CONCERT HALL AUDITORIUM GEOMETRICAL ACOUSTICS COMPUTER MODELLING

Franco Daniel Areco
Universidad Nacional de Tres de Febrero, Ingeniería de Sonido, Caseros, Buenos Aires, Argentina
email:francoareco53@gmail.com

Juan Martín Rucci
Universidad Nacional de Tres de Febrero, Ingeniería de Sonido, Caseros, Buenos Aires, Argentina
email:rucci45744@estudiantes.untref.edu.ar

Juan Almaraz
Universidad Nacional de Tres de Febrero, Ingeniería de Sonido, Caseros, Buenos Aires, Argentina
email:juan.almaraz097@gmail.com

Calquin Epullan
Universidad Nacional de Tres de Febrero, Ingeniería de Sonido, Caseros, Buenos Aires, Argentina
email:epullan44186@estudiantes.untref.edu.ar

This article presents a comprehensive report detailing the geometric and acoustic modeling of the Niccolo Paganini auditorium, located in the City of Parma, Italy, using EASE software. The acoustic parameters investigated include STI, ALcons, T20, T30, EDT, Echo Speech, Echo Music, SPL, and D/R. These parameters were calculated and compared with in-situ measurements conducted within the auditorium. Our findings reveal differences that fall under the stablished JND between the simulated and in-situ measurements, particularly in the case of T20, T30 and STI, However, there is a discrepancy on the results when analyzed by third octave bands. A 3 JND was obtained for the EDT, and 2 JND for the D/R parameter. For the ALcons parameter a -0.87 difference was found, and for Echo speech and Echo music an increase of 49.6% and 66.1% respectively was registered.

Keywords: EASE, geometric acoustics, modeling and auralization

1. Introduction

The field of room acoustics involves examination of sound in various environments and architectural spaces, be it for music or speech [1]. Contemporary advancements in simulation software help architects and engineers to predict and optimize acoustic characteristics within buildings.

Concert hall behavior has been extensively studied by experienced professionals such as Long [1] and Beranek [2], driving innovations in room acoustic modeling techniques. This article's primary objective is to evaluate EASE software [3] simulation results against in-situ measurements conducted at the Niccolo Paganini auditorium, situated in the City of Parma, Italy. The comparison centers on acoustical parameters related to reverberation time and speech and music intelligibility, employing the just noticeable difference (JND) as a parameter. The simulation conditions closely match the measurement conditions, ensuring a reliable basis for comparison.

2. State of the art

Research on the field of room acoustics saw significant advances with the pioneering work of individuals like W. Sabine, L. Beranek, or P. Eyring. Sabine's groundbreaking experiments in the early 20th century laid the foundation for the field by introducing the concept of reverberation time [4]. He developed a formula to calculate this parameter by measuring the time it took for sound to decay to a specific level after a sound source was turned off. This work formed the basis for architectural acoustics and the design of concert halls and auditoriums. Paul Eyring's research further expanded our understanding of room acoustics, including the effects of materials and shapes on sound propagation [5]. Leo Beranek, in collaboration with Harris Norris, contributed extensively to the field by developing practical methods for acoustic measurements and providing guidelines for room design to achieve desirable acoustic qualities [2].

Today, with contemporary computational power, several room acoustic simulation softwares and techniques were developed and expanded our understanding of the field. Several articles have studied the results obtained from computer simulations of the acoustics characteristics of rooms and compared them with those obtained from in-situ measurements of these spaces. According to a study conducted by Mahjoob, M. in 2008 [6], the major factor causing significant error is the size of space taken for acoustic measurements.

In a different investigation [7], where a round robin was employed to compare different acoustic simulation software packages, a key observation was made regarding the primary flaws of these algorithms. It was revealed that these deficiencies were mostly due to assumptions and constraints made when modeling the room. These assumptions result in an oversimplification of absorption and diffusion caracteristics of the surfaces.

These acoustic simulation softwares follow two main room simulation methods: image source models and ray tracing methods. The image source method is based on the premise that a specular reflection can be generated by geometric construction by replicating the position of the source in the plane of the reflecting surface [8]. In contrast, the ray tracing method employs a multitude of particles emitted from a source point in various directions, with each particle following a trajectory within the room and experiencing an energy loss at each reflection, determined by the absorption coefficient of the surface it encounters [9]. Subsequently, a hybrid model was introduced to take advantage of the strengths of both approaches [10]. The central concept is to efficiently identify imaging sources with high probabilities of validity by tracing rays from the source and recording the surfaces they intersect.

In contemporary acoustic modeling, more computationally intensive techniques such as the finite element method (FEM), the boundary element method (FEM) and finite difference time domain methods (FDTD) have emerged and are increasingly being developed [11].

3. Theoretical framework

3.1 Reverberation Time (*RT*)

The Reverberation Time (RT) is defined as the period required for the sound intensity to decrease by 60 decibels (dB) after the interruption of the sound source. This measurement standard, which uses a 60 dB reduction as a reference, provides an objective metric for evaluating the acoustic characteristics of a room. The RT is essential for the characterization of room acoustics as it influences the sound quality in various applications such as concert halls, recording studios, and classrooms.

To obtain the Reverberation Time T_{30} and T_{20}, dynamic ranges less than 60 dB are considered and extrapolated from T_{60}. T_{30} is defined as the time it takes for the sound decay curve to drop from -5 dB to -35 dB below the initial level, while T_{20} is defined as the time it takes to drop from -5 dB to -25 dB.

These reverberation time parameters provide a detailed understanding of how sound behaves in a space and are essential in the design and evaluation of acoustical environments for various applications.

It is important to note that when the sound decay is linear, i.e. when the decay curve of the sound intensity follows a constant path, the values of T60, T20 and T30 are identical. This means that if one knows the value of T60 in a given room, one can accurately infer the values of T20 and T30, which greatly simplifies the process of acoustic measurement and analysis. This phenomenon is clearly reflected in Figure 1 , where the equality of these parameters is observed when the linear decay condition is met.

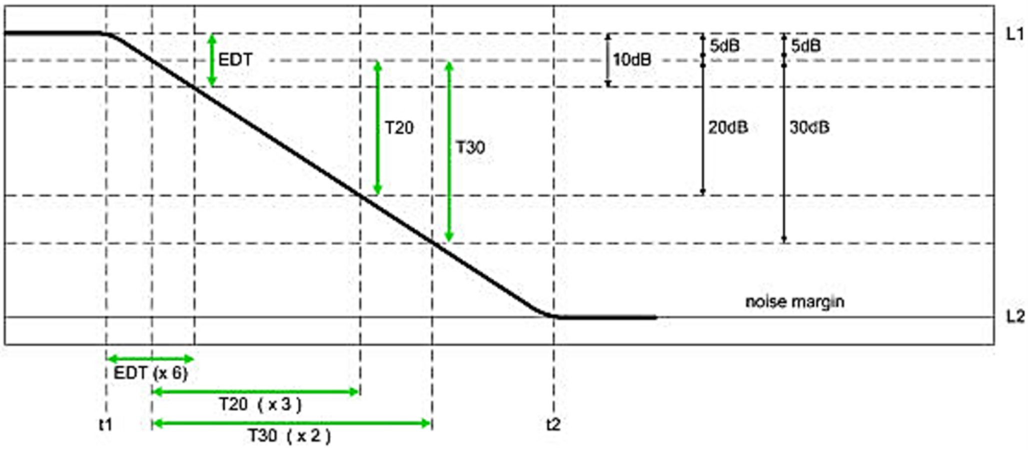


Figure 1: Diagram illustrating the comparison of EDT, T20, and T30.

3.2 Speech Transmission Index (*STI*)

Speech Transmission Index is a magnitude related to the speech intelligibility in a room. It is obtained through the determination of the modulation transfer function (MTF), and it is calculated for octave bands from 125 Hz to 8 kHz. The STI descriptor takes into account band-pass limiting, noise, reverberation, echoes and nonlinear distortion. Then, returns a value from 0 to 1 whose subjective meaning can be interpreted from Table 1. More details and specifications of this descriptor can be found in the standard IEC 60268-16 [12].

Table 1: STI values graded on a subjective scale.

Intelligibility	STI value
Poor	0 to 0.3
Satisfactory	0.3 to 0.45
Good	0.45 to 0.6
Very Good	0.6 to 0.75
Excellent	0.75 to 1

3.3 Articulation Loss of Consonants (*AlCons*)

Like the STI, the AlCons is also a parameter related to speech intelligibility. This descriptor is an objective measurement of the loss of the sounds of the consonants, and it is obtained according equation 1 [13].

$$Al_{Cons} = 200 \cdot \frac{T_{60}^2 \cdot r^2}{V} + \alpha \tag{1}$$

Where T_{60} is the calculated reverberation time of the room, V is the volume of the room, r is the distance between the listener and the sound source, and α is a correction factor. This parameter is also tied to a subjetive scale as shown in Table 2.

Table 2: AlCons values graded on a subjective scale.

Intelligibility	STI value
Ideal	$\leq 3\%$
Very good	3 to 8%
Good	8 to 11%
Poor	11 to 20%
Worthless	$\geq 20\%$

3.4 Echo Speech and Echo Music

Echo Speech and Echo Music are two descriptors formulated by Dietsch and Kraak in 1986 [14] in order to determine whether a reflection or set of reflections are perceived as an echo. The echo criterion is given by equation 2

$$EK(\tau) = \frac{t_s(\tau) - t_s(\tau - \tau_E)}{\Delta\tau_E} \tag{2}$$

where $\Delta\tau_E$ is the critical time interval over which the ratio of increase of energy is analysed (9 ms for speech and 14 ms for music), and t_s is a modified version of the center time formula, as it appears in equation 3

$$t_s = \frac{\int_{t=0}^{\tau} t|p(t)|^n dt}{\int_{t=0}^{\tau} |p(t)|^n dt} \tag{3}$$

where $p(t)$ is the amplitude of the sound pressure over time and the exponent n is given by the values found in Table 3, which also includes the echo criterion thresholds for speech and music beyond which the reflections are perceived as a disturbing echo. There are two values for each motif, where the lower (stricter) ones (EK [10%]) apply for people with trained hearing.

Table 3: Echo threshold according Dietsch and Kraak's criterion.

	Bandwidth of test signal [Hz]	n	$\Delta\tau_E$ [ms]	EK [10%]	EK [50%]
Speech	700-1400	2/3	9	0.9	1.0
Music	700-2800	1	14	1.5	1.8

3.5 Lateral Fraction (*LF*)

The lateral fraction is an indicator introduced by Barron and Marshall [15] and it quantifies the amount of lateral sound energy that arrives at a seat in a hall. This parameter can be obtained measuring the arriving energy using a figure-8 microphone and an omnidirectional one, with an integration time of 5-80 ms and 0-80 ms respectively. Then, following equation 4, LF can be obtained.

$$LF = \frac{\int\limits_{5ms}^{80ms} p_8^2(t)dt}{\int\limits_{0ms}^{80ms} p^2(t)dt} \tag{4}$$

where $p_8(t)$ is the pressure measured by the figure-8 microphone with its null axis pointed towards the source and $p(t)$ is the pressure measured by an omnidrectional microphone at that same point. The bidirectional microphone picks up only the lateral sound waves while the other transducer captures all the acoustic energy that gets to the measurement point.

3.6 Direct to Reverberant Energy Ratio (*D/R*)

The direct to reverberant ratio (D/R) expresses the ratio, at a given location, of the sound pressure level of a direct sound emitted by a directional source to the sound pressure level of the reverberated sound originating from the same source. The D/R is calculated through equation 5. The distance from the source at which this parameter is 0 dB (i.e., the direct sound and reverberant sound pressure levels are equal) is called the critical distance.

$$D/R\,[\text{dB}] = 10 \cdot log\left(\frac{\int\limits_{0}^{q} p^2(t)dt}{\int\limits_{0}^{\infty} p^2(t)dt}\right) \tag{5}$$

where $p(t)$ is the sound pressure and q indicates the time where the direct sound ends.

3.7 Just Noticeable Difference (*JND*)

The just noticeable difference (JND) is a threshold that indicates the magnitude of the minimum variation required to subjectively detect a change in some acoustical parameter. They provide guidance on the accuracy with which objective acoustic parameters should be measured and establishes the accuracy with which computational models should be able to simulate an enclosure.

Standard ISO 3382-1 [16] proposes JND values for EDT and Lateral Fraction. In the case of RT, the results obtained in a study by Blevins et al. [17] were used, in which the JND was found to be 24.5%. For the STI, the results obtained by Bradley J et al. [18] were used, and for the direct to reverberant energy ratio, results obtained by Werner, S. [19]

Table 4 shows the JND values used to evaluate the results of the acoustic parameters obtained in this work.

Table 4: JND values for RT, EDT, STI, LF and D/R.

Parameter	Frequency average	JND	Typical Range*
RT	500 to 1000 Hz	24.5%	1.0 to 3.0 s
EDT	500 to 1000 Hz	Rel. 5%	1.0 to 3.0 s
STI	-	0.01	0.03
LF	125 to 1000 Hz	0.05	0.05 to 0.35
D/R	-	2 to 20 dB	2 dB (D/R = 0 dB) 20 dB (D/R = 20 dB)

*Frequency average values in single positions in non-occupied concert halls.

3.8 Absorption and Scattering

When a sound wave reaches a surface on its way, some of its energy is absorbed. Within the phenomenon of absorption two types can be described: a porous absorption and another reactive absorption [20]. Porous absorbers are materials where sound propagation occurs in a network of interconnected pores that generate tortuous paths for the acoustic wave. When passing through this type of material, the acoustic energy is dissipated as it has been transformed into thermal energy. Instead, reactive absorption is achieved by resonance phenomena (by membranes or cavities). In this way it is possible to obtain absorption at low and medium frequencies, where absorption is difficult to achieve with porous materials because, for reasons of the size of wavelengths at said frequencies, very large thicknesses and sizes of porous material would be needed. The absorption coefficient is represented by α and is measured at $0 < \alpha < 1$, with 1 being the maximum absorption. The scattering coefficient (s) of a surface is the ratio between reflected sound power in non-specular directions and the total reflected sound power [20]. This coefficient may take values between 0 and 1, where s = 0 means purely specular reflection and s = 1 means that all reflected power is scattered according to some kind of ideal diffusivity. Diffuse reflections can be simulated in computer models by statistical methods [10]. By comparison of computer simulations and measured reverberation times in some cases where the absorption coefficient is known, it has been found that the scattering coefficient should normally be set to around 0.1 for large plane surfaces and to around 0.7 for highly irregular surfaces. The correct simulation of the diffusion phenomenon and therefore the configuration of the diffusion coefficient in the modeling have an impact on the calculation time of the software as well as on the results of the acoustic parameters obtained.

4. Procedure

4.1 Description of the concert hall and modeling

The auditorium used for this work is the Niccolo Paganini auditorium, designed by Renzo Piano and located in the city of Parma, Italy. It was built on the remains of a 19th century factory, and was inaugurated in 2001. It has a seating capacity of 780 and a total floor area of 7580 m^2. The main areas of this auditorium consist of a single audience area, a foyer, a stage area and three rehearsal rooms.

The Niccolo Paganini Auditorium was built inside the former Eridania sugar factory, an industrial complex consisting of several buildings of different types. The factory complex is located near the historical center of Parma, in a park populated by native trees and shrubs. The conversion of the factory into an auditorium was possible thanks to the original dimensions of the buildings, which made it possible to comfortably install all the equipment, and also to the particular location of the site - in the middle of the park - which simplified the soundproofing of the auditorium.

The existing facades were replaced by large glazed walls that provide light and views of the entire building, and even from the seats in the auditorium one can enjoy magnificent views of the surroundings. A system of panels covered with acoustic insulation hanging from the beams completes the spatial organization of the interior.

The empty shell of the refinery turned out to be the perfect "music box" for the front stage, both in terms of size and volume. New foundations were laid, old walls were strengthened and the roof was reinforced with trusses. Three floor-to-ceiling acoustic glass walls were installed in place of the old transverse partitions: two placed inside at the ends to enclose the building on both sides, and the third as a divider separating the hall from the vast open entrance area on the south side, which includes the ticket office and the two-level lobby. The transparency of the glass turned the 90-meter long auditorium into a single spatial unit. Under a large holding roof that sits softly against two long walls is a 780-seat concert hall with slightly sloping seating. The large raised stage acts as a natural resonance chamber and creates a quiet backdrop to the alternating views of the park and its tall trees behind the glass wall. The natural acoustic properties of the room are complemented by cherry wood panels suspended from the ceiling beams above the stage, glass covers attached to the glass wall of the building and wooden panels placed behind the orchestra.

The public enters from the south end of the building and then crosses a covered patio to access a double-height vestibule leading to the large concert hall. The stage, facing north and with a surface area of 250 m^2, can accommodate large orchestras and choirs.

In relation to the information concerning the Niccolo Paganini Auditorium, several graphic documents were provided by architect ***Erika Arellano***. These documents included the layout, the general ground floor, as well as a side elevation, a front elevation and a longitudinal and cross section of the auditorium. Architect Arellano's work consisted of obtaining the original plans, followed by the task of accurately redrawing them, incorporating precise measurements and appropriate graphic scales, work that was carried out within the context of the Drawing II course, which is part of the Architecture curriculum at the Pontificia Universidad Católica del Ecuador. The project was carried out during 2017 as part of the academic requirements of that course and the front view, side view and plan view of the auditorium that was used for modeling are presented in Figure 2.

We would like to express our sincere thanks to architect ***Erika Arellano*** [21] for her valuable contribution in obtaining and redrawing the original plans for the Niccolo Paganini Auditorium. The accuracy and quality of the graphic documents he provided were essential to our modeling process. His generosity and professionalism are highly valued, and we are grateful for his support in this project.

For this work it was necessary to obtain previous in-situ measurements of the auditorium. For this purpose, the auditorium administration was contacted via email, obtaining the results of measurements conducted in 2001 by Angelo Farina. The method by which they were obtained consists of using a sine sweep of 40 Hz to 22 kHz with a duration of 10 seconds as a test signal. A dodecahedron with a sub-woofer was used as the source, while *Neumann KU100* microphones were used at the listening points.

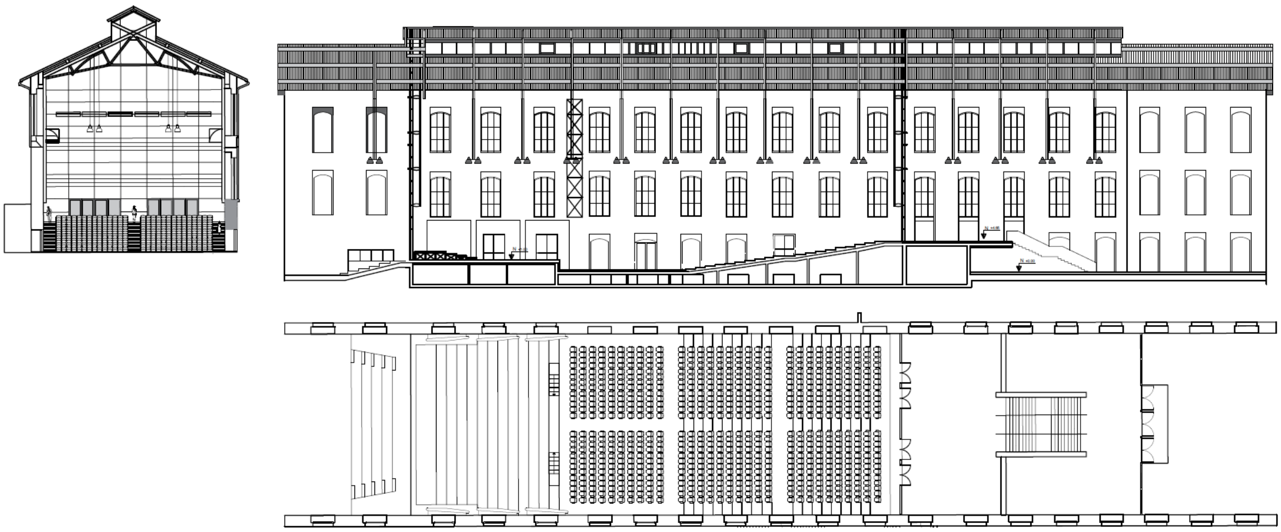
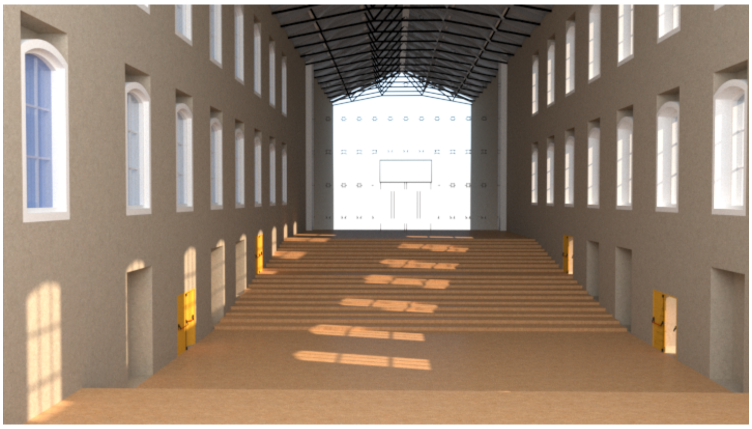
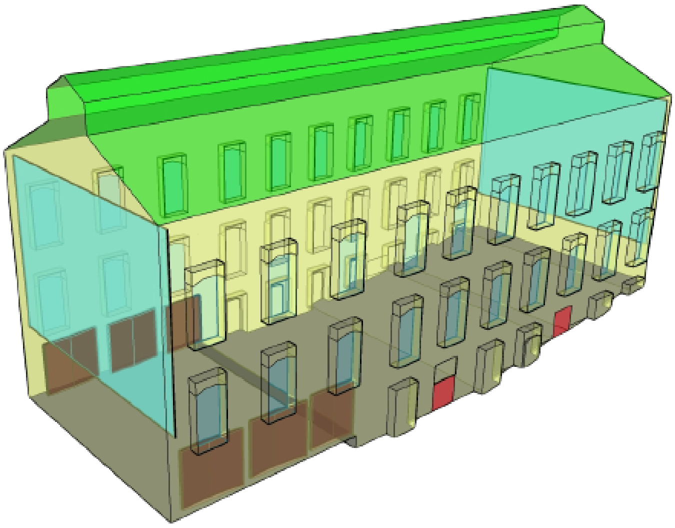


Figure 2: Front, side and top views of the Niccolo Paganini Auditorium

A three-dimensional model of the auditorium was generated using *SketchUp* design software and imported into the *EASE* acoustic simulation program. In order to optimize the computational efficiency of the model, a simplification was carried out by eliminating details such as window slots, steps in the bleachers, acoustic panels and complex geometric features of the auditorium.



(a) Real model created in Sketchup.



(b) Simplified model created in Sketchup.

Figure 3: Modeling in *Sketchup* software.

Figure 3 shows the simplified *sketchup* model. For the definition of the materials used in the simulation, as well as for the assignment of dispersion coefficients, use was made of the functionality provided by the EASE software itself. The layout of loudspeakers and microphones in the *EASE* model was designed to match the locations used in the in situ measurements carried out by Angelo Farina. The positions of the loudspeakers and microphones in the *EASE* model were arranged to match the positions used by Angelo Farina in his in situ measurements. This arrangement consists of one row of 16 measurement points, and two source positions.

4.2 Acoustic materials, absorption and scattering coefficients

When modeling the auditorium in a 3D simulation software, each surface must be assigned to a layer corresponding to the material they are made of. Each material has both absorption and scattering coefficients that vary with frequency. Taking into account that the real absorption coefficients are not available, an approximation must be made in order to choose the materials and coefficients of the surfaces. They were set by looking at pictures and videos of the auditorium. This must be taken into account when analyzing the obtained results, because a significant part of the differences between the measured and simulated acoustical parameters may be attributed to this approximation.

Six surfaces were used to model this auditorium, corresponding to the wooden floor, the sheet metal ceiling, the audience area, the glass windows, the brick walls and the wooden doors. Tables 5 and 6 show the absorption and scattering coefficients per octave band for each material [20].

Table 5: Absorption coefficients for every surface material in octave bands.

Material	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz
Wooden floor	0.15	0.20	0.10	0.10	0.10	0.10
Sheet metal ceiling	0.05	0.10	0.10	0.10	0.07	0.03
Simple glass window	0.2	0.15	0.10	0.07	0.05	0.05
Painted concrete wall	0.10	0.05	0.06	0.07	0.09	0.08
Wooden door	0.14	0.10	0.06	0.08	0.10	0.10
Unoccupied seats	0.19	0.37	0.56	0.67	0.61	0.59

Table 6: Scattering coefficients for every surface material in octave bands.

Material	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz
Wooden floor	0.10	0.10	0.10	0.10	0.10	0.10
Sheet metal ceiling	0.10	0.10	0.10	0.10	0.10	0.10
Simple glass window	0.10	0.10	0.10	0.10	0.10	0.10
Painted concrete wall	0.10	0.10	0.10	0.10	0.10	0.10
Wooden door	0.10	0.10	0.10	0.10	0.10	0.10
Unoccupied seats	0.30	0.30	0.30	0.30	0.30	0.30

4.3 Listener seats and sound source

In order to carry out the simulation, it was taken into account that the room was unoccupied, thus guaranteeing a suitable environment for the acoustic study. The seating arrangement has a layout of 13 rows and 10 columns, providing an optimal spatial configuration for the auditorium in question. In addition, six audience areas were modeled in the *EASE* software, where the position of the audience was simulated.

Previous in-situ measurements provide information on the position of the source and the listening points. Taking advantage of this fact, these same points were used for the simulation in *EASE*. These positions, both in measurements and simulation, are shown in Figure 4.

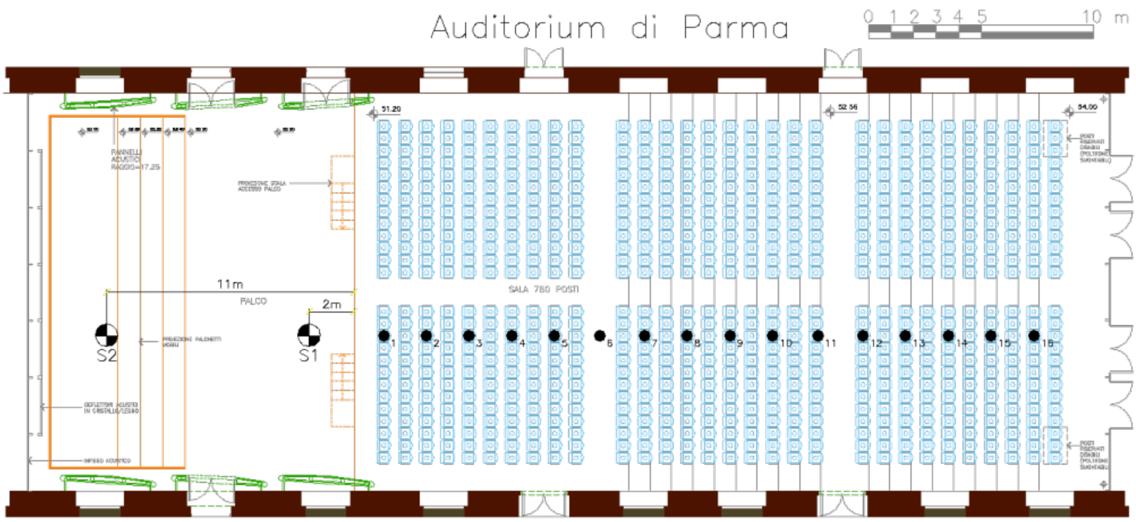
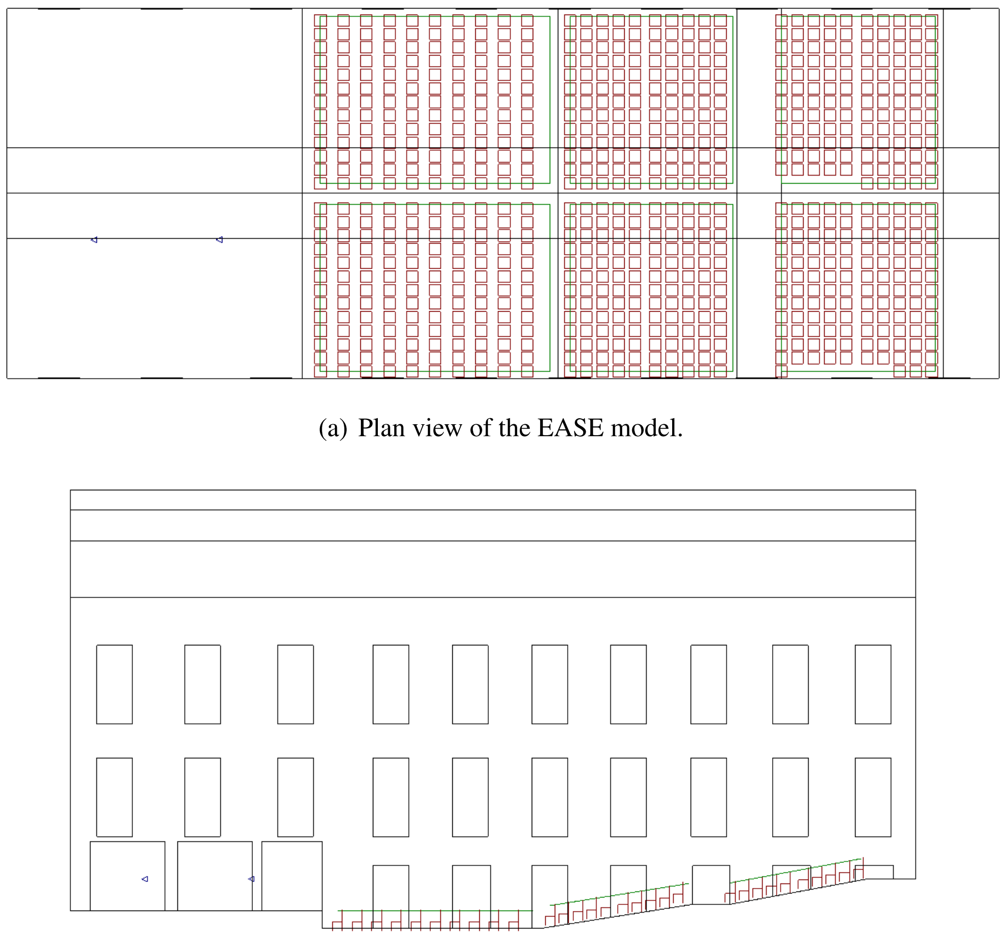


Figure 4: Source and listening points used for in-situ measurements.

In this case, to perform the in-situ measurement, 16 measurement points were taken, where these points are used in the simulation with the software. It is worth mentioning that the height of the microphones is not known, so in the modeling it was taken into account that this height is related to the listening plane of the listener. While that of the loudspeakers, the height of an average person was taken as a reference.

Similarly, the loudspeakers used at the time of the measurement are not found in the simulation program, so we try to imitate the loudspeaker used, using as data the frequency response of the system, as well as adding low pass and high pass filters of different orders to simulate the behavior of the loudspeaker.

In Figure 5 shows two views of the model made in *EASE* software of the Niccolo Paganini auditorium.



(a) Plan view of the EASE model.

(b) Side view of the *EASE* model.

Figure 5: Modeling views in *EASE* software.

4.4 Simulation configuration: Ray tracing and mapping

After the adjustments made to the auditorium model, the 'Find Impacts' function of the *EASE* is used to calculate the acoustic parameters. A number of rays per source of 200,000 and an order equal to 20 was set. With this configuration an impact probability of 88% is obtained, which is the minimum recommended in the *EASE* manual. Figure 6 shows the complete configuration used for ray tracing. After the simulation is complete, a statistical tail is added to complete the reflectogram. The start time of the statistical tail is left as calculated by the EASE. The amplitude of the added gaussian noise and gain is also adjusted to a value of 10 dB and the amplitude gain is also adjusted by visual inspection of the reflectogram. The obtained results are exported as an impulse response in wav format.

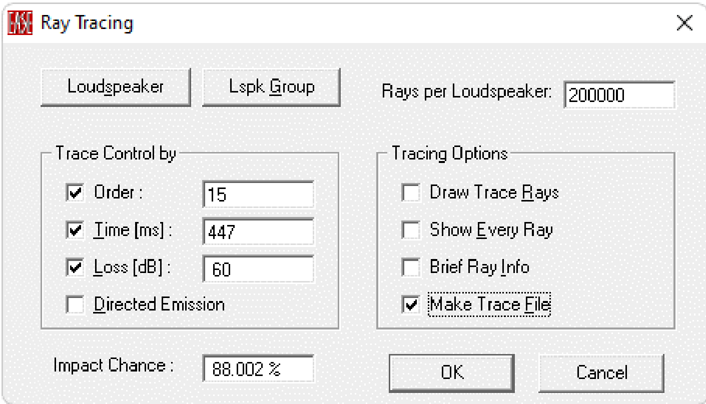


Figure 6: Configuration for the ray tracing processing.

4.5 Obtaining acoustical parameters

To approximate the model to the in situ measurements, two simulations are performed in the Ease software, adjusting and modifying the wall materials in each case in a comparative way. The same process is performed with the absorption coefficients and ray tracing configuration (cut-off time, order and number of emitted rays). After running the first simulation and obtaining the first set of acoustic parameters, it is observed that the reverberation times are notably lower in the simulation compared to the measured values, especially those corresponding to the lower part of the spectrum. In the subsequent iterations, specific adjustments are made paying attention to the correction of the absorption coefficients globally and more so in the mentioned bands in order to obtain more approximate results.

Then, all the synthesized RIRs are loaded into the EASERA software to obtain the acoustic parameters: EDT, T20, T30, STI, Alcons, D/R, Speech Echo and Music Echo. The same process is performed with the RIRs obtained by the on-site measurement performed by Angelo Farina in order to compare the results. Once both calculations are completed, we proceed to compare the magnitudes and make the necessary adjustments to the model to bring it within the JND range.

5. Results

In this section, the results obtained from the simulation are presented and compared with the in-situ measurements performed by Angelo Farina. Table 7 shows the comparison of reverberation time, STI, ALcons, Echo Speech, Echo music and D/R parameters. It is worth mentioning that Table 7 shows the averages of the 3 measurements, and simulation together with their associated uncertainty.

Table 7: Global values of measured and simulated parameters

Parameter	Measurement	Simulation	Difference	JND
STI	0.51 ± 0.08	0.52 ± 0.06	0.01	0.01 - 0.03
ALcons [%]	11.31 ± 3.87	10.44 ± 2.67	-0.87	-
Echo Speech	0.68 ± 0.05	1.02 ± 0.12	49.6 %	-
Echo Music	0.64 ± 0.12	1.07 ± 0.19	66.1 %	-
D/R [dB]	0.38 ± 3.48	-3.71 ± 1.12	4.09 dB	20
EDT [S]	2.13 ± 0.12	1.83 ± 0.31	-14.05 %	5 %
T_{20} [S]	2.21 ± 0.08	2.24 ± 0.24	0.64 %	24.5 %
T_{30} [S]	2.17 ± 0.03	2.19 ± 0.18	0.01 %	24.5 %

Table 7 shows that the difference between simulation and measurement for the STI is less than 1 JND, shows the averages of the 3 measurements position 2, 8 and 14. These values are shown in Annex A1. In addition, according to Table 1, 0.51 and 0.52 values are associated with good inteligibility.

Regarding the ALcons parameter, it is not possible to compare it with respect to the JND, but a -0.870 variation from the measurement, represent a difference in the intelligibility scale shown in Table 2. According to this grading, both the measured and simulated values lie on the threshold between good and poor intelligibility, with the simulated value falling just inside the good intelligibility category.

For Echo Speech and Echo Music, it is also not possible to compare based on the JND, but according to Table 3, a 49.6 % increase in the simulation leaves the echo speech value above the noticeable value for both trained and untrained ears. However, a 66% increase keeps the echo music parameter below 1.5 as shown in the table 3, which means that neither in the measurement nor in the simulation the reflections will be interpreted as echoes for both trained and untrained ears.

In the case of the direct-to-reverberant ratio parameter, a difference of 4.09 dB can be observed. This value is associated with a value of 2 JND, according to the table 4. This means that in situ there is a greater incidence of direct sound than reverberant field, while in the simulation it is the other way around.

For EDT, the difference between the measured and simulated value is almost 3 JND, while the T20 and T30 are almost identical in both cases, falling far below the JND value. In Figures 7, 8 and 9 the results in third octave bands for each parameter are shown.

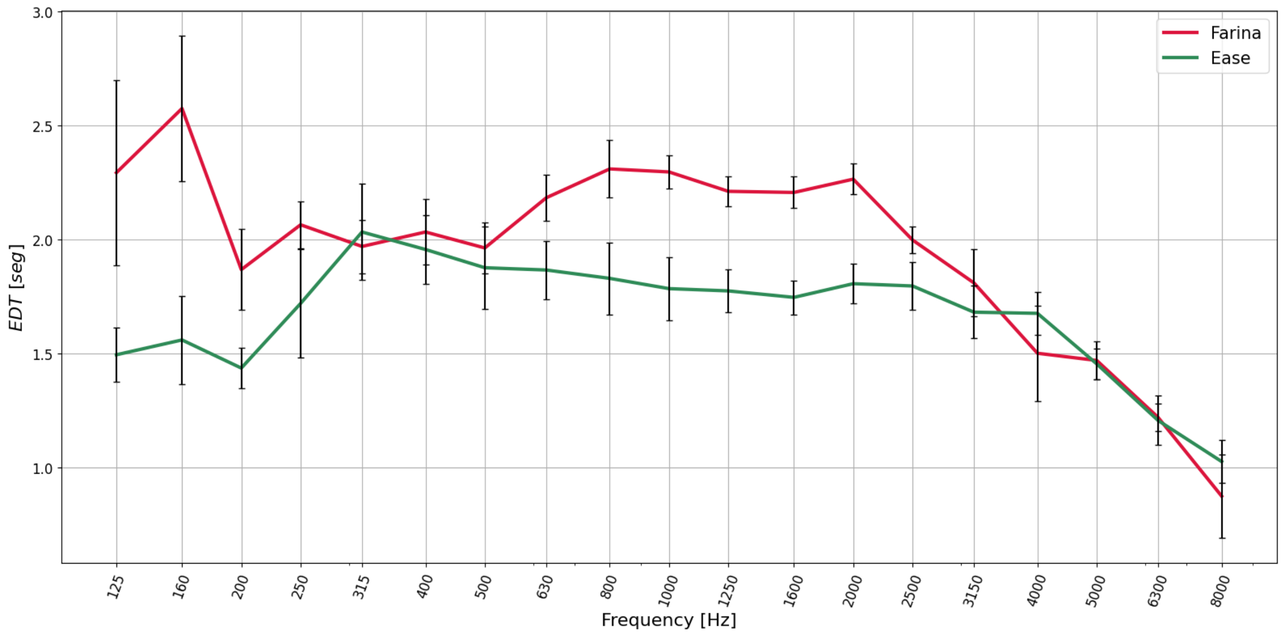


Figure 7: Comparison of the EDT parameter, between the Farina auditorium measurements and the EASE modelling of the hall.

Figure 7 shows the comparison of the EDT parameter, between the measurements made by Angelo Farina and the simulation in EASE.

The EDT curves obtained show that there is a difference between the model and the measurements, mainly at low frequencies than at high frequencies. It is also observed that their EDT values from the original measurements show higher values at low frequencies, which may indicate a higher concentration of early reflections at low frequencies. Furthermore, it is known that EDT tends to decrease at the higher end of the frequency spectrum.

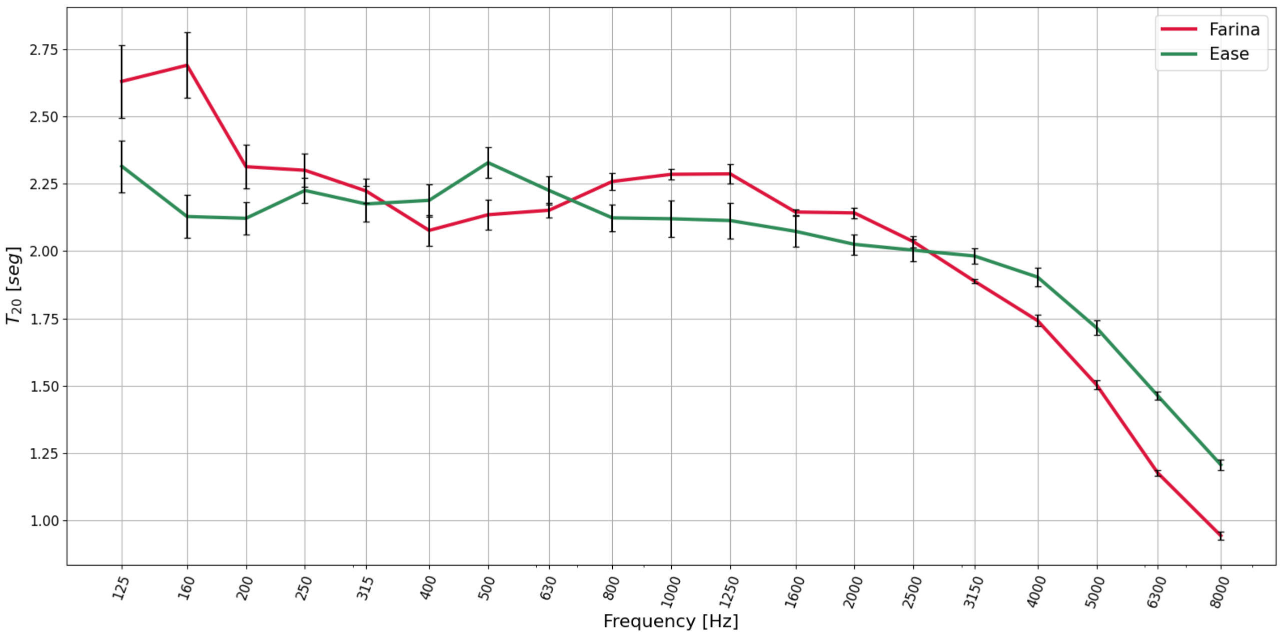


Figure 8: Comparison of the T20 parameter, between the Farina auditorium measurements and the EASE modelling of the hall.

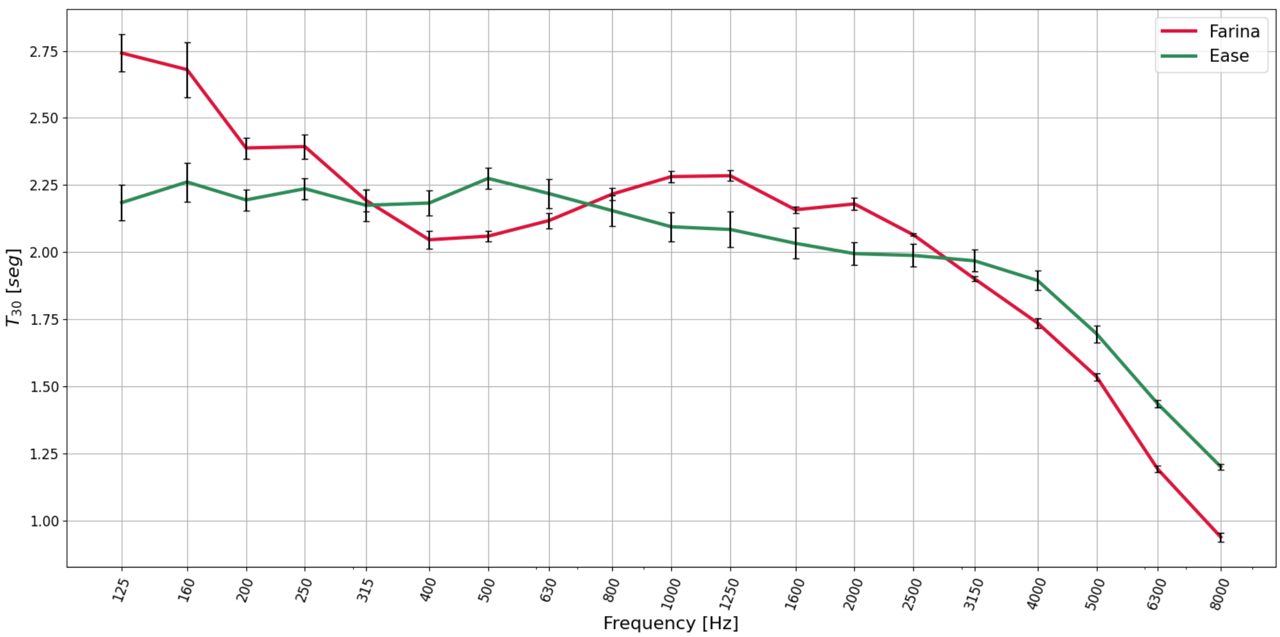


Figure 9: Comparison of the T30 parameter, between the Farina auditorium measurements and the EASE modelling of the hall.

In Figure 8 and 9 it is observed that the T_{20} and T_{30} of the modelled auditorium behave in a similar way to that measured in-situ, except at low frequencies, where a difference of 0.50 seconds can be seen, this is due to the fact that the materials used were estimated since the precise information of the absorption and scattering coefficients of the materials is not available. At high frequencies it is observed that the *TR* measured by Farina, is lower than the simulated one, this is related to the above mentioned and in addition, the room has absorbents panels in the ceiling, which were not simulated in the ceiling.

In order to visualise the behaviour of the acoustic parameters across the audience area, a mapping of the direct SPL, the STI parameter and AlCons was made. These projections were made with the EASE software, which means that they correspond to simulated values of the mentioned parameters.

Figure 10 shows the distribution of the direct SPL over the audience. As expected, the highest SPL level is found near the stage, then it decays towards the back of the auditorium following the Inverse Square Law thus obtaining a difference of 10 dB between the front and the back of the audience area.

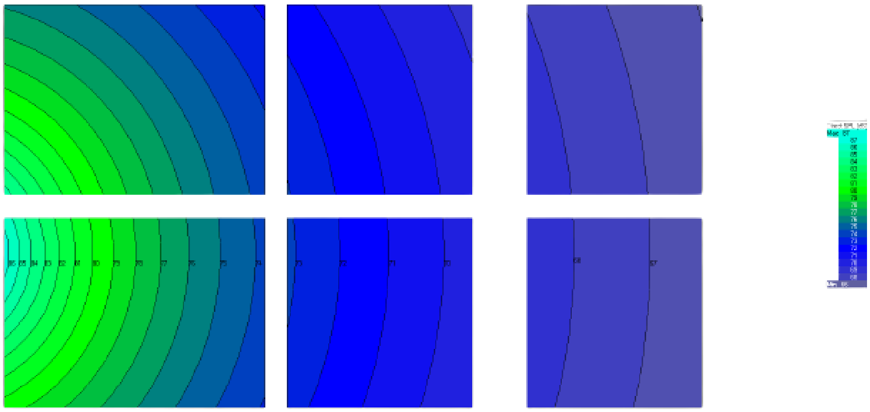


Figure 10: Spatial representation of direct SPL.

Figure 11 shows the STI parameter, which ranges from 0.58 at the front of the audience to 0.32 at the back. According to the values given in the table 1 this is good intelligibility at the front of the auditorium, while at the back of the room it refers to satisfactory intelligibility.

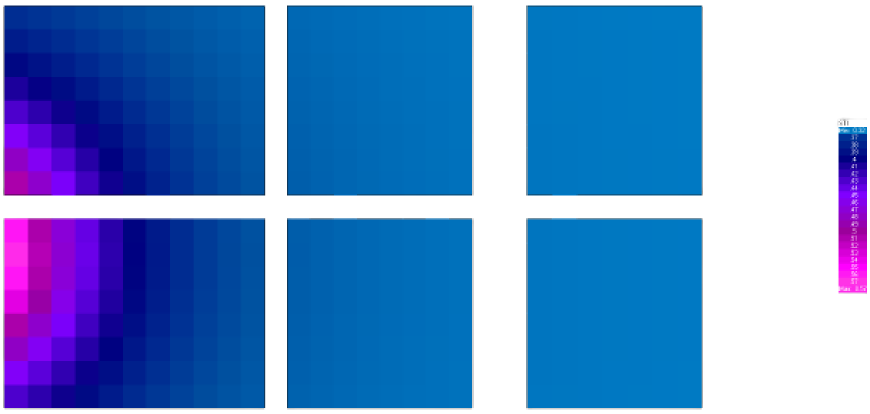


Figure 11: Spatial representation of direct STI.

Finally, figure 12 shows how the AlCons parameter is distributed in the audience area. In the front part, values between 8 % and 11 % are found, which, according to the table 2 corresponds to good intelligibility values, however, these values decrease rapidly as one moves away from the stage. Even in the first block of seats, there are already values higher than 11 %, that is poor intelligibility, but the rest of the auditorium has AlCons values exceeding 20 %, which means that the intelligibility is worthless.

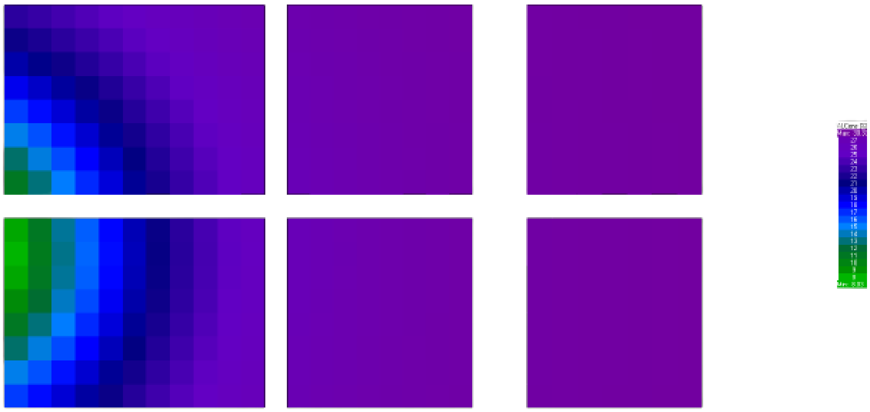


Figure 12: Spatial representation of direct ALcons.

6. Conclusion

Using the EASE software, a model of the Niccolo Paganini auditorium was simulated to obtain different acoustic parameters which were then compared with in-situ measurements carried out by Angelo Farina.

First of all, the difference between simulation and measurement for the ITS is less than 1 JND. Then, the simulated value of the ALcons parameter is just within the good intelligibility category, although both measured and simulated values are above the threshold between good and bad intelligibility. On the other hand, in the case of the direct-to-reverberant ratio parameter in the simulation, a higher incidence of the field over the direct sound was obtained. For EDT and reverberation times, very similar results were obtained between measurement and simulation.

Finally, although specific information on the materials used in the construction of the auditorium was not available, it was possible to approximate the simulated acoustic values with those measured by adjusting the absorption and scattering coefficients. The geometrical acoustics simulation based on ray tracing in EASE allows for a close approximation to a representation of the real room, although necessary adjustments have been made to achieve this. It is important to note that the EASE model was deliberately simplified because an increase in the number of surfaces and polygons would result in a prolongation of the calculation time. The determination of the acoustic parameters is not only influenced by the dimensions of the room, but also by the characteristics of the materials and the coefficients associated with each surface. These coefficients had to be adapted in each calculation in order to reach the required JND. In summary, a total of four calculations were carried out to obtain results considered satisfactory. However, despite these efforts, it was not feasible to obtain a discrepancy, in the mean frequencies, of less than 5% between the calculated and simulated EDT parameter. The simulation of geometric acoustics based on ray-tracing in EASE. Furthermore, it is important to note that in order to obtain reflectograms with sufficient density to synthesise high quality room impulse responses (RIRs), considerable computational time must be taken into account. In the context of this work, the ray tracing time amounted to 4.10 hours. It was also found that EASE only generates RIRs corresponding to omnidirectional microphones at predefined listening positions, excluding the possibility of using other types of microphones, such as those with figure-of-8 patterns. Therefore, it was impossible to obtain the LF parameter from the RIRs synthesised in this simulation environment. An approximation of the simulation with lower dispersion could be made if a detailed study of the auditorium materials and their coefficients were carried out. On the other hand, information about the frequency response of the source used could be loaded into the simulation for future measurements.

References

1. Marshall Long. *Architectural acoustics*. Elsevier, 2005.
2. Leo Leroy Beranek. *Concert halls and opera houses: music, acoustics, and architecture*, volume 2. Springer, 2004.
3. AFMG EASE 4.4 User's Manual. `https://www.afmg.eu/en/ease-44-users-manual`. Accessed: 2023-09-30.
4. Wallace Clement Sabine and M David Egan. Collected papers on acoustics, 1994.
5. Carl F Eyring. Methods of calculating the average coefficient of sound absorption. *The Journal of the Acoustical Society of America*, 4(3):178–192, 1933.
6. MJ Mahjoob and S Malakooti. Acoustic simulation of building spaces by ray-tracing method: Prediction vs. experimental results. In *23rd International Conference on Noise and Vibration Engineering*, pages 2303–2312, 2008.
7. Fabian Brinkmann, Lukas Aspöck, David Ackermann, Steffen Lepa, Michael Vorländer, and Stefan Weinzierl. A round robin on room acoustical simulation and auralization. *The Journal of the Acoustical Society of America*, 145(4):2746–2760, 2019.
8. Jont B Allen and David A Berkley. Image method for efficiently simulating small-room acoustics. *The Journal of the Acoustical Society of America*, 65(4):943–950, 1979.
9. Andrzej Kulowski. Algorithmic representation of the ray tracing technique. *Applied Acoustics*, 18(6):449–469, 1985.
10. Jens Holger Rindel. The use of computer modeling in room acoustics. *Journal of vibroengineering*, 3(4):219–224, 2000.
11. Toru Otsuru, Takeshi Okuzono, Reiji Tomiku, Kusno Asniawaty, and Noriko Okamoto. Large-scale finite element sound field analysis of rooms using a practical boundary modeling technique. In *Proceedings of the 19th International Congress on Sound and Vibration, Vilnius, Lithuania*, pages 8–12, 2012.
12. IEC 60268-16:2020. Sound system equipment - part 16: Objective rating of speech intelligibility by speech transmission index. Standard, International Electrotechnical Commission, 2020.
13. VMA Peutz. Articulation loss of consonants as a criterion for speech transmission in rooms. In *Audio Engineering Society Convention 2ce*. Audio Engineering Society, 1972.
14. L Dietsch and W Kraak. Ein objektives kriterium zur erfassung von echostörungen bei musik-und sprachdarbietungen. *Acta Acustica United with Acustica*, 60(3):205–216, 1986.
15. Michael Barron and Arthur Harold Marshall. Spatial impression due to early lateral reflections in concert halls: the derivation of a physical measure. *Journal of sound and Vibration*, 77(2):211–232, 1981.
16. ISO 3382-1:2009. Acoustics — measurement of room acoustic parameters — part 1: Performance spaces. Standard, International Organization for Standardization, Geneva, CH, 2009.
17. Matthew G Blevins, Adam T Buck, Zhao Peng, and Lily M Wang. Quantifying the just noticeable difference of reverberation time with band-limited noise centered around 1000 hz using a transformed up-down adaptive method. 2013.
18. John S Bradley, R Reich, and SG Norcross. A just noticeable difference in c50 for speech. *Applied Acoustics*, 58(2):99–108, 1999.
19. Stephan Werner and Judith Liebetrau. Adjustment of direct-to-reverberant-energy-ratio and the just-noticable-difference. In *2014 Sixth International Workshop on Quality of Multimedia Experience (QoMEX)*, pages 1–3. IEEE, 2014.

20. Trevor Cox and Peter d'Antonio. *Acoustic absorbers and diffusers: theory, design and application*. CRC press, 2016.

21. Erika Arellano. Perfil de linkedin de erika arellano. `https://www.linkedin.com/in/erika-arellano-a10106129/?original_referer=https%3A%2F%2Fwww%2Egoogle%2Ecom%2F&originalSubdomain=ec`, 2023.

22. RF Norris and CA Andree. An instrumental method of reverberation measurement. *The Journal of the Acoustical Society of America*, 1(3A):366–372, 1930.

Annex

A. Data

Table A 1: Results of measurements and simulations at position 2, 8 and 14.

Farina

Source 1	Parameters	100 Hz	125 Hz	160 Hz	200 Hz	250 Hz	315 Hz	400 Hz	500 Hz	630 Hz	800 Hz	1000 Hz	1250 Hz	1600 Hz	2000 Hz	2500 Hz	3150 Hz	4000 Hz	5000 Hz	6300 Hz	8000 Hz	RTmid
Position 2	EDT	1.20	0.98	1.87	2.17	1.76	2.01	2.47	2.18	1.85	2.30	2.19	2.21	2.20	2.33	1.75	1.13	0.55	1.28	1.14	0.03	2.19
	T20	2.16	2.02	2.41	2.21	2.19	2.11	1.89	1.97	2.26	2.19	2.23	2.19	2.16	2.06	1.97	1.88	1.75	1.51	1.14	0.92	2.10
	T30	2.49	2.58	2.57	2.35	2.34	2.09	2.03	2.05	2.17	2.16	2.22	2.23	2.18	2.08	2.07	1.89	1.70	1.49	1.16	0.91	2.14
Position 8	EDT	0.83	2.08	2.70	1.59	2.03	1.69	1.69	1.64	2.14	2.70	2.58	2.46	2.35	2.40	2.02	2.15	2.02	1.76	1.47	1.26	2.11
	T20	2.74	2.52	2.71	2.62	2.48	2.31	2.15	2.29	2.15	2.15	2.28	2.22	2.12	2.12	2.12	1.89	1.70	1.45	1.17	0.97	2.29
	T30	3.11	2.60	2.70	2.53	2.55	2.22	2.08	1.98	2.12	2.14	2.34	2.30	2.12	2.17	2.07	1.90	1.70	1.52	1.18	1.00	2.16
Position 14	EDT	2.38	2.07	1.81	1.41	2.27	2.23	2.46	1.62	2.46	2.51	2.21	2.01	2.18	2.33	1.98	1.87	1.52	1.43	1.12	1.10	1.92
	T20	2.88	2.84	2.76	2.20	2.23	2.20	2.01	2.24	2.08	2.26	2.27	2.29	2.12	2.18	2.01	1.92	1.77	1.51	1.19	0.95	2.26
	T30	3.56	2.84	2.50	2.38	2.45	2.08	1.90	2.07	2.02	2.25	2.27	2.29	2.18	2.20	2.06	1.92	1.77	1.56	1.22	0.94	2.17

Source 2	Parameters	100 Hz	125 Hz	160 Hz	200 Hz	250 Hz	315 Hz	400 Hz	500 Hz	630 Hz	800 Hz	1000 Hz	1250 Hz	1600 Hz	2000 Hz	2500 Hz	3150 Hz	4000 Hz	5000 Hz	6300 Hz	8000 Hz	RTmid
Position 2	EDT	1.19	2.18	3.89	1.67	1.91	2.00	1.91	2.04	2.05	1.87	2.15	2.16	1.95	1.98	2.15	1.94	1.59	1.28	1.08	1.04	2.10
	T20	2.87	2.80	2.61	2.20	2.49	2.14	2.04	2.17	2.18	2.27	2.28	2.24	2.18	2.15	2.06	1.87	1.68	1.47	1.18	0.89	2.23
	T30	4.63	2.69	2.81	2.25	2.39	2.18	2.06	2.06	2.17	2.22	2.29	2.23	2.16	2.24	2.06	1.88	1.71	1.52	1.18	0.89	2.18
Position 8	EDT	3.46	2.47	2.80	2.56	1.99	1.60	1.86	2.11	2.12	2.41	2.43	2.31	2.42	2.17	2.10	1.91	1.58	1.41	1.27	0.87	2.27
	T20	2.61	2.85	2.44	2.15	2.16	2.39	2.08	1.98	2.09	2.33	2.28	2.36	2.13	2.17	2.02	1.90	1.82	1.57	1.17	0.98	2.13
	T30	2.87	3.02	2.41	2.38	2.22	2.32	2.11	2.09	2.18	2.26	2.24	2.32	2.13	2.22	2.08	1.93	1.81	1.58	1.19	0.96	2.17
Position 14	EDT	3.25	3.98	2.38	1.81	2.43	2.29	1.81	2.19	2.48	2.07	2.22	2.12	2.14	2.38	1.99	1.86	1.75	1.66	1.24	0.95	2.21
	T20	3.04	2.75	3.21	2.50	2.25	2.19	2.29	2.16	2.15	2.35	2.37	2.42	2.16	2.17	2.03	1.87	1.73	1.51	1.21	0.95	2.27
	T30	6.17	2.72	3.09	2.44	2.41	2.27	2.10	2.11	2.05	2.27	2.33	2.34	2.18	2.17	2.05	1.89	1.73	1.55	1.23	0.94	2.22

EASE

Source 1	Parameters	100 Hz	125 Hz	160 Hz	200 Hz	250 Hz	315 Hz	400 Hz	500 Hz	630 Hz	800 Hz	1000 Hz	1250 Hz	1600 Hz	2000 Hz	2500 Hz	3150 Hz	4000 Hz	5000 Hz	6300 Hz	8000 Hz	RTmid
Position 2	EDT	1.02	1.64	1.71	1.64	2.51	2.28	2.15	2.11	2.01	2.43	2.17	1.86	1.66	1.85	2.13	1.86	1.79	1.68	1.60	1.46	2.14
	T20	2.25	2.46	2.34	2.13	2.46	2.61	2.49	2.59	2.54	2.35	2.38	2.40	2.35	2.16	2.19	2.12	2.04	1.77	1.45	1.13	2.49
	T30	2.34	2.27	2.46	2.20	2.41	2.52	2.36	2.41	2.47	2.43	2.35	2.40	2.33	2.15	2.14	2.16	2.07	1.82	1.46	1.21	2.38
Position 8	EDT	1.37	1.12	1.34	1.25	1.25	1.70	1.56	1.62	1.89	1.60	1.65	1.53	1.63	1.78	1.78	1.86	1.72	1.28	0.96	1.03	1.64
	T20	2.19	1.95	1.87	2.06	2.10	1.97	1.86	2.02	2.08	1.81	1.79	1.86	1.88	1.93	1.82	1.90	1.77	1.60	1.40	1.13	1.91
	T30	1.97	1.93	2.06	2.19	2.11	2.03	2.14	2.16	2.06	1.89	1.91	1.89	1.84	1.91	1.84	1.87	1.81	1.62	1.40	1.13	2.04
Position 14	EDT	1.23	1.76	1.08	1.12	1.66	1.63	1.50	1.45	1.33	1.52	1.47	1.55	1.62	1.54	1.72	1.27	1.28	1.40	1.01	0.92	1.46
	T20	2.20	2.14	2.02	2.01	2.13	2.01	2.15	2.19	2.29	2.05	2.05	2.00	1.91	1.94	1.89	1.88	1.84	1.62	1.41	1.19	2.12
	T30	2.08	1.97	2.02	2.18	2.11	2.00	2.00	2.09	2.10	2.06	2.00	1.92	1.88	1.89	1.87	1.83	1.77	1.60	1.38	1.17	2.05

Source 2	Parameters	100 Hz	125 Hz	160 Hz	200 Hz	250 Hz	315 Hz	400 Hz	500 Hz	630 Hz	800 Hz	1000 Hz	1250 Hz	1600 Hz	2000 Hz	2500 Hz	3150 Hz	4000 Hz	5000 Hz	6300 Hz	8000 Hz	RTmid
Position 2	EDT	1.47	1.20	1.08	1.64	2.01	2.57	2.00	2.04	1.90	2.17	2.24	2.03	1.95	2.13	1.77	1.85	1.89	1.52	1.45	0.98	2.14
	T20	3.08	2.95	2.66	2.58	2.40	2.36	2.41	2.55	2.36	2.30	2.48	2.46	2.34	2.23	2.20	2.10	2.09	1.89	1.51	1.20	2.52
	T30	2.63	2.60	2.72	2.47	2.44	2.41	2.44	2.49	2.51	2.40	2.38	2.41	2.29	2.24	2.24	2.17	2.08	1.88	1.52	1.23	2.44
Position 8	EDT	2.33	1.47	2.06	1.54	1.95	2.58	2.41	2.55	2.26	1.68	1.56	2.02	2.00	1.89	1.98	1.86	1.82	1.56	1.12	0.94	2.06
	T20	2.27	2.37	1.95	1.98	2.01	2.05	2.07	2.23	2.06	2.12	2.00	2.03	2.03	2.03	1.98	1.97	1.86	1.69	1.53	1.29	2.12
	T30	2.13	2.30	2.23	2.08	2.12	2.04	2.10	2.20	2.04	2.10	1.98	2.02	1.97	1.94	1.95	1.93	1.83	1.61	1.43	1.22	2.09
Position 14	EDT	2.35	1.78	2.09	1.43	0.94	1.44	2.12	1.49	1.81	1.58	1.62	1.66	1.62	1.65	1.40	1.39	1.56	1.29	1.10	0.83	1.56
	T20	1.82	2.02	1.93	1.97	2.25	2.05	2.15	2.39	2.02	2.11	2.02	1.93	1.93	1.86	1.94	1.92	1.82	1.72	1.48	1.30	2.21
	T30	1.79	2.04	2.08	2.05	2.23	2.05	2.06	2.30	2.13	2.05	1.95	1.87	1.89	1.84	1.89	1.85	1.81	1.65	1.43	1.25	2.13

B. Questions

1. **How many beams did you use and why?** The number of rays used defines the chance of im-

   pact that the modelling will have. Through the user manual provided by *EASE*, this parameter determines the accuracy of the simulation. The recommendation suggested by the manufacturer is a chance of impact greater than 88%. On the other hand, this generates an increase in processing and, as a consequence, an increase in simulation time.
   Too reach the 88% chance of impact, an order 20 was used, with 200,000 rays per speaker. It is worth mentioning that order 20 was used in order to have more information from the room.
2. **In how many positions on the stage did you place the omnidirectional source? Why?**

   Two (2) omnidirectional sources positioned on the stage were used. These positions were chosen because of the measurements made by Angelo Farina, where he gives the positions used in the measurement, so the same positions are taken for the simulation of the enclosure. In the same way, the same positions of microphones are taken, as these positions are also available.
   It is worth mentioning that the positions of the sources are not in the centre of the room. In addition, the positions of the microphones are located on the seats in order to simulate the listening position of the audience. This was simulated in the same way in the modelling of the auditorium.
3. **Why is there a difference between the statistical calculation of reverberation and the value of reverberation presented by the RIRs?**

   The difference between the statistical calculation of reverberation and the value of reverberation presented by Reverberation Impulse Responses (RIRs) is due to the way these two parameters are calculated and represented.
   Reverberation Impulse Responses are detailed representations of how sound reflects and decays in a room as a function of time. The Ray Tracing method follows the path of each sound beam from the source to the receiver, taking into account all reflections, absorptions and scatterings that occur along the way. It is a more detailed and accurate approach that models the physics of how sound interacts with surfaces and objects in the room. However, it is also more computationally intensive and can require a large amount of processing time and resources to calculate.
   The Sabine Method approach uses formulas and statistical relationships to approximate reverberation time. It is based on simplifying assumptions about room sound absorption, dispersion of reflections and other acoustic factors. Although less accurate than the Ray Tracing method, it is faster and easier to apply.
   Both approaches are useful in different contexts. The statistical calculation is valuable for providing an overview of the room acoustics, while the values of the RIRs are useful for understanding how sound behaves at specific times, which may be relevant for designing sound systems or understanding the effect of reflections on sound quality.

Produce a clear and simple question related to the topic under study, which does not have a direct answer (neither in papers nor in books), and which requires concrete research, achievable by the student with his/her own equipment and/or university, in a period of no more than 6 (six) months. research, which can be carried out by the student with his/her own equipment and/or that of the university, in a period of no more than 6 (six) months. The wording of the question does not have a limit of characters and can contain all the explanations, justifications and conditions of the case.

Is it feasible to develop a software or method using optimisation algorithms based on machine learning techniques, such as neural networks, genetic algorithms or particle swarm optimisation, to automatically adjust acoustic parameters according to actual measured values in specific spaces, with the aim of improving the accuracy of simulations and their correspondence with in situ measurements? Could this automation also lead to an improvement in the computational speed of acoustic simulations?