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Large and small room acoustics

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Abstract

The final degree project is consisted on studying the acoustics of class rooms of differnet dimensions. Study has been carried out on how to improve these class rooms acoustics by using absorbent material such as resonators and porous abosorbents. Reverberation is one of the most important phenomenon occurring in rooms that is important to correct. In this regard, one possibility could be the absorption of acoustic energy and in certain cases the only way to correct this phenomenon as well as to control the noise.

Finally, some special diffusers have been used as well in the class rooms to see how they impact on the reverberation time and their acoustics.

Resumen

El trabajo final de grado consiste en el estudio de acústica de salas de diferentes dimensiones cuyo uso es aulas de universidad. Se ha llevado a cabo un estudio para mejorar la acústica de estas salas mediante el uso de materiales absorbentes como resonadores y absorbentes porosas. La reverberación pues, es un fenómeno producido en la acústica de salas que es importante corregir. Así pues, la absorción de la energía acústica es una de las posibilidades y en ocasiones la única para la corrección de este fenómeno así como para el control de ruidos

Por último, algunos difusores especiales también se han utilizado en estas salas de clase para ver cómo afectan al tiempo de reverberación y en su acústica en general

Resum

El projecte de fi de carrera es consisteix en l'estudi de l'acústica de sales de classe de dimensions differnet. Estudi s'ha dut a terme la forma de millorar aquestes sales de classe acústica mitjançant l'ús de materials absorbents com ressonadors i abosorbents poroses. La reverberació és un dels fenòmens més importants que tenen lloc a les habitacions acústiques que és important corregir. En aquest sentit, una possibilitat podria ser l'absorció d'energia acústica, en alguns casos l'única manera de corregir aquest fenomen, així com per controlar el soroll.

Finalment, alguns difusors especials s'han utilitzat també a les sales de classe per veure com impacten en el temps de reverberació i de la seva acústica.

Acknowledgements

Doing the final project of the bachelor's degree means finishing the career and getting to this stage means a lot, it is the end of one period of studying and starting of work, another one. For this reason it is important to remember everyone who has made it possible for me to get to this point.

Firstly, the crucial role my parents who have always been there for any kind of help. Not only they have been there supporting me economically to pay for my stay in Valencia, but they have been the mainstay essential in times when I most needed them.

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Finally, in this short mention I can not forget Mr Libor Husnik for accepting me and giving me the opportunity to do this project with him at the Czech Technical University in Prague. He has guided me all the way in this project and has been there to help me whenever I needed.

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List of Abbreviations

RT	Reverberation Time
MLS	Maximum Lenght Sequence
LTI	Linear Time Invariant
PIR	Periodic Impulse Response
S/N	Signal to Noise ratio
FHT	Fast Hadmard Transform
PSD	Power Spectrum Density
ESS	Exponential Sine Sweep
IR	Impulse Response
CTU	Czech Technical University
TSD	Time Delay Spectrometry
FEM	Finite Element Technique
BEM	Boundry Element Methods

Chapter 1

Introduction

The acoustical properties of small rooms (recording studios, class rooms, home cinemas etc.) differ considerably from the large ones (concert halls, cathedrals or lecture halls etc.) in that the absorption in such rooms is not located just on walls, but acoustically active elements may protrude into the volume of the room. When a sound source located in a room is activated, it generates a sound wave which propagates in all directions. A listener located at any point of the room receives two types of sound: the so-called direct sound, the one that comes directly from the source without any interference. And the indirect sound or reflected one, originated because of the different reflections suffered by the sound wave to impinge on the boundary surfaces of the room. Having said this we can say that there are two areas with markedly different characteristics: the first zone includes all those reflections that arrive immediately after the direct sound, and are called "early reflections", and a second one formed by late reflections that are called "reverberant tail". From what we said previously, it is easy to understand that when a soundwave impinges in to a closed room we can have the following cases; total absorption of the soundwave; total reflextion; parcial absorption or parcial reflextion. The first sound that reaches any receiver is the direct sound. Then, after the direct sound, some reflected waves impiges over the receiver.

The phases and the amplitudes of the reflected waves are randomly distributed to the degree that cancellation from destructive interference is fairly negligible. If a sound source is operated continuously the acoustic intensity builds up in time until a maximum is reached. If the room is totally absorbent so that there are no reflections, the room operates as an anechoic chamber, which simulates a free field condition. With partial reflection, however, the source continues to add acoustic energy to the room, that is partially absorbed by the enclosing surfaces (i.e, the walls, ceiling,floor and furniture) and deflected back into the room.

At any point of the room, the energy corresponding to the direct sound depends only on the distance to the sound source, whereas the energy associated with each reflection depends on the path traveled by the sound beam, and the degree of the absorption of the acoustic materials used as coatings of the surfaces involved. And so its easy to understand the greater the distance and more absorbent materials we use, the lower will be the energy associated with both the direct sound and the successive reflections. Normally, the early reflections have a higher energy level comparing to the

reverberant tail as they get to the receiver without any obstacle. In addition, the fact that the early reflections depend on the geometric shape of the room, they are specific for each point and therefore determine the own acoustic characteristics same together with the direct sound.

The distribution of acoustic energy, whether originating from a single or multiple sound sources in a room, depends on the room size and geometry and on the combined effects of reflection, diffraction, and absorption. With the appreciable diffusion of sound waves due to all of these effects it is no longer germane to consider individual wave fronts, but to refer to a sound field, which is simply the region surrounding the source. A free field is a region surrounding the source, where the sound pattern emulates that of an open space. Neither reflection nor diffraction occurs to interfere with the waves emanating from the source. Because of the interaction of sound with the room boundaries and with objects within the room, the free field will be of very limited extent.

A diffuse field is said to occur when a large number of reflected or diffracted waves combine to render the sound energy uniform throughout the region under consideration. The degree of diffusivity will be increased if the room surfaces are not parallel so there is no preferred direction for sound propagation. Sound reflected from walls generates a reverberant field that is time dependent. This is one of the reasons why rooms (small or large) must be treated acustically and have some kind of diffusers or other elements in order to have good acoustics. Moreover, in this paper we will do experiments in a small room, first without any furniture or absorbing elements and afterwards with some special acoustic diffusers whose especifications will be given in the following chapters.

Chapter 2

Room Acoustics

2.1 Introduction

To study a room even if it is big or small and its characteristics, there are different types of methods. The most commonly used in acoustic field we have three methods: Wave acoustics, statistical acoustics and geometrical acoustics. The field of application of each method is somwhow in the following way: for a small room and at low frequences is more apropiate to use wave acoustics; for the same enclosure but at hight frequencies is more suitable geometric and statistical acouctics and for the lage rooms at high frequencies it is not very normal to use wave acouctics but all other options as valid. Each of the three methods has it's own importantce and evaluate and correct the enclosure from a different aspect and order of using any of it is irrelevent. With these methods it is possible to know with rigor sound behavior in any enclosure.

2.2 Wave Acoustics

Definig in a simple way, wave acoustics is based on the resolution of the differential equation in each room in question and applying the boundary conditions. It is used to study the resonances of the room. The combination of incident and reflected waves in a room creates the standing waves or that is same as eigenmodes of the room. Each eigenmode is associated with a frequency, and it is characterized by a sound pressure level, SPL, which varies depending on points of the room which we consider. So Wave Acoustics is most commonly used for studying the eigenmode of a room. The number of eigenmodes is unlimited and presence of all of them cause in each point of the room a concentration of energy around different eigenfrequencies, which gives each room a distinctive sound. This is what called sound colouration and usually appears in relative small size rooms.

The values of the eigenfrequencies associated to the different eigenmodes dependends on the geometry and dimensions of the enclosures. Only when it comes to enclosures with parallelepiped form with totally reflecting surfaces it is possible to calculate eigenfrequencies realated to the eigenmodes in a simple way with Rayleigh formula:

$$f_{k,m,n} = \frac{C_0}{2}\sqrt{\left(\frac{k}{L_x}\right)^2 + \left(\frac{m}{L_y}\right)^2 + \left(\frac{n}{L_z}\right)^2} \qquad (2.1)$$

[2]

where: Lx , Ly and Lz represent the dimensions of the room (in meters) k , m, n can take any integer (0 , 1 , 2 , 3 , ..). Each combination of values k,m,n produces a frequency and it's associated eigenmode. For example , the combination : k = 0 , m = 1 , n = 1 results in the eigenmode 0, 1 , 1

Moreover, the density of eigenmodes increases with frequency. This means that from a certain frequency, the concept of sound colouration no longer makes sense, since a large density of eigenmodes is equivalent to the absence of these.

$$f_{max} = 1849\sqrt{\frac{RT_{mid}}{V}} \tag{2.2}$$

[2]

The (2.2) formula permits us to calculate, for each room, the upper limit frequency from which the eigenmodes have practically no influence. And the parameters are as follow:

-RT_{mid} is the value of the reverberation time(explained after in 2.3.0.1) obtained as average values corresponding to the octave bands centered at 500 Hz and 1 kHz , expressed in seconds

-V is the room volume, expressed in m^3

So now we can say that the effect of the eigenmodes have more influence as smaller the room is, this is the case of recording studios, control rooms etc. The practical way to minimize the effect of eigenmodes is by using etectronic systems such as equalization or by installing resonators.

2.3   Statistical Acoustics

Statistical Acoustics evaluates the acoustic energy altogether. It is used for the study of reverberation. Fundamental hypothesis of Statistic Acoustics is that the sound field is a diffuse field and the sound particles are inconsistent and therefore energy can sum.

A diffuse field is a sound field in which a large number of reflected waves from all directions are combined so that the average energy density is uniform at any point of the field. So in a diffuse field energy distribution is homogeneous and isotropic at any time and point of the enclosure. Reverberation and diffusion are related in such way that the greater the diffusion the greater the reverberation is and vice versa.

To understand statistical acoustics it's easy to understand somehow with the following example; we put an omnidirectional sound source and let it emit continuously. From the initial instant, the sound waves goes in all directions and every sound beam travels a different path, reflecting again and again on different surfaces of the room. In each reflection, part of the energy is absorbed and part is returned to the enclosure in a greater or lesser

extent; depending on the degree of sound absorption corresponding to the coating surface involved. The constant supply of energy from the sound source makes the total energy received at any point in the room, obtained as the sum of direct and indirect or reflected one, increases gradually to achieve the balance point. That balance point is what called steady-state.

And off course, when the sound source stops abruptly, the sound pressure level (we can talk about sound pressure level or energy) begins to decrease progressively untill it disappears. The speed of sound attenuation depends on the degree of absorption of the enclosure surfaces: the higher the absorption, faster the attenuation. The degree of permanence of sound once the sound source has been disconnected is called reverberation. This is why the reverberation of a room is greater when sound takes more time to be attenuated which means when less absorbent the enclosure is.

All that explained in the previous paragraphs can be observed in the following graphs, where we can see how after starting the sound source the steadty-state is established and after the sound source's cessation how sound pressure levels decays.
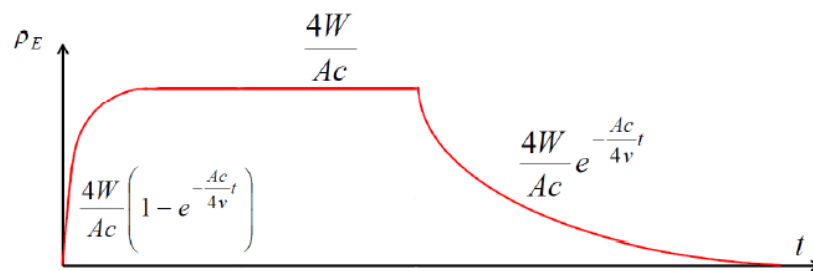
FIGURE 2.1: How sound pressure level changes in an enclosure after starting a sound source and then after its cessation

2.3.0.1 Reverberation time of Sabine

Wallace Clement Sabine after many years of research obtained an expression that relates the degree of reverberation of the rooms with absorbing surfaces. This great discovery allowed to have control over the acoustics of the room and the name of the parameter was called reverberation time. The reverberation time is directly related to the time constant and is defined as the time elapsed for the noise level to reach $60dB$ value lower than its steady-state value. The reason for choosing such a wide decay margin has to do with the dynamics of sound concert halls (difference between maximum and minimum SPL) and therefore the perceived duration of reverberation in this type of rooms

In other words, the reverberation time is the time it takes to fall $\rho_E(t)$ to the thousandth of its steady-state value. From the rise and fall curve of the

acoustic energy, figure 2.1 we know for the rising segment

$$\rho_0 = \rho_E^{sta} = \frac{4W}{Ac} \tag{2.3}$$

For the defination of thousandth of steady -state

$$\rho_{(t)} = 10^{-6} \rho_E^{sta} \tag{2.4}$$

If we substitute the above expressions to it's respective expression of rise and fall curve, the following equation is observed:

$$\frac{4W}{Ac} = 10^{-6} \frac{4W}{Ac} e^{\left(\frac{Ac}{4V}\right)} \tag{2.5}$$

By clearing the equation we get the reverberation time:

$$RT60 = \frac{0.161 \cdot V}{S \cdot a} \tag{2.6}$$

[4]

Following what we just explained and with hypothesis of steady-state it turns out that the total sound energy present in any point of the room is obtained as the sum of a variable energy value, which depends on the location of the point, and another one that is a constant value. Variable energy value corresponds to the direct sound, and decreases as the receiver moves away from the source, while constant energy value is associated with the indirect or reflected sound. The fact that this energy does not depend on the position is because we are applying statistical theory to all reflected sound and therefore by treating equally all reflections; early ones and the late ones.

Finally, we can say that the total sound pressure at any point of an enclosure is obtained from the contribution of the pressures of the direct sound that decreases with the distance from the source and the reflected sound which remains constant. The area where the direct sound predominates is called direct field area and area where the reflected sound dominates is called reverberant field area.

2.4   Geometrical Acoustics

Geometrical acoustics studys the early reflections of an enclosure and its useful for analyzing the distribution of sound field in the room to detect possible risks of echo or focalizations.

For the application of geometrical optics laws for acoustics enclosures, we will consider that the sound field is formed by beam combination, through which the sound energy propagates. And it is essential to keep in mind the relationship between wavelength and the size of the surfaces where it impinges. In depending on how this is related, we can have the following cases: diffraction, diffusion or reflection.

The wavelength of the sound must be small compared to room dimensions and objects present in it otherwise diffraction phenomena occurs. The dimensions of relief surfaces must be clearly less than the wavelength of sound considered if not the sound is reflected diffusely. And the difference of impedance between the air and the enclosures must be high so that the reflection phenomena predominates.

Moreover just to say that in geometrical room acoustics, the concept of a wave is replaced by the concept of a sound ray. The latter is an idealisation just as much as the plane wave. As in geometrical optics, we mean by a sound ray a small portion of a spherical wave with vanishing aperture which originates from a certain point. It travels in a well-defined direction and is subject to the same laws of propagation as a light ray, apart from the different propagation velocity.

2.5 Room considerations

The acoustical properties of small rooms differs considerably from the large ones. The first parameter that is different in each case is the RT60. For small rooms normally its less then 0.5 seconds. And it is because the sound absorption

$$A = S \ a (m^2) \tag{2.7}$$

[4]

in a room depends primarily on the frequency-dependent absorptive properties associated with the materials used for the six surfaces of the room, walls, floor and ceiling. The Sabine formula tells us that the reverberation time RT60 is proportional to the volume to surface area ratio,

$$\frac{V}{S} \tag{2.8}$$

[4]

For small listening rooms, the volume to surface area ratio (2.8)
is usually small (compared to a concert hall or auditorium) and hence the reverberation time RT60 for a typical small room is usually quite short as just mentioned.

The total sound that we hear in any given room at a given instant in time are a combination of direct sound from sound sources in the room + indirect, reverberant sounds from multiple reflections in the room associated with direct sounds output from the sound sources that were produced at earlier times. The reverberant sound also does not have the same frequency spectrum as that associated with the direct sound, for two reasons – frequency-dependent absorption of the sound by various internal surfaces in the room and also the excitation of room modes. Additionally, in small rooms oftentimes the sound at a given frequency f is absorbed before a uniform energy density $w(f)$ of reverberant sound is obtained throughout the room. Thus, the dynamical evolution of the reverberant sound field in a

small room in evolving from the initial direct sound to a steady-state can be quite different than for large rooms. Furthermore, in a small listening room (a living room in a house) almost always the room is filled with other items – sofas, coffee tables, lamps, chairs, etc. all of which reflect and absorb the sound in a myriad of ways, from these additional objects located at different places in the room, resulting in even more complexity associated with the reverberant sound field in a small listening room.

Then the direct sound, the early reflected sound and the reverberent sound is also different in small and large rooms. In a large room, first-arrival times of the early reflected sound are typically on the order of 50-80 ms after the direct sound, whereas for small rooms, the first-arrival times of the early reflected sound are typically on the order of just a few ms after the direct sound. The judicious use of sound diffusers in a small room helps in creating a more uniform reverberant sound field in a small listening room, hopefully approximating that associated with a larger room. Whereas flat walls and concave surfaces tend to direct the sound, convex and/or rough surfaces will instead scatter the sound in several directions, thereby helping to even out/make more uniform the reverberant sound field. Geometrical shapes attached to room surfaces (walls, floor and/or ceiling) help to scatter and diffuse the sound. Triangular, rectangular and/or semi-cylindrical protrusions on these room surfaces help to scatter the sound in many directions, thereby helping create a more uniform reverberant sound field

Additionally in home environments, the sound absorption properties of the room often are significantly higher than in large rooms, due to the presence of home furniture, window curtains on walls, etc. Porous materials such as curtains, carpets, glass fiber and acoustical tile absorb sound energy very well at high frequencies, whereas materials commonly used in home construction such as wood, glass, gypsum board (drywall) and plaster on lath absorb sound energy very well at low frequencies. Thus, the acoustic "intimacy" of the small room often makes it difficult to emulate the acoustics associated with that of a larger space.

The short and long decay time associated with sound in small and large rooms provides important auditory information/clues to the listener about the size and nature of the room. In a small listening room, at low frequencies, nearly all listeners are in the reverberant sound field, whereas at high frequencies, the effect would depend on where the listener was seated – closer (or not) to the direct sound(s) emanating from the L/R stereo speakers. Reflections from large room walls and ceilings, all add up to increased reverberation times and it mainly impacts on low frequencies. This is termed the speech intelligibility range. If reverberation times exceed 1 second, then speech begins to become unintelligible very quickly. If we exceed 1 second reverberation times for speech, the reflections in our large room become superimposed over the direct sound we need to hear. The direct sound is the sound that leaves from a source such as our loudspeaker and travels directly to the receiver's ear. It is the straight line sound that does not have any room reflections interlaced with it to confuse and muddle the direct sound.

Chapter 3

Methods of measurements in room acoustics

3.1 Introduction

Among other methods of measurements here in this work we are going to see *Sweep* and *MLS Maximum length sequence*. In our case we will use *MLS* because we have linear system and because it has immunity to signals not correlated with excitation singal. And to understand these methods in a better way first we will talk and discuss some basic concepts such as *Octave band*, *Third octave band* (1/3), *White noise* and *Pink noise*

3.2 Octave band

When more detailed information about a complex sound is needed, the frequency range of 20Hz to 20kHz can be split into sections or bands. In audio-engineering applications, sound spectrums are usually represented in octave or one-third octave frequency bands rather than in narrow frequency bands. This frequency representation is linked to the perception of sound by a human ear and it allows a compression of the amount of information.

The audio spectrum from 20Hz to 20kHz can be divided up into 11 octave bands. If we define the 7th octave band's center frequency to be f=1000 Hz then all lower center frequencies for octave bands can be defined from each other using the formula $f_{n-1}^{ctr} = f_n^{ctr}/2$. Conversely, all higher center frequencies for octave bands can be defined from each other using the formula $f_{n+1}^{ctr} = 2f_n^{ctr}$. Then for each center frequency, the half-octave low and high frequency for each octave band are given by: $f_n^{low} = f_n^{1/2}$, $f_n^{high} = 2^{1/2}f_n$ respectively.

3.3 1/3 Octave band

Here the audio spectrum from 20Hz to 20kHz can be divided up into 31 octave bands. And here if we define the 19th 1/3 octave band's center frequency to be $f_{19}^{ctr} = 1000Hz$ then all lower center frequencies for 1/3 octave bands can be defined from each other using the formula $f_{n-1} = f_n/2^{1/3}$. Conversely, all higher center frequencies for 1/3-octave bands can be defined from each other using the formula $f_{n+1} = 2^{1/3}f_n$. Finally, same as above for each center frequency, the 1/3 octave low and high frequency for

each 1/3 octave band are given by: $f_n^{low} = f_n/2^{16}$, $f_n^{high} = 2^{16}f_n$ respectively.

3.4 White Noise

Noise is used to mean a lot of things in engineering, and is an extremely important concept, since a lot of the technical challenges in, say, building a cell phone system, or an ultrasound machine, come from trying to extract a desired information 'signal' from a background of unwanted 'noise'. In general, 'noise' can refer to anything that interferes with what we want: it might be a single voice of someone sitting next to us, but the 'purest' form of noise comes from a totally random source. In engineering terms, this randomness corresponds to unpredictability: knowing one part of a signal tells you almost nothing about the future of the signal.

This kind of minimum-information noise is called white noise, by analogy with white light which is a uniform mixture of all the different possible colors. In the frequency (Fourier) analysis often used in signal processing, white noise is a uniform mixture of random energy at every frequency.

White noise is a random signal (or process), and has equal power in any band of a given bandwidth (power spectral density) when the bandwidth is measured in Hz. For example, with a white noise audio signal, the range of frequencies between 40 Hz and 60 Hz contains the same amount of sound power as the range between 400 Hz and 420 Hz, since both intervals are 20 Hz wide. White noise is also used to mask background noises in the office, or to aid in sleep. And as we can see in the graphic below (Figure 3.1) it has got equal evergy per Hertz. And it is power spectrum density, PSD, is constant. [9]
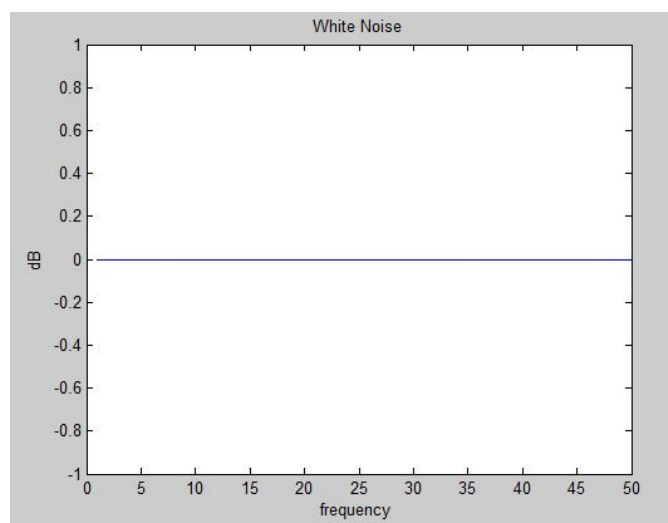

FIGURE 3.1: Constant PSD of white noise spectrum estimated with Welch's method

To end with the white noise, we can mention some of it is aplications. It is commonly used in the production of electronic music, usually either directly or as an input for a filter to create other types of noise signal. In this thesis work we have also made a matlab programme to obtain pink noise from the white noise. It is used extensively in audio synthesis, typically to recreate percussive instruments such as cymbals or snare drums which have high noise content in their frequency domain. Moreover, what's more preeminent for our experiments, white noise is also used to obtain the impulse response of an electrical circuit or enclosures.

3.5 Pink Noise

The frequency spectrum of pink noise has equal power in bands that are proportionally wide. This means that pink noise would have equal power in the frequency range from 40 to 60 Hz as in the band from 4000 to 6000 Hz because it has got equal energy in all octaves. Since humans hear in such a proportional space, where a doubling of frequency (an octave) is perceived the same regardless of actual frequency (40–60 Hz is heard as the same interval and distance as 4000–6000 Hz), every octave contains the same amount of energy and thus pink noise is often used as a reference signal in audio engineering. **[9]**

The PSD, compared with white noise, decreases by 3 dB per octave (density proportional to $1/f$). For this reason, pink noise is often called "$1/f$ noise". Pink noise can be obtained from white noise by means of a low-pass filter designed so the output spectral power density (that is, the sound power contained within a narrow frequency band of a certain fixed width, such as 1 Hz) drops by 50 percent with each octave as the absolute frequency rises. Pink noise can also be directly generated by a computer-controlled acoustic synthesizer. We can see the the PSD of pink noise the next figure:
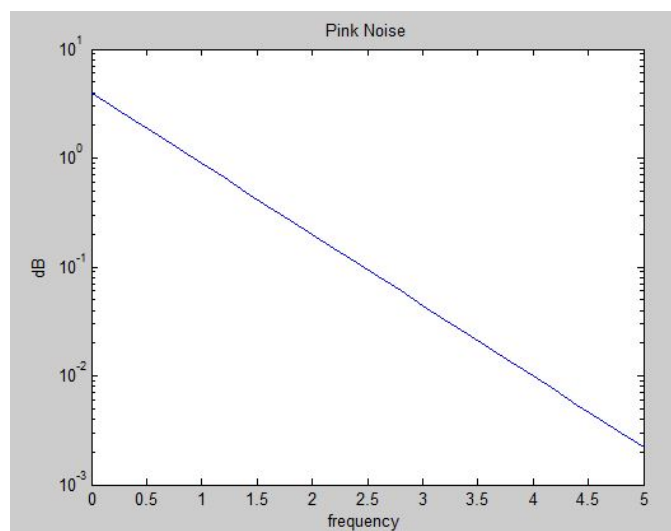

FIGURE 3.2: PSD of pink noise spectrum and how it decreases 3 dB per octave

Finally, some of the aplications of pink noise are that it is useful because it is the same as humans hear. It's not that pink noise is calibrated to the human ear's frequency response its only that each time the frequency double we hear that as an octave. And for our interest from acoustic view, it is mainly used for acoustic measurements, to obtain the frequency response of the room. And for our measurements for the large and small rooms we will also use pink noise beause we will be using a loudspeaker with a tweeter and a woofer and it is more reliable to use pink noise in that case then white noise.

3.6   MLS (Maximum length sequence)

3.6.1   Defination

MLS is an abbreviation for Maximum Length Sequence and it is a pseudo random signal consisting of the values 1 and -1. It is periodic with the period $P = 2^N - 1$ where N is number of digital shift registers; in acoustics 16 digital shift registers are normally used implying 65.535 samples for a typical MLS room-acoustics measurement signal. For our interest from acoustic point of view we use it for measuring the impulse response of a linear system as an input to the system.

3.6.2   How it is generated

For any LTI (Linear Time Invariant) system we know that if we have an input $x(t)$, the output of that system is $y(t)$ convolved with $h(t)$ (impulse response of the system) in time domain. And in the frequency domain its's the multiplication $Y(s) = X(s) \cdot H(s)$ as shown in the following diagram:

Time   domain $x(t)$ $\longrightarrow$ $\boxed{\text{h(t)}}$ $\longrightarrow$ $y(t)$ $=$ $x(t)$ $*$ $h(t)$

Frequency domain $X(s)$ $\longrightarrow$ $\boxed{\text{H(s)}}$ $\longrightarrow$ $Y(s)$ $=$ $X(s)$ $\cdot$ $Y(s)$

[5]

MLS signal is useful to measure any input-output system asmost equipment and room acoustics are and it is one of the basic room acoustics measurement methods recommended by ISO standard ISO 3382. MLS is comparable to white noise and as we know white noise is non-periodical and random. It can be used to measure the response of any type of LTI system. The impulse response of the system can be easily obtained by computing the cross-correlation between input and output signals.

3.6.3 Obtaining the impulse response

From what we explained above we know in time domain for a LTI system the output is:

$$y(t) = x(t) * h(t) = \int x(\tau)h(t - \tau)d\tau \tag{3.1}$$

The autocorrelation of the output and input signals is expressed as

$$\rho_{xy}(t) = \int x(\tau)y(\tau - t)d\tau \tag{3.2}$$

Substituting (3.1) in (3.2) we get:

$$\rho_{xy}(t) = \int x(\tau) \int x(\tau)h(\tau - t)d\tau d\tau = \int h(\tau) \int x(\tau)x(\tau - t)d\tau d\tau =$$
$$\int h(\tau)\rho_{xx}(\tau - t)d\tau = h(t) * \rho_{xx}(t) \tag{3.3}$$

[6]

And an important characteristic of MLS is that it is autocorrelation function of the input signal approximates the Dirac delta (see figures 3.4 and 3.5) and the crosscorrelation of the output and input signals approximates the system impulse what indeed we are interested to calculate.

$$\rho_{xy}(t) \approx h(t) \tag{3.4}$$

[6]
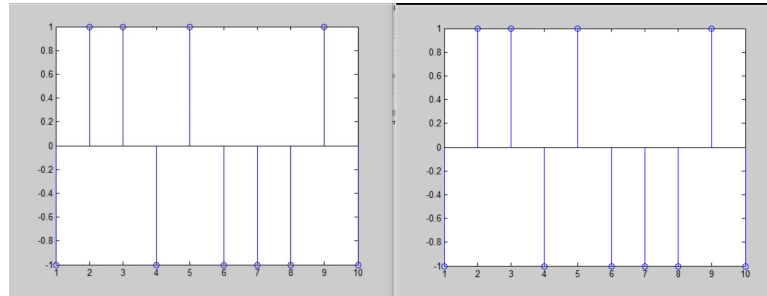
FIGURE 3.3: Two pseudorandom MLS signals

And the auto-correlation of these two pseudorandom MLS signals aproximates to just a Dirac Delta.

What explanied above for LTI system is complety analogous to discrete time domain linear systems as well. In a discrete time domain linear system, the output signal $y[n]$ has the form of the following discrete convolution sum of the input signal $x[n]$ and the system impulse response $h[n]$:

$$y(t) = x(t) * h(t) = \sum_{m=0}^{L-1} x[m]h[n - m] \tag{3.5}$$

[7]
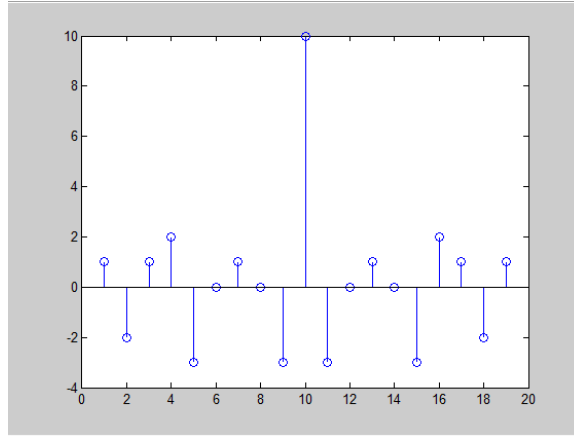
FIGURE 3.4: How the auto-correlation of two pseudoran-
dom MLS signals aproximates to just a Dirac Delta

The correlation function $\rho_{xy}[n]$ of the two signals is expressed as:

$$\rho_{xy}[n] = \frac{1}{L} \sum_{m=0}^{L-1} x[m]y[m-n] \tag{3.6}$$

Substituting Eq. (3.5) into (3.6) yields:

$$
\begin{aligned}
\rho_{xy}[n] &= \frac{1}{L+1} \sum_{m=0}^{L-1} x[m] \sum_{k=0}^{L-1} h[m]x[m-n-k] = \\
&= \frac{1}{L+1} \sum_{k=0}^{L-1} h[k] \sum_{m=0}^{L-1} x[m]x[m-n-k] = \\
&= \sum_{k=0}^{L-1} h[k]\rho_{xx}[n-k] = h[n] * \rho_{xx}[n]
\end{aligned}
\tag{3.7}
$$

[7]

3.6.4   MLS properties

Finally, some of the important properties of MLS systems are:

1)MLS technique has strong immunity to all kinds of noise because it has a very high Signal/Noise ratio ($\frac{S}{n}$). The cross-correlation used to compute the impulse response reduces all background noise (uncorrelated with MLS), so that measurements can be performed also in noisy environments. The use of averaging techniques can furhter increase the $\frac{S}{n}$ ratio.

2) MLS technique rejects the DC component of the sampled signal because it is sequence is used as the input signal and the crosscorrelation function of the input and output signals approximates the system impulse response. As the digital signals and calculations are periodic, the result is periodic as well, being called Periodic Impulse Response (PIR).

3)   The   MLS   has   a   quasi-flat   power   spectrum.

4) Because of signal and time convolution computational simplicity, it only   requires   Fast   Hadamard   Transform   (FHT).

5) The power spectrum density of MLS is constant for all the discrete frequencies in the covered frequency band, except for the DC offset. So the average power spectrum density of MLS is 2N times greater then that of a single test pulse. **[7]**

3.7   Sine sweep method

3.7.1   Basic concepts

The swept sine measurement technique is also used for measuring impulse responses. It has advantages when measuring weakly non-linear systems in that the linear and non-linear components can be separated to some extent. Firstly there was introduced linear sine sweep method and then exponential sine sweep, ESS. Here, in this work we are going to talk about exponential sine sweep method. The usage of exponential sine sweep, compared with previously-employed linear sine sweeps, provided several advantages in term of signal-to-noise ratio and management of not-linear systems.

3.7.2   Characteristics and advantages

The ESS method uses an exponentially swept sinusoid for room excitation, and aperiodic deconvolution to extract the impulse response from the recorded room response. Unlike the linear sine sweep method, which relies on synchronous averaging of multiple sweeps to increase the IR signal to noise ratio, the ESS method uses a single, long sweep. The primary advantages of this technique over other IR measurement methods are as follows:

-Better   S/N   than   the   MLS   method.

-Near-perfect separation of non-linear effects from the desired linear response

-For systems that may have time variance, such as those involving sound propagation in air, a long sweep utilizing no synchronous averaging is useful for avoiding artifacts in the reverberant tail and high frequency phase errors. **[8]**

3.7.3   Development of ESS

The general form of an exponentially swept sine signal is:

$$s(t) = sin[\theta(t)] = sin[K(e^{-t/L} - 1)] \tag{3.8}$$

here:

$$K = \frac{w1T}{ln(\frac{w1}{w2})}, L = \frac{T}{ln(\frac{w1}{w2})} \tag{3.9}$$

[10]

Starting frequency is $w1$, an ending frequency $w2$. T represents the time duration of the sweep. The range of the recorded sine wave sweep will be limited by the range that can be physically be produced by the speakers used, and to a lesser extent the frequency response of the microphones.

Given a signal that varies in frequency, the energy at a specific frequency is proportional to the time duration during which the signal oscillates at that specific frequency. In other words, energy is related to the rate of change of the instantaneous frequency. Instantaneous frequency $w(t)$ is defined as the rate of change of $\theta(t)$ which for (3.8) take the form:

$$w(t) = \frac{d\theta(t)}{dt} = \frac{K}{L}e^{-t/L} \tag{3.10}$$

Since the energy equation $E(t)$ is proportional to the time at a given frequency, it follows that $E(t)$ is inversely proportional to the rate of change in $w(t)$

$$E(t) \propto \frac{1}{[\frac{dw(t)}{dt}]} = \frac{L^2}{K}e^{-t/L} \tag{3.11}$$

By taking the Fourier transform of (3.11), energy can be expressed as a function of frequency $E(w)$

$$E(jw) \propto \frac{L^2}{K}\frac{1}{L + wj} \tag{3.12}$$

Defining k to be a constant of proportionality, $E(w)$ can be expressed as (3.10)

$$E(jw) \propto \frac{kL^2}{K}\frac{1}{L + wj} \tag{3.13}$$

[10]

From here we conclude something really important that with the exponential sine sweep, energy decreases as frequency increases. More specifically, if the frequency is doubled. And that means a drop of 3dB in energy for each doubling of the frequency ($3dB/octave$)
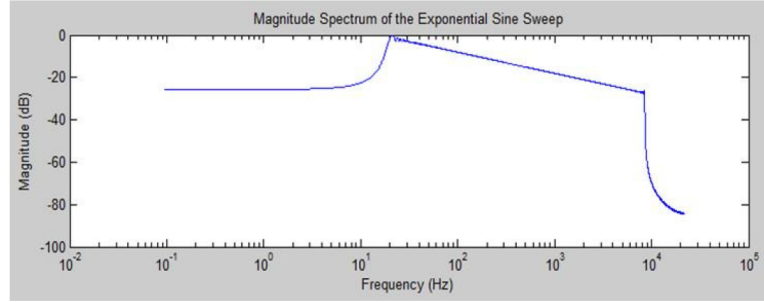
FIGURE 3.5: Magnitude espectrum of ESS

3.7.4 Inverse filter

We create an inverse filter $f(t)$, which, after being convolved with the system response, yields to the impulse response:

$$h(t) = r(t) * f(t) \tag{3.14}$$

The room impulse response $h(t)$ can be extracted by convolving $r(t)$ with the inverse filter of $s(t)$ The exponential sweep is temporally reversed and then delayed to obtain a causal system. However, if we time-reverse the excitation signal $s(t)$, it still exhibits $-3dB/octave$. Therefore, we need to compensate this energy drop by modulating the amplitude of the time-reversed signal with a $+6dB/octave$ envelope so that the inverse filter exhibits a $+3dB/octave$ slope.

This is termed post-modulation, in opposition to a pre-modulation suggested by, which modulates the input signal directly so that it has a flat spectrum and the reversed-time signal exhibits a flat spectrum alike. The post-modulation term which is to be multiplied with the time-reversed input signal is of the form :

$$m(t) = \frac{A}{w(t)} = A[\frac{K}{L}e^{t/L}]^-1 \tag{3.15}$$

[11]

where A is a scalar representing the modulation amplitude. At time $t = 0$, the instantenous frequency w equals $w1$. In this condition, we can solve for A the equation (3.15) assuming arbitrarily that $m(t) = 0$ at $t = 1$ it seems that $a = w1$ Modulating the time-reversed signal gives:

$$f(t) = \frac{w1}{w(t)}sin[\theta(T - t)] = w1\frac{L}{K}e^{-t/L}sin[K(e^{\frac{T-t}{L}} - 1)] \tag{3.16}$$

[11]

And finally the magnitud spectrum of the inverse filter can be seen in the following figure:
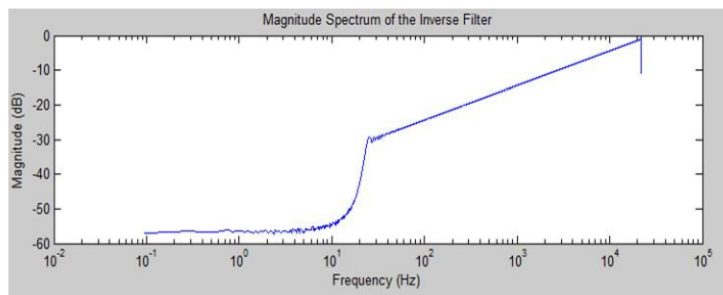
FIGURE 3.6: Magnitude espectrum of the inverse filter

Chapter 4

Measurements

4.1 Introduction

All the measuremnets have been done in the faculty of electrical engineering, Prague, Czech Republic. The measurements were done in two rooms, one of them of $940x540cm$, classroom 205 of the second of the faculty (big room) and the other of the dimesnions $740x540cm$, classroom $202A$ of the second floor of the faculty (small room). In this chapter the behaviour of the both rooms will be analyzed. Firsty, the big room with and without diffusors with high resolution will be treated. This means, points close to each other by 5cm and in total 225 points. Secondly, the big room will be studied with and without diffusers but this time with the microfone with two heights, 111cm and 155cm. Moreover, here the measurements have been done with low resolution, which means, points separated to each other by 60cm.

4.2 Project development

4.2.1 Material Used

The microphone used for the measurements was a professional Earthworks M50. This microphone is remarkably stable with respect to temperature and atmospheric conditions and is optimized for clean, very fast impulse performance providing accurate wideband response with virtually no handling noise. It's ideally suited for on-location acoustical measurements including loudspeaker design and quality control, sound system setup and troubleshooting, room acoustics, or any application where an accurate free-field measurement microphone is required. And this microphone can be seen in the next figure:

FIGURE 4.1: Microphone of Earthworks

For the source we used a loudspeaker of VMAUDIO and it can be seen in the next figure:

FIGURE 4.2: loudspeaker VMAUDIO

Other material used for the measurements was a sound card, M-AUDIO, a voltage generator and a computer with a specified program that with be explained forward. All that mentioned can be seen in the next figure:
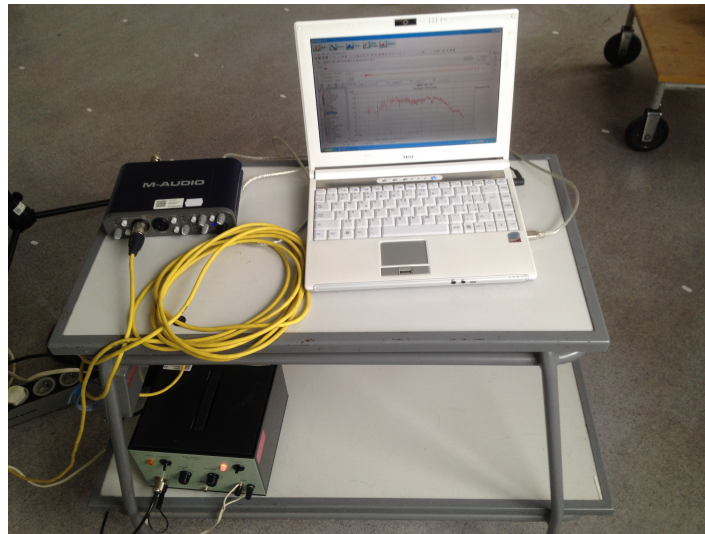
FIGURE 4.3: M-AUDIO sound card, a voltage generator and a laptop computer. Among other materials used for the measurements.

4.2.1.1 Easera

The software used for the measuremnte is called *Easera*. It is a payment software and for the study of this project it is used as a license of the CTU university. Easera provides both data acquisition with a variety of stimulus signals including Time Delay Spectrometry, TDS, sweeps MLS or noise excitation signals and a post processing engine to calculate all acoustic functions or measures. The real time analyser provides multiple ways to perform a fast onsite analysis or to obtain a precise view of the surrounding acoustic environment. And Easera consists of four logic parts: signal generator, measurement, real time analyzer and pot processing.

In experiments done for this proyect a pink noise MLS signal was generated, with the sampling rate of $44100Hz$ as it is showed in the figure 4.4. Pink noise was used because the loudspeaker employed had a tweeter and a woofer and it's more practical pink noise for that kind of loudspeaker then any other noise.
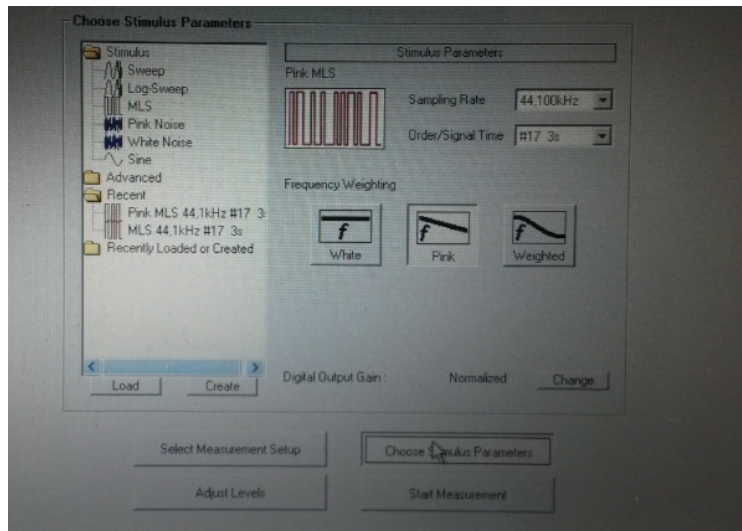
FIGURE 4.4: Easera display to choose the signal form for the
measurements

4.2.2   Measurements in the large room without diffusers

4.2.2.1   Room characteristics and grid of the measurements

The room of dimensions $940x540cm$ where the measurements have been
held can be seen in the following figure:

FIGURE 4.5: Large room of CTU, room number 205

Eventhough in that picture it is not appreciated that the room is not rect-
angle like a shoe box but it has irragularities. And that can be seen sketch
of the room in the next figure 4.6. There are also appreciated all exact di-
mensions as well:

205
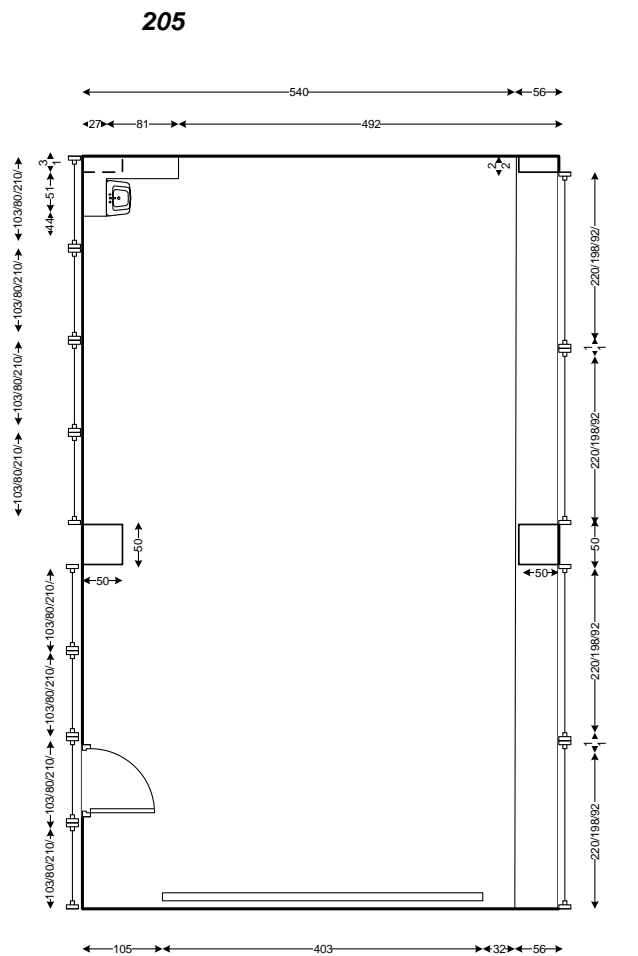
FIGURE 4.6: Sketch of the Large room of CTU, room number 205

Afterwards, as said above, a grid was made in the floor with 225 points for the measurements. All these points were contained in a surface of $80cm$ x $80cm$. Every point with a distance from eachother of $5cm$. The grid was made with white stickers so it was easy to recognize the place where to put the microphone. And to calculate the distances geometry rulers were used. In the following figures the grid is appreciated with sketch figure and also after it was made.
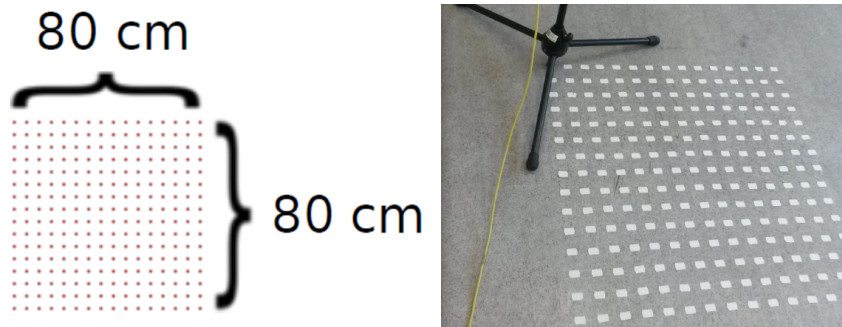
FIGURE 4.7: Sketch and aftermade of the grid of the Large
room of CTU, room number 205

4.2.2.2   Analysis of the data

After having moved the microphone over these 225 points with Easera the
impulse response was obtained for each point. And for each point *.wav* and
.emd files were saved in order to obtain wide range of information. So with
these files it was posible to have all the necesary information from where it
is posible to get all the important parameters of room acoustics and analyze
the data with *matlab*.

FIGURE 4.8: Reverberation decay of the large room 205

In the picture 4.8 we can see the reverberation decay calculated with
schroeder's reverse integration in one of the points in the class room.

In this case of class room 205 we can appreciate that reverberation decay
starts decaying from $104dB$ till $79dB$ that is $25dB$ decay in total.

4.2.3 Measurements in the small room without diffusers

4.2.3.1 Room characteristics and grid of the measurements

The room of dimensions $740x540cm$ where the measurements have been held can be seen in the following sketch:
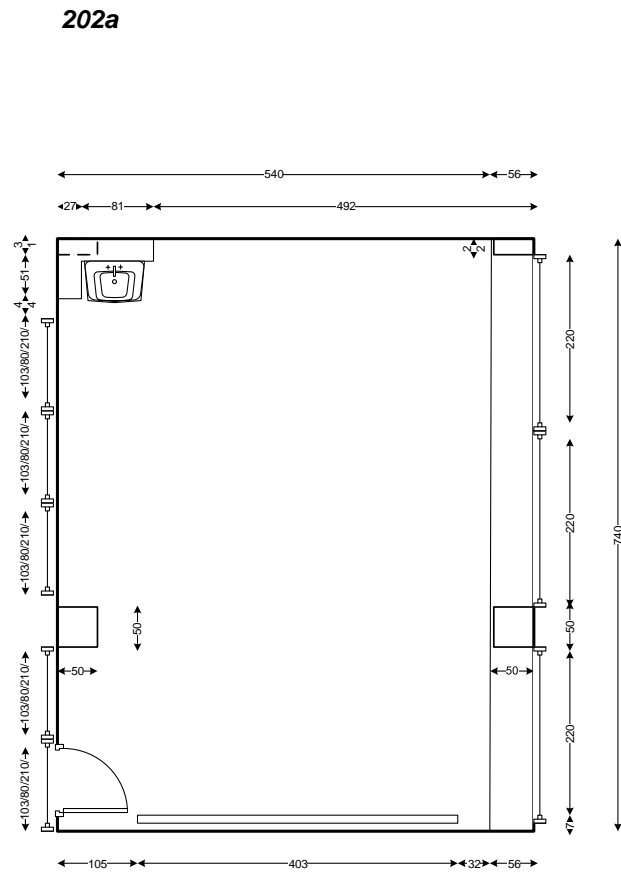
202a

FIGURE 4.9: Sketch of the Small room of CTU, room number $202a$

Afterwards, as in the 205, a grid was made in the floor but this time with high resolution, that means, with points separated from each other $5cm$ all over the room. The grid was made with white stickers so it was easy to recognize the place where to put the microphone.

4.2.3.2 Analysis of the data

After having moved the microphone over these 165 points with Easera the impulse response was obtained for each point. And for each point $.wav$ and $.emd$ files were saved in order to obtain wide range of information. So with

these files it was posible to have all the necesary information from where it is posible to get all the important parameters of room acoustics and analyze the data with $matlab$.
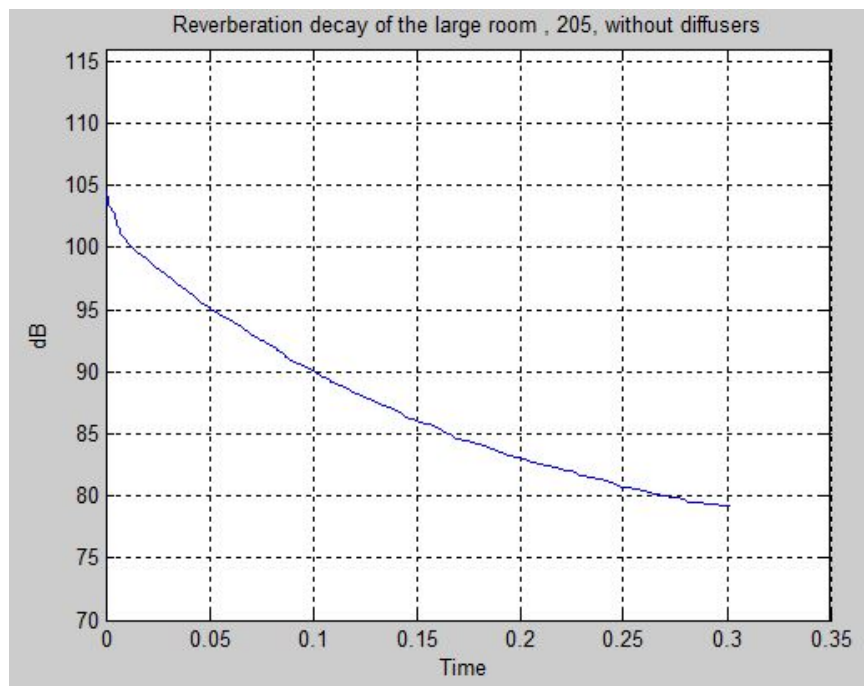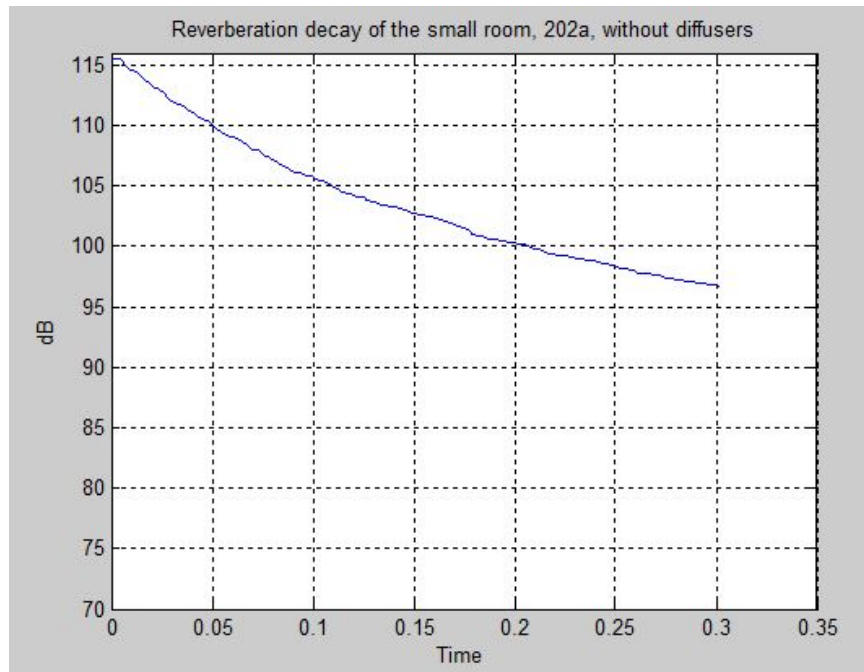

FIGURE 4.10: Reverberation decay of the small room $202a$
without diffusers

In the picture 4.10 we can see the reverberation time calculated with schroeder's reverse integration in one of the points in the class room.

Now in the case of class room $202a$ we can see that reverberation decay starts decaying from $116dB$ till $97dB$ that is $19dB$ decay in total. As we can see in the case of small room the reverberaion decay starts decaying much earlier, $12dB$ in concrete, comparing to the other class room and also stops earlier. So the difference between the reverberation time of both class rooms is of $6dB$ where small room is having smaller reverberation time.

4.3   Quality requirement parameters for the acoustics of classrooms

The acoustic characteristics of a class room are defined mainly by:
• The reverberation time
• The sound distribution
• The speech intelligibility. RASTI (Rapid speech transmission index) is the measure of speech intelligibility. It indicates how well a speaker can be understood from a seat in a class room. Other important parameters are the shape of the enclosure, the materials chosen, the arrangement of reflective and absorbent surfaces, the sound source(in case of using microphones ans amplifiers systems). And reflection, diffusion, diffraction and absorption are responsible for the final quality of the sound distribution in the room. But we must take in count that all parameters are interrelated. **[16]**

4.3.1   Subjective and objective sound quality parameters [17]

Subjective parameters that the listener wishes to have in a class room, apart from the absence acoustic problems like echoes, excessive background noise, resonances, focalizations are numerous, but the main ones are:

•**Intimacy**: The listener perceives music as if he/she found itself in a small room, having the impression that the Music surrounds him/her and thus is immersed in it. these feelings are related to the advent of direct sound and time that separates from the first reflected sound to reaches the listener. All this is related completely with the lateral reflections generated in each ear a different signal, and this is how an subjective space impression originates

•**Clarity**: Rooms must sound clear, when listening distinctly successive sounds and simultaneous sounds, thus allow the hearing of separate tones in time and also separate hearing sounds that emit various instruments. It is closely related with the relationship between direct energy and reverberation.

•**Warmth**: Related to low frequencies, which share the room reinforce the bass sounds that will make the room sounds more heaty. If the room is playing poorly the bass sounds, music seems fragile, lacking vitality and strength. It is closely related to the time reverberation at low frequencies ($< 250Hz$).

•**Diffusion**: This is how to explain because the sound seems to come from any direction with equal intensity. It depends reverberation, surfaces and diffusing power.

•**Equilibrium**: corresponds to the fact of perceiving the loudness balanced of different instruments belonging to a same orchestra. This element will depend largely on surfaces that are close to the orchestra. [17]

4.3.2   Standards of quality requirements of class rooms

There a lot of standards of quality requirements for different type of rooms and depending on country these standards changes. Among other we can found the "The ANSI S12.60-2002" which is "American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools"

Some of the following are highlight points in the mentioned standard:

•Background noise in unoccupied classrooms should not exceed 35 dB(A)

•Reverberation times in unoccupied, furnished classrooms with a volume under 10,000 cubic feet may not exceed 0.6 seconds. Classrooms between 10,000 and 20,000 cubic feet may have a maximum reverberation time of 0.7 seconds

•Use wall, floor, and ceiling assemblies with a minimum STC rating of 45 in adjacent corridors, 50 STC for general adjacent enclosed classrooms, and 60 STC in louder adjacent rooms

•An additional point can be attained by limiting the unoccupied background noise to below 40 dB(A) and constructing all core learning environment walls with the minimum STC rating per room purpose as specifi ed by ANSI S12.60-2002 **[15]**

4.4   Improvements of the measured rooms with absorbing materials

Absorbers can be classified into two distinct groups listed below and are classified according to the region of space where they can work. The first ones are the absorbers that work in the area of high-speed and second ones are absorbers that work in high-pressure zone. Within the high speed ones there are all systems or elements that have a complex internal structure, from industrial porous abosrbers to any textile. The main feature of these absorbers is that the vibration velocity of the particles must be very high in order to get a significant friction losses. Meanwhile, absorbers of high pressure zone should be located close to hard surfaces and are mostly resonant systems. The most common absorbers of this second group resonators are Helmholtz resonators and of membranes. This type of absorbent is characterized by using high levels of steady pressure generated in the vicinity of the walls in order to activate the vibration of all moving parts and armature movement is damped by friction effects, which cause losses of sound energy. **[18]**

Currently, there are a variety of models of absorbers on sale in the market. Because of the diversity of uses that can be given to the use of these absorbers, there are numerous versions of these with really different characteristics. These characteristics are among others the degree of absorption or working bandwidth. It is interested to have more properties then just purely acoustic ones and these are such as: possibility of cleaning them, manufacturability on hard surfaces, resistance to extreme environments, thermal resistance or thermal insulation, among others.

Reverberation is the most important acoustic phenomenon indoors acoustic rooms, since good listening is essential regardless of the type of room or activity carried on it. Getting adequate reverberation is essential whether intelligibility is of the spoken message, of musical performance or just want to have a pleasant acoustic environment will. Thus, the reverberation can be controlled in two ways: by loudness and timbre. Absorbers are designed to work in different frequency ranges, with the sole purpose to compensate excess reverberation in small and medium spaces.

And now eigenmodes for their part, are basically acoustic resonances that occur at different frequencies, which form a stationary spatial pattern and a response in erratic rate, resulting in impaired uniformity of the sound field

that converts to coloration of the signal. Eigenmodes cause problems in small rooms and at low frequency because appearing as independent resonance they make a distribution of maximums and minimums in the space and in frequency. That distribution depends primarily on the geometry of the room. For this reason, it becomes that, for the frequency response as balanced as possible, the dimensions of the room, the position of the sources and the listening position must be chosen by criteria. These solutions are necessary to achieve the best frequency response of a small room and particular surfaces, but are not useful for combating acoustic resonances. Absorbent that give best results give for our case of the class rooms and combat the problems of the modes are low frequency resonators. These systems resonators are placed on the walls and corners and, unlike the porous materials, these require some space in the room. The spectral range of work of these resonators is relatively narrow, so if we want the spectral range of absorption with higher bandwidth, we should combine several modules tuned to different frequencies to cover the desired area. **[18]**

On the other hand, the echoes are late reflections that have a significantly higher level than the global reverberation level, they are perceived as separate sounds. And that means that they are separate sounds of the direct signal. The echoes contain a high level of middle and high frequencies so to absorb the most harmful reflections it is only necessary porous absorbers sheets of a few centimetres.

4.4.1 Porous absorbers

Porous materials are present in almost all acoustic isolation systems and are also an important part for acoustic and thermal insulation in buildings. Such materials are formed by a network of small dimensions, which are connected with the outside air. There are various porous absorbents, the main difference between them lies in the diversity of materials, manufacturing methods and on the application that we want to give to them. This diversity has made all ranges of specialized products currently on the market. Some examples of these absorbent are fibrous absorbent, more industrial ones such as rock wool or glass wool, porous systems as acoustic ceiling tiles, and other usual elements of common rooms and properties absorbents such as carpets and curtains.

The absorption that these materials performing is due to viscous and thermal processes occurring in the immediate vicinity of its surface and on the walls of internal passageways. The region where these losses occur is called acoustic boundary layer. This boundary layer can be found even in completely smooth surfaces, which have neither pores or roughness. That is why the walls are made reflective intentionally. Thus, the reverberation chamber walls are produced in this way to have minimal absorption(aproximatelly $\alpha = 0.02$). There are two types of boundary layer: the viscous layer and thermal layer, which are associated with the respective effects of losses. The thickness of both is inversely proportional to $\sqrt{(f)}$. **[20]**

The first type of boundary layers, viscous losses result from the conversion of mechanical energy into heat. That is, when an acoustic wave incident obliquely on a rigid wall, the normal component of the velocity vanishes at the surface but even it is not obvious, the parallel component to it is also cancelled. Viscous forces are preventing the free movement of tangential to the surface wave. Since the wave attenuation occurs due to stresses caused by the lateral friction with the surface, so, if the sound is perpendicular to it there will be no viscosity losses unless the surface is porous or irregular.

The second type of boundary layers, thermal ones, here losses are caused by the heat conduction between the sound wave and the material. Sound waves experiment temperature variations in phase with their respective pressure variations. However, if the wave form had an impact on a wall, it would not be able to adapt immediately to temperature changes of the sound field, because of the high thermal capacity of the surface manner. So what appears is strong temperature gradients in the vicinity of the wall giving rise to a heat flow already named. Such exchange of thermal energy occurs at the expense of the mechanical energy of the sound wave, which fails to convert all the heat generated in the compression phase in motion.

Absorption caused by these effects is minimal when the surface is completely smooth [see Figure 4.11]. However, if the aspect is more irregular or rough the sound attenuation will be higher, increasing the space occupied by the region of losses. The difference is greater when the material is porous, as variable air flow associated with the sound wave will flow through the internal channels, so the energy is loss as switch through the boundary layer from the walls. **[20]**
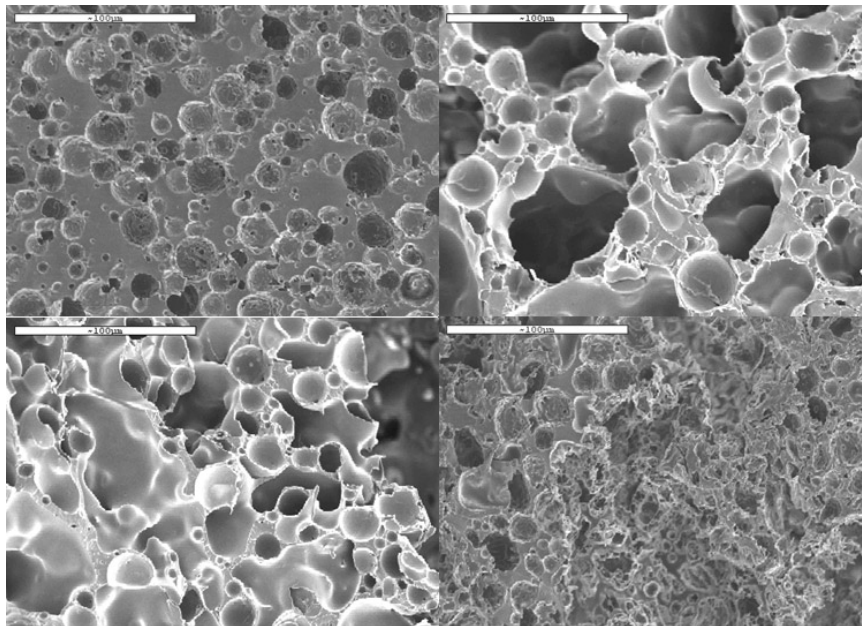

FIGURE 4.11: Boundary layer of smooth, rough and perforated surface (porous)

4.4.2 Membrane resonators

As we explained above, the porous absorbent materials are not very good for low frequencies. If we want such porous materials to adequately absorb at low frequencies, it would leave a distance to the surface or have a thickness of material that would occupy too much space, which means loss of a lot of space in the room. Moving diaphragm resonators are the ones who could play the explained absence, because they would provide quite acceptable results for medium and low-frequencies.

A resonator membrane consists of a fairly light and smooth panel. Normally the main manufacturing materials are plywood or chipboard, plaster or plastic, which would be mounted at a certain distance from a rigid and heavy surface. Thus, the existing air cavity between the two parts has an elastic behaviour above which the membrane oscillates freely. **[19]**

The membranes, as already explained are resonators, which is why these systems have a range of efficient absorption spectrum in a narrow range around its resonant frequency. If we would like to extend frequency absorption potential we would need to expand their internal losses or distribute around the room some resonators that are tuned to different third octaves.

Unlike porous materials, which only have effect to a certain distance from the walls as the vibrational wave velocity is higher, the resonators panels work in areas of high pressure, or what is the same, stuck on the walls of a room or enclosure. This property makes the membrane resonators to have to be mounted near the boundary surfaces of the rooms, which makes it possible to maximize the space itself. Because of pressure differences between both sides of the panel, a displacement occurs which achieves its maximum speed to reach the resonant frequency. This way, the internal resistance of the system show opposition to the free movement of the panel, turning mechanical energy into heat. Thus, the membrane is the responsible for passing the pressure of the sound wave to movement to dissipate a portion of it by the movements that occur inside. **[19]**

Therefore, mechanisms that are present in the absorption phenomenon of membrane resonators are the friction produced in the anchor points and the losses that occur within the membrane because of the reflection waves. These reflection waves are higher for panels of wood or plastic, and are smaller for metal panels. Eventhough all of them are small if compared with derived assembly, since they have 10 times lower level. If you would like to increase the losses it would be very interesting to partially fill the cavity with porous material.

Membrane resonators have a frequency behavior that is difficult to predict accurately. The main cause is due to how difficult it is experimentally to reproduce the values of some structural parameters the in the previous study have been assumed, concretely the damping of the edges. In addition to this cause they have to theoretical models are quite complex, so this does not facilitate the task. However, finite element mothod, FEM and boundry

element method, BEM, is a good outline for the design of these resonators theoretical model.

So, it should be noted that because of the use of space that this type of resonators offers and its good performance in bass or low frequencies, it makes the resonators membrane an useful element in the treatment of acoustic modes, and particularly in small rooms like in our case of small room, where they are more problems
[19]

4.4.2.1   Porous material in the internal cavity of the resonator

As just mentioned, the membrane resonators are characterized by relatively low absorption coefficients, about 0.3. So, if we want to increase this ratio, a good solution is to introduce porous material within the cavity in which until now had only air. This will cause the coefficient values of absorption alpha to be three times higher than without the porous material. Even if the cavities are filled partially a truly significant improvement would occur.

In any other situation it should be avoided as far as possible to have contact with the membrane, so it vibrates freely. There exists other types of systems such as those using absorbent material as elastic element or comprising the absorbent as a part of the membrane, for example the acoustic plaster. In these designs, therefore it exists the contact between the material and the panel.

This absorbent material is also able to eliminate acoustic resonance of the cavity and attenuate bending modes, but that the resonance frequency of the system will be modified. The absorbent thus causes state changes of air which is inside the cavity that are isothermal, comparing to the adiabatic behaviour occurring in the empty chamber, which has only air. Thus, the main consequence of this it has a 20% drop in the resonance frequency as shown in the following expression:

$$f_0 = \frac{50}{\sqrt{m'd}} \tag{4.1}$$

[19]

Where d is the thickness of the cavity in m and m' mass per unit area. How is placed and how the form of the porous material is into the cavity, will make the system more or less in frequency selective. This way, if the cavity is completely filled with the absorbent material, absorbent capacity of the resonator will focus on $f0$. If we want a system that is less selective and has a flatter response, we should fit it with the porous material occupying a third of the surface and the inner edges

4.4.2.2   General considerations for the design of membrane resonators

For the design of the membrane resonators it is necessary to follow the guidelines detailed in this section. This allows how to easily design a membrane resonator in a few steps and to find out which model best suited to the

required needs. However, we need to know that the guidelines are only going to detail criteria for a good design, which means that its effects will not be known until the resonator is built. Therefore, a final level of optimization is required in order to ensure the effectiveness of the designed system. Therefore, to design a resonant system, we have to follow three basic steps: choose what resonance frequency $f0$ is required, what dimensions of the plate we want and in this way maximize the absorption.

Thus, the first guideline to follow is to choose the resonant frequency, that known data is the most important and fundamental to the design organization.

First, we have to choose both the material and its thickness m' and the depth of the cavity. Second, we have to add the porous material into the cavity by applying the equation (4.1) so the desired resonant frequency will be found. Normally, $f0$ is a known fact (or chosen by the user), as it is usual that the peak absorption occurs at the desired frequency. **[19]**

And the second pattern is the dimension of the plate, which is chosen so that the first bending mode occurs at a position as close as possible to the resonance frequency $f0$. It is important, panels are chosen usually rectangular rather than square. This is done to have a more uniform distribution of structural acoustic resonances.

The last pattern is therefore optimizing absorption. This is to achieve maximum absorption at the resonance frequency $f0$ that the system is able to offer.
    - To achieve this there has to be the match of internal resistance with the radiation resistance of the system. This optimizes absorption, does not ensure that the system is frequency selective, since the selectivity depends on the internal cavity losses.

- For optimal absorption condition there should be a modification of internal resistance or radiation system.

-To change the resistance, we have to modify the conditions of fixing the plate or the presence and distribution of the absorbent material.

Control absorption of the resonator system through its internal resistance and radiation is not an easy task, since the variation of one parameter can affect the other. Although there are no strict rules for the implementation of this procedure for adjusting the impedances, it is important to take this into account when making design because it could be very useful when solving a possible malfunction of the system resonant.

4.4.3 Resonators

We can define a resonator as a container having a volume V, which communicates with the outside through a narrow and elongated opening. It works

somehow in a way that when a pressure wave penetrates into the entrance the air of the mass at the beginning starts to oscillate. In this way, power is consumed because of the air present inside provides an opposition (force) to these movements.
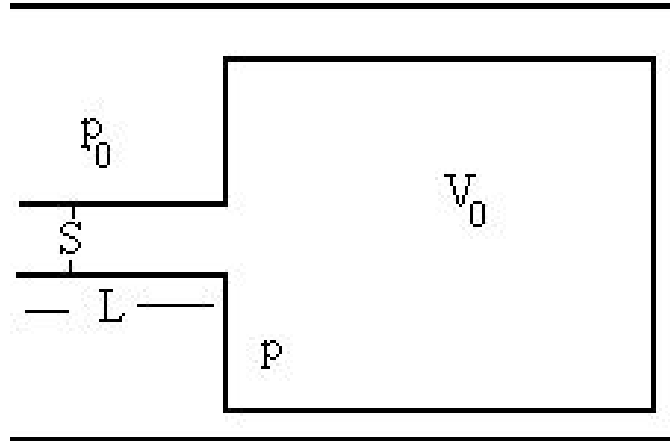

FIGURE 4.12: A simple resonator diagram

The resonance frequency is set as in the following expression:

$$f_0 = \frac{c}{2\pi} \sqrt{\frac{S_a}{VL}} \tag{4.2}$$

[19]

Where c is the speed of sound, Sa, section holes and VL the parameters shown in our figure 4.10.

The German physicist Helmholtz built one of the main resonators: the Helmholtz resonator. This is an individual absorbent formed by a volume of air enclosed within a thick cavity surface and connected with the outside through an open conduct. Its main absorption occurs in low frequencies or which are tuned to a single frequency, where all absorbent capacity is very selectively focuses on an absorption peak. Because the resonator is very selective, we have to to expand the bandwidth of the system so there has to be some internal losses.

The Helmholtz resonators are used to control the acoustic modes as problematic low frequency in small room, which is our case. As a selective resonator the bandwidth is small, and also because of enclosure modes that makes difficulties for precision in work. It is for this reason that if the working frequency is widen in the frequency of the system, it might be possible that the resonance frequency gets between two modes, worsening the acoustic conditions.

The working functionality as a resonator is caused by the vibration of air produced in the neck of the resonator. The air of the neck of the resonator behaves as if it were a mobile mass that interacts with the air within the cavity. This has a behaviour of a spring because of its elastic properties.

Thus, the resonator has the typical behaviour of the mass-spring system, so that the neck has a resistance to air flow which entails corresponding internal losses. At the same time, the outside environment creates a resistance to vibration of the moving mass.

As membrane resonators, the areas of maximum pressure (walls for example) is where the resonator has a more efficient behaviour. Because of the small size of the aperture, which is much smaller than the wavelength and that makes it possible to radiate the unabsorbed sound in all directions (diffuse radiation). It is important to high light, the Helmholtz resonator has an absorbent surface that is not flat or uniform. These resonators, like all absorbers room systems, are characterized by the equivalent absorption area A, which does not depend on the object surface unlike the absorption coefficient α

Helmholtz resonator's other feature is that it is able to radiate much of the incident acoustic energy and keeps it in a fairly narrow bandwidth. Because of the process of free vibration of the resonator sound is emitted so damped over a period of time, thus an "artificial reverberation" is generated. The large absorber and selective capacity of Helmholtz resonator can produce a reverberation time several times higher than the usual rooms.

4.4.4 Commercial absorbers systems

We already analyzed the main characteristics and theoretical models of the families of sound absorbers. Therefore, now we will proceed to describe examples of actual absorbers which provide an overview of the current market situation for the acoustic conditioning systems for our case of the class rooms.

4.4.4.1 Porous absorbers

One of the most prominent absorbing systems on the market today are porous absorbers. These are all materials that due to its internal structure of micro-cavities, which connects to the outside, achieve absorption and therefore attenuation of sound. Among the most prominent examples include mineral wool, textiles or foams.

Absorbent materials are typically coated with a layer of another material. The reasons for this are hygienic, aesthetic appearance and maintenance of equipment and exposure. That layer must be transparent to sound and thin so they do not influence on the absorber functionality. However, there are also other types of thicker coated, even though absorptions achieved with them belong to another class of membrane resonators.

Porous absorbers that are currently on the market are divided into two groups: non absorbent outer covering and those who do have outside coverage.

The first, absorber without outer covering are mineral wool that primarily use sand and recycled glass basaltic rock as raw material. The production of these absorbers has to be subjected to high temperature raw material
which causes these materials to melt and therefore that filaments of small
diameter later be joined by resin forming. Finally, as a result we obtain an
absorbent piece with a woollen texture. Rock wool mainly consists of materials such as basaltic rock. Meanwhile, the glass consists essentially of
recycled glass. [18]

Meanwhile, foams are usually found based on polyurethane or melanin,
these materials are characterized as flame retardants. Foams have to be
opened-cell, if we want to have a good sound absorption foam should have
interconnected pores and communicated with the outside, because if the
foams have closed cell will not produce sufficient absorption even whatever
the porosity they have. It is interested to say that the foams are easy to
meld and cut, for this reason, in today's market we can find various foam
panels with different forms, which will serve to improve absorption for our
studied class rooms

FIGURE 4.13:   Rockwool (left) and polyurethane foam
(right)

It is important to high light that textiles materials also have some absorbent capacity as they have fiber composition. So some examples of absorber textiles are the curtains, upholstery and carpets among others.

The second type of material found in the market nowadays are absorbers
with outer layer. Some examples of these covers are plastic films and veils.

4.4.4.2   Resonators

Finally and as we have seen so far, the resonators are acoustic absorption
systems whose main absorption occurs at a particular frequency or resonance frequency and are quite selective in frequency. Therefore, the main
resonators are Helmholtz resonators. These are formed by a large volume
cavity communicating with the outside through a neck, also filled with air,
but with a small volume compared to the volume of the cavity. The size
of the cavity makes it possible to work at low frequencies and with great
efficiency ($f < 100Hz$). Together with membrane resonators, Helmholtz
resonators the are best working in low frequency absorption. [21]

Therefore Helmholtz resonators are highly absorbent in a certain frequency range and are very selective. If we want the absorption bandwidth larger, we have to fill the cavity (so far filled with air) of porous material, so thats how the band frequency would extend. These resonators are mainly present in small rooms size, recording studios, where control of acoustic modes is really important. For very large rooms Helmholtz resonators are not too suitable solution, and not only because of its high cost, but because it would be difficult to adapt to space. However, in the current market there are models Helmholtz resonators that are integrated into bricks or even pieces to build walls.

In today's market, perforated panel resonators consist of a plate with perforations and a cavity which contains air or absorbent material. Each perforation, with the portion corresponding to the volume of the cavity resonator equivalents to a Helmholtz resonator. Therefore, the absorption capacity of these systems is concentrated around the fundamental resonance frequency, however, they are less selective than an individual Helmholtz resonator. **[21]**

4.5 Diffusers

In the other part of the project we used some special diffusers to atenuate room modes so we could have better acoustics in the class rooms. The diffusrs have been tuned in the recording studio and not in any common room beacause the recording studios's sound reinforcement is much better so it is more reliable there.

There were four diffusers that we used for our measurements and we appreciate how they look like in the following fiugure:

FIGURE 4.14: Diffuers used for the measurements

Close view of one of the diffusers to see the holes and how we can tune them by using corks:

FIGURE 4.15: close look to one of the diffusers

As seen in the figure 4.12 the diffuers have holes in them so we can use

corks to fill in them. Depending on the distribution of the corks we could atenuate some frequncies or others.

4.5.1 Diffusers in the class room 205

With diffusers we used the same methodology of work as in the class room 205 but without them. In the class room 205 the grid we made was of low reslution with 225 points for the measurements (see figure 4.7).

Also we calculated the impulse response of the class room with Easera program by going through with a microphone over all the 255 points in the class room as appreciated in the following figure:

FIGURE 4.16: Passing the microphone over all the points of
the grid of the class room 205

And the distribution of the diffusers were putting two of then at the end on of the class room one over another, and one at each side of the lateral walls as shown in the panoramic figure 4.15

FIGURE 4.17: Distribution of the diffusers in the class room
205

4.5.1.1   Results of the measurements

After having moved the microphone over these 225 points with Easera the impulse response was obtained for each point. And for each point *.wav* and *.emd* files were saved in order to obtain wide range of information. So with these files it was posible to have all the necesary information from where it is posible to get all the important parameters of room acoustics and analyze the data with *matlab*.
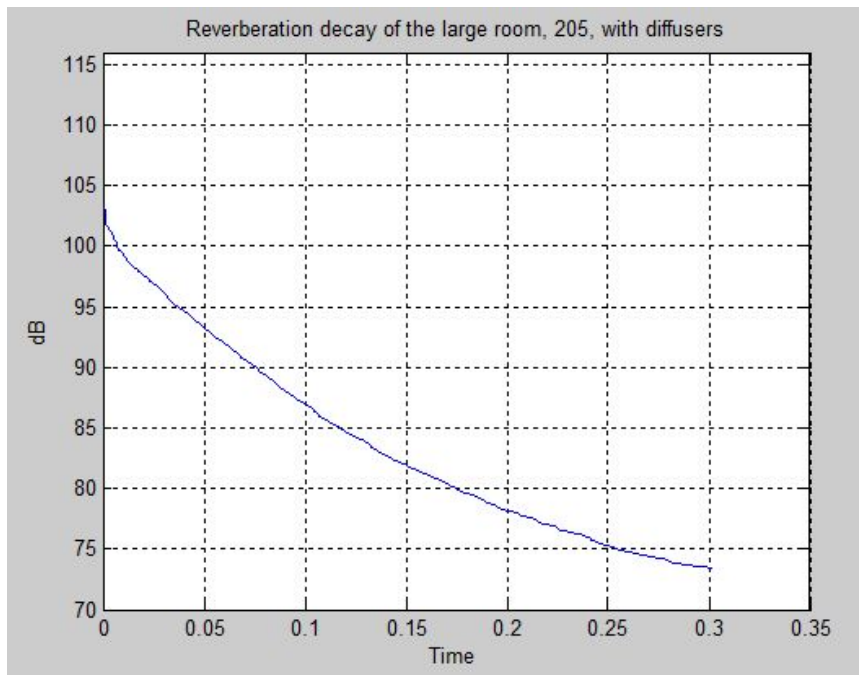

FIGURE 4.18: Reverberation decay of the large room 205
with diffusers

In the picture 4.18 we can see the reverberation decay calculated with schroeder's reverse integration in one of the points in the class room.

With the diffuser in the class room 205 the reverberation time starts decaying at $103dB$ untill $73dB$. It is clearly appreciated that with the diffuser the reverberation time starts decaying $1dB$ earlier and stops $6dB$ after. Thats means the difference to the same class room without diffusers is of $5dB$ as now the total decaying is of $30dB$ in the class room and before without diffusers it was $25dB$ in total.

4.5.2 Diffusers in the class room 202a

And for the class room $202a$ the high resolution grid was used with 165 points to measure on the floor.In this class room there was a room mode in the frequecy of $f = 46Hz$ so we tuned the diffuers to that frequecy in order to attenuate that room mode.

In in this room $202a$ the distribution of the diffusers were as shown in the next figure, four of them at the end of the class room and we passed microphone over all the 165 points for the measurements.

FIGURE 4.19: Distribution of the diffusers in the class room
$202a$

4.5.2.1 Results of the measurements

After having moved the microphone over these 165 points with Easera the impulse response was obtained for each point. And for each point *.wav* and *.emd* files were saved in order to obtain wide range of information. So with these files it was possible to have all the necesary information from where it is posible to get all the important parameters of room acoustics and analyze the data with *matlab*.
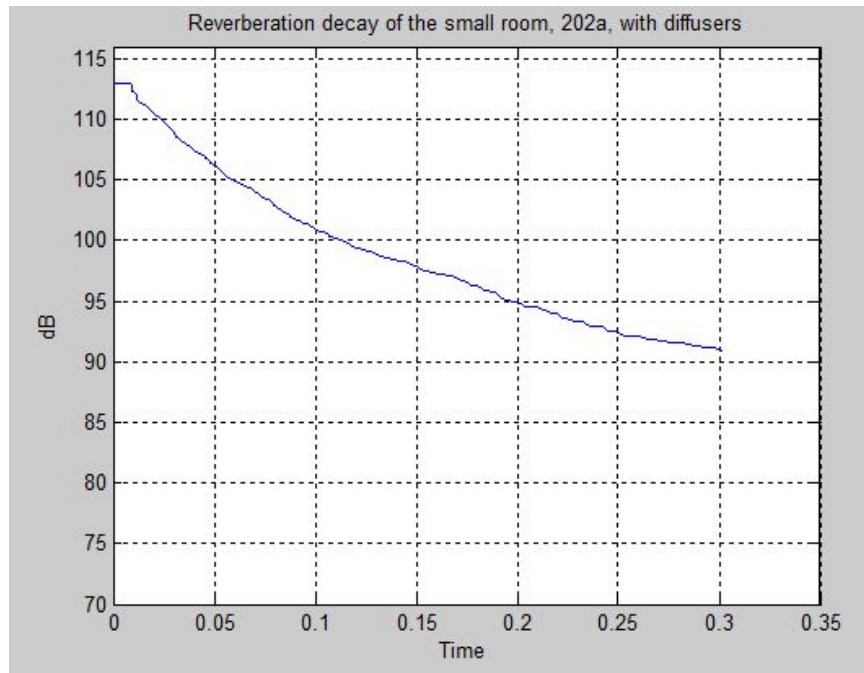
FIGURE 4.20: Reverberation decay of the small room $202a$
with diffusers

In the picture 4.20 we can see the reverberation decay calculated with schroeder's reverse integration in one of the points in the class room.

With the diffuser in the class room $202a$ the reverberation time starts decaying at $113dB$ untill $91dB$. It is clearly appreciated that with the diffuser in the same room the reverberation time starts decaying $3dB$ earlier and stops $6dB$ after. Thats means the difference to the same class room without diffusers is of $3dB$ as now the total decaying is of $22dB$ in the class room and before without diffusers it was $19dB$ in total.

Chapter 5

Conclusions

Once exposed the theoretical foundamentals, project development and analysis of results, it is very interesting to high light some of the conclusions and achievements that have been reached in this project and expose a possible proposal for future work.

High frequencies are easy to absorb with absorbent material (porous for example), but the low frequencies are the ones difficult to absorb. That is why we have analyzed that in the class rooms, even if it is 205 or $202a$, for the absorption of low frequencies we can use membrane resonators or Helmhotz resonators.

Room modes, which could be another problem in the class rooms, we experimented that it could be solved by using the diffusers. In concrete, in the class room $202a$, a room mode at the frequency $f = 46Hz$ has been atenuaded by tunning the diffuser to that frequency in order to get a clean sound without reverb replicas.

The reverberation time in small rooms is shorter that is also demonstrated. Normally, if the reverberation time is shorter in a room, the inteligibility is better and it has better acoustics. Sabines reverberation time expression works better for large chambers but it is not really reliable for small rooms.

As a proposal for the future work, it would be interesting to see the practical application of the diffusers. For example, placing them in a recording studio to check how they affect on its acoustics. And also place the absorbing material in to the class rooms to calculate the absorbing coefficients.

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