Experiments in Room Acoustics: Modelling of a Church Sound Field and Reverberation Time Measurements

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Abstract: - In this paper a study of the acoustical response of a new built church is shown. This study is based on the measurement of reverberation time, adopting the noise interrupted method, according to the International Standard. This method allows to evaluate the reverberation time by means of acoustical sound level acquisition and analysis. The reverberation time is one of the principal parameters to be optimized in order to design and/or verify the acoustical behaviour of a room and consequently to guarantee a good people hearing sensation. In a post-opera intervention, the reverberation time can be improved modifying the reflecting surfaces of walls, floor and roof, in order to reduce the energetic contributions of late reflections. This improvement can be achieved by replacing or covering reflecting surfaces with absorbing panels or carpets. The design of an appropriate intervention can be aided by a dedicated simulation software, as it is shown in the last part of the paper.

Key-Words: - Acoustical Field, Church Acoustics, Reverberation Time, Simulation Software.

1   Introduction

Church, and in general cult's place, represents an excellent experimental object of the sound propagation study in a closed space. In these places, in fact, many acoustical sources are present: loudspeakers, musical instruments, air conditioners, people talking and moving etc. In addiction, in the design phase, one must insure that people who are supposed to hear the sound, are able to hear it from the right direction and at the right time. Once a sound source launches sound into the air, the acoustics of the church "processes" the sound, until the sound dies out.

A good signal is usually thought of, as coming from a good loudspeaker, arriving loud and clear. Generally people prefer the sound of speech to be about the loudness of natural speech, about 65 dBA on the average.

The sound contractor has to hang and align the loudspeaker so it delivers sufficient direct sound into the congregation or audience. The audio tech has to properly set the loudness level and the equalization so that the speaker is not too quiet. However, in order to obtain a good signal, simply adjusting the direct sound from the speaker is not enough. To improve the intelligibility of signals, in fact, early reflections must be taken into account. Any reflection that arrives at the listener's ear within the first 50-60 ms after the arrival of the direct sound is considered part of the "direct signal" [1], i.e. brain cannot separate the direct sound from any of the very early reflections (Haas effect). They merge together and become one sound. One of the big challenges in the design of large auditoriums is to create early reflections. It is not so easy to position sound reflecting surfaces near enough to so many people and still have a nice open architectural space to look at.

In addition, during the planning one must also take into account echoes and late reflections that strongly interfere with understanding of speech. In particular late reflections are those reflections that are too strong and arrive just at the wrong time, when our listening/understanding process is most vulnerable to interruption. This sensitive time period begins just after the end of the early reflection time period. It is a time period when our understanding of

what we hear is very susceptible to being easily confused. One of the most notorious of these late reflections is echo. Designing an auditorium is, to a significant degree, a process of creating an acoustic space that eliminates, or at least temper, late reflections, especially echoes, for all listeners.

Late reflections can be moved forward in time into the early reflection time period, by repositioning the reflecting surface. They can also be shifted backwards in time into the reverberation time period. Late reflections can be simply eliminated by absorbing them. They can also be split into numerous small reflections by diffusing them.

Regardless of how the late reflections are managed, it is essential to reduce their strength to have an intelligible space. The resulting moment of relative quiet, the early time gap, is a quiet interlude that provides for the understanding of what is being said. This paper is devoted to the acoustical characterization of a new built church by means of Reverberation Time measurements that is universally considered as the main marker of acoustical quality of a closed space.

2 Experiment description

The church under study is "*S. Maria di Costantinopoli*" church, placed in Angri (SA), Italy, and the measurement was performed on the 20th of June 2008, during day time, from 4 p.m. to 7 p.m. .

The architectonical and geometrical features of the hall are the following:

- Volume $\cong 4000$ m^3 ;
- Inner surface = 465 m^2 ;
- Average height = 8,50 m ;
- Full occupancy capacity: about 200 people.

In order to get a correct analysis, we performed the measurements campaign according to the International Standard (ISO 3382 [2]) regarding measurements of Reverberation time in a closed environment.

First of all we measured the temperature of the room, according to the cited Standard; in particular we found an internal temperature of 25°C.

We performed the measurement in an empty occupancy condition in order to best evaluate the Reverberation Time, since the presence of people influence the measurement. In fact the acoustical response of the hall changes in fully occupancy since the absorbing surface increases and so the Reverberation Time decreases.

The noise source, which will be described later, has been placed on the presbytery of the church, as shown in Fig. 1.

The number and the positions of measurement points have been chosen in order to have a complete mapping of the environment, since this set of data can be used for different purposes. However the choice fulfils the ISO Standard cited above [2], which requires that:

- the positions of microphones must be at least half wave length far from each other, i.e. a minimum distance of about 2 m for the common frequency range;
- the distance between each position and the nearer reflecting surface, including the floor, must be at least one quarter of wave length, i.e. generally about 1 m ;
- the microphones must not be too much close to a source position, in order to avoid a big influence from direct sound. In particular, in [2] a minimum distance, d_{min}, is defined as:

$$d_{\min} = 2\sqrt{\frac{V}{cT}}$$

where V is the volume of the hall, c is an approximated value for the speed of sound and T is an estimation of the foreseen Reverberation Time. In our calculation we consider $V \cong 4000$ m^3, $c \cong 300$ m/s and $T = 2$ s, that result in: $d_{min} \cong 5$ m .

We can resume these considerations on the position of microphones in the following table:

Relative Position	Theoretical Distance	Chosen value
Microphone-Microphone	> $\lambda/2$	2 m
Microphone-Reflecting surface	> $\lambda/4$	1 m - 1,5 m
Microphone-Floor	> $\lambda/4$	1,5 m
Microphone-Source	$d_{\min} = 2\sqrt{\dfrac{V}{cT}}$	5 m

Table 1

Following these criteria we chose the measurements points reported in Fig.1. As suggested in the ISO Standard [2], in each position we performed three different measurements and then we calculated the mean value.
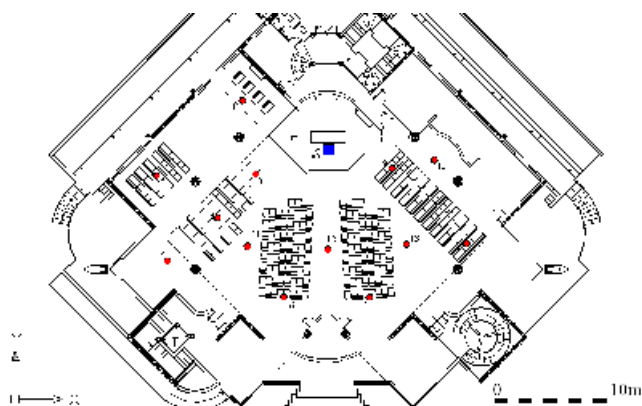
Fig.1: Distribution of measurement positions (red points) and of the source (blue square).

With this measurement points distribution we can distinguish three different zones:

- *Forward zone*, where the direct acoustic field overtake the delayed one;
- *Intermediate zone*, where the field can be considered "semi delayed";
- *Backward zone*, where the delayed field is predominant on the direct one.

3 Measurement details

3.1 Measurement instruments

The sound data acquisition has been performed by a two channel SINUS Analyzer, SOUNDBOOK (Serial Number 0614), equipped with a pre-amplifier Larson Davis PRM902 (Serial Number 3217), a microphone Larson Davis 2451 (Serial Number 8183) and an acoustical calibrator Larson Davis CAL200 (Serial Number 4874).

The microphone has been oriented with an angle of 80° with respect to the noise source, even if in diffuse field conditions, the results are not affected by the orientation.

Once the measurement has been started, the software is able to detect and record the sound pressure level for each frequency.

3.2 Noise source

The noise source is used in order to produce a relevant amount of sound energy in the environment in which the measurement of the Reverberation Time must be performed.

According to the ISO Standard on the measurement of the Reverberation Time in a closed environment [2], the noise source can be a peace gun or an isotropic source.

In our case, the chosen source is the OMNI12 Metravib 01dB^{TM} Italia noise source, shown in Fig.2. This isotropic source is made of 12 speakers embedded in a robust dodecahedron box, which produce an acoustic field very close to the spherical approximation.

The chassis, equipped with wheels, contains the amplification, the power supply system, the noise generator and the mechanical support.

The instrument can be switched on and off with a remote control, so that the noise source can be controlled also without the presence of an operator which could affect the results of the measurements.

Fig.2: Isotropic source and its operating position.

The major technical features of OMNI12 are:

- Standards: UNI EN ISO 140-3 ISO 3382
- Sound power: 350 W
- Frequency range: 70Hz – 16KHz
- Sound power level: 108 dB at 1 meter distance (L_W = 119 dB)
- Mechanical support: telescopic adjustable bar
- Speakers: 12 x 5" bi-cone
- Diameter: 50 cm
- Weight: 18 Kg

4 Measurement method

The International Standard cited above [2], describes two different methods for the Reverberation Time measurement:

- "interrupted noise" method;
- "integrated response to the impulse" method.

For our measurements, we chose the first method [3], taking also into account the features of our instruments.

In particular we use the following experimental procedure:

1. Switch on the source, by remote control, so that the acoustical field can go in a stationary state;
2. Start the measurement and recording of data;
3. Switch off the source, so that the acoustical field starts its decay;
4. Stop the measurement and data recording.

5 Background noise

The background noise is a fundamental parameter for the church acoustics. Its measurement has been performed in three positions (11, 12, 13, see Fig. 1). Each measurement lasted 3 minutes.

The average values of sound pressure level, L_{eq}, defined in [4], with A weighting, are the following:

Position	L_{eqA} [dB]
11	40,2
12	38,7
13	38,3

Table 2: average background noise level in three different position of the church.

So we can resume that the background noise level is approximately 39 dB. This value is relevant and contributes to the loss of intelligibility of speech and music.

The background noise, in fact, can be considered an acoustical polluting factor that, if combined with an incorrect reverberation time, causes many problems to the acoustics of the environment.

In our case study, the background noise is originated by the acoustical context in which the church is placed, and depends on many reasons. The principal sources are the external noises (car traffic, industrial plants, other human activities, etc.) and the internal activities (meetings and other issues in the nearby rooms, internal implants operating, etc.).

The greater the background noise is, the lower is the capability of distinguish a signal mixed with noise with similar frequencies and levels.

6 Reverberation Time calculus

In order to evaluate the Reverberation Time, the strategy is to calculate the Reverberation Time for each frequency, to mediate on the different measurement points, and finally to mediate on the frequencies.

So, data detected and recorded by the instrument have been processed with the following procedure.

The Reverberation Time is calculated according to the ISO Standard. In particular we consider the descending part of the plot, starting from the point where the level is lowered of 5 dB (with respect to the average maximum value), in order to cut the rippled zone of the signal, until the point in which the level is lowered of 65 dB. We refer to this value as RT_{60}.
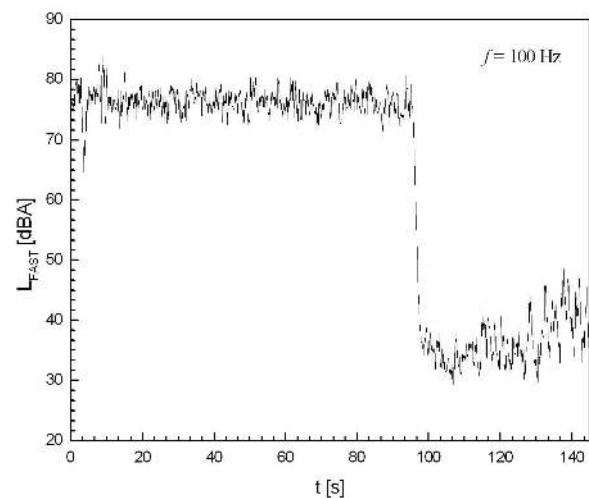

Fig.3: Time history of the detected signal for a fixed frequency (100 Hz).

Since quite always the signal is not more than 60 dB higher than the background level, the ISO standard [2] provides a mathematical procedure for the calculation of the Reverberation Time. The method consists first in the linearization of the decay of the noise signal in a smaller range and then in an extrapolation of the required value directly from the plot. In our case, we chose a 30 dB decay range, from -5 dB to -35 dB, and then extrapolate the Reverberation Time at -65 dB, typically called RT_{30}.

This procedure is well summarized in Fig. 4, for two different frequencies in the same measurement position (Point 9). The dashed lines represent the average maximum level decrease of 5 dB and 35 dB for each frequency, while the solid lines are the linear regression of the experimental data in the range between the two dashed lines. This leads us to the estimation of the RT_{30} directly from the plot (see Fig. 4).
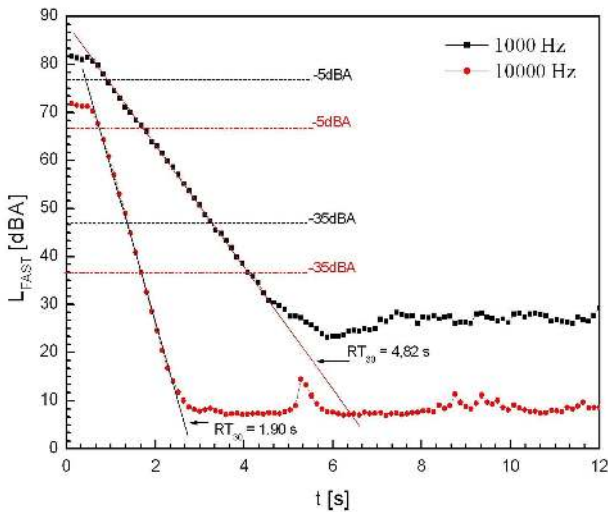

Fig.4: Signal Level and Reverberation Time evaluation for 1000 Hz (black squares) and 10000 Hz (red circles) signals, related to Measurement Point 9.

The calculation of the RT_{30} has been performed for each frequency at a fixed measurement points, resulting in a complete set of data for the whole hall.

In the following table we report the average values of the Reverberation Time, mediated on the different measurement positions.

Frequency [Hz]	RT_{30} [s]
100,0	3,03
200,0	3,19
500,0	4,26
1000,0	4,73
2000,0	4,34
5000,0	2,75
10000,0	1,88
20000,0	1,89

Table 3: Reverberation time values mediated on the different measurement positions for each frequency (octave)
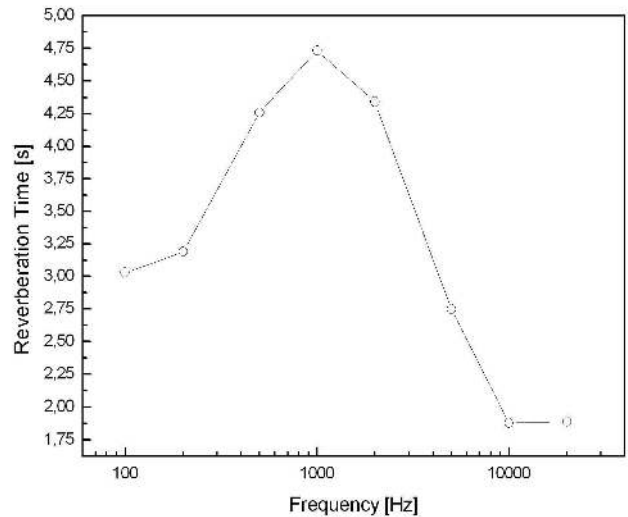

Fig.5: Reverberation Time values versus Frequency, mediated on the measurement positions.

Finally we can evaluate an average Reverberation Time for the environment, mediated on the octaves, and the result is 3,26 s.

This value has to be compared with the preferred values suggested in literature. We report in Fig. 6 the plot of Reverberation Time versus the room volume, for different purposes [5, 6]. It is clear from the plot that our value is not suitable with the purpose of the church; in fact the Reverberation Time is too high to ensure a good intelligibility. This is in agreement with our personal experience during the measurements campaign.
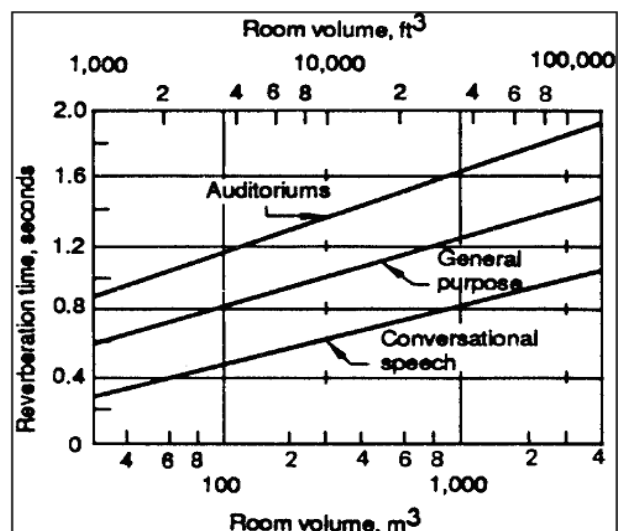

Fig.6: Reverberation Time preferred values versus room volume for different purposes [4].

7 Acoustical correction proposal

Measurement results, reported in previous paragraphs, exploit the need of an acoustical correction design for the church.

Data analysis, in fact, allows us to observe the presence of a high Reverberation Time, especially for medium-low frequencies. This configuration is not suitable for the intelligibility and perception of speech and music. Thus the correction proposal has the aim of reducing the Reverberation Time.

The audio system of the church is made of different speakers, placed in front and on the side of the ceremony hall. Regarding the side speakers, it is important to remark some geometrical features. They are six (three per side) and they have a circular shape (see Fig.7). Their height is 3 m. Moreover, other speakers are present on the side walls and on the light places.

If one compare the speakers number and their placement in the hall with the dimensions of the church, it can be observed that these parameters have been tuned in order to favour the diffusion of direct sound in the seats direction, but also in the walls one, favouring the reflections. In fact this configuration leads a surface to be hit many time by many sound rays coming from different sources. Thus, the high number of speakers and the geometrical configuration of side walls, that focus sound rays towards the centre of the hall, produce a high reverberation. Therefore, many reflected rays continuously hit the auditor, i.e. within a very small time interval, with relevant sound pressure level values due to a low absorption of surfaces (see next paragraphs).

Fig.7: Side speakers position and geometry in the church under study

Thus a first solution to be proposed is the reduction of the number of the speakers, since the maximum distance source-receiver (about 10 m) is not high enough to justify the presence of so many speakers.

In order to better model the behaviour of sound rays in the church, the Autodesk© Ecotect[tm] software has been used. It is a building design and environmental analysis tool that give the possibility to simulate and analyze the thermal and acoustical response of a building. Moreover it allows designers to work in 3D and furnish an impressive presentation of the final design and acoustical behavior of the hall under analysis.

The real configuration of the church has been fully implemented in the Ecotect software framework. The first step consists in the drawing of the project, together with the definition of the different construction materials, very important in order to model the sound absorption.

Then, the simulation of the source must be performed, by means of supposing an isotropic source placed in the centre of the altar, which is the position of the OMNI12 source we used.

Now different analysis can be performed, such as Reverberation Time evaluation, reflections study, materials analysis, etc.

8.1 Geometrical analysis

A first observation that can be done regards the shape of the church, which results to be well suitable for a good acoustical diffusion, since the conformation of side walls is such that sound is focused towards the central zone of the hall.

Another aspect related to the geometrical features of the building regards the "*spatiality*" of sound.

Fig.8: Front view of the church, in the reality (up) and in the simulation (down)

Fig.9: Entrance the church, in the reality (right) and in the simulation (left)
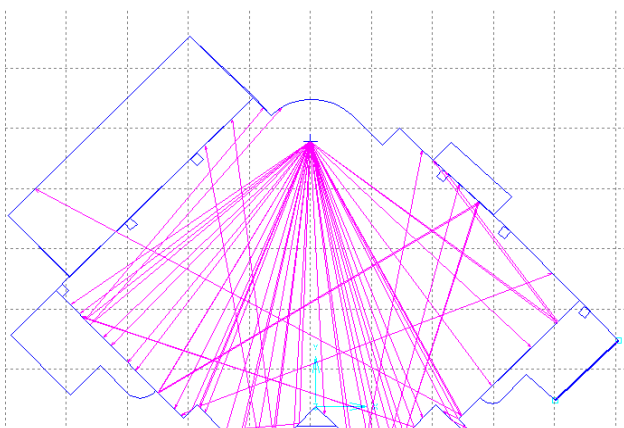

Fig.10: Sound reflection pattern. The cross represents the source.

The shape of the church assures that first reflections of sound come to the auditor from different directions, also different from the source. In such a way the perception at the same time of direct and reflected sound leads to different stimulations to the two ears, creating the so-called "*spatial sensation*" or spatial impression.

8.2 Materials analysis

The major problem from the acoustical point of view is given by the presence of relevant surfaces, in terms of dimensions, but with low acoustical absorption coefficient. Therefore, this produces a reduction in the equivalent surface values.

In particular, the two walls close to the entrance, are made of concrete, so they contribute in a significant way to the reflection of sound in the environment, because of the absorption coefficient of concrete, which is particularly low for each frequency (average value around 0.1). Moreover, the floor is made of slate which also contributes to the diffusion of acoustical waves.

Regarding the apse, the surface behind the priest position is made of a calcareous stone (with low

absorption coefficient) and has a particular curved geometry which focus the reflected rays to the altar (see Fig. 11). This leads to the production of an additional reverberation, since many rays reflect on this curved surface and are collected by the microphone with a quite significant delay. In fact, switching off the microphone, one can notice a reduction in the reverberation.

Finally we can say that, since the different surfaces don't attenuate sound rays, as it should be, the reflections present low reduction in the sound pressure. This means that the secondary rays (delayed with respect to the direct sound) that reach the auditor, cause a disturb in the perception and a loss of intelligibility. Moreover the high values of Reverberation Time, especially for medium-low frequencies, are quite fully explained by these considerations.
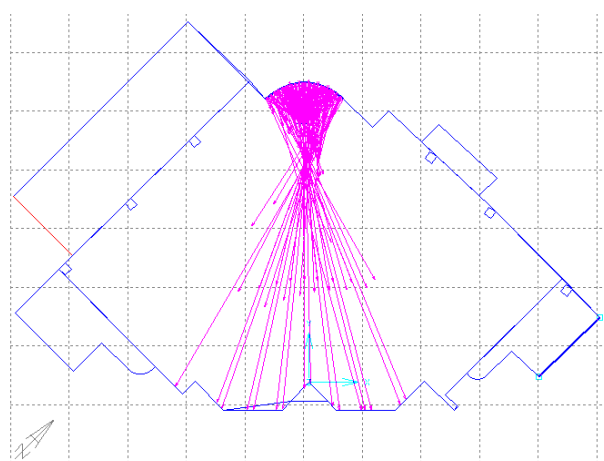

Fig.11: Reflection on the curved surface behind the altar

8.3 Reverberation Time evaluation

In order to give an estimation of the Reverberation Time, once the position of the source has been fixed, the software produces a bunch of acoustical rays, which interact with environment surfaces in terms of reflections, absorptions, etc., until their sound pressure level is lowered of a certain quantity (e.g. 40-60 dB). From that point, the ray is considered completely absorbed and it doesn't affect anymore the acoustics of the hall. The collective behaviour is resumed in the estimation of the Reverberation Time for each frequency, showed in Fig. 12, which takes in account the geometrical features of the hall and the absorption coefficient of the materials.
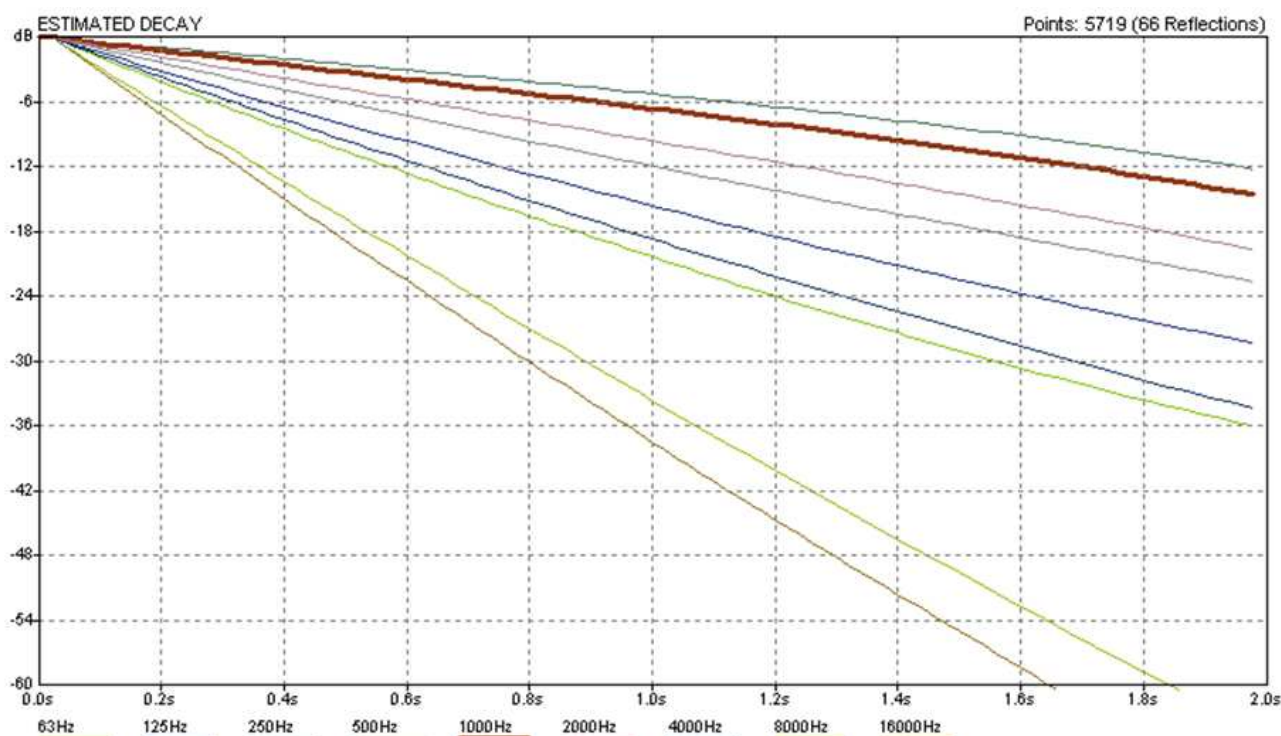
Fig.12: Reverberation Time of the church for each frequency

Starting from the considerations of previous paragraphs, an acoustical intervention can be performed, by means of inserting an absorbing material on one or more surfaces. Considering the geometry of the hall, we proposed the insertion of two absorbing panels on the concrete walls placed near the entrance, as shown in Fig. 13.
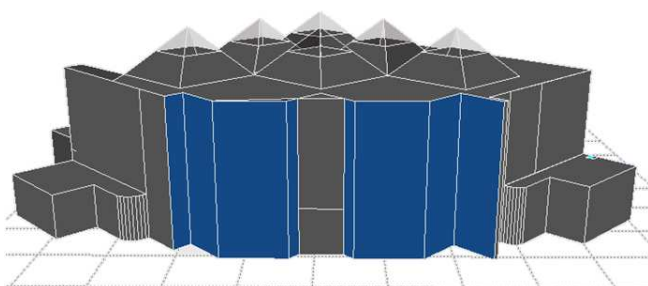

Fig.13: Ecotect modelling of the church, with, in blue, the walls covered by the absorbing panels

Once inserted these panels, we can observe a relevant attenuation of sound level when particles reflect on them. This solution strongly affect the Reverberation Time, which is shown in Fig. 14.

8 Conclusions

In this paper we reported an experimental activity regarding the measurement of the Reverberation Time in a new built church. We made measurements in different points, so that a kind of acoustical internal mapping can be achieved. Such as in [6, 7] (measurement of noise in external environment), the time history of the signal has been deeply investigated in terms of spectral analysis and parameters evaluation.

The Reverberation time has been calculated according to the ISO 3382 Standard [2], for each frequency octave and for every position. A mean value has been calculated in order to give an overall estimation of the Reverberation Time. This value is about 3 s, and, even if an optimal value cannot be defined because it depends on the personal sensibility, it appears to be higher than the preferred range of values [5], for the purposes of a church. In fact, a long reverberation time results in a loss of intelligibility which is quite evident also according to our experience.
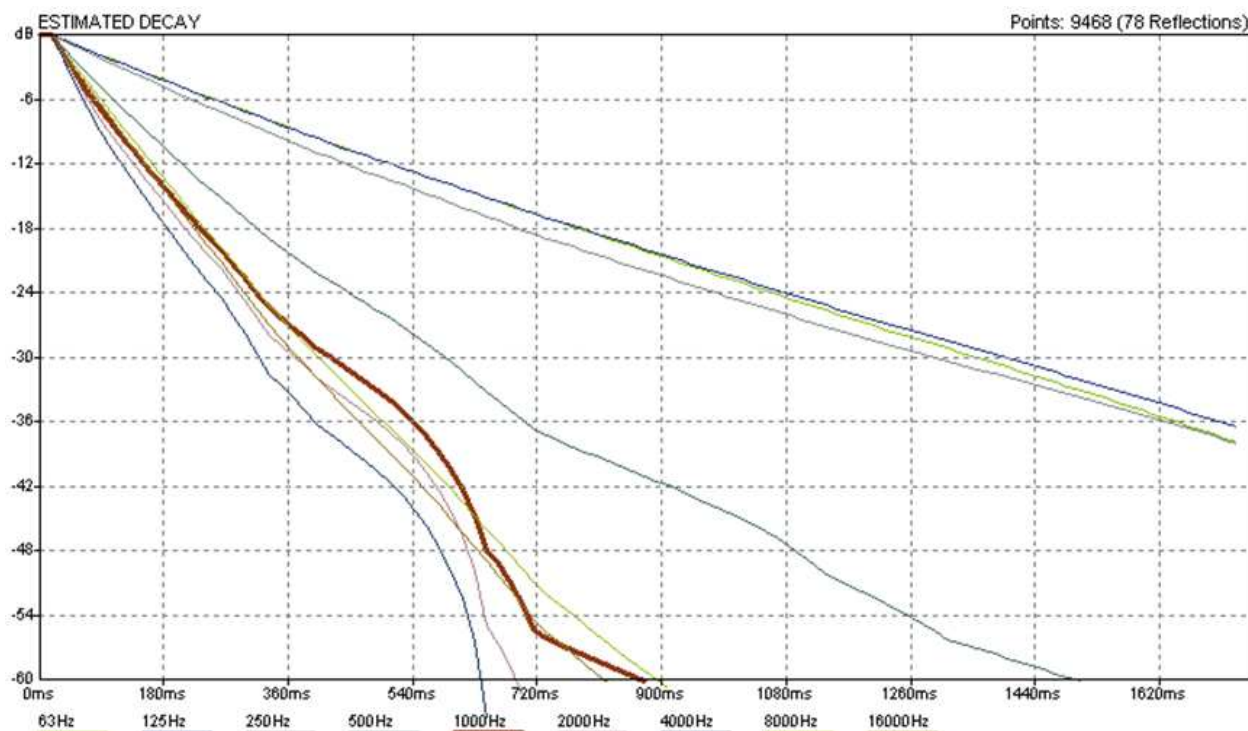
Fig.14: Reverberation Time of the church for each frequency, after the insertion of absorbing materials

The improvement of the acoustical parameters of the church can be achieved by increasing the absorption coefficient of the internal surfaces with some appropriate operations, like adding suitable absorption materials and/or architectonical inserts. In this way, in fact, we can reduce the energetic contributions of late reflections, that are the ones that contribute to the loss of intelligibility.

This intervention has been simulated and validated with the Autodesk© Ecotect[tm] software, by means of modelling of the church and estimation of Reverberation Time, before and after the insertion of absorbing panels on two concrete wall close to the entrance.

In future works, this methodology can be applied to more churches or similar environment, in order to perform a significant comparison between different case studies.

Acknowledgements

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