Acoustic characterization of rooms

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Abstract

Acoustic characterization of rooms deals with the evaluation of the acoustic quality within a certain space. This is made by a set of acoustic characteristics divided through their objective and subjective nature. The former correspond to measurable physical parameters which strongly relate to the room´s architectural characteristics, while the later correspond to acoustic attributes subjectively identified by a listener. The objective characteristics are correlated to the subjective ones, and nowadays there exists acoustic quality criteria well defined for different types of spaces, in particular performance halls and auditorium. The physical methodologies used in room acoustic analysis are based in measurements as well as in modeling techniques. The approximated impulsive response of a room can be obtained via these methods. This response is ultimately used to retrieve the most relevant acoustic parameters in order to characterize conveniently the acoustical quality of the room. This work intends to study the acoustical characterization parameters as well as methods for their computation. From the analysis of these parameters one was able to:

- Taking into account a subjective analysis of the acoustical quality of a space, it is possible to identify the main causes for the "poor" acoustical quality of that space, based on the values assumed by the objective acoustical characteristics and established quality criteria.
- With the application of acoustical conditioning measures, it is possible to verify, based on the values assumed by the objective acoustical characteristics and established quality criteria, the effective improvement of the acoustical quality of a certain room.

Key-words: room acoustics, acoustic qualities of rooms, acoustical parameters of rooms.

Introduction

The acoustic conditions of a certain room should provide a sound environment suited to the activities developed within that room, particularly in places intended for words and music performances. The "form" or "quality" with which the sounds are perceived by a listener within a room depends, essentially, on four factors:

- Physical characteristics of the sound source;
- Relative position of the sound source and the hearer;
- Architectural characteristics of the space;
- Subjective nature with which humans assess the sensation caused by a sound stimulus.

The first three factors are possible to quantify objectively, while the latter presents difficulties in being expressed by objective physical parameters. This is due to the inherent subjectivity associated to personal opinions on the acoustical conditions of a room. The personal judgment about the acoustical quality of an area is usually related to the sensation of greater or lesser quality of the perceived sounds, and depends on the training of the listener regarding the nature of sounds.

Various subjective attributes were identified in psychoacoustical studies and have been considered as subjective acoustic characteristics of rooms. These attributes may be correlated to measurable physical quantities, considered as objective indicators of acoustic quality within rooms [2] [3].

The main objective of this work is to study these acoustic parameters and how do they allow characterizing the acoustics of a space by means of the evaluation of the values assumed by these indicators, which, in principle, will agree with the subjective impression about acoustic quality of that space. These parameters are divided on time and energy criteria of, and are associated to the way the sound energy develops on time in a room. The acoustical parameters are calculated on the basis of the room's impulse response, which corresponds to the temporal development of sound pressure detected at a certain reception point, due to the emission of a sound pulse (of delta Dirac type) at another point of the room [11][12][13][14].

This work was conducted by the case study of the acoustical characterization of the multipurpose room Arena D`Évora. The study case presented is based on the hypothesis that the acoustical conditions of the room are not acceptable for music performances and other events where sound transmission within the room takes place. The admitted assumption is evidenced by the *in situ* measurements results and by the computational modeling results, whose analysis was focused on the reverberation and clarity characteristics, since these were the attributes plus referred on the subjective opinions given by the listeners, about the acoustical quality of Arena D`Évora. Measurements are used for calibration and validation of the digital model of the room created for modelling the room's acoustic characteristics. The computational model allows the assessment of the room's acoustic performance and its performance after the application of hypothetical acoustic conditioning.

Subjective attributes and objective measures

Reverberation

The reverberation is the persistence of sound in a room after the source has been disconnected. Our perception to this phenomenon is quantified by the objective parameter reverberation time T_r. A reverberant room is said to be a "live" room and a room with reduced reverberation is call a "dead" or "dry" room. The sensation of "liveness" is associated with the reverberation time at mid-frequencies T_{mid} (average of 500, 1000 and 2000 Hz one octave frequency bands); although in some rooms it may be related to the parameter early decay time EDT [9]. A certain zone of the audience area that presents an EDT significantly inferior to T_r, usually, results in a less "live" zone of the room. The reverberation influence directly the intelligibility of the sound content perceived by a listener, hence the clarity and definition of sounds.

Clarity and definition

Clarity is a subjective parameter to evaluate the degree of separation of successive sounds (horizontal definition according with Beranek [3]) and the ability to distinguish between overlapping tones (vertical definition according with Beranek [3]). The usual physical measurement of clarity is the ratio of the energy in the early sound to that in reverberant sound, designated by C_{80} for music and C_{50} for speech.

T_r e EDT

According to the model of diffuse sound field the reverberation time is estimated, for each frequency band under analyses, considering that the decay of sound energy within a room has an exponential variation over time [1].

The theoretical model for calculating the reverberation time resulting from this approach is equivalent to the empirical model of Sabine [1]. These models are the basic tools in modeling the reverberation characteristics of a room, although imprecise in certain situations (when the sound field departs from the diffuse conditions), they relate the reverberation time parameter with the architectural features of the room, as the volume of space and sound absorption characteristics of the materials constituting the interior surfaces. As the theoretical model of diffuse sound field, the reverberation time measured *in* *situ* is calculated based on the decay curve of the sound energy.

In the standard document ISO 3382-1 [14] are presented two methods for obtaining the decay curve, the interrupted noise method and the integrated impulse response method. The latter is the method used in this study and corresponds to the method developed by Schroeder [10], demonstrated that the spectrum of sound energy associated with the average of infinite decay curves is identical to the energy spectrum associated with the impulse response obtained with a single measurement.

In general, a linear fit is performed over the points of the decay curve obtained and the reverberation time, defined as the time necessary for the sound level to decrease of 60 dB, is given by:

$$T = \frac{60}{m} \qquad (1)$$

where m is the rate of sound energy decay given by the slope of the line in dB/s. According to [14] the linear fit should be performed by using a 30 dB decay (in the range of 5 to 35 dB below the stationary state level). That slope is used in the computation of the reverberation time, T_{30}. The same can be done for a dynamic range of 20 dB (-5 to -25 dB) and 10 dB (-5 to -15 dB) and the reverberation times are T_{20} and T_{10}, respectively.

EDT is a parameter similar to the reverberation time, but being only retrieved via the slope of a line fitted to a 10 dB decay below the maximum sound level [9]. Jordan [9], reported that the EDT seems to be particularly sensitive to changes in the geometry of the room. In rooms where a large part of the first reflections are directed to the audience and a small part directed to the stage, the EDT presents lower values while in rooms where the sound field is more diffuse (reflections over the audience and the stage), the EDT presents higher values and closer to the reverberation time Tr. Jordan found that these last conditions were preferred by musicians. EDT has been considered as a more relevant parameter of the sensation of reverberation and sound clarity.

C_t - C_{80} e C_{50}

This parameter is a measure of the definition with which the sounds are perceived in a room; it's defined by the ratio between the initial sound energy ("early sound"), received between the instants 0 and t, and the received reverberated energy after t. It's expressed as follows:

$$C_t = 10 \log \frac{\int_0^t p^2(t)\, dt}{\int_t^\infty p^2(t)\, dt} \qquad (2)$$

where t is the initial time of the sound arrival, i.e., is the duration of the time interval considered as "early sound". In a space for voice it's defined as t = 50 ms and for music t = 80 ms. The function p(t) correspond to the measure impulse response.

The "early sound" is a fraction of the total sound energy received in a particular receiver's position and, regarding the subjective evaluation of acoustic quality of a room, is an important factor that influences the intelligibility of the perceived sounds. In the case of a speech room the "early sound" must be higher than the reverberated energy and C50 will be positive the higher this value is the better is the intelligibility of the speech. In a room for music negative values for C80 are acceptable. The value of t for music is greater than the value for speech, which means that part of the initial energy, received later (between 50 and 80 ms) is still useful for the definition mixture of musical sounds.

The acoustic quality criteria strongly depend on the type of performance in question. Next, different quality criteria are presented that are suitable for concert halls, regarding to the characteristics of reverberation and clarity:

Reverberation time

Symphonic music: 1.8 – 2.0 s;

Baroque and Classical music: 1.6 – 1.8 s;

Chamber music: 1.3 – 1.7 s;

Opera: 1.3 – 1.5 s;

Rock, pop, traditional and other styles with PA system: 0,6 – 1.2 s;

Clarity

- 4 dB ≤ C_{80} ≤ -1 dB (acceptable for music)

C_{50} ≥ 0 dB (acceptable for speech)

- 5 dB ≤ C_{50} ≤ 0 dB (acceptable for reverberating rooms)

Methodology

Computational simulation

In a design stage it can be used modeling techniques to predict the acoustical characteristics of a room by means of computational simulation, which implements algorithms based on physical and mathematical models.

The digital model of the architecture of the room can be done in autoCAD. The file created by AutoCAD software is imported by simulation software EASE. The input data correspond to the materials acoustic characteristics, sound sources and the areas of audience. The characteristics of sound absorption of the materials constituting the interiors of the room are from a database, which is based on values of sound absorption coefficients (with frequency dependence) for different materials. The physical characteristics of sources, in terms of sound power (dynamic response and in frequency) and direction, are possible to be defined through a database where different loudspeakers types currently manufactured are available. The system of loudspeakers may be simulated for different positions and orientations.

The software allows estimating the ideal impulse response of a room for a particular relative position between the source and receiver, by using the "ray-tracing" method or method of source images [1]. The ideal impulse response is called reflectogram. The parameters of acoustic characterization are computed by EASE. The goal is to know the values entered into by those parameters on the areas of audience of the room. EASE computes the parameters and allows the visualization of the results in a colour map of their spatial distribution.

Measurements

With the knowledge of the impulse response is possible to derive all the fundamental acoustic characteristics of a room. One intends to obtain the response of the acoustic space only, free of distorting effects, caused by non-linear responses of the equipments, and noise (acoustic, thermal and electrical), to describe more precisely the acoustic characteristics of the room. Ideally, the system must have a linear behaviour and time invariant, in order to increase the precision obtained. In a real measurement system is necessary to consider the effects of distortion due to equipment non linearity's changes and drifts in their behaviour.

In order to collect suitable experimental data of impulse responses, the sound source must produce signals with enough energy for obtaining high signal-noise ratios in the whole frequency range of interest. The produced signals must have the shape of short term pulses without significant pressure variations. Furthermore they should have repeatability over time and space [11].

There are different techniques for measuring the impulse response, which are based on deterministic excitation signals. Currently the most popular techniques are: The excitement by pulses (alarm gun or pulse generated electronically and emitted by an amplification sound system), the sinusoidal static excitement (stepped tonnes, multitone), the excitement by a "sweep" signal in which the frequency varies linearly over time (Time Delay Spectrometry TDS), the excitement by a random impulse sequence (Maximum Length Sequences MLS) and excitement by "sweep" signals in which the frequency has an exponential variation over time. The latter method is the most recently developed technique [13] and presents important advantages if compared with the most used methods, TDS and MLS. These advantages are related to the susceptibility of each method to the presence of noise and to the deviations of the systems from the ideal conditions (time invariant linear system). The exponential Sweep method allows to separate the impulse response from the components of signal distortion due to non-

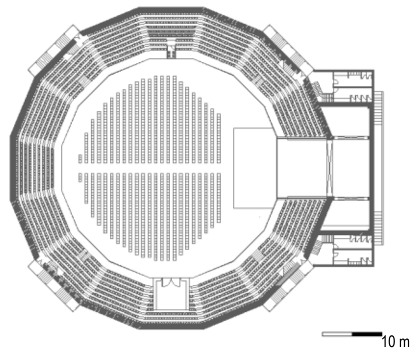
linearities and to improve the signal to noise ratio, in particular at low frequencies [13].
The use of a digital analyzer allows a fast processing of the received signals. The impulse response is computed directly after the measurement and is analyzed for the determination of the acoustic parameters. This work uses the software EASERA (Electronic and Acoustic System Evaluation Response Analysis) for recording, processing and analysis of measured signals carried out in the Hall used as case study.

Case study "Arena D` Évora"

This section focuses in the case study of the Arena D´Évora, a multipurpose room mainly used for musical performances of different styles. This works was motivated by several negative opinions and complains about the acoustic quality of this room (the opinions were collected both from direct conversation and from several sites and blogs on the internet). Such opinions can be summarized in the following subjective appreciations:

- "the room acoustic performance is not good";
- "the room has too much echo and the sounds mix";
- "the room has too much "reverb" and the sound has no definition";
- "the room is reasonable for orchestrated music, but for rock, pop and electronic music it is very bad".

Almost all the opinions collected can be associated to the first two topics, while the third and fourth are mostly given in opinions by musicians and experimented music listeners. Considering the relative validity of such subjective and qualitative appreciation, one can conclude that the room is relatively reverberant, with low sound definition and significant echoes. Although the acoustic characteristics required for a performance room depend on the musical style, it would be realistic to consider that, assuming the previous characteristics, the Arena D´Évora does not have reasonable acoustic conditions for musical performances.



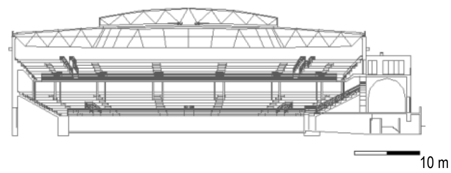


Figure 01 - Plan of the inferior balcony and view over the arena and stage (above). Longitudinal section (below).

The floor plan of the building is a regular polygonal structure with 16 sides. At centre of the ground level there is the arena zone, with a 35 m diameter delimited by a concrete wall (with six doors) up to approximately 2.3 m; starting from this height, the lower balcony rises up to 5 m from the Arena level. This lower balcony is a concrete ring with approximately 7.5 m width and an exterior diameter of approximately 52 m. The higher balcony rises up to about 11 m above the Arena level and corresponds to a concrete ring of about 3.5 m width. The exterior walls are built in stone masonry and are reinforced with concrete layer in both surfaces; these walls rise up to 11 m above the Arena floor level. The Arena roof is partly built in metal sheet supported by a metal structure. In the centre of the roof there is a detachable elevated zone in polycarbonate sheets.
The following table shows the technical details of the Arena with relevance to acoustics.

Table 1 – Technical details of Arena D`Évora.

V – volume (m³)	30 000 m³
H – average height	15 m
L – average with	52 m
C – average length	52 m
D – distance from the stage to the most remote listener	38 m
S – total surface area	8000 m²
S_A – total audience area	400 m²
N – number of seats	5000

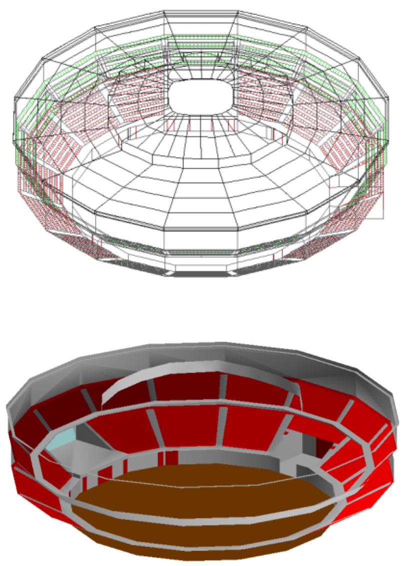


Figure 02 – Illustration of the digital model of Arena: AutoCAD (above) e EASE (below).

Results

Situação actual

The characteristics of reverberation measured in the Arena D`Evora were quantified by the parameters T_{30}, T_{20}, T_{10} and EDT (fig.03).

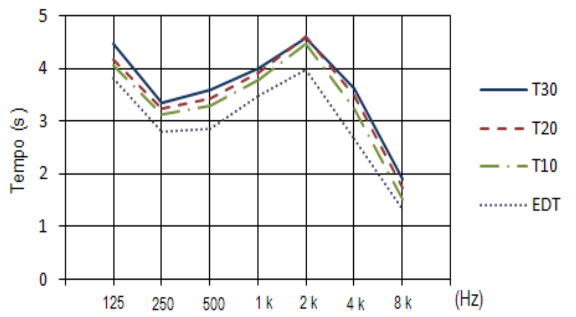


Figure 03 – Measured reverberation characteristics of Arena D`Évora.

The reverberation time of the Arena D`Evora was measured in situ and the average values of T_{30} obtained for the frequency bands of one octave validated the computational model for the room. Thus, the reverberation time predicted for the unoccupied room corresponded to the time of reverberation measured for the parameter T_{30}. Fig.04 shows the curve of reverberation predicted considering the arena zone fully occupied (about 900 seats) and for the whole room fully occupied (arena zone, upper and lower balcony).

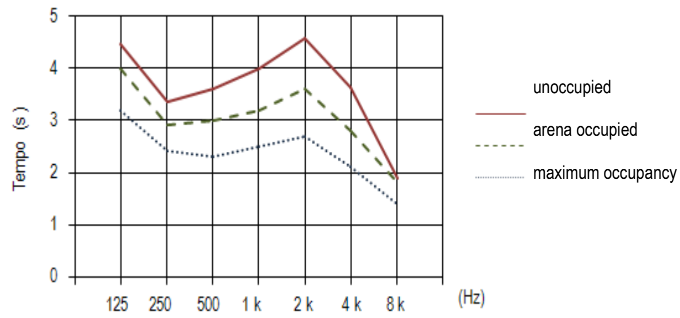


Figure 04 – Prevision of reverberation characteristics for different occupancies of Arena – actual situation of Arena.

The distribution of values taken by parameters C_{50} and C_{80} is shown in fig.05, for an unoccupied state of the Arena.

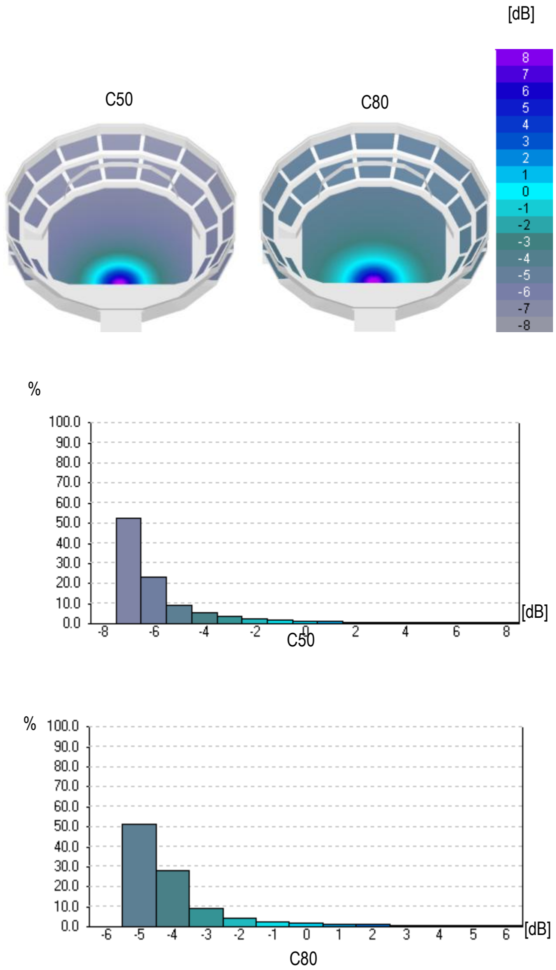


Figure 05 – Spatial distribution of C_{50} e C_{80} values (1000 Hz) - Arena unoccupied.

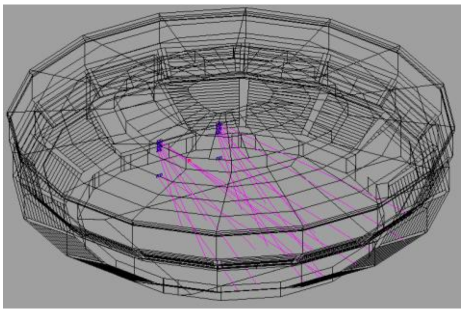


Figura 06 – Illustration of the public address system typically used in Arena.

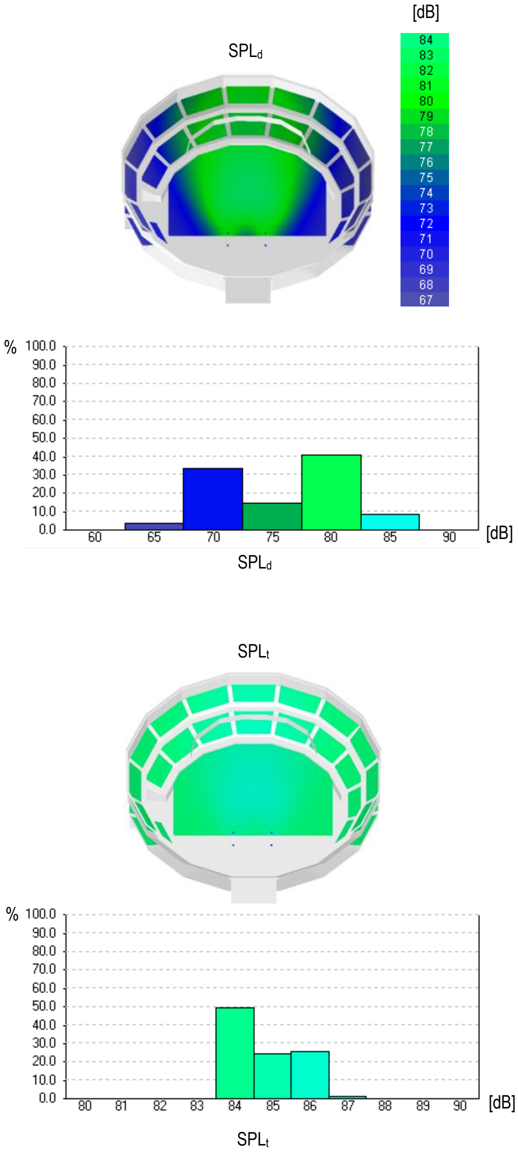


Figure 07 – Spatial distribution of SPL_d e SPL_t values (1000 Hz).

Hypothetical situation

To reduce the reverberation of the Arena it was simulated the application of sound absorption system for the vertical elements, including the walls of the arena zone, and the walls of the lower and upper balcony. To increase the initial sound energy ("early sound") received in the audience was simulated a reflective surface suspended over the stage area and also a distributed sound reinforcement system (fig. 08).

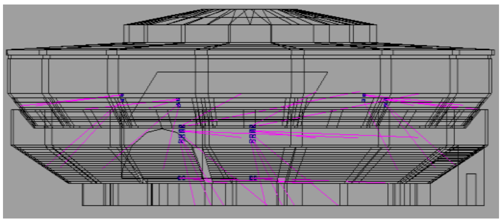


Figure 08 – Illustration of the distributed public address system propose for Arena.

The reverberation time predicted in Arena after the implementation of the acoustical solutions mentioned above is shown in Fig.09.

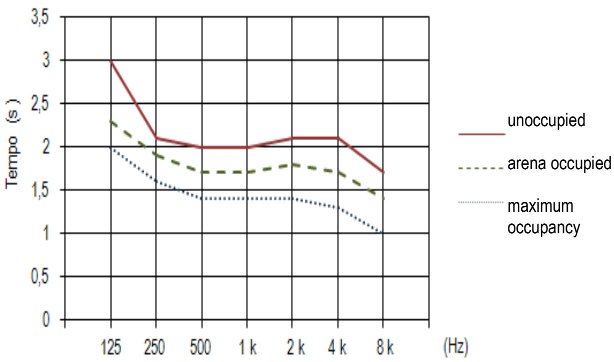
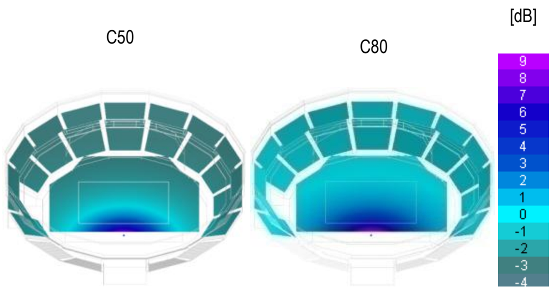
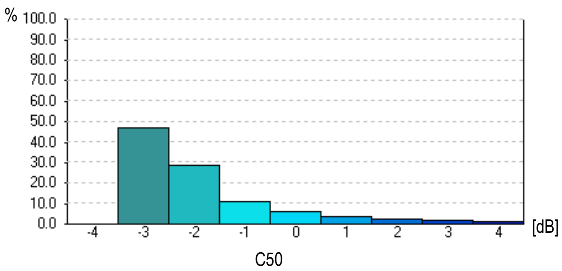


Figure 09 - Prevision of reverberation characteristics for different occupancies of Arena – hypothetic situation of Arena.

The distribution of values taken by parameters C_{50} and C_{80}, with the application of the reflector, is shown in Fig.10 for an unoccupied state of the Arena.





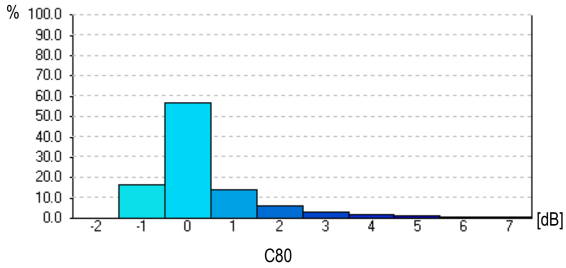
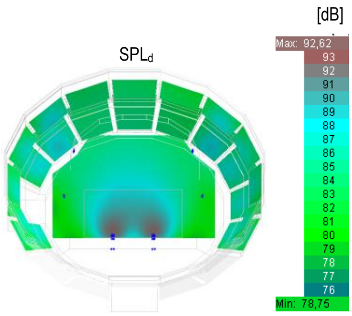
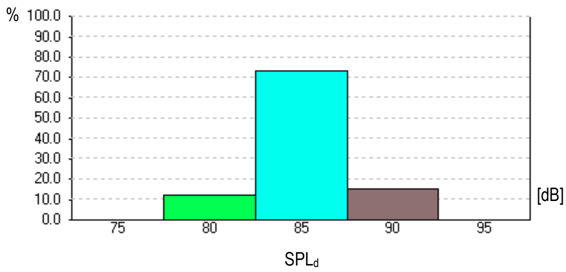
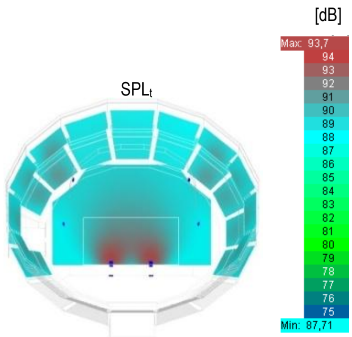


Figure 10 – Spatial distribution of C_{50} e C_{80} values (1000 Hz) - Arena unoccupied.

The distribution of values taken by parameters SPL_d (sound pressure level of the direct sound field) and SPL_t (sound pressure level of the reverberant field) with the application of distributed sound reinforcement system is shown in fig.11.







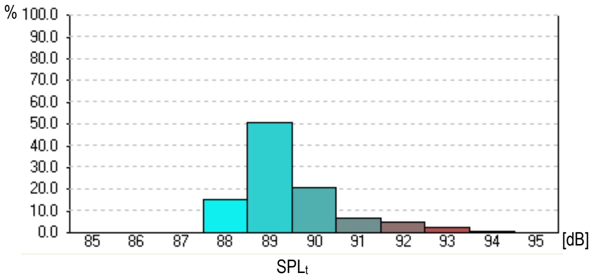


Figure 11 – Spatial distribution of SPL_d e SPL_t values (1000 Hz).

Discussion and conclusions

The reverberation time T_{mid} measured without occupation was 4.1 s (Fig.03). This value, even for a unoccupied Arena, reflects the site is excessively reverberant.

The digital modelling of the Arena required the construction of an architectural 3D model of place (performed in autoCAD) so that all the interior surfaces relevant for sound absorption and reflection processes, could be represented (Fig.02). The computational simulation allowed to model the current situation of Arena and a hypothetical situation, considering a set of acoustic conditioning measures for acoustic correction in this room. The digital model for the Arena's acoustic and electro acoustic analysis was validated by the reverberation time measured in situ. In this way, it was possible to calibrate the sound absorption characteristics of the building's coverage, assuming that the time of reverberation of Arena without occupation provided by the model is equivalent to the measured T_{30}.

The modelling of the current situation allowed to evaluate on the influence of the audience in the reverberation, where the predicted T_{mid} with maximum occupancy of arena zone was 3,3 s, and with maximum occupancy, was 2,5 s (Fig.04). The Arena's reverberation time is significantly high for the attainment of any performance based on music or voice, although it can be considered that the reverberation is acceptable to those performance that occur, normally, in places of worship (churches and cathedrals), as the liturgical music, music coral and organ. Symphonic music concerts, depending on the music content, may be improved or not by a higher reverberation time. For the characteristics of clarity and sound definition, the predicted values for the parameter C_{80} are in the range of -5 to -2 dB and for the parameter C_{50} are in the range of -7 and -6 dB, with the exception of areas of audience very close to the stage (Fig.05).

The performance of the PA system was tested in the actual situation of the Arena using the digital model (Fig.06). Regarding the spatial distribution of sound pressure level of direct field, it was found that the maximum difference in sound level between different listening places of the Arena is approximately 20 dB. This value reflects a large non uniformity of the sound pressure levels spatial distribution which usually translates into very unfavourable acoustic conditions for different places of the audience (Fig.07). Regarding the spatial distribution of sound pressure levels of the reverberant sonorous field, one observed a difference of about 3 dB between minimum and maximum values. Although a uniform spatial distribution occurs, this result reflects the reverberant character of the Arena.

The results of the hypothetical situation demonstrated a significant improvement in room acoustics, in terms of reverberation, clarity and spatial distribution of sound pressure levels of direct field. The predicted reverberation time, T_{mid}, without occupation was 2.0 s, 1.7 s with maximum occupancy of arena zone and 1,4 s with maximum occupancy of Arena (Fig.09). The modelled values are acceptable for orchestrated music concerts and considering the dimensions of the room one fairly concludes that these reverberation values are close to the minimum acceptable for rock concerts and other rich in rhythmic composition styles. The predicted values of C_{80} are in the range of 0 to 3 dB, quite acceptable for musical performances. The predicted values of C_{50} are between -3 and -1 dB, which for reverberating rooms indicate a reasonable clarity for voice (Fig.10). For the spatial distribution of direct field sound pressure levels, the distributed system of sonorous strengthening allowed to decrease the difference between maximum and minimum direct sound pressure level field down to 10 dB, increasing the uniformity of the sound levels spatial distribution (Fig.11).

The simulated acoustic conditioning measures intended to highlight the effective improvement of Arena D`Evora acoustic quality, based on objective parameters for acoustic characterisation. The hypothetical acoustic correction of the Arena should follow the guidelines pointed out, however the adequate dimensioning of the systems of sound absorption, of the reflecting surfaces and of the system of distributed sound amplification requires a more detailed analysis, where several solutions should be assessed taking into account their feasibility, given the architectural concept, the structural conditions and the different purposes of the room. This last factor, indicates the need to consider the application of systems of acoustic conditioning with variable characteristics. One stresses that Acoustics should be taken into account since the early stages of a project, allowing the development of solutions with better benefit/cost ratio.

References

[1] CROCKER, M. – Handbook of Acoustics. USA, John Wiley & Sons, Inc., 1998.

[2] BERANEK, L. L. – Music, Acoustics and Architecture. USA, John Wiley & Sons, Inc., 1962.

[3] BERANEK, L. L. – Concert and Opera Halls: How they sound. New York, Acoust. Soc. Amer., 1996.

[4] ANDO, Y. – Concert Halls Acoustics. Berlin, Springer, 1985.

[5] CAVANAUGH, W.; WILKES, J. – Architectural Acoustics, Principles and Practice. USA, John Wiley & Sons, Inc., 1999.

[6] CARRIÓN, A. – Diseno acústico de espacios arquitectónicos. Barcelona, Edicions UPC, 1998.

[7] G. M. HULBERT, D. E. BAXA, A. SEIREG, "The use of quantitative criteria for optimum design of concert halls", J. Acoust. Soc. Amer., vol. 67, 2045-2054, 1980.

[8] D. E. BAXA, A. SEIREG, "Criterion for quantitative rating and optimum design of concert halls", J. Acoust. Soc. Amer., vol. 71, 619-629, 1982.

[9] V. L. JORDAN, "Acoustical Criteria for Auditoriums and Their Relation to Model Techniques", J. Acoust. Soc. Amer., vol. 47, 408-412, 1970.

[10] M. R. SCHROEDER, "New Method of Measuring Reverberation Time", J. Acoust. Soc. Amer., vol. 37, 409-412, 1965.

[11] A. J. BERKHOUT, D. de VRIES, M. M. BOONE, "A new method to acquire impulse responses in concert halls", J. Acoust. Soc. Amer., vol. 68, 179-183, 1980.

[12] S. MÜLLER, P. MASSARANI, "Transfer-function Measurements with Sweeps", J. AES, vol. 49, p.443, 2001.

[13] A. FARINA, "Simultaneous measurement of impulse response and distortion with a swept-sine technique", J.AES, vol. 48, p. 350, 2000.

[14] ISO 3382-1:2009, Acoustics – Measurement of room acoustic parameters – Part 1: Performance spaces.