AUDIO LOCALIZATION

FINAL YEAR PROJECT UG 2019

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ABSTRACT

This report presents a method for localizing a sound source. The project is implemented to locate a sound source not visible to the naked eye. The source is localized using the TDOA (time delay of arrival) technique. The time delay of arrival technique calculates the delay of arrival of sound on to two microphones on an array and derives an angle from the delay calculation. The hardware consists of an array of two microphone being used at sampling rate of 192000 Hz to determine the general direction of sound by carrying out cross co-relation between both mics to find out TDOA. The project is intended to locate a gunshot or sniper in real time in the battlefield. The hardware was tested using intense sounds to replicate the gunshots as real gunshots could not be used. The hardware worked marvelously in noisy and reverberant environments. The angles calculated lay between +5 and -5 of the actual angles.

ENDORSEMENT OF CORRECTNESS AND APPROVAL

It is affirmed that data presented in this thesis "**AUDIO LOCALIZATION**" carried out by 1) Capt. Muhammad Atif Javed 2Capt. Zeeshan Arshad 3) Capt. Hassam ud deen 4) Capt. Umer Waqas under the direction of Asst. Prof. Dr. Muhammad Imran is in complete satisfaction of our level of Bachelor of Telecommunication Engineering, is right and endorsed. Percentage of plagiarism found in document as per software Turnitin available on LMS NUST comes out to be ____

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____

Asst. Prof. Dr. Muhammad Imran

Project Supervisor

DECLARATION

We hereby declare that no content and variety of work bestowed during this thesis has been submitted in support of another award of qualification or degree either during this course or anyplace else.

This proposition is devoted in thanks to **ALLAH ALMIGHTY,** our Creator, who has blessed us with wisdom, knowledge and understanding, then to our parents for their direction and their endless support. I would also like to thank our faculty for their guidance and supervision. Without their help and supervision this project would not have been made possible.

ACKNOWLEDGEMENT

The accomplishment of this project would not have been possible without the contribution of different people the names of which cannot be reckoned with. Their assistances are sincerely respected and thankfully recognized. We would like to thank following:

- To Allah Almighty who is the most Powerful and Merciful and everything is in His obedience.
- Assoc HoD Dr. Muhammad Imran for his support and guidance during the entire project.
- We acknowledge my sincere indebtedness and gratitude to my parents for their love, dream and sacrifice throughout my life. We are thankful for their sacrifice, patience, and understanding that were inevitable to make this work possible. Their sacrifice had inspired me from the day we learned how to read and write until what I have become now. We cannot find the appropriate words that could properly describe my appreciation for their devotion, support and faith in my ability to achieve my dreams.
- Lastly, we would like to thank any person who contributed to my final year project directly or indirectly. We would like to acknowledge their comments and suggestions, which was crucial for the successful completion of this study.

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ABBREVIATIONS

1.	TDOA	Time Delay of Arrival
2.	AOA	Angle of Arrival
3.	CC	Cross Correlation
4.	TDA	Time Delay of Arrival

Chapter 1

1.1 Introduction

1.1.1. Background

The audio source localization is a central and relatively new technique for defining the location of a target using its audio signal [1]. The project forms the basis a passive localization method, for tracing gun shots which produce noise. This project was employed to achieve a means for tracing a gunshot source and its potential applications for tracing a sniper in an open battle field scenario as well as other defense and civilian applications.

1.1.2. Problem statement

"Design and implement a real time audio source localization method prototype." The project is designed for applications in military, civilian, Police and paramilitary forces and conference rooms.

1.2. Project description

The audio source localization method consists of three parts, an audio sensor, in our case, microphones, a processing unit, in our case a Laptop, and an out display, a laptop or an android device. The location of a source are estimated using the method of cross correlation in time domain. The audio from the source arrives the microphones at a speed of 329 ms^{-1}. If more than one microphones are used and placed apart, the audio from a single source arrives both the microphones at different times, reliant upon the location of the audio producing source and the microphones. To determine the angle of arrival of the audio, relative to the microphone array, we first need to calculate the time difference in the received signals. This difference is then converted into the bearing angle of the source. To calculate the time delay, the method of cross correlation is used. In this project the location of a source is calculated by calculating the angle of the source relative to the microphone array. The calculated angle is then being displayed on a GUI.

The microphones receive the audio from the source and relay it to the laptop. The Laptop processes the signals from the microphones to get the source bearings and display the result in a GUI.

1.3. Prospective Application Areas

1. Defense and military applications to localize artillery fire and sniper fire etc.
2. Police and paramilitary forces, to enable precise and quick response.
3. In buildings to pinpoint potential break-ins.
4. Conference rooms.
5. Telepresence technique.
6. This project will add a new surveillance system to better locate sources.

1.4 Scope, Objectives, Specifications and Deliverables

1.4.1. Scope and Objectives

The scope of this project was to achieve the following objectives,

1. Planning a passive localization method
2. Understanding the behavior of audio in noisy environments
3. Designing an algorithm for calculating the time delay integral to audios received from different microphones
4. Implementing the algorithm on android application.

Following are the goals we achieved with the project

1. Efficiently determining the location of source in lateral plain
2. Modular design, we can replace the microphones without making any major change in the setup
3. Different sources may be localized, e.g. gunshots, claps, voice etc.

1.4.2. Specifications

1. Omni directional microphones.
2. 2 x microphones to acquire audio
3. Python IDLE.
4. Laptop GUI

1.4.3. Deliverables

1. Microphone array
2. Tripod stand
3. Microphone interface

Chapter 2

2.1. Literature review

2.1.1. Overview of existing literature

The audio source localization method is responsible for detecting a source, based upon its audio signature, and displaying the results in a GUI. Following are the functions performed by the method,

1. Detect audio in the environment where the method is placed
2. Determine whether the detected audio is useful or not
3. Evaluate time delays between the signals received from multiple sensors
4. Determine angle of arrival based upon the time delays in received signals
5. Display angle on GUI

2.1.2. CC

Cross-correlation is a quantity of similarity of two series as a function of the lag of one relative to the other [3][4]. It is used for examining a long signal for a shorter, known feature. It has applications in pattern recognition, single particle analysis, electrontomography, averaging, cryptanalysis, and neurophysiology.

For functions *f* and *g*, the cross-correlation is given in (i):

$$(f \star g)(\tau) \stackrel{\mathrm{def}}{=} \int_{-\infty}^{\infty} f^*(t)\ g(t+\tau)\, dt, \qquad \text{(i)}$$

Where f^* is the complex conjugate of f and τ is the delay

Similarly, for discrete functions, the cross-correlation is defined in eq (ii)**:**

$$(f \star g)[n] \stackrel{\mathrm{def}}{=} \sum_{m=-\infty}^{\infty} f^*[m]\ g[m+n]. \qquad \text{(ii)}$$

In this part the time-domain implementation of CC is briefly reviewed. First, two time-domain signals are separately converted into the frequency-domain in which they are multiplied with each other, giving Cross Correlation. This result is then converted back to the time-domain to acquire a correlation function. The position corresponding to the maximum cross correlation will be the TDOA [2]. On its turn the TDOA will indicate the angle of audio source, given the array geometry.

Theoretically, when converting back to the time-domain, this should make the correlation function a unit impulse function (neglecting noise, echo,). As a result, additional peaks in the correlation will not affect the spike from the direct path, giving better location approximations.

2.1.3. TDOA

The TDOA or time difference of arrival is the difference in arrival time of a excitation on to multiple distant sensors [3]. In this case the microphones are placed 1m apart on the array, the audio from a source arrives at the microphone at different times, reliant upon the location of the audio source as well as the array relative to it. The TDOA then calculates the angle of arrival or AOA. A minimum of two sensors are required for calculating the delay in a single dimension. We employ two microphones as only two are supported by the audio card. One being left channel and other being right of stereo input.

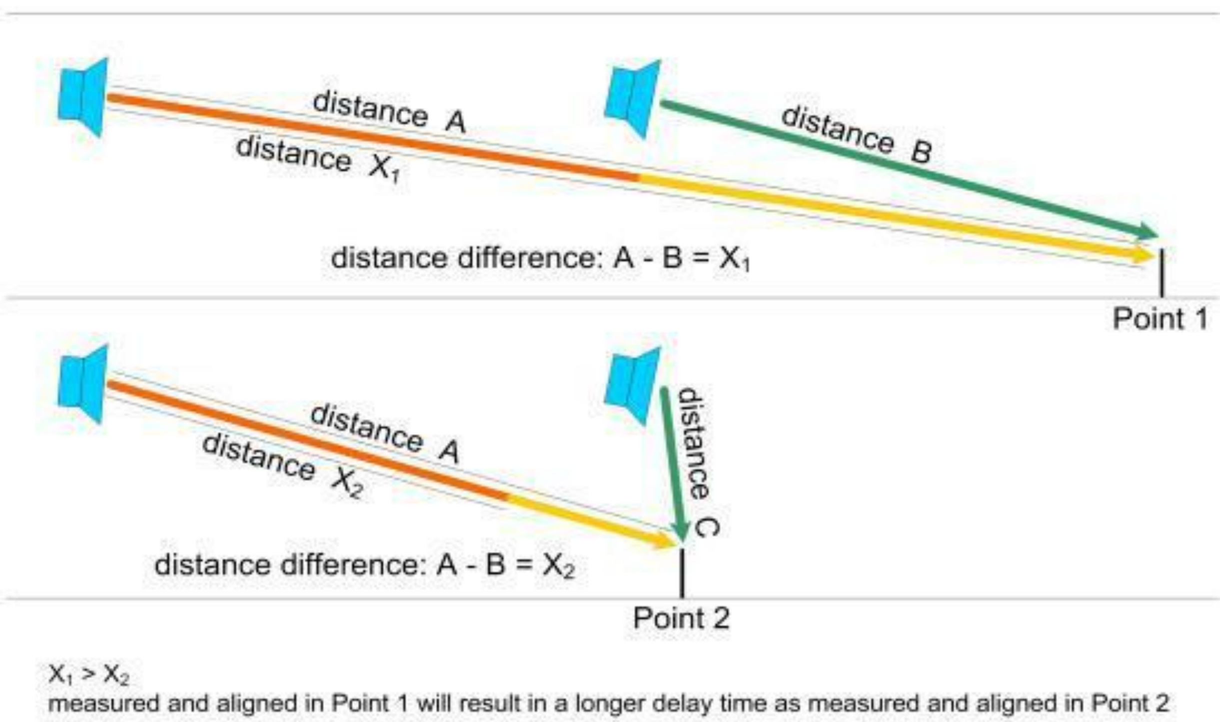

Figure 2. 1: TDOA

TDOA of a signal can be used to calculate the actual angle from where the signal is coming. One of the microphones is considered as the primary and the other, the secondary. The TDOA is calculated relative to the primary, and reliant upon its magnitude, the AOA is calculated. If the source is exactly in front of the array, the TDOA is zero, in this way, towards one side we get the maximum positive TDOA, and on the other we get the minimum possible TDOA.

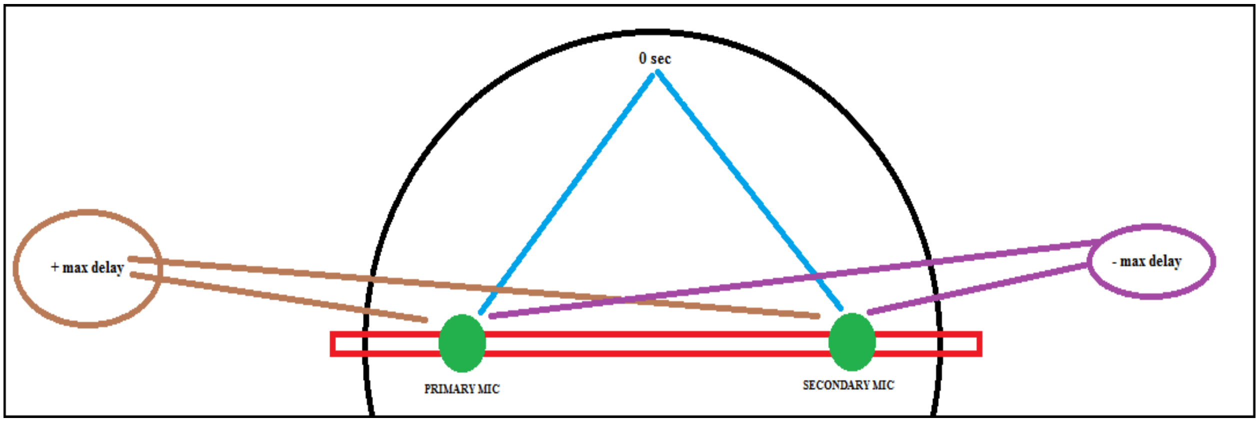

Figure 2.2: TDOA

The figure shows the TDOAs calculated, relative to the primary microphone.

2.1.4. AOA

AOA or the angle of arrival is calculated using the TDOA, by employing trigonometry. The AOA correctly gives the location of the source [4].

The AOA is directly related on the TDOA. The angle is zero if the source is directly in front of the array, it is +90 degrees for the +maximum delay and -90 degrees for the maximum delay.

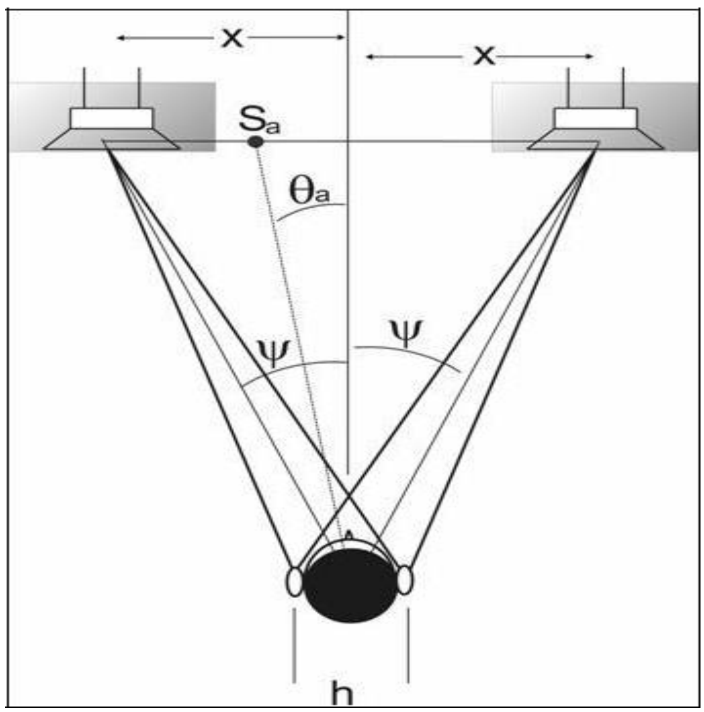

Figure 2.3: Angle of Arrival

2.2. Problem formulation

The audio source localization method is a new surveillance and localization method. To achieve the localization, we selected the minimum number of sensors and efficiently calculating the angle of the source relative to the microphone array in a 360 degrees field of view. The method would detect audio through the microphone array mounted a-top a tripod stand. The detected signals are then subjected to A/D conversion. The digitized signals are then processed, and angle of arrival is taken out from them. The results are then displayed on a GUI.

Chapter 3

3.1. Detailed Design

The project is divided into two major parts, the hardware and the software. The hardware consists of Laptop and two Omni directional microphones. The software part consists of the algorithm for calculating the angle as well as the code for the GUI made in Python.

The sensors would send the signals that they pick up, to the laptop and the Python IDLE algorithm in the laptop would determine the angle of arrival based upon various factors in the detected signals.

The angle, so determined by the algorithm are then be displayed on a GUI.

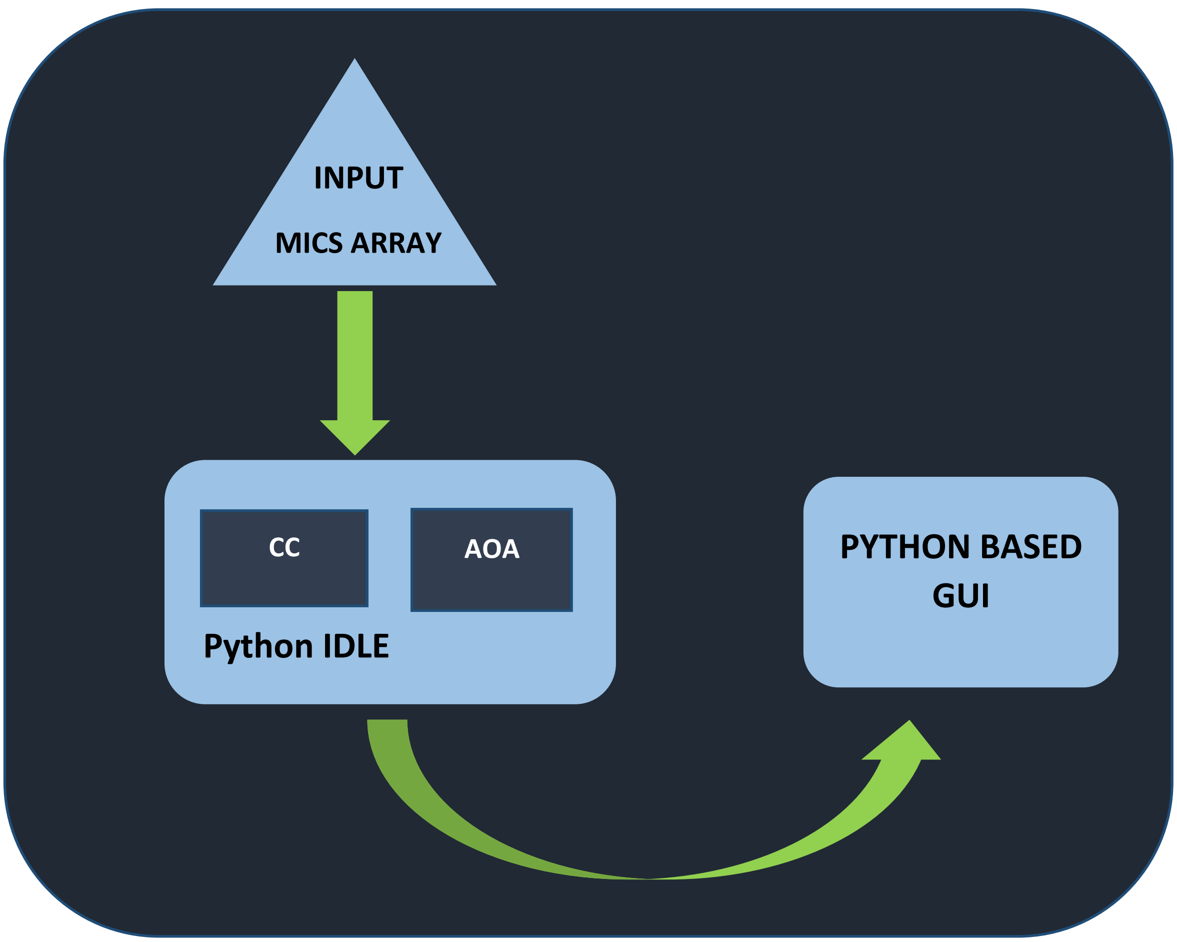

Figure 3.1: Method Flow Design

The figure 3.1 illustrates the flow of the project, i.e. the audio is captured by the microphones and sent to laptop where it is processed upon. The laptop then displays the results on GUI and shows the live stream of the source.

3.2. Hardware design

The hardware part consists mainly of the sensors. The sensors are mounted on an array and attached onto a tripod stand. A Laptop. The details about each hardware component are illustrated below.

3.2.1. Microphones

The sensors used for detecting the audio of the source are the microphones. The microphones work on the same principle as speakers, only in reverse. The vibrations in the air caused by the audio source travel towards the microphones and cause the filament to move, this motion is the same as the audio produced. These vibrations are converted to electrical signals. These signals are perceived as audio by the audio card of the laptop.

The microphones used for the project are Omni directional microphones. They are readily available in the market.

To get the optimum delay between the audios captured by the three microphones, we keep the microphones 1m apart, this would mean that the maximum delay experienced would be given by the relation

$$S=VxT$$

Here ‘S’ is the distance among the microphones, kept at 1m, ‘V’ is the speed of audio in air, i.e. $344ms^{-1}$, and ‘T’ is the time difference. Now, for maximum time delay, the source must be exactly perpendicular to the array, on either side, and thus the signal would travel at least1m further after reaching one of the microphones.

Figure 3.2: Microphone

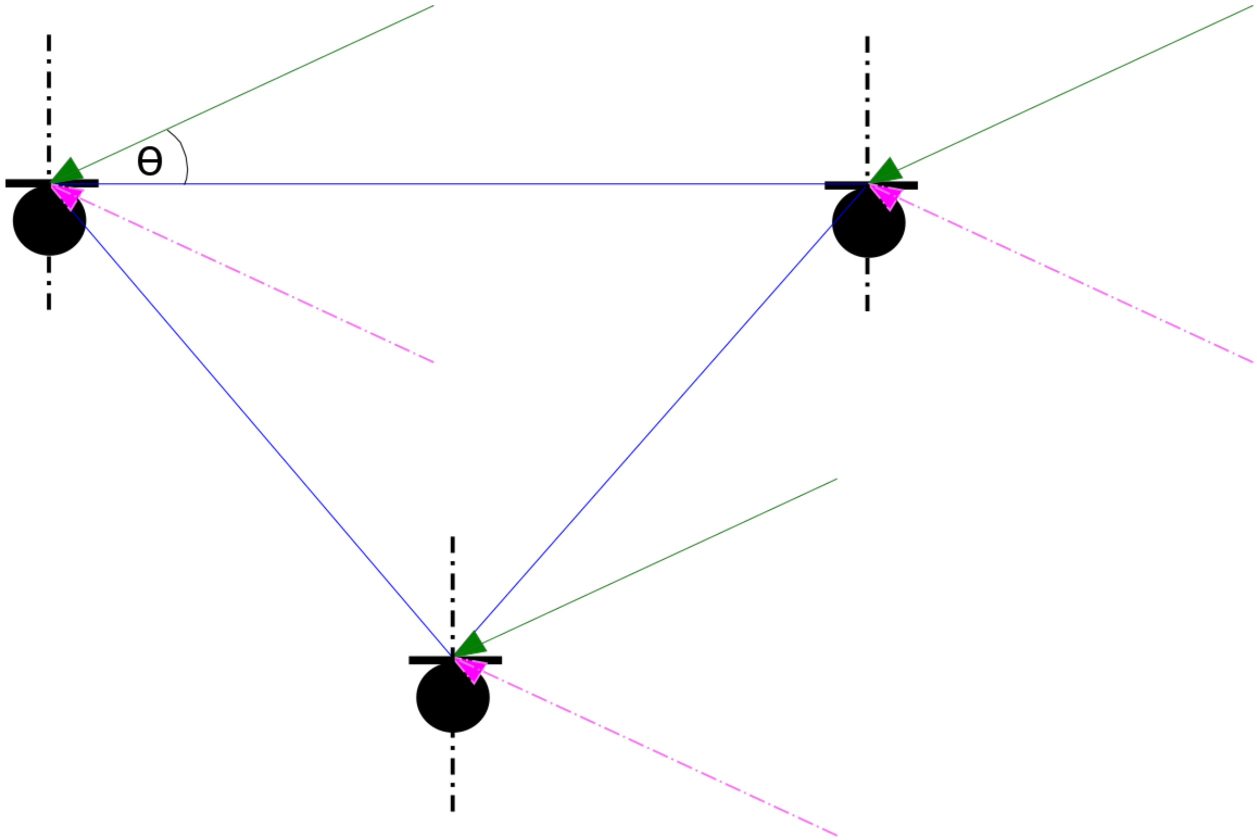

Figure 3.3: Angles of Arrival relative to array

The figure 3.3 shows the angle of arrivals which are calculated relative to the microphone array. The three microphones used are only able to provide the location of the source in one dimension, in our case the isosceles triangle configuration.

3.2.5. Linear Stand

The microphone assembly is mounted on to a linear stand. The height is variable, and this feature is required so that the height of the microphones is high enough to avoid the echoes caused by the floor.

3.3 Software development

The algorithm is implemented on Python IDLE. Many methods are available for calculating time delays. The CC, cross correlation [2] is chosen for the project. The choice of the TDOA [3] algorithm is predisposed by the environment. The very noisy environment causes delay calculation, which are useless for us. Therefore, a robust method is required for calculating the TDOA in noisy environment. Hence the CC algorithm is chosen.

The main function of the CC algorithm is to get a correlation graph of the signals detected by the two sensors. This correlation graph would enable us to determine the delay between the two signals. The flow of the algorithm is shown in the figure.

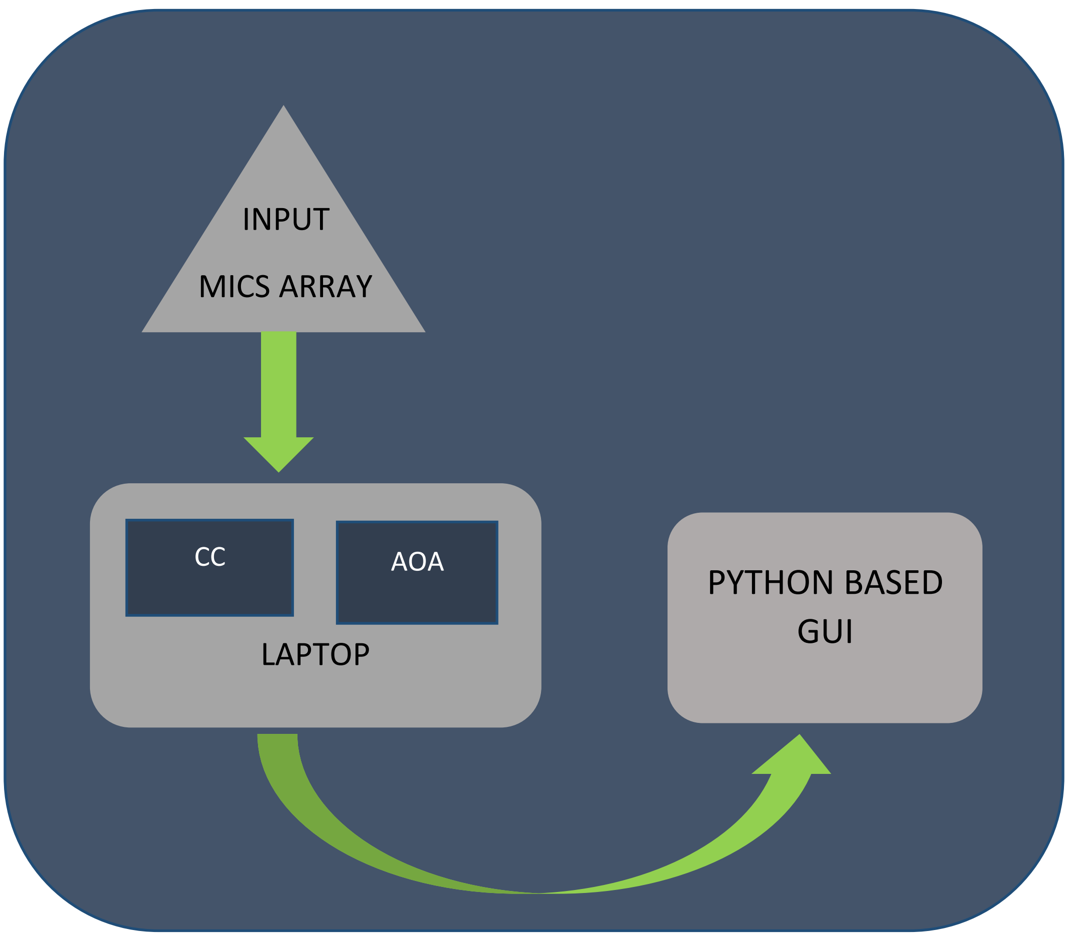

Figure 3.4: Flow Chart

The figure 3.4 illustrates the following steps followed in the development of the algorithm,

1. Acquire input from microphones
2. Check whether the signals have requisite energy
3. The selected frames are sent for processing
4. The TDOA is derived from the frame, and in turn the AOA
5. Display the AOA on GUI

The microphones, once on, pick up all kinds of noise, EM waves, Wi-Fi, mobile cellular signals etc. There are ways to filter out most of the noise, however the signals obtained are not pure. Therefore, the only option is to use the received signals. A threshold is kept filtering out speech and noise. As gunshot is an impulse and louder than normal sound. On crossing threshold of all connected microphones, the audio processing starts. Also, since the assembly is turned on, it must collect information about the surroundings. This is done to process only those signals that have an energy higher than a previously obtained threshold value, this saves both time and helps us get rid of cumbersome angle calculations.

In the hardware setup two microphones separated by 1m are soldered onto a single aux cable being left and right channels to the stereo input of laptop respectively. The audio processing then begins by getting the real time stream of microphones using pyaudio module in Python.

3.3.1 Acquiring device index of microphones

For acquiring the device indices of both microphones, a getaudiodevices() function of pyaudio is used. The python code designed for the above-mentioned purpose is given below:

```
import pyaudio

def getaudiodevices():
    p = pyaudio.PyAudio()
    for i in range(p.get_device_count()):
        print("\n")
        print("Index "+str(p.get_device_info_by_index(i)))
        #print (p.get_device_info_by_index(i).get('name'))


getaudiodevices()
```

Figure 3.5: Python code excerpt

The figure 3.5 shows the code excerpt for recording and acquiring the audio signals from the microphones. The following commands are used,

1. **PyAudio**: Python module for audio processing
2. **get_device_count** : It gives the count and index of all devices connected to laptop.
3. **pyaudio.PyAudio().open() :** It opens stream for microphone recording
4. **scipy.signal.correlate() :** It carries out cross-correlation of left and right channel stream

Few important parameters of our project are as follows:

1.	Mic distance	1 m
2.	Sampling Rate	192 kHz
3.	Speed of Sound at normal room temperature (22 ℃)	344 m/s
4.	Maximum sample difference between mics	N = 560

The BLOCK SIZE gives the length of the buffer and can e.g. be set to the length of the window that will be processed. FORMAT gives the resolution of bits in each buffer. CHANNELS represent the number of channels being used i-e left and right. RATE represents the sampling rate. CHUNK determines the number of frames per buffer. The THRESHOLD is set to check to filter out speech and noise. The configuration of PyAudio is comparatively easy and can be explained in Fig. 3.6.

```
BLOCK_SIZE = 2400
FORMAT = pyaudio.paFloat32
CHANNELS = 2
RATE = 192000
CHUNK = 1024
THRESHOLD = 0.95
MIC1_COORDS = (0,0)
MIC2_COORDS = (1,0)
sound_speed = 344
mic_dist = 1
flag = True
```

Figure 3.6: Py Audio

The mic stream is opened for recording using the code shown in Fig. 3.7

```
stream = self.pa.open(   format = FORMAT,
                         channels = CHANNELS,
                         rate = RATE,
                         input = True,
                         input_device_index = 2,
                         frames_per_buffer = CHUNK)
```

Figure 3.7: Stream self.pa.open()

Mic stream is read where 2048 bits are being processed at an instance. The resulting block is converted to float numbers. Stereo output is separated into left and right channel. A High pass and low pass filter is applied to incoming audio to discard speech and noise signals and detect only impulse signals(gunshot sound). The code for this purpose is shown in Fig. 3.8.

```
block = self.stream.read(BLOCK_SIZE, exception_on_overflow = False)

'''
The resulting stream from mics is being converted to float numbers
'''
decoded = np.frombuffer(block,dtype=np.float32)

'''
The left and right channels from stereo output are being separated for
further processing
'''
result = np.reshape(decoded, (BLOCK_SIZE, 2))

#Left and right channels maximum amplitude being obtained
amp1 = np.amax(result[:,0])
amp2 = np.amax(result[:,1])

#Left and Right channels being checked for Threshold

if amp1>THRESHOLD and amp2 >THRESHOLD:
    print("\n\nPlease wait................")
    print("Stream 1 latency")
    g = self.stream.get_input_latency()
    a = rfft(result[:,0])
    b = rfft(result[:,1])
```

Figure 3.8: Audio stream

The obtained audio signals are segregated and stored in different variables. After this the processing is carried out.

3.3.2 CC

After processing, the signals are forwarded to the main CC function from where the TDOA of the signal is obtained and the angle of arrival is obtained. The CC is based on the approximation that the correlation of the late varieties of the same signal is maximum at the point of the delay. Therefore, when the CC of the two delayed signals is plotted, there seems a distinct peak which shows the maximum correlation of the two signals.

Considering the signals from the two microphones as the late varieties of a single signal, we can say

S1= s(t+d1) + n1 ------------------(i)

S2= s(t+d2) + n2 ------------------(ii)

Here, it can be said that signals s1 and s2 are late varieties of the signal s(t) from a single audio source. N1 and N2 are the noise pathways of the two microphones. The time delay to be calculated for source localization is essentially, "d1-d2".

The signals from the two mics 'i' and 'j' are cross correlated as depicted,

$$\boxed{R_{ij}(\tau) = \sum_{n=0}^{N-1} x_i\,[n] x_j[n-\tau]}$$

This equation shows the time domain cross correlation, but because of noise, the time domain cross correlation gave bad results, thus cross correlation was done in frequency domain, as follows,

$$\boxed{G(f) = X_i(f) X_j(f)^*}$$

Here, Xi(f) and Xj(f) are the frequency domain depictions of the two signals from mics 'i' and 'j', and Xj(f)* gives the complex conjugate of Xj. the signals in frequency domain are multiplied with each other and the product graph is called the CC plot, giving the correlation of the two signals.

In Python IDLE, the algorithm is implemented in time domain. The signals are first converted to their respective time components. The signals now give their respective magnitudes and phases. After multiplication of one signal with the conjugate of the second, the phases are contracted, and a maximum is shown, which when converted back into time domain, yields the cross-correlation plot showing the sample delay.

The sample delay is converted to time delay by dividing it with the sampling frequency, in our case 192000 Hz is used. The TDOA is then converted into the AOA.

3.3.3. AOA

The location of the source is approximated in terms of angle of arrival. The angle is obtained from the TDOA calculated from the two signals. The two microphones used, give the angle in the horizontal plane only and in front of the assembly. The angle is obtained by using trigonometry as shown in Fig. 3.9.

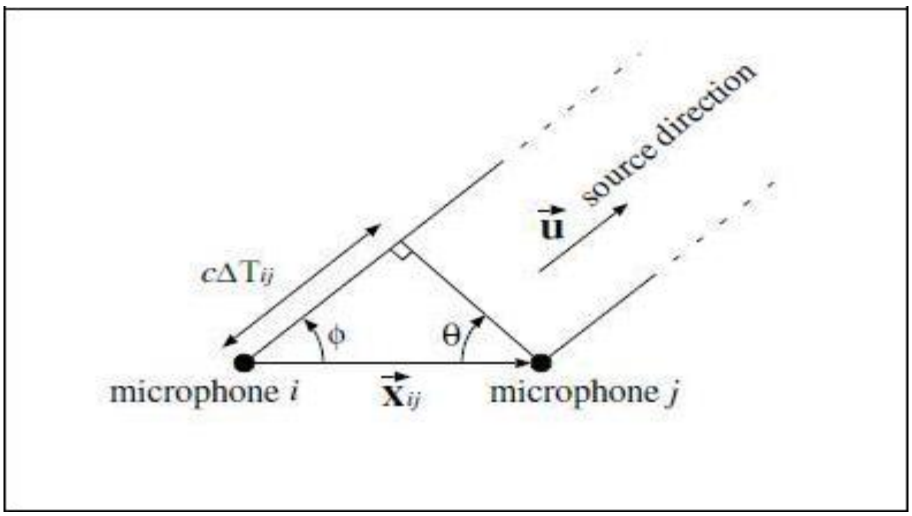

Figure 3.9: Angle of Arrival

From the figure, the ‘**xij**’ is known to us as the parting between the two microphones, this is the hypotenuse. The distance **‘c*Tij’** is the base for the angle that is the angle of arrival. Applying the formula relation,

$$\boxed{\cos\varphi = \sin\theta = \frac{c\Delta T_{ij}}{\|\vec{X}_{ij}\|}}$$

In this way the angle of arrival is obtained. This angle is then showed on a compass in android studio and a live video stream is also shown [4]. Fig 3.10 shows the hardware of the project in controlled environment.

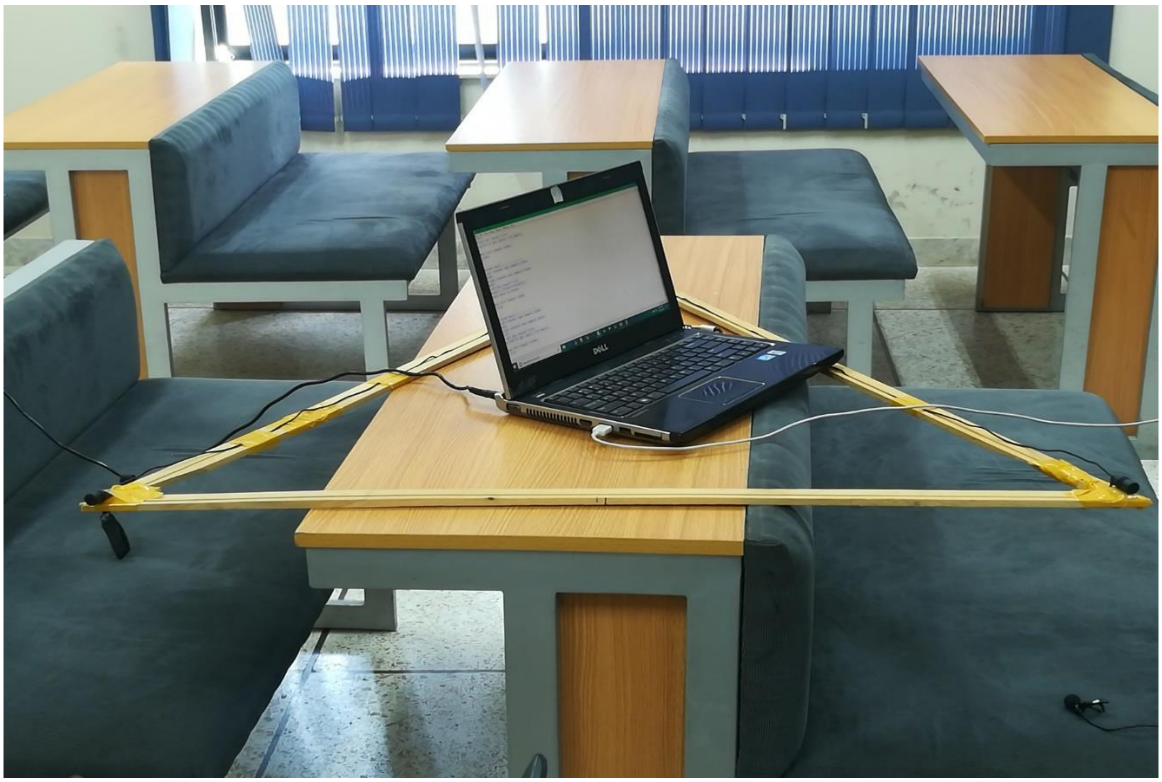

Figure 3.10: Hardware Setup

Chapter 4

4.1. Project Analyses and Evaluation

The simulation results are illustrated below.

- **Acquiring signals from microphones:**

The figure 4.1 shows the time domain signals captured by the microphones. The two signals from the microphones look the same, however, the change is in the time component, i.e. the signals are delay against the time.

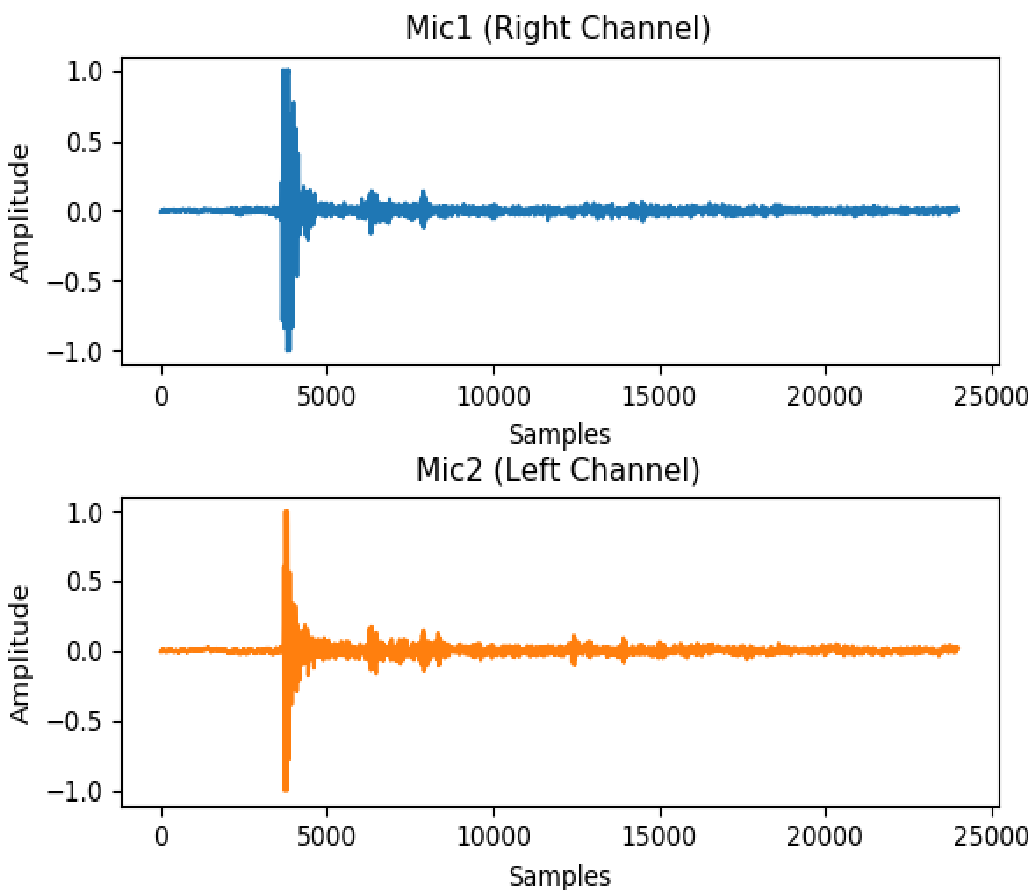

Figure 4.1: Acquired signals from microphones

- **Correlation:**

Fig. 4.2 shows the correlation between left channel and right channel. It shows peak at the maximum correlation between Mic1 and Mic2.

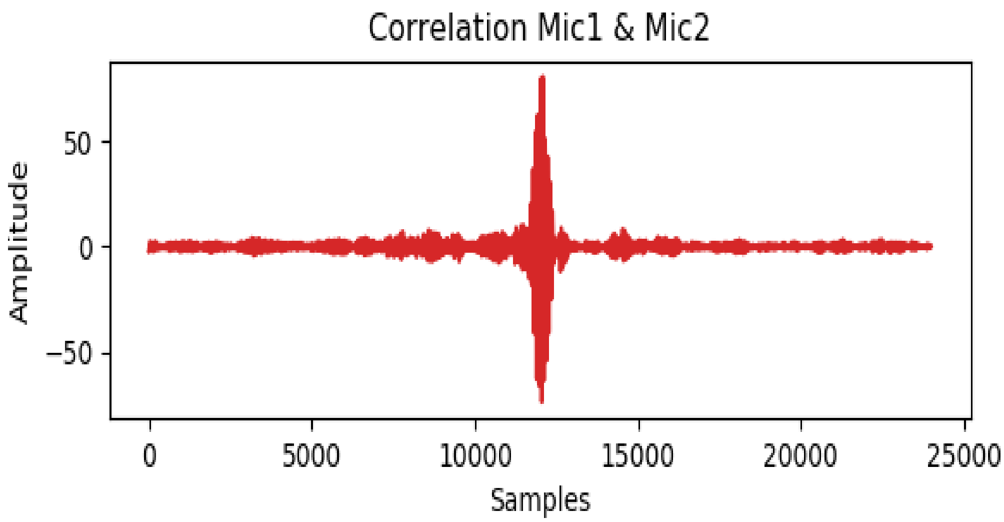

Figure 4.2: Correlation

4.2 AOA

The angle of arrival is simply obtained by the formula,

$$\boxed{\cos\varphi = \sin\theta = \frac{c\Delta T_{ij}}{\|\vec{X}_{ij}\|}}$$

The microphone assembly and the audio cause are well-thought-out to make a right-angled triangle. The angle of arrival is then simply the ‘phi’ or the ‘theta’. Fig. 4.3 shows the angle of the incoming audio signal.

Figure 4.3: Compass

Chapter 5

5.1 Recommendations and future work

1. The method prototype has been employed on Laptop, it may be gradually designed on to an explicit hardware like an FPGA kit or a DSP kit, to make it more accurate and integrated.'
2. The microphone used are the basic audio sensors, these may be replaced with the more modern pressure sensors, to improve range and sensitivity.
3. A PTZ camera may be used to track the source if it is mobile.
4. Several methods like this prototype may be connected to get the exact location of the source.
5. Increasing the number of arrays would enable azimuthal angle calculation.
6. Increasing the number of microphones per array may yield better approximation of the calculated angle.
7. Robust Signal Processing, if used, can enable us in segregating a single source amongst a bunch.
8. Multiple audio sources may also be located with a single method.

5.2 Conclusion

1. The audio source localization method guarantees the timely and exact reply of the Quick Reaction Force and Para Military Forces.
2. Helping in indicative the source in live operations helps the action to be quick and accurate.
3. In a conference room, audios can be localized, and the actual microphone and camera can be pointed towards the speaker.
4. In battlefield scenario, the prototype will help in locating snipers.
5. Bomb explosions can be gladly located.
6. It is a passive technique, therefore is not detectible.

5.3 Drawbacks

1. The method is successfully able to localize impulsive audios. The test is conducting by clapping firmly, and the method locates the clap noise successfully.
2. Very noisy environments wreak havoc with the calculations, this is realized after comprehensive testing, and this may be improved by using more expensive equipment.
3. The method is designed on Laptop, replacing it with a dedicated method would enable more efficient results and increase precision.

5.4 Advantages

1. The method is standardized for recognition of impulsive audios, at present ‘finger clapp’ noise.
2. The method can localize voice, gunshots etc.
3. The method displays the area from where the audio is originated.
4. A passive technique, undetectable by the enemy.
5. Jamming techniques do not work in this system.

References

[1]'Audio Source Localization Using Time Delay Estimation', Supercomputer Education and Research Center, Indian Institute of Science, Ashok Kumar Tellakula, August 2007

[2] ‘Acoustic Source Localization Based on Generalized Cross-correlation Time-delay Estimation’, China University of Mining and Technology School of Information and Electrical Engineering, Xuzhou 221008, JiangSu, China, Lin Chen,Yongchun Liu, FanchengKong, NaHe

[3] ‘Subsample Time Delay Estimation via Improved GCC PHAT Algorithm’ Bo Qin, Heng Zhang, Qiang Fu, Yonghong Yan

[4] X. Lai, H.Torp, Interpolation methods for time-delay estimation using cross-correlation method for blood velocity measurement, IEEE Trans. Ultrason, Ferroelect., Freq.Contr., Vol. 46 (1999), 277-290

Appendix A

Audio source localization

Extended Title:
Real time audio source localization method
Brief Description of The Project / Thesis with Salient Specs: The project aims at tracing an audio source by means of microphones, for obtaining input (audio), and methods such as time delay estimation (time delay on arrival, TDOA) and accumulated correlation for establishing the exact bearings of the source. The acoustic source produces sound waves which are received by multiple microphones placed in such a way that they have some distance between them. This phase delay enables us to triangulate the exact location of the acoustic source. The input obtained from the microphones would be integrated. The processing would take place on Python, using user defined methods to pinpoint the target.
Scope of Work The project will render the detection, of audio sources, easier. The sensor would locate the source In real time, by providing real time processing of the multiple audios In the input and separating the audios related to our source. Thus enabling the detection of that particular source.
Academic Objectives The project will impart following skills into the syndicate: 1. Signal processing techniques 2. Python language learning 3. Digital signal processing (audio) Localizing audio sources.

Application / End Goal Objectives:

The aim of this method is to provide an accurate mode for tracing a source, using audio waves. This is a passive technique and, unlike radar methods, cannot be detected. This method has vast military application for detecting and localizing enemy combatants and/or tracing artillery batteries. Civilian applications include integration with existing security method to detect acts of theft or terrorism by tracing the source of gunfire or movement and using surveillance to visually highlight the intruders.

Our objective is to demonstrate the audio localization method and its application by integrating a video camera to visually highlight the source's locale.

Previous Work Done on The Subject:

1. S. T. Birchfield**, A Unifying Framework for Audio Localization**, Proceedings of the 12th European Signal Processing Conference (EUSIPCO), Vienna, Austria, September 2004
2. S. T. Birchfield and D. K. Gillmor, **Fast Bayesian Audio Localization**, Proceedings of the IEEE International Conference on Audios, Speech, and Signal Processing (ICASSP), Orlando, Florida, May 2002

Material Resources Required:

1. Laptop
2. Microphones
3. LCD monitor
4. Tripod stand
5. Wires
6. Misc

No of Students Required: 4

Special Skills Required:

1. Digital signal processing
2. Circuit designing and soldering
3. Python coding

Appendix B

Timeline

	Nov 18	Dec 18	Jan 19	Feb 19	Mar 19	Apr 19	May 19
Literature Study	Completed	Completed					
Array Design		Completed	Completed	Completed			
Python Code			Completed	Completed	Completed		
Interfacing				Completed	Completed		
Python GUI					Completed	Completed	
Finalization						Completed	Completed
Testing							Completed

Appendix C

Cost Breakdown

Component	Quantity	Unit price	Amount	Remarks
Microphones	2	1800	3600	
Tripod stand	1	3900	3900	
Miscellaneous	-	2600	2600	Wires, etc
Total		-	Rs. 10100/-	

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