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Variability of Acoustic Parameters over Audience Areas in Auditoria

Master's Thesis in the Master's programme in Sound and Vibration

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Abstract

Besides Reverberation Time, recently other quantities have been used to characterize sound fields inside Concert Halls, Auditoriums, Theatres or Opera Halls. Also known as Acoustical Attributes, these parameters can determine the difference between "good" or "excellent" acoustics. As an example, such parameters can also explain why two concert halls with the same Reverberation Time can be ranked differently.

Unlike the "almost" uniform Reverberation Time value, the Acoustical Parameters show a different behaviour thorough the audience area due to its fine structure. This can be observed even in modern Acoustic Simulation software where a parameter value may change from seat to seat, row to row and so on. Nevertheless, some researches tend to represent by a single number the parameter's values obtained from a whole audience area.

Nowadays, some measuring procedures imply just a few measuring positions. This fact could yield to not so accurate results when the description of acoustic parameter behaviour is attempted.

The aim of this work is to describe the variability of some of the most important acoustic parameters in audience areas by measuring in a large amount of positions inside some real auditoria.

The following text is divided in three parts: The Theoretical Background, the description of Measurements in Auditoria and the presentation of Results & Conclusions. The first part gives an introduction to the theory behind the Acoustical Parameters from its Room Acoustics basis, the structure of such quantities, the factors that may influence the values and finally to the most used and recommended values.

The second part of this work describes the measures done inside different auditoria in Spain. Finally, the third part presents the obtained results and its interpretation according to the theory introduced in the first part of this text.

Key words: Spatial Sound Perception, Subjective Impression of Sound Field Components, Room Acoustics Metrics and MLS Measuring Techniques.

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Contents

1.The Theoretical Background

1.1. Room Acoustics

Room Acoustics is the study of sound behaviour inside enclosures. In the free field, an ideal omnidirectional source produces sound waves that propagate in a spherical way. The sound level generated by the sound source reduces its intensity 6dB every time the distance is doubled in what is also known as the inverse square law, see figure 1.

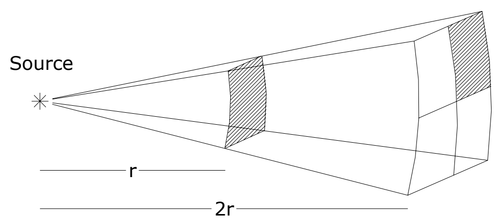


Figure 1: *Direct Sound Propagation: The Inverse Square Law.*

Indoors, the sound travelling path is influenced by the boundary conditions of the room and the way it dies out is a blend of the inverse square law for the direct sound and the energy decay from all the contributions generated by the sound waves bouncing back and forth on room surfaces.

Room acoustics is classified in two main branches: Technical Room Acoustics and Psychological Room Acoustics. While psychological acoustics documents how humans hear, perceive and judge sound fields, technical room acoustics deals with different approaches of handling sound waves in a confined space and is divided into geometrical, wave-theory and statistical acoustics.

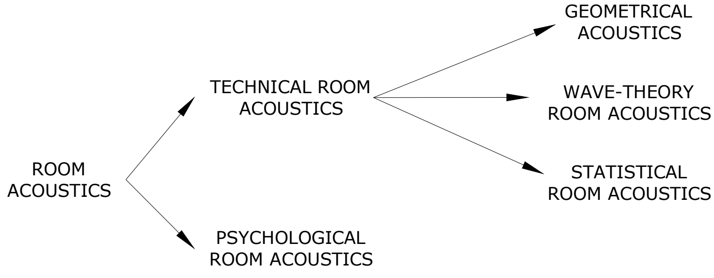


Figure 2: *The Room Acoustics Classification.*

Psychological acoustics it's related to technical acoustics through the mapping of the perceived and judged room acoustics on physically measurable room properties and associated metrics that are called subjective parameters.

1.1.1. Geometrical Acoustics (The High Frequency Approach)

Geometrical acoustics is a simplified way to describe sound propagation inside enclosures following the rules applied to geometrical optics and is limited to small wavelengths compared with the dimensions of the room. This condition is fulfilled above 1kHz ($\lambda \approx 34$ cm). To use the so-called "sound ray" approach some assumptions must be made: sources and receivers are omnidirectional, surfaces are non-scattering, the propagation takes place in straight lines etc.

For uses in auditoria, the minimum surface size to be considered as an acceptable reflector is defined by:

$$l_{\text{limit}} \approx \frac{d^2}{2\lambda} \tag{1.1}$$

Where:

d = The smallest surface dimension

l_{limit} = The distance to the observation point (listener)

λ = The wavelength of the reflected sound

When a sound ray propagates inside a room, it can either be reflected or bent by surfaces, objects or changes in propagation conditions such as wind or temperature gradients. Reflection is ruled by the mirroring principle: the incident and reflecting rays are in the same plane as the normal to the surface and the angles of incidence and reflection are equal.

If the room is bounded by various plane surfaces it is necessary to use a series of image sources obtained from mirroring the original source on the walls where the sound rays may impact. Every time a ray is reflected in a surface, it loses a part of its energy due to wall absorption. When finally reaches the listener, the ray has a different power spectrum from that originated at the source. To find the total ray tracing it is therefore necessary to follow the path to and from the image sources from source to listener, see figure below.

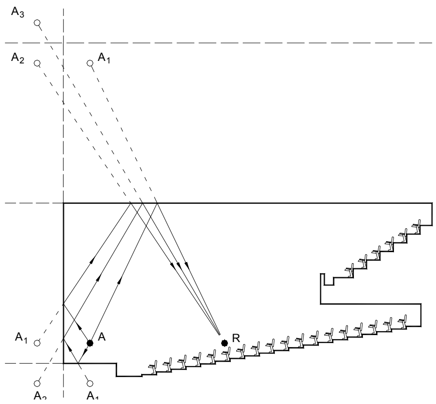


Figure 3: *Longitudinal Section of an Auditorium with Image Sources: A=Sound Source; A1=First-Order Image Sources; A2=Second-Order Image Sources, etc.*

The use of curved surfaces as reflectors requires caution: a convex surface will disperse sound, figure 4(a), whereas a concave surface will focus sound, figure 4(b), concentrating sound energy in certain spots. However, if the radius of curvature is equal or smaller than the wavelength and the source or even the receiver are not within the extended circle of the concave surface, the surface will scatter sound rather than reflect it specularly as is shown in figure 4(c).

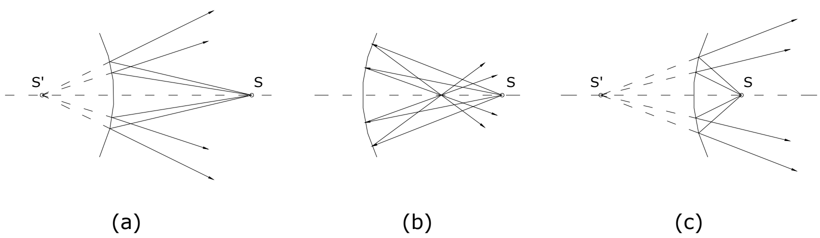


Figure 4: *Reflection of a group of rays from concave and convex mirrors.*

Perfect reflection is obtained for high frequencies. As the frequency is lowered, less and less energy is reflected along the geometrical reflection path and the distance between the reflector and the source and that between the reflector and the receiver become important. Barron uses a couple of limiting frequencies f_1 and f_2 (equations 1.2 and 1.3) to calculate the sound level change ΔL in the reflection due to the finite size of a reflector (compared to reflection from an infinite reflector along the same reflection path):

$$f_1 = \frac{c}{(1/s + 1/r) \cdot B^2 \cos^2 \theta} \tag{1.2}$$

$$f_2 = \frac{c}{(1/s + 1/r) \cdot A^2} \tag{1.3}$$

These differences are related by the speed of sound c, the distance source-reflector s, the distance listener-reflector r and the finite dimensions of the reflector (figure 5):

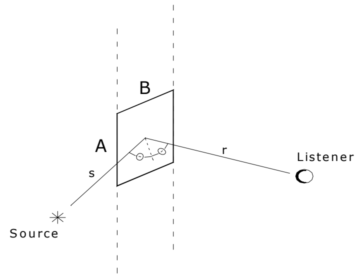


Figure 5: *Geometry of Reflection.*

Above f_1, the resulting ΔL can be ignored while below f_1 the amplitude decreases by 3dB per octave. Below f_2 diffraction effects attenuate by 6dB per octave, see figure 6.

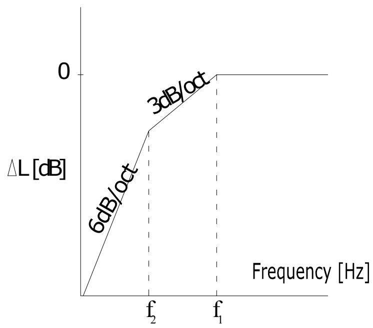


Figure 6: *Level change from reflection in a finite surface.*

Rindel provided a method to calculate the effect of curvature on reflection level by the relation of the inverse square law, diffraction (shown above) and the curvature of the surface (R). Here, the curvature effect, ΔL_{curv}, is independent of frequency. The graphical solutions for concave and convex surfaces are shown below.

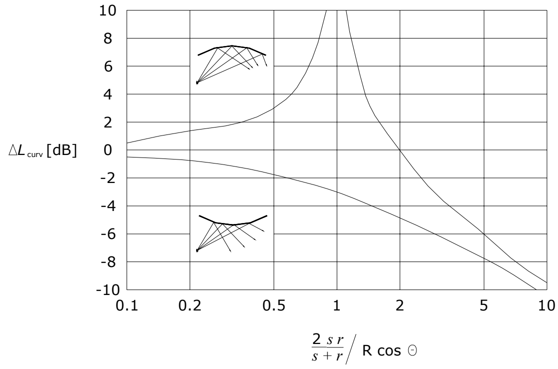


Figure 7: *Level change due to curvature of the reflector.*

In auditoria, geometrical acoustics is useful in the study of the early sound distribution: the first reflections coming from walls, ceiling or reflectors.

1.1.2. Wave-Theory Room Acoustics

The wave theory approach is used to study the sound wave behaviour in the low frequency region for simple and small room shapes. The sound field in this kind of enclosure with the presence of little damping is characterized by a resonance or modal behaviour. Resonances occur at particular frequencies called resonance frequencies normal modes or eigenfrequencies that are the natural oscillation frequencies of a room when excited by short impulses:

$$f_{q_x, q_y, q_z} = \frac{c}{2}\sqrt{\left(\frac{q_x}{l_x}\right)^2 + \left(\frac{q_y}{l_y}\right)^2 + \left(\frac{q_z}{l_z}\right)^2} \tag{1.4}$$

Where:

q_x, q_y, q_z = Natural numbers 0, 1, 2, 3, etc

l_x, l_y, l_z = Room dimensions

c = Speed of sound

At resonance and without losses (damping) the energy in the room will build up infinitely. This is the oscillation mode or eigenmode of the room. The mode shapes are the geometrical patterns of sound pressure at such resonances. For each eigenfrequency, there is at least one eigenmode. The eigenfunctions associated with such eigenfrequencies are obtained by the multiplication of three cosines, each of which describes the dependence of the pressure in one coordinate:

$$p_{q_x, q_y, q_z}(x,y,z) = C\cos\left(\frac{q_x\pi}{l_x}\cdot x\right)\cos\left(\frac{q_y\pi}{l_y}\cdot y\right)\cos\left(\frac{q_z\pi}{l_z}\cdot z\right) \quad (1.5)$$

In the expression above, C is an arbitrary constant. This formula represents a three-dimensional standing wave. When one of these cosines becomes zero the sound pressure will be zero. For a sound source, emitting at one single frequency all modes will oscillate at this "prescribed" frequency. How "strongly" any particular mode is excited depends on: how close its resonance frequency is to that emitted by the source, the damping of the mode and the localization of the source (eigenfunction value).

Interference between modes depends on how the sound pressure contributions of each mode add up with respect to magnitude and phase, varying the sound pressure from one position to another. At high frequencies, modes will "leak into another". In the figure below, dips and crests mean interference between modes.

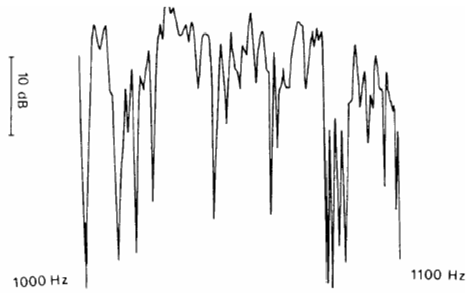


Figure 8*: Variation of the Sound Pressure Level at 1kHz.*

If the modes are weakly damped and uncoupled, once the sound source is switched off the modes will start to oscillate at their natural resonance frequencies. Modes with resonance frequencies closest to the excitation frequency will dominate the decay process.
The number of modes N_m in a room with an arbitrary shape and volume V below a frequency f can be estimated by the following equations:

$$N_m = \frac{4\pi}{3c^3}Vf^3 \quad (1.6)$$

Mode density, dependent on room volume and frequency, is defined as:

$$\frac{dN}{df} = \frac{4\pi}{c^3}Vf^3 \quad (1.7)$$

This expression explains the unsuitability of small rooms for music reproduction: they cannot support low frequencies by appropriate (and damped) room modes. A large amount of modes is required to measure the spatially averaged effective value of the sound pressure. The number of modes is dependent on both the mode density and the bandwidth of the noise.

With the excitation switched off, the energy of each eigenmode decays exponentially. The resonance curves for the effective sound pressure of a particular mode will have a certain width due the presence of the real decay constant of a specific mode, δ_q, At low frequencies, the resonance curves for various modes will overlap.

The Q-value is a metric used to describe the width of a resonance. Q is defined as the resonance frequency, ω_n, divided by the value of the half-power frequency bandwidth of the resonance at –3dB, $\Delta\omega_n$. Low and high Q-values correspond to damped and undamped resonances respectively (figure below).

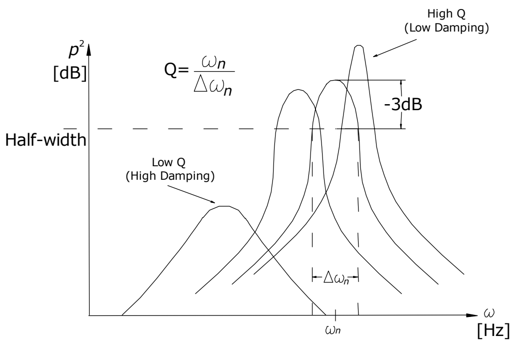


Figure 9: *Resonance Curves and the Q Value.*

Schroeder defined an upper frequency limit for the application of the wave approach. This is the frequency f_s at which at least three modes are overlapping within its resonance half-width:

$$\langle \Delta f_n \rangle = 3\,\frac{c^3}{4\pi V f^2} \tag{1.8}$$

where the angle brackets indicate averages. By introducing the average damping constant, $\Delta f_n = \delta / \pi$, its relation with the reverberation time, $\delta = 6.9 / RT$, c=340 m/s, and solving for f it is possible to find the Schroeder Frequency:

$$f_s > 2000\sqrt{\frac{RT}{V}} \tag{1.9}$$

Above this frequency limit, the sound field is considered diffuse with the damping of the modes being so high that resonance curves are wider and with a large modal overlap. It is then difficult to look for individual modes. For large halls, the Schroeder frequency is typically about 20 or 30 Hz whereas in small rooms the acoustic properties are determined by the values of the individual eigenfrequencies. The number of eigenfrequencies from zero up to f_s can be calculated by the following relation:

$$N_{f_s} \approx 900\sqrt{\frac{RT^3}{V}} \tag{1.10}$$

1.1.3. Statistical acoustics (The Diffuse Field Approach)

Statistical acoustics involves the presence of a diffuse sound field, e.g. above the Schroeder frequency limit. A diffuse sound field features: the same sound pressure (energy density) everywhere in the room, but close to the source, the arrival of equal intensity from all angles and the presence of a broadband source allows for the study of a wide frequency band rather than a single frequency. Nevertheless, in practice there is no perfect diffuse conditions; close to walls the sound field is very coherent and it is for distances beyond one half wavelength from all surfaces that the sound field is considered diffuse.

Sound build up is the process of increasing energy density in a room determined by the source's radiated power and the way in which sound is absorbed by both the surfaces and objects inside the room.

The sound absorption in a surface is represented by its mean sound absorption coefficient, α which is frequency dependent, relating both the incident and absorbed power as:

$$\alpha = \frac{\mathrm{W}_{abs}}{\mathrm{W}_{\inf}} \tag{1.11}$$

The product of α and the surface of the room, S, is the absorption surface A with units in m²S (Square Meter Sabine). The average absorption coefficient (α_m) is the mean value for various surfaces with different properties and is defined as:

$$\alpha_m = \frac{\sum_i S_i \alpha_i}{\sum_i S_i} \tag{1.12}$$

The product of the total wall area and the average absorption coefficient of the walls ($S\alpha_m$) is called the total room absorption, A'.

During the sound build up process, the energy in the room increases as:

$$E \propto \left(1 - e^{\frac{-cAt}{4V}} \right) \tag{1.13}$$

Where:
A = the total room absorption
c = The speed of sound
V = The volume of the room
t = Time

On the other hand, the rate at which the energy decreases once the sound source has been turned off is called the rate of decay, δ. The value of the decay constant is an average of all the decay constants from the modes present inside the room that are very close to each other. It is more common to express the sound decay by the reverberation time of the room, RT, which is the time that it takes the sound to decay 60dB.

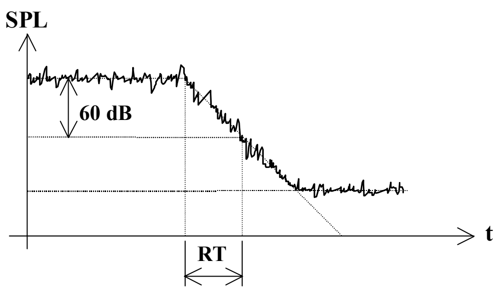


Figure 10*: Definition of reverberation time.*

Both, RT and δ are related by:

$$-60 = 10\log_{10} e^{-2\delta T} \tag{1.14}$$

By solving for RT:

$$RT = \frac{3\ln 10}{\delta} = \frac{6.9}{\delta} \tag{1.15}$$

The mean free path, l_m, represents the distance between two successive sound wave collisions and depends on room properties as geometry and on the diffusion provided by surfaces and objects In a room with volume V and a wall surface area S, l_m is given approximately by:

$$l_m = \frac{4V}{S} \tag{1.16}$$

Every time a wave is reflected, it loses an amount of energy equal to $1-\alpha$. Adding the attenuation due to travelling a distance ct in a medium, such air (with a frequency dependent attenuation value m) the remaining energy at each instant of time is approximated as:

$$E \propto (1-\alpha)^{\frac{ct}{l_m}} e^{-mct} = e^{-2\delta t} \tag{1.17}$$

By taking the logarithm on both sides of the former expression,

$$2\delta = mc - \frac{c}{l_m} \ln(1-\alpha) \tag{1.18}$$

Substituting the expressions for δ and l_m,

$$\frac{6 \ln 10}{RT} = c\left(m - \frac{S}{4V}\right) \ln(1-\alpha) \tag{1.19}$$

And solving for RT Eyring's formula is obtained:

$$RT = \frac{24 \ln 10 V}{c(4mV - S \ln(1-\alpha))} \tag{1.20}$$

By introducing A' in the expression above results in Sabine's formula for calculating RT:

$$RT = \frac{24 \ln 10 V}{c(4mV + A')} \tag{1.21}$$

Here m is important for the high frequency absorption in large volume spaces such as concert halls.

1.2. The Room Impulse Response

1.2.1. Structure of the RIR

A room can be considered as a linear system (linearity here means that multiplying the input signal by any factor results in an output signal which is augmented by the same factor). The main characteristic of such systems is represented in the figure below.

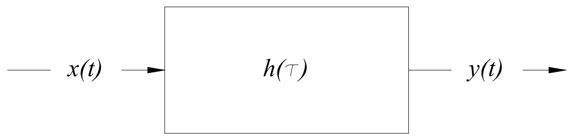


Figure 11: *A Linear system with an input and output signals and its impulse response* $h(\tau)$.

Here, an input signal, $x(t)$, yields an output signal, $y(t)$, characterized by its impulse response, $h(\tau)$. The behaviour of any input signal can be predicted in an accurate way by convoluting the output signal and the impulse response:

$$y(t) = \int_{0}^{\infty} h(\tau)x(t-\tau)d\tau \tag{1.22}$$

In an analogy, the room's impulse response RIR is the pressure-time response function at a receiver position, $y(t)$, in the room because of an excitation by a source, $x(t)$. The RIR gives a complete description of the changes a sound signal undergoes when it travels from one point in a room to another. Those changes are generated, among others, by geometry, source and receiver positions (and its respective directivities), and characteristics of the room: absorption, diffraction, reflection and losses. A RIR diagram shows the temporal distribution of sound and it contains all significant information on the temporal structure of the sound field. Whilst the vertical axis represents the sound level, the horizontal axis represents the times of arrival of impulses (see figure 12). The RIR consists of three main parts:

Direct Sound: The sound travelling directly from the source to listener. Usually the first impulse at leftmost of the graph.

Early Sound. - Early part of the energy coming from reflections on walls and ceiling that includes the direct sound plus the reflected sound up to the first 50 msec for speech or the first 80 msec for music.

Reverberant Sound. - Late part of the energy that includes all the reflections arriving either after the first 50 or 80 msec.

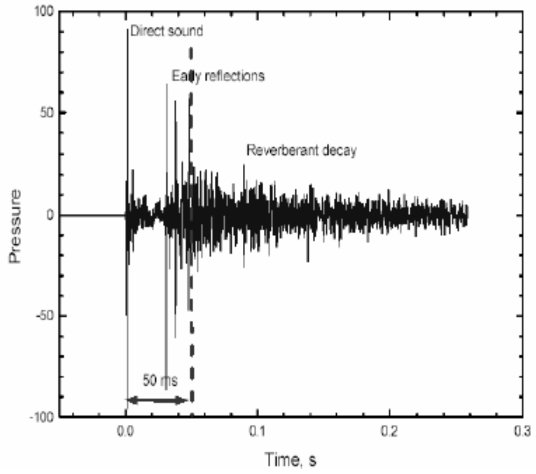


Figure 12: *Typical Impulse Response Plot.*

Normally, the plotted impulses show a gradual decrease in sound level over time. When this is the case (no presence of echoes, e.g. impulses with higher SPL than its neighbours), a listener does not perceive each impulse individually but reverberation instead. The energy-time function (figure 13) is defined as the squared RIR:

$$E(t) = [h(t)]^2 \quad (1.23)$$

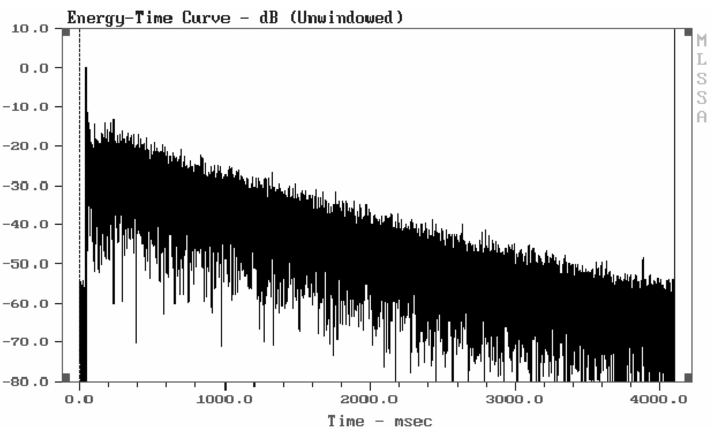


Figure 13: *Energy-Time Plot.*

For the purposes of detailed analysis, different types of impulse responses are required. To do so, microphones with different directionalities are used. Among those recurrent measures there is the binaural room impulse response, $BRIR$, whose main aim is to measure the RIR corresponding to each ear of a listener. In practice, this is obtained by placing microphones in each side of a dummy head (figure 14).

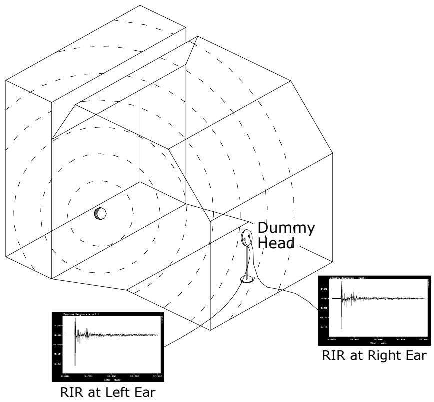


Figure 14: *Typical Binaural Room Impulse Response Measurement Set Up.*

1.2.2. Factors that influence the Room Impulse Response in halls

Besides, those related to listener's position within a hall (which will be analysed in depth in the following pages), some other factors have certain influence on the RIR, especially in the late "reverberant" part. These are air temperature and relative humidity, which influence sound propagation and absorption. Most rooms are inhomogeneous as regards to temperature owing to air handling for heating and cooling, the heating due to sunlight or electronic equipment and the diffusion of heat in and out of the room boundaries. Thus, there are many sources of heat causing local and global variations in temperature.

When sound propagates through such inhomogeneous medium, there are some fluctuations in sound speed due to local density transitions and the presence of secondary effects such as absorption, refraction and diffraction. The equation of state relates the variables of fluid pressure and density and explains the relation between temperature changes and the speed of sound:

$$p_0 + p' = p(\rho_0 + \rho', s_0) \tag{1.24}$$

Where p_0 and ρ_0 are the ambient fluid pressure and density respectively with p' and ρ' being the fluctuating acoustic contributions. The specific entropy term s_0 indicates that the ambient is homogeneous. Since the sound pressure levels used in room acoustics are below the 130dB ($p_{ref} = 20\mu Pa$) and by knowing the values of the barometer pressure, $p_0 \approx 0.1MPa$ (around 194dB) and $\rho \approx 1.2kg/m^3$ is possible to state:

$$p = p_0 + p' \qquad p' << p \tag{1.25}$$

$$\rho = \rho_0 + \rho' \qquad \rho' << \rho \tag{1.26}$$

The latter expression since $p \propto \rho^{\kappa}$ (where $\kappa = c_p / c_v$ is the ratio of specific heat). By using a Taylor series, the linearized equation of state is:

$$p' = c^2 \rho' \qquad c^2 = \frac{\partial p}{\partial \rho} \tag{1.27}$$

For sound propagation in gases and making the Laplace adiabatic assumption:

$$c^2 = \frac{\kappa p}{\rho} \tag{1.28}$$

With κ=1.4 for air. By applying the ideal gas, equation $p = \rho TR$, with the universal gas constant $R = c_p - c_v$ equal to 287 for dry air, is possible to obtain the speed of sound in terms of the temperature (in Celsius):

$$c \approx 20.03\sqrt{273.15 + \mathrm{T}} \tag{1.29}$$

Experiments have shown that small temperature fluctuations can lead to relatively large changes in the impulse response. The effect of the temperature diminishes as the distance between loudspeaker and microphone (or source and listener in a real-life set up) becomes smaller.

Since the late parts of the impulse response (reverberant sound energy contributions) propagate over a much longer distance, they are subject to the largest relative time changes. Thus, variation will be greater on long reverberation since a slight sound speed change will alter significantly the later reverberation path times.

Beside that absorption provided by surfaces in the form of their sound absorption coefficient, α, air itself also produces absorption The latter is achieved by two types of attenuation: classical attenuation and molecular attenuation The former involves viscous friction and heat transfer whereas the latter requires energy exchange by molecular movement of rotation, vibration and displacement. In the range of frequencies below the 30kHz, molecular attenuation dominates due to its dependency of frequency, temperature and relative ambient humidity. The latter is the concentration of water in the air expressed as a percentage relative to the maximum admissible at such temperature without causing condensation.

As stated in the equations above, density fluctuations also affect the impulse response. Absorption of energy as sound travels through the air reduces RT above 2kHz. The major absorption effect is caused by simultaneous presence of oxygen and water vapour in the air. Humidity also increases diffusion of high frequencies. From Sabine formula, the air absorption is implicit in the term $4mV$, where V is the volume of the room in m3, times the quantity $4m$ (in m-1).

Some values for the factor $4m$ are shown in the table below as a function of relative humidity assuming an ambient temperature of 20ºC.

Relative Humidity (%)	$4m$ (Frequency, Hz) 500	1,000	2,000	4,000
50	0.0024	0.0042	0.0089	0.0262
60	0.0025	0.0044	0.0085	0.0234
70	0.0025	0.0045	0.0081	0.0208
80	0.0025	0.0046	0.0082	0.0194

Table 1: *4m coefficients at 20ºC for different relative humidities (ISO 9613-1).*

Nevertheless, it is difficult to find a definitive value for m: several authors, bibliographies and even the standard ISO 9613-1 give different values for the same frequency and relative humidity values. Below is shown a graph to evaluate the single value m as a function of relative humidities up to 80% for 20ºC of temperature.

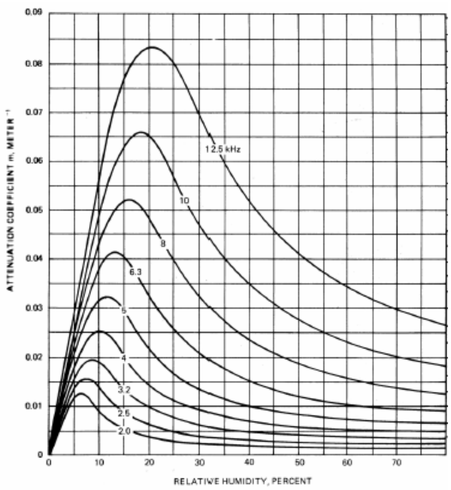


Figure 15:*Values of the total attenuation coefficient* m *vs. percent relative humidity for air at 20ºC and normal atmospheric pressure for frequencies between 2kHz and 12.5kHz.*

1.3. The Acoustical Parameters

The acoustical parameters (also known as acoustical attributes) are metrics adapted through the years to describe the subjective evaluations given by audiences regarding the suitability of acoustic environments. These were applied in the very beginning with techniques derived by psychologists since listening is primarily a physiological process.

As the *RIR* shows it, energy contributions from the room are composed of the direct sound and multiple reflections. A single reflection has the following characteristics: a sound level, a delay relative to the direct sound and a certain direction. For reflections arriving from the side, the subjective effects associated with a single reflection can be described by the figure 16 where the direct sound is used as a reference in both level and delay.

Many reflections are below the threshold of hearing whereas the subjective response to strong reflections depends on the delay. Loud and late reflections are perceived as disturbing echoes. Early loud reflections cause the so-called "image shift" with the sound apparently coming from a direction between the direct and reflected sound directions. For short delays, (less than 7 ms) a lesser image shift occurs to a lesser extent.

For delays around the 20 ms, the tone of the music appears to sharpen and become harsher, known as coloration; this occurs especially with violin tones. Tone coloration is noticeable in halls with reflections from large plane surfaces, with an increase for overhead reflections.

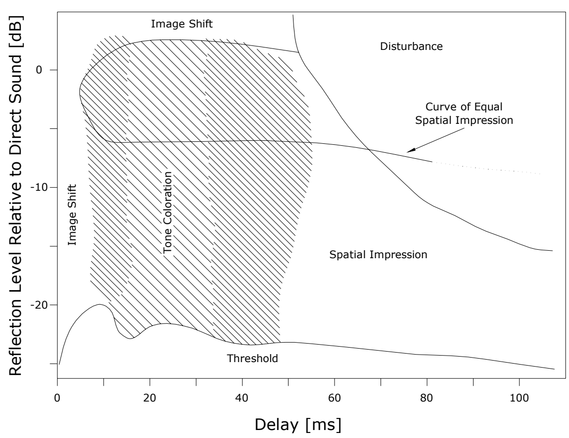


Figure 16:*Subjective effects with music for a single lateral reflection (azimuth angle 40º).*

The principal subjective effect of an early reflection is called spatial impression. With lateral reflections, the subjective impression is of a broadening of the source: a three-dimensional sound experience with a sense of envelopment by the sound. In addition, the hearing system responds favourably to dissimilarity at the two ears by performing a cross-correlation type analysis.

Beranek was one of the first to list a series of subjective attributes: warmth, liveness, intimacy, brilliance and ensemble. Nowadays, for design purposes, it is convenient to be able to relate a physical characteristic of the hall to a subjective quality. Hawkes and Douglas established the first five dimensions for describing concert hall acoustics: adequate clarity to enable musical detail to be appreciated, suitable reverberant response of the room so that the listener feels surrounded or enveloped by sound, sensing the acoustic experience as intimate and having the adequate loudness.

Most parameters can be derived from the analysis of the RIR. A classification of such parameters can be done by setting relationships between the three main stages of the impulse response, as is shown in the figure 17. However, the adequate numerical values for each of those parameters are not enough to guarantee an optimal listening experience inside concert halls. Among others, two main conditions have to be accomplished: a suitable background noise level and visual inspection that may give an idea of the existence of acoustical defects such as echoes or coloration even though it is possible to evaluate these from the RIR.

In the following pages are shown the main objective parameters evaluated from the RIR. They are classified as; strength parameters, initial time delay gap, reverberation time and early decay time, warmth and brilliance, parameters derived from the balance from early and late energy, envelopment and spaciousness, stage support factor and those used to evaluate speech intelligibility. Acoustical defects such as echo and coloration are also named together with the adequate background noise level requirement.

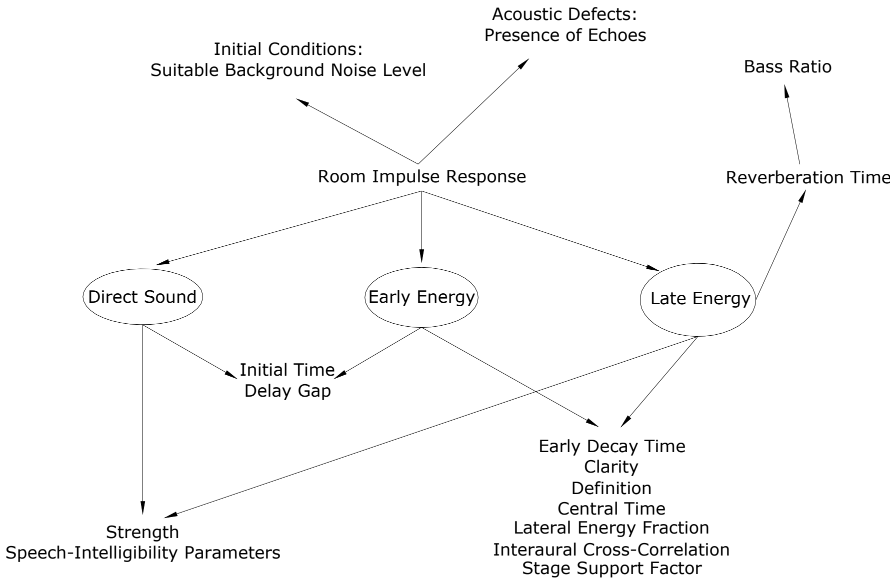


Figure 17:*Relationship between Parameters derived from the Room Impulse Response.*

1.3.1. Strength Parameters

The Strength Index (G) is the difference in SPL obtained by subtracting the sound level produced by a non-directive source in an anechoic room at a distance 10 m, $L_{p\,E,\,10}$, from that observed in the hall, $L_{p\,E}$. In terms of energy, such difference is the ratio between the energy-time function (the source's squared impulse response, $[h(t)]^2$) measured in the hall and that $[h(t)]^2$ for the same source at 10 m in an anechoic environment, $[h_A(t)]^2$.

$$G = L_{pE} - L_{pE,10} = 10\log\left(\frac{\int_0^\infty [h(t)]^2\,dt}{\int_0^\infty [h_A(t)]^2\,dt}\right)[\text{dB}] \tag{1.30}$$

Experiments have shown that G shows a linear decrease from the front to the rear of any hall, which corresponds to 1.0dB - 3.3dB per distance doubling. This is lower than the value range obtained from the equation above (2dB - 3dB). The Strength Index does not depend in a simply way on the reverberation time nor on the geometrical data, which reinforces the idea that, inside concert halls, sound fields are not perfectly diffuse.

Actually is used G_{mid} (the mid-frequency strength factor) that is the average of the G's measured in the 500Hz and 1kHz frequency bands. At low frequencies, G is very important. At the listener, position (according to the inverse square law) direct sound contributions are small. Furthermore, the equal loudness contours (figure 18), which describe the frequency response of hearing, show the unevenness for low frequencies at low sound pressure levels, relative to higher frequencies. For music, parameters as warmth (bass ratio) involve RT at low frequencies, thus the low frequency content must be enhanced. As an evaluator of the low frequency region, G_{low} (an average at 125 and 200 Hz) and G_{125} are used.

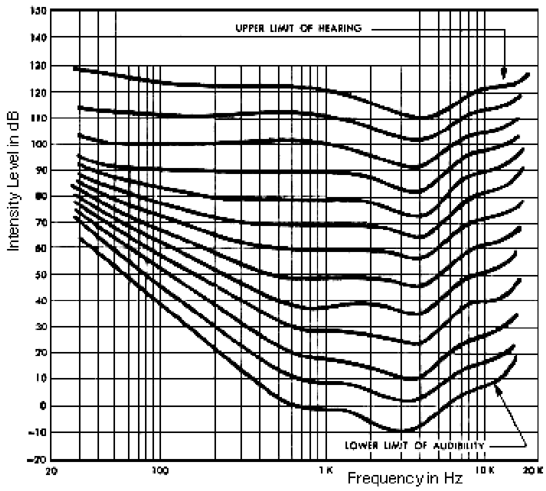


Figure 18: *Equal Loudness Contours.*

The Strength Factor, LG_{80}^{∞}, is related to late energy arriving from lateral directions:

$$LG_{80}^{\infty} = 10\log\left(\frac{\int_{0.08}^{\infty}\left[h_{10m}(t)\cos\theta\right]^2 dt}{\int_{0}^{\infty}\left[h_A(t)\right]^2 dt}\right)[\text{dB}] \tag{1.31}$$

Where $\left[h_{10m}(t)\cos\theta\right]^2$ is the squared impulse response measured by a figure-of-eight microphone at a standard distance from the source (10 m) and $\left[h_A(t)\right]^2$ is the squared impulse response measured in an anechoic environment at 10 m.

Distance source – listener is the main influence on G; this attribute decreases as long as the distance increases. For G_{low}, the position of measurement is important due to attenuation in audience areas where the sound had travel at grazing incidence over a highly absorbent surface: the audience seated in front of the listener. Such phenomenon known, as seat-dip attenuation does not occur when sound reaches, in a straight path, the audience seated in balconies.

1.3.2. Initial Time Delay Gap (ITDG)

Intimacy is the subjective impression of listening to music in a large room and hearing it as if in a smaller room. The objective measure of this impression is the Initial Time Delay Gap ($ITDG$); the interval between the direct sound arriving to the listener's ears and the first reflection arriving from the walls or ceiling, see Figure 19.

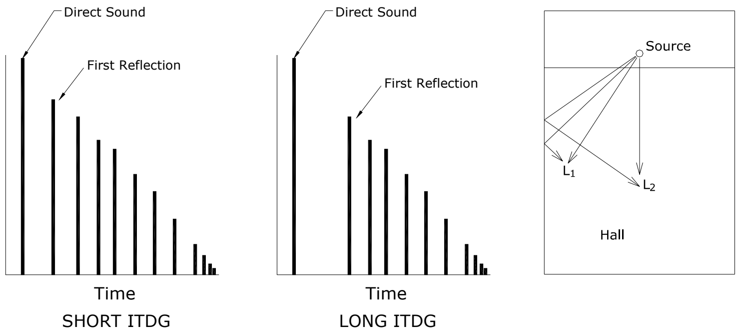


Figure 19: *Variation of ITDG with location of the listener in a hall.*

A short $ITDG$ means a higher level of intimacy, thus narrow halls, regardless of their length, are more intimate than other shaped halls, due to the proximity of walls to the listeners. However, intimacy can be achieved in large and wide halls by using reflectors on walls or ceilings in order to reduce the first reflection path, and consequently obtaining a shorter $ITDG$. A recommended value for $ITDG$ is 30 ms or less. Since $ITDG$ is very dependent on the receiver position, it is standard procedure to measure it at a position roughly in the centre of a hall for comparison purposes.

1.3.3. Reverberation Time and the Early Decay Time

As stated above, the reverberation time RT is defined by Sabine's equation:

$$RT = \frac{24 \ln(10) \mathrm{V}}{c\,(4\,m\,\mathrm{V} + A')} \tag{1.32}$$

In practice, RT can be measured by either steady state or impulsive excitations. The former process involves a constant sound power radiated into a room until the energy density approaches an asymptotic value. Once the steady state has been reached, with the energy being absorbed as fast as it is supplied, the source can be stopped and then it is possible to measure the decay rate of energy inside the room.

Although RT is defined over 60dB decay, a listener in a hall is not able to hear decays of 60dB in continuous speech or music. Thus, measurements of RT use decays from –

5dB to –35dB (T_{30}) or –5dB to –25dB (T_{20}) to determine the decay rate and then extrapolate to obtain RT over -60dB. Reverberation time, RT, is presented in the frequency bands of interest but usually it is the averaged RT_{mid} (500Hz and 1kHz), measured in an occupied (2 positions) and empty (8 to 24 positions) hall that describes the reverberation time in a hall.

Another method to determinate RT from the impulse response analysis is the Schroeder's backward integration. This technique consists in obtaining the backward integration of the square of the impulse response. The obtained slope (Figure below) gives an equivalent ensemble average of energy decay by random noise excitation, $< g^2(t) >$.

$$< g^2(t) >= \int_t^\infty [h(t)]^2 \, dx \tag{1.33}$$

The backward integrated curve is much smoother and accurate than those obtained by random noise excitations, allowing a better estimation of RT.

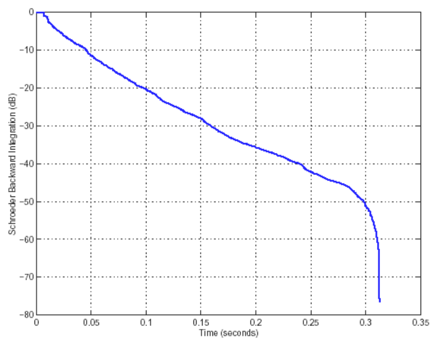


Figure 20:*RT Slope Obtained From Applying The Schroeder Backward Integration.*

The Early Decay Time (EDT) is the RT measured for the first –10dB decay multiplied by a factor of 6. According to Beranek, EDT shows a better correlation with the ranking of the rooms than RT. Although it may differ from Sabine's time, listening tests based on binaural impulse responses recorded inside concert halls confirmed that the perceived reverberance is more closely related to EDT. Furthermore, owing to the unevenness of some decays, EDT characterizes in a better way the behaviour of the whole sound decay slope, (Fig 21).

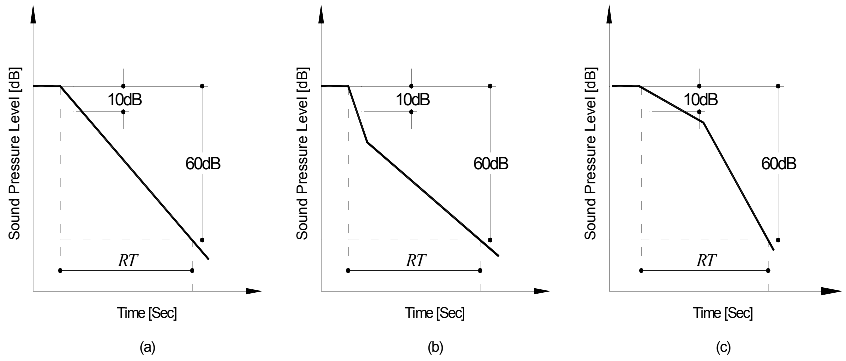


Figure 21: *Simplified sound decay outlines. (a) A uniform decay. (b) A rapid initial decay followed by a slow later decay. (c) A slow early decay followed by a rapid later decay.*

By visual inspection in Sabine's formula it is easy to see how the volume and the presence of absorption inside a hall changes RT and consequently EDT. The amount of reverberant energy inside large halls is bigger than in those with low cubic volumes. Nevertheless, a large hall may also imply larger losses both by the presence of absorbent audience area and larger travel paths where the high frequency content in the impulse response will spread diffusely and eventually die faster than low frequencies.

Room shape establishes a big difference between EDT and RT. The overall reverberation time does not show substantial variations because the decay process as a whole is made up of numerous reflections with different delays and strengths coming from all the walls whereas early decay time is strongly influenced by early reflections, and therefore depends noticeably on the measuring position and details of the room geometry. This first group of reflections arrives at listener's position either from non-absorbent lateral walls in narrow and high ceiling halls (shoebox shaped) or from reflecting low ceilings, suspended panels or from the presence of balconies in other shaped halls.

Regarding age, there is no proof of a mellow in RT and EDT. In nine halls that could be measured after 30 years, differences on the reverberation times measured in five octave-frequency bands are 0.08 sec. This difference is within the 0.1-sec accuracy of the measurements themselves.

1.3.4. Bass Ratio & Brightness

The subjective sensation of an increment in the reverberation time at low frequencies is known as warmth of musical sounds and has been identified as an important requirement for music halls. This attribute is determined by comparing low and high frequency reverberation times, known as the bass ratio, BR.

$$BR = \frac{\left(RT_{[125]} + RT_{[250]}\right)}{\left(RT_{[500]} + RT_{[1,000]}\right)} \tag{1.34}$$

Low frequency absorption is dependent on the materials used in the construction of internal surfaces. Air cavities will perform as resonators or Helmholtz absorbers decreasing BR whereas hard and heavy surfaces are recommended for music hall finishing. In relation to RT, the recommended BR values are as follows:

$$1.1 \text{ to } 1.25 \quad \Rightarrow RT \geq 2.2 \text{ sec}$$
$$1.1 \text{ to } 1.45 \quad \Rightarrow RT < 1.8 \text{ sec}$$

On the other hand, music performed with a very low EDT (or RT) at high frequencies lacks brilliance or liveness. Music is considered bright when doesn't lacks of high frequencies, thus producing a clear and ringing sound. By defining the EDT_{mid} as the average of de EDT values at 500 and 1,000 Hz, the brilliance in a music space is measured by the following ratios with the recommended minimum values:

$$\frac{EDT_{2,000}}{EDT_{500} + EDT_{1,000}} \geq 0.9 \tag{1.35}$$

$$\frac{EDT_{4,000}}{EDT_{500} + EDT_{1,000}} \geq 0.8 \tag{1.36}$$

1.1.4. Balance between Early and Late Arriving Sound

Music and speech are within the frequency range shown below. As they are made up of the same bursts of sound and silence structures their physical behaviour is practically the same inside a concert hall. Hearing discriminates them through: tonality and harmony for music and tone recognition and phrasing for speech. In concert hall design, tonality has little influence since frequency is not affected by reflection.

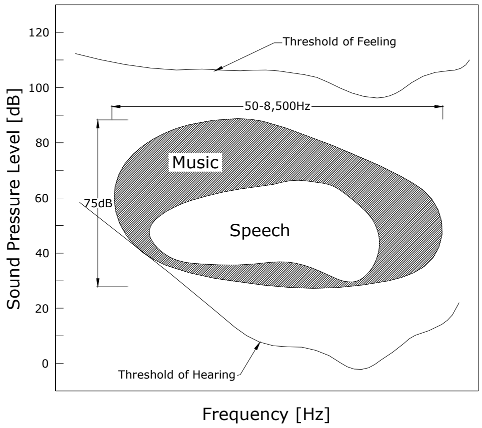


Figure 22*: Frequency range for musical and speech sounds.*

The balance between the early energy fraction and the late energy fraction can evaluate the level of transparency for music and speech inside halls. The metrics used for that balance are Definition (D), Clarity (C_{80}) and Centre Time (t_s). These are the ratios of different early time limits (also called critical delay times) and the late part of the impulse response. According to ISO 3382:1997:

$$C_{t_c} = 10\log\left(\frac{\int_0^{t_c}[h(t)]^2\,dt}{\int_{t_c}^{\infty}[h(t)]^2\,dt}\right) \quad [\text{dB}] \qquad (1.37)$$

With:
C_{t_c} =Early - late index
t_c = Early time limit, 50 or 80 ms

C_{50} is used when it is necessary to qualify the suitability of rooms for speech by setting the first 50 ms as the "critical delay time" of the reverberation process, e.g., the amount of sound, which is beneficial for speech intelligibility. Although a concert hall is not used for speech as in theatres and conference rooms, musical performances may include operas and singing.

$$C_{50} = 10\log\left(\frac{\int_{0}^{0.05}[h(t)]^2\,dt}{\int_{0.05}^{\infty}[h(t)]^2\,dt}\right) \quad [\text{dB}] \tag{1.38}$$

According to Marshall, a representative C_{50} value is calculated by applying 15%, 25%, 35% and 25% weighting factors to the 500Hz, 1kHz, 2kHz and 4kHz octave bands respectively. This value is known as C_{50} ("speech average").

$$C_{50(\text{"speechaverage"})} = 0.15 \cdot C_{50(500\text{Hz})} + 0.25 \cdot C_{50(1\text{kHz})} + 0.35 \cdot C_{50(2\text{kHz})} + 0.25 \cdot C_{50(4\text{kHz})} \; [\text{dB}] \tag{1.39}$$

Thiele proposed the commonly used value of Definition (Deutlichkeit). D_{50} is somewhat similar to C_{50} but the former considers the whole impulse response energy instead of only the late part beyond the early time limit and expresses everything in percentage.

$$D_{50} = \left(\frac{\int_{0}^{0.05}[h(t)]^2\,dt}{\int_{0}^{\infty}[h(t)]^2\,dt}\right) \quad [\%] \tag{1.40}$$

D_{50} is related with C_{50} by the following relation:

$$C_{50} = 10\log\left(\frac{D_{50}}{1 - D_{50}}\right) \quad [\text{dB}] \tag{1.41}$$

The optimum values of D_{50} are beyond the 50%, which means positive C_{50} values (0 up to 6dB or 7dB).

The Clarity (C_{80}) index is used to characterize the transparency of music in a concert hall. Due to differences in the range of modulation between speech and music (in music, longer times between reflections are allowed without perceiving echoes) the critical delay time has been set in 80 ms for this index.

$$C_{80} = 10\log\left(\frac{\int_{0}^{0.08}[h(t)]^2\,dt}{\int_{0.08}^{\infty}[h(t)]^2\,dt}\right) \quad [\text{dB}] \tag{1.42}$$

In practice the C_{80} "music average" value for the 500Hz, 1kHz and the 2kHz centred octave bands is used:

$$C_{80(\text{"music average"})} = \frac{C_{80(500\text{Hz})} + C_{80(1\text{kHz})} + C_{80(2\text{kHz})}}{3} \quad [\text{dB}] \qquad (1.43)$$

The optimum values for C_{80} are dependent on the kind of music to be performed but common values are around the –2dB to +2dB with tolerable -3dB values. In front, seats where the direct sound contributions are bigger, it is possible to find values from +3dB to +8dB.

To find a middle point for speech and music auditoria (such as an opera house) is not always possible. With C_{50} and C_{80} being positives the early arriving sound will dominate. This is optimum for speech purposes but music may require reverberant field contributions (C_{80} <= 0) and this will muddle speech.

In order to avoid the discrete division, or "cut-off", of the impulse response into early and late periods (inherent in the definitions of C_{80} and D) Kürer has suggested the concept of Centre Time (t_s). This is the centre of gravity along the time axis of the squared impulse response.

$$t_s = \frac{\int_0^\infty \mathrm{t} \cdot [h(t)]^2 \cdot \mathrm{dt}}{\int_0^\infty [h(t)]^2 \cdot \mathrm{dt}} \quad [\text{ms}] \qquad (1.44)$$

Low values of the centre time t_s indicate high transparency or high speech intelligibility. Recommended t_s values are below the 140 ms for the averaged octave bands from 125Hz to 4kHz.

1.3.5. Envelopment and Spaciousness

In order to achieve a sense of envelopment, one or more strong reflections are required from the side. A reflection is considered lateral when there is a time difference between its arrival at the near and far ear in what is called Inter-aural time. There should be sufficient angle between the reflection path and the plane bisecting the face when the listener faces the source (median plane) as is shown in the figure below. A minimum angle of 20º is considered but the larger the angle the better the spatial effect.

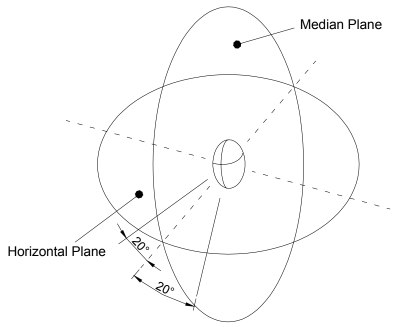


Figure 23:*Planes involved measuring the Lateral Energy Fraction.*

Figure 24 shows three different hall shapes with a sound source and different listeners. The distance source-receivers are the same for the three different cases. From visual inspection, it is possible to see that the angles for the rectangular shape are the largest but smaller than those at the back of an inverse fan shaped hall. Finally, laterality is low in fan shaped halls, which makes them unsuitable for lateral contributions.

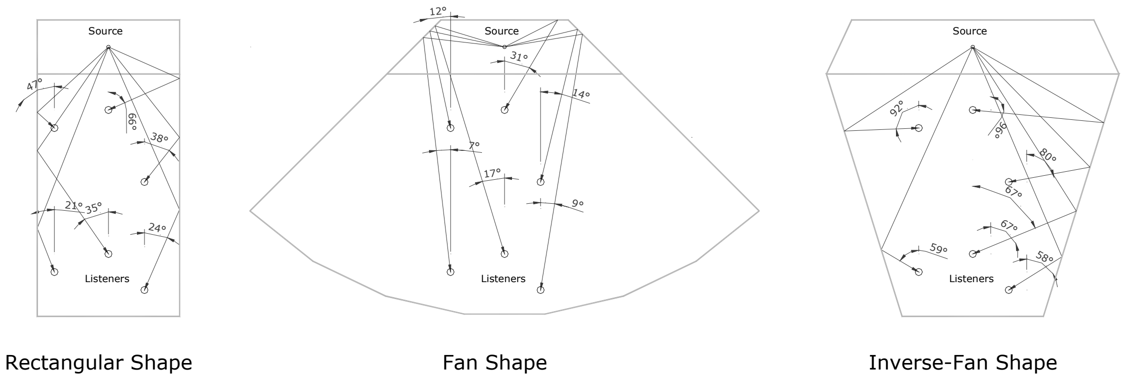


Figure 24:*Laterality in three different plan shapes.*

The sound arriving in and out of the median plane is measured by the Early Lateral Energy Fraction, L_{ef}. This measure is the ratio between the energy arriving from lateral reflections in the interval from 5ms to 80 ms and the total arriving at the listener's position from 0 to 80 ms.

The basic formulae for L_{ef} is:

$$L_{ef} = \frac{\int_{0.005}^{0.08} [h_L(t)\cos\theta]^2 \, dt}{\int_{0}^{0.08} [h(t)]^2 \, dt} \qquad (1.45)$$

Here, $[h_L(t)\cos\theta]^2$ is the impulse response measured by a bi-directional (figure of eight) microphone and $[h(t)]^2$ is the impulse response measured by an omni directional microphone. In large halls, L_{ef} may vary from zero to about 0.5.

In practice the averaged value from the 125Hz to the 1kHz centred octave bands is used:

$$L_{ef\,(E4)} = \frac{L_{ef\,(125\text{Hz})} + L_{ef\,(250\text{Hz})} + L_{ef\,(500\text{Hz})} + L_{ef\,(1\text{kHz})}}{4} \quad [\text{dB}] \qquad (1.46)$$

There are also other quantities related with the binaural response of the hearing. The Apparent Source Width (ASW), the Listener Envelopment (LEV) and the Binaural Quality Index (BQI), all defined by a metric called Interaural Cross-Correlation Coefficient $(IACC)$. This is based in the Interaural Cross-Correlation Function, $IACF_t(\tau)$, defined as:

$$IACF_t(\tau) = \frac{\int_{t_1}^{t_2} h_r(t)\cdot h_l(t+\tau)dt}{\sqrt{\int_{t_1}^{t_2} [h_r(t)]^2 dt \cdot \int_{t_1}^{t_2} [h_l(t)]^2 dt}} \qquad (1.47)$$

Where $h_r(t)$ and $h_l(t)$ are the impulse responses measured at the right and left ear respectively. By setting t_1 as 0, the integration from 0 to t_2 ms will include the energy of the direct sound and the portion of early reflections and early reverberant sound within the t_2 period. Since the difference in arrival time between the directly exposed ear and the opposite ear (figure 25) is around 1 msec, it is customary to vary τ over the range of –1 to +1 ms.

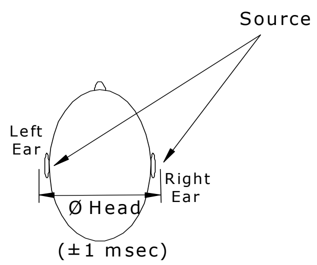


Figure 25:*Time difference between both ears when a sound wave reaches the head in lateral direction.*

Sound waves arriving from lateral directions can only create the binaural difference. Sounds that arrive from straight ahead do not contribute to $IACC$. Different integration periods result in different $IACC$ values: $IACC_A$ for a period from 0 to 1,000 ms, $IACC_{E(arly)}$ from 0-80 ms and $IACC_{L(ate)}$ from 80-1,000 ms.

Interaural fluctuations arise from interference between lateral reflected energy from different directions and interference of direct lateral sound. The physics of sound interference shows that the minimum delay needed is inversely proportional to the bandwidth of the signal. The minimum delay is approximately 0.5/bandwidth thus, for a 100Hz bandwidth, a signal with a minimum of 5ms is required to develop fluctuations. As long as the bandwidth increases, the minimum delay required will decrease.

The spatial impression in a room (Spaciousness) through G_{low} and L_{ef}, is described by the Apparent Source Width. ASW is the subjective impression of an increase in the size of the orchestra produced by early reflections and is defined by $IACC_E$. The Listener Envelopment (LEV) is the degree at which the reverberant sound surrounds the listener coming from all directions rather than from limited directions and is defined by $IACC_L$. High values of these parameters introduce a low level of spaciousness and envelopment whereas values close to zero will give a sense of diffuseness.

Beranek has proposed a quantity called the Binaural Quality Index, BQI, given by the next formula:

$$BQI = \left[1 - IACC_{E3}\right] \tag{1.48}$$

Where $IACC_{E3}$ is the Early Interaural Cross-correlation Coefficient averaged over three octave bands with centre frequencies at 500Hz, 1kHz and 2kHz.

1.3.6. Stage Support Factor

The Support Factor $ST1$ is the degree of support that the hall gives to performers in the stage due to reflections from the walls, ceiling and from the enclosure surrounding the players. This parameter is the ratio between late energy arriving in the interval from 20 to 100 ms and the early energy arriving in the time interval from 0 to 10 ms. The sound arriving in the late interval has been reflected from one or more surfaces back to the musicians on the stage.

$$ST1 = \frac{\int_{0.02}^{0.1} [h(t)]^2 \, dt}{\int_{0}^{0.01} [h(t)]^2 \, dt} \qquad (1.49)$$

The measurement is made without musicians on stage, but with chairs, music stands and percussion instruments in place. A sound source emits a pulse and a microphone receives it at 1 m away from the centre of the source averaging the results over several positions on the stage.

1.3.7. Other Parameters Related with Speech Intelligibility

Besides parameters related with transparency named above there are others specifically used for measuring speech intelligibility. Classified in two groups they are: those derived from the Signal-to-Noise Ratio and those derived from the impulse response analysis.

Any acoustic system is subject to either background or ambience noise and to particular characteristics induced by the acoustic system itself (reflections, echoes, resonance, reverberation, etc). The SNR (expressed in dB) is the logarithmic ratio of the signal and noise levels.

The Articulation Index uses the weighting factors in five frequency bands shown in Table 2 and multiplies them by the SNR in such bands. The total sum of those values gives the AI, which must be a value between 0 to 1 with values lower than 0.3 indicating unintelligible speech and those above 0.7 indicating excellent intelligibility.

Frequency (Hz)	Weighting Factor
250	0.0024
500	0.0048
1,000	0.0074
2,000	0.0109
4,000	0.0078

Table 2: *Weighting Factors Used in Calculating Articulation Index.*

However, AI is not appropriate for calculating speech intelligibility in rooms since SNR is a function of the distance source-receiver, the reflections in the room, speaker orientation, individual speaker differences, etc.

The Speech Transmission Index (STI) involves both SNR and those aspects related with the impulse response that affect intelligibility. As AI, the STI is also a number between zero and one derived from a Modulation Transfer Function (MTF) curves shown in the figure 26.

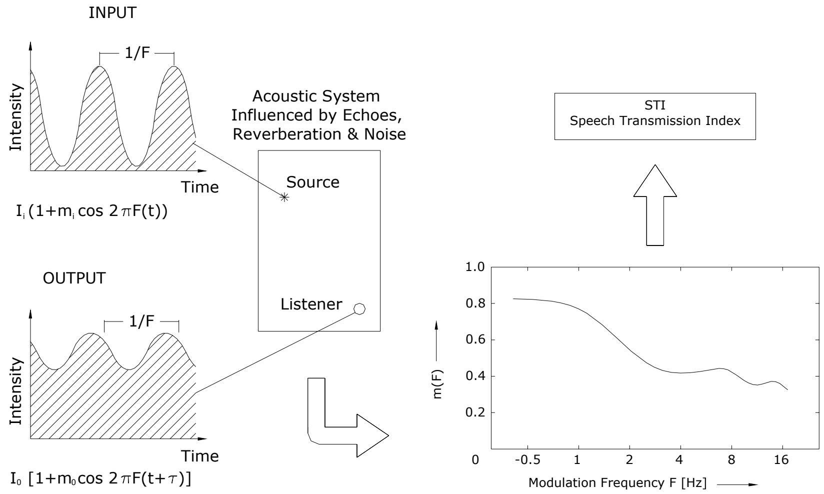


Figure 26: *Modulation Transfer Function Principle for a Modulated Noise Signal.*

The MTF is a measure of the reduction of speech signal modulations generated by the transmission through an acoustic system such a concert hall. These changes in the depth of modulation (due to the presence of noise, echo, reverberation, etc.) are described along seven octave bands, from 125Hz to 8kHz, which are divided into fourteen modulation frequencies ranging from 0.63Hz to 12.7Hz in 1/3-octave steps in order to simulate speech modulation.

Then, the modulation reduction factors are converted into effective signal-to-noise ratios that are weighted and averaged to obtain STI. A method for determining MTF from a single impulse response measurement $h(t)$, is based on the following formula suggested by M. Schroeder and refined by D. Rife:

$$MTF = \frac{\int_0^{\infty} \left[h_f(t)\right]^2 \cdot e^{-jwt} dt}{\int_0^{\infty} \left[h_f(t)\right]^2 dt} \qquad (1.50)$$

Where $h_f(t)$ is the octave-band filtered impulse response at carrier frequency f. Here the STI can be deduced from the magnitude of the complex MTF, $m = |\overline{m}|$. A simplified method of the STI is RASTI (RApid Speech Transmission Index) and it is used in portable measurement equipment. RASTI calculates m simultaneously for only two octave bands: 500Hz and 2,000Hz. For the octave band centred at 500Hz, there are four modulation frequencies whereas for the octave band centred at 2kHz there are five. 'The applied modulation frequencies range from 0.7Hz to 11.2Hz. Each of these nine values of m is converted into an "apparent signal-to-noise ratio":

$$(\mathrm{S/N})_{\mathrm{app}} = 10\log_{10}\left(\frac{m}{1-m}\right) \tag{1.51}$$

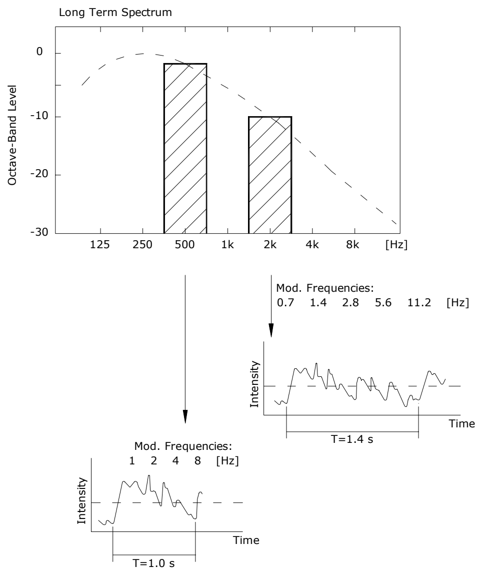


Figure 27: *Illustration of the RASTI test signal.*

These figures are averaged after truncating any that exceeds the range of ± 15: The final parameter is obtained by normalizing the average $\overline{(\mathrm{S/N})}_{\mathrm{app}}$ into the range from 0 to 1.

$$\mathrm{RASTI} = \frac{1}{30}\left[\overline{(\mathrm{S/N})}_{\mathrm{app}} + 15\right] \tag{1.52}$$

Quality Score	RASTI
Bad	<0.32
Poor	0.32-0.45
Fair	0.45-0.60
Good	0.60-0.75
Excellent	>0.75

Table 3: *Scores of Speech Transmission Quality and RASTI.*

1.3.8. Existence or absence of echoes

There are others aspects related with the sound field inside auditoria that must be carefully watched. One of them is the existence of echoes. An echo is a loud reflection (compared with those reflections surrounding it in the reverberation decay process) that arrives with a large time delay.

Echoes are produced either by strong reflections coming from a surface with a shorter path than other weak reflections or by focusing generated by large concave surfaces.

A reflection is perceived or not as an echo depending on its delay with respect to the direct sound (some authors recommend a minimum delay of 50 ms), on its relative strength, on the nature of the sound signal and on the presence of other reflections which eventually mask the reflection under consideration. Some echoes can be traced by the impulse response plot, figure 28.

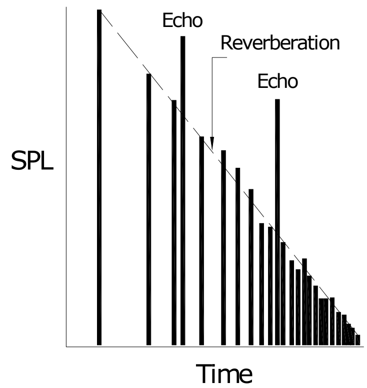


Figure 28:*Impulse diagram of a room with sound-focusing elements.*

For speech, the so-called Threshold of Echo Perception (figure below) has been obtained from experiments. Here are shown the percentage of subjects who felt disturbed by an echo of a relative level as a function of the time delay between the un-delayed signal (or primary sound) and the delayed one. The numbers next to the curves indicate the level of reflections in dB relative to that of the primary sound. The RT is 0.8 sec and the speech rate was 5.3 syllables per second.

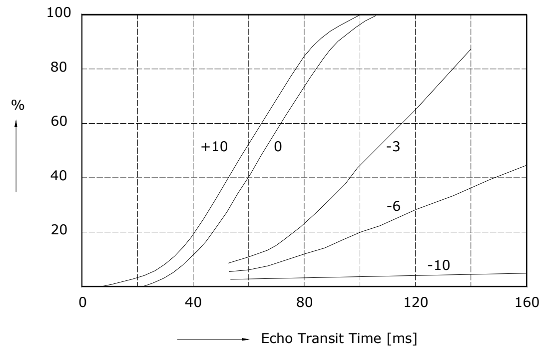
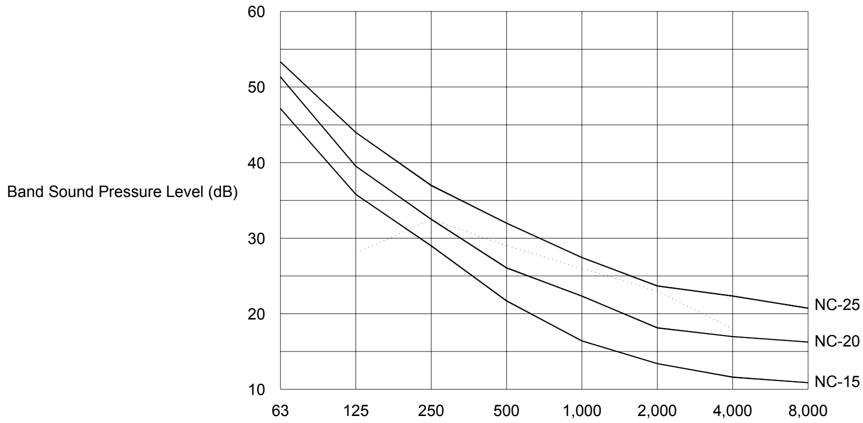


Figure 29:*Percentage of listeners disturbed by a delayed speech signal.*

From the graph above it is possible to see that for a delay time of 80 ms only 20% of the observers felt irritated by the presence of a reflection with relative level of –3dB. The percentage of annoyance increases as the relative level increases. Echoes are more intrusive in concert halls with short reverberation times.

1.3.9. Background Noise Level

One of the most important issues inside auditoria concerns the background noise levels. If these are too high, they will mask all the information generated by the sound sources. The noise generated by external sources, equipment, air conditioning and the same audience affects the natural response of halls by means of masking. The criteria for background noise are specified as NC (Noise Criterion) or NR (Noise Rating) used in America and Europe respectively although they are virtually equivalent. Octave-band noise levels are compared with the target curve that cannot be exceeded at any octave. Criteria for different auditoria are listed in table and figure below. There also appears (as dotted line) the Audience Noise Spectrum that is the contributions to background noise from audience breathing.



Valor NC	Octave Band Centre Frequency (Hz)							
Large Concert Halls								
NC-15	47	36	29	22	17	14	12	11
Opera Houses And Drama Theatres								
NC-20	51	40	33	26	22	19	17	16
Small Auditoria (500 seats max)								
NC-25	54	44	37	31	27	24	22	21

Figure 30: *Noise Criterion Curves relevant to auditoria and (dotted line) audience noise spectrum.*

Beranek introduced the concept of Dynamic Range as the spread of sound levels over which music can be heard inside concert halls. This range extends from the low level of background noise produced by audience and air-heating system up to the loudest levels produced in a performance. From avoiding all extraneous sources of noise, (including traffic and aircraft noise) it is possible to obtain a wide dynamic range.

1.3.10.Concert Hall Size and Shape

A hall should provide sufficient early reflections to all the audience within the early time limit. Distance source-receiver must be close enough to keep the direct sound at a good level as well as offering good visual sites. A maximum length of about 40m is generally adopted. Early side reflections are important for spatial impression and the width of the hall has to be limited to allow lateral reflections. For this purpose, a maximum width of about 32m is recommended.

To improve spatial impression, strong reflections from sidewalls, ceiling and side balcony cornices directed towards the audience are desirable. Low frequencies should be present in lateral reflections for optimum spatial effect and since low frequencies are attenuated by the seat-dip effect mentioned above, these require lateral reflection paths remote from audience.

Auditoria have been designed through the past years with different shape plans. Some shapes are certainly better than others but solutions can be found to improve the acoustics even in rounded or circular shaped plans like the famous Royal Albert Hall in London.. Geometrical acoustics shows that curved surfaces produce focussed reflections, which are a potential source of echoes as well as generators of uneven distribution of sound levels.

Nowadays, the most common concert hall plans used for halls are: the rectangular "shoebox" shape, the fan shape and the hexagonal (inverse-fan-based) shape (figures below). The horseshoe shape, so popular in theatres, is not recommended for music; when enlarged enough to achieve a satisfactory reverberation time, the reflection pattern becomes poor and there are focusing problems associated with the concave rear wall.

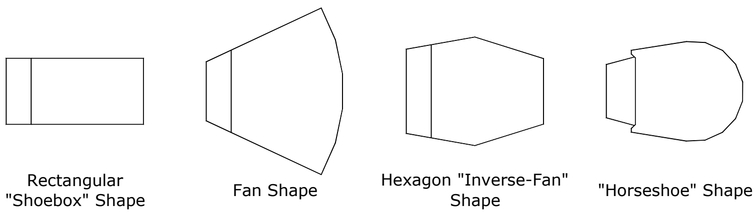


Figure 31: *Simple Plan Shapes for Concert Halls.*

Through the years, the rectangular plan has established a considerable reputation for its acoustics. Its shape does not create focusing and its parallel sidewalls produce a high reflection density, assuring early reflections to the main audience. Most of the seats are near reflecting walls and the small width of the hall enhances spatial impression. Spaciousness in this kind of halls can be further improved by shaping sidewalls near the proscenium and the sides of the performing space in order to direct the sound more uniformly to the audience areas.

On the opposite side are the fan-shape plans. The main difference with the shoebox shape is in the splaying of its lateral walls that can be an advantage in terms of audience capacity but a handicap when reflections from those walls are required to achieve lateral energy contributions. In such halls, the angle of the sidewall reflection relative to the direct sound is small and the spatial effect is weak. Hence, a fan-shaped hall is not good for spatial impression.

A good balance is achieved on inverse-fan shaped plans. The elongated hexagon compromises the visual advantages of the fan-shape and the acoustic advantages of the inverse splay. Here a good spatial impression can be obtained but problems derived from stage design and overall hall plan may arise.

Finally, horseshoe shapes have been popular in theatres for a long time but they are not good for symphonic music. A modern tendency is to place the audience close to the performers in order to improve direct sound contributions. Modern design includes the existence of audience areas surrounding the stage like those in the West Berlin Philharmonic, Madrid Auditorio Nacional or Valencia Palau de la Musica, in what is called terraced concert hall design.

When a concert hall plan shape cannot bring spaciousness by itself the spatial effect can be improved by the use of reflectors. A combination of suspended or sidewall-splayed panels and the interspersing of seating areas with "walls" that are located to provide lateral reflections can be a good solution and raking steps will preserve bass energy. However, overhead reflectors must be used with caution since strong reflections coming from the ceiling produce tone coloration.

The use of a diffuse treatment on lateral reflectors and ceiling is recommended. In the same way, reflectors too close to audience may create image shifts (see figure 16) thus the use of balconies as reflectors involve the use of proper angling and diffuse treatment.

2.Measurements Inside Auditoria

2.1. Measurements Description

In order to obtain the variability mapping of the most important parameters introduced in chapter 1 of this text, a series of measurements inside different types of Auditoria were proposed.

GARCIA-BBM made contact with contractors, administrators and directors of several Theatres, Concert Venues, Auditoriums, Opera Halls, etc where they had been or are currently involved in consulting so as to obtain permission and arrange acccess to the halls to carry out the proposed measurements. From those available during the last 6 months, one representative example of different plan shape, capacity, cubic volume, etc. was selected The following are the selected five halls:

- Teatro Nacional de Catalunya, Barcelona, Spain.
- Laguna de Duero Auditorium, Laguna de Duero, Spain
- Palau de Congresos de Peñíscola, Peñíscola, Spain
- Teatro Serrano's Big Hall, Gandía, Spain.
- Teatro Serrano's Small Hall, Gandía, Spain.

The main idea was to obtain a manageable number of evaluated positions and with this to be able to describe in detail the parameters transitions. The e Maximum Length Sequence (MLS) technique was chosen because it produces fast and accurate results. To do so a fully programmable NORSONICS Real Time Analyser 840 with an inner MLS-signal generator was used.

Details about the RTA840 MLS implementation can be found in Appendix C of this thesis. In summary, the following are the equipment and setup (figure 32) used for the measurements:

BRÜEL & KJAER Omnidirectional Microphone
BRÜEL & KJAER Microphone Preamplifier
BRÜEL & KJAER Microphone Power Supply
AKG Condenser Microphone with selectable directivity pattern
AKG Phantom-Power Supply
ACOUSTICAL Power Amplifier

CUSTOM Hexahedron Loudspeaker
NORSONIC Nor-840 Real Time Analyser
Cables

Two microphones (*D* and *E*) through their respective power supplies were plugged into the NORSONIC RTA840 (*A*) channel 1 and 2 inputs. The BRÜEL & KJAER microphone (*E*) has an omnidirectional pick up pattern whilst in the case of the AKG condenser microphone (*D*) its bi-directional "figure of eight" pick-up pattern was used to register sound coming from lateral direction.

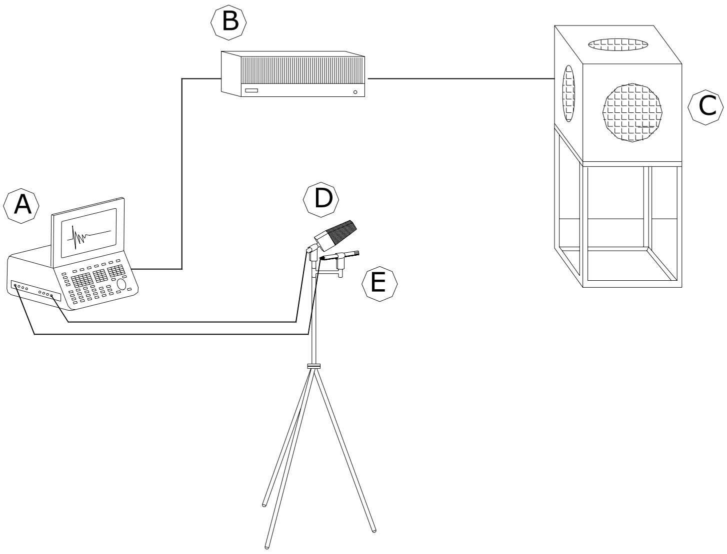


Figure 32: *Measurement Setup.*

An internal generated MLS-signal was sent from the RTA840 (*A*) noise output to the Acoustical power amplifier (*B*) input and, once amplified, played back through the CUSTOM loudspeaker (*C*). The loudspeaker was placed in one position on stage whereas the two microphones were clamped on the top of a height-and-angle adjustable tripod. This tripod was placed in every measured position. Figure 33 shows the loudspeaker's frequency response.

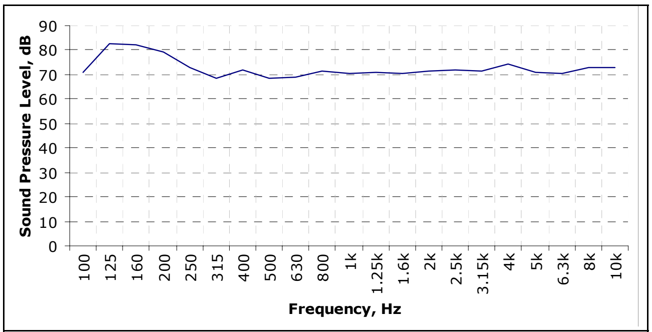


Figure 33: *Source's frequency response.*

Owing to the plan symmetry of all the halls it was decided to measure only one half of the audience area assuming a similar sound field behaviour on the other half. The measurement points in each hall are shown in figure 50.

All measurements were performed under the following conditions:

- All measurements were carried out without audience.
- The frequency range measured was from 100Hz to 10kHz.
- The loudspeaker used as sound source was placed in the centre of the stage at a distance of two meters from the frontal edge and raised at 1.25m above stage's floor except where indicated.
- The source radiation pattern was assumed omnidirectional.
- The tripod holding the microphones was adjusted in order to keep a height of 1 m from the floor in all the seats.
- Temperatures and relatives humidities were variable in all halls.

The following pages detail the five tested halls.

2.2. Teatro Nacional de Catalunya

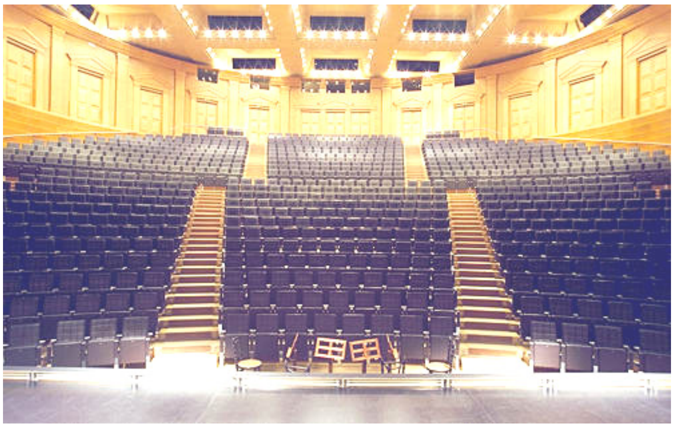


Figure 34: *Sala Gran, Teatro Nacional de Catalunya.*

The first measured hall was the SALA GRAN of the TEATRO NACIONAL DE CATALUNYA in Barcelona, Spain on February 8th and 9th, 2005. This is mainly a drama and comedy theatre although sometimes it features small opera plays since there is an orchestra pit. The hall has a semi-circular plan shape and a volume of 5,800m^3. Plan and longitudinal section are shown in figure 35.

For measurements, the fire curtain up mode was chosen since that represents a real performance situation. There were only a few pieces of furniture on stage with the air conditioning system fully functioning. Temperature was 21ºC and the relative humidity was around the 62%.

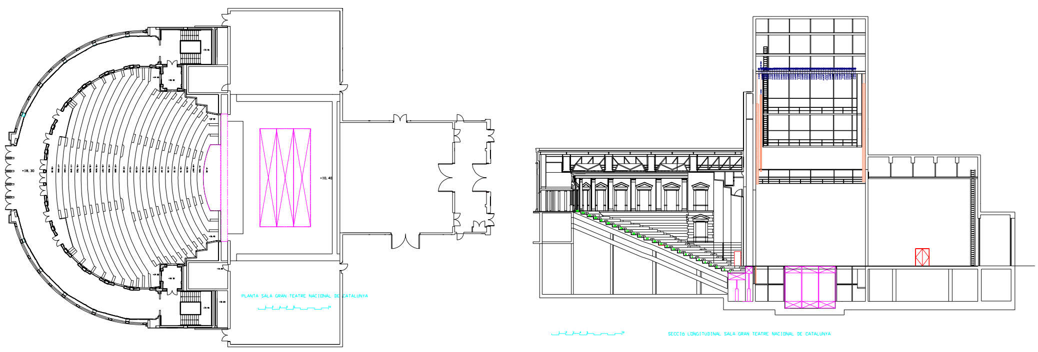


Figure 35: *Teatro Nacional de Catalunya's plan and longitudinal sections.*

The total audience is divided into 6 sections with 885 seats. From that number 114 were selected as measuring points on the left half of the audience area, which is equivalent to 13% of the total capacity.

Walls, ceiling and floors are made of wood and the seats are leather covered.

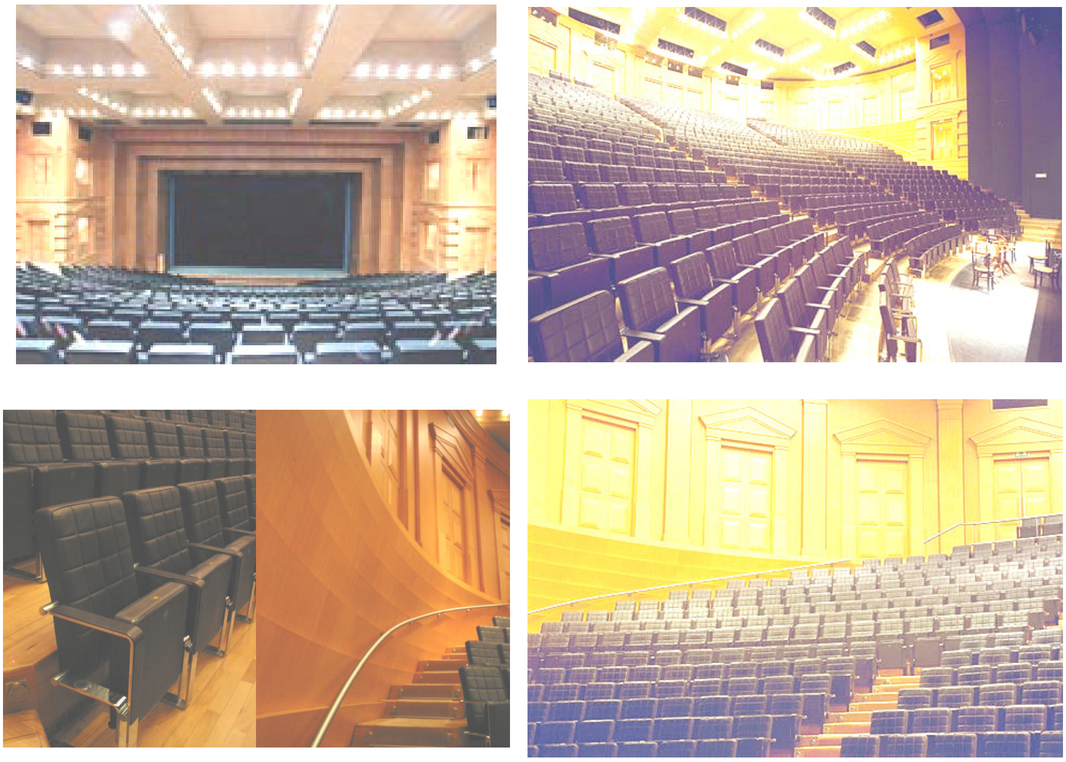


Figure 36: *Sala Gran, Teatro Nacional de Catalunya.*

The theatre does not have a variable acoustic system to adapt its acoustics to the different uses and the only movable surfaces that may modify the room are those open spaces for lighting systems located on the ceiling (see figures above) and the orchestra pit, which in this case was closed for measurement purposes.

2.3. Laguna de Duero Auditorium

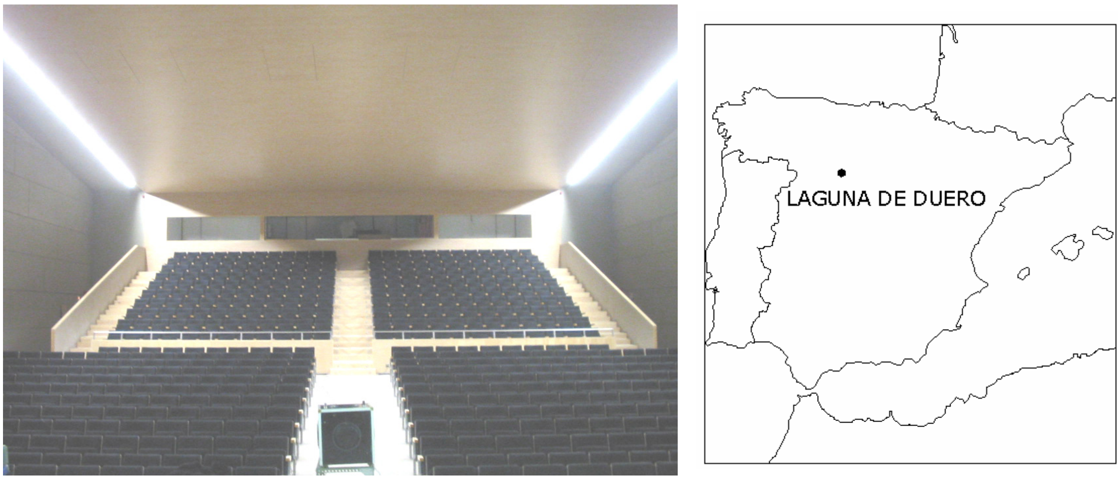


Figure 37: *Laguna de Duero Auditorium.*

On February 16th, 2005 was measured the LAGUNA DE DUERO AUDITORIUM as a part of the final measures done by GARCIA-BBM before inauguration. Located in Valladolid, Spain, this auditorium will be used for speech exclusively although it cannot be disregarded the possibility of any musical performance. This auditorium has a rectangular "shoebox" plan shape and a volume of 3,700m^3. Plan and longitudinal section are shown in figure 38.

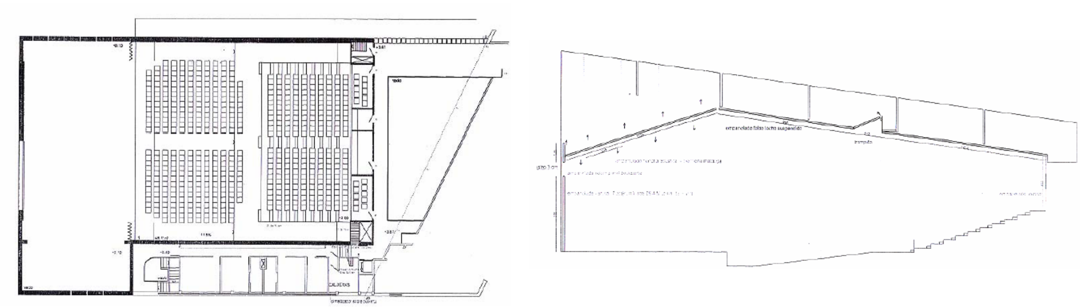


Figure 38: *Laguna de Duero plan and longitudinal sections.*

Since it was in pre-inauguration stage, the recommended absorbing material and curtains in the stage were not installed Practically the only absorption was provided by the seats upholstering. The only objects on stage during measurements were the instrumentation used for measurements. Temperature was 25ºC and the relative humidity was around the 50%.

The total audience is divided into four sections: two closer to the stage and two beyond the transversal corridor The total number of seats is 562 of which 54 were measured on the left part of the audience area, approximately 10% of the total capacity.

The walls and first part of the audience area were made of concrete although planned a carpet under the first sector of the audience area was to be installed a few days before inauguration. The floor on the second audience area and the completely flat ceiling were made of wood.

At that moment, there was no sign of the implementation of variable acoustic systems and the only movable surfaces that might modify the room were lighting system openings located on the ceiling (see figure below).

Figure 39: *Laguna de Duero Auditorium.*

Owing to its large volume and reverberant conditions, (due mostly to the materials used and surface's finishes) it was necessary to use electroacoustic equipment to obtain good speech intelligibility.

2.4. Palau De Congressos de Peñíscola

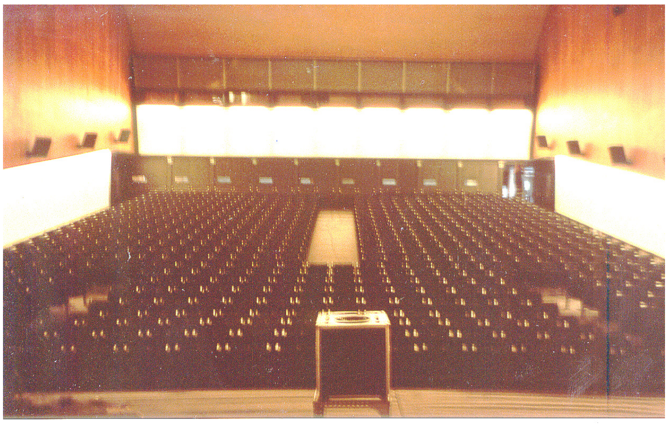


Figure 40: *Palau de Congressos de Peñíscola.*

The PALAU DE CONGRESSOS DE PEÑISCOLA is located in Castellón, Spain and the measurements in this auditorium were carried out on March 24th, 2005.

This is a multipurpose hall used as a conference centre, cinema and eventually concerts: The measurements were obtained in the cinema mode, with the screen deployed. It has a rectangular plan and, unlike Laguna De Duero, the audience area and the stage share the same height: a waving ceiling made of concrete as can be observed on the longitudinal section of figure 41. With a rectangular plan and a middle height of 9.5meters, this auditorium has a volume of 7,600m^3.

The total audience area is divided into five sections: two in raised position at the back and divided by a central corridor and three frontal sections divided by two corridors next to stage. The total capacity is 702, of which 78 seats were measured on the right part of the audience area, approximately 11% of the capacity.

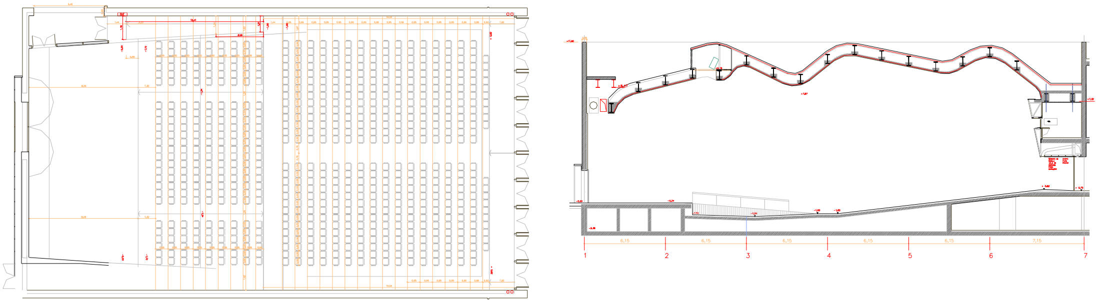


Figure 41: *Palau de Congressos de Peñíscola, plan and longitudinal section.*

As stated above, during measurements the electroacoustic equipment used in the cinema mode was also tested: lateral loudspeakers and three full-range sets placed behind a plastic perforated projection screen. Besides those and the hexahedron loudspeaker, there was no more equipment or furniture on stage. Temperature was 25ºC and the relative humidity was around 70%.

Figure 42: *Palau de Congressos de Peñíscola.*

Absorption in the hall is provided by adjustable carpet screens covering the back wall on stage, carpet on the floor and seat upholstering. Walls are made of wood and the external corridors have glass plates, which divide corridors from frontal audience areas. The intricate concrete ceiling provides some overhead reflections although it is not a low frequency absorber.

2.5. Teatro Serrano Small Hall, Gandía

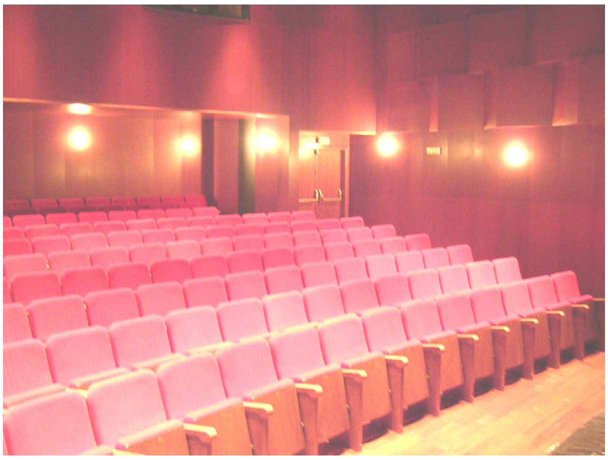


Figure 43: *Teatro Serrano' Small Hall, Gandía*

The TEATRO SERRANO Small Hall in Gandía was measured on April 6th, 2005 as part of the tests before its opening. The main uses of this hall will be: chamber music, conferences, cinema, etc.

The hall has a -fan-shape plan and an approximated volume of 760m^3. Plan and longitudinal section are shown in figure 44.

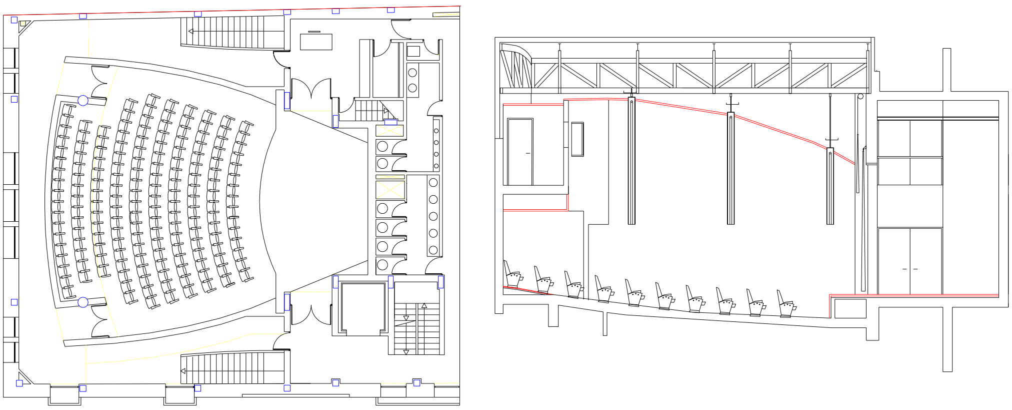


Figure 44: *Teatro Serrano Small Hall plan and longitudinal sections.*

Besides the hexahedron loudspeaker, there were pieces of furniture on stage and the air conditioning system was not functioning. Temperature was 25ºC and the relative humidity was around 70%.

The audience is on the same plane with 171 seats. From that number 15 seats were selected on the left half of the audience area as measuring points, equivalent to 9% of

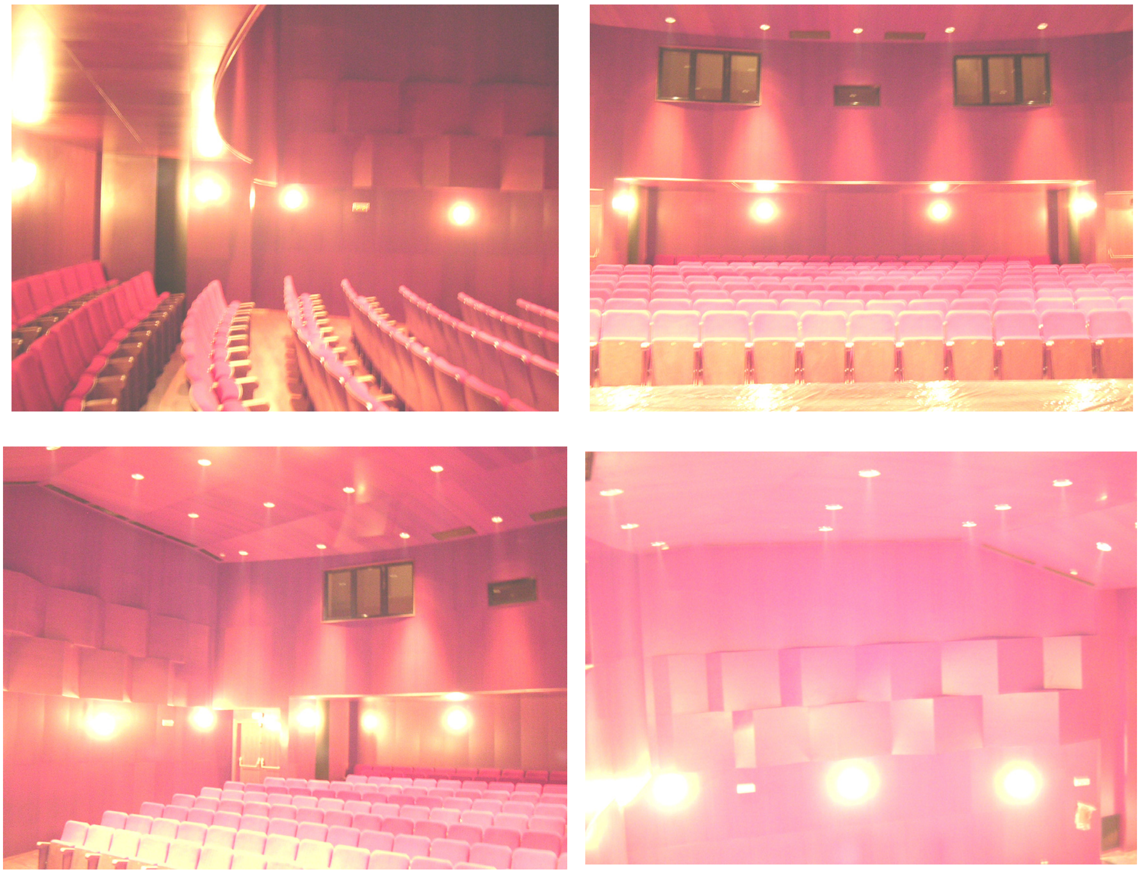


Figure 45: *Teatro Serrano Small Hall, Gandía*

the total capacity.

Walls, ceiling and floors are made of wood and the seat upholstering is made of fabric. In this hall a variable acoustic system is not contemplated and lateral reflections are enhanced by diffusers placed on the lateral walls, see figure 45.

2.6. Teatro Serrano Main Hall

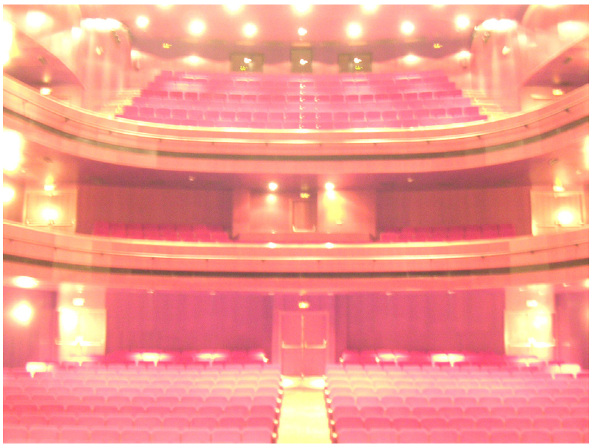


Figure 46: *Teatro Serrano's Main Hall, Gandía*

The TEATRO SERRANO's Main Hall in Gandía was measured on April 6th, 2005 before its inauguration. Plays and Opera in small format together with Symphonic Concerts (with the appropriate acoustic shell) among others will be the main uses in this hall.

The theatre comprises three well defined volumes: the Stage volume with 3,846 m^3, the Orchestra Pit with 203m^3 and the audience volume of 3,250m^3 with a horse-shoe plan shape as can be observed in figure 48.

The audience is divided into three sections: stalls, amphitheatre and upper balconies, and the total seating capacity is 611. The measuring positions were 57, which means just over 10% of the total audience capacity. Such positions were distributed evenly over amphitheatre, balconies and stalls audience areas.

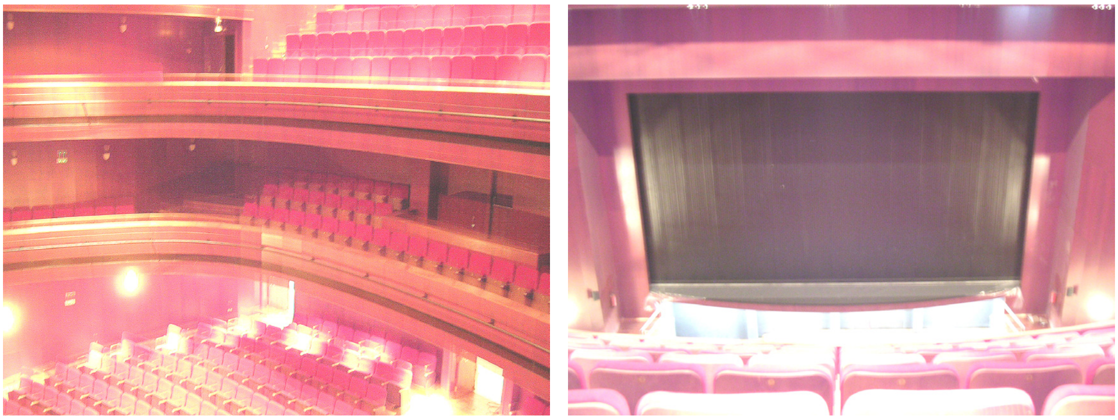


Figure 47: *Teatro Serrano Main Hall, Gandía.*

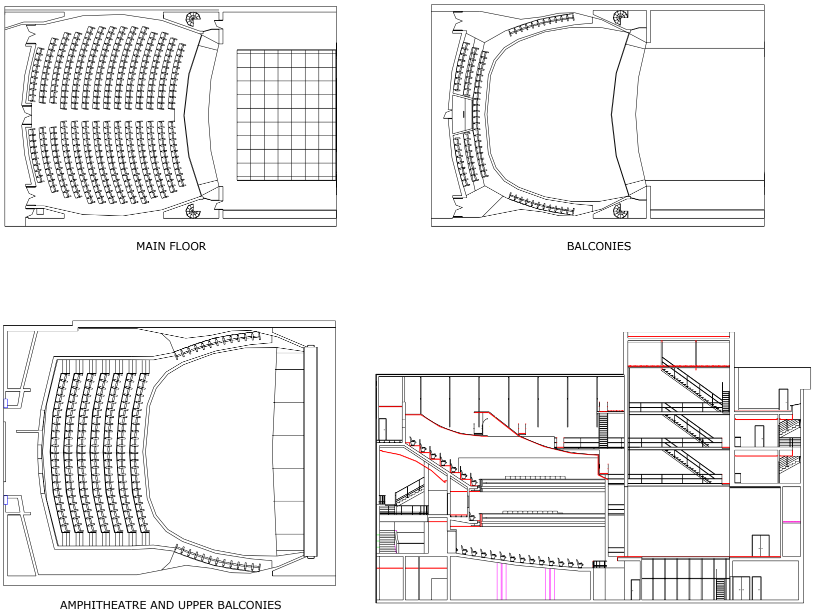


Figure 48*: Teatro Serrano Main Hall Plans and longitudinal sections*

During measurements, fire curtain was down, orchestra pit was empty and the sound source had to be placed on the central corridor just before the frontal handrail as it can be seen in figure 49.

Air conditioning systems were off and the temperature was 24ºC with a relative humidity around the 70%. Regarding materials, walls, ceiling and floors are made of wood and the seat upholstering is made of fabric. At moment of measurements, inside the theatre there was no implementation of variable acoustics.

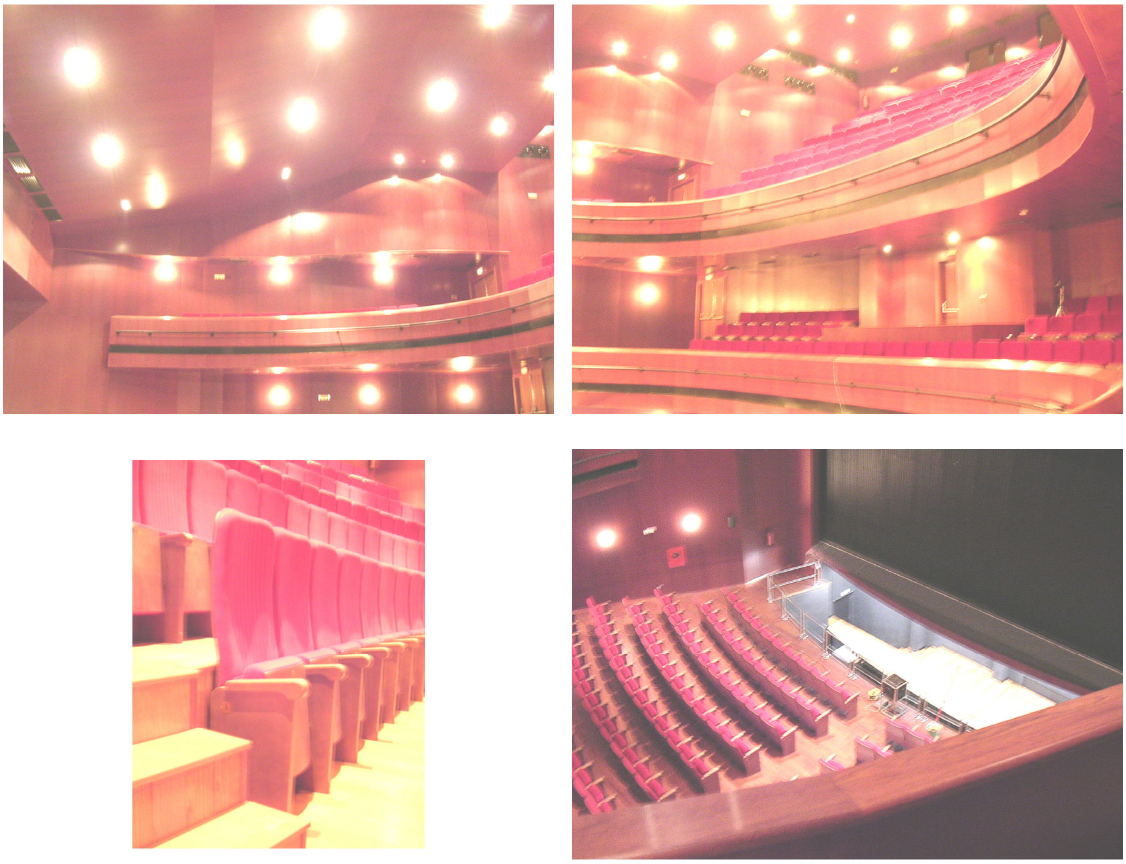


Figure 49: *Teatro Serrano Main Hall, Gandía*

Figure 50 shows the measuring positions in all the halls.

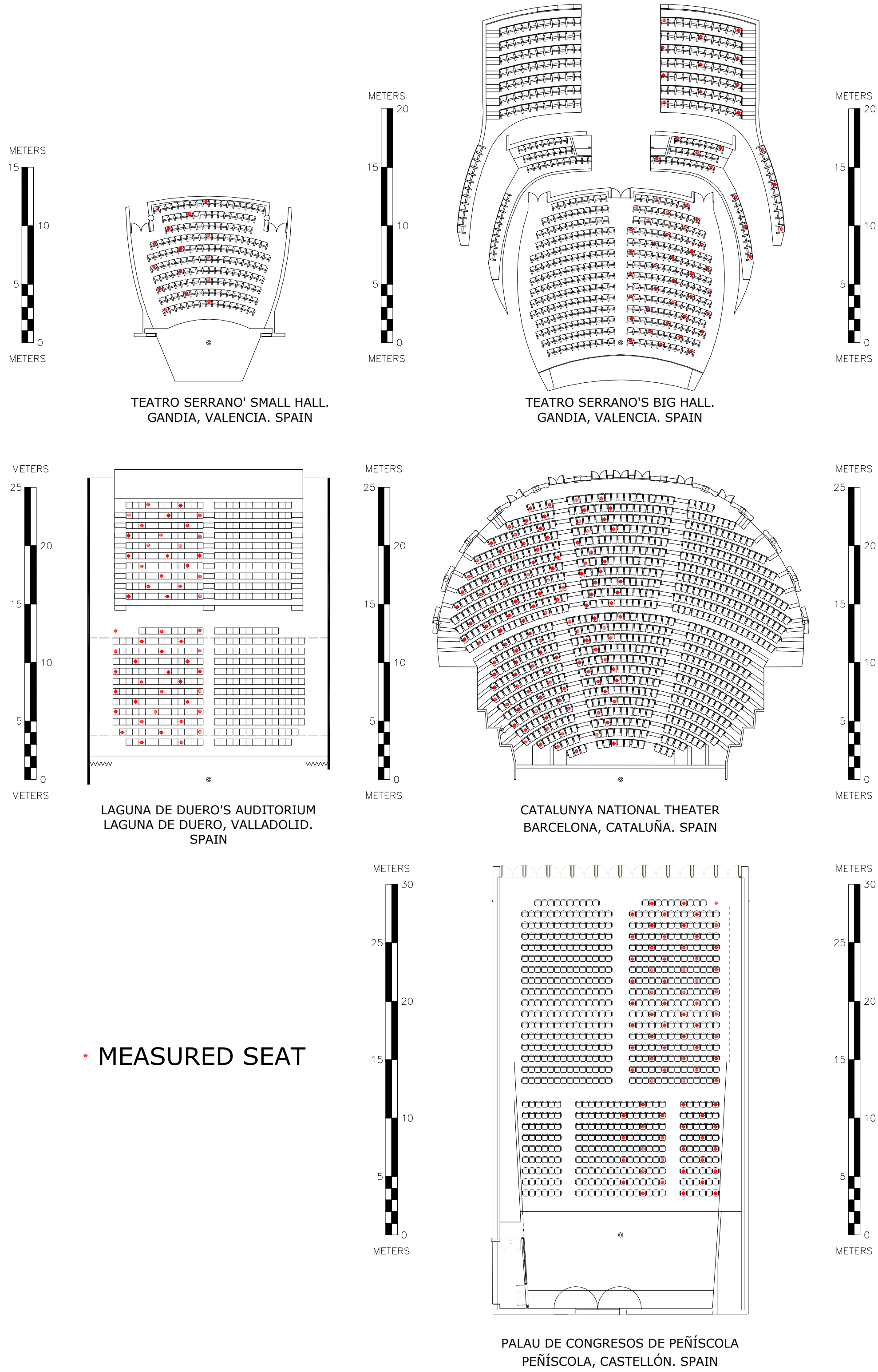


Figure 50: *Measurement Positions.*

3.Results & Conclusions

All the data obtained from measurements described in chapter 2 were analysed by means of NORSONICS Nor-Rac Software (see Appendix A). This software only requires the frequency response of the source, the calibration between omnidirectional and 'figure of eight' microphones and the .SDF files obtained from the 840RTA in each measured position to compute acoustic parameters. There are two types of .SDF files: the originated from one/two channels impulse responses and those obtained by the Schroeder Backward Integration inside the RTA840. The latter provides the EDT and RT results (more details in Appendix C).

Of all the acoustic parameters described in chapter 1, only the following will be analysed here:

- G_{mid} and G_{mid} Strength Parameters
- EDT and RT for the 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz and 8kHz frequency bands.
- Bass ratio Parameter
- Brightness at 2kHz and 4kHz Parameters
- C_{50}, "Speech Average", Clarity Parameter
- D_{50}, Definition Parameter
- C_{80}, "Music Average", Clarity Parameter
- t_s, Centre Time Parameter
- $L_{ef\,(E4)}$, Lateral Energy Fraction Parameter

With all the parameter values measured in each position, the following task was to illustrate them in a clear graphical way in order to be able of see variations over the whole audience areas. To do so, graphical extrapolation software (Arc View Gis, 3.1) was used and assuming symmetrical sound field behaviour the measured, half of the audience area was mirrored.

The following pages show the graphical results with an explanation for the values obtained. Results are classified by acoustic parameter for all the measured halls.

Nevertheless, it must be taken into account the fact that no measurement process is, in any way, safe from possible errors. Source directivity, equipment failure, environmental influences like those described in chapter 1 or human influences on data manipulation may affect in whatever manner the sound field description

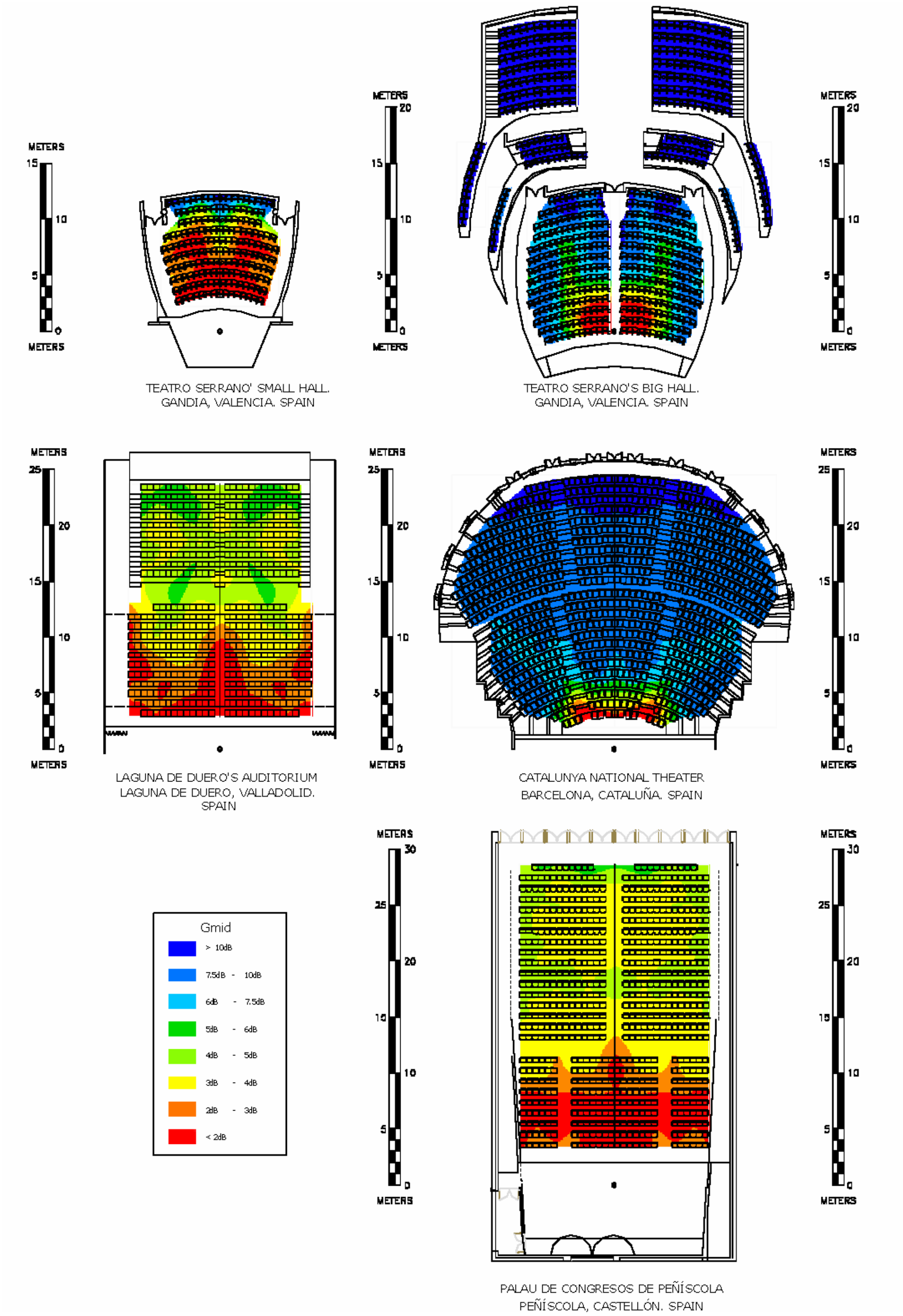


Figure 51: *Results: Acoustic Parameter* G_{mid} .

3.1. G_{mid} and G_{125} Strength Parameters

The variability of the G_{mid} parameter is shown in figure 51. The first obvious conclusions are: a) there is no relationship between cubic volume, and therefore reverberation time, and G_{mid}, b) source directivity plays a major role in this parameter.

Inside Teatro Serrano Small Hall there is not a notorious difference in the first five meters from the source but changes happen rapidly in the back of the hall especially in the last two rows below the projection room.

In the Teatro Nacional de Catalunya, things are a little bit different; only a few front seats may notice a decrease on G_{mid} whereas most of the hall experiments differences in the order of 10dB.

G_{mid} spread out is defined by source's directivity. Inside Laguna de Duero Auditorium high reflecting surfaces help to keep a good level in almost all the front audience area (10 meters long) whilst the rest of the hall never experiments differences bigger than 6dB.

Something similar occurs in Peñiscola where highly reflecting surfaces (as those glass plates dividing the first audience areas from external corridors) keep a good sound level distribution. Once more, the sound level differences are not bigger than 5dB with just a few seats in the back with differences between 5dB and 6dB. It can be said that Laguna's and Peñíscola show transitions at a distance of 15 meters from the source for this parameter.

Teatro Serrano Main Hall is more similar to Catalunya than the rest; by placing the source in practically the first row, it can be observed its directivity pattern in detail. However, a difference from 2dB to 6dB happens in a narrow area in the form of some dark spots (differences between 8dB and 10dB) over the main floor.

Situations on higher levels are even more critical; neither the first line of balconies nor amphitheatre shows differences below the 10dB.

According to the theory introduced in Chapter 1, a common variability range for this parameter is from 1.0dB to 3.0dB every time the distance is doubled but it is only in the two auditoriums measured here that that is fulfilled happen. Unlike the three theatre examples, both auditoriums have not abrupt transitions, diffusers elements or intricate geometrical plan shape. Flat highly reflecting walls may have some relation with the G_{mid} behaviour.

High levels of the G_{125} strength parameter (figure 52) are desirable to enhance music performances. Once more, the variability of a strength parameter is highly influenced by the source's radiation pattern. Although it can be said that there is no relationship between room volume- and, of the five halls, Catalunya's big cubic volume seems to offer the greatest absorption at low frequencies. Furthermore, Serrano's Small Hall shows a higher G_{125} .

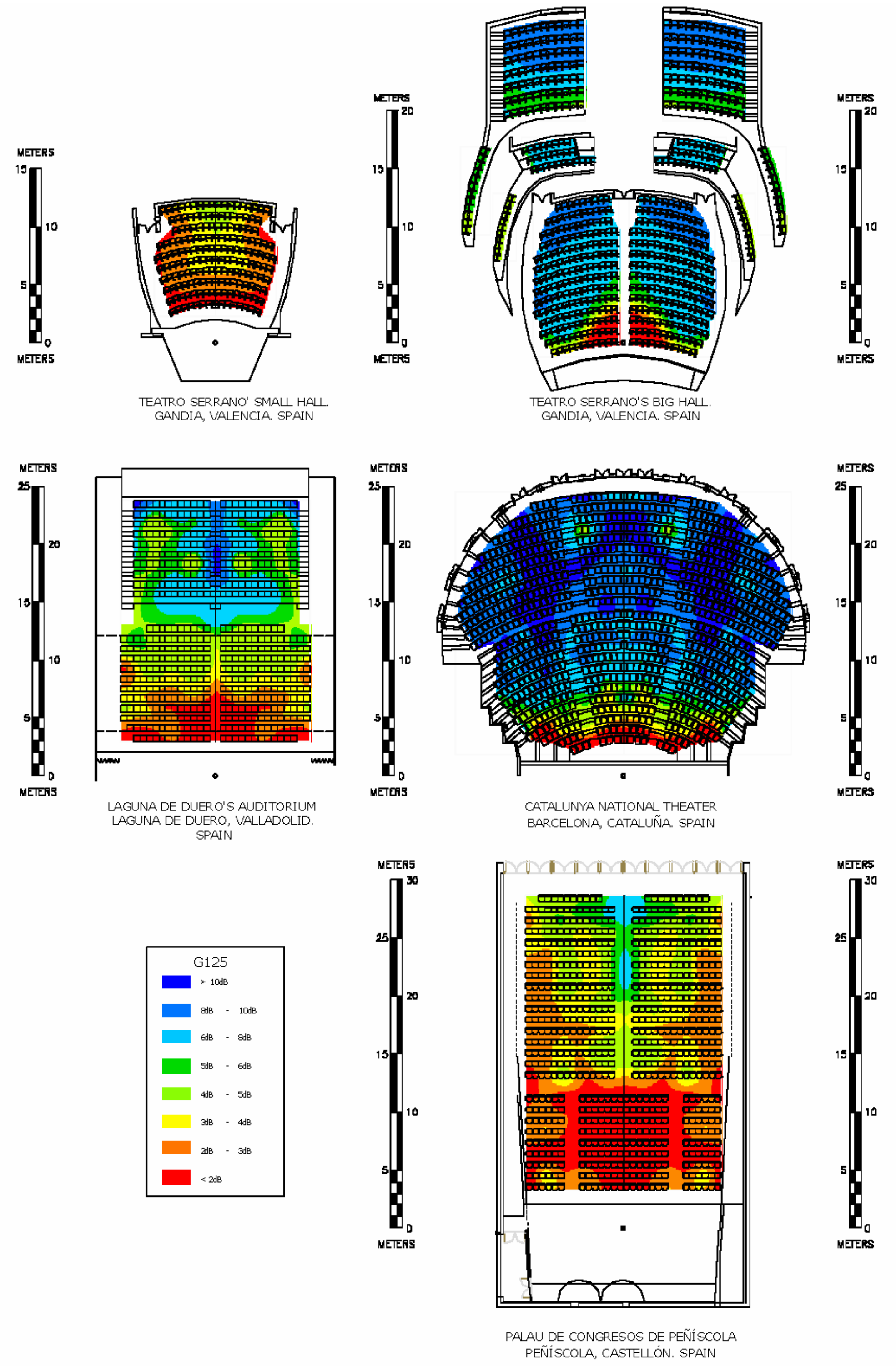


Figure 52: *Results: Acoustic Parameter* G_{125}.

The two auditoriums show higher (optimal) G_{125} values than theatres although they present larger volumes than the theatres.

Wall assemblies provide absorption at low frequencies and grazing incidence as it can be verified in Serrano's Main Hall. In there, lateral balconies and amphitheatre's first two rows show a better G_{125} value than the main floor's area at five meters from the source and the frontal balconies, all in blue.

Since grazing incidence involves absorption by audience, (or seat upholstering on this measurements) it explains why around 12 meters from the source the levels decrease abruptly inside auditoriums. Such distance corresponds to the end of the front of audience areas. Level distribution could depend on source's radiation pattern, though.

For the small hall, an optimal G_{125} level is found in the way of a red edge surrounding the whole audience area.

3.2. EDT and RT

In figures 53 to 59 the Early Decay Time values obtained inside the five halls at 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz and 8kHz octave-band frequencies are presented graphically. The 8kHz octave-band appears here only for illustrative purposes.

As it was introduced in Chapter 1, the EDT is compound from the first group of reflections and therefore is more sensitive to the hall's geometries than Reverberation Time.

EDT variation with frequency is quite different for the cases studied. Theatres experiment an EDT increase from the low frequency region to mid-frequencies and then a fast decay as frequency increases. On the other hand, measured auditoriums show a smooth time decrease from their highly reverberant low frequencies to small high frequency EDTs. In the mid-frequency region theatres, show higher EDT values than auditoriums.

In order to find a better explanation to the way that EDT varies over audience areas, it was assumed that microphones picked-up the early reverberation time corresponding just to the cubic volume in front of them. By doing this, it is possible to do an interpretation of the colour distributions.

In halls with two significantly different volumes between the stage enclosure and the rest of the hall (in Serrano's Main Hall the source was placed before the orchestra pit), the coupled rooms phenomena seems to take effect, where the existence of two contiguous enclosures of different cubic volumes yields to different reverberation times depending on measurement position.

If the total volume of a hall is the sum of a series of smaller different volumes delimited by changes in its plan or longitudinal sections, the EDT values will change according to those transitions.

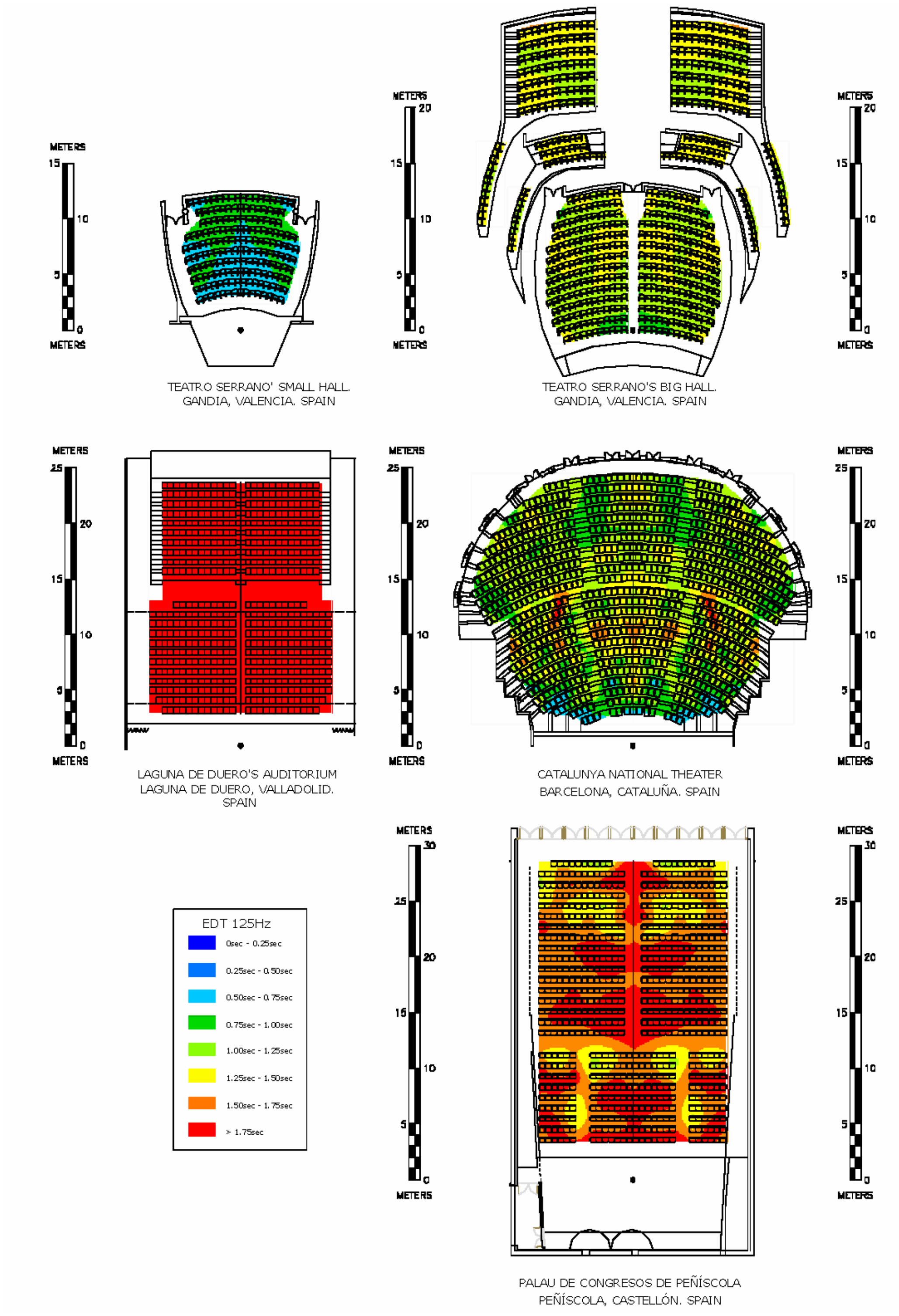


Figure 53: *Results: Acoustic Parameter* $EDT_{125\,Hz}$.

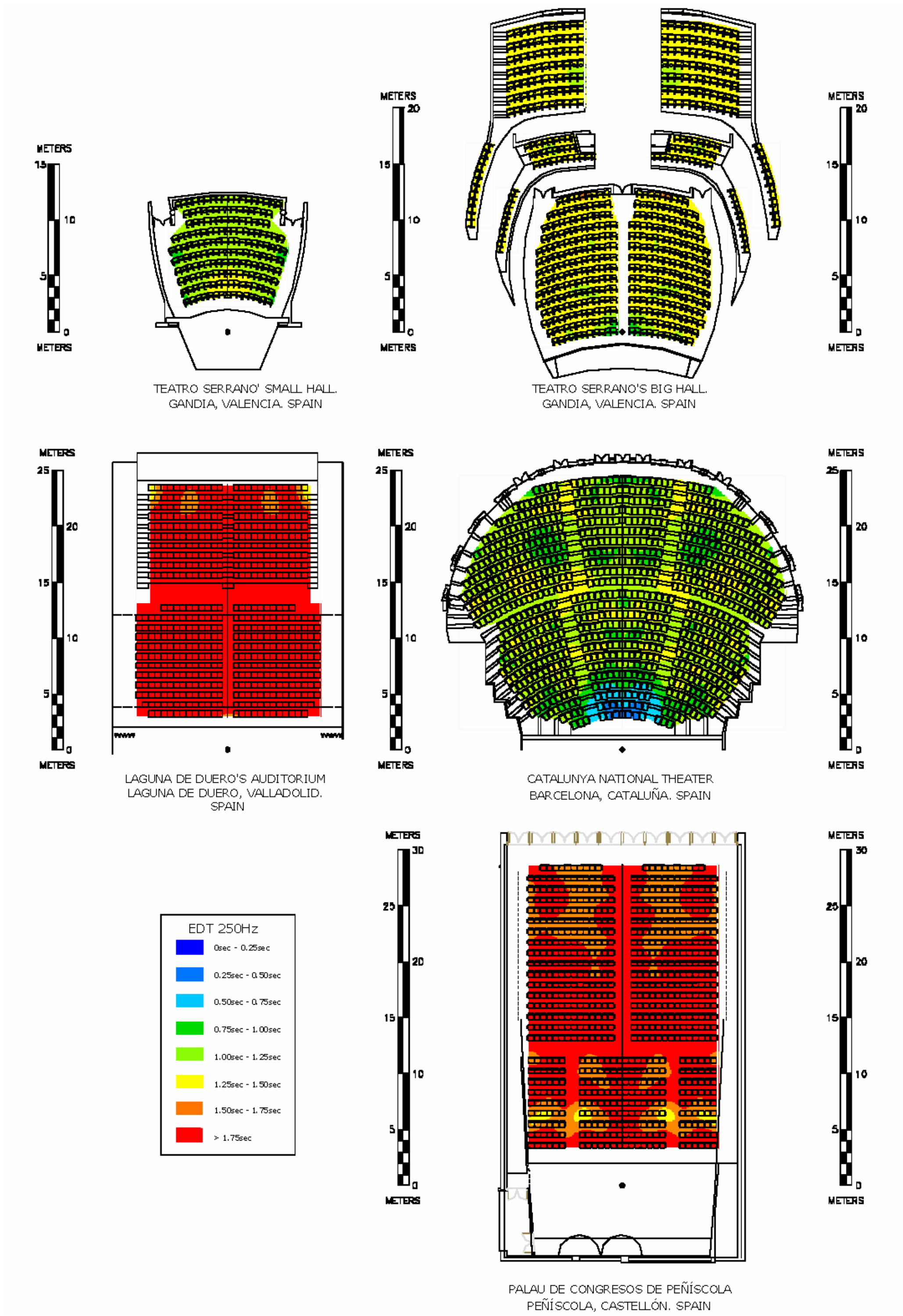


Figure 54: *Results: Acoustic Parameter* $EDT_{250\,Hz}$.

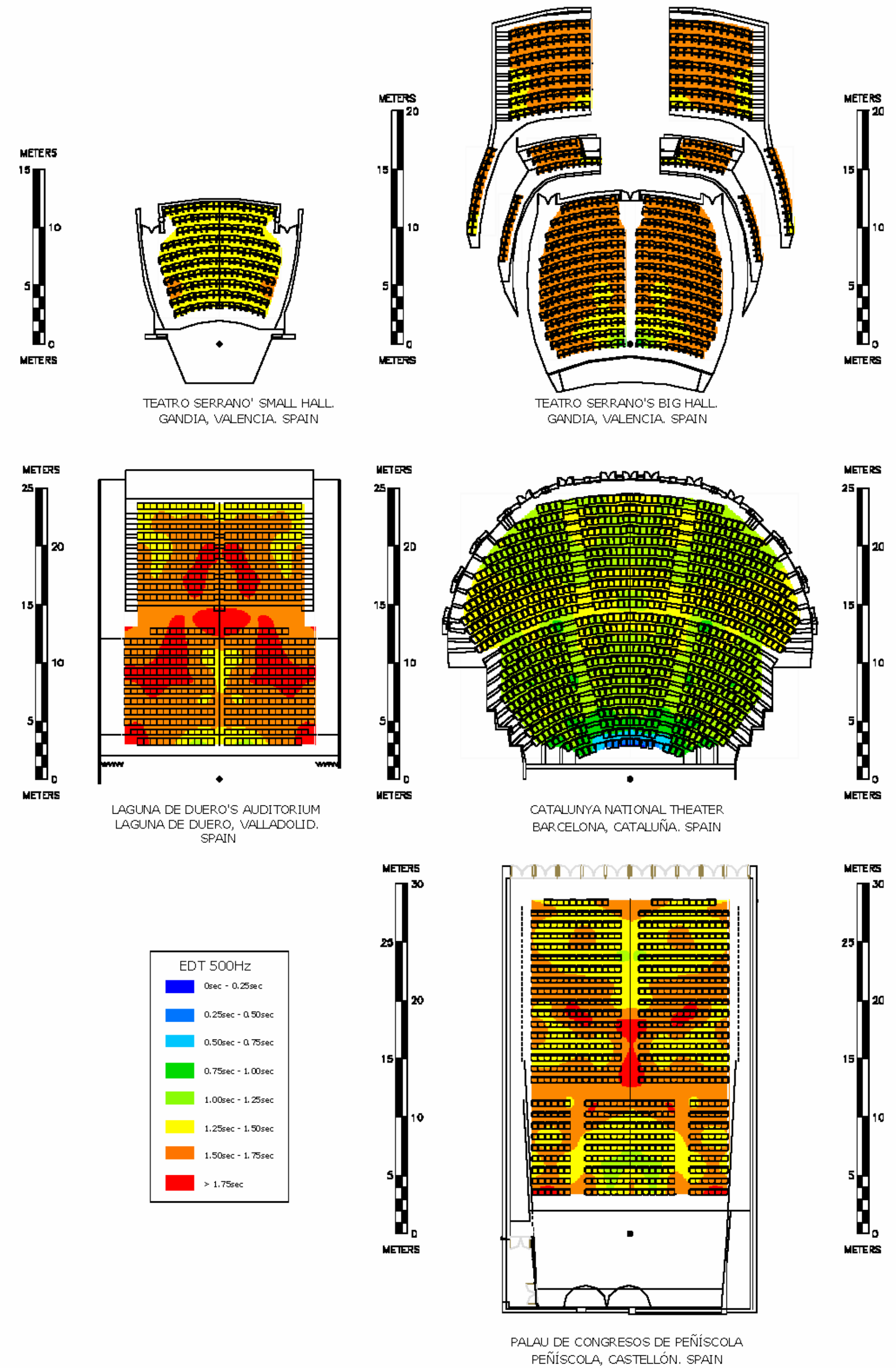


***Figure 55**: Results: Acoustic Parameter* $EDT_{500\,Hz}$.

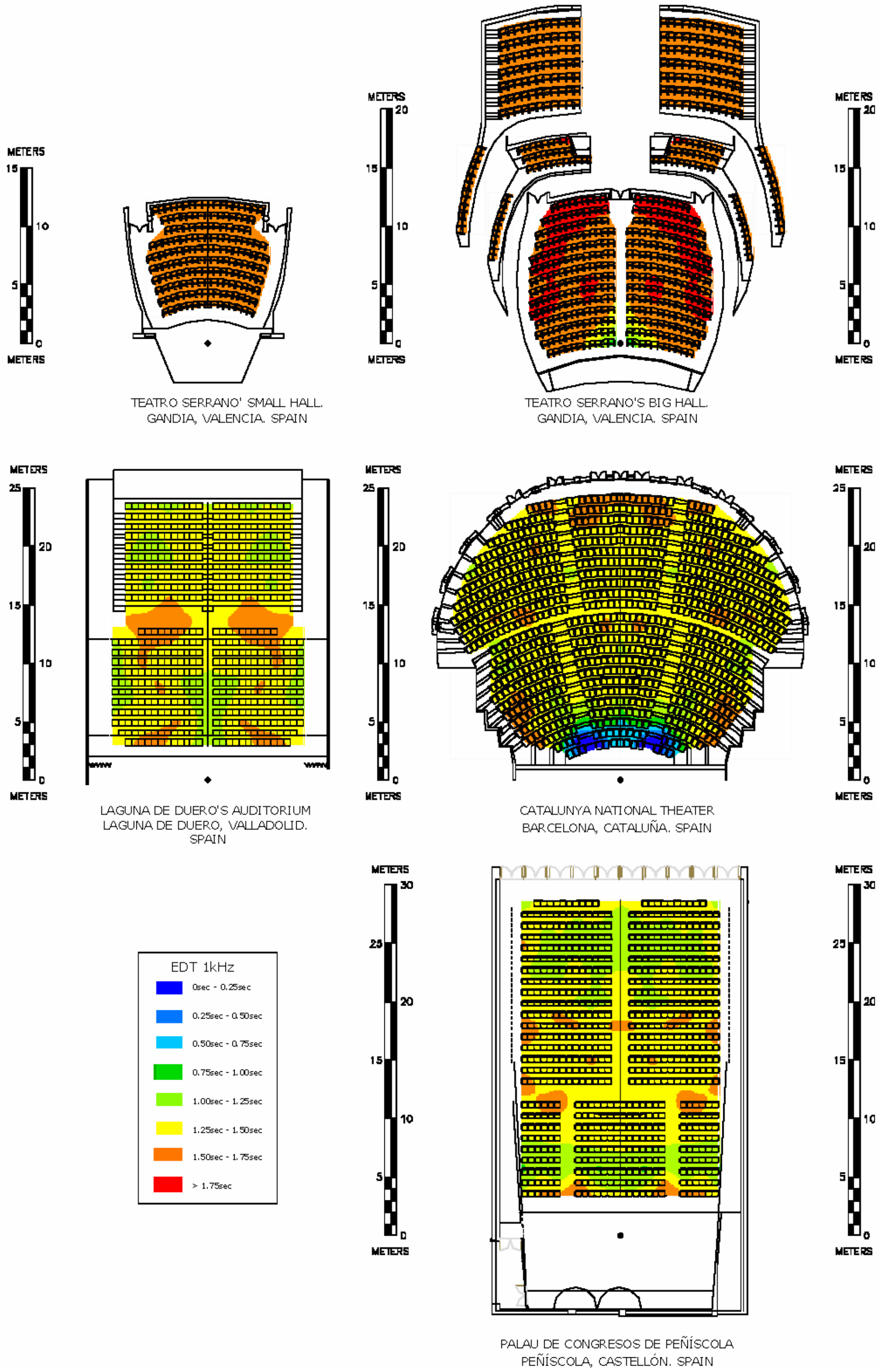


Figure 56: *Results: Acoustic Parameter* EDT_{1kHz}.

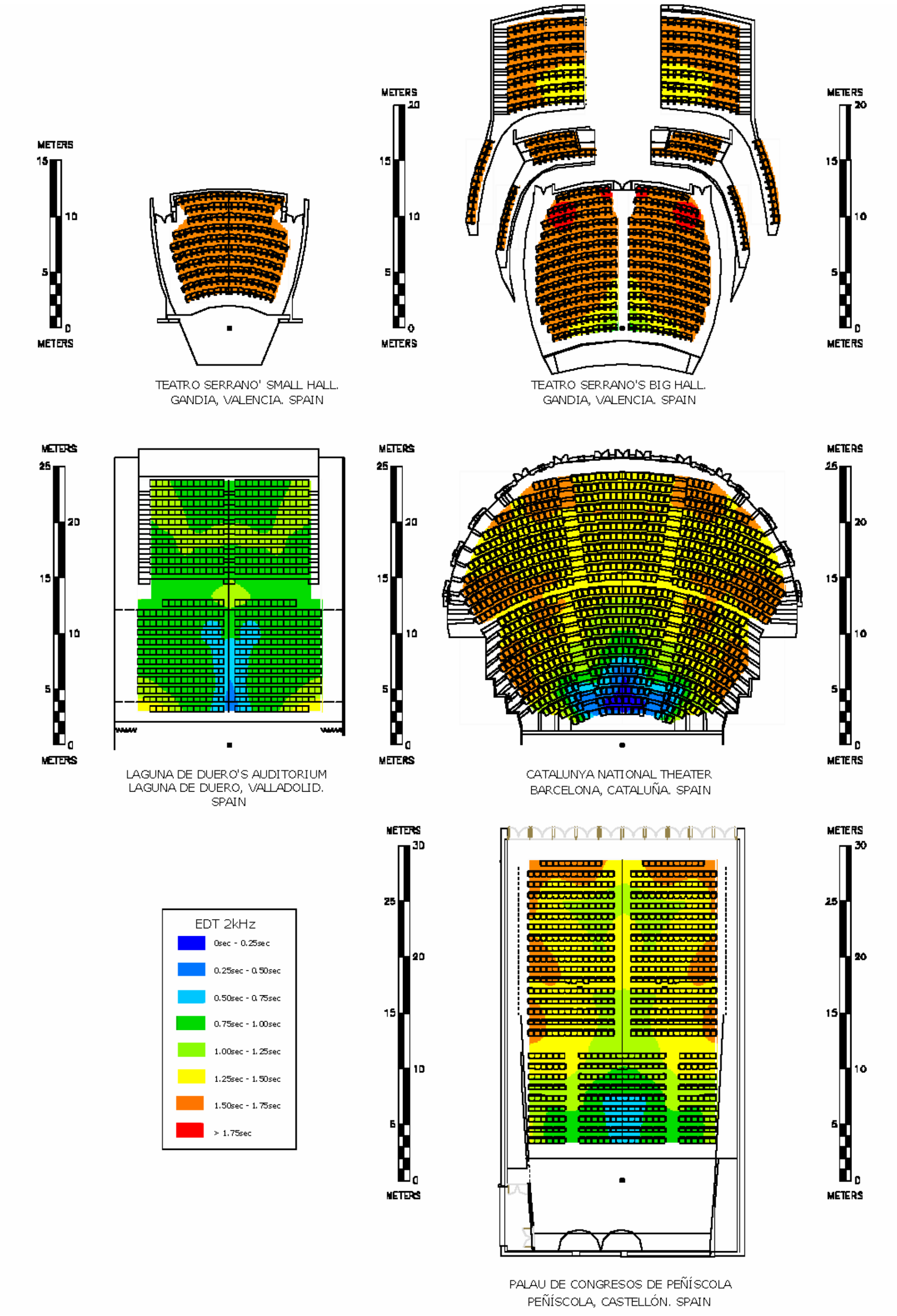


Figure 57: *Results: Acoustic Parameter* EDT_{2kHz}

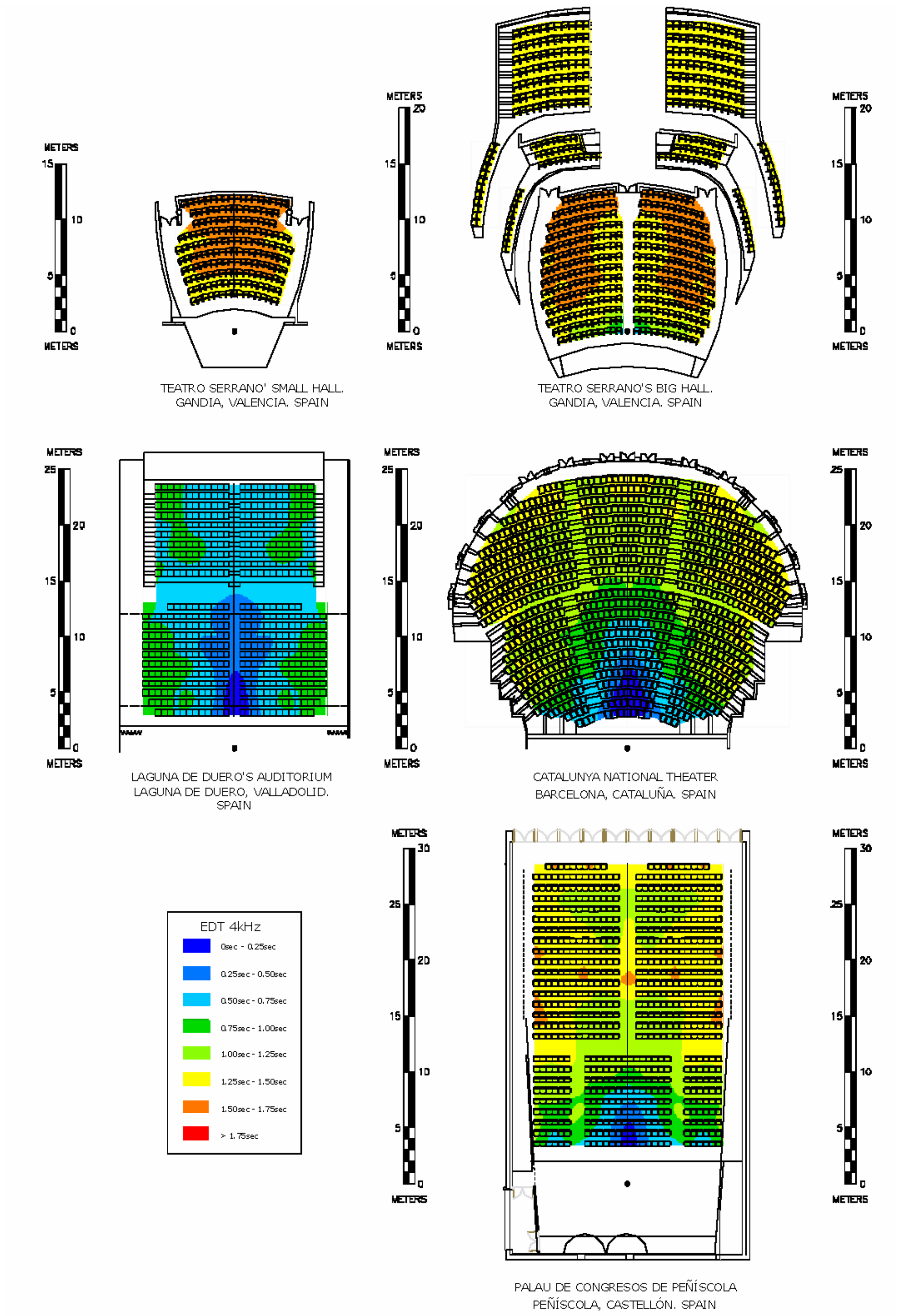


Figure 58: *Results: Acoustic Parameter* EDT_{4kHz}

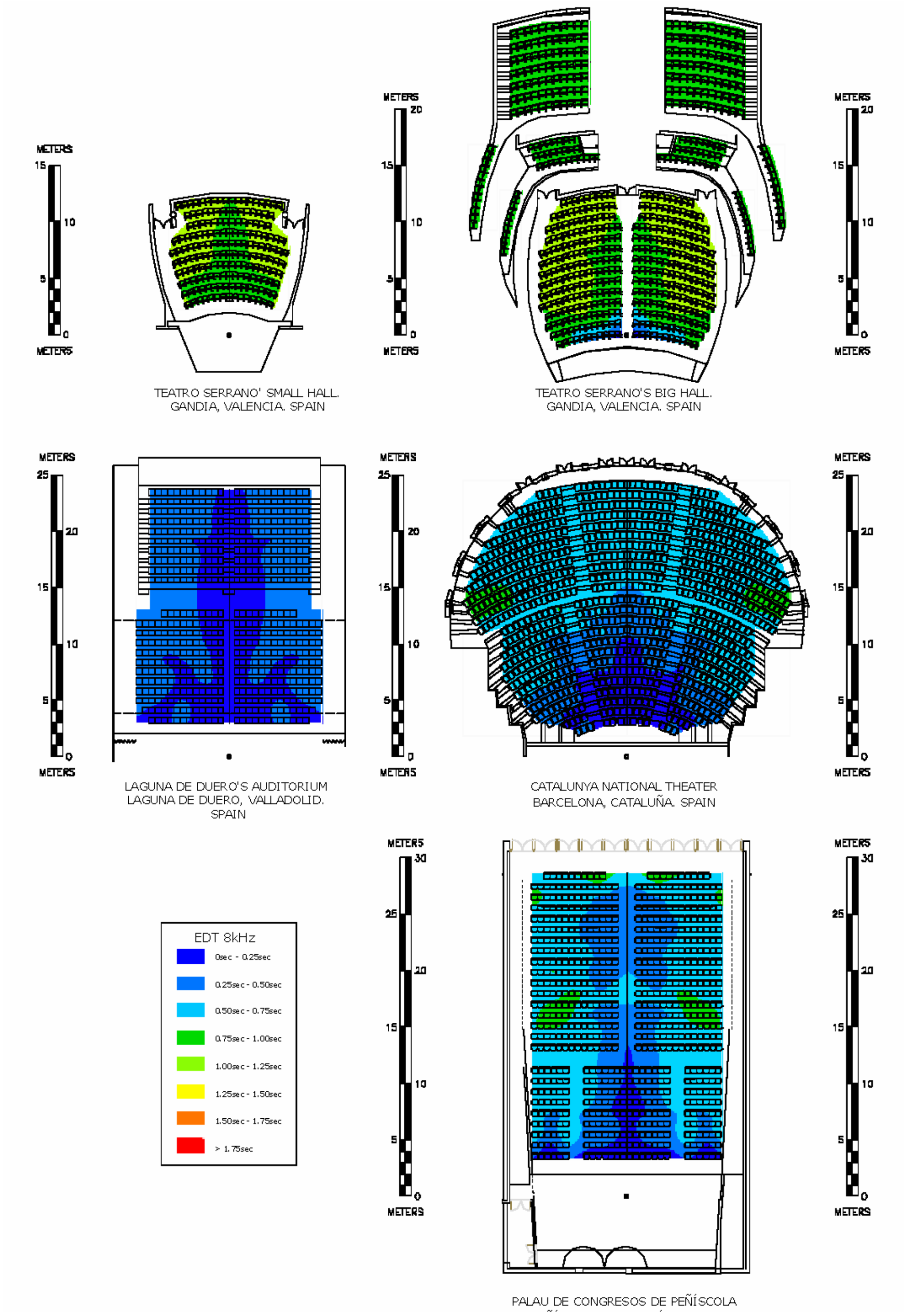


Figure 59: *Results: Acoustic Parameter* EDT_{8kHz}

That could explain why in the front rows (in the first five meters away from the source) of Teatro Serrano Main Hall and Catalunya's the lowest EDT values appear, just before volume changes.

The same could apply in the low frequency region in Teatro Serrano Small Hall, in the wet area in the form of a "yellow-belt" at 500Hz in Catalunya, around the mid frequency region in some surfaces under the first balcony line in Teatro Serrrano Main Hall and in those green areas at 1kHz in Peñíscola.

Figures 60 to 66 present the corresponding reverberation times in the same octave-band frequencies analysed above.

The first conclusion is that auditoriums show a similar RT behaviour to their EDT since there are high low frequency reverberation times that gradually decrease as frequency increases.

Position of the source influences Reverberation Time spread especially at high frequencies. Such focalisations of low RT values start around the 1kHz in Teatro Nacional de Catalunya and Teatro Serrano Main Hall. In the latter, well-defined areas appear under balconies at 2kHz and, at 4kHz, progressive lower RT values spread out over more extensive audience areas. At mid-frequencies (500Hz and 1kHz), Laguna Auditorium, Catalunya and Teatro Serrano Small Hall show the more stable and evenly distribution.

In a height divided Hall, as Teatro Serrano Main Hall, the following features are shown: Reverberation times are different in main floor, first balcony line and amphitheatre levels.

At 125Hz, RT gradually increases from values around the 0.4sec in front of the source (probably orchestra pit's RT) up to numbers around the 1.4sec on the last amphitheatre rows. This is a difference of about 1sec from one position to another.

At 250Hz, the Main Hall by itself varies from the 0.5sec to the 1.3sec (a 0.8sec difference) whereas at the frontal balconies there are fluctuations from 0.8sec to 1.3sec on the last rows.

At 500 Hz, the main floor RT values are in the 1.3sec to 1.6sec range, which means a 0.4sec difference with higher values concentrating under balconies. Upper seating levels show a uniform RT value below the numbers found down the hall.

At 1kHz, a highly reverberant zone surrounds a narrow area of approximately 7 or 8 meters length in front of the source. Seats in lateral balconies next to proscenium and amphitheatre seats show values around the 1.4sec. Middle seats in frontal balcony have bigger RT than those close to exit corridors at the same level.

Values from 0.8sec to 1.5sec (a 0.7sec difference) can be found around the 2kHz and 4kHz. At 8kHz, values from 0.3sec to 0.9sec are founding almost all the halls. At the same frequency, auditoriums experiment lower values thorough their middle axis than those existing in areas close to walls. In the case of the smallest hall, there is variability concentrated in seats close to the stage and below the projection room. Here EDT and RT values are with the same time ranges thorough all octave band frequencies.

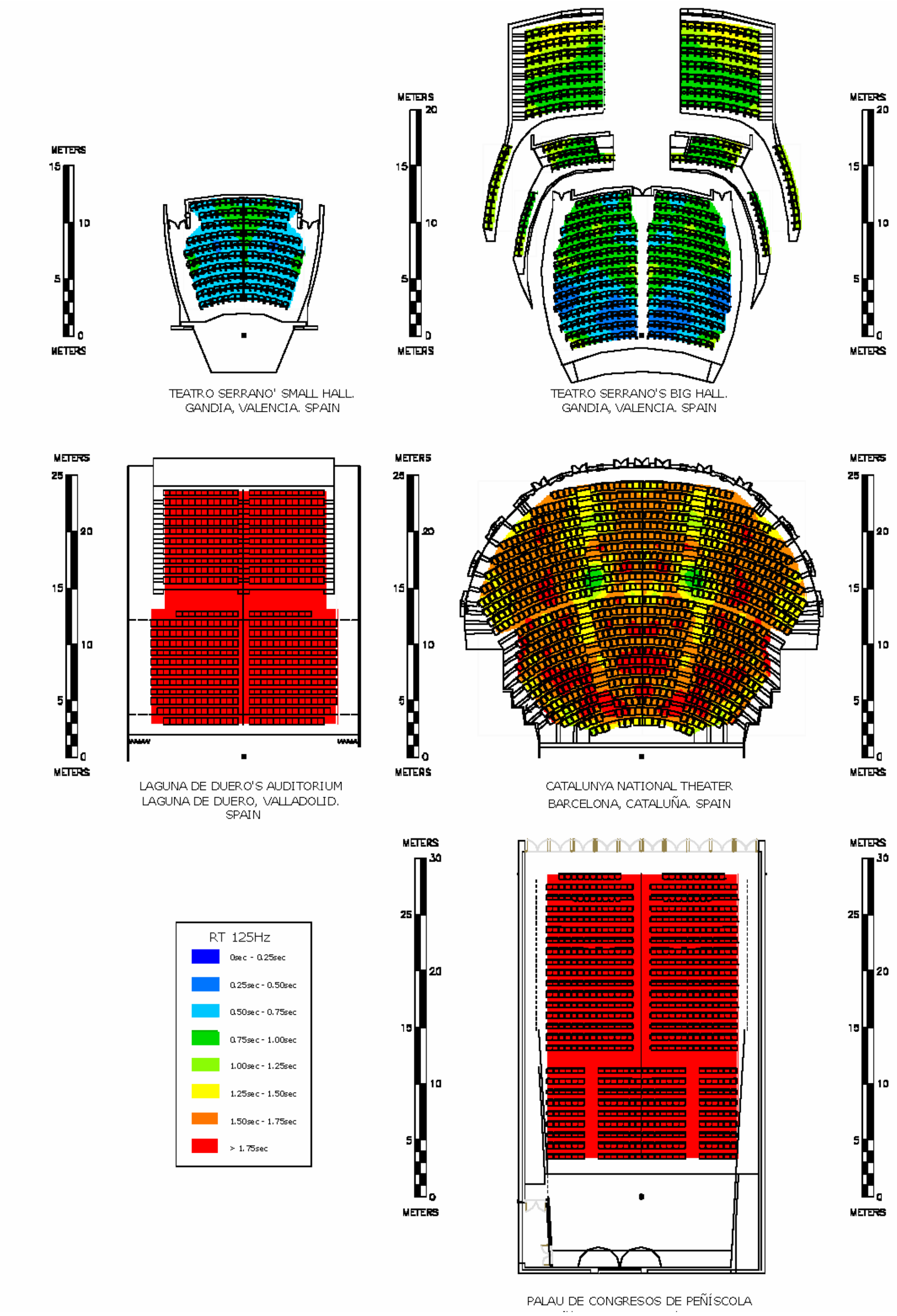


Figure 60: *Results: Acoustic Parameter* $RT_{125\,Hz}$.

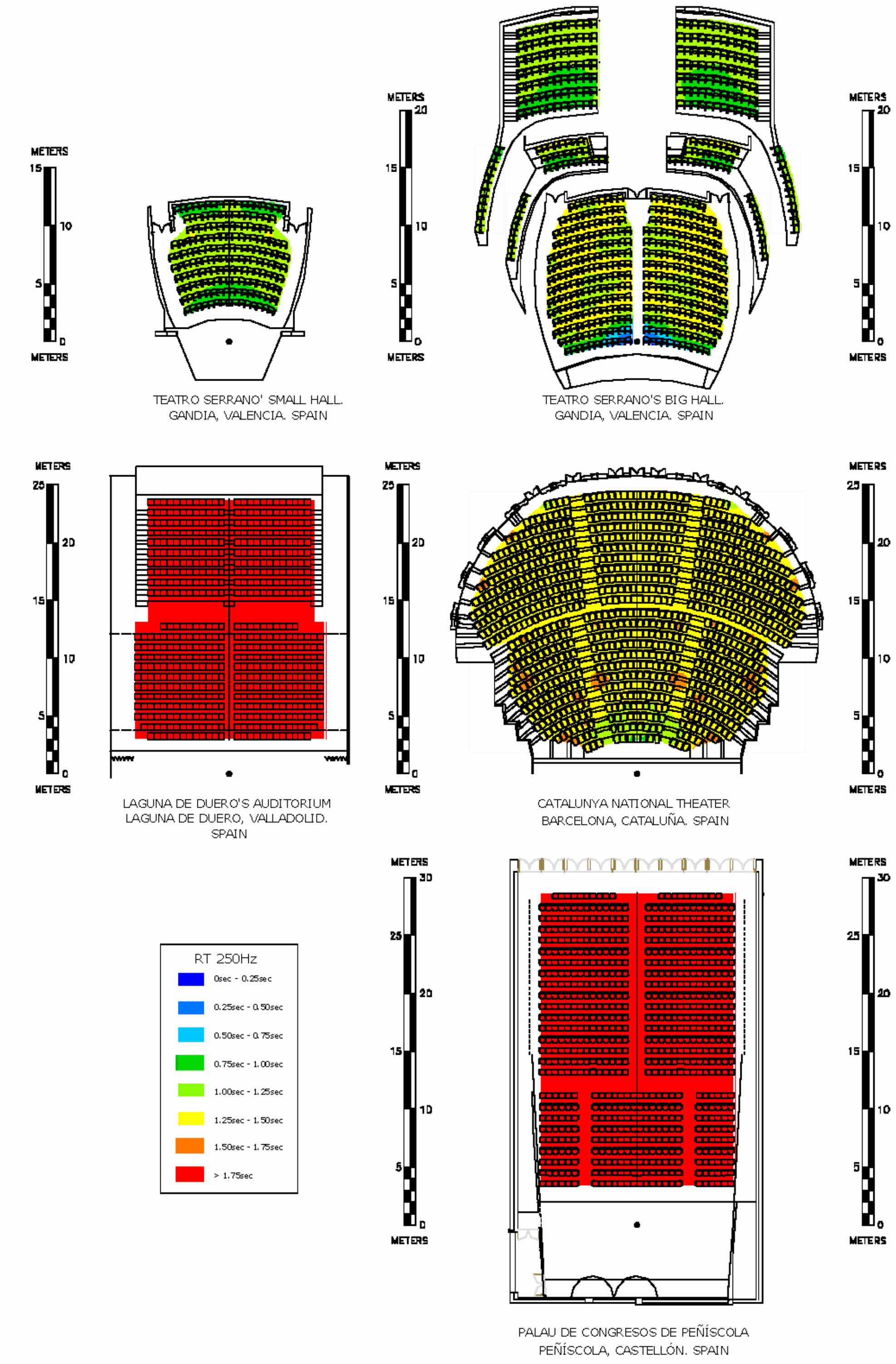


Figure 61: *Results: Acoustic Parameter* RT_{250Hz} .

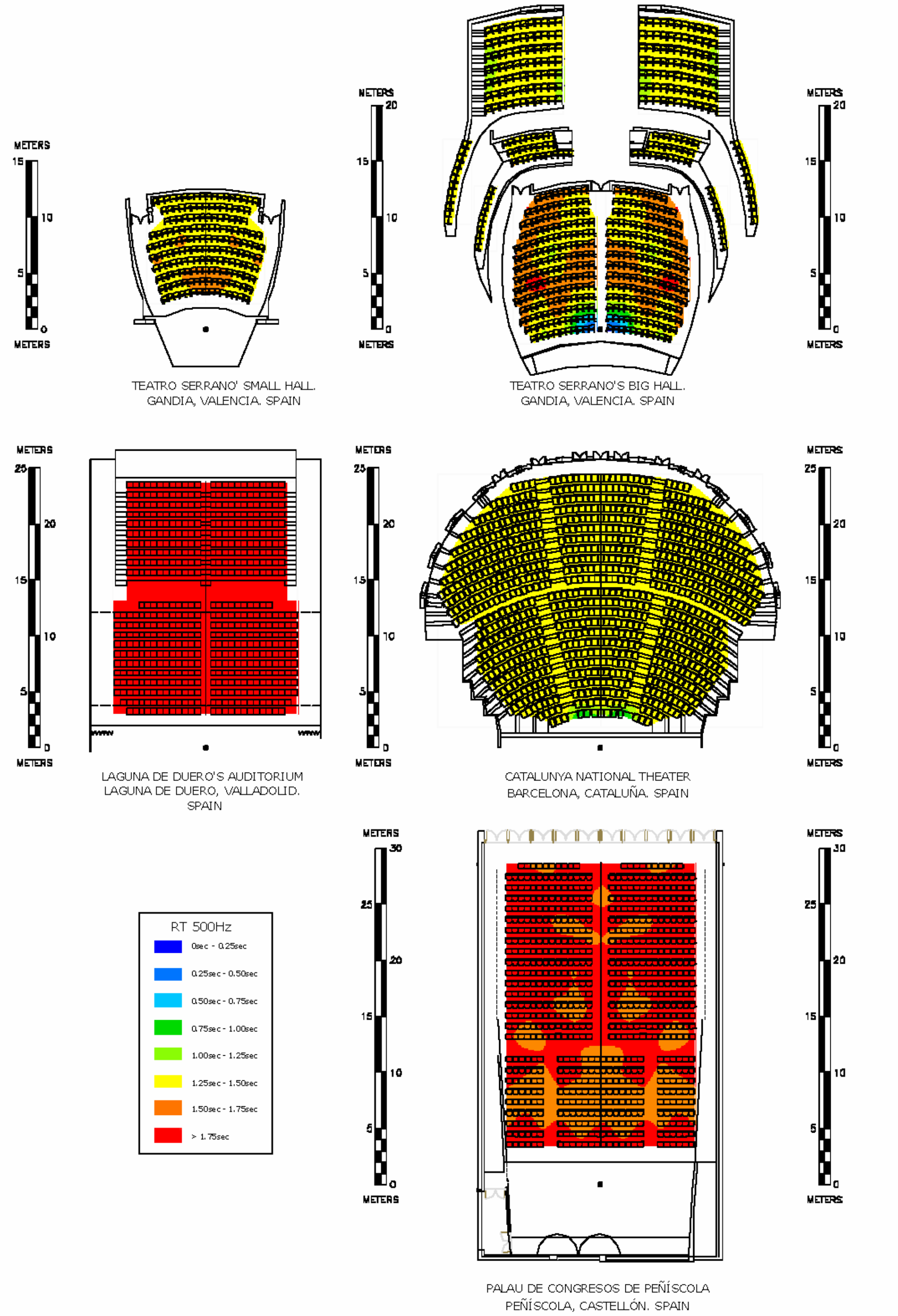


Figure 62: *Results: Acoustic Parameter* $RT_{500\,Hz}$.

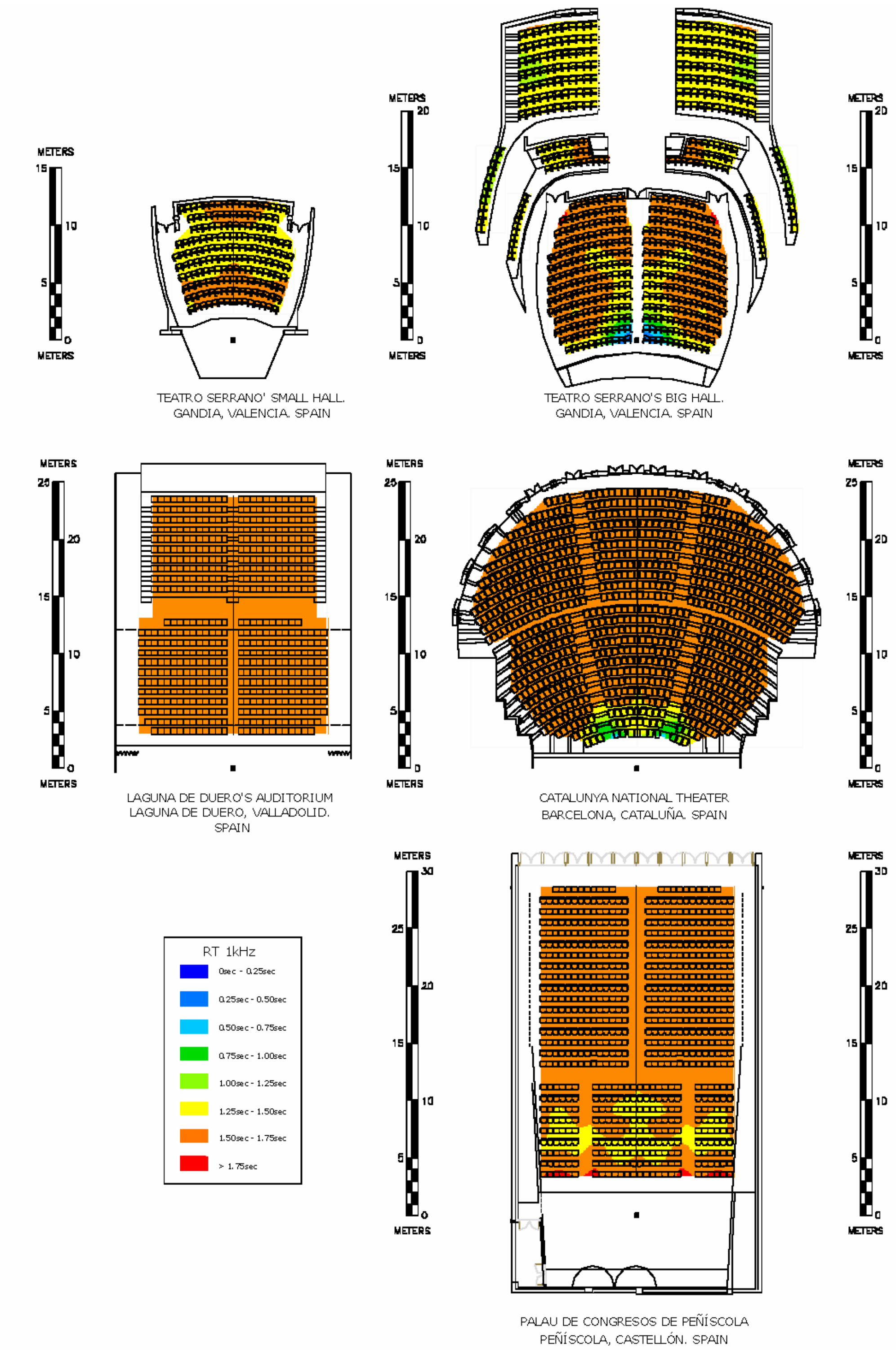


Figure 63: *Results: Acoustic Parameter* RT_{1kHz} .

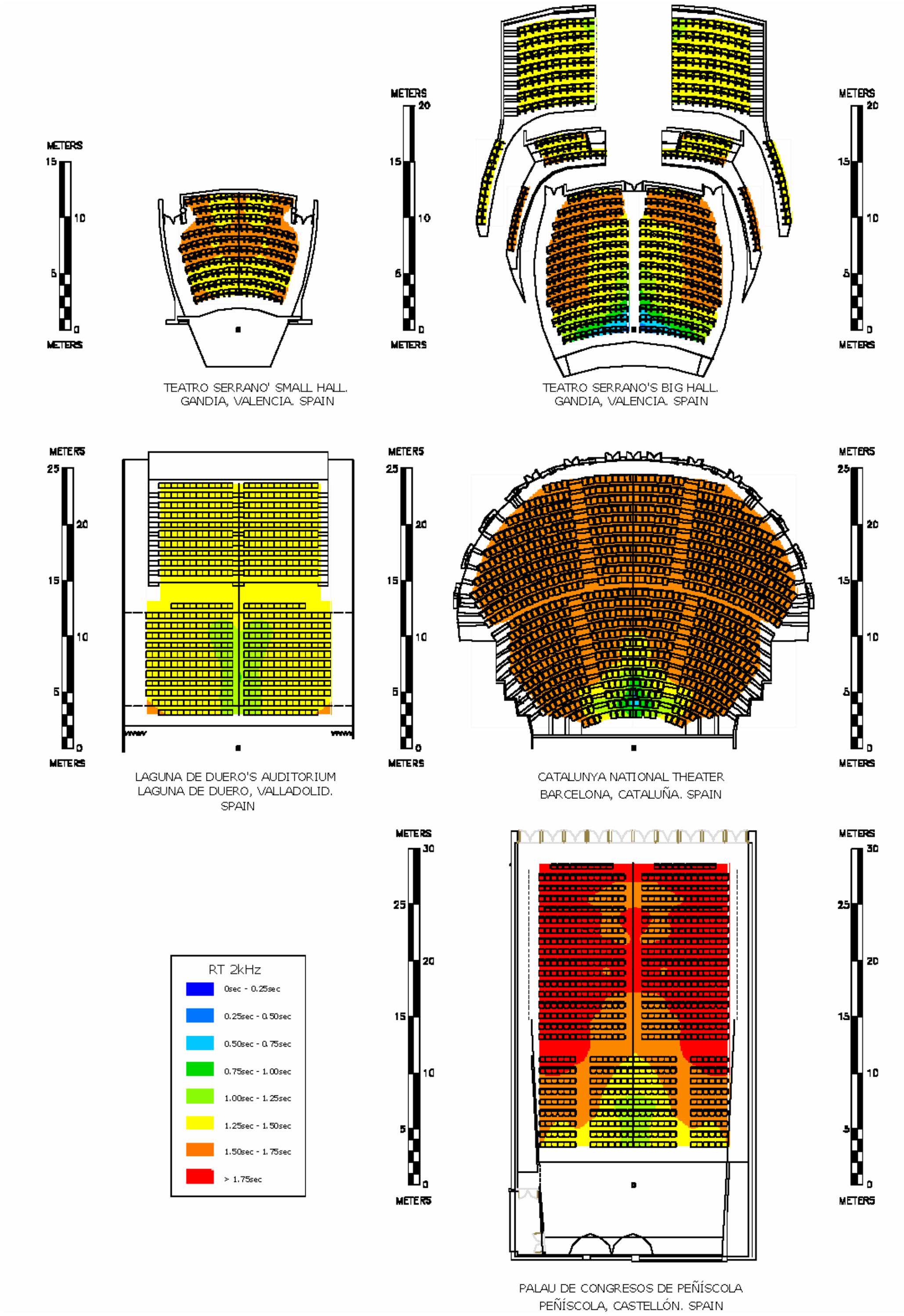


Figure 64: *Results: Acoustic Parameter* RT_{2kHz}.

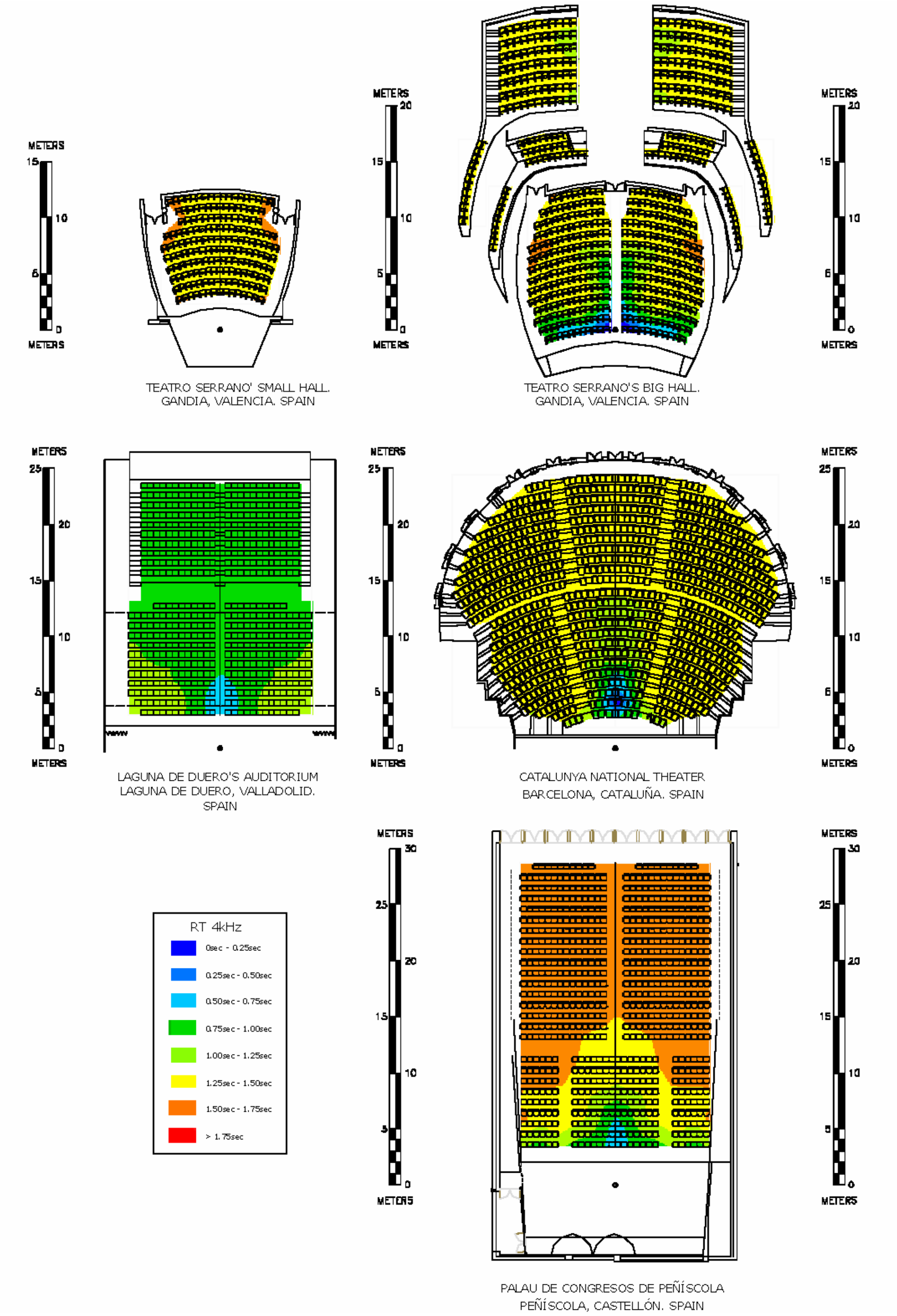


Figure 65: *Results: Acoustic Parameter* RT_{4kHz}.

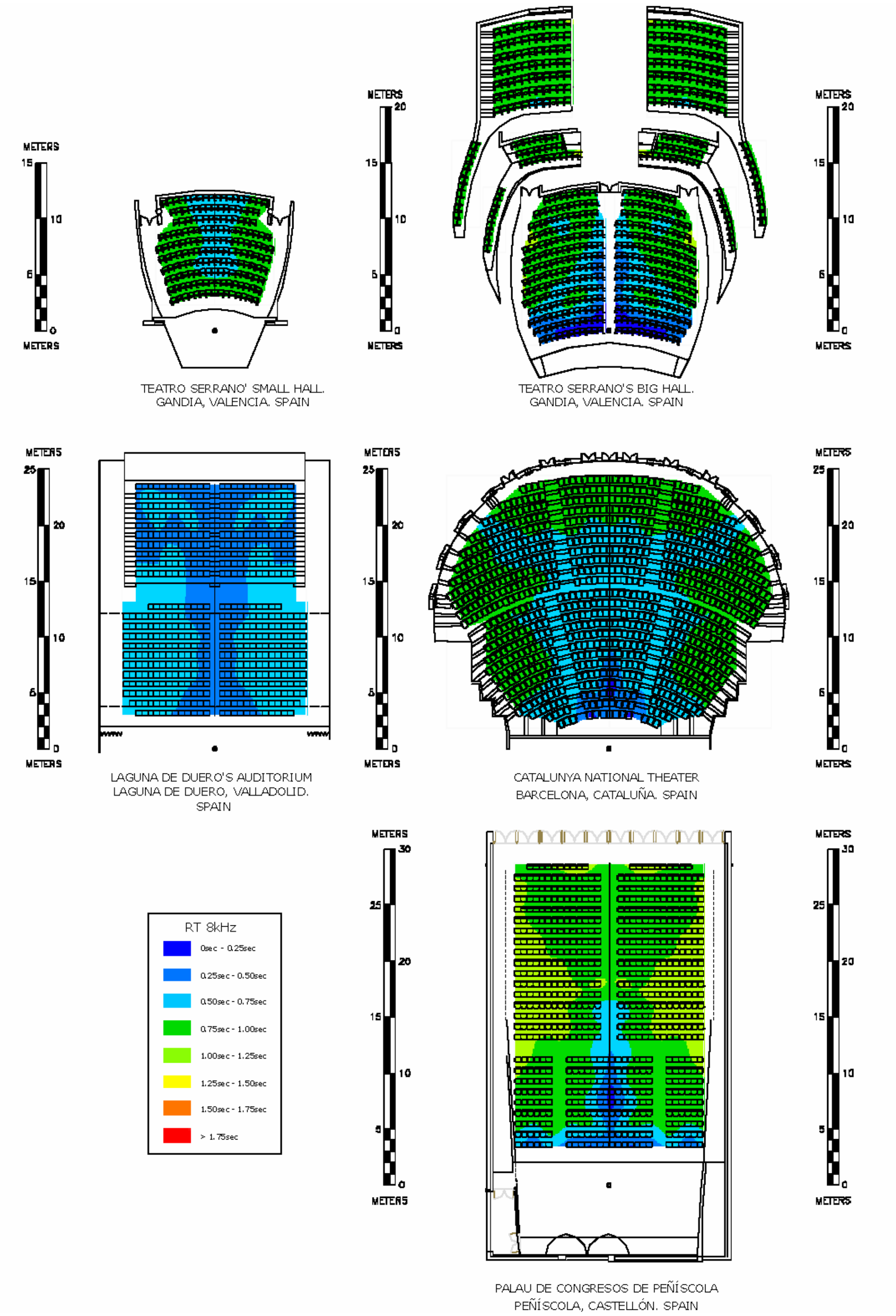


Figure 66: *Results: Acoustic Parameter* RT_{8kHz}.

3.3. Tonal Parameters (Bass Ratio and Brightness)

The Bass Ratio (figure 67) is the ratio of low frequency RT to mid frequency RT. The bigger the reverberation time at low frequencies the bigger the Bass Ratio number.

Variability of this parameter, however, has no direct relation with reverberation times as the frequencies vary. As it can be observed on figure 60, only Catalunya seems to repeat the unstable variability distribution in contrast to the uniformity observed in figures 61,62 and 63.

The Biggest values are obtained in Laguna Auditorium where there are some hot spots (values above 1.5). Catalunya shows numbers in the range of 0.8 to 1.15. Halls featuring the smallest and biggest cubic volumes (Serrano Small Hall and Peñiscola, respectively) have values going from 0.60sec to the 1.00sec.

As an important fact, the biggest values of this parameter are located away from walls in the five halls.

Brightness parameters are ratios of EDTs at high frequencies (2kHz and 4kHz) to those measured at mid frequencies. Both measured parameters are shown in figures 68 and 69. In Serrano Small Hall values over the recommended as optimal (0.9) are found with a decrease at about 5 meters from the source. Such variability has no relationship at all with its EDT equivalent at 500Hz, 1kHz or 2kHz.

Laguna Auditorium shows the lowest but still, acceptable values of brightness at 2kHz with some increasingly areas in green that match with high EDT_{2kHz}. Dark blue colours, the lowest brightness at 2kHz values, also coincide with high EDT values at 500Hz and 1kHz.

Furthermore, blend of highest EDT values at 2kHz corresponds to either orange or red values of brightness at 2kHZ whereas big EDTs at 500Hz and 1kHz mean not so highly bright values.

Things are a little bit different inside Peñíscola where any correspondence among parameters seems to be not so clear especially from the middle corridor onwards.

In the Serrano Main Hall, the main floor does not show a direct relationship with their EDT mapping. Besides rapid transitions in all balconies, (with changes almost every three seats) the amphitheatre low values match with the low yellow values found in figure 57.

Figure 69 show the Brightness at 4kHz, resembling that of Serrano Main Hall, Laguna Auditorium and Catalunya the EDT_{4kHz} follows the same tendency as for Brightness at 2kHz and once more variability has not a direct link with that shown in figures 55,56 and 58.

Distribution for brightness at 4kHz is too intricate to described it precisely but a sign of similarity with its EDT values is that the higher numbers (hot spots) are found next to lateral or rear walls.

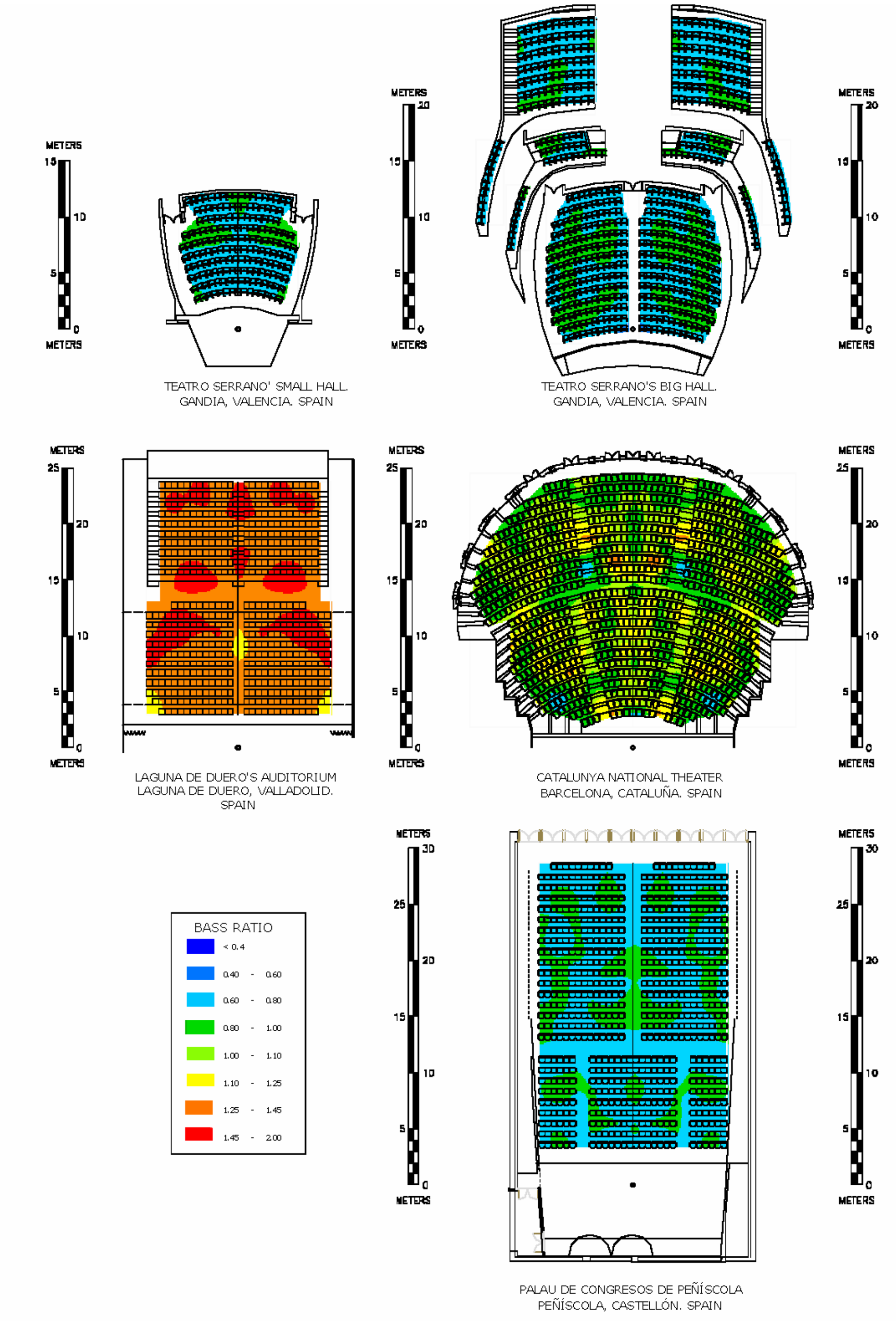


Figure 67: *Results: Acoustic Parameter Bass Ratio.*

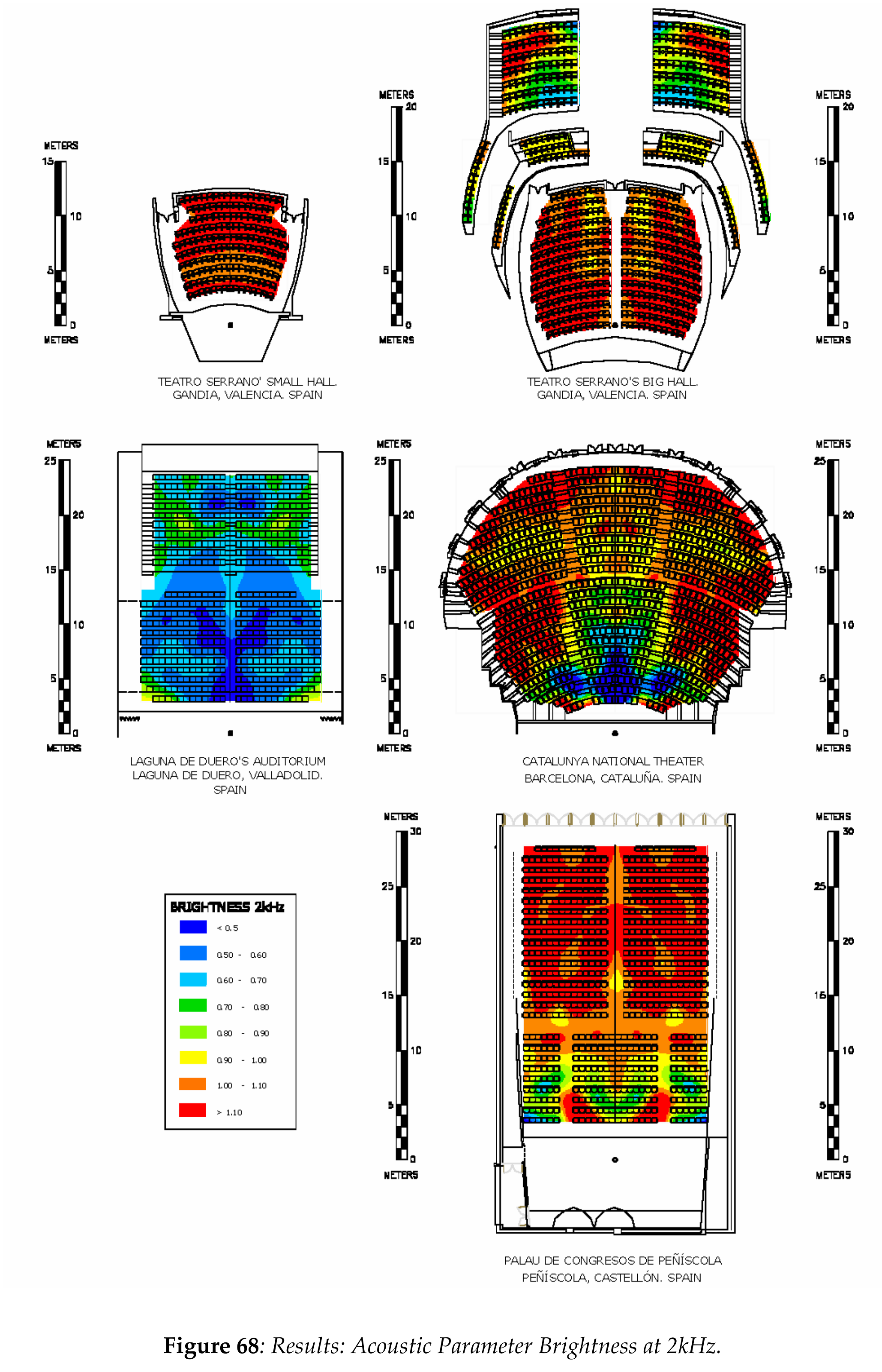


Figure 68: *Results: Acoustic Parameter Brightness at 2kHz.*

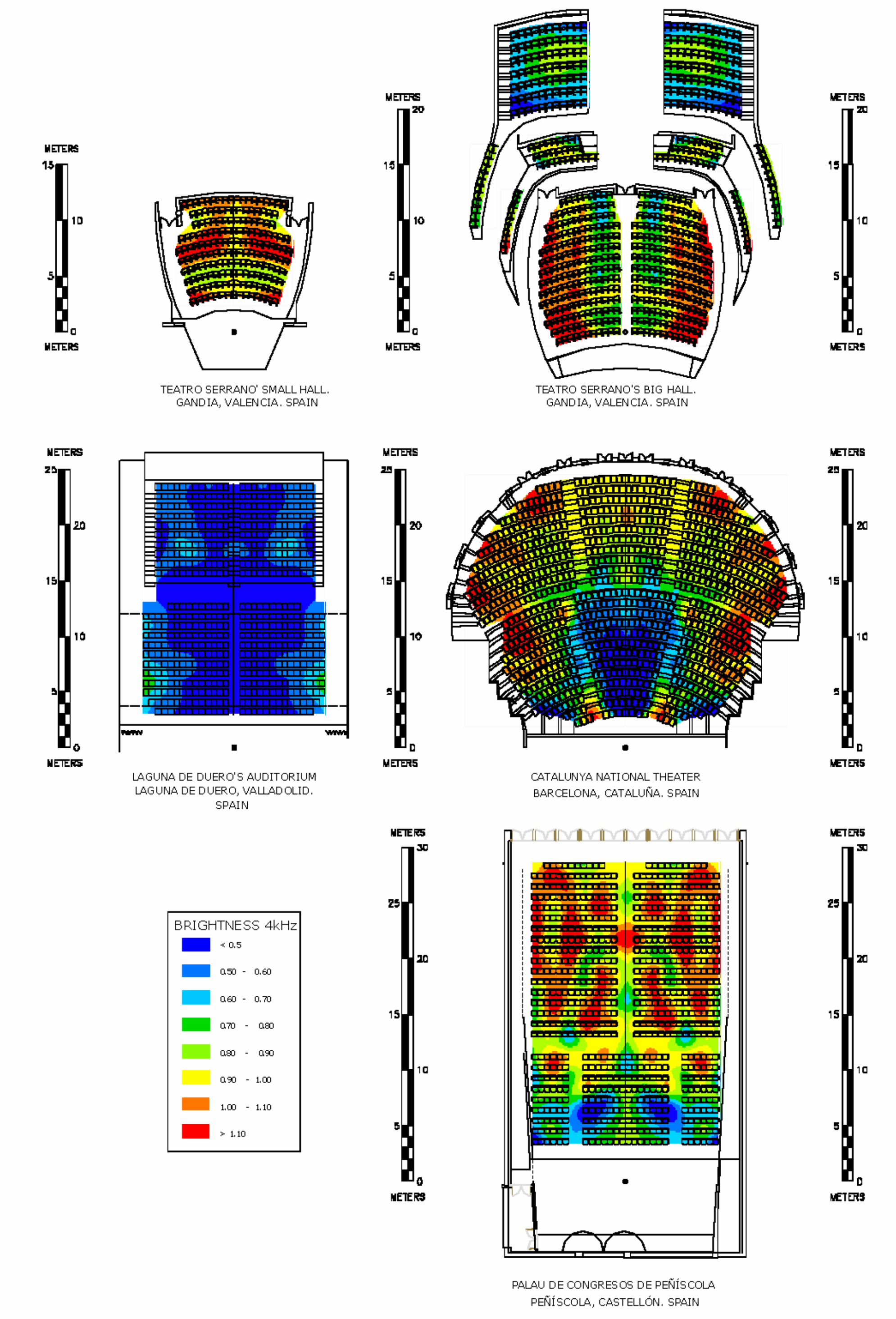


Figure 69: *Results: Acoustic Parameter Brightness at 4kHz.*

Meanwhile, in the upper amphitheatre the recommended optimal values are achieved in almost all its area except in those seats located at frontal corners (dark-blue coloured.)

3.4. C_{50} and D_{50} Parameters

The Speech Average C_{50} clarity is shown in figure 70. It can be seen that most halls have a good (above the 40%) level of clarity in the first 10 meters away from the source.

In auditoriums, however, this parameter value decays rapidly between the 15 and 20 meters away from the source, although in Peñíscola optimal levels are obtained close to the back wall. Inside the same hall, reflecting glass plates contribute to keep levels in the recommended optimal values (-2dB as the minimal). Flat concrete walls in Laguna Auditorium keep the same range over a wide area.

Nevertheless, in theatres things are a little bit different. The hot areas correspond to high levels of SPL generated by proximity to the source and, in the case of Catalunya; decreases appear at geometrical changes (after 10 meters).

Teatro Serrano Small Hall seems to be reverberant for speech with values around the –3dB whereas in the Main Hall the same behaviour exists with: a red beam in front of source’s position in all halls. Such area reaches a distance of 3 meters and strong early reflections coming from the closed fire curtain or the proscenium walls keep optimal C_{50} values over the first three rows and frontal seats in all balcony levels.

The same behaviour seems to apply also for lateral frontal areas at the amphitheatre but low levels of this parameter in the form of dark-blue spots can be found in the back upper seats.

On the main floor C_{50} decreases at –4dB close to back walls and at the same distance from the source (around 12 meters) at higher seating levels. Even lower values are found in the middle area of the main floor next to walls and disappear as long as high SPL’ s are found towards the middle corridor or at source position.

Another way of expressing speech clarity is its analogue Definition D_{50} illustrated in figure 71 and evaluated in terms of percentage values. The range used (increments of around 12.5%) smoothes some isolated spots that were found in the C_{50} representation.

By doing this is possible to get rid of the darkest zones at Catalunya and at the main floor in Serrano Main Hall. The rest of the halls keep invariably their distribution.

Recommended values over the 50% are found in almost all the halls at a distance between 10 and 15 meters away from the source.

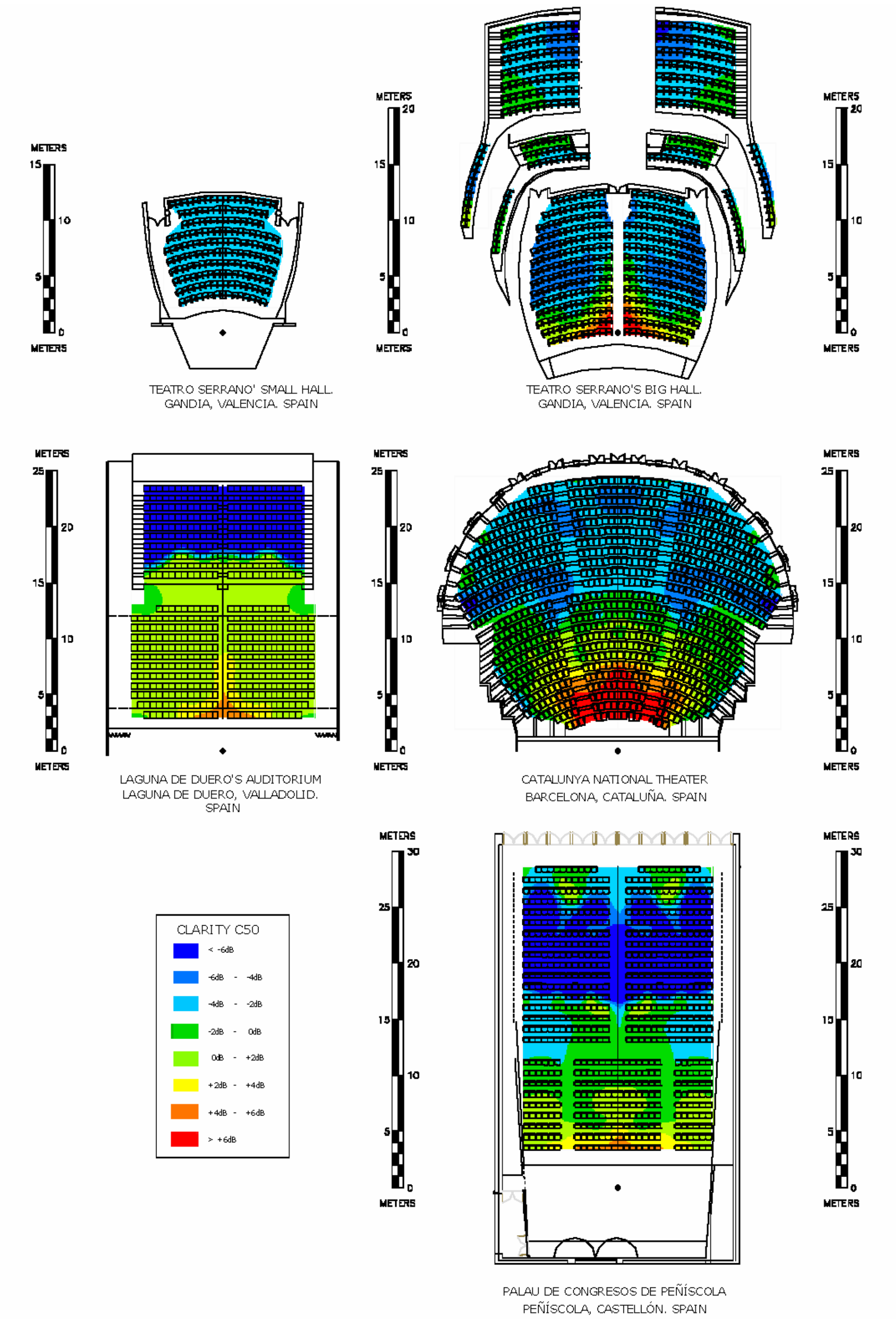


Figure 70: *Results: Acoustic Parameter* C_{50}.

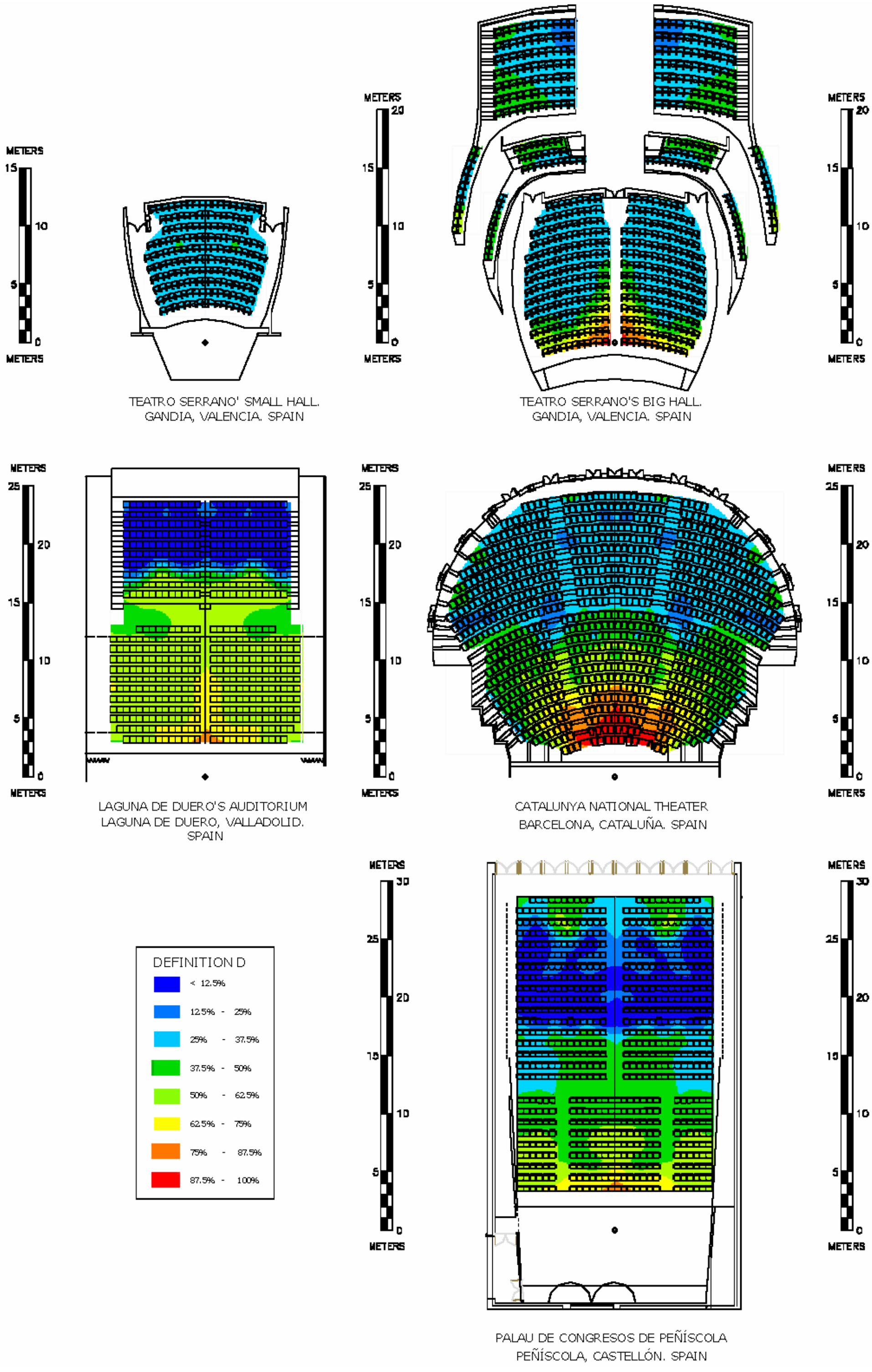


Figure 71: *Results: Acoustic Parameter* D_{50}.

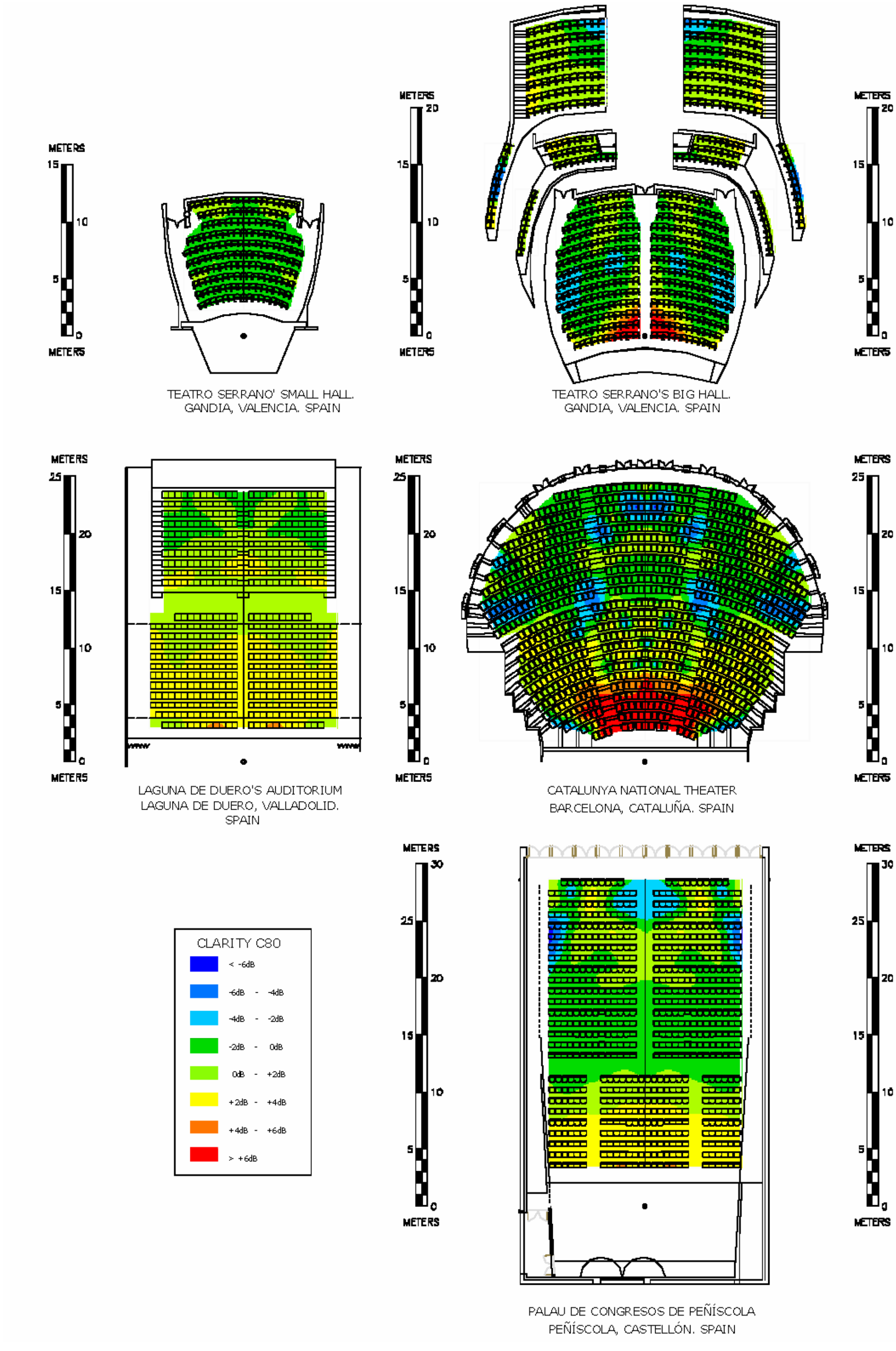


Figure 72: *Results: Acoustic Parameter* C_{80}.

3.5. C_{80} and t_s Parameters

The average Musical Clarity C_{80} for the five halls is illustrated in figure 72. According to the recommended range of values, those below the –2dB are found in only small spots at middle main floors or at the back wall in Peñíscola or at amphitheatre levels in Teatro Serrano.

High levels of clarity are found at 7 meters from the source in auditoriums and Catalunya where values can be inclusive over the 6dB. In Teatro Serrano's Main Hall those values can be found at 2 to 3 meters from the source.

With the exceptions described above and at the Laguna Auditorium raked back seats it is possible to say that 80% of the areas of the halls were inside the recommended optimal range, with values of C_{80} between –2dB and 0dB where performance of slow musical pieces can be enjoyed.

Middle seats at balconies and at amphitheatre level with their projection at main floor level receive the greatest contribution of reflections arriving at listener position within the 80msec compared with the arriving of overall sound. Light-blue spots represent this situation

The recommended values (figure 73) for Centre Time parameter are below the 140ms. It is possible to find graphical correspondence among C_{50}, C_{80} and t_s mapping over the five halls audience areas. Similarities with C_{80} variability are even well defined.

Low centre times, those close to zero, mean high transparency and as t_s increases, reverberant sound contributions become bigger.

Thus, is possible to relate the C_{80} +2dB to +4dB range with a t_s range of 80ms to 100ms. These are light-blue zones next to stage that also have C_{50} around +2dB. Light-green t_s values mean C_{80} values equal to –2dB to 0dB whereas centre time values of 140msec to 160msec correspond to C_{80} values of –4dB to -2dB.

Existence of hot-orange spots means highly reverberant areas that may not be so concentrate as they appear since results are influenced by data manipulation and an amount of global error. Nevertheless, their symmetrical appearance and localization (some of them appear close to back walls or middle corridors) are interesting.

Once more in Teatro Serrano Main Hall, there are different values for this parameter as a function of height. . Such differences are dramatically bigger than those featured in figures 70 and 72. Here, there are highly reverberant areas on the main floor (good for music reproduction) and positive values of C_{80} at balcony and amphitheatre levels.

Finally, most of halls accomplish the recommended optimal t_s value range below the 140ms.

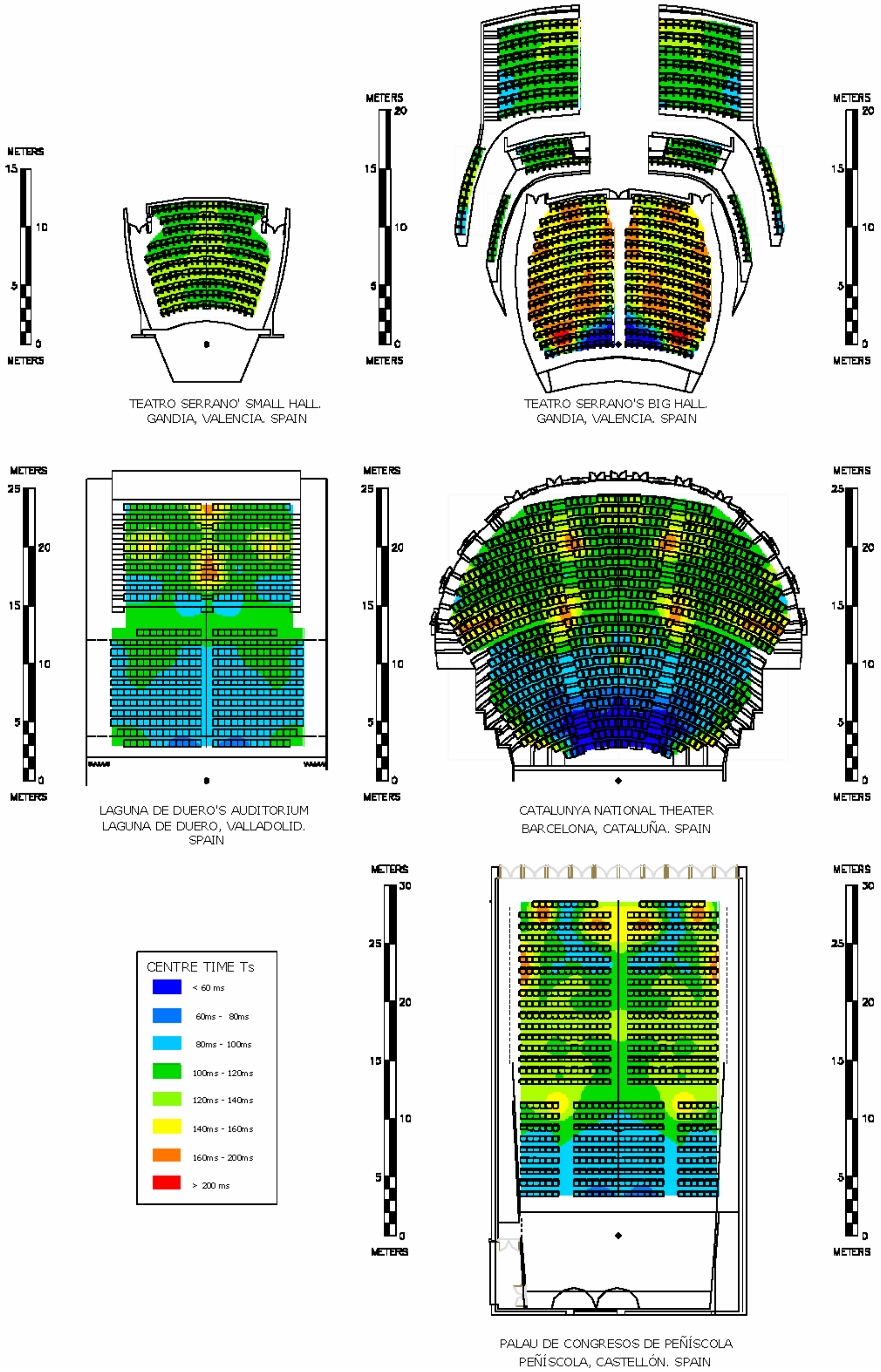


Figure 73: *Results: Acoustic Parameter* t_s.

3.6. L_{ef} Parameter

Figure 74 presents the results for the Lateral Energy Fraction Parameter. According to some authors, recommended values for this parameter are around 0.300 or 0.400 although lower values are found on measurements.

A semi- circular shape hall like Teatro Nacional de Catalunya shows the lowest L_{ef} values of all the five halls. The hot spots (high L_{ef} numbers) on lateral front seats are due to reflections originated by furniture or walls inside the stage enclosure. Since lateral proscenium walls are in a straight 90º angle, sound impelling there cannot be directed towards the audience located on the middle front areas. That, combined with high levels of direct sound due to source proximity, results in dark-blue (low L_{ef}) coloured seats.

Once geometry changes to semi-circle plan shape (approximately crossing the middle corridor) things improve, especially on the back of the hall.

Inside Serrano Small Hall, things could be similar but here diffusers placed on lateral walls (figure 45) improve early lateral reflections (those with more energy) over an important part of the audience. As in Barcelona, front middle seats also present a lack of L_{ef}, though.

The two Auditoriums are being, on average, the more benefited from early lateral reflections coming from flat walls. Both rooms have narrow shoebox plan shapes and in the case of Peñíscola, it can be observed how glass plates close to audience areas improve L_{ef}. Lateral front areas in auditoriums experiment L_{ef} values close to the 0.700. Once more without directing, sound by the appropriated panels or angled lateral walls and with a high level of direct sound, "dead zones" on the middle front seats are present. Such dark-blue areas extend to about seven meters from the source

In Serrano Main Hall, the source is not placed on stage but among the audience area instead. Frontal hot spots over the main floor are produced by early lateral reflections coming from the closed fire curtain, proscenium lateral walls or the combination of both. This can be observed also in lateral balconies green-coloured.

On the back of the main floor, changes in geometry concentrate lateral reflections as in frontal balconies deep corners. In the frontal rows of the amphitheatre, there is a low L_{ef} whereas the parameter increases in the upper seats. Once more, those seats located in front the source have a low L_{ef}.

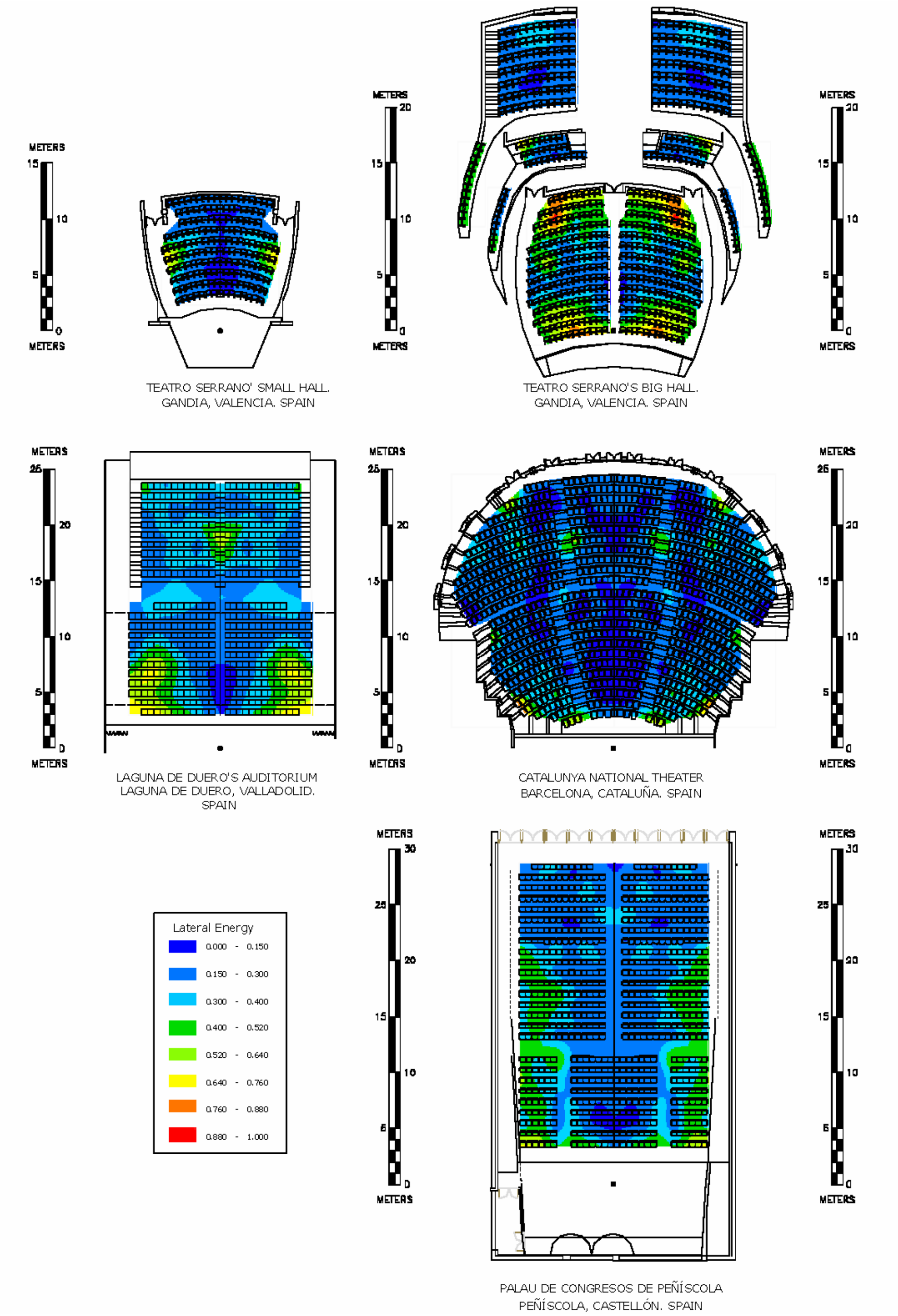


Figure 74: *Results: Acoustic Parameter* L_{ef} .

3.7. Conclusions and Recommendations

From all the descriptions presented in this text, it is easy to see that it is not possible to represent acoustical parameters by a single number. Even reverberation time, well known as a stable number in most audience areas, gives different values depending on frequency and horizontal and vertical measuring positions.

Some parameters could be averaged in order to obtain a number that makes it possible to rank halls but given the results obtained in this study it would be considered quite risky to take such values as technically accurate.

Sound fields are not uniform and even at a different floor levels high parameter variability is found. After this study, it can be seen that some parameters describe that lack of uniformity better than others do. As can be seen, G parameter and those related to clarity whether for speech or music (C_{50} and C_{80}) show more variability with the receiver position than the rest of parameters particularly C_{50} with D_{50} or both with the Central Time.

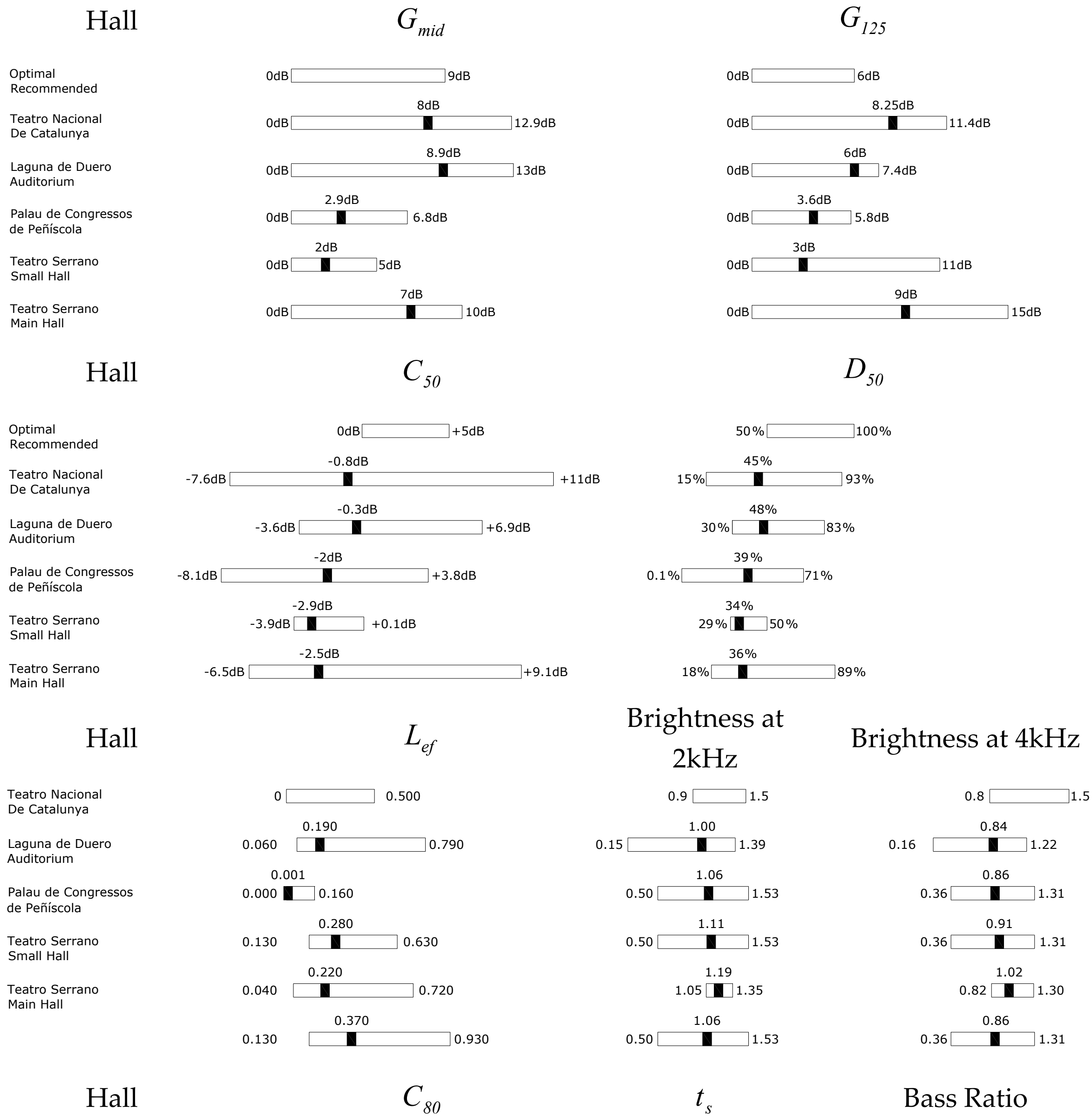
In other words, to describe the acoustic conditions of a hall one can use some parameters (*EDT*, *RT*, t_s, Bass Ratio, etc.) but to detail differences between halls a more seriously analysis should be carried out showing comparisons between more accurate parameters like G_{mid} and Clarity C_{50} and C_{80} and certainly with IACC related parameters.

Figures 75 and 76 show that values found after measurements have wider ranges than those expected and certainly wider than those recommended as optimum. In the figure 75 the values of different parameters are shown, with the white upper box indicating the optimum recommended values for each parameter, by some authors, whereas below are the measured values classified by hall. For early decay and reverberation times (figure 76) only the range of results measured are presented since optimum values are established as a function of cubic volume, use of the hall, etc.

All the results show the range between minimum and maximum values along with the median value, which gives an indication about the tendency of the results.

As an example, from the 114 measured points inside Teatro Nacional de Catalunya; the C_{50} values are between –7.6dB - +11dB with a median value equal to –0.8dB. This means that 57 values fall between –7.6dB and -0.8dB and 57 between –0.8dB -+11dB with the latter showing the biggest dispersion.

This explains why the number of measurements influences the tendency of results and why the results from a few measurements cannot be a statistically correct descriptor of the complete sound field in a hall



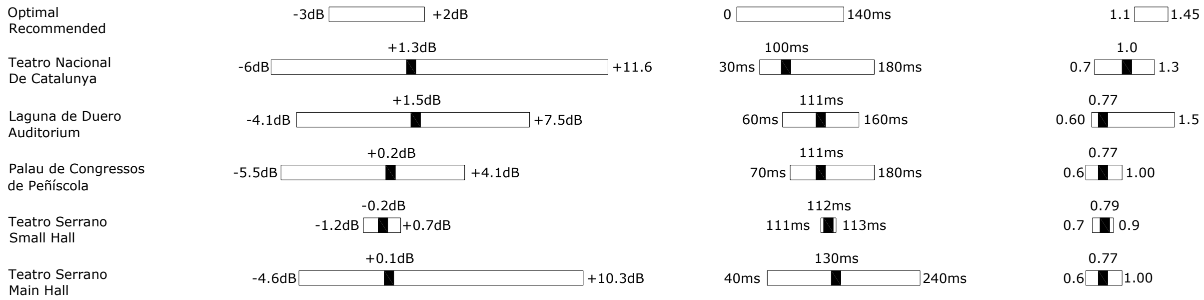
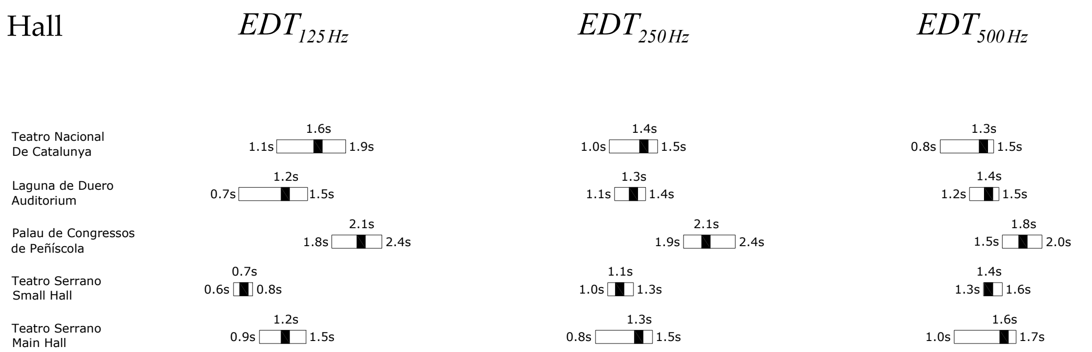
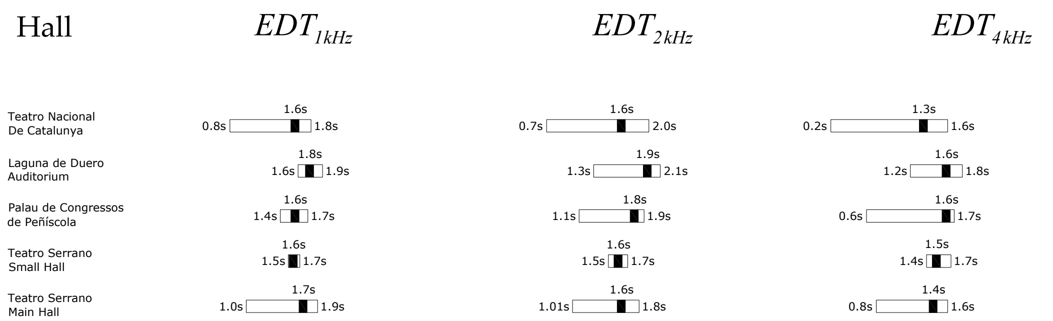
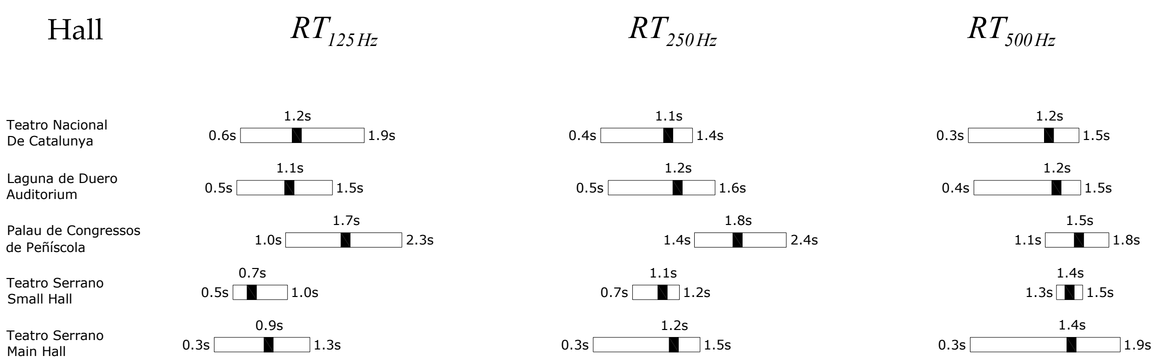


Figure 75: *Range of results, Parameters.*







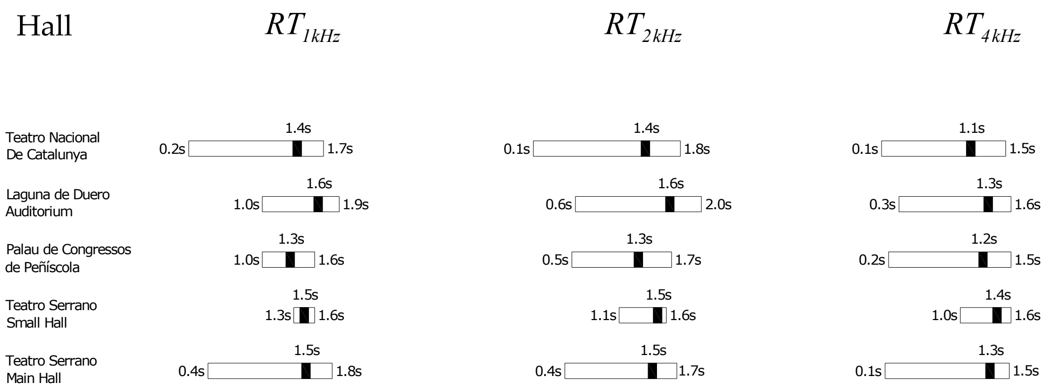


Figure 76: *Range of results, Reverberation Times.*

The most important factor in parameter variability is the localization of the sound source and the receiver together with other factors such as the sound level of the source, geometry and absorption. This is why an important further analysis should vary the location of the sound source over the stage. A more exhaustive analysis of the same sound source (testing its radiation pattern at different frequencies) may help to discard possible influences on sound spreading.

Different results may be obtained from measuring halls with different levels of occupancy since absorption by audience is a critical factor, especially at high frequencies.

In order to consider influences of temperature and relative humidity it will be necessary to perform measurements in a hall over periods of time and with different climatic conditions

By locating measuring positions on stage, parameters as Stage Support introduced in Chapter 1 could be studied. Measuring in corridors next to walls can give details of how sound behaves there.

An attempt to measure the IACC related parameters was made but unfortunately some limitations found on the instrumentation made it difficult and unreliable. The measurement of such parameters and to match them to those related by lateral sound may give a complete description of this genre.

Appendix A

LIST OF EQUIPMENT AND SOFTWARE USED ON MEASUREMENTS:

- Brüel & Kjaer Type 4192 Pressure-field ½" Microphone (Serial No: 2097391)
- Brüel & Kjaer Type 2669 Microphone Preamplifier (Serial No: 2082179)
- Brüel & Kjaer Type 2804 Microphone Power Supply (Serial No: 1318520)
- AKG Type C414 B-TL Condenser Microphone (Serial No: 3890)
- AKG Type B18 Battery Phantom-Power Supply
- ACOUSTICAL Type QUAD 405 Two-Channel Power Amplifier (Serial 094344)
- 6x8" AG/N Type Custom-Made Hexahedron Loudspeaker w/Stand
- NORSONIC Type 330A Nor-840 Real Time Analyser (Serial 23195)
- Cables

- NORSONICS Nor-Rac Type 1012 Program to Compute Room Acoustic Parameters According to ISO3382
- Environmental Systems Research Institute, Inc. Arc View GIS Version 3.1
- AutoDesk, AutoCAD 2004
- MICROSOFT Office 2000

Appendix B

SETUP PHOTOS

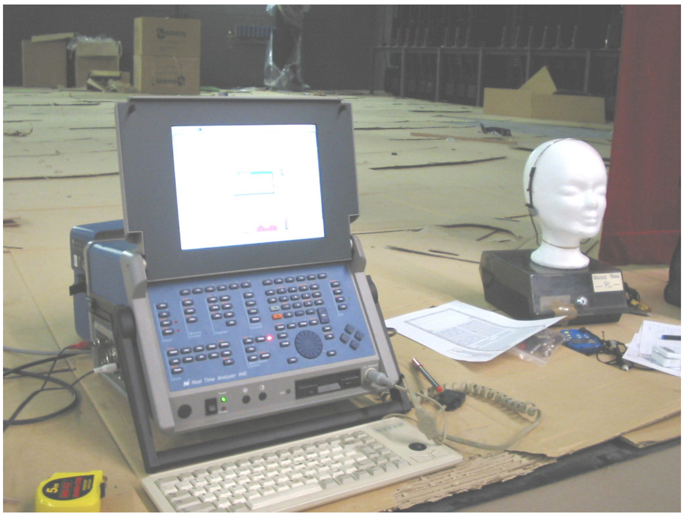


Figure 77: *NORSONIC Nor-840 Real Time Analyser.*

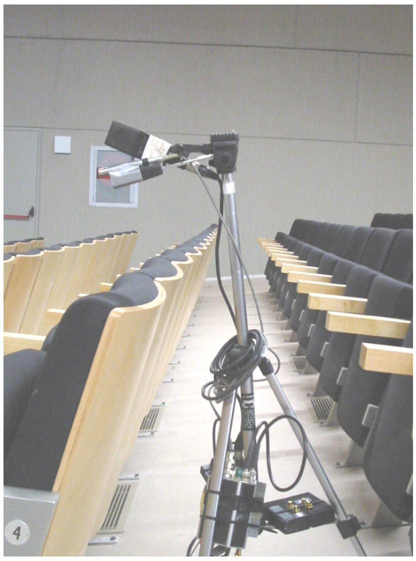


Figure 78: *BRÜEL & KJAER and AKG Microphones.*

Figure 79: *Preparing Setup for IACC measurements*

Figure 80: *Custom-Made Hexahedron Loudspeaker.*

Appendix C

THE RTA840 MAXIMUM LENGTH SEQUENCE FEATURE

With the Maximum Length Sequence (MLS), method is possible to measure the impulse response with a great amount of accuracy and repeatability. This method is based on the cross-correlation technique and uses a maximum-length sequence-which is a periodic pseudo-random binary sequence-as the source signal. The binary sequence $x(k)$ is represented by +1 and –1 and may be generated by a shift register with feedback.

The cross-correlation between the input $x(k)$ and the output $y(k)$ of linear systems is related to the auto-correlation of the input by a convolution with the impulse response:

$$R_{xy}(k) = R_{xx}(k) * h(k)$$

An important property of any MLS is that its auto-correlation function is essentially an impulse. This impulse is represented by the Dirac delta function:

$$R_{xx}(k) = \delta(k)$$

The result of convolving a sequence with a Dirac delta function is the sequence itself. Thus, the impulse response $h(k)$ can be found by cross-correlating the noise input $x(k)$ and the output $y(k)$:

$$R_{xy}(k) \approx \delta(k) * h(k) = h(k)$$

Hence, it is possible to measure the impulse response of linear systems by calculating the cross-correlation between the MLS and the system output signal. Since $x(k)$ is a known pseudo-random sequence, there exists an efficient and very fast way to calculate the cross-correlation function $R_{xy}(k)$, called Fast Hadamard Transform, also known as FHT.

A benefit of the FHT is that requires, like the Fast Fourier Transform, only $n\log_2(n)$ operations. Since the MLS is represented by +1 and –1, the FHT consist of additions and subtractions only.

An MLS is actually a deterministic signal, but it has similar spectral properties as true random white noise. A great advantage is that since the sequence is deterministic, it can be repeated precisely. It is therefore possible to increase the signal-to-noise ratio (S/N) by synchronous averaging of the response sequences $y(k)$. Any extraneous, uncorrelated background noise will then be reduced for each averaging. The S/N ratio will increase by 3 dB for every doubling of the number of averages.
Since the MLS is a periodic signal, the autospectrum for the sequence consists of lines separated by the inverse of the period T_p. It can be shown that the lines for frequencies well below the clock frequency are equal in magnitude and that the spectrum thus approximates white bandlimited noise. Since the total power is independent of the clock frequency, lowering the clock frequency increases the spectral density.

If the sequence length is longer than half the reverberation time of a room, it can be shown that at least one spectral line will fall within every mode of the room; A common requirement is that the length of the MLS should be at least equal to the reverberation time.

The MLS method is implemented in the Nor-840 in the following way:

The length of the MLS is fixed to $2^{17}-1$, which gives an impulse response of 131071 samples. It is possible to change the duration T_p of the MLS by changing the sampling frequency f_s. When changing f_s, is necessary to satisfy the Nyquist sampling theorem. Therefore, we have chosen to let the upper frequency f_h select the sampling frequency. The relationship between f_h and f_s is shown in the table below:

Upper frequency f_h	**Sampling Frequency** f_s	**Sequence Period** T_p
$12.5\text{kHz} \le f_h \le 20\text{kHz}$	64kHz	2.048s
$6.3\text{kHz} \le f_h \le 10\text{kHz}$	32kHz	4.096s
$3.15\text{kHz} \le f_h \le 5\text{kHz}$	16kHz	8.192s
$1.6\text{kHz} \le f_h \le 2.5\text{kHz}$	8kHz	16.384s
$800\text{Hz} \le f_h \le 1.25\text{kHz}$	4kHz	32.768s
$0.1\text{Hz} \le f_h \le 630\text{kHz}$	2kHz	65.536s

By making MLS measurements with the RTA840 is possible to achieve the following:

The MLS is used as the excitation signal and is available at the generator output socket. To eliminate initial transients, the measurements start after one MLS period, or in other words, after T_p seconds.

If the number of averages is greater than 1, the responses $y(k)$ of the MLS will be synchronously averaged in order to reduce the influence of the background noise.

When the measurement is finished, the post-processing tasks will be started. The response $y(k)$ will first be transformed by the Hadamard Transform to obtain the broadband impulse $h(k)$. This impulse response is used as the input signal for a normal multi-spectrum measurement in the RTA840. When this multi-spectrum measurement has been made, the signal processing of the results will start. The signal processing part can be divided into two groups; one for level measurements and one for reverberation time measurements.
On the Level measurements, the level of each one-third octave band is calculated by integrating only the part of the impulse response where the signal energy is significant, skipping the part where the background noise dominates. This gives a better estimate of the signal. By integrating the last part of the measurement is possible to get the estimate of the background noise level.

On the other hand, reverberation time calculations are based on short time L_{eq} values obtained by filtering the broadband impulse response $h(t)$. The reverberation time parameters EDT, T_{20} and T_{30} are calculated from the filtering impulse response by application of the Schroeder method. The backward integration starts at the intersection point between the linear signal decay and the background noise level. The starting point will be selected individually for each frequency band to achieve optimal dynamic range.

- T_{20} is measured as time between the –5dB and –25dB crossing
- T_{30} is measured as time between the –5dB and –35dB crossing
- EDT, is measured as time between the –1dB and –11dB crossing

Regarding limitations, the MLS technique allows measurements with large dynamic ranges. Thus, care must therefore be taken to eliminate influence from unwanted signal paths such as electrical crosstalk. A signal caused by electrical crosstalk will have very short reverberation time and may be found by studying the impulse response.

The MLS may be described as a series of pulses of equal amplitude but appearing in positive or negative direction in a random manner. The measured signal will be the response of these pulses. The effect of the Hadamard transform is to transfer the individual responses at different time and in different direction so they appear as a response to impulses appearing simultaneous and in one direction. It is therefore important that the system remains unchanged during measurement. The requirement to time-invariance is very important and should always be considered when MLS techniques are applied.

Unlike other building acoustics measurements, is not possible to move microphones during MLS measurements due to the fact that responses with different phases will be averaged together and the mean value will be too low. This will happen if more periods of the sequence is averaged, but the effect will also apply for a measurement based on one possibility of the sequence only.

Electroacoustic components such as microphones, loudspeakers and power amplifiers are so stable that few problems due to time-invariance are to be expected. Time-invariance due to environmental conditions is more pronounced if measurements are performed over a long period.

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