ACOUSTICAL PARAMETERS

AMDEN, PABLO[1] - DE BORTOLI, LUCIANO[2] - FREIDKES, JONATHAN[3]

Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina
[1]pabloamden@gmail.com
[2]luciano.nicolas.de.bortoli@gmail.com
[3]jonifreidkes@gmail.com

Abstract - *The aim of the current report is to perform an extensive analysis of various acoustical parameters of a concert hall, namely Teatro Roma in Buenos Aires. Such parameters are: RT, BR, EDT, C_{50}, C_{80}, D_{50}, D_{80}, G, STI, %AlCons, LF_E, LF_L, IACC and τ_e. A total of 132 measurements were carried out to obtain the different values according to the procedures in ISO 3382. A spatial distribution of every parameter through a mapping algorithm is accomplished for global and third-octave band values. Results are then analyzed taking into account the influence of systematic errors. Finally, a datasheet is completed for the theatre characterization. In broad terms, results show low RT values, especially for high frequencies. There are defined zones of high intelligibility, which make Teatro Roma's current acoustics best fit for speech applications, rather than intended Opera. According to this report findings, this could be solved by changing seat upholstering and curtains to lighter materials.*

1. INTRODUCTION

Room acoustics is a science applied in architectural designs since ancient times for Greek and Roman theatres and throughout the recorded human history; despite this, recently has become to be mathematically described. Wallace C. Sabine [1] introduced his RT_{60} formula in the early 1900s and since then, many people have came up with new parameters and better methodologies to describe the room behaviour when excited with sound waves. Theatres and concert halls have all been designed meeting architectural and acoustical requirements for their time and intended application, allowing to receive large audiences that can listen clearly, whether it is about a speech, a voice, instrumental music, or loudspeakers in modern times. It is important to remark that any room with sound reinforcement should also have good acoustics in order to have the best possible interaction.

Room acoustics can be the best ally and the worst enemy i.e. by being able to amplify an important speech to be heard by many people, and at the same time by adding too much reverberation, drastically deteriorating intelligibility. By and large, a balanced room acoustics design leads to better audition. It is able to generate the perception of a wider source and even provide support for dramatic movements in an opera or symphony.

Over the years, many concert halls were measured in order to conclude in recommended values for each acoustic application. New parameters have been defined in order to help characterize the room performance, which can never be described with a single criterion. Many technical instrumentation and computational development brought new and powerful assistance for acoustical analysis and design. This made the acoustical analysis more complex for its great variety of parameters and room behavior indicators. Nowadays, relatively simple measurements, following ISO 3382 guidance [2], are the basis of a complete analysis based in room impulse response for most of the usual parameters and binaural measurement for IACC-related measurements.

As in many disciplines, there is another point of view of the phenomenon. During the 1970s, PhD Yoichi Ando developed some new set of acoustical parameters used to describe subjective room acoustical parameters in relationship with human hearing. This new type of study requires a set of audio files recorded in an anechoic chamber to be reproduced in a two-way speaker and then it's room response to be registered. This is used to analyse the room influence in the human perception thru an analysis of the variation of the τ_e, one of Ando's most famous parameter (which is explained in next sections). Ando stands for 4 orthogonal parameters that lead to subjective preference: Listening Level (LL), Initial Time Delay Gap (ITDG), Reverberation Time (RT) and InterAural Cross Correlation (IACC).

This paper analyses the objective and subjective parameters of Teatro Roma located in Buenos Aires, Argentina. Its structure consists of an theoretical framework section, where parameters are defined along with its recommended values according to the acoustic application. Then, a description section, where the concert hall's historical background, material and architectonic properties are detailed; followed by the methodology section, where measuring, recording, processing and post processing methods applied are exhaustively illustrated. Afterwards, the results and discussion section exhibits the values obtained by employing intuitive global and third-octave band graphs with its correspondent analysis and discussion. Finally verdicts from this work are revealed in the conclusion section, as well as future research aspirations.

2. ACOUSTICAL PARAMETERS

2.1 Reverberation Time (RT)

The reverberation in a room is very important as it adds fullness of tone to the music, and at the same time, adds loudness and a sense to the listener of being enveloped by the sound [3].

The reverberation time is one of the first parameters introduced in room acoustic studies, and probably the most important one. In 1927 W.C Sabine introduced the first equation 1 to calculate RT.

$$T_{60} = \frac{0.161V}{A} \quad (1)$$

where RT_{60} is the time it takes for a sound to decay 60 dB.

At the time, the room was excited to a stationary state by sound source and the sound was cut off (nowadays impulsive excitations are more common to be used instead). The logarithm of the decaying RMS sound pressure was calculated and plotted against time, as illustrated in Figure 1. From this plot, the time that takes the deacy to reach 60 dB is the RT_{60}.

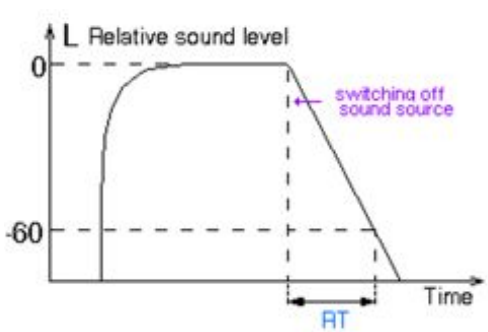

Figure 1: Sound level over time for RT calculation.

The problem with this method was the repeatability of the procedure due to the differences between the beginning of the decay's curves, making quite difficult to ensemble a good average. Nevertheless, in 1965 Schroeder developed a new procedure that allowed to determine the average with greater accuracy. To calculate RT with the Schroeder method the equation 2 is used.

$$\int_t^{\infty} p^2(t)dt = \int_0^{\infty} p^2(t)dt - \int_0^{t} p^2(t)dt \quad (2)$$

where ρ is sound pressure level and t is the selected time period.

Background noise usually prevents a signal from decaying 60 dB, and then RT is often achieved by doubling the time that it takes the sound to decay from -5 dB to -35 dB, which defines RT_{30}, or by tripling the time it takes to decay from -5 dB to -25 dB, which defines RT_{20}, in this case.

By filtering the signal and calculating octave band RT values, other useful parameters can be defined to describe a concert hall; such is the case for *Bass Ratio* (3) and RT_{MID} (4).

$$BR = \frac{RT_{125Hz} + RT_{250Hz}}{RT_{500Hz} + RT_{1000Hz}} \quad (3)$$

$$RT_{MID} = \frac{RT_{125Hz} + RT_{1000Hz}}{2} \quad (4)$$

According to Beranek [4] good values of RT_{MID} for a theater goes from 0.7 s to 1.2 s.

Because Reverberation Time is correlated with the perception of space, there are optimal RT values depending on the concert hall volume and the musical application, as illustrated in Figure 2.

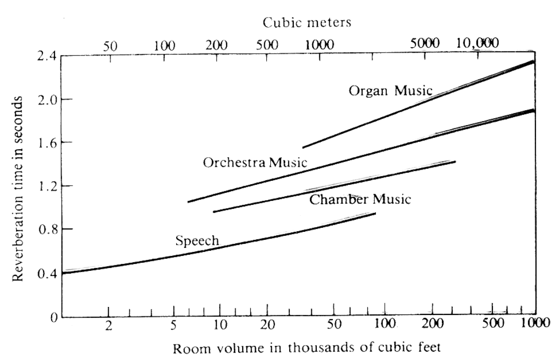

Figure 2: Optimal RT vs concert halls volume.

From the Figure, it can be observed that for volumes closer to 3000 m^3, optimal RT values for chamber music sit around 1.4 s, whereas for orchestral music is about 1.6 s and for speech, around 1 s.

2.2 Early Decay Time (EDT)

The EDT, Early Decay Time, is defined as six times the time period where the sound initially decays the first 10 dB. And, as the RT, it can be calculated for third-octave bands.

Furthermore, the early decay time is currently the preferred measure relating to perception of running reverberance [5], which is the measure of the reverberance perceived during while music is being played. This perception is not only important for the listener, it is also important for the musicians and their ensemble support. Usual values of EDT_{MID} for different unoccupied concert hall are shown in Figure 3.

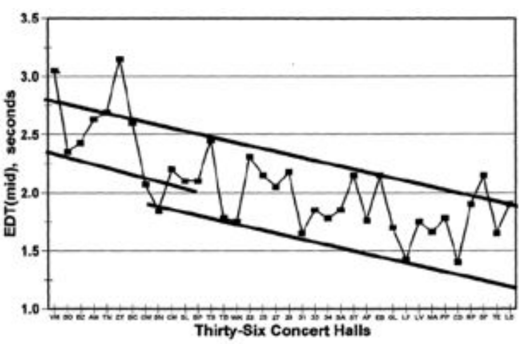

Figure 3: EDT for 36 unoccupied concert halls

From the Figure, it can be observed that EDT values often ranges from 1.5 s to 2.5 s in the case of concert halls with mainly lightly upholstered seats.

2.3 Direct to Reverberant Energy Ratio (D/R)

D/R ratio or DRR, is the proportion between the direct sound that goes straight from the source to the listener position without any reflection, and the reverberant portion of the sound, that comes only from reflecting surfaces. This is a parameter that relates very closely to the human perception of source distance [6]. Besides the fact that only the direct sound (DS) provides information about the localization and distance of a sound source it can be discussed how the perception of DS in a reverberant field depends on the D/R and the time delay between the DS and the reverberant energy.

2.4 Clarity (C_{50} - C_{80})

Sound clarity can be approached from different perspectives. From the subjective point of view sound clarity can be associated to *how clear the sound quality is*; i.e. if every separate note of a fast-tempo music passage can be distincted. Sometimes the blending between notes is desired for some styles of music, so it is not the only indicator. However, for speech and opera greater clarity leads to better intelligibility.

Analyzing clarity objectively leads to the need of establishing a parameter. Clarity is produced when a room has a high ratio of early sound energy to later reverberant energy. For music it is considered that early sound energy is that which arrives at the listener within 80 milliseconds of the direct sound from the source to the listener, and for speech 50 milliseconds is the time considered for early sound.

Clarity is defined as the logarithmic ratio of early sound energy, arriving in the first 80 or 50 ms, to late sound energy, arriving after this time period. Leading to the following equations:

$$C_{50} = 10 \log \frac{\int_0^{0.05} p^2(t)dt}{\int_{0.05}^{\infty} p^2(t)dt} \tag{5}$$

$$C_{80} = 10 \log \frac{\int_0^{0.08} p^2(t)dt}{\int_{0.08}^{\infty} p^2(t)dt} \tag{6}$$

Like many of the acoustic parameters, C_{80} and C_{50} are dependent upon frequency. Therefore, an average value is used, it is common to use the values at frequency octave bands centered at 500, 1000, and 2000 Hz, and this average would be called $C_{80\ (3)}$ or $C_{50\ (3)}$. Some typical clarity values are exposed in the following Table 1.

Concert hall	$C_{80\ UNOCC\ (3)}$ [dB]
Amsterdam, Concertgebouw	-3.3
Boston, Symphony Hall	-2.7
Vienna, Gr. Musikvereinssaal	-3.7
Basel, Stadt-Casino	-2.3
Berlin, Konzerthaus	-2.5
Cardiff, St. David's Hall	-0.9
Tokyo, Hamarikyu Asahi	-0.2
Zurich, Großer Tonhallesaal	-3.6

Table 1: Typical C_{80} values for concert halls. [7]

According to Beranek, if octave bands of 500, 1000 and 2000 Hz are averaged, C_{80} results from -4 dB to -1 dB are good values for reverberant halls, while 1 dB to 5 dB results are good values for less reverberant halls.

2.5 Definition (D_{50})

As it is known, reflections are not perceived as something separated from the direct sound as long as its delay and relative strength do not exceed certain limits. This makes the sound source seem bigger and charged with more loudness. In this case, these early reflections can be considered useful. On the contrary, if the delay is too long or if the reflections have too much energy, these reflections can be perceived like discrete echoes. Beside these effect, long delayed reflections tend to blur the time structure and mixes up the spectral characteristic of spoken syllables, which end up deteriorating speech transmission. Therefore, it can be said that the critical delay time separating useful from detrimental reflections is somewhere around 50 to 100 milliseconds [8].

The first attempt to define an objective criterion for how good the sound is distincted, derived from an impulse response was made by Mayer and Thiele, and they called it definition (D_{50}):

$$D_{50} = 100\frac{\int_0^{0.05} p^2(t)dt}{\int_0^{\infty} p^2(t)dt} \tag{7}$$

Both integrals must include the period between the beginning of the impulse response and 50 milliseconds, meaning that both include the direct sound. The definition parameter (D_{50}) is expressed in percentage, being D = 100% only direct sound without any reflections.

2.6 Speech Intelligibility Index (STI)

Nowadays the most common objective parameter used to determine speech intelligibility is STI. This parameter was introduced by Houtgast and Steenken in 1971 [9] and accepted by the Acoustical Society of America (ASA) in 1980.

This method is based on the principle that interferences such as background noise and reverberation existing in any room produce reduction in modulation indexes, that are crucial for speech transmission.

The STI parameter studies particularly 14 modulation frequencies (f_m) with third of octave intervals from 0.63 to 12.5 Hz. These frequencies modulates 7 octave bands centered in the frequencies between 125 Hz and 8 KHz (f_0).

By studying and analysing the modulation transfer function (*MTF*) obtained by modulating each f_0 with every f_m, which leads to 98 values that are averaged according to the method explained by the EN 60268-16 standard. This leads to seven new values, one for each f_0 frequency band.

Finally to obtain the STI final result, each band values is weighted differently to obtain two STI: one for male and another for female voices. This is not arbitrary, it is made like this due to the difference in frequency spectrum that female and male voices have.

STI predicts the likelihood of syllables, words and sentences being comprehended following values expressed in Table 2:

STI value	Speech intelligibility
0 to 0.3	Bad
0.3 to 0.45	Poor
0.45 to 0.6	Fair
0.6 to 0.75	Good
0.75 to 1	Excellent

Table 2: Speech intelligibility according to STI values (EN 60268-16).

2.7 Articulation Loss of Consonants (%Alcons)

%ALcons stands for Percentage Articulation Loss of Consonants. The higher %ALcons gets, the worse the intelligibility. V.M.A Peutz introduced this concept in a study in 1971, where listeners tried to guess invented words, conformed by a vowel between two consonants and the percentage of wrong guesses would determine the value of %ALcons by the following equation.

$$ALCONS = 200\frac{(rRT)^2}{V} \tag{8}$$

where r is distance to the source, RT_{60} is the averaged RT over 1 and 2 KHz, and V is volume of the room where the measurement is being held.

The assumptions made in this calculation are that there is 25 dB of signal to noise ratio at the listening position (low background noise) and that the frequency response is uniform in critical speech intelligibility bands.

Peutz states in his study that if the AL is below 10%, then the intelligibility is very good. Between 10 and 15%, the intelligibility is good and only if the speaker/talker and/or listeners are not good will the intelligibility be insufficient. Above 15% the intelligibility is only good for good listeners and good speakers, being 15% considered as the practical working limit [10]. Optimal Alcons values are summarized in Table 3.

Alcons[%]	Speech intelligibility
< 3	Excellent
3 - 8	Good
8 - 11	Fair
11 - 20	Poor
> 20	Bad

Table 3: Alcons intelligibility subjective perception.

2.8 Echo Speech & Echo Music

Disturbing echoes can be easily detected by the Echo criterion developed by Dietsch and Kraak. It can be determined from the energy impulse response, but first the center time function has to be calculated using the equation built-up by Kürer [11].

$$t_s(\tau) = \frac{\int_0^t t|p^n(t)|dt}{\int_0^t |p^n(t)|dt} \tag{9}$$

where n = 2 and $\tau = \infty$ are the values to provide the center time.

Exchanging the upper integration limit from infinity to a variable one where the exponent n equals 2 in the traditional center time formula. The final criterion is given by the ratio of increase of energy to a given time interval, $\Delta tS(\tau)\,/\,\Delta\tau$.

An echo will appear if ΔtS would be above the echo threshold ΔtSE in a given period $\Delta\tau$. The maximum value that ΔtSE can be depends on whether music or speech is concerned. Table 4 shows the critical maximum values of the difference quotient for speech and music as well the recommended values of the exponent *n* and the time interval $\Delta\tau$, all the values proposed by Dietsch and Kraak are the result of extensive listening test [12].

$$EK(\tau) = \frac{t_s(\tau) - t_s(\tau - \tau_E)}{\Delta\tau_E} \tag{10}$$

Signal	n	$\Delta\tau_e$	EK 50%	BW [Hz]
Speech	2/3	9 ms	1.0	700-1400
Music	1	14 ms	1.8	700-2800

Table 4: Recommended values for Echo criteria, regarding music and speech [13].

2.9 Strength

Loudness is one of the most important subjective parameters as it leads to correct appreciation of music. A high definition sound or a good level of clarity is quite difficult to reach if the sound is too weak to be heard. For this reason the strength factor (G) was developed: it relates very closely with loudness, and allows to know how strong the sound level is perceived in every position in a concert hall.

The strength factor (G) is the measurement of the SPL, given a certain omnidirectional source situated on the stage, in a particular place minus the SPL level measured at a distance of 10 meters of the same source at the same power level inside an anechoic room.

The equation that lead to this parameter's value is the following:

$$G = 10\log\left(\frac{100}{r^2} + 2.08e^{-0.02r}\right) \tag{11}$$

where r is the distance from the measurement position to the source.

It is important to mention that G = 0 dB is established for a distance of 40 meters, considering that it is recommended for most auditoriums that the longest distance from the stage to the last row do not exceed 40 meters. Beranek suggests that values from -1 dB to 2 dB are optimal for opera houses.

2.10 Sound Pressure Level Distribution (SPL)

Sound Pressure Level is defined as the local pressure deviation from ambient atmospheric pressure, measured in logarithmic scale. The reference value used for calculations in the air is p_0= 20 µPa. Equation 12 shows SPL calculation given a known pressure 'p' value.

$$L_p = 20\log_{10}\frac{p}{p_0} \tag{12}$$

Sound pressure intensity decreases with distance for spherical sound waves. However, this is not the case inside a room, where walls reflect sound energy and enclose a reverberant field. SPL tends to be constant for positions inside this field, which boundary radius is wavelength dependant.

For a representative measurement of SPL distribution in a room it is required a rather constant signal source, that contemplates all frequencies belonging to the frequency range of interest. For this reason, white or pink noise are usually used. White noise signals have equal energy distribution for each frequency, while pink noise present equal energy distribution for each frequency band.

SPL measurements allow to determine a Signal To Noise Ratio parameter, which can validate sound source power level used for all measurements. A common criteria applied by several international standards consists in guaranteeing a 10 dB difference between signal and noise levels.

2.11 Lateral Fraction (LF)

Barron and Marshall in 1981 have defined the apparent source width (ASW) as the subjective perception associated with early lateral reflections [14], meaning that as the lateral energy level increases, the source appears to be wider and the music gains body and fullness. After many laboratory experiments, they came to the conclusion that listeners could relate the ASW to a monaural acoustical measurement, and they called it lateral energy fraction (LF), which is mathematically defined for Early and Late versions according to equation 13 and 14.

$$LF_E = \frac{\int_{0.005}^{0.08} \left[p_8^2(t)\cos\theta\right] dt}{\int_0^{0.08} \left[p^2(t)\cos\theta\right] dt} \tag{13}$$

$$LF_L = \frac{\int_{0.08}^{\infty} \left[p_8^2(t) cos\theta\right] dt}{\int_0^{\infty} \left[p^2(t) cos\theta\right] dt} \tag{14}$$

where θ is the lateral angle (where θ is 90° from straight ahead), $p_8\ (t)$is the sound pressure measured with and figure-8 microphone with the null axis pointed at the source, and $p\ \ (t)$is the sound pressure measured with an omnidirectional microphone.

2.12 Interaural Cross Correlation (IACC)

The IACC, meaning InterAural Cross-Correlation coefficient, is nowadays discussed more than ever before. For Prof. Yoichi Ando, one of the most important investigators regarding the topic, IACC is determined considering reflections at every angle, with integration limits from zero to several seconds (depending on the auditorium RT), using various music motifs and sound sources.

According to Ando, IACC is one of the four orthogonal acoustical parameters that explains the subjective preferences of the listeners that are involved in his experiments.

It is reasonable to believe that a sound wave arriving at a listener position from a lateral direction will excite differently the two ears, and a sound wave coming from straight ahead will excite both ears in the same way. A binaural measure of this difference, is the interaural cross-correlation function, that stands for:

$$IACC(\tau) = \frac{\int_{t1}^{t2} p_L(t)(t+\tau)dt}{\sqrt{[\int_{t1}^{t2} p_L^2(t)dt][\int_{t1}^{t2} p_R^2(t)dt]}} \tag{15}$$

where R and L designates the entrances to left and right ears.

The maximum value possible for equation 15 is 1. As the time it takes for a sound wave with frontal incidence to reach both ears is approximately one millisecond [15], it is customary to vary τ from -1 to 1 ms. Besides, to obtain a single number that measure the maximum similarity of all waves arriving at the two ears within the integration limits and the range of τ, it used the maximum value, named IACC.

$$IACC = \begin{cases} |IACC(\tau)|_{MAX} \\ -1 < \tau < 1 \end{cases} \tag{16}$$

Furthermore, different IACC can be calculated depending on the interval time range (in seconds) used such as:

- $IACC_{Early}\ (t_1 = 0\ \&\ t_2 = 0.08\ )$
- $IACC_{Late}\ (t_1 = 0.08\ \&\ t_2 = 0.5\ ||\ 2\ )$
- $IACC_{ALL}\ (t_1 = 0\ \&\ t_2 = \infty)$

On the other hand, by averaging $IACC_{Early}$ across 500, 1000 and 2000 Hz octave bands, $IACC_{E3B}$ can be calculated.

It was found by Ando in 1998, that it listeners preferred lower IACC values, meaning less correlation between the sound wave that arrives to each ear.

2.13 Listener Envelopment (LEV)

Soulondre, Lavoie and Scott [16] developed an experiment to quantify the listener envelopment (LEV).

This experiment consisted in surrounding the listeners by the sound of five loudspeakers, one frontal, two +/- 30° and two +/- 110°. The stimulus was a 20 s segment of anechoic music, Handel's Water Music specifically.

Direct sound came from from loudspeaker and early reflections and reverberant sound came from the other four speakers. The reverberant sound and some of the early reflections were varied as well as the strength and reverberation time.

The subjects that concurred to the test were asked to rate only their perception of being enveloped or surrounded by the sound. Three parameters were measured, late lateral fraction (LF_L), measured with a figure-eight microphone and integrated after 80 ms, and the two others were late total energy (G_L) and reverberation time (*RT*). All three parameters were calculated in octave bands.

Leo L. Beranek [17] presented results of this calculation for 22 concert halls in Table 5

Concert hall	LEV [dB]
Zurich, Groser Tonhallesaal	2.42
Vienna, GMVS	2.04
Basel, Stadt-Casino	1.90
Amsterdam, Concertgebouw	1.42
Berlin, Konzerthaus	1.24
Tokyo, TOC Concert Hall	1.03
Vienna, Konzerthaus	0.9
Tokyo, Suntory Hall	0.44
Boston, Symphony Hall	0.35
Kyoto, Concert Hall	0.11
Lenox,Tanglewood Music Shed	0.06
Baltimore, Symphony Hall	-0.02
Munich, Phnilharmonie	-0.11
Costa Mesa, Orange County	-0.12
Berlin, Philharmonie	-0.15
Tokyo, Met. Arts Space	-0.45
Tokyo, Bunka Kaikan	-0.54
Tokyo, Orchard Hall	-1.09
Sapporo, "Kitara" Hall	-1.46
Salt Lake City, Abramavel Hall	-1.48
Buffalo, Kleinhans Hall	-2.16
Tokyo, H-Auditorium	-2.60

Table 5: Beranek's LEV results for concert halls

The equation 17 correlates highly with the listeners judgments.

$$LEV = 0.5G_{L,mid} + 10\log LF_{L,mid} \tag{17}$$

where $G_{Late,mid}$ can be replace by equation 18.

$$G_{L,mid} = G - 10\log\left(1 + \log^{-1} 0.1C_{80}\right) \tag{18}$$

where C_{80} refers to the clarity factor.

2.14 Subjective Parameters

Objective parameters can be easily calculated by applying different mathematical equations and derived algorithms to a signal of interest. However, this results do not reflect the human perception of sound. For this reason, subjective parameters are needed. The human hearing system requires to process sounds in real time, nevertheless, this can not be the case since signal analysis is always time dependant. Therefore, it requires to work on the signal at pieces, with a defined starting and ending times. These points determine an integration time window, while there are two distinct views regarding its duration: on one hand, those who establish a fixed time window, like Rossing [18]; on the other hand, those who claim that it depends on temporal and spatial processes in the brain.

Ando [19] proposed a neuronal processing model for continuous autocorrelation function analysis, as illustrated in Figure 4.

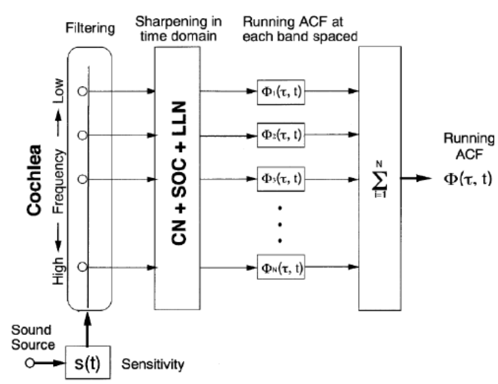

Figure 4: Ando's neuronal processing model.

A sensitivity factor models the outer and middle ear influence over the sound source, then in the cochlea the signal is filtered into discrete frequency bands. After a sharpening process, a running autocorrelation function is applied to each band spaced, then summed for each ear channel individually and sent to the left hemisphere, where temporal information is analyzed for both ears, as opposed to the right hemisphere which takes care of spatial information through the analysis of the cross correlation between left and right ear.

2.15 Effective ACF envelope duration (τ_e)

The autocorrelation function (ACF) describes a signal similarity to itself as a function of time. There are two main equations that describe long and short term behaviour. The long-term autocorrelation function is defined following equation for a 2T integration time interval.

$$\phi_p(\tau) = \lim_{T\to\infty} \frac{1}{2T} \int_{-T}^{T} p'(t)p'(t+\tau)dt \qquad (19)$$

where p' is the sound pressure level convoluted with the previously described ear sensibility function.

Similarly, the short-term autocorrelation function is described following equation.

$$\phi_p(\tau) = \frac{\frac{1}{2T}\int_{-T}^{T} p'(s)p'(s+\tau)ds}{\sqrt{\phi_P(0;t,T)\phi_P(0;t+\tau,T)}} \qquad (20)$$

where the ACF is computed during the time window, following the running step.

τ_e is defined as the time instant in which normalized ACF amplitude is -10 dB respect to its maximum at t = 0, which corresponds to the signal's sound pressure level. By taking the percentile 95%, and $\tau_{e\ min}$ parameters can be derived, which can describe sound source information density, as lower $\tau_{e\ min}$ values correspond to higher information density, according to Kato et al [20].

According to Ando, appropriate integration time windows range from 30 ms to 1000 ms, depending on the sound source. Table 6 shows recommended integration period according to music genre.

ACF	Recommended 2T [ms]
Vocal	40 - 80
Jazz	80 - 150
Orchestral	600 - 1000
Organ	1000 - 1500

Table 6: Recommended integration intervals for ACF calculations depending on genre.

It is worth mentioning that using a longer integration time window will generate a low pass filter effect on the signal. [21]

3. ROMA THEATRE

3.1 Characteristics

Roma Theatre is situated in the city of Avellaneda, located in Buenos Aires, Argentina.

It was finally built after eleven months in 1904, by architect Primitivo Gamba, of italian roots, inspired by european concert halls of the "golden century of opera". The building was constructed over two adjacent lands belonging to "Sociedad Italiana de Socorros Mutuos (1888)", later integrated into the complex. Its characteristical dome and allegories of Dante Alighieri and Pirandello were designed by artist Antonio Epifani, while the facade is ornamented in the italian renaissance style.

Until Teatro Colon was built in 1908, Teatro Roma was considered the best concert hall in América.

Initially recognized as "the theatre of the south", Teatro Roma originally had room for 400 people and after its remodeling in 1925, its seating capacity grew to over 500 seats. It was later remodeled in 1984 by Avellaneda municipality, after confiscating the building as a "public facility" in 1980.

Figure 5: Teatro Roma, historical picture (1957).

It was last restored in 2015. Repairs mainly consisted in masonry, repainting, chair upholstering replacement, new carpet installation, stage restoration, dressing room and bathroom reconditioning, as well as heating service integration. According to municipal officials, seat number and facade should not be modified in order to preserve the original characteristics.

Now declared provincial and national cultural and architectonic monument in Argentina, Teatro Roma constitutes a historical heritage. Its acoustical properties make it one of the best of the country still.

Figure 6: Teatro Roma, actual picture (2017).

The concert hall is used for around 250 shows per year, mainly for theatrical plays, jazz sessions, tango, celtic music, folklore, orchestral ensembles ballet and opera. Moreover, each month didactical concerts take place with special activities for schools. Every year Teatro Roma gathers around 180.000 people.

The concert hall is illuminated by a central chandelier made out of bronze with glass bulbs as it can be observed in Figure 7.

Figure 7: Teatro Roma in concert.

Floor carpet from ground, first and second level are made out of velvet material. The main audience area chairs upholstering are made out of a velvet. On the other hand, chairs in the balconies are made out of wood. The ceiling walls are made of concrete. There are plaster moldings, with bronze ornamentation. The wooden stage outer layer has a surface rubber material which prevents light reflections. There are 7 curtains which are made of heavy velvet in this case. Between the stage and the first row there is the orchestra pit, made out of wood.

An approximation of the concert hall volume had to be made for parameter analysis, it is estimated to be 3875 m^3. Consequently, its volume per person is calculated to be 7.75 m^3. According to Table 7, Teatro Roma fits in between the opera, multipurpose hall and the concert hall category.

Purpose	Volume / Person [m^3]
Conference Room	3 - 5
Drama Theater	4 - 6
Opera Theater	6 - 8
Concert Hall	8 - 12

Table 7: Recommended volume per person

3.2 Theatre Blueprints

Figure 8 shows Roma Theatre's blueprints.

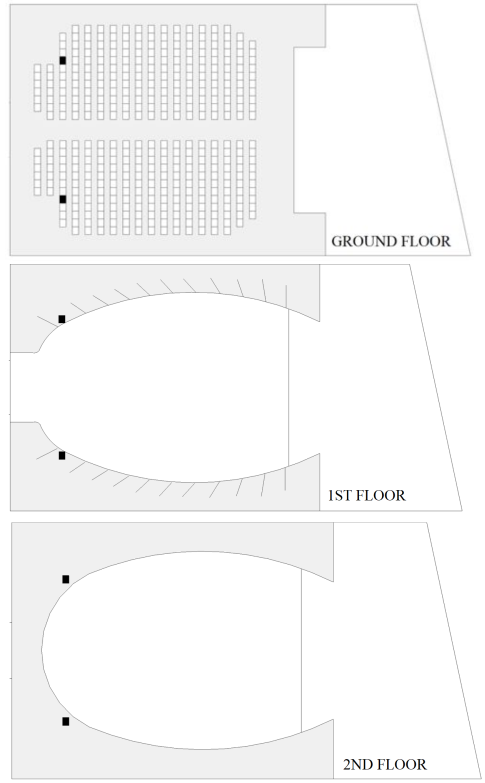

Figure 8: Roma Theatre's blueprints.

3.3 Seat Distribution

The analysed hall is a three-story building. The main audience area is on the ground floor, with 18 seats row with six to twelve seats each, depending on its location in the theatre. There are two balcony levels, two rows each, distributed following the building shape. It is important to remark that balconies were not analysed in this report because of a setback during measurement preparations which prevented the full mapping of the 1st and 2nd floor. However, the ground floor was mapped successfully with two sources positions and an array of 48 microphone positions evenly distributed. Since this could not be made in the balconies, instead of displaying a non-representative analysis, it was prefered to focus on the main audience area, shown in Figure 9.

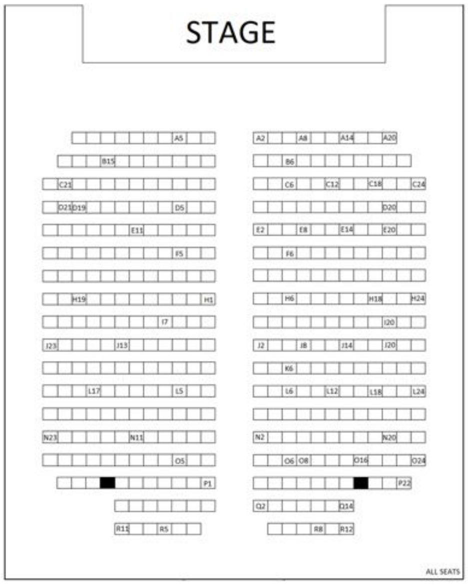

Figure 9: Roma Theatre's main audience area.

A combination of letters and numbers are shown in the seats that were measured. The letter represents each row, ascending alphabetically from the stage towards the exit doors; the number represents the seat column, ascending in odd numbers from center to the left wall, and even numbers from center to right wall.

4. METHOD

4.1 Equipment Used

The measurement was carried out by 20 people approximately, organised in little groups with specific tasks: recording and playing stimuli, loudspeaker sources management, microphone positioning, and architectonic measurements. The following list presents the main equipment used in this measurement:

- 1 RME Fireface UFX sound interface (4 ch)
- 2 RME ADAT sound cards (8 ch each)
- 1 MacBook Pro 2010 13" notebook
- Adobe Audition CS6 DAW
- 16 Earthworks M50 microphones (matched)
- 2 DPA 4060 In Ear microphones
- 1 SPS200 soundfield microphone
- 17 KRV microphone stands
- About 30 XLR cables
- 1 Zoom H4N recorder
- 1 Svantek SV 30A calibrator
- 1 Bruel & Kjaer 4231 calibrator
- 1 Outline GSR-Globe Dodecahedron
- 1 Yamaha MSR250 two-way speaker
- 1 Bosch DLE 70 laser distance meter

4.2 Recording Procedure

The measurements were carried on with two independent recording configurations. The 16 Earthworks M50 and the soundfield microphone (which consists of 4 microphone capsules in a specific array) were connected to the RME system. The main unit has 4 analog XLR input with microphone preamplifiers and phantom power, that were used for the soundfield microphone. The remaining 16 matched microphones were in the additional inputs (also including high quality microphone preamplifiers and phantom power sources) connected via ADAT to the master sound card, as shown in Figure 10.

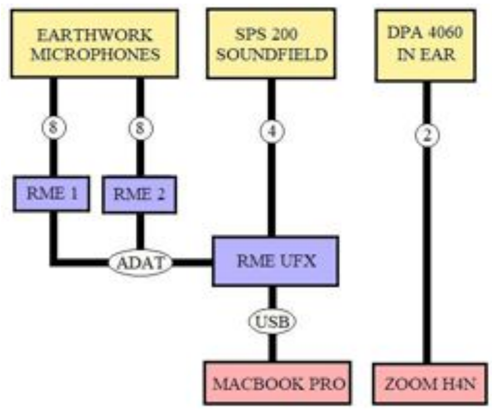

Figure 10: Sound sources, measurement set up

The recording was taken in Adobe Audition CS6 using a Macbook, creating a session with the test signals routed to the RME soundcard outputs and simultaneously recording the 20 microphone inputs. Then, 20 independent WAV files were created for each measurement in each microphone, containing the 6 test signals distributed over time.

The other recording station was dedicated to binaural measurements with the in ear microphones using the Zoom H4N as the recording system. Both signals were connected to the recorder, operated by a technician that simultaneously recorded the signal in sync with the other recording system.

Every microphone was calibrated before and after the measurement with its corresponding calibrator. Svantek SV-30 with ±0.3 dB SPL accuracy and ±0.2% frequency accuracy was used for M50 microphones, whereas Brüel & Kjær with ±0.2 dB SPL accuracy and 0.1% frequency accuracy was used for soundfield microphones.

4.2.1 Test Signals & Sound Source

The test signals were chosen and synthesized in order to meet the analysis requirements for the parameters. There were used a total of 6 different signals:

- Log sine sweep (90 s)
- Pink noise (15 s)
- Anechoic sample N°1 (18 s)
- Anechoic sample N°2 (50 s)
- Anechoic sample N°3 (16 s)
- Anechoic sample N°4 (25 s)

The log sine sweep was generated in Audacity with Farina's plugins. The total length was of 90 seconds, and the sweep covered from 30 Hz to 18 KHz.

This plugin also synthesized the inverse filter that was necessary to generate the IR for each audio file. It is important to notice that the log sine sweeps broadband is wider than the useful frequency range of the sound source.

This is intentional, in order to reach a stationary stage while in the useful range and have some wide boundaries in the post-processing.

The pink noise signal consisted of Gaussian white noise with the corresponding filter with a total length of 15 seconds.

The musical test signals are: a drum set, a baritone singer, a string trio and a reduced-orchestra passage, respectively.

There were two sound sources used in this measurement: a dodecahedron source (with omnidirectional radiation pattern) and the Yamaha 2-way active loudspeaker.

The first one was used for the objective parameters, so it reproduced the log sine sweep and the pink noise.

The anechoic audios were reproduced in the Yamaha loudspeaker, as prefered for the analysis of Ando parameters.

Employed sound sources are illustrated in Figure 11, picture taken the day of the measurement.

Figure 11: Sound sources, measurement set up.

4.2.2 Recording Set-up

The setup was based in an Adobe Audition CS6 multitrack session, with an input channel for each microphone (with its monitor output muted) and an additional track for the playback of the mentioned signals. The 6 signals were reproduced in a sequence: log sine sweep in dodecahedron, pink noise in dodecahedron and then the 4 musical passages in the Yamaha loudspeaker.

The set with the Zoom H4N recorder consisted on the in ear microphones connected to the handheld recorder. Both recording systems were set in WAV format, 48 KHz of sampling rate and 24 bits of depth.

The measurement was carried on during the morning and afternoon of September the 11th. The weather conditions at the beginning of the measurement were regular for this metropolitan area with humid subtropical climate. The stage curtains were opened, nothing but the sound sources and a table with accessories for the measurement was placed at the stage. The hall was in unoccupied conditions, there was just about 20 technicians with the equipment. All exit doors, including backstage were closed.

4.2.3 Microphone Array Distribution

The audience area was divided into three sections for its mapping in different positions that could represent any acoustic variation thru the hall. All microphones were placed in three different arrays, recording two times, one for each source position.

While the Earthworks microphones were moved into a new array, the soundfield and the in ear were used for extra measurements in order to obtain a better mapping of the hall. The measurement positions were chosen to map the audience area, so they were distributed all along the seats, not assuming symmetry in the audience area. Every microphone position meets the ISO 3382 requirements. Figure 12 shows Earthworks microphone positions.

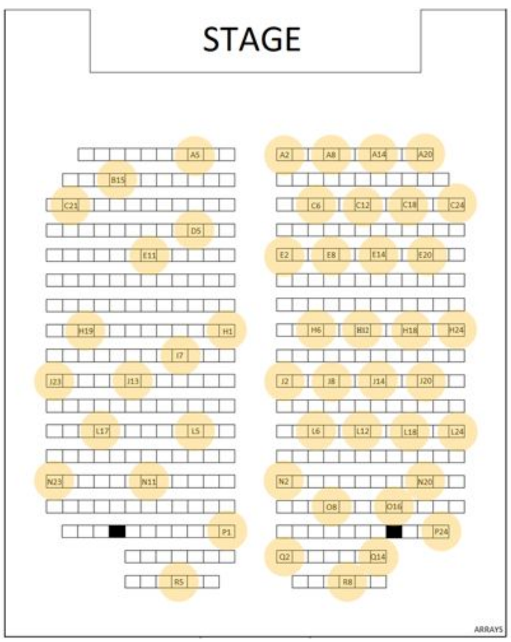

Figure 12: Measurement positions for Earthworks

4.2.4 Soundfield Measurement

The soundfield microphone consists of an array of 4 capsules in individual channels. This kind of measurement allows post-processing in order to simulate different polar patterns for the microphone. This microphone has 4 XLR outputs that carry the signal in each capsule to be recorded independently.

Since sound source measurement instances were limited, the amount of soundfield recordings was a total of 7. For this reason, microphone positions were arranged in such a way that most of the audience area can be mapped if symmetry is assumed. The positions are shown in Figure 13.

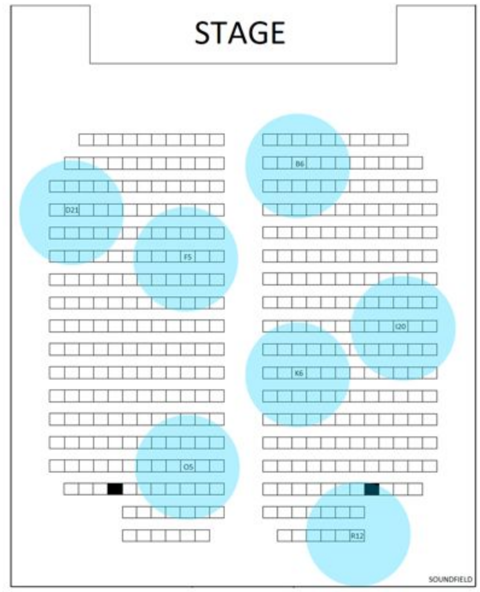

Figure 13: Soundfield microphone positions

4.2.5 Binaural Measurement

Figure 14 show the measurement positions, set with similar guidelines to the sound-field mics.

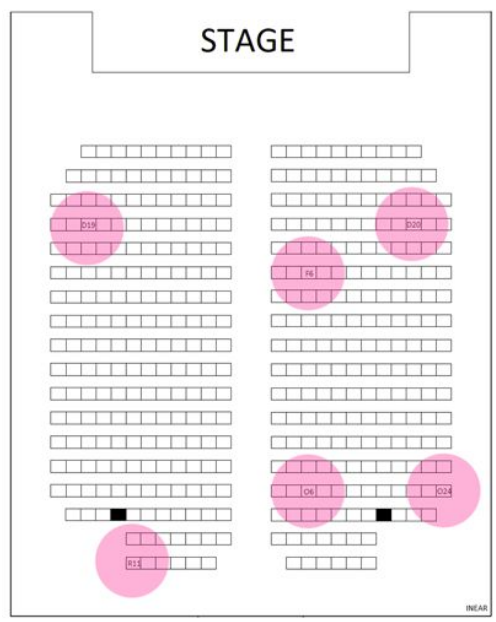

Figure 14: In-ear microphone positions.

The binaural measurement consists on a representation of human hearing. A technician is equipped with in-ear microphones taped to its head, like a dummy head. This is important to obtain parameters that relate left and right listening. For every change in sound source position, the technician changes seats twice. A total of 6 recordings were made.

4.3 Post-Processing

After the measurement, more than 10 GB of data was recorded. Useful audio files and failed takes were mixed, waveforms presented noises and other undesired information. All issues needed to be solved manually.

The first stage was labeling each audio (related to the microphone number and the position) and ordering them in folders depending on the source position ('1' or '2'). Once in order, each take of each microphone was a series of 6 audio signal that had to be separated, so this was the next stage: cropping audio, renaming and reordering in new folders, depending on the test signal contained. At this point, only anechoic audio files are ready to be processed for IACC and τ_e analysis. The other parameters require an impulse response as an input signal. At the end, the total number of files to process was 880.

4.3.1 Impulse Response Recovery

The impulse response recovery method consists in reproducing a log sine sweep signal *x[n]* through a loudspeaker and then recording the room signal *y[n]* with a microphone. In post-processing, *y[n]* signal is convoluted with the inverse filter $x^{-1}[n]$, synthesized using the original the log sine sweep signal *x[n]*, in order to get the room impulse response *h[n]*, which characterizes the system, in this case the theatre (with added noise mainly due to allinealities in the measurement equipment).

Audio files that contained a log sine sweep were manually convoluted using Farina's plugin *Aurora Convolver* and then cropped to have a clean impulse response (IR) with an average of 6 seconds each.

Once this stage is complete, IR files for both earthworks and in-ear measurements are ready to be analysed using other programs in order to get the desired parameter values.

4.3.2 Soundfield processing

In order to work with the soundfield files, a process is needed to be done using the SPS200 Surround Zone plugin in the Adobe Audition platform.

By using the plugin shown in Figure 15, the 4 channels *RF*, *RB*, *LF* and *LB* can be transformed into a stereo mid-side file (left channel: omni, right channel: side) that can be read by the *Aurora* plugin to obtain lateral fraction parameters.

Figure 15: SPS200 Surround Zone plugin.

4.3.3 Obtaining parameter values

The obtained impulse response, pink noise and anechoic audio files recorded have to be processed in specialized software in order to obtain each parameter value for each seat position. Table 8 shows the software used for every parameter.

Parameter	Software	Module
RT_{20}	Aurora	Acoustical Parameters
EDT	Aurora	Acoustical Parameters
D/R	Matlab	Dir2Rev_Ratio.m
C_{50}	Aurora	Acoustical Parameters
C_{80}	Aurora	Acoustical Parameters
D_{50}	Aurora	Acoustical Parameters
STI	Easera	STI module
Alcons	Easera	Alcons module
EC speech	Easera	Echo Speech module
EC music	Easera	Echo Music module
G	Aurora	Acoustical Parameters
LEV	Sheets	Aux_calculations
BR	Sheets	Aux_calculations
LF_E	Matlab	lateralFraction_ldb.m
LF_L	Matlab	lateralFraction_ldb.m
IACC	Aurora	Acoustical Parameters
SPL	Matlab	SPL.m
$\tau_{e\ min}$	Matlab	tauE_ldb.m
$\tau_{e\ mean}$	Matlab	tauE_ldb.m
IACC	Matlab	IACC.m

Table 8: Parameter calculations

As it is shown in Table 8, RT_{20}, EDT, C_{50}, C_{80}, D_{50}, and G parameters were obtained by using *Acoustical Parameters* module part of the Aurora plugin in the Audacity environment. Results could be appended to a external file which can be imported into an excel file, showing every parameter in third-octave bands, as well as its global value.

Parameters such as STI, Alcons, Echo Speech and Echo Music were processed using *Easera* modules, which presented results in a global fashion.

Parameters that could not be processed by *Aurora* nor *Easera,* had to be obtained by developing Matlab scripts. Such was the case for D/R, Late

Lateral Fraction and SPL distribution. In each case, the script consisted in calling the IR files with the *audioread* function, applying a third-octave filter according to ISO 61260 guidelines, and then using the respective algorithm to compute each parameter, according to Equation 12, 13 and 14.
In this work the D/R was calculated using a developed software in Matlab, where the time limit between DS and reverberant sound was 2 ms assuming that all the energy arriving after that period is reverberant sound.
For anechoic audio files such as IACC, $\tau_{e\text{-}min}$ and $\tau_{e\text{-}mean}$ were processed also developing special Matlab script based on autocorrelation algorithms.
Finally, LEV parameter was obtained by cell calculations using Equation 17 between G and IACC parameters using Google Sheets platform.
Parameter results were stored in an collaborative *.gsheets* file where every team member could easily upload their results. All scripts used for calculation are annexed to this document folder, including the final results *.xlsx* file.

4.3.4 Running Impulse Response Window

In order to obtain subjective parameters, IR signals must be temporally integrated first. The script responsible for applying the integration window and autocorrelation processes to each audio file is *integrationWindow_ldb.m*, which is a specially developed script that adapts digital signal processing code developed by Shin-Ichi Sato and Alejandro Bidondo.
The adaptation mainly consisted in file multi-selection for automated iterative processing. Moreover, an input dialog was included at the beginning to specify several integration time values, which were set to 10 ms, 100 ms and 350 ms. After DSP is applied, each windowed IR audio file is generated by the use of the *audiowrite* function. The script was left working for 8 hours until every running impulse response file was finally created for every window of interest.
All windowed impulse responses presented a low frequency boost when played, this may be due to RC integration analogy, which smoothes transients and acts as a low pass filter. In order to characterize the code behaviour, an spectrum analysis was done to several IR files and their windowed versions for 10, 100 and 350 ms integration times.
Figure 16 shows a comparison between IR and its windowed spectrums from seat A2.

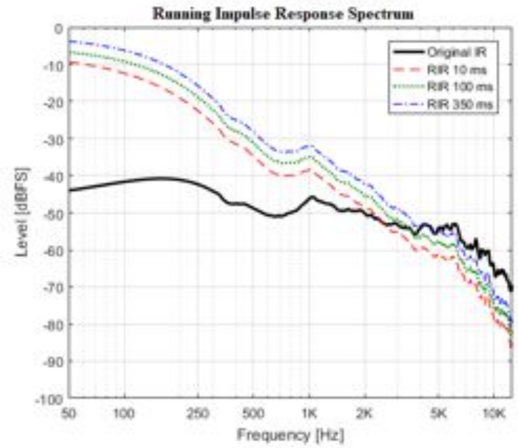

Figure 16: IR windowing spectrum for seat A2.

It can be observed that windowing process applies a boost of 40 dB for low frequencies, that decreases intensity until 2 KHz when magnitudes equalize. The observed effect is not of a LPF, but rather more of an active low boost.
It is worth to notice that audio files with integration times of 350 ms and 100 ms presented undesired noise at the end of the IR tail. To solve this issue, every 100 ms audio file was cropped to 4 s length, while every 350 ms file was necessary to be cut at the 2 s mark, losing valuable tail information. The 10 ms audio files did not present any detectable noise, conserving 6 seconds length.

4.4 Mapping Algorithm

After calculating all parameters, the "*.gsheet"* file was downloaded to a computer hard drive as an "*.xlsx"* file.
In order to generate graphical information for every parameter, a script was needed to be developed. *ParameterMapper_ldb* was written from scratch. If the "Global Mapping" option is selected, the program prompts the user to load a floorplan image of the theater, which later saves into a variable in unit pixels. Afterwards, the user is prompted to draw the main audience area with the *roipoly* function which generates a logical matrix. Then, the *readPoints*.m custom function is used to select each measurement seat position. This information is saved into an excel file *Seats.xlsx* that can be called back instead of selecting each seat again for every parameter. Parameter data is then imported into the Matlab platform by using built in *xlsread* function. Each cell array was named and saved into a variable labeled according to the parameter. The user is now prompted to select which global parameter to map from a context menu list. Global

values are calculated from taking energetic or linear averages accordingly from third-octave band information between 50 Hz and 10 KHz. Frequency bands outside this range are not considered in this report, due to sound source frequency response limitations. Once this is done, a border treatment is applied: values close to the audience area boundaries are mapped to the closest seat value in order to avoid typical corner distortion that occurs in the next step: interpolation. Every pixel is mapped using two stage interpolation methods: *scatteredinterpolant* and *griddedinterpolant* functions are used to fit every seat value and its surroundings. Then 2D matrices are built using *meshgrid* function and finally the graph is generated by the *surface* plot function. The floorplan map is then overlaid to render the final mapping graph, and saved into a *.png* file.
This process is repeated for each calculated global parameter, using different colormaps depending on the parameter of interest.
It is important to remark that even though the border treatment was successful to prevent destructive corner distortion while allowing accurate representation of values for every seat, a collateral effect is produced where pixels at the image border, that are very close to a mapped seat value, present a significantly wider and distorted representation in the final image. This must be taken into consideration when analyzing mapping figures in this paper.

4.5 Sonogram Algorithm

Within the *ParameterMapper_ldb.m* script, it was possible to include an instance where third-octave parameters can be represented for each analyzed seat position, shown as a sonogram. Seat positions are no longer represented in a map, but as an axis. *X* axis show third octave bands center frequency information; *Y* axis show each measurement position labeled according to the seat name, alphabetically ascending; and *Z* axis (color-scale) shows the parameter value intensity.

5. RESULTS & DISCUSSION

In this section, objective and subjective parameters are analyzed independently. Each global parameter is shown as a map distribution over the main audience area. On the other hand, third-octave information is displayed through sonograms.

5.1 OBJECTIVE PARAMETERS

5.1.1 Reverberation Time

As previously described in section 2, Reverberation Time is one of the most important parameters to describe a concert hall and compare between different theaters. By analysing RT_{20} and RT_{30} values, an idea of the room absorption may be acquired while a correspondence between varying sound energy decay and seat position can be observed. If the decay of the reverberation is homogeneous and regular, RT_{20} and RT_{30} values should be similar. For non regular decay curves, the sound disappearance could differ between -25 and -35 dB for both measurements.
Randall F. Barron [22], in his famous book "Industrial Noise Control and Acoustics", presents a formula, show in equation 21, to calculate objective RT_{500} depending on room volume and its application.

$$RT_{500} = (a) + 2.306(b)\log_{10}(v) \qquad (21)$$

where v is the room volume, which in this case was calculated and approximated to 3875 m^3.

As nomenclature expresses, the Roma Theater's main use is intended to be for *Music studio and opera house*. Therefore, "a" and "b" values should be -0.352 and 0.2, respectively.
Table 9 show the objective (RT_O) and measured (**RT_M**) octave band reverberation time parameters for Roma Theatre.

F [Hz]	125	250	500	1 K	2 K	4 K	8 K
RT_O [s]	1.9	1.5	1.3	1.4	1.4	1.4	1.4
RT_M [s]	1.5	1.5	1.1	0.9	0.9	0.8	0.6

Table 9: Objective and measured octave band RT

It can be observed right away that room averaged measured RT values are close to objective targets for the low frequency range, but this is not the case for high frequencies, as measured RT values are significantly lower.
A gradient map can be generated by using every seat position coordinates assigned to the corresponding global RT values.
Figure 17 and 18 show global RT_{20} and RT_{30} distribution among the theatre.

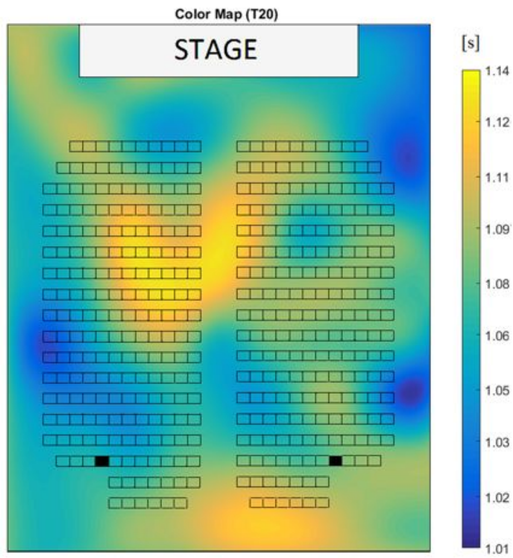

Figure 17: RT_{20} colormap for Roma Theatre.

In RT_{20}, the range is very narrow, between 1.01 and 1.14 for minimum and maximum levels. The higher RT values are displayed in the central audience area, that could be explained by a sound reinforcement provided by the painted central dome of great importance in this hall. This structure might also lead to a flutter echo or an acoustic glare, so the RT values obviously should be affected.

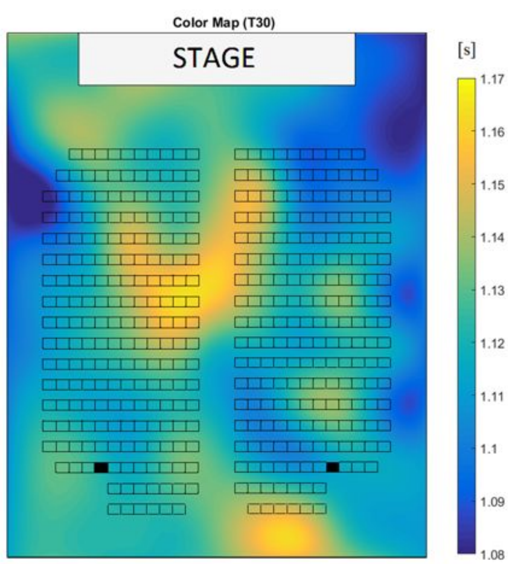

Figure 18: RT_{30} colormap for Roma Theatre.

RT_{30} values are narrower, displaying an almost insignificant difference of 90 ms. It also show the increment in the central area and the lowest values near the sidewalls. This seems to be a strange behavior, considering the audience seats as the main absorptive surface (mainly in mid-high frequencies), and the wall as almost totally reflective, as previously described. Despite this, some cancellation effect due to stationary waves can be the cause of these phenomenon. There are slight differences between the colormaps, that could indicate a regular decay curve between -25 and -35 dB.

Figures 19 and 20 show the RT_{20} and RT_{30} for each measured seat for every frequency.

In both cases, values for frequencies over 500 Hz decrease, even more so for higher frequencies. This might be due to be easiness for any material (specially porous ones) to transform sound waves into heat at this frequency bands. An increment in low frequencies can also be seen in those figures, as expected. There seems to be no acoustic treatment for low frequencies, leading to higher RT values. Between 100 and 250 Hz, RT values vary from 1,2 to 2 seconds for every seat, with preponderance of higher values in RT_{30}. This might mean that the decay of energy curve for lower frequencies is less regular than for mid and high.

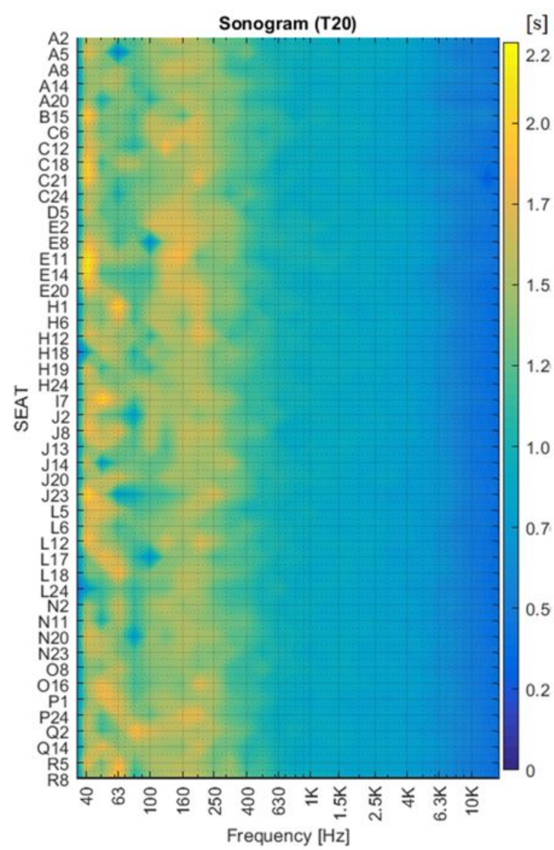

Figure 19: RT_{20} third-octave display for each seat.

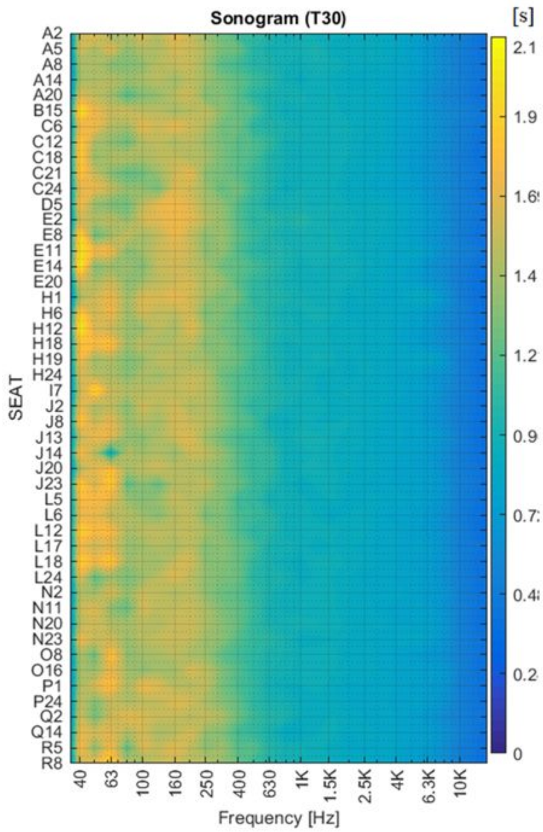

Figure 20: RT_{30} third-octave display for each seat.

As can be seen in wider band figures, values for frequencies below 80 Hz are the highest, showing a typical issue at this band; wavelengths are so big that the hall proportions and lengths have a great impact in sound behavior. In the highest range, the RT values also fall down to the lowest values, mainly because of some air absorption as well as higher absorption values for any material, regardless its position. There is also some drop in RT values around 80 Hz, that might be due to stationary waves or some acoustical effect caused by the balconies.

This results may meet Barron's recommendations for objective RT, just up to 500 Hz, where in the calculations, values remain almost constant towards 1.4 s, while in this hall it falls to 0.8 s for frequencies up to 10 KHz. This also leads to a higher Bass Ratio value for most seats, due to significant RT differences between mid and low frequencies.

Also, an extra analysis of Bass Ratio is presented. Figure 21 shows the distribution in a colormap.

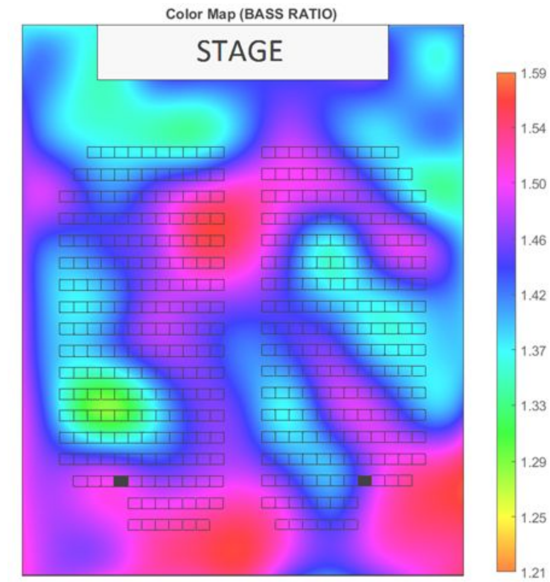

Figure 21: Colormap for Bass Ratio

Bass Ratio may be analysed as an indicator of frequency content in Reverberation Time.

There are some zones with high RT, as shown in RT_{20} and RT_{30} colormaps, that match with high bass ratio values. This means that the mapped value, the global for the parameter is highly influenced by low frequency reverberation, as noticed in the third octave figures.

There are two main spots on the rear of the concert hall where there seems to be high values of low frequency reverberation and a special spot where the dome is placed. This might indicate some stationary waves or room modal problems in the vertical axis (between floor and ceiling).

In the whole audience area, the bass ratio varies from 1.21 to 1.59, meaning that the 125 and 250 Hz band has about 20 to 60% more energy than 500 and 1000 Hz, as could be seen before in this section in RT analysis.

5.1.2 Early Decay Time

As mentioned before, EDT is an important factor for running reverberance perception. It is an additional parameter that describes the room reverberation time, but in this case, based on the first stage of the sound pressure decay curve. If the decay is totally regular, EDT, RT_{20}, RT_{30} and every reverberation parameter should be the same; but this is not very common, mainly due to sound distribution in the room, measurement points and diffusion in the room.

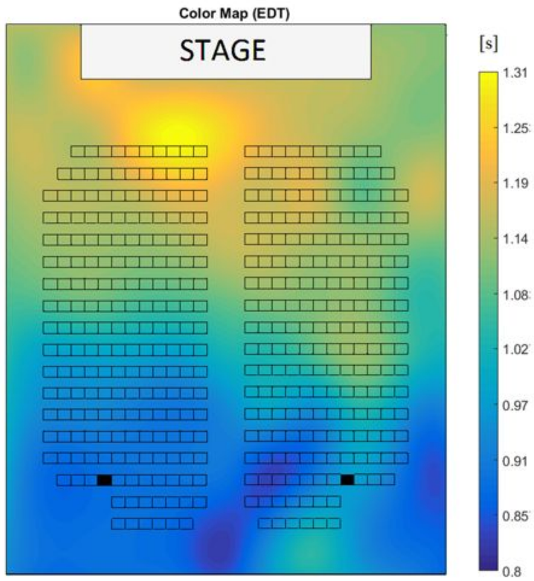

Figure 22: EDT distribution in Roma Theatre.

As it can be observed in Figure 22, this distribution is quite different than the ones for RT_{20} and RT_{20}, with a greater maximum (1.31) and minimum (0.8). Also, the highest values are closer to the stage, over the first rows, up to the middle of the hall. Rear seats present EDT values of 0.9 seconds. This could be explained by some reinforcement provided by the stage area, with harder materials that provide some additional reflections that affect the first milliseconds after the excitation ceased and truncate the decay slope. It might indicate that the type of reverberation in this concert hall is conformed by a considerable amount of early reflections, mainly in the front seats.

Figure 23 shows the distribution over frequency for EDT in each seat. As analysed in section 5.1.1 for RT, the EDT behavior is similar, with higher values in low frequency and almost constant values for mid-high frequencies. Below 80 Hz, the values rise up to almost 3 seconds, and fall below 0.3 for frequencies higher than 8 KHz. However, some *spots* are shown and can be interpreted as a specular reflection, which is usual for EDT. Also, the sound field is less diffuse and there are more variations between seats, less homogeneous than in RT_{20} and RT_{30}. Whether by analyzing global, mid, octave or third octave results, it can be concluded that EDT values for Roma Teatre are low, compared to typical values previously shown in Figure 3 in the theoretical framework section, with a similar trend to RT values.

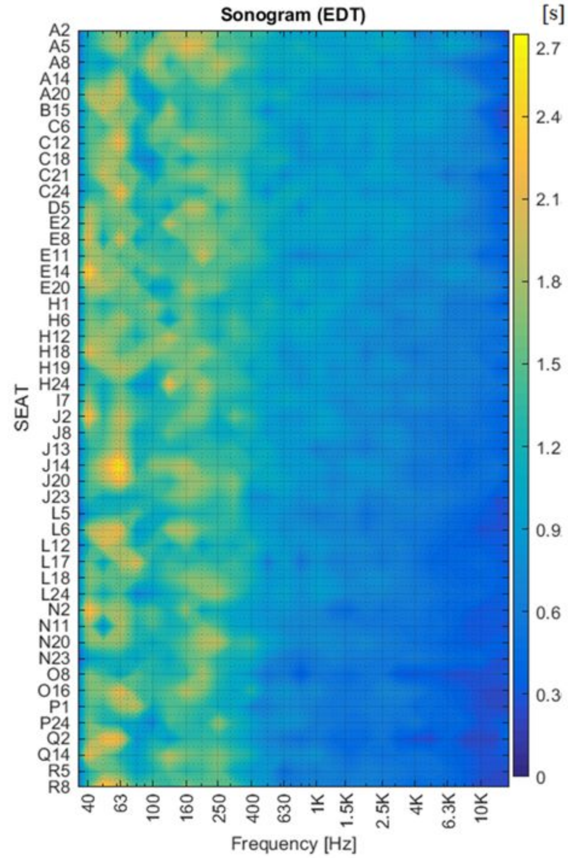

Figure 23: EDT third-octave display for each seat.

5.1.3 Direct to Reverberant Energy Ratio

Figure 24 shows the D/R Ratio in global values for the concert hall.

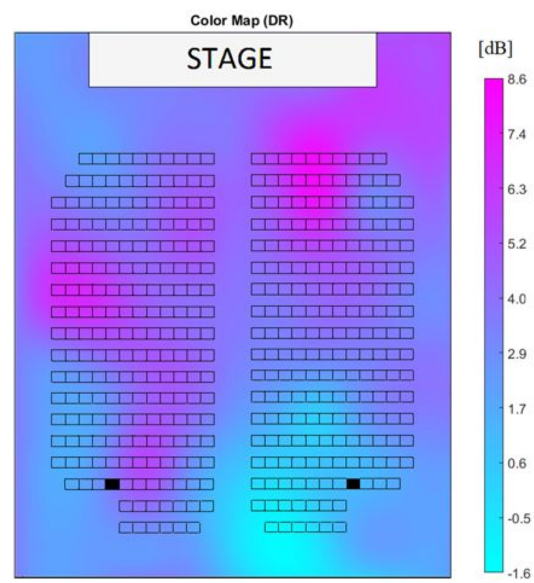

Figure 24: D/R distribution in audience area.

As seen on Figure 24, there is a 10 dB range in the room with higher values in some front seats and the worst in the rear. Some lateral seats; on the left; show an increased value that can be explained by some reinforcement provided by a hard surface as a wall. It can be seen that D/R ratio decreases with distance, as expected. Figure 25 displays the information in third-octave sonograms for each seat measured.

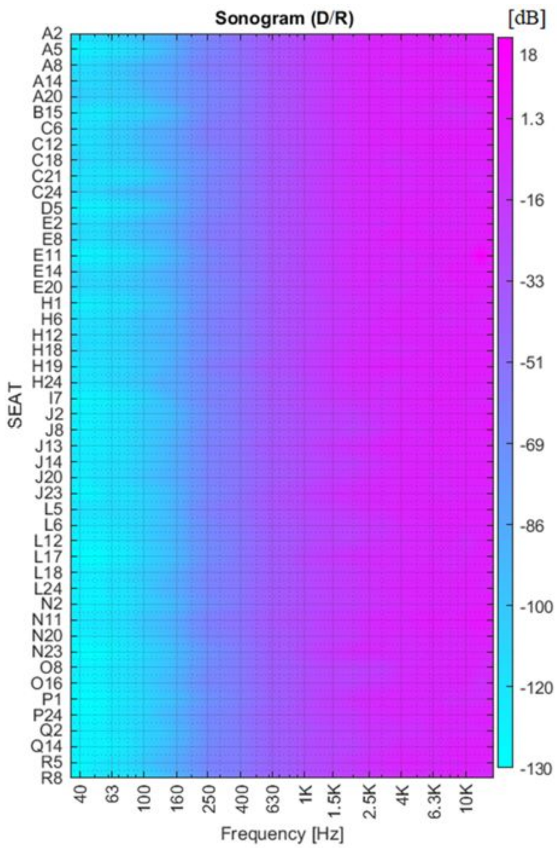

Figure 25: D/R third-octave display for each seat.

This Figure shows a progressive behavior for every seat in the whole spectrum. For frequencies below 160 Hz, the ratio is considerable negative, corresponding to much higher values of reverberated sound, rather than direct sound.

Up to 1.5 KHz, the ratio tends to 0 dB, meaning same energy for 'initial' and 'late' time (after 2 ms in this study).

In the highest range, the reverberation time drops and the direct sound becomes preponderant. As seen on section 5.1.1, the bass ratio at those areas is higher, so the results are as expected in this zone.

It is important to notice that the minimum resulted in more negative values than expected, up to 130 dB. This could be explained by the phenomena in signal widowing, where an undesired LPF is applied. It means that the signal has an undesirable process that can explain the affection in the results. Some phenomenon related to wavelength can be related, because of the short time of considered direct sound.

5.1.4 Clarity

Figures 26 and 27 show the distribution of C_{50} and C_{80} for speech and music, measured in Roma Theatre's audience area.

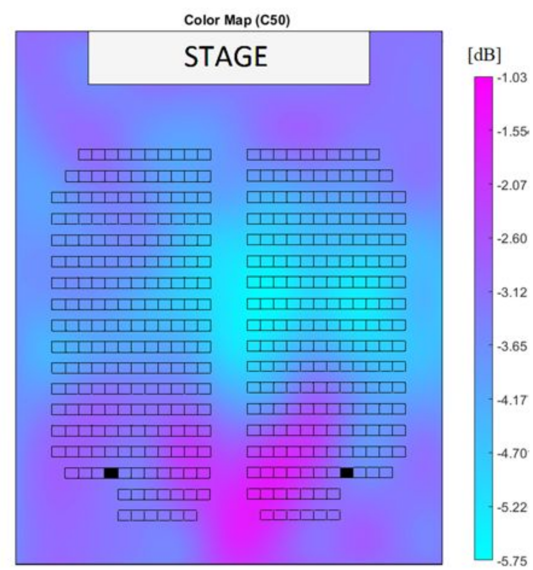

Figure 26: C_{50} distribution in Roma Thatre.

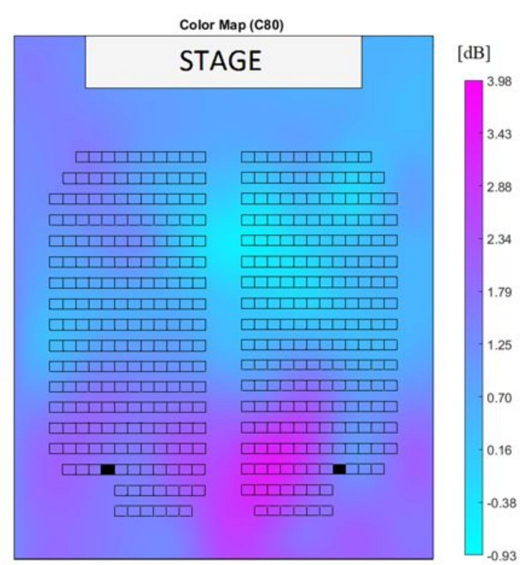

Figure 27: C_{80} distribution in Roma Theatre.

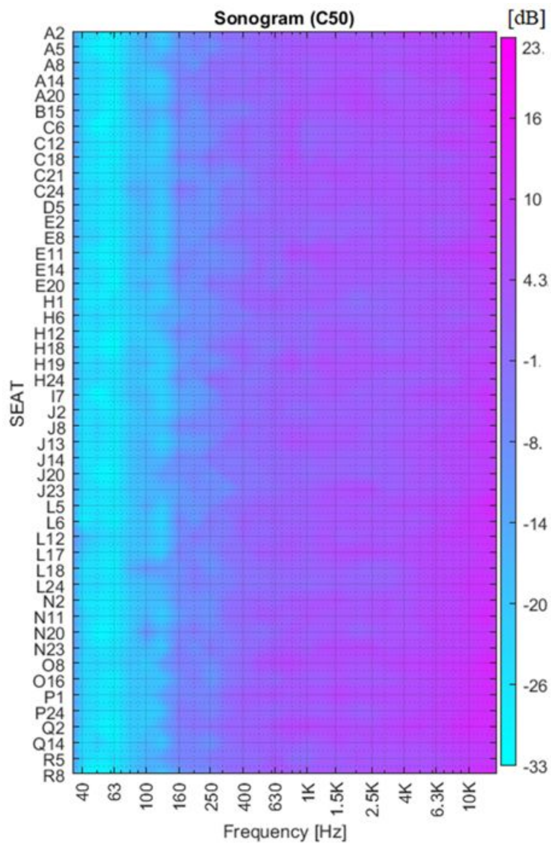

Figure 28: C_{50} third-octave display for each seat.

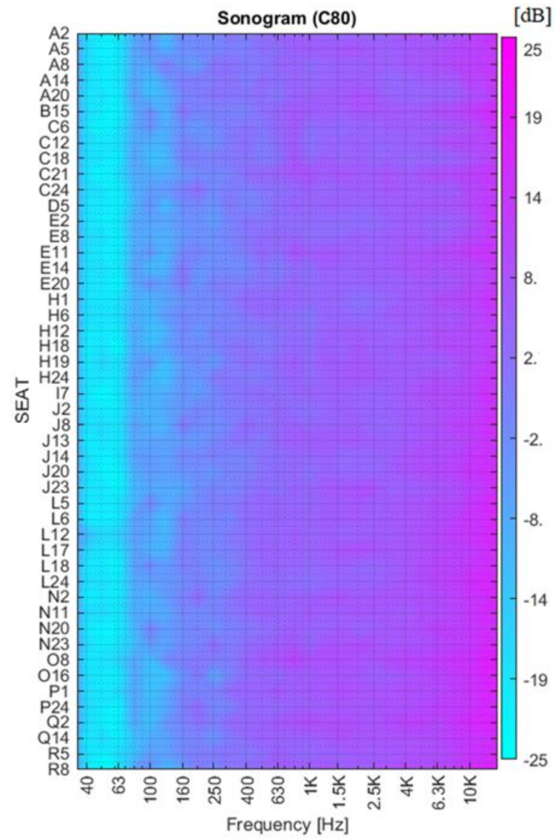

Figure 29: C_{80} third-octave display for each seat.

For Figures 26 and 27, a strange fall of the parameter is presented in the dome zone. It might indicate that the energy after 50 milliseconds for C_{50} and 80 milliseconds for C_{80} is higher than before this time. This specially affects the C_{50}, which varies between -5.75 and -1.03 dB; C_{80} has a range that is almost positive, from -0.93 to 3.98 dB. At the rear seats, the highest values of this parameter are presented. This is not as expected, because these are the areas with preponderant reverberation, rather than direct sound. This could be explained by the calculation of these parameter that sets a threshold for 'direct' and 'reverberant' fields, and some 'direct' sound could be taken as part of the 'reverberant' because of the highest distance to the source at those seats. It can be also explained as a zone with presence of a focal spot of direct sound with reverberation of long duration but little energy. Figures 28 and 29 show the distribution of the parameters in third octave band.

There seems to be an important roll-off frequency both for C_{50} an C_{80} third octave sonograms. This occurs around 63 Hz in C_{80} and 100 Hz for C_{50}; both present the same progression.

Highest levels are presented in high frequency, where RT drops, so the lowest part of the fraction to calculate Clarity is smaller.

The table presented in section 2.4 can be used to establish some comparison, by taking in account that values are part of an average of clarity in 500, 1000 and 2000 Hz. This parameter is defined as C80 MID, which value for Teatro Roma was calculated to be 4.92 dB. Teatro Roma $C80_{MID}$ value is comparably higher than every unoccupied concert hall in the list, being closer to the ones presented in St. Davis Hall and Hamarikyu Asahi.

This information shows that Teatro Roma presents high clarity values overall, and indicates that the theatre could have a good performance for voice applications.

5.1.5 Definition

The parameter D_{50} describes how the speech intelligibility is in every position inside the concert hall. As it stands for the Roma Theater, the results are shown in Figure 30.

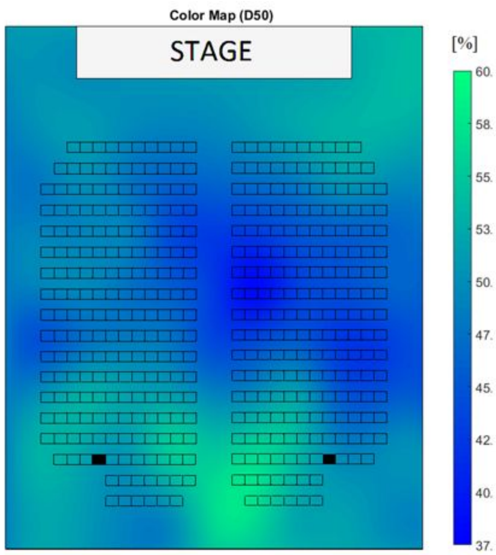

Figure 30: D_{50} for Roma Theater's audience area.

In the Figure above, it can be seen that the hall do not complies with the criterion of D_{50}>50% for all the seating positions, especially in the center area where the values are around 45%. This particularity could be associated to the strong influence of late reflections provided by the dome in the ceiling, and the walls of the balconies that affect more the central area rather than the rear seats or the area below the balconies, where the D_{50} is higher, reaching almost values of 60%.

Regarding the frequency analysis, results in third of octave bands are exposed in Figure 31.

A clear change is presented around 300 Hz, where D_{50} rates turns over 60%. This abrupt modification can be related to some diffusion provided by the chandelier hanging from the roof, particularly for mid-high frequencies. Another cause for this phenomenon could be that the absorption provided by the seats and carpet do not diminish the energy in low frequencies so the reverberant energy is mainly situated in that range, and affect directly the speech intelligibility, and consequently the D_{50} percentage.

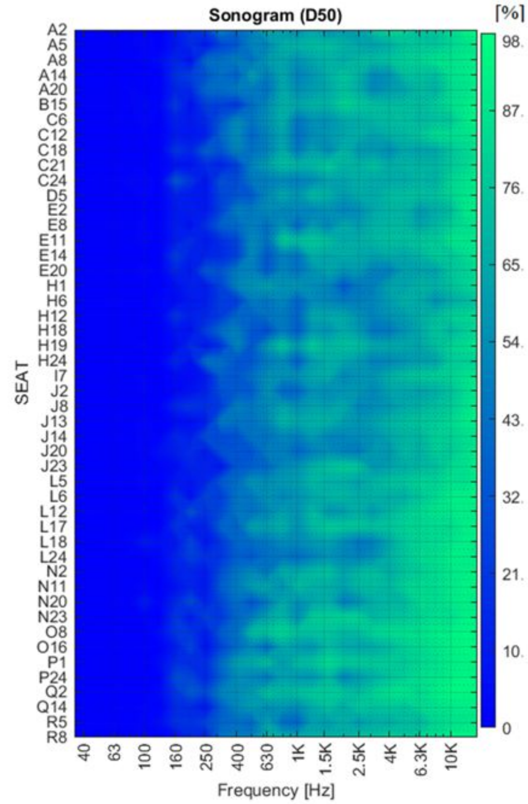

Figure 31: D_{50} distribution for Roma Theatre

5.1.6 Speech Intelligibility Index

In order to analyze the speech transmission index in every seating position a map was made show the corresponding values. Results are shown in Figure 32.

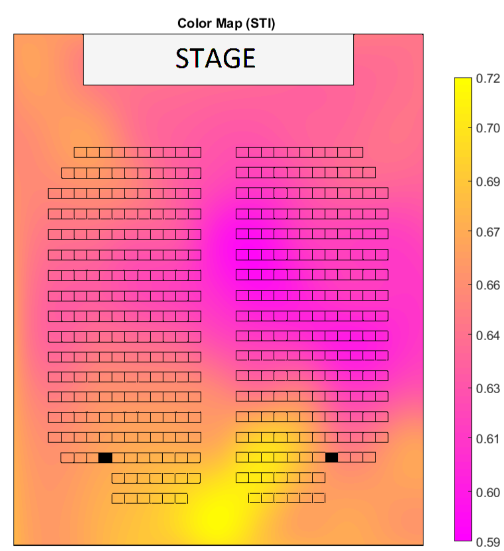

Figure 32: Roma Theatre STI for the seating area.

As the Figure 32 shows, all the STI values in the auditorium classify as “good” for the transmission of speech, according to the EN 60268-16 standard, as they lay between 0.6 and 0.75.

The lower results appears in the center area, meaning that there, whether the background noise is too high or that there is an important amount of reverberation energy reaching those seats, probably coming from the ceiling or the balconies walls. The probable existence of these reflections also appeared when the C_{50}, C_{80} and D_{50} were analyze, which are parameters associated to sound clarity and definition.

5.1.7 Articulation Loss of Consonants

As well as the STI, the %ALcons also presents good results of intelligibility, staying under 7% for all the theater’s audience area. These results are exposed in Figure 33.

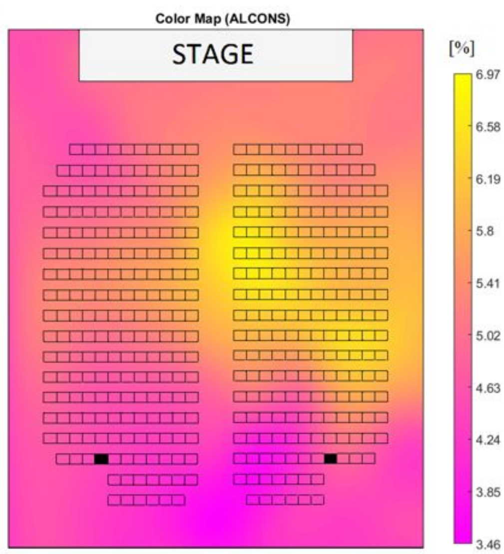

Figure 33: %ALCONS in the audience area of Roma Theater.

Contrary to the STI, Alcons shows lower percentages for good speech intelligibility as it was expressed in section 2.7. Moreover, there is consistency between the results of %Alcons and STI, both showing the worst intelligibility at the right center area. The cause of these loss of intelligibility could be the same that same that was analyzed for STI in section 5.1.6.

5.1.8 Echo Criterion

Figures 34 and 35 show the Echo speech and Echo Music values obtained for the concert hall. Both figures show similar behaviour and range in the whole audience area. Comparing with the recommendations in section 2.8, the values of the measurement are below the recommendation, comparing 0.54-0.78 in echo speech to 1 and 0.52-0.75 in echo music to 1.8. The pattern is similar to the one shown for RT distribution, clarity and DRR, so the acoustic phenomenon producing this is also reflected in this parameter. It is also consistent with STI, ALCONS and C_{50}/C_{80}; the problematic seats are almost the same.

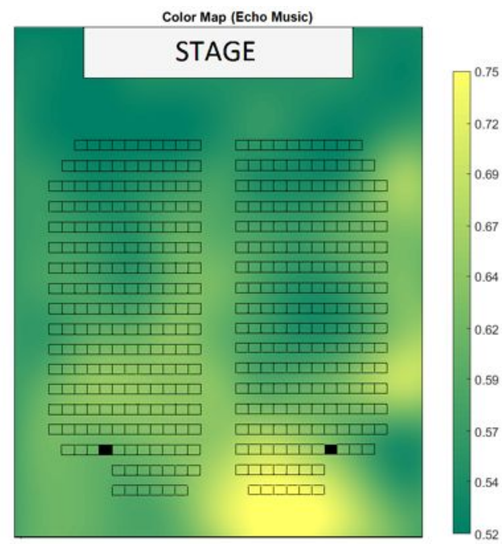

Figure 34: Echo music display for audience area.

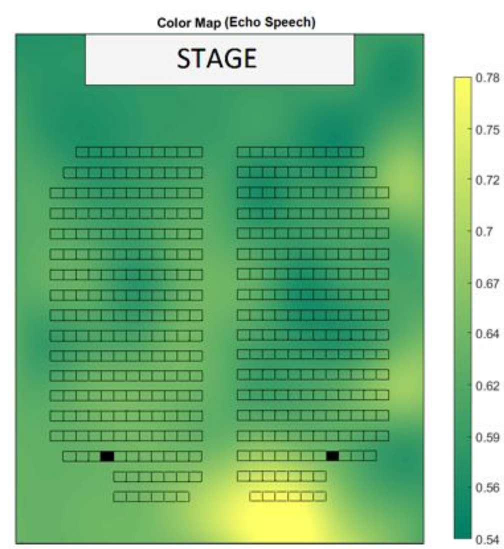

Figure 35: Echo speech display for audience area.

5.1.9 Strength

According to Roma Theatre's blueprints provided, the maximum length of the concert hall is 20.27 m, therefore, the distance from the last seat to the sound source is less than 20 m. Nevertheless, using 20 m as the radius in equation 11, G is calculated to be 2.16 dB. According to Beranek, strength values around 2 dB are optimal for small opera houses; which is satisfied in the case of Roma Theatre.

Nevertheless, when G was calculated using the Aurora plugin some problems occurred. The software did not ask for any any reference signal, which could be the cause to the misleading results obtained.

Besides this, when the measurement in the theater was performed the sound source was not set as the procedure for the calculation of G parameter indicates. The test signal used was not reproduced with the same sound source and sound power that the one that was used in the audio's recording inside the anechoic chamber, as it should have been done, like the procedure explained in section 2.9 indicates.

The controversial results are exposed in Figures 36 and 37.

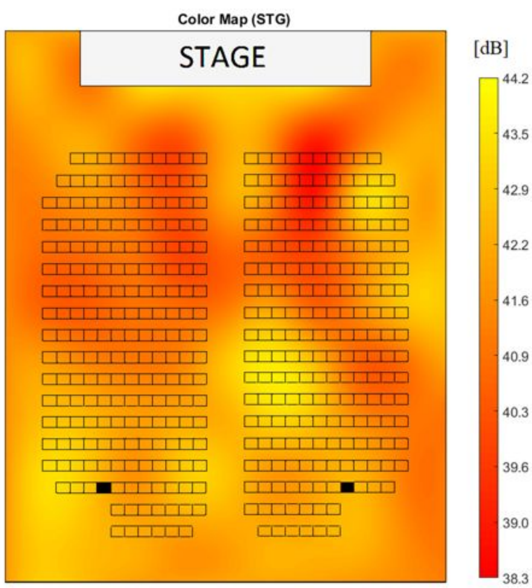

Figure 36: Strength colormap in Roma Theatre.

As it can be seen in Figure 36, there are higher values of strength in the rear seats than in the front area closer to the stage. Consequently, these results should be discarded as well as the results obtained for the third-octave band analysis that are shown in Figure 37.

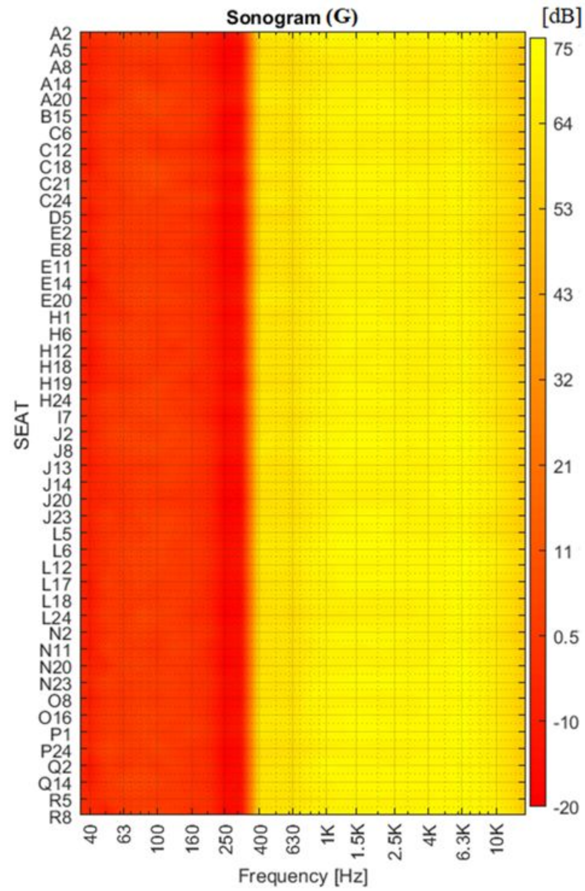

Figure 37: Strength in third-octave bands.

5.1.10 SPL Distribution

Figure 38-a shows the colormap of sound pressure level distribution in Roma Theatre.

The values are obtained by analysing pink noise signals for every seat position. For this measurement, it is very important that all microphones are correctly calibrated.

While this procedure was successfully achieved, resulting audio files waveforms showed a significant amount of amplitude variations; and noteworthy unbalance.

The obtained audio files show variation during the emission, related to mechanical noise and interaction between the technician, calibrator, adapter to ¼" and the microphone.

Despite the fact, for every signal used for calculating SPL values was carefully cropped to avoid irregular waveforms.

Obtained values are then mapped to each seat position accordingly.

Map from Figure 38-a shows an irregular distribution with high levels at the front and lower at the rear, as expected.

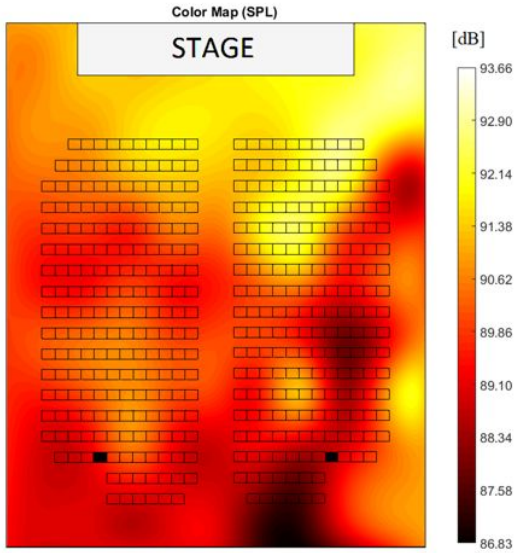

Figure 38-a: SPL Distribution in Roma Theatre.

By using the audio between each take, a background noise file can be generated for every seat position. The files recovered are 15 seconds long, enough to obtain a reliable measurement according to most international standards. This information also can be processed with the matlab script, and values for background noise can be achieved. Global noise values were 45 dBZ at the time of the measurement. On the other hand, an in-situ measurement using a sound meter was recorded to measure 25 dBA. By computing the difference between signal and noise, a new parameter can be derived. The S/N reflects available headroom for each seat position, helping validate all measurements, shown in Figure 38-b.

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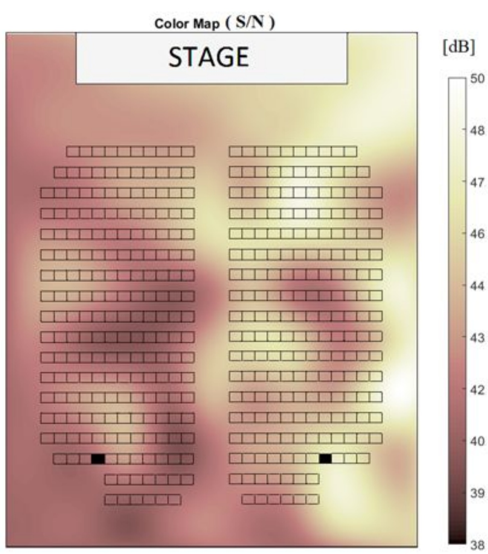

Figure 38-b: SPL Distribution in Roma Theatre.

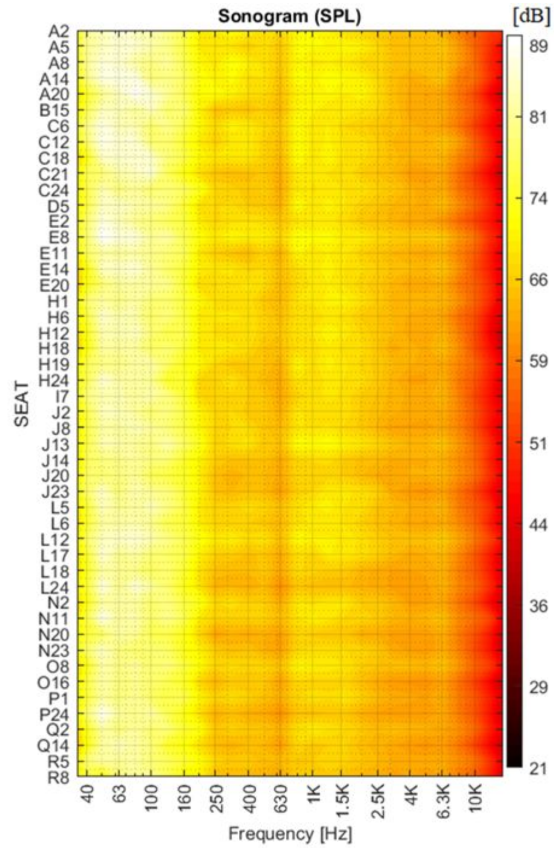

Figure 39 shows the SPL over third octave band frequencies.

Figure 39 shows an interesting level drop in 200 Hz, that could be the crossover point between the dodecahedron sub-low and the main ball of 12 loudspeakers. There are also displayed some differences in non-harmonic related frequencies, so this might indicate failures in the source's frequency response. As expected, RT supports the SPL level, especially in mid and low frequencies, leading to lowest values in high frequency.

SPL sonogram helps to validate all measurements for every seat in third-octave bands.

5.1.11 Lateral Fraction

Regarding LF parameter, which relates the energy that comes from the lateral sides with the energy arriving from the front, it can be said that there is plenty of sound pressure coming from lateral reflections in the seats area that are closer to the stage. In the contrary, the rear area has lower LF values, the explanation for this fact could be that mainly the direct sound arrives to the posterior seats, and just few reflections reach the area.

Moreover, the difference between front and rear area regarding the lateral fraction, could be generated by a great sum of reflections provided by the dome, the balconies walls and the stage walls, that add a lot of lateral energy to the center a front part of the auditorium. A colormap of the early lateral fraction is presented in Figure 40 and 41.

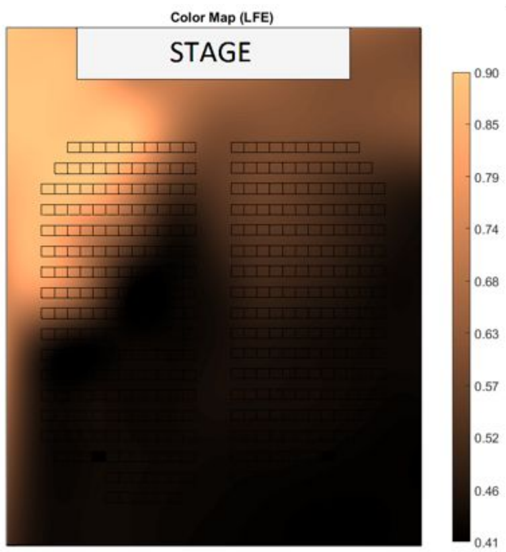

Figure 40: Early Lateral Fraction colormap in Roma Theatre.

In Figure 41 a colormap of late lateral fraction is shown, where it can be seen that the same pattern repeats.

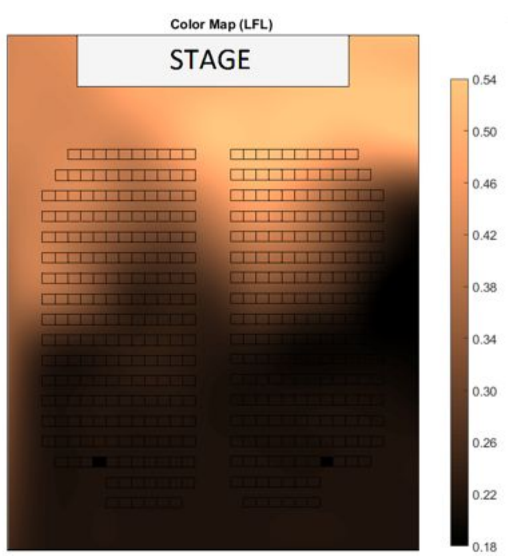

Figure 41: Late Lateral Fraction colormap in Roma Theatre.

There were very few measurement positions using the soundfield microphone, therefore, mapping could not be representative of the Roma Theater's acoustic behaviour. Increasing the number of soundfield measurements are thought to solve this issue.

Figure 42 show the early lateral fraction in third of octave bands, and Figure 43 shows the results for late lateral fraction.

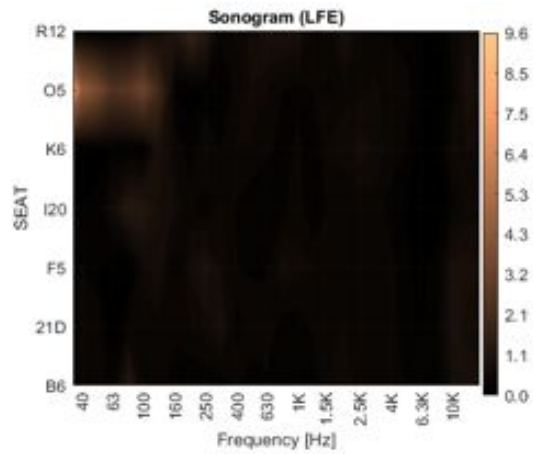

Figure 42: Early Lateral Fraction in third-octave bands.

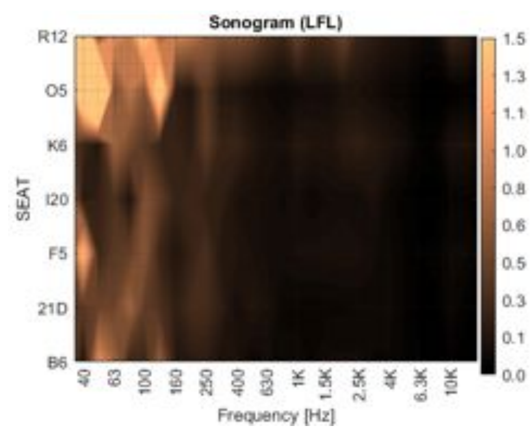

Figure 43: Late Lateral Fraction in third-octave bands.

A slight difference between the results for late LF and early LF, and it can be associated to the low frequency energy arriving after 5ms, making the ratio get higher values in low frequencies. Even though, as it was said before, these results can not be analyzed deeply because of the lack of reliability in the plugin provided by the soundfield microphone manufacturer.

5.1.12 Interaural Cross Correlation Coefficient

Figure 44 shows the seat mapping of Teatro Roma IACC values using impulse response signals.

On the other hand, Figure 45, Figure 46, Figure 47 and Figure 48 show IACC seat mapping using recorded anechoic samples in Roma Theatre main audience area.

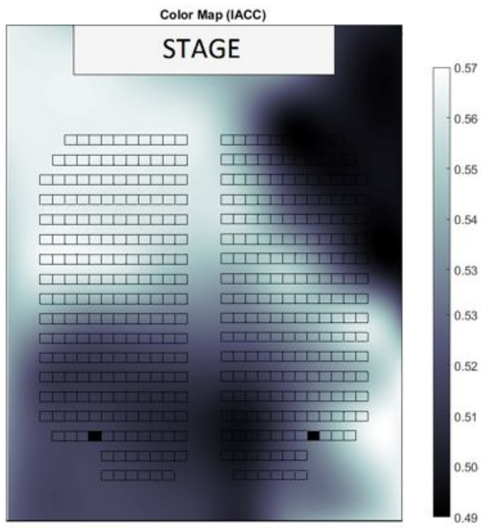

Figure 44: IACC colormap with IR analysis.

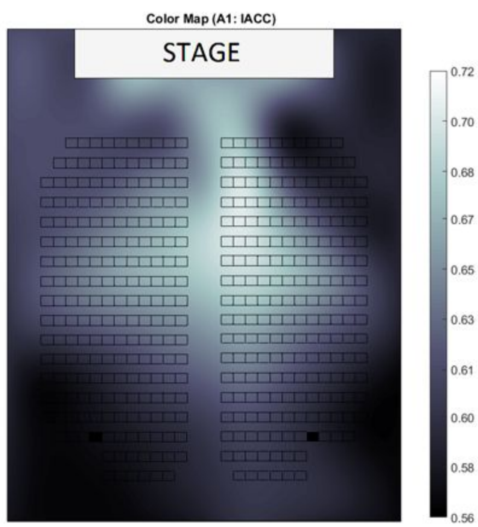

Figure 45: IACC mapped for anechoic audio 1.

In this case, a notable difference between the calculation with IR and audio files is shown. The colormaps are similar in the 4 musical passages, but slight different from the obtained by the IR.

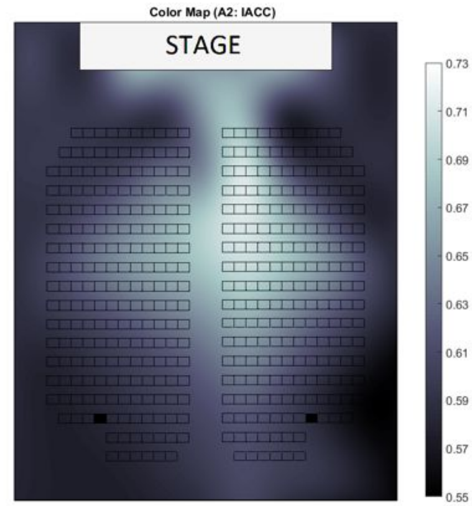

Figure 46: IACC mapped for anechoic audio 2.

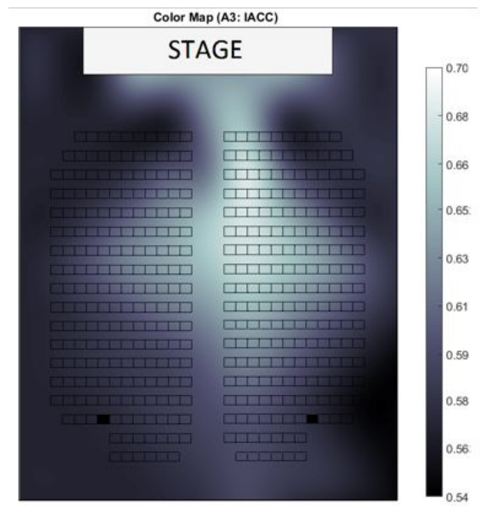

Figure 47: IACC mapped for anechoic audio 3.

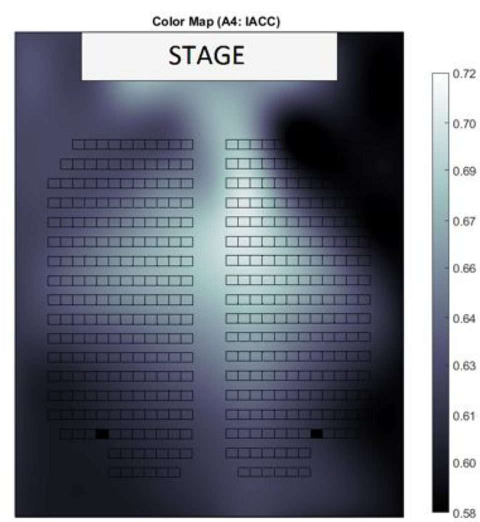

Figure 48: IACC mapped for anechoic audio 4.

This last one, has a narrow range, just between 0.49 and 0.57 meaning a mid-correlation between left and right ear. There is a strange display in the upper left seats, where the IACC value rises, but it seems to be a fake-display error. When analysing the audio files reproduced in the 2-way speaker, the IACC values are higher, up to 0.72, almost always in higher range than the first Figure. This means that the listener should perceive less difference in the information between ears while playing different musical passages, rather than with the test signal.
The shape is as expected, with higher correlation for central seats and lower for the one closer to the walls with high information from lateral reflections. The colormap should be analysed considering the mapping algorithm (and its solutions for the boundaries) and the small number of measurement points with in ear microphones, mainly due to lack of time during the measurement. Figure show the values in third octave bands using IR, and then Figure 50, for the anechoic audio files.

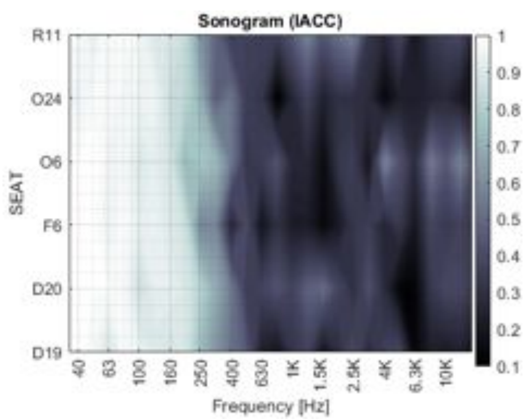

Figure 49: IACC in third octave bands for IR.

The displayed shape for the anechoic audios and IR are similar, with a high value in 2.5 KHz in O6 seat and a drop in correlation around 250 Hz, where it drops drastically. This can also be explained by the physical principle, where low frequency waves tend to be more omnidirectional in it's displacement and higher frequencies tend to be more directive. However, this phenomena is shown independently on the musical program, so the different audios and the IR show a similar behavior in this case, everything has the same drops and highs pattern, with close values. It might seem like an over-interpretation, but the high value in 2.5-3 KHz might be explained considering the ear's natural frequency, close to 3 KHz, where the sensibility is maximum, and could lead to an interaction with the in ear microphones taped over the technician ears.

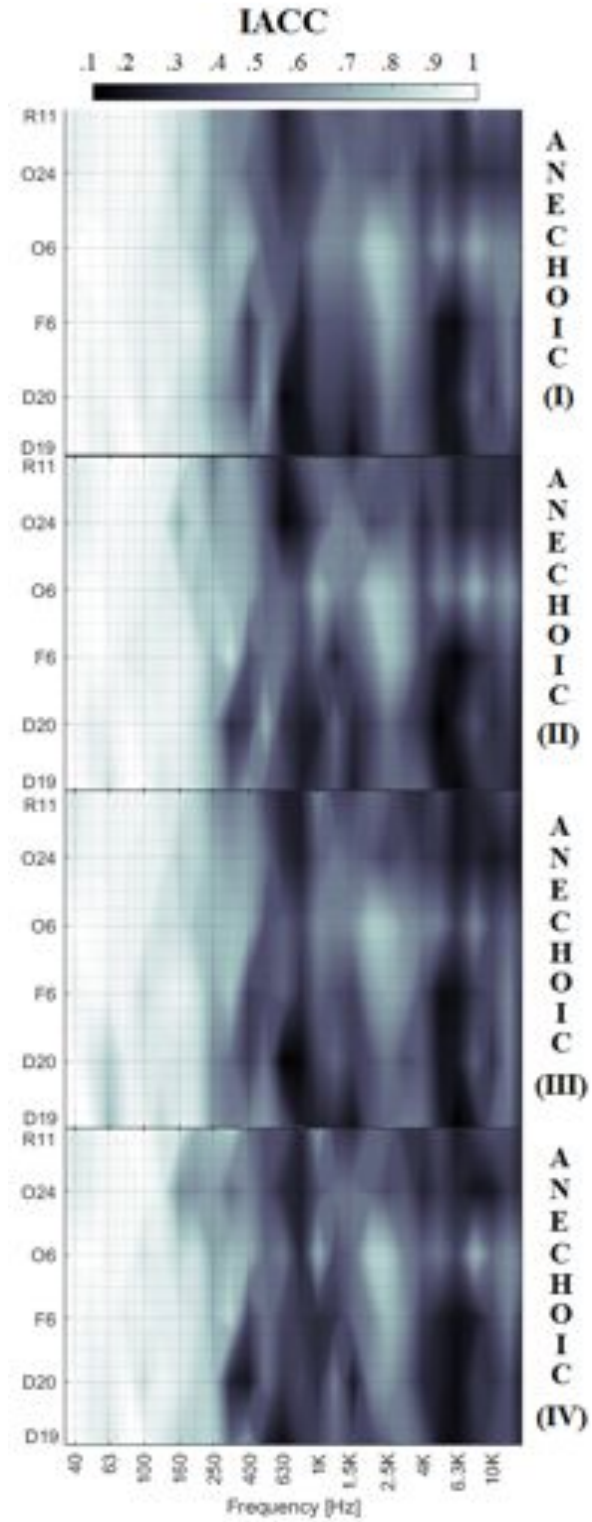

Figure 50: IACC for anechoic audios

5.1.13 Listener Envelopment

Figure 51 shows the colormap obtained for LEV.

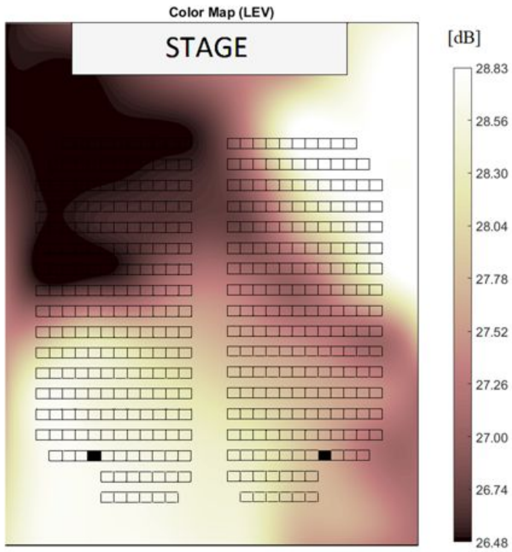

Figure 51: Listener Envelopment in Roma Theatre.

As this parameter is based in strength (G), this parameter cannot be trustly analysed. LEV values are considerably higher than the one presented, with a maximum of 2.42 dB, and in this case 27.76 dB. The displayed range is not consistent with the information provided in section 2 and is trusted to be something related to a bad calculation of the parameter.

5.1.14 τ_e Analysis

Figure 52 shows the analysis of the anechoic samples and its τ_e values.

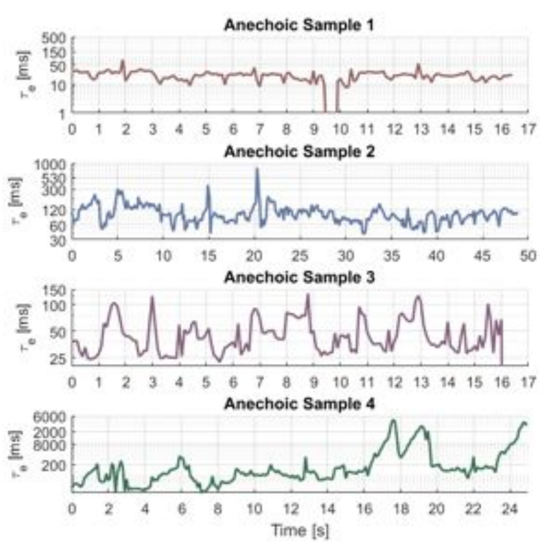

Figure 52: τ_e values for the anechoic samples.

Table 10 shows the $\tau_{e\ min}$ average for each anechoic sample and then Figures 53 to 56 show the distribution of $\tau_{e\ min}$ in the audience area for each sample.

Sample #	$\Delta\tau_{e\ min}$
1	12.5 ms
2	7 ms
3	4.5 ms
4	18.3 ms

Table 10: $\tau_{e\ min}$ values for anechoic samples.

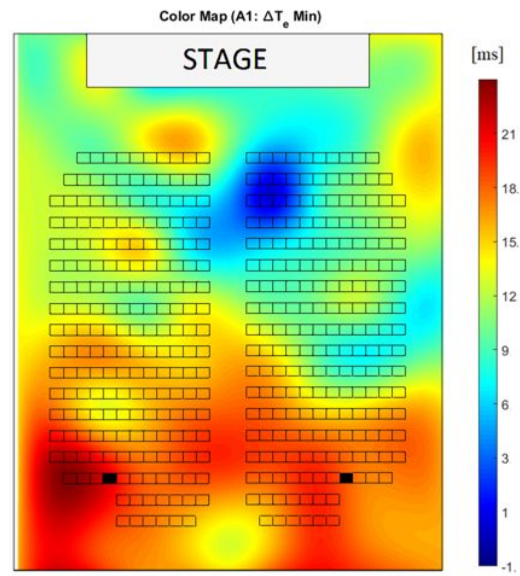

Figure 53: $\tau_{e\ min}$ values for the anechoic sample 1 in Roma Theatre.

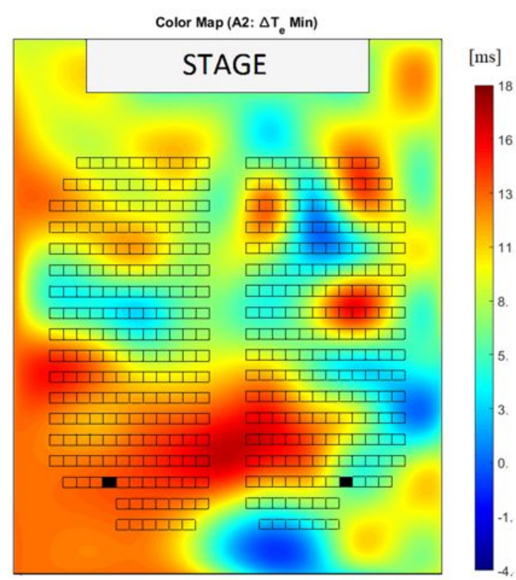

Figure 54: $\tau_{e\ min}$ values for the anechoic sample 2 in Roma Theatre.

As can be seen in every colormap, the rear seats are characterized with the mid to high values, where the RT is proved to be higher and the room affects the perception. As anechoic sample 1 was the most percussive and the RT values are not extremely big, it was expected to have the lowest values of $\boldsymbol{\tau}_e$, but the influence of the room shows a different result, with big differences, up to 20 ms. In anechoic audio 4 there are big differences in a wide range, that might be due to the different parts that form this musical passage with different instruments and dynamics, that could require 'more attention' from the listener's ear, specially while played in this room.

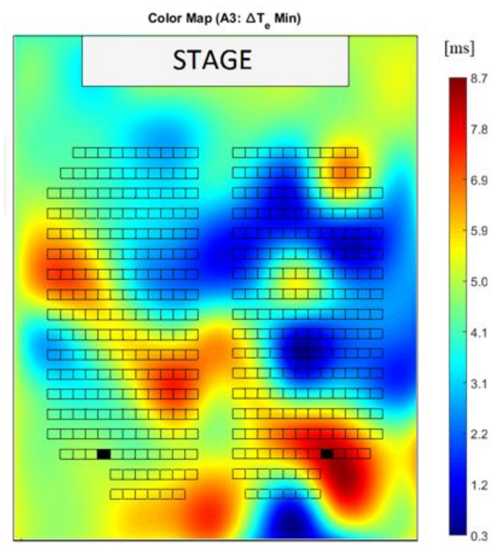

Figure 55: $\boldsymbol{\tau}_{e\ min}$ values for the anechoic sample 3

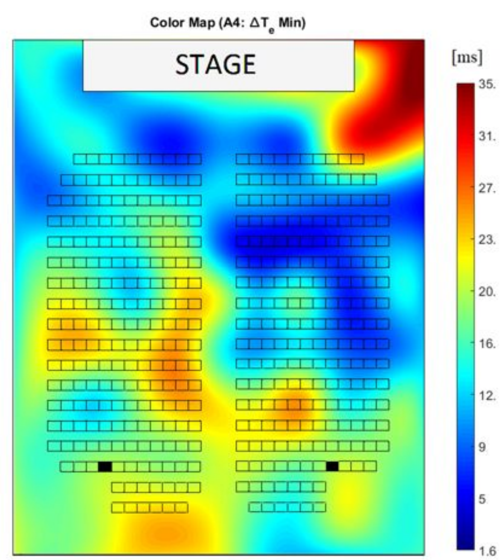

Figure 56: $\boldsymbol{\tau}_{e\ min}$ values for the anechoic sample 4

There could be some relation by the shape of audios 1 and 2, and then apart between 3 and 4, but it does not seem to indicate anything. The maps for audios 1 and 3 show the influence of the dome, and ir can also be noticed for audio 4. As expected, the most affected areas are the ones with higher RT values and lower intelligibility, so the $\boldsymbol{\tau}_e$ displays the same performance as the previous parameters. Table 11 shows the $\boldsymbol{\tau}_{e\ avg}$ for each anechoic sample and then Figure 57 to 60 show the distribution of $\boldsymbol{\tau}_{e\ avg}$ in the audience area.

Sample #	$\Delta\boldsymbol{\tau}_{e\ avg}$
1	33.5 ms
2	40 ms
3	35 ms
4	66.5 ms

Table 11: $\boldsymbol{\tau}_{e\ avg}$ for anechoic samples

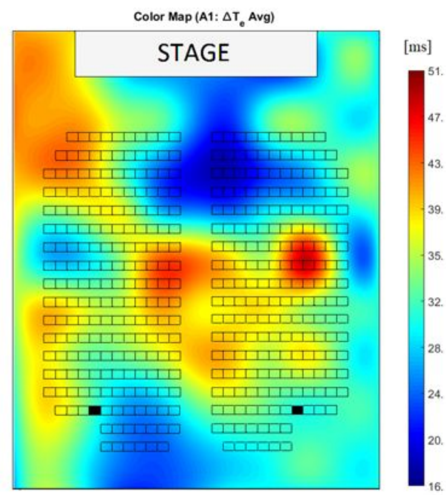

Figure 57: $\boldsymbol{\tau}_{e\ avg}$ values for the anechoic sample 1

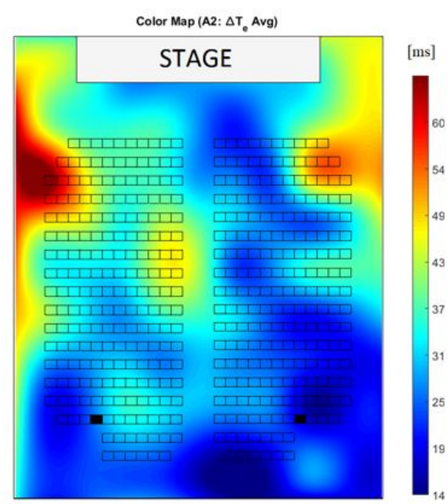

Figure 58: $\boldsymbol{\tau}_{e\ avg}$ values for the anechoic sample 2

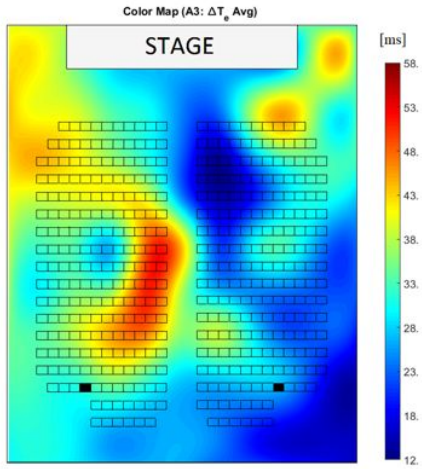

Figure 59: $\tau_{e\ avg}$ values for the anechoic sample 3

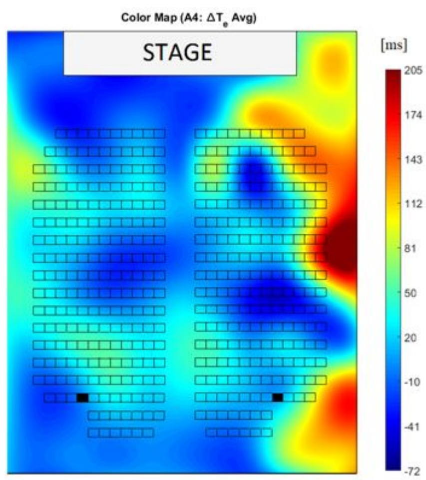

Figure 60: $\tau_{e\ avg}$ values for the anechoic sample 4

The τ_e colormap displayed are quite similar for audio files 2 and 3, with a close mapping range between 10 and 60 ms. For sample 1, the range is wider and sample 4 presents preponderance of negative values, which can be interpreted as a bad calculation in this case. The τ_e variation is, in every case, very little for the main portion of the audience area, resulting in little spots of higher values than are presented sometimes near the walls and other, under the dome. This figures are harder to interpret and lead to less information than the ones for $\tau_{e\ min}$, so little analysis can be performed here.

The less affected is the audio 2, which is the baritone singer. Recalling that the main application (as designed) for this concert hall is opera, this is a good indicator that this type of music will not be affected by the room as much as a symphonic orchestra or a percussive group.

5.1.15 Objective Parameters Data Sheet

By taking an average of every seat for each global parameter, a room value could be achieved, which may be useful to easily characterize the theatre. With this information, a datasheet can be created, shown in Table 12, which reflects Teatro Roma general behaviour and could be used to compare parameters with other theatres. However, it is worth noting that, in this case, global values were mostly obtained by accordingly averaging third-octave bands from 50 Hz to 10 KHz.

Parameter	μ_{ROOM}	σ_{TYP}
RT_{20} [s]	1.08	0.03
RT30 [s]	1.12	0.02
EDT [s]	1.05	0.11
RT_{MID} [s]	1.02	0.04
BR	92.65	0.07
D / R [dB]	3.11	2.13
C_{50} [dB]	-3.83	1.03
C_{80} [dB]	0.97	0.95
$C_{80\ MID}$ [dB]	4.92	1.32
D_{50} [%]	43.52	4.26
STI	0.64	0.03
ALCONS [%]	5.33	0.77
EC SPEECH	0.62	0.05
EC MUSIC	0.60	0.06
LF EARLY	0.53	0.18
LF LATE	0.29	0.13
G_{AURORA} [dB]	41.66	1.34
$G_{THEORIC}$ [dB]	2.16	-
NOISE [dB_Z]	45.37	2.56
IACC	0.5	0.1
LEV [dB]	27.76	0.92

Table 12: Teatro Roma Data Sheet.

From this data, some analysis can be inferred in order to compare and characterise this concert hall. RT_{MID} is between 0.7 and 1.2 s, which is desired. Considering Figure 2, shown in a previous section, that relates volume and RT for each application, Roma Theatre could be useful for speech or chamber music.

$C_{80\ MID}$ values are favorable for a less reverberant concert hall, which is recommended to be between 1 and 5 dB; in this case, 4.92 dB. This value is really high comparing with the values presented for well-known concert halls. G values are higher than recommended (-2 to -1 dB), in this case according to theoretical calculations 2.16 dB. Also the IACC value could be smaller, as recommended by Ando's work, to be low, in this case is just mid, 0.5. Nevertheless, STI and ALCONS values are *good* as the recommendations show, so this support the speech applications for this hall.

5.2 SUBJECTIVE PARAMETERS

Subjective parameters are obtained by windowing IR signals using integration intervals of 10, 100 and 350 ms as described in section 4.3.3. All parameter mappings are shown in Figures 61 to 73.

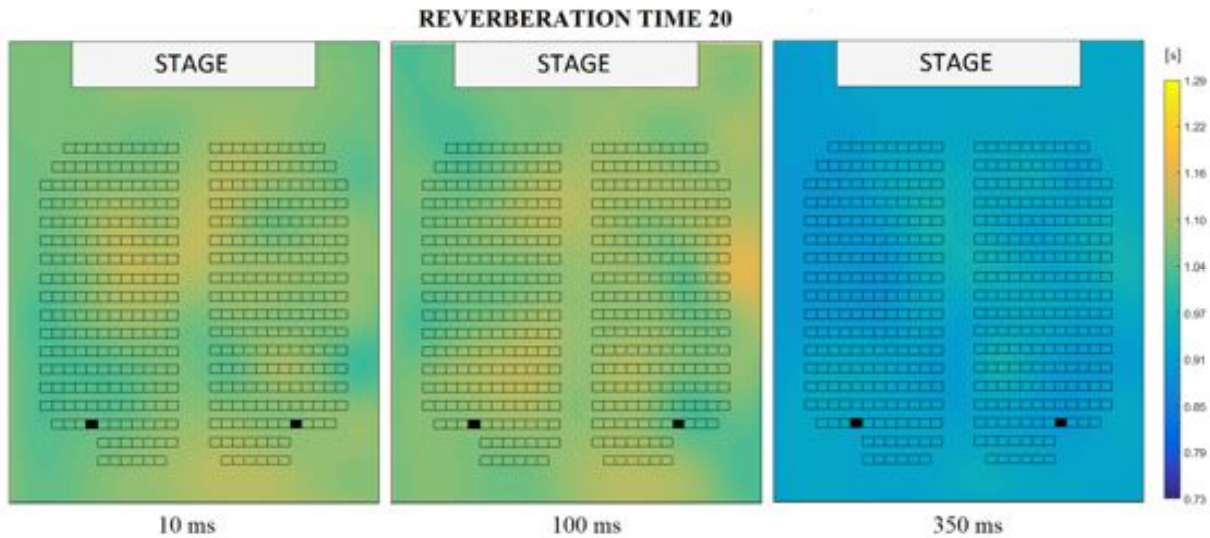

Figure 61: RT_{20} values for each integration window, maped in audience area.

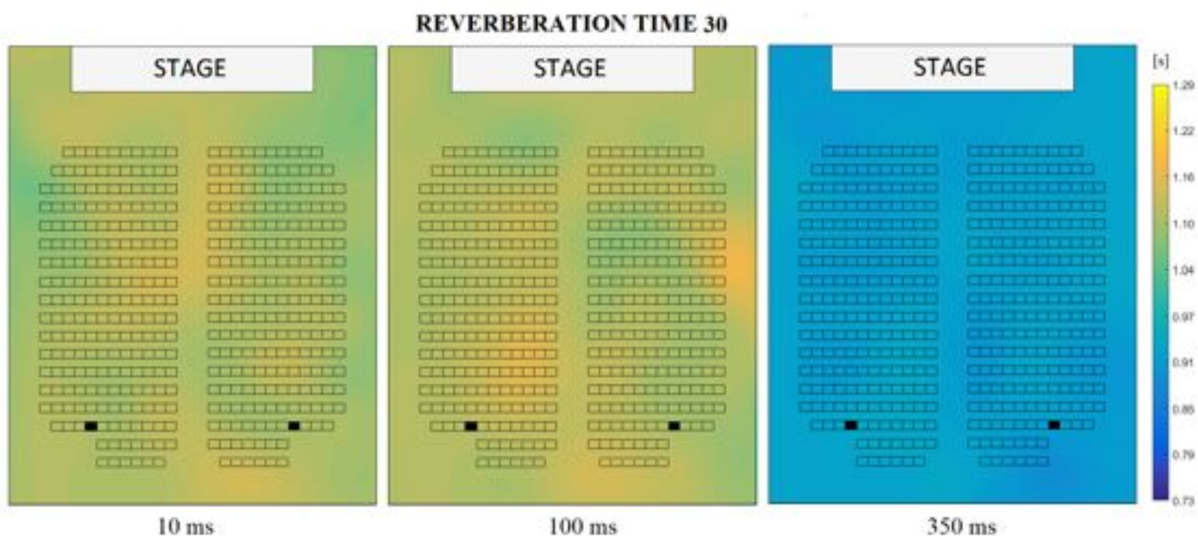

Figure 62: RT_{30} values for each integration window, maped in audience area.

As it can be seen from RIR mappings, differences between 10, 100 and 350 milliseconds integration times are generally not very significant. But this is not always the case. Parameters calculated using 10 milliseconds windows are often very similar to the non-windowed version.

As observed in RT_{20} and RT_{30} graphs, reverberation time is longer for seats at the centre of the concert hall; similar to non-windowed results, with minimal differences between 10 milliseconds and 100 milliseconds windows. However, for 350 milliseconds RT values are drastically lower. This is because IR processing impurities at the tail end forced the cropping of 350 milliseconds signals to 2 seconds. While this did not affect most parameters in a significant way, it certainly was the case for the reverberation time determination, since the chopped tail contained valuable information.

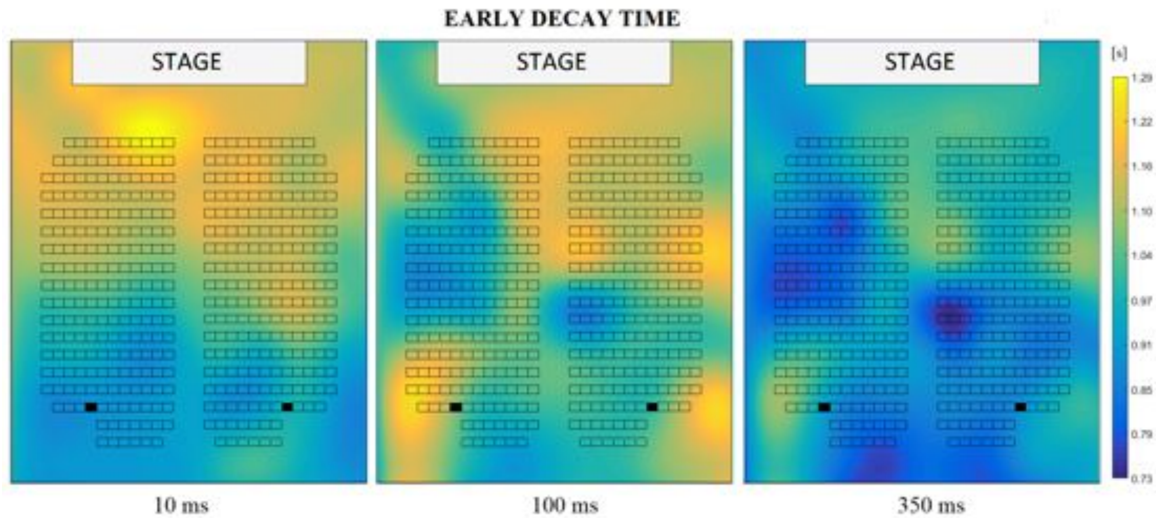

Figure 63: EDT values for each integration window, maped in audience area.

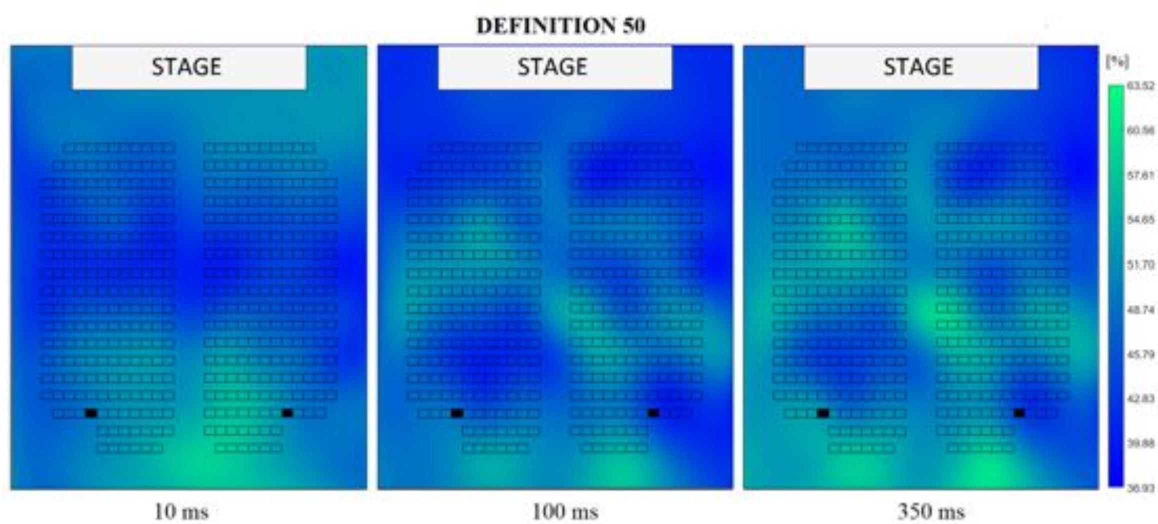

Figure 64: D_{50} values for each integration window, mapped in audience area.

EDT shared a similar fate, as 350 milliseconds results show shorter times; nonetheless, because early decay time measures the time it takes to the signal to decrease the first 10 dB, the missing tail was not a compromise, and results are more reliable than RT_{20} and RT_{30}. In this case, every window tells a different story: 10 milliseconds show a well-defined EDT gradient from front to back, identical to non-windowed mapping. As integration time gets longer, the RC low pass filter effect becomes more noticeable, smoothing transients and turning EDT calculation somehow erratic. This can be observed as 100 milliseconds windowing shows a rather random EDT distribution, accentuating back left and back right seating positions, similar case for 350 milliseconds window, only with lower values.

D_{50} analysis present the same trend. All last parameters have one thing in common: they are all calculated from the ratio of an integration that uses time boundaries such as 2 milliseconds, 50 milliseconds and 80 milliseconds, particularly comparable to windowing integration times. It is expected that 10 milliseconds mappings present more reliable results whereas 350 milliseconds, the least.

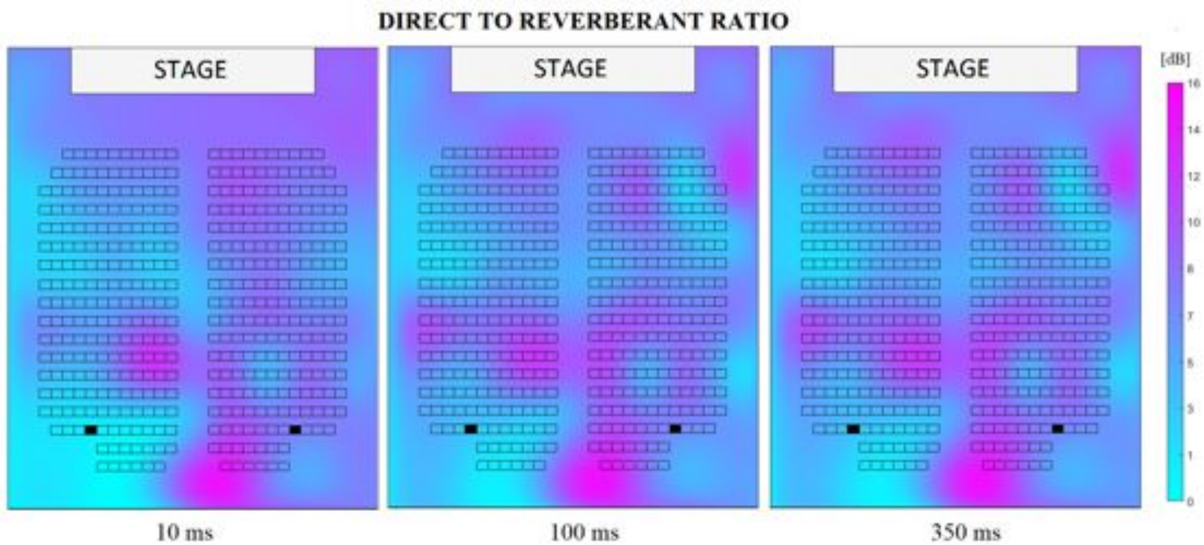

Figure 65: D/R values for each integration window, maped in audience area.

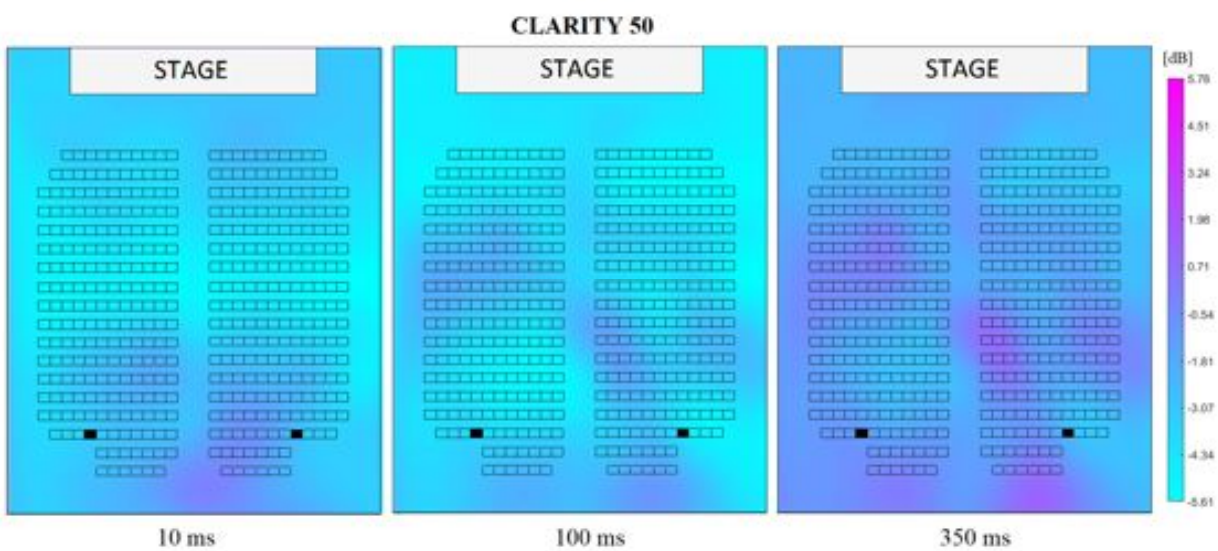

Figure 66: C_{50} values for each integration window, maped in audience area.

Direct to Reverberant Ratio present unexpected colour map results: despite all windows display almost identical seat value distribution, they all notably differ from the non-windowed results, virtually showing an opposite behaviour, as values at the back get larger rather than lower. Therefore, D/R RIR mapping shapes now appear to have an alike response to non-windowed C_{50} and C_{80} parameters. For their windowed version, a similar trend to RT takes place, where the 10 milliseconds integration time perpetrates indistinguishable information, as if non-windowed. When testing with a 350 milliseconds period boundary, clarity appears to be generally improved.

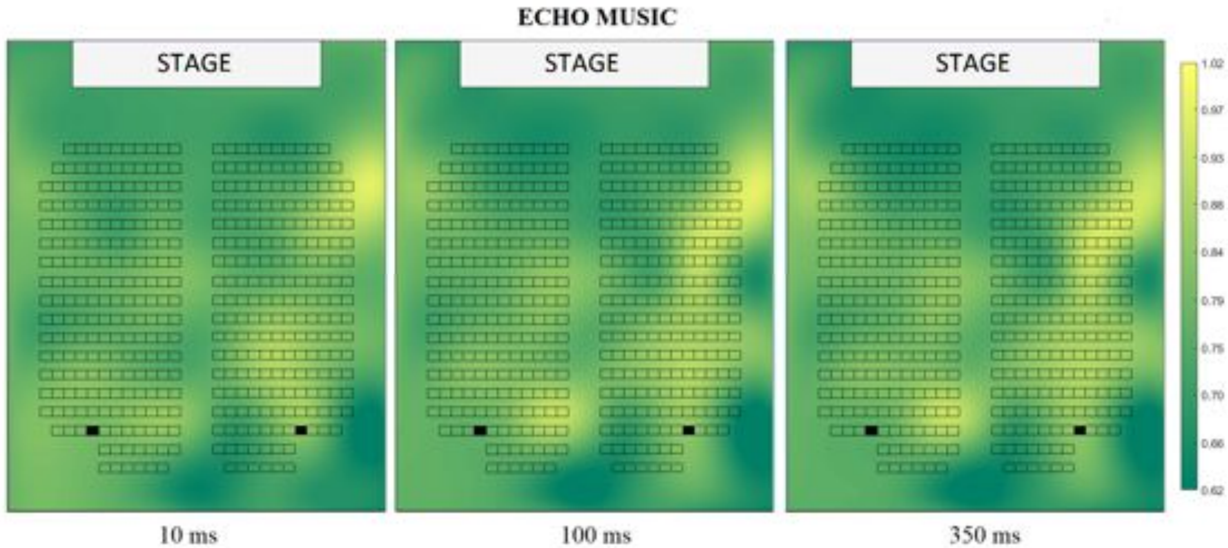

Figure 67: Echo music values for each integration window, mapped in audience area.

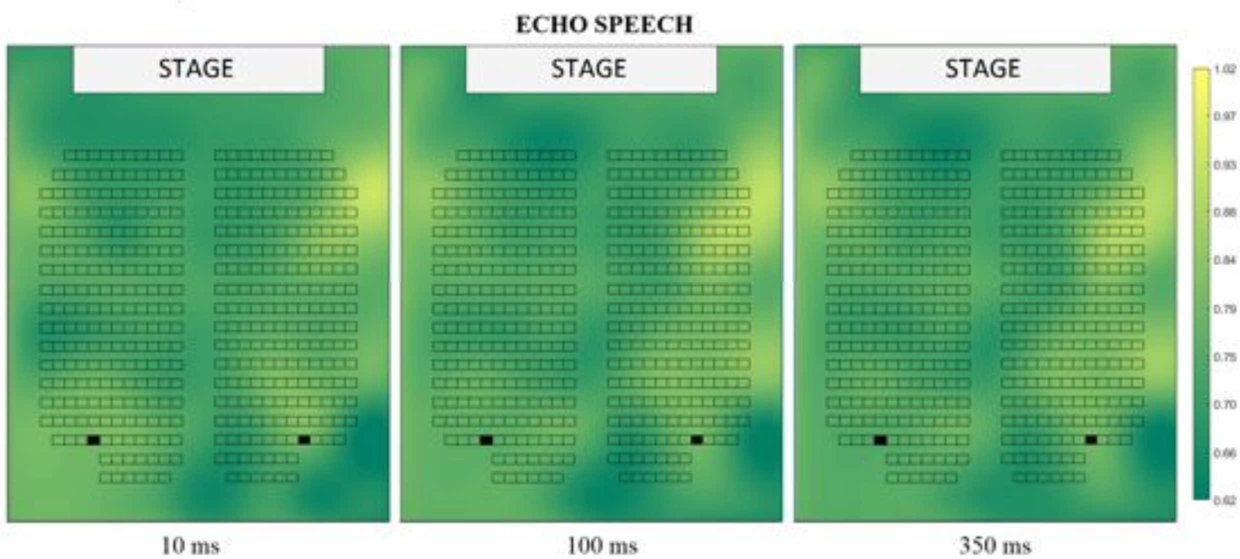

Figure 68: Echo speech values for each integration window, maped in audience area.

Because Echo Music and Echo Speech RIR mapping show very close values between each other, it is suggested that windowing does not influence those parameters. Nevertheless, they all differ slightly to the non-windowed IR mapping: both parameters become lower for the back seats.

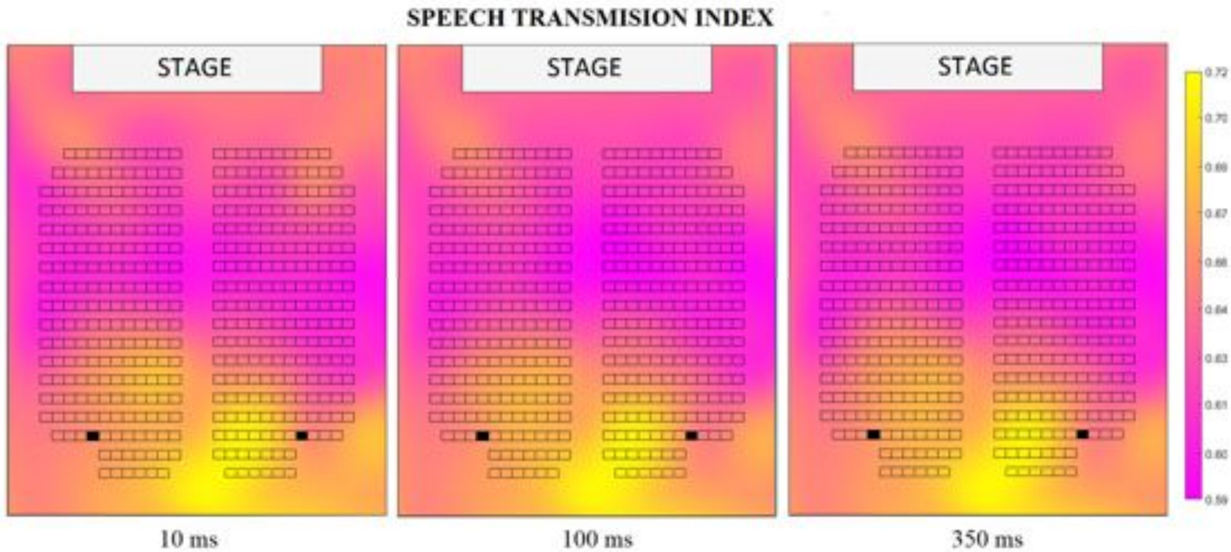

Figure 69: STI values for each integration window, mapped in audience area.

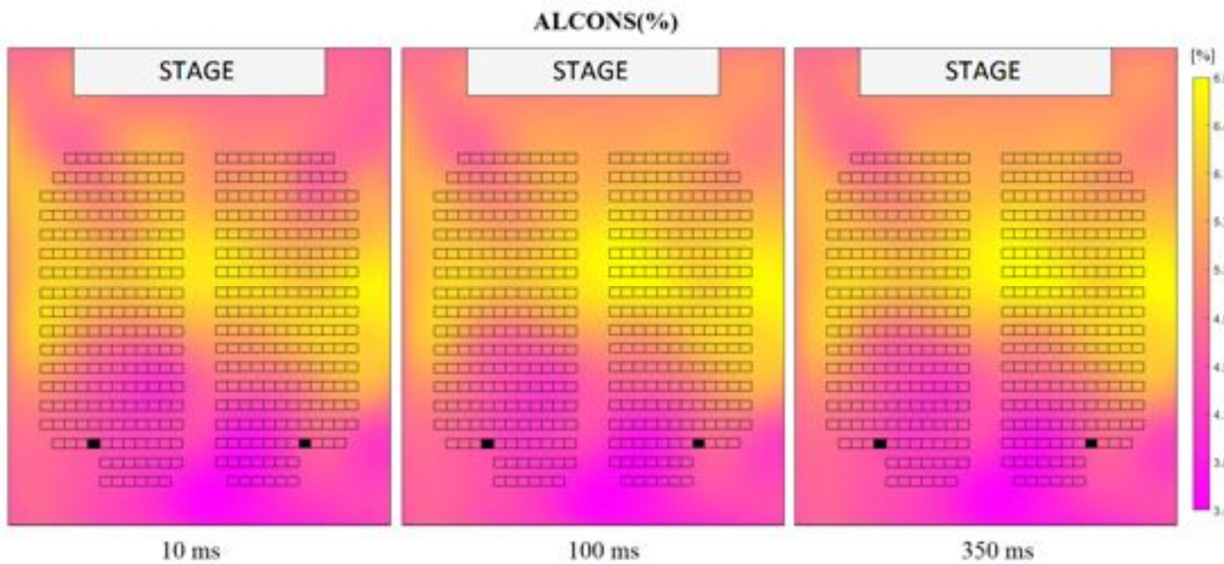

Figure 70: %ALCONS values for each integration window, mapped in audience area.

Since STI and ALCONS are not particularly classified as temporal parameters, and they are computed by applying complex intelligibility algorithms, the tail information becomes not quite relevant. Despite that fact, both STI and ALCONS present undistinguishable results for every window tested, suggesting that neither time integration influences windowed results.

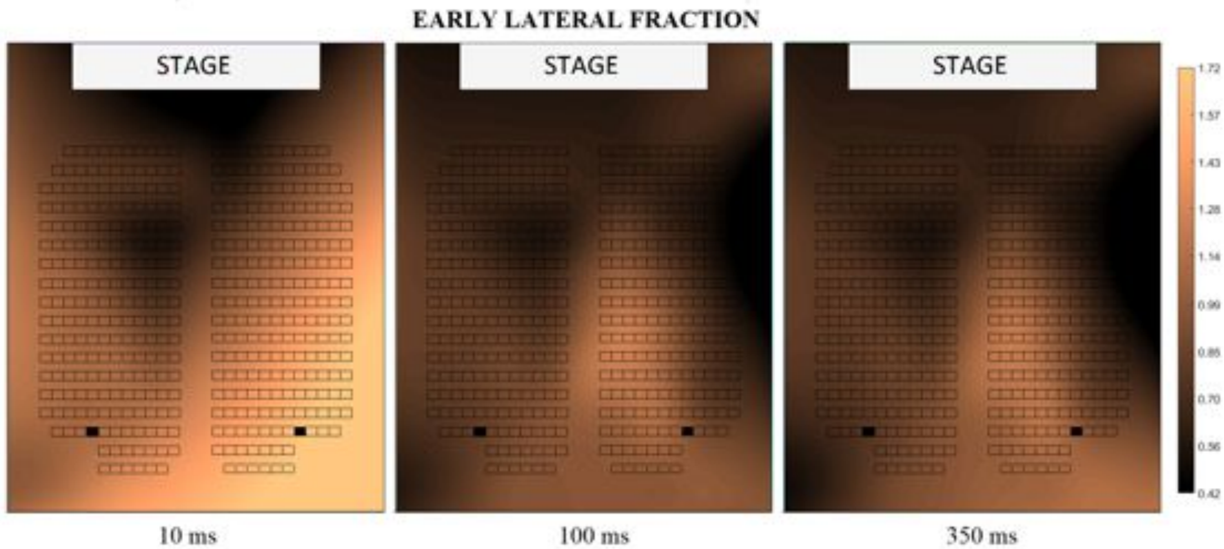

Figure 71: Early Lateral Fraction values for each integration window, mapped in audience area.

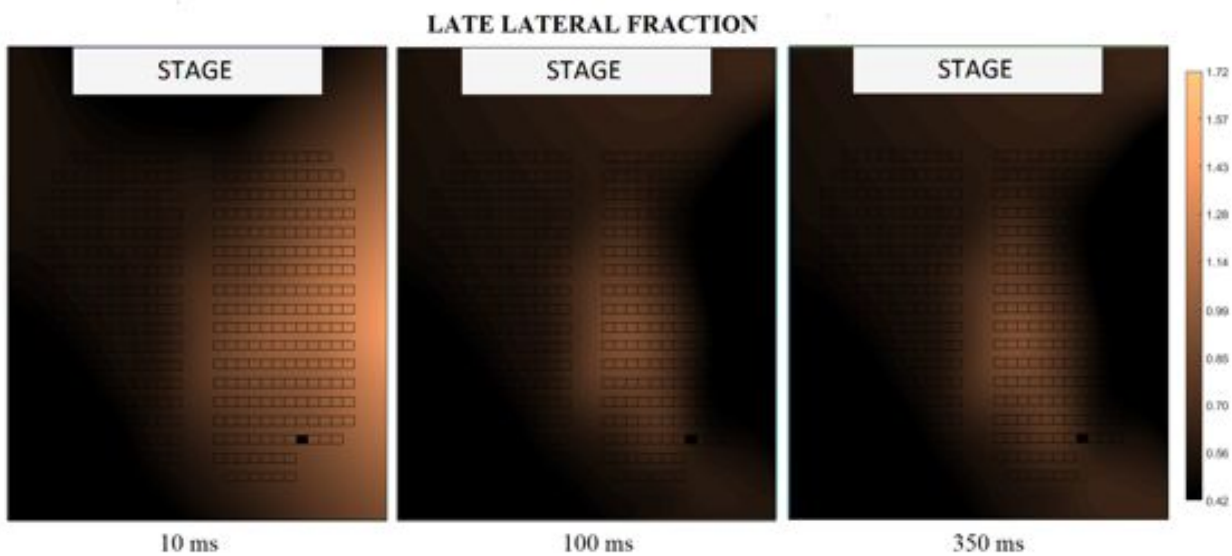

Figure 72: Late Lateral Fraction values for each integration window, mapped in audience area.

While Lateral Fraction parameter describes spatial information, its calculation is based on a temporal computation. It uses 80 milliseconds integration time, which is also comparable to the windowing integration time. For that reason, windows of 100 millisecond and 350 milliseconds show almost identical graphs, while 10 milliseconds is slightly distinct. In any case, the most remarkable difference is that while IR LF values originally showed higher values next to the stage, and lower towards the back of the concert hall, now is the other way around: RIR LF show higher values for the back. However, since LF measurements were recorded using only 7 sound-field microphone positions, mapping is believed to be significantly distorted. More measurement points are required for a good spatial representation of lateral fraction.

In order to simplify and provide another comparison between RIR parameters, Table 13 compiles seat averaged global values that characterizes the concert hall main audience area for 10 ms, 100 ms and 350 ms windows.

Parameter	10 ms	100 ms	350 ms
RT_{20} [s]	1.08	1.08	0.94
RT30 [s]	1.11	1.12	0.93
EDT [s]	1.06	1.06	0.92
D / R [dB]	6.87	6.85	6.93
C_{50} [dB]	-3.56	-3.74	-1.58
C_{80} [dB]	1.05	0.91	2.89
D_{50} [%]	44.62	43.76	45.48
BR	1.44	1.44	1.36
STI	0.64	0.64	0.64
ALCONS [%]	5.36	5.46	5.47
EC SPEECH	0.75	0.76	0.76
EC MUSIC	0.78	0.79	0.79
LF_E	1.09	0.84	0.83
LF_L	0.73	0.6	0.61

Table 13: Subjective Parameters for Roma Theatre.

As can be seen in Table 13, by in large, most parameters were not drastically affected by windowing process, but rather by necessary audio file cropping due to generated impurities added by the script used. This can be observed when comparing differences between 10 and 100 ms windows which are very little, as opposed to differences between 100 ms and 350 ms, which are significantly larger for most parameters. This is because 350 ms files were the most affected by tail cropping (final RIR files had 2 s max length).

6. CONCLUSIONS

A detailed measurement has been carried out with successful results. Lots of parameters were obtained and their distribution could be displayed in relation to the measured positions. Some errors were found in calculation due to human factors and programming scripts. This could be saved rewriting some scripts and reprocessing the audio files.

With these results, the measurement could be carried out better with improved microphone positions and a new analysis, especially in problematic zones. Also this could include the mapping of the first and second levels, that could not be reached in this work.

According to the results, this theatre is very useful for speech applications, for its low RT values and high intelligibility, but is not as good as desired for opera or orchestral applications (as is was originally designed).

Some influence over the acoustics by the shape of the theatre and the dome were found, and some changes can be proposed. The RT values in high frequency are very poor, mainly due to high absorption in the seats upholstery and the lateral and rear curtains. The dome provides some undesired acoustic phenomenon that might be improved with diffusion in this zone. Also, some treatment in low frequency can be carried out in order to have better intelligibility values in these bands. IACC values are not as high as expected, so lateral energy could be increased.

Subjective analysis based on integration windows did not ended in powerfull results, non significant changes were found in most of the cases.

As expected, front and central rows could be considered as the *sweet zones* (rather than *spot*), but with some considerations caused by the dome and balconies influence.

7. FURTHER RESEARCH

Firstly, in the attempt of achieving better results to correctly characterize Teatro Roma, more measurements should be done with binaural and soundfield microphones. This will not only improve mapping resolution but also provide useful data of conflictive regions, helping to determine whether results depend on the concert hall behaviour rather than measurement errors.

For a better parameter mapping, measured seats distribution should be redesigned to accommodate for the mapping script used, which consist in recording symmetric positions. Another possibility is to only record one side of the concert hall with high density coverage, and later to mirror the missing side assuming room symmetry.

Soundfield plugin processing derived in incoherent Lateral Fraction results, suggesting midside signals were poorly generated. The use of an omnidirectional microphone in conjunction with a bidirectional microphone should grant a real midside signal for reliable results in a future measurement.

The work done in this paper reflected the need for developing more robust algorithms and digital tools to correctly analyze the recorded data. Some parameters obtained using Aurora software use inappropriate algorithms to achieve results, “strength” (Str) for instance. On the other hand, the matlab code used for windowed IR genesys contained glitches that may have compromised further processing.

After studying the subjective parameter method, new ideas arose for analysing temporal characterization, such as: windowing an anechoic signal of any length using a fixed interval, and later calculating a parameter of interest for every window, which size should be taken into account depending on the frequency band of interest. Hopefully, this would allow to study the subjective perception of a parameter as a signal evolves in time.

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10. QUESTIONS

1. *¿Qué problemas acústicos presenta la sala medida?*

Los principales problemas encontrados son:

- Bajo RT en alta frecuencia para la aplicación declarada por el diseño del teatro, en este caso ópera.
- Poca inteligibilidad en el fondo y los laterales, incluyendo ciertas zonas particulares del frente, posiblemente por influencia del domo central.
- Preponderancia de la baja frecuencia en valores de RT, que se identifican con pérdida de inteligibilidad en la misma región.
- Variadas afecciones encontradas por influencia del domo central y algún posible fenómeno acústico involucrado, como la concentración focal de energía.

2. *¿Cómo mejoraría la acústica de la sala medida? (liste 3 actividades a realizar en orden de prioridad, de mayor efecto a menor efecto).*

- Bajar la absorción en alta frecuencia: posiblemente con telones más livianos y menor relleno en las butacas.
- Difusión para la cúpula, probablemente los efectos adversos encontrados tienen que ver con concentración de energía por geometría y con reflexiones especulares.
- Reflectores laterales para mayor auralidad y baja de la correlación inter-aural.

3. *¿Cómo relacionaría la inteligibilidad de la fuente con la variación del τ_e que produce la sala en cada posición?*

En las zonas que presentan mayor variación del valor de τ_e, la inteligibilidad se ve comprometida, ya que es evidente en esta sala que el valor de RT afecta al desplazamiento del mismo. Eso se corresponde con lo analizado en valores de STI y ALCONS, con lo cual, una posible baja de RT en frecuencias más bajas también sea beneficioso. Es importante también rescatar, como fue visto en el análisis, que donde mejores resultados se obtuvieron, fueron los casos con la señal del barítono, lo que apoya la idea de que la mejor aplicación, en la situación actual del teatro es para palabra hablada, posiblemente teatro.