# ANALYZING THE EFFECTS OF ROOM ACOUSTICS IN DIFFERENT MUSIC RECORDINGS

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Master thesis UPF 2011

Master. in Sound and Music Computing

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...To Santi, because none of this would have been possible without him...

# Acknowledgements

I would like to thank Emilio Molina, Àlex Rodríguez, Alfred Tapscott, Jaume Yelo and Marcus Sucram for their help evaluating the rooms. Their opinions and subjective perception were fundamental for the development of this master thesis. Also I would like to thank Enric Guaus for his help analyzing the results of the measurements, and Perfecto Herrera for his help in the correct evaluation of the surveys. Their advices were fundamental for the successful development of this project. Enric and Ferran for all their advices, help and patience across this project, and Santiago Ortega for his help in writing the report and making the presentations.

# Abstract

Everyday, in the common activities of life, we experience to listen to music or sounds in different environments (the metro, elevators, the bathroom, etc). Every environment has different acoustic properties that can be preferred or annoying for a determined kind of music or for a different purpose.

In this project we analyzed those properties for 4 different rooms, and correlated them with the purpose they have, and with the perceptual opinions of 5 musicians playing the same song in all of them. In this way we show which properties are preferred for each purpose of the room, which characteristics are annoying to the musicians, and why.

Cada día, en las actividades cotidianas de la vida, escuchamos música o sonidos en distintos ambientes (el metro, ascensores, el baño, etc). Cada uno de estos ambientes tiene diferentes propiedades acústicas que pueden ser preferidas o molestas de acuerdo a determinados tipos de música o propósitos.

En este proyecto, analizamos estas propiedades para 4 cuartos diferentes, y las relacionamos con su propósito y con las precepciones de 5 músicos que tocaban la misma canción en todos ellos. De esta forma mostramos qué propiedades se prefieren para cada propósito del cuarto, qué características son molestas para los músicos y por qué.

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# Chapter 1: Introduction

Since the beginning of the nineteenth century, humans have been concerned about architectural acoustics. Musicians of that period played their pieces according to the halls where they were located, and tried to use these physical properties to have the sound they wanted (Kuttruff, 2000). The design of concert halls always has to involve the musicians' preferences and the purpose of each one of them.

For this reason, room acoustics is a subject that has been studied for decades. Many investigations have been made about this and we have a large theoretical background where we can consult the information we need. Room acoustics is a very interesting way to correlate the music as an art with the physics, the wave theory and the properties of all the sound signals that produce it.

Even though room acoustics have been studied a lot across the years, there hasn't been a deep analysis in the ESMUC (Escola Superior de Música de Catalunya) yet about the acoustics of its halls, related to the opinion of its students. This project is interesting for the ESMUC as it provides information about the preferences from some of their students and experienced musicians about their rooms. We would like to find the objective parameters that may describe the ideal room for the analyzed songs, focusing in the acoustical configuration of the rooms of the ESMUC, and correlating these parameters with the subjective opinion of experienced musicians.

In order to do this, we will record musical pieces in halls with different acoustic treatment in the ESMUC, evaluate the preferences of the musicians who interpreted the songs, and then correlate and compare these results with objective measured characteristics of the halls.

To set the boundaries of the project, in order to comply with a scope within the reach of a master's thesis, we decided to work mainly acoustic music. For the sake of the experience portability among rooms, we decided to record just guitars, voices, basses, strings and a portable percussion set. The musicians were ESMUC students who wanted to help with the project, and the rooms were 4 different ones in the ESMUC: The chorus chamber, the organ hall, the rehearsal room A108 and the recording studio A124.

The recording method and the microphone techniques chosen will be explained later, in the chapter 3.2 of this document.

Before recording, the reverberation time of the rooms RT60 will be measured and compared with the theoretical value given by the acoustic engineer for each room during building up. We will also measure the frequency response and the transfer function of the room using the software Smaart®, so we can compare these objective properties for each one of the rooms.

## 1.1 Motivation

Personally, I have always been interested in how the music expresses feelings in such an artistic way by mixing sounds that can be analyzed with physical theories and objective measures. All music that we listen to everyday in our lives is composed by several sound waves that varies continuously or discontinuously, usually rhythmically, and can be analyzed from a physical point of view. The acoustics of the rooms is the science of sound. It plays with resonance, frequency, amplitude, reflections and delay to modify the way we feel it, and it can be analyzed objectively in order to have the sound intended for each purpose. This constant interest that I have had for a long time ago was the reason that motivated me the most to propose this Master Thesis.

Mainly what we wanted to do with this project was trying to approach two different disciplines that should not be that far away from each other: the music from an artistic and subjective point of view, represented in the musicians' interest to express feelings, and the physics of the sounds, the technology and acoustics from a more rational point of view, represented in recording room design.

Musical acoustics can be considered an art as well as a science. The science lies in applying acoustical attributes derived from measures of physical properties. The art lies in judging the attributes that are still unmeasurable. The art of the music and the science of the sound must fuse, since the experience of the music and the musician will vary according to the acoustic properties of the room (Beranek, 1996).

By analyzing and correlating these features we hope to identify the musicians' opinion and what they are expecting to get out of their musical experience. We want to extract as much information as we can about the needs of the musicians, based on a rigorous analysis as most objective as it can be with such a subjective issue. We want to learn about which are the best or the worst rooms for each of the analyzed genres, and why does this happen.

This project is also interesting for the ESMUC (Escola Superior de Música de Catalunya) to have an objective and deep analysis of their rooms and the preferences and opinions of some musicians about them. This analysis will include measurements of the reverberation time of their rooms, the transfer function of each one of them and the instantaneous frequency response. Also will include recordings of the same song in each one of these rooms, and an analysis of surveys made to the musicians. With this master thesis they could detect how is the behavior for the different rooms according to different genres and purposes, and use them properly for a best performance of the musicians.

## 1.2 Goals and objectives

### *1.2.1 Main Goal*

The main goal of this project is to correlate and compare the subjective opinion of experienced musicians about the ideal room acoustics for acoustic pop/folk music with the objective characteristics and properties of the recording sound.

### *1.2.2 Measurable Objectives*

1. To measure the reverberation time RT60 of each room, and compare it with the theoretical reverberation time given by the architects of the ESMUC. Measure the transfer function and frequency response of the different rooms.

2. To record two different pieces of music in the four different rooms with different acoustic treatment.

3. To make surveys with experimented musicians with the goal of analyzing the subjective opinion about the different acoustic treatment and the existing preferences.

4. To analyze the recorded sounds and the objective differences between the rooms.

5. To make a deep analysis with the preferences and the objective characteristics of the sounds for each genre. This will be done by correlating and comparing the objective characteristics of the sound with the subjective preferences of the musicians.

## 1.3 Structure of this document

The structure of this document has one chapter dedicated to each phase of the thesis. First chapter states about the motivation of the project, the goals, what do we expect to get in the end and how are we going to proceed with the work. The second chapter presents the state of the art on room acoustics describing previous studies, and how are important as fundaments and starting points in the development of this thesis.

The third chapter is related to the procedure followed to develop this study. It defines and describes the methodology used to make the measurements, to record the songs and the surveys to get the subjective opinion of the musicians.

The fourth chapter makes the analysis of the results, and the correlation we are expecting between the objective analysis and measurements, and the subjective opinions and preferences of the musicians.

The fifth chapter talks about the conclusions reached from the analysis of the data and the connection with the information. It also presents the findings that arise from this study, and a discussion about future work and the research questions that derive form it.

Bibliography and references used for the development of this project are listed in chapter six. The appendixes are presented later.

# Chapter 2: State of the art

We perceive different acoustic treatment in all the rooms we visit everyday in our lives. The living room, our bedrooms, the stairs, the classrooms, and the different environments we occupy everyday have different acoustic features that most of the times we don't even notice. Few people actually know the reasons behind different acoustics, but most of the people are able to judge if the acoustics of a place are good or bad depending on its architectural purpose.

In his book *Room Acoustics*, Heinrich Kuttruff compares concert halls with large musical instruments. The shape and the material used in the construction of both are very important to determine their sound. The sound generation, definition of pitches, timbre and resonances are physical processes that can be analyzed objectively and rationally. Nothing is a matter of magic, and what happens can be explained with mathematic equations (Kuttruff, 2000).

The difficult part when analyzing room acoustics, and what makes it different from other technical disciplines, is that it is very important to take into account the subjective impressions of the musician, and how the materials, dimensions and sound field affect them. It can be stated that the sound in a real room cannot be a matter of exact mathematic treatment because particular subjective sensations could affect the results. In this area is where we want to focus our project: How these subjective sensations are involved in the rooms analyzed, related to the measurements and recordings.

The purpose of each room is also very important when we are going to analyze its acoustics. Rooms may have different acoustic treatment if they are made for speeches, films, orchestras, chorus, pipe organs, rehearsal of bands, live concerts or even if made for sports games. In our case, we will study four different rooms that are used daily for different purposes in the ESMUC: the organ hall, the chorus chamber, a rehearsal room for bands and the recording studio. It is interesting to analyze also how they sound according to their purpose.

As it is a very important subject, it has been studied with thoroughness across many years to develop new information and knowledge. Important disciplines of the physics are dedicated to study the sound waves and how they behave to have different acoustics, and it has become a science during the past century. We can find much information about room acoustics in different books and papers.

## 2.1 Objective terms: the hall

### *2.1.1 Reverberation time*

The reverberation can be defined as the “hanging on” of a sound when the exciting signal is removed. It is an important term when we are going to analyze the quality of a room. It occurs because of the reflections of the sound in the room, and they slowly decay as the air and the walls absorb the sound. A room with very low reverberation, such as anechoic chambers, sounds like *dead* or *very dry,* and music does not feel natural. Rooms with a lot of reverberation affect the intelligibility of the sound.

Reverberation adds richness and supports musical sounds. It also helps to integrate all the sounds from an instrument so the listener hears them all incorporated, including the directional parts of it. For this reason, in rooms with low reverberation it is difficult to play music because you don’t have a proper feedback (Angus, 2001).

The reverberation time is defined as the time that is required for the sound pressure level in a room to decay 60dB. This represents a change of sound pressure level of 1000 (because 20*log(1000)=60dB). The value of 60dB was selected from the psychophysical point of view, because is in that level where the sound is perceived as inaudible (Everest, 2001).

Wallace Clement Sabine made several experiments in the 19th century to prove the effects of absorption in the reverberation time. He measured the reverberation of the source to inaudibility, by using organ pipes, a stopwatch and his ear, and he found that the reverberation time is directly proportional to the dimensions of the room and inversely proportional to the absortion of it (Lloyd, 1970). Then, he empirically developed the following equation.

$$RT_{60} = -\frac{24 \cdot V \cdot \ln 10}{c \cdot S \cdot \ln(1-\overline{\alpha})}$$

**Equation 1: Sabine's reverberation time**

Where V is the volume of the room, c is the speed of sound, S is the total surface of the room and $\overline{\alpha}$ is defined as the absorption coefficients of the various portions of the wall, averaged arithmetically using the respective areas as weighting factors, as can be observed in Equation 2.

$$\overline{\alpha} = \frac{1}{S}(S_1 \cdot \alpha_1 + S_2 \cdot \alpha_2 ...) = \frac{1}{S}\sum S_i \cdot \alpha_i$$

**Equation 2: Absorption Coefficients.**

Where $\alpha_i$ are the different absorption coefficients and $S_i$ its corresponding surface in the room. The absorption coefficients generally change depending on the frequency, and that is the reason why reverberation time is commonly evaluated for the different bands of frequency. Most materials absorb less energy in the lower frequency ranges; hence they have more reverb in them.

If we insert the numerical values of the speed of sound, and with some mathematical treatment, we can get the final expression shown in Equation 3. The 4mV term is added to take into account the air absorption of the sound; some times it can be omitted (Kuttruff, 2000).

$$RT_{60} = 0.163\frac{V}{S \cdot \overline{\alpha} + 4mV}$$

**Equation 3: Final Expression for Sabine's Reverberation Time**

### *2.1.2 Impulse response and Transfer Function*

The impulse response, as its name says, is the response of certain system due to an impulsive stimulus. In the acoustical field, the impulse response of an environment is its behavior under an impulsive sound. An acoustic impulse is considered as a short sound with arbitrary shape and duration less than 50ms. Normally it is plotted in the time domain, showing the changes of the signal as it passes through the system.

An impulse response of an acoustic environment provides a lot of information about it such as its propagation delay, reflections reverberation and decay. It yields a complete description of the changes of a sound that travels through an acoustic environment. Due to these reasons, the experimental measurement of the impulse response of a room is one of the most important tasks in room acoustics. It must be done with high quality standards for all the measuring components, trying to guarantee the most objective measurement as possible (Kuttruff, 2000).

In the other hand, the transfer function of a system compares a reference signal with a measured one, normally the input and the output of the system under analysis. Usually is defined as the Laplace transform to the impulse response (Mikio Tohyama, 1995). The transfer function is always calculated in the frequency domain, with the FFT data.

It can be plotted both in frequency or time domains though. A graphic representation of this can be seen in Figure 1.

For this thesis we are going to plot the transfer function in the frequency domain, to see the frequency response of the room in magnitude and phase. Also we will watch the **coherence plot,** that represents a complex mathematical function calculated by Smaart®, that represents the coherence between two signals. Smaart displays this function in values from 0 to 100%. When nonlinear issues appear between the two signals, the coherence decreases. Also when there is a delay between the two signals, insufficient energy in the reference or acoustical issues like reverberations, reflections or noise (Calvert Dayton, 2007).

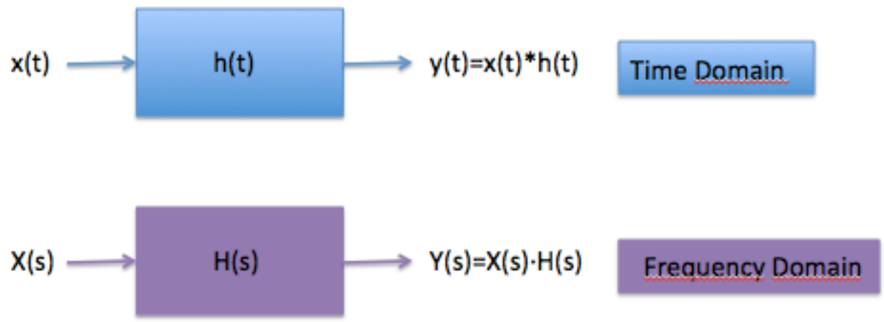


**Figure 1: Impulse response h(t) and Transfer function H(s) in a system**

### *2.1.3 Real Time Analysis*

The real time analysis, also called RTA, shows different audible frequency ranges and displays the energy that is present in them. This function works as a real time Fast Fourier Transform analyzer for the spectrum of the captured sound. In the y-axis the display plots the magnitude of energy, and in the x-axis it plots the different bands of frequencies organized in fractional octave bands (Calvert Dayton, 2007).

The real time analyzer is implemented in the software Smaart v.6 ® that we used for the measurement process of this project. When the function is activated, the software uses the analog to digital (A/D) converter of the sound card, and starts to transform continuously the time domain signal captured by the microphone, in a frequency domain spectrum using a fast Fourier transform (FFT). The information given by the FFT process is plotted in real time, and that is the visualization that the software displays to the user.

The measures of the real time analysis depend directly on the position of the loudspeaker and the microphone used because the reflections of the room captured by the microphone are not the same if it moves, and hence the instantaneous frequency response changes. This is important to take into account to have objective measurements.

### *2.1.4 Measurement techniques*

Measurements in room acoustics are necessary because they allow us to determine the objective factors that influence the subjective impressions of the acoustics. In this way we can increase our knowledge about the qualities of the room and give useful information for the design of large halls.

During the past decade, the analog systems used before for measurements are being replaced by digital components, especially computers connected to analog to digital converters, transient recorders, among others. Nevertheless, some of these analog systems are still used extensively, for example the transducers such as loudspeakers or microphones. The computers allow us to make more powerful, precise and flexible measurements, while they are cheaper than traditional analogue equipment.

To measure reverberation, W.C. Sabine used modest equipment: he excited the room with a few organ pipes, and then measured the reverberation using his ear and a stopwatch. But after the 1920s all the measurements became electrical, replacing the ear for a microphone, and the pipes for a loudspeaker. The fall in the sound level was observed with electromechanical level recorders such as oscilloscopes (Kuttruff, 2000).

Actually there are two main techniques that can be used to measure reverberation. The first one is called Interruption method, and consists in a generation of white or pink noise, and a loudspeaker ideally omnidirectional and with flat frequency response for the bands of analysis. The room is excited for some seconds with the noise at a level enough to overcome the background noise, and then the sound is interrupted abruptly. The recording continues until the sound completly falls down. The main advantage of this method is that it gives the capability to control which bands of frequency are being excited and the recording levels to avoid the saturation.

The other method is called Impulse Noise method. It consists in the emision of an impulse sound able to excite all the frequencies in the room, with short duration but large amplitude. The most convenient way to make this impulse sound is to shoot a pistol, since it is easy to operate and it sounds loud enough to overcome any background noise. The pistol can also be replaced for bursting air balloons or wooden hand clappers, since they yield high excitation especiallly at low frequencies (Guaus, 2011).

No particular requirements concerning the uniformity of sound radiation are needed to measure reverberation, since various components will be mixed during the decay process.

## 2.2 Subjective terms: The music and the musician

In the area of acoustics, we can't be satisfied with the objective measurements of the rooms and with the reduction of the reverberation. In the end, the final consumer of the acoustics is the listener, and his opinion is an important variable to take into account. He might be the one who wants to enjoy a concert, attend a lecture or a theatre performance in a given hall. The listener expects the room and its acoustics to support the music or to render the speech easily intelligible.

In the particular case of our project, the subjective impressions of the musicians are the variables we are going to analyze, and how they are related to the measurements we took for each of the rooms, with the instrument they were playing and with each genre. The perceptual terms we used to evaluate the subjective impressions of the musicians will be explained along this section.

Then, we have to pay attention to the properties of the sound field that are related to the hearing impressions. How a particular reflection is perceived and which reverberation is preferred for each performance and for each instrument according to the musician's opinions. Considering all this variables, we enter into the realm of psychoacoustics, and combine it with the physical information we presented in section 2.1.

Basically, the methods used to evaluate the subjective perceptions of people are the experiments according to what it is intended to evaluate. Testing the opinions of the listeners and processing that information gives us a hint on their preferences. In our case, we tested the musicians after they played in each environment and evaluated how they felt in that room, and their perception about some basic terms.

In the beginning, reverberation was the only variable that was taken into account in the measurement of room acoustics. It was introduced by W.C. Sabine in the $19^{th}$ century, as was explained in section 2.1.1. It refers to the decay of the signal, and it is still considered as the most important objective quantity in room acoustics. Anyhow, it has been proved nowadays that the early reflections of the signal are also very important for the human perception, because they give the coloration and the timbre of the sound (Kuttruff, 2000). The effects of reflections will be explained more in detail in the section 2.2.3.

### *2.2.1 Survey design to determine psychoacoustical parameters*

To evaluate the perceptual quality of a hall, the most accurate way is to choose some acoustical attributes, explain them to the listeners, and test them by giving rating points and using scales.

There have been several studies that identify and expand a set of attributes that can be measured by listeners in a concert hall. Those attributes change from study to study, new attributes appear and some disappear in an effort of trying to understand better the variables that affect perception.

One of the most remarkable studies was the one that Barron made in 1988. He performed some listening experiments using 27 experimented listeners (most of whom were acoustical consultants). They went to concerts in 11 different concert halls across the United Kingdom, and were changing the sits during the intermission. When the intermission was finished, they filled a questionnaire shown in the Figure 2, and the overall impressions of each one of the halls was correlated closely to the scales of reverberance, envelopment and intimacy. There were high similitudes between the reverberance and the envelopment, and between the envelopment and the intimacy, but low correlation between the reverberance and the intimacy (intimacy is explained in detail in section 2.2.2).

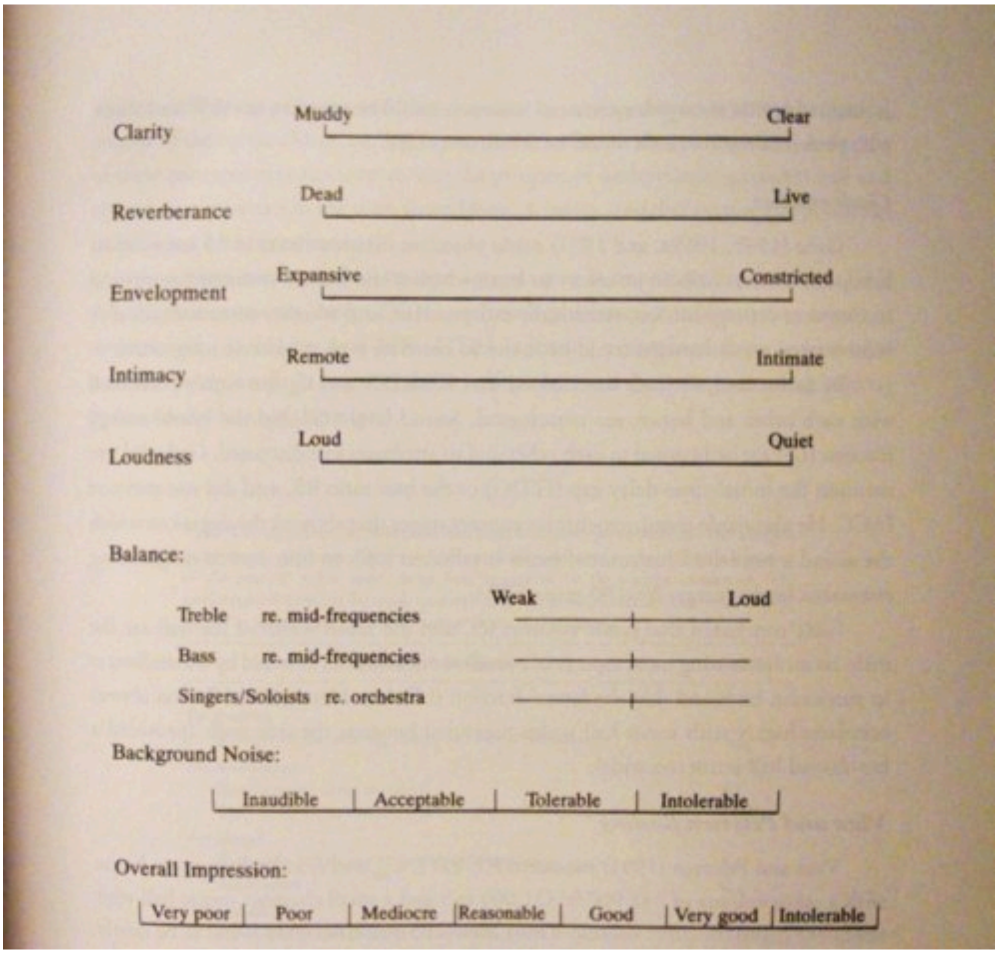


**Figure 2: Subjective questionnaire used by Barron in 1988 (Beranek, 1996)**

The listeners were divided in two groups according to their opinions: those who preferred intimacy and those who preferred reverberance, and they seemed to judge the envelopment differently, associating it with reverberance or with intimacy, as a problem of semantics and misunderstanding the instruction.

All this experiments show a close relationship between the measured and the judged attributes. However it is important to avoid the different interpretations of the attributes, and to take into account which music the listener is used to hear and how it is related to their judgment (Beranek, 1996).

The recommendations of the ITU-R BS.1284-1 for testing the audio quality in acoustic rooms, can be found in Appendix A: ITU-R BS.1284-1 Recommendations.

### *2.2.2 Perceptual Terms*

**Clarity:** it is defined as the degree to which discrete sounds stand apart from each other. Usually, clarity depends on the intention of the performers, the musical factors and the skills, but it is also closely related to the acoustic of the rooms.

The physical measurement of the clarity is the ratio of the total energy in the early sound, to that in the reverberant sound. The early sound is defined as the one that is heard in the first 80ms after the arrival of the direct sound, and reverberant sound is the one that is listened afterwards. Clarity is designated by $C_{80}$ and measured in decibels.

Different amounts of clarity are preferred for different situations. For rehearsals, more clarity is desirable, as long as for live concerts, more reverberation is preferred. The listeners according to their musical experience can judge clarity qualitatively. It is easy to decide whether if the music is to clear or if the reverberation is too strong, or when the balance between the early and the reverberant sounds is not good enough (Beranek, 1996).

**Intimacy:** A hall has acoustical intimacy when the music gives the impression of being played in a small place. It is also called “presence”. In 1988, Barron coined a definition for intimacy saying “Intimacy refers to the degree of identification between the listener and the performance, whether the listener feels acoustically involved or detached from the music”.

The intimacy is determined by the initial-time delay gap. The initial-time delay gap is the time between the direct sound and the first reflection. The shorter the initial-time delay gap, the more intimate the hall is. How ever, this must be treated carefully because if the initial-time delay gap is very short (tending to zero), the intelligibility of the sound is affected. In large concert halls, usually the musical quality is improved by adding horizontal suspended panels to shorten this initial-time delay gap and make them feel smaller than they are (Beranek, 1996).

**Reverberance or liveness:** as it was explained in the section 2.1, reverberation refers to a sound that persists in a room after a tone is suddenly stopped. A hall that is reverberant is also called a “live” hall. A room with short reverberation is also called a “dead” or “dry” hall.

Liveness is related to the reverberation times at middle and high frequencies. A room can sound live and still be deficient in the reverberation of the low frequencies. In that case, the hall can be called “warm” (Beranek, 1996).

The effects on reflections and reverberation will be explained in detail at section 2.2.3.

**Envelopment:** It describes the listener impression of the directions and the strength from which the reverberant sound arrives. The envelopment is highest when the reverberant sound seems to arrive equally from all the directions, and lowest if it is very directional (Beranek, 1996).

It is also called Listener envelopment o LEV, and will be explained in detail in the section 2.2.3.

**Loudness:** As Beranek says in his book of 1996, loudness hardly needs a definition. It states how intense the sound is felt in that room. A room is louder if the room is reverberant than if it is dry. Three architectural factors affect the loudness: the distance between the listener and the stage, the reflecting surfaces that send early sound energy to the audience and the area where the audience sits over (Beranek, 1996).

**Balance:** The balance evaluates the loudness for different bands of frequency. A concert hall with good balance is when the different sections of the orchestra are felt equally loud. Balance can be achieved if the surfaces located near to the players help to emphasize certain sections of the orchestra, and also when the conductor knows the stage, and plays with the location of the musicians (Beranek, 1996). Usually people prefer a balance with more loudness in the lower and higher frequencies, and less in the middle, to compensate the perception of the ears according to the Fletcher and Munson curves for equal loudness.

**Background Noise:** The background noise in a hall is the sound that can be listened when there are not sound sources active reproducing sound (i.e. an instrument or a loudspeaker). It can be produced by the air handling system, electric noise of the lightning, the audience or errors in the acoustic isolation of the hall. It is marked as better if it is inaudible (Beranek, 1996).

### *2.2.3 THE EFFECTS OF REFLECTIONS IN A ROOM*

To understand the perceptual perception of the music according to the room where it is played, it is important to understand some definitions about the effects of reflections in a room.

Let us suppose we are seated in an anechoic chamber, or close to an instrument in an open field, where we perceive no reflections. When the musician plays a note in the instrument, we perceive the sound precisely as the instrument has produced it[1]. There is no enclosed space, and qualities as loudness, the decay and the timbre are not affected for the reflections, and what we hear is only the **direct sound**. In an acoustic room or a concert hall what we call the direct sound is the sound that first reaches the listener, before any reflections from the walls or the ceiling (Beranek, 1996).

[1] It is only affected by the air and its properties. The speed of sound can change because of the temperature or the humidity.

The **early sound** is considered as the direct sound plus those reflections that arrive after 80ms of the direct sound. Those reflections that arrive after the 80ms of the direct sound are considered the reverberant sound, which is created for many reflections that occur subsequently.

The main factors about reflections have to do with where do they come from and when do they arrive (early or late). If they arrive from the lateral directions of the listener, they appear to broaden the source and increase the **apparent source width (ASW).** An increase in the ASW gives more quality to the music heard in that concert hall, so, it is important to direct these reflections to the listener (Beranek, 1996).

Another important attribute perceived because of the reflections is the **listener envelopment (LEV).** It is defined as the sense of being surrounded by different sound images that are not associated with particular sound locations. They are associated with the late reflections (those that arrive after 80ms of the direct sound). The reverberant sounds from the sides of the listener are more important to define a strong LEV.

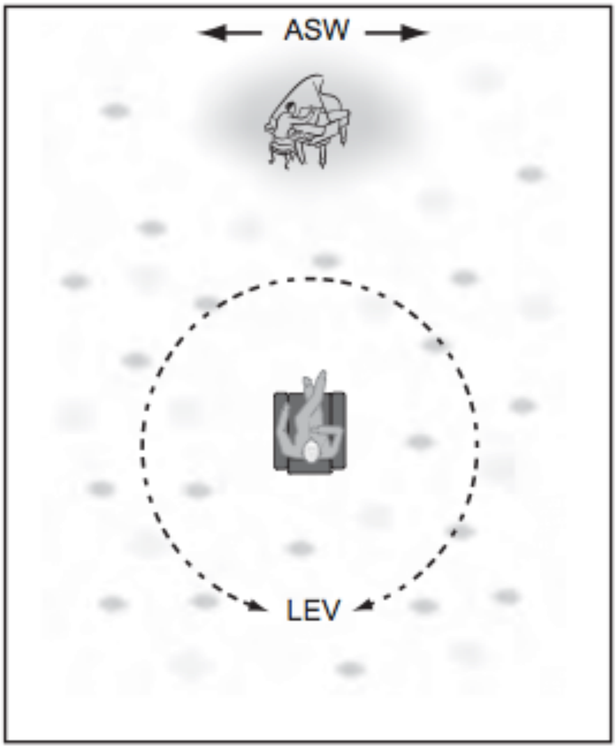


**Figure 3: Illustration of ASW and LEV in a room. (Toole, 2008)**

An illustration of what ASW and LEV are can be observed in Figure 3. Usually, LEV and ASW coexist in the rooms, in proportions related to the specific acoustics (Toole, 2008). In our particular case, we evaluated the perception of the LEV under the name of "envelopment" as we defined in section 2.2.2.

When we listen to music generally our ears are less sensitive to reflections than when we listen to speech. The lateral reflections help the music to feel more width, as it surrounds the listener. The reflections of the ceiling are masked by the direct sound more effectively than lateral reflections.

A reflection can be perceived without reaching the consciousness of the listener. At low levels it only increases the loudness of the signal, changes the timbre or increases the apparent size of the sound source. But at a higher level a reflection can be considered as a separate event with the repetition of the direct sound. The common name for this repetition is an **echo,** which is very unpleasant in concert halls because distracts the attention of the listeners. It reduces the intelligibility of music and speech due to the subsequent sounds mixed up.

Echoes commonly are heard outdoors, because indoors they are commonly masked by the general reverberation of the room. A reflection can be considered as an echo depending on its delay respect to the direct sound, its relative strength, and the absence of other reflections that could mask it. Usually the delay between both sounds varies around 100ms for dry sounds, until some seconds for complex sounds (Kuttruff, 2000).

# Chapter 3: Methodology and procedure

In general, the methodology of the project can be observed in Figure 4. The methodology for measurement, recording and surveys are explained across this chapter. The objective analysis and correlation is explained in detail in chapter 4.

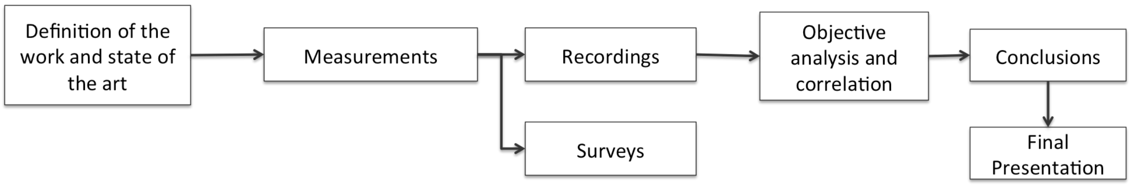


**Figure 4: Methodology and procedure of the project**

## 3.1 Rooms measurement and Characterization

### *3.1.1 Equipment*

**Loudspeaker:** To do the measurements we chose the loudspeaker L-Acoustics 112-P. It is a portable loudspeaker powered by a 1000W power amplifier with dedicated on-board digital signal processing. It includes four different presets with different frequency responses to make it flexible to many applications. For this purposes, we chose the FILL preset, which has a nominally flat frequency response that can be observed in Figure 6 (L-Acoustics, 2011). We can conclude from the figure that the frequency response can be considered flat from 100Hz to 20kHz. It is important to take into account which frequencies we can trust when we measure the reverberation time or the transfer function using this loudspeaker.

**Figure 5: Amplifier L-Acoustics 112-P (L-Acoustics, 2011).**

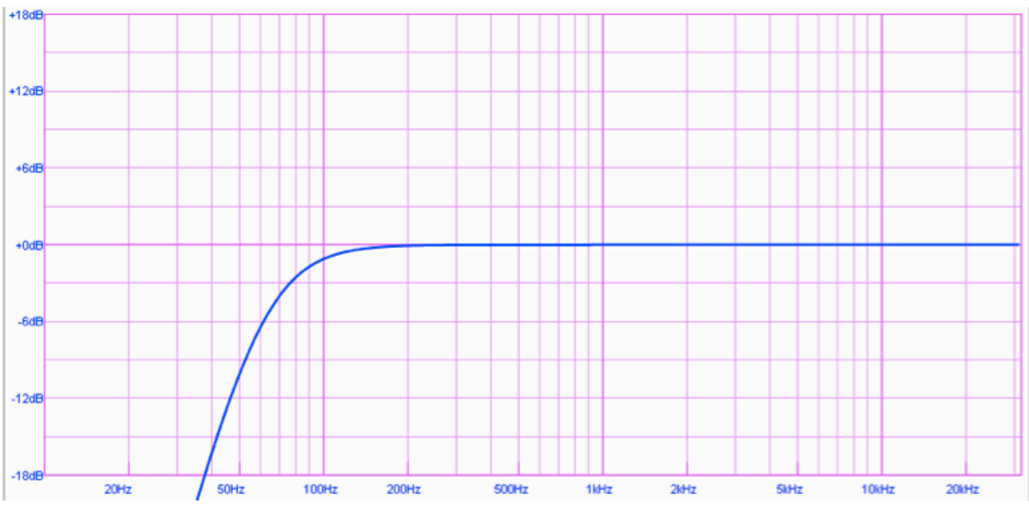


**Figure 6: Frequency Response of L-Acoustics 112P (L-Acoustics, 2011).**

**Microphone:** For performing the measurements we used a Behringer ECM8000 measurement microphone. This is a condenser microphone with a linear behavior, flat frequency response and ultra-high sound resolution. This microphone is especially designed to measure room acoustics with real time analyzers, to provide an accurate acoustic picture of the room. It has an omnidirectional polar pattern and a flat frequency response from 15Hz to 20kHz. A picture of the microphone can be observed in Figure 7 (Red Chip Company Ltd. , 2011).


**Figure 7: Behringer ECM8000 microphone (Red Chip Company Ltd. , 2011).**

**Sound Card:** To connect the microphone to the computer and record the response of the room, we used a sound card TASCAM US 122. It has two XLR-based microphone inputs with phantom power. Two lineal 1/4'' TRS inputs are also included, and are switchable for guitar input level or microphone input level. This sound card also provides a dedicated control for adjustable zero-latency, two level outputs and a dedicated headphone output (TASCAM TEAC Professional, 2011). It can be observed in Figure 8.

**Figure 8: Sound Card TASCAM US-122**

In order to have more neutral and objective measures, we wanted to prove that the system was flat itself, and we were not adding a particular frequency response with the sound card. For this reason we generated some pink noise and measured the reference by connecting directly an output to an input.

At the beginning, we started using the sound card M-Audio Mobile Pre, but we noticed that its frequency response was not flat, so we were not having the RTA we were expecting. In Figure 9 we can see both responses, with the sound card M-Audio Mobile Pre (up) and the TASCAM US-122 (down). The blue signal represents the reference, connected from the output to the input. The green signal represents the response of the room measured with the ECM8000 microphone.

It can be observed how the first sound card added color to the pink noise, while in the second one we have a flat response in the reference, and hence a more trustable measured signal. A “smiley” frequency response is obtained with the M-Audio Mobile Pre card, which may be tailored for musical purposes but must be discarded for measurement tasks.

**Audacity:** Audacity is a free open-source software to record and edit audio signals. Several volunteers around the world develop it thanks to SourceForge.net. It is available for Windows, Mac OS X and Linux. It can be downloaded at http://audacity.softonic.com/.

In this project we used Audacity to generate the white noise used to measure the reverberation time, and to record the response of the room to a ballon pop and to the white noise. It was also used later to record the different musicians playing in the different rooms.

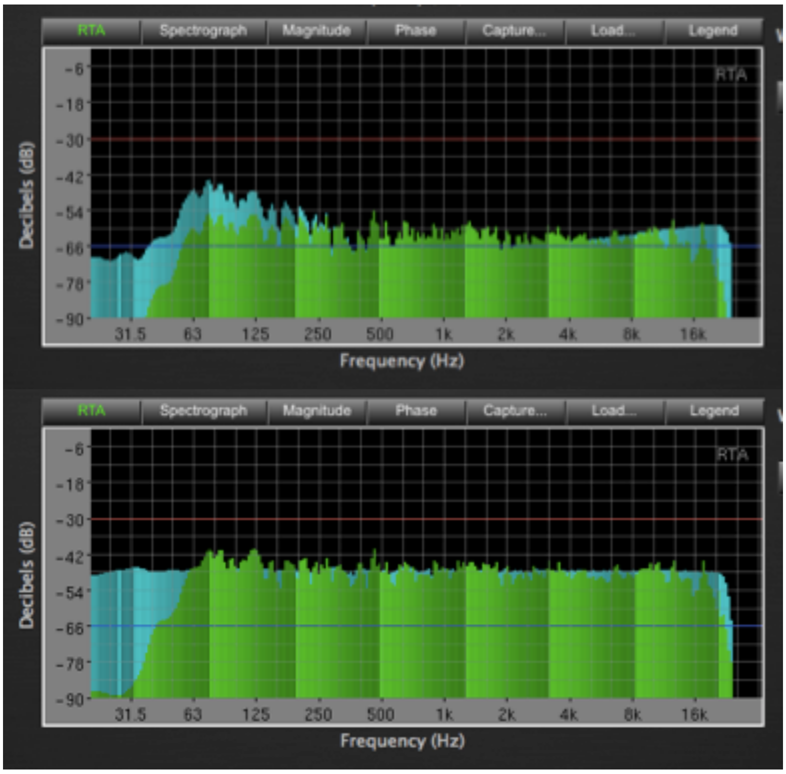


**Figure 9: Instantaneous frequency response for pink noise, using M-Audio Mobile Pre (up) and Tascam US-122 (down)**

**Smaart:** It is a software developed by EAW Software Company® Inc. for real time sound system measurement, optimization and control. It performs dual channel FFT based measurements with an intuitive interface (Calvert Dayton, 2007). Using this software we perform the measurement of the Real Time Analysis of the room, the spectrogram, and the transfer function in magnitude and phase.

**Matlab:** Matlab® is a high-level software developed by The MathWorks Inc., that allows to perform complex computations in a faster way than with other programming languages (The MathWorks Inc. , 2011). In the project we use this software mainly to calculate the reverberation time by using the Reverberation_Time_Calculator tool, and its function reverb_time. In the following section we will explain in detail how the code works.

### *3.1.2 Procedure*

To perform the measurements of the four different rooms in the most neutral and objective way possible, we followed a strict procedure that can be observed in **Figure 10**. It is important to guarantee that the measurements are taken with objective criteria and repeating the same procedure in the different rooms, in order to get trustable results that allows us to make conclusions out of them. In the present section we will explain in detail each one of the blocks on **Figure 10**.

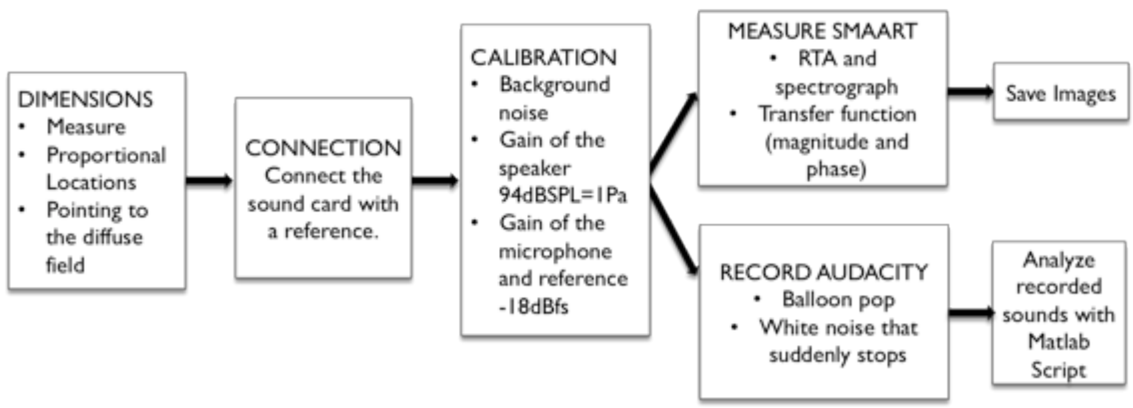


**Figure 10: Block diagram of the measurement procedure**

The first step in all the rooms was to measure the **dimensions** of each one of them, and then decide proportional locations for the microphone ECM8000, and the Loudspeaker 112P. The idea was to keep them equally separated from the walls and from each other. As a rule, for all the rooms, we pointed the microphone and the loudspeaker to the diffuse field, so we captured more the reflections of the room and less the direct sound. The diffuse field can be understood as the sound coming from everywhere simultaneously, or from different directions in succession with no time in-between their arrival (Martin, 2006). Typically, we think of reverberation as a diffuse field, as it bounces around the room for some seconds. So, we picked those positions for the microphone and the loudspeaker in an attempt for capturing more the response of the room and trying to avoid the direct sound.

Then we **connected** the TASCAM US-122 sound card to the computer. In the output1 we connected the loudspeaker 112P, which we used to reproduce the noise. In the input1 we connected the microphone Behringer® ECM8000 we used to measure the response of the rooms to that noise. The output2 was directly connected to the input2 in order to have a reference signal, as can be observed in Figure 11.

In our particular case the microphone preamplifier was included in the sound card, and the power amplifier was included in the loudspeaker. We connected the right output of the soundcard directly to the right input, to have a clan reference to compare, without the influence of the room.

Then, we started to do the calibration of the equipment. In the first place we measured the background noise in dBSPL to have a reference. Then we set the gain of the loudspeaker to 94dBSPL, in order to have the same reference in all the rooms. We chose that value because it is the standard of 1Pa. Using the software Smaart® we set the gain of the measured signal and the reference to -18dBfs (full-scale decibels), to have it equal for both signals and to be able to compare the two plots.

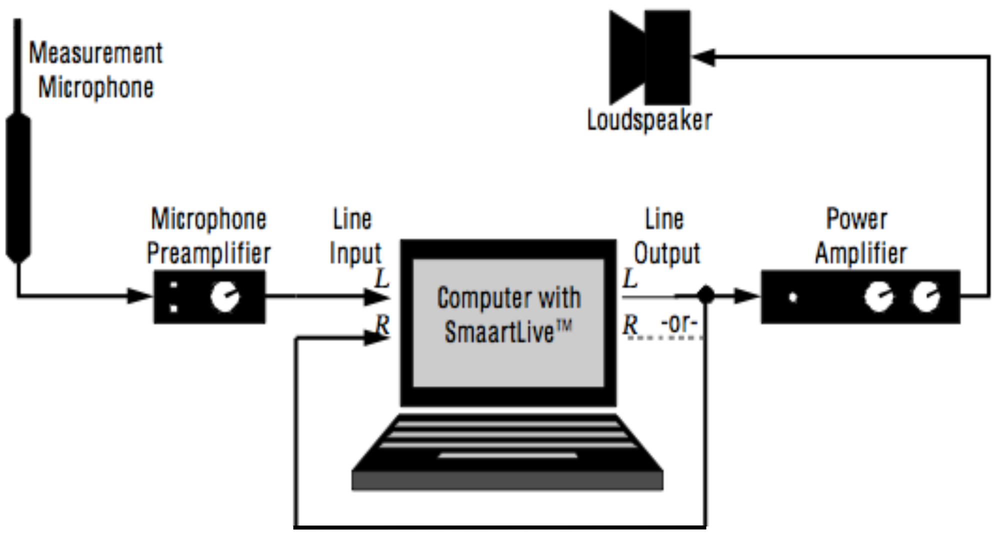


**Figure 11: Microphone, loudspeaker and reference connections (Henderson, 2004).**

Then we started to measure by using the Smaart® software. First we measure the Real Time Analysis curve and the Spectrogram curve and saved the images. As we said in the definition of Real Time Analysis, it is important here to take into account the position of the loudspeaker and the microphone, because as they change, the RTA also changes.

The Spectrogram and the Real Time analysis curves are useful to see the behavior of that room with those locations of the equipment, but we are not able to compare the rooms with these measurements.

However, we can compare the different rooms by using the transfer function of them, because it compares the measured signal with the reference and it is not dependent on the position of the equipment. We measure them also using the Smaart® software and saving the images.

Then, we measured the Reverberation Time using the two different methods explained in section 2.1: balloon pop and white noise suddenly stopped. The position of the balloon was the same position of the loudspeaker. We recorded both responses of the room with the ECM8000 microphone, and then analyzed them using the Reverberation_Time_Calculator Matlab® tool.

The Reverberation_Time_Calculator tool includes a function called reverb_time which we used to do the calculation. It supports the two different methods we wanted to use, they recommend the speaker on-speaker off method though.

The reverb_time function calculates the reverberation time for 1/3 octave broad band noise for multiple microphones, so we used a script that runs it for different bands of frequency, prints the results in the command window and in the end plots the reverberation time versus frequency.

We ran this script for each one of the rooms with balloon pop and with the white noise suddenly stopped, and saved the results which will be compared and discussed in section 4.1. The source code can be found in the Appendix D: Matlab® source codes.

## 3.2 Recording in the different rooms

In this phase of the project we recorded two different bands playing acoustic music in each one of the four rooms at the ESMUC. The first one was a guitar player alone with a voice, who played two different cover songs in acoustic versions: “Volver” by Carlos Gardel and “Nowhere man” by The Beatles. The other one was a band called Kibo, composed by four musicians, including: guitar and voice, cello, bass and a snare as a portable percussion. They played a song of their own called Equilibri that can be listened at their MySpace site http://www.myspace.com/kibomusica.

The idea is to record them in each one of the rooms, and analyze their opinions about the hall, its properties and how they improve or worsen the songs. To do this we made a survey that is explained in detail in section 3.3. The rooms are explained in detail in chapter 4.

The recordings were made only with the purpose of having a file that supports the opinions of the musicians while music is played in the rooms, related to the results of the measurements. The purpose of these recordings is not to do them in an artistic or commercial way, in the other hand we want to capture the music, as close as possible to the way the musicians perceive it, and also how the properties of the room influence the music. In this section we will explain the methodology we used in order to have those results.

### *3.2.1 Equipment*

**Microphones:** As we explained before, the purpose of these recordings is not to do an artistic mix for selling, but to capture the sound as the musician is listening to it, and to capture the influence of the room acoustics in the song. In order to have those results, we selected two different stereo microphone techniques, one for each purpose.

To record the song as the musician perceives it; we selected the ORTF technique developed by the Office de Radiodiffusion Télévision Française (ORTF) around the 1960s. It was applied by using a MSTC 64 U microphone from SCHOEPS Mikrofone Inc. We selected this technique because it simulates the way as the human being listens, with two cardioid microphones separated 17 centimeters from each other an in an angle of 110º. It has a T-shaped body with two built-in microphone amplifiers, and two MK 4 cardioid capsules. The stereophonic recording acceptance angle is 95º (SCHOEPS Mikrofone Inc. , 2011). It can be observed in the left side of Figure 12.

**Figure 12: Microphone techniques used. ORTF (left) and AB (right)**

To capture the sound affected by all the reflections of the room, we used an AB stereophonic microphone technique with two AKG C414 B-ULS microphones in the omnidirectional mode of operation. They were separated 50 centimeters from each other, to help with the decorrelation of the signal and avoid phase problems, and they were located at 1,70 meters over the floor. They can be observed in the right side of Figure 12. As they are omnidirectional, they are capturing as well the direct sound, as the reflections that come from the walls, the floor and the ceiling.

This is one of the most used condenser microphones in the world. It can be used for many purposes and many recording techniques due to its capability of selecting 4 different polar patterns. It has a smooth frequency response, low self-noise and a 126dB dynamic range (AKG Acoustics GMBH, 2011).

**Sound Card:** For the recordings we used the same sound card TASCAM US-122 explained in section 3.1.1. We have already proved that it is flat and it works very well for our purposes.

**Audacity:** To record the groups we used the software audacity explained in section 3.1.1. It is free, easy to use and worked very well for our purpose.

### *3.1.2 Procedure*

To perform the recordings, we also followed a strict procedure that was repeated for each one of the rooms. A block diagram of this procedure can be observed in Figure 13.

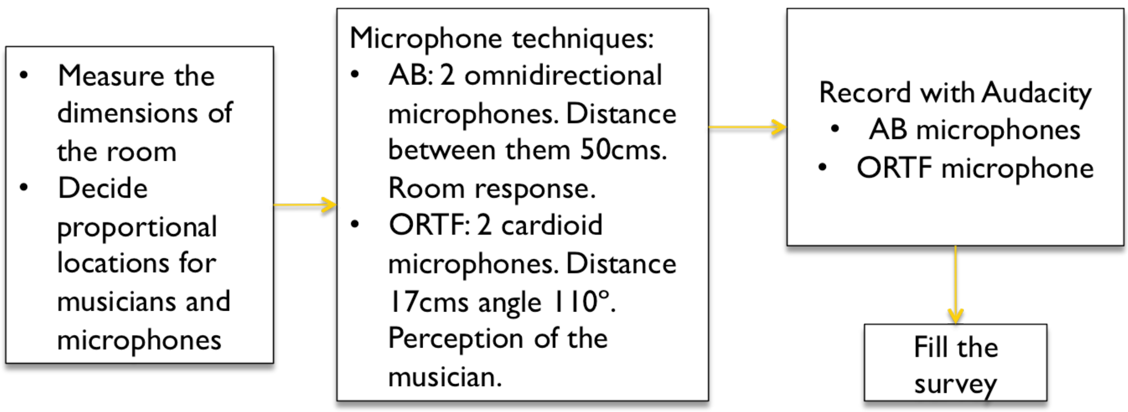


**Figure 13: Recording Methodology**

The first step consists set the locations for the microphones and the musicians in the room. We took into account the same dimensions used in the measuring phase, and located the AB configuration in the same position of the measuring microphone. Also we located the ORTF microphone approximately in the same position of the loudspeaker, and located the musicians near to it.

As can be observed in Figure 14 when we recorded the guitar player, the ORTF microphone was located above his head. When we recorded the band, this microphone was in the middle of them at 1,60 meters over the floor, simulating the height of the ears of an average human being.

**Figure 14: Illustration of microphone locations for the guitar player and for the band**

Then we connected the ORTF microphone arrangement to the Left and Right inputs of the TASCAM US-122 sound card, and then recorded the song using Audacity. After that, we connected the two microphones of the AB configuration to the same inputs and recorded the song again.

### 3.3 Subjective questionnaire

According to the previous questionnaires studied and the recommendations explained in section 2.2.1, we designed our own survey with the attributes we considered more important to test. Our test can be observed in the Appendix E: Surveys.

The listeners we chose for our project were the same musicians that played in the rooms and were recorded, because we are trying to analyze the preferences of them for each room. They were: a single musician who played guitar and voice, and a quartet of pop/folk music, including voice and guitar, cello, bass and percussion.

The tests were performed to the musicians after they played and were recorded in the room. We couldn't handle to make the surveys with sessions without interruption, or separated by periods of the same length, as was recommended by ITU-R BS.1284-1, because in our case we depended on the availability of the musicians or the rooms in the ESMUC. To avoid the problems in the comparision of the different rooms, we allowed the musicians to listen to the recordings again, or to read again the questions they filled for the other rooms, before they continue with the experiment. The results and the processing of the surveys data are discussed in the section 5.2.

The attributes we tested were clarity, reverberance, envelopment, loudness, background noise and balance, all explained with detail in the section 2.2.2. Also we evaluated the overall impression of the hall for different purposes: as a live stage, as a recording space or as a rehearsal room, according to the different attributes of the hall.

Also, we left a blank space for the musicians to explain the results, to allow them to give a subjective assessment of the room, and explain the ranking he gave for each one of the attributes. This was made by taking into account the fact that the people we are going to ask are all musicians, and as artists they can express better their feelings and perceptions in an open space than ranking with numbers.

Basically, we based our survey in the one developed by Barron in 1988, explained with detail in the section 2.2.1. We changed some of the attributes choosing those we found more relevant, we explained them, and also selected a scale from 1 to 5, taking into account the recommendations made by the ITU-R BS.1284-1[2]. In each survey the meanings of the scale for each attribute are explained, as can be observed in the Appendix E: Surveys. In the analysis of the results the scale was treated as a continuous value with one decimal point.

[2]Details of this recommendation can be seen in the Appendix A: ITU-R BS.1284-1 Recommendations

# Chapter 4: Description of the analyzed rooms

To analyze the differences in the acoustic of the rooms and the preferences of the musicians, we chose 4 halls in the ESMUC with different acoustic treatment: the chorus chamber, the organ hall, a rehearsal room and the recording room. These rooms and the theoretical data we had will be described in detail in the present chapter, before we analyze the results of the measurements.

## 4.1 Chorus Chamber

The Chorus Chamber of the ESMUC is the room A309, located in the third floor. It is a parallelepiped with the following characteristics:

- 9.2 meters long, 8.75 meters wide.
- Ceiling built with RPG diffusors
- The total volume of the room is 582,9$m^3$.

The RPG diffusors are commercial scattering surfaces, with standardized values to enhance the architectural acoustics with high performance and accurately documented acoustical palette in the industry. (RPG Diffusor systems, Inc. , 2000)

The theoretical data given by the architect for the RT60 of the room can be observed in Table 1, if the room is empty: no audience is considered.

Can be observed in the table that the curtains effect for absorption is evident after 500Hz. This is one of the most appreciated rooms in the ESMUC, and since its construction there have not been any acoustical changes. Curtains do not change the modal zone of the rooms.

Some pictures of the Chorus Chamber can be observed in Figure 15.

**Table 1: Theoretical reverberation time for the Chorus Chamber**

| **Frequency range (Hz)** | **125** | **250** | **500** | **1000** | **2000** | **4000** | $T_{LOW}$ | $T_{MID}$ | $T_{HIG}$ |
|---|---|---|---|---|---|---|---|---|---|
| Without Curtains | 1,03 | 1,22 | 1,61 | 1,76 | 1,65 | 1,54 | 1,12 | 1,68 | 1,60 |
| With Curtains | 1,03 | 0,92 | 0,77 | 0,79 | 0,75 | 0,75 | 0,97 | 0,78 | 0,75 |

**Figure 15: Chorus Chamber Images**

## 4.2 Organ Hall

The Organ Hall of the ESMUC is the room A311, located in the third floor. It is a parallelepiped also, but with the following characteristics:

- 9.24 meters long, 5.82 meters wide.
- Ceiling built with RPG diffusors
- The frontal wall is covered with marble slabs, to increase the RT60 of the room. This is the most important characteristic of this room.
- The total volume of the room is 367m$^3$.

**Table 2: Theoretical reverberation time for the organ hall**

| **Frequency Range (Hz)** | **125** | **250** | **500** | **1000** | **2000** | **4000** | **$T_{LOW}$** | **$T_{MID}$** | $T_{HIG}$ |
|---|---|---|---|---|---|---|---|---|---|
| Without curtains | 1,26 | 1,32 | 1,34 | 1,43 | 1,39 | 1,30 | 1,29 | 1,39 | 1,35 |
| With curtains | 1,04 | 1,16 | 0,89 | 0,86 | 0,82 | 0,81 | 1,10 | 0,87 | 0,81 |

The theoretical data given by the architect for the RT60 of the room can be observed in Table 2, if the room is empty: no audience is considered. Can be observed in the table that the curtains effect for absorption is evident after 500Hz.

Some pictures of the organ hall can be observed in Figure 16.

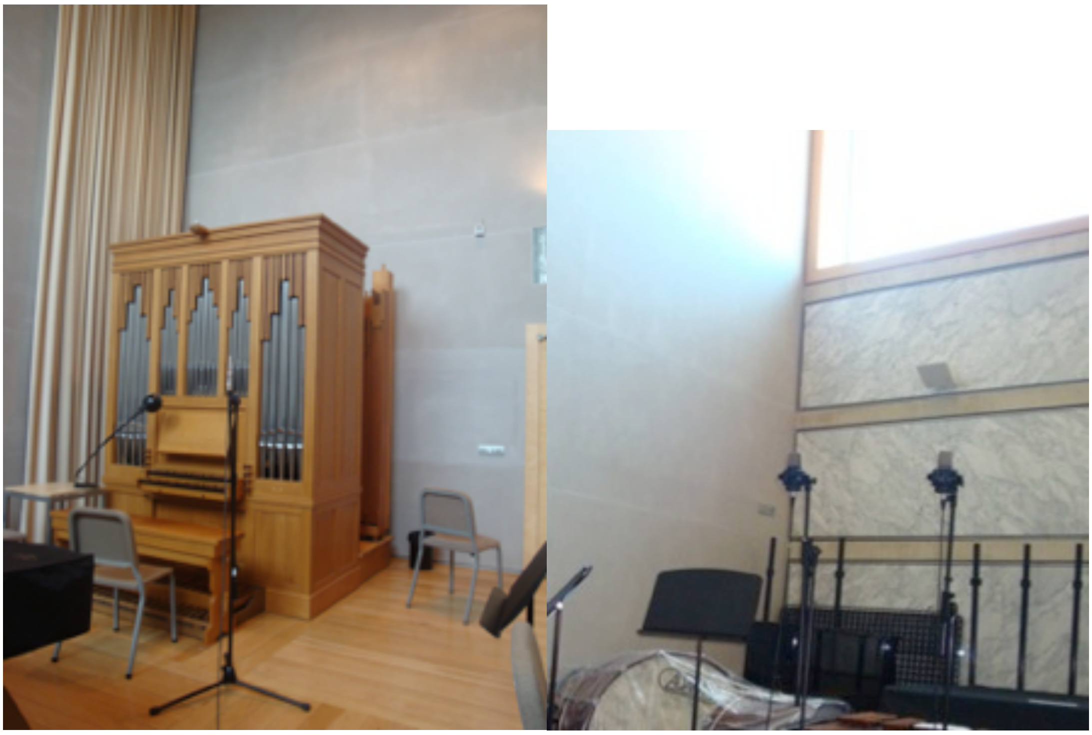

**Figure 16: Organ Hall Images**

## 4.3 Rehearsal Room

To study a rehearsal room of the ESMUC we chose the room A108, located in the first floor. It is a parallelepiped, with the following characteristics:

- 9.2 meters long, 5.9 meters wide.
- Total volume of 167.8 $m^3$
- The audience expected is 15 people.

**Table 3: Initial measurements of reverberation time for the Rehearsal room**

| **Frequency Range (Hz)** | **125** | **250** | **500** | **1000** | **2000** | **4000** | $T_{LOW}$ | $T_{MID}$ | $T_{HIG}$ |
|---|---|---|---|---|---|---|---|---|---|
| Without curtains | 0.86 | 0.88 | 1 | 1.15 | 1.03 | 1.15 | 0.87 | 1.08 | 1.09 |

The data given at

**Table 3** are the results of measurements made to this room before us, provided by the ESMUC. These measurements were made with white noise, and the room empty without curtains.

The room A108 is a parallelepiped with plywood walls, ceiling and floor.

Some pictures of this room can be observed in Figure 17

**Figure 17: Rehearsal Room Images**

## 4.4 Recording Room

The Recording room of the ESMUC is the room A124, located in the first floor. It is the cabin of the studio, and it is designated for recording musicians in it. It is a parallelepiped with the following characteristics:

- 5.8 meters long, 4.3 meters wide.
- Ceiling and walls built with diffusors and absorbers to get a dry sound.
- It has windows in one of the walls, and natural lightning.

The measurements of the reverberation time for each frequency range are shown in Table 4. Some pictures of the recording studio can be observed in Figure 18.

**Table 4: Theoretical reverberation time for the recording studio**

| **Frequency Range (Hz)** | **125** | **250** | **500** | **1000** | **2000** | **4000** |
|---|---|---|---|---|---|---|
| RT60 Sabine | 0,64 | 0,54 | 0,32 | 0,30 | 0,31 | 0,30 |

**Figure 18: Recording Room Images**

# Chapter 5: Results and analysis

This chapter presents the results of the measurements obtained for each one of the rooms and the surveys applied to the musicians. Also it contains the correlation between the objective properties of the hall and the subjective preferences of each one of the musicians that played in them.

## 5.1 Measuring and characterizing results

As it was explained in the chapter 3, each one of the rooms was measured and characterized according to an objective procedure; in the same conditions we recorded the musicians. The results of the measurements are presented in this section, and some analysis relating them to the theoretical data we presented in chapter 4.

### *5.1.1 Reverberation time*

In chapter 3 we commented how the reverberation time was measured by using the Reverberation_Time_Calculator Matlab® tool and its reverb_time function. We used two different methods, the balloon pop and the white noise suddenly stopped. All the results of the measurements, such as values and original plots, are shown in the Appendix C: Reverberation time measurement; complete results, but in this section we will present a comparison and analysis of the different results by using plots and graphics. We only analyzed the data above 100Hz because with the actual measuring technique, this as the data that is considered reliable.

**Chorus Chamber:** In Figure 19 the measurements of the reverberation time with the balloon pop and the white noise are shown, and compared with the theoretical data given by the ESMUC. The three plots are actually very similar, but the theoretical reverberation time is higher than the ones measured by us. This occurs because the theoretical data was calculated with the room empty. The measurements we took for the chorus chamber were made with the curtains closed and chairs in the room, in the same conditions we recorded. The chairs absorb some of the reflections, reducing the reverberation time of the room. It can be noticed also that for the low frequencies (around 100Hz) the reverberation of the balloon pop is lower, because is does not excite equally all the frequencies, especially the low ones.

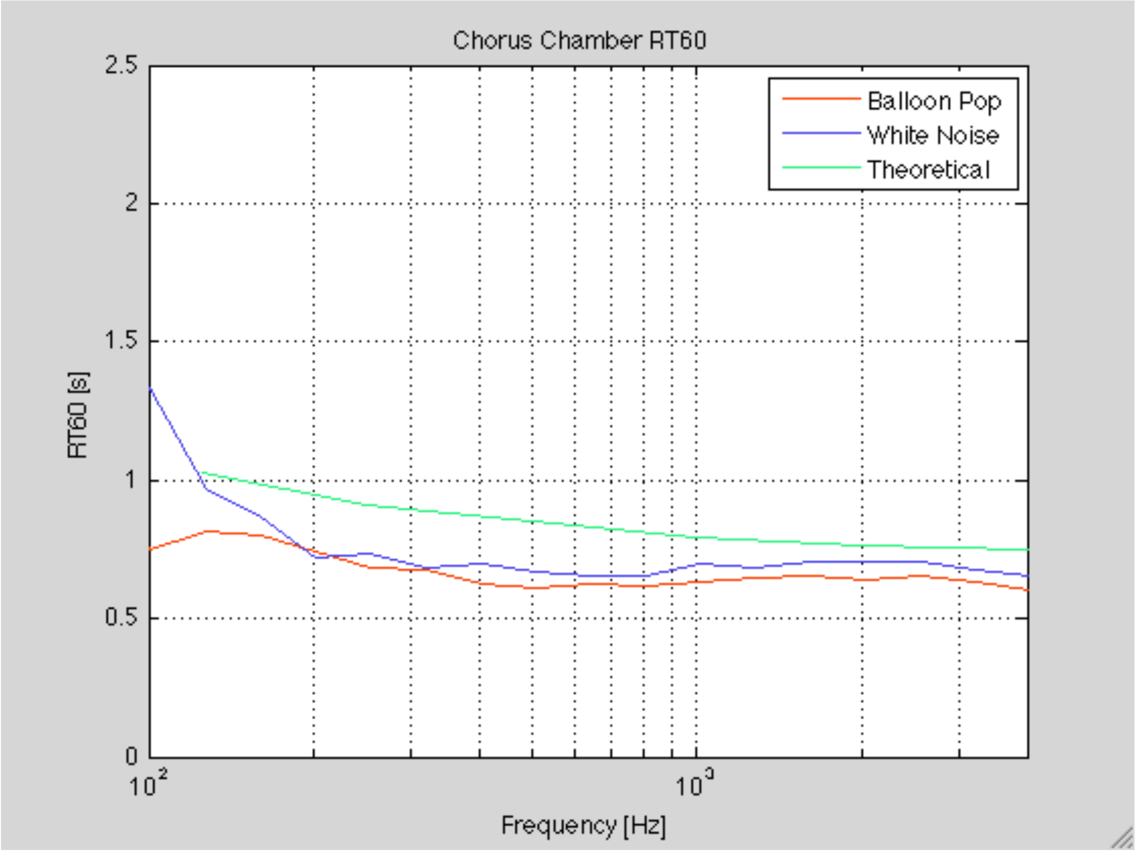


**Figure 19: Reverberation time for the chorus chamber**

**Organ Hall:** The organ hall, as it was commented in chapter 4, is characterized for having a marble wall to increase the reverberation time, and gives the feeling of being in a church.

In Figure 20 we can see a plot of the reverberation time in the organ hall, comparing the measured values with balloon pop, white noise, and the theoretical data. Across the plot we can see that the white noise curve has more reverberation time than the balloon pop one, although they have the same shape. This occurs because with the balloon pop we cannot guarantee the same excitation and amplitude for all the frequencies, while with the white noise we have more control.

Anyhow, we can notice the measured plots here are different that the theoretical data. For the frequencies below 500Hz a prominent decay can be observed in both measured plots. This occurs because the theoretical data was calculated for an empty room without curtains. Though in this case we made the measurements without the curtains, the room had some instruments and chairs that affected the results.

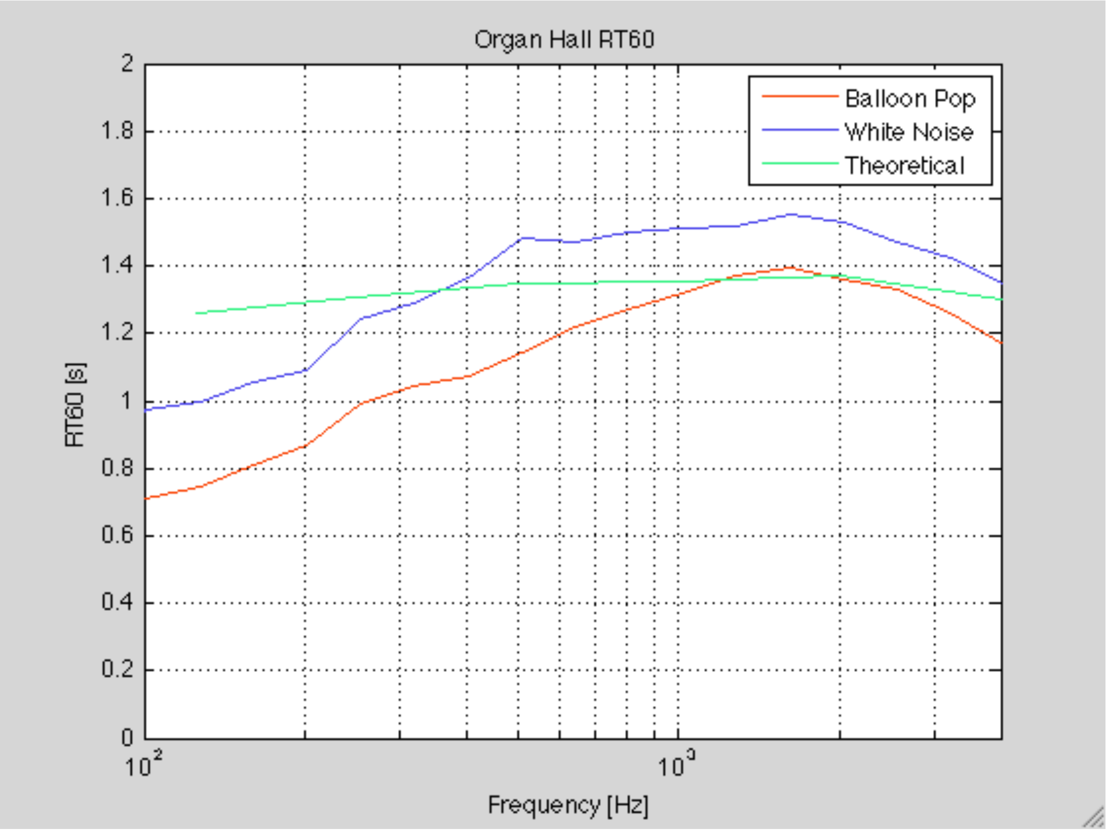


**Figure 20: Reverberation time for the organ hall**

The most important effect of the instruments on the reverberation time is given by the organ and its pipes. They absorb the sound; especially the frequencies below 500Hz, acting as a **Helmholtz resonator.** A Helmholtz resonator is a container of gas (usually air) with an open neck. The volume of air contained vibrates because of its springiness. An example of this is when you blow air into an empty bottle and obtain some sound. The Helmholtz resonators can be tuned also to act like absorbers, and are commonly used in architectural acoustics to reduce undesirable low frequencies.

The frequency of resonance of a Helmholtz resonator can be calculated using Equation 4, where c is the speed of sound, S is the cross-sectional area of the neck, V is the volume of the container and L is the length of the neck (The university of South Wales: School of physics, 2011).

$$fr = \frac{c}{2\pi}\sqrt{\frac{S}{V \cdot L}}$$

**Equation 4: Frequency of resonance in Helmholtz resonators**

In Figure 21 and Figure 22 we present the spectrogram of the balloon pop and white noise recorded in the organ hall. In both figures can be observed that the decay occurs faster for low and high frequencies, while the frequencies between 200Hz and 4000Hz decay slowly. In Figure 22 we can notice that during the whole time of the sound, the sound is quite weak for those frequencies below 50Hz, because of the frequency response of the loudspeaker presented before in Figure 6.

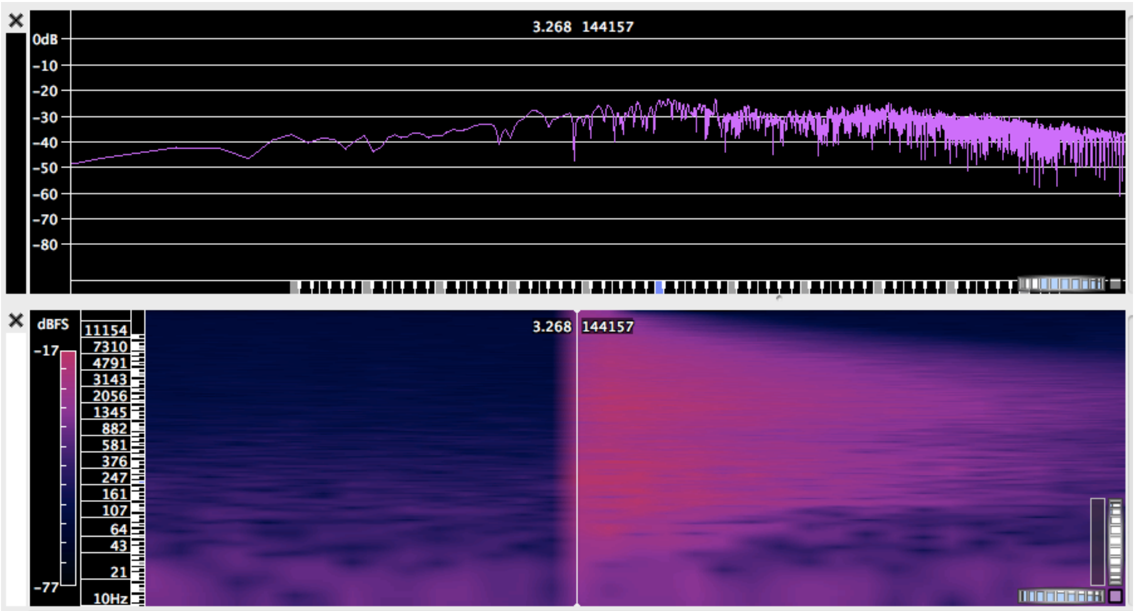


**Figure 21: Spectrum and spectrogram visualization for balloon pop in the organ hall**

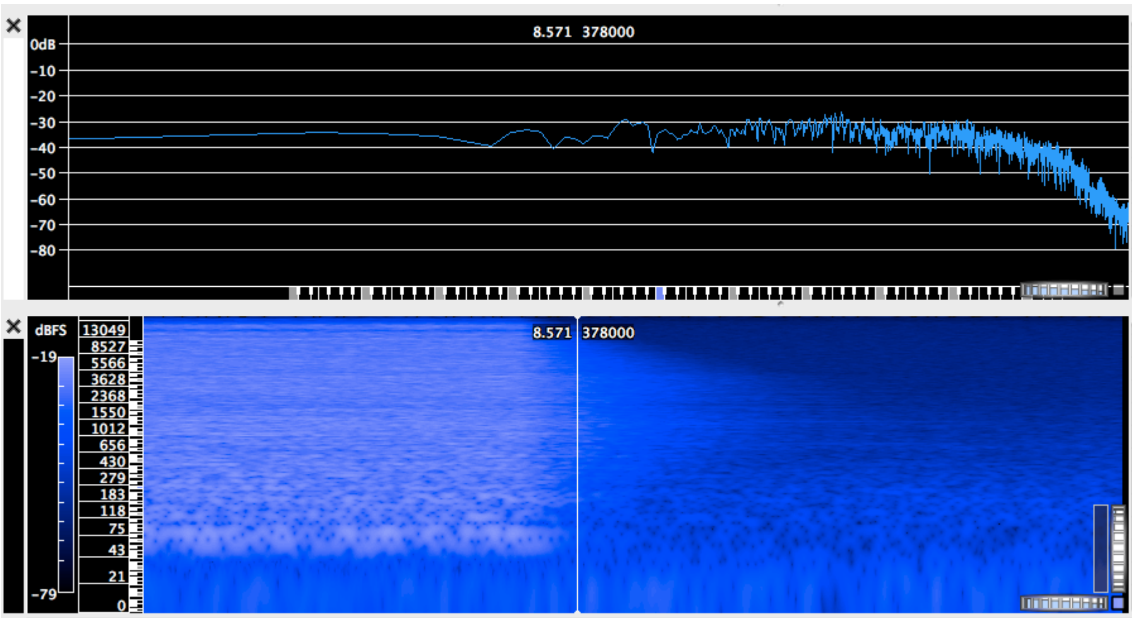


**Figure 22: Spectrum and spectrogram visualization for white noise in the organ hall**

To get these images, we analyzed the sounds using the software Sonic Visualiser® developed at the Queen Mary, University of London. It is free and can be downloaded at http://www.sonicvisualiser.org/.

**Recording Studio:** The comparison between the measurements made with the balloon pop, the white noise and the theoretical data can be observed in Figure 23. We can notice that the three plots are quite similar in shape and values. In general, the reverberation time for all the frequencies is lower in the recording studio than in the two rooms analyzed before, hence the sound is expected to be dryer.

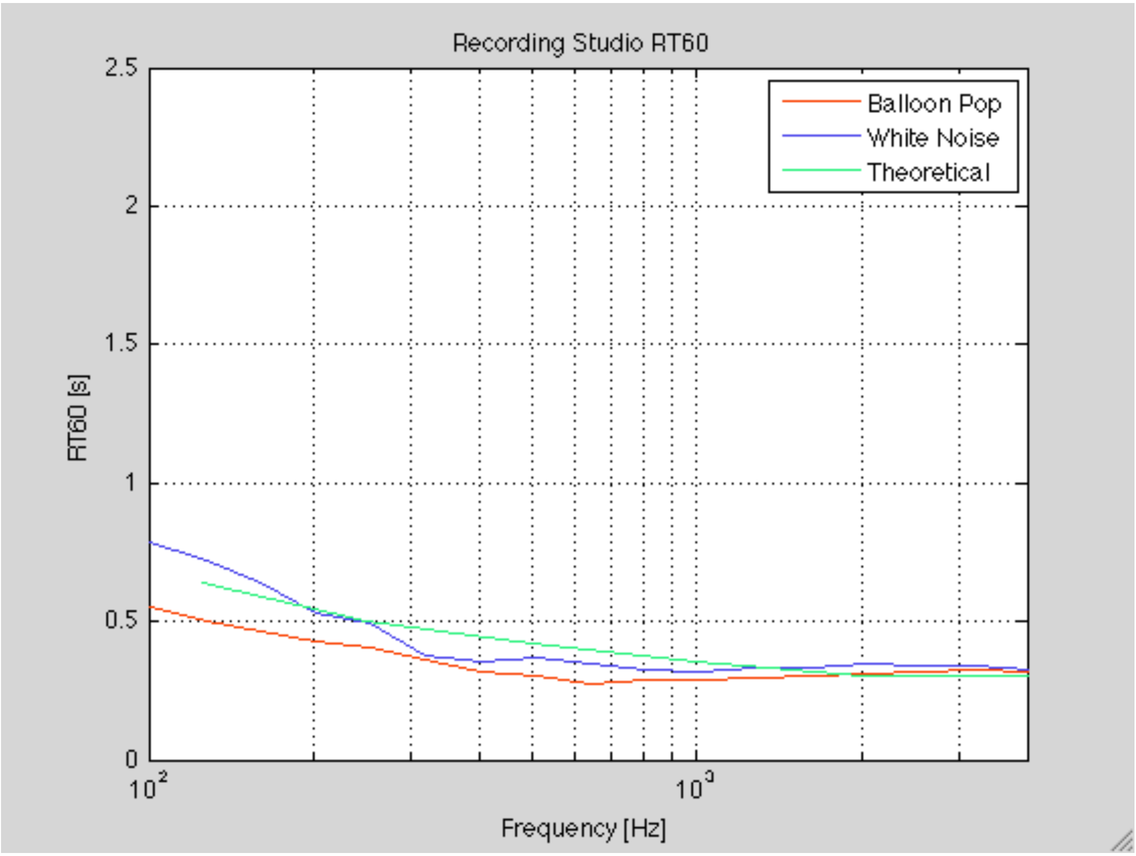


**Figure 23: Reverberation time for the recording studio**

**Rehearsal Room:** For the rehearsal room we are plotting the measurements with balloon pop, white noise and the initial given measurements in Figure 24. These measurements were made without curtains and the room empty, but with the chairs and instruments in the room, in the same way we recorded.

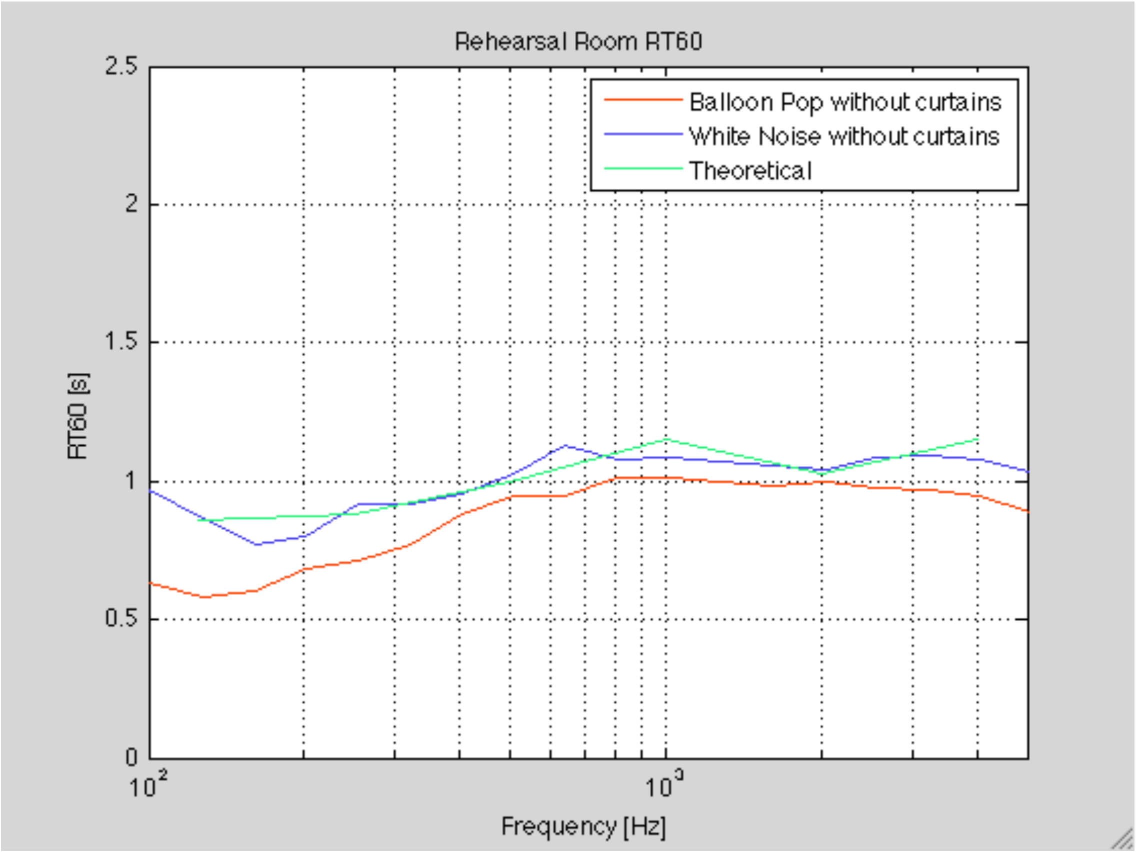


**Figure 24: Reverberation Time in the Rehearsal Room**

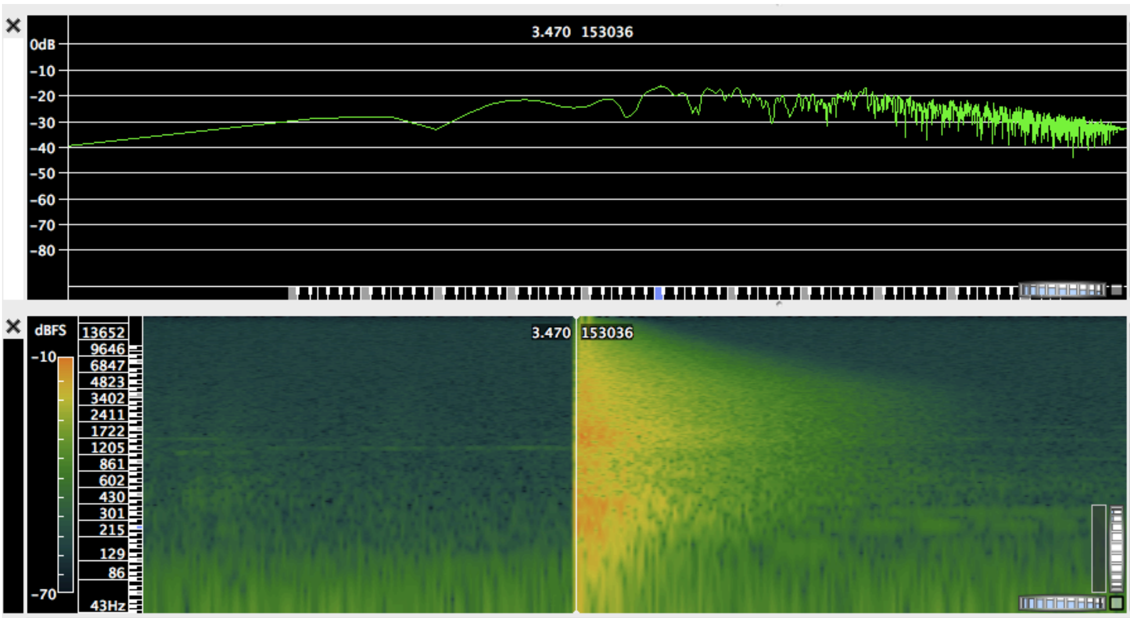


**Figure 25: Spectrum and spectrogram visualization for balloon pop in the rehearsal room**

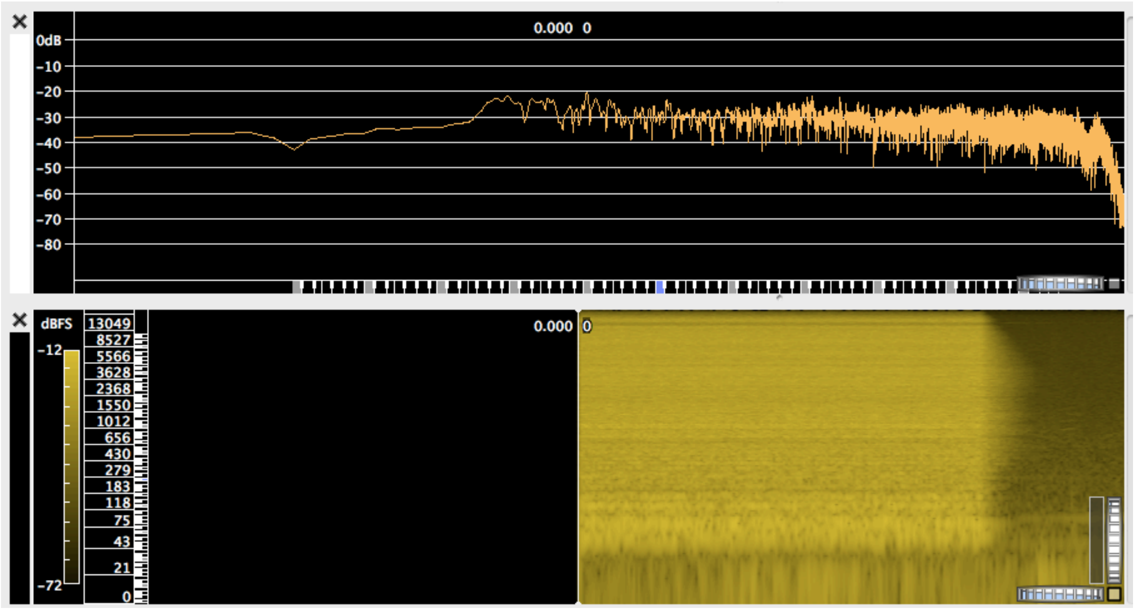


**Figure 26: Spectrum and spectrogram visualization for white noise in the rehearsal room**

Both signals have similar shapes and values, but the balloon pop is lower for the frequencies below 200Hz, because it does not excite equally the low frequencies. This can be seen in Figure 25 and compared with Figure 26, where the white noise only decays below 50Hz, because of the frequency response of the loudspeaker showed in Figure 6.

In Figure 24 can be noticed that the middle-high frequencies have more reverberation than the low frequencies. In contrast, we can observe in the spectrograms of Figure 25 and 26 how the low frequencies are absorbed first than the middle and high ones. Can be observed in Figure 24 that the behavior of the measured plots (blue and red) is similar to the initial measurements given by the ESMUC.

**Table 5: Absorption coefficients of plywood (SAE institute, 2011)**

| Frequency | 125 Hz | 250 Hz | 500 Hz | 1000 Hz | 2000 Hz | 4000 Hz |
|---|---|---|---|---|---|---|
| Absorption coefficient $\alpha$ | 0.28 | 0.22 | 0.17 | 0.09 | 0.1 | 0.11 |

As it was described in chapter 4, the rehearsal room is a parallelepiped, with walls, ceiling and floor in plywood. According to Table 5, the absorption coefficients of plywood are bigger for the frequencies below 500Hz than for the frequencies above

1kHz. For this reason, the wood absorbs more energy in the middle low frequencies, and the reverberation time is higher for the middle high frequencies.

### *5.1.2 Real time analysis and spectrograph*

As it was explained in sections 12.1 and 13.1, the RTA is a measure of the instantaneous frequency response of the room. It changes according to the location and position of the loudspeaker and microphone, hence, we cannot conclude about the room characteristics with these measurements, but we can see its behavior under certain conditions.

In the other hand, the spectrogram plot displays another form of RTA, where the frequency content of the signal is showed in colors versus a period of time. Instead of showing the instantaneous frequency response of the room, it shows a record of the most recent 100 (or even more) RTA updates (Calvert Dayton, 2007).

In section 13.1 the RTA display is explained, as how the measurements are done with a reference signal and a measured signal. Both signals are compared in the RTA display, with blue we can see the reference signal, and with green the measured one.

In the other hand, the spectrograph shows three dimensions of data. The time is showed on the x-axis, the frequency is showed in the y-axis, and the magnitude of the frequency is represented by colors.

As can be seen in the figures, the reference signal (blue) was pink noise (same magnitude for all the frequency octaves). The same pink noise reference signal was used for all the rooms, and it was generated with the Smaart ®.

In all the rooms we can see that the blue signal (reference) is flat, because is a pink noise with the same magnitude for all the octaves. The green plot corresponds to the RTA measured by the microphone, and we can see the behavior of the frequencies at the room under those conditions.

For all the rooms, we divided it by trying to locate the loudspeaker and the microphones to the same distance from the walls and from each other, if it was possible. We also pointed them to the walls (avoiding the direct field), and took note of the locations attached in the Appendix B: Location of the microphone and the loudspeaker for each room. Then this configuration was repeated by locating the musicians (and ORTF microphone) where the loudspeaker was, and the AB microphones where the omnidirectional microphone was.

**Chorus Chamber:** In Figure 27 we can see the spectrograph and RTA display for the chorus chamber with the locations shown in the Appendix B: Location of the microphone and the loudspeaker for each room and pointing to the walls. We can remark that the behavior under these conditions is very flat, with a little more energy in the low frequencies and less energy in the high frequencies.

**Organ Hall:** In Figure 28 we can see the spectrograph and RTA display for the organ hall with the locations shown in the Appendix B: Location of the microphone and the loudspeaker for each room and pointing to the walls. In this case, the behavior is not that flat, the low energies have more amplitude while the energy decay in the high frequencies.

**Recording Studio:** In Figure 30 we can see the spectrograph and RTA display for the recording studio with the locations specified in the Appendix B: Location of the microphone and the loudspeaker for each room and pointing to the walls. We can see it has a boost in the low frequencies and decays a lot for the high frequencies.

**Rehearsal Room:** In Figure 30 we can see the spectrograph and RTA display for the rehearsal room with the locations specified in the Appendix B: Location of the microphone and the loudspeaker for each room and pointing to the walls. We can see it has also a boost in the low frequencies but the decay in the high frequencies is not as prominent as in the recording studio.

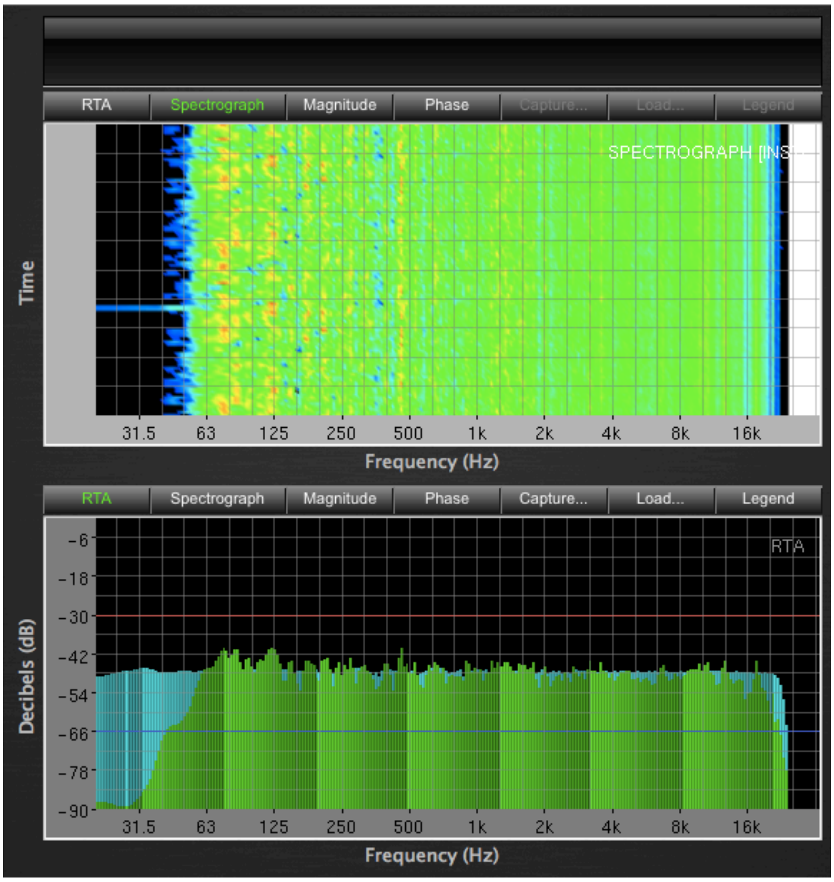


**Figure 27: Spectrograph (up) and Real Time Analysis (down) for the chorus chamber**

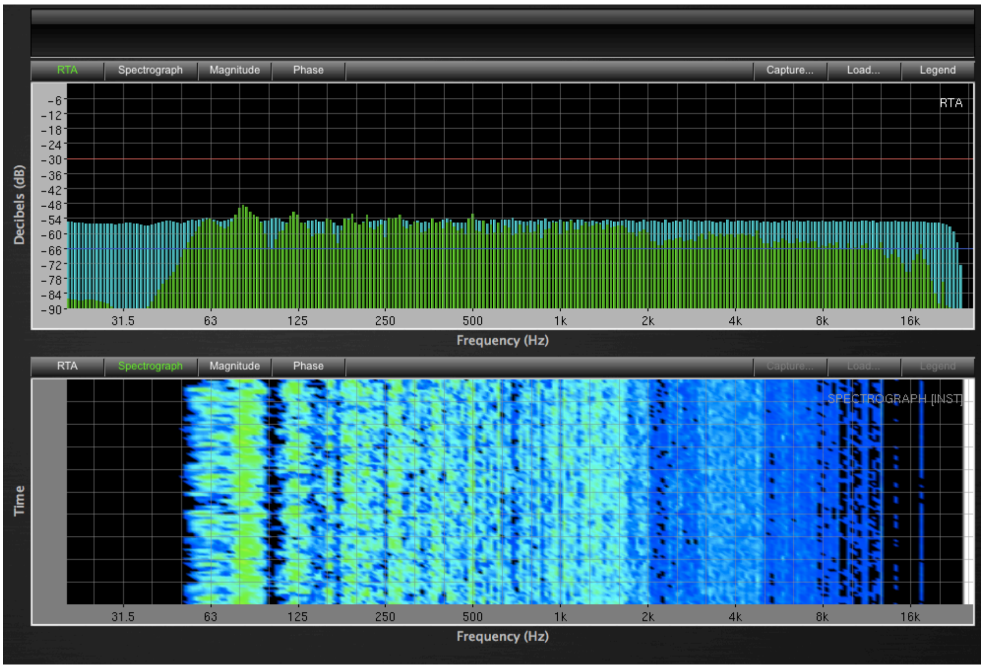


**Figure 28: Real time analysis (up) and spectrograph (down) for the organ hall**

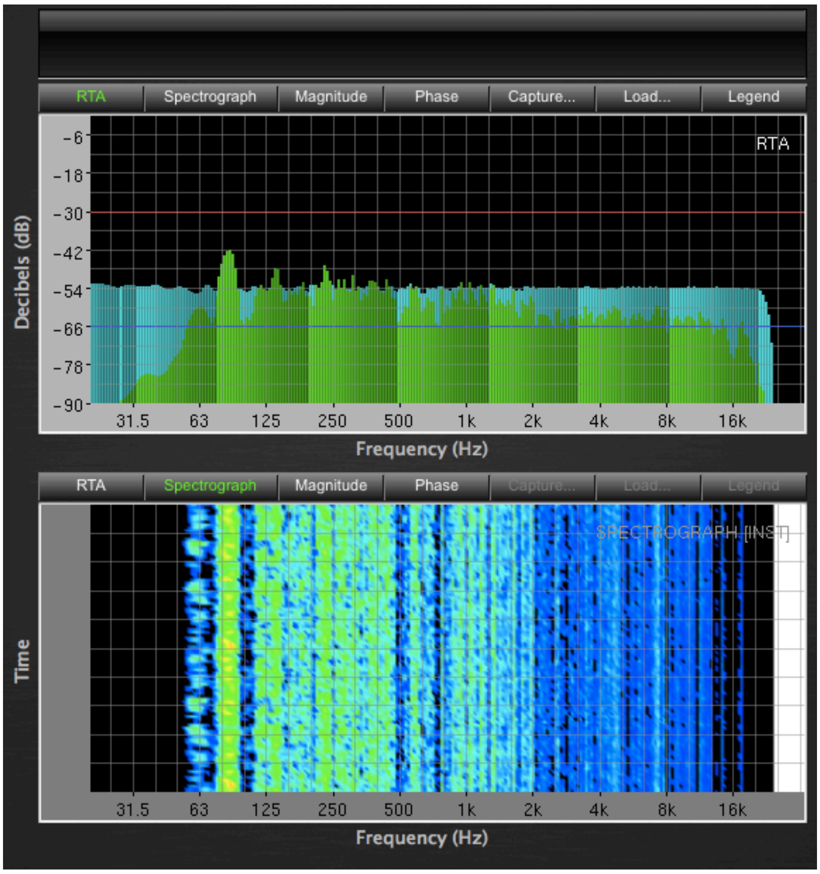


**Figure 29: Real time analysis (up) and spectrograph (down) for the recording studio**

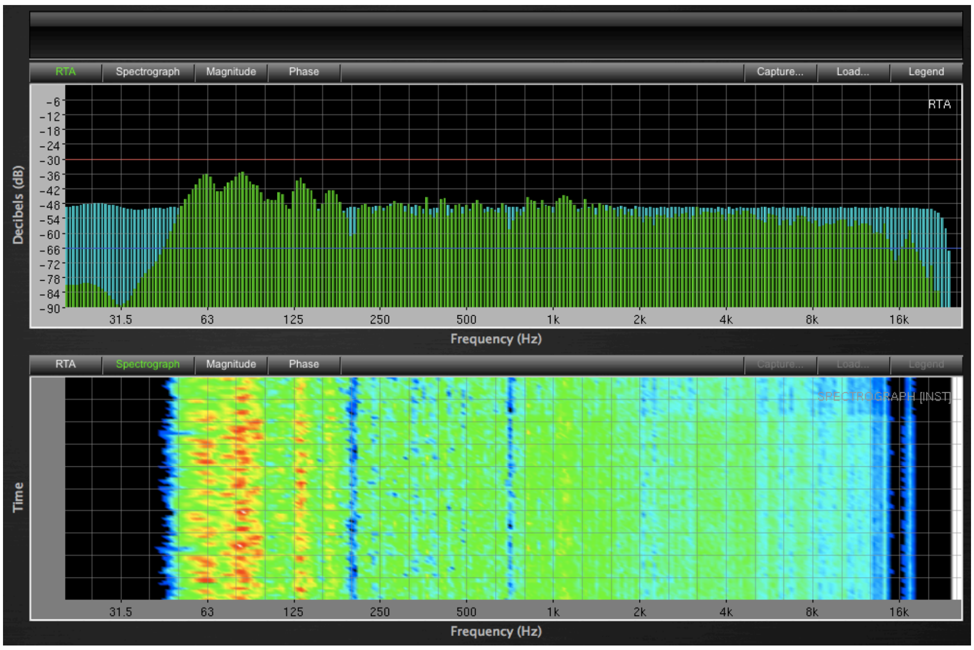


**Figure 30: Real Time Analysis (up) and spectrograph (down) for the rehearsal room**

### 5.1.3 Transfer function

As we mentioned in section 2.1 the transfer function compares a reference signal with a measured one (output versus input). In this case, we can see the transfer function for each one of the rooms in magnitude and phase in yellow, and the coherence trace in red.

The coherence trace was also explained in section 2.1, and it is the representation of the linearity between two signals. Issues such as reverberation, reflections or noise affect the coherence of the signals.

Smaart® is also able to provide an internal delay compensation, up to 750ms, to provide signal alignment between the reference and the measured signal for frequency domain plots. This delay must be used, and the distance between the loudspeaker and the microphone should be set to have the right measurements of phase in the transfer function. The delay compensation can be automatically measured by Smaart®, and introduced to the system.

To explain the concepts of coherence and delay compensation, and understand the behavior in the rooms analyzed, we made the following experiment in the recording studio (the driest room). First, we measured the pink noise by using absorbers around the microphone and the loudspeaker (simulating an anechoic chamber), with the loudspeaker and the microphone pointing to each other. Then, we removed the absorbers (to see the response of the room), and analyzed the response with and without delay compensation. Then we analyzed the response with the microphone and the loudspeaker pointing to the walls, with and without delay compensation. The results can be seen in the following figures.

In Figure 31 and Figure 32 we can see the transfer function for a pink noise in the recording studio, with the microphone and loudspeaker pointing to each other, without using the absorbers. The results with and without absorbers were similar in coherence and phase, but one showed the response of the room and the other did not. Figure 31 shows the results without using the delay compensation while Figure 32 shows the results of the transfer function with the delay compensation set.

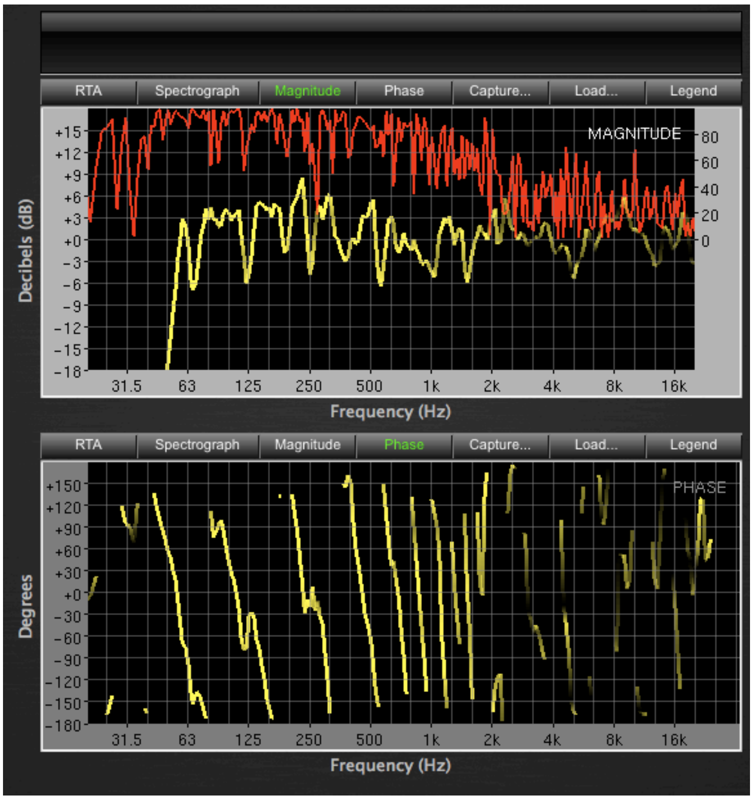


**Figure 31: Recording Studio experiment, without absorber, without delay and pointing to each other**

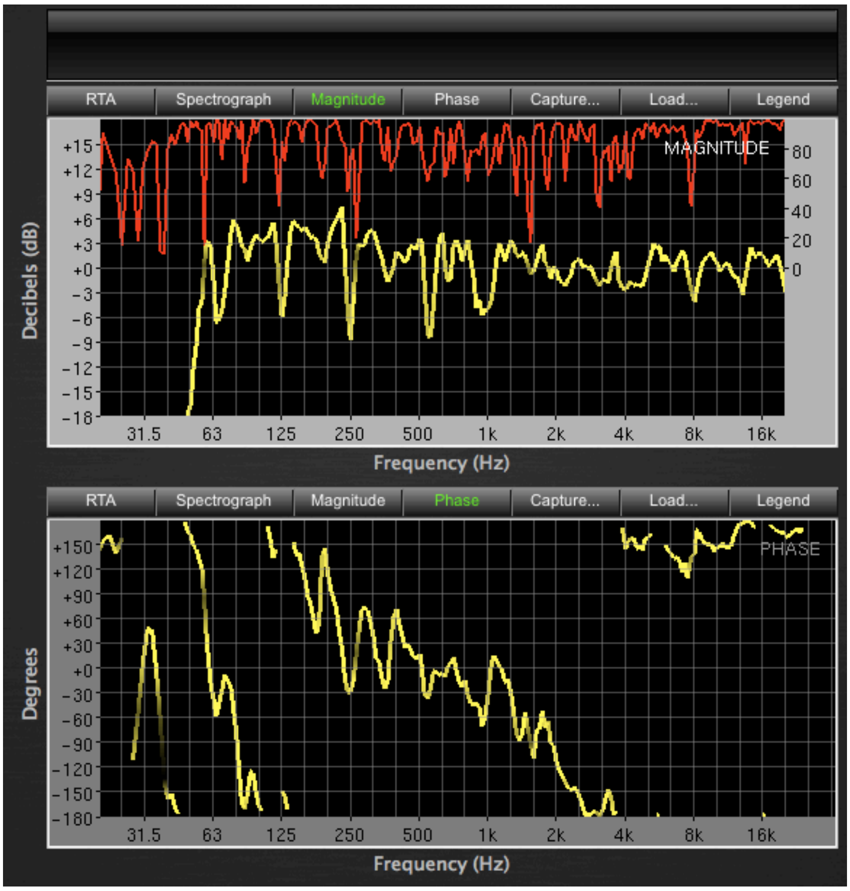


**Figure 32: Recording Studio experiment, without absorber, with delay and pointing to each other**

In Figure 31 we can observe that the coherence is low especially for the high frequencies, and the phase of the transfer function is not so clear. Once we automatically set the delay compensation, the coherence is higher and the phase can be observed clearer in Figure 32.

Once we turned around the microphone and the loudspeakers pointing to the walls (as we did the measurements), the delay compensation was not captured automatically by Smaart® and it had to be set manually. Anyway, we can notice in Figure 33 and Figure 34 that even though the delay compensation was set manually, the response was very similar, and it was confusing for the high frequencies.

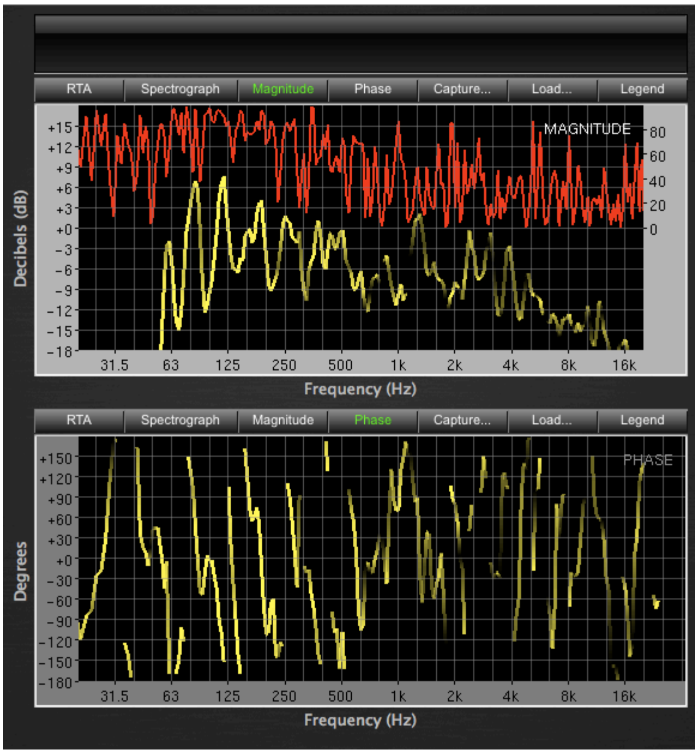


**Figure 33: Recording Studio experiment, without absorber, without delay and pointing to the walls.**

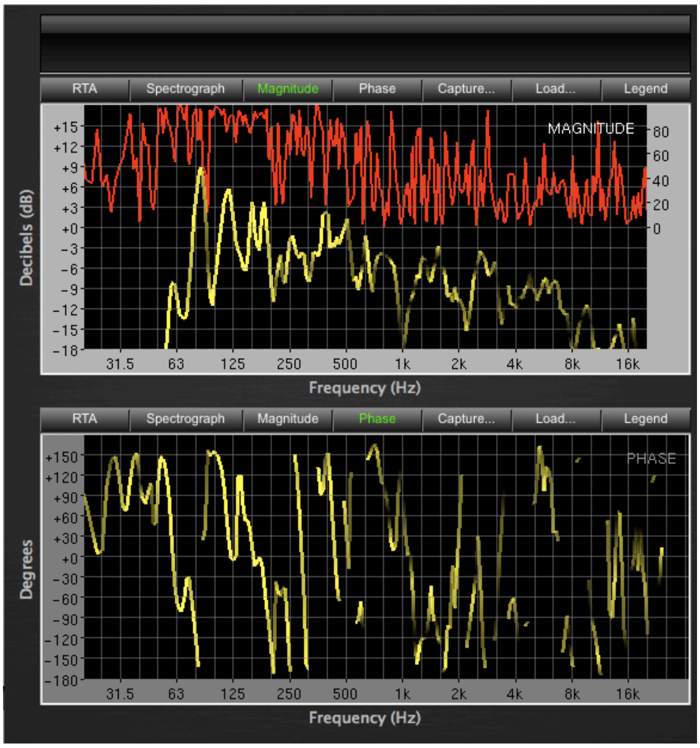


**Figure 34: Recording Studio experiment, without absorber, with delay and pointing to the walls.**

We can conclude that it is very important to set the delay compensation for the measurements of the transfer function, when the work is going to be done at direct field. In order to have correct results, it is necessary to set the distance between the microphone and the loudspeaker, and use the software to calculate the delay compensation between them and draw the proper phase of the two signals.

Anyhow it is irrelevant when we are going to work in the diffuse field. In our project we did all the measurements by setting manually the distance between the loudspeaker and the microphone. Even though there was no difference in setting or not the delay compensation for the diffuse field, we set it because it was conceptually correct, and to avoid any problems in the phase that could be caused by the distance between loudspeaker and microphone.

When the microphone and the loudspeaker are pointing to the walls, the measured signal is not capturing the direct field, but only reflections and reverberation (as was explained in section 3.1.2, the diffuse field). Hence, the coherence will always be low, especially for the high frequencies, because their wavelength is shorter.

For our case, in the following figures can be observed that the coherence is quite low, especially for the high frequencies, hence, the phase will always be confusing and we could not get good conclusions of it. It is interesting to analyze better the magnitude and the modes of frequencies of each one of the rooms.

The microphone was capturing only the sound reflected in the walls of the room. For this reason can be observed that the most reverberant room (the organ hall) was the one with lowest coherence. The rehearsal room, which is also quite reverberant, had low coherence as well, while the chorus chamber and the recording studio, which are dryer, had higher coherence traces. This makes sense because if we are capturing more reflections across the time, the linearity with the reference signal will be lower.

As we said before, because of the low coherence due to the diffuse field, we cannot conclude on the behavior of the phase for each room or the magnitude for the high frequencies.

Smaart® is able to show the phase trace in two different ways: **degrees or group delay**. For the experiment we did at the recording studio, we could see the phase in degrees, but in the rooms we showed it using the group delay.

To understand the difference between the phase in degrees or in group delay, let us refer to the Figure 35, and suppose a sinusoid of 1kHz in blue, and another sinusoid of 1kHz in black, 90 degrees out of phase from the first one.

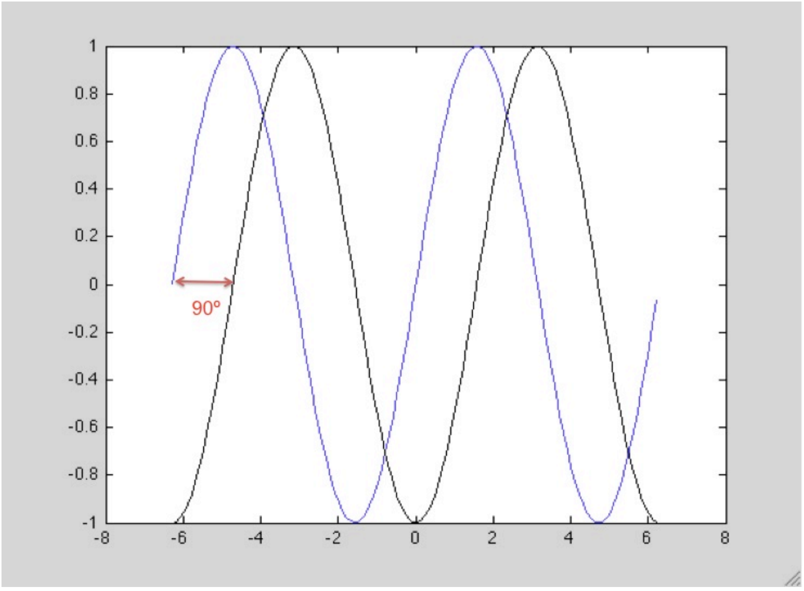


**Figure 35: Example of phase in sinusoids**

For the transfer function plot, at the 1kHz frequency, the phase between these two signals would be 90 degrees. The period of a sinusoid is 360 degrees, so 90 degrees correspond to a quarter of the period.

For a 1kHz frequency signal, and according to Equation 5, the period is 1 millisecond. As we said before the phase in this case is a quarter of the period, so, in milliseconds, the phase would be 0,25ms.

$$f = \frac{1}{T}$$

**Equation 5: Frequency and period relationship**

To convert a phase in degrees to a group delay phase, Equation 6 could be used, where x correspond to the phase in degrees, and f correspond to the frequency in Hz.

$$GD = \frac{x}{360^{\circ} * f}$$

**Equation 6: Group Delay**

If the phase in degrees were a straight line, the group phase would be a constant line. The higher the inclination of the phase in degrees, the higher the value of the group delay trace. (Troxel, 2005)

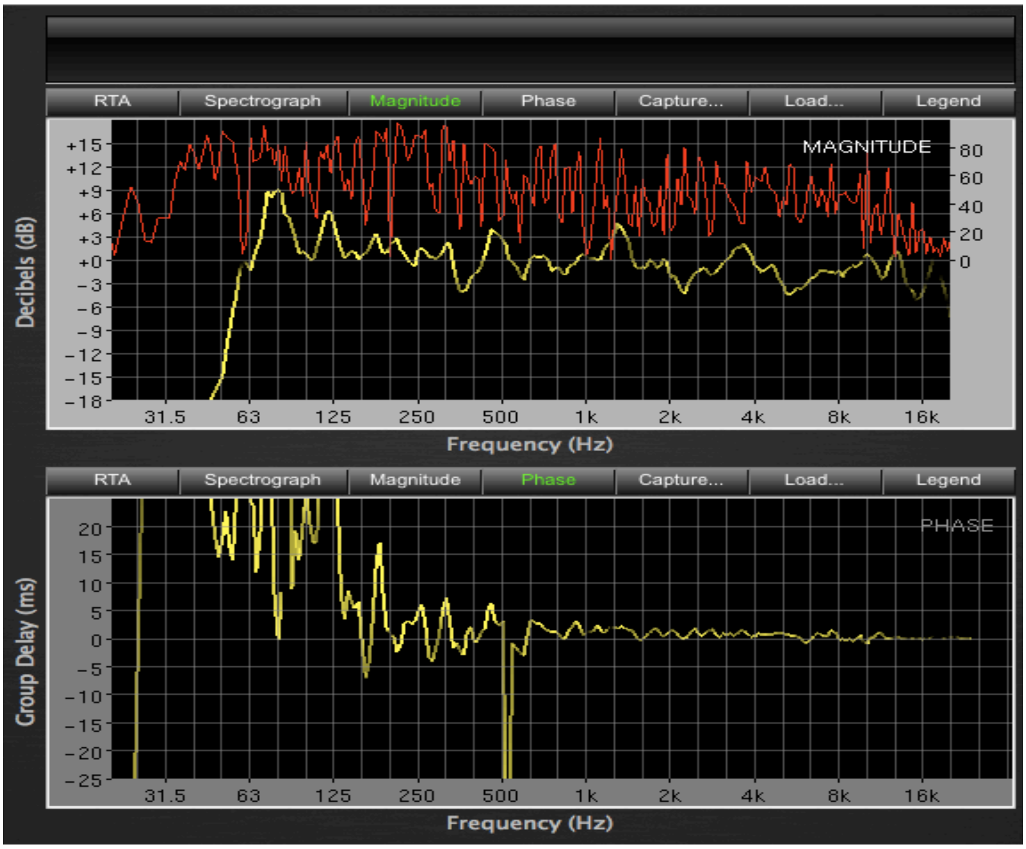


Figure 36: Transfer function in magnitude (up) and phase (down) for the chorus chamber

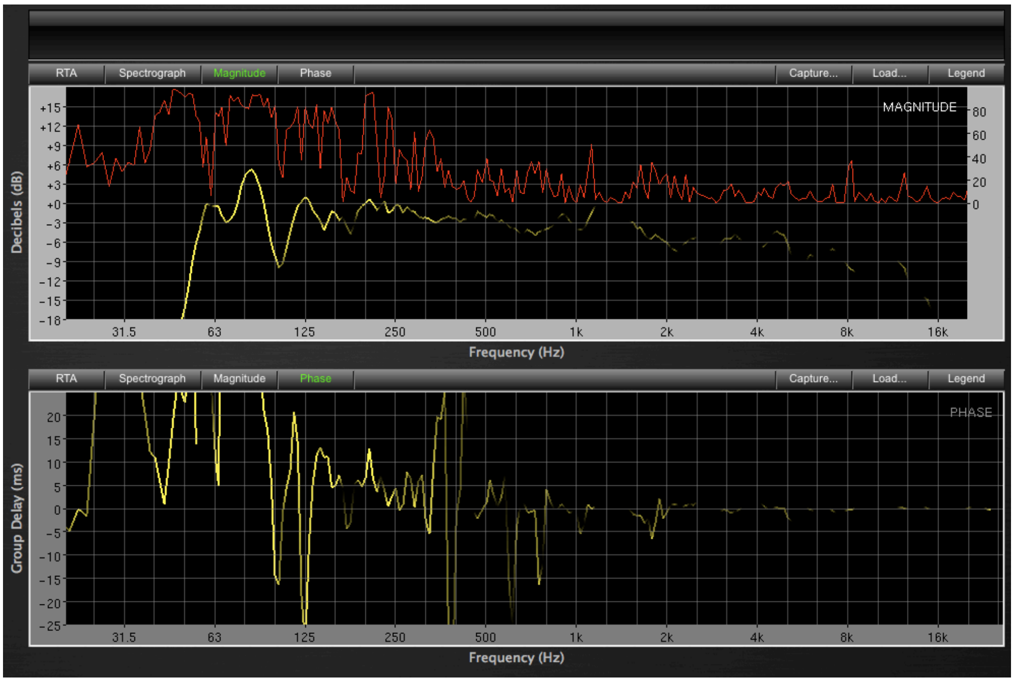


Figure 37: Transfer function in magnitude (up) and phase (down) for the organ hall

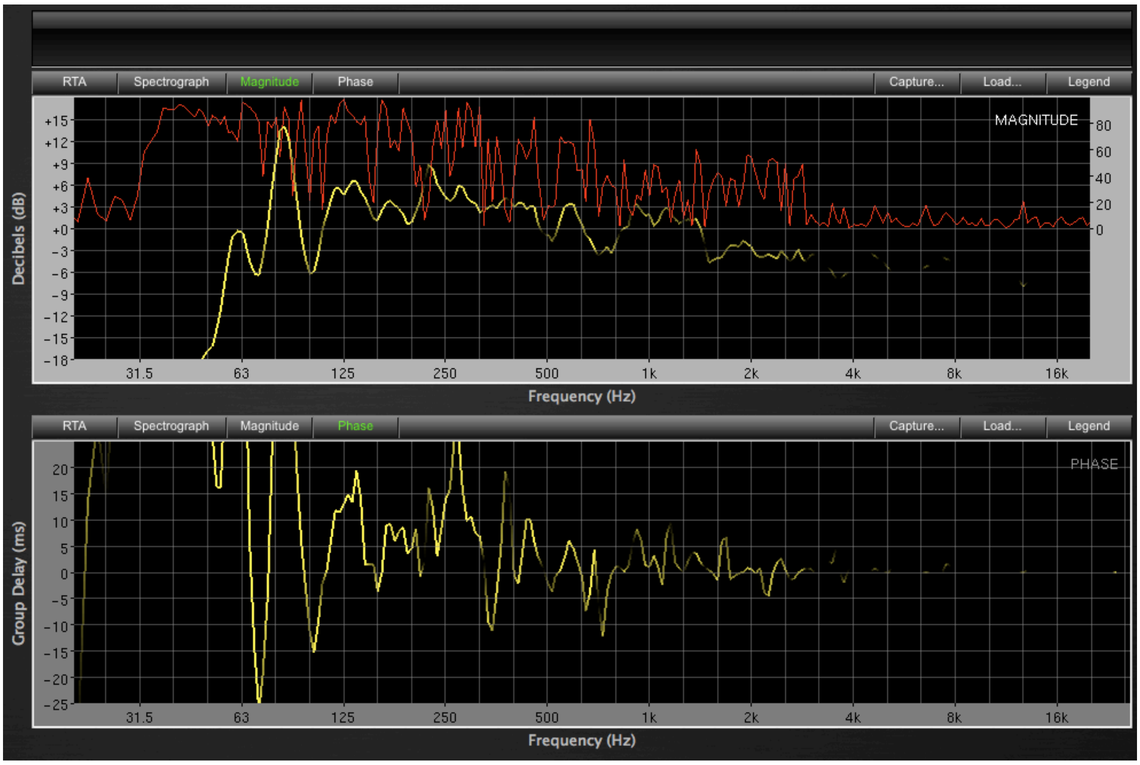


**Figure 38: Transfer function in magnitude (up) and phase (down) for the recording studio**

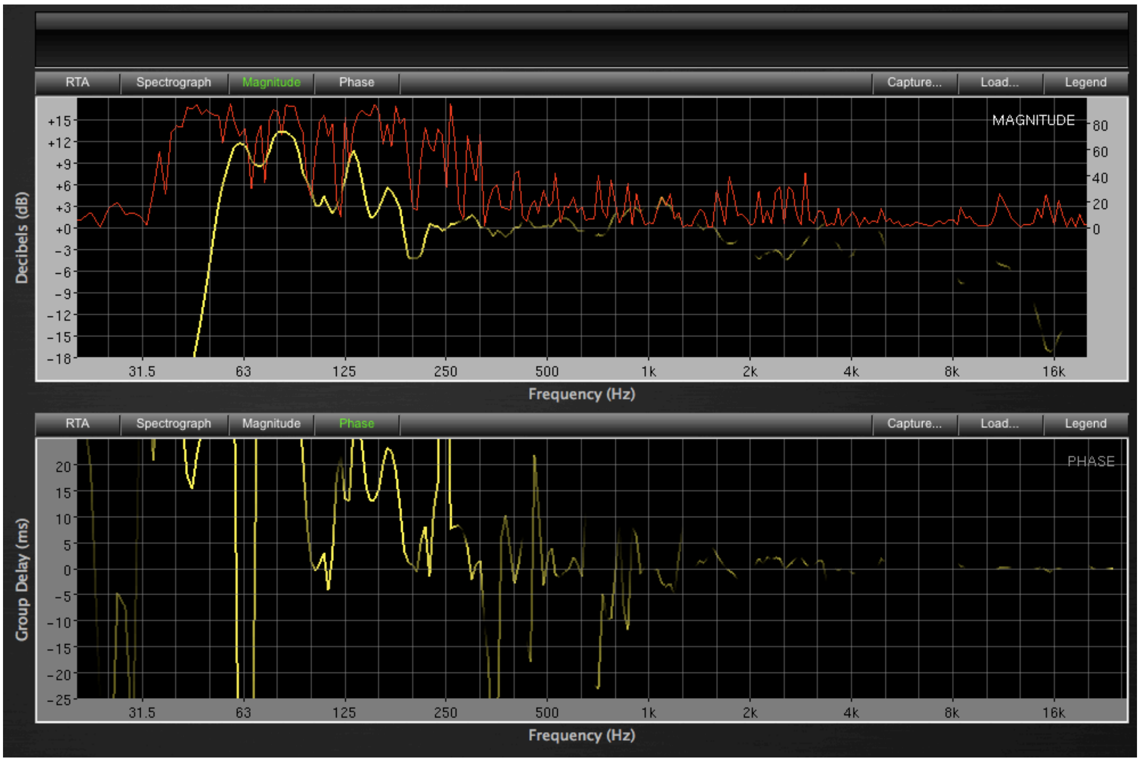


**Figure 39: Transfer function in magnitude (up) and phase (down) for the rehearsal room**

Anyhow, for group delay or degrees, the phase was confusing in all cases because we were measuring the diffuse field, and we cannot get enough conclusions out of these plots.

The plot with the higher coherence is the chorus chamber transfer function at Figure 36, this occurs because is a dry room, with low reverberation and big dimensions. In this case, the microphone was capturing more direct sound than reflections, the coherence is higher and the phase is not as confusing as in the others. We can see that for frequencies higher than 600Hz the reference and measured signals are in phase, and the group delay is 0ms approximately.

## 5.2 Surveys results

The surveys shown in the Appendix E: Surveys were applied to the musicians just right after they played in each one of the rooms. Using them, they gave their opinion on the acoustics, and judged them for different purposes. As can be observed, all the instructions were written in the survey, but someone was present during the process to explain more and answer questions regarding it.

As was mentioned in section 3.3, the time between the different surveys could not be fixed because we depended on the availability of the musicians and the rooms, so we refreshed their memory by showing the previous surveys when it was considered necessary. In some occasions we also allowed them to hear the recordings, in order to compare the different sounds.

No extra training was carried out for the listeners; we only took into account how they felt playing in the rooms, evaluating factors such as the feedback, the intimacy and the clarity.

The results of the surveys are presented in the Appendix F: Surveys results in detail, but in this section we will discuss only the mean results, and compare the plots obtained.

The data generated was analyzed for each musician and for the average values, and each case was plotted. In Figure 40, Figure 41, Figure 42 and Figure 43 we can see the results of the surveys for each room, according to each musician,

In Figure 44, Figure 45, Figure 46 and Figure 47 we can see the mode values (most frequent opinions for each question). The values that are marked with red are those where the mode contains less than 60% of the opinions (less than 3 musicians that agreed in the opinion)..

In the surveys shown in the Appendix E: Surveys, there is an open question called “hall valuation”, where the musicians wrote what they thought about each room, without the boundaries of the numbers. Their opinions are attached in the Appendix F: Surveys results in detail.

### *5.2.1 DESCRIPTION OF THE MUSICIANS WHO EVALUATED THE ROOMS*

To evaluate the rooms, we applied the surveys to 5 musicians, who played the same song in the four rooms, with the same instrument, and judged their preferences by filling the survey presented in Appendix E: Surveys.

All of them were musicians with more than 15 years of experience playing an instrument. They dedicated an average of 3.5 hours of active listening to music. One of them had problems of hearing, because his ears hurt with loud sounds. To see details of the demographic data of the musicians please refer to Appendix F: Surveys results in detail.


**Figure 40: Results of the surveys for each musician at the Chorus Chamber**

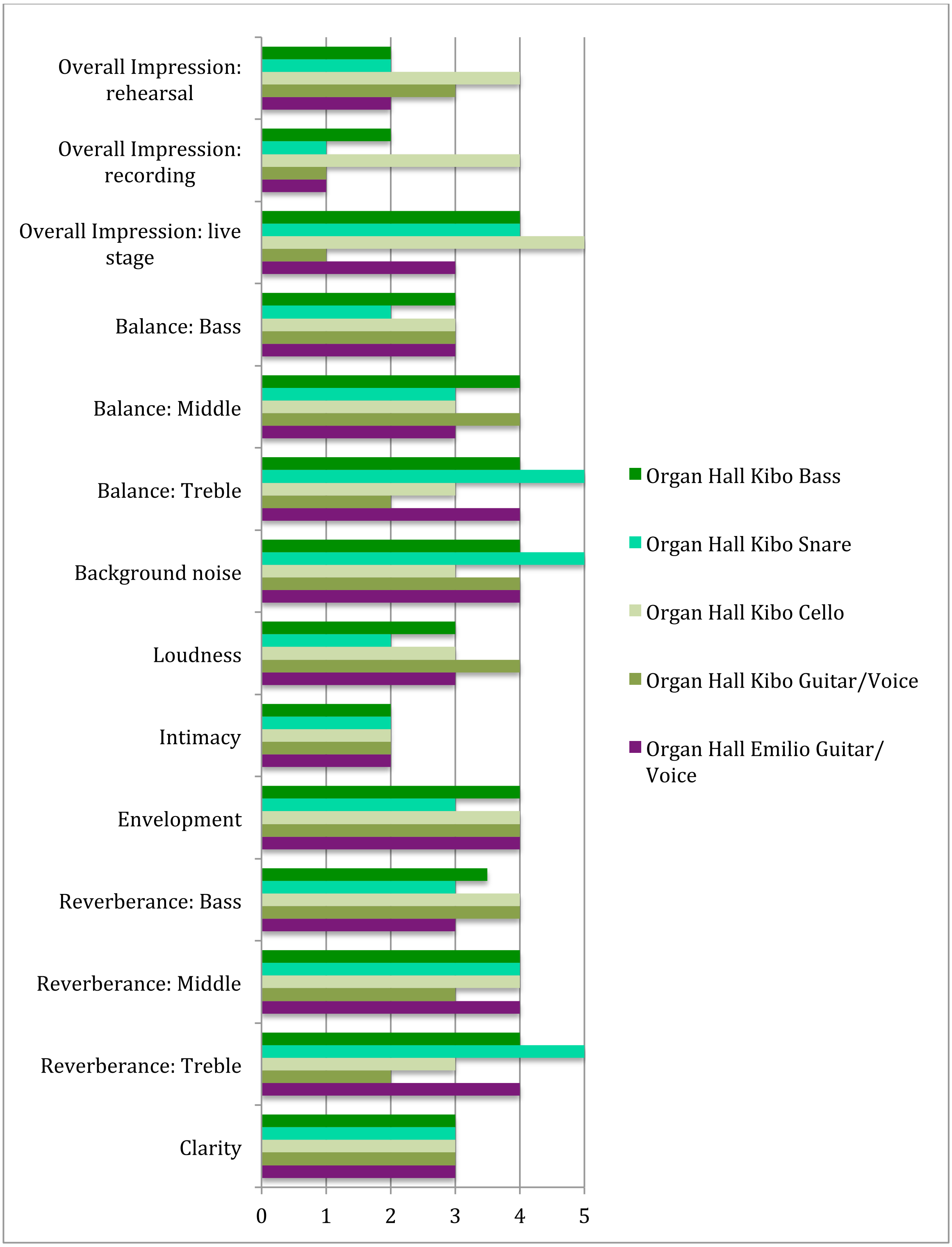


**Figure 41: Results of the surveys for each musician at the Organ Hall**

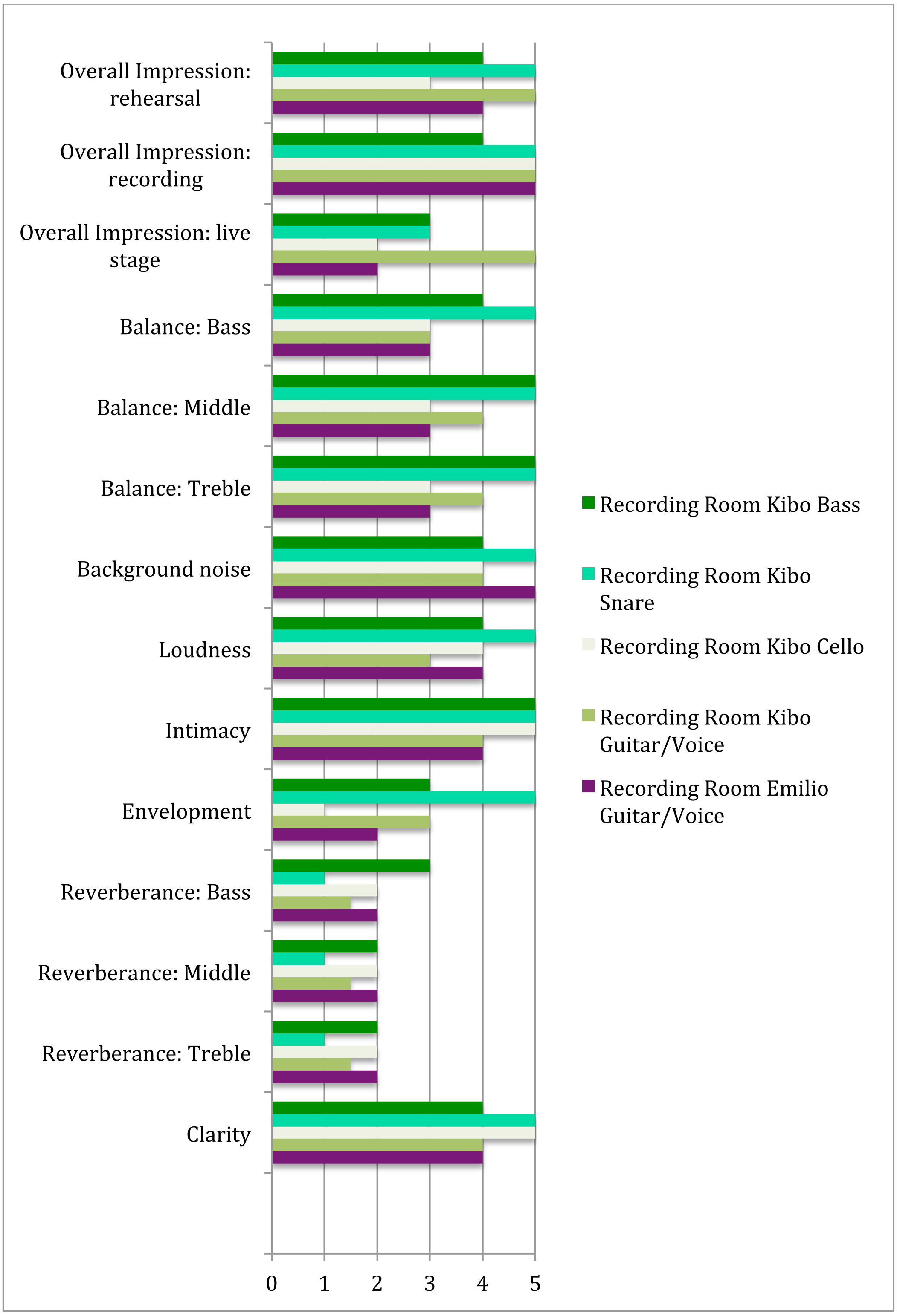


**Figure 42: Results of the surveys for each musician at the Recording Room**

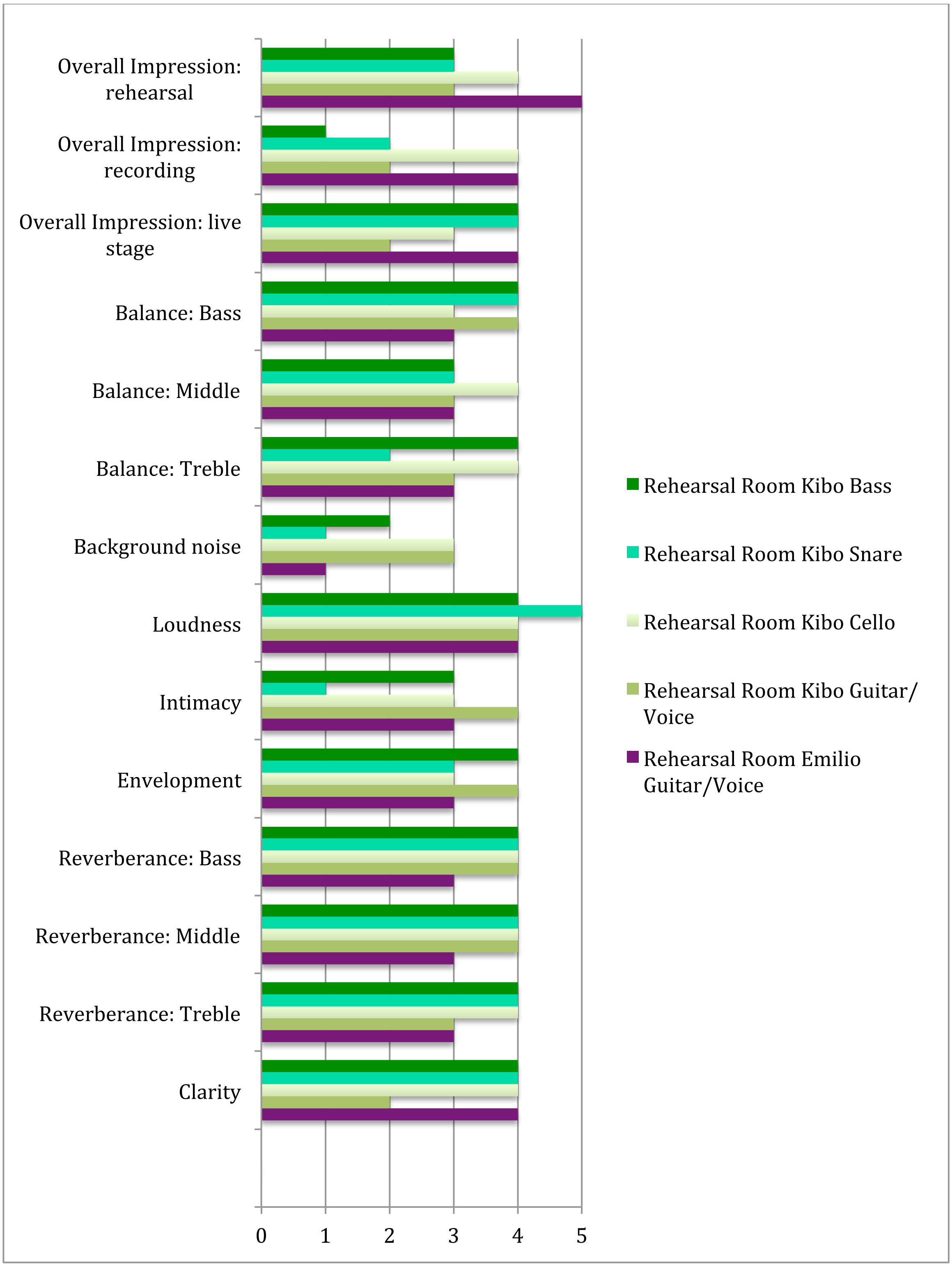


**Figure 43: Results of the surveys for each musician at the Rehearsal Room**

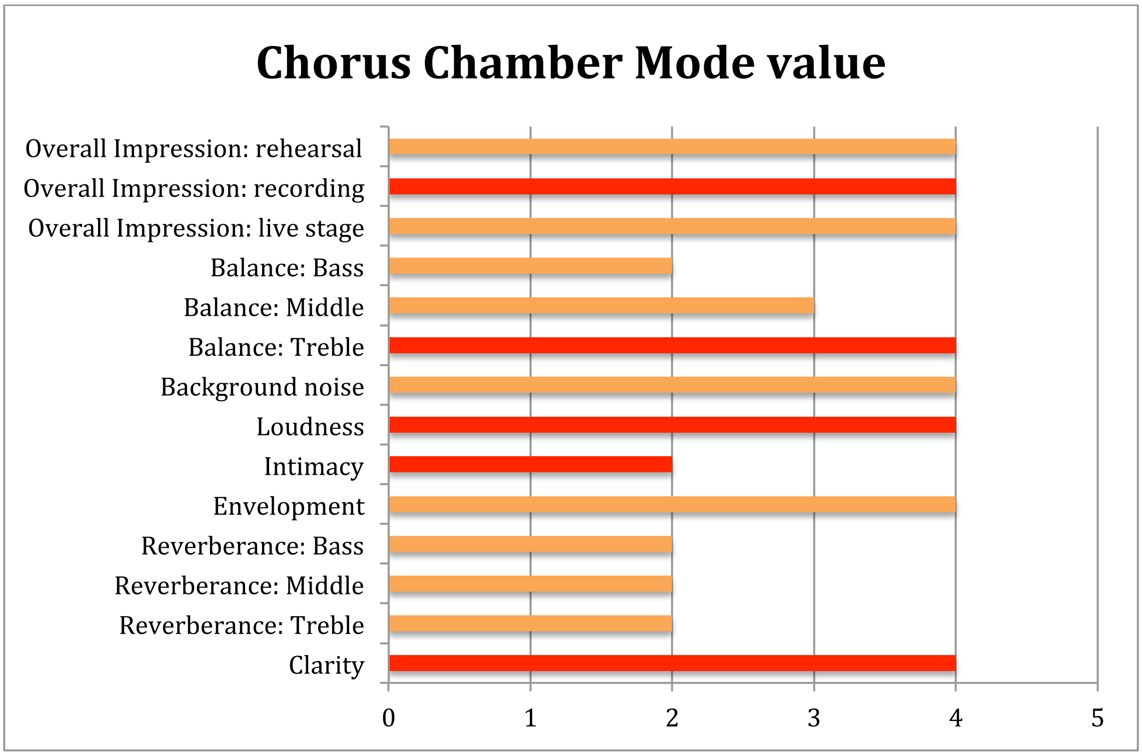


**Figure 44: Mode values of the surveys for the Chorus Chamber**

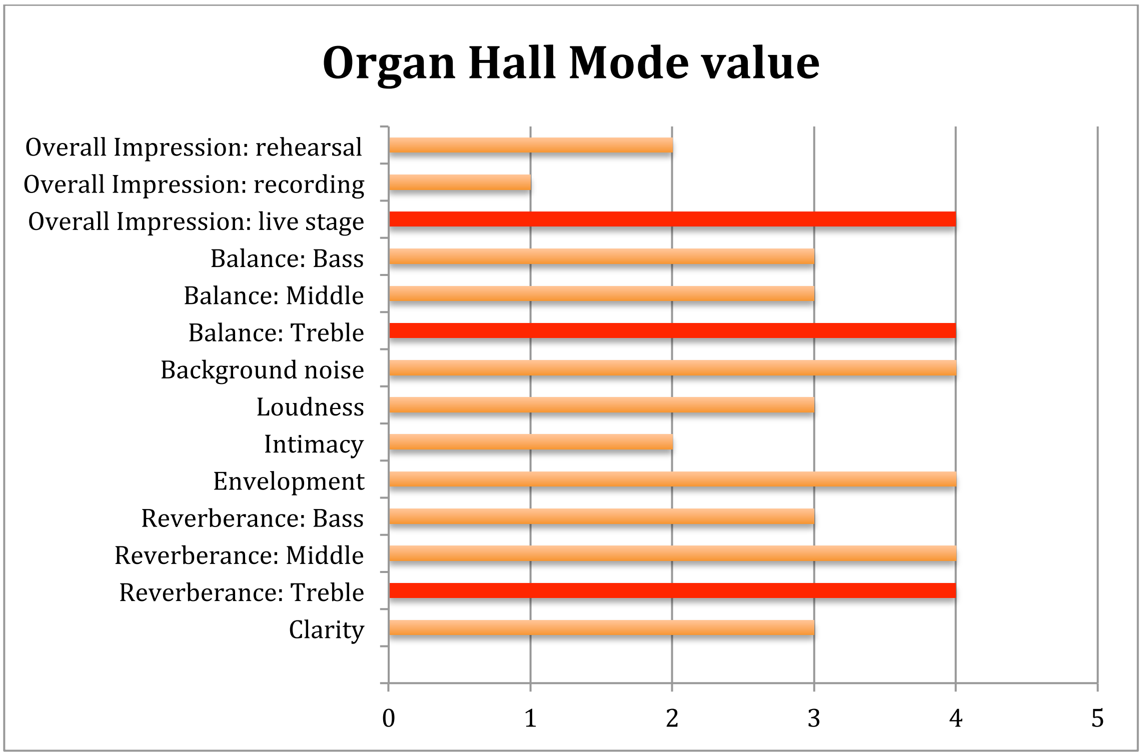


**Figure 45: Mode values of the surveys for the Organ Hall**

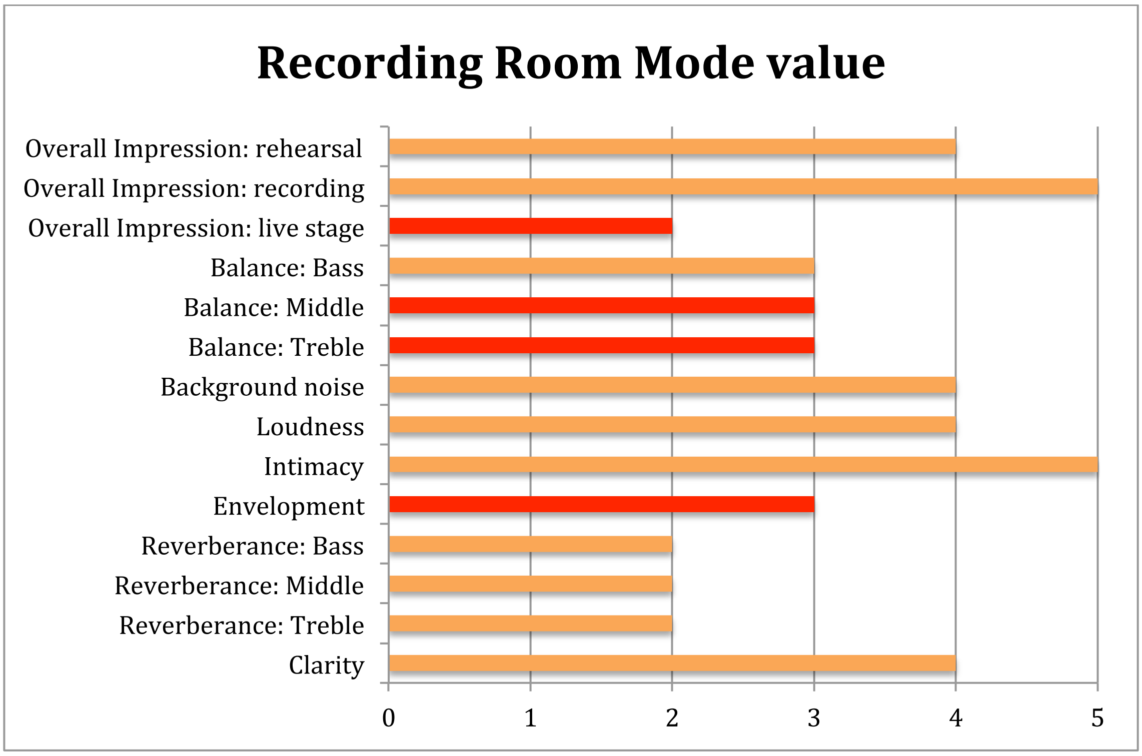


**Figure 46: Mode values of the surveys for the recording room**

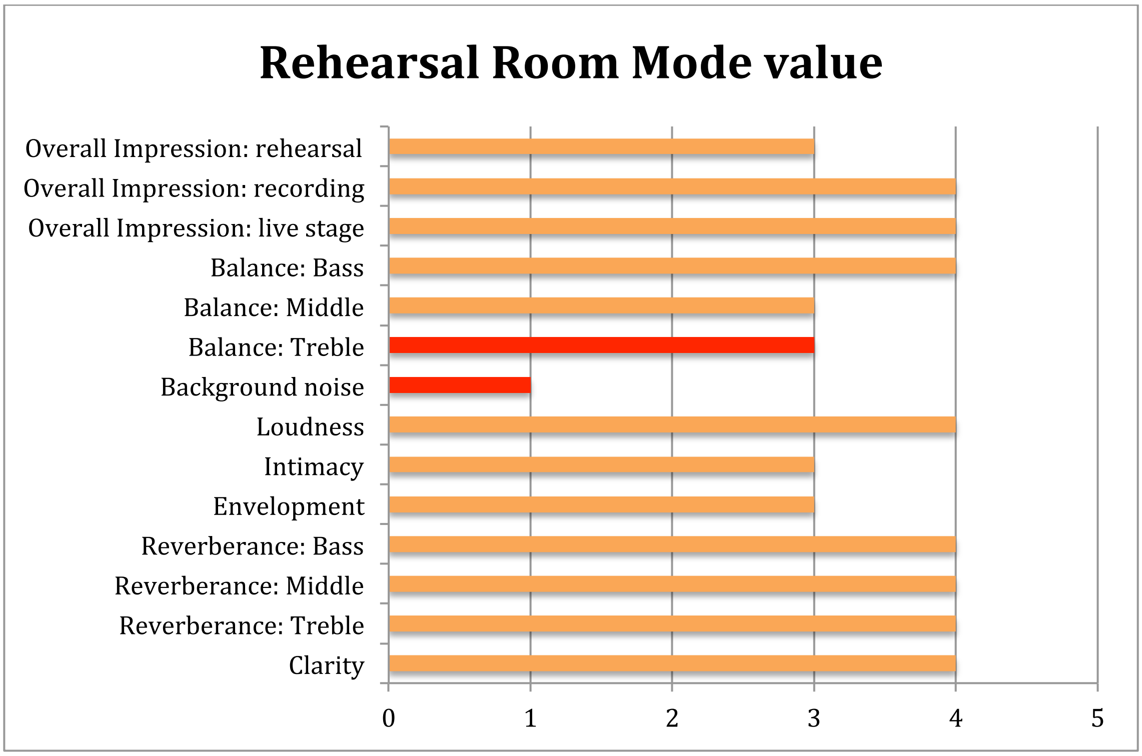


**Figure 47: Mode values of the surveys for the rehearsal room**

## 5.3 Correlation and analysis

### *5.3.1 Chorus Chamber*

In Figure 19 it can be observed that in general, the reverberation in the Chorus Chamber is quite low, and according to the surveys this was also perceived. This room is very big and quite dry, and besides it had the curtains closed and chairs in it, which absorbed more the sound. This room is more appropriate for loud sounds, produced either with many instruments, as with loud ones such as the snare or the cello. In general, the musicians felt that some sound was lost in the room; the bass player felt he did not have enough return and feedback, while the alone guitar player complained that it was very difficult for him to fill the whole room, and because of its big dimensions it does not vibrate with his voice. In general is a good qualified room; it is preferred for rehearsal than for live concerts or recordings, which had divided opinions.

### *5.3.2 Organ Hall*

As it was observed in Figure 20, the reverberation we measured for the organ hall was quite high compared to the other halls, and was more prominent in the middle range of frequencies. This occurred because of the Helmholtz resonator phenomena explained in section 5.1.1 and in Figure 41 can be observed that the musicians perceived it, the reverb was perceived higher for the middle range of frequencies. The bass player found uncomfortable that the pipes absorbed his sound, he felt a lot of reverberation but he didn't felt he had enough feedback of what he was playing. He felt the sound was reflected in the marble and got lost in the pipes, just as we measured and explained before. The amount of reverb was annoying while playing, so in general this room got bad qualifications. It is preferred for a live scenario than for recording or rehearsing. The intimacy and the clarity were lost with the high reverb.

### *5.3.3 Recording Studio*

In the measurements we took, the recording studio was the driest room, as can be observed in Figure 23. In Figure 42 can be seen that the reverberance is qualified low for all the musicians, but the bass player felt good return as he classified it with more reverberance for the low frequencies. Both guitar/voice musicians thought that the room was good in general; it was not excessively dry and vibrated with the voice as they were singing. For the string instruments, such as the cello, the guitars and the bass, they felt instrument was quite precise, harsh and difficult to control. The dynamics of the room are wide; musicians agreed that you could play softly and hear very well the pianissimos, while the room vibrates when you play loud. The intimacy and clarity were very good qualified for the musicians. In general it is a good qualified room; preferred for recording than for rehearsing. In general it was not preferred for live concerts because of the lack of envelopment it has and the dry sound.

### *5.3.4 Rehearsal Room*

As was mentioned before, the measurements and the recordings were done without curtains, so this room had a considerable reverb that was measured and perceived by the musicians, as can be observed in Figure 24 and Figure 43. The musicians in general thought this reverberance was pleasant, because it vibrated with the music, and gave a good sense of envelopment and loudness. The alone guitar player thought the room rounded the sound and he had good return of what he was singing. Other musicians, such as the cello and the snare players, complained that the loudness of the room was too big because of the reverb. This room had very bad isolation and musicians rehearsing in other rooms could be heard. All the musicians found this annoying and complained in the surveys, as can be seen in the bad qualifications for the background noise and for the intimacy. This room was preferred for rehearsing, or playing live, but the musicians did not like to record in it.

# Chapter 6: Conclusions and future work

## 6.1 Conclusions and analysis of the correlation

It is possible to relate and conclude a lot from the opinions of the musicians about the measured rooms. Reverb, intimacy, and when can it be nice or disturbing and why.

According to the characteristics of the rooms, they have different purposes that were preferred for the musicians, such as playing live, recording or rehearsing. The rooms with more reverb used to have better return, more loudness and were preferred to play live, while the dry rooms were preferred to record, because they sound more “pure” and allows to work better with the recordings.

The chorus chamber had big dimensions and low reverberation. It was preferred for the musicians who played louder instruments, while the low volume instruments felt they did not have enough return. It is important to take into account the size of the room, its reverb and what is going to be played in it

Marble walls increase the reverberation of the room, while a pipe organ located in it absorbs the low range of frequencies. This causes a room with a bright reverberation, and poor return for the bass frequencies. Pipe organs at rooms usually act like Helmholtz resonators, absorbing the low frequencies.

In general, the isolation of the rooms is very important to the intimacy and comfort of the musicians playing in them. At the rehearsal room they all complained of hearing other musicians rehearsing outside. This fact devolved them and damaged the quality of the music played. When designing a room, the isolation is a very important issue to take into account for every purpose.

The plywood material is better absorber for the low frequencies than for the middle and high frequencies. This causes reverberations more bright but natural, and the musicians happen to like them more than the marble ones. They felt comfortable, as if the reverberation was enhancing the music with more return. It works both as an absorber and as a resonator.

The more reverberant a room is, the less linearity between the reference and the measured signals. Because of the reverberation, the signals are more out of phase, and hence the coherence trace is lower

The delay compensation is very important to take measures in the direct field, to avoid the differences of phases due to the distance between the microphone and the loudspeaker. Nevertheless, when the measure is made in the diffuse field, this compensation is not relevant because the microphone is only capturing the reverberations of the room. Anyway it is recommended to use it, as it is a good practice.

The phase can be measured in group delay or in degrees. Both scales show the information of phase between the measured signal and the reference, but in different scales with different meanings. Any of them can be used to understand the behavior of the room, by analyzing the meaning of the trace for the different frequencies.

## 6.2 Future work

As future work, in order to improve the work we have done and get better and most reliable results, it would be very interesting to analyze the opinions of more people. We could work with more acoustic experts who listen to artists playing in the rooms and give their opinion and preferences. This would allow us to generate more accurate statistics.

Also it would be very interesting to analyze more musicians who played different musical genres, such as electric rock, or classical music, and analyze how the preferences of the rooms are related to the genres and its characteristics.

It would be interesting also to analyze more rooms, not only rooms in the ESMUC but in other schools, and compare them to have a better understanding of how a room should be designed according to its purpose.

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# APPENDIX A: ITU-R BS.1284-1 RECOMMENDATIONS

The ITU-R BS.1284-1 recommends testing the quality of an acoustic room by making an experiment and managing the results using statistical methods. The attributes of the experiment should be assessed using one of the five-grade scales showed in the following table. The nature and purpose of the tests usually determine which one of the two scales is more the appropriate.

| Quality | | Impairment | |
|---|---|---|---|
| 5 | Excellent | 5 | Imperceptible |
| 4 | Good | 4 | Perceptible, but not annoying |
| 3 | Fair | 3 | Slightly annoying |
| 2 | Poor | 2 | Annoying |
| 1 | Bad | 1 | Very annoying |

The scales should be treated as continuous with a resolution of 1 decimal place. The sessions of the listeners to test the quality of the sound should last at least 15 or 20 minutes without interruption, and they must be consecutive or separated by rest periods of the same length.

The subjective data should be processed and it is important to get the mean values, to describe the data and discriminate it to satisfy the objectives of the test. If it doesn't satisfy them, it is important to carry out further processing.

The presentation of the data should be done in a way that an expert or naive readers are both able to evaluate the relevant information. Graphical forms to present the data are preferred. Detailed quantitative information and numerical analyses should be presented in appendices. To present the mean values gives a good initial overview of the data.

As far as possible, all the aspects of the test should be reported (For example if no training was carried out for the listeners). The reports should be as clear as possible, and they must show the rationale of the study, the methods used and the conclusions drawn. It is important to put special attention to the following (ITU-R BS.1284-1 Recommendation, 2003):

- The selection of subjects and excerpts.
- The physical details of the listening environment and equipment.

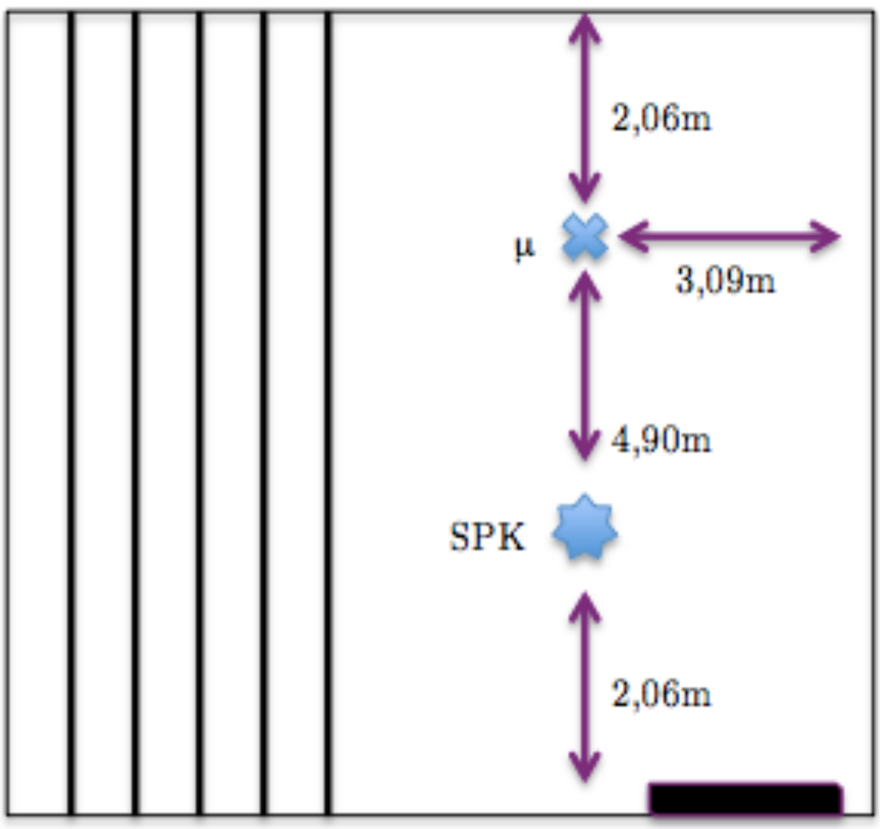
- The experimental design, surveys, training, instructions, sequences, procedures and data generation.
- The processing of data, descriptive and analytic inferential statistics.
- The basis of all the conclusions that are drawn.

# Appendix B: Location of the microphone and the loudspeaker for each room

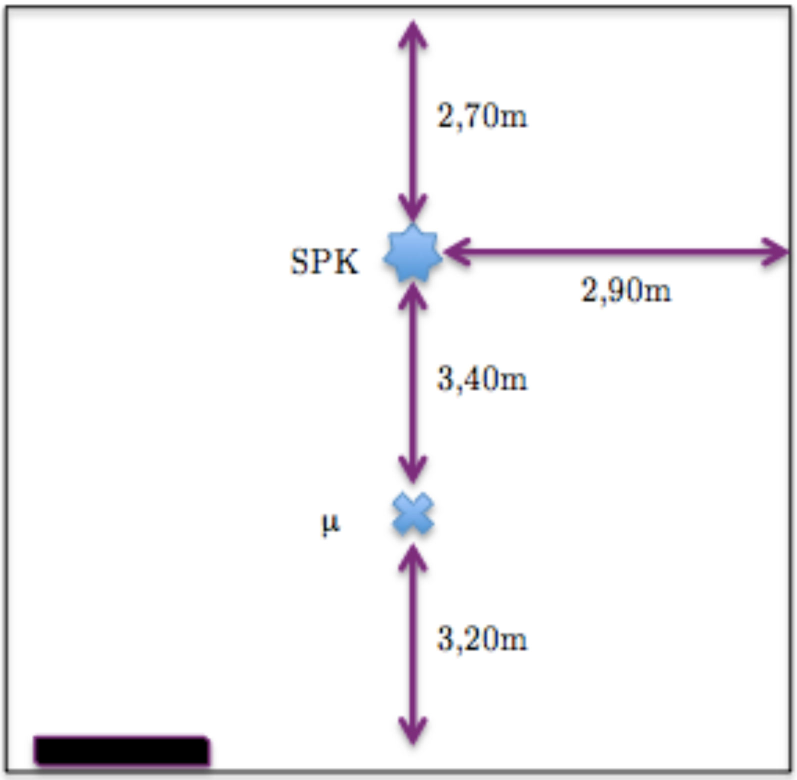
As we have mentioned before, when we measured the rooms, all the positions of the microphone and loudspeaker were annotated, in order to have a similar configuration later, when the recordings were carried out. Where the omnidirectional microphone was located in the measurements, we positioned the center of the AB configuration in the recordings. Where the loudspeaker was located in the measurements, we positioned the ORTF microphone, and the musicians sat around it.

In this appendix, the configurations are shown for each one of the rooms. The X represents the microphone, and the star represents the loudspeaker.

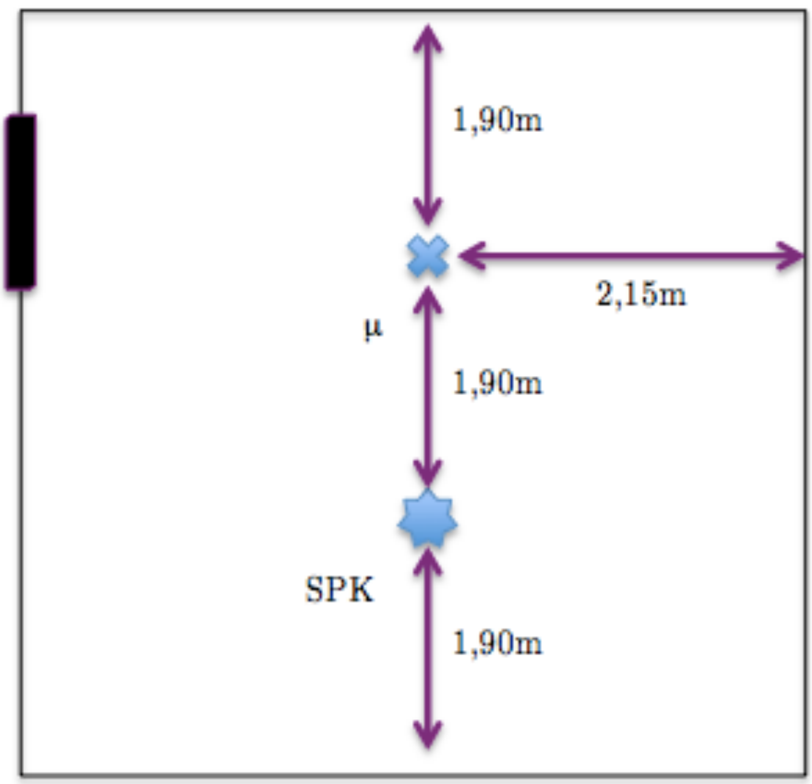
**Chorus chamber:**



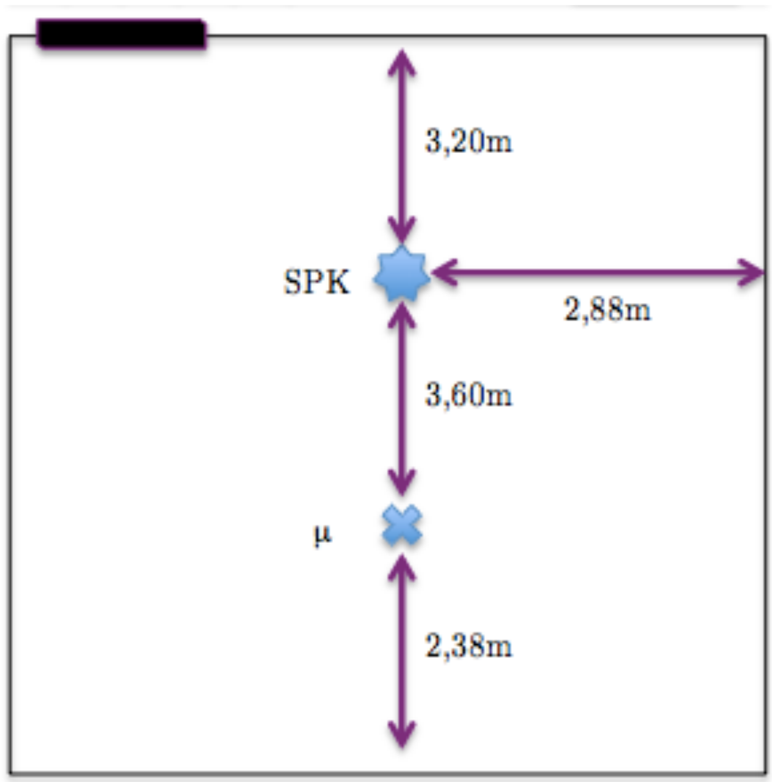
**Organ Hall:**




**Recording Studio:**



**Rehearsal Room:**



# Appendix C: Reverberation time measurement; complete results

In this appendix is attached the complete results for the measurements of the reverberation time, using the Reverberation_Time_Calculator Matlab® tool and its reverb_time function. In chapter 5 we only analyzed the frequencies above 100Hz, but here we can see the complete results that the software returned and its plots.

**Chorus chamber:**

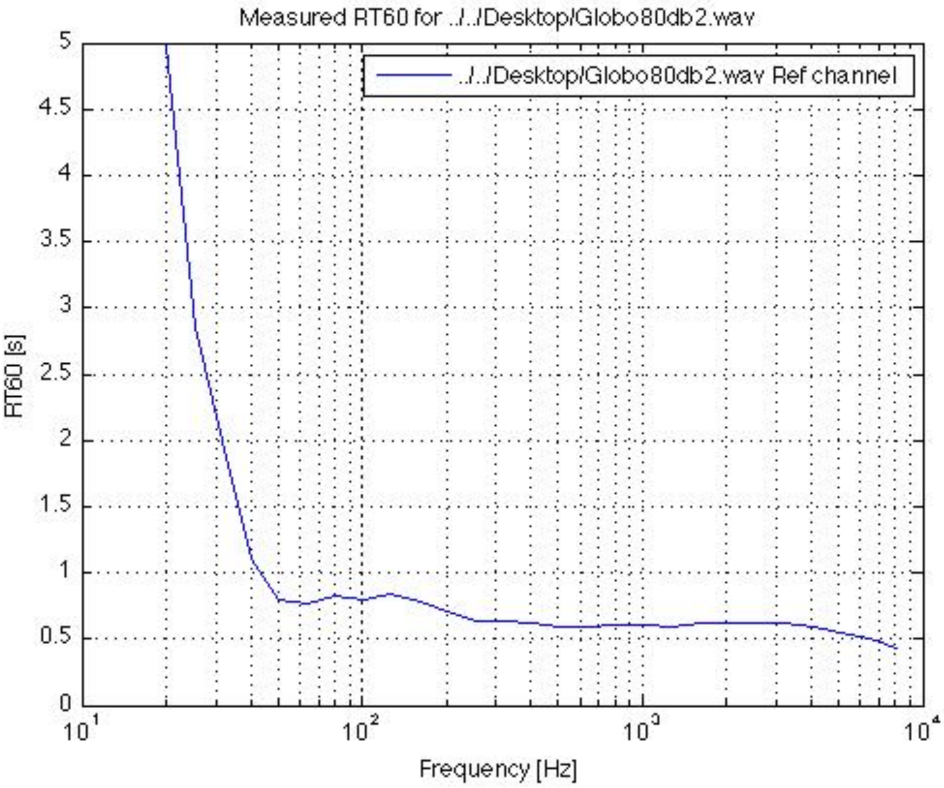
Analyzing the balloon pop recording:

Band 1; f=20Hz -> 4.9821s
Band 2; f=25Hz -> 2.5663s
Band 3; f=31Hz -> 1.044s
Band 4; f=40Hz -> 0.69775s
Band 5; f=50Hz -> 0.72376s
Band 6; f=63Hz -> 0.46418s
Band 7; f=80Hz -> 1.0617s
Band 8; f=100Hz -> 0.92147s
Band 9; f=126Hz -> 0.94848s
Band 10; f=160Hz -> 0.59682s
Band 11; f=201Hz -> 0.67104s
Band 12; f=253Hz -> 0.73813s
Band 13; f=320Hz -> 0.62034s
Band 14; f=403Hz -> 0.53057s
Band 15; f=507Hz -> 0.63156s
Band 16; f=640Hz -> 0.59365s
Band 17; f=806Hz -> 0.55116s
Band 18; f=1015Hz -> 0.62588s
Band 19; f=1280Hz -> 0.61088s
Band 20; f=1612Hz -> 0.62597s
Band 21; f=2031Hz -> 0.58267s
Band 22; f=2560Hz -> 0.67956s
Band 23; f=3225Hz -> 0.60876s
Band 24; f=4063Hz -> 0.6034s

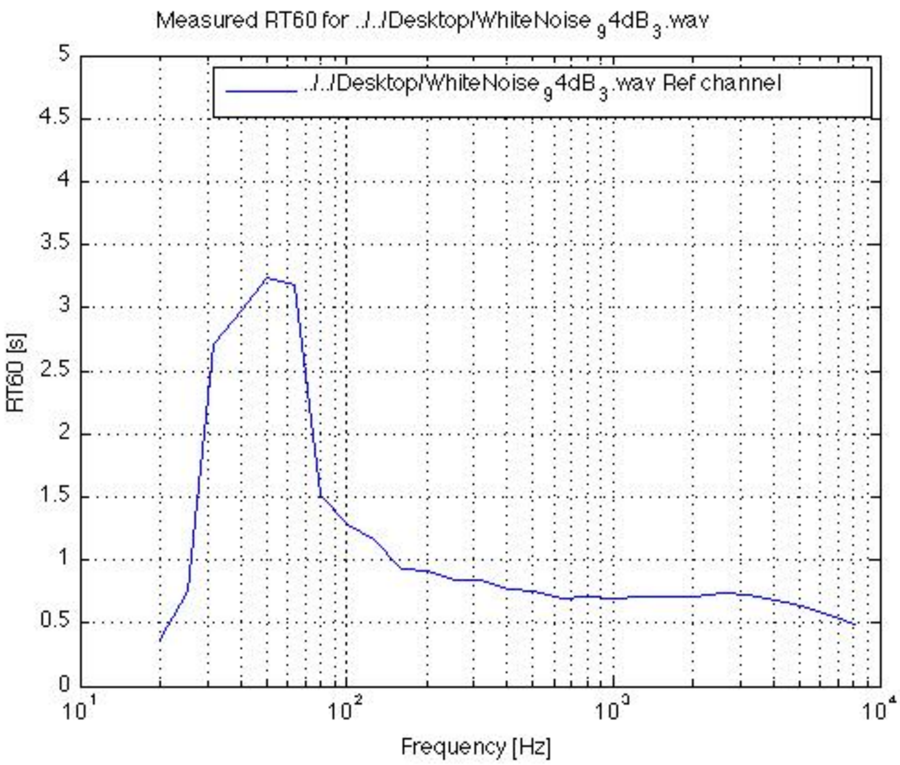
Band 25; f=5120Hz -> 0.59595s
Band 26; f=6450Hz -> 0.50035s
Band 27; f=8127Hz -> 0.42521s

Analyzing the white noise recording
Band 1; f=20Hz -> 0.37669s
Band 2; f=25Hz -> 0.59685s
Band 3; f=31Hz -> 1.3069s
Band 4; f=40Hz -> 9.3322s
Band 5; f=50Hz -> 1.931s
Band 6; f=63Hz -> 1.6814s
Band 7; f=80Hz -> 1.9169s
Band 8; f=100Hz -> 1.068s
Band 9; f=126Hz -> 0.95033s
Band 10; f=160Hz -> 0.77775s
Band 11; f=201Hz -> 1.0627s
Band 12; f=253Hz -> 0.76509s
Band 13; f=320Hz -> 1.0063s
Band 14; f=403Hz -> 0.59854s
Band 15; f=507Hz -> 0.80671s
Band 16; f=640Hz -> 0.64532s
Band 17; f=806Hz -> 0.71654s
Band 18; f=1015Hz -> 0.68834s
Band 19; f=1280Hz -> 0.69398s
Band 20; f=1612Hz -> 0.72707s
Band 21; f=2031Hz -> 0.72278s
Band 22; f=2560Hz -> 0.70773s
Band 23; f=3225Hz -> 0.66856s
Band 24; f=4063Hz -> 0.8385s
Band 25; f=5120Hz -> 0.64851s
Band 26; f=6450Hz -> 0.56366s
Band 27; f=8127Hz -> 0.4922s

Plot for the balloon pop:




Plot for the white noise




**Organ Hall:**

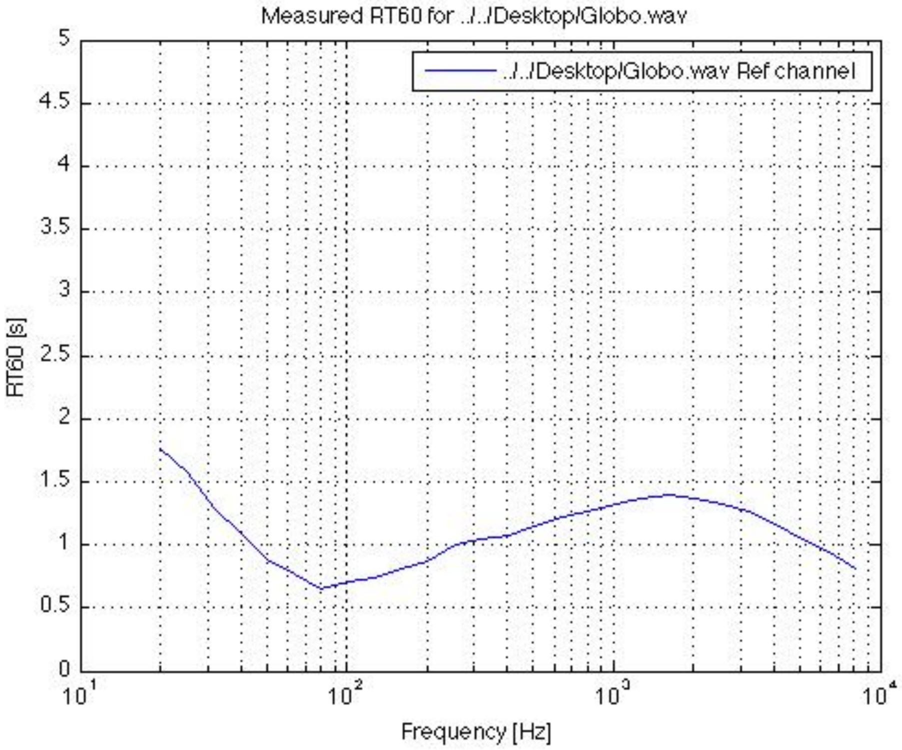
Analyzing the balloon pop recording:

Band 1; f=20Hz -> 1.7642s
Band 2; f=25Hz -> 1.695s
Band 3; f=31Hz -> 1.2508s
Band 4; f=40Hz -> 1.1007s
Band 5; f=50Hz -> 0.6237s
Band 6; f=63Hz -> 0.80091s
Band 7; f=80Hz -> 0.61688s
Band 8; f=100Hz -> 0.70086s
Band 9; f=126Hz -> 0.49169s
Band 10; f=160Hz -> 0.9465s
Band 11; f=201Hz -> 0.96689s
Band 12; f=253Hz -> 0.93843s
Band 13; f=320Hz -> 1.0218s
Band 14; f=403Hz -> 1.0988s
Band 15; f=507Hz -> 1.2091s
Band 16; f=640Hz -> 1.1106s
Band 17; f=806Hz -> 1.2659s
Band 18; f=1015Hz -> 1.4152s
Band 19; f=1280Hz -> 1.3485s
Band 20; f=1612Hz -> 1.4462s
Band 21; f=2031Hz -> 1.3925s
Band 22; f=2560Hz -> 1.3713s
Band 23; f=3225Hz -> 1.2448s
Band 24; f=4063Hz -> 1.1953s
Band 25; f=5120Hz -> 1.1066s
Band 26; f=6450Hz -> 0.89556s
Band 27; f=8127Hz -> 0.81708s

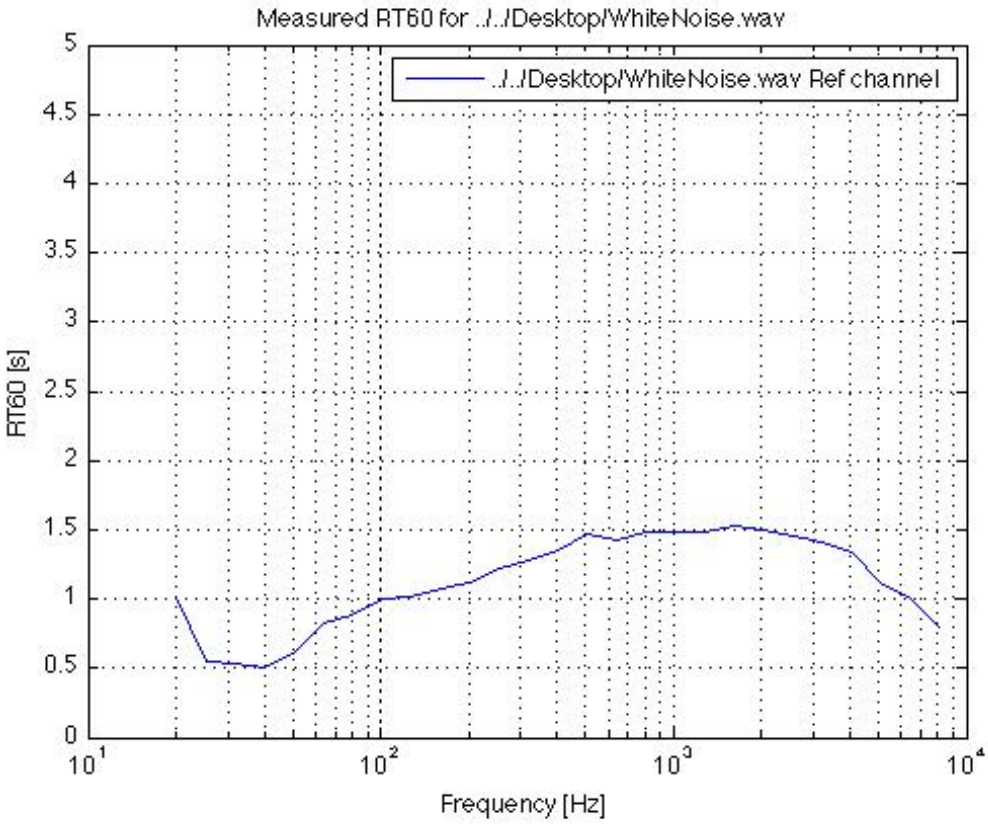
Analyzing the white noise recording:

Band 1; f=20Hz -> 1.0109s
Band 2; f=25Hz -> 0.27301s
Band 3; f=31Hz -> 0.35332s
Band 4; f=40Hz -> 0.40863s
Band 5; f=50Hz -> 0.60233s
Band 6; f=63Hz -> 0.88021s
Band 7; f=80Hz -> 0.80944s
Band 8; f=100Hz -> 1.4031s
Band 9; f=126Hz -> 0.6927s
Band 10; f=160Hz -> 1.1976s
Band 11; f=201Hz -> 0.98003s
Band 12; f=253Hz -> 1.0805s
Band 13; f=320Hz -> 1.625s
Band 14; f=403Hz -> 1.2356s
Band 15; f=507Hz -> 1.4414s
Band 16; f=640Hz -> 1.3722s
Band 17; f=806Hz -> 1.6502s
Band 18; f=1015Hz -> 1.4404s
Band 19; f=1280Hz -> 1.539s
Band 20; f=1612Hz -> 1.4399s
Band 21; f=2031Hz -> 1.3807s
Band 22; f=2560Hz -> 1.8296s
Band 23; f=3225Hz -> 1.319s
Band 24; f=4063Hz -> 1.3142s
Band 25; f=5120Hz -> 1.2222s
Band 26; f=6450Hz -> 0.96916s
Band 27; f=8127Hz -> 0.79881s

Plot for the balloon pop




Plot for the white noise



**Recording Studio:**

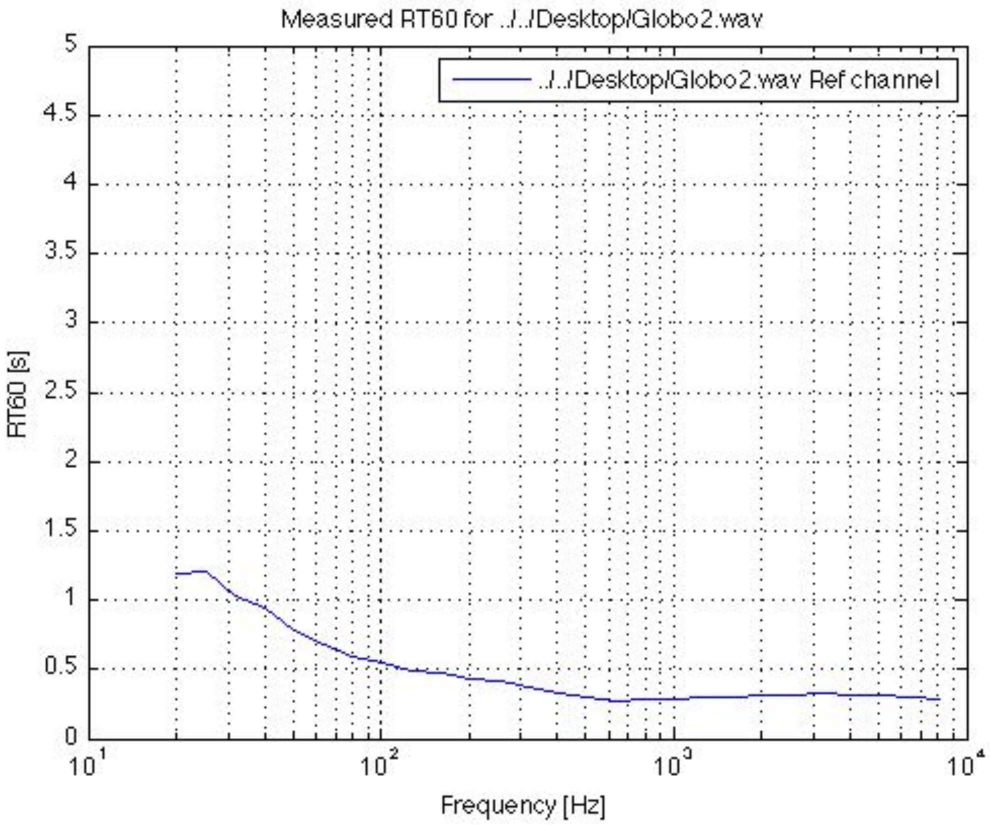
Analyzing the balloon pop recording:

Band 1; f=20Hz -> 1.1899s
Band 2; f=25Hz -> 1.3267s
Band 3; f=31Hz -> 1.1117s
Band 4; f=40Hz -> 0.77309s
Band 5; f=50Hz -> 0.73879s
Band 6; f=63Hz -> 0.77019s
Band 7; f=80Hz -> 0.51797s
Band 8; f=100Hz -> 0.56549s
Band 9; f=126Hz -> 0.38887s
Band 10; f=160Hz -> 0.50917s
Band 11; f=201Hz -> 0.48954s
Band 12; f=253Hz -> 0.39846s
Band 13; f=320Hz -> 0.38797s
Band 14; f=403Hz -> 0.27025s
Band 15; f=507Hz -> 0.30645s
Band 16; f=640Hz -> 0.2627s
Band 17; f=806Hz -> 0.28129s
Band 18; f=1015Hz -> 0.2649s
Band 19; f=1280Hz -> 0.3231s
Band 20; f=1612Hz -> 0.29576s
Band 21; f=2031Hz -> 0.30212s
Band 22; f=2560Hz -> 0.31742s
Band 23; f=3225Hz -> 0.33379s
Band 24; f=4063Hz -> 0.34825s
Band 25; f=5120Hz -> 0.31259s
Band 26; f=6450Hz -> 0.29202s
Band 27; f=8127Hz -> 0.29202s

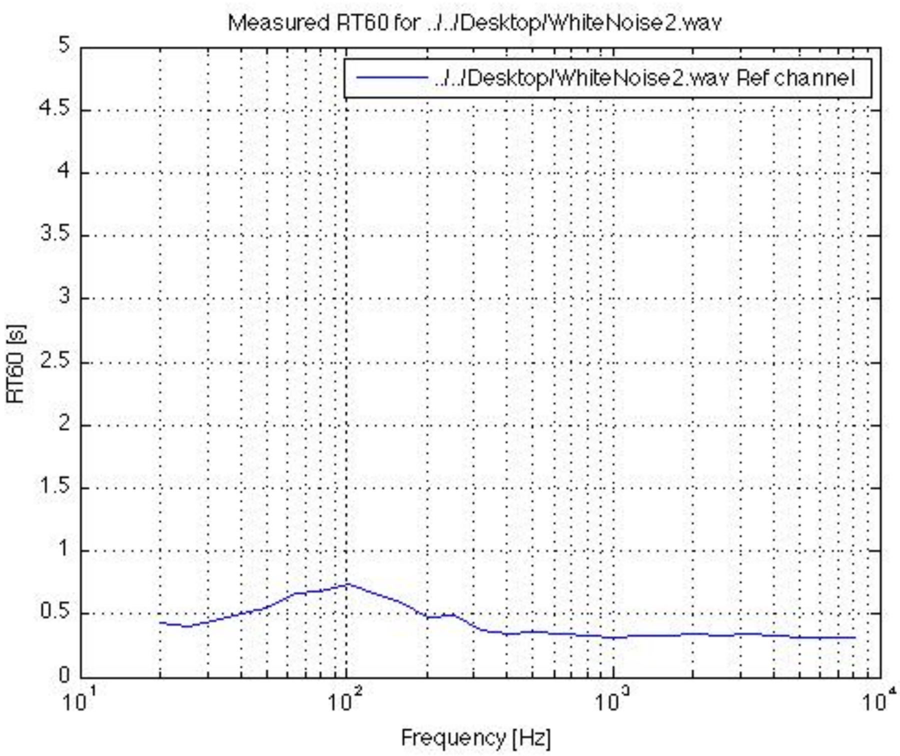
Analyzing the white noise recording:

Band 1; f=20Hz -> 0.43237s
Band 2; f=25Hz -> 0.46587s
Band 3; f=31Hz -> 0.29383s
Band 4; f=40Hz -> 0.34361s
Band 5; f=50Hz -> 0.68163s
Band 6; f=63Hz -> 0.73204s
Band 7; f=80Hz -> 0.67891s
Band 8; f=100Hz -> 0.91891s
Band 9; f=126Hz -> 0.38676s
Band 10; f=160Hz -> 0.96245s
Band 11; f=201Hz -> 0.40594s
Band 12; f=253Hz -> 0.27957s
Band 13; f=320Hz -> 0.36741s
Band 14; f=403Hz -> 0.44291s
Band 15; f=507Hz -> 0.35726s
Band 16; f=640Hz -> 0.29787s
Band 17; f=806Hz -> 0.34308s
Band 18; f=1015Hz -> 0.27477s
Band 19; f=1280Hz -> 0.35474s
Band 20; f=1612Hz -> 0.31654s
Band 21; f=2031Hz -> 0.35749s
Band 22; f=2560Hz -> 0.35286s
Band 23; f=3225Hz -> 0.32702s
Band 24; f=4063Hz -> 0.31511s
Band 25; f=5120Hz -> 0.33384s
Band 26; f=6450Hz -> 0.28437s
Band 27; f=8127Hz -> 0.31837s

Plot for the balloon pop




**Plot for the white noise**



**Rehearsal Room:**

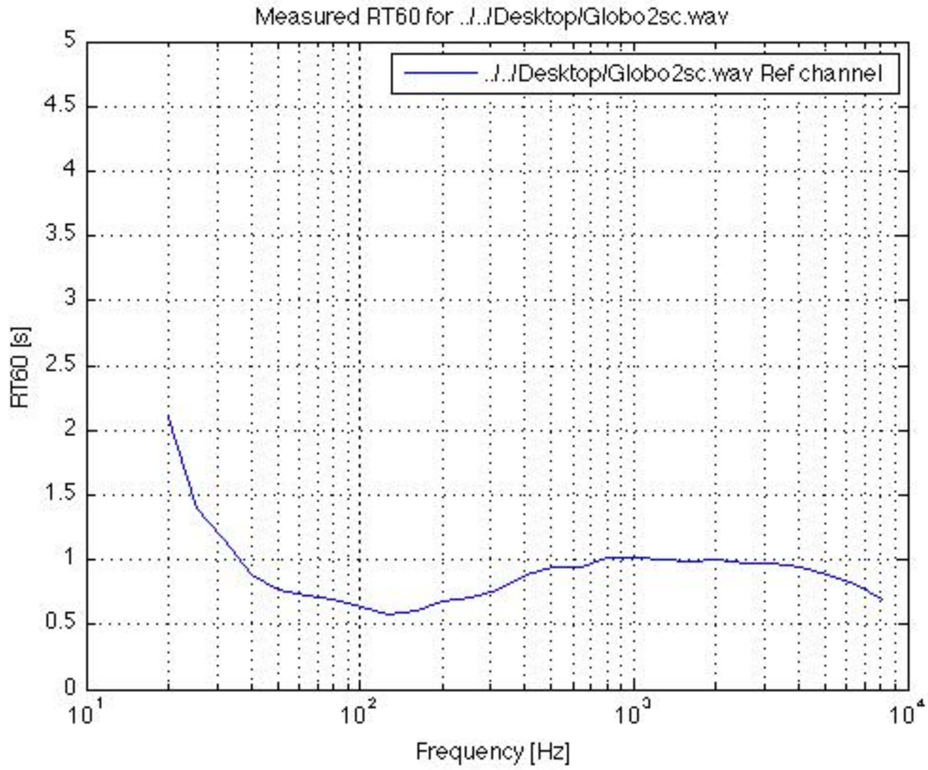
Analyzing the balloon pop recording:

Band 1; f=20Hz -> 2.0949s
Band 2; f=25Hz -> 1.2237s
Band 3; f=31Hz -> 0.87269s
Band 4; f=40Hz -> 0.65755s
Band 5; f=50Hz -> 0.93183s
Band 6; f=63Hz -> 0.7581s
Band 7; f=80Hz -> 0.60934s
Band 8; f=100Hz -> 0.64222s
Band 9; f=126Hz -> 0.51926s
Band 10; f=160Hz -> 0.64015s
Band 11; f=201Hz -> 0.48495s
Band 12; f=253Hz -> 0.71601s
Band 13; f=320Hz -> 1.0312s
Band 14; f=403Hz -> 0.67751s
Band 15; f=507Hz -> 0.95698s
Band 16; f=640Hz -> 1.0237s
Band 17; f=806Hz -> 1.0331s
Band 18; f=1015Hz -> 1.0243s
Band 19; f=1280Hz -> 1.0382s
Band 20; f=1612Hz -> 0.93146s
Band 21; f=2031Hz -> 0.94785s
Band 22; f=2560Hz -> 0.98374s
Band 23; f=3225Hz -> 1.0746s
Band 24; f=4063Hz -> 0.93747s
Band 25; f=5120Hz -> 0.91172s
Band 26; f=6450Hz -> 0.81607s
Band 27; f=8127Hz -> 0.69413s

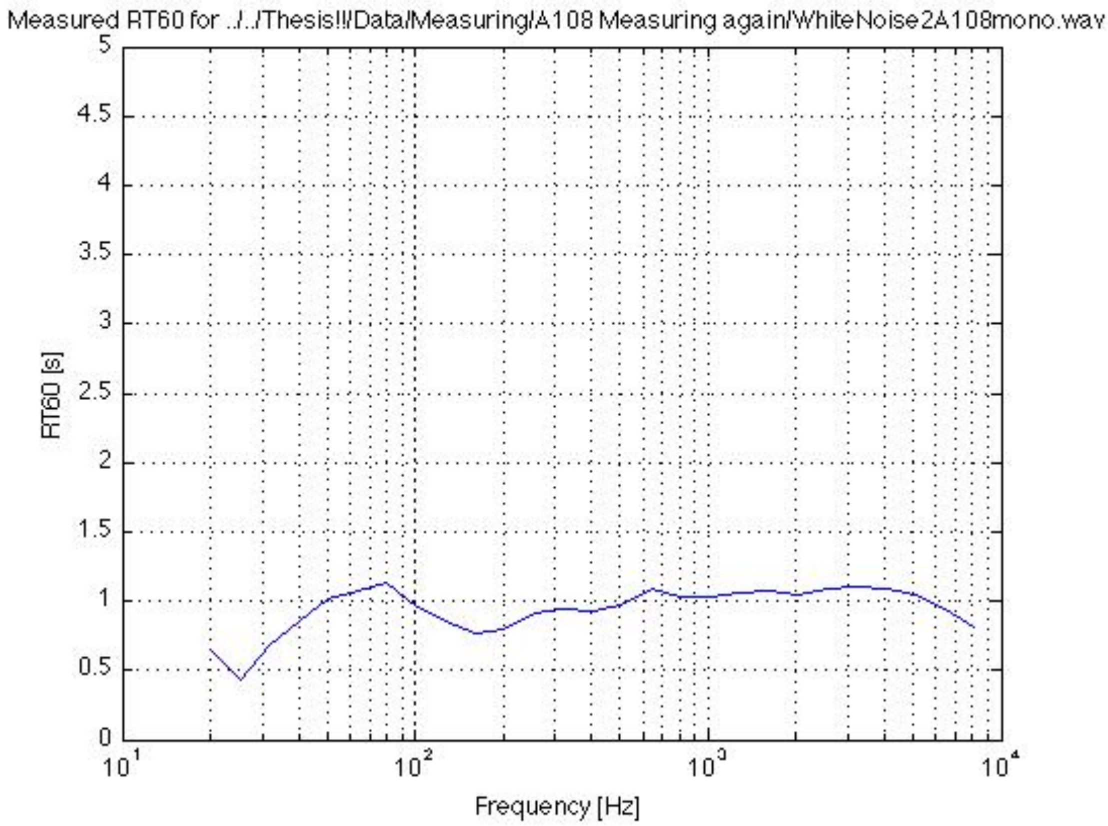
Analyzing the white noise recording

Band 1; f=20Hz -> 0.64893s
Band 2; f=25Hz -> 0.364s
Band 3; f=31Hz -> 0.27684s
Band 4; f=40Hz -> 0.35823s
Band 5; f=50Hz -> 1.7568s
Band 6; f=63Hz -> 1.4987s
Band 7; f=80Hz -> 1.1911s
Band 8; f=100Hz -> 0.56101s
Band 9; f=126Hz -> 0.64257s
Band 10; f=160Hz -> 0.86826s
Band 11; f=201Hz -> 1.0192s
Band 12; f=253Hz -> 0.74494s
Band 13; f=320Hz -> 0.71286s
Band 14; f=403Hz -> 1.2303s
Band 15; f=507Hz -> 1.0043s
Band 16; f=640Hz -> 0.92416s
Band 17; f=806Hz -> 0.98142s
Band 18; f=1015Hz -> 1.2892s
Band 19; f=1280Hz -> 0.96773s
Band 20; f=1612Hz -> 1.0222s
Band 21; f=2031Hz -> 1.0217s
Band 22; f=2560Hz -> 1.0379s
Band 23; f=3225Hz -> 1.1781s
Band 24; f=4063Hz -> 1.217s
Band 25; f=5120Hz -> 1.0781s
Band 26; f=6450Hz -> 0.93315s
Band 27; f=8127Hz -> 0.80635s

Plot for the balloon pop




Plot for the white noise



# Appendix D: Matlab® source codes

To calculate the reverberation time, we used the following Matlab® function that works with the Reverberation_Time_Calculator Matlab® tool. There we had the visualization of the RT as is showed in Appendix C: Reverberation time measurement; complete results.

```
function rta=RT60_Lina(filename,method)
%This function calculates the Reverberation time for different bands of
%frequency, displays the values in the command window and plots it.
%input parameters:
%filname: name of the file to analyze
%method: 1 for speaker, otherwise for balloon pop

%Defining variables-----------------------------------------------------------

N_BANDS=27; %number of bands
f0=20; %lowest frequency to analyze

t_high=0.5;
t_low=0.5;
num_x_filter=2;
pbs=15;
pab=15;

disp(['--> Computing file ' filename]);
[x,Fs]=wavread(filename);
[len,channels]=size(x);

%Calculating the RT for each band------------------------------------------


for i=1:N_BANDS,

        fc(i)=f0*2^((i-1)/3);
        clear tmp;
        [rta(:,i), Lya, tmp]=reverb_time(x, Fs, fc(i), t_high, t_low, ...
            num_x_filter,pbs,pab, method); %calculates RT
        disp(['     Band ',num2str(i),'; f=',num2str(floor(fc(i))), ...
            'Hz -> ',num2str(rta(1,i)),'s']); %displays the RT in cmd

end

%Visualization ------------------------------------------------------------

%%%For Mono
semilogx(fc,smooth(rta(1,1:N_BANDS)));
grid on;
title(['Measured RT60 for ',filename]);
xlabel('Frequency [Hz]');
ylabel('RT60 [s]');
axis([10 10000 0 5]);

end %end of the function
```

To compare the values with white noise, balloon pop and the theoretical data, we developed the following Matlab® script. We developed a different script for each room, but I attach here only the Chorus Chamber one as an example.

```
%Script for plotting Chorus Chamber
close all;
clear all;

%Defining variables ---------------------------------------------------

N_BANDS=27; %number of band of frequency to analyze
f0=20; %frequency to start analysing

fref=[125,250,500,1000,2000,4000]; %frequencies for theoretical data
theoreticalRT=[1.03,0.92,0.77,0.79,0.75,0.75]; %theoretical data

for i=1:N_BANDS,
    fc(i)=f0*2^((i-1)/3);%Definition of the bands of frequency
end

%Calculating the reverb time--------------------------------------------

CC_G=RT60_Lina('Globo80db1.wav',2);
%Calculating the reverberation time with the balloon pop
CC_WN=RT60_Lina('WhiteNoise82dB.wav',1);
%Calculating the reverberation time with the white noise

%Plotting -------------------------------------------------------------

semilogx(fc,smooth(CC_G(1,1:N_BANDS)),'r'); %semilogaritmic axes
hold on; %to plot all in the same figure
semilogx(fc,smooth(CC_WN(1,1:N_BANDS)));
hold on;
semilogx(fref,smooth(theoreticalRT),'g');
hold off;

grid on;
title('Chorus Chamber RT60 ');
xlabel('Frequency [Hz]');
ylabel('RT60 [s]');
axis([100 4000 0 2.5]);

legend('Balloon Pop','White Noise', 'Theoretical');
```

It can be observed that besides plotting the white noise, balloon pop and theoretical data in the same figure, we limited the axis only to the part we were interested to analyze.

# Appendix E: Surveys

Surveys for the musicians to fill in each room

**Listener** ______________________ **Date** ______________________

**Room** ______________________ **Genre** ______________________

**Clarity**

Is the degree in which discrete sounds stand apart from each other. Classify the sound: 1 is muddy sound, 5 is clear sound.

| | | | | | | |
|---|---|---|---|---|---|---|
| Muddy | 1 | 2 | 3 | 4 | 5 | Clear |

**Reverberance**

Is when the sound persists in the room after the sound is suddenly stopped. Classify for different bands: 1 is dead room, 5 is live reverberant

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| Treble | Dead | 1 | 2 | 3 | 4 | 5 | Reverberant |
| Middle | Dead | 1 | 2 | 3 | 4 | 5 | Reverberant |
| Bass | Dead | 1 | 2 | 3 | 4 | 5 | Reverberant |

**Envelopment**

Impression of the strength and directions from which the reverberant sound seems to arrive. 1 is directive sound, 5 is equally from all directions

| | | | | | | |
|---|---|---|---|---|---|---|
| Directive | 1 | 2 | 3 | 4 | 5 | Equal |

**Intimacy**

When the music gives the impression of being played in an small hall. Also called presence. 1 is remote sound, 5 is intimate.

| | | | | | | |
|---|---|---|---|---|---|---|
| Remote | 1 | 2 | 3 | 4 | 5 | Inimate |

**Loudness**

How loud sounds the music in each hall. Classify the sound: 1 is lower, 5 is louder

| | | | | | | |
|---|---|---|---|---|---|---|
| Low | 1 | 2 | 3 | 4 | 5 | Loud |

**Background Noise**

How the noise is felt in the background when the music is playing. 1 is intolerable background noise, 5 is inaudible background noise.

| | | | | | | |
|---|---|---|---|---|---|---|
| Intolerable | 1 | 2 | 3 | 4 | 5 | Inaudible |

**Balance**

A good balance of frequencies can also depend on the hall. Please classyfy each range of frequencies, 1 is low level, 5 is louder level.

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| Treble | Low | 1 | 2 | 3 | 4 | 5 | Loud |
| Middle | Low | 1 | 2 | 3 | 4 | 5 | Loud |
| Bass | Low | 1 | 2 | 3 | 4 | 5 | Loud |

**Overall Impression**

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| As a live stage | Bad | 1 | 2 | 3 | 4 | 5 | Good |
| As a recording space | Bad | 1 | 2 | 3 | 4 | 5 | Good |
| As a rehearsal space | Bad | 1 | 2 | 3 | 4 | 5 | Good |

**Hall valoration**

How did you feel in the room. How will you assess it for the purposes you want (live/recording/etc...). How did you feel according to the genre you played. Feel free to write, if you want you can use the back of the sheet

Demographic data of the musicians

**Listener** ______________________ **Date** ______________________

**Male/Female** ______________________ **Age** ______________________

Are you a musician?

Yes ☐ Years of experience ☐
No ☐

Approximately, in average, how many hours a day do you actively listen to music?

Less than 2 ☐
2 hours ☐
5 hours ☐
7 hours ☐
10 hours ☐
More than 10 ☐

Do you consider yourself an active music listener? Pay much attention to music when it is played?

Yes ☐
No ☐

Have you ever had, or do you actually have hearing problems or diseases?

Yes ☐ Which ones? ______________________
No ☐

# Appendix F: Surveys results in detail

To make the plots showed in chapter 5, the following tables were developed, with the results of the surveys for each musician and for each room.

| Subjective Issue Evaluated | Chorus Chamber | | | | |
|---|---|---|---|---|---|
| | Emilio Guitar/Voice | Kibo | | | |
| | | Guitar/ Voice | Cello | Snare | Bass |
| Clarity | 4 | 4 | 5 | 5 | 3 |
| Reverberance: Treble | 3 | 2 | 3 | 2 | 2 |
| Reverberance: Middle | 3 | 2 | 3 | 2 | 2 |
| Reverberance: Bass | 3 | 2 | 3 | 2 | 2 |
| Envelopment | 4 | 4 | 3 | 5 | 4 |
| Intimacy | 2 | 2 | 4 | 1 | 3 |
| Loudness | 4 | 1 | 4 | 1 | 2 |
| Background noise | 4 | 5 | 4 | 4 | 3 |
| Balance: Treble | 4 | 3 | 3 | 4 | 2 |
| Balance: Middle | 4 | 4 | 3 | 3 | 3 |
| Balance: Bass | 4 | 2 | 2 | 2 | 2 |
| Overall Impression: live stage | 4 | 2 | 4 | 4 | 4 |
| Overall Impression: recording | 2 | 4 | 5 | 4 | 3 |
| Overall Impression: rehearsal | 3 | 4 | 5 | 4 | 4 |

| Subjective Issue Evaluated | Organ Hall | | | | |
|---|---|---|---|---|---|
| | Emilio Guitar/Voice | Kibo | | | |
| | | Guitar/ Voice | Cello | Snare | Bass |
| Clarity | 3 | 3 | 3 | 3 | 3 |
| Reverberance: Treble | 4 | 2 | 3 | 5 | 4 |
| Reverberance: Middle | 4 | 3 | 4 | 4 | 4 |
| Reverberance: Bass | 3 | 4 | 4 | 3 | 3,5 |
| Envelopment | 4 | 4 | 4 | 3 | 4 |
| Intimacy | 2 | 2 | 2 | 2 | 2 |
| Loudness | 3 | 4 | 3 | 2 | 3 |
| Background noise | 4 | 4 | 3 | 5 | 4 |
| Balance: Treble | 4 | 2 | 3 | 5 | 4 |
| Balance: Middle | 3 | 4 | 3 | 3 | 4 |
| Balance: Bass | 3 | 3 | 3 | 2 | 3 |
| Overall Impression: live stage | 3 | 1 | 5 | 4 | 4 |
| Overall Impression: recording | 1 | 1 | 4 | 1 | 2 |
| Overall Impression: rehearsal | 2 | 3 | 4 | 2 | 2 |

| Subjective Issue Evaluated | Recording Room | | | | |
|---|---|---|---|---|---|
| | Emilio Guitar/Voice | Kibo | | | |
| | | Guitar/Voice | Cello | Snare | Bass |
| Clarity | 4 | 4 | 5 | 5 | 4 |
| Reverberance: Treble | 2 | 1,5 | 2 | 1 | 2 |
| Reverberance: Middle | 2 | 1,5 | 2 | 1 | 2 |
| Reverberance: Bass | 2 | 1,5 | 2 | 1 | 3 |
| Envelopment | 2 | 3 | 1 | 5 | 3 |
| Intimacy | 4 | 4 | 5 | 5 | 5 |
| Loudness | 4 | 3 | 4 | 5 | 4 |
| Background noise | 5 | 4 | 4 | 5 | 4 |
| Balance: Treble | 3 | 4 | 3 | 5 | 5 |
| Balance: Middle | 3 | 4 | 3 | 5 | 5 |
| Balance: Bass | 3 | 3 | 3 | 5 | 4 |
| Overall Impression: live stage | 2 | 5 | 2 | 3 | 3 |
| Overall Impression: recording | 5 | 5 | 5 | 5 | 4 |
| Overall Impression: rehearsal | 4 | 5 | 3 | 5 | 4 |

| Subjective Issue Evaluated | Rehearsal Room | | | | |
|---|---|---|---|---|---|
| | Emilio Guitar/Voice | Kibo | | | |
| | | Guitar/Voice | Cello | Snare | Bass |
| Clarity | 4 | 2 | 4 | 4 | 4 |
| Reverberance: Treble | 3 | 3 | 4 | 4 | 4 |
| Reverberance: Middle | 3 | 4 | 4 | 4 | 4 |
| Reverberance: Bass | 3 | 4 | 4 | 4 | 4 |
| Envelopment | 3 | 4 | 3 | 3 | 4 |
| Intimacy | 3 | 4 | 3 | 1 | 3 |
| Loudness | 4 | 4 | 4 | 5 | 4 |
| Background noise | 1 | 3 | 3 | 1 | 2 |
| Balance: Treble | 3 | 3 | 4 | 2 | 4 |
| Balance: Middle | 3 | 3 | 4 | 3 | 3 |
| Balance: Bass | 3 | 4 | 3 | 4 | 4 |
| Overall Impression: live stage | 4 | 2 | 3 | 4 | 4 |
| Overall Impression: recording | 4 | 2 | 4 | 2 | 1 |
| Overall Impression: rehearsal | 5 | 3 | 4 | 3 | 3 |

## Evaluation of the musicians about the rooms

### *Chorus Chamber*

**Emilio Guitar/Voice:** In general the sound is very pleasant, but when you sang with the face pointing to the wall you could feel a strange reflection. Nevertheless the sound is quite sharp but natural at the same time. In the other hand, as the room is too big, it is difficult to "fill" it only with one voice. The room does not vibrate as the recording room did. In any case is much better than the organ hall for all the purposes.

**Kibo Cello:** For my taste, is the best one of the rooms in an acoustic level. It has the exact quantity of reverberation to do not bother but does not sound dry either. The basses are lost because the great dimensions of the room.

**Kibo Snare:** The sound is dispersed because the room is very big, and that makes it good for instruments with great volume level.

**Kibo Bass:** Because the room is very big, and with the curtains closed it is quite dry, it does not help to the sound of the bass. The low frequency sounds are quite slow, and they are lost in this room. I did not have good feedback.

### *Organ Hall*

**Emilio Guitar/Voice:** There's a type of reverb that is not confortable to sing in this room. Some consonants, such as the s and the r, result in strange echoes and you have the sensation of not filling the room properly. The acoustics of the room does not help to feel confortable in her for this intimate style of music. The fact of singing headed to the marble wall might have affected the sound, because a metallic echo was felt, exaggerating the sound in some consonants, specially the s. In the other hand is that due to the reverb you feel more "live" and not so much in a room special for recording. Though, this would not be the room I would choose for a concert. Another detail: In a concert only with guitar, without the voice, this room would be more appropriate because it rounds the sound and has less aggressiveness. I had the sensation that the sound didn't respond well to the dynamics of my voice. If I sing loud or soft I sounded in a similar way due to the acoustics of the room.

**Kibo Guitar/Voice:** In general this is a bad room. The reverb is not too good. I did not use it that much and the sound gets a little dirty.

**Kibo Cello:** It is pleasant to play in this room. Maybe the reverberation is a little excessive. Could be good to live concerts or rehearsals.

**Kibo Snare:** The room can be good for playing alive, to rehearsals you feel uncomfortable because there is not good definition of the sound. Could be good for a recording room to get a natural reverb.

**Kibo Bass:** Excellent room for a tube pipe organ, but maybe not so much for a string quartet. Personally I felt that the sound got lost between the ceiling and the pipes.

### *Recording Studio*

**Emilio Guitar/Voice:** Is confortable due to the silence and the acoustics. The room enhances the voice at the time of singing, without losing intelligibility in the lyrics. The guitar is a little difficult to control because it sounds aggressive more easily than in rooms with more reverberation. The response of the room is more precise, so as a musician more control is needed. Is very sensitive to the way of playing. In the other hand the room responses to the voice dynamics. When I sang strongly it appeared to vibrate with my voice and when I sang soft everything was quiet more easily, to appreciate the color of the sound and play with it. However, for live concerts I feel that the room is too precise

**Kibo Guitar/Voice:** Very good. Is not excessively dry, absorbs the sound in a natural way...

**Kibo Cello:** It is a dry room so it is uncomfortable to play, specially a string instrument. Even so, the room is more isolated so the pianissimos notes can be present, which is very good.

**Kibo Snare:** Perfect for recording and rehearsing. Poor resonance so it is not very good to play live.

**Kibo Bass:** Good, isolated, confortable, good visual communication, complete sense, maybe a little rounded.

### *Rehearsal Room*

**Emilio Guitar/Voice:** The room has a compensated acoustics, perfect for this kind of music. The voice is enhanced with the acoustic wood objects, as if it vibrates with the room. Besides, the reverberation feels very natural. Maybe it is a little bit more reverberant than how it is desired, and in the soft passages the intimacy is not as good as in the recording room. The noticeable background noise (other musicians playing in other rooms) can influence also in the intimacy. Besides, the room feels cozy for all the purposes and you easily feel comfortable in it.

**Kibo Cello:** It is a good room but maybe too reverberant for my taste.

**Kibo Snare:** The volume level of the room is too loud according to the moment. It has a lot of reverb and that amplifies the sound.

**Kibo Bass:** Is a good room in general, but it is a shame that you can hear a lot of noise from the outside.

## Demographic data of the musicians who answered the survey

| | Emilio | Alfred | Alex | Jaume | Marc |
|---|---|---|---|---|---|
| Are you a Musician? | YES | YES | YES | YES | YES |
| Years of experience | 18 | 15 | 18 | 17 | 17 |
| Hours listening to music | 2 | 1 | 5 | 5 | 5 |
| Active music listener? | YES | YES | YES | YES | YES |
| Hearing problems? | NO | YES | NO | NO | NO |