Faculdade de Engenharia da Universidade do Porto

On the Analysis and Improvement of Hearing Aid Devices

Diogo Rafael Amado Almeida

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Orientador: Professor Artur Capelo Cardoso

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Abstract

Hearing Aid devices have come a long way since the implementation of DSPs, but their users still feel the disadvantages of having to use one on a daily basis. These devices fail to simulate the ability of the human hearing system, mostly because of the complexity of the acoustic factors involved, and also, sound cannot be reproduced exactly how it is due to limitations of digital sound signal reproduction. Among the several problems reported by the users the most common are occlusion effects, sense of disorientation and discomfort in loud environments due to the amplification of all the sounds around the user. The focus of the research presented in this report is on directional sound capturing which will be highly valuable for loud environments since this is the biggest complain of hearing aid users. The final prototype consists of two cardioid microphones connected to a FPGA that processes the two input signals and outputs the sound that the user will be listening. The ultimate goal is that in a noisy situation the user only listens to his point of interest, and everything else is filtered out.

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Introduction

Hearing aids are a perfect example of a current technology that really stresses the need for digital signal processing. These devices are systems that combine acoustics, electronics and human hearing.

Initially, non-digital hearing aids only amplified the sound they captured. This brought advantages because the sound was more natural and disadvantages because users only loose hearing capabilities on certain frequencies, so amplifying all of them cause even more damage to the hearing system. Also the performance was not the best since the user still needed to be really close to the source of the sound.

After the introduction of digital hearing aids, problems that did not existed before started to arise, like feedback, occlusion effects, background noise among others. Many problems like feedback were fixed, but there is a lot of space for improvement, since it is very rare to find a satisfied hearing aid user. The main problem is that natural acoustic sound and the way it is captured by our human hearing system is hard to reproduce digitally. This project focus on the research of directional sound capturing, since one of the biggest complains of hearing aid users is the felling of disorientation and discomfort in loud environments. A healthy human hearing system can detect, on a three dimensional space, where the sound is coming from, so in a situation where multiple sound sources are active, the system can focus on the one that matters since it knows and feels where it is coming from. On a hearing aid all the sound is captured by the microphones and then outputted from only one source, the speaker. Thus the user cannot identify were the sound is coming from, and on the situation where multiple sources of sound are present, the user gets disorientated.

1.1 - State of the art

In the market there are around ten different main types of Hearing Aids. The main difference between those types is the size of the device and how they approach acoustic treatment.

The size is important because the bigger it is, the most processing capacity can be implemented, but the portability decreases and the comfort for the user is reduced. The shape of the earpiece is going to affect acoustics, so there are different approaches depending on the needs of the user. An example of how the shape can affect acoustic is the occlusion effect. This occurs when the earpiece blocks the ear canal making it hard for air to flow freely, so the sound generated by our body, for example chewing, is going to reverberate much more than it should.

- “body worn aids” are the largest type of Hearing Aids and are not used as much nowadays due to the existence of much smaller devices. They have the potential for much greater battery life and better processing capabilities, but they are not good for children or individuals with a very active life style.
- “behind the ear” are one of the most famous Hearing Aids, they have medium size and are supported by the ear itself. It is practical to remove and operate, but is visible to the naked eye. Esthetics is a concern.
- “In the ear, In the canal and Completely in the canal” are a much smaller type of Hearing Aids that are used depending on the age and preference of the user.

- “Open-fit devices” are a type of Hearing Aids specifically designed to improve the acoustics, where the earpiece is not blocked thus reducing the occlusion effect discussed previously. Due to their small size “Invisible in canal hearing aids” also reduce this effect a lot.

Digital and Analog Hearing Aids:

- Analog- These Hearing Aids do not use any type of digital processing like the name implies. All the sound treatment is achieved with hardwired circuits that can achieve amplification, directional sound capturing, frequency filtering, noise and feedback reductions, but due to their nature they are not reconfigurable and are not as effective as digital Hearing Aids. They have a few options to adjust the sound, like volume control and triggering directional capturing on/off. Some users still prefer this technology since the analog sound is considered more natural that artificially generated sound by the DSP in the digital Hearing Aids.
- Digital- Widely available since 1997, this have become the most popular type of Hearing Aids, they have an integrated microprocessor that processes the sound allowing for an automated set of functions like automatic volume control among all the following features.

Regarding the sound treatment there are several approaches used in Hearing Aids.

- “Digital Feedback Reduction” solved a very common problem in the past: feedback, which usually occurred when the volume of the device was to loud or if the user was close to the source of the sound.
- “Digital Noise Reduction” a process that analyses the sound and reduces the noise. This process is done by averaging the input information making it possible to recognize a signal with the same characteristics over time, usually unwanted sound, or noise. Amplification of certain frequencies is also used with some Hearing Aids having the capability of being customized for the end user.
- “Digital Speech Enhancement” is a recent technology that analyses the captured sound analyzing the temporal occurrence and spectrum and amplifies certain areas of the captured speech considered more relevant for the user.

- “Gain Processing” is an important area of the Hearing Aids, since the main objective is to amplify the sound, using Gain Processing technology, this sound is not uniformly amplified, but instead thresholds of amplitude are used, this thresholds can be applied to frequency variations or fluctuations on the surrounding environment.
- “Signal generators” produce sound generated by the DSP in the Hearing Aid, this can be very useful for the personalization of the device, work that could be only done previously by a specialist. The system generates a sound that can be adjusted by the user to meet the desired result.
- “Directional Sound Capturing” is commonly used in Hearing Aids, this system is usually triggered on or off automatically by the device when the user enters a loud environment, and using two directional microphones it adds their signals, thus, rejecting a certain amount of lateral sound. A lot of false positives are generated since a loud situation can last a few seconds, minutes or hours. This is the area of focus on this Project.

1.2 - Motivation

Sound is an area of engineering with a lot of progress to be made. It is information that can be translated digitally, although not completely accurately and always with information loss. One of the initial ideas of this project was to create a system that would capture sound information from two different sources, compare all the characteristics from those sources and locate the differences and then apply a filter that would modulate source 1 in order for it to have all the characteristics from source 2. But after some consideration it was found to be more valuable to apply this principle to a technology that is essential to a lot of individuals, hearing aids. These two areas are similar since we are trying to reproduce the characteristics of the sound that a healthy human being hears on a digital device.

After the analysis of the state of the art of hearing aids, there were a lot of fields to be studied, from frequency modulation, amplitude treatment to directional sound capturing, but the work of Alan Dower Blumlein about Binaural sound was an inspiration that oriented the interest of this project towards directional sound capturing.

Blumlein, inventor of stereo was one of the first individuals that knew that reproduced sound had to have in consideration the human hearing, so if humans have two inputs why should the reproduced sound have only one output. [9]“One day in 1931, Alan Blumlein took Doreen to the cinema and said to her during the film: ‘Do you realize the sound only comes from one person?’ Doreen, by her own admission, was not a techniqueal person and so replied to him, ‘Oh does it?’ and he said, ‘Yes. And I’ve got a way to make it follow the person. Alan Blumlein had just tried to describe his first thoughts about the system he would always call ‘binaural sound’, but which we have to know better as stereophonic or stereo sound. Blumlein explained to Doreen that, if she could imagine being blind, and sitting in the cinema, she would be able to point out exactly where the person was on the screen with his system. This, of course, was what he was trying to achieve, not this ‘terrible effect’ where the sound comes from one side of the screen when the actor was at the other side.” After some more research it was obvious that the path to follow to achieve directional sound capturing was through signal phase interpretation, which is going to be discussed in detail in this report.

1.3 - Objective

The objective of this project is to study techniques that will solve the problem of disorientation that hearing aid users feel in loud environments. This will be achieved with directional sound capturing.

Several fields will be studied, from signal correlation to signal modulation, in order to determine the advantages and disadvantages of each one and determining the most efficient.

Noise reduction and angle control will also be taken into consideration.

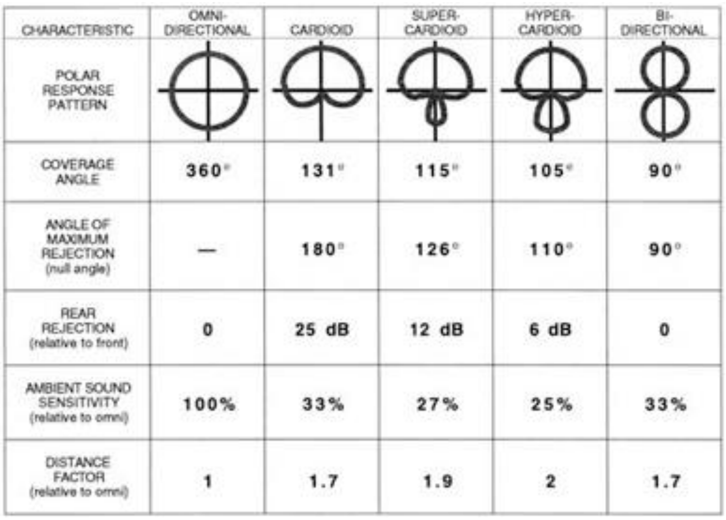
No fixed angle of rejection was set as an objective because the goal is to have the best angle of rejection.

2

Approach

Due to the fact that there are multiple different environments with which a hearing aid user is confronted, it is not always the best solution to have directional sound capturing. Because of this the system will have the possibility of being turned on or off by the user, giving more potential on directional sound capturing since it is not necessary to take in consideration environments other than loud ones. A lot of hearing aids have dynamic directional sound capturing, where the system tries to identify loud situations, triggering the directional microphones. This is not always good because it creates false positives.

After the analysis of different types of microphones available in the market it was obvious that cardioid microphones were the best choice, since they are the ones with most angle rejection.



CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BI-DIRECTIONAL
POLAR RESPONSE PATTERN					
COVERAGE ANGLE	360°	131°	115°	105°	90°
ANGLE OF MAXIMUM REJECTION (null angle)	—	180°	126°	110°	90°
REAR REJECTION (relative to front)	0	25 dB	12 dB	6 dB	0
AMBIENT SOUND SENSITIVITY (relative to omni)	100%	33%	27%	25%	33%
DISTANCE FACTOR (relative to omni)	1	1.7	1.9	2	1.7

Figure 2.1 - Microphone polar patterns compared [12].

2.1 - Theory

What is the main difference between the signals captured by the microphones of a sound source that may vary in space? If the source is closer to microphone 2 theoretically the amplitude will be higher, but if there is a spike of noise coming from the left and being captured by microphone 1, both microphones might capture the same amplitude. This makes amplitude relation unreliable for directional sound capturing.

Human voice can be mostly filtered from other sounds using frequency filters, but since human voice is also part of the background noise, frequency filters are not a valid option as the main engine for directional sound capturing.

The answer to this question is time, in other words, phase, a signal that is exactly in front of both microphones will be captured by them at the exact same time (common phase), but a signal that is closer to one of the microphones, will reach one of them before the other, so, although the signals captured from the microphones have approximately the same characteristics, since they are captured from the same source, they will have different phases.

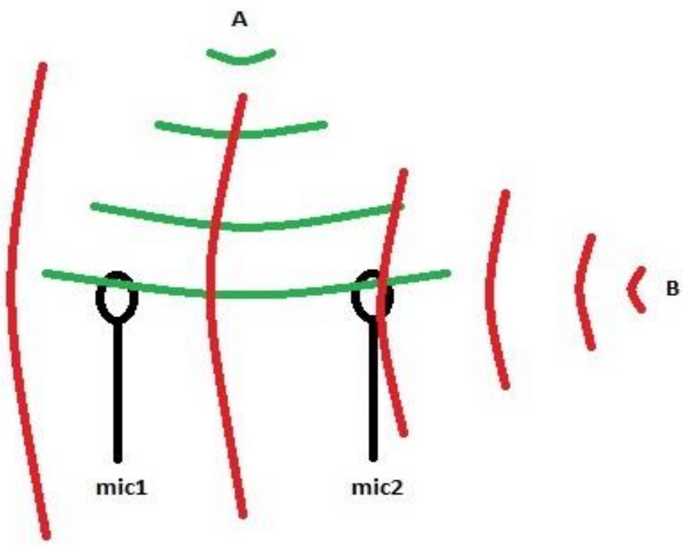


Figure 2.2 - Sketch of two signals being captured.

This is the main element that will be studied. In the example above, two sound sources are happening at the same time. The Objective is to reject sound source B and keep sound source A. Two main approaches were made in the development of this project: first signal correlation; second, signal adders.

2.1.1 – Signal correlation

To compare the phase using signal correlation, it is needed more than one point of the signal since time differences are being dealt with.

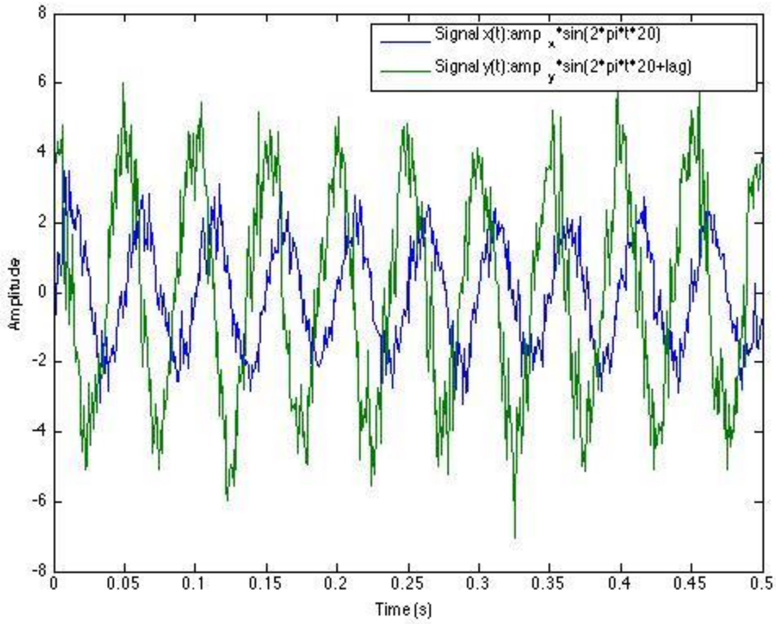


Figure 2.3 - Similar signals with different phase.

Above is an example of two similar signals with a phase discrepancy. Although these signals are not captured from the real world, they represent a similar situation where one of them actually has higher amplitude and they a have a difference in phase.

The approach to compare the phases between these two signals was to represent the signal in frequency domain, find the max absolute value and then calculate the angle. This way, the main source of information will be the one analyzed.

At this point a problem was foreseen, the amount of processing power was unknown, as well as how the buffers would impact the quality of the sound.

Nevertheless if signal correlation is achieved successfully it would be the most accurate way to achieve directional sound capturing.

2.1.2 – Signal adder

This technique has several advantages: the future of the signal is irrelevant; the level of processing power is minimal; and multiple sources can be processed at the same time.

If the same signal captured by the two microphones is in phase and an addition is made, the amplitude will be doubled, but if the signals are out of phase, the amplitude gain will be inferior as shown in the simulation bellow

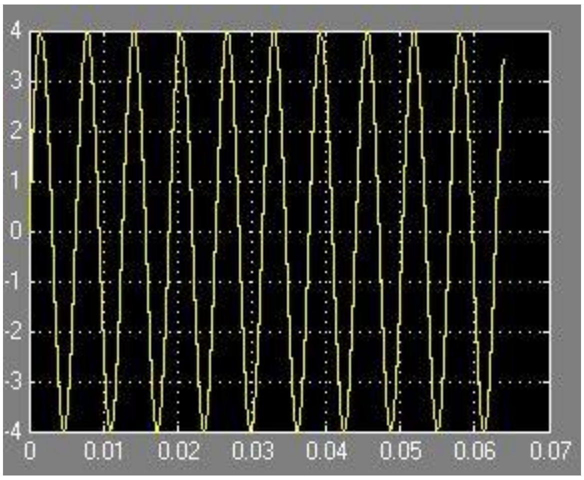


Figure 2.4 – Signals in phase.

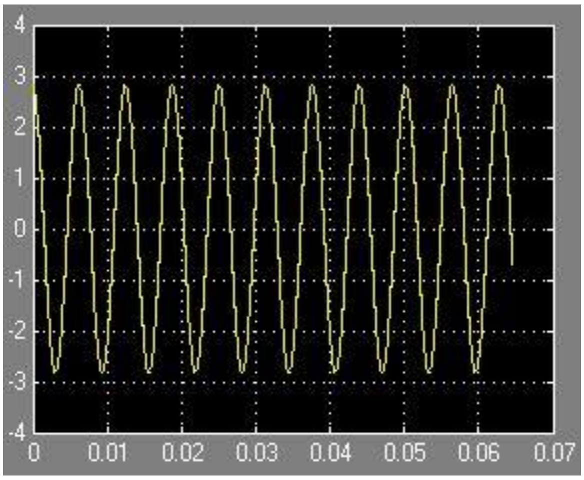


Figure 2.5 – Signals pi/2 out of phase.

In this example two sinusoidal of amplitude 1 were added as X=sin1+sin2, and then Y=X1+X2. In figure 4 they were in phase so the final amplitude is 4, in figure 5 they were pi/2 out of phase, resulting in amplitude bellow 3.

During the project tests with real voice signals were made, noise filters were implemented as well as an angle control technique.

The base formula for the signal adder is the following (situation were no extra gain is desired):

$$Output = \frac{Mic1 + Mic2}{2}$$

2.2 - Block Diagram

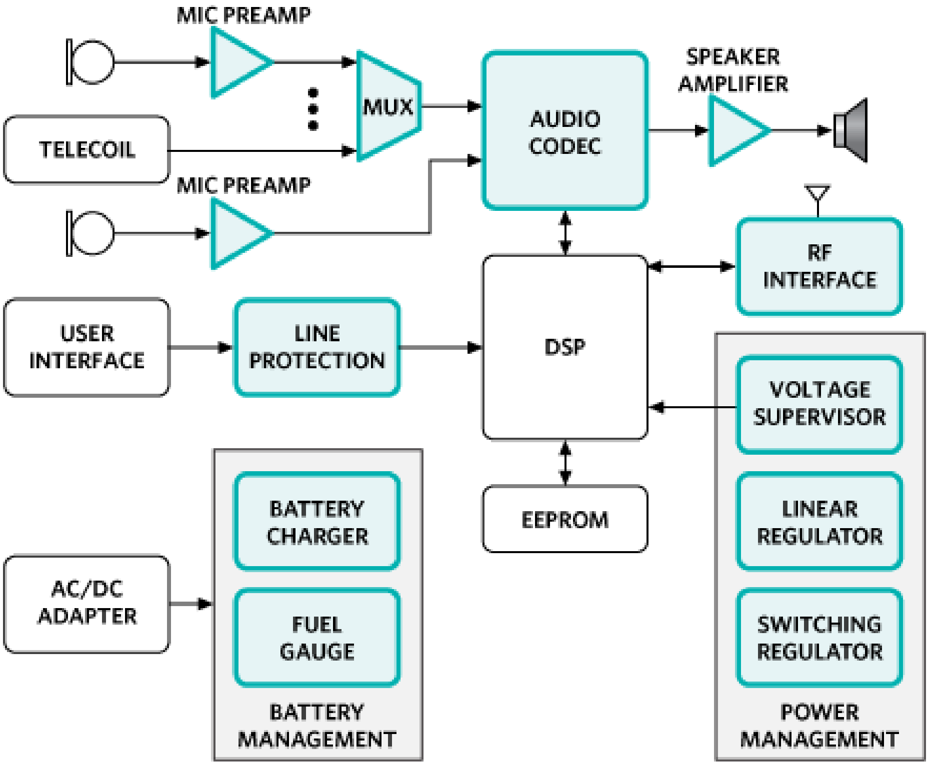


Figure 2.6 – Typical block diagram of a hearing aid [13].

The image above represents the typical block diagram of a hearing aid.

The blocks used in this project are the following:

- Microphone Preamps
- Audio Codec
- Speaker Amplifier

These blocks are integrated on the board Virtex 2 Pro.

- User Interface- On the test model the user interfaced is simulated using Matlab and System Generator.
- DSP will be the FPGA on the Virtex 2 Pro board.

2.3 – Conceptual design

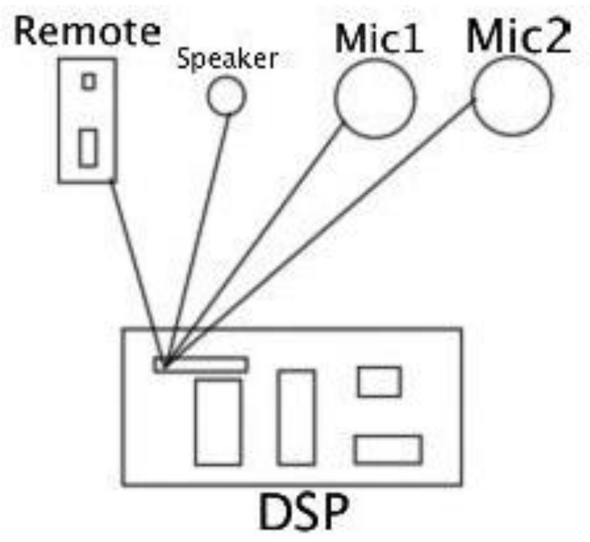


Figure 2.7 – Conceptual design for the main system.

The image above represents a sketch of the system used in this project; two mono cardioid microphones connected to the board Virtex2pro with an FPGA by Xilinx, a speaker and an interface, that in this case was the computer to which this FPGA was connected.

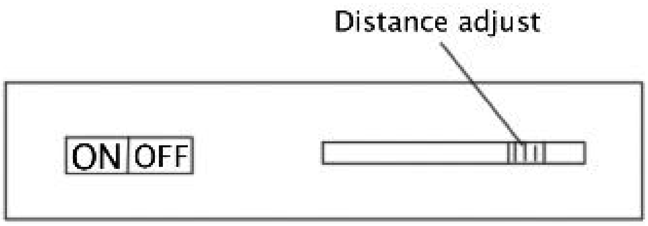


Figure 2.8 – Conceptual design for the remote.

The sketch above represents the functionalities that the required remote needs to have, an on/off button and an angle adjustment slide.

3

Implementation

As stated previously, this project had two main approaches, the implementation of signal correlation, and the signal adders.

Because signal adders are commonly used to achieve directional signal capturing, no pre-simulation was considered necessary, but regarding the signal correlation some studying and tests needed to be made. Therefore, the first implementation began with Matlab/Simulink tests.

3.1 – Matlab stimulation

An initial Simulink model was built, composed of two sources: one processing block and two sources. The two sources were the microphones connected to the computer, speakers and a scope.

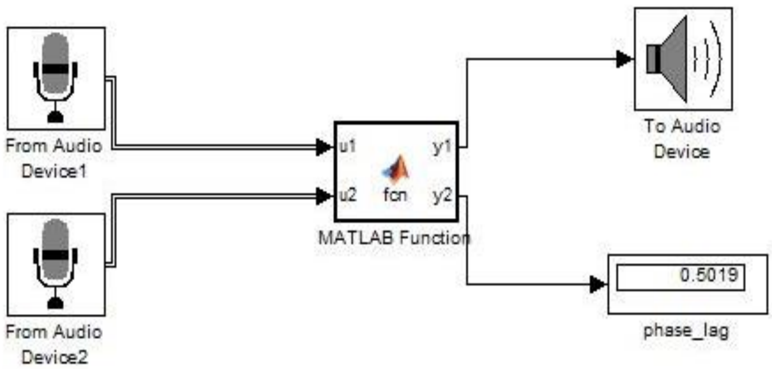


Figure 3.1 – Simulink model.

The Matlab function needed to be coded had to first remove the bias of the signals, then acquire the phase information from each one and calculate the difference between them, if this difference was considered acceptable the signal would be outputted, if not, no sound would go through.

```
function o = fcn(u1,u2)
x = u1;
y = u2;
u1size=size(u1);
% remove bias
x = x - mean(x);
y = y - mean(y);
%fast fourier transform
X=fft(x);
Y=fft(y);
% Determine the max point
[point_x] = max(abs(X));
[point_y] = max(abs(Y));
% phase difference at the maximum point
px = angle(X(point_x));
py = angle(Y(point_y));
phase_lag = py - px
%phase similiarity test
if (abs(phase_lag)>1.5)
   u1=single(zeros(u1size));
end
o = u1;
```

Figure 3.2 – Matlab function code.

The code is divided in seven main areas, the storage of the signals as variables, removing the bias from these signals, applying a fast Fourier transform, determining the point of these new graphs where the max value is found, acquiring the angle of the points, comparing these values. Represented in radians, and finally if the value is within boundaries, output the original signal.

In the comparison stage is where the value can be changed, this value controls the angle of capturing. If a value 0 is inserted, only signals completely in phase are going to be outputted.

When the code was first tested, with microphones and speaker, the results were very bad, the reason for this was considered to be that since the code needed to be running on real time, not enough processing time was given in order to achieve complete processing of the sound. In order to prove the code was working, a solution had to be found for testing purposes. This solution consisted in recording the voice of an individual on a file, then duplicating that file and editing in order that the sound would be 0°, 20°, 60°, 80°, 100°, 120°, 140°, 160° and 180° out of phase in relation to the original one (more explanations are given on the Chapter Results and Analysis).

A new Simulink model was built.

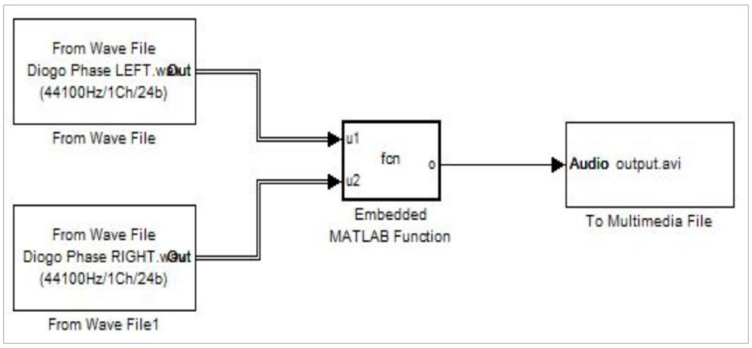


Figure 3.3 – Second Simulink model.

After this new implementation, the results were much better than expected. The sound was completely processed and the level of angle control was close to 100%, actually achieving this value if the phase tolerance would be set to 0, where only the portions of the signal completely in phase would go through.

More details are given in the fourth chapter.

3.2 – FPGA implementation

The FPGA used was a Virtex 2 pro by Xilinx. The respective board has four stereo 3.5mm jacks and uses ac97 audio codec. Since two microphones and one speaker needed to be connected, the first step was to analyze the circuits where these jacks were connected.

On the following images, it is possible to realize that none of these inputs have a similar construction, therefore, the two microphones, both mono, had to be connected to a stereo cable that would separate the left signal from the right signal.

The left signal would correspond to the microphone on the left and the right signal to the microphone on the right.

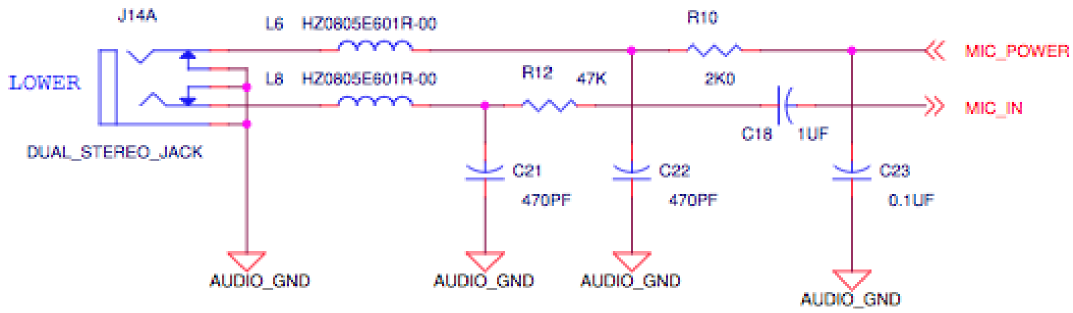


Figure 3.4 – MIC_IN circuit.

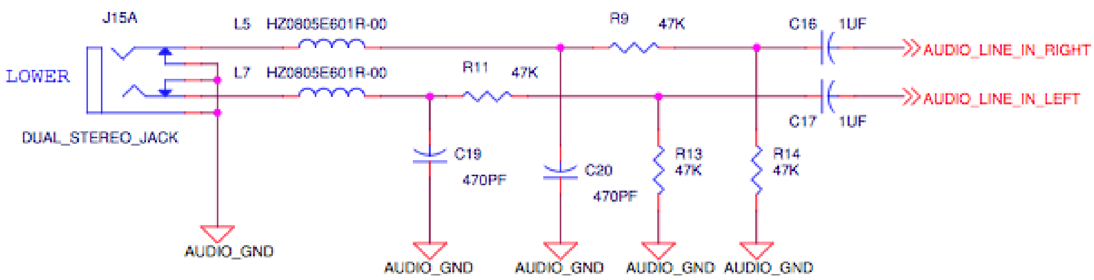


Figure 3.5 – LINE_IN circuit.

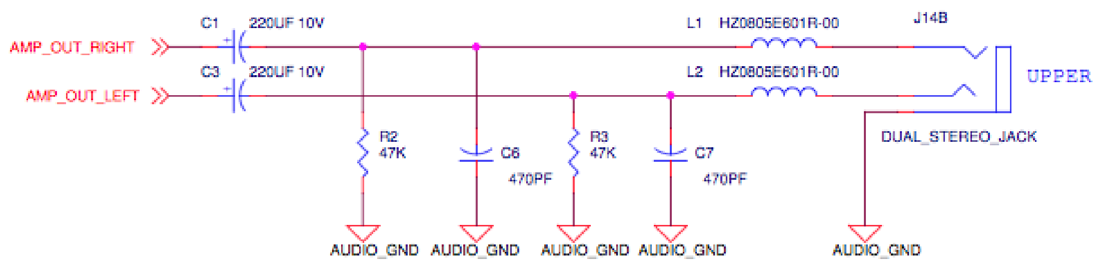


Figure 3.6 – AMP_OUT circuit.

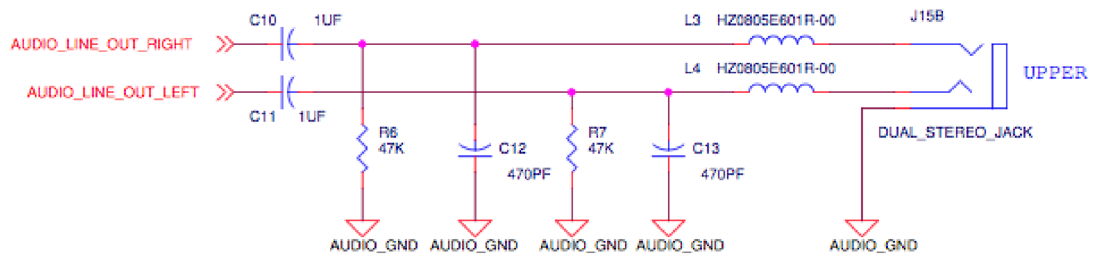


Figure 3.7 – LINE_OUT circuit.

After the analysis, the choice made was to connect a stereo jack to the LINE_IN, this stereo jack would carry the signals of the two microphones, and the output would be connected to the AMP_OUT circuit.

The reason that this careful analysis needed to be performed was because it was crucial to make sure that both signals from the microphones would be confronted with an exact same type of circuit since the objective is to compare the characteristics of these two signals.

The next step was to configure the net list. Xilinx provides on the Virtex 2 pro website documentation a pre-built System Generator for Matlab that outputs the signals received from the LINE_IN interface, with a user block where it is possible to program Xilinx specific blocks, and then outputs the signal to the OUT channels.

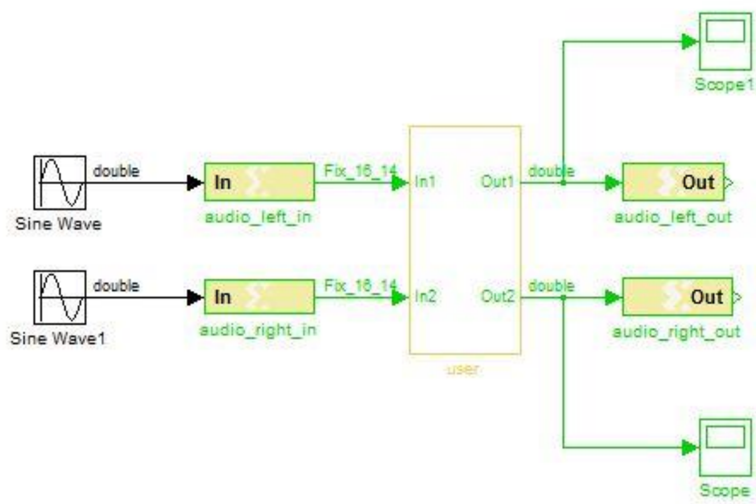


Figure 3.8 – System Generator connections circuit.

The Sine Wave blocks and the Scope blocks were manually inserted for testing purposes.

After the compilation of the blocks, System Generator generates the VHDL files, which include all the necessary net lists that make the interface between the board processor and the audio codec processor.

These files are then loaded into the board using iMPACT software.

3.2.1 – Signal adder

The first System Generator implementation was the addition of the two input signals. As mentioned before, this is the conventional way to achieve directional sound capturing.

The following image shows the block construction located inside the user block. This block can be interpreted as a DSP block since the signal operations are realized inside it.

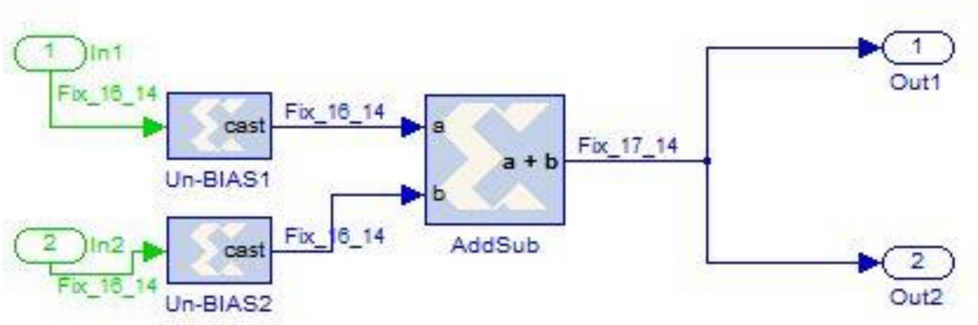


Figure 3.9 – Signal adder block set.

The two first blocks remove the BIAS of the signal, and then, a simple ADD block adds the two signals, the signals are outputted with a gain if in phase with each other, but the gain reduces gradually to zero as the signals get more out of phase, being completely canceled if 180º out of phase..

3.2.2 – Signal adder with noise cancelation

On situations where a hearing aid user is waiting to be engaged in a conversation or other type of sound interaction, there is a waiting period with background or static noise that is of interest to be rejected. For this reason, a noise cancelation technique was implemented.

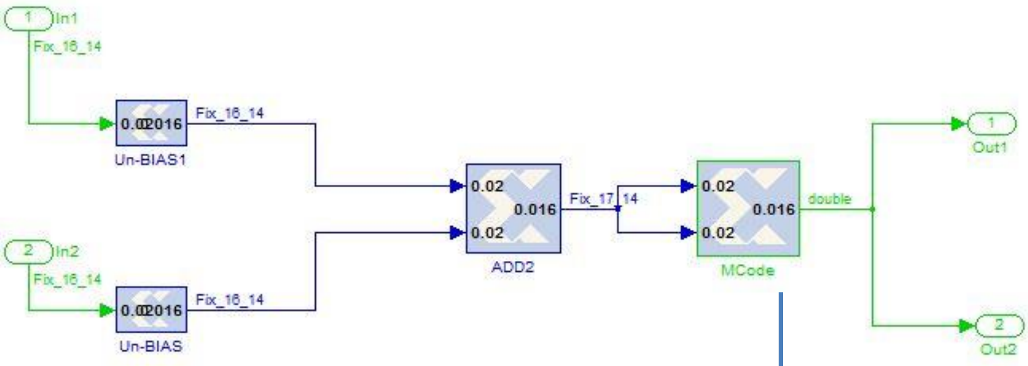


Figure 3.10 – Signal adder with noise cancelation block set.

```
function z = phase(x, y)
  if (-1 > x) || (x > 1)
    z = y;
  else
  z = 0;
  end
```

Figure 3.11 – Noise cancelation code.

The numerical values on figure 19 represent the boundaries of the amplitude that the user can change. If the signal is within boundaries, it goes through to the output, if not, the output is 0.

These values must be changed inside the VHDL in order to configure for rational numbers.

3.2.3 - Signal inverter for angle control

Since the attenuation of the output signal depends on how much the signals are out of phase, it is not possible to have an exact angle value of tolerance like in the signal correlation scenario studied before.

The alternative found was the following.

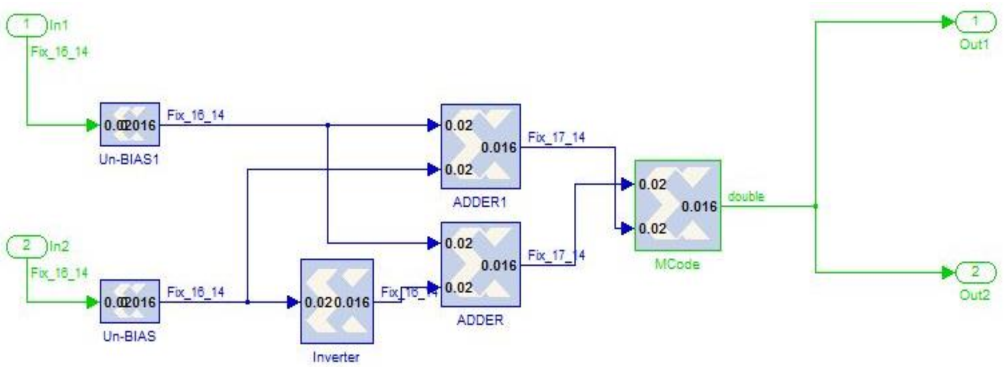


Figure 3.12 – Signal inverter for angle control block set.

On this situation, the signal 2 is inverted and then added to the signal 1, what happens in this situation is the opposite of the normal adder. If the input signals are in phase, the signals will be cancelled, if they are out of phase, their amplitude will be doubled.

This was used to test how much out of phase these signals are. Inside the Matlab code block, a test was made. If the signals after the inversion and adder are above a certain threshold they would be rejected. This means that the signals are too much out of phase, if the signals are inferior to the threshold, then, the original non inverted signals will be outputted after being added like in the normal signal adder blocks.

In the situation where the signals are completely in phase the addition of them after one of them is inverted will be always 0. Therefore, this situation always passes the test.

The user on the practical implementation can change this numbers remotely.

These values must be changed inside the VHDL in order to configure for rational numbers.

```
function z = phase(x, y)
  if (-1 > x) || (x > 1)
    z = 0;
  else
  z = y;
  end
```

If after adding the inverted signal 2 to signal 1 and the result is within the boundaries, we output our normal added input signals 1 & 2.

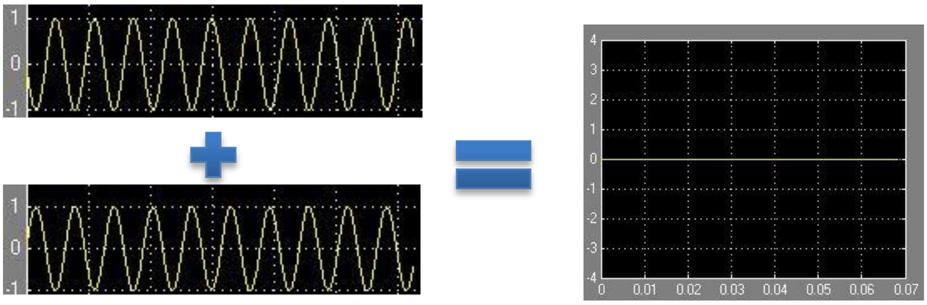


Figure 3.13 – Phase inverter code and explanation.

3.2.4 – Signal inverter for angle control

Hearing aid users have different types of hearing handicaps, and a lot of those vary in the frequency range. For this reason was considered valuable to implement a low band pass filter and a high band pass filter. Both of this can act has band pass filters depending on the values of the constants.

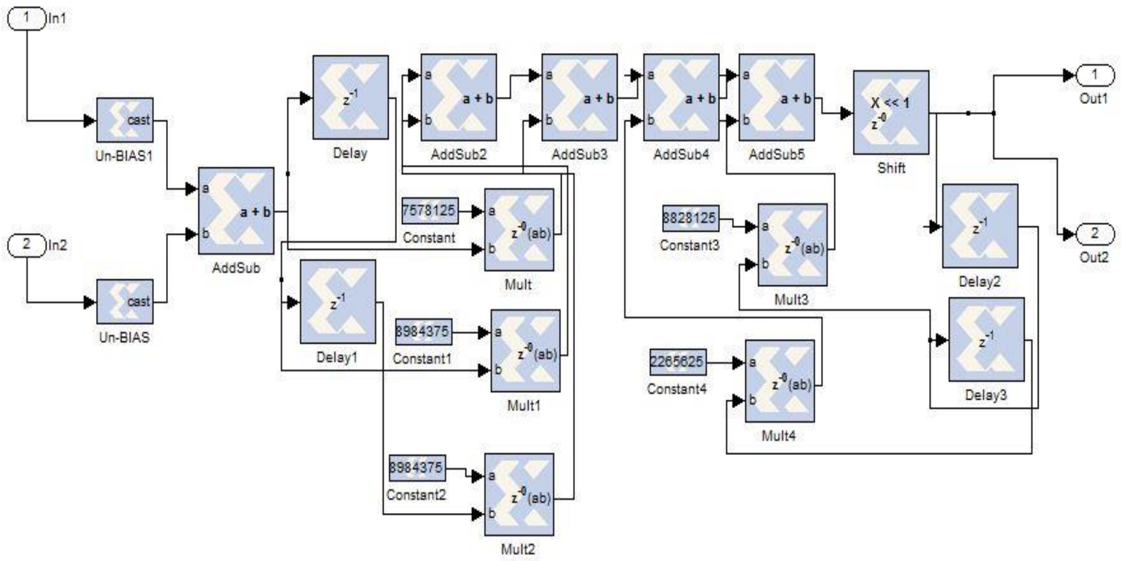


Figure 3.14 – High band pass filter block set.

The values that can be changed are the “constants” and the number of constants are related to the order of the filter. In this example the filter is of order 6, and the values for the constants were gathered from the FDA tool available on the Xilinx block set showed below.

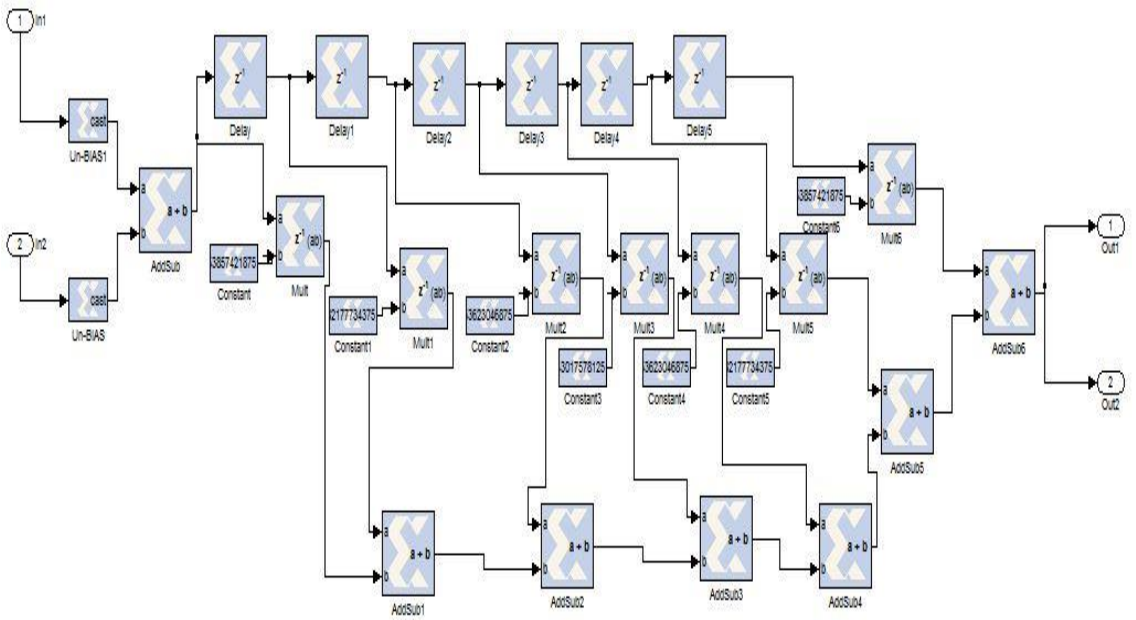


Figure 3.15 – Low band pass filter block set.

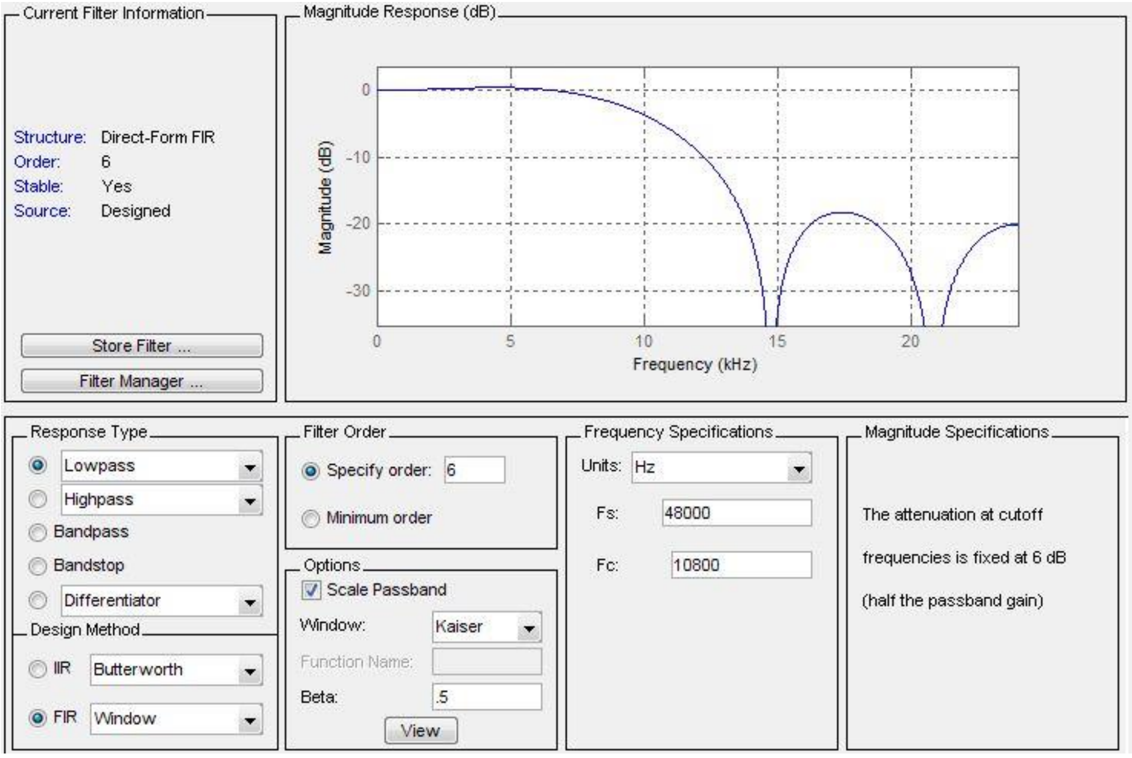


Figure 3.16 – FDA tool.

Human voice is located below the 4000 Hz range. Therefore the configuration of the low band pass filter had a cut off around this value.

The high band pass filter was tested for frequency above 10 kHz.

Because this section is not the main focus of this project, not a lot of testing and improvement was made, but it still was interesting to see how these filters interact with the directional sound techniques studied.

4

Results & Analysis

Two different tests were made for all the scenarios, one was on real time, using the cardioid microphones, and other was using two mono .wav files, the first file was the recorded voice of an individual saying 10 sentences and the second file was the first file duplicated where each one of the sentences were out of phase 20 degrees in relation to the previous one. This was achieved using a plug in for the software Qbase.

The graphic of these two files is found on the image bellow.

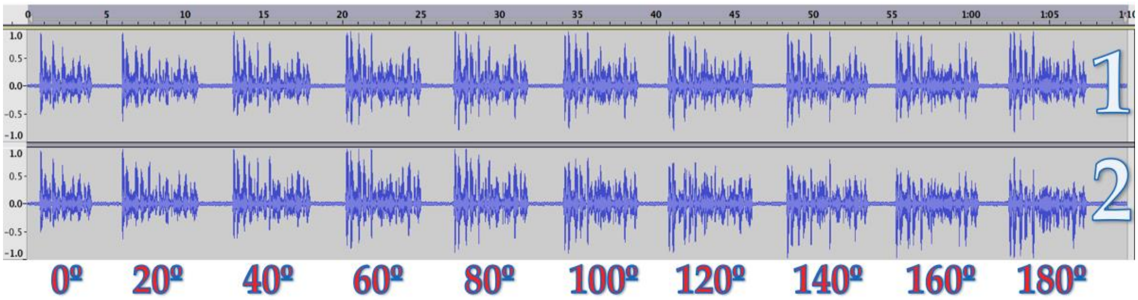


Figure 4.1 – Graph of the two mono .wav files. 1-Left input 2-Right input with signals out of phase from left input.

The output of these tests would be a cable connected to a computer Line In port. Using Audacity this computer would capture the processed signals.

These tests were applied to all scenarios, from signal correlation to the signal adders using the FPGA connections.

The sample rate of these files was of 44,100 with a 16 bits per sample. This way the quality loss was neglectable.

The first file would always be the left input and the second file the right input.

It is important to mention that only one channel is shown in the output because the output signal is mono, therefore the left and right signal are the same in all scenarios.

4.1 - Matlab/Simulink signal correlation results & analysis

As explained before, the results obtained using the model from figure 9 had very low quality. This was due to the fact that the computer could not process the code in real time without parts of the input signals being skipped, and delay between the comparison of the signals was present.

The solution for this was testing signal correlation using the model on figure 3.3.

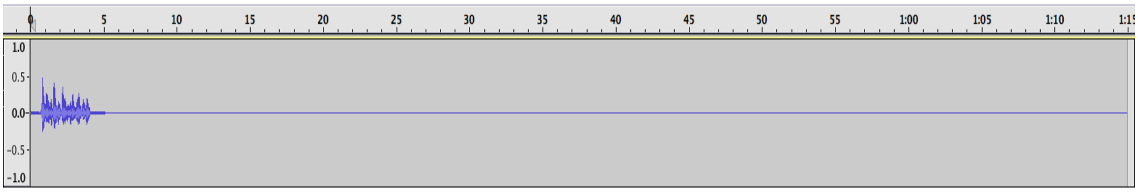


Figure 4.2 – Matlab output after signal correlation with 0rad phase tolerance.

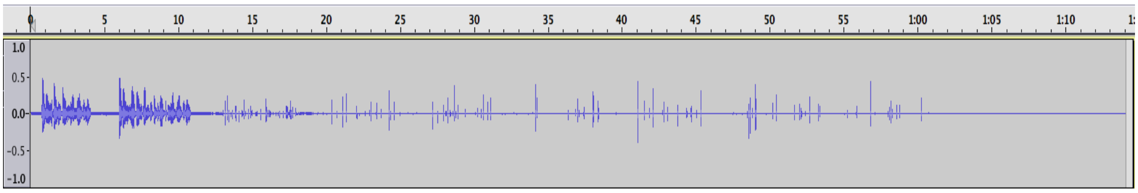


Figure 4.3 – Matlab output after signal correlation with 0.5rad phase tolerance.

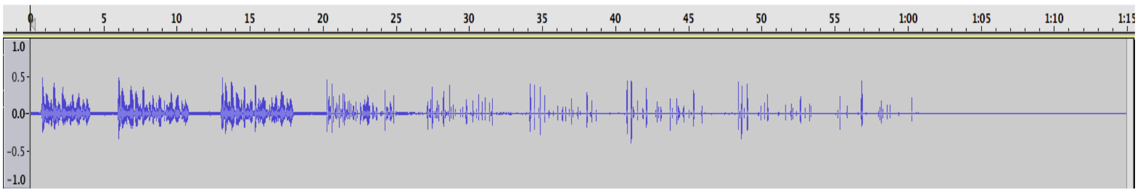


Figure 4.4 – Matlab output after signal correlation with 1rad phase tolerance.

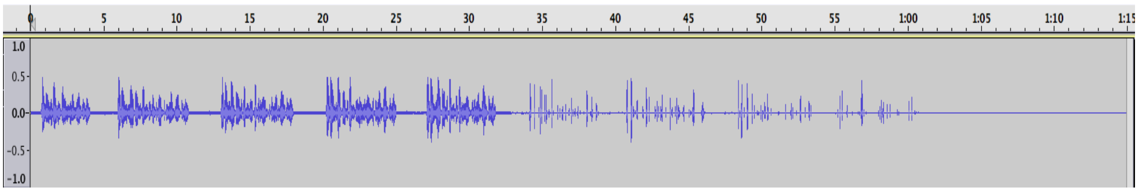


Figure 4.5 – Matlab output after signal correlation with 1.5rad phase tolerance.

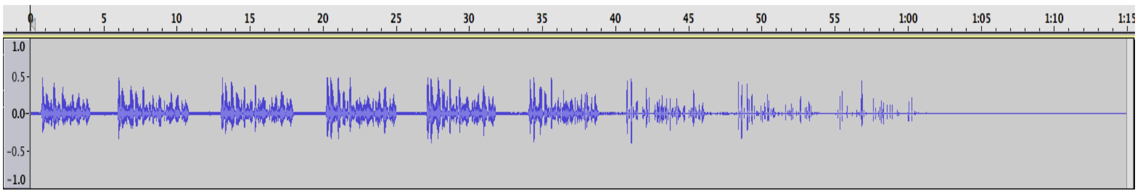


Figure 4.6 – Matlab output after signal correlation with 2rad phase tolerance.

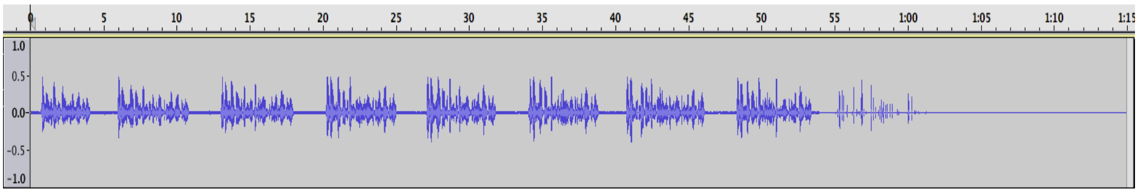


Figure 4.7 – Matlab output after signal correlation with 2.5rad phase tolerance.

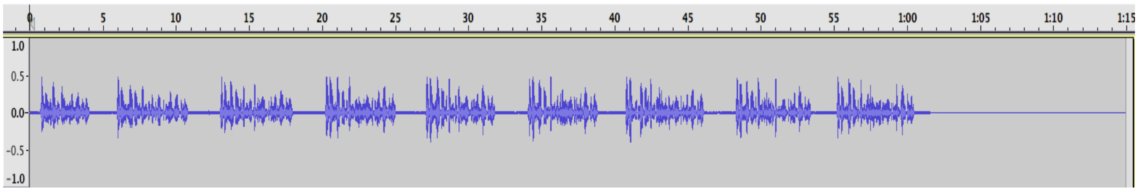


Figure 4.8 – Matlab output after signal correlation with 3rad phase tolerance.

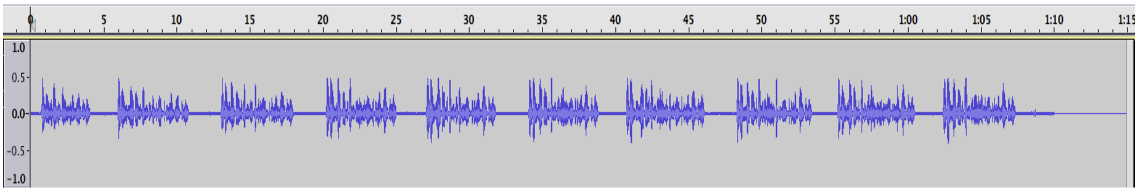


Figure 4.9 – Matlab output after signal correlation with $>\pi$ phase tolerance.

The results obtained were impressive, since the level of angle control was nearly perfect; with just some artifacts occurring between the phase's discrepancies between 20º and 160º. These artifacts are easily removed with noise filters.

The quality of the sound was lossless with the 0rad and the >pi phase tolerance, but some quality was lost on the other tests.

Among all tests, signal correlation proved to be the most efficient directional sound capturing technique, but also the one that requires the most processing demands.

Real time tests had poor results, but the design of a specific circuit for this implementation would solve the problem.

4.2 – FPGA implementation results & analysis

The audio files sound was transmitted to the FPGA the same way the microphones are connected, through the LINE_IN port. This cable was connected to the line out port of the computer and the AMP_OUT port of the FPGA was connected to the line in port of the computer.

The signals were captured by Audacity.

The results from the tests performed using the microphones and the speakers are, naturally, subjective, but a group of individuals gave the opinion on the results being their average response considered unanimous, and therefore stated here as valid results.

4.2.1 - Signal adder

The results of the signal adder were expected and satisfying, with completely less loss quality and a considerable attenuation of the signals out of phase, as shown in the image bellow.

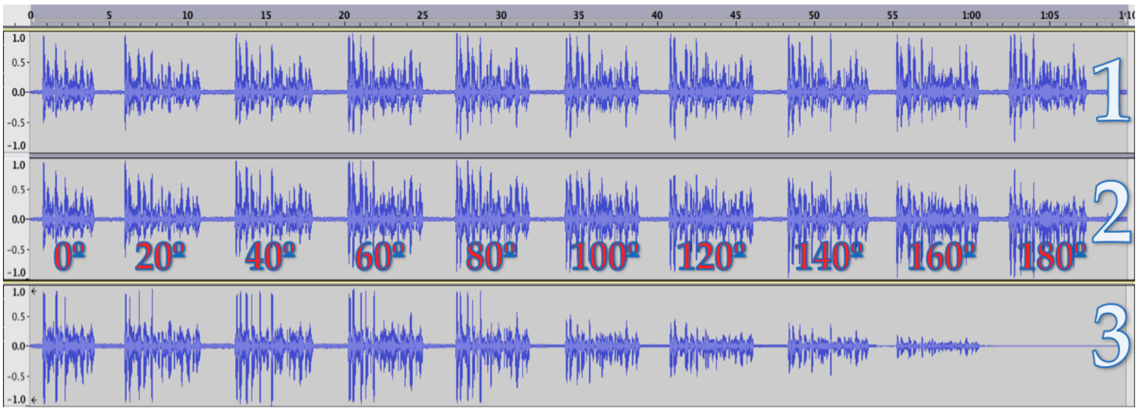


Figure 4.10 – 1-Left input 2-Right input with signals out of phase from left input 3-Output from the FPGA after Adder.

A gain is present, this happened because the signals were not divided after the adder; this was intended since the goal was to try to simulate the amplification of the signal by a hearing aid.

It is clear in the graph that the signal starts to attenuate considerably at 100º, and it is completely rejected at 180º, exactly as expected.

The results obtained by the real time tests using the microphones and the speakers were even more satisfying, since the microphones already have a big angle of rejection by default. The angle of capture was even lower, with a big attenuation above approximately 40º of capturing.

A special test was made where speakers playing loud music were positioned next to the right of the microphones, and an individual spoke in front of them.

The result was that the voice from the individual was completely clear, and the loud music coming from the speakers was attenuated almost to a complete rejection.

Signal adders proved to be efficient in directional sound capturing, but without the possibility of exact angle control.

The great advantage of this technique is that it requires almost zero processing power.

4.2.2 - Signal adder with noise cancelation

The results obtained from these tests determined that this technique is not adequate to use during conversation. Instead, on a situation where the user is on a loud environment waiting to be engaged in a conversation, since when this situation occurs, the user can set the boundaries of the signal rejection to 0, and the system will operate as usual.

The first test with the noise cancelation technique implemented was with the normal signal adder with the same out of phase signals used before, and the results are shown below.

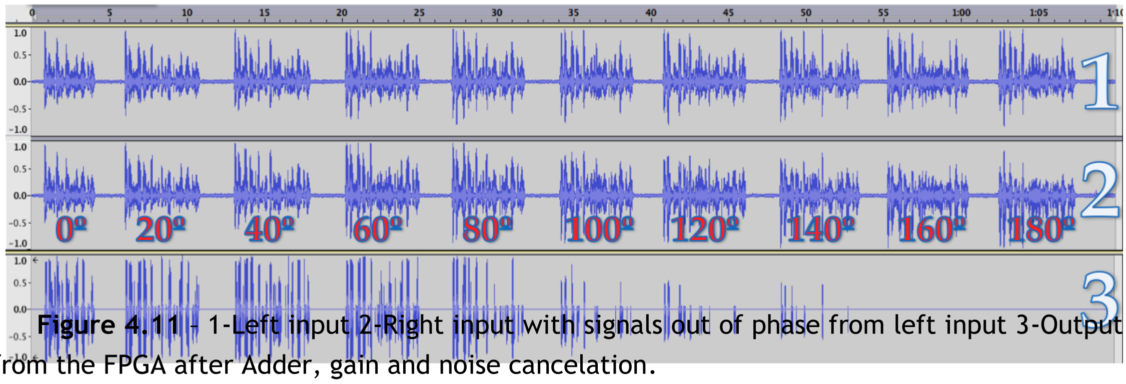


Figure 4.11 – 1-Left input 2-Right input with signals out of phase from left input 3-Output from the FPGA after Adder, gain and noise cancelation.

This technique is not suitable for conversation because human voice is very rich in harmonics that have much more amplitude than the rest of the signal. For this reason, another test was made with a sound from a cymbal swell that increased gradually in amplitude.

This test represents random noise rejection of an environment composed by sounds other than human voice.

The results are shown in the image below.

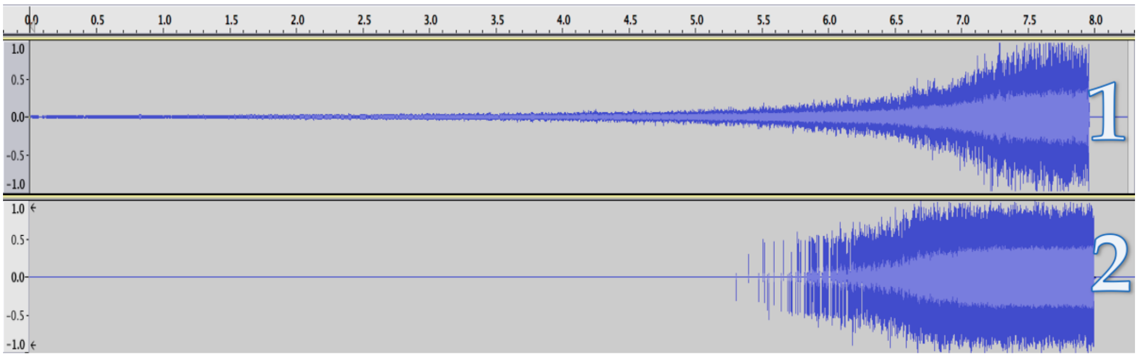


Figure 4.12 – 1-Cymball swell increasing in amplitude 2- Output from the FPGA after Adder, gain and noise cancelation.

For this purpose the noise cancelation technique is useful, where the signal does not vary so much in amplitude.

The tests made on real time where satisfying. Where static noise was completely canceled, and if the threshold was high enough, surrounding noise was also, completely rejected.

No plans are made for implementing this technique in the potential final product, since hearing aids nowadays already have better noise cancelling techniques, so it is not necessary to develop this one further.

4.2.3 - Signal inverter for angle control

On this test the block set present on figure 20 was implemented and the results are shown below.

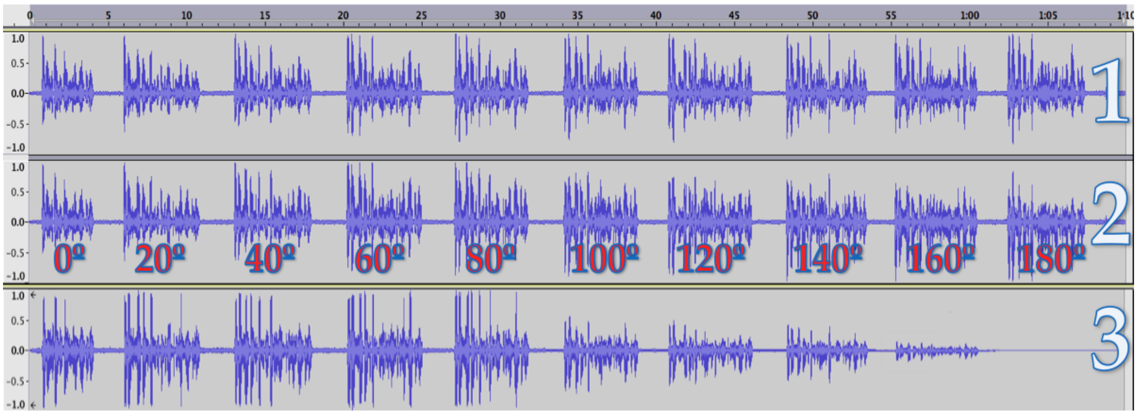


Figure 4.13 – 1-Left input 2-Right input with signals out of phase from left input 3-Output from the FPGA after signal inverter for angle control.

The threshold on this test was very high. Despite this fact, comparing to the simple adder it is possible to notice a higher reduction in the outputs from 140º and 160º, this is due to the fact that after the addition of the original signal and the inverted signal, the harmonics surpass the threshold, being rejected. This is a satisfying result, because opposite to the noise cancelation technique. This one doesn't keep the harmonics but rejects them. This makes the signal still audible but with much less emphasis.

The results were especially noticeable on the real time tests, where the sound that was around the microphones decreased even more in amplitude when compared to the usage of the adder alone.

5

Other Issues

5.1 – Head-related transfer function (HRTF)

This topic will be particularly important to study when the techniques above are to be implemented in actual hearing aids; the reason for this is because the choice to use one or two hearing aids will affect how the microphones capture the sound and how that will affect the phase of the signals.

If the user only uses one hearing aid, the two directional microphones need to be very close to each other, in this scenario, they must be extremely precise because the gap of time that the sound takes to reach the second microphone after reaching the first one will be much smaller.

In the other scenario where each microphone is located in each ear, more tests need also to be made, because the head will be in the middle of the two, this will create a considerable delay and attenuation on the signals. This is not necessarily bad, it can be even beneficial, but for that, binaurally studies must be made.

5.2 – Frequency filters

Since the frequency filters where not the main topic of study of this project they were not developed to a great extent, but they are important for the final implementation, since the users have different needs of amplification in different frequencies, and this will affect how the amplitude of the gains.

Ideally each hearing aid must be configured for each individual, but this increases implementation costs. A solution would be to have also an adjustable band, where the user could change the gain on each frequency, but tests regarding safety and easy of use should also be made.

5.3 – Circuit board for signal correlation

Signal correlation was the test that had the best results; the amount of control over the angle of capturing was nearly perfect. For this reason, this is the most valuable result of this project. Signal adders are effective, but they are already used in present technology, the next step is to design an embedded circuit to process the code for signal correlation.

This circuit must be capable of doing two fast Fourier transforms and the respective following operations in real time, without delaying the output of the signal.

If successful this might very well be the future of directional sound capturing not only for hearing aids, but also for other applications that can take advantage of directional sound capturing.

5.4 – Tests on hearing handicaps individuals

On a final stage of development, it will be necessary to test the prototype on users that need hearing aid devices. This will be valuable to compare the efficiency of the implemented techniques on already accustomed users of present hearing aids.

Probably the best usage is to merge the already existent hearing aid technologies with the signal correlation technique developed in this project.

Conclusion

In terms of efficiency the most effective approach was the signal correlation, the amount of angle rejection was maximum. This is due to the fact that the analysis was numerical, so it left no space for error if the signals could be completely processed. This alternative is excellent compared to the current technologies used on the market, and is of best interest to explore.

The signal adder proved to be effective as expected, and the technique of signal inversion for angle control is simple but practical, allowing for a control of angle capturing to an extent. To increase the angle of rejection using the signal adders, one would have to increase the number of microphones used. It is useless to increase the number of adders because after the addition is made one time, the signal looses the difference in phase so a new addition would only increase it in amplitude, and adding the input signal again would just create delay and errors on the signal, but with more microphones we have more inputs and the signals out of phase can be attenuated even more.

After testing on a real life situation with the cardioid microphones the results were very satisfying, with the individuals that witness the results agreeing that at least a 40º angle of capturing was achieved.

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