spatial sensation occurs (step 0). Figure 5.13 shows a spatial sensation diagram for a seat in the Hamburg Musikhalle. It is evident that below a level of 76 dB no spatial sensation occurs and that above this level-boundary, three spatial steps are reached up to a *forte* level of 91 dB.

The level-boundary for the onset of spatial impression can be different for different halls. The value of this level-boundary becomes lower as the sound level coming from the side increases. Figure 5.14 shows this relationship. Particularly low values, and thus pronounced spatial impressions, are found in long stretched rectangular rooms. This also makes the requirement understandable that, for sound esthetic reasons, the first side wall reflections should arrive earlier than the first ceiling reflections and reflections from the back wall (Marshall, 1967).

Fundamentally, all spectral components of the early lateral reflections contribute to the subjective spatial impression, and the broader the frequency region contained in these reflections, the more strongly do these reflections enhance the spatial impression. However, the frequency contributions contained in the reflections can evoke a difference in the nature of the spatial impression: if the reflections contain only low and middle frequencies, a predominant sensation arises that the sound is broadened in the depth of the room (to the front and back). Many listeners connect the feeling of being immersed in the sound with that effect. When the spectrum of

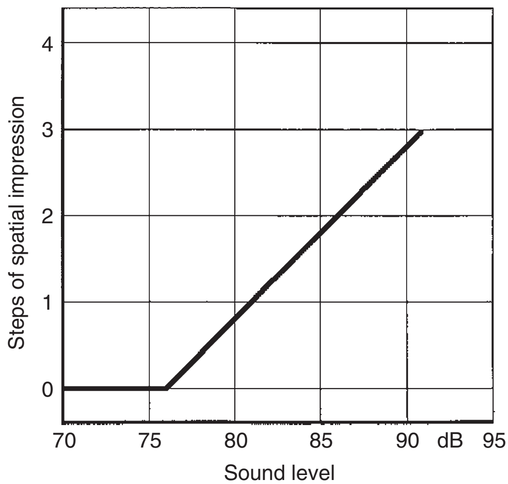


Fig. 5.13 Spatial impression of the orchestral sound in the Hamburg Musikhalle in dependence on sound level (after Kuhl, 1978)

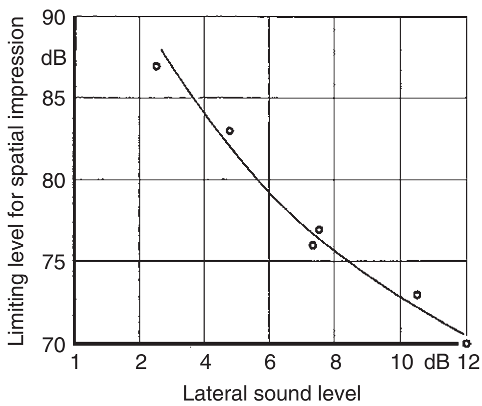


Fig. 5.14 Limiting value of sound levels for onset of spatial impression. Data points for halls (After Kuhl, 1978) from *top* to *bottom*: Stadthalle Hannover, Rheinhalle Düsseldorf, Philharmonie Berlin, Musikhalle Hamburg, Stadthalle Wuppertal. The lowest data point refers to hearing tests with electroacoustic instrumentation

the reflections also includes sufficiently high frequencies (above 3,000 Hz) a preferential sensation of tonal broadening occurs (Blauert and Lindemann, 1986). These differing dimensions of spatial impression can be influenced within certain limits by construction modification of the relevant reflection surfaces. They belong to specific tonal characteristics which differentiate large halls from each other.

By changing reflector arrangements, delay times of individual sound reflections can occasionally be influenced. However, it should be noted, that the ear responds more sensitively to reflections from the side than to those which come from the front, from above or behind (Reichhardt and Schmidt, 1967). Thus for lateral reflections in a test series, changes in delay time of 7 ms were noted; this corresponds to a lengthening of the sound path of approximately 2.4 m. For ceiling reflections, in the same test, changes of approximately 12 ms corresponding to an additional path of 4.1 m were required. It can be assumed, however, that for music with strong rhythmic structure, these values are reduced by approximately one-half (Schubert, 1969). This is also supported by the fact that musicians sense a clear difference in the spatial effect, depending on whether the orchestra is located on the same level as the concert hall floor or on a 1.5 m high podium.

Based on the very different sound field developments at different locations in the hall, specifying a uniform initial transient, as is done for musical instruments, is difficult. If the time required for the sound level to reach a value of only 3 dB below the steady state value is calculated, a relatively short initial transient time is obtained under the assumption of statistical conditions: it amounts to 1/20 of the reverberation time. After approximately 1/14 of the reverberation time, the level lies only 2 dB below the final value. In reality, in such a short time, the requirements for a diffuse sound field are certainly not met. Accurate studies of the spatial sound field development in a concert hall however, have given the result, that for most locations the level of 3 dB below the final value was already reached within three quarters of the time span which would have been calculated from statistical considerations; only in the least favorable locations were these two values the same (Junius, 1959). From this, one can generalize that the initial transient times of a room can be assumed to be somewhat shorter than 1/20 of the reverberation time for the majority of the locations.