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FUNDAMENTALS OF ROOM ACOUSTICS

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Abstract

This text has the objective of providing some basic knowledge of room acoustics phenomena and its relation to music performance, necessary for the understanding of the subsequent lectures which will introduce more advanced concepts on architectural acoustics. We will focus on large halls for classical or erudite music performances for which the room acoustics play a vital role. Small rooms, in which the low-frequency resonances emerge in an important part of the audible frequency range, are not considered.

We start by presenting an overview of the relation between music, its history, composing styles and the evolution of some architectural features of the performing space. The interaction between musicians and the musical instruments with the room is also considered. Several important characteristics of these sources of musical sound are described, particularly those which are relevant for the room response to this acoustical excitation. We then proceed to a more theoretical perspective considering sound propagation in the free-field and also the main phenomena resulting from its interaction with surfaces. This leads us to the treatment of sound propagation in closed spaces, which sets the appropriate background for the following and final chapter, in which the main parameters for the evaluation of the acoustical quality of rooms for music are described.

1 INTRODUCTION

The performance of music is inevitably related to the space where it is presented. Concert halls, opera houses or theaters are the "bridges" that bring together the source (musical instrument) and the receiver (audience), changing and hopefully enriching the musical experience. But how does this transmission take place? What effects does it impose on the sound radiating from the instruments? What frequency ranges are important in room acoustics? How do musicians interact with the room acoustical response?

The answer to these questions can be found in quite scientific terms but, for musicians, it only finds its relevance if it is transmitted in a common language level. This means that both researchers and musicians have to learn the meaning of each other's terms in what concerns the acoustics and the its relation with the performers. The most classical term used in room acoustics to qualify its acoustical quality, the *reverberation time*, is defined as the time that will take a sound which is abruptly stooped to decrease in level 60 dB. This definition finds a most clear translation if we refer to the *sound decay* in a room to inaudibility, of which the associated subjective impression is called reverberance. This and other terms will be mentioned and clarified in this text with the objective of achieving a clear correlation between both musical and scientific terminology.

2 MUSIC, MUSICIANS AND THE PERFORMANCE SPACE

The quality of the performance of musicians depends to a high on their interaction with the performance space. Musicians playing different instruments usually ask for different acoustical conditions which are more adequate to the characteristics of their musical instruments. For example, it is understandable that a classical guitar and a violin player require for a more reverberant space than a pianist. A classical guitar played in a large hall usually lacks enough sound power to reach the most distant listeners and therefore benefits from higher reverberation times which increase the perceived loudness. The violinist, on the other hand, can produce higher sound levels but requires the room reverberation to give "sustain" to some musical passages due to the somewhat higher damping characteristics of the instrument. The pianist is in a more comfortable position since its instrument is not only capable of producing high sound levels but also allows the musician to control the decay time of the produced sound. The latter situation is generally in opposition with the former ones as the pianist usually asks for a less "live" room, in which the musical details are clearly perceptible and controllable by the musician.

The previous explanation is valid for most musicians playing solo, but ensemble playing is quite different. Orchestras are already composed of a number of instruments which allow an adequate tonal balance over the whole group. Clearly, a higher number of stringed instruments are necessary to cope with the high sound levels that can arise from the brass section or percussion instruments[1] (see Figure 1). In this case, the relation between the performers and the room is not so dependent on the instrument played but more associated with the type of music being executed. Depending on the adequacy of the room, the conductor and the musicians usually have to adapt, with more or less effort, the performance of a particular kind of music to the specific acoustics.

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[1] Other reasons are also associated to the high number of instruments such as the beneficial *chorus effect* that occurs with a large number of instruments playing vibratos with relative different phases or *tuning masking* with choral groups.

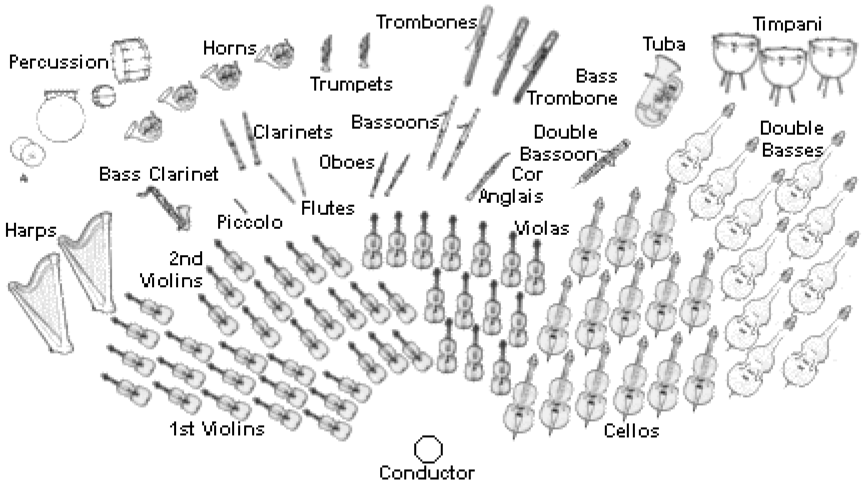

Figure 1 – Example of an orchestra seating plan

Along the several periods in history, architecture and music have been intimately related. The reverberation a composer imagined while working depended on the architecture that dominated the musical performance in his surroundings [1]. During the Medieval Period (1000-1450), music was mostly performed for religious purposes, such as the plain-chant or the Gregorian chant, for example. The long, monophonic musical style associated with high reverberation times of large cathedrals induced a feeling of deep spiritual consciousness and divinity awareness. However, the development of music and particularly polyphony during the Renaissance (1450-1600), gave rise to musical styles which needed quite different architectural features. The contrapuntal style characteristic of the Baroque, explored by composers such as Bach, Handel and Vivaldi, is formed by deeply articulated groups of musical notes which need to be clearly perceived. Although music of this period was also written for more reverberant spaces (consider for example Bach's organ works which brilliantly explore the liveliness of large churches), it was in the relatively dry and intimate acoustics of palace rooms or small theatres (when occupied) that most Baroque music developed [1].

Additionally, the Baroque period is intimately related to the beginning of Opera performances. The need for theatre sets and orchestras to inhabit the same space gave origin to very distinctive "horseshoe" shapes, which are still preferred nowadays (La Scala de Milan, for example), with an audience distributed over the floor and lateral boxes and the orchestra "hidden" in the stage pit. Due to the need for high speech intelligibility, the acoustical characteristics of these spaces are still less reverberant than for music halls.

The Classical period (1750~1820) brought a different composing concept, with melodies being accompanied by the full harmony of orchestral chords such as in the classical symphonies of Mozart and Beethoven. This new style benefited from a more live and slightly less clear acoustical character than Baroque music. This fact associated to the growth of popularity of orchestral performances could indicate a natural increase in room sizes to accommodate larger audiences. However, it was only towards the second half of the 19th century that large concert halls were built [1]. The longer reverberation times of these halls enhanced the fullness of the musical harmonies while the narrow rectangular shapes provided the adequate sound reflections for some degree of articulation clarity.

The last symphonies by Beethoven define a different path for music composition, using large orchestras, choirs and vocal soloists in the same performance. Ultimately this style would develop into the Romantic period, which strived for the expression of strong emotions developing thick textures with full chords and complex rhythms. In this musical style, the need for accurate definition of musical notes, as in the Baroque and the Classical period, was less important. The room acoustics should allow the transmission of a great emotional or dramatic power which implied much longer reverberation times. Composers of this period sometimes wrote with a specific concert hall in mind. Wagner, for example, designed an Opera house for the needs of his compositions (the Festspielhaus in Bayruth, Germany), which in contrast to the Opera styles of the previous periods were best supported by high fullness of tone and relatively low definition. Nowadays, some of the most famous concert halls share the characteristics which favoured the music of this period. The Musikverein in Vienna (1870), the Concertgbow in Amsterdam (1888) and the Boston Symphony Hall (1900) are rated as some of the best in the world with reverberation times between 1.9 and 2.0 seconds [1].

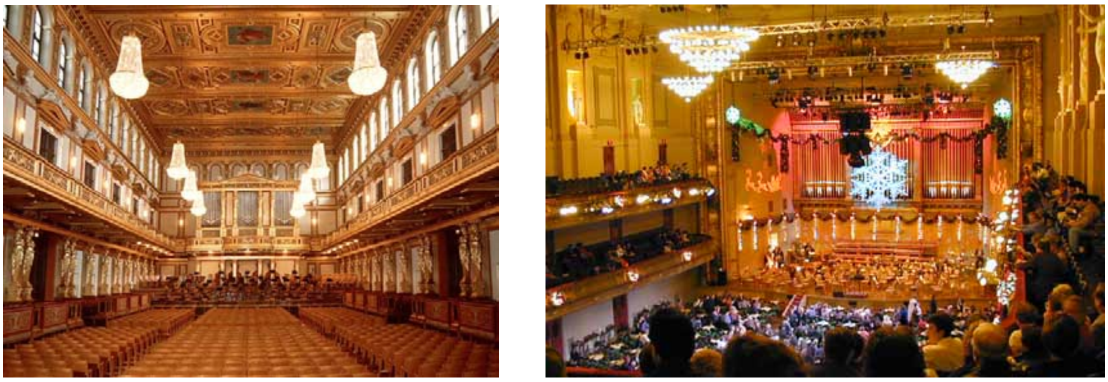

Figure 2 – The Musikverein in Vienna and the Boston Symphony Hall in Boston.

It should be noted that the design of the first two rooms previously mentioned (as well as those built before that time) was developed without any help of solid scientific background. Only with the work of Wallace Clement Sabine (responsible for the acoustical design of the Boston Symphony Hall) in the beginning of the 20th century, and his formulation of the reverberation equation have the foundations of architectural acoustics started to develop. Nevertheless, the concepts of reverberation, interference, echo disturbance and clarity of voice had already been described in Vitruvius work (ca. 25 B.C) [2].

At present, music of all the previous musical periods is performed regularly as is contemporary music. If those styles have quite definite acoustical requirements which are generally common to the compositions over the corresponding period, the same is not true to contemporary works. Not only the variety of styles written after the end of the 19^{th} century is enormous, but also composers understand much better the interaction between their music and the room in which it is performed, allowing them to write for very different acoustical environments. These facts, together with the progress of Acoustics as a science, gave origin to the development of rooms designed for different purposes, allowing their acoustical characteristics to be changed depending on the programmed performance. For this purpose, halls that could accommodate theatre and opera would have orchestra shells installed specifically designed for the performance of music, allowing an effective physical and acoustical separation between the stage tower with the stage and the audience volume. Similarly, several variable acoustics devices are used, such as absorptive panels, heavy curtains or coupled reverberation chambers to either damp a live room or rise the reverberation time of dead halls, respectively.

Although we have mainly discussed the importance of the reverberation time (as describing the acoustics of a room) on the music performance of a particular period, it should be noted that this acoustical parameter is not usually sufficient to guarantee what could be considered good acoustics, as will be seen later. It has been, however, the most used parameter in acoustical design of rooms for music and speech and is definitely one of the most important acoustical requirements.

A paradigmatic example of the importance of room acoustics on the musical performance and on the perceived acoustical characteristics of musical instruments is a statement by the pianist Claudio Arrau. When asked if there was a specific piano he preferred to any other, he answered that his choice would go to a piano of a small concert hall in La Chaux-de-Fonds (Switzerland). Although similar in model to many others it was the particular acoustics of the hall that enhanced the quality of the instrument [3].

3 SOUND SOURCES: MUSICAL INSTRUMENTS AND SPEECH

Musical instruments and the singing voice – also considered as a musical instrument – represent most of the natural sound sources used in musical performances (other sound producing devices sometimes used in contemporary music are not usually associated with musical sounds, but in that context can nevertheless be considered as musical instruments). From the variety of acoustical characteristics that these instruments reveal, some are extremely important when considering the interaction between the instrument and the room. The sound power radiated by these sound sources varies with frequency, time and the direction of radiation. The first two properties are related to the timbre of the instrument sound and therefore can be dramatically altered by the frequency or time-dependent character of the room acoustics. The source directivity is usually less problematic, since sound reflections in the room surfaces can compensate for a very directional or too omnidirectional radiation. However, knowledge of the sound radiation patterns of musical instruments and of the human voice is essential when wanting to simulate the sound of a specific instrument in a virtual room, for example.

The human ear is sensible to frequencies ranging from approximately 20 Hz to 20 kHz and can perceive sound pressures as small as 20 μPa. Therefore, it is very important to know the frequency distribution of the sound levels radiated from a musical instrument. Figure 3 shows examples of the approximate frequency ranges of the sounds radiated by some instruments. The ranges represented do not imply that these are the only frequencies radiated by those instruments which are perceived by the human being. In fact, each note is composed of a fundamental and higher frequency partials (either harmonic or not) which extend over the ranges represented and even well above 20 kHz – see [4].

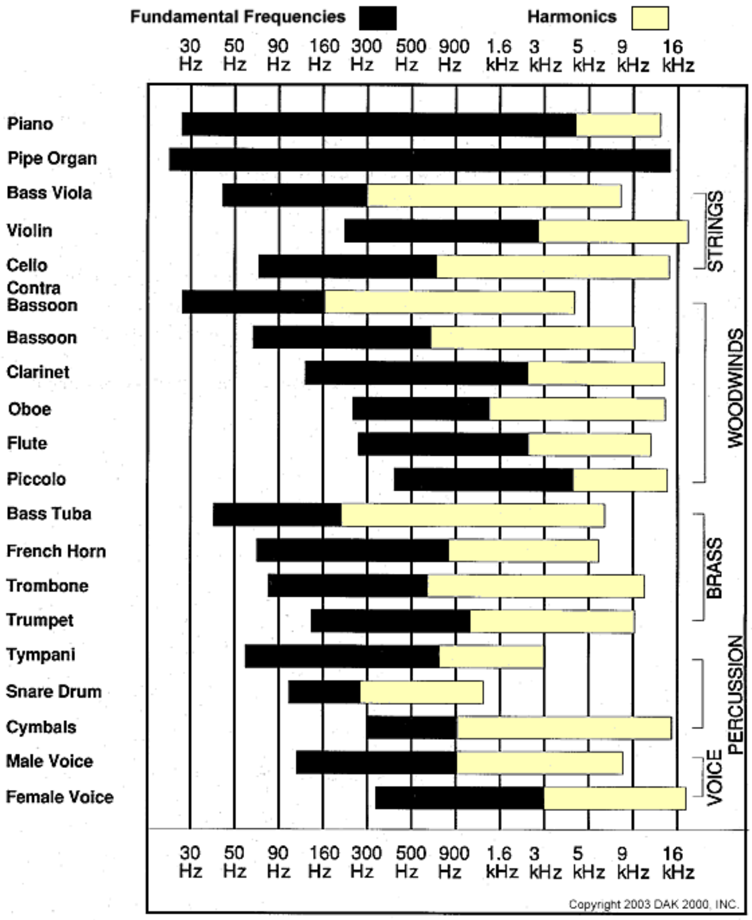

Figure 3 –Frequency ranges of several musical instruments [30]

The analysis of Figure 3 shows how important is the control of a concert hall acoustics over a large frequency spectrum. The frequency variation of the room response will affect the radiated sound by enhancing or attenuating determined frequency ranges. One well-known phenomenon that illustrates the importance of the room design on the spectrum of the sound reaching the listeners is the "seat dip effect". The "seat dip effect" results in selective low-frequency attenuation, which can reach 20 dB, of sound propagating across the audience at grazing incidence. Depending on the seating arrangement it can affect frequencies ranging from 75 to 300 Hz, covering the fundamentals of voices and some musical instruments, which justified, for some authors, the phenomenon of the "missing cellos". Apart from increasing the slope of the audience plane [6], it has been found that the seat dip effect can be rendered inaudible by introducing a cavity under the seats [2] or by changing the impedance of the seats upholstery [10].

For normal speech, the fundamental frequency lies between 50 Hz and 350 Hz. However, the higher frequency partials (overtones) are much more characteristic than the fundamental tone. These components are particularly strong in certain frequency ranges called "formants" (also present in other musical instruments), extending up to 3500 Hz. Some consonants have broadband spectral components up to 10 kHz or higher. It is therefore essential that a room designed for speech or opera can transmit these frequencies with great fidelity [7]. Interestingly, in contrast with other musical instruments, the singer (mostly sopranos) can change the central frequencies of these formants in order to concentrate the sound power in a preferred range (*formant tuning*). An example of a particular characteristic of the male singing voice is the so-called *tenor formant* (or *extra formant*), which the singer uses to make his voice come out of the sea of sound radiated by an orchestra playing *tutti* [28].

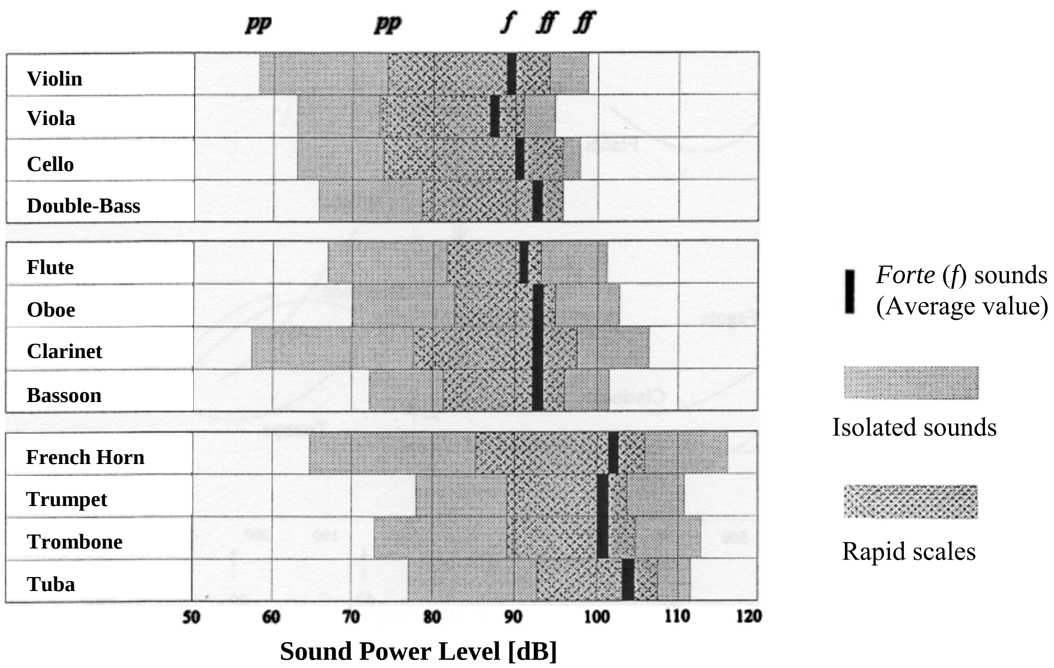

Figure 4 – Dynamic range of the sound power of orchestral musical instruments (from [5])

This last example shows how important is the balance between the sound of the orchestra and the singer in opera performances. For this balance to be achieved, the acoustical design of the room has to take in consideration the loudness of the signal that reaches the listeners. The dynamic range of the sound power level of musical instruments is very large (in some cases testing the limits of our perception) as can be seen in Figure 4. Therefore, reflections from the room boundaries should be able to compensate for the attenuation of sound with distance (see next section) allowing the more distant listeners to perceive the faintest sounds, without over-powering the strongest levels.

The sound level reaching the listener is not only dependent on the sound power of the source but also on the radiation characteristics from which directionality is of utmost importance.

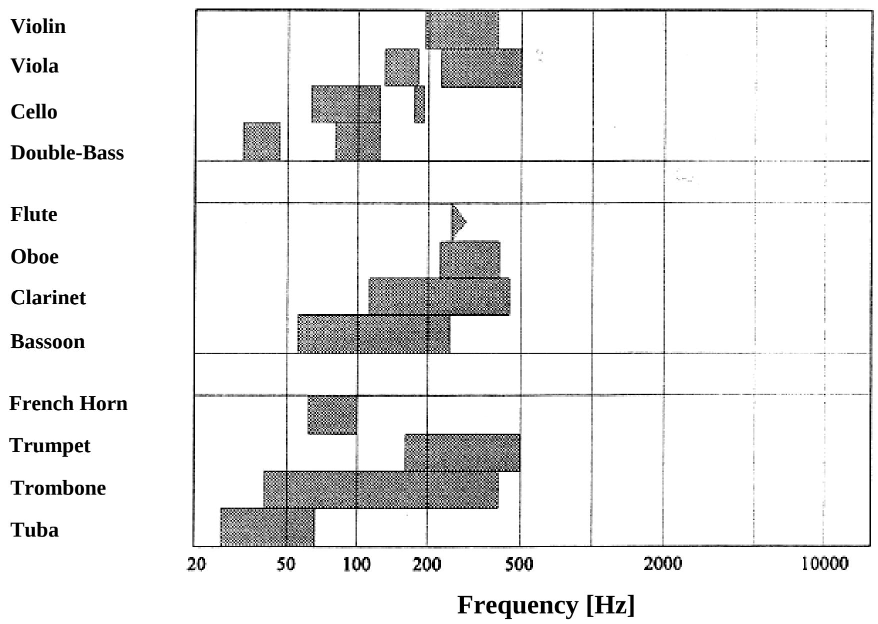

Figure 5 – Frequency ranges for which instrument radiation is omnidirectional (adapted from [5])

The radiation directionality of musical instruments is frequency-dependent, mostly because of its dimensions, usually large when compared to the wavelengths involved, and because of the complex vibration patterns that develop in their radiating surfaces, as in string instruments. In wind instruments such as the trumpet for example, the radiation intensity at an angle θ away from the axis of the horn with a mouth radius a, is proportional to

$$\left[\frac{2J_1\left(ka\sin\theta\right)}{ka\sin\theta}\right]^2 \tag{1}$$

where $k=\omega/c$ and J_1 is a Bessel function of order 1 [8]. At low frequencies

($ka << 1$), the radiation is nearly omnidirectional; however, for higher frequencies the energy is largely concentrated in a primary lobe centered on the horn axis. This frequency dependence of the radiation pattern of these instruments indicates that listeners located in different positions will perceive their sound with different timbre characteristics. Figure 5 shows the frequency ranges for which the radiation of some instruments is independent of direction. Other instruments, as the horn, the tuba or most woodwinds, radiate sound in directions which are not pointed towards the audience, and it is the shape of the room boundaries which will allow sound to be redirected to the listeners. In rooms which enable a diffuse sound field usually associated with a long reverberation, the listeners are less affected by the directionality of the musical sources.

Figure 6 shows an example of the directionality of the sound radiated by a trumpet for different frequency octave bands, on the horizontal plane. As can be seen, this instrument is almost omnidirectional for frequencies up to 500 Hz (compare this data with the results presented in Figure 5). For higher frequency bands, however, sound radiation is predominant to the front of the instrument and along its axis. If one considers a listener located at approximately 60º away from the axis of the instrument, it is clear that the timbre of the instrument will be perceived with less brightness due to the low level of the high-frequency components in this direction. This is an aspect quite frequently explored by sound engineers when trying to capture a particular sound character of an instrument.

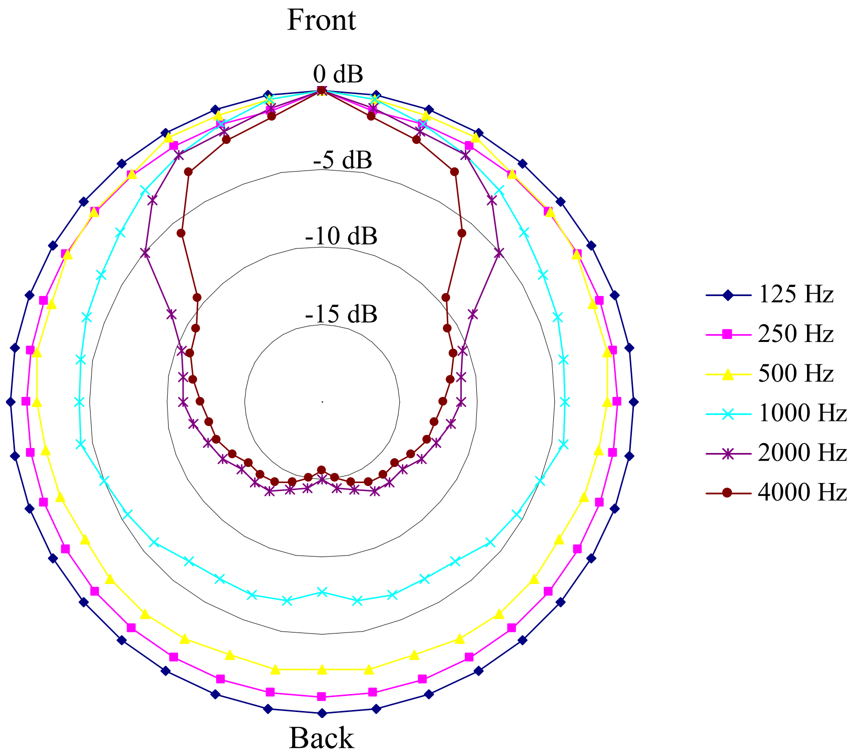

Figure 6 – Directional distribution of a trumpet sound in a horizontal plane for different frequency bands (adapted from [31])

4 SOUND PROPAGATION IN FREE FIELD

Although architectural acoustics mainly focuses on the way the room boundaries interact with the propagation of sound, it is the first wavefront reaching the listener – the direct sound – that defines the most important characteristics of the perceived sound. Interestingly, different musical styles benefit from the presence or, sometimes absence of this component. For romantic operas, for example, the orchestra is hidden in a deep pit with the sound reaching the listeners being mainly due to reflections, which provids a less defined sound and a more adequate balance with the singers.

Consider the time-averaged sound pressure level perceived by a listener at a distance *r* from a sound source. If the distance *r* is large enough to place the listener in the far-field and assuming free-field propagation, a generalized expression can be written which relates the sound pressure level, L_p, to the sound power level, L_w of the source [8],

$$L_p = L_w - 10\log(4\pi) - 20\log(r) + 10\log Q - A_E \tag{2}$$

where Q is a directivity coefficient which takes into account the directional properties of the source (or its location, when close to one or more surfaces) and A_E is an excess attenuation factor which accounts other attenuation effects such as air absorption or refraction phenomena, for example. The second and third term of the right member in Equation 2, describe the attenuation due to geometrical spreading of the energy carried by the sound wave, which represents a sound pressure level decrease of 6 dB per each doubling of the distance to the source. If we consider a source with a sound power level of 80 dB with omnidirectional radiation (Q = 1) and neglecting other attenuation effects, the sound pressure level at 20 m from the source would be 43 dB. The other attenuation effects comprised in term A_E are also very important to consider, particularly air absorption (sound refraction inside a room can occur mainly if large temperature gradients exist, but this is generally not the case). Sound attenuation by air absorption is dependent upon temperature and relative humidity. These effects are due to different phenomena, such air viscosity, thermal diffusion and molecular relaxation. As can be seen in Figure 7, one of the principal characteristics is a stronger attenuation at high frequencies (above 2 kHz). For large rooms this effect is responsible for an unavoidable decrease in sound level at these frequencies.

Although usually used for outdoor sound propagation, this simple approach allows the calculation of the sound pressure level of the direct sound reaching a listener inside an auditorium. As is well known, the human ear has a frequency-dependent degree of loudness perception, which associated with the effect if the directionality of musical instruments and the attenuation by air absorption and geometrical divergence, allows us to recognize that the direct sound in a concert hall, might not be capable of providing the most satisfying listening experience. Therefore, the help of sound reflections in a closed room provided by adequately designed room surfaces is extremely important.

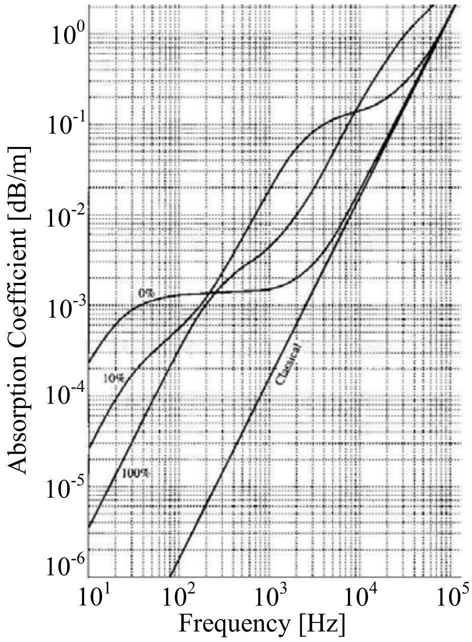

Figure 7 – Air absorption at 20º and 1 atm for different percentages of relative humidity [29].

4 SOUND REFLECTION, DIFFUSION, ABSORPTION AND TRANSMISSION

Before proceeding to the study of sound propagation in closed spaces, it is necessary to review some of the basic properties of the interaction between a sound wave and a surface. Consider a sound wave with energy E_i, incident over a surface S, schematically represented in Figure 8.

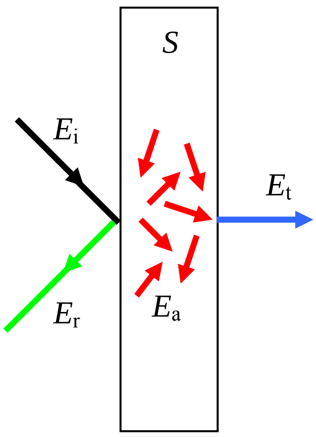

Figure 8 – Schematic representation of the interaction between a sound wave and a surface.

The wave interaction with the surface occurs in three different ways: part of the incident energy is either reflected from the surface (E_r), absorbed by the surface (E_a) or transmitted through the surface (E_t). The energy balance equation (3) allows the definition of three basic coefficients which define the surface acoustic properties.

$$E_i = E_a + E_r + E_t \qquad \text{or} \qquad 1 = \frac{E_a}{E_i} + \frac{E_r}{E_i} + \frac{E_t}{E_i} = \alpha + r + \tau \tag{3}$$

where α, r and τ are the absorption coefficient, reflection coefficient and the transmission coefficient, respectively, varying between 0 and 1.

The absorption coefficient is then defined as the ratio between the acoustic energy absorbed by a surface to that incident upon it. The value of this coefficient is dependent on the angle of incidence of the sound wave, (towards grazing incidence the absorption coefficient tends to zero). Therefore it is usually presented in the form of a random incidence absorption coefficient, measured in diffuse field conditions. Figure 9 depicts the frequency distribution of the random incidence sound absorption coefficient for different materials.

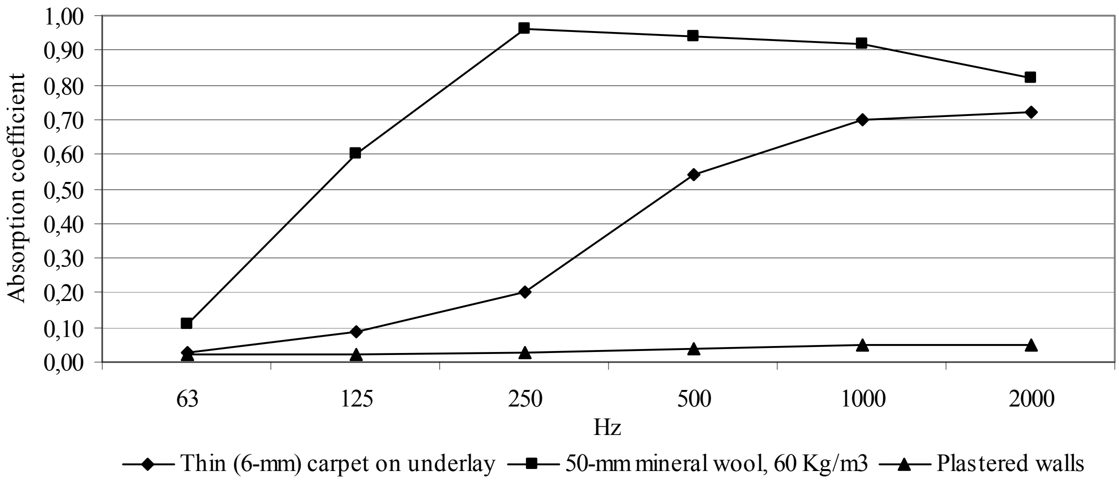

Figure 9 – Absorption coefficients of different materials (see [32]).

The transmission coefficient characterizes the resistance of the surface to the passage of sound waves. This is the important quantity to refer to when considering sound isolation between rooms. Although this is an important component of the design of rooms for music performance, it concerns the transmission between two rooms and therefore it will not be considered in this text.

The reflection of sound from a surface may not occur only as depicted in Figure 8 – specular reflection –, where the angle of incidence of the sound wave is equal to the angle of reflection. Depending on the surface geometry or its impedance distribution, the reflected wave can be scattered in different directions, dispersed not only over space but also with different time delays. Consider the interaction between a sound wave and a rough surface (with $\tau = 0$) as represented in Figure 10. In this case, some parts of the incident wavefront will travel further to more deep regions of the surface, while other segments will be reflected sooner. Although this is not the only process which allows a reflected wave to be spread over different directions, the phase differences between each reflected part of the incident wavefront will interfere constructively and destructively to form a space distributed reflected (scattered) wave. The scattering property of the surface can then be described by a scattering coefficient, δ .

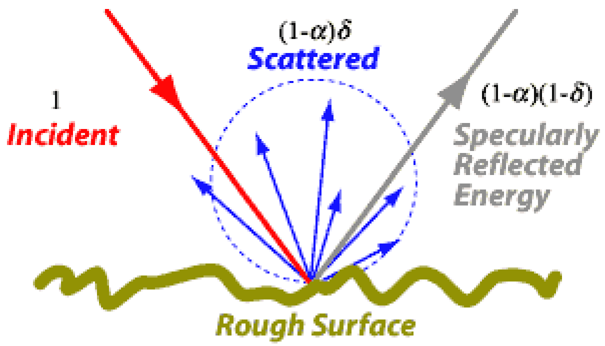

Figure 10 – Schematic representation of the interaction between a sound wave and a rough surface [33].

Considering a unitary normalization of the incident energy, the available energy to be reflected is $(1-\alpha)$ and the scattered energy will be given by $(1-\alpha)\delta$. As most of these surfaces (diffusers) are not able to scatter all of the non-absorbed energy, the specular reflected energy will be $(1-\alpha)(1-\delta)$.

Due to the common use of the expressions diffusion and scattering to describe the same surface property, it is important to distinguish these two terms which have different meanings. A diffusion coefficient measures the quality of reflections produced by a surface, by measuring the similarity between the scattered polar response and a reference uniform distribution. A diffusor that scatters sound uniformly in all directions will have a diffusion coefficient of 1. When the scattered level is concentrated in one angular location (even if different from the specular direction), the diffusion coefficient approaches zero.

On the other hand, the scattering coefficient is a measure of the amount of sound scattered away from a particular direction or distribution. If $\delta = 0$ than a pure specular reflection takes place; however, if $\delta = 1$ than all reflected power is scattered according to some kind of 'ideal' diffusivity. One weakness of the definition is that it does not say how the directional distribution of the scattered power is; even if $\delta = 1$ the directional distribution could be very uneven. This coefficient has the greatest similarity to the coefficients required as inputs to geometric acoustic models [10].

Figure 11 shows the frequency distribution of the scattering and diffusion coefficients of a sound diffusor QRD734, from RPG Diffusor Systems®.

The acoustical characteristics of the surfaces of a room, described by the previous coefficients, can be used in an appropriate combination so as to beneficiate a particular acoustical application. For example, for concert halls and auditoria where high reverberation times are expected, there is a more need for specular and diffuse reflections than for sound absorption. However, rooms for cinema or recording studios require a high degree of sound absorption combined with some diffuse reflections. When noise control is the problem, then the emphasis is most entirely given to sound absorption.

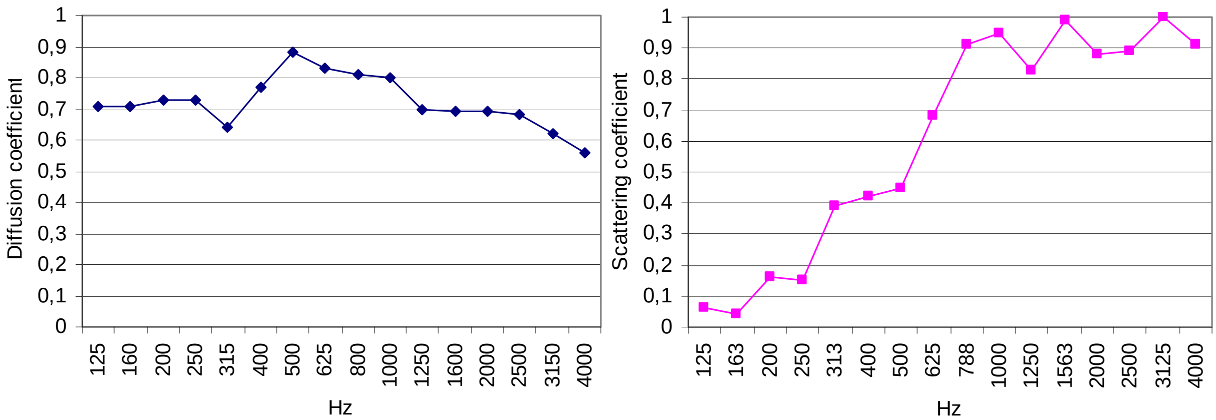

Figure 11 – Diffusion and scattering coefficients for the same diffusor (adapted from [34]).

4 SOUND PROPAGATION IN CLOSED SPACES

As seen in the section 2, the first wavefront reaching a listener without any interaction with any boundaries or obstacles is designated as direct sound. In a closed room, the sound wave propagating in all directions will sooner or later interact with a surface and will consequently be reflected.

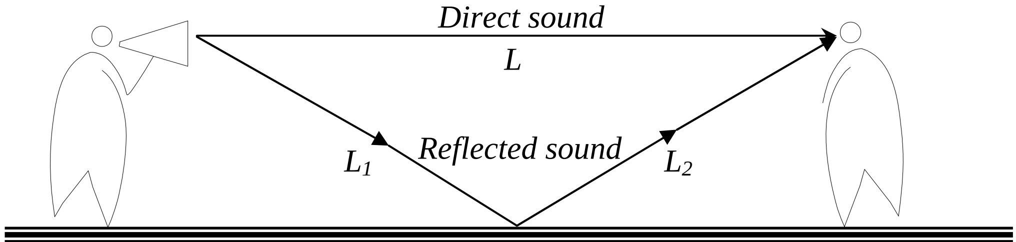

Figure 12 – Schematic representation of the direct and reflected sound propagation paths.

Consider the case represented in Figure 12. The reflection from the ground will reach the listener after the direct sound with a time delay equal to:

$$\Delta t = t_1 - t_0 = \frac{(L_1 + L_2) - L}{c} \tag{4}$$

where t_0 and t_1 are the time instants of arrival of the direct and reflected sound wave, respectively, and c is the sound speed (~344 m/s). Although simple, Equation 4 is extremely important to predict how this sound reflection will affect the perception of the listener. If the reflected sound arrives within a short time interval after the direct sound (≤ 50 ms), our ear will integrate both sounds as one and add the corresponding intensities. However, larger time delays will cause us to understand two separate sounds (echo).

In ancient Greek and Roman open-air theatres, the increase in sound level due to a single reflection from the ground had an extremely beneficial effect. However, in closed rooms such as concert halls sound interacts with the various surfaces of the surrounding architectural environment in different ways, imposing different amplitudes to reflections, different directions of propagation, etc. The result of this interaction, from the perspective of a listener sitting in an audience or for a musician on the stage, is a series of consecutive sound waves arriving after the direct sound with different amplitudes, time delays and directions. This time-history is extremely important for the evaluation of the acoustical quality of a room, because it represents the room acoustical “signature”, which we call the **room impulse response**. Note that there is one impulse response for each position of the source and receiver considered.

Figure 13 represents schematically the arrival times of direct and reflected sound waves, with their corresponding amplitudes. This representation, also called echogram, shows how the reflections (R_i) have lower amplitudes than the direct sound (D) because of the longer propagation path and the interaction with the room surfaces. The time delay between the direct sound and the arrival of the first reflection – the **Initial Time Delay Gap** (ITDG) – is extremely important to define the acoustical quality of the room, as will be seen later.

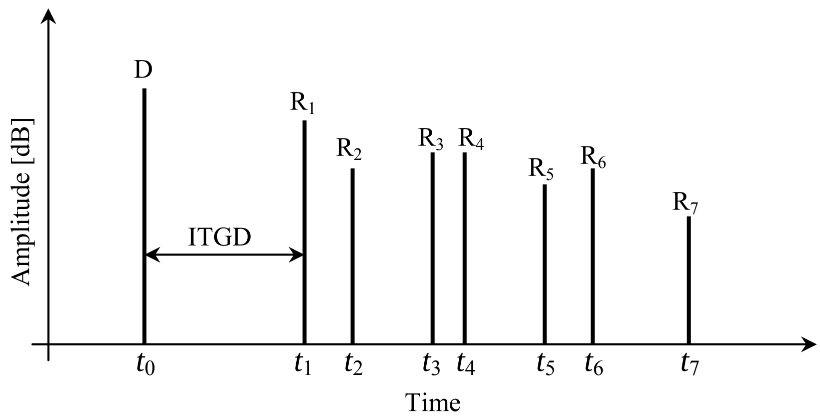

Figure 13 – Example of the arrival times and amplitude of early reflections.

The previous figure shows the early sound reflections which usually arrive, for large halls, within 80–100 ms after the direct sound, mainly due to the first reflections from the ceiling, ground or walls. However, the reflection pattern is much more complicated with such a high reflection density, particularly after the early sound, that individual reflections are not distinguishable. These late reflections compose what is called the reverberant field. Its main characteristic is that, in simple room geometry and diffuse field conditions, the corresponding sound pressure decays exponentially, with the reverberation time being defined as the time needed for these reflections to become practically inaudible (corresponding to a 60 dB sound level decrease). The equation that allows its calculation at each frequency band, introduced by Wallace Sabine at the beginning of the 20th century [11], remains still today a vital parameter in concert hall design,

$$T_{60} = 0.16\left(\frac{V}{A}\right) \tag{5}$$

where V is the room volume and A the total absorption (m^2) which results from the product of the total area of the surfaces of the room (S) and the average absorption coefficient of the total surfaces, $\bar{\alpha}$. Further formulations to the calculation of the reverberation time have been developed which refine Sabine's formula for situations of rooms with a high average absorption coefficient (Eyring formulation [12]), or with unequal distribution of absorption materials around the room (the Arau formulation, to quote one of the most recent [13]), among many others [7,14,15,16]. All these prediction techniques are based on a statistical approach of the evolution of the decay rate of the reverberant field.

When more accurate calculations are necessary a geometrical approach is usually applied. This approach describes sound propagation as sound rays, which can be a valid assumption when considering only wavelengths which are smaller than the characteristic dimensions of the room and of surfaces and obstacles. Although this is a technique mainly used for numerical computations, a revised theory as been developed which describes the decay of sound in rooms, where the reverberant field is found not to be constant, but decreasing constantly with increasing distance from the sound source [17].

In real rooms the reverberant field is only approximately diffuse, with reflections arriving from several directions with a similar pattern in most of the audience positions. However, the degree of diffuseness is dependent on the uniformity of distribution of absorption surfaces around the room. As we have seen, the received sound in a room can thus be divided into three components: direct sound, early reflections and late reverberant sound. Figure 14 represents an impulse response measured in a real room with these components represented. It should be noted that the time limits mentioned for the first reflections is dependent on the size of the room, being larger as the room size increases.

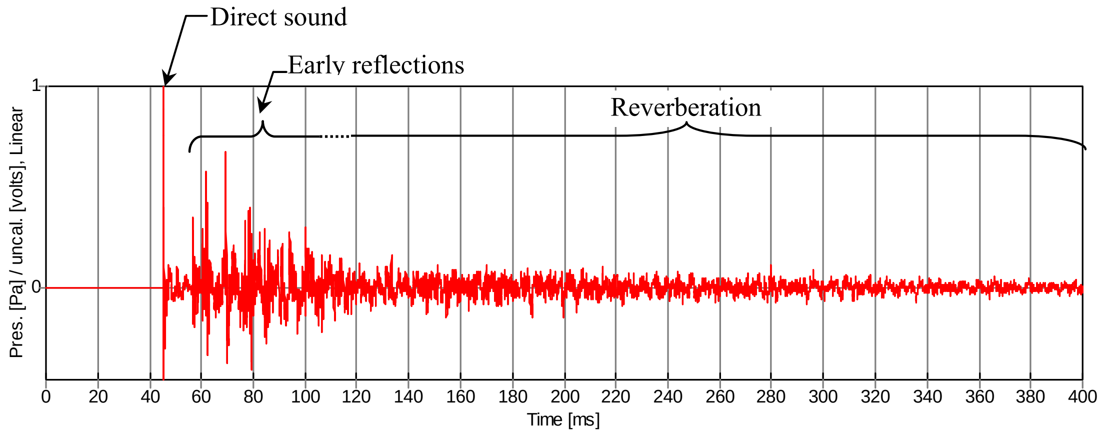

Figure 14 – Example of an impulse response in a medium size auditorium.

After the early reflections, the exponential decrease of the sound pressure with time, (seen in Figure 14), turns to a linear decay if one considers the respective logarithmic quantity, sound pressure level. This (approximately) linear decay can be seen in Figure 15. Presence of the early reflections, imposing steps on the decay, is clearly apparent between 50 and 80 ms.

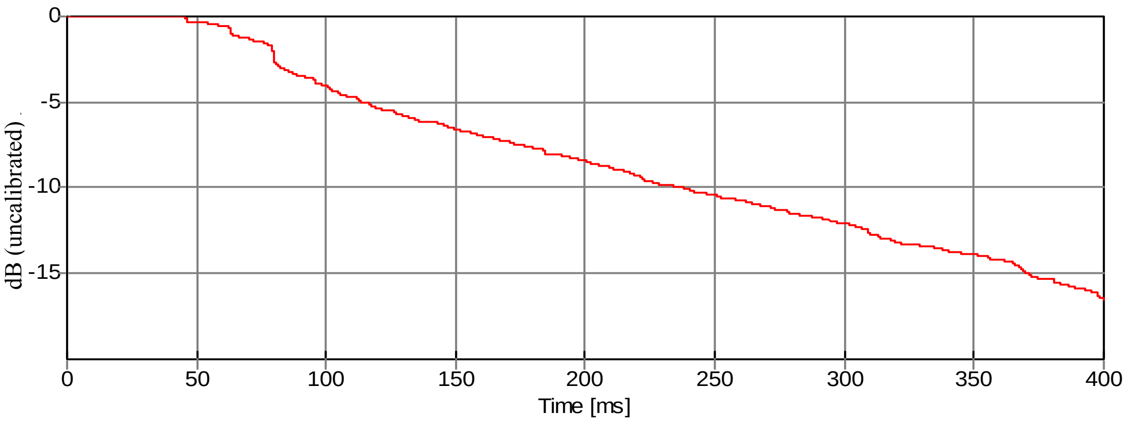

Figure 15 – Sound level decay of the impulse response in Figure 14.

The calculation of the sound pressure level at a distance *r* from a source in the free-field (Equation 2), i.e. due to the direct sound, can now be adjusted to introduce the influence of the reverberant field,

$$L_p = L_w - 10\log(4\pi) - 20\log(r) + 10\log Q - A_E + 10\log\left(\frac{4}{R}\right) \qquad (6)$$

where R is the room constant, given by $R = \dfrac{S\bar{\alpha}}{1-\bar{\alpha}}$, where S is the physical surface of the boundaries.

In the presence of this semi-reverberant sound field, a listener located closer or further away from the source will perceive a different ratio of direct to reverberant sound level. The distance at which the direct sound level equals the reverberant sound level is called the **critical distance** and is given by,

$$r_c = \sqrt{\frac{Q \cdot R}{16\pi}} \tag{7}$$

According to Barron's revised theory of sound decay in a room mentioned before, where the reverberant sound level is not constant but decreasing linearly with distance, Equation 6 can be deduced in the expanded form [18] as,

$$L_p = L_w - 10\log(4\pi) - 20\log(r) + 10\log Q - A_E + 10\log\left(\frac{25T_{60}}{V}e^{-0.04\frac{r}{T_{60}}}\right) \tag{8}$$

The results for the total reflected sound level according to this revised theory have been found to correlate better with measurements carried in several concert halls [18], than the traditional formulation.

In Figure 16 are represented the locations of the impulse response measurements whose results are shown in Figure 17 (attention should be paid to the fact that the instant of arrival of the direct sound was shifted to $t = 0$ s, so that comparisons could be more easily carried out). The sound source was located near to the first violin position. As the distance to the sound source increases, the impulse responses show an increasingly higher density of reflections and a correspondingly lower ratio of direct to reverberant sound level.

In a general form, the response $p_r(t)$ of a room to a particular acoustical excitation $p_e(t)$ (a musical piece or speech) is determined by the convolution integral,

$$p_r(t) = p_e(t) * h_{er}(t) = \int_{-\infty}^{\infty} p_e(t)h_{er}(t-\tau)d\tau = \int_{0}^{t} p_e(t)h_{er}(t-\tau)d\tau \tag{9}$$

where $h_{er}(t)$ is the room impulse response function and τ represents a time delay. This operation is a well-known property of linear time-invariant systems and is entirely applicable to sound propagation in closed spaces (within the amplitude limits for linear propagation to occur). The second formulation extends from the causality of real physical systems.

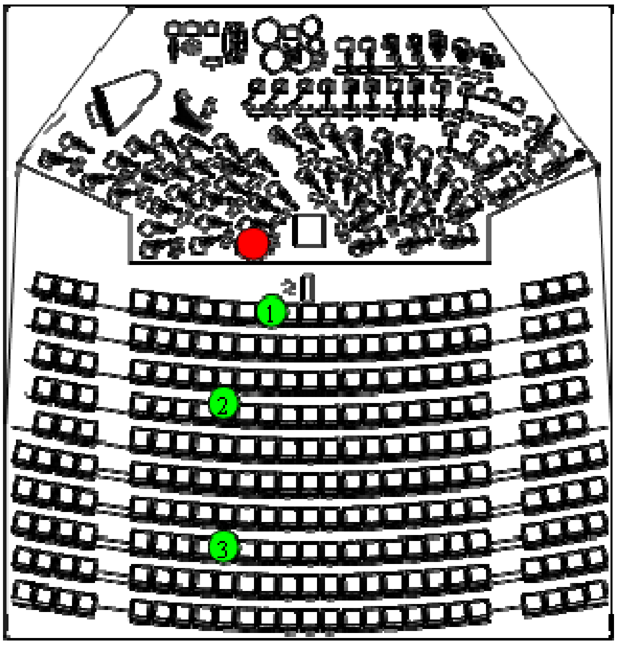

Figure 16 – Sound source (red) and measuring positions (green) in small 240 seat hall.

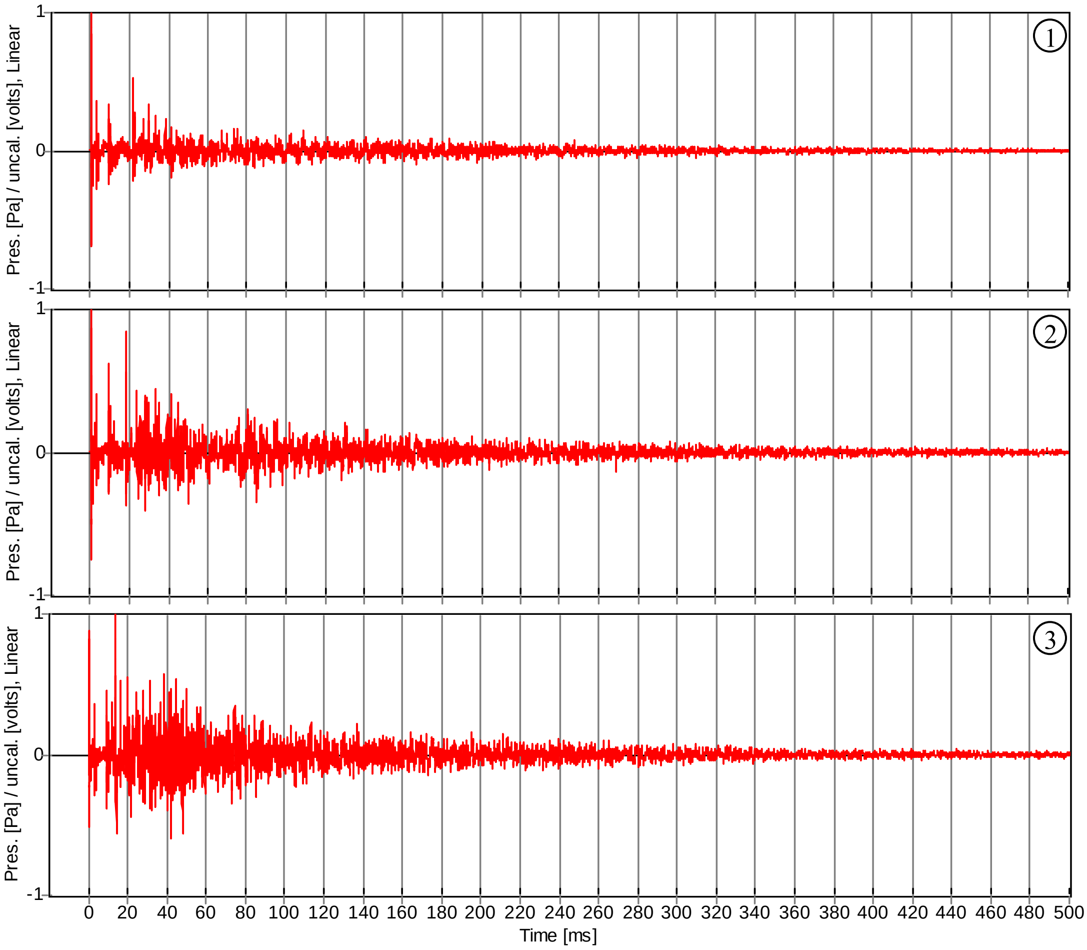

Figure 17 – Impulse responses corresponding to the 3 measurement positions in Figure 16.

6 ROOM ACOUSTICAL QUALITY EVALUATION

For decades, researchers have tried to correlate the subjective dimension of the acoustical quality of concert halls, based on the opinions of listeners of real performances, with physical attributes which can be measured and possibly predicted. If it is clear that for spoken communication the important result is that it should be intelligible, which can be fairly easily evaluated by the number of words understood in a speech, music takes a much more demanding effort. Several studies have been reported on this subject (see [19] for a thorough review), from which a few subjective (psychoacoustic) impressions resulted as extremely important for the acoustical quality evaluation of concert halls:

- **Clarity** (or definition) names the degree to which a listener can distinguish sounds in a musical performance, either for a succession of notes or for notes played simultaneously, i.e. musical detail. The influence of the room acoustics on this parameter, particularly the reverberation time, can be understood by analyzing Figure 18, where two notes played consecutively are schematically represented. If the reverberation time is long enough (Figure 18a) the second note will be masked by the remaining reverberant decay of the first note, making it difficult to distinguish between them. As the reverberation is diminished (Figure 18b) both sounds are clearly perceptible. Notice however, that the reverberation is not the only physical parameter that influences clarity. If the level of the second reflection is high enough (Figure 18c), even if the reverberation time is long, both sounds would still be discernible. It is the relation between the sound energy provided by the early reflections and the energy of the late reverberant field that affects most the impression of clarity. In what concerns the distinction between notes played simultaneously, one has to consider the frequency distribution of both reverberation time and of the early-to-late energy relation. If this distribution is not similar at the different frequency bands, some notes of a chord will sound longer or with higher level than the others, giving origin to a low "vertical" definition [1].

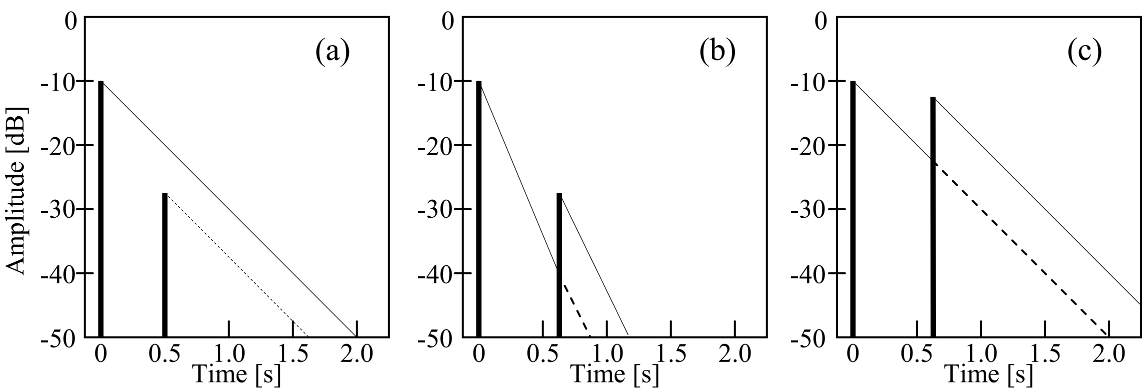

Figure 18 – Representation of two notes played consecutively for different room acoustical conditions.

- **Reverberance** is the subjective impression related to one's degree of perception of reverberation. It is quite easy to distinguish between a "dead" or "live" room, however, during a musical performance the feeling of reverberance can be distinguished in two parts: an early reverberance, associated with the initial decay of the first reflections, and late reverberance related to the decay of the actual reverberant, more diffuse field. When a musician or an ensemble plays rapidly, it is essentially the early part of the sound decay which is audible between successive notes. As was seen in Figure 15, the first reflections can alter substantially the decay rate of the sound level in the first 50 milliseconds or even up to 200 ms in larger rooms. This early reverberation is correlated to the feeling of "running liveness" [1].

- **Intimacy** refers to the one's degree of identification with the performance, whether acoustically involved or detached from it [20]. A hall will have "acoustical intimacy" if sounds seem to originate from nearby surfaces. This means that early reflections should arrive in the first milliseconds after the direct sound, i.e. the initial time delay gap defined in section 4, should be small. However, this is not the case for seats close to the orchestra, where the early reflections take much longer to arrive to a listener than for seats further away, and yet, as shown in [18], the feeling of intimacy is high. These conclusions demonstrate that not only the ITDG is important to describe this impression (with preferable values around 20 ms) but also the actual proximity to the players which is directly related to loudness. This parameter is described by some authors as part of the spatial impression of the hall [24].

- **Envelopment** is part of the **spatial impression** of the perceived sound [20]. As the word clearly suggests, it refers to the degree to which the reverberant sound seems to surround the listener coming from all directions rather than from limited directions [1]. It should be noted that this does not mean that the direction from which the sound initially originated is not important and this will be almost always perceptible if the direct sound arrives unobstructed to the listener. Nonetheless, if reflections arrive from lateral directions, not only will they contribute to the feeling of envelopment, but also to another subjective attribute related to the spatial impression (or **spaciousness**) designated by **apparent source width (ASW)**. If strong lateral reflections arrive within 20º to 90º of an imaginary vertical plane that connects the listener to the performers, sound will seem to emanate from a wider source than its actual physical limits [19]. These strong lateral reflections, arriving in the early part of the room response, have been shown to be one of the most important aspects that determine the high acoustical quality of a concert hall [20].

- **Loudness** is an obvious choice for a subjective attribute, since it can be used to determine whether even the faintest sounds arrive to the listener with enough level to be clearly perceived. This parameter is a function of the energy in a sound wave divided by the number of people who must share it [21]. The energy is also decreased by the absorption in the hall. Notice that in most halls the audience absorbs approximately 80% of the sound energy radiated by an orchestra.

The list of parameters just presented could be extended to further details that are subjectively apparent to a listener, like **warmth** (related to the relation between reverberation time at low and higher frequencies), **timbre**, **acoustical glare**, **brilliance**, among other which can be found in the literature (see [1]). Additionally, it is also important to understand the acoustical conditions that musicians prefer for their performance. For this purpose, a few other perceptual aspects have been studied by several researchers:

- **Balance** between sections of the orchestra and between the orchestra and singers has been reported as a major characteristic of good halls from the musicians' perspective. This balance refers particularly to a uniform distribution of sound level between parts of the orchestra. We can therefore evaluate if the sound from the strings of the left and right side of the orchestra is evenhanded or the brass and percussion at the rear of the orchestra overwhelm the instruments at the front, especially at *fortissimo* passages [19]. In opera houses, this parameter is extremely important when comparing sound levels from the orchestra in the pit and from the singers on stage.

- **Ensemble** refers to the facility of performers to play in unison and depends on the ability for them to hear their fellow performers. The sound reflecting surfaces near and above the performers should carry the sound the sound from the players on one part of the stage to those in other parts [1].

- **Immediacy of Response** is related to the degree of acoustical feedback that the musicians receive from the room response. The hall should give the performers the feeling that it responds immediately to a note. This characteristic depends on how the first reflections from surfaces in the hall arrive back to the musician's ears. If reflections occur too long after the note is sounded, the players will hear them as an echo. Conversely, if the musicians hear reflections only from the nearby surrounding stage walls, they will have no sense of the hall's acoustics [1].

Although we have mainly referred to the acoustical aspects that influence the opinion of the hall quality, it is well known that other non-acoustical factors such as visual references, the music characteristics and good performance are also very important.

Several <u>physical (objective) parameters</u> have been proposed that correlate with the previously listed subjective impressions. If the degree of correlation is high one can establish the corresponding appropriate values that contribute to the best possible acoustical conditions. These parameters are calculated on the basis of the one or more room impulse responses which can be predicted using appropriate numerical calculations or scale models during the design stage, or measured *in situ* after the construction of the hall in order to evaluate the results of the predictions. Some of these parameters will be described in the next paragraphs.

- **Reverberation Time** measures can be distinguished in two types which are related to the different kinds of reverberance (early and late) mentioned previously:

  - The classical **Reverberation Time, T_{60}**, is obtained by calculating the time interval needed for the sound level to decrease 60 dB. However, during measurements it is usually difficult to obtain a signal-to-noise ratio high enough to enable a 60 dB decay. This means that extrapolations are necessary and usually the T_{30} (corresponding to a 30 dB decay) is measured, between -5 dB and -35 dB, and then multiplied by 2 in order to make it equivalent to the T_{60}. Other decay intervals can be used in order to calculate the T_{20} or T_{15} but attention should be paid that if the decay is not linear along the whole time history, these values will not be good estimates of the T_{60} measure.
  - The **Early Decay Time (EDT)**, which correlates to the early decay of reflections (early reverberance), is calculated as the time it takes for the sound to decay from 0 to -10 dB and then multiplied by a factor of 6.

In order to obtain the decay curve that allows the previous calculations, it is usual to use the Schroeder backwards integration scheme applied to the impulse response and then apply a least-square fit of a straight line to the portion of the decay curve needed to determine the previous parameters. Figure 19 shows two impulse responses and the corresponding reverberation time measures.

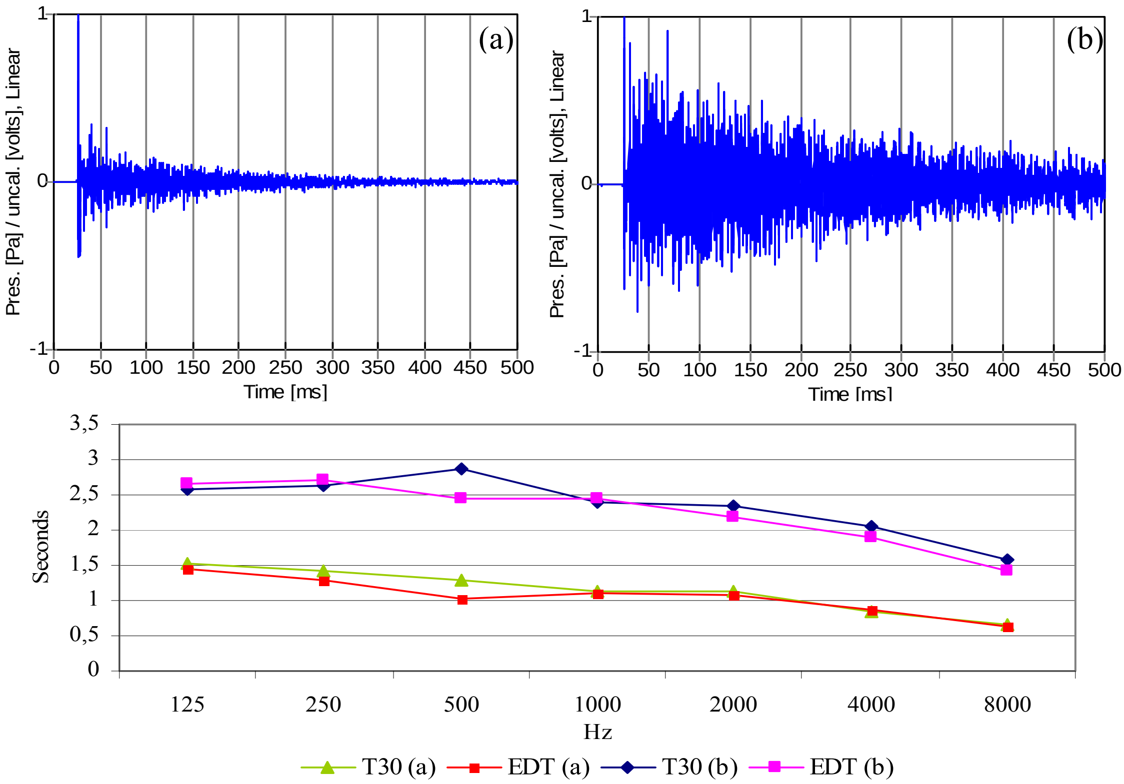

Figure 19 – Impulse responses measured in very different rooms (but with similar source to receiver distances), and the corresponding T_{30} and EDT.

The optimum values for the previous parameters, depend naturally on the use of the room in consideration. Figure 20 show some examples of rooms and the range of optimum values for the reverberation time. It should be noted that the minimum and maximum values of this range usually depend on the volume of the room. Arau [22] describes some of these limits based on the graphs presented by Cremer [23], with the reverberation time being represented by a single number value T_{mid} equal to the average of T_{30} values over the 500 Hz, 100 Hz and 2000 Hz octave bands:

- Concert Halls: $0{,}4245\ V^{0{,}1331} \leq T_{mid} \leq 0{,}6\ V^{0{,}1325}$
- Opera Houses: $0{,}396\ V^{0{,}1273} \leq T_{mid} \leq 0{,}509\ V^{0{,}1335}$
- Theatres and conference rooms: $0{,}264\ V^{0{,}1394} \leq T_{mid} \leq 0{,}368\ V^{0{,}1505}$

The Early Decay Time optimum values are also reported in this work [22] defined in relation to the T_{mid}:

- Concert Halls: $0{,}9\ T_{mid} \leq EDT \leq T_{mid}$
- Opera Houses: $0{,}75\ T_{mid} \leq EDT \leq 0{,}9\ T_{mid}$
- Theatres and conference rooms: $0{,}6\ T_{mid} \leq EDT \leq 0{,}75\ T_{mid}$

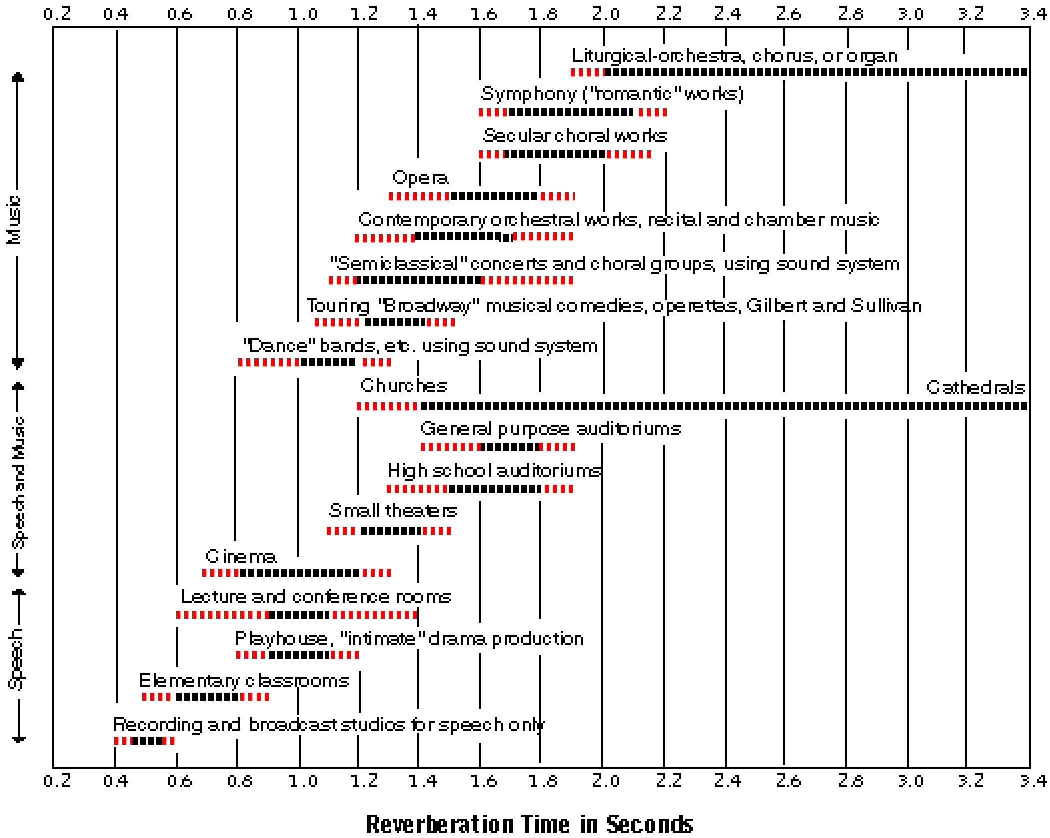

Figure 20 – Optimum values for the reverberation time for different types of rooms [35].

- **Early-to-late energy ratios** are based on a time subdivision of the impulse response between early energy provided by the first reflections and the energy due to the late reverberation or to the whole impulse response. These parameters are correlated with the impression of clarity or definition described earlier. The **Clarity Factor** (***C_{80}***) is expressed in decibels and defined as the ratio of the early energy, between 0 and 80 ms, to the late reverberant energy (after 80 ms):

$$C_{80} = 10 \log \frac{\int_0^{0.08} p^2(t)\, dt}{\int_{0.08}^{\infty} p^2(t)\, dt} \tag{10}$$

where $p(t)$ is the sound pressure impulse response of the room. This value can be determined in frequency bands or by a single number quantity $C_{80}(3)$ which the average of the C_{80} values over the 500 Hz, 100 Hz and 2000 Hz octave bands [25]. The 80 ms second limit is the appropriate value when evaluating halls for music performance while 50 ms is usually chosen for speech.

The appropriate values for the C_{80} depend on the style of music and are usually in the range between -4 dB to 2 dB. Music from the Romantic period will sound better in the low limit of this range while Classical and Baroque music prefer the higher limit. Higher values than 2 dB usually denote a “dry” room, since this parameter is inversely related with the reverberation time.

Another parameter very correlated with clarity is the ***D_{50}***, designated as **Definition** (**Deutlichkeit** or **Distinctness**), and is the ratio of the sound energy in the first 50 ms after the arrival of the direct sound at the listener’s position to the total sound energy, and is expressed in percentage:

$$D_{50} = \frac{\int_0^{0.05} p^2(t)\, dt}{\int_0^{\infty} p^2(t)\, dt} \tag{11}$$

This parameter is mostly used to evaluate quality of speech perception in rooms and values higher than 50% are preferable for this purpose.

- The **Centre Time** (***t_s***) is the so-called centre of gravity time (see Equation 12) of the decaying sound field, and is also correlated with clarity, particularly with C_{50} and C_{80} which means that it doesn’t bring much more information [26]. It is given by the following expression, with preferred optimum values of 0 ms < t_s < 50 ms for speech and approximately 50 ms < t_s < 250 ms for music:

$$t_s = \frac{\int_0^{\infty} t \cdot p^2(t)\, dt}{\int_0^{\infty} p^2(t)\, dt} \tag{12}$$

- The subject of spatial impression has received great attention from researchers, particularly when defining the acoustical measures to which it can have the best correlation. Several early lateral energy measures have been developed for this end from which the **Lateral Fraction** (**LF**) is one of the most used. While the previous parameters can be measured by the use of a single omnidirectional microphone, the LF needs additionally a figure-of-eight microphone (with the null axis facing the source) to differentiate lateral reflections from reflections that arrive to the listener from all directions. It is given by:

$$LF = \frac{\int_{0.005}^{0.08} p_L^2(t)\ dt}{\int_{0}^{0.08} p^2(t)\ dt} \tag{13}$$

where $p_L(t)$ is the instantaneous pressure in the impulse response measured with a figure-of-eight microphone. The 5 ms limit of the numerator integral is introduced to guarantee that the direct sound energy is not accounted for. A single value parameter, LF_{E4} is calculated by the LF values in the 125 to 1000 Hz octave bands.

For concert halls the optimum values should be higher than 0.2 or 0.25. Although a maximum limit is not usually defined, it is understandable that it should not be high enough for a listener to lose the perception source localization.

Other parameters, such as the **Lateral Efficiency** or a lateral fraction parameter (**LFC**) corrected to take into account the directivity effect of the figure-of-eight microphone [26] have also been proposed.

- The lateral parameters just described are monaural, i.e. the impulse response is measured in a single point. However, the fact that the process of hearing is binaural lead to the creation of a family of parameters which are based on the **Interaural Cross Correlation Function**. The **IACF** is a binaural measure of the difference in the sounds arriving at a listener's ears, produced by a source on stage (see [2] for a detailed analysis) and can be calculated by,

$$IACF_{t_1,t_2}(\tau) = \frac{\int_{t_1}^{t_2} p_l(t)\cdot p_r(t+\tau)\ dt}{\left(\int_{t_1}^{t_2} p_l^2(t)dt \cdot p_r^2(t)\ dt\right)^{1/2}} \tag{14}$$

where $p_l(t)$ is the impulse response measured at the entrance of the left ear canal and $p_r(t)$ designates the impulse response measured at the entrance of the right ear canal [26]. The maximum possible value of Equation 13 is unity. As can be seen, the IACF is a function of the parameter τ , which is usually varied between -1 ms and 1 ms since the time that a sound wave takes to propagate from one side of the human head to the other is approximately 1 ms [1]. To determine a single value quantity that measures the maximum similarity between waves arriving at the two ears, the **Interaural Cross Correlation Coefficient** (**IACC**) is used:

$$IACC_{t_1,t_2} = \max\left|IACC_{t_1,t_2}(\tau)\right| \text{ for } -1 \text{ ms} < \tau < 1 \text{ ms} \tag{15}$$

The most general form of IACC is defined with $t_1 = 0$ and $t_1 = \infty$ although early and late measures are also possible, correlating with ASW (Apparent Source Width) and listener envelopment [26]. The lowest values of IACC are always preferred, with values of 0.3 having been obtained for rooms of excellent acoustical quality.

- Loudness is one of the most important attributes that define the acoustical quality of a hall. It can be evaluated by a measure of the 'amplification' that the hall gives to the sound source, for which a reference level has to be established. This objective measure has been named **Sound Strength** (***G***) and compares the sound pressure level at a point in a hall, with an omnidirectional source on stage with the sound pressure level that would be measured at a distance of 10 m from the same sound source in free-field:

$$G = 10\log\left(\frac{\int_0^\infty p^2(t)\,dt}{\int_0^\infty p_{10}^2(t)\,dt}\right) \text{ [dB]} \tag{16}$$

where $p_{10}(t)$ is the impulse response measured at a distance of 10 m in free-field conditions. This value can also be obtained by sound pressure level measurements with a sound level meter, if the sound power of the source, L_w, is known. In this case,

$$G = L_p - L_w + 31 \tag{17}$$

with the 31 value being a correction factor related to the 10 m distance imposition. A single number quantity, G_{mid}, can be used by calculating the average of the G values in the 500 Hz and 1000 Hz octave bands.

Optimum values for this parameter can be found in Table 1, for different types of sound sources [27].

Table 1 – Optimum values for the Sound Strength parameter

Sound Source	***G***
Symphonic orchestra	≥ -4 dB
Chamber orchestra	≥ 1 dB
Trained speaker	≥ 6 dB
Normal speaker or weak instruments	≥ 11 dB

- Another important feature in concert hall design is the appropriate choice of the balance between low frequency reverberation and medium to high frequency reverberation. This aspect is correlated with the subjective impression of warmth and fullness of bass sound. It can be evaluated by the **Bass Ratio** (**BR**) which is expressed by:

$$BR = \frac{RT_{125} + RT_{250}}{RT_{500} + RT_{1000}} \tag{18}$$

  where RT is the reverberation time at the octave band centre frequencies shown in the subscripts. The optimum values vary between 1.1 and 1.25 for rooms with high reverberation times, and between 1 to 1.45 for rooms with reverberation times below 1.8 s [22].

- In Opera houses, the singers and the musicians are placed in different levels with relation to the audience. This fact can generate high level differences between the sound arriving to the listener from the stage and from the pit. It is therefore essential to estimate the **Balance** (***B***) between the singer and the orchestra. This parameter can be expressed by the following formulation developed by Prodi and Pompoli [27]:

$$B = L_{stage} - L_{pit} \tag{19}$$

  where L_{stage} refers to a sound level measurement in the audience (stalls and boxes) due to a directional source placed on stage and L_{pit} to the same measurement but with an omnidirectional source in the pit [27]. Omnidirectional sources in both positions can also be used, solving the problem of choosing the directional source angular position. To have the orchestra balanced with the singers the values of B should lay between -2 and 2.3.

The previous parameter referred essentially to evaluation of the acoustical quality of the hall from a listener's perspective. Other parameters have been formulated which try to measure the acoustical conditions of musicians on stage or pit, from which we refer the Support Factor.

- The **Support Factor** (**ST1**) quantifies the energy of reflections which is useful for the performers located on orchestra platforms, which correlates with the easiness of playing ensemble. It is given by the following expression:

$$ST1 = 10 \cdot \log\left(\frac{\int_{0.02}^{0.1} p^2(t)\, dt}{\int_{0}^{0.01} p^2(t)\, dt}\right) \tag{20}$$

  The impulse response $p(t)$ is measured at 1 m distance from the centre of the sound source, which is placed either on stage or in the pit. Although other limits are found in the literature, accepted desirable values vary between -12 to -14.4 dB [1], however optimum values up to ± 1 dB have also been reported [19].

7 CONCLUSIONS

The last element in the chain for the transmission of musical sound to a listener is the room where the music is performed. If the design of a hall is not considered with the same degree of detail that a *luthier* or a musical instrument researcher gives to his work, it can totally degrade the musical experience. This is why musical acoustics also embraces the study of the acoustics of concert halls and opera houses.

Although a recent discipline (~100 years old), the acoustics of rooms has always been regarded, by musicians and composers, as an important feature which influenced the style of musical composition and performance along the history of music. However, it was only with Sabine's work at the end of the 19th century that the design of concert halls starts to have a solid scientific background, with the development of the reverberation time formula.

The reverberation time depends on the interaction of the sound waves radiated by the musical instruments with the surfaces that compose the room. Both the radiation properties of the instruments as well as of the room materials need to be accurately characterized for a complete understanding of the perceived sound. Several other acoustical parameters were and are still being developed, which help the accurate design of rooms to optimize both the musical performance and the listener experience.

These aspects were covered in the present text in order to provide an overall perspective of the fundamentals of room acoustics. Several other concepts have been left unsaid, and therefore the reading of the references is strongly recommended for a more deep understanding of room acoustics for music performance.

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