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"Comparison of Acoustic Simulation Tools for Shoebox-Shaped Rooms"

TRABAJO FINAL DE GRADO

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Abstract

In this project, the acoustic characteristics of a shoebox-shaped room will be studied. It is focused on the evaluation of the RIRs (Room Impulse Response) and the reverberation time. These parameters are needed in order to improve hearing sensation. The characteristics of the room can be modified, so that different types of room can be simulated. By modifying the reflection surfaces of floor, ceiling and walls the desired parameters can be recalculated. For this purpose, three toolboxes (AKtool, Polarch and MCRoomSim) will be used, making use of MATLAB (Matrix Laboratory) software. For the simulation, sound sources and receivers will be placed inside the room. The audio will be transmitted through the sound sources and the receivers will be used to collect the simulated information through the desired room. In order to analyse the data collected, the results obtained by each of the three toolboxes will be analysed together. The data will then be displayed graphically and analysed at the same time and commented on at the end of the project.

Zusammenfassung

In diesem Projekt werden die akustischen Eigenschaften eines rechteckigen Raumes untersucht. Konkret geht es um die Bewertung von RIR (Room Impulse Response) und Nachhallzeit. Diese Parameter sind notwendig, um das Hörempfinden in einem Raum zu verbessern. Die Eigenschaften des Raumes können modifiziert werden, so dass verschiedene Raumtypen simuliert werden können. Durch Modifikation der reflektierenden Oberflächen von Boden, Decke und Wänden können die gewünschten Parameter neu berechnet werden. Zu diesem Zweck werden drei kostenfreie Tools (AK, Polarch und MCRoomSim) mit der Software MATLAB (Matrix Laboratory) verwendet. Für die Simulation werden die Quellen und Empfänger innerhalb des Raumes platziert. Der Ton wird durch die Tonquellen übertragen, und die Empfänger sind für die Erfassung der Informationen der Simulation des gewünschten Raumes zuständig. Um die gesammelten Daten zu analysieren, werden die Ergebnisse der drei einzelnen Toolboxen miteinander gemeinsam analysiert. Anschließend werden die Daten grafisch dargestellt und zeitgleich analysiert.

Resumen

En este proyecto se estudiarán las características acústicas de una sala rectangular. En concreto, se centra en la evaluación de los RIRs (Room Impulse Response) y el tiempo de reverberación. Estos parámetros son necesarios para mejorar la sensación auditiva de una sala. Las características de la habitación pueden ser modificadas, de manera que se puedan simular diferentes tipos de habitación. Modificando las superficies de reflexión del suelo, techo y paredes se pueden recalcular los parámetros deseados. Para

ello se utilizarán tres herramientas de uso libre (AK, Polarch y MCRoomSim), utilizando el software MATLAB (Matrix Laboratory). Para la simulación, las fuentes y los receptores se situarán en el interior de la sala. El audio se transmitirá a través de las fuentes de sonido, y los receptores serán los encargados de recopilar la información de la simulación de la sala deseada. Con el fin de analizar los datos recopilados, se analizarán conjuntamente los resultados obtenidos por cada una de las tres toolboxes. Posteriormente se visualizarán gráficamente y analizarán los datos a la vez.

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Acronyms

MATLAB Matrix Laboratory

ISO International Organization for Standardization

dB Decibel

t Time

T Reverberation time

EDT Early Decay Time

V Volume in m^3

S Surface Area of the Room in m^2

α Absorption Coefficient

SPL Sound Pressure Level

RIR Room Impulse Response

H Relative Humidity in percent

P Atmospheric pressure in Pa

Fs Sampling rate in Hz

Src Source

Rec Receiver

ISM Image Source Model

HRTF Head Related Transfer Function

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Contents

1 Introduction

The room acoustic simulation has many applications, such as music research, acoustic room conditioning, game audio or virtual acoustic reality. When designing a room whose main function is the interpretation of music, the reproduction of the word (speech room) or the performance of other types of shows, the acoustic of the room is of vital importance. A theatre, conference room or auditorium that does not have the right acoustics will overshadow the performances and will decrease the intelligibility of the word.

To ensure that the acoustics of a room are optimal, the geometry of the room and the materials that make it up must be considered. To assess this, it is convenient to use computer tools that allow to predict the sound field that will exist in the room, and even to hear as if someone was located inside such room. This can be done, prior to its construction or to improve the acoustics of an already built room that does not have good acoustics. By means of measurements made in the room and a subsequent acoustic study, solutions are proposed and implemented that allow for good acoustics. These acoustic studies ensure the best results. Acoustic measurements are made to certify that the acoustic conditions of the room are the most appropriate for it, depending on the purpose of the room to be tested.

During the simulations, reflections will occur in the room, as the walls of the room are finite, therefore the sound will at some point meet a wall, and reflections will occur. Depending on the acoustic function of the room, an increase or decrease of the reverberation time will be needed. In this project, the room acoustics of Figure 1 will be studied. This study will be carried out by different positions of omnidirectional microphones and omnidirectional sound sources.

A comparison of different toolboxes will be carried out, for this purpose simulations will be run, testing the possible parameters. Once the simulations are done, the possible variations obtained between the different toolboxes will be studied. Finally, it is necessary to highlight the relevant elements to be tested, such as some of the room acoustic parameters described in the ISO 3382 standard [1]. This ISO standard stablishes the reference parameters for the objective evaluation of acoustics in rooms from impulse responses. The evaluation of some of the ISO 3382 parameters for performance spaces is an important part of an acoustic report, either for a new or existing room. In this paper, a selection of some important parameters is analysed in terms of measurements and simulations.

Currently there are many Acoustic toolboxes that help processing signals. Several free toolboxes are available, such as AKtool, MCRoomSim and Polarch.

The AKtool

It is a toolbox used for signal processing. It is made up of tools such as SUpDEq (Spatial Upsampling by Directional Equalization), Room Simulation or Spherical harmonics, among many others. It is an easy to download and install toolbox. It contains a README document, which briefly explains how to get started. From Matlab a script can be executed in order to add all needed folders to the end of the Matlab search path. AKtools also contains demo scripts, these scripts are named according to their purpose and include a description of their content at the beginning [2]. In addition, when the tool is installed in Matlab, a folder is also generated, which contains demo data on microphones and sound sources.

MCRoomSim

It is a multichannel shoebox room acoustics simulator that allows the simulation of multichannel microphone arrangements, HRTF's and spherical harmonic expansions in reverberant environments. It works as an extension function of Matlab (mex). This toolbox simulates the geometrical acoustics of a perfect rectangular room volume using the image-source model (ISM), in order to create an IR (Impulse Response) from the omnidirectional source to the omnidirectional receiver. This simulation software package can model specular and diffuse reflections in a shoe-box type environment [3].

In order to successfully simulate the room, the tool contains specific high-level functions and secondary functions to create the simulation of the room it wants to be tested.

Polarch

Polarch toolbox is a fast Matlab shoebox room simulator using the image source model, tuned for spatial sound processing [4]. This toolbox simulates the geometrical acoustics of a rectangular room volume using the image source model.

In order to handle the simulations easier and effectively, it has been created a main script for each of the toolboxes, through which it could be modify each parameter used in the room simulation. The script starts by declaring the dimensions of the room [8 x 6 x 3] (m), followed by the absorption coefficient in octave frequency bands, showed in Table 2, and the position of microphones and sound source. If desired, the direction in which both microphones and sound sources point, in angles, can be modified.

2 Literature Review

2.1 Room Acoustics

The design of the room in which the sound is perceived is of great importance for good acoustics. There are many methods and criteria for the correct design of the room and for its acoustic's evaluation. Acoustic spaces can have a variety of uses (speech, music chamber, opera, concert and more).

Sound Propagation in a room

The propagation of a wave through a room is of great importance for this study. It is necessary to understand how a wave propagates and behave through a room in order to be able to carry out the study. The following are characteristics and properties of room wave propagation that need to be understood [5], [6].

- The sound waves propagate around 345 m/sec, due to its velocity, the sound that a sound source emits in a room will reach the receiver after around 10 to 20 milliseconds.
- For each doubling of propagated distance, direct sound will decrease by 6 dB.
- In a short time after the arrival of the direct sound, several reflections from the reflecting surfaces (walls, ceiling and floor) will reach the listener.
- Sum location describes the phenomenon that two very close sounds (time difference < 1 millisecond) arrive at the receiver from different directions. The direction of hearing event is determined by both the time difference and the level difference. The location of the hearing event can only be moved along a line between the speakers.
- Law of the First Wave front (Predicting Effect): If the transit time is greater than 1 millisecond, then no longer is the sum location, but the direction of the audio event is determined by the first wave to reach the receiver. Reflections and reverberation do not influence directional perception. The predicting effect helps greatly in extracting useful sound from a sound source from the noise.
- Haas Effect: The direct sound determines the direction of the hearing event, although it is approximately 6 dB – 10 dB weaker than the reflected sound.
- Echo: If the time difference between the direct sound and the first relevant reflections exceed a period of about 50 to 80 milliseconds, an echo is perceived.
- The sound wave propagates until it meets a wall and reflections occur. The receiver receives two types of reflections, early and late reflections. Late reflections are usually smaller in amplitude and more spaced out in time than first reflections. These reflections end up forming the reverberant sound.
- Remote hearing: The hearing system is also able to estimate the distance between a sound source and the listener. The mechanism depends on the distance from the sound source:

- Distance < 0.25 m: In the case of hearing system, combinations of the time difference, the level difference and the spectral properties are evaluated by the hearing system. Interaural level differences increase due to the Head-Shielding.
- Distances between 0.25 m -15 m: The signal level decreases as the distance from the sound source increases.
- Distance > 15 m: Due to dissipation there is a frequency dependent attenuation of the sound. The dissipation increases at high frequencies.
- Secondary characteristics: Reflections. The higher the level relationship between the direct sound and the reflections, the smaller the distance to the sound source.

Octave Bands

Octave bands and fractional octave bands are special designed filter based on. Octave Bands are very important and widely used in the engineering field.

An octave band is a frequency band where the highest frequency is twice the lowest frequency. They can be divided into three ranges, referred to as one-third-octave [7].

Octave Bands		
Lower Band Limit (Hz)	Center Frequency (Hz)	Upper Band Limit (Hz)
11	16	22
22	31.5	44
44	63	88
88	125	177
177	250	355
355	500	710
710	1000	1420
1420	2000	2840
2840	4000	5680
5680	8000	11360
11360	16000	22720

Table 1 Octave Band Frequencies

Reflections

The reflections are of great importance in acoustic simulation. The acoustic information of the room will be given by the reflections through the surfaces, until it reaches the receiver. The sound sources emit audio in all directions (omnidirectional, 360º). Therefore the receivers (omnidirectional, 360º) will collect information not only coming directly from the sound source (direct sound), purple ray in Figure 2, but the reflections (early and late reflected) will be produced in the room until they reach the receiver with the information, or until the wave disappears. The wave disappears when it bounces repeatedly around the room. For each reflection on a surface it loses energy until it disappears, Equation (2.1).

$$\alpha = 1 - |r| \tag{2.1}$$

The wave sound is reflected according to the Snell's law (2.2), if it can be assumed that the reflecting surface is flat. This can be seen in Figure 1. Because the speed of sound is considerably greater than the wave propagation speed, sound delays are produced. These sound delays are proportional to the length of the surface. If the surfaces of the room were not flat, but slightly curved, it would happen that the sound is dispersed when it reaches the surface [8].

$$k_1 * \sin\theta_1 = k_2 * \sin\theta_2 \tag{2.2}$$

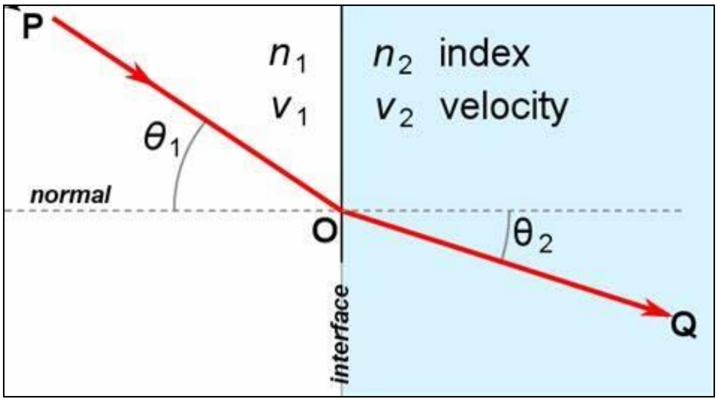

Figure 1 Snell's Law

Since the information collected by the microphones, which are located inside the room, will be the information that need to be collected in order to know how the room behaves acoustically. If few reflections are received, it means that the surfaces of the room have a very high absorption coefficient (short reverberation time), while if many reflections are received it can be bad for the acoustics of the room, creating echoes for example [9].

$$r = \frac{Z_0 - Z_w}{Z_0 + Z_w} \tag{2.3}$$

$$r \leq 1 \tag{2.4}$$

Reflections can be differentiated between two types:

1. Undesirable reflections.
   Sometimes appear reflections, which cause echoes that we do not want to hear in the room. When echoes occur, they can produce the speech less intelligible. Echoes can be controlled by modifying the absorption values of the room surfaces. To prevent the echoes from appearing, we can add more absorbent materials to the room's surfaces.

2. Useful reflections.
It is often discussed that reflections are bad for the acoustics, but sometimes it is better that reflections appear because the reverberation time is very short. If the absorbent materials of the surfaces are very high, it may be that the desired audio does not propagate as much as we wanted. In order to prevent this from happening, less absorbent materials can be used, which will cause the desired reflections to appear.

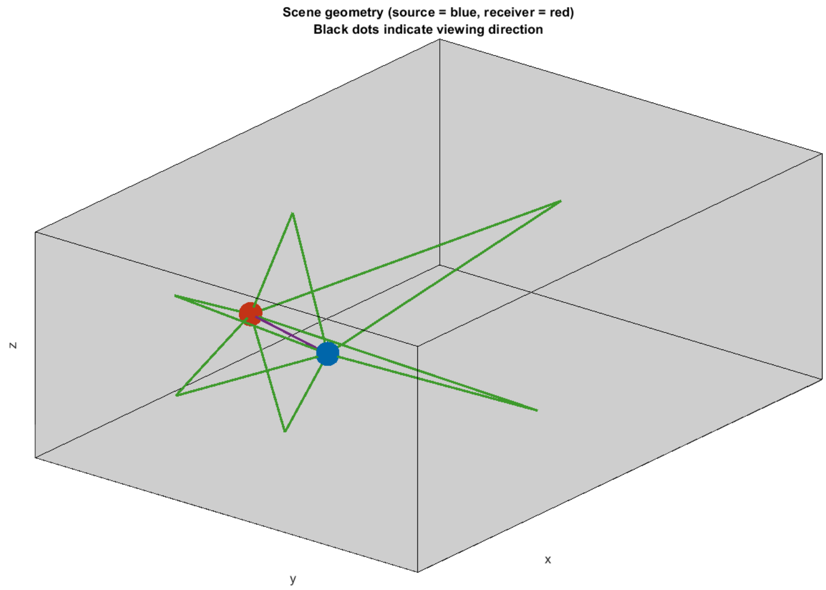

Figure 2 Reflexion Path Example Receiver 1

Reflections can be divided in two groups: early and late reflections.

Early Reflections

Early reflections are a significant part of the RIR. They have an important influence on the subjective impressions of the listener. Early reflections can lead to different kinds of effects on the perception of sound [10].

According to standard definitions, early reflections are the sound the listener receives after the wave has been reflected once or twice across the room, such as the walls, floor and ceiling. They arrive later than direct sound, often in the range of 5 to 100 ms. The early reflections give to the listener the information about the dimensions of the room. They have an important role in determining the general character and sound of the room. The duration of the early reflections determines the impression of the size of the room, this applies for natural hearing [11].

The reflections that arrive shortly after direct sound can influence the sense of sound direction. The detection of the early reflections in the RIR is an important process in the evaluation of room acoustic quality.

Late Reflections

Late reflected sounds are those that arrive at the receiver after multiple reflections. They make up the reverberant tail of the response. Late reflections consist of a dense succession of echoes, it cannot be integrated with the direct sound and are perceived as separate echoes or as reverberation. Early reflected sounds are much more important than late reflected sounds.

Room Impulse Response

The room impulse response is one the most important parameters to study the acoustics of the rooms. The room impulse response (RIR) is the transfer function (H(f)) of the room between the sound source and the microphone. The information that is given by the RIR is needed, informs about how the room behaves when faces with a certain sound, for an input X and an output Y, the room H, modifies input X and becomes output Y, (2.5).

$$Y(f) = H(f) * X(f) \tag{2.5}$$

Or:

$$H(f) = \frac{Y(f)}{X(f)} \tag{2.6}$$

The reflection density (number of reflections per time unit) (2.7), increases with time. In many rooms the increase in density is about t^2 [12].

$$Reflection\ density = {}^{dn}/_{dt} \tag{2.7}$$

All sources radiate a short impulse in time t=0, within a period, all receivers located within the range of (2.8) are reached by the sources.

$$r = c * t \tag{2.8}$$

For the density of reflection:

$$\frac{dn}{dt} = \frac{\frac{4}{3} * \pi * c^3 * t^2}{V} \tag{2.9}$$

The impulses that reach the receiver thus become more and more dense in time, and the directions of incidence become more and more varied. The energy of the individual impulses also gets closer and closer.

An example of such an impulse response is shown in Figure 3.

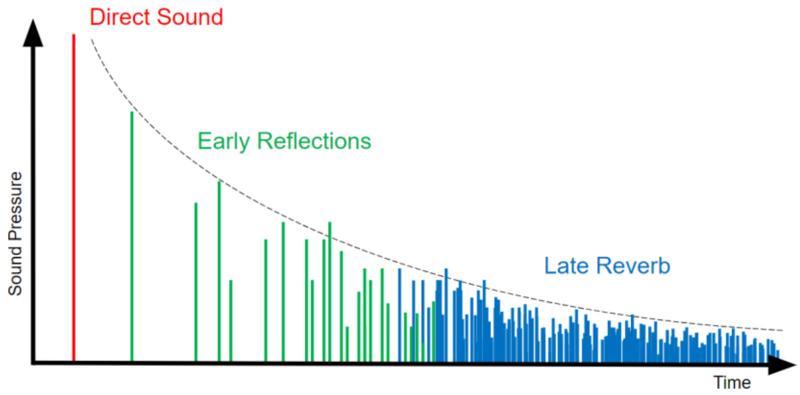

Figure 3 Room Impulse Response Example

Each peak in the Figure 3 represents a single reflection path starting from the direct sound that has not undergone any reflection. The different phases of the impulse response can be clearly distinguished. Direct sound (red) and early reflections (green) can still be easily separated. Over time, the density of the discards increases more and more, and the result is a reverberation curve that decreases exponentially (blue).

The amplitude of the individual discards decreases with each reflection of the wave propagated with an absorbing surface of the room. The wave loses energy each time it reaches an absorbing surface. The energy is reduced by a factor of α for each individual reflection. The amount of energy with which the wave continues after reaching the surface can be calculated with the following formula:

$$|r| = 1 - \alpha \tag{2.10}$$

Algorithm of room acoustic computer simulation

The computer simulations for room acoustics are based on geometrical acoustics. There are mainly three methods, ray tracing, image sources model and hybrid method. The ray tracing algorithm takes an image made of pixels. For each pixel in the image, it shoots a primary ray into the scene. It is a method for calculating the path of waves through a system, while image source method is based on the image source principle [13]. The hybrid method combines ray tracing and image source model. In this project ISM has been used.

The image source model is a well-known technique in geometrical room acoustic if it is desired to generate early reflections (ER). The ISM tries to find, between the source and the receiver, paths of specular reflections. It can consider the source and receiver directivity, as well as the absorption properties of the room. ISM can be used in order to generate a synthetic room impulse response (RIR). The ISM offers a quick and simple way to originate RIRs with different characteristics, like reverberation times. Therefore, the image source model technique is widely used in many fields of application in room acoustics and signal processing [14].

The objective of ISM is to search and locate the purely specular reflection paths between a source and a receiver. The process can be simplified by assuming that sound propagates only straight rays. The rays are then reflected perfectly to the surface in the room. When a ray is reflected, it creates a secondary source behind the surface. This source is located on a perpendicular line to the wall, at the same distance from it as the original source [15].

A reflection on a wall can be simulated by mirroring the source at the wall, as shown in Figure 4. This creates an image source, whose distance and angle in relation to a receiver are identical to the reflected sound path.

A specular sound source is a virtual sound source that emits the same sound as the real sound source [6]. It embodies a single reflection of the sound on a wall. A mirror sound source is built with vertical reflection of the real sound source on the wall. Both sound sources emit the same sound signal at the same time t=0.

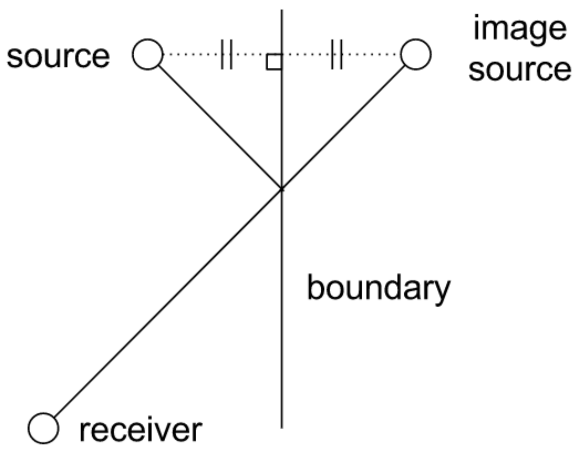

Figure 4. Mirror Sound Source

With the help of mirrored sound sources, higher order reflections can also be represented, so a second order mirrored sound source can be built by reflecting a first order mirrored sound source on an additional wall.

If the surfaces of a room are not flat, but slightly curved, the sound can be scattered or focused

Reverberation Time

Reverberation Time (T) is the time required for the reverberant sound to decrease by 60 dB, since the sound source is off. It generally varies with the frequency. In a room, where all surfaces (walls, ceiling and floor) absorb the same quantity of the sound that reaches them, theoretically, the reverberation time is proportional to the ratio of volume to surface area. The absorption coefficients used to be determined for the following frequencies:

125, 250, 500, 1000, 2000, 4000 Hz. A low absorption coefficient indicates a more reflective material.

The reverberation time is generally recognized as the most important acoustical parameter for any type of rooms.

There are three important reverberation times RT_{60}, RT_{30} and RT_{20}. They are all valid and applicable.

RT_{60} is the time it takes for the level to decay from -5 dB to -65 dB.

RT_{30} is the time it takes for the level to decay from -5 dB to -35 dB, multiplied by 2.

RT_{20} is the time it takes for the level to decay from -5 dB to -25 dB, multiplied by 3.

Reverberation time is used as the most important and often decisive parameter in room acoustic design. If the reverberation time is too long, the individual components of the signal are diffused, if it is too short, a high energy input is required (or decrease the absorption coefficients of the room's surfaces).

The following table shows the variation of the optimal reverberation time values for a frequency range of 500 Hz – 1000 Hz.

Room type	V (Thousands of m³)	T (sec)
Speech	0 - 4	0.7 – 1.2
Music chamber	0.3 - 11	1.4 – 1.6
Opera	2 – 20	1.5 – 1.7
Concert	3 – 15	1.9 – 2.1
Organ music	1 – 25	2.5 and more

Table 2 Reverberation Time for different type of room

Reverberation Time Calculation

If V is the volume of the room and α the absorption coefficient of the walls, floor and ceiling, then the reverberation time can be estimated according to the following formulas [16]:

The Sabine Equation

Wallace Sabine proposed a formula (for $\alpha \leq 0.2$) to calculate reverberation time from the dimensions of the room. It is a simple equation; this equation relates reverberation time to the room volume, and inversely to the absorbing coefficient of the audience, walls and objects inside along the room.

$$RT_{60} = \frac{0.161 * V}{\alpha * S} \tag{2.11}$$

Where,

RT_{60} is the time in seconds required for a sound to decay 60 dB since the sound source is off.

V is the volume of the room,

S is the boundary surface area,

α is the average absorption coefficient.

The value of α is:

$$\alpha = {(S_1 * \alpha_1 + S_2 * \alpha_2 + \cdots S_n * \alpha_n)}/{S}$$

The Eyring Equation

Eyring argued that Sabine's formula could be improved, as Sabine's equation for calculating reverberation time varies with the shape of the room. Eyring created a more general equation than Sabine's. In Eyring's equation it is assumed that the sound field within the room is ideally diffuse. When the field of a room is ideally diffused, all the walls of the room are hit by the same amount of energy per second.

The equation can be seen below:

$$RT_{60} = \frac{0.161 * V}{(-S * \log(1 - Am))} \tag{2.12}$$

Where,

$$V = S_x * S_y * S_z$$

$$S = L * W * H$$

$$S_x = W * H$$

$$S_y = L * H$$

$$S_z = L * W$$

$$S_{tot} = 2 * S_x + 2 * S_y + 2 * S_z$$

$$A = S_x * \big(\alpha(1) + \alpha(2)\big) + S_y * \big(\alpha(3) + \alpha(4)\big) + S_z * (\alpha(5) + \alpha(6))$$

$$A_m = {A}/{S_{tot}}$$

The Fitzroy Equation

In 1959, Fitzroy presented the problem of a more accurate calculation of the reverberation time with non-uniformly distributed absorption

$$RT_{60} = \frac{0.161 * V}{(S_{tot})^2 * (t_x + t_y * t_z)} \tag{2.13}$$

Where,

$$S_{tot} = 2 * S_x + 2 * S_y + 2 * S_z$$

$$t_x = \frac{-2 * S_x}{\log(1 - mean(\alpha(1:2)))}$$

$$t_x = \frac{-2 * S_y}{\log(1 - mean(\alpha(3:4)))}$$

$$t_x = \frac{-2 * S_z}{\log(1 - mean(\alpha(5:6)))}$$

3 Methodology

3.1 Description of the Room

The room studied in this project is a shoebox shaped room, the room was chosen to have the dimensions L=8, W=6 and H=3. This corresponds to the 3D coordinates system *[x, y, z]* [8, 6, 3] (m). It is supposed to be a completely sealed room. First it was necessary to find a correlation between the room coordinate system of each toolbox. Inside each tool there is a script in charge of defining the dimensions of the room to be tested. MCRoomSim uses the right-hand coordinate system, while AKtool and Polarch are defined to use the first octant of a right-handed three-dimensional cartesian system coordinate.

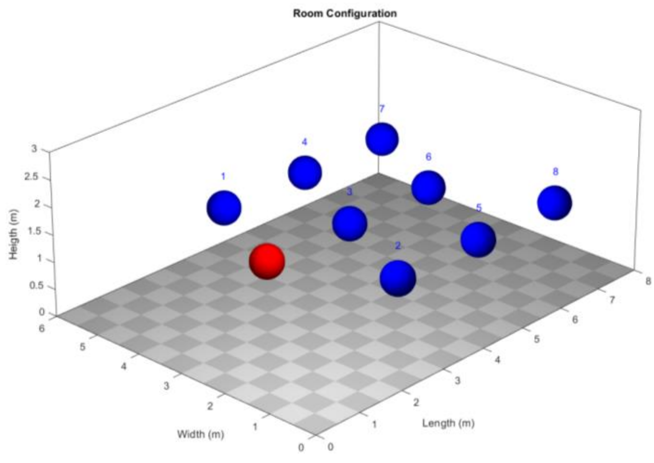

Figure 5. Setup of the Simulated Room

In all the toolboxes an absorption value can be assigned for each of the walls that make up the room. With 1 being the maximum possible absorption value (total absorption), there is no reflexion of the wave on the surface. In the case where the absorption value is 0, when the wave arrives at a surface occurs total reflexion.

Frequency (Hz)	Absorption coeff.
125	0,1735
250	0,1732
500	0,2403
1000	0,2255
2000	0,1992
4000	0,1924

Table 3 Absorption Coefficient per Frequency

A certain absorption coefficient is attached to the walls of the room. The absorption values were assigned independently for each of the frequencies. The frequencies to be studied are those described in the ISO 3382-1 standard, these values and absorption coefficient values are shown in Table 3. The acoustic study of the room will be carried out using omnidirectional sound source and microphones. The configuration of the room can be visualized in Figure. 5, the red ball is the omnidirectional sound source and the blue balls are the receivers.

$$V(m^3) = L * W * H \tag{3.1}$$

$$V(m^3) = 8 * 6 * 3 = 144 \tag{3.2}$$

Parameters	Value
Surface (m²)	90
Volume (m³)	144
Humidity (%)	42%
Temperature (Celsius)	20
Atmospheric pressure (Pa)	101.325
Sampling Frequency (Hz)	48000

Table 4 General Parameters

3.2 Measuring Position

To take the correctly measurements of the room, the receiver’s positions should be placed where the real listeners would be in, for example students in a classroom. For a good reverberation time measurement, the room must have receivers all over the room, like shown in Figure 5, in order to be able to collect all the room information through the receivers.

The receivers were carefully placed, based on the ISO 3382-1 standard [1]. The measurements obtained by the different receivers will vary systematically from one seat to another. It is therefore necessary to include an adequate number of sound source and receiver positions to characterise the room. The receivers must be separated by a minimum distance of 2 metres between the receivers themselves, and a minimum distance of 1 metre between the receiver and any type of reflecting surface (walls, floor, ceiling).

In order to prevent the receiver from obtaining a very loud direct sound, the receivers must be placed at a minimum distance of 1.2 metres from the sound sources.

The virtual sound sources positions should be placed where the natural sound source would be in the room, for example a teacher in a classroom. As it is a small space only one sound source needs to be used. The study should be carried out with a sound source with a height of 1.5 metres, simulating the height of the real sound source [1].

The Figure 5 shows the geometrical representation of the room.

Receiver	Position (x, y, z) (m)
Rec 1	[3, 5, 1.2]
Rec 2	[3, 1, 1.2]
Rec 3	[4, 3, 1.2]
Rec 4	[5, 5, 1.2]
Rec 5	[5, 1, 1.2]
Rec 6	[6, 3, 1.2]
Rec 7	[7, 5, 1.2]
Rec 8	[7, 1, 1.2]

Table 5 Receivers Position

4 Results and Conclusions

Once all the simulations were successfully carried out, they were stored in ".mat" files in order to compare the simulations of the three toolboxes data at the same time, both analytically and graphically, and to be able to see what the different Matlab toolboxes have in common.

4.1 Comparison of the Room Impulse Responses

By examining the initial part of the measured and simulated RIR shown in Figure 6 and Annex Figure 1, both graphs are identical. This is due to several reasons, the first reason being that the room is geometrically perfect, Figure 5. The second reason is that the two microphones (receiver 1 and 2) are at the same distance from the sound source (2,23 metres), the front wall (3 metres), the back wall (5 metres) and finally it is at the same distance from the nearest wall (1 metre). This last wall is the surface that returns the reflections with the greatest amplitude in the room, since is the closest to this wall than any other. It should be noted that al the surface (ceiling, floor and walls) have the same absorption values for the same range of frequencies, values which can be seen in Table 3.

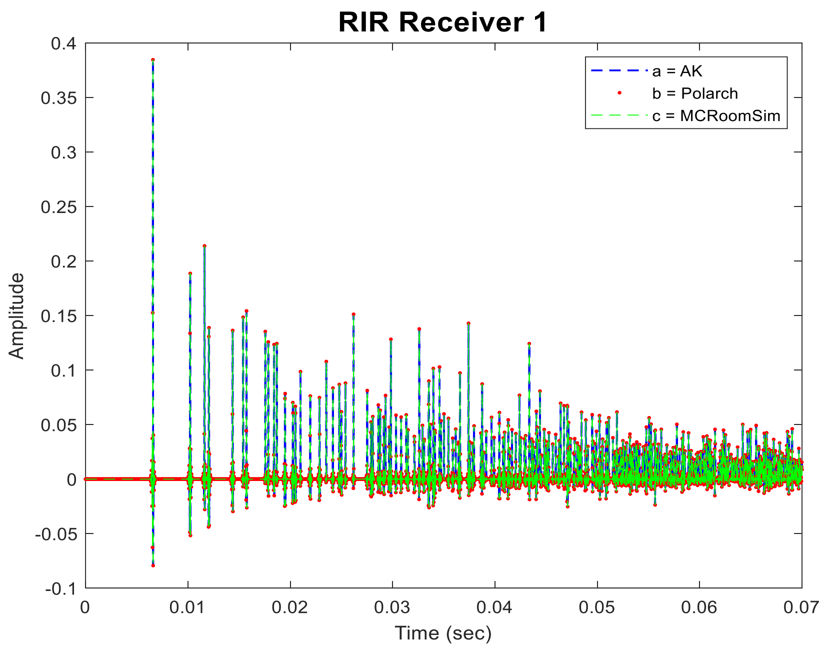

Figure 6 Receiver 1 RIR

When two receivers share the same coordinates on the X and Z axis, and are at the same distance from the centre of the room (Y=3), it can be assumed that the two receivers will receive the same data, with the same amplitude level, and at the same time. Assuming, of course, that the sound source is on the imaginary line of the centre of the room (Y=3). The direct sound in both graphics shows they are identical; they have the same level of amplitude with which the receiver collects the information. Also, receivers 1 and 2 get the direct sound at the same time (ms).

In Figure 7, the graph of receiver 3, the direct sound arrive before than in receivers 1 and 2. This is because receiver 3 is located 0,236 m closer to the sound source. It can also be seen that the reflections from the different walls reach it more broadly than those from receivers 1 and 2, since it is in the centre of the room. The greater the distance from the receiver to the sound source, the longer the time it takes for the signal to reach the receiver, and the lower the amplitude value of the wave. Receiver 3 is the receiver that reaches the direct sound with the highest amplitude level and the shortest time between sound source and receiver in the whole room.

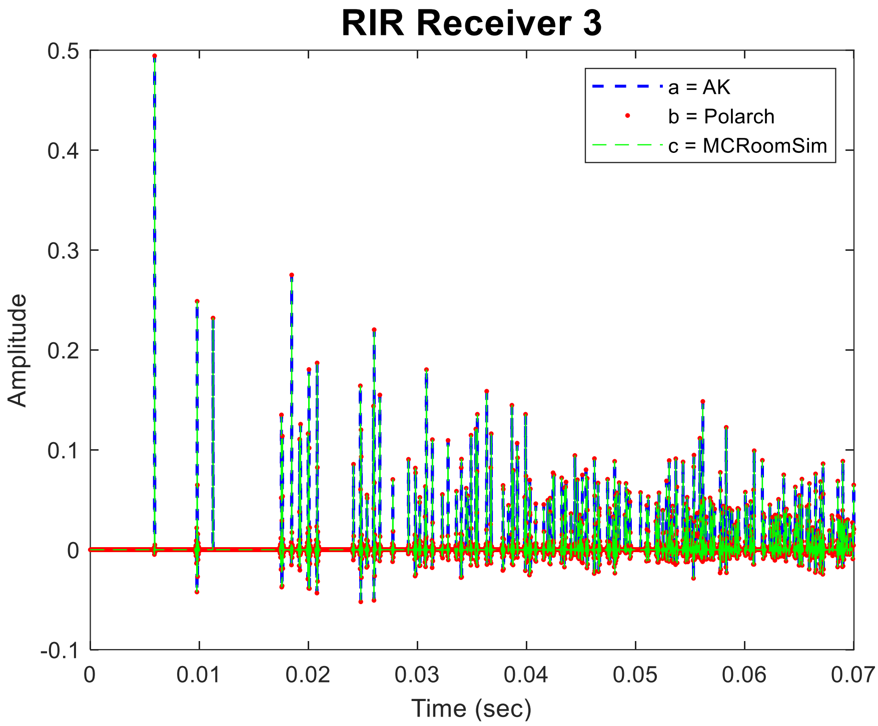

Figure 7 Receiver 3 RIR

Looking at the receivers 4 and 5, Figure 8 and Annex Figure 2, it can be seen that occurs the same as the data of the receivers 1 and 2. As they are symmetrically located in a geometrically perfect room, the collected data by both receivers is identical. The same applies to receivers 7 and 8, which are also located symmetrically in the room. They will obtain the same sound at the same time and the same amplitude, due to the same reflections on the walls.

Receivers 4 and 5 are a clear example of the fact that the greater the distance from the receiver to the sound source, the longer it takes for the signal to reach the receiver. As shown in Figure 8 and Annex Figure 2, the direct sound arrives 10 ms later than on receiver 3, due to the distance from the sound source. By examining the first reflections, they arrive with an amplitude of 8 ms less than the direct sound. In other receivers, the amplitude of the first reflections may be less abrupt than for the direct sound. After the first reflections, no other superior peak in amplitude appears, the signal decays exponentially, no reflections are seen that amplify the received signal.

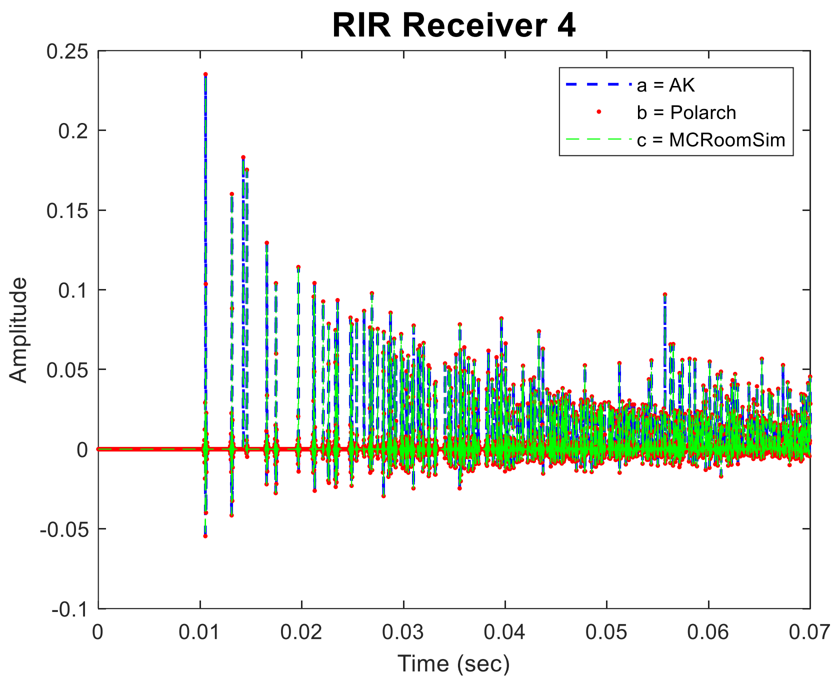

Figure 8 Receiver 4 RIR

Figure 9 and Annex Figure 4 shows the RIR of receivers 7 and 8. They are receivers with the lowest direct sound amplitude level. They are also the receivers that take the longest time for the sound to reach them. The first reflections are of similar amplitude level to the direct sound. Unusual levels of amplitude can be seen between 30 and 40 milliseconds. This is due to the reflections produced on the walls and the time it takes to reach the farthest receiver. The back and the left wall are the closest walls to the receiver; therefore, they offer a greater amplitude reflection than the rest of the walls.

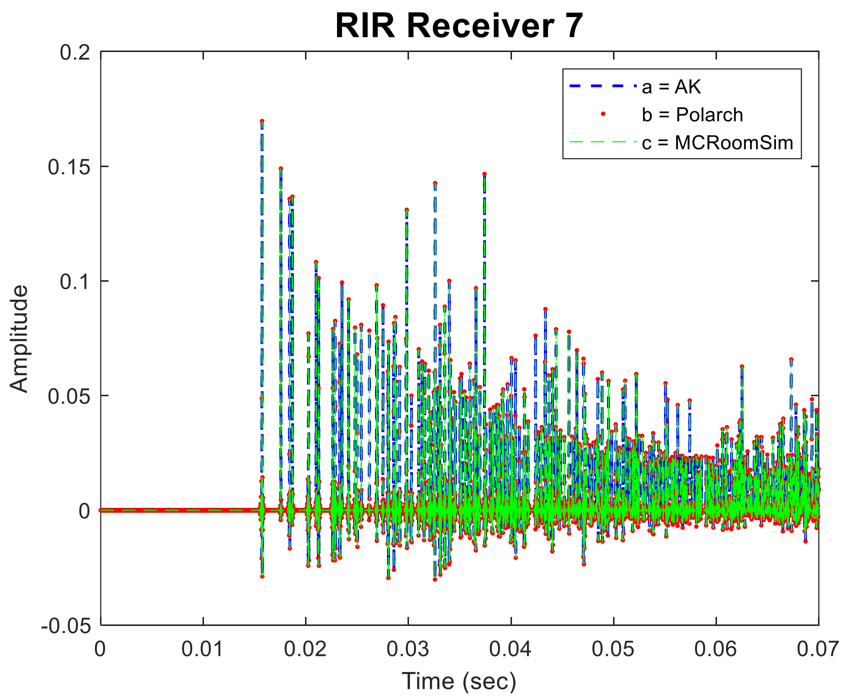

Figure 9 Receiver 7 RIR

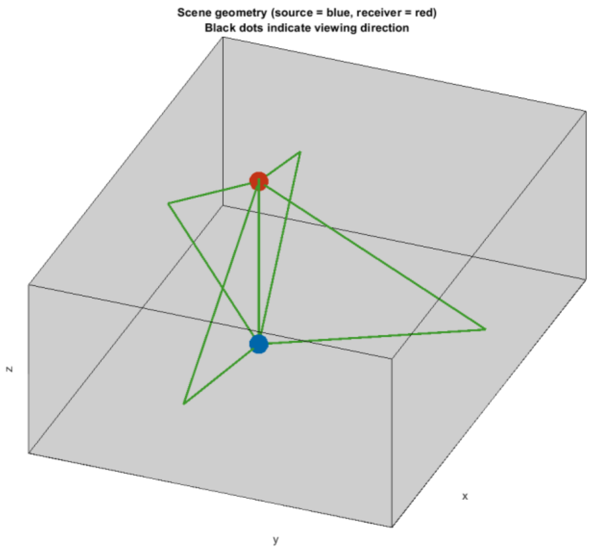

Figure 10 Receiver 7 Reflection Paths

Looking at the previous figures, it can be stated that the three toolboxes to be tested behave in an identical way. In fact, when obtaining the RIR, the microphones receive the information at the same time. This shows that the propagation of the wave in the room is the same for the three toolboxes.

Looking closely at the RIR graphs, on a temporal level it can be stated that no calculated RIR of the 8 microphones positioned along the room is echoed, as previously explained, if the time difference between the direct sound and the first relevant reflection exceed a period of about 50 to 80 milliseconds, an echo is perceived.

In order not to shown so much graphics, the receivers that behaved the same way have been assigned to the Annex A. These graphs contain the data of the three toolboxes collected by a receiver and drawn at the same time, in order to be able to compare them successfully. In the graphics the different toolboxes are differentiated. Each toolbox has been assigned a colour. AKtool is drawn with blue dashes, Polarch is shown with red dots and MCRoomSim with green dashes.

The graphs belong to the 8 receivers distributed throughout the room, one graph per receiver, a graph showing the first 70 ms of the receiver’s RIR.

4.2 Comparison of the Reverberation Times

The analysis demonstrated that there is no significant difference between the three measurement methods: Sabine, Eyring und Fitzroy Formula.
Comparing Eyring with Fitzroy it is easy to visualize that they match perfectly in the simulation. Regarding the reverberation time calculated using Sabine's method, it should be noted that it behaves in an identical manner. Sabine's reverberation time is always a little higher (0.13 sec) for each frequency that the other two methods, but always maintains constant the value above them.

As can be seen in Figure 11, the reverberation time values calculated by the formulas 2.11, 2.12 and 2.13 are like reverberation time values of a speech room. In order to use such a room for speech, a brief acoustic conditioning could be done by increasing the absorption values to the walls, floor and ceiling of the room. Therefore, in this way, we would be able to decrease the reverberation time to the desired reverberation time for voice (0.7 – 1.2 sec).

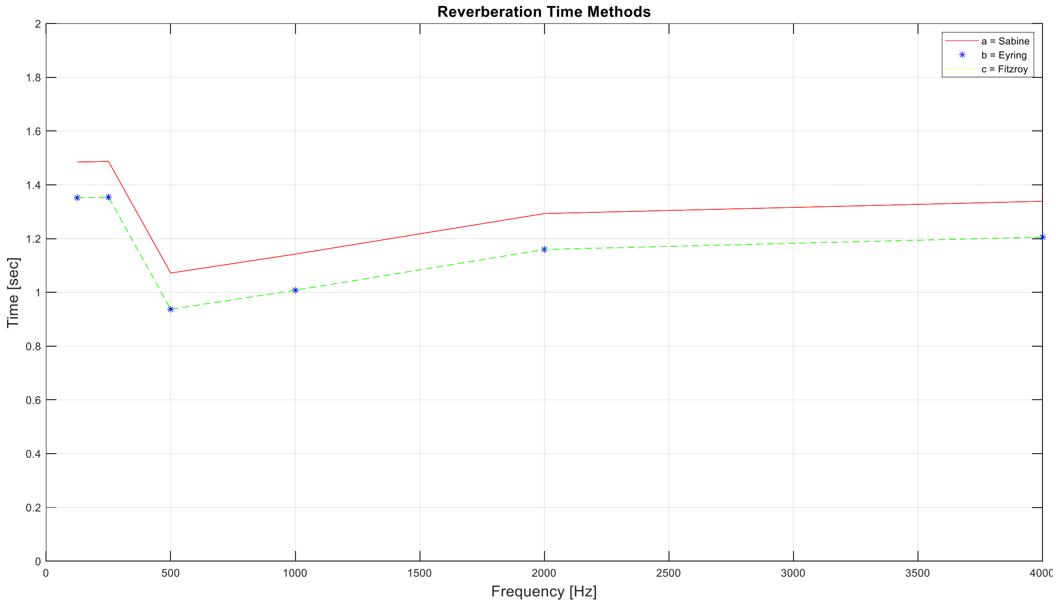

Figure 11 Reverberation Times Comparison

Looking at the figure above, Figure 11, it can be stated that as far as the calculation of reverberation times is concerned, it is logical that the values obtained for the reverberation times are identical. This is because the same formulas are used to calculate the reverberation times. The reverberation times consider, among other parameters, the volume, the surface area of the room, the absorption coefficient, atmospheric pressure… These are identical parameters for the three toolboxes.

5 Future Research

In this chapter, future research, different topics and characteristics are proposed to carry out the acoustic study of the room used in this project, in order to be able to analyse an acoustic room in a much more complete and professional way.

An acoustic study could be made of an identical room to the one used for this project but using directional microphones and sources.

Another alternative would be to carry out the same acoustic study but using a directional sound source and directional receivers. The sound source would be rotating around, pretending to be a professor in motion during a university class.

It could be also made the same acoustic study but using microphone arrays.

In the future, a very interesting study could be to carry out the same acoustic study of a room, comparison of different toolboxes for a shoebox-shaped room, but for binaural receivers. Binaural receivers can be implemented by heads with two microphones (left and right ear) that collect the information of the room. Using binaural channels, making use of the Head Related Transfer Function (HRTF).

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7 Annex

The graphs belong to the receivers distributed throughout the room, a graph showing the first 70 ms of the receiver's RIR.

Annex A. Room Impulse Responses

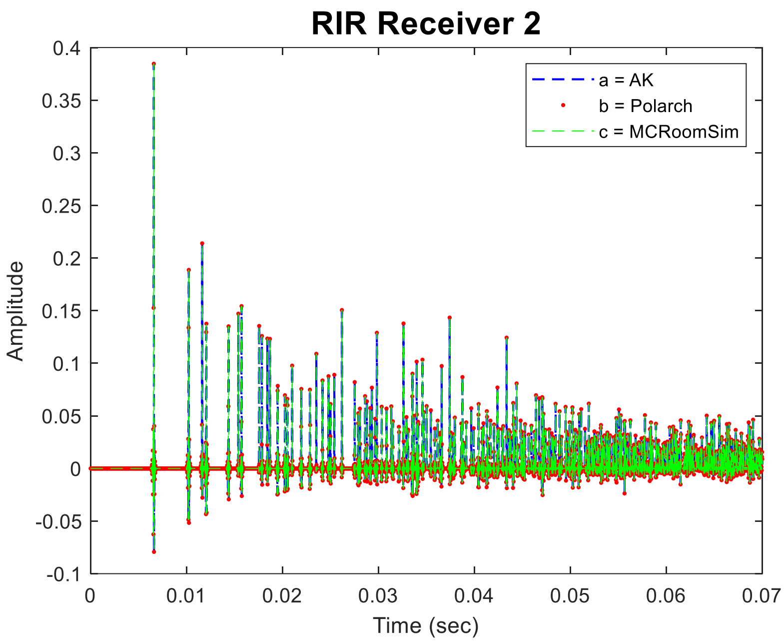

Figure 1 Receiver 2 RIR

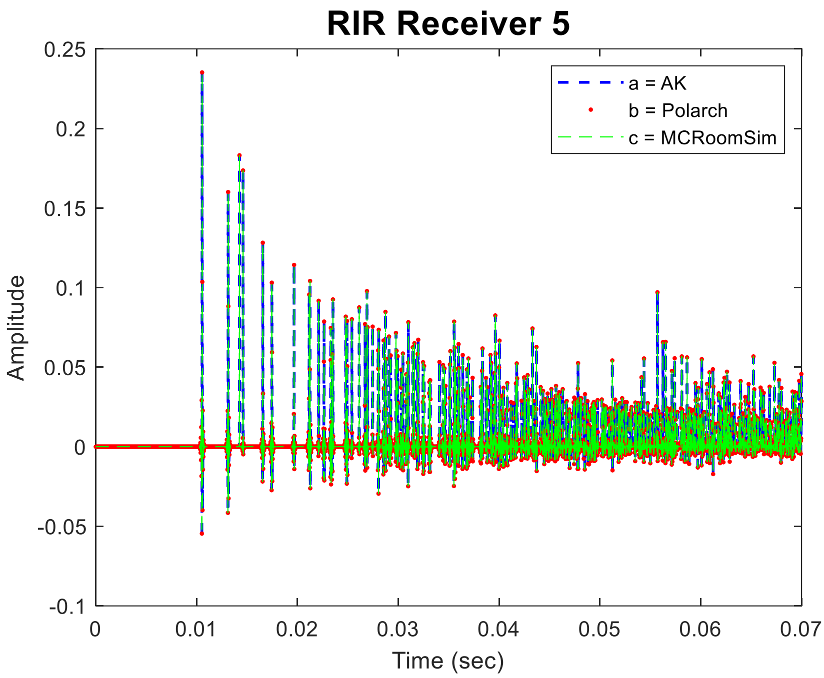

Figure 2 Receiver 5 RIR

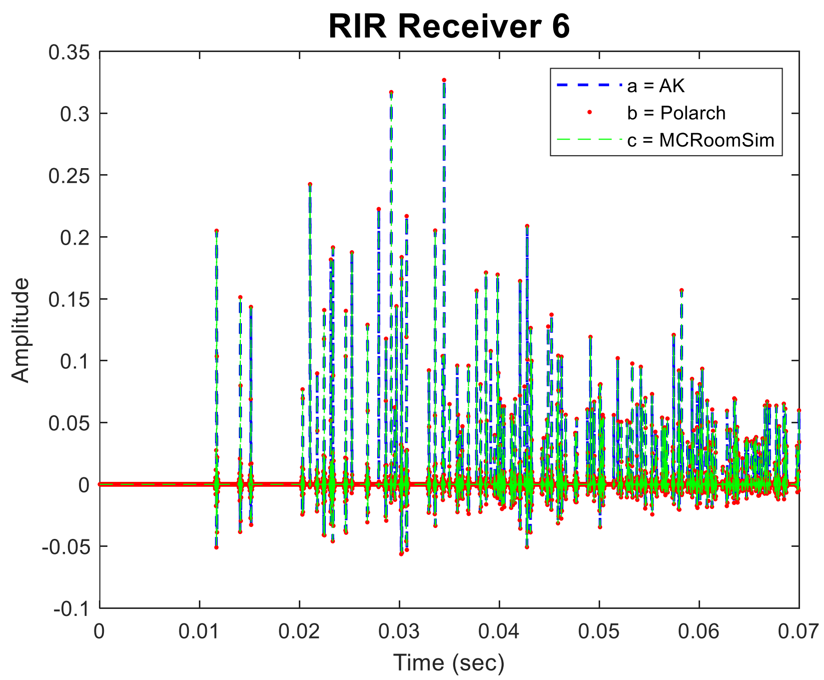

Figure 3 Receiver 6 RIR

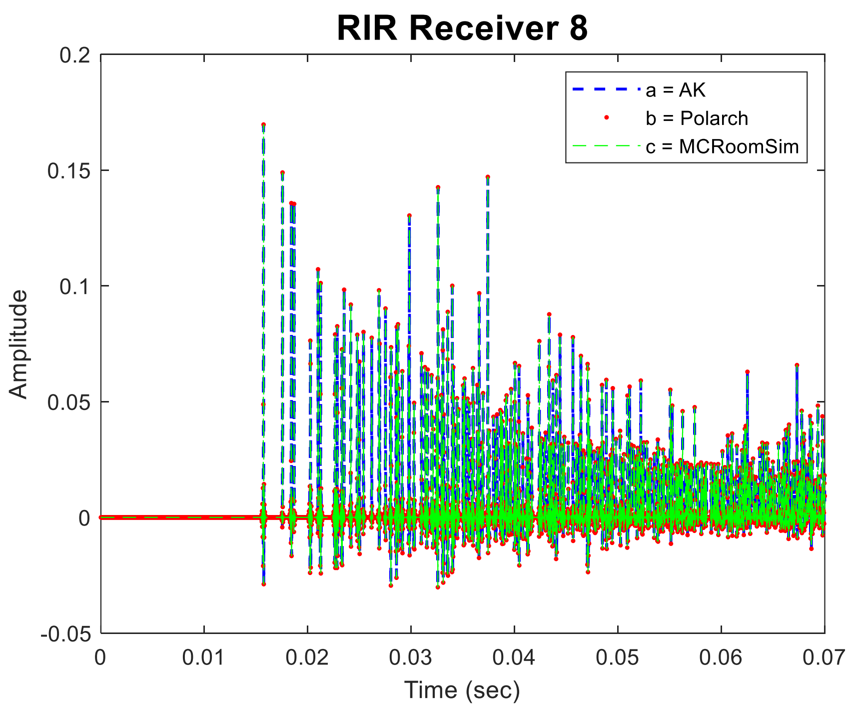

Figure 4 Receiver 8 RIR

Annex B. Reflection Paths

The following figures shown the paths of reflection from the sound source to the receivers.
It can be easily seeing how the sound leaves the omnidirectional sound source (blue ball) and reaches the receiver (red ball). The receiver does not receive a single signal, but rather reflections from the ceiling, floor and walls.

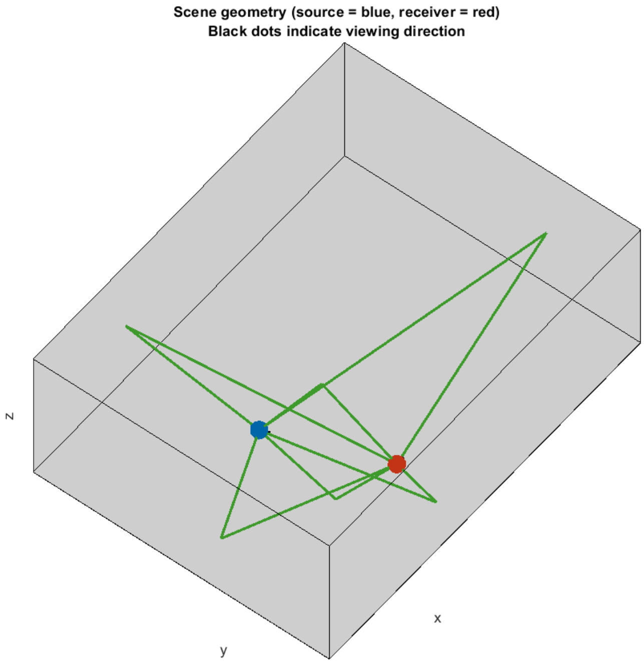

Figure 1 Receiver 2 Reflection Path

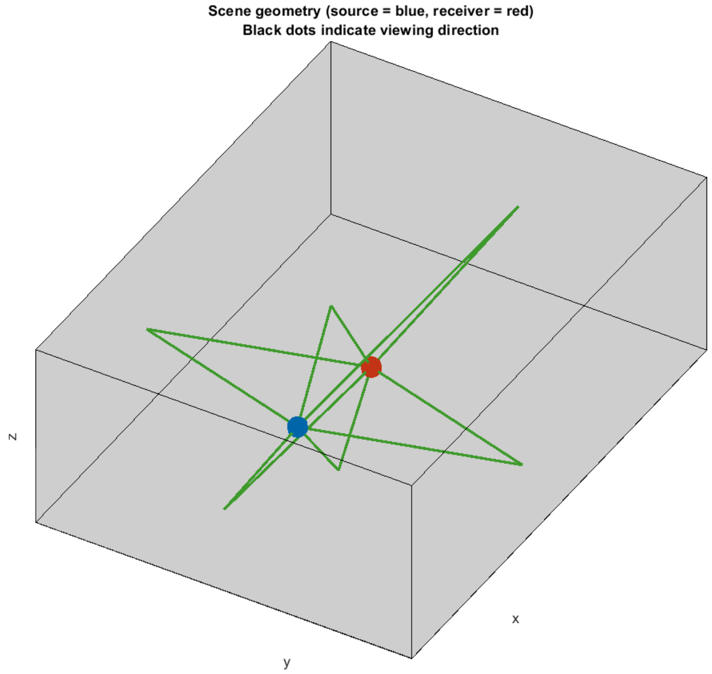

Figure 2 Receiver 3 Reflection Path

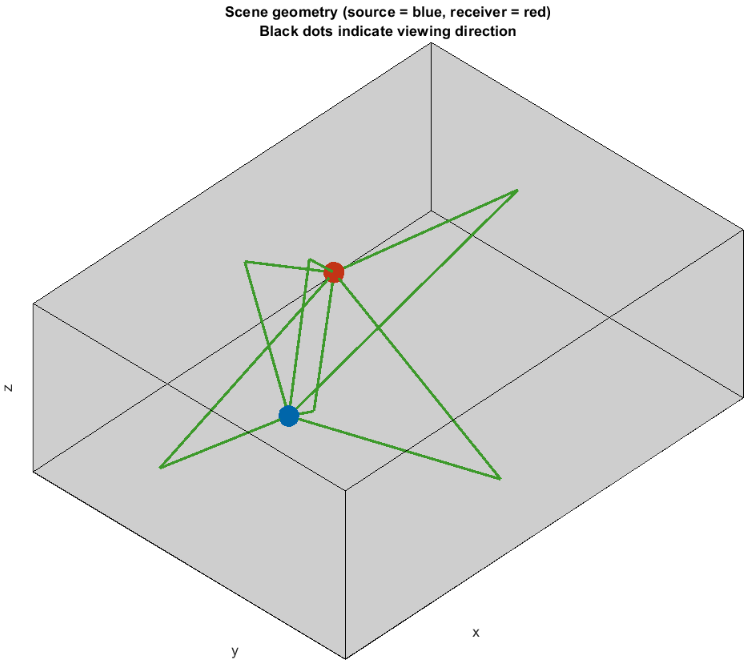

Figure 3 Receiver 4 Reflection Path

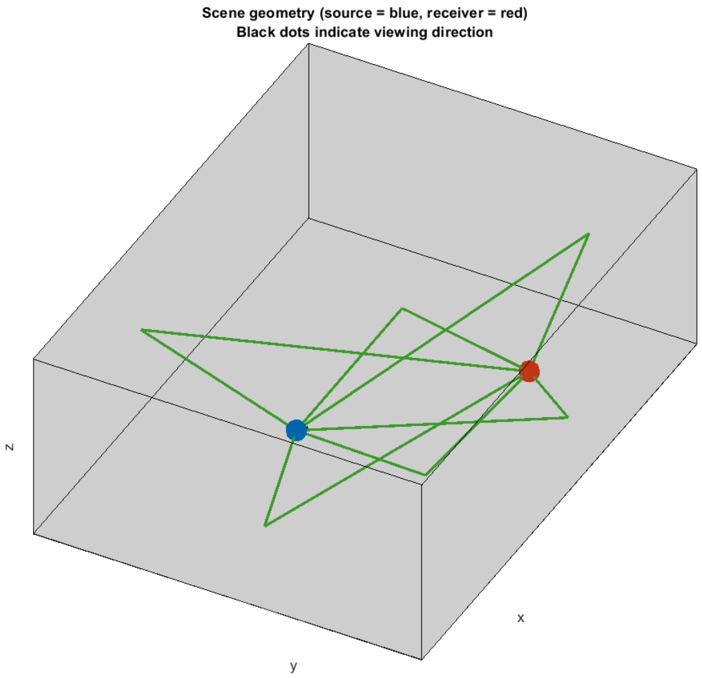

Figure 4 Receiver 5 Reflection Path

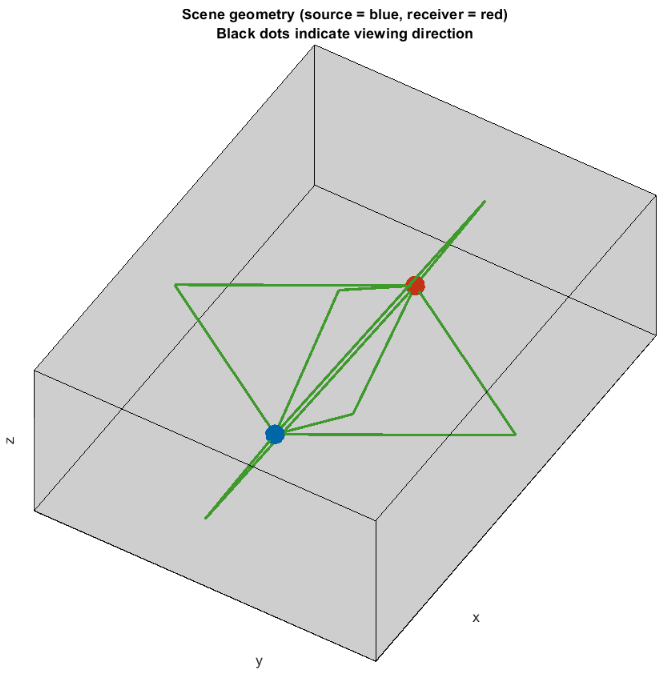

Figure 5 Receiver 6 Reflection Path

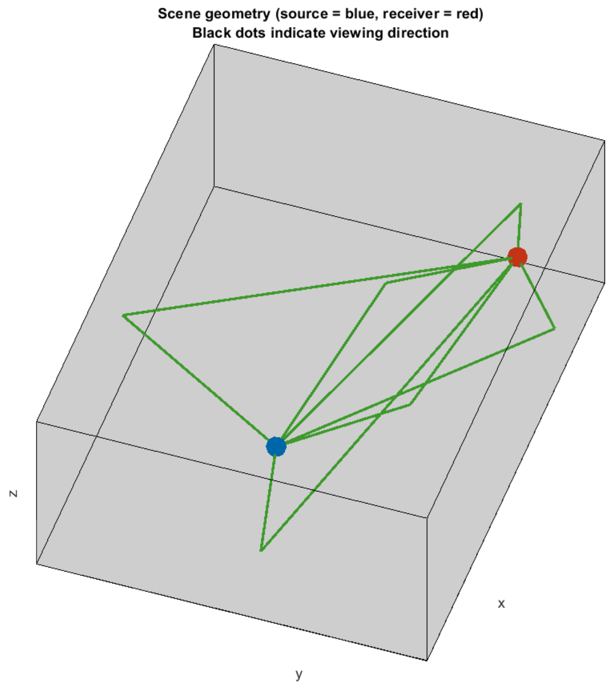

Figure 6 Receiver 8 Reflection Path