CHALMERS
UNIVERSITY OF TECHNOLOGY

Assessment and improvement of the acoustical properties of a testing room for bone anchored hearing aids

Master of Science Thesis in the Master's Programme Sound and Vibration

MÓNICA LORÍA GAMBOA

Department of Civil and Environmental Engineering
Division of Applied Acoustics
Rooms Acoustics Research Group
CHALMERS UNIVERSITY OF TECHNOLOGY
Gothenburg, Sweden 2015
Master's Thesis 2015:93

MASTER'S THESIS 2015:93

Assessment and improvement of the acoustical properties of a testing room for bone anchored hearing aids

Master of Science Thesis in the Master's Programme Sound and Vibration

MÓNICA LORÍA GAMBOA

Department of Civil and Environmental Engineering
Division of Applied Acoustics
Room Acoustics Research Group
CHALMERS UNIVERSITY OF TECHNOLOGY
Gothenburg, Sweden 2015

Assessment and improvement of the acoustical properties of a testing room for bone anchored hearing aids
Master of Science Thesis in the Master's Programme Sound and Vibration
MÓNICA LORÍA GAMBOA

Master's Thesis 2015:93

Department of Civil and Environmental Engineering
Division of Applied Acoustics
Room Acoustics Research Group
Chalmers University of Technology
SE-41296 Gothenburg
Sweden

Tel. +46-(0)31 772 1000

Reproservice / Department of Civil and Environmental Engineering
Gothenburg, Sweden 2015

Assessment and improvement of the acoustical properties of a testing room for bone anchored hearing aids
Master of Science Thesis in the Master's Programme Sound and Vibration
MÓNICA LORÍA GAMBOA
Department of Civil and Environmental Engineering
Division of Applied Acoustics
Room Acoustics Research Group
Chalmers University of Technology

Abstract

Subjective testing of bone anchored hearing aid users allows the bone conduction hearing aid technology to be improved. Properly calibrated measurement equipment and a good sound field in a testing room are the keys to good quality measurements.

The work was carried out for Cochlear Bone Anchored Solutions (BAS) located in Mölnlycke. The hearing laboratory at Cochlear consists of a test room, a sound booth and a control room. The room of interest for this work is the test room called "Soundroom". This room is acoustically isolated and contains an array of 16 loudspeakers, placed on a circle. The room is used for tests with bone anchored hearing aid users such as speech-in-noise, sound scenarios, threshold of hearing among others. It is also used to perform engineering tests.

The purpose of this work is to measure and improve the equalization of the speakers and the sound field in the room. It includes an assessment of the acoustical parameters of the room and evaluating enhancements. Further it presents suggestions for the improvement of the calibration procedure itself.

The results show that the acoustical parameters of the room are almost good enough according to standards and literature. The main problem in the room was a strong reflection between two parallel surfaces, the floor and the ceiling which can potentially lead to coloration of the test signals. With an appropriate treatment of the floor the problem can be avoided.

In order to equalize the frequency response of the speakers, an inverse filter was tested and proved to be fruitful. A potential downside of the procedure is that the reproduced sound can be affected by the corrections.

Keywords: room acoustics, sound field in small rooms, bone anchored hearing aids, sound insulation, room impulse response, inverse filtering.

Contents

Acknowledgements

Thanks be to God in the first place, since his will was for me to have this amazing opportunity, and because of his mercy, I conclude this step in my education. I also thank my parents, the only ones who believed this was possible, for allowing me to know my true strength and courage and for their constant support.

My greatest gratitude for my supervisor at Chalmers, Erkin Asutay for all his valuable help, support and reassurance through all the steps of the way. Also many thanks to my supervisor at Cochlear, Jona Hoffmann for trusting me with this project and always giving me a hand when I needed it.

Many thanks to the personnel at Cochlear, for giving me such a warm welcome and bearing with me every time I needed to be in the lab for long hours.

I would like to sincerely thank Wolfgang Kropp for being so supportive, for all the advice and for being such a good professor, my career is completely marked because of you.

Thank you family and friends in Costa Rica, for the prayers and support. Thank you all my dear friends in Sweden, you have a piece of my heart.

I dedicate this work to Per Engström, see you soon my friend.

1 Introduction

The work was carried out for Cochlear Bone Anchored Solutions (BAS), a medical device company which provides bone conduction hearing solutions. It is located in Mölnlycke and employs more than 200 people whose responsibilities include research, product design and development, marketing, quality and regulatory, manufacturing, distribution and administration of products such as Baha® and Vistafix®.

The hearing laboratory at Cochlear consists of the "Soundroom", the "Soundbooth" and the "Control Room". The room of interest for this work is the Soundroom, which is acoustically isolated and contains an array of 16 loudspeakers, placed on a circle. This room is used for technical measurements and clinical studies for the development of bone conduction hearing aids. All kind of technical measurements, like feedback measurements, input and output characterisations, parameter tuning, signal to noise measurements and multi factor analysis are performed this room.

The main purpose of the room is the evaluation of new products with bone conduction hearing aid users. Their hearing abilities are evaluated using an audiometer, where the main speaker used is calibrated yearly by certified personnel. A touchscreen interface allows the easy selection of the active speakers. The tests include evaluations of hearing thresholds, speech perception in quiet and noise, directionality testing, loudness perception and subjective evaluation of sound scenarios. The frequency range of interest for these kind of devices is between 250 Hz and 7000 Hz.

Properly calibrated measurement equipment and a good sound field in the room are the keys to good quality measurements. That is why the frequency response functions (FRF) of all the speakers should be as flat and similar as the calibrated speaker as possible, therefore parametric equalizers have been used to correct the responses in third octave bands.

The purpose of this work is to improve the equalization of the speakers and the sound field in the room. It includes an assessment of the acoustical parameters of the room, evaluating enhancements, and it also presents suggestions for the improvement of the calibration procedure itself.

1.1 Objectives

- Measure the acoustic properties of the Soundroom: reverberation time, ambient noise, attenuation of the sound from the Control Room.
- Measure the combined frequency responses of the room and the loudspeakers.
- Assess the reflections in the the combined impulse responses.
- Compare to earlier investigations and standards.
- Investigate improvements of the setup to achieve a flat frequency response from all speakers, by acoustic and electronic means.
- Recommend improvements of the yearly calibration procedure.

1.2 Description of the rooms

The rooms are located on the third floor and are part of the inner core of the building, figure 1.1 shows the neighboring rooms.

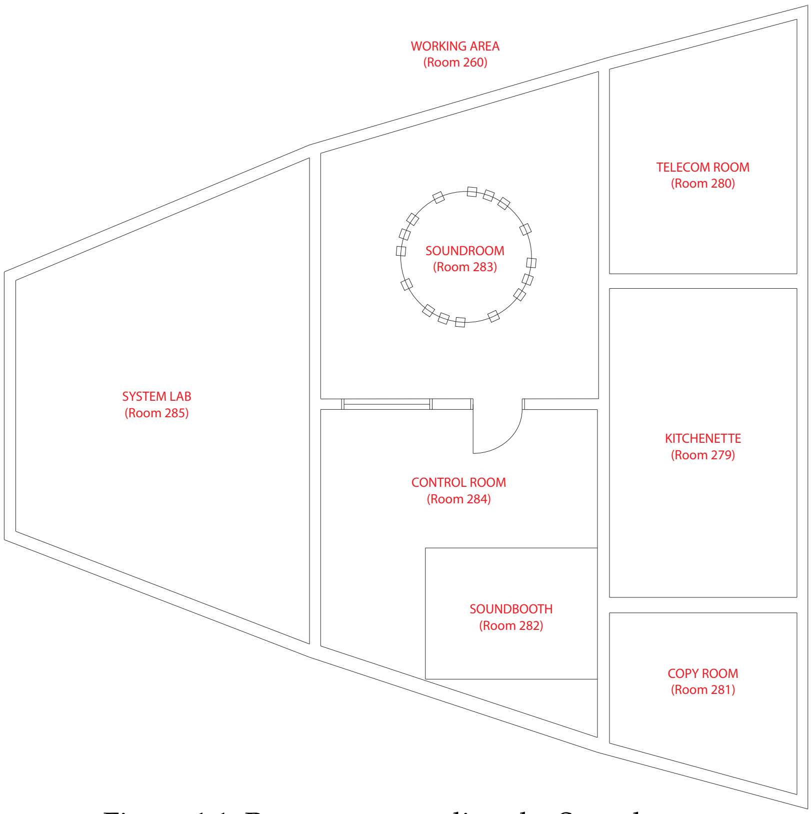


Figure 1.1: Rooms surrounding the Soundroom.

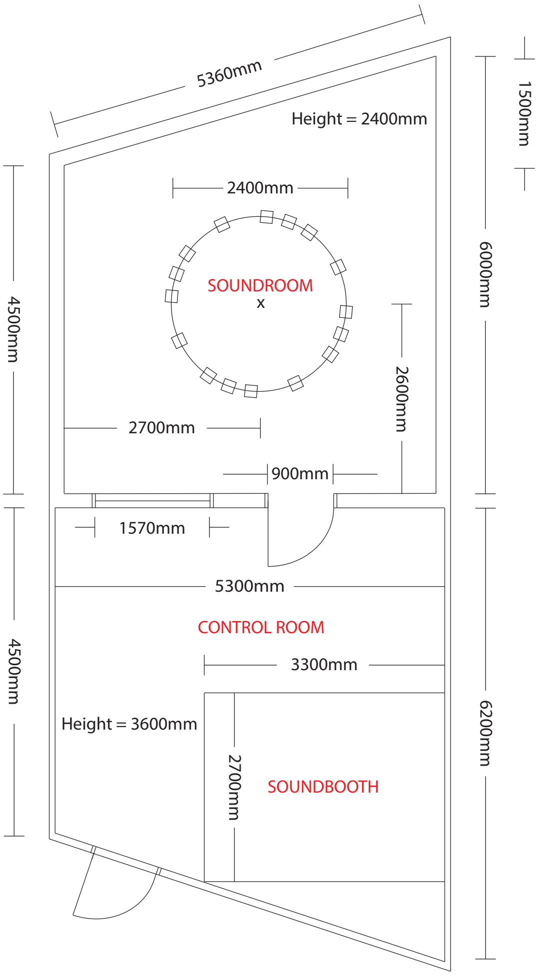


Figure 1.2: Dimensions and layout of the rooms.

The Soundroom was built first, then the Soundbooth was added. The dimensions of the rooms are shown in 1.2.

The Soundroom has a volume of 64 m^3. The area of the wall (partition) between the Soundroom and the Control Room is of 12 m^2. This partition is a cavity wall, consisting of two leaves with an air gap in between. It is 17 cm thick, made of gypsum, glass wool and some other insulation materials.

The Soundroom has treated walls and ceiling. The walls are type *Ecophon Super G™ A* or similar and the ceiling is type *Ecophon Master™ Ds* or similar. The ceiling construction can be seen in figure 1.3.

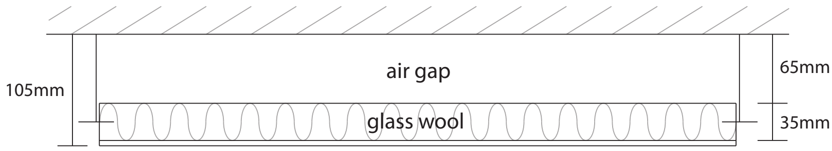


Figure 1.3: Ceiling section.

The ceiling of the Control Room is much higher than the actual ceiling of the Soundroom. There are water and ventilation pipes running above the Soundroom as seen in figure 1.4.

Figure 1.4: Pipes coming from the Control Room into the Soundroom.

The floor is made of wood and gypsum. It was raised from the concrete floor using wooden bars and has a carpet on top, its section is shown in figure 1.5.

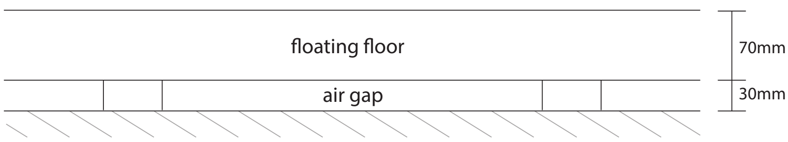


Figure 1.5: Floor section.

The Soundroom contains a loudspeaker array at a distance of 1.3 m from the floor, which can be seen in figure 1.6.

Figure 1.6: Loudspeaker array.

There are other items of importance in the room, the window and the desk with the computer which are shown in figure 1.7. The window is 1.57 m x 0.67 m. It has two glass panes, separated 7 cm between them, one of the leaves with a 5 degree inclination.

Figure 1.7: View of the window and the desk.

The Control Room has a volume of 102 m^3 and has no treatment, except for the ceiling as shown in figure 1.4. The walls and the floor are reflective and there are several pieces of reflective furniture. The Soundbooth's outer walls are also reflective. All of this could produce a non-diffuse sound field. In figure 1.8, some views of the Control Room are shown.

Figure 1.8: Control Room.

2 Theory

2.1 Basic fundamentals of hearing

2.1.1 Anatomy of the ear

The ear consists of three parts: the outer, the middle and the inner ear (figure 2.1). The external and middle ears capture sound energy and couple it to the cochlea.

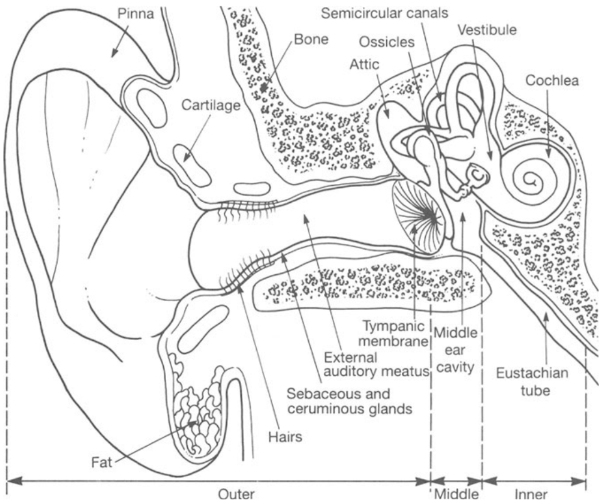


Figure 2.1: Representation of the anatomy of the human ear [Tate, 1994, p. 21].

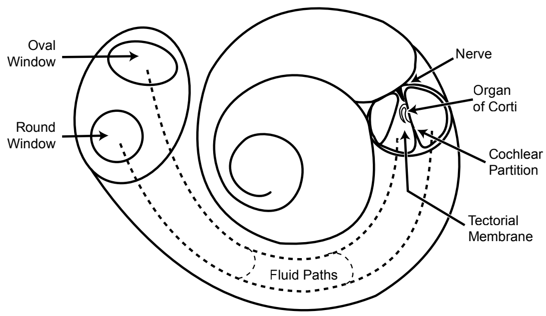
The outer part of the ear consists of the *pinna*, and the *external auditory meatus* or *canal*. The *pinna* gathers the sound and helps localizing the sound source and the *auditory canal* conducts the sound waves to the *tympanic membrane* (ear drum). The ear drum divides the outer from the middle ear.

The middle ear contains the *ossicles*, the three bones which act as an impedance matching and mechanical link to transfer the power from the eardrum to the inner ear. These bones are connected to the cochlea, through the *oval window* which is a membrane located at the entrance of the cochlea.

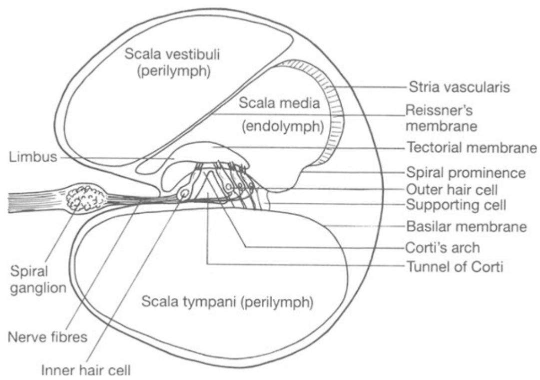
The cochlea is a rolled up tube filled with fluid. It contains hair cells that sense sound. Two membranes divide the cochlea. The thicker one is the *basilar membrane*, it divides the cochlea in two galleries, the upper one which is called *scala vestibuli* and the lower one, called *scala timpani*. In this membrane lies the *auditory nerve* which conducts the information to the brain. The upper gallery is divided from another section called *scala media* by the *Reissner's membrane*. Here is located the *organ of Corti*, which has the hair cells that act as transducers between the motion of the fluid to electrical impulses.

In the end of the cochlea (near the *apex*) there is an opening called the *helicotrema* that connects the two galleries and allows the fluid to flow into both of them. At the opening side there is other membrane called the *round window* and it is a pressure reliever for the fluid inside of the cochlea.

Figure 2.2 shows two diagrams of the cochlea, the first one (2.2(a)) shows the whole cochlea and the second one (2.2(b)) its cross section.



(a) Cochlea [Wikimedia Commons, Cochlea].



(b) Cross section of the cochlear duct [Tate, 1994].

Figure 2.2: Diagrams of the cochlea.

2.1.2 Physiology of hearing

The hearing system has two main functions, reception and conduction of the sound and the analysis of it [Tate, 1994, p. 35].

The reception and conduction of the sound is performed by the *pinna*, the *external auditory canal*, the *tympanic membrane*, the *ossicles* and the cochlear fluids. Any damages or malformations to these parts, produces a conductive hearing loss.

For the perception of the sound, the important part is the *organ of Corti* where the sound is analyzed into its frequency components and then sent to the brain via the *auditory nerve*. Damages in the perceptive section causes sensorineural hearing loss.

2.1.3 Conductive hearing loss

Conductive hearing loss can be caused by some abnormality of the outer or middle ear that blocks the sound from being transferred from the outside to the *tympanic membrane*. The principal causes are:

- wax blocking the *auditory canal*
- infections or inflammation of the outer or inner ear (otitis)
- abnormal growth of bone (otosclerosis)
- absence of the *auditory canal* or complete or partial closure (atresia or stenosis)

2.1.4 Bone anchored hearing aid

The bone anchored hearing aid works on the principle of bone conduction. It is the conduction of the sound to the inner ear by the excitation of vibrations on the skull (figure 2.3). It is mostly used in the cases of conductive hearing loss.

This type of hearing aid uses a sound processor. It first captures sounds in the air, then it turns the sound into vibrations and sends them through the abutment (magnetic connection) to an implant (titanium prosthesis screwed to the skull), which transmits the vibrations to the inner ear [Baha® , Cochlear].

The frequency range of operation of the sound processors lay in between 250 Hz and 7000 Hz [Baha® 5, Cochlear].

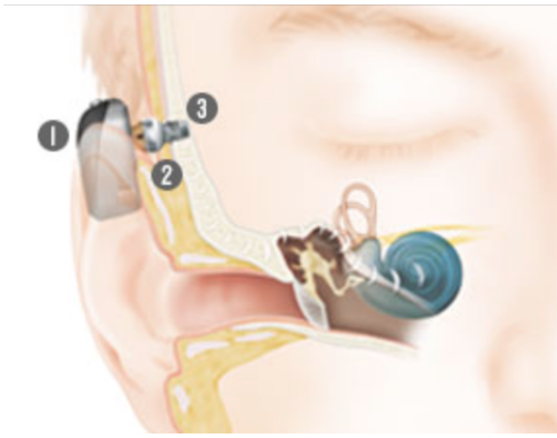


Figure 2.3: Bone anchored hearing aid - 1. Sound processor, 2. Abutment, 3. Implant [Baha®, Cochlear].

2.2 Room acoustics

Room acoustics can be divided into three branches, *wave theory*, *geometrical* and *statistical room acoustics*.

The *wave theory* describes how the sound pressure in a room is distributed. Since it is about the exact mathematical solution to the wave equation it is impractical for high frequencies (due to a high modal density) and for complex room geometries (due to complex boundary conditions).

Geometrical acoustics describes the reflections in a room. The same modelling principles as in geometrical optics can be used (ray tracing and mirror images) to study the propagation of sound. To apply this theory, the surfaces should be larger than the wavelengths of the sound and so it would not work for low frequencies.

Finally the *statistical room acoustics* is based on the analysis of the energy distribution in rooms, and how this sound is absorbed under stationary conditions [Kleiner, 2012].

2.2.1 Statistical room acoustics

At higher frequencies, the modes increase in density and so the *wave theory* cannot longer be used and so this statistical approach is a good option to get an accurate description of the behavior of a room. A completely diffuse field is assumed. This method has its limitations since the sound field in a real room is usually far from ideal (not completely a diffuse field) [Dietze, 2010].

Diffuse sound field theory

At higher frequencies there is a high modal density and so room behavior is difficult to analyze without using energy density or statistical considerations. At this region "it is customary to describe the space in terms of a statistical model known as a diffuse field. A diffuse field is one in which there is an equal energy density at all points in the room. This implies that there is an equal probability that sound will arrive from any direction" [Long, 2006].

An ideal diffuse sound field does not exist in reality, but the diffuse sound field theory can be reliable for the evaluation of the behavior of a room if some things are taken into consideration (recommendations from standards [ISO 3382-2]).

Reverberation time

The reverberation time (T_{60}) is defined as the time it takes for the energy in the room to drop to one-millionth of its value (sound pressure level drop of 60 dB) after the source has been switched off [Kleiner, 2012].

It can be calculated as shown in figure 2.4. To ensure that the reverberant sound is diffuse the start of the measurement is usually 5 dB below the steady-state value as seen in the figure.

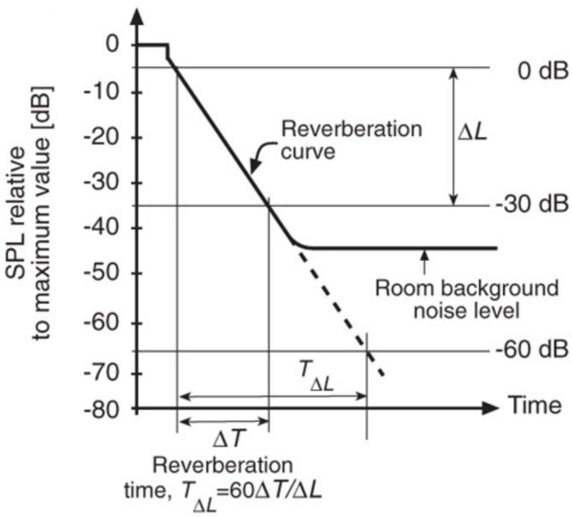


Figure 2.4: Example of the determination of reverberation time when room background noise level is high [Kleiner, 2012].

It is usually hard to have an excitation 60 dB above the background noise, and so it can be calculated with the estimation of the decay over a 15 (T_{15}) or 20 dB (T_{20}) span and then extrapolated (the T_{15} would then be multiplied by 4 and the T_{20} by 3).

Schroeder Frequency

The sound in rooms can be analyzed in different regions according to its frequency and therefore its wavelength. At low frequencies (pressure zone) only plane waves are formed (when half the wavelength is greater than the longest dimension of the room). Plane waves travel in the form of wave fronts, normal to the wave propagation. Higher in frequency, the modes start to form (*normal modes region*) [Long, 2006]. At this region the *wave theory can be used*.

At higher frequencies (statistical region) the *statistical room acoustics* is used. Figure 2.5 shows the regions mentioned before.

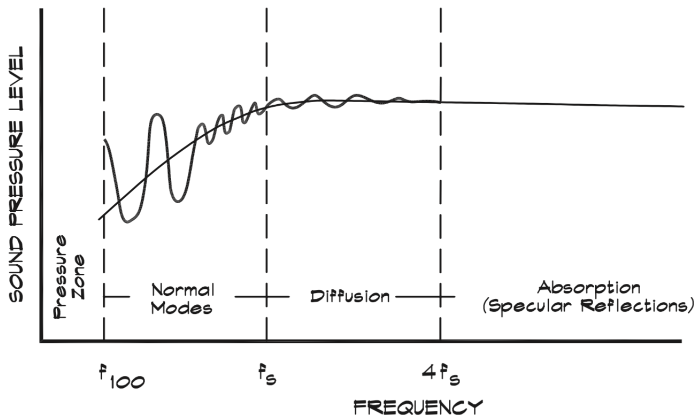


Figure 2.5: Room response divided by frequency regions [Long, 2006].

As seen in figure 2.5, f_{100} or$f_{1,0,0}$ is the frequency where half of the wavelength fits in the largest dimension of the room (first mode) and therefore indicates where the *normal modes region* begins. One can also see the transition between the *normal modes region* and the *statistical region*, which is called the *Schroeder frequency* and it can be estimated as in equation (2.1) where V is the volume of the room and T_{60} is the reverberation time. It is the boundary where individual modes cannot longer be seen, but several modes add up.

$$f_S = 2000\sqrt{\frac{T_{60}}{V}} \tag{2.1}$$

2.3 Acoustics of small rooms

A small room is a room with low reverberation, which does not have a completely diffuse sound field. The sound field in a small room consists of a direct sound, several strong early reflections and weaker late reflections [Dietze, 2010]. The sound reproduction in a small room can be affected in several ways, depending on the frequency region.

2.3.1 Room impulse response

When recording a sound in a room, we can have information from the direct sound coming from the source and also the sound that has been reflected on all the surfaces. The room can therefore be modelled as a black box with the direct sound as the input and the recorded sound as the output.

The room IR is only valid for the specific speaker and microphone location used in the room [Granbom, 2014].

From an IR, one can obtain information on the reflections the sound experienced. In figure 2.6 a reflectogram can be seen, which in this case could be considered as the simplification of an impulse response.

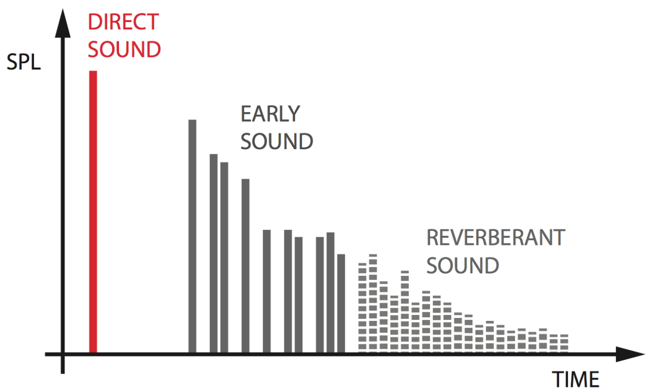


Figure 2.6: Reflectogram displaying the direct sound, the early sound, and the reverberant sound [Gjers, 2014].

The direct sound is seen as the component with the highest amplitude, then the reflections come later in time, depending on the distance between the loudspeaker, the different artifacts or boundaries and the microphone.

Using the Fourier transform, one can estimate the frequency response function (FRF) of the room from the IR. The FRF has two parts, the magnitude response and the phase response. The magnitude response describes the behavior of the amplitude through the frequency and the phase response describes the delay of the system.

"In room equalization it is more suitable to treat the whole chain of sound source, amplifier, cables, loudspeakers and the room as a single system because all of these parts will alter the sound that the listener perceives in the end" [Granbom, 2014].

It is important to know that reflections affect the direct signal and this effect depends on their magnitude and phase. The magnitude is related to the distance traveled by the wave and also the sound absorption coefficient of the surfaces. The phase is also related to the distance, also the wavelength, the surfaces (reflective or absorptive) and the angle of incidence of the wave.

Looking at it in a simplified way, the time it takes for any reflection to arrive at the listening position (after the direct sound) can be calculated as in equation (2.2).

$$t = \frac{d_2 - d_1}{c} \tag{2.2}$$

Where d_2 is the reflected path, d_1 is the direct path and c is the speed of sound in air.

And so the relative phase between the direct and the reflected wave for a specific frequency (hence wavelength) can be calculated as in (2.3)

$$\phi = 360 \times \frac{d_2 - d_1}{\lambda} \tag{2.3}$$

Where ϕ is the phase shiftand λ is the wavelength.

If the phase difference is 180^o, there would be a cancellation or antiresonance (since the direct sound's phase is opposite to the reflected sound's phase). If it is 0^o then both pressures would add up.

In real life the absorption coefficient is frequency dependent, angle of incidence dependent, and may itself affect the phase of the reflected wave.

Energy decay curve (EDC)

From the impulse response, one can obtain the EDC to calculate the reverberation time T_{60}. The EDC is the integral of the squared impulse response (h) at time t as seen in equation (2.3.1).

$$EDC(t) = \int_{t}^{\infty} h^2(\tau)\, d\tau \tag{2.4}$$

The EDC decays more smoothly than the impulse response itself, and so it is useful for estimating T_{60} [Smith, 2010].

2.3.2 Room modes

Modes are standing waves with regions of high and low pressure. They are the pathways available for a wave to travel. The resonances associated with the modes are called *eigenfrequencies* (natural frequencies of resonance in the room). There can be *axial modes* (between two parallel surfaces), *tangential modes* (between four surfaces and travel parallel to the other two) and *oblique modes* (between all six surfaces) [Newell, 2012].

The frequency of a specific mode f_{q_x,q_y,q_z} in a rectangular room can be calculated using equation (2.5), where q is the order of the mode in each direction, l is the length of the room in each direction and c is the speed of sound [Kleiner, 2012].

$$f_{q_x,q_y,q_z} = \frac{c_0}{2}\sqrt{\left(\frac{q_x}{l_x}\right)^2 + \left(\frac{q_y}{l_y}\right)^2 + \left(\frac{q_z}{l_z}\right)^2} \tag{2.5}$$

For *axial modes*, only the x term is used; for *tangential modes*, the x and y terms are used, and for *oblique modes* all the terms are used.

The existence of room modes can affect the listening when they are not damped, also when the modal density is low and the spacing between modes is large, the dips in the frequency spectrum can be easily audible, producing a coloration and unevenness of the sound.

2.3.3 Interferences in the frequency response

When having coherent signals being summed up, positive and negative interferences can lead to an uneven frequency response. When there are several interferences (due to strong reflections), the *comb filter effect* is produced.

Figure 2.7 shows in (a) and (b) an impulse response and its frequency response. In the impulse response, one can see that there is a direct sound and a delayed sound by Δt with a gain of g. In (c), it is displayed the value of the gain g in dB for which the coloration is just perceivable, for a certain delay (in this case for white noise).

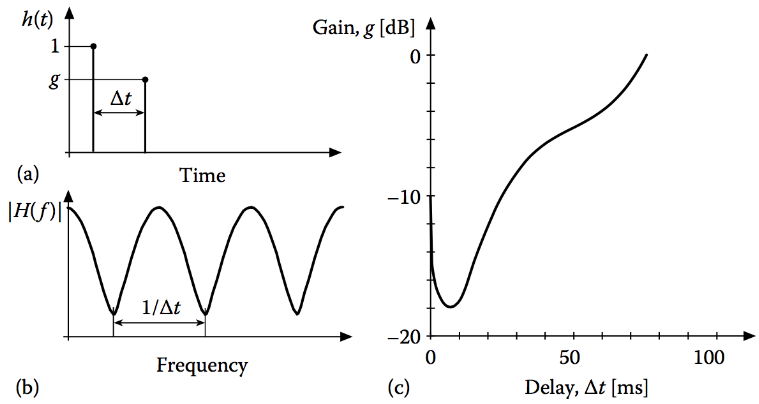


Figure 2.7: An impulse response and frequency response magnitude with a delayed signal. (a) Impulse response $h(t) = \delta(t) - g\delta(t - \Delta t)$, (b) frequency response, (c) gain factor g that gives just perceptible coloration of white noise [Kleiner, Tichy, 2014].

In general the interferences themselves (dips) are only perceivable when the time interval between the reflections is more than 30 ms, and when the time is shorter, the ear tends to integrate the sound into a continuous one. Since 30 ms is the limit for the effect to be audible, this means that the path length (between the reflective surfaces) should be less than 10 m, but even if the dips are not perceived, the coloration will remain [Newell, 2012].

When one has reflective parallel surfaces in a room, a repetitive infinite IR is produced, and this causes a stronger *comb filter* in the frequency response as shown in figure 2.8. The effect is more noticeable when only two reflective surfaces exist, for example, when having hard ceiling and floor while the walls are damped.

When the direct sound is long, the effect of having repetitive reflections produce a change in the timbre (due to coloration). When the direct sound is short (shorter than Δt), the effect produced is called *flutter echo*, causing the sound to have a metallic ring character.

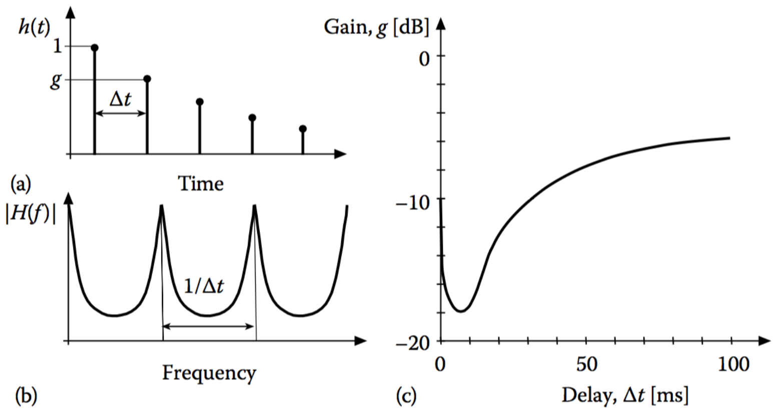


Figure 2.8: An infinite impulse response and frequency response magnitude. (a) Impulse response $h(t) = \sum g^n \delta(t - n\Delta t)$ $(n = 0, 1, 2, 3, 4, ...)$, (b) frequency response, (c) gain factor g that gives just perceptible coloration of white noise [Kleiner, Tichy, 2014].

2.4 Improving the sound reproduction in a small room

The sound reproduction can be improved by acoustical and electronic means (and also a combination of them). The acoustical means are related to adding absorption, adding diffusion, changing the position of the loudspeakers (or the listening position), having non-parallel walls, among others. The electronic improvements can be using equalizers, filters, trying to correct the effects of the room by adding loudspeakers, or other means to try to flatten the frequency response.

All methods have limitations and advantages. In some cases some of them could be useless or detrimental to the overall response or the sound being reproduced.

2.4.1 Acoustic correction of rooms

Sound absorption

Absorption converts sound into heat. Absorbers are used mainly to reduce the sound levels or to improve the sound field of a room. The absorbers can be resonant (membrane, Helmholtz) or non-resonant (porous). The ratio of absorbed to incident sound energy is called absorption coefficient α, and it varies with the frequency. Porous materials are much more sound absorbing at high frequencies than at low frequencies. For frequencies below 250 Hz, resonant absorbers are generally used [Newell, 2012].

Sound diffusion

As mentioned before, the sound reflection in planar surfaces can produce the *comb filter effect*. In some cases, difussion can remove this unwanted effect. For a diffuser to work as wanted, it needs to have surface irregularities the size of at least a quarter of a wavelength (at the frequency of interest), therefore the sound diffusers can be used for mid-frequency ranges [Kleiner, 2012].

Parallel surfaces

Having parallel surfaces produce the repetitive reflections that can cause a strong *comb filter effect* as mentioned before. The term *parallel* is frequency dependent, this means that non-parallel surfaces (by angling one of the surfaces) can seem to be parallel at low frequencies.

The angling of a surface can be effective as long as the path length differences for the reflections must be a significant part of the wavelength. When considering the frequency of 50 Hz, the wavelength is about 7 m and so the angle needed for it to be non-parallel (at 50 Hz) would be quite large.

At higher frequencies, the modes are changed from *axial* to *tangential* (because of the angling) and these kind of modes have more complicated paths to travel and they hit the surfaces at oblique angles, therefore they loose more energy than when they are perpendicular (*axial modes*).

The angling is more useful to control the mid and high frequencies. For low frequencies it is better to combine it with absorption.

2.4.2 Electronic correction of rooms

A room can be improved by electronic means but hardly completely corrected. Next are described two methods, by equalization and by applying an inverse filter.

The equalization objective is to increase the sound power output if a speaker cannot provide it. It can be used when a room affects directly the response of a loudspeaker, such as by loading the diaphragm. If the response is affected by reflections (resonances), the equalization is not useful.

These two situations are minimum phase and non-minimum phase respectively. "A minimum phase response is one where every change in the amplitude response has a corresponding change in the phase response, and vice versa. When the restoration to flatness of either response does not restore the other, the response is said to be non-minimum phase, and cannot be corrected by a causal inverse filter" [Newell, 2012].

Also for the case of non-minimum phase, every different listening position would have a different response, due to the difference in path length and reflections from the boundaries, therefore this complex behavior cannot be fixed by equalizing the response. If one tries to correct one of these responses using for example third octave filters, the amplitude response can look better but the phase response can develop anomalies which can distort the time response.

An inverse filter can be calculated and used to correct either the minimum or the non-minimum phase components of the response but the results are useful over a narrow area.

Two methods can be used for improving a room response:

- Correcting the magnitude response (no correction of the phase): minimum phase filtering.
- Correcting both the magnitude and the phase responses: non-minimum phase filtering.

According to Dietze [Dietze, 2010], the minimum phase filtering is good enough for basic correction tasks and the phase response errors are not usually perceivable. Hence in general there is no need for non-minimum phase filtering aiming at a phase response correction.

The desired frequency response can be an ideal system response (flat), or a version with the room acoustics effect minimized, or also a version with the effects of bad loudspeaker response corrected.

2.5 Apparent sound reduction index R'

The apparent sound reduction index R' is the ratio of the sound power incident on a partition (from the sending room) to the sound power transmitted through it into the receiving room. It is used to rate the insulation provided by a partition.

The equation used to calculate R' is shown in (2.6) [ISO 140-4, p. 3].

$$R' = L_{sending} - L_{receiving} + 10log\left(\frac{S}{A_{receiving}}\right) \text{ dB} \tag{2.6}$$

Where L is the level in the sending room and the receiving room respectively, S is the partition area and $A_{receiving}$ is the equivalent sound absorption area in the receiving room. To make this calculation, it is needed to have a diffuse field in both rooms.

2.5.1 Weighted apparent sound reduction index R'_W

This index is the value in dB of the reference curve at 500 Hz after shifting it according to the method explained below.

To carry out the comparison, the standard ISO 717-1 was used. The values obtained in the previous section are compared with the reference values in table 2.1.

Table 2.1: Reference values for airborne sound [ISO 717-1, p. 4]

Octave band center frequency (Hz)	Reference values (dB)
125	36
250	45
500	52
1000	55
2000	56

To evaluate the results, the reference curve is shifted towards the measured curve in steps, until the sum of the deviations is as large as possible but less than 10 dB. these deviations are considered when the measurement is less than the reference at a certain frequency. The value of this reference curve at 500 Hz in its final position is the apparent weighted sound reduction index R'_W.

2.6 Room requirements for audiometry tests

2.6.1 Ambient noise

The standard ISO 8253 states the maximum permissible ambient sound pressure levels (SPL) for Hearing Level (HL) measurements.

If an individual's hearing threshold measurements (HTL) are to be measured, the ambient noise of the room should be low enough so the HTL is actually measured instead of it. "If measuring to 0 dB HL is the objective then the background levels must be significantly below the SPL of the applied test signal over the whole frequency range measured and for bands that may mask the measured bands" [Williams, 2009].

All test environments should meet the ambient noise requirements for audiometric testing as specified in table 2.2.

Table 2.2: Maximum permissible ambient noise levels L_{max} for audiometry testing for hearing thresholds to 5 dB, with 5 dB uncertainty over the range 500 to 8000 Hz, using typical supra-aural earphones (adapted from the Standard [ISO 8523]).

Octave band center frequency Hz	L_{max} (typical) (re 20 μPa) dB
125	34
250	24
500	21
1000	20
2000	19
4000	15
8000	22

2.6.2 Reverberation time

According to Tate [Tate, 1994, p. 122], a room for audiometry must have a low reverberation time and for audiometric purposes a reverberation time between 0.2 and 0.25 seconds is ideal.

3 Methodology

3.1 Measurement of the acoustic properties of the Soundroom

The measured acoustic properties of the Soundroom were the ambient noise, the reverberation time and the attenuation between the Control Room and the Soundroom.

3.1.1 Ambient noise

The ambient noise was measured in three different random positions and then space and time averaged. The results were then compared to the requirements according to the Standard [ISO 8523].

3.1.2 Reverberation time

The reverberation time measurement was performed using a sine sweep as the excitation. The sine sweep had a length of 10 s and a full frequency range from 1 Hz to Nyquist frequency (24000 Hz). The impulse response was obtained, and the decay curve was calculated from the IRs for each octave band. From the data, it was determined if it was suitable to estimate the T_{20} or the T_{15}.

The equipment setup used can be seen in figure 3.1.

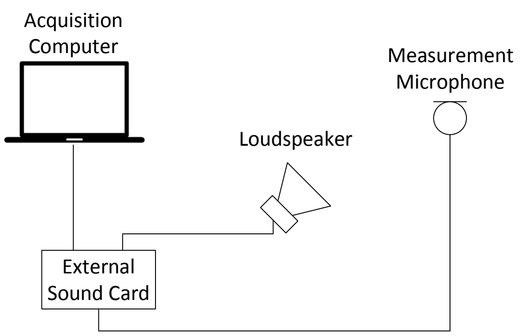


Figure 3.1: Equipment settings for the RT measurements.

Measurement positions

According to the standard ISO 3382-2, the sound source should be omnidirectional, but it allows some deviations, as listed on table 3.1 [ISO 3382-2]. The source needs to produce enough sound pressure level so the decay curves obtained have the dynamic range needed (over the background noise levels). The microphone should be omnidirectional as well.

Table 3.1: Maximum deviation allowed in the directivity of the source.

Frequency [Hz]	125	250	500	1000	2000	4000
Maximum deviation [dB]	±1	±1	±1	±3	±5	±6

The standard recommends that the source should be placed in a corner of the room. Two source positions are recommended.

The minimum number of microphone positions per source position is two, but in rooms with more complicated geometry, more microphone positions are needed. The positions should be at least half a wavelength apart. The recommendation says 2 m for the usual frequency range, but since the range of interest starts at a much higher frequency, the distance used was 1 m.

The distance from the microphone positions to reflecting surfaces must be at least a quarter of a wavelength, but it was also considered to be 1 m.

The minimum distance between the source and the microphone positions should be calculated as in equation (3.1). Where V is the volume of the room, c is the speed of sound and T is the estimate of the expected reverberation time. In this case, the calculated minimum distance was 2.7 m.

$$d_{min} = 2\sqrt{\frac{V}{cT}} \tag{3.1}$$

The chosen measurement positions are shown in figure 3.2. It basically is a grid, with almost all of the points at 1m distance from each other and from the walls, and more than 1 m from the floor and the ceiling. The heights were chosen randomly.

The source was placed in two different positions, $x4$ and $x15$ (two corners of the room), each of them with 11 different microphone positions, following the recommendations of the Standard ISO 3382-2 mentioned before.

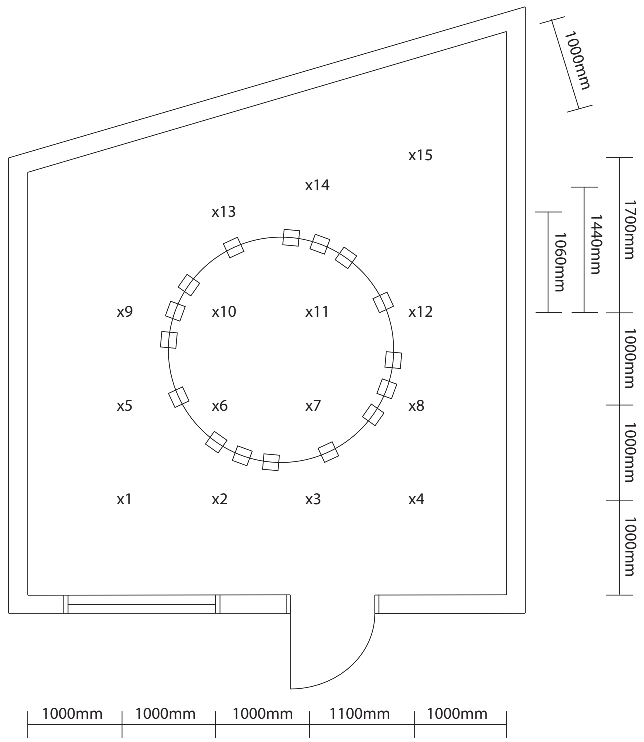


Figure 3.2: Measurement positions for the RT measurements.

3.1.3 Low frequency response

According to measurements performed by an acoustic consultant in 2007, the reverberation time of the room is 0.1 s [Akustikon, 2007 (2007060E)], and so the Schroeder frequency of the room is close to 79 Hz. For the application of this lab, the lower limit of the frequency range of interest is 250 Hz, which is located in the high modal density region, therefore the modal response of the room is not expected to cause coloration at this range.

In spite of this, the low frequency response of the room was assessed as well. Figure 3.3 shows the microphone positions chosen for the three experiments, first for the x-direction, then for the y-direction and finally for a diagonal direction. In the figure one can see that the loudspeaker was placed on one corner of the room.

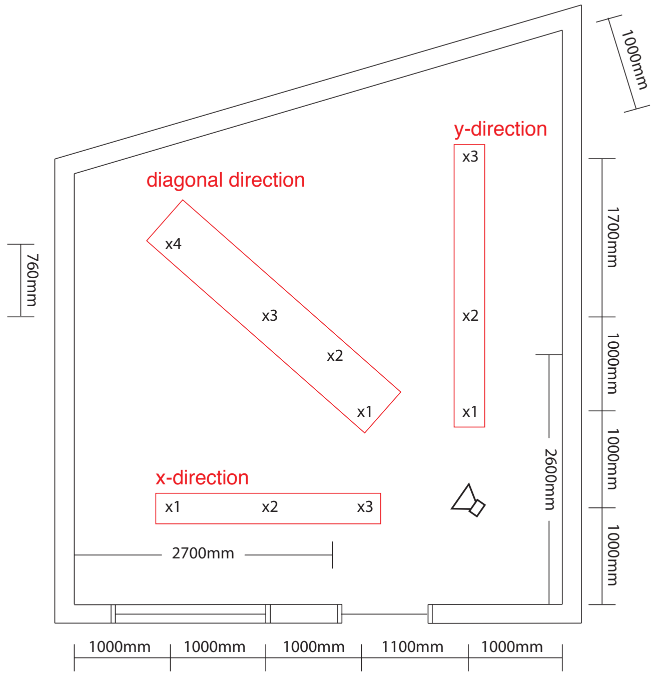


Figure 3.3: Microphone positions for the low frequency response analysis.

3.1.4 Attenuation of the separating walls

To calculate the attenuation produced by the separating walls, the airborne sound insulation was measured.

In the past, an acoustic consultant measured the reduction in dB from all the walls surrounding the Soundroom. Even the impact sound insulation from the room on top was measured [Akustikon, 2007 (2007060D)].

The walls or ceiling have not been changed since then. The only difference is that by that time, there were missing wall panels on the door and it was not completely sealed. Therefore the wall of interest for the present work is the one dividing the Soundroom and the Control Room.

In this case, the Control Room was chosen to be the sending room and the Soundroom the receiving one. The equipment setting used for this measurement is shown in figure 3.4.

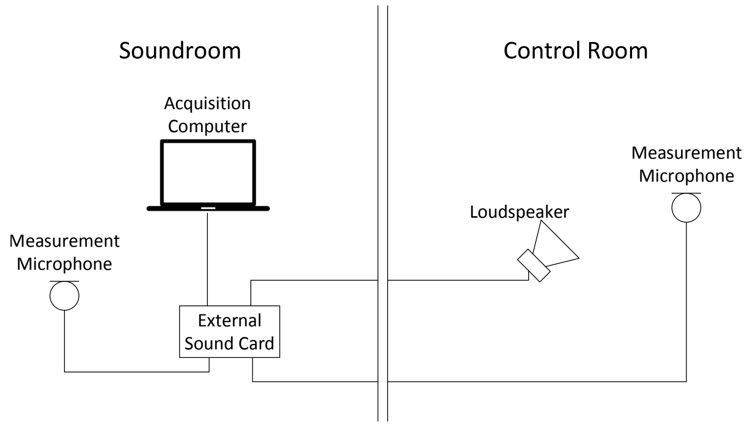


Figure 3.4: Equipment settings for the attenuation measurements.

Measurement positions

To carry out these measurements, the standard ISO 140-4 was followed.

It states that the sound source should be preferably white noise with a sufficiently high sound pressure level so that in the receiving room the measured sound pressure level is at least 10 dB higher than the background noise in any frequency band.

If the difference between the measured data and the background noise is less than 10 dB but higher than 6 dB, the correction shown in equation (3.2) needs to be applied [ISO 140-4, p. 7].

$$L = 10log\left(10^{L_{sb}/10} - 10^{L_b/10}\right)\ \text{dB} \tag{3.2}$$

Where L is the corrected signal level, L_{sb} is the measured data (that includes the background noise), and L_b is the background noise.

When the difference in levels is less than 6 dB, 1.3 dB should be subtracted but the results should indicate that the reported values are the limit of measurement.

The sound source should be in at least two positions and they should be located in a way that the produced sound is as diffuse as possible and at a distance from the flanking elements so the direct radiation upon these elements is not so strong.

The Control Room has an unconventional shape, mostly because of the Soundbooth. It produces a hallway-like room, and there are many close reflective surfaces and so following this last recommendation was not easy and so some deviations from reality could be expected from the results.

Related to the microphone positions, the standard suggests that the distance between microphone positions should be at least of 0.7 m, and that microphone positions and walls, floor and ceiling should be at least 0.5 m apart. The sound source and the microphone should never be closer than 1 m.

A minimum of five microphone positions is recommended for each loudspeaker position, and each measurement needs to be at least 6 s long.

The chosen measurement positions are shown in figure 3.5. All of the points are located at 1m distance or more from each other and from the walls, and more than 1 m from the floor and the ceiling. The heights were chosen randomly.

The source was placed in two different positions, $x1$ and $x6$ (Control Room), each of them with 5 different microphone positions, following the recommendations of the Standard ISO 140-4 mentioned before.

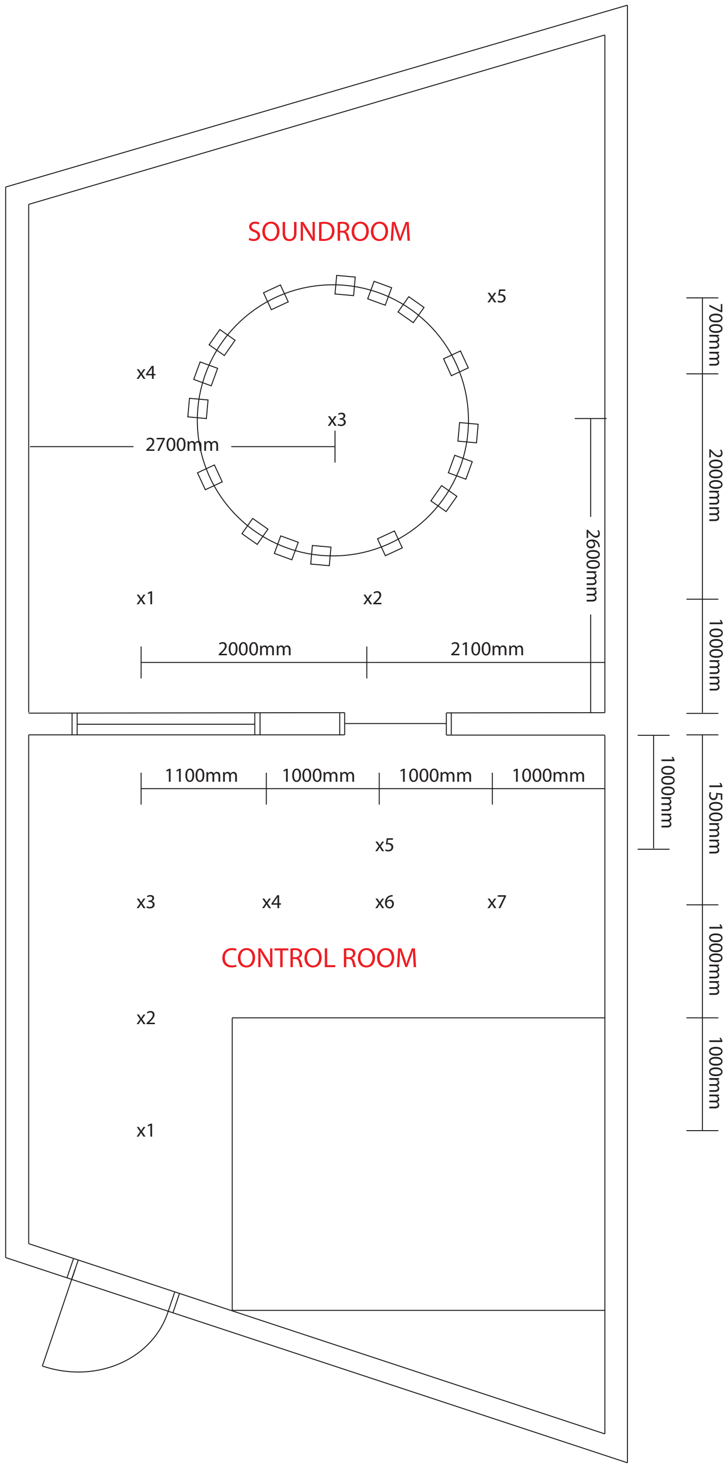


Figure 3.5: Measurement positions for the attenuation measurements.

3.2 Measurement of the combined frequency responses of the room and the loudspeakers

The IRs of the combination of the room and the loudspeakers were measured, using a sine sweep, in order to obtain the FRFs. The equipment setup used can be seen in figure 3.6.

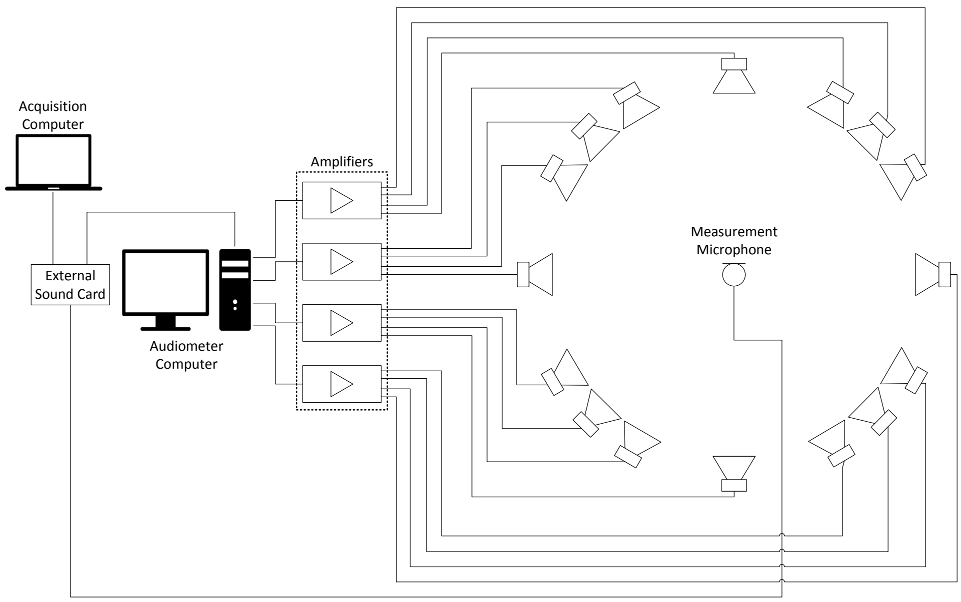


Figure 3.6: Equipment settings for the of the combined frequency responses of the room and the loudspeakers.

In the figure 3.6, one can notice that the excitation signal was delivered to the loudspeakers via the audiometry system. The signal coming from the acquisition computer was sent through the amplifiers. This amplifiers have embedded equalizers, but the settings were bypassed, so the signal would not be modified.

3.2.1 Assessment of the reflections in the responses of the loudspeakers

This was done by reviewing the IRs and relating the reflections with the distances in the room, from the floor, the ceiling, the walls, and the other loudspeakers.

3.3 Investigating improvements of the setup to achieve a better frequency response from all speakers

Several different tests were carried out. Glass wool was used and also pieces of foam found in the office building.

The items were placed according to the assessment of reflections and the IRs and FRFs were compared, to confirm that the potentially problematic reflections were attenuated.

Other tests performed involving the use of the parametric equalizer, which comes incorporated into the amplifiers software and also a minimum-phase inverse filter was implemented.

3.4 Equipment list

The main pieces of equipment used for the measurements are listed in table 3.2.

Table 3.2: Equipment list

Name	**Model**	**Brand**	**Serial Number**
USB 2.0 Audio/MIDI Interface	E-MU 0404	Creative	MAEM8761750R00151E
Pro Low-Noise MIC Preamp	MPA-102	Img Stage Line	-
Electret Microphone	-	-	-
Sound Level Calibrator	Type 4231	Brüel & Kjær	2136623
DIY Laser Range Finder	PLR 30	Bosch	786785504
CTs Multi-channel Power Amplifier	CTs 4200	Crown	8001262630 - 8001291521 - 8001291527 - 8001291511
Bi-Amplified Monitoring System	8030 A	Genelec	8030APM6030460
Power Amplifier	iP 2100	Lab.gruppen	00040664
Full-Range Mini Reference Monitors	MixCubes	Avantone Pro	-
Hexahedron speaker	-	-	-
Active Direct Inject Box	AR-133	BSS Audio	073924
Laptop with FuzzMeasure Pro, Audacity® and Matlab®			
Computer with HiQnet System Architect™ 3.4 and BSS HiQnet London Architect™			
Cables, power strips and microphone and loudspeaker stands			
Pieces of glass wool, pieces of foam			

4 Results and Analysis

In this chapter, all the results obtained are shown and analyzed, following the same order as for the *Methodology* (Chapter 3).

4.1 Measurement of the acoustic properties of the Soundroom

4.1.1 Ambient noise

The measured ambient noise is shown in table 4.1, along with the recommended values according to the Standard [ISO 8523].

Table 4.1: Measured and recommended ambient noise.

Octave band center frequency (Hz)	Measured (re 20 μPa) (dB)	Recommended (re 20 μPa) (dB)
125	30	34
250	28	24
500	25	21
1000	24	20
2000	22	19
4000	22	15
8000	22	22

One can notice how the ambient noise in the Soundroom is 4dB to 7dB higher than the recommended values for almost all the frequency bands. This is important to consider reducing, maybe except for the lower band, since it is out of the range of interest (250 Hz - 7000 Hz).

The ambient noise limits in the Sound Room could be exceeded due to some water pipes running above both the Soundroom and the Control Room. Another source of problems is the hole to run the cables through the wall between the Soundroom and the Control Room. To stop the noise from coming in, some foam has being used but there is a need for a sound isolating cable grommet.

4.1.2 Reverberation time

The reverberation time is shown in table 4.2. The two-point average reverberation time was calculated (for 500 Hz and 1000 Hz) as 0.1 seconds, which is lower than the recommended 0.2 to 0.25 seconds according to Tate [Tate, 1994, p. 122]. In general all the bands comply with this recommendation except the band with center frequency 125 Hz. This could be improved by adding low frequency absorption, but since this band is out of the frequency range of interest this does not require any further actions.

Table 4.2: Reverberation time of the Soundroom.

Octave band center frequency (Hz)	Reverberation Time (s)
125	0.39
250	0.22
500	0.12
1000	0.08
2000	0.06
4000	0.07
8000	0.08

With the reverberation time and the volume of the Soundroom the Schroeder frequency obtained was 79 Hz, which is quite low and therefore there is not a clear area of low modal density in the range of interest as expected.

4.1.3 Low frequency response

The low frequency analysis was done by measuring in the positions shown in figure 3.3 (previous chapter). The results at two of the positions were plotted up to 150 Hz to show the behavior of this positions at a lower frequency range. These results can be observed in figure 4.1.

As mentioned in section 3.1.3, the room is not rectangular and so the positions in the x-direction could be affected by modes in the y-direction and vice versa.

In the figure, the changes for the two positions are displayed. One can see that, the dip and resonances below 70 Hz are due to a modal behavior, which could be improved to have a flat frequency response using bass traps or resonators, but since there is only one point of interest in the scope of this work (listening position) and the frequency region is higher than the modal region, this will not be considered any further.

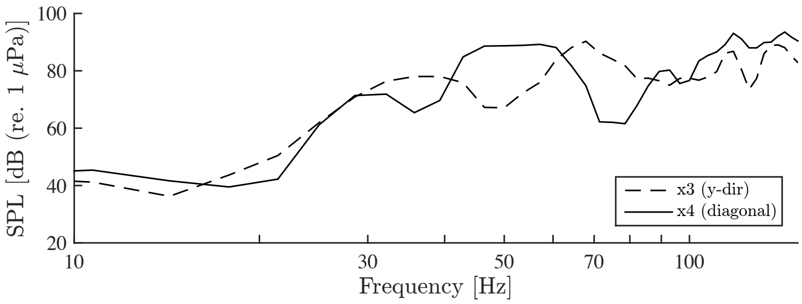


Figure 4.1: Variations on the low frequencies.

4.1.4 Attenuation of the separating walls

As mentioned in section 3.1.4, an acoustic consultant carried out the measurement from the walls surrounding the Soundroom, including the one which was found most important for this thesis. In table 4.3, the results obtained by the consultant are shown.

Table 4.3: Attenuation from the walls surrounding the Soundroom (measured by the consultant) [Akustikon, 2007 (2007060D)]

Octave band center frequency (Hz)	Room 284 (Control Room) (dB)	Room 260 (dB)	Room 280 (dB)	Room 285 (dB)
125	29.6	30.8	38.0	29.4
250	45.3	54.0	59.0	47.8
500	50.1	65.8	72.2	64.0
1000	56.5	65.2	68.1	65.2
2000	56.3	61.6	65.9	62.6
4000	57.1	56.8	61.0	58.0
8000	55.5	52.7	56.6	52.8

In table 4.4, the results shown were calculated for the purpose of this work (partition between the Soundroom and the Control Room).

Table 4.4: Attenuation from the walls surrounding the Soundroom (measured for this work).

Octave band center frequency (Hz)	Control Room (Room 284) (dB)
125	30.3
250	46.5
500	52.6
1000	58.2
2000	58.7
4000	61.5
8000	58.1

From the measured data in table 4.4, the apparent sound reduction index R' was calculated using the equation (2.6) in section 2.5, also considering the data obtained from the reverberation time (two-point average at 500 Hz and 1000 Hz), the volume of the Soundroom and the area of the partition.

The index R', the reference curve and its weighted value R'_W are shown in figure 4.2. There it can be seen that $R'_W = 45$ dB. These values were calculated as explained in section 2.5.1.

The R'_W was also calculated for the data obtained by Akustikon, using the same value of reverberation time found in this present work and the value obtained is 44 dB, therefore the changes done afterwards (door seal) were good for the insulation but it is not a significative improvement.

Some of the data for these calculations needed a correction for the background noise as stated in section 3.1.4, and so the reported values are the limit of measurement.

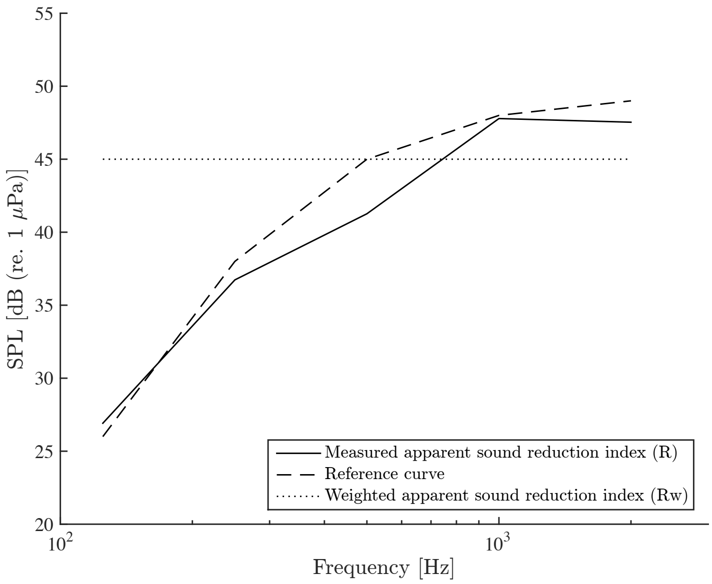


Figure 4.2: Apparent sound reduction index.

4.2 Measurement of the combined frequency responses of the room and the loudspeakers

The figure 4.3 shows the distribution of the loudspeakers in the array.

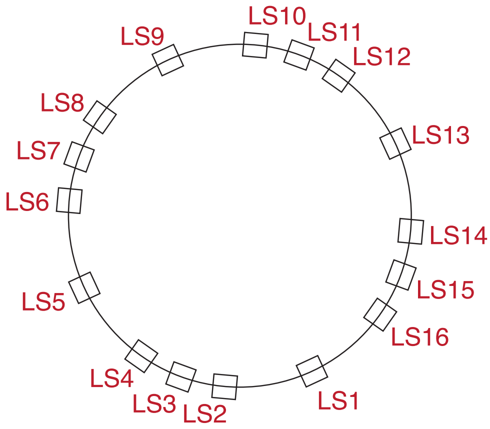


Figure 4.3: Loudspeaker array.

The frequency response of the 16 loudspeakers at their position in the room can be seen in figure 4.4. Those were obtained as explained in section 3.2, measuring the impulse responses first.

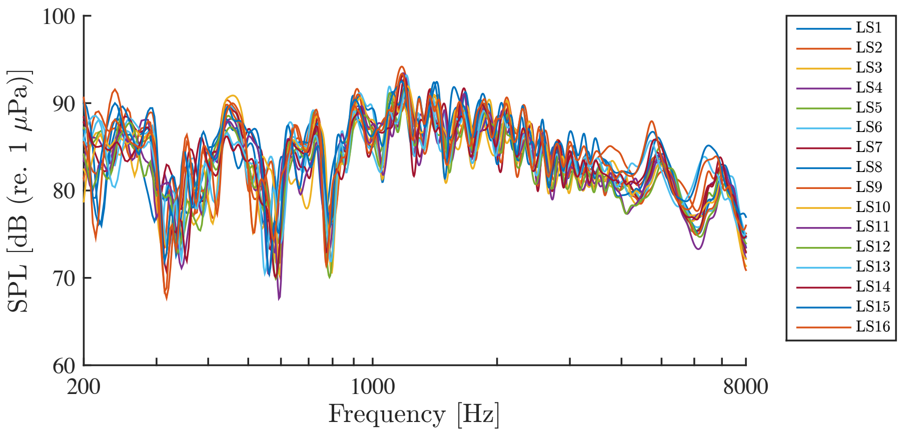
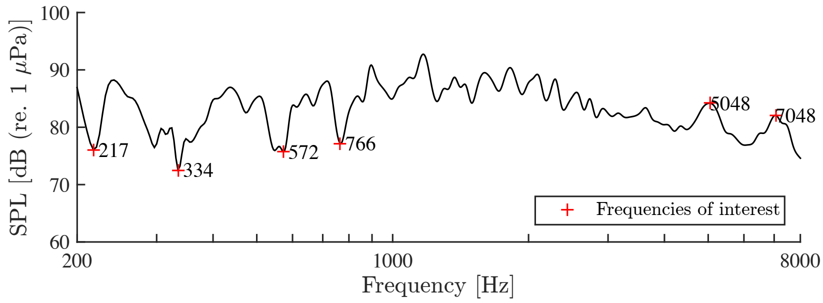


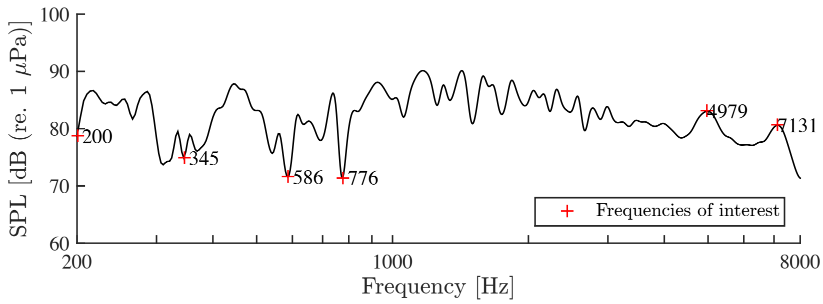
Figure 4.4: Frequency response in narrowband (1/6 octave smoothing).

One can notice that the behavior of the responses are quite alike, there are three predominant anti-resonances or dips at frequencies lower than 1000 Hz and also there is a strange behavior at frequencies higher than 4000 Hz (this last occurs too high in frequency to be a strong reflection or a modal pattern, so it it likely to be due the combined loudspeaker-amplifier response).

Looking at the detail of the responses of some of the loudspeakers (as an example loudspeakers 1 and 3) shown in figure 4.5, it is more clear to see that the dips at lower frequencies occur at very similar frequencies. The resonances at the higher frequencies are also occurring at almost the same frequencies for both loudspeakers. This indicates that the effect is present for all the loudspeakers, and so it must be due to something that is equal for all loudspeaker-microphone paths, like the floor, the ceiling and the amplifiers.



(a) LS1



(b) LS3

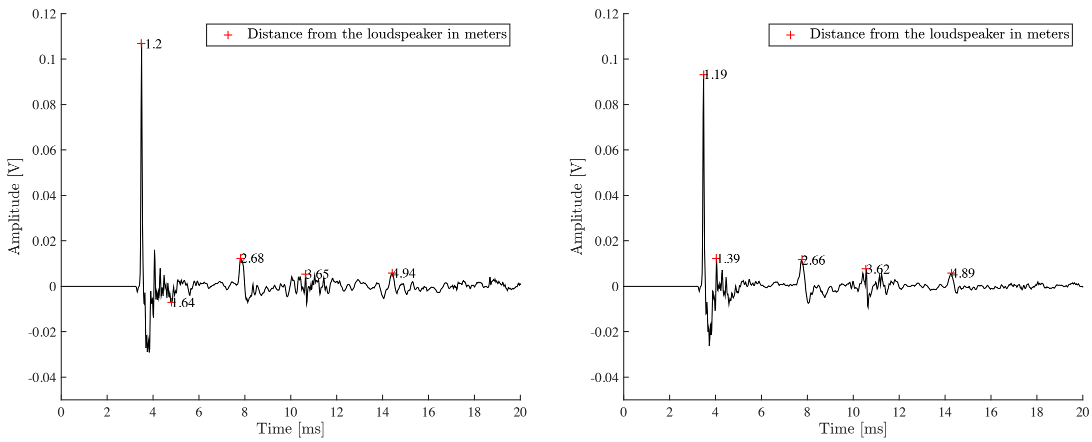
Figure 4.5: Frequency response of LS1 and LS3 (1/6 octave smoothing).

The first dips seen in this frequency range for LS1 (figure 4.5(a)), are spaced from each other 117 Hz, 238 Hz and 194 Hz respectively. The spacing of this dips can be translated into spacing in the time domain, therefore the dips are ”delayed” from the direct sound. The direct sound arrives at around 3.5 ms, therefore the anti-resonances could correspond to strong reflections at 12 ms, 7.7 ms and 8.7 ms. For LS3 (figure 4.5(b)) the values are very similar, 10.4 ms, 7.7 ms and 8.8 ms. This will be confirmed next.

Another thing of interest from these figures is the effect produced by strong reflections, explained in section 2.3.3. Since this effect happens because of having strong reflections, it is more likely for the effect to be produced by a first order reflection on the floor or a second order reflection on the floor and the ceiling. This will also be confirmed in the next section.

4.2.1 Assessment of the reflections in the responses of the loudspeakers

The main reflections of the impulse responses for the loudspeakers 1 and 3 were analyzed and are shown in figure 4.6. The distances from the loudspeaker to the microphone are displayed in the figures. The direct sound arrives first and corresponds to a distance of around 1.2 m for both cases.



(a) LS1

(b) LS3

Figure 4.6: Impulse responses of LS1 and LS3 (first 20 ms).

The second value marked on the figures corresponds to the reflection on the neighboring loudspeakers, this is the first possible reflection due to the geometry of the room and the settings of the loudspeaker array. Data between the direct sound and the first reflection could be produced by the loudspeaker response itself.

As seen in figure 4.3, LS1 has LS2 and LS16 next to it, at a distance of around 0.45 m (0.45 m + 1.2 m = 1.65m). For LS3, the neighboring loudspeaker are LS2 and LS4 which are at a distance of around 0.16 m (0.16 m + 1.2 m = 1.36 m). The distances may vary since the microphone could've been positioned at a distance a little bit more or less than 1.2 m from all loudspeakers.

The third mark corresponds to the first order reflection from the floor. The microphone and the loudspeakers are located at 1.3 m from the floor, therefore this peak occur at almost the same time for all the loudspeakers, changing only because of the deviations in the placement of the microphone.

The fourth reflection also occurs at similar times for all the loudspeakers and this one is due to the loudspeaker located in front of it, for LS1 it is LS9 and for LS3 it is LS11. Finally the last important reflection is also similar for all of them, this could be related to a second order reflection from the floor-ceiling combination.

As mentioned before, the first order reflection from the floor and second order reflection from the floor-ceiling are more likely to be the ones producing the *comb filter effect*.

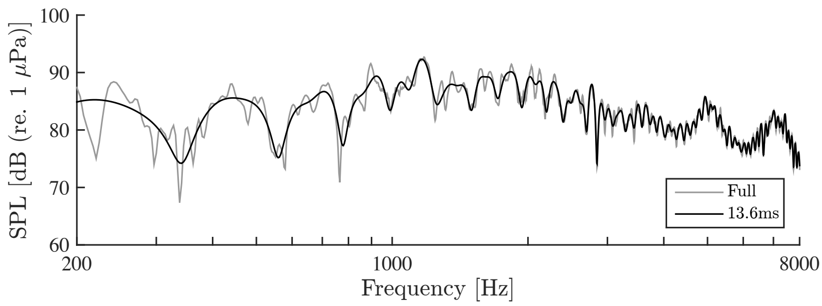
Next, the impulse response of LS1 was modified to justify some of the behavior of the frequency response. This was done by eliminating certain reflections by trimming and zero-padding the rest of the impulse response. The FRF with no smoothing was used to get a better understanding. Figure 4.7 show the FRF for four different "cuts", 4.7(a) when eliminating the second order floor-ceiling reflection, 4.7(b) when eliminating the opposite loudspeaker reflection, 4.7(c) when eliminating the first order floor reflection and finally 4.7(d) when eliminating the first possible reflection.

4.7(a) show some significant improvement at frequencies below 1000 Hz, and so it can be assumed that the *comb filter effect* in this frequency range is caused by this reflection. In the previous section it was assumed that the anti-resonance occurring at 217 Hz was due to a reflection at 12 ms, but instead it was produced by the second order floor-ceiling reflection a little later in time. The improvement at the higher frequency range is not significative.

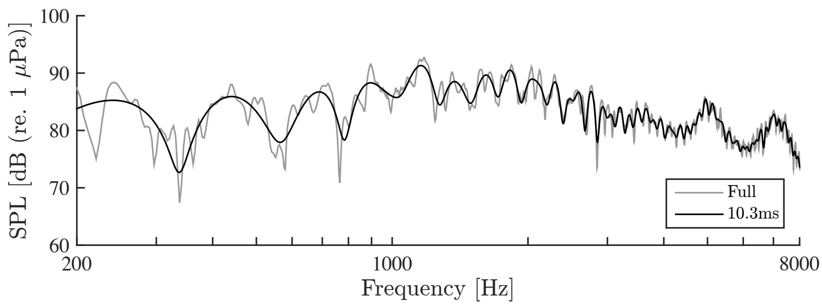
4.7(b) show the same response at low frequencies than before, but some improvement in the amplitude of the *comb filter effect* at the frequencies above 4000 Hz.

4.7(c) show that the *comb filter effect* is mostly gone, therefore removing these reflections (mainly on the floor) is extremely important in order to improve the sound field of the room.

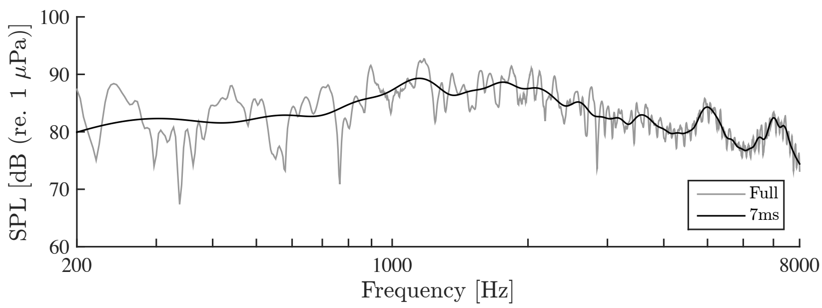
The importance of 4.7(d) is that it shows that the strange resonances at high frequencies are still present when considering only the direct sound. And so this effect could be produced by the loudspeaker itself and by the amplifiers. This could be considered for the improvement of the sound field.



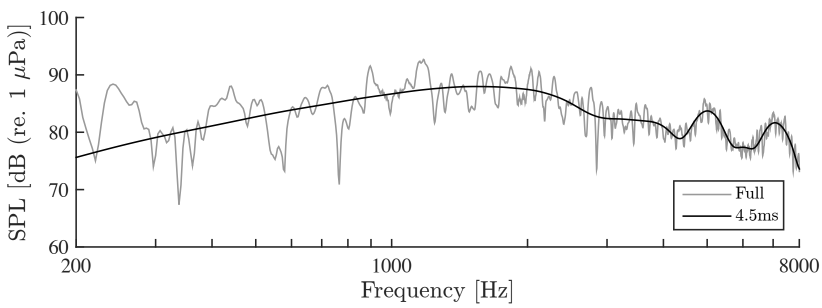
(a) Eliminating second order floor-ceiling reflection.



(b) Eliminating opposite loudspeaker reflection.



(c) Eliminating first order floor reflection.



(d) Eliminating first reflection.

Figure 4.7: Non-smoothed FRF of LS1 with different impulse response lengths (no smoothing).

In section 2.3.3, the *comb filter effect* was explained and in figures 2.7(c) and 2.8(c), the gain factors of the reflection in comparison with the direct sound which are in the limit of giving coloration to white noise can be seen.

In this case the strongest reflection producing the effect is the floor one. Figure 4.8 shows the values in dB for the normalized impulse response of LS1. One can see that the direct sound and the reflection of interest are marked. Here the reflection has a delay from the direct sound of 4.3 ms and a level of -18.7 dB relative to the direct sound.

From figure 2.7(c) one can see that for the delay mentioned, a level of around -17 dB relative to the direct sound gives just perceivable coloration. A level of -18.7 dB is not so close from -17 dB and so there is a risk of having an audible coloration.

There are more strong reflections after the first one thus they could mask it. This could decrease the risk of having coloration, but since the reflection is near the limit of being perceived, it is important to damp it.

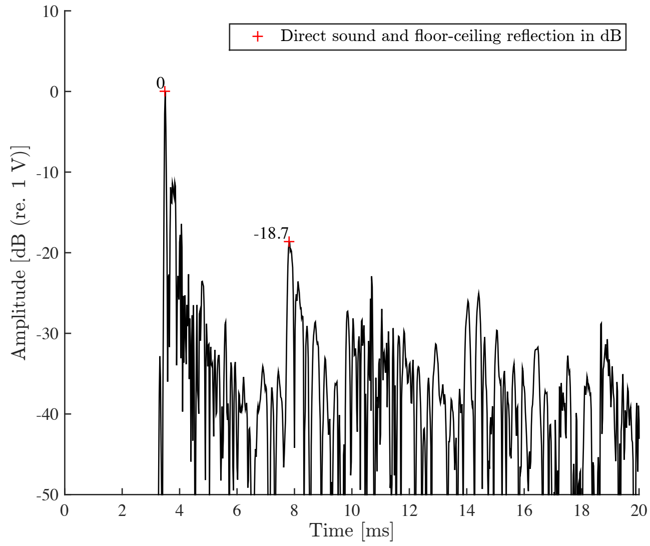


Figure 4.8: Normalized impulse response of LS1 in dB.

4.3 Investigating improvements of the setup to achieve a better frequency response from all speakers

By now, it is known that the main issue that could affect the sound field of the Soundroom is the floor and the ceiling. Some tests were carried out, some of them using absorbers (acoustical means) and others with signal filtering and equalizing (electronic means). The damping of the floor reflection and other reflections was attempted.

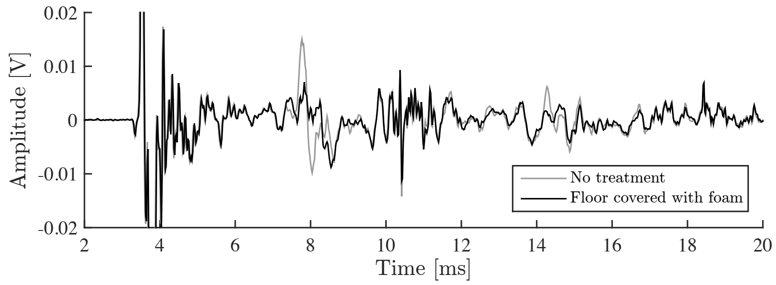
4.3.1 Acoustical improvements

A way to improve a room response acoustically is to put absorption on the floor between the loudspeaker and the listener.

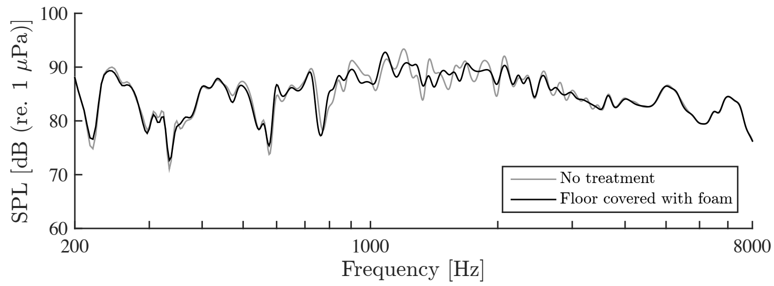
Figure 4.9: Polyurethane foam piece.

First, polyurethane foam pieces (figure 4.9) were placed in layers (6 cm thick total) covering the area on the floor where the reflection could be produced (in the middle between the microphone and the loudspeaker). Next, glass wool (25 cm thick) was used in the same position. Figure 4.10 show the IRs and the FRFs obtained from those tests for the loudspeaker LS1.

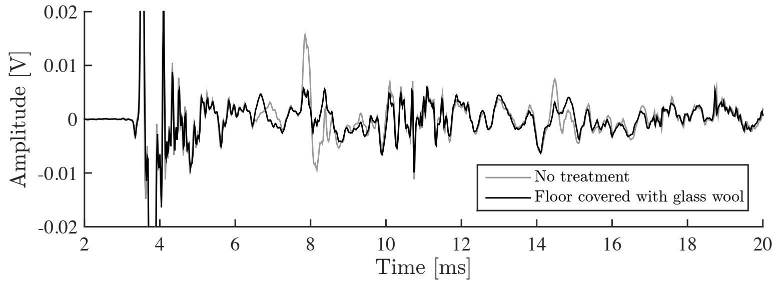
In 4.10(a) and 4.10(c), one can observe how the reflections due to the floor are a bit reduced but they are not completely gone (around 8 ms and 14.5 ms). The FRFs show the impact in the frequency domain. In 4.10(b), the main three dips discussed in previous sections are reduced close to 2 dB which is not substantial, and the response between 900 Hz and 2000 Hz seem to be a little improved. In 4.10(d), the dips are mainly shifted, but there is an improvement at the dip corresponding to 330 Hz. The rest of the response show a little improvement (in the mid-frequencies).



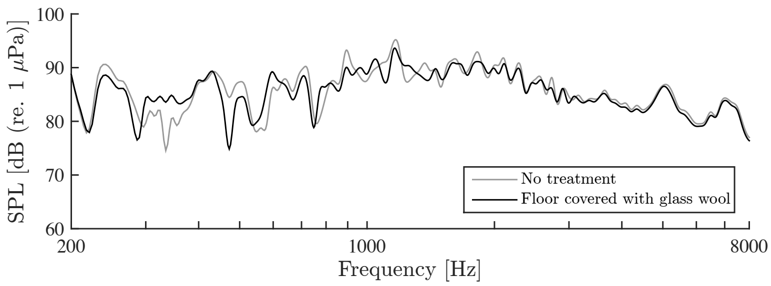
(a) Impulse response (foam) (zoomed in for detail)



(b) Frequency Response (foam) (1/6 octave smoothing)



(c) Impulse response (glass wool) (zoomed in for detail)

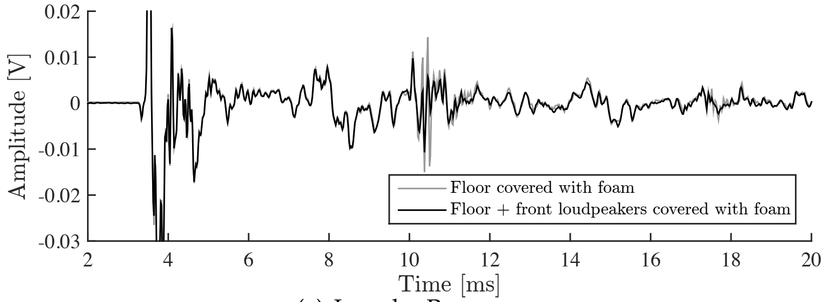


(d) Frequency Response (glass wool) (1/6 octave smoothing)

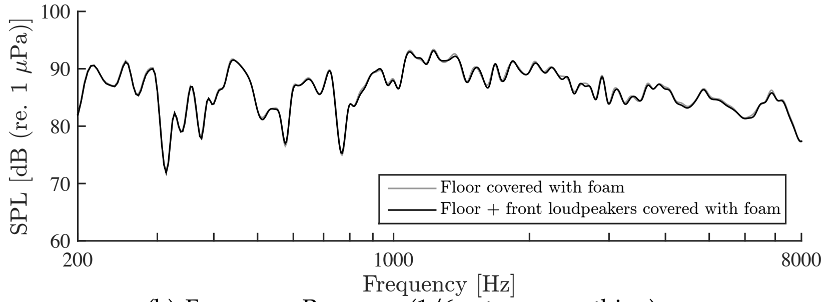
Figure 4.10: Responses for the modifications on the floor for LS1.

Further tests were carried out with LS1, for example placing absorption on the walls and covering the desk but the results did not give much improvement, only reducing up to 2 dB at the first dip.

Another test was performed, while keeping the floor covered with foam as explained, the loudspeakers at the opposite side from LS2 (LS10, LS11, and LS12), as seen in figure 4.3, were also covered with the foam and the results are shown in figure 4.11. One can notice that in the impulse response (4.11(a)), the reflections around 10 ms are reduced, but in frequency (4.11(a)) there is no clear improvement from what the foam on the floor already improved.



(a) Impulse Response



(b) Frequency Response (1/6 octave smoothing)

Figure 4.11: Responses for the modifications on the front loudspeakers for LS2.

After these tests it can be assumed that the main needed improvement is placing absorption of the floor (inside the array) and/or changing the ceiling angle, in order to get a better frequency response.

4.3.2 Electronic improvements

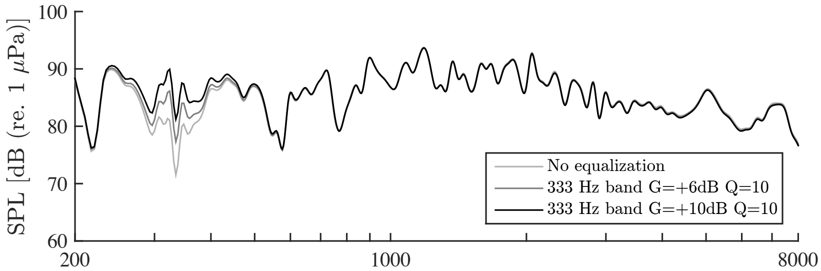
The main objective here is to equalize the sound output since it's not optimal as found in figure 4.7(d). Two methods were used, the first one is the equalization by using a parametric equalizer and the second one is the implementation of an inverse filter.

Parametric equalizer

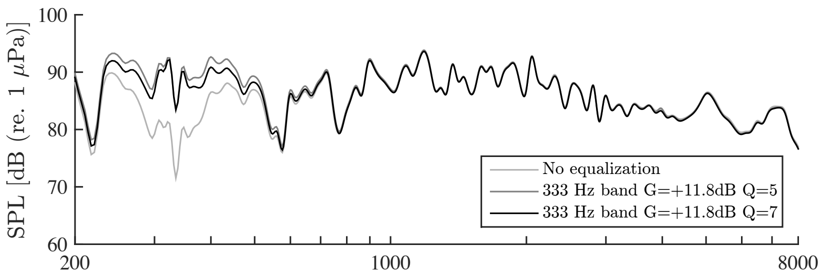
The parametric equalizer embedded in the amplifiers was used to test LS1. This mainly to understand the behavior of the FRF when changing the parameters of the equalizer: Q (or the bandwidth), the center frequency f_c and the gain G, figure 4.12 show the results.

In figure 4.12(a), the gain was changed from +6 dB to +10 dB and in figure 4.12(b), the Q was changed from 5 to 7. In both cases the f_c is 333 Hz.

One can notice that the antiresonance (at 334 Hz) will be higher in dB for both cases, but the effect itself is not removed. The antiresonance is not going to be eliminated with the equalization since it is product of a reflection and trying to equalize it and flatten it completely may produce unwanted changes in the sound.



(a) Change in the gain



(b) Change in the Q

Figure 4.12: One band ($f_c = 333$ Hz), changes in G and Q (LS1) (1/6 octave smoothing).

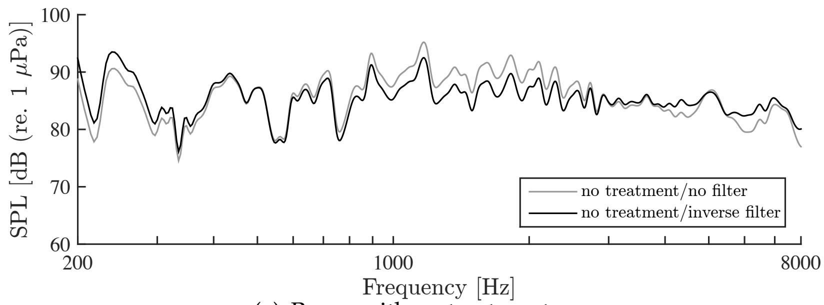
Minimum-phase inverse filtering

An inverse filter is created from a measured IR, such that when the inverse filter is convolved with the system IR, the result is a target response. In this case, the target response corrects the negative effect shown in figure 4.7(d) which could be produced by the loudspeaker and more probably from the amplifiers. Therefore the inverse filter is designed from the direct sound (first 4.5 ms).

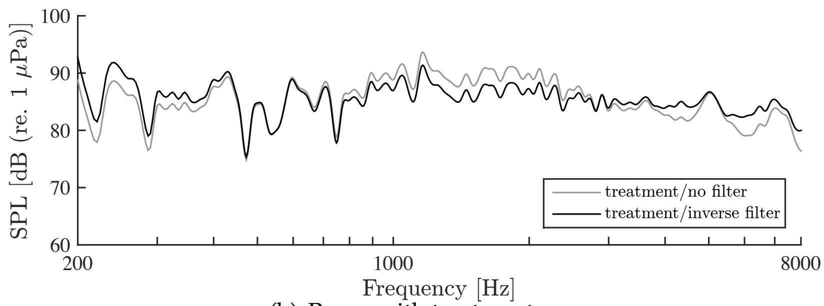
This filter should be embedded in the system (or fed to the amplifiers' equalizers) and it should be adaptive (it should be modified each time the calibration is done). For the case of this thesis, the filter was not implemented due to the lack of flexibility of the system, since there is no constrain in processing capacity, each loudspeaker can have its own inverse filter.

A minimum phase filter was used since it does not attempt to correct the phase and therefore it avoids having abnormalities which can lead to time domain issues.

Figure 4.13 shows the differences between the un-filtered and the filtered signal for LS1 (in the frequency domain). In 4.13(a), the signals were measured without any treatment and in 4.13(b), the signals were measured with glass wool on the floor.



(a) Room with no treatment



(b) Room with treatment

Figure 4.13: Effects of filtering in LS1 (1/6 octave smoothing).

The inverse filter effectively improves the response at high frequencies, and in general flattens the whole response, but there is a boost at the low frequencies. It compensates for the low frequency behavior of the loudspeakers.

The deviations between the maximum/minimum value and the median value of the frequency responses were calculated for filtered and unfiltered signals and are compared in table 4.5. The results show that the deviations for both the room with and without treatment are similar (max 1 dB difference), and that there is around 2 dB of reduction in the highest peak and 1 dB of reduction in the lowest peak (with respect to the median value). This confirms there is a flattening on the frequency response product of the filtering.

Table 4.5: Deviations from the maximum/minimum value and the median value.

	Unfiltered signal (dB)	Filtered signal (dB)
Room with no treatment	+11/-9.5	+8.9/-8.4
Room with treatment	+10/-8.8	+8.1/-9.2

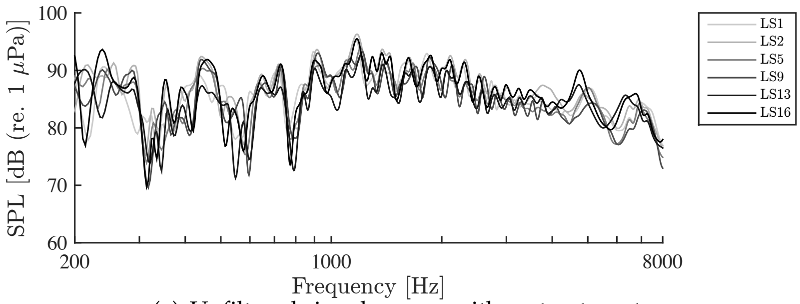
It was chosen not to use the full IR to correct the room effects since the dips are cancellations due to reflections. Fixing the dips implies a boost in the respective frequency and a worsening of the sound field. Even when the FRF looks flat, it does not mean that the sound is optimal.

Figure 4.14 shows as examples the responses from LS1, LS2, LS5, LS9, LS13 and LS16. It can be seen that the filtered signals (4.14(b) and 4.14(d)) in average look flatter than the unfiltered ones (4.14(a) and 4.14(c)). The dips that have been analyzed are reduced, but only a few dB as expected. One can also note a boost at frequencies below 400 Hz, this due to the shape (roll-off al low frequencies observed in the direct sound response (figure 4.7(d)).

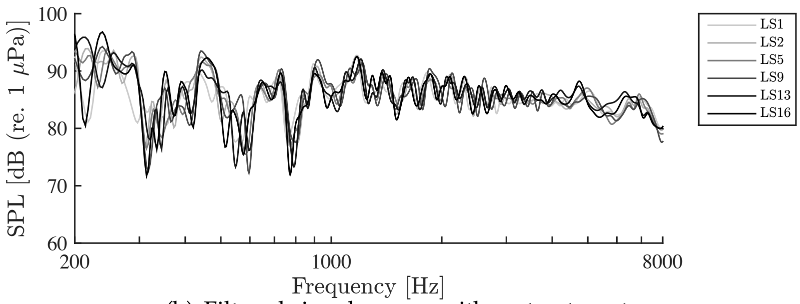
The best responses are definitely the ones shown in 4.14(c). The boosts at low frequencies are not so high and at high frequencies, the responses are much flatter.

To diminish the low frequency boost, a high-pass filter can be used. It it a good idea to remove the boost because the loudspeaker response was shown to be not so good below 500 Hz (figure 4.7(d)).

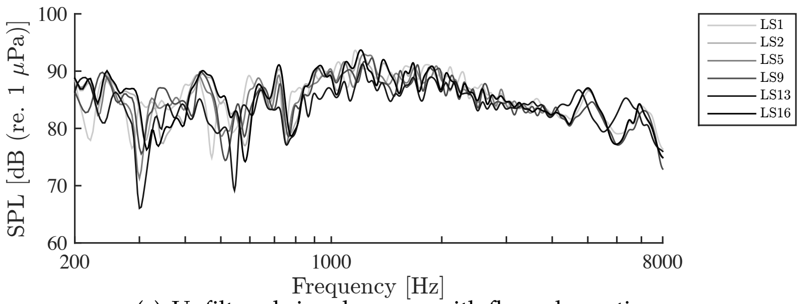
Some sounds were filtered and played in the Soundroom (white noise, pink noise, a song). These sounds do not seem to be affected too much, except at low frequencies where the boost can be perceived. A better assessment should be carried out.



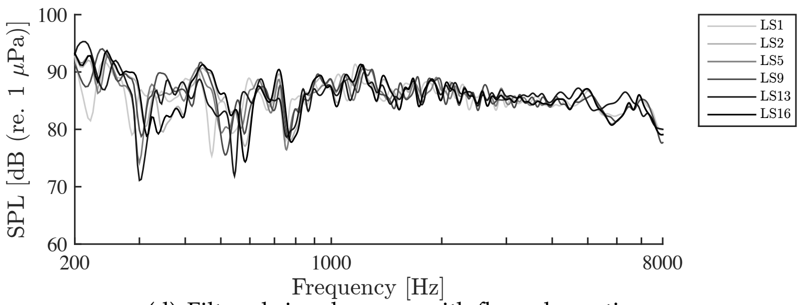
(a) Unfiltered signals, room with no treatment



(b) Filtered signals, room with no treatment



(c) Unfiltered signals, room with floor absorption



(d) Filtered signals, room with floor absorption

Figure 4.14: Effects of filtering (1/6 octave smoothing).

5 Recommendations

Make the room more silent by improving insulation

According to the measurements, background noise was slightly higher than the recommended values. In order to reduce it one can take several measures. The hole used to run the cables in between rooms should be sealed, possibly using a sound isolating cable grommet or similar.

It is important to evaluate the noise coming from the water pipes, which are installed on top of the Soundroom. It would be important to see if there is enough insulation on top of the false ceiling and add more insulation if necessary.

Also more absorption can be added as a thick carpet (or carpet underlay), which would help with other room acoustic issues.

Make the ceiling not parallel to the floor to avoid unwanted reflections

According to Newell [Newell, 2012], "for geometric angling to be acoustically effective, the path length differences for subsequent reflexions must be a significant part of the wavelength". He also says that when having a 9 degree inclination between two surfaces, there is a reduction in modal energy compared to the parallel case but only above 200 Hz (because the modes are being changed from axial to tangential). Below that there is not significant difference and therefore the surfaces are acoustically parallel. Also more absorption should be included.

Figure 5.1 shows a type of ceiling construction. One can see there is also diffusive elements and absorption in the construction. A 9 degree inclination implies the lower part of the ceiling to be 2.2 m high and the highest part 2.4 m. The lower part of the structure should not be located over any loudspeaker, so the sound coming from all of them can have a chance to hit one of the two parts of the ceiling.

A more complicated ceiling structure could be implemented, having for example a shape of a square or triangular pyramid. This structure does not need to cover the entire room since the area of interest it the area inside the loudspeaker array.

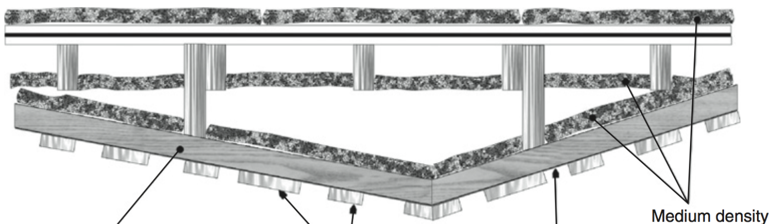


Figure 5.1: Ceiling construction to avoid the *comb filter effect* [Newell, 2012].

Add absorption on the floor

To have a better sound field, the area of the floor inside the loudspeaker array needs to be covered with an absorptive material covered in acoustically transparent fabric. The material can be glass wool or similar. If the ceiling changes are implemented, there is only the need to eliminate the reflections produced from the sound coming directly from the loudspeaker, and so 2.5 cm of mineral wool (density of 24 kg/m^3) would do the job [Aso et al., 1966].

It is important to consider that the reverberation time is shorter than the recommended, and adding absorption might reduce it in some of the frequency bands. An assessment on the effects of a shorter reverberation time should be performed.

Changing the loudspeakers for more directional ones

The directivity of the actual loudspeakers is unknown, but trying to increase the directivity of the loudspeakers would reduce the strength of the reflections on the floor, so it is recommended to purchase better quality monitors with a known high directivity and also with a better response at low frequencies.

Implementing a minimum phase filter

The objective of the filtering is to make the responses more flat but also to try to reduce differences between them. The main reason they are not alike is the room itself. When the room is corrected acoustically and the reflections are removed, a filter could be implemented to try to make the responses more alike between them.

Then the filter can be used to correct the response of the loudspeakers and the amplifiers.

Improve the calibration procedure

The actual calibration procedure includes the use of a software for audio post production. As a recommendation, an acoustical measurement software could be used instead or any other powerful tool.

After correcting the room acoustically, each combined room-loudspeaker FRF could be measured (by measuring the IR with a sine sweep or noise) and corrected with an inverse filter.

If after fixing acoustically the room, the comb filter effect is eliminated and there are not so many anti-resonances, a graphical equalizer could be used. In this case, a hand-held analyzer and sound level meter can be used to measure the responses in third octave bands while playing pink noise on each of the loudspeakers. Then the graphical equalizer can be adjusted to attempt for all the bands to have the same amount of energy. Of course, after equalizing, it is important to assess the final settings with test sounds to make sure that they are not colored.

6 Conclusions

According to the literature and the standards, the room is suitable for audiometric testing; there is only a bit of deviation in recommended ambient noise and the reverberation time is a little low in most frequency bands according to the recommended values, but even if the literature only mentions that the room should be free of echo, there is not much information about the flatness of the frequency response.

In a small room, the modal region extends to a higher frequency than for a large room (the Schroeder frequency is higher), but since the reverberation time also depends on the size this is not necessarily true. For this case, the room is quite damped with a very low reverberation time and therefore the modal region stays at low frequencies. It is important to know that even the modes affect the whole sound field of the room, the frequency range of interest lays at a high modal density region thus correction of modes was not considered.

The room should be as neutral as possible and this is often assessed by a flat frequency response. But even though the frequency response is flat, it does not mean that the sound field sounds neutral. It is important to also assess the time domain response.

To achieve neutrality in a room, the surfaces should not be parallel but what needs to be taken under consideration is what is geometrically non-parallel may be far from acoustically non-parallel at low frequencies.

There is a balance between what can be corrected by electronic means and what cannot. The effect of bad loudspeakers of other problem in the amplifiers can be corrected, but the dips and resonances need to be fixed by acoustic means.

The inverse filter proves to be a good method for correcting minor loudspeaker badness and for flattening the frequency response.

Listening tests should be carried out in the room after applying filtering or equalization, to confirm that the procedure followed improves the sound quality or if it deteriorates it.

References

[Aso et al., 1966] Aso, S. et al. (1966). *Sound Absorption Coefficient of Glass Wool*, (Journal of Textile Engineering 12(3) 101-106). Japan: The Textile Machinery Society of Japan.

[Akustikon, 2007 (2007060D)] Westbrandt, A. (2007). *Solsten Cochlear 1:16. Mät PM luftljudsisolering, stegljudsisolering och efterklangstid*, (Document PM 2007060D). Sweden: Akustikon AB.

[Akustikon, 2007 (2007060E)] Westbrandt, A. (2007). *Solsten Cochlear 1:16. Mät PM efterklangstid och bakgrundsnivå*, (Document PM 2007060E). Sweden: Akustikon AB.

[Baha®, Cochlear] Cochlear. *Baha® bone conduction implants*. http://www.cochlear.com/wps/wcm/connect/intl/home/discover/baha-bone-conduction-implants

[Baha® 5, Cochlear] Cochlear. *Cochlear™ Baha® 5 Sound Processor*. (Datasheet).

[Dietze, 2010] Dietze, B.M. (2010). *Room Response Equalization and Loudspeaker Crossover Networks*, (Diploma Thesis). Austria: Technical University of Graz.

[Gjers, 2014] Gjers, E.L. (2014). *Stage Acoustics in Concert Halls: a study of musicians' acoustical environment*, (Master of Science Thesis). Sweden: Chalmers University of Technology.

[Granbom, 2014] Granbom, V. (2010). *Digital frequency compensation for loudspeakers used in rooms using adaptive equalization*, (Master of Science Thesis). Sweden: Chalmers University of Technology.

[ISO 140-4] International Organization for Standarization (1998). *Acoustics - Measurement of sound insulation in buildings and of building elements - Part 4: Field measurements of airborne sound insulation between rooms*, (Norm No. ISO 140-4). Switzerland: ISO.

[ISO 717-1] International Organization for Standarization (1996). *Acoustics - Rating of sound insulation in buildings and of building elements - Part 1: Airborne sound insulation*, (Norm No. ISO 717-1). Switzerland: ISO.

[ISO 3382-2] International Organization for Standarization (2008). *Acoustics - Measurement of room acoustic parameters - Part 2: Reverberation time in ordinary rooms*, (Norm No. ISO 3382-2). Switzerland: ISO.

[ISO 8523] International Organization for Standarization (1989). *Audiometric test methods - Part 1: Basic pure tone air and bone conduction threshold audiometry*, (Norm No. ISO 8253-1). Switzerland: ISO.

[Kleiner, 2012] Kleiner, M. (2012). *Acoustics and Audio Technology*, (Third Edition). USA: J. Ross Publishing.

[Kleiner, Tichy, 2014] Kleiner, M., Tichy, J. (2014). *Acoustics of Small Rooms*. USA: CRC Press.

[Long, 2006] Long, M. (2006). *Architectural Acoustics*. United Kingdom: Elsevier Academic Press.

[Newell, 2012] Newell, P. (2012). *Recording Studio Design*, (Third Edition). USA: Focal Press.

[Smith, 2010] Smith, J.O. (2010). *Physical Audio Signal Processing for Virtual Musical Instruments and Audio Effects*, (Stanford University). USA: W3K Publishing.

[Tate, 1994] Tate, M. (1994). *Principles of hearing aid audiology*. United Kingdom: Springer-Science+Bussiness Media, B.V.

[Williams, 2009] Williams, W. (2009). *Using ISO 8253 to calculate the maximum permissible background sound pressure levels for audiometric testing*, (Technical Note). Australia: Acoustics Australia Vol. 37 (No. 1-27)

[Wikimedia Commons, Cochlea] *Cochlea*, from Wikimedia Commons, the free media repository, http://commons.wikimedia.org/wiki/File:Cochlea.svg.