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PHASED ARRAYS OF MICROPHONES – SOUND LOCALIZATION

UNDER THE SUPERVISION OF PROF. WALID ALI-AHMAD

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ABSTRACT

Phased Arrays of antennas are devices that capture electromagnetic waves coming from a certain direction, by using the concept of phase construction and destruction. This concept was extrapolated to sound waves, in order to generate a device based on phased arrays of microphones that capture a sound coming from a certain direction, while ignoring other less significant sources, emanating from other directions. A practical application would be a speech tracking device for a conference room.

Previous research has shown that this extrapolation led to poor results, mainly because of the wideband characteristic of sound, and the poor directivity of the designed phased arrays, especially at the array boundaries. Moreover, all previous designs focused on static and preprogrammed arrays.

To solve these issues, a device that contains three phased arrays of microphones, each with two omni directional microphones at the edges and a cardioid microphone at the center, disposed in a triangular structure was proposed. The scanning algorithm used is dynamic and continuous: it detects and amplifies the loudest sound source in the room, and repeats the process automatically when the main sound source changes. Theoretical simulation of this device predicted a high directivity main lobe (-10 dB at 20° from the lobe’s center).

We have simulated a single sub-array using a single tone sound source. This sub-array covering 120° accurately detected the sound source in 5 different regions (24° each).
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CHAPTER 1: INTRODUCTION

Our project is entitled “Phased Arrays of Microphones – Sound Localization” and is supervised by Prof. Walid Ali-Ahmad.

Moreover, this project has been part of the 5th FEA Student Conference and in the Virtual Instrumentation Academic Paper Contest- National Instruments which is in conjunction with the Annual Arabia Academic Event. This technical paper contest aims at showcasing the best NI-based application papers.

This first chapter introduces the project problem and its scope. The practical applications to this problem and the previous attempts to solve similar issues (literature survey) follow. The report then focuses on the analysis design and implementation stages. It finally gives a thorough analysis and appraisal of the work performed. Finally, conclusions and report overview are stated.

A. PROBLEM STATEMENT

This project can be divided into three main problems. The first is to constantly localize a main sound source among many. The second is to capture the sound emitted by that source and amplify it, while ignoring noise and other less significant sources of sound. The last is to ensure an automatic transition among the sources.
B. SCOPE

The problem described above is tightly related to the Audio Engineering field (speech, sound wave propagation and characteristics, microphone technology, room acoustics, etc.). It also requires knowledge in Wave Theory (beam forming, phasing, etc.) as well as Signal Processing (sampling, analog/digital signals, noise processing, etc.). As indicated, this project allows us to apply acquired theoretical information in finding an optimal solution for a given problem.

In addition, the implementation phase of this project allows the use of many hardware and software components. The main software tools are Matlab (for simulation) and Ni LabView (for implementation), while the hardware counterpart encompasses microphones, preamplifier, connection box and DAQ.

C. PROJECT SUGGESTION

Many reasons drove us to choose this research topic. First of all, the technical aspects of this project (audio engineering, wave theory and signal processing) are of great interest to us. Furthermore, the idea of sound localization is becoming a predominant subject in several engineering applications that we will describe in the next section.

The main idea for this application came from Prof. Ali-Ahmad who suggested a possible design strategy using the concepts of phased arrays, initially used in antenna design.
D. PRACTICAL APPLICATIONS

The concept of sound localization and differentiation has many applications that belong to various fields. On one hand, it has been implemented to guide the visually impaired [1] and the hearing impaired [2]. On the other, it can be used to develop a speech tracking device for conferences. This equipment will automatically localize the individual who is speaking and amplify his/her speech while ignoring sounds coming from other directions (noise and whispers). It will also provide a smooth transition among speakers by scanning the room to detect the current speaker. When the person finishes talking, the procedure is repeated.

E. PREVIOUS ATTEMPTS

The concept of phased array has been previously applied to antennas [3] as well as microphones ([4] [5] and [6]). Yet, only a single array has been used in these implementations, but the theoretical results are very encouraging: a very narrow directivity was achieved. However, in all these cases, the direction to be amplified was predetermined. The proposed design aims at adding synchronization among several arrays, as well as obtaining a dynamically changing directivity.

Some other concepts, related to robotics and video processing, have also been used to precisely localize sound. However, these techniques require additional cost and processing overhead.
F. REPORT OVERVIEW

Section 1 provided a brief introduction to the chosen research subject.

Section 2 tackles the research conducted. It contains the review of the theoretical background and survey literature used to design our solution.

Section 3 describes analysis and design alternatives and decisions as well as budget considerations.

Section 4 showcases the implementation of the proposed design.

Section 5 evaluates the obtained results and provides a critical appraisal of the proposed solution.

Section 6 summarizes the report’s findings and provides guidelines for future research.

The appendices of this report contain simulation codes we have developed as well as datasheets of hardware components.
G. PROJECT TIMELINE

The following timelines summarize the main milestones of the project.

**Figure 1: Fall timeline**

**Figure 2: Spring timeline**
CHAPTER 2: LITERATURE SURVEY

After introducing the scope of the project, the collected literature survey is presented. The review starts with a brief overview of phased arrays, which includes its underlying principles, restrictions and effects. Afterwards, the mechanisms, efficiency and enhancements of previous experiments are showcased. Then, literature related to room parameters and noise effects and microphones is summarized.

A. PHASED ARRAYS

According to research, an efficient sound localization technique uses the concept of phased arrays of microphones [7]. This section will describe the basics behind the idea of phased arrays, the constraints to which it is subjected and its beam forming effect.

1. BASICS

The following is a simplified model of the theory behind the phased arrays, as described in reference [3].

Consider figure 1, representing K receivers, where a plane wave is incident under an angle θ (i.e. the angle formed by the wavefront and the array is θ).

Because the wavefront is not parallel to the array, the wave will not reach the elements at the same time. Assuming the receivers are equally spaced by a distance d and the wave arrives at receiver K, it has to travel an additional
distance of $d\sin\theta$ to reach receiver K-1, $2d\sin\theta$ to reach receiver K-2, and so on.

![Figure 3: Phased Array Reception](image)

Given that the waves are periodic in time and space, they will be received at different receivers with different phases. For each angle $\theta$, a different combination of phase differences is obtained. Thus, the output of each array element can be phase shifted so that the sum of all the outputs of the elements gives a constructive interference for a given $\theta$, whereas for other angles, the sum will be negligible due to destructive interference.

The **receiver spacing** plays a crucial role in the performance of the array. Visser [3] states that, “the main beam of the linear array antenna gets smaller when the elements occupy a larger area”, which means that the directivity gets narrower as the spacing increases. However, the spacing cannot be increased
indefinitely because exceeding a certain critical spacing will introduce additional main beams (this critical spacing is presented in section 3).

Because of the analogy between electromagnetic waves and sound waves, a phased array of microphones can be proposed.

2. FAR FIELD

The concept explained above relies on the fact that the incoming sound waves have planar wave fronts. This assumption is only valid when the sound wave receiver is located in the far field region of the transmitter. The boundary for this region is at:

\[ d = \frac{2L^2}{\lambda} \]  [7]

Where: L is the aperture of the sender and \( \lambda \) is the wavelength of the incoming wave.

3. BEAM FORMING

Beam forming is the process of directing an array of microphones into receiving all incoming sound sources from a given direction \( \theta \). Beam forming techniques are characterized by [7]:

**Their data dependence:**

Beam forming techniques can be either adaptive or fixed. Adaptive methods update their parameters based on received signals, while fixed methods have
predefined configurations.

**Their noise assumptions:**

The result of each beam forming technique is strongly dependent on the type of noise present. Some of them perform better with a specific type of noise while others can handle different noise types.

**The array configuration:**

As mentioned in the previous section, the spacing of microphones plays a very important role in the directivity of the array. Each beam forming technique uses a specific microphone spacing which strongly affects the directivity of the array.

**B. SOUND LOCALIZATION**

As mentioned in section A, the concept of phased arrays can be used for localization. In this section, the mechanisms of sound localization, and the efficacy of previously designed systems are assessed.

**1. SOUND LOCALIZATION MECHANISM**

Directional auditory perception is innate in humans and animals. This sensing ability is greatly dependent on the organs’ functioning, especially the two ears, as well as the spatial behavior of sound waves. In order to artificially reproduce this phenomenon, we need to delve into the processes that dictate such a behavior [4].
We are able to distinguish sounds coming from the left and right sides, and we are also able to determine if they are emanating from above or from below. This indicates that sound localization thus depends on the azimuth and the elevation angles: this gives a 3D function. If we only take into account the azimuth angle, we get a 2D function [8].

First of all, two important concepts are used to localize sound waves: the inter-acoustic differences in phase and differences in intensity level between receptors [4] [5] [6]. In fact, since the sound waves arrive at different microphones at different times we can estimate their position using the concept of phased arrays.

Moreover, data acquisition greatly affects the array’s performance: in fact, the analog vibrations producing the sounds that are recorded by the microphones should be converted into digital information. This conversion requires the use of Nyquist Theorem as well as a proper choice of sampling rate [4] [6].

Once the digital data has been obtained, it should be analyzed: the Fast Fourier Transform is used to detail the frequency content of the recorded speech [4] [5].

Furthermore, experiments have also shown that for a given fixed microphone spacing, increasing the number of microphones reduces the acoustic bandwidth of the array [9]. In other terms, as the number of microphones increases, the array becomes more frequency selective, and picks a smaller
frequency range in a given direction. This means that we will have to choose the number of microphones, large enough to provide a narrow directivity, and small enough to provide a bandwidth that accommodates the human voice range.

Finally, wave diffraction is usually ignored when the sound source is considered to be far enough to be considered propagating in a planar front.

2. EFFECTIVENESS OF BUILT SYSTEMS AND RESULTS

Many systems have used robots in order to dynamically localize a moving source and follow it [4]. They involve the use of image processing in order to obtain an accurate localization. This is worth mentioning, yet it is out of the scope of our project.

The theoretical results are generally 100% correct yet the experimental ones are poor. In fact, some conducted experiments have generated poor results because of aliasing, noise and reflections [8].

C. EXTERNAL PARAMETERS

External parameters were the main reason for the failure of most designs. The two most important factors are room acoustics and noise effects.

1. ROOM ACOUSTICS AND ENVIRONMENT

Any is usually divided into 4 regions according to frequency-band behavior:
Region X: frequencies between 0 and \( f_X = \frac{c}{2L} \) where \( L \) is the largest dimension of the room. Those frequencies cannot be captured by the microphone.

Region A: frequencies between \( f_X \) and \( f_A = 11250 \frac{T}{\sqrt{V}} \) where \( V \) is the volume of the room in feet\(^3\) and \( T \) is the reverberation time. This region is characterized by normal modes, which are resonating frequencies that depend on room dimensions.

Region B: frequencies between \( f_A \) and \( f_B = 4f_A \). This region is characterized by dominant diffraction and diffusion.

Region C: frequencies larger than \( f_B \) characterized by reflection: sound waves behaving like ray of light (array phasing is the most efficient in this region).

\[ \text{Figure 4: Frequency division of rooms} \]
2. NOISE EFFECTS

Noise interferes with the information signals, creates distortion, and yields incorrect practical results [11].

Possible noise sources are:

1. Mechanical equipment: fans, motors, machine vibration, office machinery and equipment
2. Self noise from air motor and turbulence within air conditioning system
3. Cross talk from one room to another
4. Transmission through walls, ceilings from other neighboring rooms
5. Noise from external sources: rain, fog, transportations and vehicular noise…

On the other hand, microphone arrays are typically subject to three main categories of noise. These categories are defined on the basis of the correlation between noise signals at different spatial locations [7]. These three types of noise are:

**Coherent noise:**

Coherent noise fields are noise signals that travel directly from their source to the microphone array. They are not subject to reflections or diffusions due to
the surrounding environment.

In coherent fields, noise inputs at the microphone array are strongly correlated.

**Incoherent noise:**

An example of incoherent noise is the self noise of the microphone array and processing equipment. This noise, also called spatially white can be considered as random, thus having a correlation very close to zero.

**Diffuse noise:**

Diffuse noise fields receive weakly correlated noise that has approximately the same energy everywhere in the environment. Most incoming noise can be characterized as diffuse.

Noise can also have several colorations [9]:

**White noise:**

The intensity of the power spectral density is constant and independent of frequency along the whole spectrum

**Pink noise:**

It has a uniform power density for a relative bandwidth generally octave. There is a -3dB/octave frequency response. It is also called 1/f noise.
D. MICROPHONES

The last step of our review focused on microphone design and characteristics. In this section, we analyze microphone technology and microphone directivity.

The most frequently used types of microphones are the condenser microphone and the dynamic microphone. The former operates based on a capacitor whose plate vibrates when exposed to sound waves, creating changes in voltage, whereas the latter is made of a coil inside a magnetic field, which will vibrate, generating electric current, whenever exposed to sound waves. The condenser microphone is much more sensitive to small pressure variation, much more fragile, and much more expensive. Because in our project we will be dealing with average human voice, we do not need to take advantage of the high sensitivity offered by the condenser microphone, so we will basically use dynamic microphones.

Directivity is another factor that should be mentioned. The following figures show different types of microphone directivity.
Figure 5: Omnidirectional

Figure 6: Bidirectional

Figure 7: Cardioid
Figure 8: Hypercardioid

Figure 9: SuperCardioid

Note that directivity increases as frequency increases.

E. RESEARCH CONCLUSIONS

After doing the necessary research and literature survey, we now have the sufficient knowledge to design and implement a solution for our stated problem.
This research shows that no previous attempt has been made to solve our particular issue, and any closely related design has practically failed due to simplified design (single linear array) and lack of noise consideration [11].
CHAPTER 3: ANALYSIS AND DESIGN ALTERNATIVES

After giving all the necessary background information, we now apply the reviewed concepts. This chapter is dedicated to the analysis phase of our project. We start by studying wave behavior and reception by an array of microphones. We present in detail MatLab simulations which are intended to emulate the response of the arrays. We also study the human voice characteristics with LabVIEW experiments in order to reach a proper modeling that will be used in the implementation part.

A. SINGLE PHASED ARRAY SIMULATION

Based on the technique developed in reference [3] and discussed above (Ref: Basics of Phased Array Antennas), we decided to develop many simulations involving an array of microphones. They are intended to give us a better idea about the number of microphones to be used, their geometrical disposition, the spacing between them, and their directivity. We have reached conclusions by varying different parameters.

1. SIMULATIONS EXPECTATIONS

Before going into the details of the simulation, it would be useful to present the expected array performance. Firstly, as the directivity gets narrower, the array should guarantee more focus on the set direction, and this by giving less emphasis on other sound sources present in other directions. Secondly, the
greater directivity we can achieve, the more selective our device will be. Moreover, since the sounds to be processed are mainly human voices, we need to adapt our system to a wide range of frequencies. Finally, the device will most probably be placed in the center of a room, and we thus need it to cover the entire plane (from 0° to 360°). However, note that in the next simulations, we consider only 180° for reasons that will be explained in the interpretation section.

2. IDEA BEHIND THE ALGORITHM

Each element in the array will receive the wave with a different phase, and this particular phase depends on the angle of the wavefront (assuming, of course, that the source is far enough for the wave to be considered as a plane wave). Referring to Figure 1, the wave has to travel an additional distance of $d \sin \theta$ to reach microphone K-1. Assuming a wave velocity of 350 m/s, this corresponds to a delay of $\frac{d \sin \theta}{350}$. This time delay can be converted to a phase delay, using the following formula: $\frac{t}{T} = \frac{\phi}{2\pi}$, where $t$ is the time delay, $T$ is the period of the wave, and $\phi$ is the phase delay. Thus, if we consider the phase at receiver K to be equal to zero, then, the phase difference at K-1 is $\phi \theta$ (keep in mind that, for each $\theta$, we have a different $\phi$), the phase difference at K-2 is $2\phi$, and so on…

If we want optimal reception at an angle of $x$ degrees, we have to subtract
from the output of receiver K-1 a phase angle of $\phi x$, from receiver K-2 a phase angle of $2\phi x$, and so on… This way, the outputs of all the receivers will be in phase, and their sum will be exactly the same as the input wave, with the amplitude multiplied by K.

The idea of the simulation is to vary the direction of the input wave, from 0 to 180 degrees, given a certain reception angle, and observe the directivity pattern. It is no other than the amplitude of the sum of the K waves divided by K, computed for each tenth of a degree between 0 and 180. We will next present the waveforms obtained, which will help us draw some useful conclusions. The MatLab codes of the simulations can be found in the appendix.

3. SIMULATIONS RESULTS

Case 1: The varying parameter is the number of microphones in the array. The arrays respectively 2, 3 and 4 microphones are set to receive a 500 Hz wave at 60 degrees. The following graph shows the results from 0 to 180 degrees:
As can be seen, increasing the number of microphones gives a higher directivity.

Case 2: For 4 microphones in the array, we will vary the spacing between the microphones. The array is set to receive at 60 degrees, with a wave of 500 Hz.
This figure verifies the statement made by Visser [3], which was discussed above. We can see that, as the spacing increases, the directivity gets narrower, however, after a certain critical spacing, a new main beam is introduced. The critical spacing, for a region between 0 and 180 degrees, is equal to 0.35m or $\frac{\lambda}{2}$.

Case 3: For an array of 4 microphones, a constant spacing of $\frac{\lambda}{2}$, a sine wave of 500 Hz, the reception angle will be varied: the following figures represent reception at 30º, 50º, 90º and 110º.
As one can see, the performance is quite poor at the boundaries. In fact, the directivity gets narrower as we approach 90 degrees.

**Case 4:** Finally, the following set of figures represents the combined effect of varying the wave frequency, for different reception angles. Each figure represents waves of 350 Hz, 450 Hz, 500 Hz, 600 Hz, and 700 Hz, received by an array of 4 microphones, designed to receive a wave of 500 Hz at the specific angle. In other terms, the purpose of this last simulation is to check the optimal reception angle for different frequencies, when the array is designed to receive a wave of 500 Hz at a specific angle. Note that the microphone spacing is still $\frac{\lambda}{2}$. The reception angles are 45°, 60° and 90°.
Figure 13: Varying the frequency at 45°

Figure 14: Varying the frequency at 60°
Figure 15: Varying the frequency at 90°

Figure 16: Varying the frequency at 110°
We notice that varying the frequency changes the optimal reception angle, and this effect increases as we drive away from 90°. At 90°, the reception angle is the same, no matter what the frequency is, however, higher frequencies are received with a higher directivity.

4. INTERPRETATION OF RESULTS

To sum up with, the simulation allowed us first to prove that increasing the number of microphones yields a higher directivity. However, as stated by Loeppert, P. & Wickstrom [10], we shouldn’t increase this number indefinitely, or else the receiver will be a narrowband receiving only a limited range of frequencies at the desired angle: this distorts speech. Furthermore, the element spacing is also an important factor, whose increase will improve the directivity. Again, we cannot increase this spacing indefinitely, because, after a certain critical distance, the main beam will be reproduced, resulting in two or more main beams. For 180 degrees, this distance is equal to \( \frac{\lambda}{2} \).

Intuitively, we can state that as the region to be covered decreases, we can perform several increases in the microphone spacing without introducing a new main beam. In fact, the following graph shows that, for a region of 90°, the critical spacing is equal to \( \lambda \) (in this simulation, the region is reduced to 90°, the reception angle is 45°, and the microphone spacing is increased to \( \lambda = 70 \text{ cm} \)): 
Figure 17: Region of 90°

Finally, we noticed that the performance at the region boundaries is weak in two ways: the main beam gets wider as we approach 0° and 180°, and waves with different frequencies are more distributed around the desired reception angle.

To solve these weaknesses, we thought of using more than one array, in a specific geometrical disposition (triangle, square, hexagon...). This way, we obtain the following advantages:

The room is divided into regions, each processed by one of the arrays. This way, the large beam effect at the boundaries is reduced.

The reduction of the region covered by each array also allows to get rid of the effect of receiving different angles at different frequencies, which is
accentuated at the region boundaries.

Finally, we will also be able to increase the microphone spacing to more than $\frac{\lambda}{2}$, without introducing another main beam. This new distance depends on the size of the portion to be covered by each array, which in turn depends on the geometrical disposition.

B. MULTI ARRAY CONFIGURATIONS

From the previous section, we determined that single sub-array is not a good design strategy, and we need to use alternative configurations.

In this section, we will analyze three dispositions of sub arrays which are: rectangular, circular and triangular, and discuss their advantages and disadvantages.

1. CIRCULAR ARRAY

The circular array consists of six sub-arrays placed in a hexagonal matter. The main advantage of this configuration is that each sub-array covers a narrow region, thus yielding accurate results. However, three main disadvantages make its usage inconvenient: the number of microphones needed (12), the size of the structure due to microphone spacing and the complexity of the phase-delays computations.
2. TRIANGULAR ARRAY

First of all it uses a small number of microphones (6) and has a relatively small dimension. Moreover, it eliminates the wide directivity pattern found at the boundaries of a sub-array (0 and 180 degrees). However, the range of operation of each sub array is still large (120 degrees).

3. RECTANGULAR ARRAY

This setting divides the space into four main regions. It is a good choice for our application due to its directivity (90 degree coverage for each sub array); however, it uses a large number of microphones.

C. HUMAN VOICE SIMULATIONS AND ANALYSIS

The human voice, which will be our main input, contains many varying parameters that make it unique. We need our device to work on all kinds of human voices; what we mean by work is first accurately localizing the region in which the speaker is sitting, and most importantly, reproducing this speaker’s voice with high fidelity. As discussed earlier, the wider the frequency band to be covered, the more considerations need to be taken into account. What follows is an analysis of the human voice, with its dominant and intelligible frequencies.

First of all, we need to visualize the spectrum of the human voice. To do so, we asked ten people chosen at random to speak normally a sample sentence (“Hello I am in the simulation phase of the FYP project. This is to see the spectrum of my
voice”). The voice was recorded using a National Instruments LabVIEW virtual instrument (courtesy of the Communications Lab crew).

**Figure 18: LabView Sound Record**

The figures displayed below represent the voice of one female sample and one male sample.

**Figure 19: Female Voice Sample**
Figure 20: Male Voice Sample

All the other samples are very similar to the ones displayed above, so we will omit their graphs. As we notice, most of the power is located below 700 Hz for the female voice, and below 500 Hz for the male voice (the amplitude is in log scale). These results are confirmed by our literature [9], which states that the voiced speech of a typical adult male has a fundamental frequency of 85 to 155 Hz, and the voiced speech of typical adult female has a fundamental frequency of 165 to 255 Hz.

Based on these results, we will try to determine a frequency range, so that, when applied to a human voice, this voice does not lose any of its intelligibility, and even does not sound distorted.

We started with the LabView virtual instrument that samples and records sounds.
The idea was to modify this module by adding a filter to the sound wave that is being recorded module and listen to the filtered voice samples. After several trials, the range for which the voice is not distorted, for both male and female, is 100Hz – 700 Hz.

Note however that we might modify this range in the implementation part because we will be using more sophisticated microphones than the ones currently available (Discovery Multimedia Headset DHS-613).

**D. DESIGN CONCLUSIONS**

In the previous two sections, we have discussed and analyzed the various alternatives that we can use to solve the stated problem. In this final section, we will indicate our design decisions.

We first determined the number of microphones in each array of our design.

1. **NUMBER OF MICROPHONES**

The number of microphones in each array is equal to 3; this is a good compromise between directivity and frequency band coverage especially that we will be dealing with a relatively large frequency band.

2. **ARRAY CONFIGURATION**

The linear configuration has many disadvantages, and this has leaded us to look for another geometrical disposition. The circular is too large to be
manipulated in practice, and the square involves a total of 8 microphones, we decided on the simple, but yet efficient solution, which is the triangular configuration.

The fourth decision was related to the types of microphones to be used in each sub-array.

3. TYPES OF MICROPHONES IN SUBARRAY

In each sub-array consisting of 3 microphones, the middle microphone should have a cardioid pick-up pattern, while the two side microphones should have an omni directional pattern. Since we are using a structure of sub-arrays, this choice of microphone types is the most suitable one. The omni-directional microphones are shared between 2 sub-arrays, while the cardioid microphone allows each sub-array to effectively capture the sound waves coming from its region of coverage.

Figure 19 is the result of a MatLab simulation where all the microphones of the array have an omnidirectional pattern, whereas Figure 20 shows the result when one microphone is cardioid. As we can notice, the cardioid microphone does not allow the introduction of a new main beam in the pattern. The program simulates a triangular array, with the setting shown in figure 18.

Note that, in order to reduce the amplitude of the side lobes, we put more weight on the output of the cardioid microphone.
Figure 21: Wave coming towards the triangular setting

Figure 22: Design Representation
Figure 23: Result for 6 omnidirectional microphones

Figure 24: Result for 3 omnidirectional and 3 cardioid
The last decision to be made was linked to the spacing between any two microphones in a sub-array.

4. MICROPHONE SPACING

Because we have decided on the triangular disposition, we can slightly increase the spacing between the array elements, without risking the addition of a new main beam. If for 180° the critical spacing is $\frac{\lambda}{2}$, then for 120°, the maximum spacing is $\frac{\lambda}{2} \times \frac{180}{120} = \frac{3\lambda}{4}$.

For a center frequency of 264 Hz, assuming that sound travels at 350 m/s, the maximum microphone spacing is then equal to $\frac{3}{4} \times \frac{350}{264} = 1\, \text{meter}$. However, for practical issues, we will use smaller spacing, if this does not weaken the performance.
CHAPTER 4: IMPLEMENTATION

Now that we have described the theoretical design, completed during the fall term, we turn to the practical implementation of the phased arrays of microphones. This chapter starts by describing the basic experiments conducted before the actual implementation. Then an overview of the implementation is exposed, followed by the hardware implementation strategy. Finally, the software implementation algorithm is described in detail. Our implementation was done using the NI LabView software.

A. EXPERIMENTAL ANALYSIS

This first section describes the work we have completed before starting with the actual implementation of the phased array of microphones. Since there is no previous work to base ourselves on, we had to start the implementation from scratch.

Because our equipment didn’t arrive until mid April, we had to find out inventive ways to start our implementation. In the following subsections, we describe the basic pre-implementation experiments conducted.

1. PHASE CANCELLATION VERIFICATION

This is the first practical experiment conducted. We decided to start with the main processing box of our design which is the phase shifting module.
**Goal:** Design a phase shifting module on LabView.

**Description:** This experiment was conducted using a single microphone. We used an available VI (LabView function) which stores the sound recorded by the microphone into an array. In order to phase shift that signal, we deleted a predefined number of samples from the array (this concept is explained in the software section) and added that modified array to the original one, and saved that array as a sound file.

**Results:** As expected, the phase shifting process worked. We recorded a single tone 400Hz, and stored it in an array using the given module. When we removed a number of samples equal to a 180° phase shift, thus building two destructive waves and adding them together, we cancelled the sound wave.

The following figure shows the recorded waveform of the sine wave before phase shifting:

![Figure 25: recorded 400Hz sine wave](image)

The next figure shows the sound generated after adding the original recorded signal and the 180° phase shifted signal. As you can see, the recording is
complete almost 0.

![Figure 26: recorded sine wave after phase cancellation](image)

**Conclusion:** From this experiment we conclude that our phase cancellation VI is effective. However, we still have to incorporate it in a dynamic setting, i.e. phase shift the incoming signal as it is received.

### 2. MICROPHONE ARRAY FOR SPEECH

Following this first successful trial, we decided to investigate the effect of this phase cancellation on speech.

**Goal:** Notice a phase cancellation while summing two phase shifted versions of the same speech using two separated microphones.

**Description:** We used two low quality microphones that we had at our disposition, along with a LabVIEW virtual instrument. The virtual instrument in question is based on the RecordSound VI provided by the communications lab crew. With some modifications, we were able to manipulate the signals the way we want.
First, it is interesting to note that the virtual instrument can record a stereo sound, which is equivalent to 2 different signals multiplexed and saved into the same file. We need to differentiate between these two signals in order to be able to manipulate each one separately. To do so, we needed to gather some information about how the signals are acquired and stored.

Each of the left and right signals is sampled then stored in an array, so the stereo signals is represented by a two dimensional array. We only have to take each dimension at once in order to separate the signals. The process is shown in the figures below.

Figure 27: Experiment VI (part one)
The transpose is needed because the sound samples are stored in column arrays.

We are interested in hearing the result of the sum of the two signals. But before, in order to avoid bit overflow, we divided each of the signals by two so that the addition does not exceed the 8 bits and consequently generate unwanted distortion.

Now that we can record two signals separately and add them together, we can start our experiments. The first consisted of comparing a person speaking on the axis of the array to a person standing right in front of the array, with
microphones spaced about 60 cm, corresponding to a frequency of 300 Hz.

**Results:** though not very accurate, were promising, especially that we are only using two microphones, and they are low quality. The following figures show the spectra of the same speech for two the different situations described above:

![Figure 29: Person in line for the array (180° phase shift)](image)

**Conclusion:** From this experiment, we notice that the effect of phase cancellation on speech is not very evident, so we will try to see its effect on a single sound source using the same array.
3. MICROPHONE ARRAY FOR A SINGLE TONE SOUND SOURCE

Because the effect of phase cancellation was not very evident on speech, we decided to test its efficacy on a single tone sound source using two microphones.

**Goal:** Show the phase cancellation effect of a microphone array on a single tone sound source.

**Description:** The setup was similar to the previous experiment. We played a 726Hz sine wave in two different situations: in the first, the speaker was at an equal distance of 50 cm of both microphones, whereas in the second, the speaker was at aligned with the microphones, at a distance of 50 cm from the midpoint of the microphones. In both cases, the microphone spacing was equal to \( \frac{\lambda}{2} = \frac{344}{2 \times 726} = 23.7 cm \). The dispositions are summarized in the following figures:
Theoretically, the first situation should give a perfect reconstruction of the sine wave and the second case should result in a total destruction.

**Results:** Using the FFT VI, we displayed the spectra of the recorded signals:

![Figure 32: Reconstruction](image-url)
We notice a difference of about 50dB, which clearly indicates the presence of phase cancellation.

4. MICROPHONE ARRAY FOR WHITE NOISE

The last experiment conducted was to determine the effect of this phase cancellation on white noise.

Goal: Observe a frequency dip when phase cancellation is expected because of the property of phased arrays.

Description: The setup was similar to that of the previous two experiments. The experiment is similar to the one described above, with the sine wave replaced by a source of white noise (courtesy of Audio Engineering course). Because the preliminary analysis of the white noise provided reveals a concentration around 110 Hz, we changed the microphone spacing in order for the result to be more obvious. Actually, we expect a dip to occur around this
frequency, due to phase cancellation.

**Results:** The white noise spectra in both cases are depicted in the following figures.

![Figure 34: White noise](image1)

![Figure 35: White Noise- Phase cancellation](image2)

As we can see, a noticeable dip occurs at 110 Hz and neighboring frequencies. However, we still have doubts whether this dip is significant enough to be detected in real time.
5. EXPERIMENTAL CONCLUSIONS

From these experiments, we can conclude the following:

- Phase cancellation can be done in LabView by deleting calculated number of samples of the early received signal.

- The effect of phase cancellation on the power of single tone frequencies is evident, whereas its effect is less significant when the sound source is composed of multiple frequencies. The noticeable effect is a frequency dip at the array frequency, which is determined by the microphone spacing.

B. IMPLEMENTATION OVERVIEW

This first section introduces the simulation phase of the project. It starts by describing the implementation settings, and the reasons behind this choice of simulation.

1. IMPLEMENTATION SETTINGS

In order to prove the effectiveness of our design, we decided to simulate the actual design using hardware and software components. Recall that our design is a triangular phased array of microphones, composed of three sub-arrays, each covering 120°. We simulated a single sub-array composed of three microphones, using a single tone 500Hz sine sound source, localized in 5 regions.
2. ROOM ANALYSIS

Experiments were conducted in the analog electronics lab of the Faculty of Engineering and Architecture. It is an 8.3 x 5.4 x 2.7 room, surrounded by concrete walls and glass windows. The glass is known to be very reflective, and this greatly affected our experiments.

In addition, the lab is equipped to fit 12 groups of students, so we cannot consider it as an empty room, because of the big number of tables, desks, chairs, computer, and other electronic apparels. The uneven distribution of these items in the room makes it very difficult to obtain a good directivity.

Finally, we can compute the frequencies that will be emphasized using Rayleigh’s equations that relate the room dimensions to the modes (resonant frequencies):

\[
f = \frac{c}{2} \sqrt{\left(\frac{p}{L}\right)^2 + \left(\frac{q}{W}\right)^2 + \left(\frac{r}{H}\right)^2}, \text{ where } L, W, \text{ and } H \text{ are the dimensions of the room, and } p, q, \text{ and } r \text{ are integers, and } c \text{ is the speed of sound.}
\]

Because the room dimensions are relatively large, the room response at the frequency band of interest is nearly flat, so the frequencies are emphasized in a continuous manners (all the frequencies behave the same way in our room).
3. REASONS

As described in the previous section, we have not simulated our whole design, but only a single section. The reasons behind our choice are the following:

Completeness: the three sub arrays of our design are never activated at the same time (cf. algorithm in section B). That is why simulation of a single sub-array and of the transition between sub-arrays is enough.

Budget and Equipment: since only three microphones were at our disposition, we didn’t want to increase the project budget.

Sound Source: starting the simulation of the device on speech is not a very effective strategy. Since many trials are involved, it is not possible to have a speaker (or many speakers for that matter) talk at every trial. That is why we decided to start of simulation with sine sources. The extension of our simulation to speech is described in chapter 4.

Region Choice: We found experimentally that 5 regions per sub-array is the maximum number that we can choose. In fact, this choice infers that the whole device could localize 15 different speakers, which is a very good number.

C. ALGORITHM

After giving an overview of the implementation objectives, the used algorithm is described. At first the theory behind the algorithm is explained, and then a
pseudo code is presented. Finally, the noise cancellation technique is described.

1. THEORY

As described in Chapter 1, our design is based on inter-acoustic differences between receivers and phase delays. So, our algorithm follows the same strategy. By using sound level differences between microphones in parallel with phase shifting the microphones inputs and summing them and comparing power, we were able to devise a dynamic sound localization algorithm.

2. PSEUDOCODE

Following this brief theoretical revision, the pseudo code used in our implementation is described.

The following figure summarizes the algorithm:

**Choice of sub-array:** as mentioned in the design chapter, each of the three sub-arrays has a cardioid microphone in the center. The sub-array to be activated is the one that receives highest power at its cardioid microphone.

**Phasing:** Since we have 5 regions, we determined that we needed 5 phase sets to localize sound. However, this is not the case. This is because a phased array of microphones is symmetric. If we phase shift to form a directive beam at 60° from the axis of the array, we find that the power of the same signal at 180-60 = 120° is the same. This experimental finding has led us to find the symmetric property of the array. Thus, we wanted to check if this should be the case, so
we proved our finding as follows:

![Figure 36: Mathematical Representation](image)

Let \( d \) be the distance between two adjacent microphones (34.4 cm) and let \( c \) be the speed of sound (344 m/s) and the frequency of the sound source is 500 Hz.

For \( \alpha = 60^\circ \), 
\[
T = \frac{\phi}{2\pi} = \frac{5 \times 10^{-4}}{\frac{2\pi}{500}} = \frac{\pi}{2} \text{ rad.}
\]

For \( \beta = 120^\circ \), \( \phi = -\frac{\pi}{2} \text{ rad.} \)

Sum of powers:

The sum is \([a \sin (wt) + a \sin (wt + \phi_1) + a \sin (wt + \phi_2)]\) its power is equal
to $\frac{3a^2}{2}$

Power of sum = Total power (for $\phi_2 = 2\phi_1$)

**Position x, $\alpha = 60^\circ$, $\phi = \frac{\pi}{2}$**

Sum $= [a \sin (wt) + a \sin (wt + \frac{\pi}{2}) + a \sin (wt + \pi)] = -a \cos (wt)$, its power is equal to $\frac{a^2}{2}$

**Position y, $\beta = 120^\circ$, $\phi = -\frac{\pi}{2} \text{rad}$:**

Sum $= [a \sin (wt) + a \sin (wt - \frac{\pi}{2}) + a \sin (wt - \pi)] = a \cos (wt)$, its power is equal to $\frac{a^2}{2}$

**Direction Detection:** Following this proof of symmetry, our algorithm must still detect the direction of the sound source, i.e. whether this sound source is at the left or right of the sub-array. This step can be done easily by comparing the power of the boundary microphones of the sub-array and use their inter-acoustic difference to determine finally the exact region which contains the source.

**Noise Cancellation:** Noise cancellation is done using an ambient microphone that is directed away from the array. The input of this microphone is
subtracted from the inputs of the array. Experimental verification of this strategy yielded impressive noise canceling results.

As a conclusion for this explanation, the following is a text pseudo code of the algorithm:

For 3 phase shifts (0 samples, 5 samples and 10 samples) corresponding to indexes 0, 1 and 2 respectively

Index 0 – 90 degrees
Index 1 – 60 degrees
Index 2 – 30 degrees

Store the powers in an array
Find the maximum of the array and the index of this maximum

If index=0
    Region C
Else If Index = 1
    If Power of microphone 1 > Power of microphone 3
        Region D
    Else If Power of microphone 1 < Power of microphone 3
        Region B
Else If Index = 2
    If Power of microphone 1 > Power of microphone 3
        Region E
    Else If Power of microphone 1 < Power of microphone 3
        Region A

Repeat Process

The following figure shows the region subdivisions
In order to implement the designed algorithm, hardware and software components are required. This section is dedicated to describe the hardware setup used, while the next one showcases the software implementation methodology. This section starts with detailed description of each hardware component (microphones, preamplifier, connection box, DAQ, computer.) We then show the connection setup used in order to make this hardware operational. Finally, a budget section summarizes the cost of the equipment used.

Figure 37: Sub-Array Regions

**D. HARDWARE**

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1. MICROPHONES

We have used two types of microphones:

**Beyerdynamic MC 834:** This microphone was used for noise removal. It is a condenser microphone, which is a very sensitive and accurate transducer. Its data sheet is presented as an appendix. It has a wide variety of applications, from vocal works to percussions, as well as basses, guitars, piano and brass.

![Beyerdynamic MC834](image)

*Figure 38: Beyerdynamic MC834*

This microphone was available in the Audio Lab.

**Shure Beta 57 A:** It is a dynamic microphone whose Frequency Response extends from 50 Hz to 16000 Hz. It has a super cardioid pattern. It also allows for low-frequency proximity effect to enhance bass for close-up vocals.

It can be used for acoustic and electric instruments as well as for vocals: guitar amp, bass amp, acoustic guitar, brass/saxophone, snare drums, rack/floor toms, congas, woodwinds and lead vocals.
As for the condenser microphone, we have used the three available microphones from the Audio Lab to construct our linear sub-array.

2. PREAMPLIFIER

We have used SMPro Audio PR8 8-Channel Mic Preamp

It has the following main features:

8 channels of balanced input/output

Full-range gain control from -20dB to +40dB

Phantom Power for every channel

8 XLR Inputs on front panel

8 TRS Outputs on rear panel

The following is a series of pictures of the preamplifier.
Figure 40: Preamplifier – Front View

Figure 41: Preamplifier Components

Figure 42: Preamplifier - Backside
This device was bought from Rag Time Computer and Music Technology in Hamra – Beirut – Lebanon.

### 3. CONNECTION BOX

We have used the SCB-68 National Instruments Connection Box Quick Reference Label MIO- 16E Series.

**Specifications:**

**Input**

Eight differential, 16 single-ended

**Power Requirement**

Typical: 1 mA with no signal conditioning installed

Maximum: 800 mA from host computer

**Physical**

Box dimensions (including box feet): 7.7 by 6.0 by 1.8 in. (19.5 x15.2 x 4.5) cm

I/O connectors Screw terminals: One 68-pin male SCSI connector
We have connected the Audio cables according to the following table:

<table>
<thead>
<tr>
<th>PIN NUMBER</th>
<th>SIGNAL</th>
</tr>
</thead>
<tbody>
<tr>
<td>68</td>
<td>ACH0</td>
</tr>
<tr>
<td>34</td>
<td>ACH8</td>
</tr>
<tr>
<td>33</td>
<td>ACH1</td>
</tr>
<tr>
<td>66</td>
<td>ACH9</td>
</tr>
<tr>
<td>65</td>
<td>ACH2</td>
</tr>
<tr>
<td>31</td>
<td>ACH10</td>
</tr>
<tr>
<td>30</td>
<td>ACH3</td>
</tr>
<tr>
<td>Pin</td>
<td>Signal</td>
</tr>
<tr>
<td>------</td>
<td>---------</td>
</tr>
<tr>
<td>63</td>
<td>ACH11</td>
</tr>
<tr>
<td>28</td>
<td>ACH4</td>
</tr>
<tr>
<td>61</td>
<td>ACH12</td>
</tr>
<tr>
<td>60</td>
<td>ACH5</td>
</tr>
<tr>
<td>26</td>
<td>ACH13</td>
</tr>
</tbody>
</table>

*Table 1: Connection pins of the SCB-68*

*Figure 44: SCB-68 Quick Reference Label*
This connection box was available from the Communications Lab
4. DAQ (Data Acquisition Device)

The National Instruments data acquisition device is internally linked to the PC, and behaves similarly to a sound card, as it receives analog input and converts to digital signals, ready to be processed by LabView. The DAQ was also provided by the Communications Lab.

5. DELL COMPUTER

A basic device used in any ECE FYP. It was offered by, again, the Communications Lab.

6. CONNECTIONS

We have used 6 audio cables to connect the output interface of the mic preamp with the connection box.

Those cables were 6 feet long each (1.8 m). They were shielded and had a $\frac{1}{4}$ inch stereo phone plug to $\frac{1}{8}$ inch mono phone plug. We have stripped the mono end of the cables and we have separated the two inner parts in order to plug them into the connection box.
We have bought the connectors from RadioShack store Hamra – Beirut – Lebanon.

The following figure shows the hardware connections:
7. HARDWARE ISSUES

As you may have noticed, there is quite a number of hardware components involved. Moreover, the setup of this hardware is quite unconventional.
Absence of Sound Card: this project might easily be the first audio project done without the use of a sound card. This is due to the fact that the SCB-68 connection box has replaced the need for a sound card, since it provides accurate A/D conversion.

Connectors: The connectors used between the preamp and the Connection Box have to be of Mono type, which is quite unusual. This is due to the fact that each preamplifier output delivers a mono signal.

8. BUDGET

The following table summarizes the overall budget of the project:

<table>
<thead>
<tr>
<th>Hardware Component</th>
<th>Price</th>
<th>Financial Source</th>
</tr>
</thead>
<tbody>
<tr>
<td>3 Shure Beta 57 A microphones</td>
<td>450$</td>
<td>Available in the Audio Lab</td>
</tr>
<tr>
<td>1 BeyerDynamic MC384</td>
<td>1000$</td>
<td>Available in the Audio Lab</td>
</tr>
<tr>
<td>SM Prop Audio 9 preamplifier</td>
<td>175$</td>
<td>ECE Department (not yet paid)</td>
</tr>
<tr>
<td>SCB-68 Connection Box</td>
<td>300$</td>
<td>Available in the Communications Lab</td>
</tr>
<tr>
<td>4 Mono Connectors</td>
<td>15$</td>
<td>Own resources (Diana)</td>
</tr>
<tr>
<td>4 Microphone Cables</td>
<td>120$</td>
<td>Available in the Audio Lab</td>
</tr>
<tr>
<td>Computer &amp; DAQ</td>
<td>2500$</td>
<td>Available in the Digital Lab</td>
</tr>
<tr>
<td>TOTAL</td>
<td>4560$</td>
<td></td>
</tr>
</tbody>
</table>
E. SOFTWARE

After describing the hardware setup used to implement the phased array of microphones, we will now dwell into the software implementation. This section contains a brief description of LabView, the signal processing software, the different sections of our software implementation, which are Data Acquisition, Sub-array activation and X-Y Localization, are explained in detail. Then, the chosen user interface for our software is described. As in the previous section, we will describe the implementation issues that have arisen and the method used to solve them.

1. LABVIEW

Labview is a graphical programming system that is designed for data acquisition, data analysis, and instrument control. LabVIEW can run on a number of systems including PC Windows, Macintosh and VXI systems, and is transportable from one system to another. Programming an application in LabVIEW is very different from programming in a text based language such as C or Basic. LabVIEW uses graphical symbols (icons) to describe programming actions. Data flow is ``wired" into a block diagram. Since LabVIEW is graphical and based on a windows type system it is often much easier to get started using it than a typical language.

LabVIEW programs are called virtual instruments (VIs) because the
appearance and operation imitate actual instruments. VIs may be used directly by the user or as a subroutine (called subVI's) of a higher program which enables a modular programming approach. The user interface is called the front panel, because it simulates the front panel of a physical instrument. The front panel can contain knobs, push buttons, graphs, and other controls and indicators. The controls can be adjusted using a mouse and keyboard, and the changes indicated on the computer screen. The block diagram shows the internal components of the program. The controls and indicators are connected to other operators and program structures. Each program structure has a different symbol and each data type (eg. integer, double-float etc) has a different color.

**Sound and Vibration Toolkit**

The National Instruments LabView Sound and Vibration toolkit has been downloaded. It allows the performance of audio measurements, fractional-octave analysis, swept-sine analysis, sound level measurements, frequency analysis and transient analysis. In our case, it has been used to perform single tone measurements of the obtained signals via Single-Tone VIs. The gain and the phase of the obtained signal have been measured (needed for vertical detection). It has also been used to get the power in a frequency band (power at microphone1 and 3 respectively).
2. DATA ACQUISITION

We have tried two types of data acquisition: Sequence and AI Multiply recording.

**Sequence Method:** The Sequence method consists in putting in sequence three consecutive DAQ Assistants that are configured to have Analog Inputs more specifically Voltage and that are each connected to a channel of the NI Connection Box. In the figure below, DAQ Assistant is connected to channel a0, DAQ Assistant2 to channel a1 and DAQ Assistant3 to channel a2.

Then, the setup of the voltage input is set: The input range is given a Max value and a Min value, in our case plus or minus 5V with differential terminal configuration. An N samples Acquisition Mode is chosen. The Clock Settings are also defined: we give it the number of Samples to Read and a Rate (Hz).

![Figure 49: Sequence Method Data Acquisition](image)

We have tried three different sampling rates: 44100, 22050 and 11025 with number of samples ranging from 100 to 100000.
When recording the obtained signals from the DAQ Assistants, it turned out that the signals were suffering from cancellation: chunks of the signal were swollen.

This can be explained in the following way: the sequence is included in a loop, every time the DAQ Assistant turn is to capture the signal; it reinitializes the NI Connection Box. This re-initialization time causes silent gaps. In addition to this, DAQ Assistants have to wait for each other in order to get the signals: this causes chunks of the three signals to overlap thus decreasing the audibility of the recorded signal.

**AI Multiply Method:** It acquires data from the specified channels and samples them at the specified scan rate. The AI Acquire Waveforms VI performs a timed measurement of multiple waveforms on the specified analog input channels. This method gives clear and eligible signals:

- No re-initialization is performed in this case.

- A real time acquisition is thus efficiently obtained.
3. SUB-ARRAY ACTIVATION

The designed set-up activates in a cyclical manner its three sub-arrays. It scans the first region by activating the first sub-array, checks for the presence of a sound source. If no sound source is found (silence region), it scans the adjacent sub-array by deactivating the first sub-array and activating the second one etc. The transition between arrays is based on the following logic: comparing the acquired signals levels to the silence threshold.

Transition with Next Sub-Array:

If Total Power < Silence Threshold

Move to next array
4. X-Y LOCALIZATION

After choosing the activated sub-array we move to the actual sound localization, which is based on the phase properties of sound waves. This localization is done in two steps: half-plane detection and Left/Right Panning.

After comparing the acquired sound to the threshold level of silence, the next step was to scan the 120° region to check in which sub region the sound source was located.

**Half-Plane Detection:** The scanning process is done by appropriately applying the calculated phase shifts to each signal, summing the obtained signals together and measuring the obtained power. This phase shifting is done by removing a pre-computed number of samples from the recorded array based on the region, the speed of sound and the microphone spacing.

For example for $\theta = 30^\circ$, $\phi = 2.72$ rad, so, the time delay t
\[ \frac{2.72}{2\pi} \times \frac{1}{500} = 8.658 \times 10^{-4} \text{ seconds.} \]

Since a sampling rate of 11025 (Hz) has been chosen, the number of samples reflecting the deletion portion of the array should be 11025 \times 8.654 \times 10^{-4} = 9.54 samples. Applying the ceiling function the number of samples is 10.

Because the array is symmetric, this process repeats three times, for 30, 60 and 90 degrees (regions A-E, regions B-D, region C respectively).

The maximum of the three measured powers narrows down the possible locations of the source. If the obtained region is C, no more processing is needed.

**Figure 52: Phase Shifting VI**

**Panning:** The final step used to localize the source is comparing the power at the boundary microphones. This step executes when the source is not found to be in region C. Here, the power at the two boundary microphones is measured. The location of the sound source is determined by the microphone receiving highest power.
This algorithm repeats until a Stop button is pressed.

5. NOISE REMOVAL

In order to minimize the interaction with the room, special care has been dedicated to noise cancellation.

A fourth microphone has been used to acquire ambient sounds. Its signal is subtracted from the other three microphones before phase shifting.
6. REAL TIME LIVE RECORDING AND AMPLIFICATION

The signals are continuously recorded as they are inputted and processed in the VI.

In order to do so, an extra VI, not present in LabView libraries has been designed: Write wave.vi.

The write wave.vi uses a VI in the labview library: Snd Write Wave File

![Figure 54: Recording VI](image)

The VI designed stores the waveform data and its sound format information into a file. The file is in a PC wave file (.wav) format.

Once the sound source is detected and localized, appropriate phase shifts are applied during processing: they permit to focus and amplify the wanted signals, i.e., the sound source signals.
7. SOFTWARE BLOCK DIAGRAM

All of the above processing explanation is summarized in the following block diagram.

Figure 55: LabView Block Diagram
8. USER INTERFACE

Finally, we arrive to the user interface of our phased array, whose goal is to show the results of all the processing implemented in the LabView Code. The following figure is a screen shot of the user interface.

![User Interface Diagram](image)

**Figure56: User Interface**

**Activated Array**: the LED display shows which linear sub-array is activated. Recall that only one sub-array is activated at a time.

**Localization in the X-Y plane**: The VU meter indicates which region within the activated sub-array contains the sound source.

**Localization on the vertical axis (not yet implemented)**: The bar indicator shows whether the sound source is lower than the array, in front of the array or higher than the plane of the array.
Following this extensive explanation of the implementation process of our design, we now turn to the last section of our report, which is the Evaluation section.

CHAPTER 5: EVALUATION

As for every proposed and implemented design, an evaluation phase is bound to be executed. This section shows how we, as a team, evaluated the rationale behind our design, as well as a critical appraisal of our findings. This section begins with a thorough explanation of the testing procedure. Then, our testing results are evaluated by comparing them with the stated objectives as well as with previous work on phased arrays of microphones. Finally, our design is not without limitations. That is why a section is dedicated to show the shortcomings of our proposed design.

A. TESTING

The two main types of testing we have performed are logical testing and user acceptance testing.

<table>
<thead>
<tr>
<th></th>
<th>LOGICAL TESTING</th>
<th>USER ACCEPTANCE TESTING</th>
</tr>
</thead>
<tbody>
<tr>
<td>Meaning</td>
<td>Building the system right</td>
<td>Building the right system</td>
</tr>
<tr>
<td>Timing</td>
<td>During coding</td>
<td>During Testing</td>
</tr>
<tr>
<td>Nature</td>
<td>Syntactic(properties of the system)</td>
<td>Semantic(extrinsic properties)</td>
</tr>
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</table>

Table 2: Comparison of the two used testing types
1. LOGICAL TESTING

The main purpose of this type of testing is to check the system in order to make sure it gives solutions or results correctly. It looks for anomalies in order to fix them and ensures a bug-free scheme. It ignores the external factors.

This step is divided into two main parts: Verifying formation and functionality.

**Formation:** it consists of looking for circular or redundant errors as well as taking a closer look to consistency, correctness and completeness of the system.

- Circular and Redundant errors: while building the system we avoided circular and redundancy errors by building blocks each responsible of a certain functionality and then integrating them in a larger system.

- Consistency: we have also checked that the system produces similar answers at all times with no contradiction. This part has been checked for by simulation of a single frequency signal (500Hz) inputted to the system.

- Correctness and Accuracy: the used strategy has proven to be efficient since it has localized a speaker connected to a wave generator outputting a 500Hz sine wave placed at the bisectors of the regions. This localization is instantaneous and 100% accurate. This accuracy is due to the correctness.
of the design, as well as to the signal processing capabilities of the used software.

Moreover, the designed algorithm is dynamic, which means that this phased array of microphone tracks the sound source as it moves from region to region.

Furthermore, the recorded signals that were outputted on loud speakers were perfectly audible and continuous: no interruption or overlapping occurred.

Finally, the design worked for any time constant power sound source, whether single toned or multi toned.

It is also worth noting that the frequency measurement of the inputted signal is very close to reality. When generating a 500Hz sine wave, the detected frequency fluctuated between 499.8 and 500.2 which mean less than 0.4% error. This is mainly due to LabView capabilities.

- Completeness: Only a part of the set-up has been built along with the connection between the sub-parts. Since the same logic is used in all three sub-arrays and the link between them properly works, the whole set-up if built should be normally working.

**Functionality**: It consists of looking at confidence and reliability of the system.
• Confidence: since the system is a first prototype, the confidence level cannot achieve 100%. However, the different tests have shown encouraging results, thus, the confidence in the design is relatively high for a first prototype.

• Reliability: Since the system is greatly dependant on the environment, reliability of the implementation is fairly low. The room must be fairly isolated, inside noise must be very low and region delimitation must be precise.

2. USER ACCEPTANCE TESTING

This type of testing is performed after the logical testing. It checks the set-up behavior in a real environment. It ignores the internal mechanics of the system. It is also complemented by a quality assessment. It requires checking adaptability, robustness and appeal of the system as well as design test cases.

**Adaptability:** The device cannot be moved to other rooms and be as efficient as it is in the room where we have implemented it since it is greatly dependent on room acoustics and reflections.

**Robustness:** The built set-up didn’t crash and didn’t stop processing the signals unexpectedly if not required to do so manually. It continuously acquired the signals and processed them. Robustness was mainly achieved because of the high scan rate that Lab View allowed and the absence of bugs.

**Appeal:** The user interface is clear and self explanatory. It has three types of
indicators representing the region where the speaker is localized (region A to E), the currently activated array (1, 2 or 3) as well as the height of the speaker (low, mid or high)

**Sound Generation:** To properly generate sine waves, we used the wave generator (Agilent 33120A 15MHz Function Arbitrary waveform generator), connected directly to a DELL sound speaker, as shown in the diagram below:

*Figure 57: Sound Generation*

**Region Delimitation:** In order to properly delimit the speaker regions, we place the linear array on a table and fixed the microphone positions and orientations using markers. Then, using a fixed length chord and a geometry set, we were able to design the 5 separate regions. We used removable scotch tape in order to keep these limits for future use.
**Figure 58: Phased Array- Region Delimitation**

**Hardware testing:** After making the connections inside the NI Connection SCB-68 Box, we connected the microphones to the preamp and then to the connection box. In order to make sure that proper signals with a specific amplitude level are arriving at the connection box, we have used the Tektronix TDS 220 two-channel digital real-time oscilloscope (100 MHz, 1 GS/s).
When the signals at the three microphones are added without phase shifting, we get a single lobe with 125 dB at the edge and 105 dB at the sides. This result is consistent with the phased arrays concept: there is a 20dB difference between the median and the sides.

Calibration of the three microphones: Our set-up requires that microphones
1 and 3 have a similar cardioid pattern. While implementing the set-up, we originally used two condenser cardioid microphones Beyerdynamic MC 834. While calibrating them, it turned out that the two microphones had different patterns that would greatly affect the response of our set-up. In fact, the phase shifting part is very sensitive and is greatly dependent on phase measurements and sound levels. The two rented microphones have previously received shocks that slightly changed their frequency response. We thus replaced them by Shure Beta 57 dynamic microphones. During calibration, the two microphones turned out to have similar patterns.

**Power Tables:** We have built two tables containing the power values of the signals at microphones 1 and 3 and one for the phase measurement. Then, we have moved the speaker linked to a wave generator outputting a 500Hz sine wave around the array of microphones to see how the power fluctuates. The results obtained were satisfactory since the power at microphone 1 increased greatly when the speaker was on its side and low at microphone 3, and vice versa.

**Frequency resolution:** when performing the FFT of the signals, we made sure that the resolution was narrow enough to insure smooth graphs and accurate results. We computed the df by using a df calculation vi present in LabView and we obtained 29Hz which indicates a high resolution for speech. This shows that the results and the graphs obtained were accurate.
3. NON TECHNICAL TESTING

In this part of the report we will take a closer look at non-technical aspects of the designed set-up and the project in general, more particularly economic, environmental, social, political, ethical, health and safety, manufacturability, and sustainability.

**Economic:** The built device is not more expensive than usually installed conference devices. It can thus be easily dealt with as an alternative to all already existing systems.

**Environmental:** The designed set-up doesn’t affect the neighboring environment and doesn’t pollute it or cause damage. What we aimed at was trying as much as possible to remove noise pollution from the neighboring environment (computers, cars, adjacent rooms, planes etc) that altered the audibility of the outputted signals.

**Social:** The set-up we have built can be used in conference rooms. It focuses on the speech of the person talking while ignoring other less significant sources of sound: noise, whispering etc that may interfere with the speaker’s voice and cause it to be less audible. Moreover, a dynamic transition between speakers is insured: this way, instead of waiting for a person to stop talking, then pushing the button to have the token, nothing is asked from behalf of the conference members and no pre-organization is required. Note that this system eliminates the ambiguity when two speakers want to have the token at
the same time.

Thus, the conference becomes more civilized and better controlled.

**Political:** The designed and implemented set-up is apolitical. The concept behind it is not linked in any way to political issue. Yet, the device can be used during political debates, campaigns and talk shows in order to focus on the politician during his/her speech or on the interviewer asking the questions.

**Ethical:** A minor ethical issue that could be raised is the following one: the device focuses on the signals with highest power and amplitude, meaning that it focuses on the individuals with loudest voices. Thus, if two persons are speaking and one of them has a low voice, the time slot that it should be getting will most probably be given to someone else with a highest voice. So, the device doesn’t allocate time slots equally between conference members.

More advanced versions of the designed set-up could be used for spying which, if not legally allowed, could become unethical.

**Health and Safety:** All the components used do not have sharp ends and aren’t heavy. They cannot injure the individual dealing with them. Moreover, the set-up doesn’t work on high voltages so the individual cannot be electrocuted. In order to add even more safety, we have grounded the whole set-up.

**Manufacturability:** The device we have built is an assemblage of various pre-manufactured components. We only had to strip the mono plug wires
ends in order to insure proper connections with the connection box. However, we didn’t need to shape, cut or melt any substance or material.

**Sustainability:** The built set-up is fully sustainable: in fact, it meets the needs of the present without compromising the ability of future generations to meet their own needs. It doesn’t have side ecological effects whatsoever.

**B. RESULTS**

This section summarizes the results obtained after implementing one of the sub-arrays. The detailed results versus the original goals are described along with the detailed results versus other performed set-ups described in the survey literature.

**1. RESULTS VERSUS ORIGINAL GOALS**

The strategy of a single sub-array has been proven to be efficient, since it has localized a speaker connected to a wave generator outputting a 500Hz sine wave placed at the bisectors of the regions in a plane divided into 5 sub regions. This localization is dynamic, instantaneous, and 100% accurate.

Moreover, the designed algorithm is dynamic, which means that this phased array of microphone tracks the sound source as it moves from region to region.

The results are summarized in the following table:
GOALS | ACHIEVED RESULTS
---|---
Implementation of the whole set-up: 3 sub-arrays + array activation | Implementation of the whole set-up: 1 sub-array + array activation
Noise Removal | Noise Removal
Speech Localization | Single Tone Localization
Focus and Amplification | Focus and Amplification
60° region determination | 120/5 = 24° region determination → more accurate than originally expected
Dynamic Localization | Dynamic Localization
Use of omni-directional and cardioid microphones | Use of cardioid microphones (omni-directional not available & expensive)

*Table 3: Goals versus achieved results*

2. RESULTS VERSUS PREVIOUS ATTEMPTS

The design we have implemented is different in many ways from previous attempts mainly because many features have been added.

In fact, previous attempts only used single arrays of microphones whereas our design used three arrays. No previous design has tried the triangular shape which turned out to be highly efficient: they have used linear, square or
circular shapes. Moreover, no previous attempt has combined omni-directional microphones and cardioid microphones.

All those features in our device made it more directive than previously attempted ones.

It would be interesting to compare it to paper [8] where a linear array of two microphones has been implemented. In this paper, the results have shown to be poor, only one out of the eight regions were properly detected, and only 180° surface has been covered. In our case, five out of five regions were properly detected and we aimed at achieving a 360° surface coverage.

C. LIMITATIONS

This final section of the implementation chapter will describe the limitations faced by our project. These limitations were defined during the testing period, and can be grouped into four categories: power limitations, room limitations, speaker positioning and microphone performance.

1. POWER LIMITATIONS

When the power emitted by the sound source was constant, it can be detected easily by the device. However, for sources with varying power, such as a person speaking, the task is more complicated. In fact, these variations in power can “mislead” the algorithm, as shown in the following example: as
stated previously, each phase shift is kept for one second, and the power after the phase shift is recorded. If the speaker spoke very loudly in a phase shift other than the one corresponding to his region, and then spoke softly in the phase shift of his region, then a wrong detection will occur.

This is the reason why the scheme did not work perfectly for the case of human speech, while it has shown very accurate results for sounds generated by the function wave generator.

2. ROOM LIMITATIONS

If the device is being tested in a small nonempty room, such as a computer lab or a classroom, then the accuracy of the results will be limited by the amount of reflections the environment provides. This affects both the angle detection and the panning schemes. In many cases, the power received by the left hand side microphone was greater than the power received by the right hand side microphone, when the sound source was located on the right hand side. This issue was partially resolved by placing the device in such a way to minimize the reflections.

3. SPEAKER POSITION

Because of the causes cited above, localization provided extremely accurate results at the theoretical optimum points (the bisectors of the regions), with decaying performance as we move farther away from these points. In fact, as the speaker was moved more than 5 degrees from the bisector, the results
started fluctuating between the actual region and its neighbors in some cases, and the between the actual region and its symmetric in other cases.

4. MICROPHONE QUALITY

As stated in the hardware part, the microphones used are dynamic, which is not the best type of microphones. Condenser microphones capture sound waves with higher fidelity, which could have been more accurate for phase difference measurements and power differences.

D. CRITICAL APPRAISAL

The Final Year Project was a very enriching experience that allowed us to learn many valuable lessons.

When dealing with a large scale extended project over a whole year, we should organize and give proper credit to the work. It is also essential to be constantly ready to face unexpected problems and deal with them: adaptation becomes a key element for success.

We also learned to choose between alternatives whether in design or in implementation based on specific criteria.

We have dealt with all the stages in a project from the topic selection till the testing and valuation as well as with various aspects of it: economic, technical, etc. We have thus had the chance to look at a global view of a project.

We have learned to work under the supervision of a Professor while respecting
deadlines, submitting reports, performing presentations etc.

We have also applied all knowledge acquired during four years spent at AUB in order to solve a practical problem.

Finally, we have experienced the professional side in every aspect of engineering and we approve the saying “an engineer is a problem solver”.

CHAPTER 6: CONCLUSIONS

A lengthy report as this one needs to be summarized for clarity. This chapter is dedicated to review the major findings of our Final Year Project, as well as suggestions for future work on the topic of sound localization.

A. REPORT SUMMARY

1. IDEA

The idea behind the project is the extrapolation of the phased array of antennas to microphones. The set-up designed and implemented aimed at localizing a main sound source among many, capturing the sound emitted by that source and amplifying it, while ignoring noise and other less significant sources of sound and ensuring an automatic transition among the sources.
2. DESIGN

Based on several simulations and pre-experiments, and in order to achieve the set objectives, we have designed a triangular array composed of 3 sub-arrays each covering a 120° region. Each sub-array is composed of 3 microphones, two omni-directional ones at the edges and a cardioid microphone in the center.

3. IMPLEMENTATION

We have implemented a single sub-array using several hardware components: 3 dynamic microphones, a preamp, a connection box as well as a DAQ. We have processed the obtained signals using LabView. Moreover, we have simulated the link between two adjacent sub-arrays.

4. EVALUATION

The tests conducted on the implemented set-up have indicated that the obtained results were very accurate: we were able to detect a single-tone sound source in 5 different regions (24° coverage each) and correctly link the sub-arrays together. Some limitations mainly reflections have been encountered. We have also looked at the project from a non-technical point of view and have detailed a critical appraisal of the project as a whole.
B. FUTURE WORK

In this section, suggestions for future work are proposed. The device we have designed and implemented can be used as a major building block for future sound localization and speech tracking applications. As a team, we suggest three major improvements: **building the whole triangular array**, supporting **vertical localization** and adapting the software for **speech localization**.

1. WHOLE SETUP

After we have proven the efficiency of one sub-array on a 120° sub plane and the whole vertical axis, the next step is to build the whole set up, consisting of three sub-arrays, identical to the one described above, in order to cover the whole 3-D space. The algorithm would work one sub-array at a time, so if no sound above a certain threshold is found in some region, we move to another sub-array, repeating the same process. The interface will show which sub-array is currently activated.

An additional extension would be to build a stand for the six microphones, where the distance between them is variable, in order to fit different uses of the device.

2. 3D (Vertical) LOCALIZATION

Our device currently supports plane localization of the sound source. This device can be easily improved to support 3D localization using two different
methods that aim at adding a vertical detection capability to the previously achieved x-y plane localization scheme. The first one consists of using a power measurement scheme and one additional microphone, placed in a plane parallel to that of the triangular array. The second one consists of approaching the solution in the reverse way: the phase difference between the microphones is measured and then compared to a predefined table from which the angle of the sound source is determined. This method is called the MINIMUM DISTANCE CLASSIFIER method. It is summarized in the following diagram.

Figure 61: Minimum Distance Classifier

3. SPEECH LOCALIZATION

As stated previously, the device cannot be easily applied to speech for two
main reasons: the first is the unstable power of speech, and the other is the richness of speech in frequency content. We intend to solve each of these problems separately. The first idea is to detect the peak frequency of the speech and base our computations on this estimated frequency. Then, in order to solve the issue of fluctuating power, we need to drastically reduce the scanning time; however, this will raise a new issue, because for each iteration, the DAQ “steals” a few milliseconds to configure, and if each iteration is only a few milliseconds long, then we will be losing a considerable proportion of the signal.
REFERENCES

[1] Tauchi M., An analysis of sound localization performance provided by the present sounds of audible traffic signals for visually impaired pedestrians


http://cnx.rice.edu/content/col10250/latest/


APPENDIX: MATLAB CODES

2 microphones, 0.35 cm spacing, 60º spacing, 500 Hz input wave:

```matlab
% 2 microphones, for 500Hz sine waves, 
% adjusted to receive at 60 degrees 

t=[0:0.1:180];
y2=0*t;
x=[0:0.0001:0.01];
ph=0*x;
for i=1:1800
    a=0.35*sin(pi/2-(t(i)*pi/180));
    time=a/350;
    for j=1:100
        ph(j)=pi*time*1000-0.5*pi;
    end;
    rec=sin(500*2*pi*x)+sin(500*2*pi*x+ph);
    y2(i)=max(rec)/2;
end;

plot(t,y2);
```
3 microphones, 0.35 cm spacing, 60° spacing, 500 Hz input wave:

%3 microphones, for 500Hz sine waves,
%adjusted to receive at 60 degrees

t=[0:0.1:180];
y3=0*t;
x=[0:0.0001:0.01];
ph1=0*x;
ph2=0*x;
for i=1:1800
    a1=0.35*sin(pi/2-(t(i)*pi/180));
a2=0.7*sin(pi/2-(t(i)*pi/180));
time1=a1/350;
time2=a2/350;
    for j=1:100
        ph1(j)=pi*time1*1000-0.5*pi;
        ph2(j)=pi*time2*1000-pi;
    end;
    rec=sin(500*2*pi*x)+sin(500*2*pi*x+ph1)+sin(500*2*pi*x+ph2);
y3(i)=max(rec)/3;
end;

plot(t,y3);

4 microphones, with the spacing, the reception angle and the input wave frequency specified as input parameters:

%4 microphones, for 500Hz sine waves,
%arguments are microphones spacing and
%desired reception angle
%speed of sound assumed 350m/s, so wavelength=0.7m

function y4=mic_test(spacing,an,f);
%spacing is the microphone spacing

t=[0:0.1:180];
y4=0*t;
x=[0:0.0001:0.01];
ph1=0*x;
ph2=0*x;
ph3=0*x;
    for i=1:1800
        %these are the distance differences for the received waves
            a1=spacing*sin(pi/2-(t(i)*pi/180));
            a2=spacing*2*sin(pi/2-(t(i)*pi/180));
            a3=spacing*3*sin(pi/2-(t(i)*pi/180));
        %and the following are the time differences
            time1=a1/350;
            time2=a2/350;
            time3=a3/350;
        
        for j=1:100
            %the term after the minus sign is the microphone phase setting, or
            %the phase needed for each microphone to capture a certain angle
            %so is ph1, ph2 and ph3 are zero, there's matching and constructive
            %interference
            ph1(j)=2*pi*time1*f-spacing*pi*sin(pi/2-an*pi/180)/0.35;
            ph2(j)=2*pi*time2*f-2*spacing*pi*sin(pi/2-an*pi/180)/0.35;
            
    end
end
\[ \text{ph3}(j) = 2\pi \times \text{time3} \times f - 3 \times \text{spacing} \times \pi \times \sin\left(\pi / 2 - \pi \times \text{an} / 180\right) / 0.35; \]
end;

\[ \text{rec} = \sin(500 \times 2 \times \pi \times x) + \sin(500 \times 2 \times \pi \times x + \text{ph1}) + \sin(500 \times 2 \times \pi \times x + \text{ph2}) + \sin(500 \times 2 \times \pi \times x + \text{ph3}); \]
\[ \text{y4}(i) = \max(\text{rec}) / 4; \]
end;

plot(t, y4);

Our design simulation – Triangular- 3 cardioid microphones and 3 omni microphones

t=[0:0.1:360];
y3=0*t;
x=[0:0.0001:0.01];
ph1=0*x;
ph2=0*x;
for i=1:1200
\[ \text{a1} = 0.35 \times \sin(\pi / 3 - (t(i) \times \pi / 180)); \]
\[ \text{a2} = 0.7 \times \sin(\pi / 3 - (t(i) \times \pi / 180)); \]
\[ \text{time1} = \text{a1} / 350; \]
\[ \text{time2} = \text{a2} / 350; \]
for j=1:100
\[ \text{ph1}(j) = \pi \times \text{time1} \times 1000; \]
\[ \text{ph2}(j) = \pi \times \text{time2} \times 1000; \]
end;
\[ \text{rec} = \sin(500 \times 2 \times \pi \times x) + 2 \times \sin(500 \times 2 \times \pi \times x + \text{ph1}) + \sin(500 \times 2 \times \pi \times x + \text{ph2}); \]
\[ \text{y3}(i) = 20 \times \log(\max(\text{rec}) / 4); \]
end;

for i=1201:3600
    a2=0.7*sin(pi/3-(t(i)*pi/180));
    time2=a2/350;
    for j=1:100
        ph2(j)=pi*time2*1000;
    end;
    rec=sin(500*2*pi*x)+sin(500*2*pi*x+ph2);
    y3(i)=20*log(max(rec)/4);
end;
plot(t,y3);
axis([0 360 -35 0]);