CONCERT HALL ACOUSTIC COMPUTER MODELLING: TEATRO LICEO

DE BORTOLI, LUCIANO - KAPLUN, TOMAS - ROTEA, FRANCISCO

Universidad Nacional de Tres de Febrero, Acoustics Measurements and Instruments, Buenos Aires, Argentina
luciano.nicolas.de.bortoli@gmail.com
Universidad Nacional de Tres de Febrero, Acoustics Measurements and Instruments, Buenos Aires, Argentina
tomaskaplun@gmail.com
Universidad Nacional de Tres de Febrero, Acoustics Measurements and Instruments, Buenos Aires, Argentina
francisco.rotea@gmail.com

Abstract - *With the rapid growth of speed and power of computers in the last decades, room acoustic computer models have become a reliable and efficient design tool for acoustic consultants. The aim of this paper is to simulate the Liceo theater in Buenos Aires, Argentina, using EASE 4.3 software in order to obtain its main acoustic parameters, namely: STI, ALcons, RT, EDT, Echo Speech, Echo Music, Lateral Fraction, SPL and D/R. Results obtained are then compared to previous measurements made by a research group of the Universidad Nacional de Tres de Febrero. Findings are that the computer model is accurate enough at medium-high frequencies for reverberation time and intelligibility parameters, and inaccurate for low frequencies. Regarding the others parameters, the simulation was not as precise as RT, however differences are not generally significative.*

1. INTRODUCTION

Before the beginning of the massive use of personal computers and the rise of acoustic simulation softwares, the objective acoustical parameters of a room were only possible to obtain by measurements performed *in situ* once the enclosure was already constructed, or by making a scale model. Nowadays, with consumer computers, a simulation of the acoustical behaviour of a enclosure can be easily made, allowing acoustical engineers and architects to predict acoustical characteristics and parameters. However, the accuracy of the software can be a focus of study in order to determine the validation of the simulation. Many different software environments (such as EASE, Odeon, CATT-Acoustic, COMSOL, etc) can obtain the parameters by different methods, like empirical simulation, image source simulation, ray tracing simulation or by using acoustic diffusion equations.

Even though there are no studies about the acoustic modelling of the Liceo theater, as far as we know, there are a large number of comparisons made between simulations and real measurements in the literature. In 2013, Piqueras Moreno [1] performed a comparative evaluation of two acoustic modelling softwares (EASE and CATT-Acoustic) simulating a german theater called Hoftheater Kreuzberg Berlin. The investigator found that both softwares lack accuracy at low frequencies (because both softwares use geometrical acoustics), and it is stated that EASE is considerably easier and quicker to do the graphical representations. In 2008, Christensen *et al.* [2] presented an investigation with eight different users simulating a room using Odeon as software, and then compare them with measurements. It was found that the largest differences between users was in the computation of the absorption coefficients, which led to inaccurate values for reverberation time. It was also found that most participants were able to predict STI with reasonable accuracy. Lastly, in 2005, Field *et al.* [3] presented a study on the prediction of the acoustic parameters of a church auditorium using Odeon and CATT-Acoustic, comparing them with measurements performed *in situ*. It was found that reverberation time and STI is predicted with a high degree of accuracy.

The purpose of this paper is to perform a computer model of the Teatro Liceo, starting from a SketchUp 3D model based on the theater blueprints, which is imported into EASE 4.3 (*Enhanced Acoustic Simulator for Engineers*). All the surfaces' materials and dimensions are taken into account, with its corresponding absorption and diffusion coefficients. Then, fifteen synthesized impulse responses (RIR) are obtained from different positions uniformly distributed among the audience area, so that a representative value of the acoustical parameters is acquired. Acoustical parameters are evaluated using EASERA and Aurora Modules for Audacity 2.0.5. Finally, the acoustical parameters obtained through the simulation are compared to those obtained from real measurements performed in location on April 7th, 2017 by a group of students and professors of the class of Acoustical Instruments and Measurements from the Universidad Nacional de Tres de Febrero (UNTREF).

The structure of this paper will be as follows: in the first part, we present a brief summary of the basic concepts and terms used in this paper. Then, the methodology and procedures of the simulation will be explained. Later, results will be presented and analyzed. Finally, the conclusions of the investigation will be introduced.

2. CONCEPTS AND THEORETICAL TERMS

In this section a brief introduction of the theater and the theoretical framework that supports the investigation with the main concepts and terms used are presented.

2.1 Reverberation Time (RT)

When the source of sound in a room is suddenly turned off, a certain period of time is required before the sound energy is practically all absorbed by the surfaces in the room [4]. The reverberation time is defined as the time required for the sound pressure level to decrease by 60 dB, as shown in Figure 1. Because reverberation time is frequency dependent [5], it is more precisely described in terms of frequency bands (1 octave, 1/3 octave, etc.).

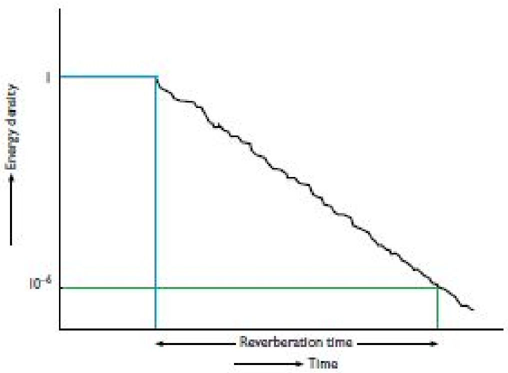

Figure 1: Reverberation time.
Source: Acoustics: An Introduction.

2.2 Early Decay Time (EDT)

Reverberation can further be considered in two parts: early and late reverberation. The ear is very sensitive to early reverberation. This is partly because in most music, later reverberation is partly masked by the following music notes. Early reverberation largely defines our subjective impression of the entire reverberation event. The early decay time (EDT) is defined as the time required for sound to decrease 10 dB, multiplied by 6 to allow comparison with late reverberation time RT. Unlike dense late reverberation, early reverberation comprises a relatively few primary reflections; these reflections arrive within the Haas fusion zone and are integrated with the direct sound, reinforcing it. This early reverberation can affect the clarity of sound. The greater the energy in the early reverberation, the better the clarity [6].

2.3 Lateral Fraction (LF)

The lateral energy fraction is the ratio of the output of a figure-8 microphone (with its null direction aimed at the source) to the output of a non-directional microphone. The figure-8 microphone weights the non-direct energy by $cos^2\theta$ where $\theta = 90º$ is in the direction of the sound. LF is given by [7]:

$$LF = \frac{\int_{0.005}^{0.080} p_8^2(t)dt}{\int_{0}^{0.080} p^2(t)dt} \qquad (1)$$

As shown, the time integration is usually performed over the interval of 5-80 ms for the figure-8 microphone and 0-80 ms for the omnidirectional microphone. The 5 ms value is introduced to make certain that the direct sound is eliminated.

2.4 Speech Transmission Index (STI)

In auditoria used for speech, such as lecture halls or theaters, the influence of the acoustics on intelligibility is a major issue. Currently, the most common way to assess objectively speech intelligibility in rooms is by measurement of the speech transmission index STI.

This measure is based on the idea that speech can be regarded as an amplitude modulated signal in which the degree of modulation carries the speech information. If the transmission path adds noise or reverberation to the signal, the degree of modulation in the signal will be reduced, resulting in reduced intelligibility. The modulation transfer is tested by emitting noise in seven octave bands, each modulated with 14 different modulation frequencies (0.03, 0.8, 1, 1.25, 1.6, 2.0, 2.5, 3.15, 4, 5, 6.3, 8, 10 and 12.5 Hz) and then calculating the ratio between the original and the received degree of modulation, the modulation reduction factor, in each of these 98 combinations. A weighted average of the modulation reduction factor then results in a number between 0 and 1, corresponding to very poor and excellent conditions respectively. A faster measurement method using only two carrier bands of noises and four plus five modulation frequencies is called rapid STI (RASTI).

Although the original method of STI or RASTI measurement employs modulated noise signals, it is also possible to calculate the modulation reduction factor from the impulse response. Thus, the modulation reduction factor versus modulation frequency F, which is called the modulation transfer function (MTF), can be found as the Fourier transform of the squared impulse response normalized by the total impulse response energy [8].

2.5 Percentage Articulation Loss of Consonants (%ALcons)

The articulation loss of consonants, expressed as a percentage, is another way of characterizing the intelligibility of speech. Similar to the articulation index, it measures the proportion of consonants wrongly understood. Peutz also found that the correlation between the loss of consonants (in Dutch) is much more reliable than a similar test with vowels. In 1971 he published a relationship to predict intelligibility for unamplified speech in rooms [9], which had been studied much earlier at Bell Labs:

$$ALcons = \frac{200r^2RT^2}{V} \tag{2}$$

Where V is the room volume (in m³) and r is the talker to listener distance (in m). Beyond the limiting distance defined by equation 3.

$$r_l = 0.21\sqrt{\frac{V}{RT}} \tag{3}$$

ALcons can be calculated as:

$$ALcons = 9RT \tag{4}$$

2.6 Direct to Reverberant Ratio (D/R)

The D/R is the ratio between the direct energy and the reverberant energy. This parameter follows equation 5.

$$D/R = \frac{\int_0^a p(t)^2 dt}{\int_a^\infty p(t)^2 dt} \tag{5}$$

where *p(t)* is the sound wave, and *a* is the end of the direct sound. The end of the direct sound is a focus of discussion, some authors say different things about the duration of the direct sound [10]. The point is that this duration depends of the signal and the enclosure, this means that define a constant fixed duration is not correct.

2.7 Echo-Speech & Echo-Music

The echo criterion (EC) was proposed by Dietsch and Kraak in 1986 [11] as a procedure to indicate whether a certain peak in a measured impulse response or 'reflectogram' hints at an audible echo and should be removed by suitable constructive measures. [12] Such criterion is based on equation 5.

$$t_s(\tau) = \frac{\int_0^\tau |g(t)|^n t dt}{\int_0^\tau |g(t)|^n dt} \tag{5}$$

With $g(t)$ being the impulse response of a room and n is a parameter related to the type of signal: if it is speech, $n = 2/3$ (echo speech), and if it is music, $n = 1$ (echo music). Finally, the quantity used for rating the strength of an echo is based upon the difference quotient of $t_s(\tau)$, as shown in equation 6.

$$EC = max\left[\frac{\Delta t_s(\tau)}{\Delta\tau}\right] \tag{6}$$

Where $\Delta\tau$ can be adapted to the character of the sound signal (9 ms for speech and 14 ms for music). The dependence of the echo criterion EC on the directional distribution of the various reflections is accounted for by recording impulse responses with both microphones of a dummy head and adding their energies.

It should be noted that echo disturbances are mainly due to spectral components of somewhat elevated frequency. For practical purposes, however, it seems sufficient to employ test signals with a bandwidth of 1 or 2 octaves (700-1400 Hz for speech and 700-2800 Hz for music).

2.8 Clarity

Clarity describes the degree to which every detail of the performance can be perceived as opposed to everything being blurred together by later-arriving, reverberant sound components [12]. Thus, clarity is to a large extent a property complementary to reverberance. When reflections are delayed by no more than 50–80 ms relative to the direct sound, the ear will integrate these contributions and the direct sound together, which means that we mainly perceive the effect as if the clear, original sound has been amplified relative to the late reverberant energy. Thus, an objective parameter that compares the ratio between energy in the impulse response before and after 80 ms has been found to be a reasonably good descriptor of clarity:

$$C = 10\ log\left[\frac{\int_{0}^{80\ ms} g^{2}(t)dt}{\int_{80\ ms}^{\infty} g^{2}(t)dt}\right] \tag{7}$$

The higher the value of C, the more the early sound dominates, and the higher the impression of clarity. C80 and C50 are defined according to the integration time limits used. Figure 2 graphically represents the reflections considered before and after 80 seconds .

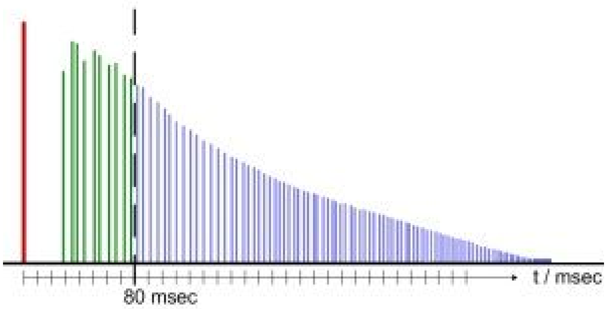

Figure 2: Definition of clarity.

2.9 Ray Tracing Method

The ray tracing method uses a large number of particles, or rays, which are emitted in various directions from a source point. The particles are traced around the room losing energy at each reflection according to the absorption coefficient of the surface. When a particle hits a surface it is reflected, which means that the new direction of the propagation is determined according to Snell's law as known from geometrical optics. This is called specular reflection. In order to obtain a calculation result related to a specific receiver position it is a necessary either to define an area or a volume around the receiver in order to catch the rays when travelling by, or the sound rays may be considered the axis of a wedge or pyramid. There is a reasonably high probability that a ray will discover a surface with an area A after having traveled the time t if the area of the wavefront per ray is not larger that $A/2$. This lead to the minimum number of rays N [13]:

$$\mathrm{N} \geq \frac{t^{2} 8\pi c^{2}}{A} \tag{8}$$

where c is the speed of sound. According to equation 8 a very large number of rays is necessary for a typical room. The result of the simulation process is impulse response in the position of reception. There are many customs sets in a software, as choose the maximum order allowed of the ray, the maximum length (seconds) of the impulse response obtained, etc.

There is another problem in the lowest frequencies. When the acoustic wavelengths are short relative to the surface sizes that take part in the reflection, sound can be modeled as rays of light that behave geometrically. The angle of incidence will equal the

angle of reflection, as with a mirror. As frequency decreases, the light model becomes increasingly inaccurate and the sound must be modeled as a wave, which is much more difficult to handle computationally. The transition of a room from "ray behavior" to "wave behavior" is a gradient with no clear single transition frequency. By an empirical method it could be said that above 4 times the Schroeder frequency the sound wave can be considered as a ray in the specific enclosure [14]. This behaviour is shown in Figure 3. Also, this method has problems on the phase of the wave sound, just because the software does not consider it.

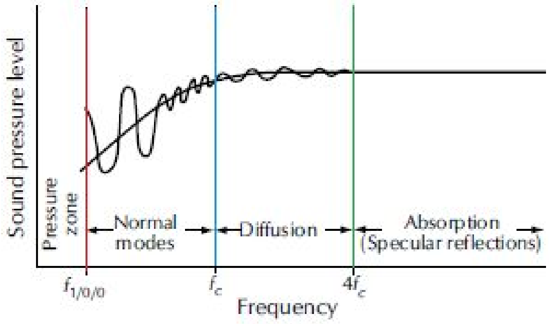

Figure 3: Sound behaviour in an enclosed space.
Source: Sound System Engineering.

The development of a room acoustical ray tracing models started about 50 year ago but the first models were mainly meant to give plots for visual inspection of the distribution of reflections [15]. The method was further developed and in order to calculate a point response the rays were transferred into circular cones with special density functions [16].

2.10 Image Source Method

The image source method is based on the principle of that a specular reflection can be constructed geometrically by mirroring the source in the plane of the reflecting surface. The advantage of the image source method is that it is very accurate, but if the room is not a simple rectangular box there are too many posibles image sources. with n surfaces there will be n possible image sources of first order, and each of these can create $(n-1)$ second order image sources. If this is continued until the reflection order i, the number of possible image sources N will be following equation 9.

$$N = 1 + \frac{n}{n-2}\left[(n-1)^{i} - 1\right] \qquad (9)$$

This means to many image sources, but if the reflection order is limited by compromise consideration, the number of reflection drops and the process calculation is less complicated. For this reason this method is commonly only used for simple rectangular rooms or in cases where low order reflections are sufficient [13].

3. THEATER CHOSEN

Teatro Liceo is located in Capital Federal, Buenos Aires, Argentina. It has been recognized by the city government as the oldest theater in the province [17]. Figure 4, Figure 5 and Figure 6 show pictures of the theater's stage and audience area.

Figure 4: Teatro Liceo, Seat view.

Figure 5: Teatro Liceo, Balcony.

Figure 6: Teatro Liceo, Audience Area.

It was built in 1872 under the name of "*El Dorado*", and subsequently changed its name many times under different ownerships ("*Goldoni*" in 1877, "*Progreso*" in 1893, "*Rivadavia*" in 1907, "*Teatro Moderno*" in 1911, and finally "*Teatro Liceo*" in 1918). In 1986 the Museum of the City of Buenos Aires awarded the theater a diploma, recognizing it as "living testimony of the memory of the city" [18]. Its style can be categorized under "Italian architecture", which reached its peak between 1830 and 1880, representing the transition between the colonial style (inherited from Spanish and Lusitanian traditions) and the academician French style that would dominate the architecture in the country until 1930 [19].

The theater has capacity for 575 people and is intended to be used for theatres operas. The theater´s blueprint and lateral cross section are shown in Figure 7 and Figure 8.

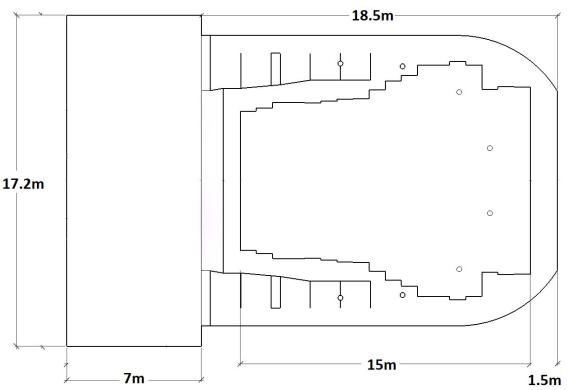

Figure 7: Teatro Liceo, Blueprint.

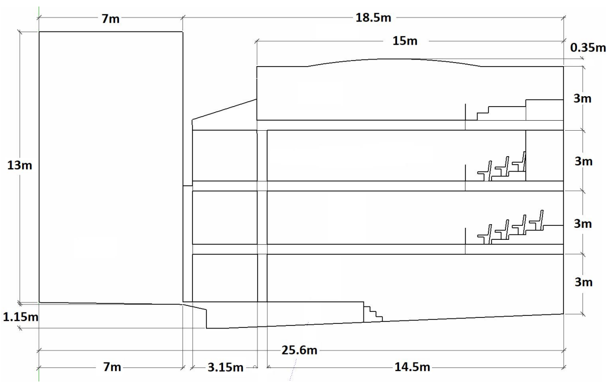

Figure 8: Teatro Liceo, Cross Section.

Liceo is a horseshoe theatre and it has a volume of 4637.5 m^3. The room itself is not a very reverberant enclosure, the lateral walls are probably made with resonant panels covered with carpet to control the low frequencies. All the floors, except the stage one that is a wooden floor, is carpet.The theatre has 3 mains floor of audience, the ground floor, the first floor and the second floor, it has also third floor or *tertulia*; all the audience area has medium upholstered seats and this provides the main control over the medium high frequencies. The stage area is 120.4 m^2 and its volume is 1565.2 m^3. As the Liceo is a italian style theatre there are many ornaments on the balconies front walls and on parts of the walls that provides diffusion.

4. PROCEDURE

In this section, measurement, modelling and simulation procedures are explained in detail.

4.1 Measurements

The measurements were performed in July of 2017. The following instruments were used:

- 16 measurements microphones: Earthworks M50 *(serial numbers: 4210H, .4211H, 4212H, 4213H, 4214H, 4215H, 4216H, 4217H, 4218H, 4219H, 4220H, 4221H, 4222H, 4223H, 4224H, 4225H)*
- 1 audio interface: RME Fireface UFX+ *(serial number: 23771444).*
- 2 audio interfaces RME Octamic XTC *(serial numbers: 23766981, 23755753).*
- Omnidirectional sound source: Outline with subwoofer *(serial number: UOABU001).*
- DAW: Ableton Live 9 digital audio workstation.
- Acoustic calibrator: Svantek SV-30-A *(serial numer: 14258 ,last calibrated in May 5th, 2017).*
- Distance Laser Meter: Bosch DLE 70.

The measurement consisted in acquiring impulse responses of different positions of the theatre, by using a 90 seconds long, log-sine-sweep, which was played through an omnidirectional source, captured by the microphones and recorded with the software. Afterwards, recorded signals were convolved with their inverse filter and finally impulse responses were reconstructed, which are analyzed with software such as Audacity and Easera in order to obtain the required acoustical parameters. Because Liceo theatre is a symmetrical enclosure, all measurement were recorded in the right side of the audience area, so later a symmetrical area mapping extrapolation can be made. From the total data available, 15 Microphone measurement positions were chosen for simulation analysis performed in this paper. Measurement positions are marked in Figure 9.

Figure 9: Measurements locations (red dots)

Two sound sources were used for the measurement procedure, one omnidirectional dodecahedron and one mono speaker. The first one is used to play a log-sine-sweep while the second one is used for the reproduction of pink noise and several musical passages. Their locations are shown in Figure 8 associated to Table 1 coordinates. Impulse responses used in this paper for analysis are gathered from Omni 1 speaker.

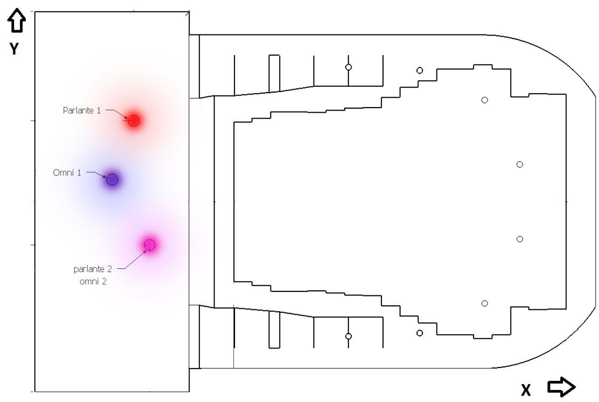

Figure 8: Sound source locations.

Table 1: Sound Source coordinates

SOURCE	X [m]	Y [m]	Z [m]
SPEAKER 1	4.49	12.25	1.20
SPEAKER 2	5.22	6.64	1.20
OMNI 1	3.51	9.60	1.50
OMNI 2	5.22	6.64	1.50

4.2 Sketch Up 3D Computer Model

The computer simulation begins with the building of the theater's 3D model in the SketchUp Pro 2015 software. All available dimensions were previously provided by blueprints made by students who carried out the measurements. Those were taken into account in order to correctly emulate the theatre. Several layers are created according to its surface material.

Various versions of the model had to be made in order to satisfy EASE's "*Check Data*" simulation requirement, each version more complex than the previous one until reaching the point where EASE could not properly read the Sketch File or check data. The final 3D Model chosen for the simulation will be the last version that is able to pass. In this case, the last design obtained is shown in Figure 9 and Figure 10. Model that passed check data is the previous one, shown in Annex.

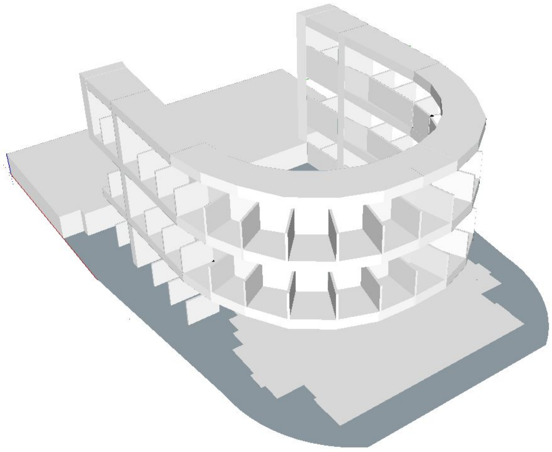

Figure 9: Designed Sketchup 3D model, isometric view.

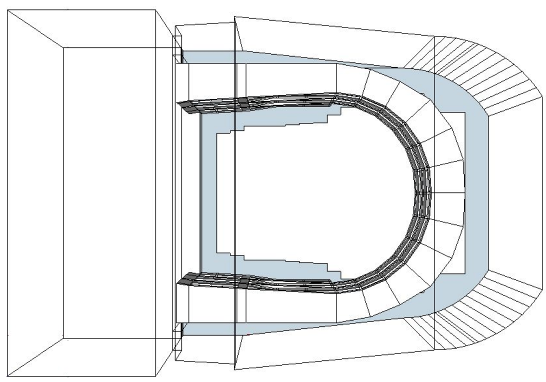

Figure 10: Designed Sketchup 3D model, upper view.

Once this is done, the 3D file is exported to the EASE platform where acoustical simulations can be made.

4.3 EASE Computer Simulation

In this paper, it is intended to use the ray tracing method and area mapping method. In order to do this, all surfaces have to be described with their own absorption coefficients. As this data was not available at the moment of this work, absorption coefficients were selected among various lists, gathered from some simulation libraries [20]. Criteria for material choices was based on detailed descriptions and observations provided by pictures of the theatre. Each material has to be created in EASE with the *"Material Editor"* tool, then loaded into EASE, and applied to each surface accordingly. The materials and coefficients are accommodated after the preliminary results in order to obtain closer RT values to those measured in situ. The process is repeated until each representative octave-band RT value simulated is less than 100 ms apart from the real measurement. Some of the materials tested in this instance are listed below, and their respective absorption coefficient is shown in Table 2.

- Concrete (M01)
- Thin Carpet Over Concrete (M02)
- Thin Carpet On Wood (M03)
- Carpeted Floors (M04)
- Heavy Curtains (M05)
- Tight Velvet Curtains (M06)
- Wooden Chairs (M07)
- Medium Upholstered Seats (M08)
- Light Upholstered Seats (M09)
- Wooden stage floor (M10)

Table 2: Material Absorption Coefficients [%]

F[Hz]	125	250	500	1K	2K	4K	8K
M01	2	2	3	3	4	7	5
M02	10	15	25	30	30	30	30
M03	20	25	30	30	30	30	30
M04	12	17	25	30	30	30	30
M05	30	45	65	56	59	71	71
M06	5	12	35	45	38	36	36
M07	5	8	10	12	12	12	12
M08	44	65	77	89	82	70	70
M09	21	48	62	75	74	74	74
M10	10	7	6	6	6	6	7

The method applied consisted in selecting the materials that represented most of the real surfaces first, and later in making changes on no more than 2 surfaces at a time in order to accurately test its impact in the overall reverberation time. The ray tracing simulation for each material change is the most time consuming procedure in this work, so deliberate decisions are advised. The simulation's reflection order limit was set to 20 because it is considered that rays orders over 20 do not contribute to the impulse response as their energy is comparatively low, while the computational cost is very high. On the other hand, a total of 5.000.000 random rays was configured to be emitted by the source in order to obtain more than 99.99 % impact probability. Finally, simulation time limit was set to 1060 ms, knowing from analysed in situ measurements that most of the expected octave-band RT values are lower than 1.5 s; meaning 700 ms should be enough for T30 calculations. Moreover, reflections after 1 s are considered random Gaussian noise. This could be later corrected by the addition of a white noise tail if it is intended to reduce computational time. In this case, processing power was not a constraint, as the ray tracing simulation was performed using various computers with multiple cores each. The software allows to take advantage of multiple cores inside the CPU in order to process different material choice case scenarios in parallel. At times, up to five simultaneous simulations were able to be calculated using a single eight-core machine.

As previously discussed, ray tracing method has phase issues which the software does not consider. This can generate problems because it recollects all rays regardless the reflection's phase difference or modal room characteristics. For this reason, the Schroeder frequency was taken into account and it was calculated using equation 10.

$$F_S = 2000\sqrt{\frac{RT}{V}} \tag{10}$$

Schroeder frequency was calculated to be 29.4 Hz. Four times F_S is the empirical lowest representative frequency that can be considered as valid in a computer simulation. Therefore, in this case, frequencies below 117.5 Hz could not be taken seriously entirely. The lowest octave-band center frequency analysed in this paper is 125 Hz, which has a minimum band frequency at 88 Hz.

Consequently, 125 Hz octave-band parameter values obtained in this research could be compromised.

Once every impulse response is obtained in every receiver position, the results were analyzed and post processed with EASERA software so that acoustical parameters can be finally obtained. *Room mapping* is used to calculate sound pressure levels in each seat position in order to make a comparison with the in-situ measurement.

On the other hand, *Area Mapping* is used to graphically represent acoustical parameters over a defined audience area, which is elevated 1.2 metres above the ground as it represents listening positions more accurately.

5. RESULTS

In this section, results obtained for each parameter are shown and discussed.

5.1 Simulation: Surface Materials

The first simulation was done using light upholstering seats, carpet on the floor and walls, wooden stage, surrounded by heavy curtains and concrete in the ceilings. Results were surprisingly close to the target right away. However, there were still RT differences larger than 500 ms in some bands. Based on this preliminary results, material variations were done, such as changing to wooden seats, adding velvet curtains, experimenting with different carpets and concrete materials, etc.

More than 10 different ray tracing method simulations were done, each taking between 10 and 14 hours approximately.

After changing the stage side walls to concrete and using tight velvet curtains, TR values became very close to target in low frequency bands, but still more than 100 ms apart in high frequencies. At this point curtains were changed back to heavy, and medium upholstering seats were introduced, resulting in TR values very close to target in high frequencies, but almost 100 ms apart in low frequencies, not considering the 125 Hz band. Since both cases were solid candidates, other parameters values were tested in more detail in order to find which one was the best approximation. Based on this criteria, the last material simulation was chosen. Final materials used for the definitive simulation are shown in Table 3.

Table 3: Materials used in the simulated theatre.

Section	Material
Audience Area	M08
Corridor Floor	M02
Main Laterals Walls	M03
Main Back wall	M03
Stage Floor	M10
Stage Side walls	M01
Stage Back Curtains	M05
Ceiling	M01
Balcony Structure	M01

5.2 Global Values

In the Table 4, the global acoustical parameters obtained by the simulation and the ones obtained by measurement *in situ* can be observed.

Table 4: Simulated vs Measured Global Parameters.

Parameter	EASE	Measurement	Dif
RT [s]	0.89	0.96	0.07
EDT[s]	1.16	0.86	0.3
STI	0.60	0.67	0.07
ALcons [%]	6.75	4.52	2.23
Echo-Speech	1.00	0.61	0.39
Echo-Music	1.03	0.51	0.52
D/R [dB]	-3.95	-6.12	2.17
C80 [dB]	3.55	5.89	2.34
SPL [dB]	81.03	81.02	0.01

5.3 Reverberation Time

In the Table 5, octave bands reverberation time obtained by the simulation and the ones obtained by measurement *in situ* can be observed.

Table 5: Reverberation time [s]

F [Hz]	Simulation	Measurement	Difference
125	0.98	1.22	0.24
250	0.99	1.06	0.07
500	0.91	1.01	0.10
1000	0.95	0.97	0.02
2000	0.91	0.92	0.01
4000	0.81	0.83	0.02
8000	0.68	0.69	0.01

For a clearer understanding, a graphic comparison is made in the Figure 11.

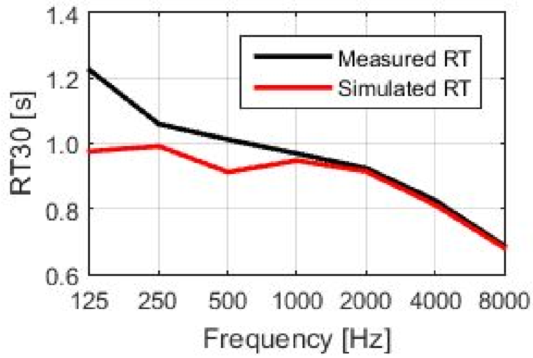

Figure 11: Measured RT vs simulated RT.

5.4 Early Decay Time

In Table 6, the early decay time by octave bands obtained by the simulation and the ones obtained by measurement *in situ* can be observed.

Table 6: Early decay time. [s]

F [Hz]	Simulation	Measurement	Difference
125	1.43	0.98	0.45
250	1.36	0.93	0.43
500	1.21	0.94	0.27
1000	1.19	0.90	0.29
2000	1.15	0.89	0.26
4000	1.02	0.77	0.25
8000	0.77	0.63	0.14

For a clearer understanding, a graphic comparison is made in the Figure 12.

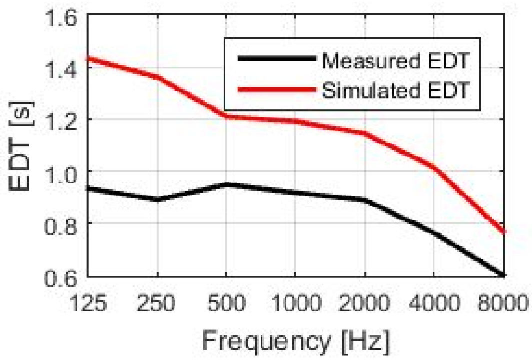

Figure 12: Measured EDT vs simulated EDT.

5.5 D/R Ratio

In the Table 7, the direct to reverberant ratio by octave bands obtained by the simulation and the ones obtained by measurement *in situ* can be observed.

Table 7: D/R Ratio. [dB]

F [Hz]	Simulation	Measurement	Dif
125	-2.79	-4.71	1.92
250	-2.25	-3.42	1.17
500	-0.78	-5.20	4.42
1000	-4.89	-7.01	2.12
2000	-5.81	-8.74	2.93
4000	-9.01	-8.69	0.32
8000	-8.19	-8.29	0.1

For a clearer understanding, a graphic comparison is made in the Figure 13.

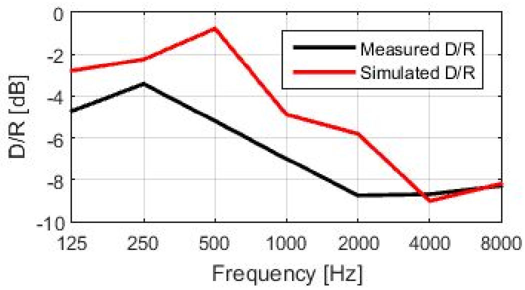

Figure 13: Measured D/R vs simulated D/R.

On the other hand, D/R ratio was obtained using area mapping simulation of the audience area for the 1 KHz octave band, shown in Figure 14.

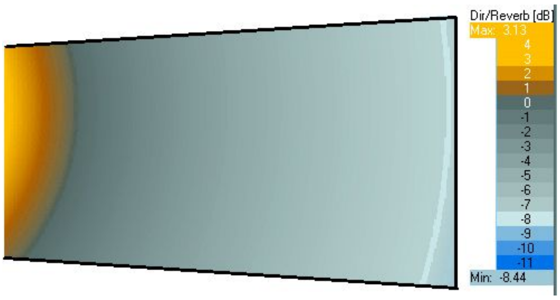

Figure 14: D/R Ratio, Area Mapping, 1 KHz

5.6 Clarity

In the Table 8, clarity parameter by octave bands obtained by the simulation and the ones obtained by measurement *in situ* can be observed.

Table 8: Clarity. [dB]

F [Hz]	Simulation	Measurement	Dif
125	-2.65	4.81	7.46
250	-0.58	5.01	5.59
500	2.38	4.93	2.55
1000	3.00	5.47	2.47
2000	3.21	5.36	2.15
4000	4.77	7.09	2.32
8000	7.28	8.57	1.29

For a clearer understanding, a graphic comparison is made in the Figure 15.

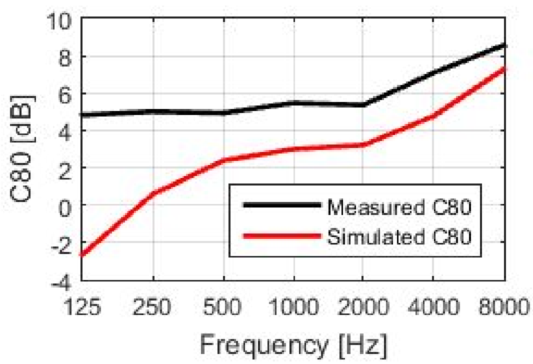

Figure 15: Measured C80 vs simulated C80.

On the other hand, C80 was obtained using area mapping simulation of the audience area for the 1 KHz octave band, shown in Figure 16.

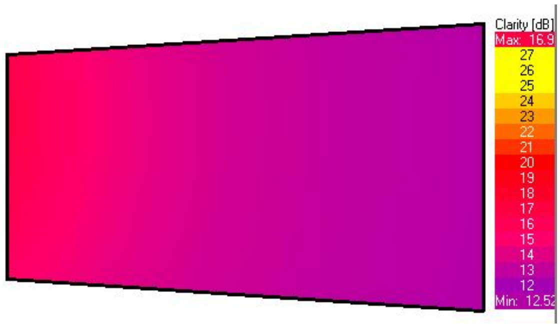

Figure 15: C80, Area Mapping, 1 KHz

5.7 Sound Pressure Level

Global Sound pressure level values are obtained by room mapping simulation for each seat analysed. Comparison between Simulated and Measured levels are shown in Table 9.

Table 9: Sound Pressure Level [dB]

SEAT	Simulated	Measured	Difference
0° S1	83.22	74.76	8.46
0° S03	83.26	78.81	4.45
0° S09	81.79	73.85	7.95
0° S16	80.75	75.85	4.90
0° S19	80.33	79.90	0.43
1° S04	81.17	79.54	1.63
1° S08	80.38	77.09	3.29
1° S10	80.25	78.08	2.17
1° S12	80.14	76.70	3.44
2° S02	81.00	78.63	2.37
2° S05	80.41	82.11	-1.70
2° S14	80.21	81.92	-1.71
2° S15	80.15	83.20	-3.05
3° S06	80.12	87.97	-7.85
3° S12	80.28	81.77	-1.49

Figure 16 shows simulated sound pressure levels for every floor, in octave bands, using room mapping method calculations.

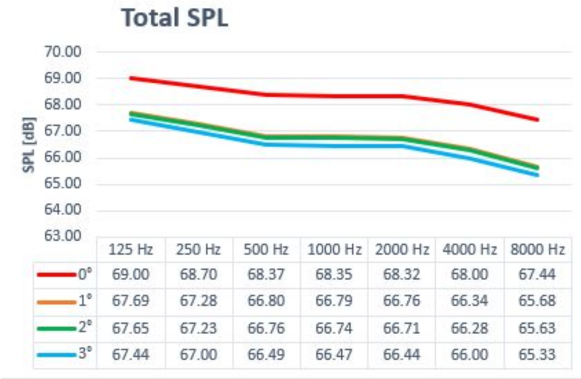

Figure 16: Simulated Total SPL for each floor

On the other hand, an area mapping was performed in order to obtain the total SPL and Direct SPL in the audience for the 1 KHz octave band. Result is shown in Figure 17 and Figure 18.

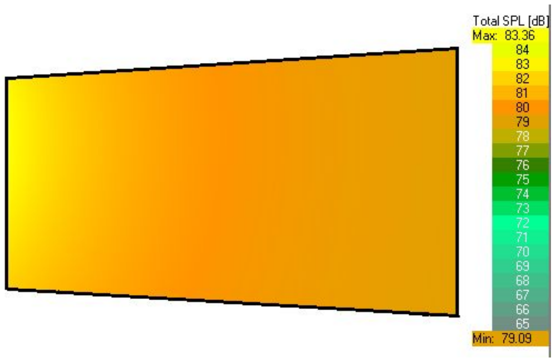

Figure 17: Total SPL. Area Mapping, 1 KHz

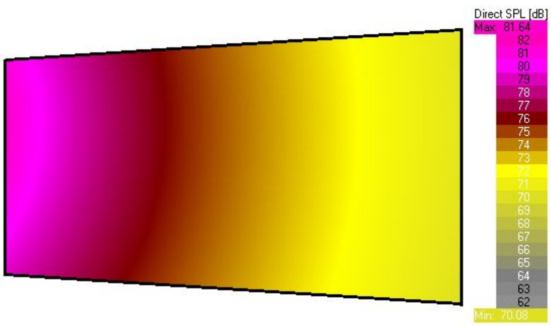

Figure 18: Direct SPL. Area Mapping, 1 KHz

By using the area mapping method, it can be calculated the time delay for the first arrival reflection for every point in the audience área. Resulting mapping is shown in Figure 19.

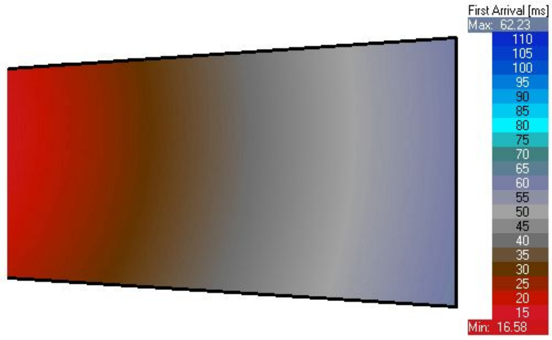

Figure 19: First Arrival Delay, Area Mapping, I KHz

5.8 STI & ALcons

STI and ALcons area mappings were simulated for the 1 KHz octave band; results are shown in Figure 20 and Figure 21.

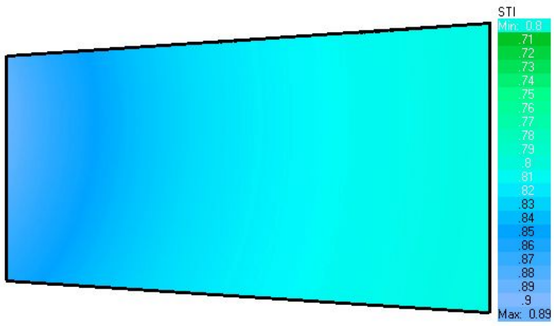

Figure 20: STI. Area Mapping, 1 KHz

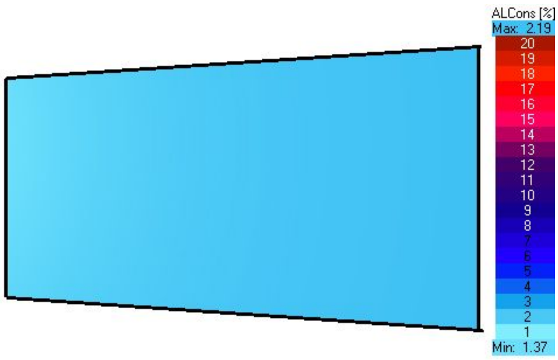

Figure 21:ALcons. Area Mapping, 1 KHz

6. DISCUSSION

6.1 Reverberation time

Results obtained for reverberation time are considered to be the most precise; this is because the model was adjusted to match the reverberation time obtained in the real measurements. As stated in previous sections, Ease miscalculates low frequencies, this simulation is no exception as the largest RT difference is found in the 125 Hz octave band. For frequencies above 1 kHz, the difference is minimal (less than 2.5 %).

According to Barron [21], reverberation time of around 1.0 seconds are generally recommended for speech use, although shorter values, such as 0.8 seconds are also common. These short RT values are achieved either by having small room volumes or by installing an absorbent material with careful suspended ceiling design. This value is in agreement with the results presented by Long [22], shown in Figure 22 (variations of 5 to 10 % from the ideal values are commonplace). For a volume of 4637.5 m³, recommended reverberation time turns out to be 0.8 to 1.2 s.

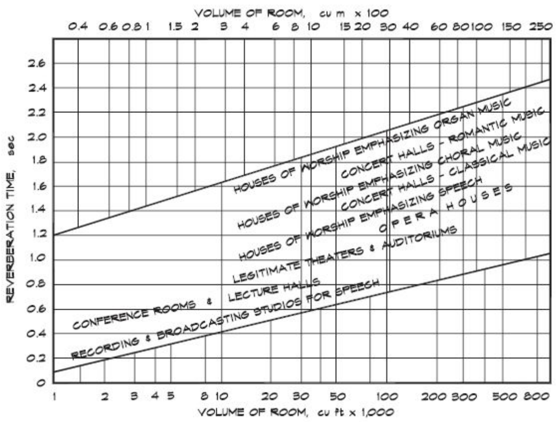

Figure 22: Recommended reverberation time for different rooms. Source: Architectural Acoustics.

6.2 Early decay time

Results for early decay time on the simulated model are significantly higher than the ones obtained through real measurements (around 20 % higher for medium-high frequencies). Nevertheless, it is observed that the both curves follow the same tendency: approximately constant from 500 to 2000 Hz and decreases linearly from 2000 to 8000 Hz.

The differences of this comparison can be associated, in one part, to the absence of scattering surfaces on the simulation.

6.3 Intelligibility

Results obtained for intelligibility measurements (both STI and %ALcons) are fairly accurate.

According to Carrion [23], the subjective valuation of the corresponding STI and ALcons values are shown in Table 10.

Table 10 : Subjective valuation for intelligibility

ALcons [%]	STI	Valuation
0 - 1.4	0.88 - 1	Excellent
1.6 - 4.8	0.66 - 0.86	Good
5.3 - 11.4	0.50 - 0.64	Acceptable
12.0 - 24.2	0.36 - 0.49	Poor
27.0 - 46.5	0.24 - 0.34	Bad

Therefore, results obtained from real measurement would be in the lower end of good speech intelligibility, while results obtained from the simulation would be situated in the higher end of acceptable speech intelligibility.

Even though these values are good, they could be slightly improved since speech intelligibility is a primary concern in theater acoustics.

6.4 Echo Criterion

By numerous subjective tests, carried out both with synthetic sound fields and in real halls, Dietsch and Kraak [11] determined the critical values EC_{crit}, which must not be exceeded to ensure that not more than 50% of the listeners will hear an echo. These recommended values are shown on table 11:

Table 11: Recommended echos values.

Parameter	EC critical
Echo Speech	1.0
Echo Music	1.8

It is observed that, even though the results of echo speech and echo music obtained through the simulation are significantly higher than the real ones, they both comply with the echo criterion.

6.5 D/R Ratio

According to Long [22], a direct-to-reverberant ratio of 0 dB is almost never achieved indoors, and even a ratio greater than -3 dB is rare. The direct-to-reverberant ratio is related to speech intelligibility, as shown in Table 12. Values between –3 and –6 yield very satisfactory intelligibility. Normally it is sufficient to design a room for good intelligibility and a small number of fair seats, particularly at the sides and rear, are tolerable.

Table 12: D/R Ratio and speech intelligibility.

D/R [dB]	Intelligibility
> -3	Excellent
-3 to -6	Very good
-6 to -9	Good
-9 to -12	Fair
-12 to -15	Poor
< -15	Very poor

Even though there is a slight difference of 2 dB between the real global values and the simulated ones, they are both close to very good intelligibility. This is not in agreement with the results obtained from STI/ALcons intelligibility parameters; this difference can be attributed to the different ways in which direct and reverberant are defined. For this project, this parameter was obtained setting the EASERA markers from the absolute maximum to the end of the impulse response.

6.6 Clarity

Results for clarity on the simulated model are significantly lower than the ones obtained through real measurements (around 20 % lower for medium-high frequencies). Nevertheless, it is observed that the both curves follow the same tendency for medium-high frequencies: approximately constant from 500 to 2000 Hz and increases linearly from 2000 to 8000 Hz. Octave band of 125 Hz shows the largest difference, considered attributed to simulation miscalculations, as previously discussed.

7. CONCLUSIONS

In first instance, an acoustical simulation by computer can be a excellent resource when a enclosure is going to be built. The simulation provides engineers and architects a tool to know how different materials configurations and shape arrangements interact and how they should be in order to obtain the desired performance of the enclosure. Because the software allows to set transmitter and receiver locations, a good mapping of the theatre can be made. However, the ray tracing method is a simulation with some flaws, meaning that some acousticals parameters should be carefully examined or ignored depending on the octave band.

On the other hand,, when the acoustical simulation is made based on an already constructed theatre, a different methodology is applied. Often, it is very hard to know every single surface materials properties and the characteristic of the audience area. Sometimes the only available information is gathered by visual inspection, which forces the designer to speculate absorption values, which ultimately compromises the resulting parameter values. Moreover, material's age and time of use is often unknown as well, which adds another layer of uncertainty. All this considerations can make a simulation somehow imprecise.

In this paper, a computer simulation was successfully carried out based on a already built Liceo theatre. The acoustical parameters were obtained by the simulation and later compared with the real measurements. After differents simulations, the finals results seems to be close enough compared with the originals measures, especially for reverberation time and intelligibility parameters (STI and %ALcons). These results are in agreement with the ones found by Field [3]. Nevertheless, it could be observed that the main differences are located in the lower frequencies.

Future work

A good point to be analyzed in the future is the comparison of the complete simulation and a simulation with a tail of gaussian noise. It is known that after the crossover time the impulse response begin to be random and that the impulse response reverberation tail can be characterized as exponentially decaying Gaussian white noise. [24][25] This configuration could be a good tool when a simulation is made because it shortens the computational time, but in this paper it was not necessary to do it, since there was enough time to do a complete process.

Another point that can be evaluated in the future is to do the simulation taking into account the diffusion coefficient of all the materials. This approach the simulation even closer to the reality and the precision of the obtained parameters can be more accurate. Furthermore, as a future topic a comparison between many source positions and the same receiver point can be made, doing this, probably differents results can be derived and later a more detailed performance of the room can be obtain.

Finally another good subject to be evaluated in the future is to make a comparison between the different RT obtained by the simulation and the ones that can be calculated by the sabine's equation.

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9. ANNEX

A comparison between 6 of the 10 different 3D made models can be observed on Table 13. For a simplified understanding, only the reverberation time is compared.

Table 13: RT Comparison Between the 3D Models

Frq.band [Hz]	125	250	500	1k	2k	4k	8k	GLOBAL
REAL T30 [s]	1,22	1,06	1,01	0,97	0,92	0,83	0,69	0,96
1 - 05 T30	0,95	0,96	0,91	0,95	0,92	0,81	0,68	0,88
2 - 06 T30	1,01	1,07	1,02	0,99	0,97	0,92	0,78	0,97
3 - 07 T30	0,98	0,99	0,91	0,95	0,91	0,81	0,68	0,89
4 - TK T30	0,94	0,98	0,93	0,95	0,92	0,82	0,68	0,89
5 - 00 T30	0,98	0,93	0,83	0,87	0,83	0,74	0,65	0,83
6 - 04 T30	1,01	0,94	0,83	0,87	0,83	0,75	0,64	0,84

REAL T30 [s] shows target values corresponding to in situ measurements in Teatro Liceo. 05 T30 Corresponds to the first simulation previously detailed, but changing stage side walls to concrete. 06 T30 is an improved version of 05 T30, incorporating velvet curtains. 07 T30 is the final version, which incorporates medium upholstered seats instead of light ones. TK T30 is a 05 T30 model which incorporates cotton over wood for the main walls. 00 T30 corresponds to the first simulation values, and finally 04 T30 shows RT values for a 00 T30 configuration just changing seats to medium upholstering.

Both model 06 and 07 were solid choices as previously explained, However, even though model 07 was the chosen because other acoustical parameters than RT, the chart shows that model 06 had the closest global reverberation time to the target measurement.

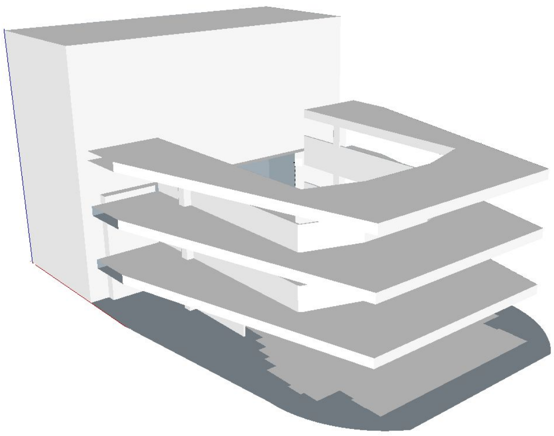


Figure 23: 3D Model validated by EASE and used for simulations. Isometric view.

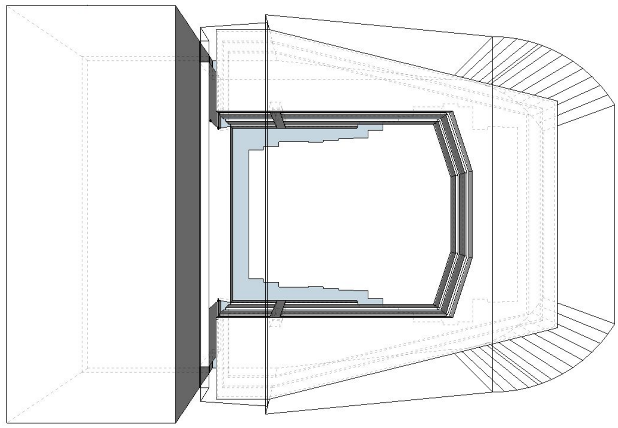


Figure 24: 3D Model validated by EASE and used for simulations. Upper view.

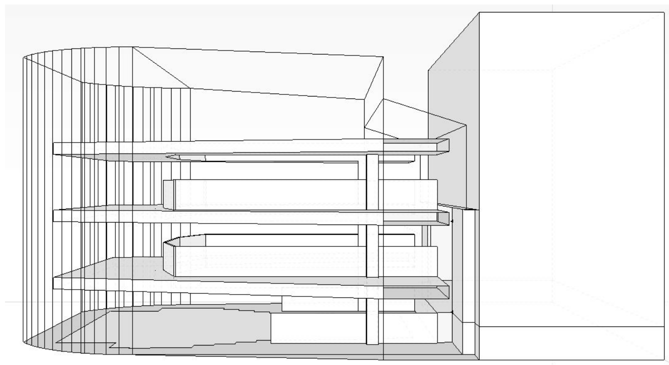
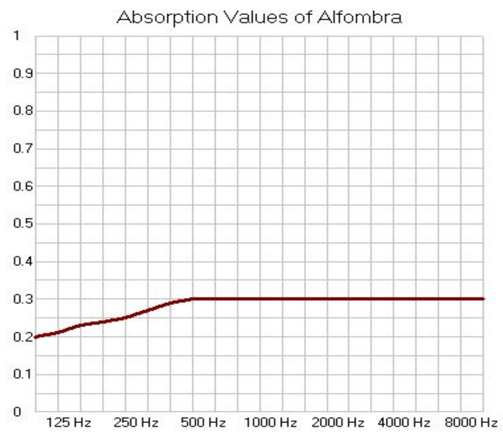


Figure 25: 3D Model validated by EASE and used for simulations. Side view.

Figure 24: 3D Model validated by EASE and used for simulations. Audience view.

The following Figures (next page) show the materials absorption coefficient curve extrapolated to third octave bands, used for the acoustical simulation.


Absorption Values of Alfombra
1
0.9
0.8
0.7
0.6
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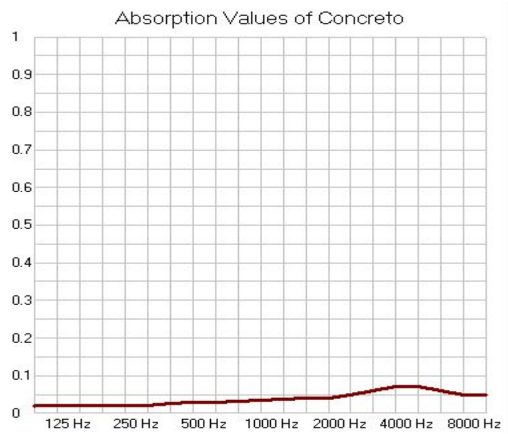
0.3
0.2
0.1
0
125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz


Absorption Values of Concreto
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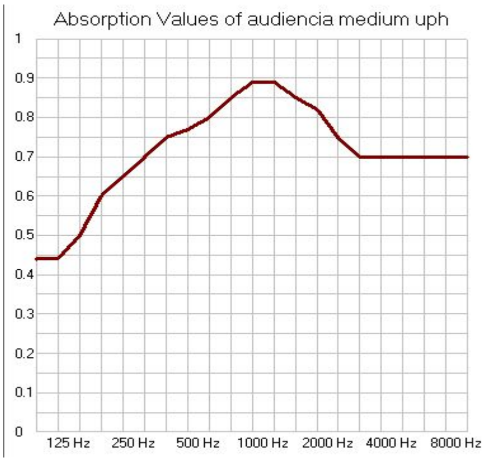
125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz


Absorption Values of audiencia medium uph
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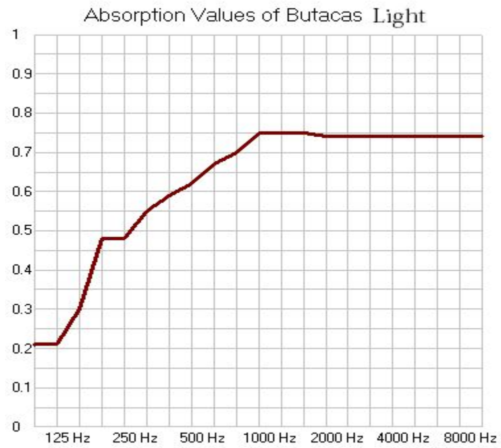
125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz


Absorption Values of Butacas Light
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0.9
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0.6
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0.4
0.3
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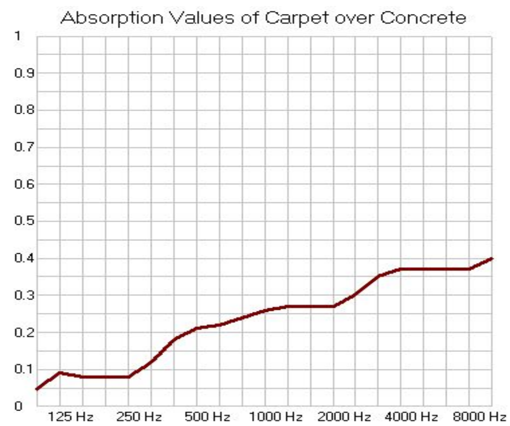
125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz


Absorption Values of Carpet over Concrete
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0.9
0.8
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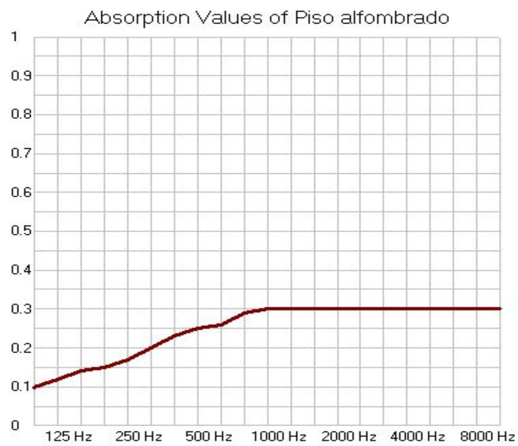
0.6
0.5
0.4
0.3
0.2
0.1
0
125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz


Absorption Values of Piso alfombrado
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0.9
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0.6
0.5
0.4
0.3
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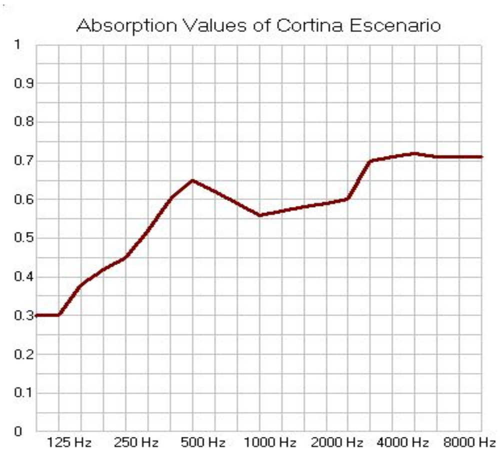
125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz


Absorption Values of Cortina Escenario
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0.6
0.5
0.4
0.3
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0.1
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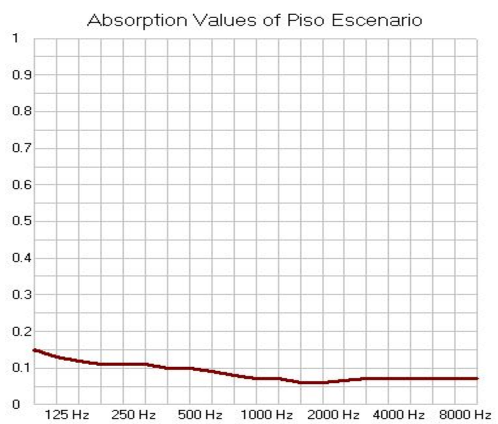
125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz


Absorption Values of Piso Escenario
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0.6
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0.4
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125 Hz 250 Hz 500 Hz 1000 Hz 2000 Hz 4000 Hz 8000 Hz