9. Acoustics in Halls for Speech and Music

Acoustics in

This chapter deals specifically with concepts, tools, and architectural variables of importance when designing auditoria for speech and music. The focus will be on cultivating the useful components of the sound in the room rather than on avoiding noise from outside or from installations, which is dealt with in Chap. 11. The chapter starts by presenting the subjective aspects of the room acoustic experience according to consensus at the time of writing. Then follows a description of their objective counterparts, the objective room acoustic parameters, among which the classical *reverberation time* measure is only one of many, but still of fundamental value. After explanations on how these parameters can be measured and predicted during the design phase, the remainder of the chapter deals with how the acoustic properties can be controlled by the architectural design of auditoria. This is done by presenting the influence of individual design elements as well as brief descriptions of halls designed for specific purposes, such as drama, opera, and symphonic concerts. Finally, some important aspects of loudspeaker installations in auditoria are briefly touched upon.

9.1 **Room Acoustic Concepts** 302

9.2 **Subjective Room Acoustics** 303
9.2.1 The Impulse Response 303
9.2.2 Subjective Room Acoustic Experiment Techniques 303
9.2.3 Subjective Effects of Audible Reflections 305

9.3 **Subjective and Objective Room Acoustic Parameters** 306
9.3.1 Reverberation Time 306
9.3.2 Clarity 308
9.3.3 Sound Strength 308
9.3.4 Measures of Spaciousness 309
9.3.5 Parameters Relating to Timbre or Tonal Color 310
9.3.6 Measures of Conditions for Performers 310
9.3.7 Speech Intelligibility 311
9.3.8 Isn't One Objective Parameter Enough? 312
9.3.9 Recommended Values of Objective Parameters 313

9.4 **Measurement of Objective Parameters** ... 314
9.4.1 The Schroeder Method for the Measurement of Decay Curves 314
9.4.2 Frequency Range of Measurements 314
9.4.3 Sound Sources 315
9.4.4 Microphones............................ 315
9.4.5 Signal Storage and Processing..... 315

9.5 **Prediction of Room Acoustic Parameters** 316
9.5.1 Prediction of Reverberation Time by Means of Classical Reverberation Theory................ 316
9.5.2 Prediction of Reverberation in Coupled Rooms 318
9.5.3 Absorption Data for Seats and Audiences 319
9.5.4 Prediction by Computer Simulations 320
9.5.5 Scale Model Predictions 321
9.5.6 Prediction from Empirical Data ... 322

9.6 **Geometric Design Considerations** 323
9.6.1 General Room Shape and Seating Layout................... 323
9.6.2 Seating Arrangement in Section .. 326
9.6.3 Balcony Design 327
9.6.4 Volume and Ceiling Height 328
9.6.5 Main Dimensions and Risks of Echoes 329
9.6.6 Room Shape Details Causing Risks of Focusing and Flutter 329
9.6.7 Cultivating Early Reflections........ 330
9.6.8 Suspended Reflectors................ 331
9.6.9 Sound-Diffusing Surfaces 333

9.7 **Room Acoustic Design of Auditoria for Specific Purposes** 334
9.7.1 Speech Auditoria, Drama Theaters and Lecture Halls 334

9.7.2 Opera Halls 335
9.7.3 Concert Halls for Classical Music .. 338
9.7.4 Multipurpose Halls 342
9.7.5 Halls for Rhythmic Music 344
9.7.6 Worship Spaces/Churches 346

9.8 **Sound Systems for Auditoria** 346
9.8.1 PA Systems 346
9.8.2 Reverberation-Enhancement Systems 348

References 349

Current knowledge about room acoustic design is based on several centuries of practical (trial-and-error) experience plus a single century of scientific research. Over the last couple of decades scientific knowledge has matured to a level where it can explain to a usable degree:

1. the subjective aspects related to how we perceive the acoustics of rooms
2. how these subjective aspects are governed by objective properties in the sound fields
3. how these objective properties of the sound field are governed by the physical variables in the architectural design

This chapter is organized in the same manner. The various subjective acoustic aspects will form the basis for our discussion and be a guide through most aspects of relevance in room acoustic design. However, some trends in current design practice based on the experience and intuition of individual acoustical designers will also be commented on. It is hoped that this approach will also have the advantage of stimulating the reader's ability to judge a room from her/his own listening experience, an ability which is important not only for designers of concert halls, but also for those responsible for creating and enjoying the sound in these halls: musicians, sound-system designers, recording engineers and concert enthusiasts.

9.1 Room Acoustic Concepts

In order to help the reader maintain a clear perspective all the way through this chapter, Fig. 9.1 illustrates the universe of architectural acoustics.

In the upper half of the figure, we have the phenomena experienced in the real world. Going from left to right, we have the auditoria, in which we can experience objective sound fields causing subjective impressions of the acoustic conditions. In all of these three domains, we find a huge number of degrees of freedom: halls can differ in a myriad of ways (from overall dimensions to detailed shaping of small details like door handles), the sound fields which we describe by the impulse response concept – as explained in a moment – contain a wealth of individual sound components (reflections), each being a function of time, level, direction and frequency. Also, every individual may have his/her own way of expressing what the room does to the sound heard.

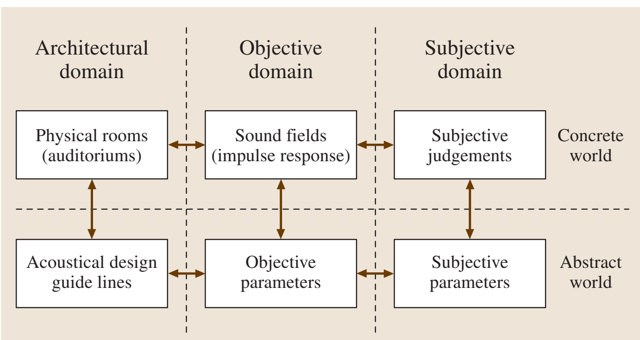


Fig. 9.1 Overview of concepts related to subjective room acoustics (after [9.1])

In other words, like in many other aspects of life, the real world is so complex that we need to simplify the problem – reduce the degrees of freedom – through definition of abstract, well-formulated and meaningful concepts and parameters in all three domains (the lower row of boxes). First, we must try to define a vocabulary to describe the subjective acoustic impression: we will isolate a set of subjective acoustic parameters that are valid to most people's listening experience (as indicated by the box in the lower-right corner), then we will try to deduce those properties from the sound fields that are responsible for our experience of each of the subjective aspects: we will define a set of objective, measurable parameters that correlate well with the subjective parameters, and finally – in order to be able to guide the architect – we must find out which aspects of the design govern the important objective and in turn the subjective parameters, so we can assist in building halls meeting the specific needs of their users.

A rough historical overview: systematic search for objective acoustic parameters peaked in the 1960s and 1970s, while the relationships between the objective parameters and design variables were the topic of many research efforts in the 1980s and 1990s. However, we cannot yet claim to have the answers to all questions regarding the acoustic design of auditoria. Hopefully, there will always be room for intuition and individual judgment, not only in room acoustic design; but also as a source of inspiration for continued research efforts. Still it should be clear that room acoustic design today is based on solid research; it is not a snake-oil business.

9.2 Subjective Room Acoustics

9.2.1 The Impulse Response

The basic source of information regarding the audible properties of the sound field in a room is the impulse response signal. Actually, this signal – when recorded with a multichannel technique preserving the information about direction of incidence – can be shown to contain all information about the acoustics of a room between two specific source and receiver positions.

Consider a short sound impulse being emitted by the source on the stage, as shown in Fig. 9.2. A spherical wave propagates away from the source in all directions and the sound first heard in the listener position originates from that part of the wave that has propagated directly from the source to the receiver, called *the direct sound*. This is shown on the left in the lower part of Fig. 9.2, which shows the received signal versus time at a given position in the room.

This component is soon followed by other parts of the wave that have been reflected one or more times by the room boundaries or by objects in the room before reaching the receiver, called *the early reflections*. Besides arriving later than the direct sound, normally these reflections are also weaker because the intensity is reduced as the area of the spherical wavefront increases with time (spherical distance attenuation) and because a fraction of the energy is being absorbed each time the wave hits a more or less sound-absorbing room boundary or object in the room.

The sound wave will continue to be reflected and to pass the receiver position until all the energy has been absorbed by the boundaries/objects or by the air. The density of these later reflections increases with time (proportional to t^2), but the attenuation due to absorption at the room boundaries ensures that eventually all sound dies out. This decay is often heard as reverberation in the room, as Sabine did, when he carried out his famous experiment in the Fogg Art Museum more than 100 years ago [9.2]. This event marked the start of subjective room acoustics as a science.

9.2.2 Subjective Room Acoustic Experiment Techniques

With each of the reflections specified in time, level, spectrum and direction of incidence, the impulse response contains a huge amount of information. Now, the question is how we can reduce this to what is necessary for an explanation of why we perceive the acoustics in different rooms as being different.

Experiments trying to answer this question have mainly been carried out in the second half of the 20th century when electro-acoustic means for recording, measurement and simulation of sound fields had become available. Such experiments take place in the four rightmost boxes in Fig. 9.1: a number of impulse responses – or rather, music or speech convolved (filtered) through these responses – are presented to a number of subjects. The subjective evaluations are collected either as simple preferences between the sound fields presented in pairs or as scalings along specific subjective parameters suggested by the experimenter. The experimenter will also choose a number of different objective parameters, and calculate their values for each of the impulse responses presented. The level of correlation between these objective values and the objective scalings will then indicate which of the parameters and therefore which properties of the impulse responses are responsible for the subjective evaluations.

As can be imagined, the results of such experiments will be strongly dependent on the range of variation and degree of realism of the impulse responses presented – as well as by the sound quality of the equipment used for the presentation. Besides, the results will always be limited by the imagination of the experimenter regarding his/her:

1. suggestions of proper semantic descriptors for the subjects' evaluation (except in cases where the subjects are only asked to give preferences) and

2. suggestions of calculation of parameters from the impulse response.

In other words, before the experiments, the experimenter must have some good hypotheses regarding the contents of the lower-mid and lower-right boxes in Fig. 9.1.

Over the years, a variety of different techniques have been applied for presenting room sound fields to test subjects:

1. Collection of opinions about existing halls recalled from the memory of qualified listeners
2. Listening tests in different halls or in a hall with variable acoustics
3. Listening tests with presentation of recordings from existing halls or of *dry* speech/music convolved with impulse responses recorded in existing halls
4. Listening tests with sound fields synthesized in anechoic rooms (via multiple loudspeakers in an anechoic room) or impulse responses generated/modified by computer convolved with *dry* speech/music and presented over headphones.

It should be mentioned here that our acoustic memory is very short. Therefore, unless the subjects are highly experienced in evaluating concert hall acoustics, comparison of acoustic experiences separated by days,

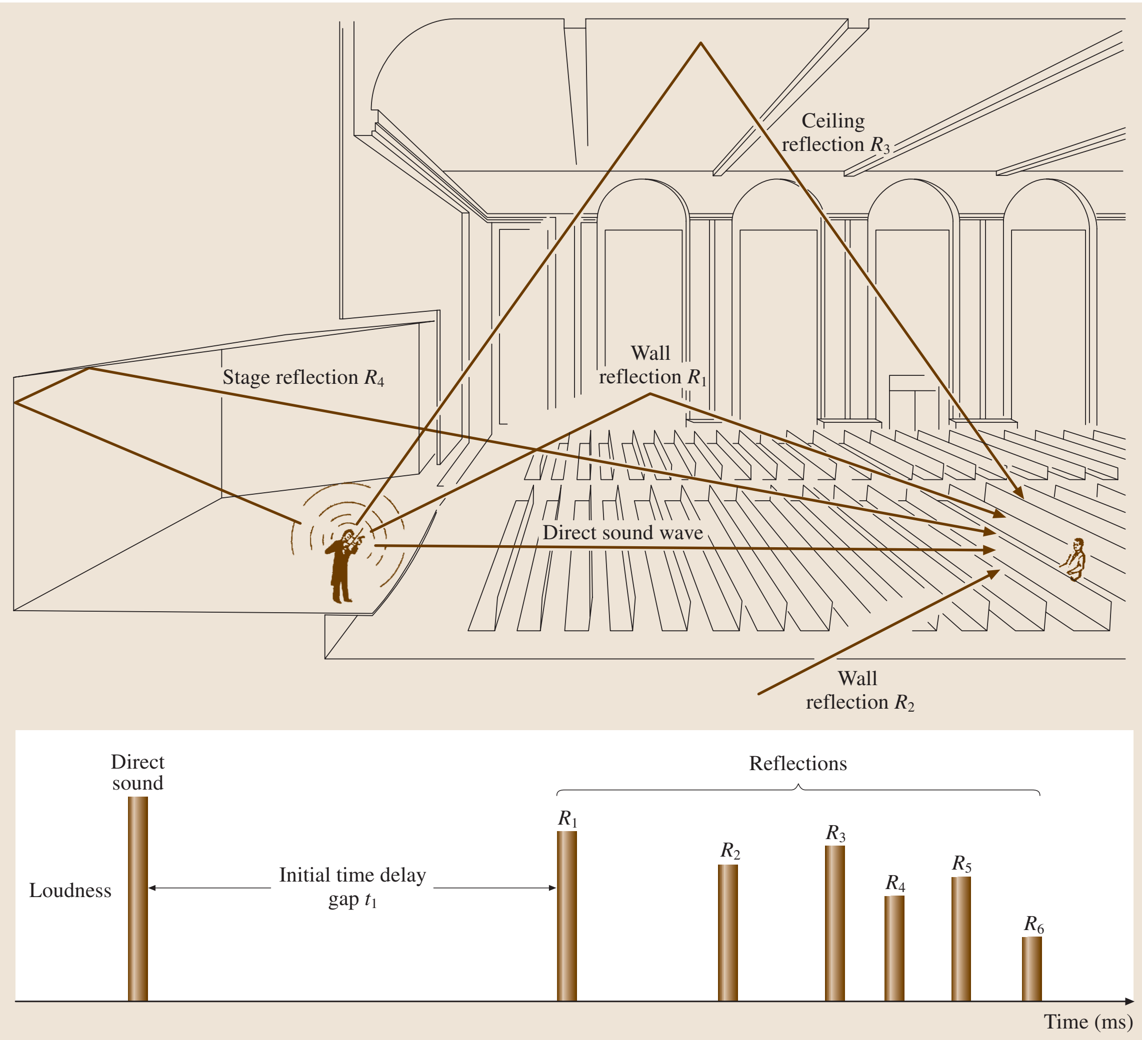


Fig. 9.2 Diagram illustrating the generation of an impulse response in a room (after [9.3])

weeks or even years is less reliable than comparison in the lab, where different recordings can be presented immediately one after the other. On the other hand, it is often difficult to get really qualified people to come to the lab to participate in scientific tests. Therefore, results from interview surveys can still be highly relevant.

These different techniques for presenting the sound fields all have different limitations. Experiments in the lab involving electro-acoustic reproduction or the generation of sound fields may lack fidelity, while listening in a real hall to a real orchestra will normally lack flexibility and control of the independent variables. Another important difference between these two methods is the absence of realistic visual cues in the lab. Enjoyment of a performance is a holistic experience, a fact which is now given increased scientific attention. In all cases elaborate statistical analysis of the results are needed in order to separate the significant facts from the error variance present in any experiment involving subjective judgements. Since the 1960s, multidimensional statistical tool such as factor analysis and multidimensional scaling have been very useful in distinguishing between the different subjective parameters which are present – consciously or not – in our evaluation. In any case, published results are more likely to be close to the truth, if they have been verified by several experiments – and experimenters – using different experimental approaches.

9.2.3 Subjective Effects of Audible Reflections

As the impulse response consists of the direct sound followed by a series of reflections, the simplest case possible would be to have only the direct sound plus a single reflection (and perhaps some late, artificial reverberation). Many experiments in simulated sound fields have been conducted using such a simple setup, and in spite of the obvious lack of realism, we can still learn a lot about the subjective effects of reflections from these experiments.

First of all, due to the psycho-acoustic phenomenon called forward masking, the reflection will be inaudible if it arrives very soon after the direct sound and/or its level is very low relative to the direct sound. Thus, there exists a level threshold of audibility depending on delay and direction of incidence relative to the direct sound. Only if the level of the reflection is above this threshold will the reflection have an audible effect, which again depends on its level, delay and direction of incidence. The possible effects are (at least):

- Increased level (energy addition)
- Increased clarity, if it arrives within 50–80 ms after the direct sound
- Increased spaciousness, if the reflection direction in the horizontal plane is different from that of the direct as this causes the signals received at the two ears to be different (If the angle between the direct and reflected sounds differs in the vertical plane only, the effect is rather a change in *timbre* or coloration of the sound.)
- Echo, typically observed for delays beyond 50 ms and at high reflection levels. If the delay is very long, say 200 ms, then the echo may even be detected at a much lower level;
- Coloration. If the delay is short, say below 30 ms, and the level is high, phase addition of the direct sound and the reflection will create a *comb filter* effect, which severely deteriorates the original frequency spectrum
- Change in localization direction, in cases where the reflection is louder than the direct sound. This may happen either due to amplification of the reflection via a concave surface or due to excess attenuation of the direct sound, for instance by an orchestra pit rail.

Audibility thresholds, echo risk and other effects also depend on spectral and temporal properties of the signal itself. Thus, with speech and fast, staccato played music (not to speak of artificial, impulsive *click* sounds), echoes are much easier to detect than when the signal

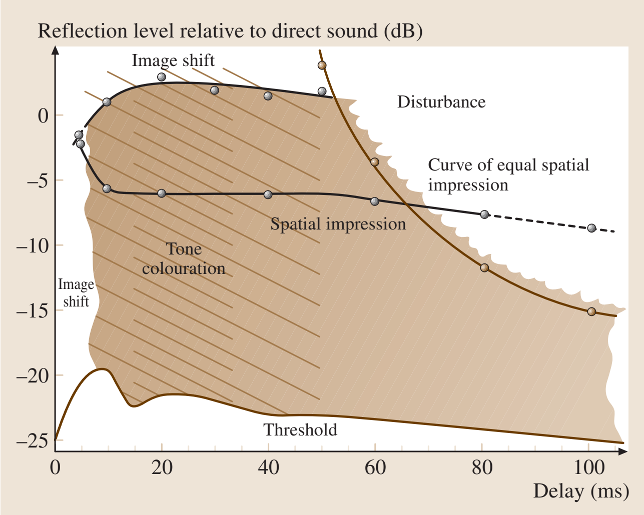


Fig. 9.3 Various audible effects of a single reflection arriving from the side. The signal used was music (after [9.4])

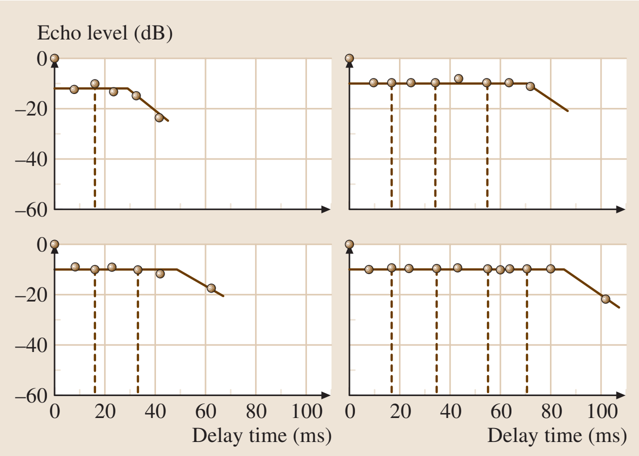


Fig. 9.4 Thresholds of perception of a new reflection (*circles* and *horizontal lines*) being added to impulse responses already containing one, two, three, or four other reflections with delay and relative levels, as indicated by the *vertical lines* (after [9.5])

is slow, legato music. On the other hand, these slower signals are more sensitive for the detection of coloration effects.

The temporal properties of sounds are sometimes described by the autocorrelation function. According to Ando (see Chap. 10 in this book) not only the thresholds of audibility but also the preferred delay of early reflections and the preferred reverberation time depend strongly on the autocorrelation of the signal. However, such a strong effect on concert hall preference – derived from listening experiments in rather simplified simulated sound fields in the lab – is not in accordance with everyday listening experiences in concert halls, in which we gladly accept listening to different music pieces of different tempo from different seats (different reflection delays) in the same hall (fixed reverberation time).

Figure 9.3 illustrates the various audible effects of a single reflection arriving from a 40° angle relative to a frontal direct sound. In this experiment music was used as the stimulus signal. Below the lower threshold (reflection level below about −20 dB and almost independent of the delay), the reflection is inaudible. In the upper-right region it causes a distinct echo, while in the shaded area we experience *spatial impression* or a broadening of the area from which the sound seem to originate. For delays lower than about 30 ms, this single reflection also causes an unpleasant change in *timbre* or *tonal color*, which is due to the combination of the direct and reflected signal forming a comb filter giving systematic changes in the spectrum. If the level of the reflection is increased to be louder than the direct sound (above 0 dB) while the delay is still below say 50 ms, we experience an *image shift* or a change in localization from the direction of the direct sound towards the direction from which the reflection is coming.

In practice, of course, the impulse response from a room contains many reflections, and we therefore need to know how the threshold of perceptibility of the incoming reflection changes in cases where the impulse response contains other reflections already. As seen from Fig. 9.4, the threshold level increases substantially in the delay range already occupied by other reflections. This phenomenon is primarily due to masking.

From this we can conclude that many of the details in a complex impulse response will be masked and only some of the dominant components or some of its overall characteristics seriously influence our perception of the acoustics. This is the reason why it is possible to create rather simple objective room acoustic parameters, as listed in the following section, which still describe the main features of the room acoustic experience. Thus, we seem to have some success describing the complex real world (the upper-middle and right boxes in Fig. 9.1) by simpler, abstract concepts. Also, it will be seen that the subjective effects caused by a single reflection remain important in the evaluation of more-realistic and complicated impulse responses.

9.3 Subjective and Objective Room Acoustic Parameters

From a consensus of numerous subjective experiments in real rooms and in simulated sound fields (representing all of the previously mentioned subjective research techniques), we now have a number of subjective parameters and corresponding objective measures available. Most of these are generally recognized as relevant descriptors of major aspects in our experience of the acoustics in rooms. This situation has promoted standardization of measurement methods and many of the objective parameters are now described in an appendix to the International Organization for Standardization (ISO) standard [9.6]. In order to maintain a clear distinction between the subjective and objective parameters in the following, the names for the subjective parameters will be printed in italics.

9.3.1 Reverberation Time

Reverberance is probably the best known of all subjective room acoustic aspects. When a room creates too much *reverberance*, speech loses intelligibility because important details (consonants) are masked by louder, lingering speech sounds (the vowels). For many forms of music, however, *reverberance* can add an attractive fullness to the sound by bonding adjacent notes together and blending the sounds from the different instruments/voices in an ensemble.

The reverberation time T which is the traditional objective measure of this quality, was invented 100 years ago by W. C. Sabine. T is defined as the time it takes for the sound level in the room to decrease by 60 dB after a continuous sound source has been shut off. In practice, the evaluation is limited to a smaller interval of the decay curve, from −5 dB to −35 dB (or −5 dB to −25 dB) below the start value; but still relating to a 60 dB decay (Fig. 9.5), i. e.:

$$T = 60\,\mathrm{dB}\,\frac{(t_{-35}) - (t_{-5})}{(-5\,\mathrm{dB}) - (-35\,\mathrm{dB})}\,. \tag{9.1}$$

In this equation, t_{-X} denotes the time when the decay has decreased to XdB below its start value, or, if we let $R(t)$ represent the squared value of the decaying sound pressure and shut off the sound source at time $t = 0$:

$$10 \log_{10}\left(\frac{R(t_{-X})}{R(0)}\right) = -X\,\mathrm{dB}\,. \tag{9.2}$$

With the fluctuations always present in decay curves, T should rather be determined from the decay rate, A dB/s, as found from a least-squares regression line (determined from the relevant interval of the decay curve). Hereby we get for T:

$$T = \frac{60\,\mathrm{dB}}{A\,\frac{\mathrm{dB}}{\mathrm{s}}} = \frac{60}{A}\,\mathrm{s}\,. \tag{9.3}$$

Ways to obtain the decay curve from the impulse response will be further explained in the section on measurement techniques.

Due to masking, the entire decay process is only perceivable during breaks in the speech or music. Besides, the rate of decay is often different in the beginning and further down the decay curve. During running music or speech, the later, weaker part of the reverberation will be masked by the next syllable or musical note. Therefore an alternative measure, early decay time (EDT) has turned out to be better correlated with the *reverberance* perceived during running speech and music. This parameter, like T, also measures the rate of the decay; but now evaluated from the initial part, the interval between 0 and −10 dB, only. Thus,

$$\mathrm{EDT} = 6(t_{-10}) \quad \text{or} \quad \mathrm{EDT} = \frac{60}{A_{(0\,\mathrm{dB}\rightarrow -10\,\mathrm{dB})}}\,\mathrm{s}\,. \tag{9.4}$$

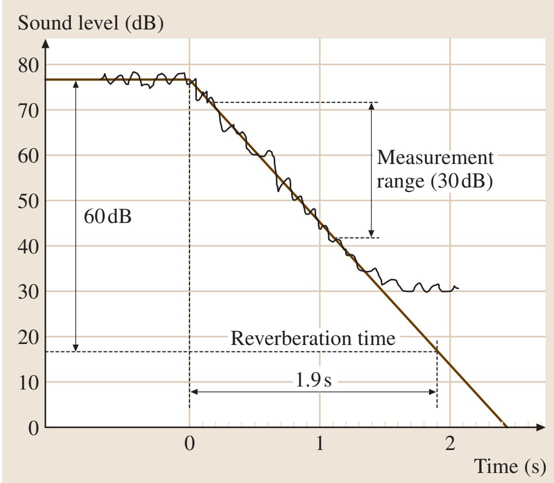


Fig. 9.5 The definition of reverberation time (after [9.4])

The detailed behavior of the early part of the reverberation curve is influenced by the relative levels and distribution in time of the early reflections, which in turn vary depending on the positions of the source and receiver in the room. Likewise, the value of EDT is often found to vary throughout a hall, which is seldom the case with T.

In spite of the fact that EDT is a better descriptor of *reverberance* than T, T is still regarded the basic and most important objective parameter. This is mainly due to the general relationship between T and many of the other room acoustic parameters and because a lot of room acoustic theory relates to this concept, not least diffuse field theory, which is the basis for measurements of sound power, sound absorption, and sound insulation. T is also important by being referred to in legislation regarding room acoustic conditions in buildings.

Talking about diffuse field theory, it is often of relevance to compare the measured values of certain objective parameters with their expected values according to diffuse field theory and the measured or calculated reverberation time. As diffuse field theory predicts the decay to be purely exponential, the distribution in time of the impulse response squared should follow the function:

$$h^2(t) = A \exp\left(\frac{-13.8}{T} t\right) , \tag{9.5}$$

in which the constant -13.8 is determined by the requirement that for $t = T$:

$$10 \log_{10}\left(\exp\left(\frac{-13.8}{T} t\right)\right) = 60\,\text{dB} . \tag{9.6}$$

With an exponential decay, the decay curve in dB becomes a straight line. Consequently, the expected value of EDT, EDT_{exp}, equals T.

When evaluating measurement results, it is also relevant to compare differences with the smallest change that can be detected subjectively. For EDT, this so-called subjective difference limen is about 5% [9.7].

9.3.2 Clarity

Clarity describes the degree to which every detail of the performance can be perceived as opposed to everything being blurred together by later-arriving, reverberant sound components. Thus, *clarity* is to a large extent a property complementary to *reverberance*.

When reflections are delayed by no more than 50–80 ms relative to the direct sound, the ear will integrate these contributions and the direct sound together, which means that we mainly perceive the effect as if the clear, original sound has been amplified relative to the later, reverberant energy. Thus, an objective parameter that compares the ratio between energy in the impulse response before and after 80 ms has been found to be a reasonably good descriptor of *clarity*

$$C = 10 \log_{10}\left[\int_0^{80\,\text{ms}} h^2(t)\,\text{d}t \Big/ \int_{80\,\text{ms}}^{\infty} h^2(t)\,\text{d}t\right] . \tag{9.7}$$

The higher the value of C, the more the early sound dominates, and the higher the impression of *clarity*.

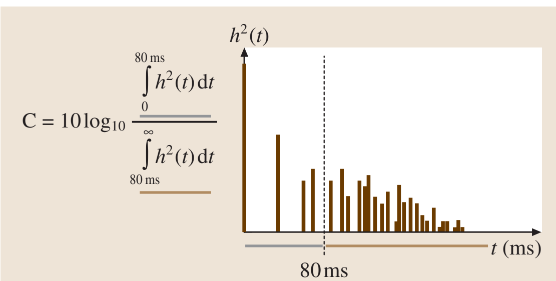


Fig. 9.6 The definition of C: the ratio between early and late energy in the impulse response

With exponential decay, the expected value of C becomes a function of T alone:

$$C_{\text{exp}} = 10 \log_{10}\left[\exp\left(\frac{1.104}{T}\right) - 1\right] \text{dB} . \tag{9.8}$$

The subjective difference limen for C (the equal best difference perceivable) is about 0.5 dB.

The definition of C is illustrated in Fig. 9.6.

Another parameter, which is also used to describe the balance between early and late sound or the balance between *clarity* and *reverberance*, is the center time t_s, which describes the center of gravity of the squared impulse response:

$$t_\text{s} = \int_0^{\infty} t h^2(t)\,\text{d}t \Big/ \int_0^{\infty} h^2(t)\,\text{d}t . \tag{9.9}$$

A low value of t_s corresponds to a clear sound, whereas higher values indicate dominance of the late, reverberant energy. The main advantage of t_s is that it does not contain a sharp time limit between early and late energy, since this sharp distinction is not justified by our knowledge about the functioning of our hearing system. The subjective difference limen for t_s is about 10 ms, and the expected diffuse field value is simply given by:

$$t_{\text{s,exp}} = \frac{T}{13.8} \tag{9.10}$$

9.3.3 Sound Strength

The influence of the room on the perceived *loudness* is another important aspect of room acoustics. A relevant measurement of this property is simply the difference in dB between the level of a continuous, calibrated sound source measured in the room and the level the same source generates at 10 m distance in anechoic surroundings. This objective measure called the (relative) strength G can also be obtained from impulse response recordings from the ratio between the total energy of the impulse response and the energy of the direct sound with the latter being recorded at a fixed distance (10 m) from the impulsive sound source:

$$G = 10 \log_{10} \frac{\int_0^{\infty} h^2(t)\,\text{d}t}{\int_0^{t_\text{dir}} h^2_{10\,\text{m}}(t)\,\text{d}t} . \tag{9.11}$$

Here the upper integration limit in the denominator t_dir should be limited to the duration of the direct sound pulse (which in practice will depend on the bandwidth selected). A distance different from 10 m can be used, if a correction for the distance attenuation is applied as well.

The expected value of G according to diffuse field theory becomes a function of T as well as of the room volume, V:

$$G_{\text{exp}} = 10\log_{10}\left(\frac{T}{V}\right) + 45\,\text{dB}\,. \tag{9.12}$$

The subjective difference limen for G is about 1.0 dB. The definition of G is illustrated in Fig. 9.7.

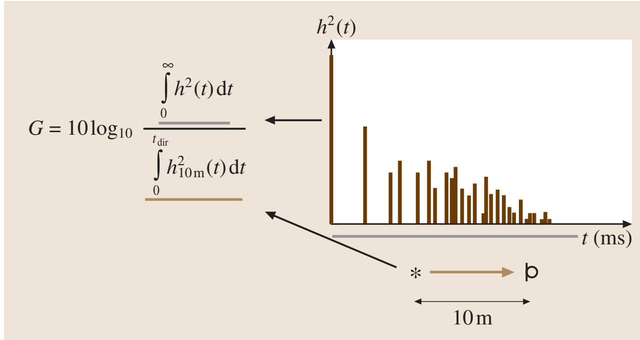


Fig. 9.7 The definition of strength G: the total energy in the impulse response measured relative to the direct sound level at 10 m distance from the source

9.3.4 Measures of Spaciousness

Spaciousness is the feeling that the sound is arriving from many different directions in contrast to a monophonic impression of all sound reaching the listener through a narrow opening. It is now clear that there are two aspects of *spaciousness*, both of which are attractive, particularly when listening to music:

- *apparent source width* (ASW): the impression that the sound image is wider than the visual, physical extent of the source(s) on stage. This should not be confused with localization errors, which of course should be avoided.

and

- *listener envelopment* (LEV): the impression of being inside and surrounded by the reverberant sound field in the room.

Both aspects have been found to be dependent on the direction of incidence of the impulse response reflections. When a larger portion of the early reflection energy (up to about 80 ms) arrives from lateral directions (from the sides) the ASW increases. When the level of the late, lateral reflections is high, strong LEV results.

The lateral components of the impulse response energy can be recorded using a figure-of-eight microphone with the sensitive axis held horizontal and perpendicular to the direction towards the sound source (so that the source lies in the deaf plane of the microphone). For measurement of the lateral energy fraction (LEF), the early part (up to 80 ms) of this lateral sound energy is compared with the energy of the direct sound plus all early reflections picked up by an ordinary omnidirectional microphone:

$$\text{LEF} = \int_{t=5\,\text{ms}}^{t=80\,\text{ms}} h_1^2(t)\,dt \Bigg/ \int_{t=0\,\text{ms}}^{t=80\,\text{ms}} h^2(t)\,dt\,, \tag{9.13}$$

where h_l is the impulse response pressure recorded with a figure-of-eight microphone, whereas h is captured through the (usual) omnidirectional microphone.

It is mainly the energy at low and mid frequencies that contribute to the *spaciousness*. Consequently, LEF is normally averaged over the four octaves 125–1000 Hz. The higher the value of LEF, the wider the ASW. The literature contains many data on LEF values in different, existing concert halls.

LEF does not have an expected value related to T. In a completely diffuse field, LEF would be constant with a value of 0.33, which is higher than that normally found in real halls. The subjective difference limen for LEF is about 5%.

The definition of LEF is illustrated in Fig. 9.8.

The ASW aspect of *spaciousness* is not only dependent on the ratio between early lateral and total early

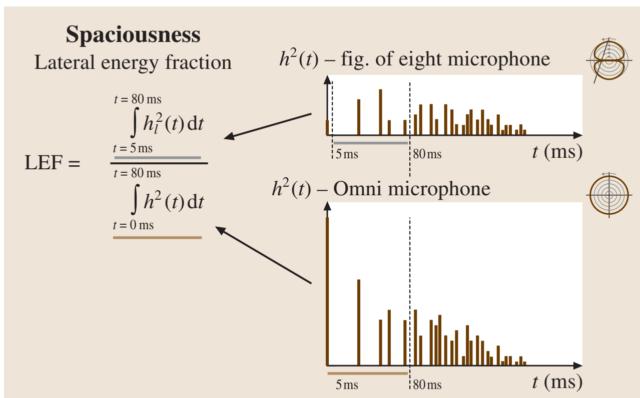


Fig. 9.8 The definition of lateral energy fraction LEF: the ratio between early reflection energy arriving from the sides and total early energy

sound; but also on the total level of the sound. The higher the G value (and the louder the sound source), the broader the acoustic image of the source. However, at the time of writing, there is no solid way of incorporating the influence of level into the objective measure.

Listener envelopment seems to be determined mainly by the spatial distribution and the level of the late reflections (arriving after 80 ms). A parameter called *late lateral strength* (LG) relating the late lateral energy to the direct sound at 10 m distance from the source has been proposed to measure this quality [9.8]:

$$\mathrm{LG} = \int_{t=80\,\mathrm{ms}}^{t=\infty} h_{\mathrm{l}}^2(t)\,\mathrm{d}t \Bigg/ \int_{t=0\,\mathrm{ms}}^{t=t_{\mathrm{dir}}} h_{10\,\mathrm{m}}^2(t)\,\mathrm{d}t\,. \quad (9.14)$$

LG is likely to be included in the ISO 3382 standard (from 2007); but so far LG has only been found correlating with subjective responses in laboratory experiments using synthetic sound fields. Therefore, it may be wise not to put too much emphasis on this parameter until its value has also been confirmed by experiments in real halls.

In contrast to reflections arriving in the median plane, lateral reflections will cause the instantaneous pressure at the two ears of the listener to be different. Such a dissimilarity of the signals at the two ears can also be measured by means of the interaural cross-correlation coefficient IACC:

$$\mathrm{IACC}_{t_1,t_2} = \max\left|\frac{\int_{t_1}^{t_2} h_{\mathrm{L}}(t)h_{\mathrm{R}}(t+\tau)\,\mathrm{d}t}{\sqrt{\int_{t_1}^{t_2} h_{\mathrm{L}}^2(t)\,\mathrm{d}t \int_{t_1}^{t_2} h_{\mathrm{R}}^2(t)\,\mathrm{d}t}}\right| \quad (9.15)$$

in which t_1 and t_2 define the time interval of the impulse response within which the correlation is calculated, h_{L} and h_{R} represent the impulse response measured at the entrance of the left and right ear, respectively, and τ is the interval within which we search for the maximum of the correlation. Normally, the range of τ is chosen between -1 and $+1$ ms, covering roughly the time it takes for sound to travel the distance from one ear to the other. As the correlation is normalized by the product of the energy of h_{L} and h_{R}, the IACC will take a value between 0 and 1. Results are often reported as $1-$IACC in order to obtain a value increasing with increasing dissimilarity, corresponding to an increasing impression of *spaciousness*. If the time interval for the calculation (t_1, t_2) is (0 ms, 100 ms), then the IACC will measure the ASW, while a later interval $(t_1, t_2) = (100\,\mathrm{ms}, 1000\,\mathrm{ms})$ may be used to describe the LEV. The literature on measured values in halls mainly contain data on IACC related to the 0–100 ms time interval.

Although the LEF and IACC parameters relate to the same subjective quality, they are not highly correlated in practice. Another puzzling fact is that LEF and IACC emphasize different frequency regions being of importance. LEF is primarily measured in the four lowest octaves, 125 Hz, 250 Hz, 500 Hz and 1000 Hz while IACC should rather be measured in the octave bands above 500 Hz. IACC values would always be high in the lower octaves, because the distance between the ears (< 30 cm) is small compared to 1/4 of the wave length (≈ 70 cm at 125 Hz).

9.3.5 Parameters Relating to Timbre or Tonal Color

Timbre describes the degree to which the room influences the frequency balance between high, middle and low frequencies, i. e. whether the sound is *harsh*, *bright*, *hollow*, *warm*, or whatever other adjective one would use to describe *tonal color*. Traditionally, a graph of the frequency variation of T (per 1/1 or 1/3 octave) has been used to indicate this quality; but a convenient single-number parameter intended to measure the warmth of the hall has been suggested [9.9]: the bass ratio (BR) given by:

$$\mathrm{BR} = \frac{T_{125\,\mathrm{Hz}} + T_{250\,\mathrm{Hz}}}{T_{500\,\mathrm{Hz}} + T_{1000\,\mathrm{Hz}}}\,. \quad (9.16)$$

Likewise, a *treble ratio* (TR) can be formed as:

$$\mathrm{TR} = \frac{T_{2000\,\mathrm{Hz}} + T_{4000\,\mathrm{Hz}}}{T_{500\,\mathrm{Hz}} + T_{1000\,\mathrm{Hz}}}\,. \quad (9.17)$$

However, in some halls, a lack of bass sound is experienced in spite of high T values at low frequencies. Therefore EDT or perhaps G versus frequency would be a better – and intuitively more logical – parameter for measurement of *timbre*. Likewise, BR or TR could be based on G rather than T values.

Besides the subjective parameters mentioned above, a quality called *intimacy* is also regarded as being important when listening in auditoria. *Beranek* [9.9] originally suggested this quality to be related to the initial time delay gap ITDG; but this has not been experimentally verified (At the time of writing, Beranek no longer advocates this idea but mentions the observation that, if the ITDG exceeds 45 ms, the hall has no intimacy [9.10]). It is likely that *intimacy* is related to a combination of some of the other parameters already mentioned, such as a high sound level combined with a clear and enveloping sound, as will naturally be experienced fairly close to the source or in small rooms.

9.3.6 Measures of Conditions for Performers

In rooms for performance of music it is also relevant to consider the acoustic conditions for the musicians, partly because it is important to ensure that the musicians are given the best working conditions possible and partly because the product as heard by the audience will also be better if the conditions are optimal for the performers.

Musicians will be concerned about *reverberance* and *timbre* as mentioned above; but also about at least two aspects which are unique for their situation: *ease of ensemble* and *support* [9.11] and [9.12].

Ease of ensemble relate to how well musicians can hear – and play together with – their colleagues. If ensemble is difficult to achieve, the result – as perceived by musicians and listeners alike – might be less precision in rhythm and intonation and a lack of balance between the various instruments. *Ease of ensemble* has been found to be related to the amount of early reflection energy being distributed on the stage. So far, the only widely recognized objective parameter suggested for objective measurement of this quality is *early support* $\mathrm{ST_{early}}$.

$\mathrm{ST_{early}}$ is calculated from the ratio between the early reflection energy and the direct sound in an impulse response recorded on the stage with a distance of only one meter between the source and receiver:

$$\mathrm{ST_{early}} = 10\log_{10}\frac{\int_{20\,\mathrm{ms}}^{100\,\mathrm{ms}} h_{1\,\mathrm{m}}^2(t)\,\mathrm{d}t}{\int_{0\,\mathrm{ms}}^{t_{\mathrm{dir}}} h_{1\,\mathrm{m}}^2(t)\,\mathrm{d}t}\,. \tag{9.18}$$

Support relates to the degree to which the room supports the musicians' efforts to create the tone on their own instruments, whether they find it easy to play or whether they have to force their instruments to fill the room. Having to force the instrument leads to playing fatigue and inferior quality of the sound.

Support is also related to the amount of reflected energy on the stage measured using only a one meter source/microphone distance. For measurement of *support*, however, one needs to consider also the later reflections. This is because for many types of instruments, (especially strings), the early reflections will be masked by the strong direct sound from the instrument itself. Consequently, it seems relevant to define a measure

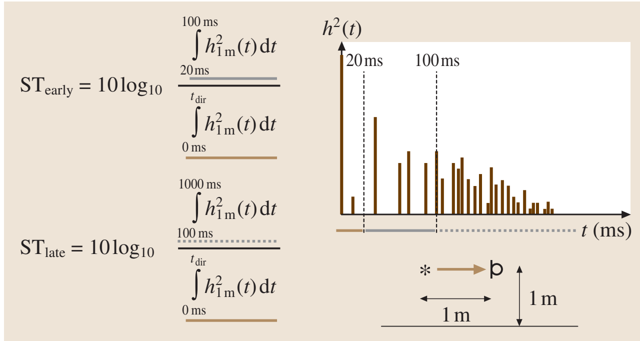


Fig. 9.9 The definition of early support $\mathrm{ST_{early}}$ and late support $\mathrm{ST_{late}}$, the ratio between early and late reflection energy respectively and the direct sound in the impulse response. Both are measured at 1 m distance from the source

such as *late support*, $\mathrm{ST_{late}}$:

$$\mathrm{ST_{late}} = 10\log_{10}\frac{\int_{100\,\mathrm{ms}}^{1000\,\mathrm{ms}} h_{1\,\mathrm{m}}^2(t)\,\mathrm{d}t}{\int_{0\,\mathrm{ms}}^{t_{\mathrm{dir}}} h_{1\,\mathrm{m}}^2(t)\,\mathrm{d}t} \tag{9.19}$$

which relates to late response (after 100 ms) from the room to the sound emitted.

$\mathrm{ST_{early}}$ and $\mathrm{ST_{late}}$ are typically measured in the four octaves 150 Hz, 500 Hz, 1000 Hz, 2000 Hz. The lowest octave is omitted primarily because it is not possible to isolate the direct sound from the early reflections in a narrow band measurement. The definitions of early and late support are illustrated in Fig. 9.9.

Also the support parameters are planned to be included in the ISO 3382 standard (from 2007); but like LG the experiences and amount of data available from existing halls are still rather limited.

The amount of *reverberance* on the stage may be measured by means of EDT, (with a source–receiver distance of, say, 5 m or more).

9.3.7 Speech Intelligibility

All the objective parameters mentioned above (except the basic T parameter), are mainly relevant in larger auditoria intended for performance of music. In auditoria used for speech, such as lecture halls or theaters, the influence of the acoustics on intelligibility is a major issue.

Currently, the most common way to assess objectively *speech intelligibility* in rooms is by measurement of the speech transmission index STI.

Theory:
Modulation reduction function of T and S/N

$$m(F) = \frac{1}{\sqrt{1+\left(2\pi F\,\dfrac{T}{13{,}8}\right)}}\;\frac{1}{1+10^{(-S/N)/10}}$$

$$(S/N)_{\text{app}} = 10\log_{10}\left(\frac{\text{m}}{1-\text{m}}\right)$$

$$\text{RASTI} = \left((\text{S/N})_{\text{app}} + 15\right)/30$$

RASTI value	<0.30	0.30–0.45	0.45–0.60	0.60–0.75	>0.75
Intelligibility	Bad	Poor	Fair	Good	Excellent

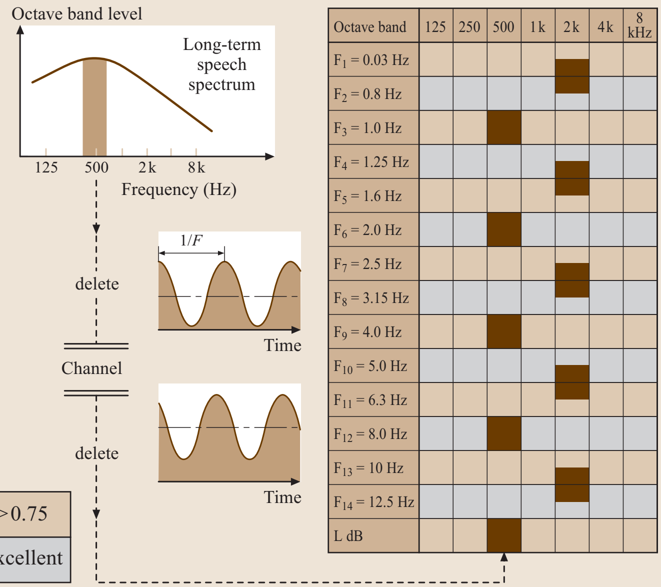


Fig. 9.10 Illustration of the theory and principle in the measurement of STI or RASTI. The scale for the evaluation of RASTI values is shown at the bottom (after [9.13])

As illustrated in Fig. 9.10, this measure is based on the idea that speech can be regarded as an amplitude-modulated signal in which the degree of modulation carries the speech information. If the transmission path adds noise or reverberation to the signal, the degree of modulation in the signal will be reduced, resulting in reduced intelligibility.

The modulation transfer is tested by emitting noise in seven octave bands, each modulated with 14 different modulation frequencies as listed in the table in Fig. 9.10 and then calculating the ratio between the original and the received degree of modulation, the modulation reduction factor, in each of these 98 combinations. A weighted average of the modulation reduction factor then results in a number between 0 and 1, corresponding to very poor and excellent conditions respectively.

A faster measurement method using only two carrier bands of noises and four plus five modulation frequencies (indicated by the dark squares in Fig. 9.10) is called rapid STI (RASTI). The STI/RASTI method is described in an International Electrotechnical Commission (IEC) standard: IEC 286-16.

Although the original method of STI or RASTI measurement employs modulated noise signals, it is also possible to calculate the modulation reduction factor from the impulse response. Thus, the modulation reduction factor versus modulation frequency F, which is called the modulation transfer function (MTF), can be found as the Fourier transform of the squared impulse response normalized by the total impulse response energy.

According to *Bradley* [9.14], a simpler parameter, called the *useful-to-detrimental sound ratio* U_{80} is equally suitable for measurement of speech intelligibility. U_{80} is simply a modified version of the clarity parameter defined in Sect. 9.3.2, in which a correction is made according to the ratio between the levels of speech and background noise.

9.3.8 Isn't One Objective Parameter Enough?

It is important to notice that all the parameters mentioned above – apart from T – may show large differences between different seats within the same hall. Actually, differences within a single hall are sometimes as large (both objectively and subjectively) as the differences between two different halls.

It should also be mentioned that the objective parameters related to the three subjectively different aspects: *reverberance/clarity*, *loudness* and *spaciousness* show low mutual correlation when measured in different seats or in different halls. In other words, they behave as orthogonal factors and do not monitor the same properties of the sound fields. Consequently, it is obviously necessary to apply one objective parameter for each of these subjective aspects.

It is also clear that different listeners will judge the importance of the various subjective parameters differently. While some may base their judgment primarily on *loudness* or *reverberance* others may be more concerned about *spaciousness* or *timbre*.

For these reasons, one should be very sceptical when people attempt to rank concert halls along a single, universal, one-dimensional quality scale or speak about the best hall in the world. Fortunately, the diversity of room acoustic design and the complexity of our perception does not justify such simplifications.

9.3.9 Recommended Values of Objective Parameters

In view of the remarks above, it may be regarded risky to recommend values for the various objective acoustic parameters. On the other hand, there is a long tradition of suggesting optimal values for reverberation time T as a function of hall volume and type of performance. Besides, most of the other parameters will seldom deviate drastically from the expected value determined by T. As we shall see later, this deviation is primarily influenced by how we shape the room to control early reflections. Designing for strong early reflections can increase *clarity*, *sound strength* and *spaciousness*. Current taste regarding concert hall acoustics for classical music seems to be in favor of high values for these factors in addition to strong *reverberance*. This could lead to suggested ranges for the various parameters in small and large halls, as shown in Table 9.1. These values relate to empty halls with well-upholstered seating, assuming that, when fully occupied with musicians and audience, the T values will drop by no more than say 0.2 s. The reason for relating the values to the unoccupied condition is that it is seldom possible to make acoustic measurements in the occupied state and almost all reference data existing regarding parameters other than T are from unoccupied halls. On the other hand, suggesting well-upholstered seats is a sound recommendation in most cases, which will ensure only small changes depending on the degree of occupancy. This will also justify extrapolation of the unoccupied values of the other parameters to the occupied condition, as mentioned later in this section.

The values listed in the table were arrived at by first choosing values for T ensuring high *reverberance*, while the correlative values for EDT, G and C stem from the diffuse field expected values; but these have been slightly changed to promote high *clarity* and high *levels*. As this is particularly important in larger halls, it has been suggested to use an EDT value 0.1 s lower than T ($=\mathrm{EDT}_{\mathrm{exp}}$) and C values 1 dB higher than C_{exp} in a 2500 m^3 hall, but 0.2 s less than T and 2 dB higher than C_{exp}, respectively, in the 25 000 m^2 hall. Likewise, it is suggested to use G values 2 dB less than G_{exp} in the small halls; but only 1 dB less than or even equal to G_{exp} in large halls. For halls of size between the 2500 and 25 000 m^3 listed in the table, one may of course interpolate to taste. Please note that, in general, G is found to be 2–3 dB lower that G_{exp} (Sect. 9.5.6) so the G values suggested here are actually 1–2 dB higher than those found in "normal" halls.

In the occupied condition, one may expect EDT, C and G to be reduced by amounts equal to the difference between the expected values calculated using T for the empty and occupied conditions respectively. In any case,

Table 9.1 Suggested position-averaged values of objective room acoustic parameters in unoccupied concert halls for classical music. It is assumed that the seats are highly absorptive, so that T will be reduced by no more than 0.2 s when the hall is fully occupied. 2.4 s should be regarded as an upper limit mainly suitable for terraced arena or DRS halls with large volumes per seat (Sect. 9.7.3), whereas 2.0 s will be more suitable for shoe-box-shaped halls, which are often found to be equally reverberant with lower T values and lower volumes per seat

Parameter	Symbol	Chamber music	Symphony
Hall size	V/N	2500 m^3/300 seats	25 000 m^3/2000 seats
Reverberation time	T	1.5 s	2.0–2.4 s
Early decay time	EDT	1.4 s	2.2 s
Strength	G	10 dB	3 dB
Clarity	C	3 dB	−1 dB
Lateral energy fraction	LEF	0.15–0.20	0.20–0.25
Interaural cross correlation	1 − IACC	0.6	0.7
Early support	$\mathrm{ST}_{\mathrm{early}}$	−10 dB	−14 dB

if the empty–occupied difference in T is limited to 0.2 s, as suggested here, then the difference in these parameters will be fairly limited, with changes in both C and G being less than one dB - given that the audience does not introduce additional grazing incidence attenuation of direct sound and early reflections (Sect. 9.6.2).

For opera, one may aim towards the same relationships between the diffuse field expected and the desired values of EDT, C and G, as mentioned above for symphonic halls; but the goal for the reverberation time will normally be set somewhat lower, 1.4–1.8 s in order to obtain a certain intelligibility of the libretto and perhaps to make breaks in the music sound more dramatic (Sect. 9.7.2).

For rhythmic music, T values of 0.8–1.5 s, depending on room size, are appropriate, and certainly the value should not increase towards low frequencies in this case.

In very large rock venues such as sports arenas, it may actually be very difficult to get T below 3–5 s; but even then the conditions can be satisfactory with a properly designed sound system. The reason is that in such large room volumes the level of the reverberant sound can often be controlled, if highly directive loudspeakers are being used. In these spaces, the biggest challenge is often to avoid highly delayed echoes, which are very annoying. For amplified music, only recommendations for T are relevant, because the acoustic aspects related to the other parameters will be determined primarily by the sound system (Sect. 9.7.5).

In order to ensure adequate speech intelligibility in lecture halls and theaters, values for STI/RASTI should be at least 0.6. In reverberant houses of worship values higher than 0.55 are hard to achieve even with a well-designed loudspeaker system.

9.4 Measurement of Objective Parameters

Whenever the acoustic specifications of a building are of importance we need to be able to predict the acoustic conditions during the design process as well as document the conditions after completion. Therefore, in this and the following sections, techniques for measurement and prediction of room acoustic conditions will be presented.

In Sect. 9.2.1 it was claimed that the impulse response contains all relevant information about the acoustic conditions in a room. If this is correct, it must also be true that all relevant acoustic parameters can be derived from impulse response measurements. Fortunately, this has also turned out to be the truth.

9.4.1 The Schroeder Method for the Measurement of Decay Curves

Originally, reverberation decays were recorded from the decay of the sound level following the termination of a stationary sound source. However, *Schroeder* [9.15] has shown that the reverberation curve can be measured with increased precision by *backwards* integration of the impulse response, $h(t)$, as follows:

$$R(t) = \int_t^\infty h^2(t)\,\mathrm{d}t = \int_0^\infty h^2(t)\,\mathrm{d}t - \int_0^t h^2(t)\,\mathrm{d}t \quad (9.20)$$

in which $R(t)$ is equivalent to the decay of the squared pressure decay. The method is called backwards integration because the fixed upper integration limit, *infinite time*, is not known when we start the recording. Therefore, in the days of magnetic tape recorders, the integration was done by playing the tape backwards. In this context *infinite time* means the duration of the recording, which should be comparable with the reverberation time. Traditional noise decays contain substantial, random fluctuations because of interference by the different eigenmodes present within the frequency range of the measurement (normally 1/3 or 1/1 octaves). These fluctuations will be different each time the measurement is carried out because the modes will be excited with random phase by the random noise (and the random time of the noise interruption). When, instead, the room is exited by a repeatable impulse, the response will be repeatable as well without such random fluctuations. In fact, Schroeder showed that the ensemble average of interrupted noise decays (recorded in the same source and receiver positions) will converge towards $R(t)$ as the number of averages goes towards infinity.

In (9.20), the right-hand side indicates that the impulse response decay can be derived by subtracting the running integration (from time zero to t) from the total energy. Contrary to the registration of noise decays, this means that the entire impulse response must be stored before the decay function is calculated. However, with modern digital measurement equipment, this is not a problem. From the $R(t)$ data, T is calculated by linear regression as explained in Sect. 9.3.1. As the EDT is evaluated from a smaller interval of the decay curve than T, the EDT is more sensitive to random decay fluctuations than is T. Therefore, EDT should never be derived from interrupted noise decays – only from integrated impulse responses.

9.4.2 Frequency Range of Measurements

T may be measured per 1/3 octave; but regarding the other objective parameters mentioned in this chapter, it makes no sense to use a frequency resolution higher than 1/1 octave. The reason is that the combination of narrow frequency intervals and narrow time windows in the other parameters will lead to extreme fluctuations with position of the measured values due to the acoustic relation of uncertainty. Such fluctuations have no parallel in our subjective experience. *Bradley* [9.16] has shown that, with 1/1 octave resolution, a change in microphone position of only 30 cm can already cause fluctuations in the clarity C and strength G of about 0.5–1 dB, which is equal to the subjective difference limen. Higher fluctuations would not make sense, as normally we do not experience an audible change when moving our head 30 cm in a concert hall.

The frequency range in which room acoustic measurements are made is normally the six octaves from 125 to 4000 Hz, but it may be extended to 63–8000 Hz when possible. Particularly for reverberation measurements in halls for amplified music, the 63 Hz octave is important.

9.4.3 Sound Sources

With the possibility of deriving all of the objective parameters described above from impulse response measurements, techniques for creating impulses are described briefly in the following.

Basically, the measurements require an omnidirectional sound source emitting short sound pulses (or other signals that can be processed to become impulses) covering the frequency interval of interest, an omnidirectional microphone and a medium for the storage of the electrical signal generated by the microphone.

Impulsive sources such as pistols firing blank cartridges, bursting balloons, or paper bags and electrical spark sources may be used; but these sources are primarily used for noncritical measurements, as their directivity and acoustic power are not easy to control.

For measurements requiring higher precision and repeatability, omnidirectional loudspeakers are preferable, as the power and directivity characteristics of loudspeakers are more stable and well defined. Omnidirectional loudspeakers are often built as six units placed into the sides of a cube, 12 units placed into a dodecahedron or 20 units in an icosahedron, see Fig. 9.11. The requirements regarding omnidirectivity of the source are described in the ISO 3382 standard.

Loudspeakers, however, cannot always emit sufficient acoustic power if an impulsive signal is applied directly. Therefore, special signals of longer duration such as maximum-length sequences or tone sweeps have been developed, which have the interesting property that their autocorrelation functions are band-limited delta pulses. This means that the loudspeaker can emit a large amount of energy without challenging its limited peak-power capability while impulse responses with high time resolution can still be obtained afterwards through post-processing (a correlation/convolution process) of the recorded noise or sweep responses.

In certain cases, other source directivity patterns than omnidirectional can be of interest. For example, a directivity like that of a human talker may be relevant if the measurement is related to the intelligibility of speech or to measurements on the stage in an opera house. A suitable talker/singer directivity can be obtained by using a small loudspeaker unit (membrane diameter about three to four inches) mounted in a closed box of size comparable to that of the human head. Using only one of the loudspeakers in a cube or dodecahedron speaker of limited size is another possibility.

9.4.4 Microphones

For most of the parameters mentioned, the microphone should be omnidirectional, while omni plus figure-of-eight directivity and dummy heads are used when the impulse responses are to be used for the calculation of LEF and IACC.

Three-dimensional intensity probes, sound field microphones or microphone arrays have also been suggested in attempts to obtain more-complete information about the amplitude and direction of incidence of individual reflections [9.17] or for wave field analysis [9.18]. However, such measurements are mainly for research or diagnostic purposes. As such, they do not fulfill the goal set up in Sect. 9.1 of reducing the information to what is known to be subjectively relevant. A selection of relevant microphones are shown in Fig. 9.12.

9.4.5 Signal Storage and Processing

For the storage of the impulse responses the following choices appear.

- The hard disk on a personal computer (PC) equipped with a sound card or special analog-to-digital (A/D) converter card

- Real-time (1/3 and 1/1 octave) analyzers or sound level meters with sufficient memory for storage of a series of short time-averaged root-mean-square (RMS) values
- Fast Fourier transform (FFT) analysers capable of recording adequately long records
- Tape recorders, preferably a digital audio tape (DAT) recorder, or even MP3 recorders given that the data reduction does not deteriorate the signal.

Of these, real-time analyzers and sound level meters will process the signal through an RMS detector before storage, which means that not the full signal, but only energy with a limited time resolution will be recorded.

Today, calculation of the objective parameters will always be carried out by a PC or by a computer built into a dedicated instrument. A large number of PC-based systems and software packages particularly suited for room acoustic measurements are available. Some systems come with a specific analog-to-digital (A/D), digital-to-analog (D/A) and signal-processing card, which must be installed in or connected to the PC, while others can use the sound card already available in most PCs as well as external sound cards for hobby music use.

9.5 Prediction of Room Acoustic Parameters

Beyond theoretical calculation of the reverberation time, the prediction techniques presented in this section range from computer simulations and scale model measurements to simple linear models based on empirical data collected from existing halls.

In all cases, the main objective is the prediction of the acoustic conditions described in terms of the acoustic parameters presented in Sect. 9.3. Just as we can record the sound in a hall, both scale models and numerical computer simulations can also provide *auralizations*, which are synthesized audible signals that allow the client and architect to listen to and participate in the acoustic evaluation of proposed design alternatives. Such audible simulations will often have much greater impact and be more convincing than the acoustician's verbal interpretation of the numerical results. Therefore, the acoustician must be very careful to judge the fidelity and the degree of realism before presenting auralizations.

9.5.1 Prediction of Reverberation Time by Means of Classical Reverberation Theory

Reverberation time as defined in Sect. 9.3.1 is still the most important objective parameter for the characterization of the acoustics of a room. Consequently, the prediction of T is a basic tool in room acoustic design. Calculation of reverberation time according to the Sabine equation was the first attempt in this direction and today is still the most commonly used. However, during the 100 years since its invention, several other reverberation time formulae have been suggested (by Eyring, Fitzroy, Millerton and Sette, Metha and Mulholland Kuttruff and others), all with the purpose of correcting some obvious shortcomings of the Sabine method.

The Sabine equation has been described in Sect. 11.1.4. In larger rooms, the absorption in the air plays a major role at high frequencies. If we include this factor, the Sabine equation reads:

$$T = \frac{0.161V}{S\alpha^* + 4mV}\,, \qquad (9.21)$$

where V is the room volume, S is the total area of the room surfaces, α^* is the area-weighted average absorption coefficient of the room surfaces

$$\alpha^* = \overline{\alpha} = \frac{1}{S}\sum_i S_i\alpha_i \qquad (9.22)$$

and m accounts for the air absorption, which is a function of the relative humidity and increases rapidly with fre-

Fig. 9.11 Two types of omnidirectional sound sources for room acoustic measurements: blank pistol (in the *raised hand of the seated person*) and an omnidirectional loudspeaker (icosahedron with 20 units)

Table 9.2 Air absorption coefficient m for different frequencies and relative humidity; valid for an air temperature of 20 °C (after [9.19])

Relative humidity (%)	0.5 kHz	1 kHz	2 kHz	4 kHz	8 kHz	Air absorption
40	0.4	1.1	2.6	7.2	23.7	$10^{-3}\,m^{-1}$
50	0.4	1.0	2.4	6.1	19.2	$10^{-3}\,m^{-1}$
60	0.4	0.9	2.3	5.6	16.2	$10^{-3}\,m^{-1}$
70	0.4	0.9	2.1	5.3	14.3	$10^{-3}\,m^{-1}$
80	0.3	0.8	2.0	5.1	13.3	$10^{-3}\,m^{-1}$

quency [9.19]. Values of m versus frequency and relative humidity are listed in Table 9.2.

The Sabine equation assumes the absorption to be evenly distributed on all surfaces, and that the sound field is *diffuse*, which means that in each point of the room:

1. the energy density is the same, and
2. there is equal probability of the sound propagating in all directions.

Among the alternative reverberation formulae, only the Eyring method will be described here, as it may give more-accurate predictions in cases where the room is highly absorptive.

In contrast to Sabine, the Eyring theory assumes that the sound field is composed of plane sound waves that lose a fraction of their energy equal to the absorption coefficient of the surface whenever the wave hits a surface in the room. Thus, after n reflections, the energy in all wavefronts and the average energy density in the room has been reduced to $(1-\overline{\alpha})^n$ times the original value. The average distance of propagation during this process is $l_m n = ct$, in which l_m is the *mean free path*, equal to the average distance traveled by the wave between two reflections, and t is the time elapsed since the wave propagation started. *Kosten* [9.20] has shown, that regardless of the room shape, l_m equals $4V/S$, if all directions of propagation are equally probable. These assumptions lead to the Eyring formula, which can be expressed by substituting α^* in (9.21) by

$$\alpha^* = \alpha_e = \ln\left(\frac{1}{1-\overline{\alpha}}\right) \tag{9.23}$$

For low values of $\overline{\alpha}$, $\overline{\alpha}$ and α_e are almost identical, causing the Sabine and the Eyring formulae to give nearly identical results; but the difference becomes noticeable when $\overline{\alpha}$ exceeds about 0.3. In general, α_e will always be larger than $\overline{\alpha}$, leading to $T_{Sabine} > T_{Eyring}$. In the extreme case of the mean absorption approaching 1, α_e converges towards ∞, whereby $T_{Eyring} \to 0$ as one would expect for a totally absorbing room. However, T_{Sabine} converges towards the finite positive value $0.16\,V/S$, which of course is unrealistic.

Both the Sabine and the Eyring theories suffer from at least two shortcomings. Neither of them consider the actual distribution of the free paths (see the comments on the *mean free path* above), which is highly dependent on the room shape, nor the often very uneven distribution of the absorption in the room, which causes different directions of sound propagation not to be equally probable. Typical situations where these factors become important are rooms with highly absorbing acoustic ceilings and hard walls and floors – or the reverse situation of an auditorium, where almost all absorption is found in the seating. In these situations the energy is certainly not evenly distributed, there is a considerable distribution of the path lengths, and there is a much higher probability for sound energy to travel towards the absorbing surface than away from it.

A common example in which the Sabine and Eyring theories fall short is the simple case of a rectangular room with plane walls, absorbing ceiling and low height compared to length and width. In such a room the main part of the late reverberant energy will be stored in

Fig. 9.12 A selection of microphones types used for recording impulse responses in rooms. *From left*: artificial head (with both internal and external microphones), *sound field microphone* (with four capsules for 3-D recordings of reflections) and a two-channel (stereo) microphone with omnidirectional and figure-of-eight capsules

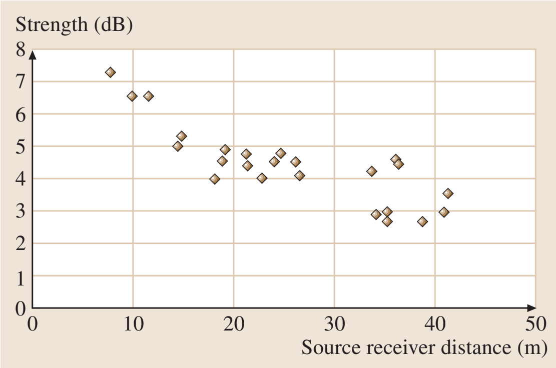


Fig. 9.13 Strength versus distance measured in the Boston Symphony Hall; values averaged over the four octaves 250–2000 Hz. The reverberation distance in this hall is about 5 m

one- and two-dimensional horizontal room modes with long free paths, whereas the three-dimensional modes with shorter free paths will soon have lost their energy by interacting with the absorbing ceiling. Hereby, different modes end up having very different decay rates, leading to bent or double-sloped decay curves and to measured T values often longer than calculated. In fact, only the unnatural situation with an even absorption distribution and all free paths of equal length will result in a straight-line decay and a well-defined T value.

From practical measurements of the distribution of G values, it is also clear that the assumption of an even distribution of the energy density is not met in large auditoria. Contrary to the Sabine theory, which predicts a constant level beyond the reverberation distance (Fig. 11.6), we observe that the level continues to decrease with distance from the source – even far beyond the reverberation distance, where the contribution of the direct sound is negligible. An example of this is seen in Fig. 9.13. (Based on empirical data, simple equations predicting this attenuation of G with distance as a function of room geometry have been derived as described in Sect. 9.5.6.)

Still, rather than relying on another of the reverberation formulas to provide a more correct result, today it may be wiser just to apply the Sabine and perhaps the Eyring equations, keeping in mind their limitations. Thus, *Beranek* [9.21] has suggested that, for Sabine calculations in concert halls, one should apply different sets of seat absorption values depending on the room shape. These values should be measured in nearly the same type of room as the hall in question.

When higher prediction accuracy is required, it is recommended to carry out simulations in a computer model as described in Sect. 9.5.4. However, also with this calculation approach, the question about which absorption values to use is relevant.

9.5.2 Prediction of Reverberation in Coupled Rooms

Auditoria with coupled spaces form another situation in which the decay curve may end up being nonlinear and the energy density will be different in different areas of the room. Typical of such cases are open-plan office floors coupled to an atrium covering several storeys, seating areas in auditoria subdivided by (deep) balconies, theaters with proscenium openings between the auditorium and the stage house, churches subdivided into main and side naves and concert halls with reverberation chambers. Diffuse field theory can be used to illuminate important properties of the reverberation conditions in coupled rooms.

We consider a sound source placed in a room, room 1, with volume V_1 and physical absorption area A_{10}. This room is coupled by an opening with area S to another room, room 2, with volume V_2 and absorption area A_{20}. From diffuse field theory (the energy balance equations) we find that the apparent absorption area of room 1, A_1, depends on the ratio between the absorption area of the attached room A_{20} and the area of the opening S:

$$A_1 = A_{10} + \frac{A_{20} S}{A_{20} + S} \,. \tag{9.24}$$

It is seen that for small openings, $S \ll A_{20}$, $A_1 \approx A_{10} + S$, and for large openings, $S \gg A_{20}$, $A_1 \approx A_{10} + A_{20}$, as we would expect.

As for the reverberation curve, a double slope may be observed, particularly if both source and receiver are placed in the less reverberant of the two rooms. The two reverberation times T_{I} and T_{II} defining the double slope can be described in terms of the reverberation times for each of the two rooms separately if we consider the opening to be totally absorbing:

$$T_1 = \frac{0.161 V_1}{A_{10} + S} \quad \text{and} \quad T_2 = \frac{0.161 V_2}{A_{20} + S} \,. \tag{9.25}$$

However, for our purpose the math gets simpler if we apply damping coefficients: $\delta_1 = 6.9/T_1$ and $\delta_2 = 6.9/T_2$ instead of the reverberation times. Then, with coupling, the two damping coefficients, δ_{I} and δ_{II}, corresponding to the slope in the first and second part of the decay, respectively, can be calculated from:

$$\delta_{\mathrm{I,II}} = \frac{\delta_1 + \delta_2}{2} \pm \sqrt{\frac{(\delta_1 - \delta_2)^2}{4} + \frac{S^2 \delta_1 \delta_2}{(A_{10} + S)(A_{20} + S)}} \tag{9.26}$$

upon which

$$T_{\mathrm{I}} = 6.9/\delta_{\mathrm{I}} \quad \text{and} \quad T_{\mathrm{II}} = 6.9/\delta_{\mathrm{II}} \,. \tag{9.27}$$

Another important feature of the double-slope decay is how far down the knee point appears. If the opening area is large and/or the source or receiver is placed close to the opening, then a substantial exchange of energy can occur and the knee point may appear so early that even the EDT can be influenced by the decay from the coupled room with longer T. If the opening area is small, then the knee point appears late and far down on the curve, in which case the influence of the coupled room can only be perceived after final cords. This is the case with some of the existing concert halls equipped with coupled reverberation chambers. Further descriptions of coupled spaces can be found in [9.5].

9.5.3 Absorption Data for Seats and Audiences

Regardless of which reverberation formula is used, the availability of reliable absorption values is fundamental for the accurate prediction of T as well as of most other room acoustic parameters. Beyond the absorption values of various building materials (as listed in Table 11.1), realistic figures for the absorption of the chairs and seated audience are of special concern, because these elements normally account for most of the absorption in auditoria. The absorption of chairs – both with and without seated people – depends strongly on the degree of upholstery. Average values of absorption for three different degrees of chair upholstery have been published by *Beranek* [9.22] and quoted in Table 9.3. As mentioned earlier, chair absorption values for use specifically in Sabine calculations and for different types of hall geometries can be found in [9.21].

Sometimes audience absorption data are quoted as absorption area per chair or per person. However, as demonstrated by *Beranek* [9.3], the use of absorption coefficients multiplied by the area covered by the seats plus an exposed perimeter correction is more representative than absorption area per person, because audience absorption tends to vary proportionally with the area covered, while the absorption coefficient does not change much if the density of chairs is changed within that area.

Minimizing the total sound absorption area is often attempted in auditoria for classical music in order to obtain a strong, reverberant sound field. One way to achieve this is not to make the row-to-row distance and the width of the chairs larger than absolutely necessary. Another factor is the seat design itself. Here one should aim at a compromise between minimum absorption and still a minimum difference between the absorption of the occupied and empty seat. This is facilitated by back rests not being higher than the shoulders of the seated person and only the surfaces covered by the person being upholstered. The rear side of the back rest should be made from a hard material like varnished wood. On the other hand, a minimum difference between the occupied and empty chair implies that the upholstery on the seat and back rest should be fairly thick (e.g., 80 mm on the

Table 9.3 Absorption coefficients of seating areas in concert halls for three different degrees of upholstery; both empty and occupied (after [9.22])

Octave centre frequency (Hz)	125	250	500	1000	2000	4000
Heavily upholstered chairs Unoccupied	0.70	0.76	0.81	0.84	0.84	0.81
Heavily upholstered chairs Occupied	0.72	0.80	0.86	0.89	0.90	0.90
Medium upholstered chairs Unoccupied	0.54	0.62	0.68	0.70	0.68	0.66
Medium upholstered chairs Occupied	0.62	0.72	0.80	0.83	0.84	0.85
Lightly upholstered chairs Unoccupied	0.36	0.47	0.57	0.62	0.62	0.60
Lightly upholstered chairs Occupied	0.51	0.64	0.75	0.80	0.82	0.83

seat and 50 mm on the back rest), the bottom of the tip-up seats should be perforated or otherwise absorptive, and when the seat is in its vertical position, there should be a wide opening between the seat and the back rest so that the sound can still access the upholstered areas.

9.5.4 Prediction by Computer Simulations

Computer simulations take into account the geometry and actual distribution of the absorption materials in a room as well as the actual source and receiver positions. The results provided are values for all relevant acoustic parameters as well as the generation of audio samples for subjective judgments. Computer simulation is useful not only for the design of concert halls and theaters, but also for large spaces such as factories, open-plan offices, atria, traffic terminals, in which both reverberation, intelligibility and noise-mapping predictions are of interest. Besides, they can be used in *archaeological acoustics* for virtual reconstruction of the acoustics of ancient buildings such as Roman theaters (http://server.oersted.dtu.dk/www/oldat/erato/).

The first computer models appeared in the 1960s, and today a number of commercially available software packages are in regular use by acoustic consultants as well as by researchers all over the world.

In computer simulations the room geometry is represented by a three-dimensional (3-D) computer-aided design (CAD) model. Thus, the geometry information can often be imported from a 3-D model file already cre-

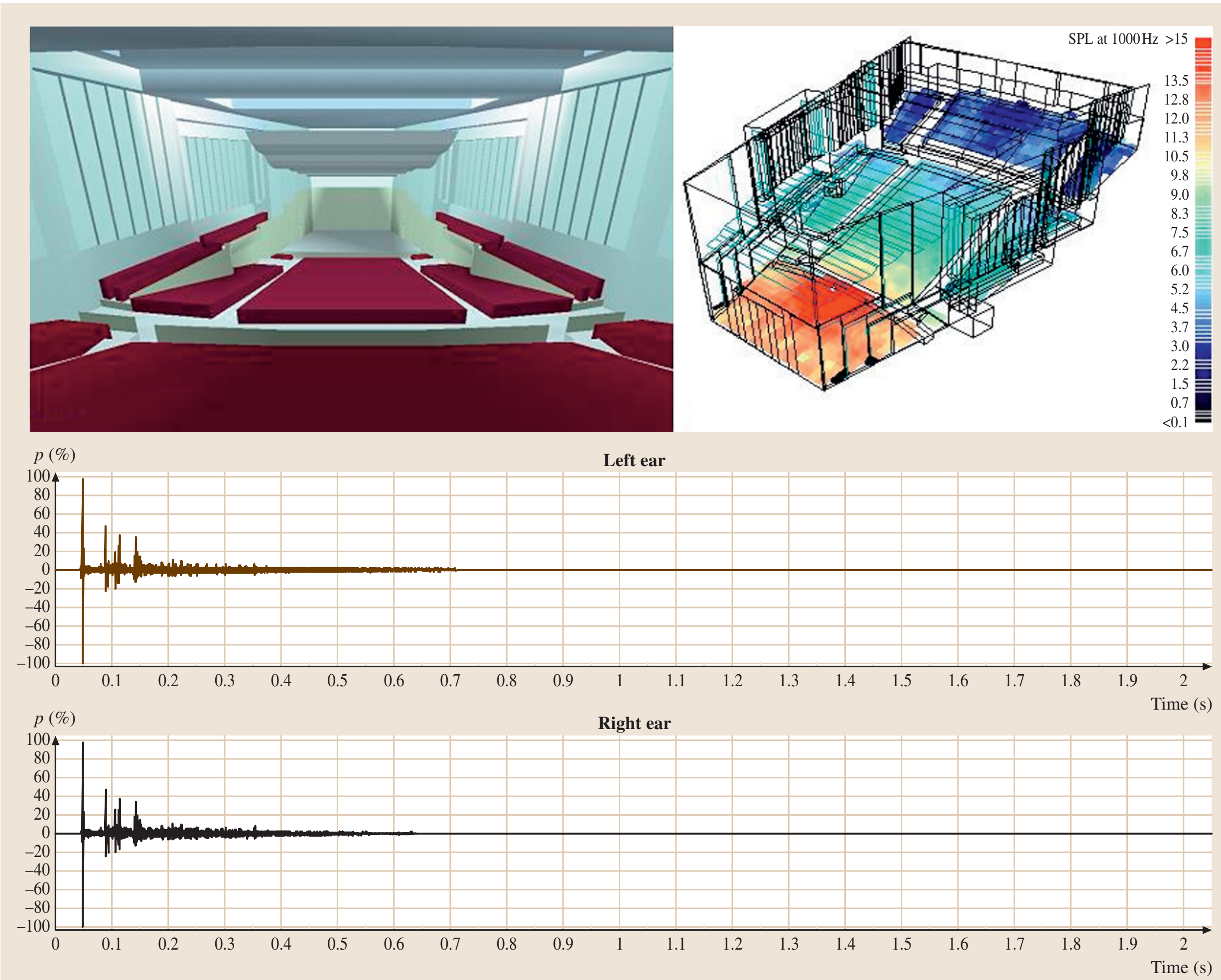


Fig. 9.14 Examples of output from a room acoustic simulation software package (ODEON); see text for explanation

ated by the architects. However, one needs to break up eventual curved surfaces into plane elements and often it is also advisable to simplify details in the geometry provided by the architect. Alternatively, the acoustician can build the room model by writing a file of coordinate points or mathematical expressions for all surface corners. Of course this will be more time consuming in cases of complex room geometry, but some programs offer intelligent programming routines for the modeling of geometric elements.

When the geometry has been completed, the absorption values for each octave band, the scatter, and the eventual degree of acoustic transparency have to be assigned to each surface. In addition, the source and receiver characteristics (power and directivity versus frequency, position and orientation) must also be entered.

In the calculation of sound propagation, most models disregard phase properties and use an energy approximation. The sound propagation is studied by means of rays (up to millions depending on the room complexity) being traced from the source position through reflections until their energy has been reduced below the dynamic range of interest (often determined by T or the signal-to-noise ratio in auralizations). Reflection directions are chosen so that angle of reflection equals angle of incidence or with more or less random angles depending on the scatter value attributed to the surface. For the calculation of early reflections, some models also apply image source theory, which actually makes phase considerations possible if the complex impedances of the various surface materials and not just their energy absorption coefficients are known. However, the late reverberation part is normally calculated by some kind of ray method, as image source calculations become impractical after a few orders of reflection. Recently, however, attempts to carry out complete calculations inspired by finite-element (FEM) or boundary-element (BEM) methods have been reported even for rather complex rooms [9.23], whereby phase and diffraction phenomena can be handled.

As a result of tracing the history of all rays emitted in the room model, source and receiver specific impulse responses are produced with detailed information about the direction of incidence, delay and level versus frequency of each reflection component. From these, all the parameters mentioned in Sect. 9.3 can be calculated. Most programs can generate a grid of receiver point over selected audience areas and code the numeric results using a color scale to facilitate a fast overview of the detailed position results for any of the parameters. Also, automatic plots of relevant statistics regarding the distribution with frequency and position are provided.

Impulse responses suitable for use in *auralization* can be generated by applying available head-related transfer functions, which have been recorded as a function of direction of sound incidence on an artificial head. Such transfer functions not only represent the directivity of the human receiver, but are also important for correct perception of direction of incidence of sound field components when listening through headphones. Alternatively, the directional information of the individual reflections making up the impulse response can be coded into a surround-sound format for listening via a multiple-loudspeaker setup.

Figure 9.14 shows a view of a room model as well as two binaural impulse responses and a color mapping of a calculated parameter (generated by the ODEON software).

9.5.5 Scale Model Predictions

Acoustic scale models have been used since the 1930s [9.24] to study the behavior of sound in rooms. With proper care in detailing the geometry and the choice of materials for building the model, this technique can provide quite accurate predictions of impulse responses and acoustic parameters. Actually, scale modeling is still regarded as more reliable than computer models in cases where the room contains a lot of irregular surfaces or objects that diffuse or diffract sound waves. However, the use of physical scale models is limited to larger, acoustically demanding projects, as building and testing the model is far more time consuming and costly than computer modeling.

The diffusion and diffraction caused by an object or surface in a room depends on the ratio between the linear dimensions and the wavelength of the sound: l/λ. Therefore, in a $1:M$ scale model, the diffraction/diffusion conditions will be modeled correctly if the wavelengths of the measurement signals in the model are chosen such that:

$$\frac{l}{\lambda} = \frac{l_\mathrm{M}}{\lambda_\mathrm{M}} = \frac{l/M}{\lambda/M} \quad \Rightarrow \quad f_\mathrm{M} = fM\,. \tag{9.28}$$

This means that we should increase the frequency used in the model measurements by the factor M. Fortunately, sound sources and microphones exist that are capable of handling frequencies up to around 100 kHz, whereby the frequency range up to 10 kHz can be handled in a $1:10$ scale model or 2 kHz in a $1:50$ scale model. The sources may be small loudspeaker units (piezoelectric or electro magnetic) and the microphones 1/4" or even 1/8" condenser types.

Ideally, the absorption coefficient α_{M} of the materials chosen for building each surface in the hall model should fulfil the equation:

$$\alpha_{\mathrm{M}}(fM) = \alpha(f)\,. \tag{9.29}$$

In practice, however, we need only to distinguish between mainly reflecting and mainly absorptive materials. The reflecting surfaces are quite easy, as these can be made of any hard material (plywood, plaster) with a couple of layers of varnish if necessary. It is far more important to build the seats/audience so that both head diffraction and absorption is correct for the scale chosen. In the end it is often necessary to fine-tune the reverberation time in the model by applying absorption to some secondary surfaces (surfaces that do not generate primary early reflections).

Another important issue is to compensate for the excessive air absorption at high frequencies in the model. This can be achieved either by drying the air in the model, by exchanging the air for nitrogen, or simply as a calculated compensation since the attenuation is a simple linear function of time once the, frequency, humidity and temperature have been specified. However, in the case of compensation the signal-to-noise ratio at high frequencies will be severely limited, which reduces the decay range for the estimation of T and the sound quality if auralization is attempted. Figures 9.15 and 9.16 show an example of a 1:10 scale model and its audience respectively.

9.5.6 Prediction from Empirical Data

In the 1980s, when most of the current objective acoustic parameters had been defined, several researchers started making systematic measurement surveys of concert, opera and multipurpose halls. Many of these measured data were published in [9.3]. Combining acoustic data with data on the geometry of the halls has made it possible to derive some simple rules of thumb regarding how, and how much, acoustic parameters are affected by the choice of gross room shape and dimensions.

Through statistical multiple linear-regression analysis of data from more than 50 halls, a number of relationships between acoustical and geometrical properties have been found [9.25]. A number of the linear regression models derived are shown in Table 9.4. In these models, the variations in the acoustic parameters are described as functions of the expected value according to diffuse field theory (G_{exp} and C_{exp}, Sect. 9.3) plus various geometrical variables such as: average room height H, average width W, room volume V, number of seats N, number of rear balcony levels 'no. rear balc.' etc. When the models were derived, the measured reverberation time (T) was used to calculate the expected values of strength G_{exp} and clarity C_{exp}. When using these models for prediction of the values in a hall not yet built, the expected values could be based on a Sabine calculation of T instead. It should be mentioned that, apart from $\Delta G(10\,\mathrm{m})$, the attenuation in strength per 10 m increase in source–receiver distance, the predicted values should be considered as representing the position average of values to be found in the hall.

The three rightmost columns in Table 9.4 contain information about how well each prediction formula matched the measured data. When comparing the amounts of variance explained by the different models, it is seen that the diffuse field prediction is responsible for the largest portion of the variance. This may be interpreted as hall volume and total absorption area being the most important factors governing the behavior

Table 9.4 Regression models for the relationships between room acoustic parameters and room design variables as derived from a database containing data from more than 50 halls (after [9.25])

Room acoustic parameter	Regression models: $f(\mathrm{PAR}_{\mathrm{expected}}, \mathrm{geometry})$	Correlation coefficient	% of variance	STD residuals
C (dB)	$-0.1 + 1.0C_{\mathrm{exp}}$	0.76	58	1.0 dB
	$-1.4 + 0.95C_{\mathrm{exp}} + 0.47W/H + 0.031$ floor slope	0.83	68	0.9 dB
	$-1.2 + 1.03C_{\mathrm{exp}} + 0.43W/H + 0.013$ wall angle	0.84	70	0.9 dB
	$-1.77 + 1.10C_{\mathrm{exp}} + 0.055W + 0.027$ stage ceil. angle	0.86	74	0.8 dB
G (dB)	$-2.0 + 0.94G_{\mathrm{exp}}$	0.91	83	0.9 dB
	$-5.61 + 1.06G_{\mathrm{exp}} + 0.17V/N + 0.04$ distance	0.94	89	0.9 dB
$\Delta G(10\,\mathrm{m})$ (dB)	$-1.85 + 0.42$ no. rear balc.	0.50	25	0.7 dB
	$-1.41 + 0.35$ no. rear balc. $- 3.93$ distance$/(HW)$	0.55	31	0.6 dB
LEF (–)	$0.39 - 0.0061$ width	0.70	49	0.05
	$0.37 - 0.0051$ width $- 0.00069$ wall angle	0.72	53	0.05

of C and G. However, as all the independent variables listed gave a significant improvement of the model accuracy, this also demonstrated that consideration of the geometrical factors can improve the prediction accuracy. Equally importantly, the acoustic effects of changes in certain design variables can be quantified quickly.

Prediction of Clarity

The first C-model illustrates that absorption and volume, as reflected in T, are responsible for the main part of the variation in C. The other three models all illustrate the positive, but sometimes unwanted, effect of average hall width on clarity. Moreover, it is seen that introducing a moderate 15° slope of the main audience floor (without changing the other variables: average width-to-height ratio, volume and absorption area) causes C to increase by about 0.5 dB on average. Similarly, changing the basic design from rectangular to a 70° fan shape makes C increase by about 1 dB. The last C-model illustrates the effect of tilting the angle of the stage ceiling towards the audience. Changing the slope from horizontal to 20° results in about 0.5 dB higher C values.

Predictions of Strength

The G model only considering G_{exp} is rather accurate. The constants in both G-models illustrate the fact that G is always a few dB lower than G_{exp}, as also predicted by *Barron's revised theory* [9.26].

The second G-model contains an independent variable: the ratio between the volume and the number of seats. The positive influence of this ratio is not immediately evident because V/N, which is strongly correlated with T, is expected to be incorporated into the variable G_{exp} already (although this result has also been confirmed by other researchers [9.27]).

As mentioned earlier, G shows a steady and significant decrease with distance in most halls, as was also found by *Barron* [9.26]. This phenomenon is described quantitatively by the two models for estimation of the rate of decrease in G per 10 m source–receiver distance $\Delta G(10\,\mathrm{m})$.

Both listed $\Delta G(10\,\mathrm{m})$ models indicate a reduced distance attenuation when the number of rear balconies is increased. This may be related to the fact that the level is increased in the more-distant seats when these are placed on a balcony, whereby they are closer to the stage and to the reflecting ceiling.

In the second $\Delta G(10\,\mathrm{m})$ model, distance/HW appears as an independent variable. This variable equals the distance from the stage to the rearmost seat divided by the product of the average room height H and average hall width W. As the coefficient to this variable is negative, the natural result appears to be that attenuation with distance will increase if the hall is long, narrow or has a low ceiling.

Predictions of Lateral Energy Fraction

At the bottom of Table 9.4 are listed two models for LEF as a function of hall geometry only, as diffuse field theory has no effect on LEF variation. The effects of the width and angle between the side walls are understandable.

Simple prediction formulae as listed in Table 9.4 are particularly useful in the very early phases of the design process, in which it is natural for the architect to produce and test many different sketches in short succession, leaving no time to carry out computer simulation of each proposal. The importance of the knowledge embedded in these rule-of-thumb equations is highlighted by the fact that many aspects of the acoustics of a new hall are settled when one of these sketches is selected for the further development of the project.

9.6 Geometric Design Considerations

With the connections between room acoustic parameters and hall geometry described in the previous section, we have made the first approach to the third question set up in the introduction to this chapter, which is of real interest to the acoustic designer: how do we control the acoustic parameters by the architectural design variables? Referring again to Fig. 9.1, we are beginning to fill the lower-left box with architectural parameters and establish the relationship between these and the objective parameters. The major geometric factors of importance in auditorium design will be dealt with in the present section: seating layout in plan (determining the gross plan shape of the room) and in section (determining sight lines), use of balconies, choice of wall structure, room height, ceiling shape and use of free-hanging reflectors.

9.6.1 General Room Shape and Seating Layout

When people stop to watch or listen to a spontaneous performance, for instance in an open square in the city, the way they arrange themselves depends on the type

Fig. 9.15 A 1:10 scale model of the Danish Broadcasting concert hall in Ørestad, Copenhagen. The model is fitted with audience and orchestra (acoustic design by Nagata Acoustics, Japan)

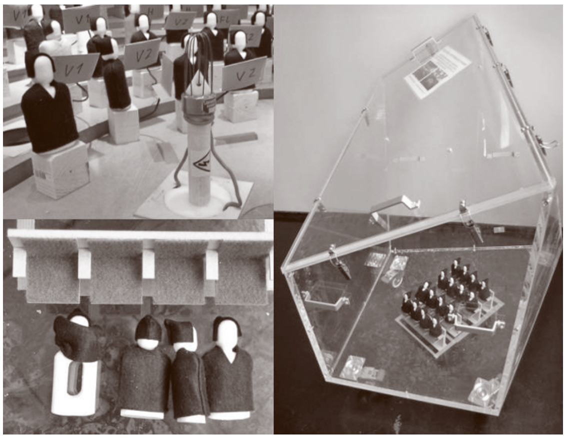


Fig. 9.16 Model audience used in a 1:10 scale model of the Danish Broadcasting concert hall shown in Fig. 9.15. *Upper-left corner*: Close view of model orchestra musicians and spark sound source. *Lower left*: close-up view of model chairs and polystyrene audience with hollow chest. Dress and hair is made from wool felt. *Right*: model audience and chairs placed in model reverberation chamber for absorption testing (reverberation room designed by Nagata Acoustics)

of performance. It will be governed by the fact that any newcomer will look for the best position available relative to his/her need to see or hear properly. The choice will be a compromise between choosing a position close to the performer(s) and next to other members of the audience or farther away but close to an eventual main center line for vision and sound radiation. In Fig. 9.17,

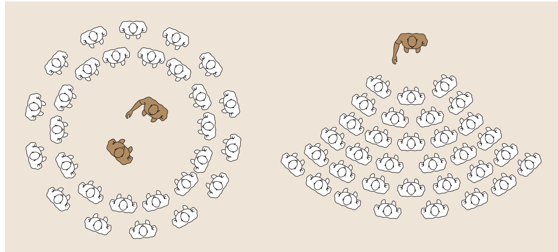


Fig. 9.17 *Organic* formations of audience depending on the type of performance

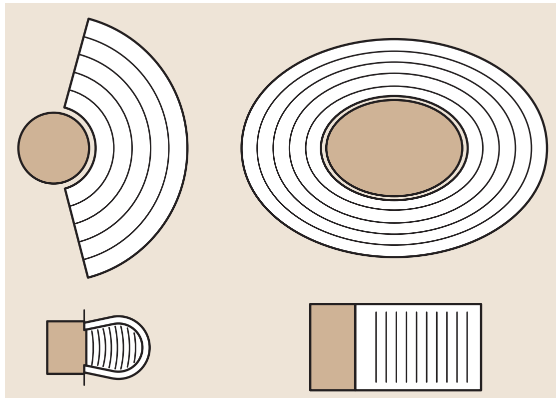
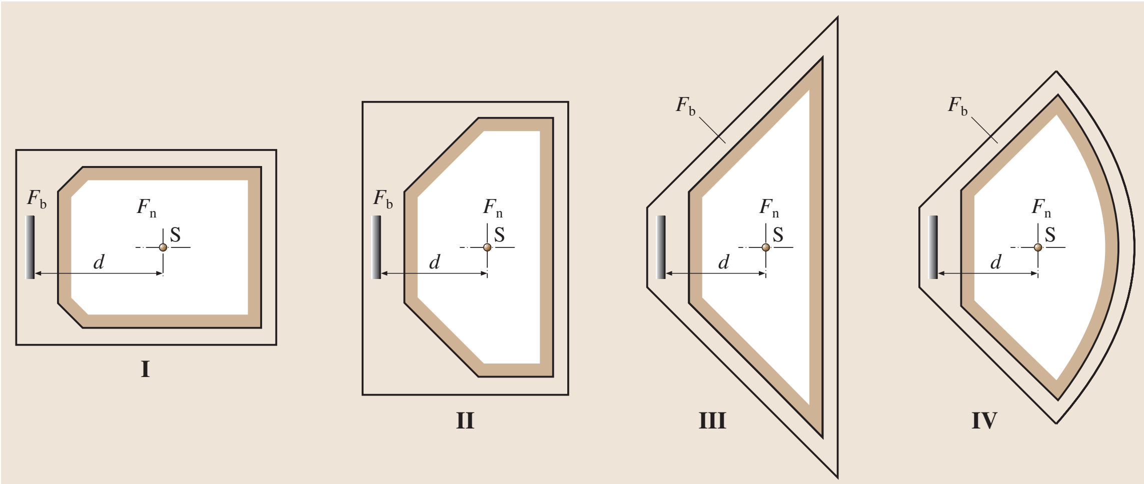


Fig. 9.18 Basic room shapes (after [9.4])

such organic audience arrangements are shown for two different types of performances:

1. action or dialogue theater such as dancing, fighting, debating, circus performance, for which the visual and acoustic emission are more or less omnidirectional and
2. monologue or proscenium stage theater performance and concerts with acoustic instruments – all of which have a more-limited visual and/or acoustic directivity.

When we set up walls around the gathered people, we arrive at two classical plan shapes of auditoria developed early during our cultural history. These are shown at the top of Fig. 9.18: the fan-shaped Greek/Roman theater (left), and the amphitheater, circus or arena (right). Both were originally open theaters; but the shapes were maintained in roofed buildings. In the bottom of Fig. 9.18, later, basic forms are shown: the horseshoe, the Italian opera plan and the rectangular concert hall. The latter form was originally a result of traditional building forms and limitations in roof span with wooden beams.



	I	II	III	IV
F_n/F_b	0.63	0.55	0.64	0.67
d/d_I	1	0.83	0.78	0.79

Fig. 9.19 Mean stage–listener distance and net efficiency of floor area for four different room shapes. F_n is the net area occupied by the audience, F_b is the total floor area, d is the average source–listener distance; $d_I = d$ for case I (after [9.28])

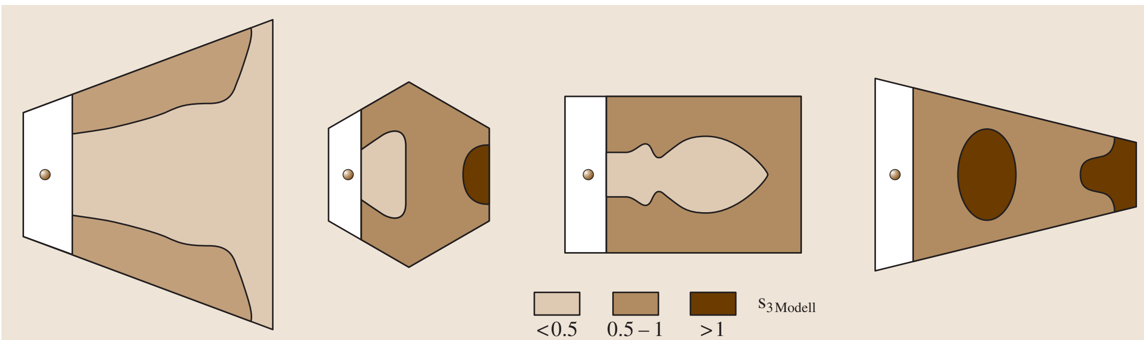


Fig. 9.20 Calculated distributions of early lateral reflection levels in rooms of different plan shape: fan, hexagon, rectangular and reverse fan. *Darker areas* represent higher LEF levels (after [9.5])

When building an auditorium, efficient use of the available floor space and the average proximity of the audience to the performers are very important parameters. Depending on the plan shape of the room different values appear as shown in Fig. 9.19.

Low average distance is the main reason for the frequent use of the fan shape (III and IV in Fig. 9.19), although the directivity of sound sources (like the human voice) as well as the quality of lines of sight sets limits on the opening angle of the fan.

Another limitation of the fan shape is that it does not generate the strong lateral reflections so important in halls for classical music. This was already seen in the empirical equations in Sect. 9.5.6 and confirmed in Fig. 9.20, showing the calculated distributions of early lateral reflection levels in rooms of different plan shapes.

The reason is that sound waves from the source hitting the side walls will produce reflections that will run almost parallel to the direct sound and so hit the listener from an almost frontal direction, therefore not contribut-

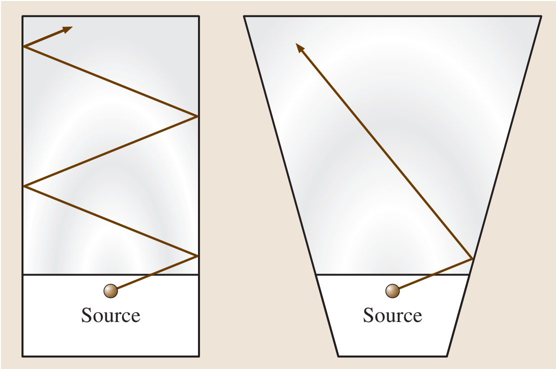


Fig. 9.21 Directions of side-wall reflections depending on room shape (after [9.4])

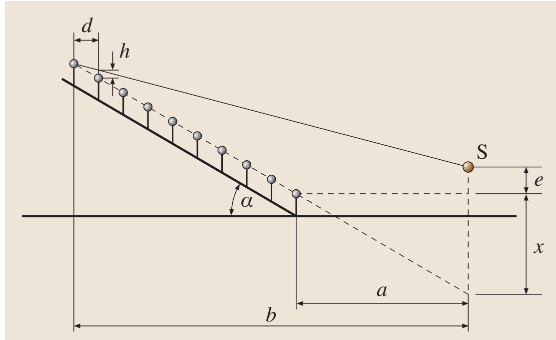


Fig. 9.22 Relevant parameters for the calculation of the necessary angle α for a constantly sloping floor section. a: distance from the source to the first seating row, b: distance from the source to the last seating row, d: distance between the seat rows, e: source height above the head of listeners in the first row (determined by the height of the stage floor over the stalls floor), h: line-of-sight clearance at the last row. Rows closer to the source will have clearance values larger than h (after [9.28])

ing to the LEF or causing dissimilarity between the two ear signals. This is also illustrated in Fig. 9.21. In addition, many of the side-wall reflections will not reach the center of the hall at all.

9.6.2 Seating Arrangement in Section

When several rows of seated audience are placed on a flat floor, the direct sound from a source on the stage and the reflection off the audience surface will hit a receiver in this audience area after having traveled almost identical distances, and the reflection will hit the audience surface at a very small angle relative to the surface plane. At this grazing angle of incidence, all the energy will be reflected regardless of the diffuse absorption value of the surface, but the phase of the pressure in the reflected wave will also be shifted 180°. Hereby the direct sound and the grazing incidence reflection will almost cancel each other over a wide frequency range. Typical attenuation values found at seats beyond the tenth row can amount to 10–20 dB in the range 100–800 Hz relative to an unobstructed direct sound. Moreover, the higher frequencies above 1000 Hz will also be attenuated by 10 dB or more due to scattering of the sound by the heads of people sitting in the rows in front.

The same values of attenuation mentioned above for the direct sound are likely also to be valid for first-order reflections off vertical side walls. The result will be weaker direct sound and early reflections in the horizontal stalls area and subjective impressions of reduced *clarity*, *intimacy* and *warmth* in the stalls seats.

To avoid this grazing incidence attenuation, the seating needs to be sloped relative to the direction towards the source. The vertical angle of the audience floor can be designed to be constant or to vary gradually with distance. If a constant slope is planned, the angle α necessary to obtain a certain clearance h at the last row (equal to the vertical distance between sight lines from this and the preceding row) can be calculated from the variables indicated in Fig. 9.22:

$$\tan\alpha = \frac{hb}{da} - \frac{e}{a} \,. \tag{9.30}$$

In particular, it is of interest to use this formula to see how far away from the source it is possible to maintain a certain slope (one obvious example being $\alpha = 0$, corresponding to a horizontal floor):

$$b = \frac{d}{h}(e + a\tan\alpha) \,. \tag{9.31}$$

Normally, a clearance value of minimum 8 cm will be sufficient to avoid grazing-incidence attenuation; but sight lines will still be unsatisfactory unless staggered seating is applied so that each listener is looking towards the stage between two other heads and not over a person sitting directly in front. However, if the clearance is increased to about 12 cm, then both the acoustic and visual conditions will be satisfactory, but this implies a steeper floor slope.

In rows closer to the sound source a linear slope will cause the clearance to be higher than the design goal – and so higher than necessary. Often this is not desirable as it may cause the last row to be elevated high above the source and the steeply sloped floor will reduce

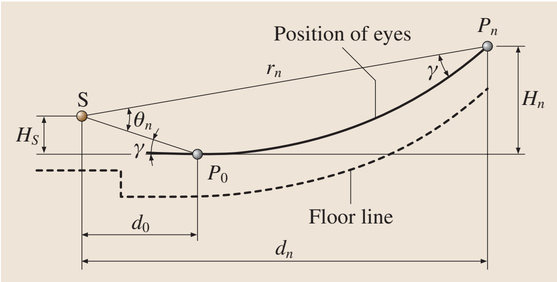


Fig. 9.23 Relevant parameters for the calculation of a floor slope with constant clearance. The variables are explained in the text (after [9.29])

the effective volume of the room. Therefore, it can be relevant to consider instead a curved floor slope with constant clearance, as illustrated in Fig. 9.23.

In Fig. 9.23, the height of a person's head in the n-th row H_n relative to the height of the head in the first row is calculated according to the formula (derived from a logarithmic spiral)

$$\begin{aligned} H_n &= d_0\gamma + d\,(\theta - \gamma) \\ &= \gamma \left[d_n \log_e \left(\frac{d_n}{d_0} \right) - (d_n - d_0) \right] . \end{aligned} \tag{9.32}$$

In this expression d_0 is the distance from the source to the first row, d_n the distance to the n-th row and γ the desired angle between the tangent to the seating plane and the line of sight to the source. For a one meter row-to-row distance, an angle γ of 8° will correspond to a clearance of about 12 cm.

A curved slope can also be obtained by simply applying the linear-slope formula to each of smaller sections of seat rows successively (e.g. for every five rows). In this case the variables a and e in (9.30) should refer to the first row in the relevant section.

As mentioned earlier, a steep slope will reduce the volume and so the reverberation time T. Besides, geometric factors may reduce T beyond what a Sabine calculation may predict. The reason is that the elevated seating – and its mirror image in the ceiling – will cover a larger solid angle as seen from the source. Hereby a larger part of the emitted sound energy will quickly reach the absorbing seating and leave less for fueling a reverberant sound field in the room. The result is higher *clarity* and less *reverberance* in halls with steeply raked seating. This is in line with the empirical equation for clarity as a function of T (C_{exp}) and floor slope listed in Table 9.4, Sect. 9.5.6. Consequently, it is a good idea to limit the clearance/angle in halls for classical music to about 8 cm or 6°, whereas for speech auditoriums and drama theaters, in which both intelligibility and good visual sight lines have priority, values closer to 12–15 cm or 8°–10° may be preferable.

9.6.3 Balcony Design

The introduction of balconies allows a hall with a larger seat count to be built where a limited footprint area is available; but often balconies are introduced simply in order to avoid longer distances between stage and listeners. The average stage–listener distance can be reduced significantly by elevating the rearmost seat rows and moving them forward on rear or side balconies above parts of the audience in the stalls. In this way not only is the direct sound increased by shortening the direct sound path, but the elevated seats also become closer to a possibly reflecting ceiling. The result is higher strength as well as higher clarity in the balcony seats compared to the alternative placement of seats at the back of an even deeper hall.

Since the line-of-sight and clearance criteria still have to be fulfilled for balcony seats, the slope often becomes quite steep on balconies, as illustrated in Fig. 9.24 showing a long section in a hall with constant clearance on main floor and balcony seats. For safety reasons the slope must be limited to no more than about 35°.

When balconies are introduced, the acoustics in the seats below the balconies need special attention. It is important to ensure sufficient opening height below balcony fronts relative to the depth of the seating under the balcony (Fig. 9.25). Otherwise, particularly the level of the reverberant energy in the overhung seats will be reduced, causing the overall sound level to be weak and lacking in fullness. Rules of thumb in design are $H \geq 2D$ for theaters (in which reverberant sound is less important) and $H \geq D$ for concert halls (in which fullness is

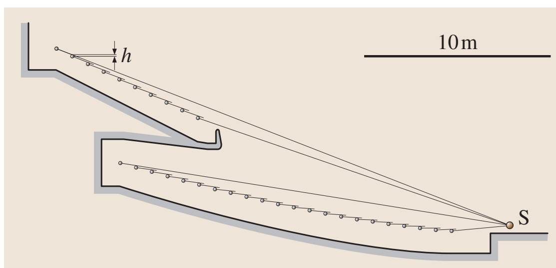


Fig. 9.24 Section of a hall with a rear balcony and constant clearance. The result is an increased slope for the elevated balcony (after [9.30])

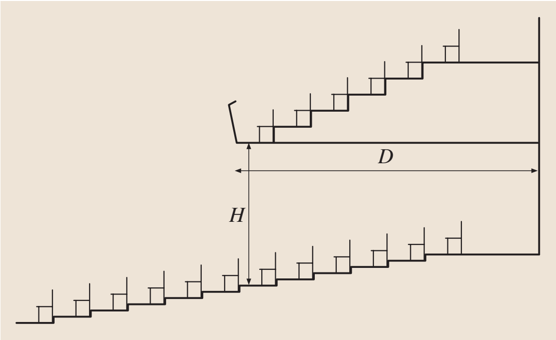


Fig. 9.25 Important parameters for maintaining proper sound in seats overhung by balconies (after [9.9])

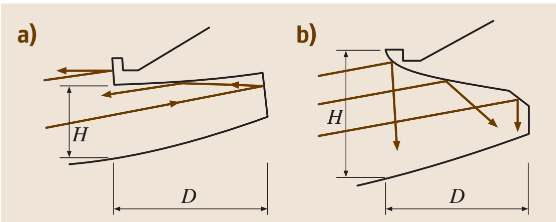


Fig. 9.26a,b Poor **(a)** and good **(b)** design of balcony profile (after [9.29])

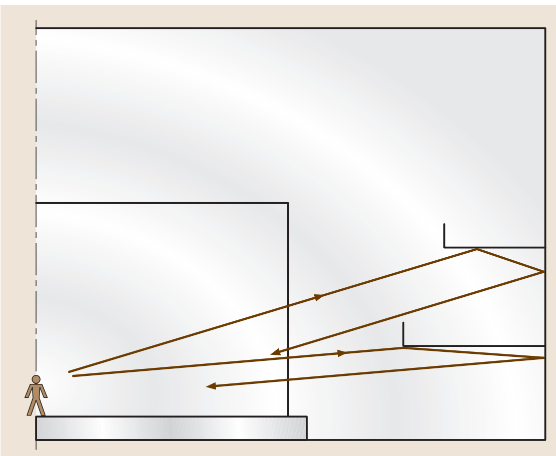


Fig. 9.27 A half cross section of a hall with side balconies that, in combination with the side wall, return early reflection energy towards the stalls area from a lateral angle (after [9.4])

more important) [9.9]. The criteria may also be stated in terms of the vertical viewing angle from the seat to the main hall volume. Recently, *Beranek* [9.3] has suggested this be a minimum of 45°.

Sometimes, the requirement for limited depth of balcony overhangs is also specified as a percentage of the ceiling area needed to be visible from the last row of seating or as a vertical opening angle β between the balcony soffit and the head of listener below the balcony front as seen from the last row. In the latter case, Barron has suggested 25° as suitable for drama halls and 35° for concert halls.

As illustrated in Fig. 9.26, it is often advantageous to let the soffit be sloped, so that it will help distribute reflected sound to the overhung seats. If a diffusing profile is added to the balcony soffit, the reflection off this surface may even gain a lateral component, so that *spaciousness* will not be reduced. The drawing also illustrates how the otherwise vertical balcony front and the right angle between the rear wall and balcony soffit have been modified to avoid sound being reflected back to the stage where it could generate an echo. However, in a rectangular hall with balconies along the side walls, vertical soffits in combination with the side-wall surface can increase the early and lateral reflection energy in the stalls area as shown in the half cross section in Fig. 9.27.

In general, as large-scale modulations in the hall geometry, balconies also provide low-frequency diffusion, which is considered an advantage.

9.6.4 Volume and Ceiling Height

As the people present are often the main source of sound absorption in an auditorium, the ratio between the volume and area covered by the audience and performers is an important factor for the reverberation time achievable. The general relationship valid for concert halls is shown in Fig. 9.28.

Because the variation in area per person does not vary significantly between halls, the volume per seat is also used as a rule of thumb for the calculation of the required volume. For speech theaters, a volume per seat of 5–7 m^3 will result in a reverberation time around 1 s, whereas at least 10 m^3 per seat is needed to obtain a T value of 2 s.

Since the audience normally occupies most of the floor area, the volume criterion can also be translated to a ceiling height criterion. In such cases, the abscissa in Fig. 9.28 can be interpreted as roughly equal to the required room height. A ceiling height of about 15 m is required if a T value of 2 s is the target, whereas a height of just 5–6 m will result in an auditorium with a reverberation time of about 1 s. It should be emphasized that, in the discussion above, we have assumed the other room surfaces to be reflective.

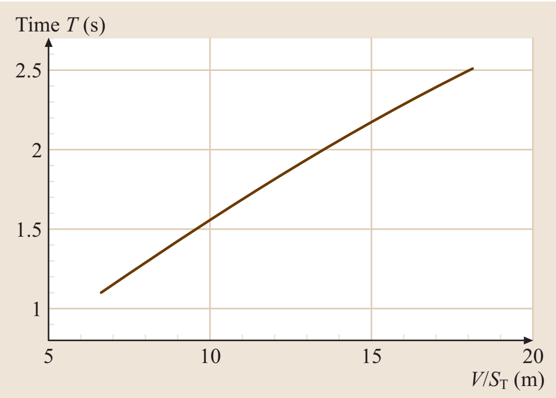


Fig. 9.28 Reverberation time T in occupied concert halls versus the ratio between the volume V and area occupied by the audience and orchestra S_T (including aisle space up to 1 m width). Derived from empirical data (after [9.9])

These guidelines on volume should be regarded as rough rules of thumb only, as room shape, floor slope, the presence of balconies and the distribution of absorption on other surfaces can cause significant variations in the objective values obtained for T and other parameters. In smaller halls (say below 1500 seats) for symphonic concerts, it is better to start with a volume slightly larger than required rather than the opposite, as one can always add a little absorption – but not more volume – to a hall near completion. On the other hand, in larger halls, where maintaining a suitably high G value is also of concern, a better strategy is to limit the volume and minimize the absorption instead. One way of minimizing absorption is to place some of the seats under balcony overhangs, where they are less exposed to the reverberant sound field. However, the other side of the coin is the poor conditions in these seats, as explained in Sect. 9.6.3.

9.6.5 Main Dimensions and Risks of Echoes

In order for early reflections to contribute to the *intelligibility* of speech or to the *clarity* of music, they must not be delayed more than about 30 ms and 80 ms, respectively, relative to the direct sound. If the first reflection arrives later than 50 ms, it is likely to be perceived as a disturbing echo when impulsive sounds are emitted (Sect. 9.2).

In large rooms it is a challenge to shape the surfaces so that reflections from the main reflecting areas arrive within 50 ms at all seats. It is possible to identify which surfaces are able to generate echoes at the receiver point

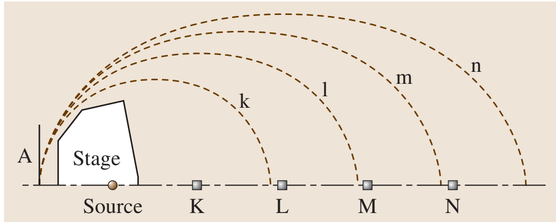


Fig. 9.29 Echo ellipses drawn over the plan of an auditorium (after [9.4])

in question by means of simple geometrical studies of plan and sectional drawings as seen in Fig. 9.29. Ellipses are drawn so that the source and relevant receiver positions are placed at the focal points and so that the sum of distances from the focal points to any point on the ellipse equals the distance between the focal points plus 17 m (times the scale of the drawing). Then, if a surface outside the ellipse marked "m" in the figure directs sound towards seats near point M, this surface must be made absorbing, diffusing or be reoriented so that the reflection is directed towards areas farther away from the source than M.

In particular, it is important to check the risk of echoes being generated by the rear wall behind the audience and from a high ceiling.

9.6.6 Room Shape Details Causing Risks of Focusing and Flutter

Concave surfaces can cause problems as they may focus the sound in certain areas while leaving others with too little sound. Thus, vaulted ceilings as seen in Fig. 9.30 are only acceptable if the radius of curvature is less than half the height of the room (or rather half the distance from peoples' heads to the ceiling) so that the focus center is placed high above the listeners. Situations with

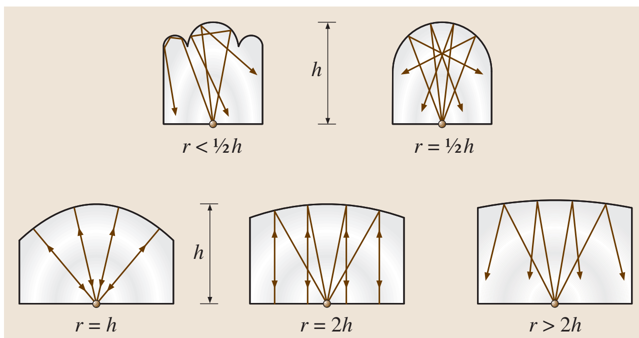


Fig. 9.30 Focusing by concave ceilings (after [9.31])

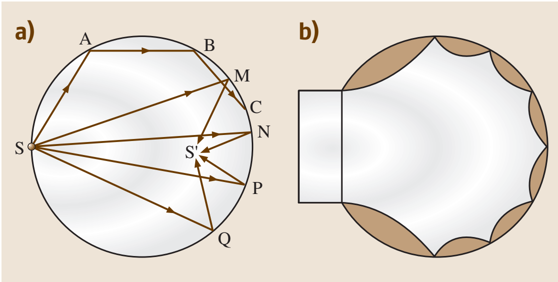


Fig. 9.31a,b Circular room with focused and creeping waves **(a)**, and with the surface shape modified **(b)** so that these phenomena are avoided

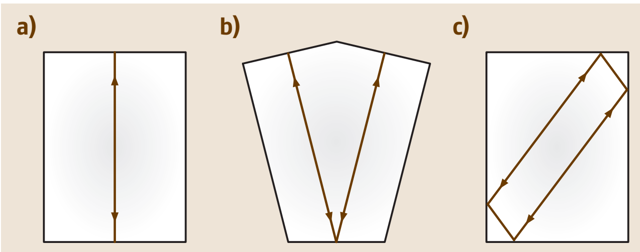


Fig. 9.32 Room shapes causing risk of flutter echoes

a slight curvature, $r \geq 2h$, may also be acceptable if the floor is covered with absorptive seating so that multiple reflections between floor and ceiling will not arise.

Circular rooms are also unfavorable both due to the risk of focusing and the risk of a whispering-gallery effect (boundary waves creeping along the wall). Modifications of the concave wall within the overall circular shape as shown in Fig. 9.31 can solve the problem, at least at mid and high frequencies.

Another frequent problem is regular, repeated reflections, so-called flutter echoes, which arise between parallel, smooth reflecting walls or along other simple reflection paths that can repeat themselves as shown in Fig. 9.32. Particularly in small rooms such reflections may cause strong coloration of the sound (through comb filtering as explained in Sect. 9.2.3) while in larger rooms they may be perceived as a long series of distinct echoes evenly spaced in time.

In any case, the flutter can be avoided by simple absorptive or diffusive treatment of at least one of the opposite reflecting surfaces, as shown in Fig. 9.33. Besides, if it is possible to change the angle between the opposite walls just by a few degrees (3–5°), the problem will also disappear.

9.6.7 Cultivating Early Reflections

In most rooms accommodating more than say 25 listeners, it is of relevance to consider how early reflections can help distribute the sound evenly from the source(s) to the audience. This will increase the early reflection energy and so improve *clarity/intelligibility* and perhaps even reduce the reverberation time by directing more sound energy towards the absorbing audience.

In rooms for speech, the ceiling height should be moderate (Sect. 9.6.4) so that this surface becomes the main surface for the distribution of early reflection energy to the audience. Figure 9.34 shows examples of how this can be accomplished in rooms with both low and high ceilings.

In this figure the two situations at the top need improvement. In the low-ceiling room to the left the echo from the rear wall/ceiling combination can be removed by introducing a slanted reflector covering the corner. In this way the sound is redirected to benefit the last rows, which may need this to compensate for the weak direct sound at this large distance. Alternatively, the energy from this rear corner may simply be attenuated by one or both surfaces near the corner being supplied with absorption.

If the ceiling is so high that echoes can be generated in the front part of the room, as shown to the right, the front part may likewise be treated with slanted reflectors that redirect the sound to the remote seat rows, or this ceiling area may simply be made sound absorbing. The lower-right section in the figure also illustrates how the

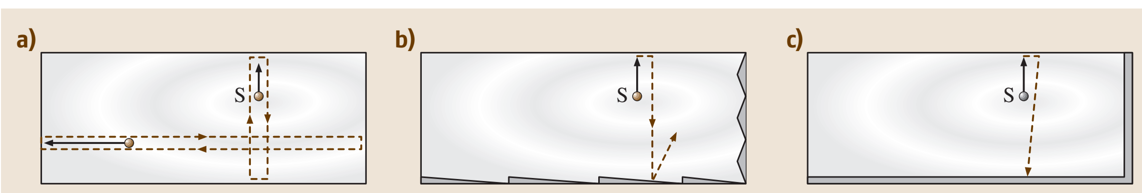


Fig. 9.33a–c Room with flutter echoes **(a)**, and measures against this by means of diffusion **(b)** or absorption **(c)** (after [9.28])

floor can be tilted to reduce volume and allow the rear seats benefit from being closer to the reflecting ceiling.

If for architectural reasons the ceiling is higher than necessary in a room for speech or rhythmic music, substantial areas of the room surfaces must be treated with sound-absorbing material to control the reverberation time. However, more sound absorption will reduce the overall sound level in the room so this is only recommended if the sound can be amplified.

If the ceiling is to be partly absorbing, the ceiling area that should be kept reflecting in order to distribute useful first-order reflections to all listeners can be generated geometrically by drawing the image of the source in the ceiling surface and connecting lines from this image point to the boundaries of the seating area. Where these lines cross the actual ceiling we find the boundaries for the useful reflecting area, as shown in Fig. 9.35.

In larger auditoria, a more-detailed shaping of the ceiling profile may be needed to ensure even distribution of the reflected sound to the listeners. Notice in Fig. 9.36 that the concave overall shape of the ceiling can be maintained if just the size of the ceiling panel segments is large enough to redirect the reflections at suitably low frequencies.

Local reshaping of surfaces while maintaining the overall form is also illustrated in Fig. 9.37, in which the weak lateral reflections in a fan-shaped hall are improved by giving the walls a zigzag shape with almost parallel sets of smaller areas along the splayed side walls. In the example shown in the photo, the panels are even separated from the wall surface and tilted downward.

9.6.8 Suspended Reflectors

In many cases it is advantageous to reduce the delay and increase the level of a reflection without changing the room volume. In this case it is obvious to suggest individual reflecting surfaces suspended within the room boundaries, as shown to the right in Fig. 9.38. If the reflector is placed over an orchestra pit, it can improve mutual hearing among the musicians in the pit as well as increase the early energy from the singing on the stage to the seating primarily in the stalls area.

Suspended reflectors are also suitable for acoustic renovation of existing, listed buildings, because the main building structure is not seriously affected (as in the case of the Royal Albert Hall in London [9.3]).

For a given source position, a small reflector will cover only a limited area of receiver positions. There-

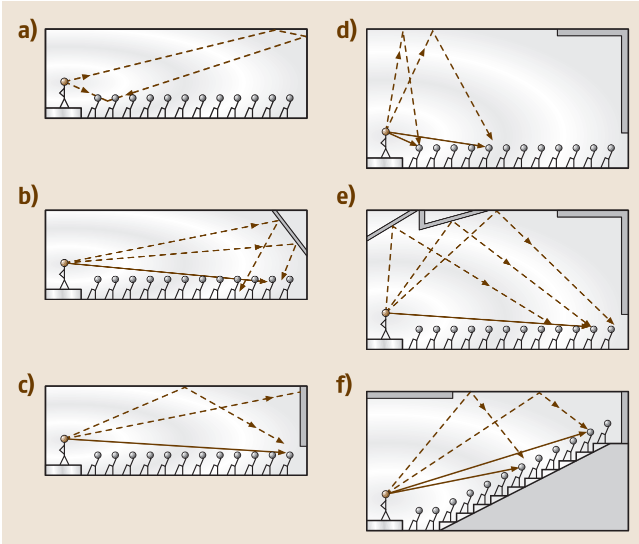


Fig. 9.34 Means of controlling early reflections in auditoria (after [9.28])

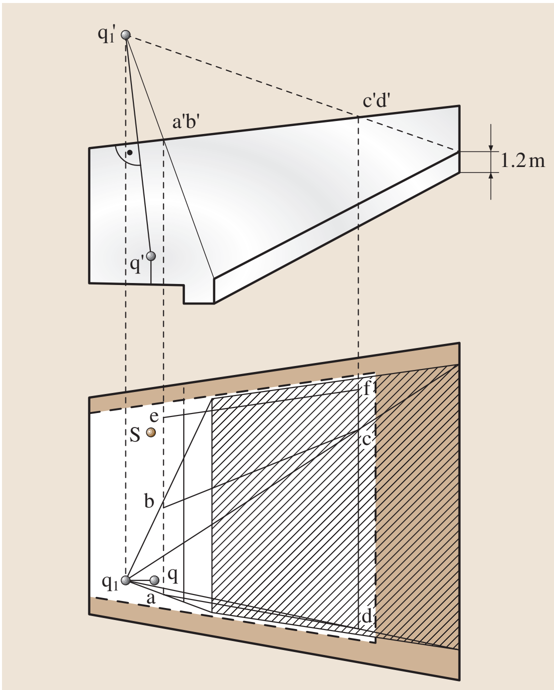


Fig. 9.35 Sketch identifying the area responsible for covering the audience area with a first-order ceiling reflection (after [9.32])

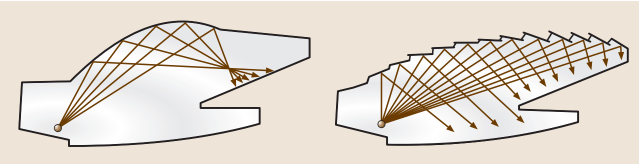


Fig. 9.36 Reshaping of ceiling profile to avoid focusing and provide more-even sound distribution

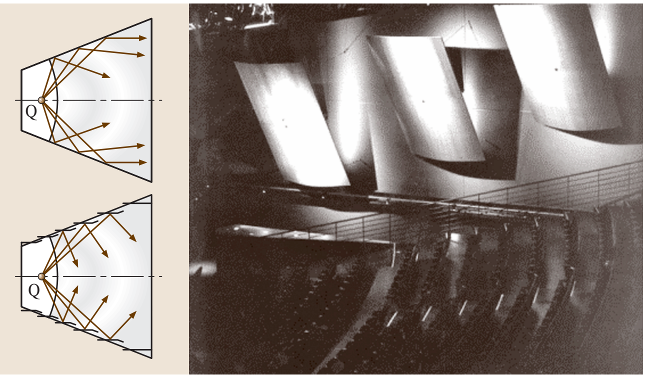


Fig. 9.37 Improvement of side-wall reflections in a fan-shaped hall by local reshaping or adding of panels parallel to the long axis of the room (drawing after [9.32])

fore, how the reflector(s) influence the balance heard between different sources such as the different sections in a symphony orchestra must be thoroughly considered. With small reflectors, the balance between sections may be perceived as very different in different parts of the audience. Often it will be advantageous to give the reflector(s) a slightly convex shape to extend the area covered by the reflection and to soften its level.

The nature of the reflection off a panel of limited size is strongly frequency dependent and may be influenced by several factors: distance to source and receiver, absorption by the panel material, diffraction governed by the ratio between panel size and the wavelength, and focusing/diffusion due to panel curvature.

The attenuation with distance is as for a spherical wave in free field: 6 dB per doubling of the total distance from the source to the reflector a_1 plus the distance from the reflector to the receiver a_2. Normally, the attenuation due to absorption is negligible, as reflectors should obviously be made from a reflective material. The only exception could be when a thin, lightweight material is used, such as plastic foil or coated canvas. In this case the attenuation ΔL_{abs} can be calculated from the mass law

$$\Delta L_{\text{abs}} = -10 \log_{10} \left[1 + \left(\frac{\rho c}{\pi f m \cos\theta} \right)^2 \right] \tag{9.33}$$

where ρc equals the specific impedance of air ($\approx 414\,\text{kg}\,\text{m}^{-2}\,\text{s}^{-1}$), f is the frequency in Hz, m is the mass per square meter of the material and θ is the angle of incidence relative to the normal direction.

As a simple design guide, diffraction can be said to cause the specular reflection to be attenuated by 6 dB per octave below a limiting frequency f_g given by:

$$f_g = \frac{ca^*}{2S\cos\theta} \tag{9.34}$$

where a^* is the weighted *characteristic distance* of the reflector from the source and receiver:

$$a^* = \frac{2a_1a_2}{a_1 + a_2}\,, \tag{9.35}$$

S is the area of the reflector, and θ is the angle of incidence as before. Above the frequency specified in (9.34), the influence of diffraction can be neglected.

If the reflector is curved, the attenuation ΔL_{curv} can be calculated as:

$$\Delta L_{\text{curv}} = -10 \log_{10} \left| 1 + \frac{a^*}{R\cos\theta} \right| \,. \tag{9.36}$$

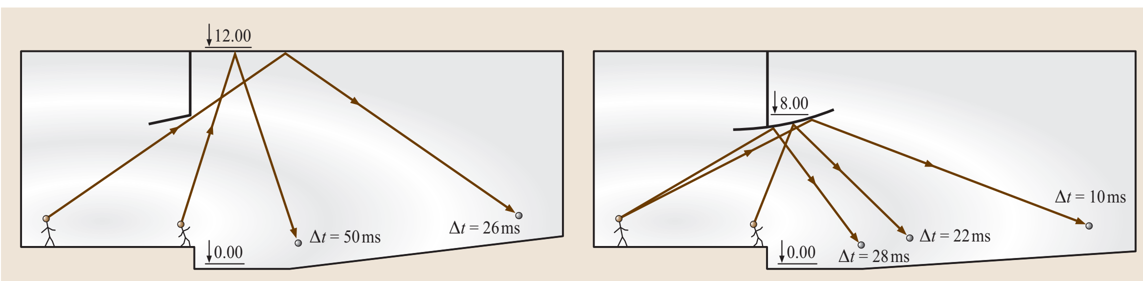


Fig. 9.38 The influence on reflection paths of a reflecting panel suspended in front of a stage (after [9.33])

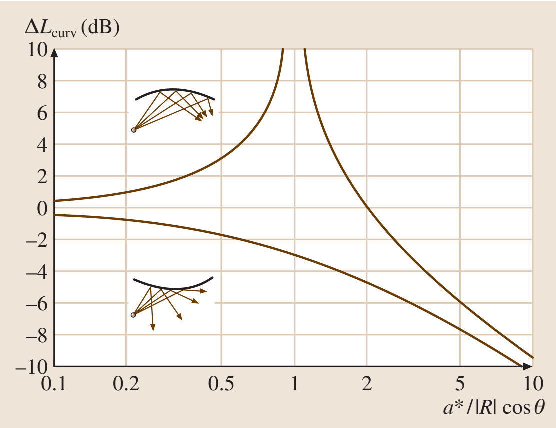


Fig. 9.39 Attenuation of a reflection caused by concave and convex reflectors (after [9.34])

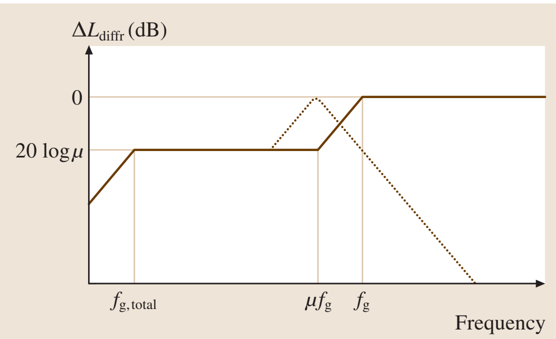


Fig. 9.40 Approximate attenuation due to diffraction from a reflector array. All non-horizontal lines have a slope of 6 dB per octave. See text for further explanation (after [9.34])

Here, the only new variable R is the radius of curvature. R should be entered positive for convex surfaces and negative for concave surfaces. When the reflector is concave, strong focussing occurs when $R = -a^*/\cos\theta$. In Fig. 9.39, graphs of ΔL_{curv} are shown for both the concave and convex cases. It is noted that the values can be both positive and negative. In the case where the surface is curved in both the x- and y-directions of the reflecting plane, the ΔL_{curv} correction should be applied twice using the appropriate radii of curvature in the two directions.

In certain cases an array of (identical) reflectors are suspended as an alternative to having one large reflector. Such an array can be characterized by the area of each element, S, the total area covered by the array S_{total} and the degree of coverage, μ, equal to the ratio between the total area of the n reflector elements nS and S_{total}. In this case, the attenuation due to diffraction depends on frequency as shown in Fig. 9.40.

Here, f_{g} is the limit frequency of the individual elements calculated according to (9.34), while $f_{\text{g,total}}$ is the lower-limit frequency related to the entire array area S_{total}

$$f_{\text{g,total}} = \frac{ca^*}{2S_{\text{total}}\cos\theta} \,. \tag{9.37}$$

First we look at the situation above the frequency f_{g} in Fig. 9.40. As in the case of the single reflector, the attenuation is zero as long as the specular reflection point falls on one of the reflector elements. If the specular point falls between two elements, the attenuation can be approximated by the dotted line, according to which we have an attenuation of 6 dB per octave as frequency increases. In the frequency range between $f_{\text{g,total}}$ and μf_{g}, the attenuation is roughly frequency independent and governed by the array density μ, and below $f_{\text{g,total}}$ the level again rolls off by 6 dB per octave. It is seen that, in order to obtain a wide range of frequency-independent reflection, one should aim to use many small reflectors rather than fewer, larger elements. Fortunately, the strong position-dependent variation in reflection level at high frequencies can be reduced by making the individual elements slightly convex.

It should be remembered that whenever surfaces are shaped with the purpose of efficiently directing sound energy from the source towards the absorbing audience, this energy is no longer available for the creation of the reverberant sound field. In other words, the more we try to improve the *clarity* and *intelligibility*, the more we reduce reverberation in terms of T and EDT (regardless of the physical absorption area), and the more the sound field will deviate from the diffuse field predicted by any simple reverberation formula. If substantial reverberation is needed as well, the solution may be to increase the volume per seat beyond the 10 m^3 as already suggested for large halls in Table 9.1. As seen in Sect. 9.7.3 this is in line with the current trend in concert hall design.

9.6.9 Sound-Diffusing Surfaces

It is often stated that diffusion of sound from irregular surfaces adds a pleasant, smooth character to the sound in a concert hall. It is a fact that reflections from large smooth surfaces can produce an unpleasant harsh sound. It is also a fact that most of the surfaces in famous old and highly cherished halls such as the Musikvereinsaal break up and scatter the reflected sound in many directions due to rich ornamentation, many window and door niches and the coffered ceiling (Fig. 9.45). Even the large chandeliers in this hall break up the sound waves.

Many present-day architects are influenced by modernism and are less enthusiastic about filling their designs with rich ornamentation. Therefore, close cooperation is often needed between architect and acoustician in order to reach a surface structure that will scatter the sound sufficiently, although this is difficult to define exactly as good guidelines do not exist for what percentage of the wall/ceiling area one should make diffusive.

Schroeder [9.35] has done pioneering work on creating algorithms for the design of surface profiles that will ensure uniform distribution of scattered sound within a well-defined frequency range. These algorithms are based on number theory using the maximum length, quadratic residue or primitive root number sequences. However, in most cases of practical auditorium design, perfectly uniform scattering is not needed, and good results can be obtained even after a relaxed architectural adaptation of the ideas, as shown in Fig. 9.41.

The primary requirement to achieve diffusion is that the profile depth of the relief is comparable with a quarter of the wavelength and the linear extension of the diffusing element (before it is eventually repeated) is comparable with the wavelength at the lowest frequency of interest to be diffused. Given this, many different solutions can be imagined such as soft, curved forms cast in plaster, broken stone surfaces, wooden sticks, irregular brick walls etc.

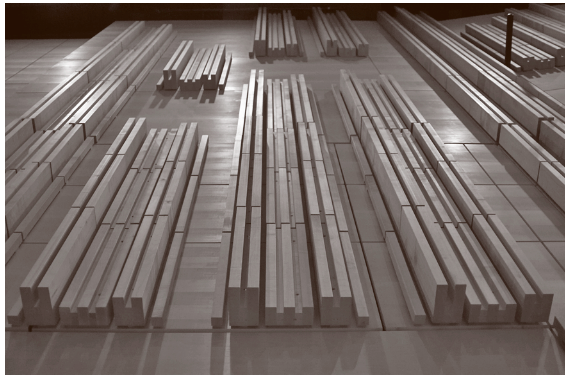


Fig. 9.41 Diffusion treatment by means of wooden sticks screwed on an otherwise plane, wooden wall surface

Finally, it should be mentioned that strict periodic structures with sharp edges equally spaced along the surface can cause problems of harsh sound at frequencies above the design frequency range. Actually, it is safer to apply more-organic, chaotic structures, or to ask the workers not to be too careful with the ruler.

9.7 Room Acoustic Design of Auditoria for Specific Purposes

This section will present some examples showing how the principles outlined in Sect. 9.5 are implemented in the current design of different types of auditoria. The main focus will be on concert halls for classical music as well as drama and opera theaters. It will also be shown that design choices are not only compromises between acoustic and other concerns, but sometimes even between different room acoustic aspects. This is often the case with multipurpose auditoria. In any case, the compromises are more serious the larger the seating capacity.

9.7.1 Speech Auditoria, Drama Theaters and Lecture Halls

In the Western world, rooms for spoken drama as well as for opera have their roots in the Greek and Roman theaters in which the audience is arranged in sloping concentric rows, the *cavea*. In Roman times, this audience area covered a 180° angle around the orchestra or apron stage. The advantage of such an arrangement is of course the possibility to accommodate a large audience within a limited maximum distance from the stage. Many modern theaters and speech auditoria have adopted this layout – perhaps extended with one or more sets of balconies.

An example of such a modern speech auditorium (the lecture hall at Herlev Hospital, Denmark) is seen in Fig. 9.42 along with a Roman theater for comparison.

In a drama theater, a moderate reverberation time around 1 s will provide a good compromise between high intelligibility and a high strength value, as both are needed for unamplified speech. The T value may be reduced to about 0.5 s in small rooms for 50–100 people and also increased slightly for audiences above 1000. To obtain a high strength value in a large theater, it is important to choose a modest volume per seat (about 5 m^3) rather than to control the reverberation in an unneces-

Fig. 9.42 Photos of Roman theater (in Aspendos, Turkey) and of a modern speech auditorium (Herlev Hospital, Denmark)

sarily large volume by applying additional absorption beyond that provided by the audience chairs. Thus, the height of the ceiling should be chosen fairly low – which is also advantageous for its ability to distribute early reflections to the audience. According to diffuse field theory, the steady-state energy density equivalent to G_{exp} is inversely proportional to the total absorption area A in the room.

High *clarity* and sound *level* are also promoted by careful shaping of the reflecting ceiling and a substantial sloping of the audience floor. Due to the directivity of the human voice, a reflecting rear wall (or stage scenery in a theatre) is of less importance, unless an actor should turn his back to the audience. Also the wall surfaces close to the stage opening are important, because they will send sound across the room so that sound also reaches people behind the speaker when he turns the head towards one side. The rear wall/balcony soffits behind/over the rear part of the audience should diffuse or redirect the sound to benefit nearby listeners.

As *spaciousness* is not a quality of particular importance in a speech auditorium, reflections from the side walls farther from the stage are often given lower priority than a low average stage-to-audience distance, which is promoted by a fan-shaped floor plan. However, as indicated by the problems encountered in the Olivier Theater in London [9.4], there should be limits to the opening angle of large fan-shaped theaters.

In a modern theater, the floor slope will often be more modest than seen in Fig. 9.42. In contrast to the lecture hall, a modern drama theater also needs to be highly flexible regarding the layout of stage and audience areas.

The classical proscenium frame inherited from the Italian Baroque theater is often considered to limit intimacy as well as the creativity of the stage-set designers and directors. If the classical proscenium stage is not needed, intimacy and audience involvement can be maximized by an arena arrangement, with audience all around the stage. In other cases, the stage can be extended into the audience chamber, for instance by removing the first rows of seats and raising the floor below to stage level. If such a flexible forestage is considered, the line-of-sight design should be adjusted accordingly, particularly if the hall features balconies.

An example of a flexible modern theater (without balcony) is seen in Fig. 9.43. In this room, the circular side/rear wall was made highly sound absorbing, leaving most of the distribution of early reflection energy to the ceiling.

Dedicated drama theaters are seldom built with a seat count higher than 500–1000, because visual as well as acoustic intimacy, including close view of facial expressions and freedom from artificial amplification, are given high priority. Thus, the maximum distance from the stage front to any seat should not exceed about 20 m for drama performances.

From this discussion it can be concluded that a reflecting ceiling of modest height is the main source of early reflection energy in theaters. However, lighting staff will often want to penetrate this surface with slots for stage lights and rigging or place light bridges across the room, which will act as possible barriers for the sound paths going via the ceiling. This challenge must always be considered in the acoustic design of drama theaters.

9.7.2 Opera Halls

There is a 350 year tradition of performing opera in halls of the Italian Baroque theater style. These are halls with a horse-shoe plan shape with several balcony levels running continuously from one side of the proscenium arch

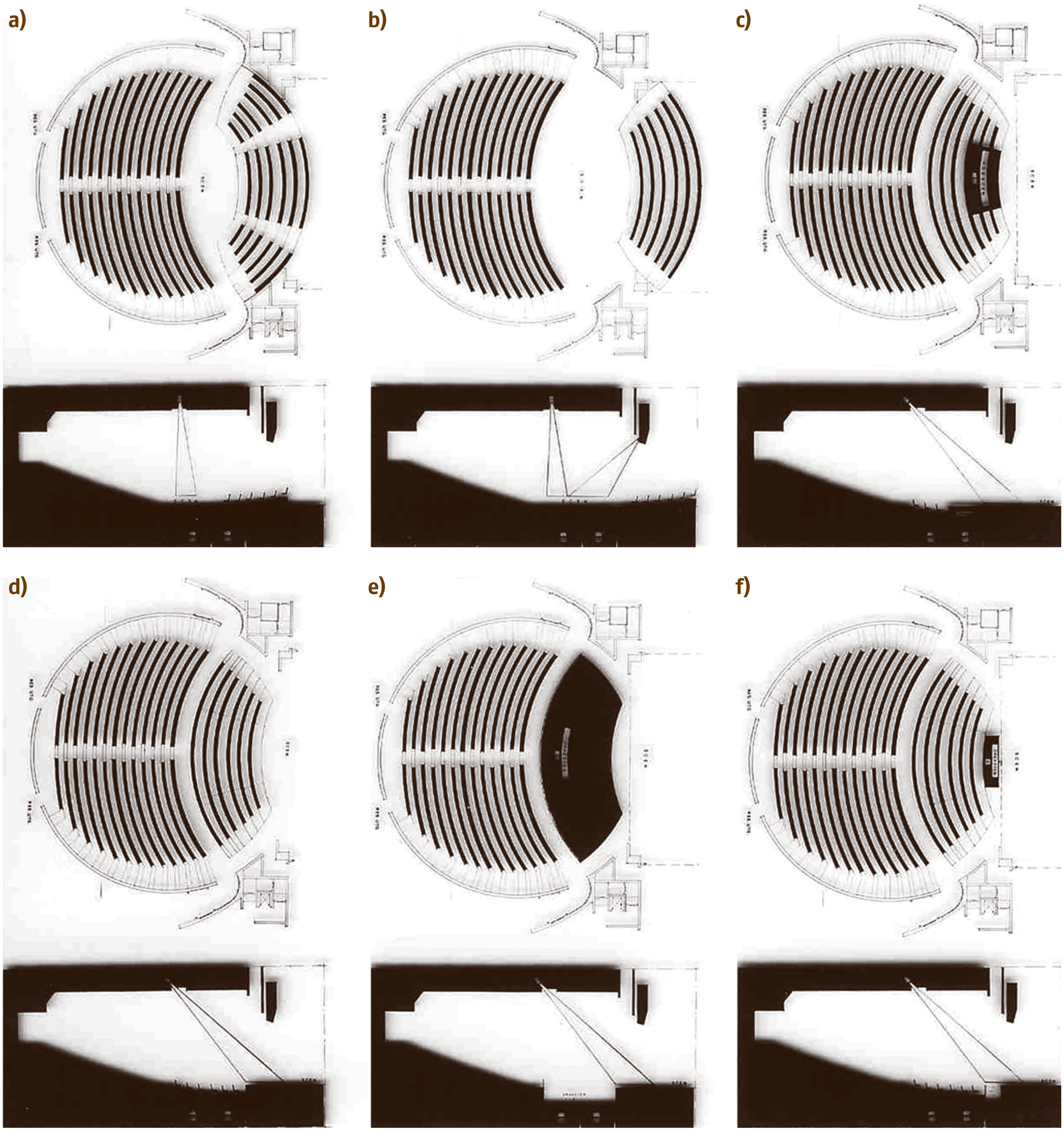


Fig. 9.43a–f Plans showing a flexible theater in Hëlsingborg, Sweden, with different layouts of the stage, orchestra pit and audience seating. **(a)** and **(b)**: arena with small and large stage respectively, **(c)**, **(e)**, and **(f)**: proscenium theater with three different sizes of orchestra pits. **(d)**: proscenium theater without pit

to the other along the curved side and rear walls. The balconies are either divided into boxes separated by thin walls or are open with long seat rows. The orchestra is placed in front of the proscenium opening that connects the auditorium with the stage house.

This tradition has undergone few changes during the past 350 years. Since the 19th century, the orchestra has been placed in a lowered pit separated from the stalls by a closed barrier. In this arrangement, seats in the stalls and lower rear balcony experience an attenuation of the direct sound from the orchestra, because the pit rail blocks the free line of sight to the orchestra. The result may be an improved singer-to-orchestra balance in these seating areas. The pit rail will also reflect sound back and forth between the stage and pit, giving improved contact between the singers and the lowered orchestra.

Another development is that the separate boxes have been exchanged for open balconies, which improves the possibility for the now exposed side walls to help distribute the sound to seating farther backwards in the auditorium. This development is the result of advances in building technology as well as of changes in social attitudes.

The multiple levels of horseshoe-shaped balconies facilitates a short average distance between the audience and the singers, promoting both the visual and the acoustic intimacy, which is important for both parties. However, balcony seats close to the proscenium and on the upper balcony levels often have a very restricted view of the stage.

The short audience–singer distance obtained by the horseshoe form gives high levels of both strength and clarity. On the other hand, the horseshoe form and the proscenium arch also cause some acoustic problems:

- with all wall surfaces covered by balconies, only small free wall areas are left to generate the early reflections that assist an even distribution of the sound, clarity and the generation of reverberation in the room;
- often focusing effects occur from the curved wall surfaces and a cupola-shaped ceiling which, along with the placement of the orchestra out of sight in the pit, give a high risk of false localization of certain instruments;
- only part of the energy generated by the singer will reach the auditorium, because the singers are placed in a large stage house coupled to the auditorium via a proscenium opening of limited size, and this energy even depends on the acoustic properties of the ever-changing stage sets.

Recent developments in the acoustic design of opera halls have mainly aimed at avoiding these problems. Only a few attempts have been tried to radically change the overall shape (e.g. the Deutsche Oper in Berlin, and the Opéra Bastille in Paris) but without much success. Therefore, we are left with tradition and acoustic properties of horseshoe halls to shape the current acoustic ideals for opera, which are

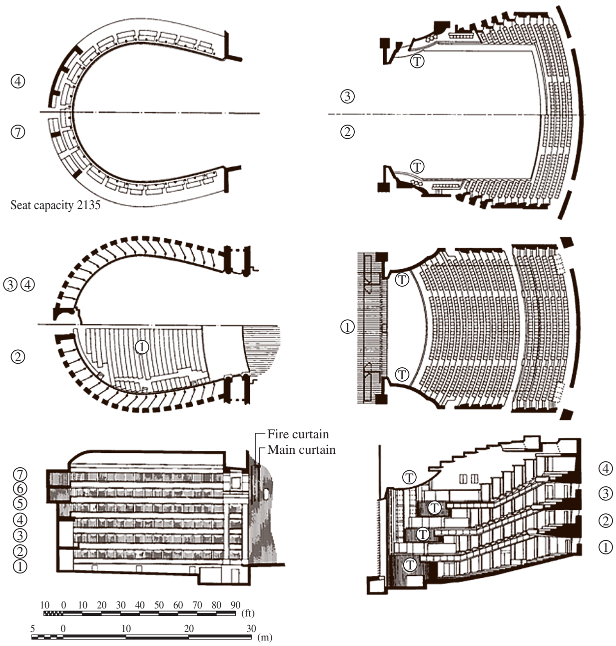


Fig. 9.44 Plan and section drawings of the Teatro alla Scala, Milan, Italy, from 1778 (*left*) and the New National Theatre Opera, Tokyo, Japan, from 1997 (*right*) (after [9.3])

- sufficient level of the singers' voices relative to the orchestra,
- some emphasis on *clarity* and *intelligibility* (although not as much as in speech auditoria),
- a certain *fullness of tone* and *reverberance*; but less than in a classical concert hall.

In recent years the trend has moved towards higher T values than found in the old halls, like Opéra Garnier, Paris (2131 seats, 1.1 s), Teatro Alla Scala, Milano (2289 seats, 1.2 s). This might well be due to the tendency to perform operas in the original language and display the translated text on a text board above the proscenium. Thus, for many recent opera halls we find T values of 1.4–1.8 s. Examples are: Göteborg (1390 seats, $T = 1.7$ s), Glyndebourne (1200 seats, $T = 1.4$ s), Tokyo (1810 seats, $T = 1.5$ s), and Semper Oper, Dresden (reconstruction, 1290 seats, $T = 1.6$ s).

As the width of the traditional horseshoe shape is about equal to its length, the lateral sound is not very pronounced and so *spaciousness* is seldom regarded a factor of particular importance in opera hall acoustics.

In order to ensure a sufficient level and *clarity* of the singers' sound, efficient sound reflections off the proscenium frame, ceiling, balcony fronts and from the corners between balcony soffits and rear/side walls are important. Of course the scenery may also be designed to help in this respect, but generally this is out of the hands of the acoustician.

If a seat count larger than, say, 1500 is demanded it is better to extend the top balcony backwards, as seen in Covent Garden in London and the Metropolitan in New York rather than increase the overall dimensions of the horseshoe, which would give poorer acoustics for all seats. As this extended balcony is close to the ceiling surface, the acoustics here can often be very intimate, clear and loud, in spite of the physical and visual distance to the stage being very large. In the Metropolitan Opera in New York the farthest seat is about 60 m from the stage.

The culmination of the old Baroque theater tradition is represented by the Teatro alla Scala in Milan, Italy, which dates from 1778. Drawings of this theater are shown in Fig. 9.44 along with a modern opera with almost the same number of seats, the New National Opera in Tokyo from 1997.

The horseshoe-shaped Milan opera contains six levels of balconies subdivided into boxes with the openings covered with curtains so that the audience could decide whether to concentrate on the play or on socializing with their guests or family in the box. The audience in the boxes receive a rather weak sound unless they are very close to the opening.

This aspect has been improved in the modern opera with open balconies. Steeper slopes on the stalls floor and even on the side balconies have also improved the direct sound propagation to all seats. Fewer balcony levels have created higher openings at the balcony fronts, allowing the reverberant sound to reach the seats in the last rows below the balconies, providing more early reflection energy from the walls to all seats.

In the Tokyo opera, much emphasis has been put on retracting the balconies away from the proscenium so that extensive side-wall areas close to the proscenium are available for projecting the sound from the stage into the auditorium. Along with the ceiling above the pit, these wall areas almost form a trumpet flare. In the section, it is also seen that the side balconies are tilted downwards towards the stage in order to improve lines of sight. Finally, the side-wall areas and the side-balcony fronts have been made straight to avoid focusing and improve lateral reflection energy.

9.7.3 Concert Halls for Classical Music

Public concerts with large orchestras started in Europe about 250 years ago in halls that resembled those in which the composers normally found audience for their nonreligious music such as the salons and ballrooms of the nobility and courts. It is likely that this tradition as well as the limited span of wooden beam ceilings are the main reasons why many of the old concert rooms were built in what we now call the classical shoe-box shape: a narrow rectangle with a flat floor and ceiling and with the orchestra placed on a platform at one end.

The most famous of the shoe-box-shaped halls is the Musikvereinsaal in Vienna, Austria, shown in Fig. 9.45. This hall, dating from 1870, has a volume of 15 000 m^3 and seats 1670 people. Today, we understand many of the factors responsible for its highly praised acoustics:

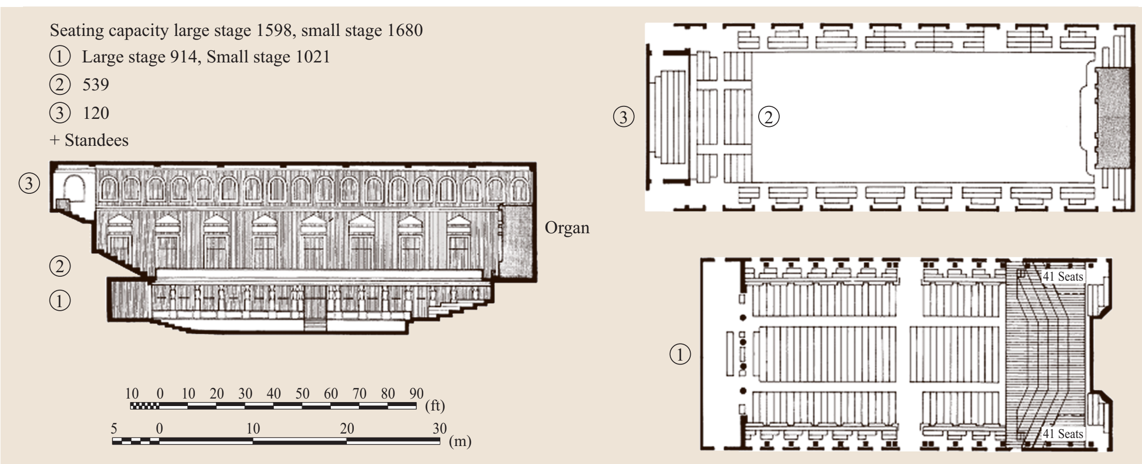


Fig. 9.45 The Musikvereinsaal, Vienna. Balcony and floor plans and long section (after [9.3])

- The volume per seat is only $9\,\mathrm{m}^3$; but T is still 2.0 s in the fully occupied hall because the seating area is relatively small (the seats are narrow and the row-to-row distance is small) and because of the flat main floor. Thus, the sound becomes both loud and reverberant.
- The reverberation also benefits from the fact that the absorbing audience (apart from a few on the top rear balcony) is placed in the lower half of the volume, whereby a *reservoir* of reverberant energy can exist in the upper part of the room.
- The close side walls create strong lateral reflections giving high *clarity* and *envelopment*, and all surfaces are modulated with rich ornamentation, which prevents echoes and causes the room reflections to sound soft and pleasant.

There are however elements of the acoustics, and lack of comfort, that we would not accept in a modern concert hall. The sound is somewhat unclear and remote in the rear part of the flat main floor due to grazing-incidence attenuation. Also, in the side balconies only people sitting close to the balcony front can see well. Other old and famous shoe-box halls (such as the Boston Symphony Hall) have too-shallow balcony overhangs, which results in weak reverberant energy below the balconies.

Because of the general and proven success of the shoe-box design, many new halls are built with this shape. However, in most of the modern shoe-box halls, the drawbacks mentioned above are avoided by a slightly sloping stalls floor and a different layout of the side balconies.

One of the modern shoe-box halls in which the above modifications have been introduced is the Kyoto Concert Hall, Japan, shown in Fig. 9.46. Comparing Fig. 9.45 and Fig. 9.46, the increased floor slope and the subdivided side balconies turned towards the stage are clearly seen.

Recalling the influence of floor slope on T (and other parameters, Sect. 9.5.6) the increased slope will result in a reduced T and higher C. In modern designs T will also tend to be reduced by more-spacious seating and less-shallow balcony overhangs, which result in a larger absorptive audience area being exposed to the sound field. All these factors call for a larger volume per seat in modern shoe-box halls than found in the classical ones, if a high T value is to be maintained. On the other hand, this might well be the reason why the newer halls seldom provide the same intensity and warmth as experienced when listening from one of the sparsely upholstered wooden chairs in Vienna.

Early in the 20th century the development of concrete building technology allowed architects to shape the halls more freely. At the same time it became popular to treat sound as light, by designing the surfaces by means of geometric acoustics and by trying to transmit the sound as efficiently as possible from the source on stage to the (absorbing) audience. Another desire was to put the audience as close as possible to the stage.

The result was the fan-shaped concert hall, which dominated modernist architecture between about 1920 and 1980. An example is seen in Fig. 9.47: Kleinhans Music Hall in Buffalo, USA. Comparison with Fig. 9.45 reveals a very wide hall with a relatively low ceiling and the stage surrounded by concave, almost parabolic, walls and with the ceiling projecting the sound into the absorbing audience area after just a single reflection. This phenomenon, as well as the low volume per seat (about $6.4\,\mathrm{m}^3$), are the reasons for the resulting low reverberation time and the weak sense of *reverberance*. The audience receives most of the sound from frontal directions or from the ceiling, so the sound is clear but lacks envelopment. Other examples of modernist fan-shaped halls are, the Salle Pleyel in Paris (1927) and the old Danish Broadcasting Concert Hall (Copenhagen, 1946). In conclusion, few concert halls from the modernist era have been acoustically successful.

Fortunately, mainly as a result of advances in room acoustic research, two alternative and more-promising designs appeared in the 1960s and 1970s: the *vineyard*

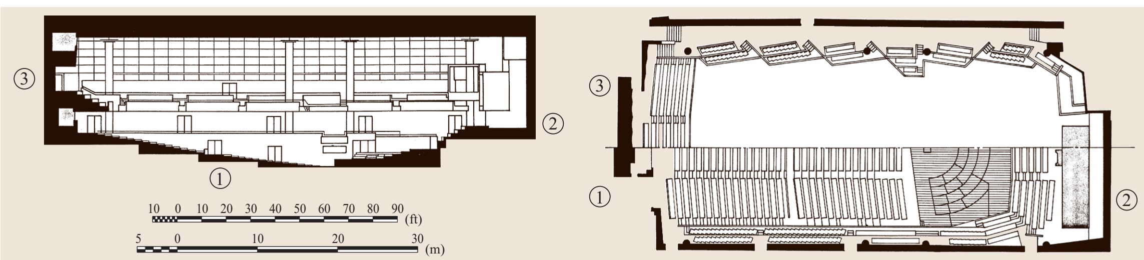


Fig. 9.46 Kyoto Concert Hall, Japan. Built 1995, volume $20\,000\,\mathrm{m}^3$, 1840 seats, RT = 2.0 s (after [9.3])

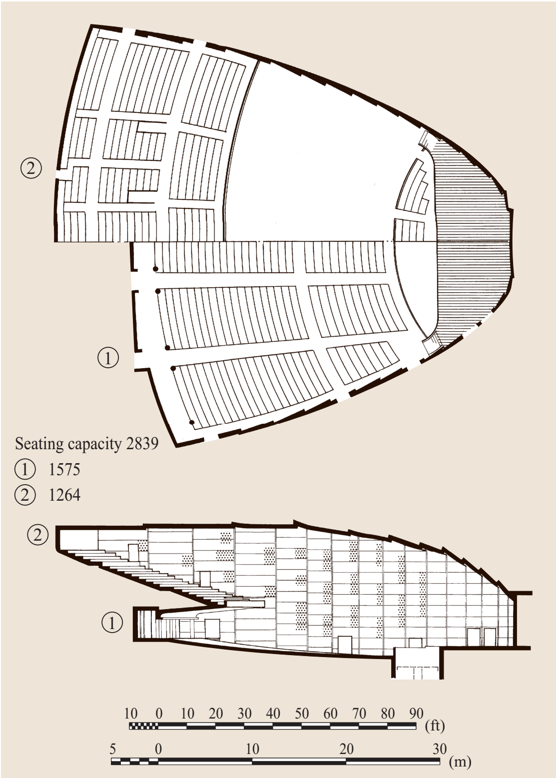


Fig. 9.47 Kleinhans Music Hall, Buffalo, USA. Built 1940, volume 18 240 m^3, 2839 seats, RT = 1.5 s. Plans and long section (after [9.3])

and the *directed reflection sequence* (DRS). Both are arena-shaped with the possibility of the audience being placed all around the orchestra platform. Even more than the fan shape, this causes the audience to come closer to the orchestra for the sake of intimacy. However, both designs are primarily relevant for larger halls, say more than 1500 seats. For halls seating more than about 2000, these designs may even be more successful than the classical rectangular shape, which, with so many seats, possess a risk of some listeners being placed too far from the stage.

The first vineyard concert hall was the Philharmonie Berlin, shown in Fig. 9.48, which opened in 1963. This hall has no balconies. Instead, the seating area is subdivided into terraces elevated relative to each other, whereby the terrace fronts and sides can act as *local* reflectors for specific seating areas. By careful design of these terraces, it is possible to provide plenty of early, and even lateral, reflections to most of the seats in an arena-shaped hall. With no seats being overhung by balconies, a large absorptive area is exposed to the sound, whereby a generous volume per seat is recommended. Thus, in the Sapporo Concert Hall in Japan (1997) as well as in the new Danish Radio Concert Hall in Copenhagen, the volume per seat is about 14 m^3. This is a quite high value for concert halls, as a 2 s T value should normally be obtained with just 10 m^3 per seat.

In *directed reflection sequence* halls such as the Christchurch Town Hall shown in Fig. 9.49, most of the early reflections are provided by huge, suspended and tilted reflectors. These are distinctly separate from

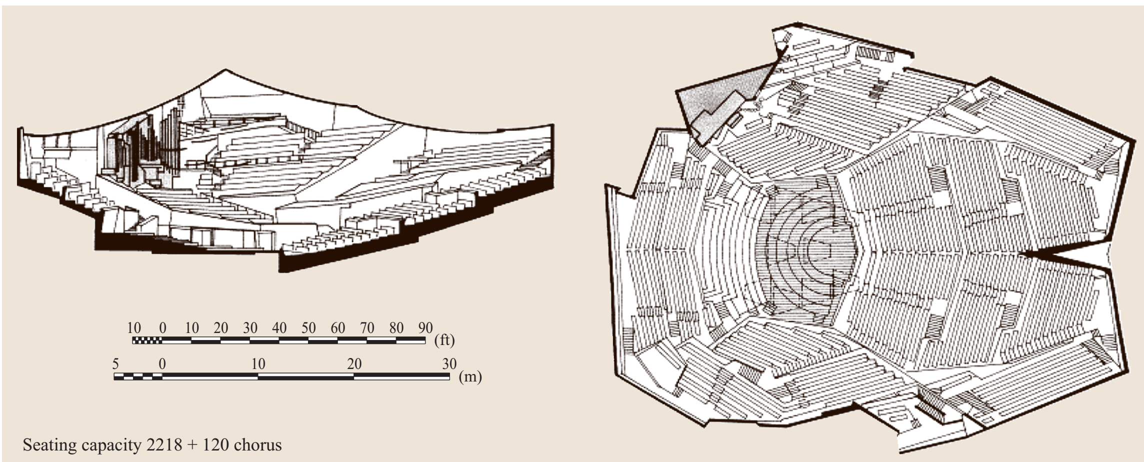


Fig. 9.48 Plan and section of Philharmonie Berlin, Germany. (Built 1963, 2335 seats, $V = 21\,000\,m^3$, $T = 2.1$ s) Acoustician: Lothar Cremer (after [9.3])

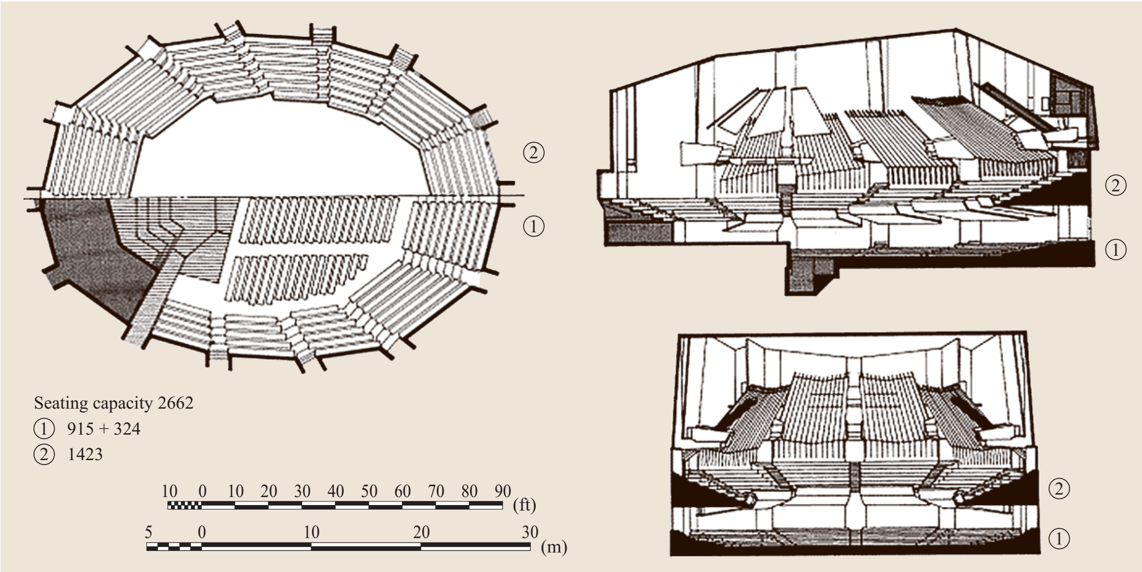


Fig. 9.49 Christchurch Town Hall, New Zealand. (Built 1972, 2662 seats, $V = 20\,500\,\mathrm{m}^3$, $T = 2.4\,\mathrm{s}$) Acoustician: Harold Marshall (after [9.3])

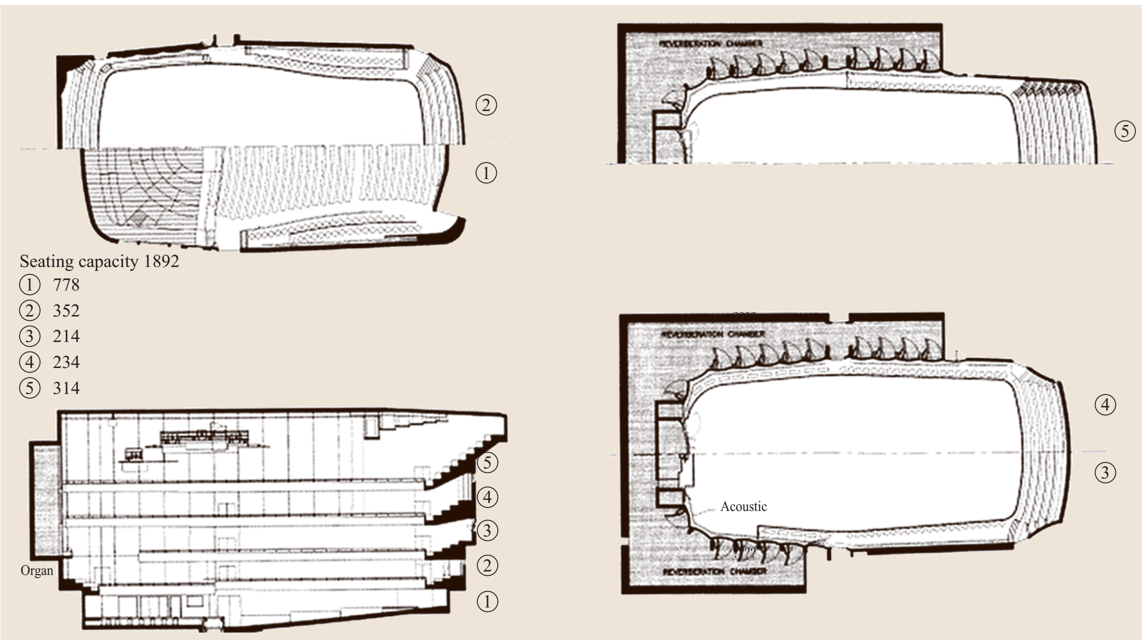


Fig. 9.50 The KKL Concert hall in Luzern, Switzerland. (Built 1999, 1892 seats, $V = 17800 + 6200\,\mathrm{m}^3$, $T = 1.8–2.2\,\mathrm{s}$) Acoustician: Russell Johnson (after [9.3])

the boundaries, which define the reverberant volume. The fronts and soffits of the sectioned balconies surrounding the main floor and stage in the elliptical plan provide additional early reflections. Because of the arena layout combined with the extensive use of balconies (containing more than half of the seats) the hall is very

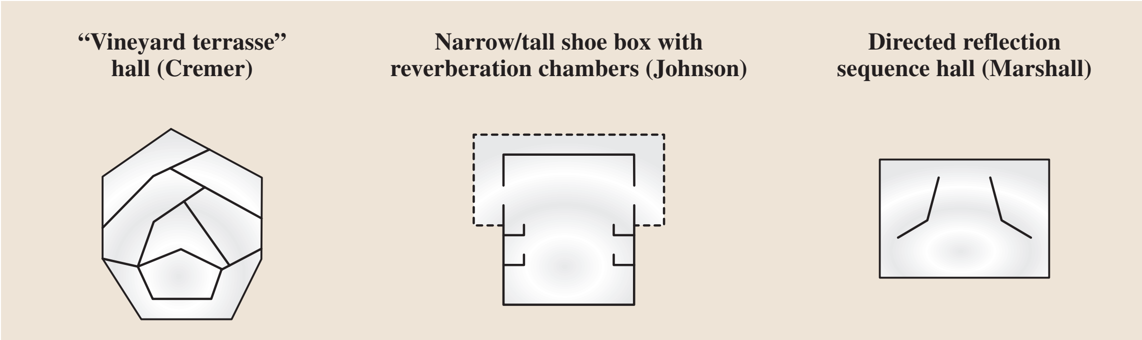


Fig. 9.51 Three concert hall concepts with the potential of combining high clarity with long reverberation. *From left to right*: vineyard (plan), shoe box with coupled reverberation chambers (cross section), and directed reflection sequence (cross section)

intimate, and the large reflecting surfaces provide a stunningly clear, dynamic and almost too loud sound. Still, the reverberation is long and rich because of the significant volume without absorption above/behind the reflectors.

This aim towards a combination of high *clarity* and high *reverberance*, probably developed through our extensive listening to recorded music, requires close reflecting surfaces as well as a large volume. The natural consequence of this demand is to separate the early reflecting surfaces from the room boundaries. Both the terrace fronts in the vineyard halls and the suspended reflectors in the DRS halls offer this possibility. A third way to achieve high *clarity* as well as high *reverberance* is a narrow shoe box with added surrounding volume, as found in a number of halls since about 1990. In these halls the volume of the narrow tall shoe-box hall providing the early reflections is quite moderate; but an extra volume is coupled to the main hall through openings that can be closed by heavy doors, so that the total volume can be varied. However, in such halls one should be careful to make the coupling area large enough for the added volume to have any significant effect (unless one sits close to one of the open doors), otherwise it is hard to justify the enormous costs of the extra volume and door system. With weak coupling, a double-sloped decay curve with a knee point perhaps 20 dB down is created, whereby the added, longer reverberation becomes barely audible except during breaks in the music.

A recent design of such a rectangular hall with a coupled volume is shown in Fig. 9.50.

The sketches in Fig. 9.51 summarize the basic design of the three types of halls mentioned. It is seen that they all possess the possibility of separating the surfaces that generate the early reflections from the volume boundaries that generate the reverberation.

9.7.4 Multipurpose Halls

Many modern halls built for cultural purposes often have to accommodate a variety of different types of events, from classical to pop/rock concerts, drama and musicals, opera, conferences, banquets, exhibitions, cinema and perhaps even sports events. From an acoustical point of view, the first concern in these cases is whether a variable reverberation time will be necessary. The answer is yes in most cases where some functions primarily require intelligibility of speech while others require a substantial reverberation, such as for classical music.

A hall in which these demands have been met by means of variable absorption is Dronningesalen, at the Royal Library in Copenhagen (Fig. 9.52). For this hall both chamber-music concerts and conferences with amplified speech were given high priority. The variable absorption is provided by means of moveable panels on the side walls as well as by folded curtains and roller blinds on the end walls. Combining these measures in different ways, T values in the range from 1.1 to above 1.8 s can be obtained (in the empty hall) as seen in Fig. 9.52.

In many cases, stage performances with extensive scenery are also required. The most common way to accomplish this is to design a stage house, mount an orchestra shell on the stage and raise the pit to stage level for concerts. If a hall is also to be used for banquets and exhibitions requiring a flat floor, it is common to place the stalls seats on a telescopic riser system on a flat floor instead of having a fixed, sloped seating. When not in use, this riser system is stored along the rear wall or under a balcony. If the reverberation time suits classical music concerts with the sloped seating in place, a substantial area of variable absorption is needed to reduce the T value when the chairs are absent. Another problem may be that the telescopic systems offer only a rather steep and linear slope, which is not optimal for classical concerts. If the chairs are fixed on the riser steps, the minimum step height is about 25 cm, corresponding to a rake of about 25%.

If a change in T is accomplished by means of variable absorption, G will be reduced along with lowering T. However, if instead T is lowered by reducing the volume, for instance by moving a wall or lowering the ceiling, G will remain approximately constant or perhaps even increase. This can be advantageous when the low-T setting is to be used, for instance for unamplified drama or for chamber music in a larger symphony concert hall.

An example of a hall with variable volume is found in Umeå, Sweden (Fig. 9.53). In this hall the auditorium volume can be adjusted by moving the proscenium wall between three positions. When the proscenium is stored against the rear wall in the *concert* format, the entire volume is available for concerts with large symphonic orchestras on an open stage. In this situation, the stage tower can be closed off by horizontal panels at ceiling level. But when the proscenium is moved forward to the *opera* setting, the auditorium becomes smaller and a proscenium stage area is created, while the ceiling panels are removed for access to the fly tower above. In the third position, *drama*, the auditorium volume is further reduced to create an intimate theater. The variable elements also include a hinged side-wall section to improve the shape of the room for theater as well as moveable reflectors and variable absorption curtains above the grid ceiling level.

It should be mentioned that many of the problems related to the successful design of speech and music auditoria become more severe with increased size of the room. In other words, it is much easier to design a hall for less than 1000 people, like the two examples presented above, than for 2000 plus. The problems become even more complicated if multipurpose function is requested.

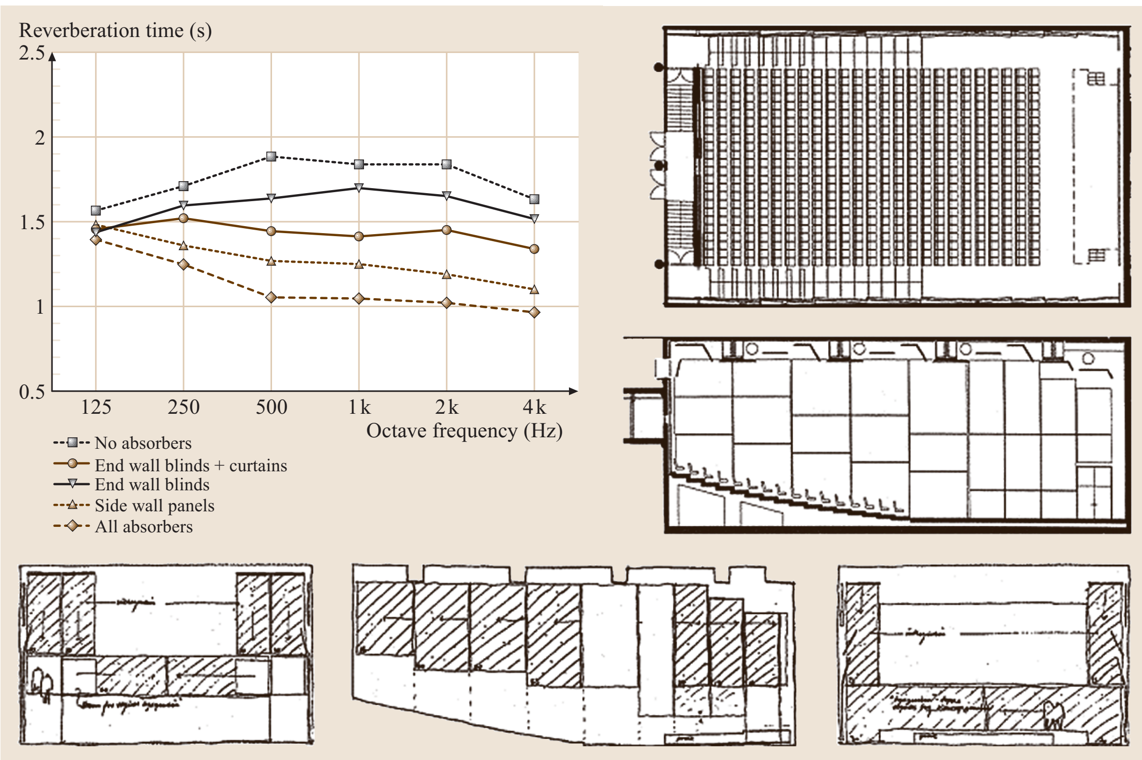


Fig. 9.52 Dronningesalen in the Royal Library in Copenhagen, Denmark; 1999, 400–600 seats. Reverberation time curves, plan, section and wall elevations with *hatched areas* indicating variable absorption

Built 1986

Drama

$V = 5000\,\mathrm{m}^3$
$T = 1.1\,\mathrm{s}$

Opera

$V = 6000\,\mathrm{m}^3$
$T = 1.3\,\mathrm{s}$

Concert

$V = 9000\,\mathrm{m}^3$
$T = 1.8\,\mathrm{s}$

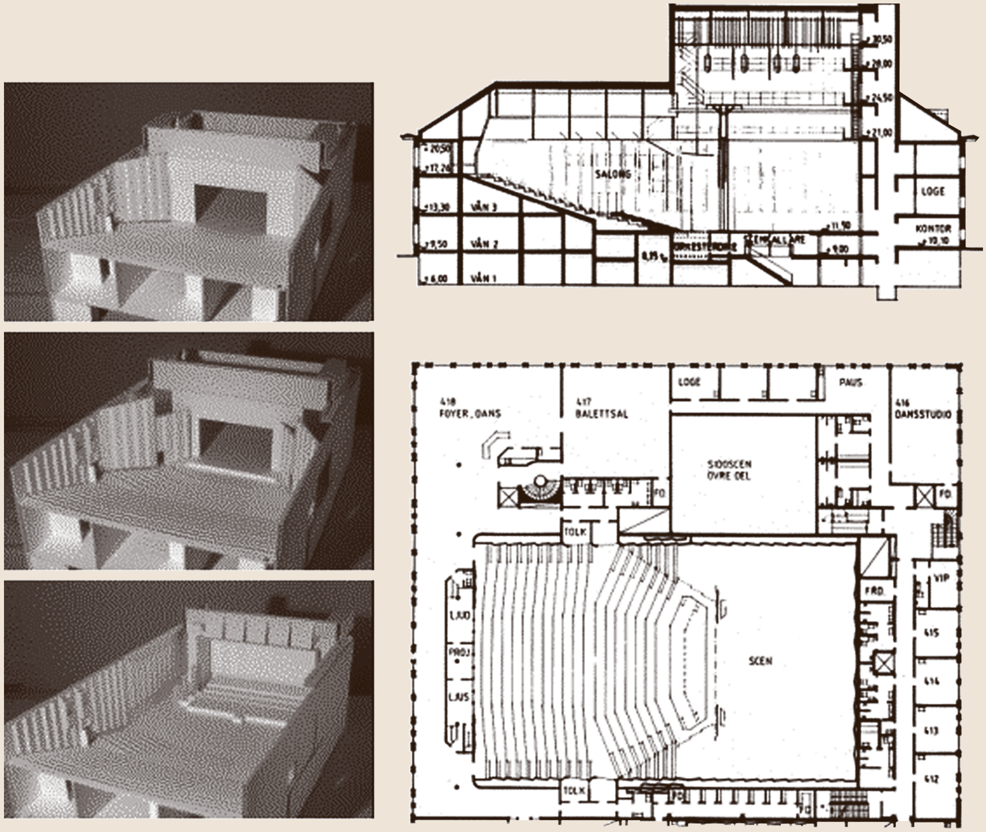


Fig. 9.53 The Idun multipurpose hall in Umeå, Sweden, with variable volume and stage configurations. The hall seats up to 860 people. Brief data (*left*), photos from cardboard model, long section and plan (*right*)

Still, a few examples of such successful designs exist. The most innovative is probably the Segerstrom Hall in Orange County, California, USA, which is shown in Fig. 9.54. This hall features a special layout of the almost 3000 audience seats, which are distributed on four levels. Each of these forms an almost rectangular, or even reverse fan, shape. However, the orientation of each level is shifted so that the overall plan becomes a wide fan. In this way the virtues of the fan shape for audience proximity to the stage are combined with the advantages of the rectangular shape for strong, early reflections from lateral directions. Thus, although the total width is about 50 m, the lateral energy level in this hall resembles that found in rectangular, 25 m-wide concert halls with typical seating capacity up to, say, 2000 people [9.36].

9.7.5 Halls for Rhythmic Music

Halls intended for rhythmic, amplified music concerts range in size from less than 100 to perhaps more than 50 000 listeners (in roofed sports arenas). The reason for this broad range of audience capacity is that the size and number of the sound amplification loudspeakers as well as of event video screens can be chosen at will to ensure good vision and adequate (often too high) sound levels for any size of audience. In any case, the reverberation time should be close to the optimal for speech, i. e., not more than 1 s if possible.

In huge arenas, of course, the reverberation time will be higher; but if efficient absorption is applied to the critical surfaces, a higher reverberation time need not compromise clarity. This is because the reverber-

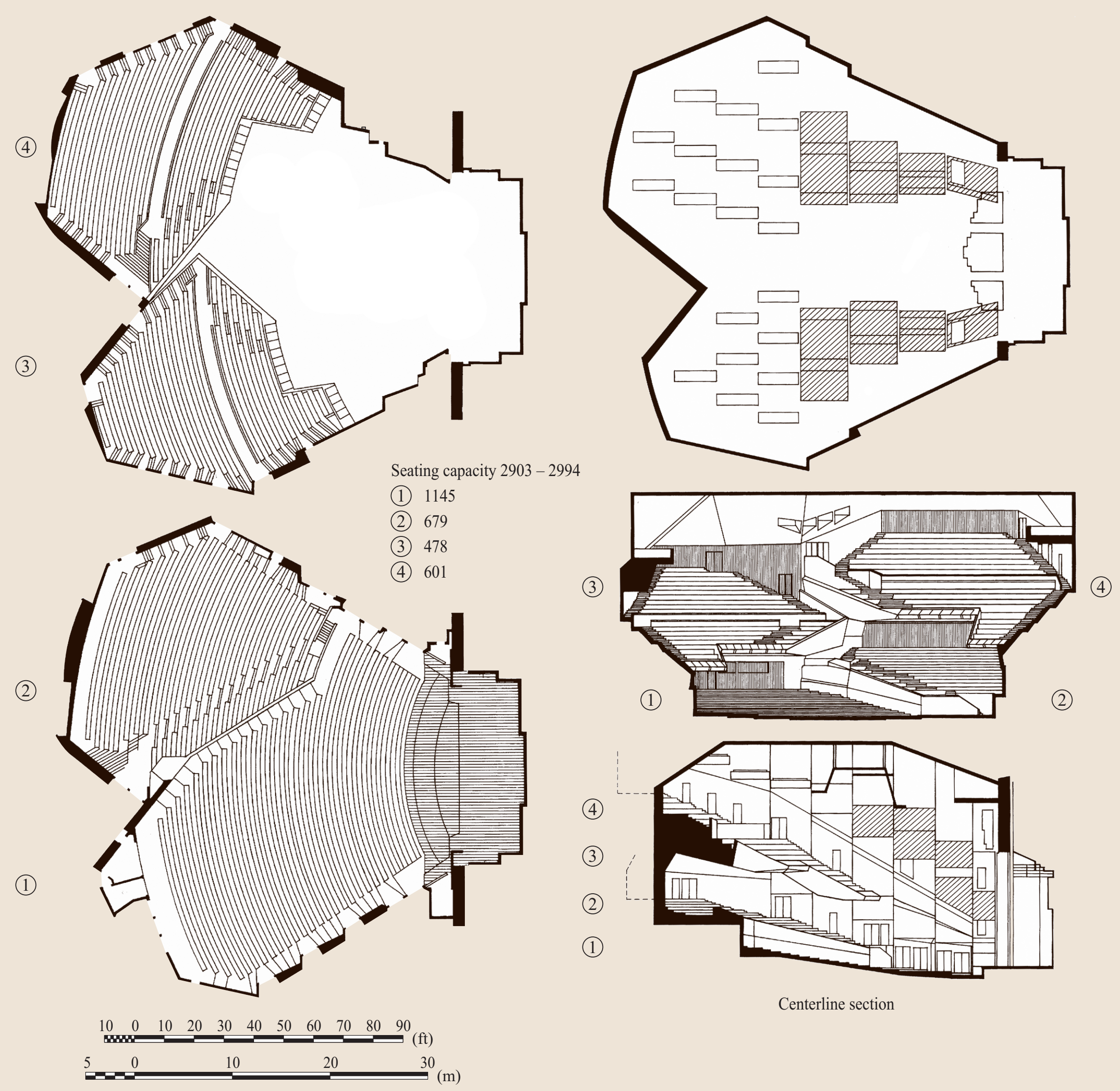


Fig. 9.54 Segerstrom Hall, Orange County, California, seating 2900 people. Acoustic design: Marshall, Hyde, Paoletti (after [9.3])

ation level will be low compared to the direct sound from the loudspeakers. In large spaces, designed to have a moderate T value, one must be careful to avoid echoes from distant, hard surfaces facing the often highly directional loudspeakers. Even small surfaces such as doors can cause echo problems if most other surfaces in the room are efficiently absorbing.

It is also very important to maintain modest low-frequency T values. A bass ratio larger than unity can create serious problems with control of the low-frequency levels. This is largely because most loudspeakers are almost omnidirectional at low frequencies, at which they willingly excite the room *reverberance*. In contrast, with high loudspeaker directivity in mid- and high-frequency range, it is easy to control the level in the audience area at these frequencies without spilling additional sound energy into the room.

Unfortunately, low-frequency reverberation control can be difficult to achieve, because low-frequency absorbers are normally less efficient (have lower α values) than mid-/high-frequency absorbers, and in a room in which these already occupy most of the available surfaces, it can be difficult to also find spaces for dedicated low-frequency absorbers. In particular, this poses a problem in buildings made from heavy concrete or masonry (which on the other hand is favorable for insulation of low-frequency sound towards the neighbors). When the low-frequency reverberation is too high the string bass and pedal drum will sound muddy. The audience area layout should be shaped to allow an even coverage by a sensible loudspeaker set up, and in large halls, a sloping floor is not suitable if a standing audience is expected. Fortunately, a sloping floor is not always mandatory, since in larger venues, the loudspeakers can be elevated to avoid grazing-incidence attenuation and elevated video projection screens improve the visual experience. It is advantageous to design the hall as well as the loudspeaker system so that zones with lower levels are also created, whereby people can have a chance to rest their ears, for instance while picking up refreshments in the bar. This may be accomplished by placing the bar in a smaller, partly decoupled, space with a shallower room height and an efficiently absorbing ceiling.

9.7.6 Worship Spaces/Churches

The liturgies in old Christian and Muslim worship were developed in highly reverberant cathedrals and mosques, both of which have their roots in Byzantine architecture. When new churches are built for the traditional Catholic or Protestant Christian denominations, it is customary to aim at a rich reverberation, which suit the pipe organ literature and congregational singing, while the intelligibility of the words by the priests is taken care of by a distributed public address (PA) sound system.

The same is true for new large mosques, in which the speech is amplified while the long reverberation supports traditional singing by the imams. Apart form these songs music is not performed in mosques.

In contrast to this, several Christian denominations, particularly in the US, have built new churches or temples (often accommodating several thousand people) and brought in newer art forms making use of large rock-music sound systems. In these spaces, therefore, the solution is often for the acoustician to create fairly dry, natural acoustics and to install both a heavy PA sound system as well as an artificial, electronic reverberation-enhancement system. Both types of sound systems are briefly described in Sect. 9.8.

9.8 Sound Systems for Auditoria

It is important to realize that sound systems will always be installed in both new and existing halls. As these systems are often used to amplify natural sound sources appearing on the stage in the hall, they may be regarded as a means for creating variable acoustics of the space. In this regard, the following will focus on how these systems interact with the natural acoustics, rather than talking about specific details or trends in the design of the loudspeakers themselves.

In principle, one may consider two different types of loudspeaker to be installed in an auditorium: traditional PA systems intended for the modification of the early part of the impulse response of the hall, and reverberation-enhancement systems intended for modification of the later part. Below, important aspects of both types will be briefly explored.

9.8.1 PA Systems

Normally, PA sound systems are installed with the purpose of increasing the sound level and/or the intelligibility of the performance. Use of the sound system for the reproduction of prerecorded sounds is of less interest in this context, as in this case it just acts like any other sound source working under the acoustic conditions provided by the natural acoustics of the room. In Sect. 9.2 we learned that, in order to improve *clarity/intelligibility* as well as the *level*, we need to increase the clarity C by adding sound components to the early part of the impulse response. (The alternative, to increase C by reducing the late part, can so far only be accomplished by adding physical absorption; but some day in the future, active systems with multiple microphones and loudspeakers integrated into *acoustic wallpaper* could be imagined as well.) Doing this requires that the main part of the sound energy leaving the loudspeakers hits the absorbing audience area. Alternatively, if substantial parts of the energy hit other, reflecting room surfaces, it may be reflected randomly and feed the reverberant field, which will reduce the clarity. For this reason, it is very important to apply loudspeakers with well-controlled directivity for this purpose, especially in auditoria with T values larger than optimal for speech.

For the acoustician, the most important technical specification for PA loudspeakers relates to their directivity, either in terms of vertical and horizontal radiation angles or in terms of the Q factor. Sometimes, the term *directivity index* $\mathrm{DI} = 10\log(Q)$ is used instead. Q is defined as the ratio between

1. the intensity emitted in the axis direction of the directive loudspeaker, and
2. the averaged intensity emitted by the same loudspeaker integrated over all directions.

The higher the Q value, the more the sound emitted will be concentrated in the forward direction (or rather, within the radiation angles specified).

When the loudspeaker is placed in a room, the sound field close to the loudspeaker will be dominated by the direct sound, whereas at longer distances, this component is negligible compared to the (diffuse) reverberant sound, as illustrated in Fig. 11.6. The (squared) sound pressure as a function of distance from the source in a room can be written as follows:

$$p^2(r) = \rho c\, P_{\mathrm{SP}} \left(\frac{Q_{\mathrm{SP}}}{4\pi r^2} + \frac{4}{A} \right) . \qquad (9.38)$$

In (9.38) the first term in the parenthesis is the direct sound component and the second one, $4/A$, the diffuse, reverberant component, P_{SP} is the emitted sound power, Q_{SP} is the Q factor of the loudspeaker, and A is the total sound absorption area of the room. Notice that in (9.38), for simplification, we assume a diffuse field intensity that is constant throughout the room, although in Sect. 9.5.1 and Fig. 9.13 we demonstrated that this is seldom the case.

Equation (9.38) shows that the direct component will dominate as long as the direct term is larger than the diffuse term. This will be the case for distances below the *critical distance* or the *reverberation distance*, $r_{\mathrm{cr,SP}}$. At this distance, the two components have equal magnitude, whereby

$$r_{\mathrm{cr,SP}} = \sqrt{\frac{Q_{\mathrm{SP}} A}{16\pi}} = \sqrt{\frac{Q_{\mathrm{SP}} V}{100\pi T}} . \qquad (9.39)$$

As seen from (9.39), we maximize the range within which we are able to provide the listeners with a clear sound from the speakers by increasing Q (as well as A).

One should be careful not to apply a larger number of loudspeakers than needed for coverage of the audience area, as each new loudspeaker will contribute to the generation of the reverberant diffuse field, while only one at a time will send direct sound towards a given audience position. Likewise, the number of open microphones should be minimized (which is often done automatically by an intelligent microphone mixer), as all open microphones will pick up and amplify the unclear reverberant field in the room.

The number of active loudspeakers and microphones should also be minimized in order to reduce the risk of feedback, which is a consequence of the microphone(s) picking up the sound from the loudspeaker(s) and the amplifier gain being set to high. If the amplification through the entire closed loop (microphone–amplifier–loudspeaker–room and back to the microphone) exceeds unity, the system will start to oscillate at a random frequency, which is emitted at maximum power level into the room, a most annoying experience.

The risk of feedback can be reduced by minimizing the loudspeaker–listener distance and the source–microphone distance, whereby a suitable listening level can be obtained with a moderate amplifier gain setting. Use of head-borne microphones represents the ultimate reduction of the source–microphone distance. Consequently, with such microphones the risk of feedback is substantially reduced. Also, the microphone–loudspeaker distance should be maximized and directional microphones and loudspeakers should be used and positioned so that the transmission from the loudspeaker back into the microphone is minimized. Thus, the loudspeakers and the microphones should be placed pointing away from each other with the microphone pointing towards the stage and the loudspeaker pointing toward the audience.

Considering the factors mentioned above, we still need to choose a configuration of the loudspeakers for a given room. Often the choice is between

1. a central system, consisting of a single or a limited number of highly directional units arranged close together in a cluster over the stage, or
2. a distributed system of several smaller units placed closer to the listeners.

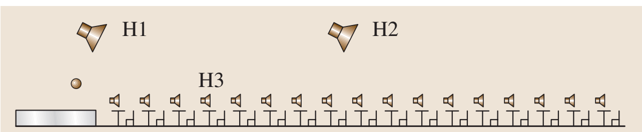


Fig. 9.55 Alternative solutions for loudspeaker coverage in an auditorium: a central (cluster) system H1, the same with a delayed secondary unit H2, or a highly distributed (pew back) system with small loudspeakers placed in each row of seats

These options are illustrated schematically in Fig. 9.55 as H1 and H3, respectively. H2 illustrates a compromise with a secondary unit or cluster assisting the main cluster, H1.

In meeting rooms or churches, highly distributed systems with small loudspeakers built into fixed meeting tables or pews can be very successful. With each loudspeaker being within a distance of say 1 m from the listener, each unit can be set to a very moderate level. Therefore, it is often possible to serve the listeners with a clear direct sound from such speakers without the total number of speakers exciting too much of the room reverberation. If the system is subdivided into independent sections, it is even possible to install automatic switches that only activate loudspeakers in zones where people are sitting. In this way, the reverberant field can be even further reduced in cases of fewer people attending the meeting or service. This is actually very fortunate as, especially in churches, fewer people also means less absorption and louder reverberation. The performance regarding intelligibility of pew-back systems with perhaps one hundred, moderately directive speakers is often equal to or better than when a highly directive central-cluster system is used. This is simply due to the moderate sound level needed from each of the closely placed loudspeakers compared to the much higher level needed for the cluster to reach the distant listeners. As can be imagined, the sound level distribution produced by the pew-back system will also be more uniform.

Of course, the highly distributed systems are not suitable for amplification of stage performances, where a high sound level of music is needed. For this purpose, a main cluster over the stage or a stereo system with loudspeakers on each side of the stage is preferable. For such systems, it is also more important to maintain a correct localization of the sound as coming from the performers on the stage.

In cases where some additional, distributed loudspeakers are needed to assist coverage in certain parts of the room, for instance on and under deep balconies, these are equipped with an electronic delay. If this delay is set so that the sound from the distributed assisting loudspeakers arrives 5–25 ms after the sound from the main loudspeakers, the listener will still perceive the sound as coming from the direction of the source. This is even true when the assisting loudspeakers are up to 5 dB louder than the main cluster at the listener's position. This very convenient phenomenon, called the precedence effect or Hass effect, is generally used as a guideline for the choice of level and delay settings in distributed loudspeaker systems.

9.8.2 Reverberation-Enhancement Systems

For auditoria where a long reverberation time is only needed occasionally or when the available room volume is simply inadequate, one may consider creating more reverberation by means of an electronic reverberation-enhancement system. Like the systems described in the previous section, these also consist of a number of loudspeakers and microphones connected by amplifiers; but there are many important differences.

The basic difference is that enhancement systems primarily attempt to add energy to the late part of the impulse response. Therefore, delays must be built into the signal processing between microphones and loudspeakers. In the first systems of this kind, *assisted*

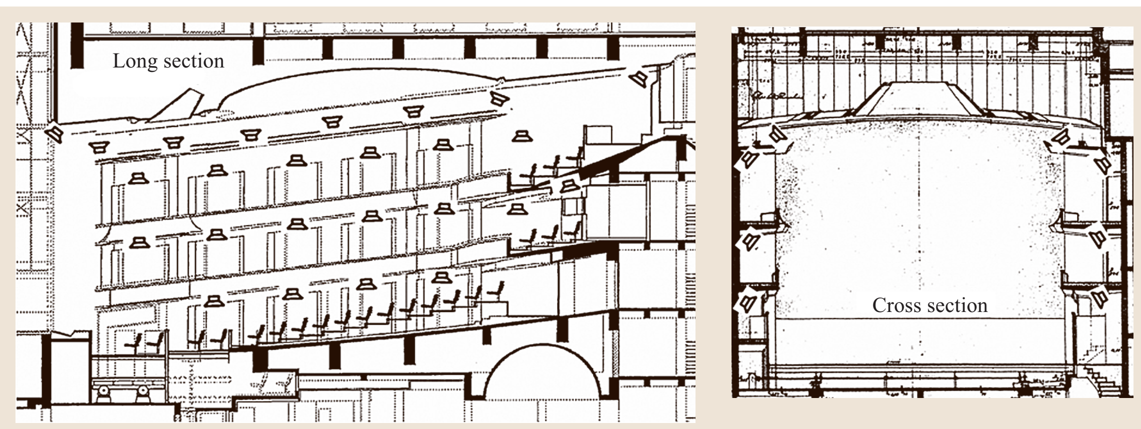


Fig. 9.56 Example of distribution of loudspeakers for reverberation enhancement in a theatre

resonance (AR) and *multichannel reverberation* (MCR), which appeared in the 1960s, this delay was provided by letting the loop gain of a large number of microphone–loudspeaker channels be tuned to just below the feedback limit. By spacing the microphones and loudspeakers far apart in the room (normally close to the ceiling), the sound would spend sufficient time traveling several times between loudspeakers and microphones for an audible prolongation of the reverberation to be perceived. In more-modern systems, electronic delays and reverberators are normally used.

In order to create the illusion of diffuse reverberation coming from the room itself, a large number of loudspeakers with low directivity must be distributed over the main surfaces in the room, as illustrated in Fig. 9.56. Since many speakers are needed, each can be fairly small and with limited acoustic power capability. The density and placement of these speakers as well as the balancing of their levels is essential to prevent localization of individual loudspeakers from any seat.

Regarding the microphones, these may be highly directive, but they must be placed quite far from the sound sources (and preferably out of sight) in order to cover the stage without the reverberant level being strongly dependent on the position of the sound source.

In view of what was explained in Sect. 9.8.1 about the measures needed to reduce the risk of feedback in loudspeaker systems, it is no wonder that most reverberation-enhancement manufacturers have to implement a solid strategy against uncontrolled feedback. One very efficient approach is to introduce time-varying delays in the sound-processing equipment, as this seems to destroy sharp room resonances when they start to build up to form a hauling frequency. Still, most enhancement systems work with a loop gain close to the feedback limit (which is actually participating actively in increasing the reverberation time), which is acceptable as long as it is under control and does not cause audible coloration of the sound.

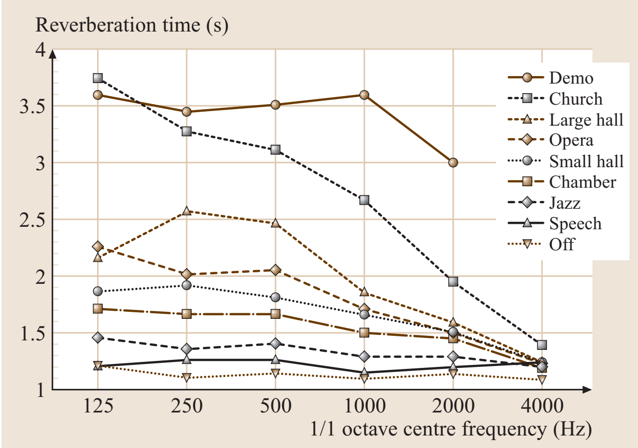


Fig. 9.57 Reverberation time values measured in a hall for different settings of the reverberation enhancement system (LARES)

It is essential for the acceptance of enhancement systems that neither audience nor performers experience the sound as coming from a loudspeaker system. Likewise, artefacts such as coloration due to feedback or the poor frequency characteristics of speakers are unacceptable.

The maximum increase in reverberation time possible with feedback-based systems like AR and MCR was typically about 50%. With modern digital reverberators, a much larger range can be achieved, as shown in Fig. 9.57. Increasing the reverberation time by a factor of three or more is not impossible. However, the illusion, when you still see the physical, much-dryer room around you, tends to break down when the value increases much above the 50%, which ironically is what the old systems could produce.

From a purist standpoint, reverberation enhancement can be regarded as an acoustic prosthesis, but in many cases a pragmatic approach to the use of these systems can increase the range of programs that a cultural venue can host.

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