Acoustic Beamforming with a MEMS Microphone Array

4th Year Project Final Report

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**Abstract**

Conference room video calls carry a lot unnecessary noise in the audio signals that are transmitted from the conference room to the receiving party. The noise can be the fan in the room or chatter that occurs from other people in the room. An array of microphones is used to capture the audio sound signals and transmit them to a microcontroller. The following paper discusses a new method of eliminating the noise that occurs in the room and minimize the number of microphones in a conference room with the use of Microelectromechanical Systems (MEMS) Microphones. Using delay sum beamforming and beam steering, a direction of arrival for the sound signals can be achieved. The paper also discusses how an engineer impacts the society, the environment and finally the economy. The consideration that engineers take to maintain public health and safety and to minimize the risks related public and environment health.

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**I. Introduction**

1. Overview and Background

A method that utilizes principles of depth, perception, triangulation and wave transmission is referred to beamforming. Beamforming is used to analyze a signal and determine the direction of transmission. The overview of the project is an analog sound signal is transmitted from any direction, then is received by an array of microphones and is then processed and analyzed digitally. The microphones used in the project are the Microelectromechanical Systems (MEMS) microphones. The signals are received by the microphones and using the Matlab software, the direction of arrival (DOA) of the sound signal to determined. The direction of arrival of the sound signal is the end goal of the project. The following report include the steps taken to achieve this, the theory and techniques used, the different methods that were considered and used to achieve the project goal and a discussion. The report will then conclude by summarizing the project and the future work that will be done with the project.

Beamforming

Beamforming refers to the technique that aims at improving captured sound quality by exploiting the diversity in the received signals of the microphone array [1]. The following technology can also be used in an active state, this is seen in sonar and radar applications. Beamforming limits the phase and amplitude at each array element to influence the direction of signal transmission. Beamforming can also track moving object by collecting the sound that the moving object creates. With the understanding of how waves act and are analyzed and the understanding the techniques of beamforming, the filtered sound is can be analyzed further to obtain a direction of where it was transmitted from.

Sensor Array

A sensor array is used to find the most efficient method for signal optimization and sound localization. The larger the effective distance of the sensor array yields a greater accuracy in signal processing during the analyzing phase. The distance between the sensors and the number of sensors used can be altered to find the most efficient configuration for the project. The distance between the sensors can be configured by changing the spacing between the MEMS microphones in the array. The following method is more effective and a more practical approach to achieving the goal of the project. With the sensor array completed and utilizing the beamforming technique, the array of microphones captures a signal from multiple MEMS microphones and is processed in Matlab.

Matlab Software

Matlab is a powerful high-performance software that focuses on matrix and array mathematics. The software aids in the processing, analyzing and plotting of the captured sound signals. The use of the Matlab libraries allows for minimization of unnecessary sound and obtaining a source location.

Direction of Arrival

The direction of arrival focuses on when and how the analog sound signal arrives at each microphone in the array. Each signal arrives at a different phase and at a different time to the microphones in the array, depending on the position of the microphones in the array and the direction of where the sound signal was transmitted from. The sound localization principles stem from the direction of arrival of the sound signal to the sensor array.

2. Motivations and Significance

The objective of the project is to investigate and examine beamforming methods using a STM32F401 Nucleo microcontroller and X-Nucleo-CCA02M1 microphone expansion board, four microelectromechanical systems (MEMS) microphones, Matlab software and audacity software. Sound localization is a very valuable tool that can be used when other observation methods have failed or have demonstrated to be less effective. Using the principles and techniques mentioned, the use of sound localization methods is vast. The applications that may benefit from the use of this technology are military, biomedical and acoustic. For example, a military application would be detecting, and the direction of a sniper shot. This technology will help determine the direction of arrival of the sniper bullet resulting in a reduction of casualties and injuries on the battlefield.

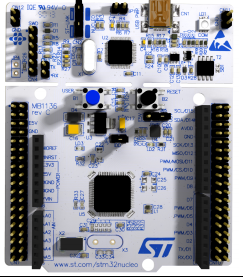
3. Objectives and Specifications

The objective of the project was to determine a direction of arrival of an analog sound signal with the use of a microcontroller and 4 MEMS microphones. The project compared simulated results that were created in Matlab to the ones generated from a built physical system in the lab. Next, the system was to be built in a way to be portable in order to use this technology in a variety of circumstances and different environments.

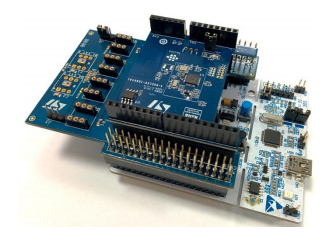
There are two main physical components that were used in the build of the project. A STM32F401 Nucleo microcontroller and four microelectromechanical systems (MEMS) microphones.

STM32F401 Nucleo microcontroller

The micro-controller used in the project is the STM32F401 Nucleo microcontroller. The nucleo microcontroller can be seen in figure 1 below. On board was attached was the X-Nucleo CCA02M1 microphone expansion board. The expansion board can be seen in the figure 2 below. The expansion board contains four on board microphones and has the option for additional microphones to be added. The jumper setting on the microcontroller board allows for external microphones to be added. Due to the formation and how close the built-in microphones are to one another; the jumper setting was turned on and the external microphones were used. The microcontroller is then connected to a computer through a USB, and then Matlab software is then used to analyze the signals created. The CCA02M1 microphone expansion board connected to the STM32F401 Nucleo microcontroller can be seen in figure 3 below.



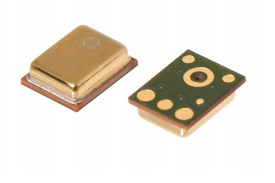
*Figure 1: STM32F401 Nucleo micro-controller [2] Figure 2*: X-Nucleo CCA02M1 microphone expansion board [3]



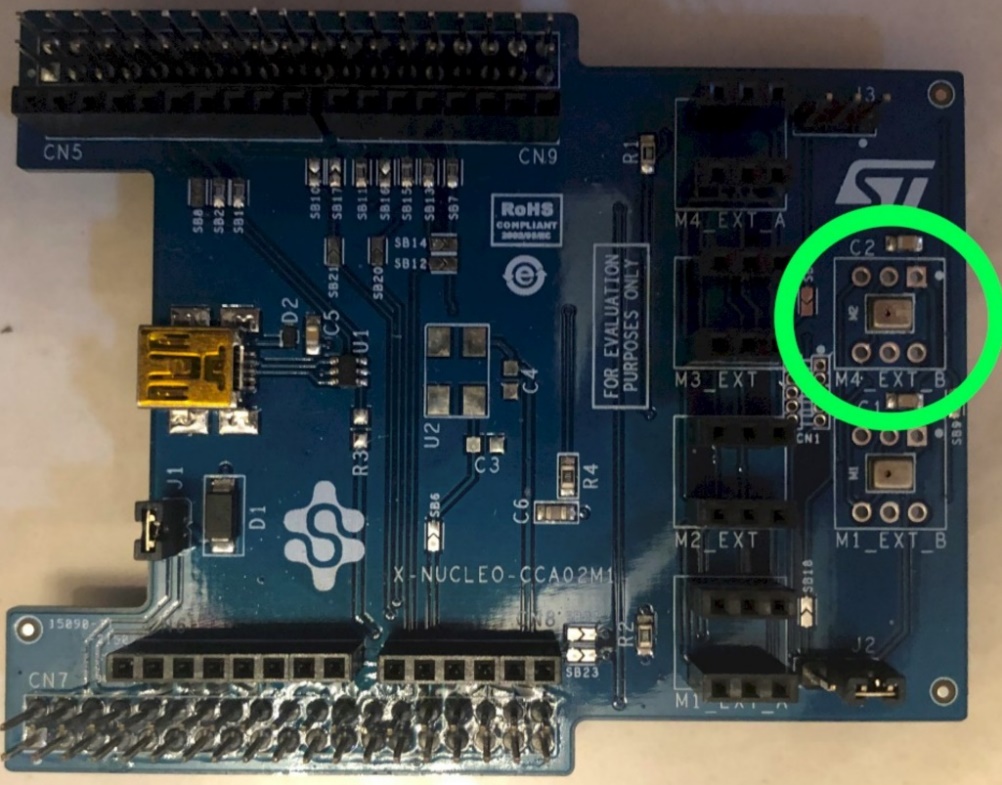
*Figure 3*: CCA02M1 microphone expansion on-board the *STM32F401 Nucleo micro-controller* [4]

Microelectromechanical Systems (MEMS) Microphones

The microphones that were used in the project are the Microelectromechanical Systems Microphones. The microphones can be seen in figure 4 below. The microphones have recently dominated the microphone market and due to that were the microphones tested and implemented into the project. The MEMS microphones offer high signal to noise ratio, low power consumption, and good sensitivity [5]. Two MEMS microphones are found on the extension board. The microphones can be seen in figure 5 below.



*Figure 4: Microelectromechanical Systems Microphones [6]*



*Figure 5: Mounted MEMS microphone on an expansion board [7]*

4. Relevant Publications

The team researched for the first month on how and what types of beamforming technique will be used. The methods of recording and the types of microphones to be used. A journal from the NRIAG Journal of Astronomy and Geophysics, titled, Optimization of MUSIC algorithm for angle of arrival estimation in wireless communications, discussed angles of arrival for elements(antennas) in an array [8]. The topic had similar attributes to the one discussed in this report.

Another publication that was researched was a paper by Iaian McCowan called Microphone Arrays: A Tutorial. The paper discussed different methods of beamforming, such Delay-sum, sub-array delay-sum, superdirectivity, near-field superdirectivity, AMNOR, Post- filtering, and finally Generalized sidelobe Canceler [9]. The result was the team settled on using Delay-sum due to the simplicity of the method compared to the others.

**II. Professional Considerations**

1. Engineering Professionalism
2. Professional certification requirements

The Professional Engineers Ontario (PEO) is an organization that regulates

the Practice of engineering in the province of Ontario [10]. The PEO is a self-regulating body that regulates and set standards to all professional engineers in the province. The reason for this is to ensure public and environmental safety. To become a certified professional engineer, there are five requirements that need to be present for an individual. The individual must have an acceptable engineering education, must pass the professional practice examination, must be of good character, must have 48 months worth of acceptable engineering experience and finally must have references from supervisors and a minimum of one professional engineer [10].

1. Health and safety at work

Health and safety in the workplace is something that should not be taken

lightly. The Occupational Health and Safety Act (OHSA) sets out the rights and responsibilities as a worker, the responsibilities of the supervisor and employer, the role of the government in enforcing the law and finally specific regulations for certain jobs. Everyone in the workplace has a role in preventing injury or illness, it is their legal duty in keeping the workplace safe and healthy [11]. There exist three types of work hazards, acute (occurs in minutes), chronic (occurs in a few years) and latent (occurs in 20 or more years) [11].

As a professional engineer their role is determining if a Pre-Start Health and Safety Review (PHSR) is required or not and performing them, examination of physical conditions or equipment, discussions and observations regarding the use of equipment of a process and examination of drawing and calculations related to the equipment to determine safe use [11].

1. Code of ethics and protection of public interest

The code that all professional engineers follow is the Code of Ethics from the

Professional Engineers Ontario (PEO). Engineers are bound by legal and Ethical responsibilities to follow the Code of Ethics. It is the duty of a practitioner to act at all times with fairness and loyalty to everyone they are associated with. Must act with fidelity to the public, devotion to high ideals of personal honour and professional integrity, knowledge of developments in their area of expertise, and competence in their performance of any professional engineering services [12]. A professional engineer should always take the public health and their needs into consideration when planning, designing and implementing their work in the field. There are penalties to acting incompetent by disregarding the code of ethics or not following them. The penalties are as follows, revoke in license, 24 months suspension, $5000 fine, restrictions, counselling and publishing the details of its findings [13].

1. Impact of Engineering on Society and Environment
   1. Place of engineering in society

Engineers have a big impact in society and the way society functions. The role

that engineers play is very important as it shapes the way society goes throughout their day. Engineers create, design and build many new inventions that society use. They need to ensure that the societies needs and safety is their top priorities. Engineers must evaluate all consequences that may arise from their work and ensure that they follow the code of ethics that they are required to follow by law. Society places a trust in engineers, and engineers must not break that trust.

* 1. Sustainable design; life-cycle planning

When engineers work on projects that require design or implementing

features to their projects, they must ensure that the product that is created for society has a long-lasting life-cycle and must be sustainable. For a product to be sustainable it must be environmentally sustainable, economically sustainable and finally socially sustainable. These three are the three pillars of sustainability [14].

* 1. Interactions (with society and stakeholders)

Engineers impact the society, the economy and the environment. There

are many uncertainties that may arise in the world around them. Uncertainties such as new environmental laws or budget cuts in the work place or even uncertainties in the evaluation of engineering projects. An engineer must always be ready for the change that may occur and adapt promptly to the changes, even when it might cost the company, or stakeholders money. An engineer will always stay true to the law, environment and the society.

* 1. Health, safety, and risk

An engineer must always understand the health and risks that may arise

from working on a project. The health, safety and risks may not only damage society but also the environment and the economy. All three factors must be considered by an engineer when working on a project. As stated above, the society and the law trust that the engineers will perform their work with the health and safety of the society, environment and economy in mind.

1. Economics and Project Management
   1. Engineering economics
   2. Risk and change management
   3. Individual and team work
      1. Personal and Group Time Management

Many factors have affected mine and the teams time to be able to provide

sufficient time to meet regularly and work together. Working two jobs to provide for myself and having five and six courses in the winter and fall semester respectively, made it hard for me to meet at the scheduled meetings. However, I was able to meet weekly to work with my team members on our part of the project together. Being able to meet with the entire team was difficult as my available time slots did not match the rest of the team.

* + 1. Group Culture, Group Dynamics

The group culture and group dynamic was one made up of frustration and poor

internal communication. The team had trust in each other to complete their own sub tasks, however, during the first semester, the lack of internal communication between the teammates, about being stuck led to frustration among the team. The physical product was with two team members for two months, and they did nothing with it. The team members never asked for help and that led to a loss in trust between the teammates.

* + 1. Leadership, Initiative, and Mentoring Teammates

I have taken many opportunities in leading the team. I have suggested many

different opportunities to get passed a roadblock. For example, I took the initiative in using of a different microcontroller and supplying the Arduino Mega and using different microphones in the project. I have also suggested different spacing measurements when I physically built the product.

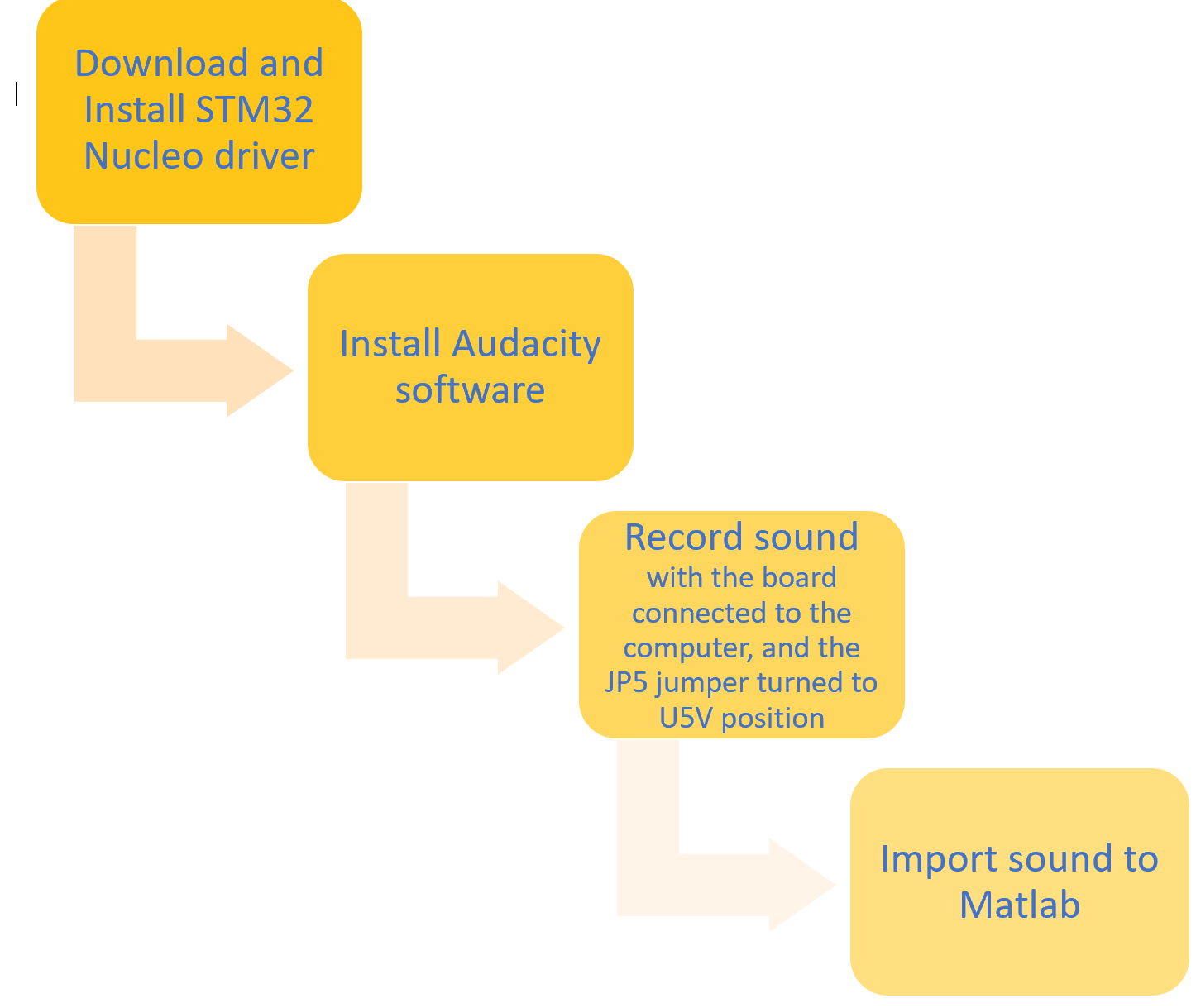
1. Project Management

To ensure that the objectives that were set out to be completed from the start of the project were met, the team members divided the work load into three subsections. The team of six set out to complete these three subsections in groups of two. This allowed the team to work in parallel to one another and ensured that each member completed the job that they were assigned from the start. The three subsections that were created within the project were as follows. The first group worked on the layout and the formation of the MEMS microphones. There was no user manual that could be found as the microphones used have recently been dominating the microphone market. Therefore, understanding and learning the microphones were a challenge. The group focused on finding the most efficient use of the number of microphones being used. The next group focused on understanding and simulating beamforming and localization algorithms such as array signal processing algorithms. Finally, the last team focused on optimizing and implementing the algorithms on the microcontroller. The project used Matlab as its main software to achieve the tasks for each group.

**III. Theory and Techniques**

The Technique used to obtain the direction of arrival of sound signals within in the project, is by creating simulated signals in Matlab and using the libraries that were available to us within the software towards the physical product. The following technique would help the team use the most useful methods and libraries in the simulated analysis into the physical device. This would eliminate any unnecessary methods from implemented resulting in a prolonged process when the code is ready to be implemented on the microcontroller.

Next, the microcontroller was then connected to a computer by a USB port. Audacity a software to record sound from microphones was installed onto the computer. Sound was then recorded from a direction and was saved and exported as a wav file onto the computer. The wav file was then imported into Matlab using an audioread () function and was plotted on Matlab to view the signal that was created from the two microphones 20cm apart. Figure 6 below shows the steps taken to complete this.



*Figure 6: Technique used*

Delay Sum Beamforming

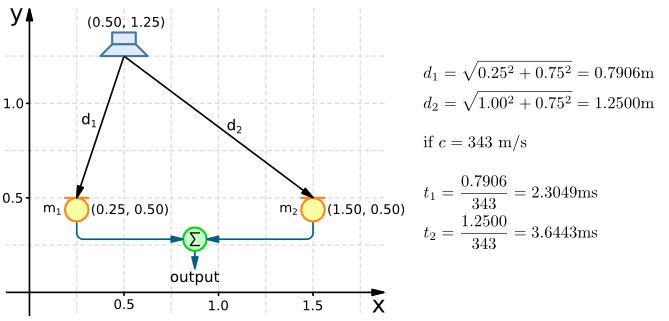
A method used in determining a beam pattern within the array of microphones, a frequency response calculation of the array, determining the microphone spacing and the number of microphones to be used and finally, beam steering.

Beam Pattern

Beam pattern is determining the sensitivity of a microphone array for signals arriving from one direction [16]. To calculate the array’s gain for one frequency, an equation is derived using wave propagation and summation. The figure below demonstrates how to calculate, the distance d1, and d2 and the time that sound will arrive at each microphone. Equation 1 and 2 below, show how distance d1, and d2 and time t1 and t2 are calculated. Distance is calculated using the Pythagorean Theorem as seen below.

(1)

(2)



*Figure 7: Wave Propagation and Summation [17]*

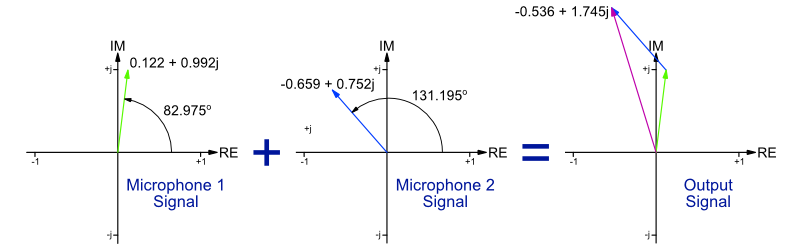
Next phasors are used to calculate the amplitude of the array’s output. The first step is to calculate the phase of both microphone signals. Using equation 3 below, the phase is calculated, and a phasor plot is generated to represent the amplitude and phase of the signal [17].

(3) where,

tx = time of sound arrival to x microphone

ᴦ = period of signal

Next the real and imaginary equation of the signal is calculated. The plots are converted from phasor plots to cartesian and polar coordinates. The output is calculated using vector addition. The figure below demonstrates phasor summation [17].



*Figure 8: Phase Summation [17]*

Finally using Euler’s Formula Representation, the output of the linear array is calculated using equation 4 below [16].

(4)

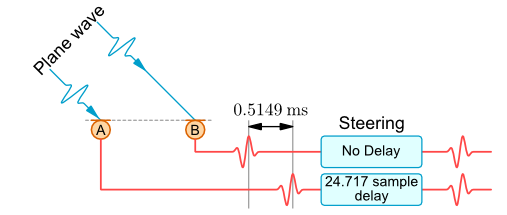
Note: The equation above is to calculate the output of the signals when the array is linear. In addition, attenuation of the signal is not considered.

Frequency Response and Microphone Spacing

Next the frequency response of the sound signal is determined. The amplitude and the phase of the sound signals are determined and this is done in Matlab using the freqz() function. The spacing of the microphone is important as the performance of the beamforming is dependant on it [16].

Beam Steering

A method to minimize unnecessary noise or interference from the array. Beam steering will steer the beam pattern electronically towards the targeted signal. By adding delay to each microphone, the signals that arrive from another direction are aligned before they are added. Figure 9 below demonstrates the use of fractional delay to an array of 2 microphones A and B.



*Figure 9: Fractional Delays to 2 microphones [18]*

1. General Approach

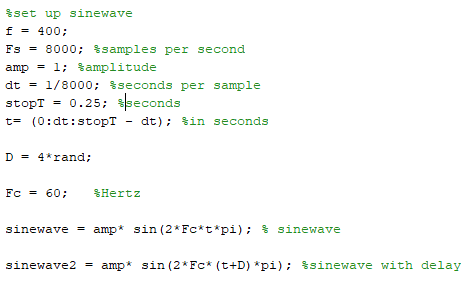
Different approaches were considered and even implemented into the project to complete the task. Firstly, the choice between what type of microcontroller was to be used in the project. A microcontroller that most of the team was familiar and had prior experience with, the Arduino Mega. On the other hand, using a microcontroller that the team was not comfortable in using due to the lack of experience, however, is a microcontroller that is used professionally, the STM32F401 Nucleo microcontroller. Another approach that was considered during the project, is what microphones were going to be used in the project. The MEMS microphones that was discussed above, or a funspark microphone that was supplied to the team.

1. Specific Methods

Matlab – Creating a simulated Signal

Before implementing code onto real time sound signals generated by the array of MEMS

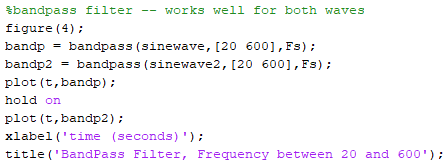
microphones on the microcontroller, two waveforms were created on Matlab using the sin () function with a sampling rate of 8000 samples per second that was chosen by the team. The reason two waveforms were generated was to mimic two microphones in an array. The sin () function generates a sine wave with parameters chosen by the team. Below is a snippet of the code used to generate both sine waves.



*Figure 10: Generating Two Sine Waves in Matlab*

The signals were then passed through a bandpass filter to minimize the noise

generated. The Matlab function to pass the signals through the bandpass filter is bandpass () functions. The reason for using filters on a simulated filter is to closely resemble what would occur if the signals were the physical sound signals that would be created by the microphones. The snippet below, is the code that implements both filter functions. The highpass cut-off frequency was set to 600Hz, and the lowpass cut-off frequency was set to 20Hz. The reason for the chosen cut-offs was the human fundamental speech ranges from 40Hz to 450Hz. Therefore, any sound signals that are outside the voice spectrum is deemed by the team as unnecessary sound and can be removed from the sound signals with the use of a bandpass filter.



*Figure 11: Bandpass Filter in Matlab*

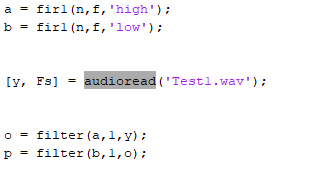
Next the cross correlation between the 2 signals was determined using a Matlab function xcorr (). The correlation between the 2 signals determines the similarities as a function of distance of the sound signals to one another. In this case the cross correlation returns the lag of which the correlations are computed [19].

After computing the cross correlation of the two signals, the microcontroller is connected to the computer and a sound signal is generated by the microphones.

Matlab - Receiving a Signal

After Audacity records a sound signal from a source, it is saved and exported as a wav

file. The wav file is then imported into Matlab and delay and summation algorithms are implemented to the signals. The method of importing the signal is done by the audioread function. The audioread function reads data from the wav file created and returns a sampled data with a sampled rate. The signal is filtered through a high and low pass filter. Below a snippet of code can be seen performing the high and low filter of the signal in addition to the audioread function. Note: The audio file is saved as Test1.wav



*Figure 12: Importing the wav file to Matlab and performing a low and high pass on the signal*

The delay and summation algorithms were implemented in Matlab. Using nested

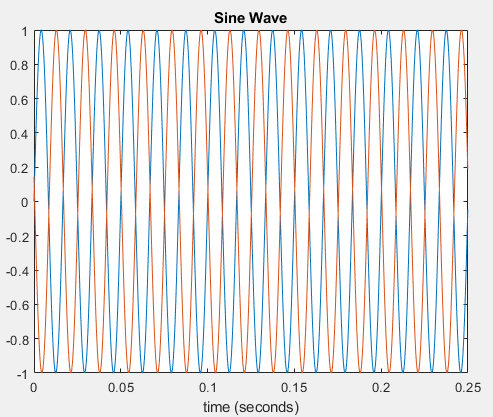
for loops. The first loop calculated the planewave arrival angle, and the second calculated the element position as it iterated through the array of microphones. The output was then calculated using equation 4 (implemented in Matlab).

**IV. Results and Discussions**

1. Results

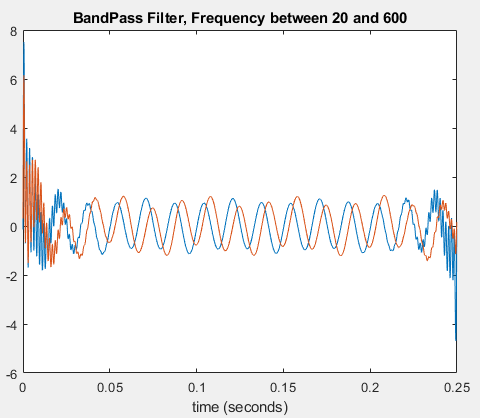
Matlab uses matrices as its basic data element. In addition, Matlab contains many

written libraries that the team uses to analyze the sound signals that are recorded and imported into Matlab. As stated in the report, Matlab code was written to create a simulation of 2 sound signals arriving at 2 different microphones. The code used to write the code to perform the operations can be seen in Appendix A. The resulting simulation plot can be seen in figure 13 below.



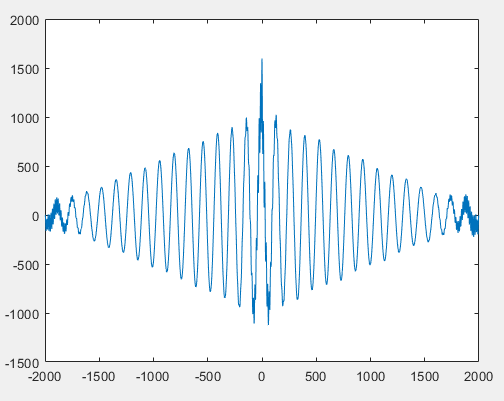
*Figure 13: Two sine waves created in Matlab*

Next, the waves were filtered through a bandpass filter to remove any additional noise that was present. The result can be seen below. The section that will be focused on are between 60 and 600. This will allow for the removal of any ambient noise that was present. Ambient noise such as fans blowing in a conference room, echoing, or even noise from the circuit that is being used. Figure 14 below shows the result obtained from the bandpass operation.



*Figure 14: Bandpass Filter, with upper and lower frequencies set to 600Hz and 20Hz*

Next the cross correlation between the 2 signals was determined. Cross correlation is a measurement that tracks the movements of the 2 sound signals relative to each other. It measures the similarity between the one sound signal to the other as a function of lag. Getting the cross correlation of the 2 waves allows us to begin calculating the Direction of Arrival of the plot using the result of the cross correlation.



*Figure 15: Cross Correlation of the Two Signals*

Next using the other team members research and findings, implementation onto the microcontroller began. The measured signals from the microphones were plotted in Matlab after importing the data from Audacity. Figure 16 below shows the generated sound signals.

*Figure 16: Sound Signals Generated by the microphones*

1. Discussion

**V. Conclusion**

1. Summary

The project analyzed beamforming with 4 MEMS microphones that were separated and spaced out 20cm apart. The array of microphones captured signals by streaming sound and through delay sum beamforming, beam pattern, microphone spacing and beam steering, the team was able to determine the direction of arrival of sound towards the array of microphones. The report explains in detail how and why the techniques and methods used in the project. The ST Nucleo board contains beamforming applications, the microcontroller was connected to Matlab via USB and Matlab was then used to perform beamforming applications.

Signals were first generated in Matlab with delays to mimic real sound signals generated by the array of microphones. Matlab functions were implemented on the generated sound signals. After performing operations on the generated sound signals, real time audio signals were generated by the array microphones while the microcontroller was connected to Audacity. Audacity is a recording software that allows sound files to be saved and exported to the computer. The file was then imported into Matlab where the operations that were done to the generated sound signals were performed on the sound file that was created in Audacity.

Unfortunately, due to circumstances outside the control of the team, the project was not completed. Two team members have left the team at a crucial time in the project. When implementing the code into the microcontroller, one team member who had the job to do the task, failed to do so and resulted in an incomplete project.

1. Design Implementation and Practical Use

The produced design was created with the intent of minimizing the number

of microphones and minimizing any unnecessary noise that results from the number of people available in a conference room. When communicating in a conference room with people outside the room, noise is often an issue at the receiving end of the conference room. A typical conference room has people in the room. Any chatter in the room between coworkers is often captured when on call with another party. The product created looks to eliminating the unnecessary noise and reduce the number of microphones in the room, using the beam steering technique discussed in the report.

1. Instructions for Using the Produced Design

The instructions to use the product designed in the report is simple. Place

the microcontroller in the center of the conference room. Then place the four (ideally) microphones around the microcontroller 20cm apart in a diamond shape. The microcontroller is then powered by the use of a computer via USB port or by connecting the power cable to a wall socket. The product will then run the algorithm without the interference of physical contact by the users (Adding code to constantly run when the microcontroller is connected).

1. Future Work

Due to two people dropping the project, the tasks assigned to them were

lost. Therefore, the first step for the project is to complete it by implementing the work done by the rest of the teammates. Next, adding more microphones to the project. The project tested the use of 2 microphones. As seen in the report, two microphones were used in the simulation phase and when performing all calculations. Therefore, optimizing the design to use more than 2 microphones and implement the design.

Next, adding code to the microcontroller to operate automatically as opposed to have to manually operate the microcontroller to perform the operations designed by the team. Another feature that would be added is a LED system that would light the direction that the sound signal is arriving to the system created.

Finally, the last addition to the product is to create a method to allow the

product to be portable. Adding an option to make the product portable would increase the use of the product to multiple fields. For example, the portable device could be used for the military as a defensive tactic to counter offensive snipers. The device could pinpoint the direction the sniper shot was taken and help the forces defend against the oncoming shots.

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VII. Appendix

1. Appendix A- Generated Code

%testing high and low pass filter using a generated sine wave.

clc;

close all;

clear;

%set up sinewave

f = 400;

Fs = 8000; %samples per second

amp = 1; %amplitude

dt = 1/8000; %seconds per sample

stopT = 0.25; %seconds

t= (0:dt:stopT - dt); %in seconds

D = 4\*rand;

Fc = 60;%Hertz

sinewave = amp\* sin(2\*Fc\*t\*pi); % sinewave

sinewave2 = amp\* sin(2\*Fc\*(t+D)\*pi); %sinewave with delay

%sine wave

figure(1);

plot(t,sinewave);

hold on

plot(t,sinewave2);

xlabel('time (seconds)');

title('Sine Wave');

% high pass filter -- works well with both delays

figure(2);

highp = highpass(sinewave,300,Fs);

highp2 = highpass(sinewave2,300,Fs);

plot(t,highp);

hold on

plot(t,highp2);

xlabel('time (seconds)');

title('Highpass Filter, The passing Frequncy is 300Hz');

%low pass filter -- works well with both delays

figure(3);

lowp = lowpass(sinewave,60,Fs);

lowp2 = highpass(sinewave2,60,Fs);

plot(t,lowp);

hold on

plot(t,lowp2);

xlabel('time (seconds)');

title('Lowpass Filter, The passing Frequncy is 60Hz');

%bandpass filter -- works well for both waves

figure(4);

bandp = bandpass(sinewave,[20 600],Fs);

bandp2 = bandpass(sinewave2,[20 600],Fs);

plot(t,bandp);

hold on

plot(t,bandp2);

xlabel('time (seconds)');

title('BandPass Filter, Frequency between 20 and 600');

%finding the delay of both plots

Delay = finddelay(sinewave,sinewave2) / Fs;

%getting the cross correlation of 2 time sequences

[Correlation, lag] = xcorr(bandp, bandp2);

[~,I] = max(abs(Correlation));

lagDiff = lag(I);

timeDiff = lagDiff/Fs;

figure(5);

plot(lag,Correlation)

alignw = bandp(-lagDiff +1: end);

talign= (0:length(alignw)- 1)/Fs;

figure(6)

subplot(2,1,1)

plot(talign,alignw);

xlabel('time (seconds)');

title('Aligned Plots');

subplot(2,1,2)

plot(t,bandp);

hold on

plot(t,bandp2);

xlabel('time (seconds)');

1. Appendix B – Implemented Code

clc;

clear;

close all;

f = 0.1;

n = 2; %filter order

a = fir1(n,f,'high');

b = fir1(n,f,'low');

[y, Fs] = audioread('Test1.wav');

o = filter(a,1,y);

p = filter(b,1,o);

subplot(2,1,1);

plot(y,'b');

subplot(2,1,2);

plot(p,'r');

d = y(:,1);

dt = 1/Fs;

t = 0:dt:(length(y)\*dt)-dt;

plot(t,y);