University of Liège

Faculty of Applied Sciences

Evaluation of the accuracy of measurements performed by a spherical array of microphones

Graduation Studies for obtaining the Master's degree in Electrical Engineering by:

Author: Iratxe Barreiro

Supervisor: Prof. Jean-Jacques Embrechts

Academic year 2015-2016

University of the Basque Country

Master in Telecommunication Engineering

Abstract

This Master's thesis consists in evaluating a spherical microphone array developed in a previous project, which will be referred to later, that aimed to extract 3D Room Impulse Responses with a measurement system composed of the referred microphone, among other tools. We will focus on the accuracy of localization of the sound as a direct continuation of the previous work. It could be considered as the main application of this system, hence it is important to emphasize the need of this evaluation.

The measurement system consists of a spherical antenna containing 16 microphones, two cards with 8 channels each and the associated signal treatment. The current aim is to evaluate the precision provided by the system, which is evaluated in several situations. Different acoustics measurements have been taken in an anechoic room and also in a reverberant room.

Adobe Audition and the Matlab software are used in order to process the information which is provided by the measurement system.

Page-summary

Title of the work: Evaluation of the accuracy of measurements performed by a spherical array of microphones.

Name and surname: Iratxe Barreiro

Affiliation: Master's degree in Electrical Engineering

Current academic year: 2015-2016

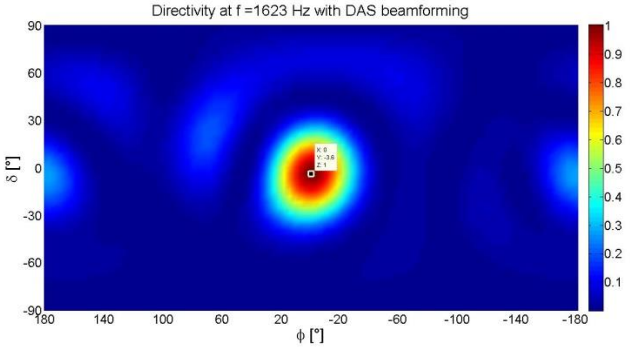
Name of the promoter: Prof. Jean-Jacques Embrechts

This Master's thesis consists in evaluating a spherical microphone array developed in a previous project, which will be referred to later, that aimed to extract 3D Room Impulse Responses with a measurement system composed of the referred microphone, among other tools. We will focus on the accuracy of localization of the sound as a direct continuation of the previous work. It could be considered as the main application of this system, hence it is important to emphasize the need of this evaluation.

The measurement system consists of a spherical antenna containing 16 microphones, two cards with 8 channels each and the associated signal treatment. Adobe Audition and the Matlab software are used in order to process the information which is provided by the measurement system.

The current aim is to evaluate the precision provided by the system, which is evaluated in several situations. Different acoustics measurements have been taken in an anechoic room and also in a reverberant room. A calibration task can be found and additionally, the first measurements with their corresponding analysis are presented. After that, measurements with pure tones and impulse responses have carried out. In both parts, we focus on the horizontal plane. We introduce the diffuse field measurements using pure tones and pink noise.

Then we treat the recorded information to show different diagrams obtained by different beamforming methods. I conclude that there are some unusual results but the microphone array works. The experimental study includes capturing the impulse responses and evaluating of different types of beamforming. We conclude that PWD beamforming works badly in the measurements of this project. We can also say that the sound field in the reverberant room is not really diffuse.



Acknowledgements

My first thanks go to Prof. Jean-Jacques Embrechts, who proposes this work and has guided me in the research and developing of this project. He has always been available to help myself and to give me many advices. He was a great supervisor and professor with plenty of patience who has shared his knowledge pleasantly.

I have to express my gratitude to Angel Calderon Jimenez, who has dedicated his time helping me and developing each device that we have needed in the developed project. Every work has been better with his presence because he always has been offering his help and knowledge in a good mood.

Furthermore, I would like to thank all the members of the jury for their time dedicated in the evaluation of this work.

I wish also thank my family, because I am far from them but they have been supporting me in each moment. I would like to thank my friends, those ones who I have left in my country as well as those who have been with me during these past months.

Contents

List of Figures

List of Tables

1. Introduction

1.1 Context

The development of systems which are able to localize the sound source direction has nowadays interest due to the implementation in robots, cameras for video conference room or using devices of defence. Due to this kind of applications, it is necessary to make experiments in room acoustics, not only to analyse the acoustic behaviour in those circumstances, also to deal with source localization. Sound source localization system based on microphone arrays is being increasingly used.

Microphone arrays are able to obtain better directivity than a single microphone due to several waves arrives at each microphone with a lightly delay. Thus, the incident waves can be combined to acquire higher performance.

Spherical network is especially useful analysing the sound field due to the capability to measure it in three dimensions. This kind of configuration is ideal for capturing and description of the sound field in 3D space.

The processing of the acoustic field is a complex area to work, it is hard to study and analyse every method, property, or technique. Therefore, in this thesis we focus in several points in order to obtain relevant and useful conclusions in order to take advantage of our measurement system.

The development of the current project is a direct continuation of a previous work, and as a consequence, these results create the starting point of this Master thesis.

1.2 Objectives

This project is based in a previous Master's thesis which was developed in the 2012-2013 academic year by Hermine Feron 48[1]. The aim of that project was to create an antenna of microphones and to analyse room acoustic.

On the other hand, the main goal of the current project is to carry out a thorough review and assessment of the implemented system in order to analyse the accuracy of the instrument and, if it were possible, to obtain a better result in the localization of the sound source.

In that previous project, the measurements in an anechoic chamber demonstrated that the system was able to measure the directional responses in the entire space and it was able to localize the first reflections in a wide band pass.

An analysis and evaluation of the system have been conducted in the present dissertation.

1.3 Structure

This document is organised in 6 sections.

Next section correspond to the state of the art, which is composed of three parts, the first one will focus on spherical microphones arrays' properties, secondly, we will refer to other works in order to be informed about several experiments and analysis of microphone arrays with a spherical configuration. The last part the chapter 2 will do special attention to main characteristics of the current project.

In the following chapter, a calibration task can be found and additionally, the first measurements with their corresponding analysis are presented.

After that, most of the carried out measures are given in the Chapter 4, the obtained results are divided into two big sections, which are measurements with pure tones and impulse responses. In both parts, we focus on the horizontal plane.

In chapter 5, we introduce the diffuse field measurements using pure tones and pink noise.

Finally, we will finish this document with some future works and conclusions about the results concerning the current project.

2. State of the art

2.1 Spherical arrays

In this section, we focus on some characteristics of spherical arrays and their advantages in relation to the aim of the current project.

Previously, H. Feron [1] talked about three categories of antennas taking into account the distribution of the microphones, linear, planar and volumetric. After the analysis of different parameters spherical geometry was chosen mostly due to the capability of providing sound field in three dimensions.

Thus, the most important thing is that with spherical array it is possible to have a 3D description of the sound field around the microphone. Nevertheless, with other kind of configuration, such as a linear array, is not possible. This is why it is interesting to use this sort of arrays in the particular application of impulse response measurements in room acoustics, which is the main application of our developed system. Thus, the spherical geometry is the most effective one for our main purpose. Due to this reason we will use a spherical array instead of other, such as circular or linear.

The advantages of using microphone arrays have to do with directivity and removing interferences. Spherical microphone array has difference sensitivity depending on the beamforming that we use. The received signals can be combined forming a beam, this help to achieve good results when we are talking about directionality.

We will use beamforming methods to process the captured signals with the spherical microphone. Hence, we first introduce the beamforming concept. Beamforming is a technique in signal processing which allows isolating a signal from a specific direction.

There are three main beamforming methods which we will use frequently during the development of this project. The first one is called Delay-and-Sum (DAS) beamforming method and it is based on applying a delay to the microphone signals and a sum of them. It is a sum of phase shifted signals to obtain all the output of the microphones in phase. The second method is the MVDR (Minimum Variance Distortionless Response) and the third one is PWD (Plane Wave Decomposition) beamforming method [2]. The reader is also referred to [1] for more details about these beamforming methods.

There are several works about sound source localization, therefore, in the following subsection we will try to focus on several works where they use a spherical microphone array.

2.2 Diverse kind of tests

We focus on applications of microphone arrays with spherical configuration because we will use this kind of array.

Room acoustics are usually analysed calculating impulse responses [1] and standard acoustic parameters. However, directional impulse response is more useful to study the spatial impressions and the room structure.

Directional sound-field information is more useful when the purpose is to analyse the sound-field in auditorium acoustics. There are suitable indicators for room acoustics that require spatial sound-field information. Different methods could be used in order to measure them, but spherical microphone arrays are nowadays more interesting due to the inherent rotational symmetry in the spatial analysis [1]. Directional room impulse response (DRIR) could be measured by directional microphones or by spherical microphone array. The last one is helpful in order to create directivity patterns with a single measure.

As we will work with a spherical microphone array, we have investigated about other projects where that kind of microphone is used. We are interested in getting to know the obtained results and the methods that they have applied to get the best effect to each purpose.

In the thesis [1] different kinds of antennas are named, as linear, planar or volumetric, each of them could classify by the positioning of the microphones, as uniform, not uniform and random.

In [3] a rigid spherical microphone array was designed with two sampling configurations, firstly with M=50 samples for order N=4 and secondly with M=98 samples for N=6. In these configurations they use a single microphone to measure the impulses responses moving the sensor to several positions.

It is shown in [3] that the optimal use happens around $kr \approx N$ when the radius r is equal to the surface radius (r=a), where k=w/c (‘c’ is the speed of sound, 343 m/s, ‘w’ represents the angular frequency and ‘k’ is the wave number). If kr is lower, transducer noise and positioning errors appear. On the other hand, when kr is higher than N, aliasing errors appear in the results.

The experiments carried out in [3] show a narrower circle in the results with N=6, this means a better spatial resolution but with N=4 there are less errors caused by measurement noise and microphone misplacement.

In this paper it is concluded that the direction of arrival direct sound and the first reflections have been detected accurately in rooms using impulse response information. Furthermore, it is said that array performance was improved when time windowing of the impulse responses was employed before plane-wave decomposition.

In [4] an open-sphere configuration is used with six spatially distributed omnidirectional microphones, where the size of the radius is 7.5 cm, in order to estimate the arrival direction and to localize the sound source. The authors used three sine sweep signals, with a length of time of 3 seconds, as simultaneous sources in different location of a reverberant chamber.

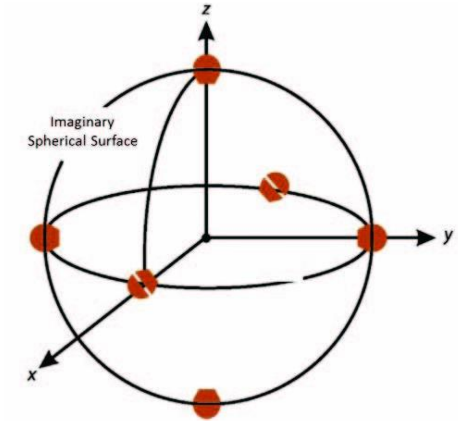


Figure 1. Conceptual diagram of the array [4]

As beamforming methods, the authors have studied MVDR steered-power response (SPR) weightings with PHAT (phase transform) as a combiner for all microphone pair GCC (Generalized Cross-Correlation) functions.

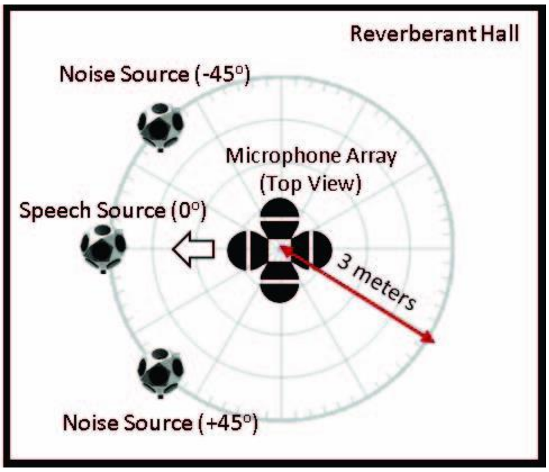


Figure 2. Measurement setup and Microphone reference coordinate system [4]

The results presented in this paper are better using MVDR weights in GCC-PHAT instead of using only GCC-PHAT. The authors obtained quite successful results in reverberant environments.

Using PWD (Plane Wave Decomposition) is common in spherical microphones but is also sensitive to random array errors from practical applications [5]. Robust maximum-directivity (RMDI) beamformer can provide balance between the maximum value of DI (Directivity Indices) and the robustness against the random errors at high frequencies, but if we want the same robustness at low frequencies is difficult to keep a high DI [5].

In the paper [6], the authors have employed a sphere microphone array in order to estimate the direction of arrival (DOA) of the main reflections in a room and the corresponding surfaces. A new more robustness method is proposed which consist on using EB-MVDR (Eigenbeam-Minimum Variance Distortionless Response) beamformer to localize the sources (direct contribution, reflections, interferences…) with a steerable spherical microphone array and then extract the localized signals by means of EB-MVDR.

This method is robust and precise, with a good performance even with interferences. They have also tested with DAS (Delay-and-Sum) and PWD but the result had a low resolution. EB-MUSIC is well-know and could offer good resolution if the number of sources is known. In order to verify the efficacy, other experiment is presented in this paper, this time with two signals (a white noise and a music) played by two loudspeakers. The result was got by EB-MVDR with a position quite near to the reality.

In the paper [7], the performance of ML (Maximum Likelihood) and WSF (Weighted Subspaces Fitting) methods are studied. Five DOA estimation methods are presented in this work. PWD, MUSIC (Multiple Signal Classification), ML stochastic (S) and deterministic (D) and WSF, which is a long sample approach of SML method.

The authors have used the em32 Eigenmike, a spherical microphone array with 32 capsules [8], in order to make the measurements and to study the estimation of the reflections' direction from spatial room impulse responses. They simulated two cases, one of them was with a single reflection in wideband and in the second one they applied four delays with two reflections in the same time-window. Concerning to the results, they were investigated with the root mean squared error (RMSE). The first conclusion was about PWD, it was poor at low frequencies (<500 Hz) in DOA estimations of several reflections.

Real situations were analysed in a semi-anechoic room, where they introduced seven reflexions including a reflective corner. These results show that ML and WSF methods provide more accurate information than MUSIC or PWD about acoustic rooms. Real experiments also show that stochastic ML performs worse than the deterministic one or the WSF. Thus, in this paper [7] the WSF was chosen as the best method in the reflections analysis.

In the paper [9] the authors have used a measurement system called 3DVMS (3-Dimensional Virtual Microphone System), which captures 32-channels impulse responses using a spherical microphone array (Eigenmike™, Figure 3) and a matrix of 32 x 32 FIR filters. The method consist of synthesising 32 directive virtual microphones in the direction of the capsules employing digital filters [10].

Figure 3. EigenmikeTM probe [10]

The scheme of the signal processing is shown in figure X, where xm corresponds to the input signals of M microphones, yv to the output of V virtual microphones and hm,y to the matrix of filters. Each output is obtained applying a sum of the results of the convolutions of the M inputs with M filters.

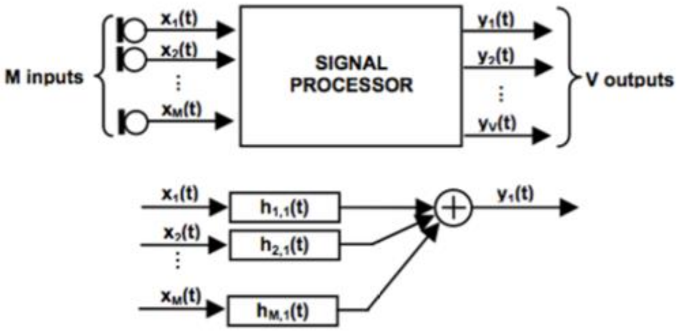


Figure 4. Scheme of the signal processing [10]

In the theatres (Figure 5 and Figure 6), the authors used a dodecahedron as omni-directional source playing a sine sweep as test signal.

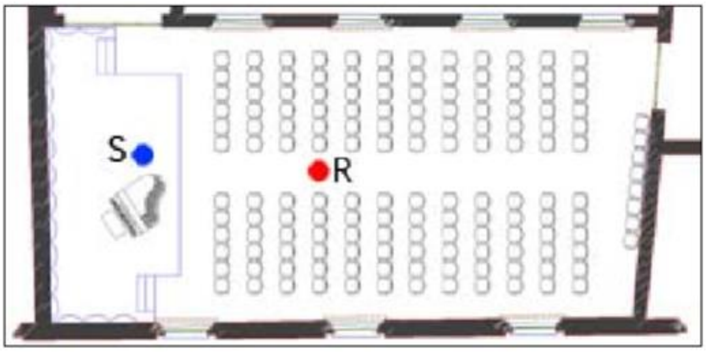


Figure 5. Figure of "Sala dei Concerti" (Parma) [10]

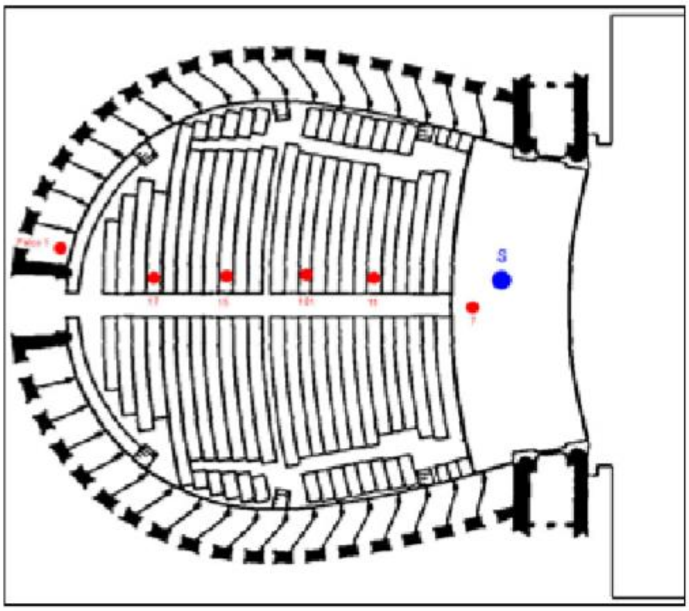


Figure 6. Map of "La Scala" (Milan) [10]

The authors in [16] chose to synthesise 24 virtual microphones on the azimuthal plane and 32 in the directions of the capsules as it is shown in Figure 7 and Figure 8.

Figure 7. 24 virtual microphones for horizontal polar [10]

Figure 8. 32 virtual microphones for 3D map [10]

The results of these experiments show the effect of the distance between source and receiver, the greater the distance, the more relevant lobes of the reflections comparing with the direct sound.

It is also interesting the fact that very low frequencies (500 Hz) appear before the direct sound due to the radiation of the wooden stage (Figure 9) and it is not the case for 4000 Hz band (Figure 10). It is noticed that there is a lot of absorbing material in the walls due to the difference between colours. The reflections of the walls are plotted in cold colours. The authors noticed also that the wave front at low frequencies is large than at high frequencies (Figure 11).

Figure 9. Direct sound in the 500 Hz band [16]

Figure 10. Direct sound in the 4000 Hz band [16]

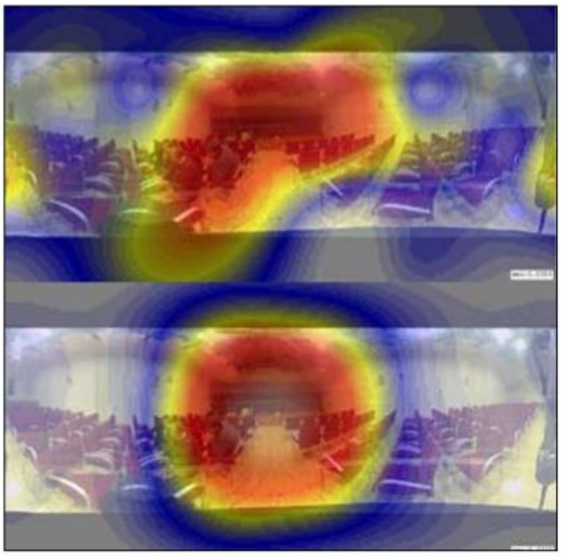


Figure 11. Maps of direct sound at 500 Hz and at 4000 Hz

The analysis shows that the direct sound has a lot of low frequencies due to the diffraction effect, which is not present at higher frequencies.

The method is able to place sound coming from any direction. Therefore, the experiments in the different concert hall were successful in the three concert halls [10]. It allows plotting the dynamic view of the impulse responses on a panoramic picture (360ºx180º) and it provides better spatial resolution than with sum and delay beamforming.

Depending on the accuracy requirements, a configuration is more suitable than others. In our case, we will continue focusing on spherical surface geometries with the aim to analyse in detail the spherical microphone array which was developed in a previous Master thesis [1]. The spherical form was the most effective one for the purpose of the thesis, i.e. room impulse response measurements.

A narrow main lobe and lower side lobes are characteristics which provide higher performance for our system in order to localise the desired sound source, and to be able to isolate the possible reflections.

2.3 Current project

As we have said before, the aim is to make a testing of the created instrument in the project of Hermine Feron [1]. Therefore, we will use the same sphere microphone array that the previous student developed few years ago. With this developed system, our work focuses on spherical characteristics with the purpose of analyse the performance and obtain new results.

A compromise between different factors is usually needed to design a microphone array. The available system contains 16 omnidirectional microphones disposed, in an almost uniform configuration, in a rigid sphere that has a radius of 10 cm. The author said in her thesis that this antenna has, theoretically, an independent directivity of the frequency and an enough noise gain. It has been shown that this system is able to measure the directional impulse responses in the whole space in a frequency range from 250 Hz to 2700 Hz.

The complete system that we will use in the most measures consists on a sphere with 16 integrated microphones, which is connected to two EDIROL sound cards [12]. 8 of the 16 microphones are connected to each sound card.

Concerning my work, I will make several tests using the spherical array of microphones. Then we treat the recorded information to show different diagrams obtained by different methods which have already been implemented on Matlab [15] in the previous project. With a summary of the results then we can conclude that there are some unusual results but the microphone array works. An automatic rotated table is available and we have made some measure but we do not include in the project results finally.

The data of the experiments were managed to test the performance of the device. Almost every experiments in our project are took place in an anechoic room. Last measures were taken in a reverberant chamber, where it is extremely hard to deal with all effects of reverberant environment.

3. Calibration and first measurements

3.1 Calibration

As we have said, the measurement system, which is going to be used in the whole project, consists of a rigid sphere which has 16 integrated microphones. Each microphone has a sensitivity of -8 dB (tolerance: ±3 dB) and an environment SNR (Signal-to-noise ratio) of 60 dB. This sphere, which is placed on a manageable support, is connected with two soundcards. These cards, which have a sampling frequency of 48 kHz, are connected with a PC where we have installed the desired software. The input signal is played by a loudspeaker which is always situated at the same height as the centre of the sphere.

The sound is received at each microphone with different intensity, and then all the measurements (in a WAVE format) have been recorded by Adobe Audition software [11], where it is shown the intensity of different signals, related to each microphone. Adobe Audition is used in order to play and/or record the desired signal. Furthermore, it is useful to verify the proper operation of each microphone and also the received intensity. Then, the signal processing is made by Matlab software [15].

As we have said, we have two sound cards (Edirol FA-101 FireWire Audio Capture [12]) with 8 input in each one (Figure 14). A single card have also 2 input sensitivity knob to adjust the input level of inputs 1 and 2 (Figure 12 and Figure 15), and also another input sensitivity knob which adjust the input level of input 7 and 8 at the same time (but not in the same way). In the following paragraphs this step will be explained.

Figure 12. Edirol FA-101 Fire Wire Audio Capture [13]

The first point before starting the desired measures is making the calibration of the cards which control the microphones and the loudspeaker’s signal. In order to calibrate the system, we have used a single microphone instead of the sphere to verify that each input signal of both cards have the same influence, this is, to obtain the same amplitude in each output for the same input signal (which is a sinusoidal wave at 250 Hz, 1 kHz and 4 kHz). The frequencies of sine waves were chosen considering the theoretical frequency range corresponding to the geometrical structure of our microphone array [1].

These measures have been taken in an anechoic room, where the noise level is minimal and the direction of the sound is already known a priori.

We have calibrated the soundcards as follows:

1. Play the same signal with the loudspeaker and capture this sound signal with the same single microphone. We use Adobe Audition to record each signal, which are saved in ".wav" and ".pkf" format. This software has the following appearance (Figure 13):

Figure 13. Appearance of Adobe Audition software

2. Connect this microphone successively to each of the 16 inputs of the soundcards (Figure 12 and Figure 14).

Figure 14. Back of the Edirol FA-101 FIreWire Audio Capture [14]

3. Fix the sensitivity knobs of input 1 and 2 (Figure 15) due to that two inputs have a different behaviour. Therefore, we will turn the knobs with the aim of obtaining the same intensity level as the ordinary inputs.

Figure 15. Front of the soundcards

4. Fix the sensitivity (unique) knob of inputs 7 and 8 (Figure 14). Doing this, we realized that it is not possible to change the sensitivity of inputs 7 and 8 independently. Consequently, we adjust the sensitivity knob to obtain the same amplitude in the 7^{th} input and we must apply a correction to equalise the level of 8^{th} input.
5. Measure for each input the amplitude of the signal on the graph (Figure 12) displayed by Adobe Audition. We can see the wave form and the lecture of signals is in dB, we transform in RMS (Root mean square) value to obtain the results in Table 1.

In the end of the calibration procedure we have got 16 measures per each frequency. We have calculated the RMS **¡Error! No se encuentra el origen de la referencia.**) values, at different frequencies, belonging to the sampled output data of every microphone in order to determine if all the inputs jacks give us the same value when the same input is applied. The parameter xi (i=0, 1…, n, where n is the length of signal x) is the recorded value by Adobe Audition which corresponds to relative voltage amplitudes. We can see that all the values obtained are quite similar except for the input 8 in each sound card (corresponding with microphone 3 and 8). For this reason, a correction will be applied in the further results as a calibration parameter in some Matlab functions which were used by Hermine Feron in her project.

Micro	RMS value at 250 Hz (10^{-3})	RMS value at 250 Hz (dB)	RMS value at 1 kHz (10^{-3})	RMS value at 1 kHz (dB)	RMS value at 4 kHz (10^{-3})	RMS value at 4 kHz (dB)
1	0.653	-63.7	0.716	-62.9	0.468	-66.6
2	0.661	-63.6	0.724	-62.8	0.468	-66.6
3	0.767	-62.3	0.841	-61.5	0.543	-65.3
4	0.661	-63.6	0.716	-62.9	0.473	-66.5
5	0.676	-63.4	0.733	-62.7	0.479	-66.4
6	0.646	-63.8	0.708	-63.0	0.462	-66.7
7	0.653	-63.7	0.724	-62.8	0.468	-66.6
8	0.759	-62.4	0.832	-61.6	0.543	-65.3
9	0.661	-63.6	0.724	-62.8	0.473	-66.5
10	0.653	-63.7	0.716	-62.9	0.468	-66.6
11	0.661	-63.6	0.724	-62.8	0.473	-66.5
12	0.653	-63.7	0.708	-63.0	0.462	-66.7
13	0.653	-63.7	0.716	-62.9	0.468	-66.6
14	0.661	-63.6	0.724	-62.8	0.473	-66.5
15	0.661	-63.6	0.724	-62.8	0.473	-66.5
16	0.661	-63.6	0.724	-62.8	0.473	-66.5

Table 1. Linear and RMS values for each microphone's output signal

3.2 Directivity measurements

The first measurements with the sphere microphone (Figure 16) would be taken also in an anechoic room, as shown in Figure 23, removing the single microphone and connecting the 16 microphones of the sphere with the sounds cards in the correctly way (Figure 17). We have tried different locations of the sound source and computed the following distribution of the sound pressure (by beamforming): see Figure 19 - Figure 22. We then searched for the maximum intensity and checked if it was in the right position.

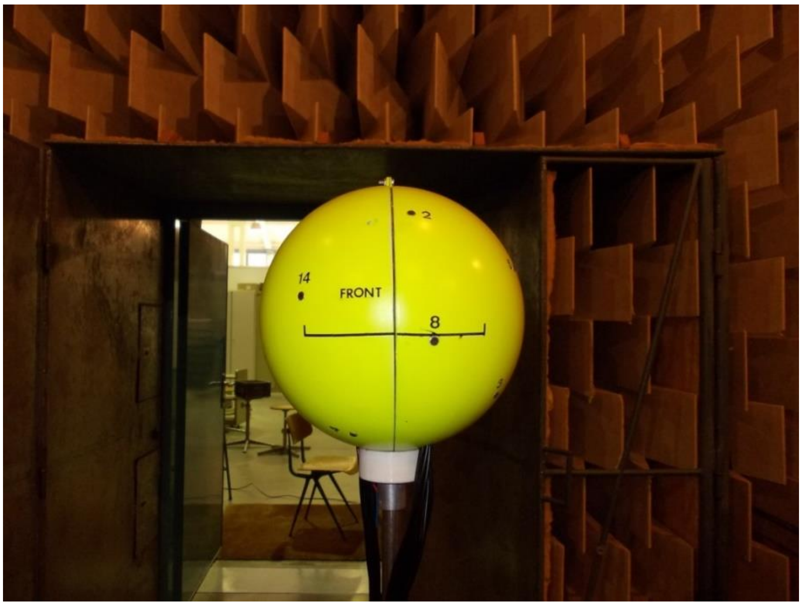


Figure 16. Spherical array microphone

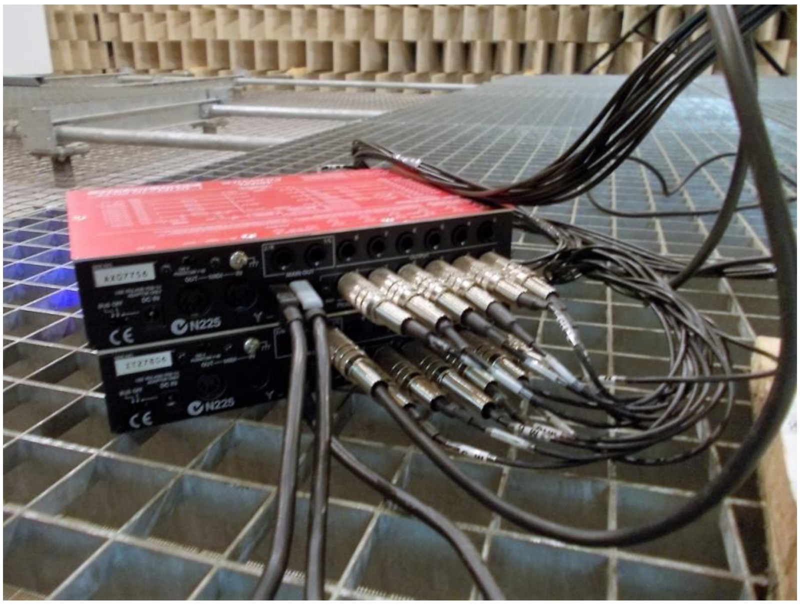


Figure 17. Back setup of the soundcards

With the measurement setup installed in the anechoic room the following cases were carried out at 1623 Hz:

- M0: Sphere and loudspeaker are aligned face to face with a distance of 2.3 meters.
- M1: Sphere and loudspeaker have an angle of 14.6º (Figure 18).
- M2: Sphere and loudspeaker have an angle of -75.4º.
- M3: Sphere and loudspeaker are aligned face to face with a distance of 2.3 meters but with the head turned 10º.

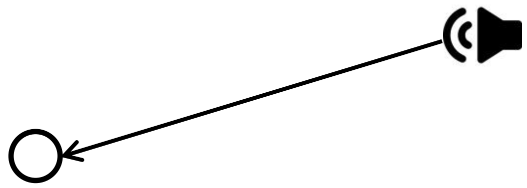


Figure 18. Case M1

What it can be seen in figures is a directivity map where the pressure amplitude is represented in relative units, this is, the colour code attribute 1 value to the maximum result. The darkest red colour represents the maximum pressure level. Related to the axes, the angle of elevation θ is transformed into latitude δ (δ = 90º - θ) and the azimuthal angle ϕ varies from -180º to 180º.

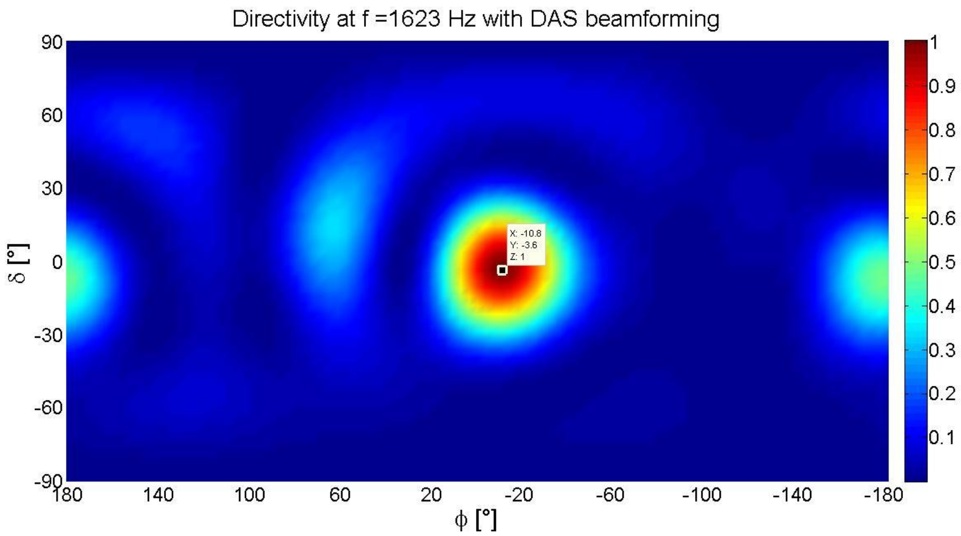


Figure 19. Case M0

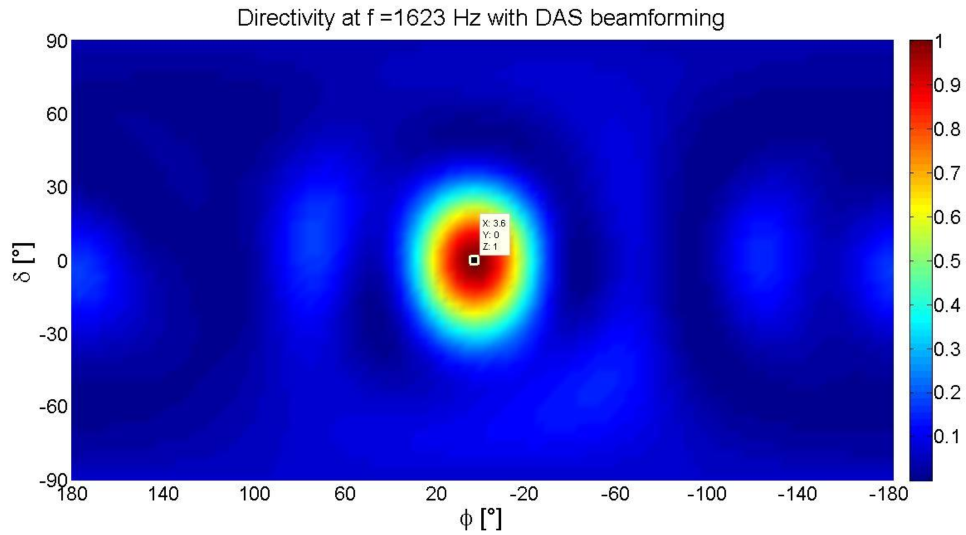


Figure 20. Case M1

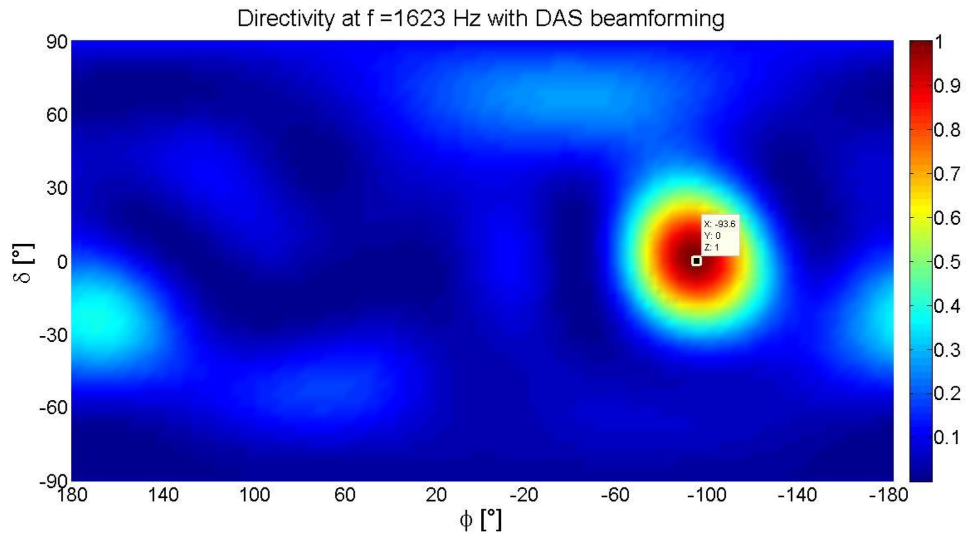


Figure 21. Case M2

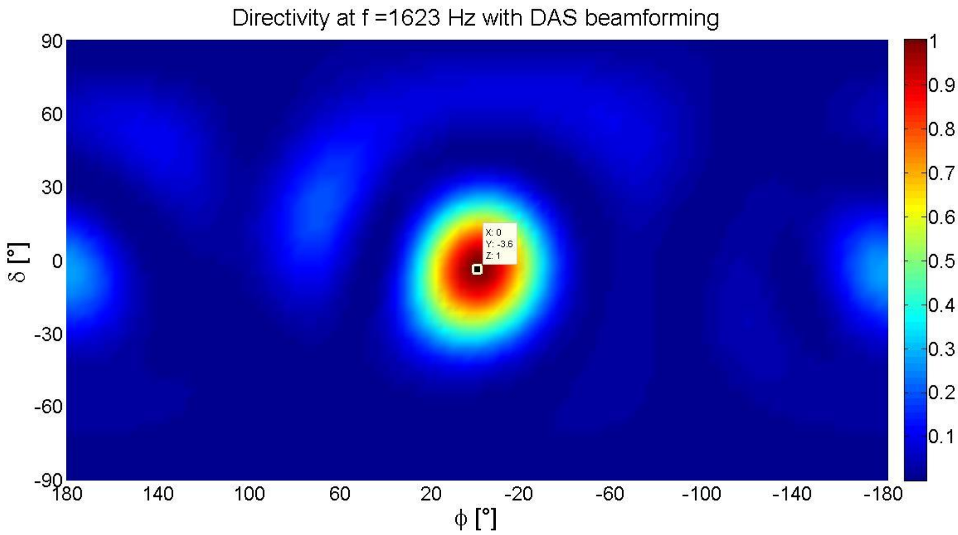


Figure 22. Case M3

After these measurements, which are summarised in Table 2, we could notice a deviation in relation to expected positions, this may be due to human errors, because we do not have any device to align both instruments. It could also appear some errors in positioning on account of real deviation of the microphones' position.

Experimental case	Expected position (δ, ϕ)	Obtained position (δ, ϕ)	Deviation (Δδ, Δϕ)
M0	(0, 0)	(-3.6, -10.8)	(-3.6, -10.8)
M1	(0, 14.6)	(0, 3.6)	(0, -11)
M2	(0, -75.4)	(0, -93.6)	(0, -18.2)
M3	(0, 10)	(-3.6, 0)	(-3.6, -10)

Table 2. Summary of the position of the maximal value

The expected position column in Table 2 shows the approximate direction of the source in elevation and azimuth angles.

The following column shows the obtained position in the represented graphics. The deviation between the expected angles and the obtained results is shown in the last column. It is also important to take into account that in signal processing the values of the pressure level are saved in a matrix, where between each array position of rows and columns have a difference of 3.6 degrees. Thus, this precision needs to bear in mind in order to interpret the results.

3.3 Distance to the source and delay

It is interesting taking measurement of the same input signal but with different distance between the source and the sensor (sphere) because we have seen that there are several seconds in excess in time domain. These results could help us to discard some reasons or to determine the origin of this delay.

It might be due to the sum of the delay of the microphone, the soundcards and the loudspeaker. To be sure that it could be for that reason, we measure the same impulse response but in several distances of the loudspeaker and we will attempted to make a correspondence between the distance of the loudspeaker and the delay.

We have fixed ϕ equal zero and changed the distance between the loudspeaker and the microphone from 4m to 1m each 0.5m.

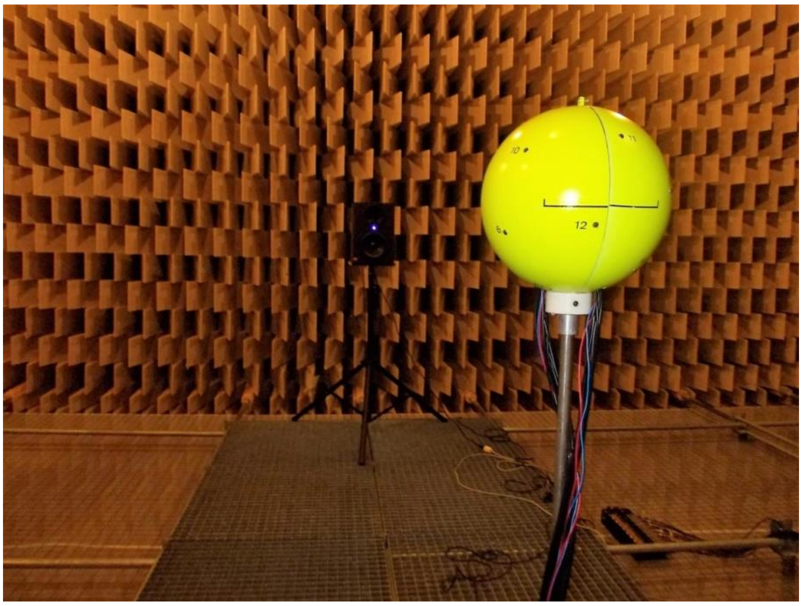


Figure 23. Equipment to make the measurements

Loudspeaker is not a time invariant system, thus, for each measurement we avoid the initial state waiting few seconds before I start the record of the signals. The more you play the loudspeaker, the more increase of the elements' temperature. We want to keep the loudspeaker in the same conditions to avoid different behaviours.

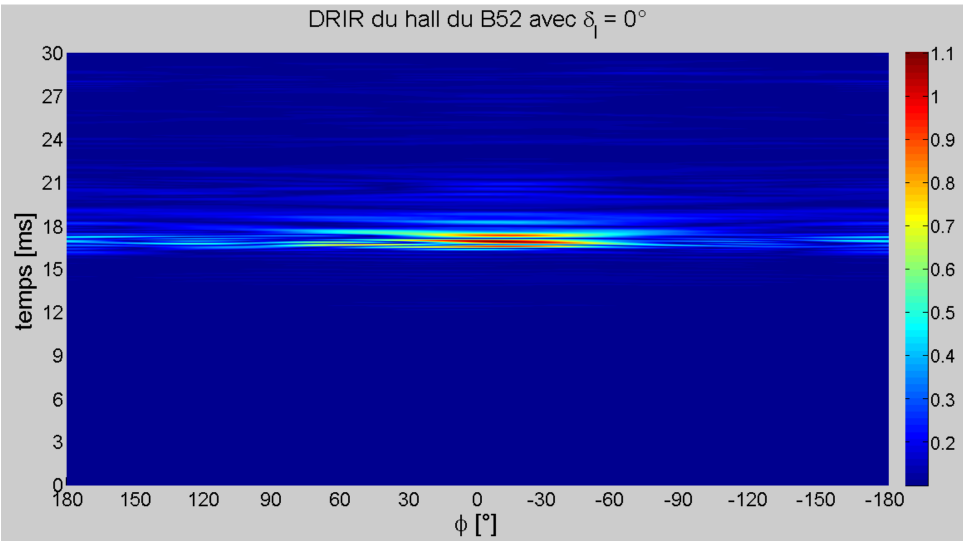


Figure 24. DRIR with δ=0° at 4m to the loudspeaker

The goal was to check that firsts impulse occur at a specific number of milliseconds (approximately 16ms in Figure 24), which was not correspond to the distance (delay · c = 16 ms · 343 m/s = 5.5 m), therefore, we want to show if there is a linear relation between the first impulse in the direct sound and the distance between the source and the microphone.

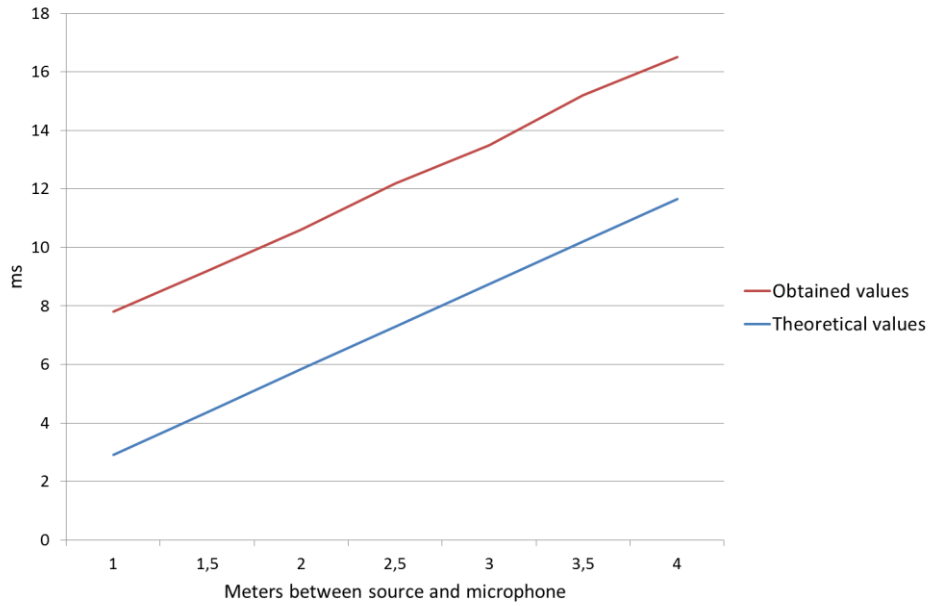


Figure 25. Delay related to the distance

After computing each result and as we can show in Figure 25, for each measure the system has the same delay, 4.9 ms, this is approximately 1.7 m. Fortunately, there is a linear correspondence between the distance to the loudspeaker and the delay of the signal.

3.4 With a panel

The next measurement consists in including a rectangular reflecting panel in the anechoic room whose length is around 1 metre. The Figure 26 shows the location of the panel relation to the measure equipment. The panel is located in parallel with an angle of 56.3˚ related to the sphere (Figure 27).

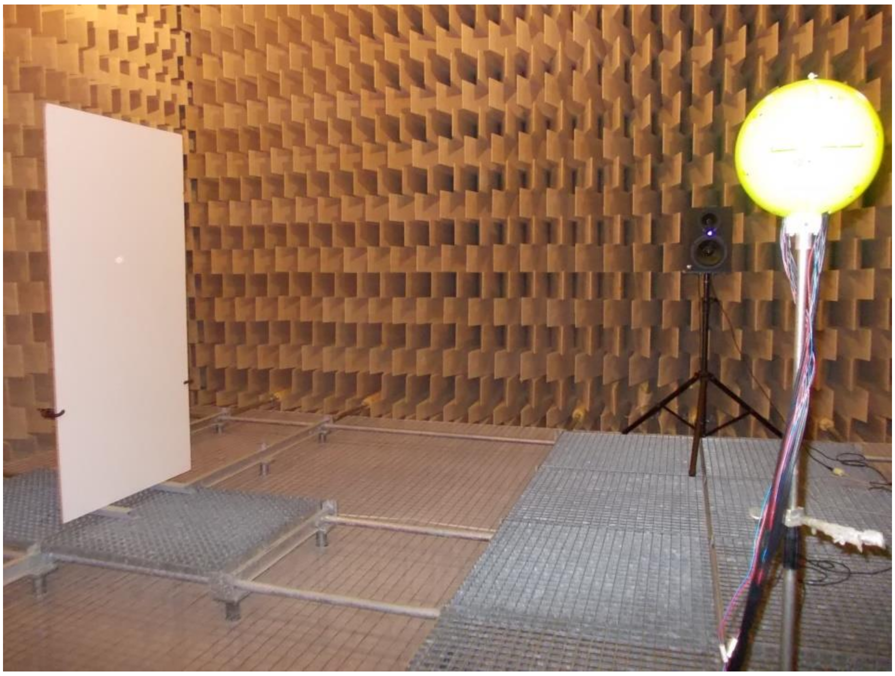


Figure 26. Measurement setup with a panel

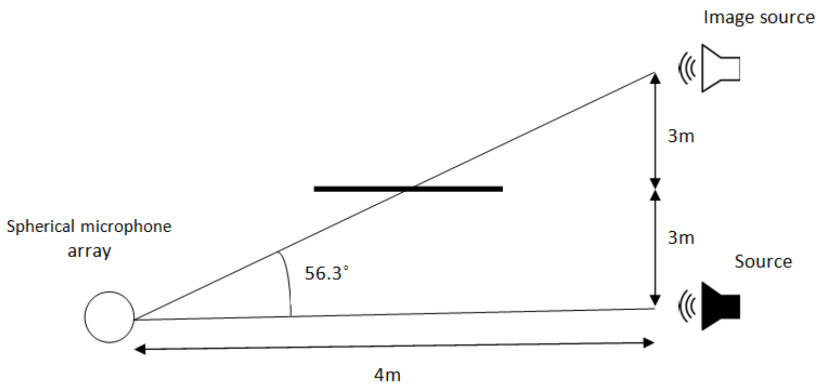


Figure 27. Angle of image source

The measurements have been taken using three sinusoidal signals of 400 Hz, 1623 Hz and 2500 Hz respectively. The information has been processed in order to obtain the directivity relating to the azimuthal angle using three methods, DAS (Delay and Sum), MVDR (Minimum Variance Distortionless Response) and PWD (Plane Wave Decomposition). These results are indicated in Table 4 and Figure 31 to Figure 33.

We also are interested in playing a “logarithmic sweep out” (a logarithmic chirp which frequency varies between 250 Hz and 4000 Hz) in the same conditions. As we can see in the Figure 28, the signal is composed of three sweeps separated by a silence second. This kind of signal allows us to process the data in order to obtain impulse responses.

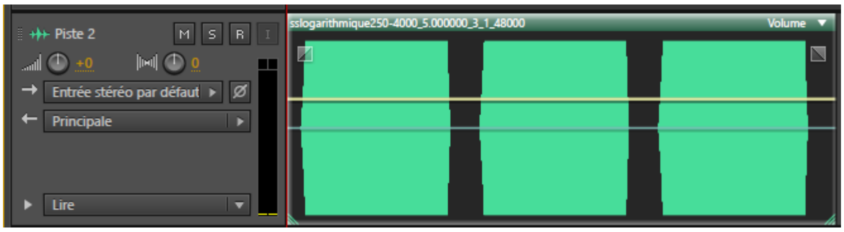


Figure 28. Three logarithmic sweep played by Adobe Audition

In the previous work [1] DAS beamforming was chosen in order to compute DRIRs. Figure 29 shows the DRIR in the horizontal plane for the previous logarithmic sweep using a time window of 50 ms, where two contributions are illustrated. Once we know the time interval of interest, we compute the directivity with a time window by MVDR method (Figure 30).

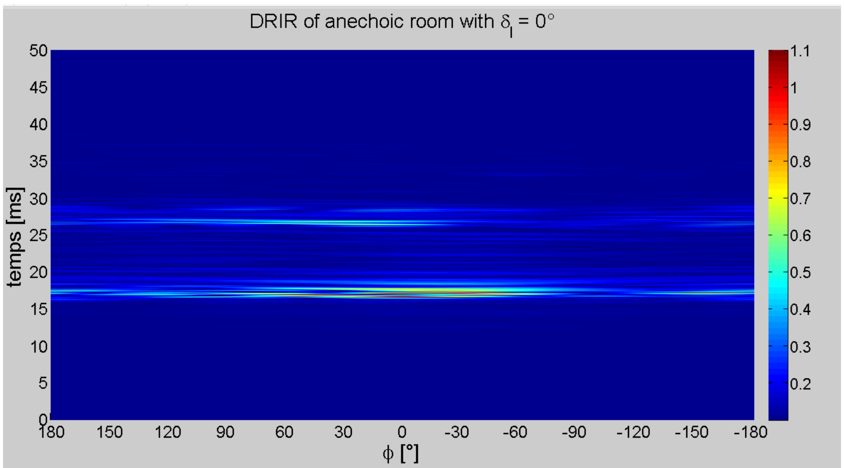


Figure 29. Directional Room Impulse Response in horizontal plane by DAS beamforming

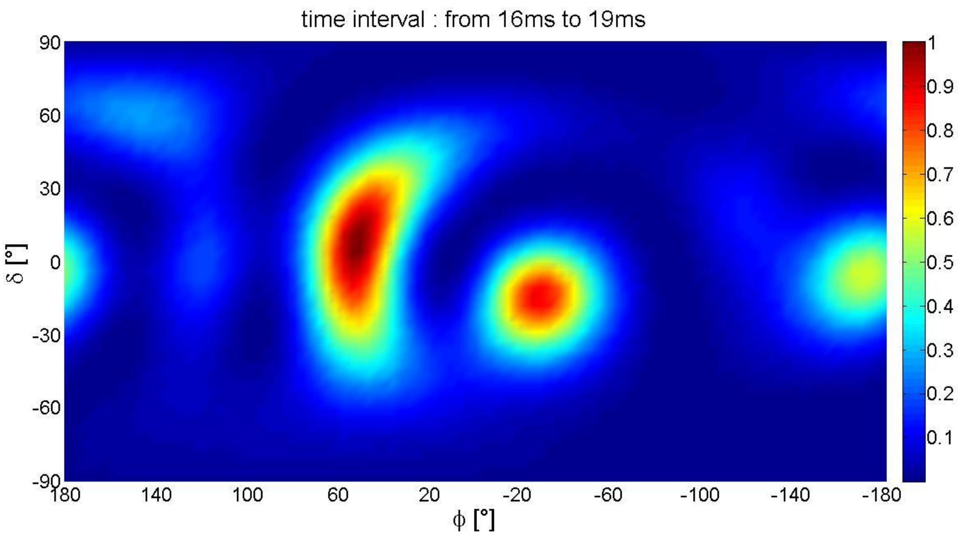


Figure 30. Directivity with time window

On the one hand, analysing all the results we could conclude that the method PWD is not good if we want to obtain this kind of diagram, the provided results are degraded because some incoherent reflections appear in the graph. In the case of DAS method we have the same problem, although to a lesser extent. Therefore, we choose MVDR method to graph these directivity diagrams (with time windowing).

Relating to the frequencies, the results obtained playing a sinusoidal of 2500 Hz are completely deformed. The graphs belonging to 400 Hz are not appropriate for our aim.

MVDR beamforming	Contributions (degrees)
400 Hz	-21.6 y -79.2
1623 Hz	54, 21.6 y 180
sweep (window 16-19 ms)	50.4 y -28.8
sweep (window 25-30 ms)	25.2

Table 3. Contributions

The real angle of the reflection due to the reflector panel is 56.3˚. As we can show, the Table 3 provides up a summary of the obtained results, therefore we could confirm that the system does not work properly at low frequencies as 400 Hz. At 1623 Hz (Figure 35) we have approximately the desired result, despite some reflexions at 180 degrees. If we use a sweep as input signal and then we apply a window function, focusing on the time window corresponding to the desired signal, the result is also quite acceptable.

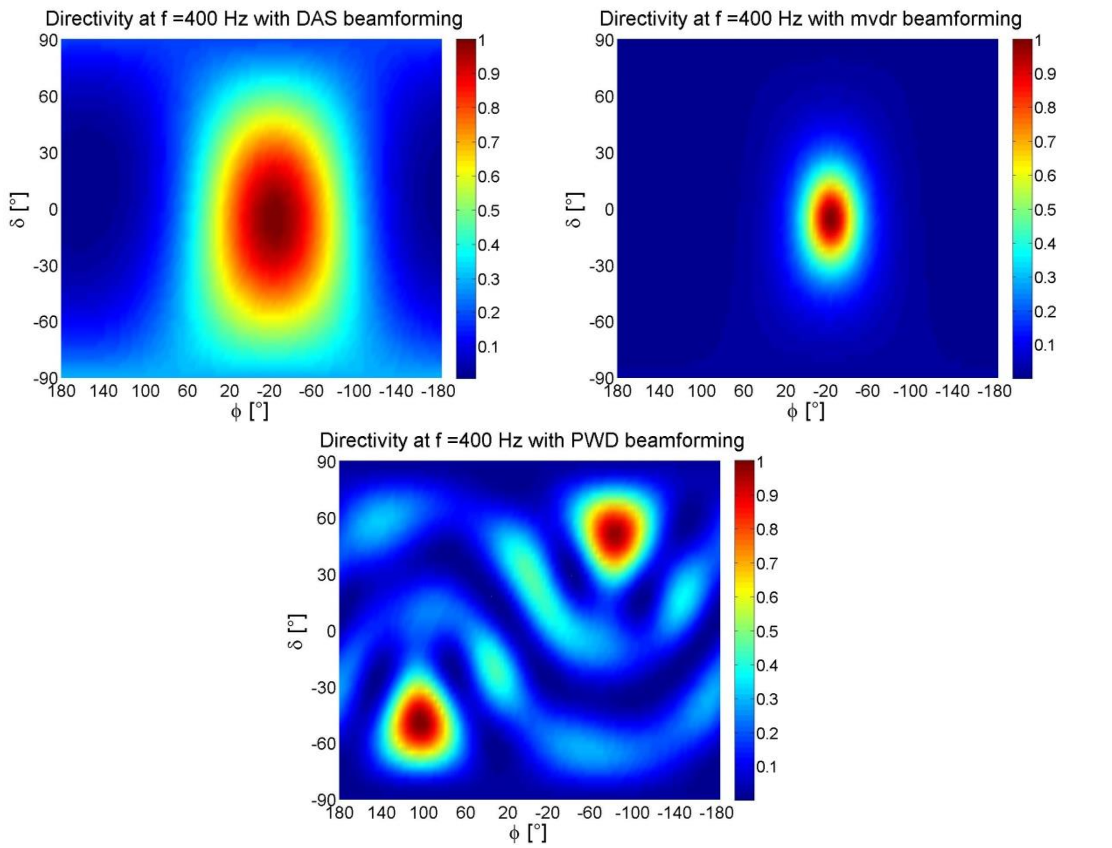


Figure 31. Results of a reflection at (0˚, 56.3˚) at 400 Hz with different methods

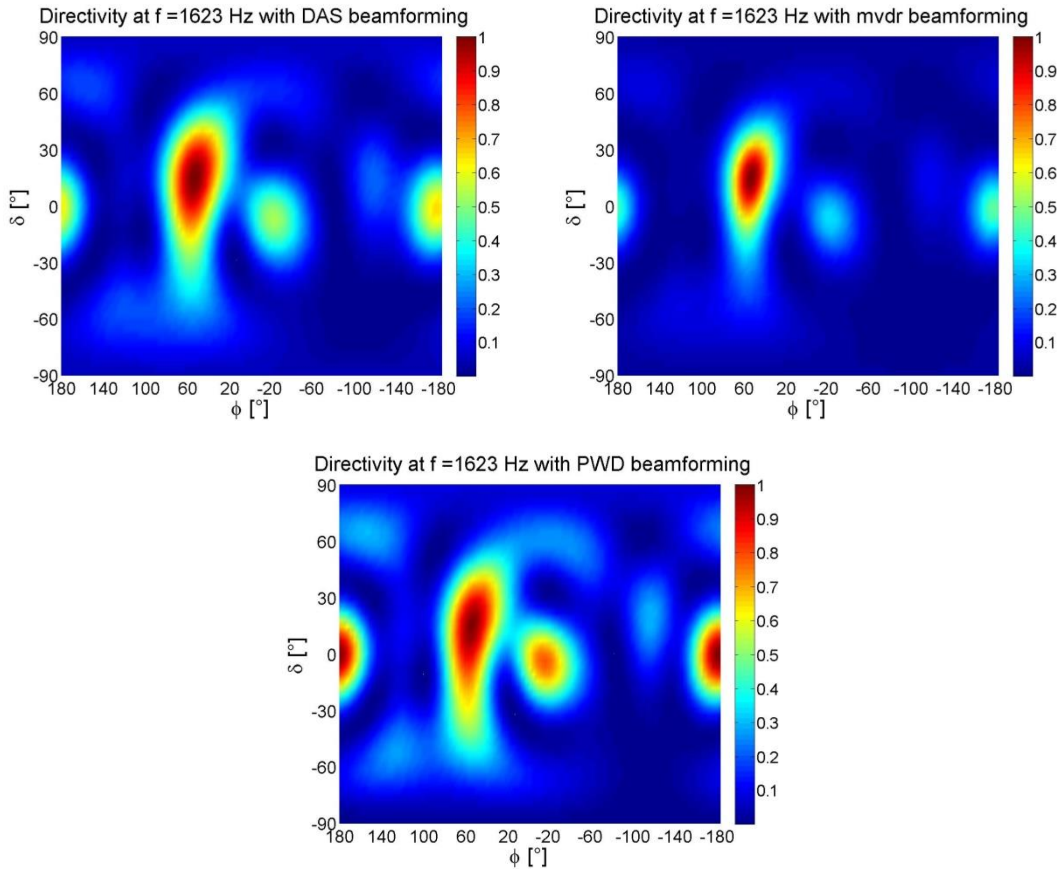


Figure 32. Results of a reflection at (0˚, 56.3˚) at 1623 Hz with different methods

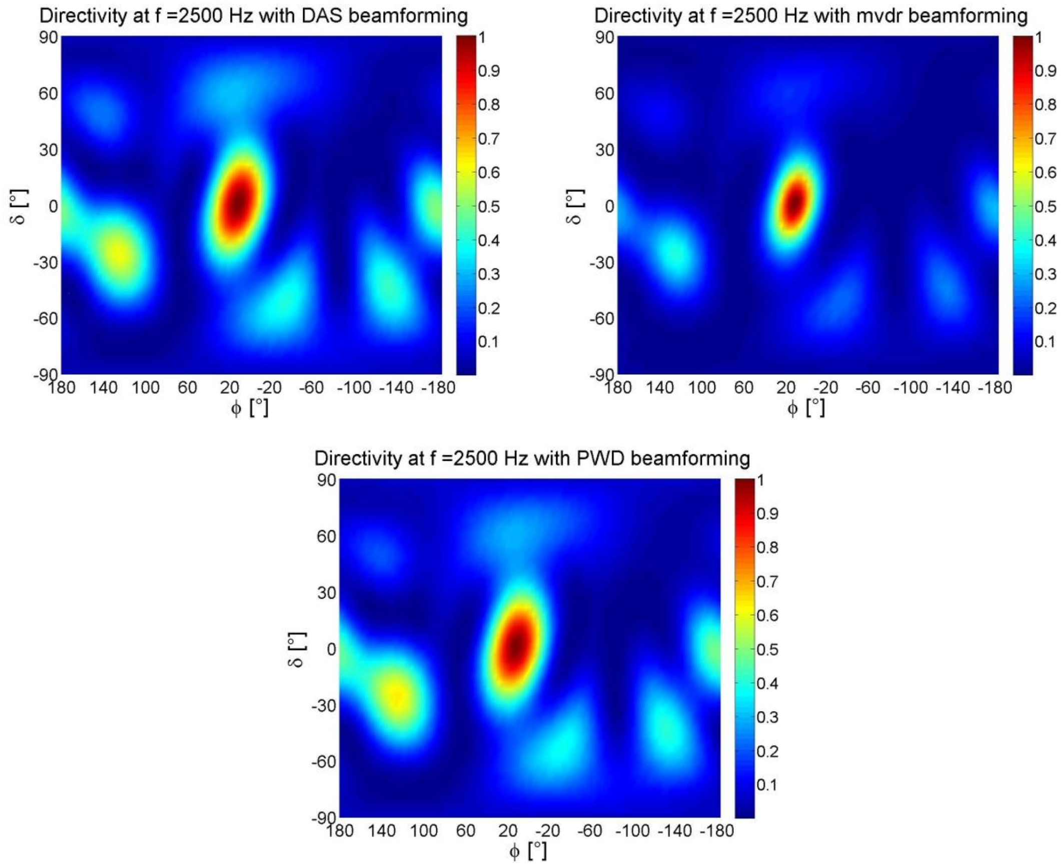


Figure 33. Results of a reflection at (0˚, 56.3˚) at 2500 Hz with different methods

Frequency (Hz)	Method	Desired position (δ, ϕ)	Obtained position (δ, ϕ)	Deviation (Δδ, Δϕ)
400	DAS	(0, 56.3)	(-3.6, -21.6)	(-3.6, -84.3)
	MVDR	(0, 56.3)	(-3.6, -21.6)	(-3.6, -84.3)
1623	DAS	(0, 56.3)	(14.4, 54) (-7.2, -21.6)	(14.4, -8.7) (-7.2, -84.3)
	MVDR	(0, 56.3)	(14.4, 54) (-7.2, -21.6)	(14.4, -8.7) (-7.2, -84.3)
2500	DAS	(0, 56.3)	(3.6, 10.8) (-25.2, 126)	(3.6, -51.9) (-25.2, 63.3)
	MVDR	(0, 56.3)	(3.6, 10.8) (-25.2, 126)	(3.6, -51.9) (-25.2, 63.3)

Table 4. Results with the panel reflector for each method and frequency

3.5 Main conclusions

We have to take into account that the measurement of the location of the sphere related to the loudspeaker is not always as precise as we would like, since we do not use any equipment capable of providing us that desired precision. Both equipment are aligned each time manually, therefore some human errors could appear in the results.

The size of the highest value region should be as small as possible to estimate the arrival sound direction accurately.

We can notice from the figures that in general PWD provides bad results, therefore, even though we have computed most of the result with the three methods, from here on out, we are not going to show the results of PWD because they do not offer us useful information. In contrast, we have seen good results for MVDR beamforming.

Related to the time domain, we have seen that first impulse arrived some milliseconds after that we expected to, but the distance between the source and the receiver is linearly related with the delay.

4. Directivity measurements in far field

In order to proceed, two types of measurements were performed in the anechoic room: in the first one we use pure tones and in the second one we treat with impulse responses (sweep). In both cases a mechanic rotating table is employed.

A pure tone is represented by a sine wave (single frequency), whereas the sweep is described by a range of frequency, in this case this range varies between 250 Hz and 4000 Hz.

4.1 Pure tones in the horizontal plane

After introducing the reflecting panel, we have measured same signals without reflections to compare the applied methods. From Figure 34 to Figure 36 the directivity graphs that we have obtained for each frequency and each method are shown. We have displayed PWD beamforming in order to confirm its corrupt behaviour. It works adequately at 1623 Hz, which is the optimal frequency for our microphone. We have filled a table with the angles corresponding to the maximal value of these figures. This table is included in the Appendix.

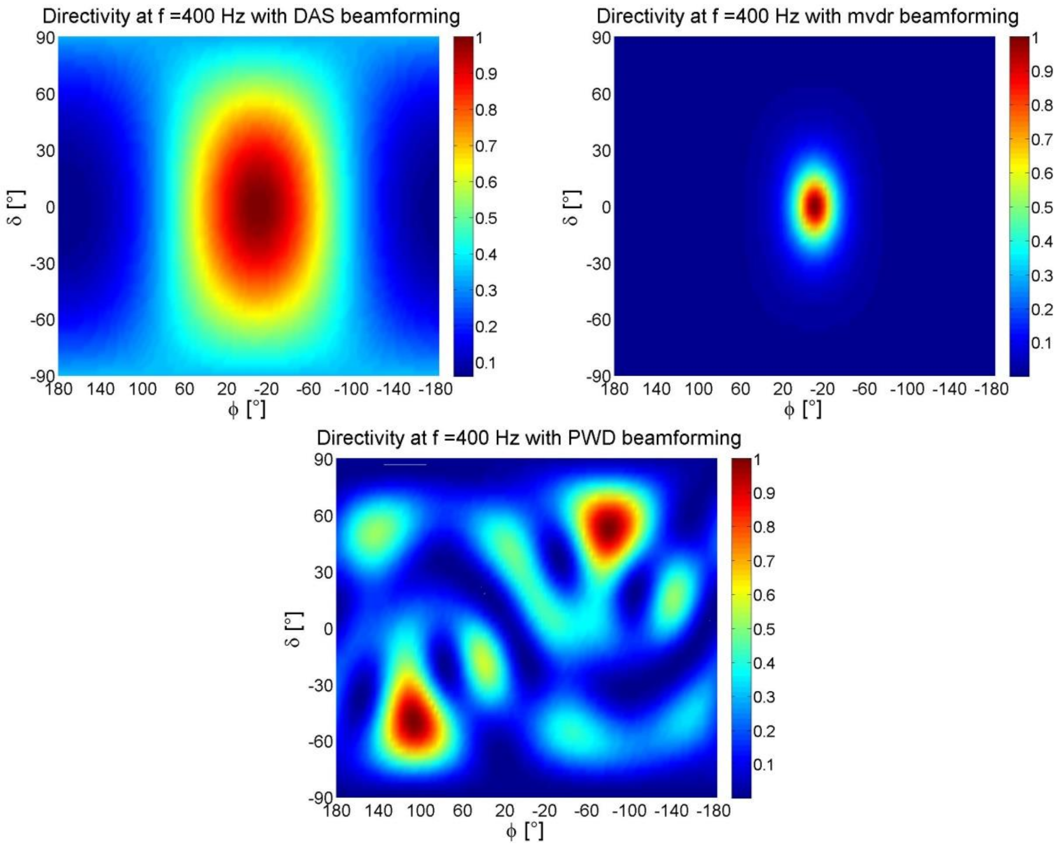


Figure 34. Results at (0°, 0°) without reflector panel at 400 Hz with different methods

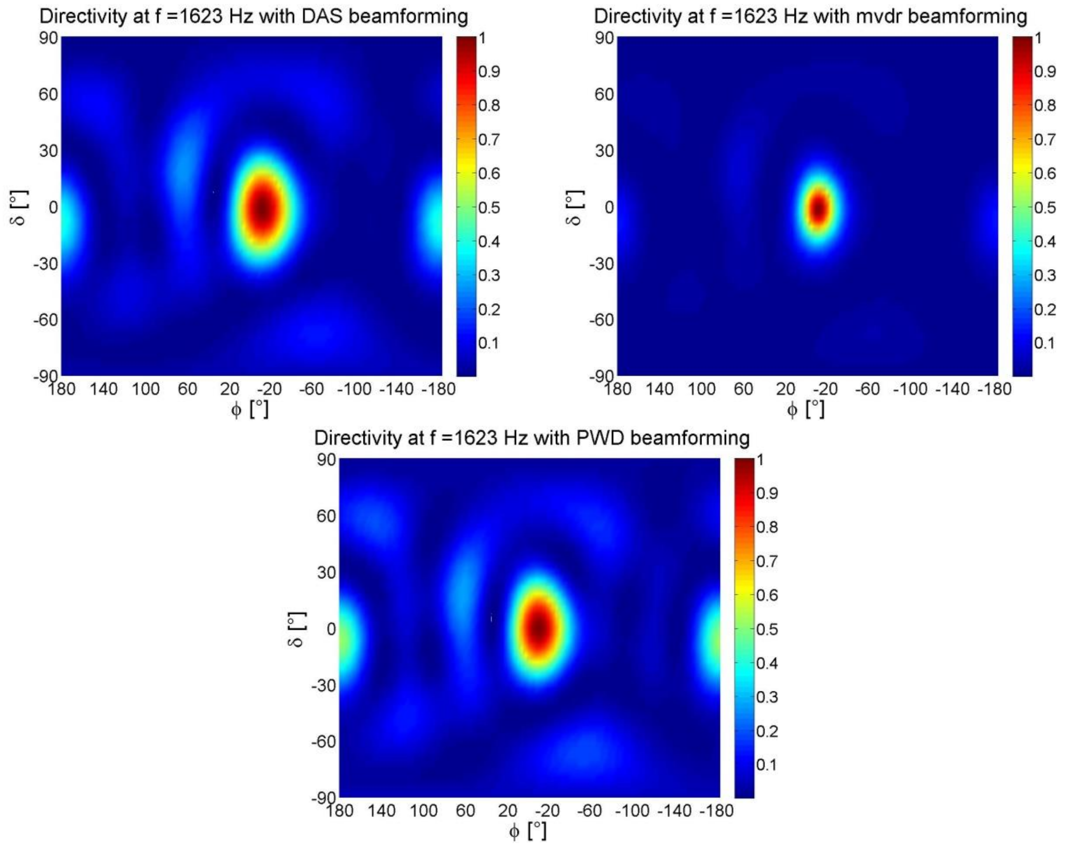


Figure 35. Results at (0°, 0°) without reflector panel at 1623 Hz with different methods

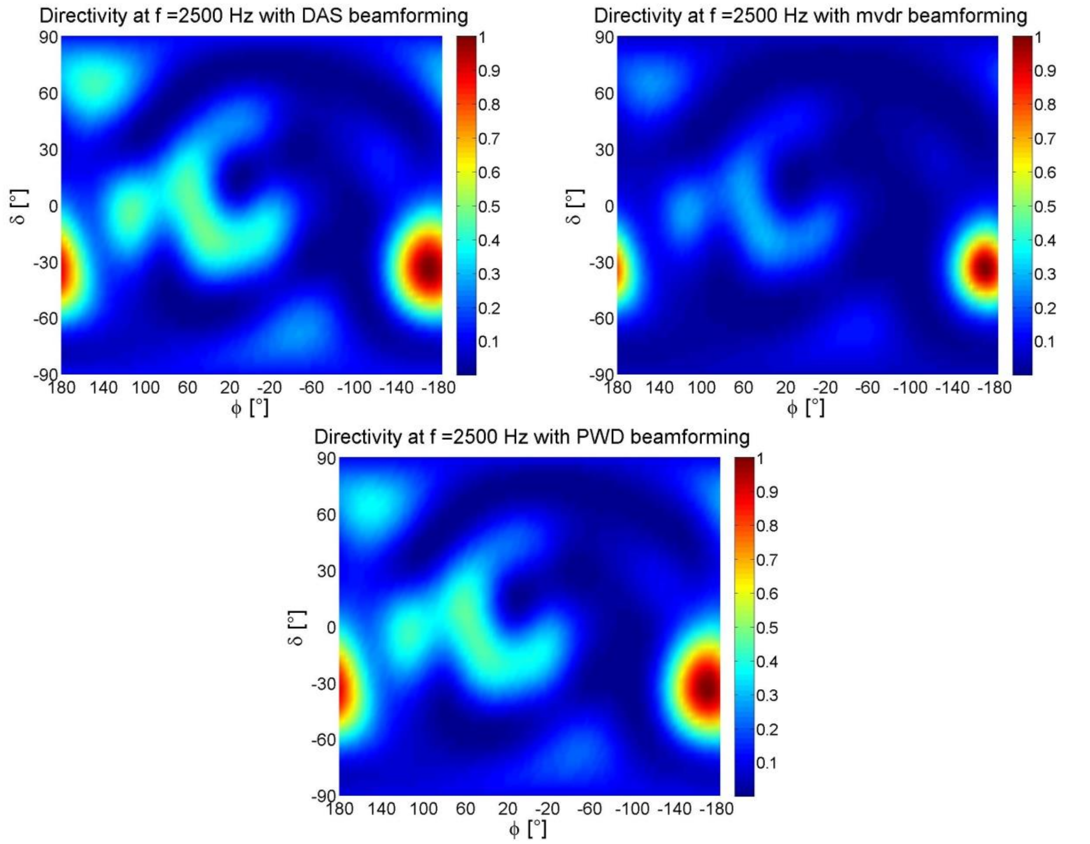


Figure 36. Results at (0°, 0°) without reflector panel at 2500 Hz with different methods

We can say that at high frequencies the deviation between the expected and the obtained angles is too high, just see Figure 36.

It could be interesting measure the sound field turning the head of the microphone each 20˚, this means 18 steps to complete the revolution.

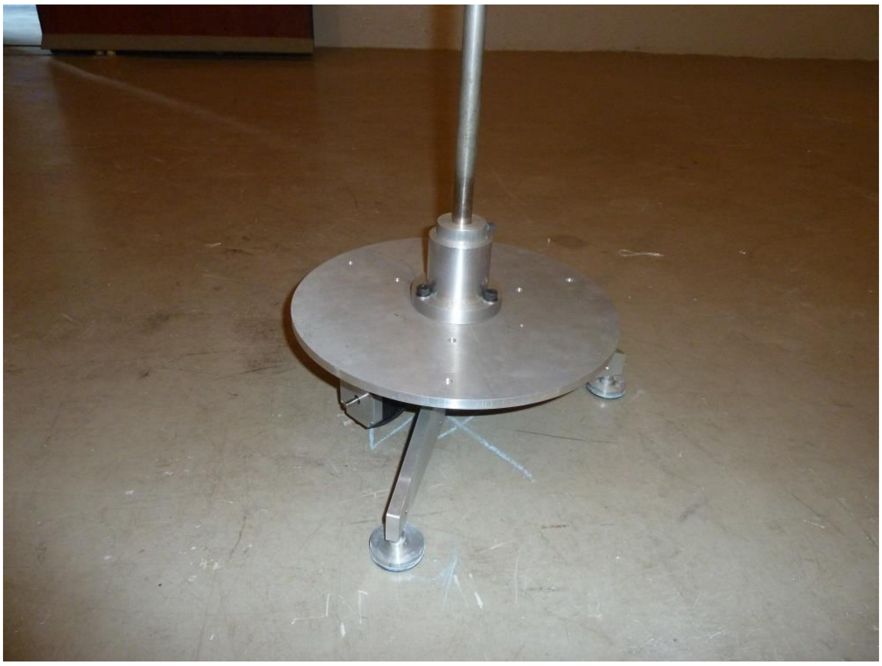


Figure 37. Rotating table

The task that we dealt with is making measurements for different positions of the sound source obtained by placing the microphone array on a rotating table as is shown in Figure 37. These measurements were taken in a smaller anechoic room, due to an issue of availability and to verify the right accuracy in different rooms, with a programmable rotated table. Firstly, we have taken measurements each 20 degrees taking into account the incident angle in order to analyse the directivity diagram. The input signal chosen was a sinusoidal wave of the optimal frequency [1], 1623 Hz. Three methods have been used in order to graph the directivity, DAS, MVDR and PWD.

We decided to measure each several degrees instead of doing a continuous rotation. This implies more time making the measurements but the mechanical noise is avoided. An electronic circuit is used in order to rotate the table automatically by the interface that we can see in Figure 38, where we could fix the speed, the angle and direction of rotation (clockwise or anti-clockwise) etc. It needs to consider a small degree deviation in each rotation.

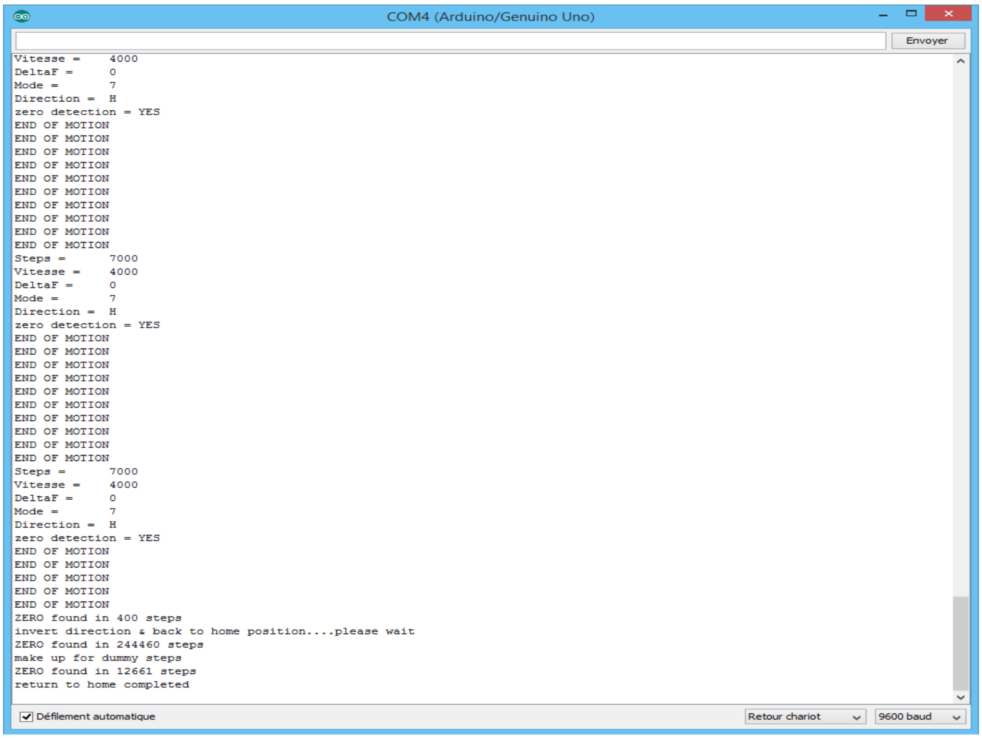
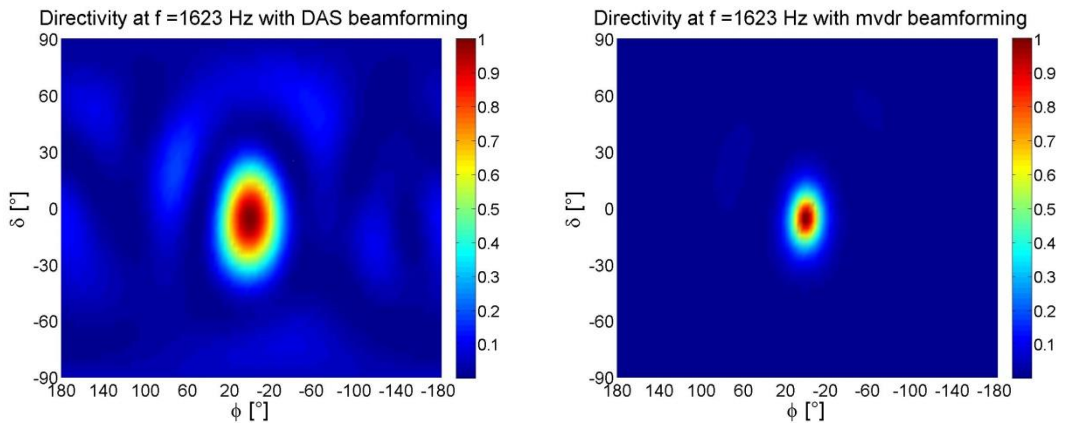


Figure 38. Interface to rotate the table

The following images (Figure 39) show the directivity diagram at 340 degrees (meaning -20˚), that is to say the input signal strikes at 340˚ of the sphere. If we use the PWD method, some reflexions appear in the graph, it is almost the same for DAS method. Consequently we will prefer MVDR as the appropriate method to compute the directivity graph in this way.



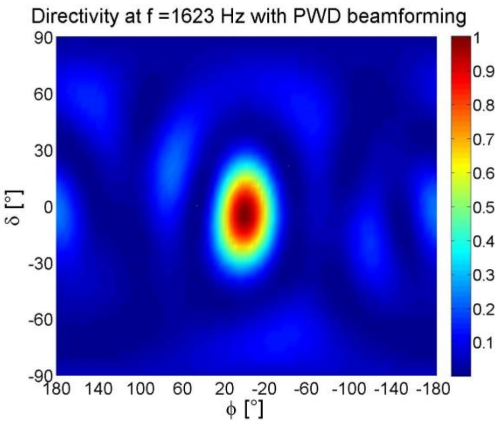


Figure 39. Directivity diagram at -20° and 1623 Hz with DAS, MVDR and PWD methods

We have enough information after measuring each 20 degree to compute a directivity diagram in the horizontal plane depending on the azimuth angle (ϕ) by DAS, MVDR and PWD beamforming.

Figure 40 illustrates an example of the directivity diagram depending on ϕ, where we can show that even in this kind of representation, PWD shows higher secondary lobes than DAS or MVDR. We can also see that the secondary lobe has low level than others, which we also call secondary lobes. Therefore, the idea is to have the maximum value of the secondary lobes.

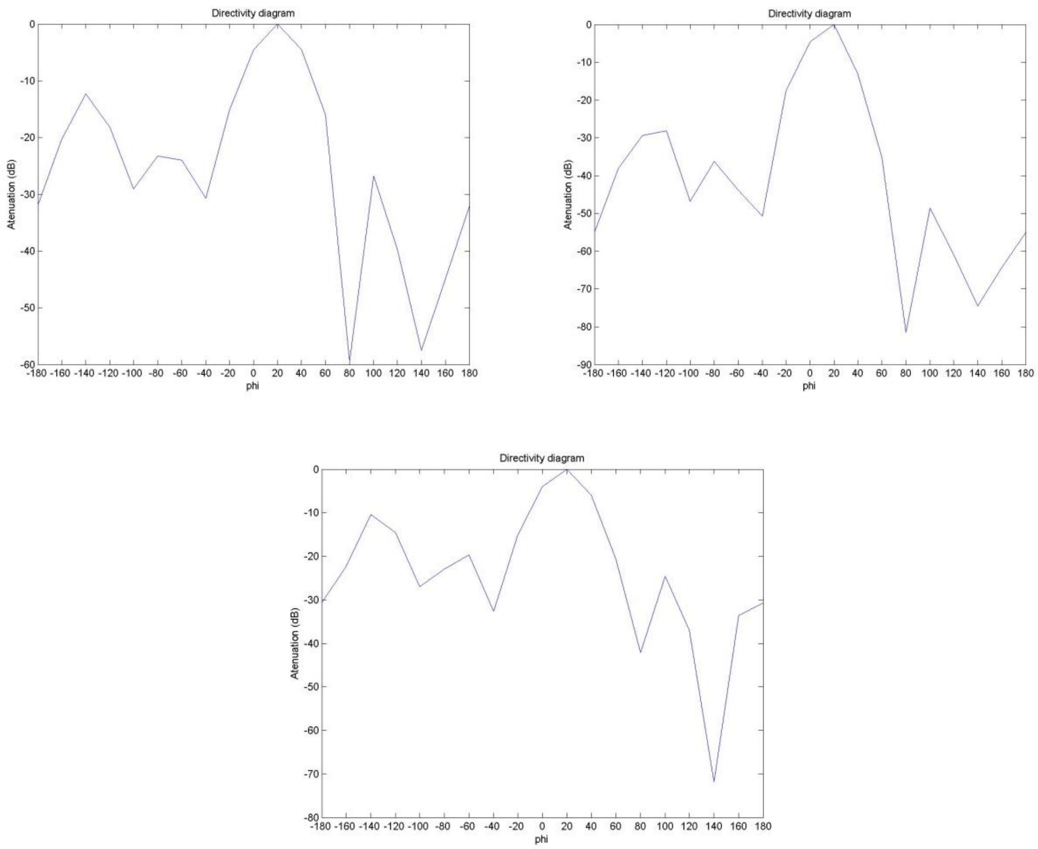


Figure 40. Directivity diagram depending on ϕ

After computing every directivity diagram we have focus on several interesting parameters as the intensity of the secondary lobes, the angle deviation respect to the real one (if it is negative means that the resultant angle is been shifted in an anti-clockwise direction, it should be located |∆| degrees more in a clockwise direction), half-power beamwidth or the beamwidth at -10 dB. The Table 5 shows this information for MVDR beamforming.

Degrees	∆ degree	BW 3 dB	BW 10 dB
0	0	0-20	20-40
20	0	0-20	0-20
40	0	0-20	20-40
60	0	0-20	20-40
80	-20	0-20	20-40
100	-20	0-20	20-40
120	-20	0-20	0-20
140	-20	0-20	20-40
160	-20	0-20	20-40
180	-40	0-20	0-20
200	-40	0-20	20-40
220	-40	0-20	20-40
240	-40	0-20	20-40
260	-20	20-40	40-60
280	-20	0-20	20-40
300	-20	40-60	60-80
320	0	0-20	0-20
340	-20	0-20	20-40

Table 5. Parameters for each 20˚ with MVDR beamforming

We have shown just the MVDR case because the previous results have not got enough information to be able to have a clear conclusion, for this reason, I have measured in the horizontal way again but each 10 degrees. We want to compare with different frequencies, taking into account that the system operates in a frequency range from 250 Hz to 4 kHz. Therefore, we will chose a low frequency (300 Hz) a middle one (1623 Hz) and a high frequency (3000 Hz).

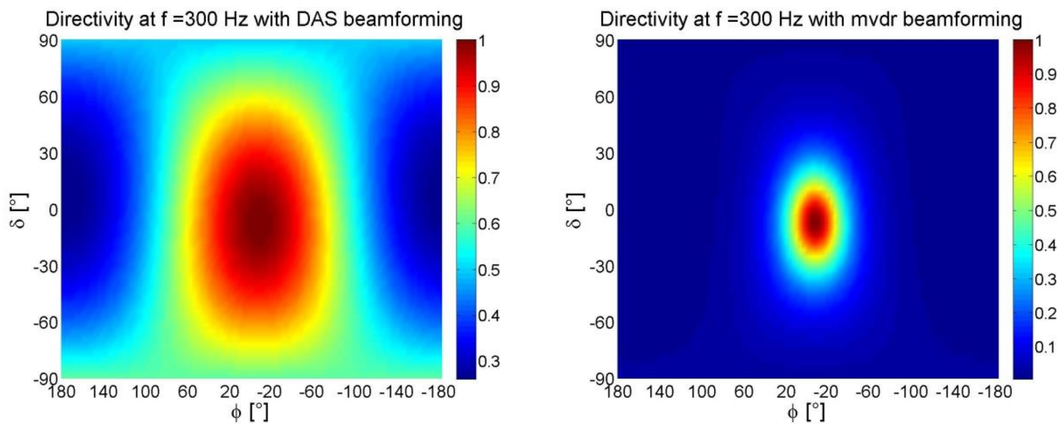


Figure 41. Directivity diagram at 0˚ at 300 Hz with DAS, MVDR and PWD methods

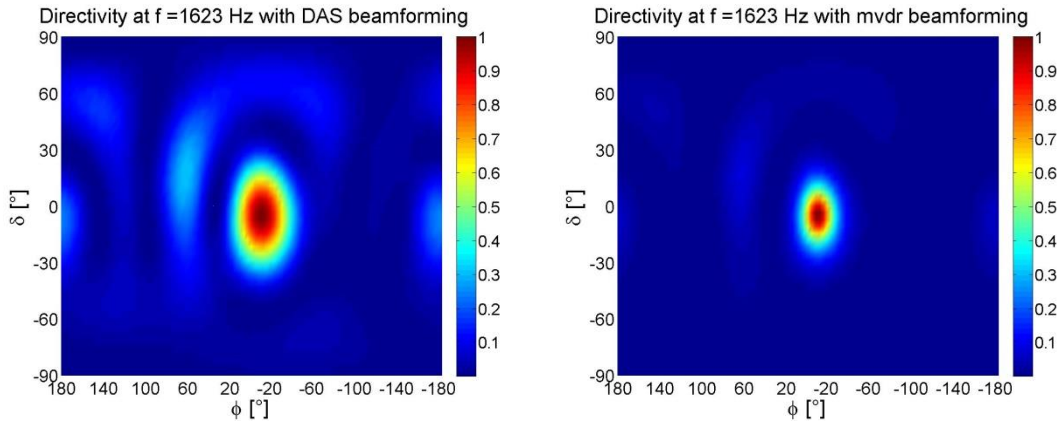


Figure 42. Directivity diagram at 0˚ at 1623 Hz with DAS, MVDR and PWD methods

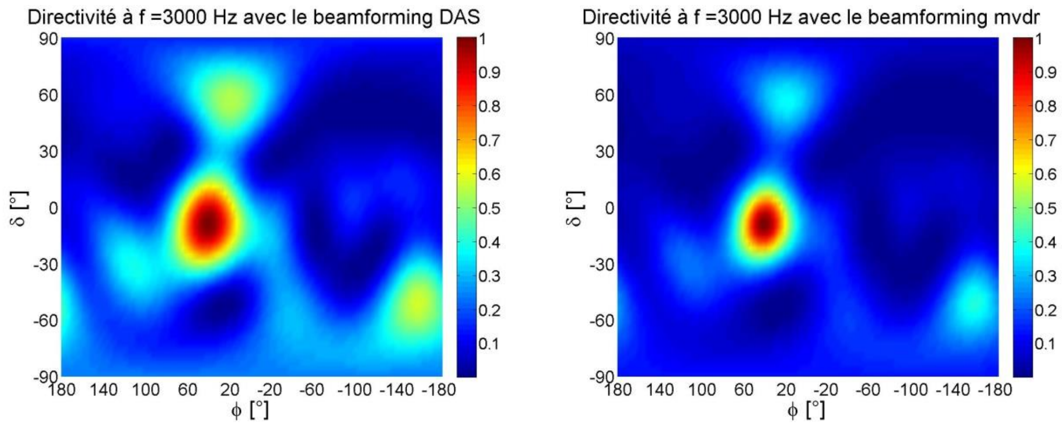


Figure 43. Directivity diagram at 0˚ at 3000 Hz with DAS, MVDR and PWD methods

A summary table has been added to the Appendix with the bandwidth at -3 dB and at -10 dB for each 10˚. For MVDR beamforming we obtain in the majority of cases a bandwidth at -3 dB of 30˚ at 300 Hz, and 120˚ with DAS, this value is unusual but it is what I have obtained. At -10 dB we get 60˚ with MVDR method.

We could show some graphs displaying the directivity diagram related to the ϕ angle in the case when the front of the microphone is looking to the source. See Figure 44: DAS and MVDR consecutively.

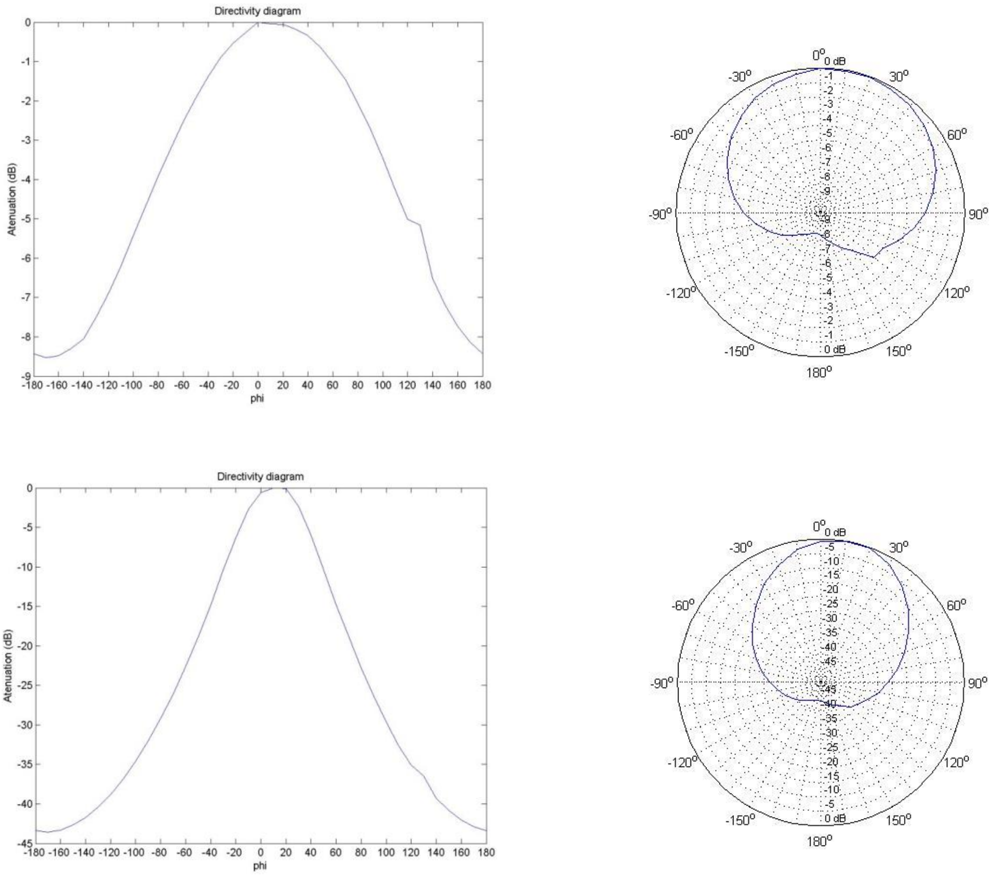
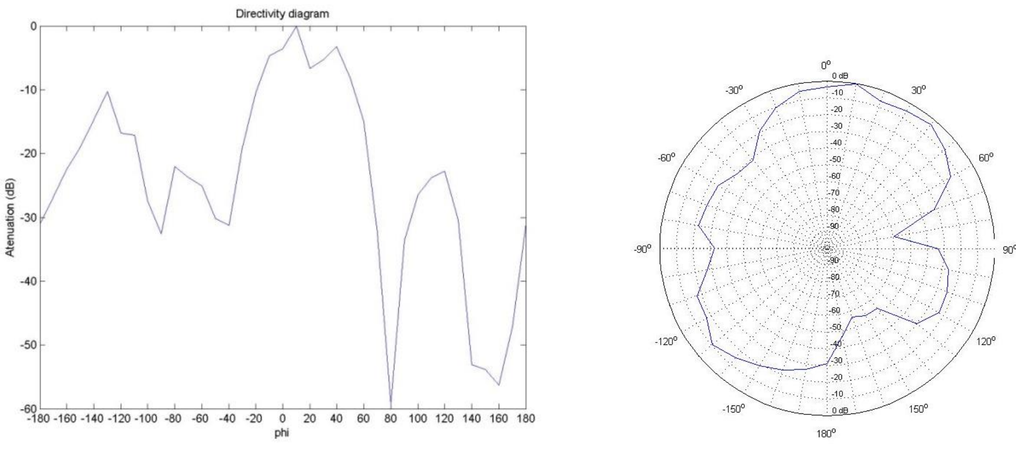


Figure 44. Directivity diagram depending on ϕ for DAS, MVDR at 300 Hz

Intensity of second lobes has been computed for the data obtained at 1623 Hz and 3000 Hz in order to have knowledge of the influence of them. The values of the column "2nd lobe (dB)" correspond to the high lobe regarding the main lobe, regardless of being the nearest one or not. This table is in the Appendix, in Figure 45 and Figure 46 the respective directivity diagrams are shown, for 1623 Hz and 3000 Hz.



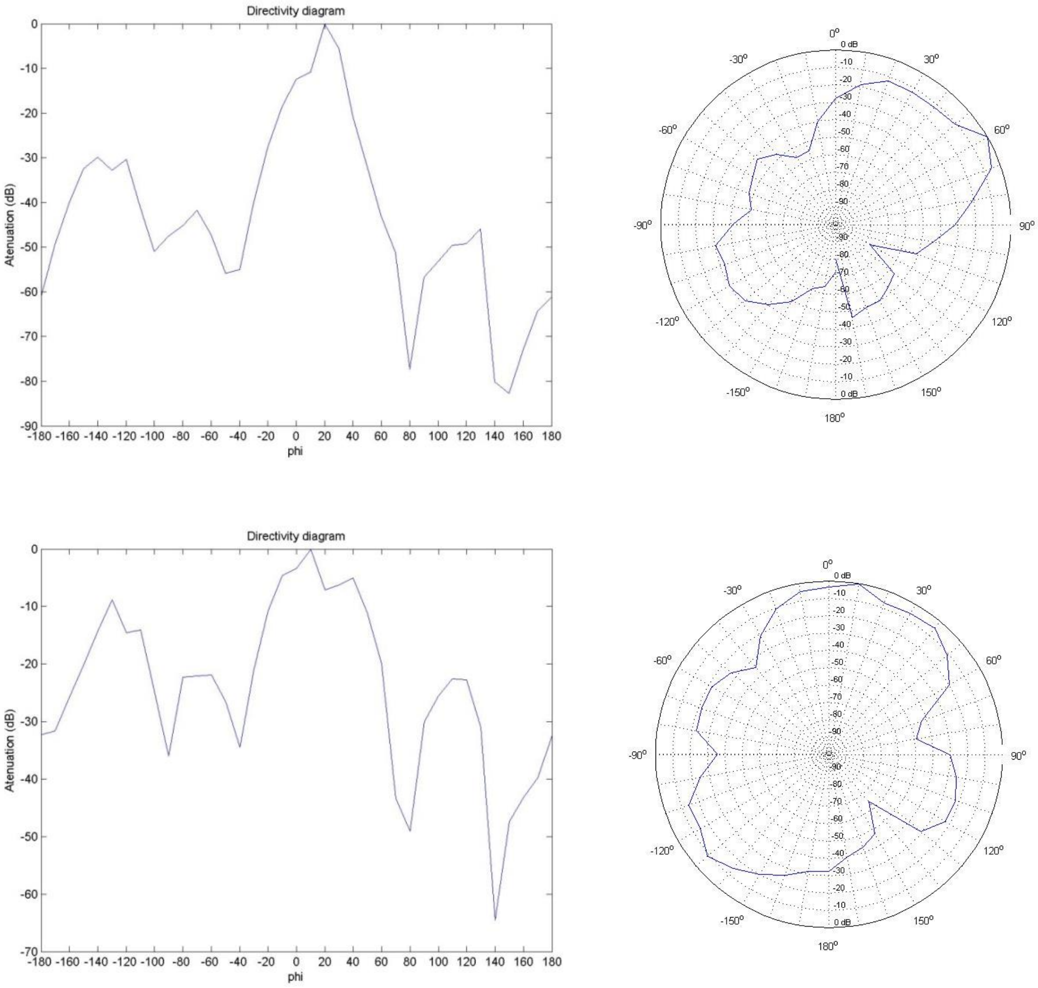
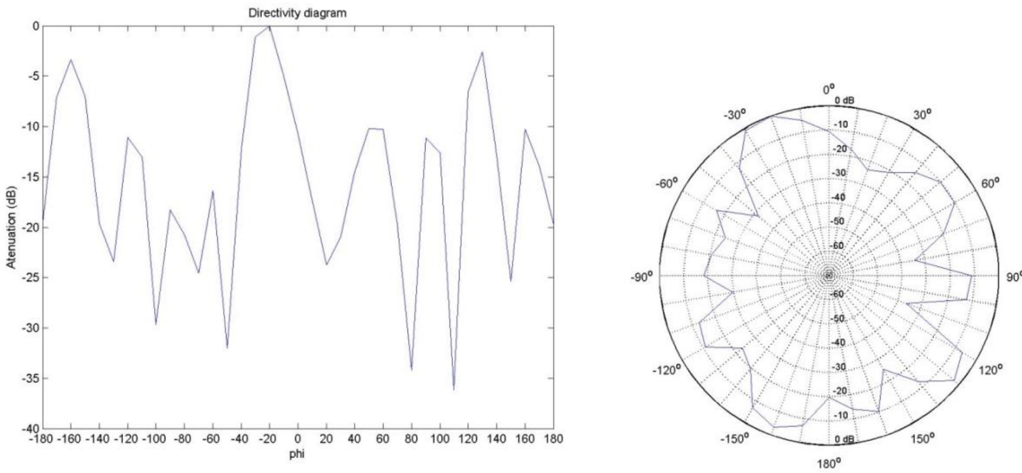


Figure 45. Directivity diagram depending on ϕ for DAS, MVDR at 1623 Hz

It is also noticeable numerically that DAS is not a proper beamformer because the intensity of secondary lobes is too high not to disturb the main lobe.



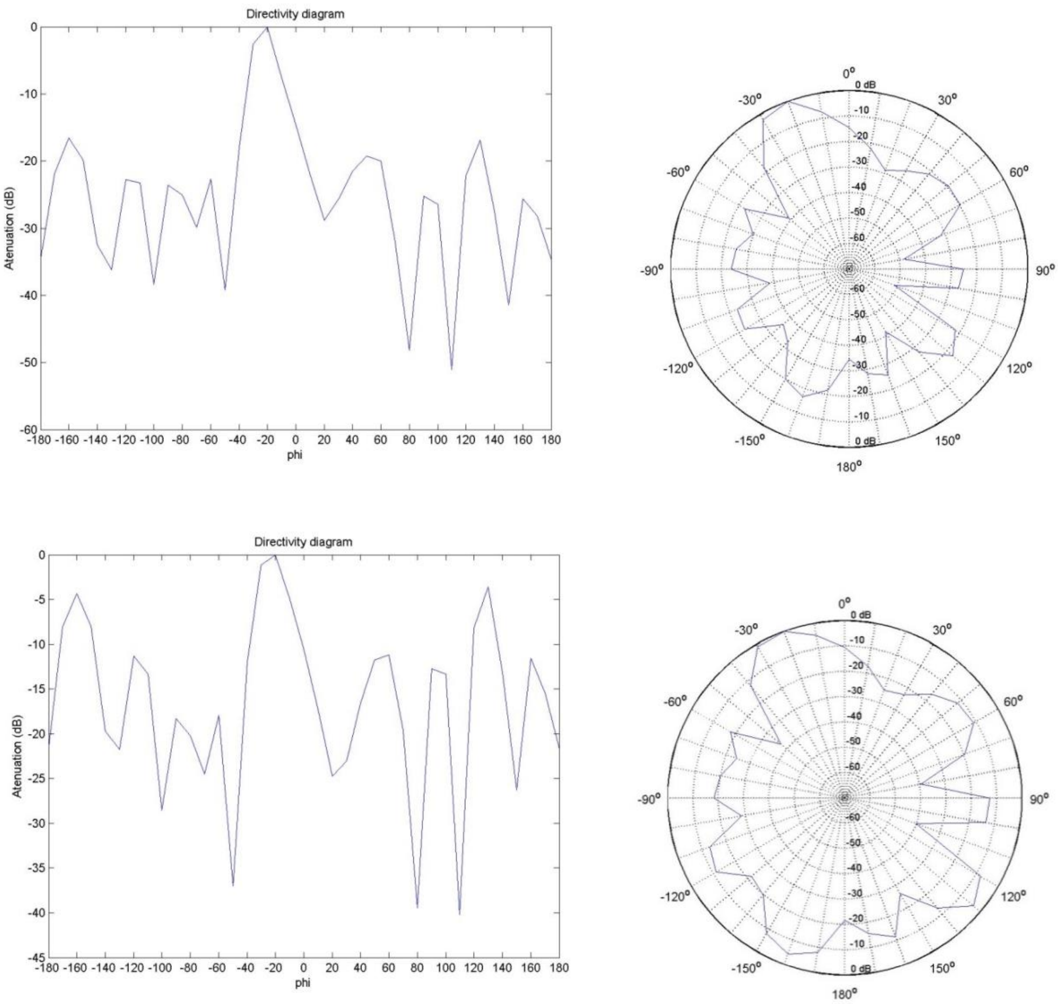


Figure 46. Directivity diagram depending on ϕ for DAS, MVDR at 3000 Hz

4.2 Impulse responses in the horizontal plane

We want to try the same measurements with a logarithmic sweep, which contains a variety of frequencies, and then we will compute the impulse response in the time domain in order to isolate the signal of interest and to compute the directivity applying the pertinent time window.

Several of Impulse Responses were measured at each angular position (a sweep is played). We should concentrate only on the first direct contributions, others could be reflections. The goal is to isolate the direct contribution.

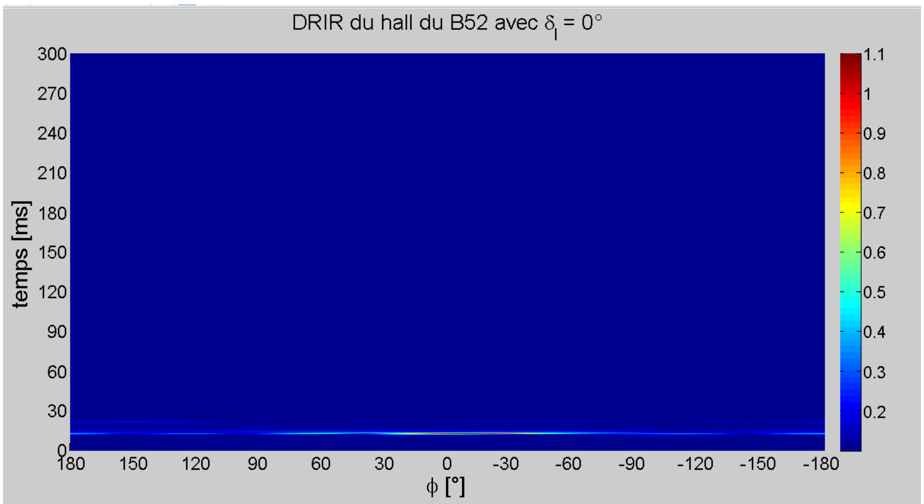


Figure 47. Impulse response

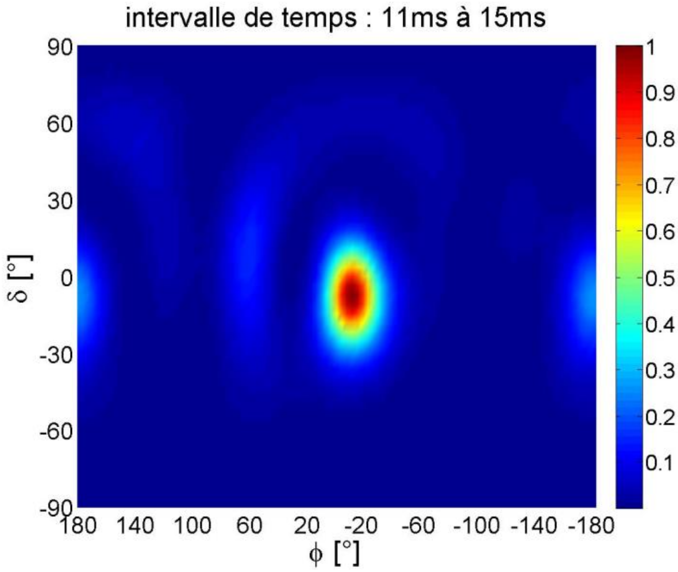


Figure 48. Directivity in a time interval

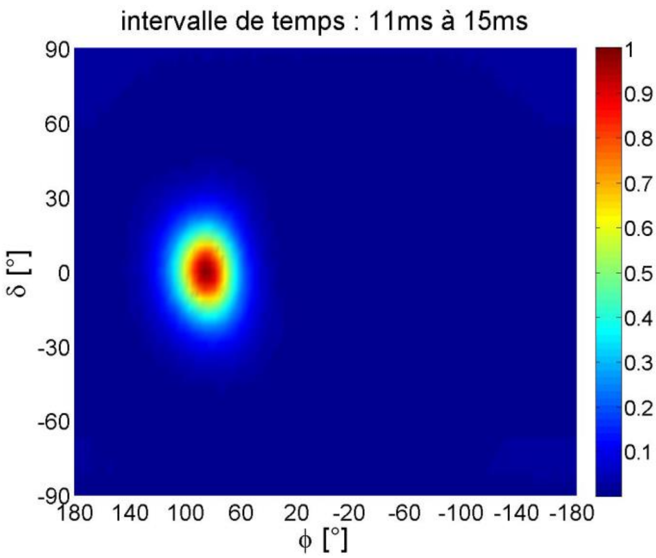


Figure 49. Directivity in a time interval when the source is at 90°

The following table has been computed with a time window from 11 ms to 15 ms:

Incident angle	Angle of max intensity	BW 3 dB (º)	BW 10 dB (º)	2º lobe (dB)
0	20	20-30	50-60	11.30
10	20	10-20	40-50	13.52
20	20	10-20	40-50	15.39
30	10	10-20	40-50	20.05
40	10	10-20	50-60	14.38
50	10	10-20	40-50	12.72
60	10	10-20	40-50	11.96
70	10	10-20	40-50	12.47
80	10	10-20	40-50	14.72
90	10	10-20	30-40	15.59
100	0	10-20	30-40	19.67
110	0	0-10	20-30	19.94
120	0	10-20	30-40	19.28
130	-10	10-20	30-40	18.75
140	-10	10-20	40-50	17.24
150	-10	10-20	40-50	16.31
160	-10	10-20	30-40	16.75
170	-10	10-20	30-40	12.91
180	-20	10-20	30-40	11.70
190	-20	20-30	40-50	11.21
200	-20	20-30	40-50	11.50
210	-20	20-30	50-60	12.00
220	-20	20-30	50-60	12.43
230	-20	20-30	50-60	12.03
240	-20	20-30	80-90	11.06
250	-20	50-60	90-100	10.87
260	0	30-40	80-90	15.40
270	0	20-30	60-70	17.53
280	-10	20-30	70-80	14.86
290	-10	50-60	90-100	12.15
300	20	50-60	80-90	12.02
310	20	20-30	60-70	15.98
320	10	20-30	40-50	21.58
330	10	10-20	30-40	16.78
340	0	10-20	20-30	13.30
350	20	40-50	70-80	10.31

Table 6. Summary of the parameters playing a sweep

In order to compute the impulse response we will use DAS beamforming. However, we will apply MVDR to compute the directivity with windowing.

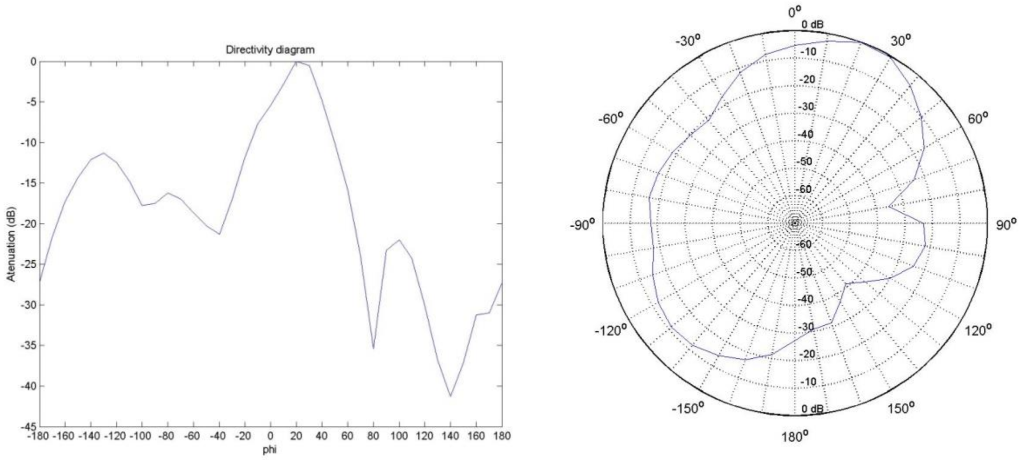


Figure 50. Directivity depending on ϕ in (0, 0)

4.3 Main conclusions

As we have already said, it works better for DAS beamforming than for PDW. PWD is really sensitive to the position of the microphone, particularly at low frequencies. It could be explained that these kind of result can arise from PWD beamforming, which means that we do not trust it for our purpose, we prefer anyone of other two cases.

If a window is taken, then we will have a better directivity with MVDR than with DAS beamforming but, if an impulse response is plotted on the whole length of the response, then we could have weird results with MVDR. In this case, we will have much better result with DAS, because MVDR does not really respect the time sequence of the signals.

Measuring pure tones, it is not really important which of them you want to use because it is a periodical signal. Nevertheless, if we measure an impulse response then the time sequence becomes important and MVDR does not respect this in every situation. The best way to proceed is to measure a complete impulse response with DAS and then if we need more details in the impulse response, such as to acquire the direction of the first direct impulse, then we will perform the MVDR on the small desired window.

A lot of time is invested measuring the sound field each several degrees with the rotating table not only to obtain the values, but for the following processing of these values. The table can make a complete revolution continuously or step by step. We have done discrete steps to avoid the noise of the rotation and because the rotating table is not time invariant. However, we have to assure of waiting enough time until the microphones become stable.

With unusual values we have to be careful when we use or interpret the results. Based on the results, in some cases is hard to achieve only one main lobe, just see Figure 45. At 3000 Hz some problems have arisen, such as Figure 46, there are several secondary lobes in a single graphic, for this reason, we have to use the maximum lobe as the main lobe, and take from the other maximums, which we have called secondary lobes, the maximum value of all secondary lobes.

5. Diffuse field measurements

The last part of the project is more focused on the analysis of reverberant sound field taking measurements by means of our spherical microphone. Therefore, this step was performance in a reverberant room with the same equipment. In a reverberant room all the surfaces are reflective, thus, incident sound is reflected.

In this reverberant room the sound propagation could be affected by the reflections mainly from the wall, roof and floor.

In this case it is interesting to locate the microphone in several positions to see the caused effect. We will play several sinusoidal signals, at different frequencies, and a pink noise. Pink noise is a white noise that has been filtered to give equal power per octave, while white noise is a sound with equal power per Hertz. Pink noise is often used for testing auditoriums.

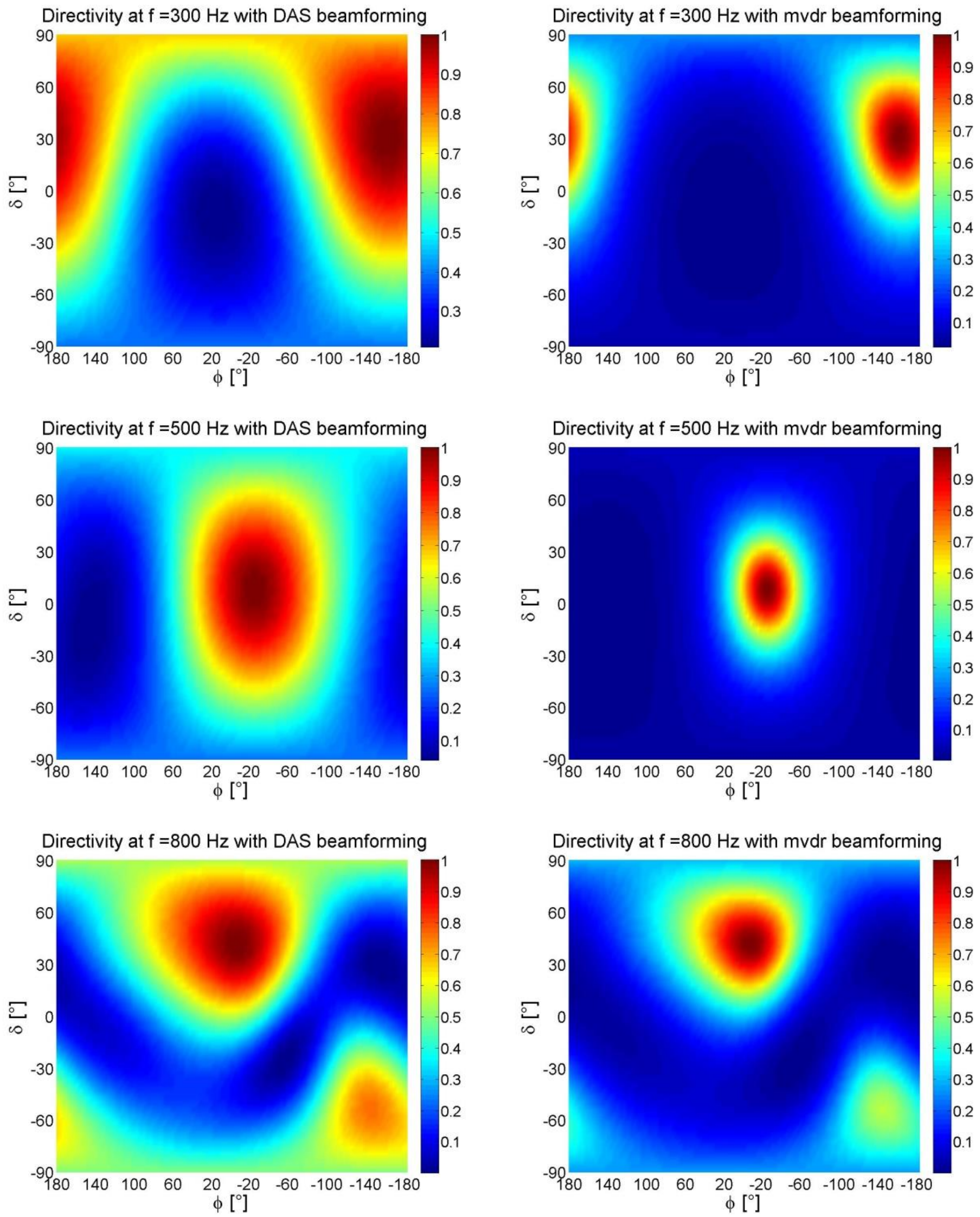
Therefore, in this chapter the diffusivity is evaluated in four points of the reverberant room for several pure tones and a pink noise (see Figure 51). We fix the microphone array in the horizontal way because the sound field should be close to be diffuse, so we will not turn the head. Furthermore, we will fix the azimuth and elevation angles at (0˚, 0˚) looking to the sound source.

Figure 51. Setup in the reverberant room without absorbing material

The location of the spherical microphone in the reverberant room is shown in Figure 53.

In these reverberant conditions there is no special interest in time domain analysis because there are so many reflections. However, it should be more useful measure pure tones or pink noise in order to assess the diffusivity of the room. The length of the wave field is much greater than the period of the signal, so we will have several periods in the recorded signal. But what is also important, we could have many seconds of reverberation.

We take 5 second of the middle of the record, it should not be too long to compute the beamforming. We avoid the beginning because we still do not have the reverberation, thus the interval will star at 10 seconds in order to have reverberation in the room until 15 seconds, before the end of the sinusoidal signal. We also avoid the end of the record, where the play was stopped and we only have the reverberation tail In conclusion, we make the beamforming of 5 seconds.



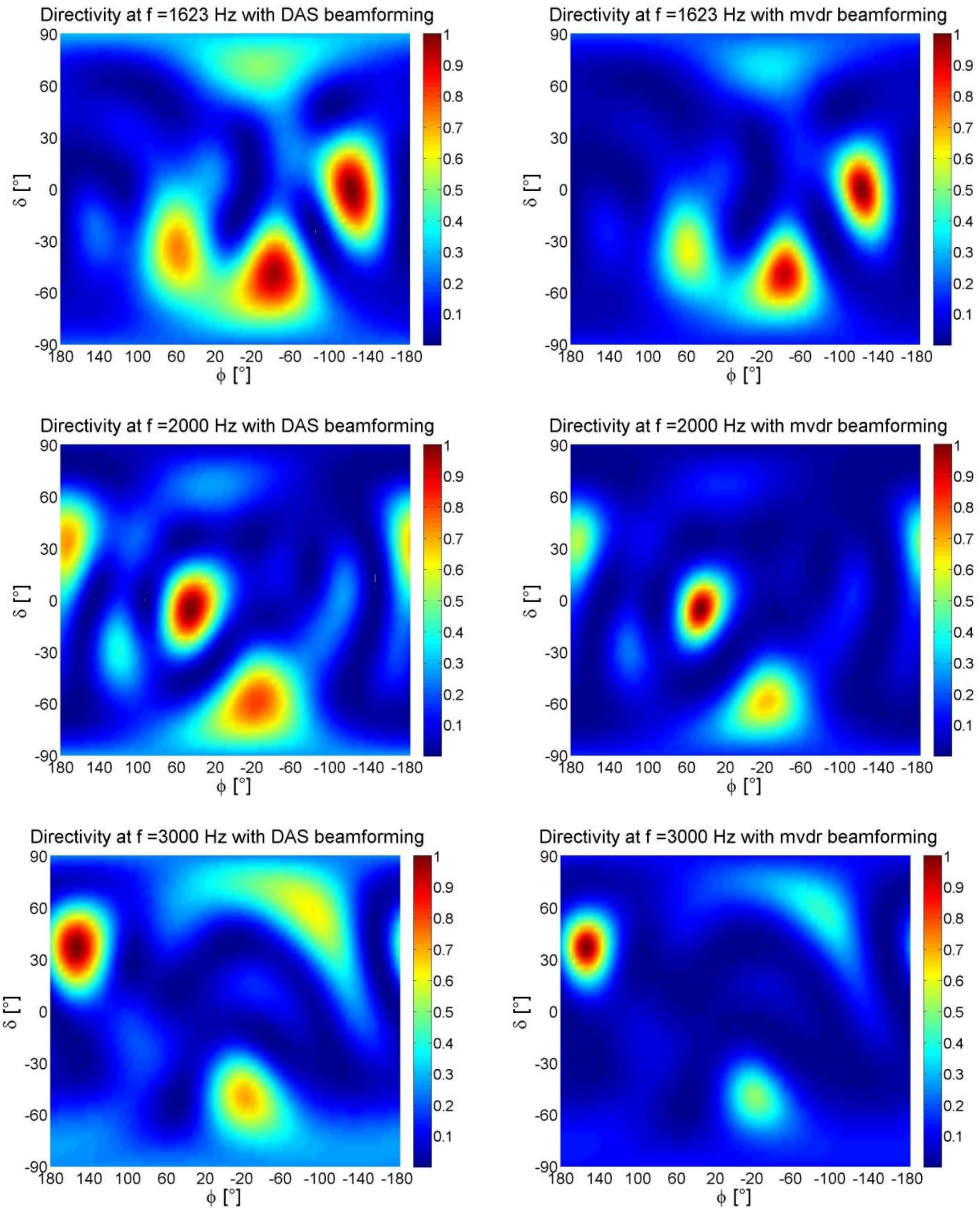


Figure 52. Directivity at several frecuencies and beamforming in the reverberation room in the location 1

As we can see in the previous figures, we have large variation of the signal, which means that we have some modes of propagation in the room, which could be the case at 300 Hz for example, Figure 52.

A sound field is diffused if the sound is propagated with the same intensity in every location and direction, which means that the sound level does not depend on the place or the direction of the measurement system.

I could have some reflection from the wall, but if the sound field is completely diffuse, then we should have approximately the same intensity, that is why is quite interesting to have this kind of results. We expected to have approximately the same value everywhere, nevertheless, these are the results of the measurements and this means that the sound field is not really diffuse.

Persisting with the purpose of evaluating the diffuse character of the room, we have added absorbing materials in the reverberant room in order to analyse if the results are consistent with the theory. We have carried out the same measurements in the same four locations. The absorbing material has been distributed as is shown in Figure 53.

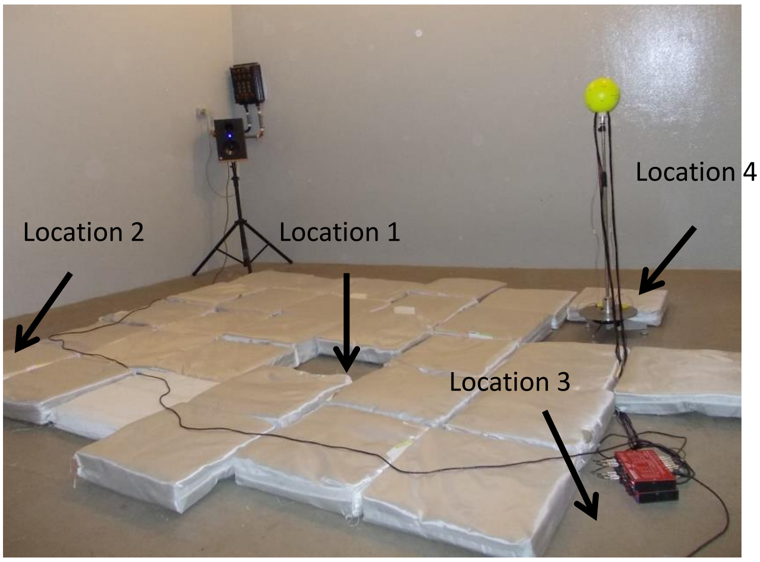
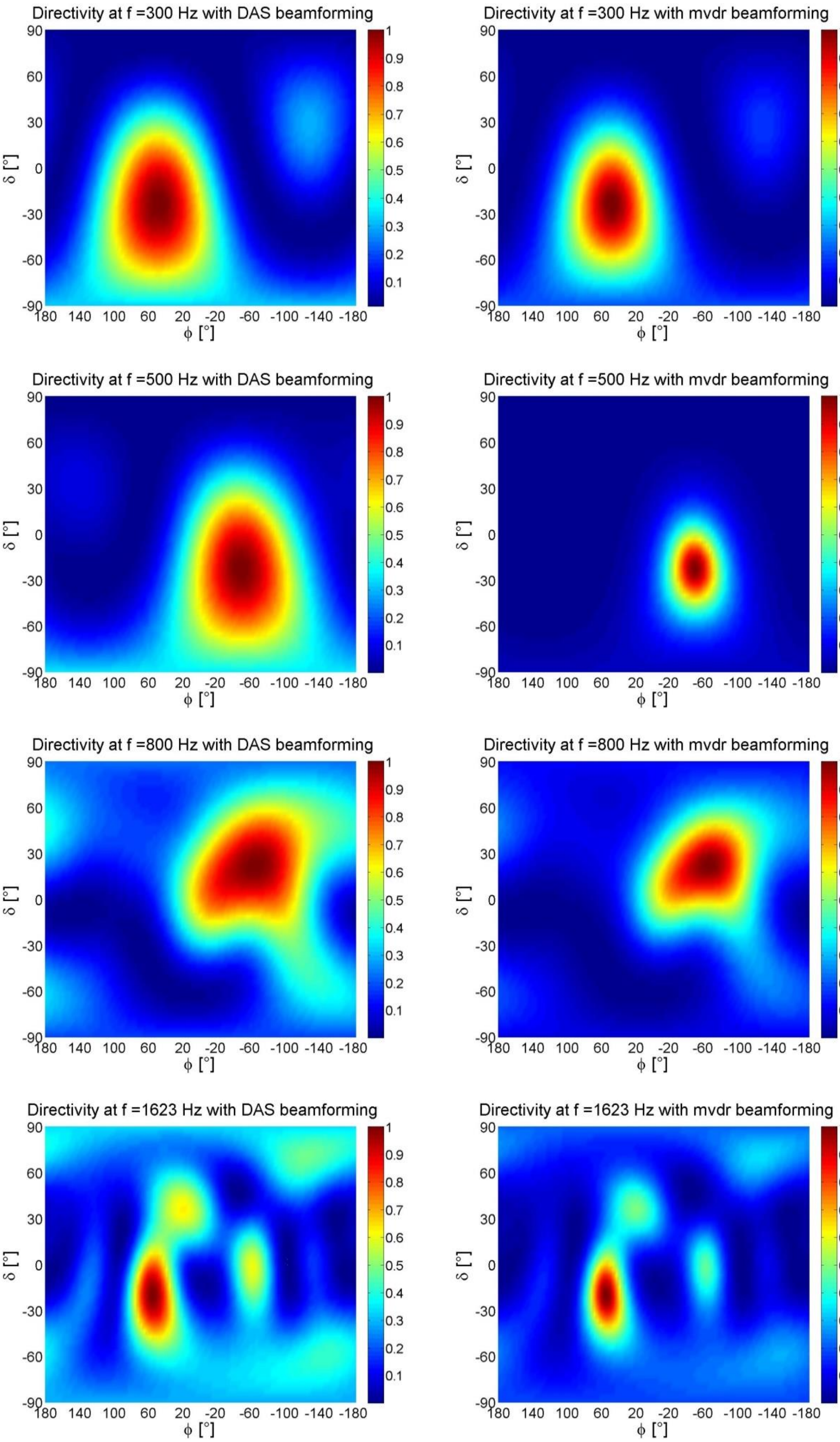


Figure 53. Setup in the reverberant room with absorbing material

Theoretically, we should have better directivity in these conditions because the reverberation decreases and we suspect that in that case the source could be more visible to the microphone array than in a “completely” diffuse field. In fact, we should have results that tent to the achieved results in the anechoic room. Obviously, is not an anechoic room, but as we have said, the fact that we introduce absorbing materials on the floor of the room means that the reverberation should decrease and then the source might be more visible. We could also expect directivity in the direction of the material. In that case, it would not be a source, it would be, what is called a sink. The source emits sound, whereas the sink absorbs sound. The material which we have added is a surface that attracts the soundwave instead of emitting soundwave.

Far from the source and the material the intensity vector will be approximately zero because the sound field is diffuse. There is no particular direction, therefore we have a very small intensity vector, but close to the absorbing material you could have significant vector in the direction of the absorbing materials and it could be also important when we are close to the source.


Directivity at f =300 Hz with DAS beamforming
Directivity at f =300 Hz with mvdr beamforming
Directivity at f =500 Hz with DAS beamforming
Directivity at f =500 Hz with mvdr beamforming
Directivity at f =800 Hz with DAS beamforming
Directivity at f =800 Hz with mvdr beamforming
Directivity at f =1623 Hz with DAS beamforming
Directivity at f =1623 Hz with mvdr beamforming
δ [°]
ϕ [°]
90
60
30
0
-30
-60
-90
180 140 100 60 20 -20 -60 -100 -140 -180
1
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1

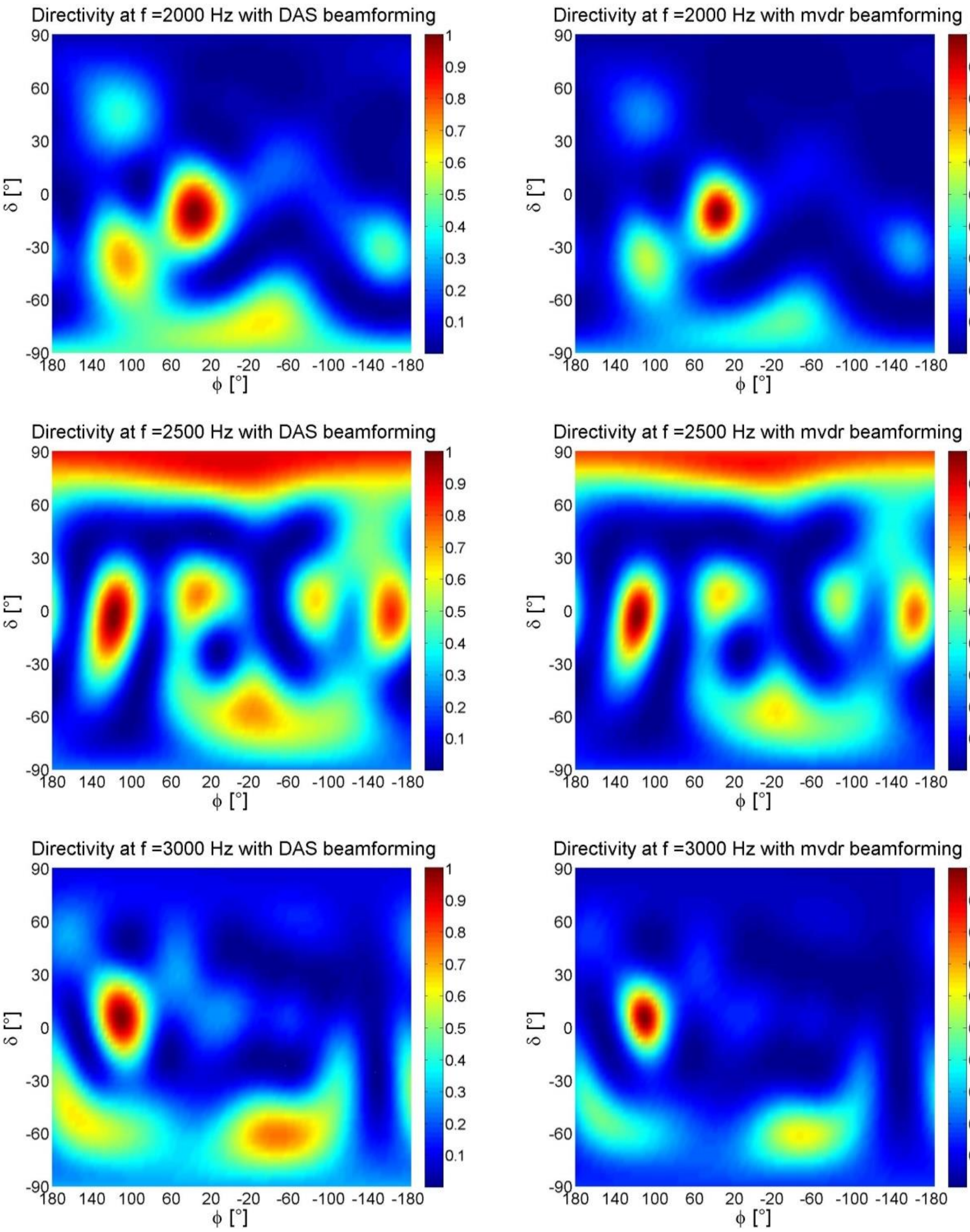
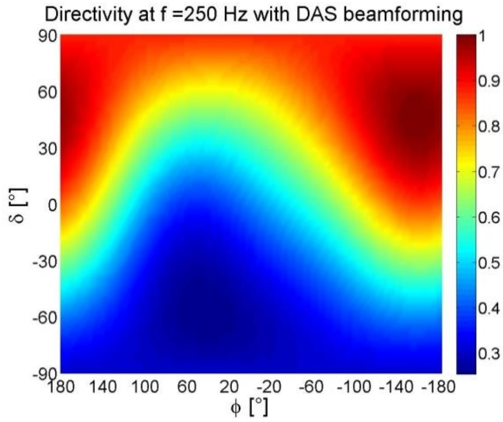
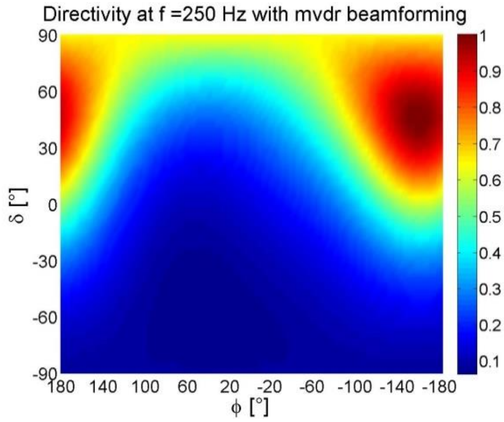


Figure 54. Directivity at several frecuencies and beamforming in the reverberation room in the location 1 with absorbing materials

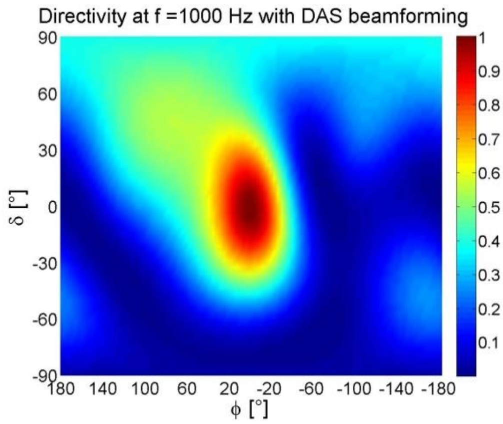
As in the previous task, without absorbing materials, we also measure a pink noise, where I can specify a band pass filter. With a pink noise we will have many frequencies which contribute to the results. This means that before computing the results, the program will first filter the microphone signals in the specific band. We have used close to octave band filters.

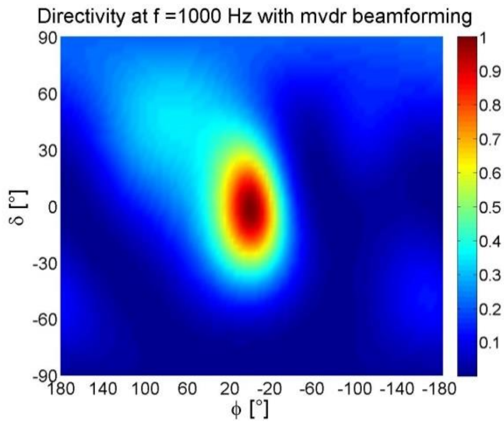
180-315



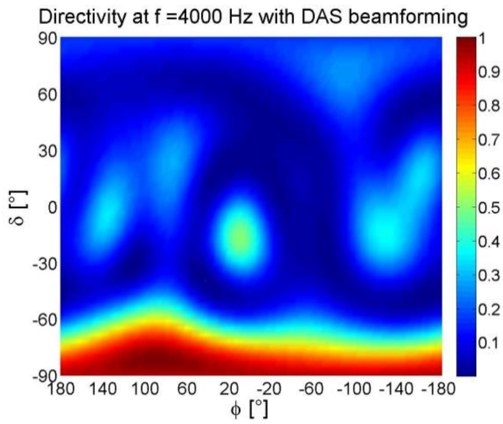


700-1500





3k-6k



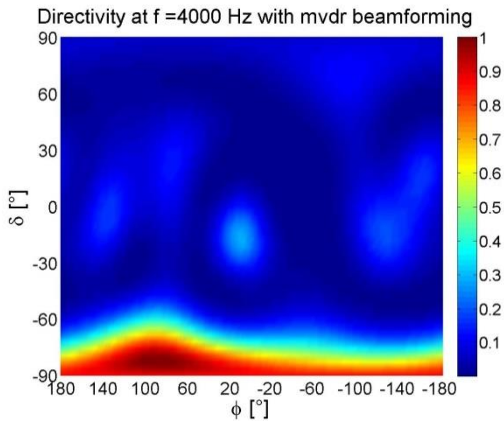


Figure 55. Directivity at several frecuencies and beamforming in the reverberation room with absorbing materials

The direct sound it should be always in (0˚, 0˚). The fact that we find a specific reflection that it is stronger than the other, it is could be quite questionable because it is an unusual behaviour in this kind of rooms. The other conclusion could be that beamforming method is not really accurate in rooms with reverberant characteristics.

It is clear also in this case, that PWD give us worse problems, therefore, these results are not showed either.

We have decided to evaluate the percentage of a section of intensity plotted in the pink noise diagrams by selecting a small interval to avoid the highest intensity. We will do it just for the location number 3 and a range between 0.1 and 0.5 of the colour bar. We compute this percentage for DAS and MVDR beamforming. In average, the obtained percentage has been 20%, therefore we can’t say nothing relevant.

According to my measurements, the main conclusion in the reverberant room if that for pure tones and pink noise, even if the pink noise is filtered in some bands, the sound field is not really diffuse, since we have some directions from which the intensity is more important than others. We cannot conclude anything relevant about the diffuseness of the sound field.

The issue in these measures is that the Matlab function ask for a specific frequency, thus in such space (reverberant) we can have specific effects at this frequency. Perhaps it would a better idea to use temporal analysis in this case (between 15 and 20 s) because the signal is quite stationary and we are not forced to specify a given frequency. The drawback is that we only have an azimuthal analysis, between -180˚ and +180˚.

Furthermore, we cannot rely on the PWD beamforming method because it gives us strange results in almost all frequencies.

6. Conclusions and applications or future work

Microphone arrays are being studied and employed in various acoustic applications such as sound source localization or the description of main characteristics of reverberant sound fields. There are several recent applications based on microphone arrays such as virtual reality or related to improving in robot audition.

Sound source localization using microphone arrays has had a high success in room acoustics.

This thesis develops a continuation of a previous work in order to analyse the performance of the spherical microphone array that the authors had created. The performance of the measurement system is studied by several kinds of measures which have been carried out in different rooms by displaying the sound pressure distribution.

The results of this evaluation are also illustrated in this document. The estimation of the direction and localization of acoustic sources was not really good.

The experimental study includes capturing the impulse responses and evaluating of different types of beamforming. We conclude that PWD works badly in the measurements of this project.

For future works, it might suggest measure not with continuous signal but with a sine sweep and then obtain the impulse response where we will could suppress more easily the direct sound by taking specific time section. However, with a continuous signal like sine wave, we cannot remove the direct sound, it is not possible. It would be better, looking at this diagram, to make impulse response measurements and remove the direct contribution. Then we continue analysing the rest of the impulse response. Perhaps should be more diffuse.

In some tests, the obtained results have not been as good as we expected but all the measurements, the corresponding process and obtained results are useful despite the fact that we have also obtained unexpected results.

What is interesting in the reverberant room is to show that the level is the same in any direction, which is not the case but we could try in pink noise (because even if I filter in a specific octave band we have many frequencies which contribute to the result which is shown in my diagrams.

References/Biography

[1] Hermine Feron, “Conception et applications d’un réseau de microphones pour l’acoustique des salles”, Master’s thesis, University of Liège, 2013.

[2] M. Imran and J.Y. Jeon, “Plane Wave Decomposition and Beamforming Directional Spatial Sound Location”, *International Bhurban Conference on Applied Science & Technology,* (Islamabad, Pakistan), January 2016

[3] Munhum Park and Boaz Rafaely, “Sound-field analysis by plane-wave decomposition using spherical microphone array”, The Journal of Acoustical Society of America, vol. 118, pp 3094-3103, 2005.

[4] Akhtar Hussain, Nassir M.Qazi, Muhammad Sadiq, “A methodology for sound source localization and tracking”, *International Bhurban Conference on Applied Science & Technology,* (Islamabad, Pakistan), January 2016.

[5] H. Sun, S. Yan, U.P. Svensson, A .Solvang and J.L. Nielsen, “Achievable maximum-directivity beamforming for spherical microphone arrays with random array errors”, *European Signal Processing Conference*, (Aalborg, Denmark), August 2010.

[6] H. Sun, E. Mabande, K. Kowalczyk and W. Kellermann, “Joint DOA and TDOA estimation for 3D localization of reflective surfaces using eigenbeam MVDR and spherical microphone arrays”, *International Conference on Acoustics, Speech and Signal Processing*, Prague, 2011.

[7] Sakiri Tervo and Archontis Politis, “Direction of arrival estimation of reflections from room impulse responses using a spherical microphone array”, *Transactions on audio, speech, and language processing,* vol. 23, no. 10, October 2015.

[8] mh acoustics, “EM32 eigenmike microphone array release notes (v17.0),” *mh acoustics*, Summit, NJ, USA, Tech. Rep., Oct. 2013.

[9] A. Farina, L. Tronchin, “3D Sound Characterisation in Theatres Employing Microphone Arrays”, Acta Acustica united with Acustica, vol. 99 (2013), pp 118-125.

[10] A. Farina, A.Amendola, A. Capra and C. Varani, “Spatial analysis of room impulse responses captured with a 32-capsules microphone array”, Audio Engineering Society 130th Convention, London, UK, May 2011.

[11] Adobe Audition, URL http://www.adobe.com/es/products/audition.html

[12] Roland Corporation, “Edirol FA-101 FireWire Audio Capture, Owner’s Manual”, November 2006.

[13] Roland website, URL http://www.rolandus.com/products/fa-101/

[14] Promidi website, URL http://promidi.ru/shop/UID_429.html

[15] Matlab website, URL http://uk.mathworks.com/