ACOUSTIC OBJECT LOCATING SYSTEM USING TRIANGULATION METHOD

By

HAZILAH BINTI HAMZAH

FINAL PROJECT REPORT

Submitted to the Electrical & Electronics Engineering Programme
in Partial Fulfillment of the Requirements
for the Degree
Bachelor of Engineering (Hons)
(Electrical & Electronics Engineering)

Universiti Teknologi Petronas
Bandar Seri Iskandar
31750 Tronoh
Perak Darul Ridzuan

CERTIFICATION OF APPROVAL

ACOUSTIC OBJECT LOCATING SYSTEM USING TRIANGULATION METHOD

By

Hazilah Binti Hamzah

A project dissertation submitted to the
Electrical & Electronics Engineering Programme
Universiti Teknologi PETRONAS
in partial fulfilment of the requirement for the
Bachelor of Engineering (Hons)
(Electrical & Electronics Engineering)

Approved by:

DR LEE YEN CHEONG
Lecturer
Fundamental & Applied Sciences Department
Universiti Teknologi PETRONAS
Tel: 05-368 7685 Fax: 05-365 5905
Email: lee_yencheong@petronas.com.my

DR LEE YEN CHEONG
Project Supervisor

UNIVERSITI TEKNOLOGI PETRONAS

TRONOH, PERAK

May 2011

CERTIFICATION OF ORIGINALITY

This is to certify that I am responsible for the work submitted in this project, that the original work is my own except as specified in the references and acknowledgements, and that the original work contained herein have not been undertaken or done by unspecified sources or persons.

Hazilah Binti Hamzah

ABSTRACT

In this paper we study the acoustic object locating system using triangulation method. This system consists of a set of acoustic sensors aligned in a straight line with equal distance apart. The sound intensity is measured by the acoustic sensor at different positions. Since the distance between the source and the sensors vary, the location of the source can be determined by the discrepancy of the intensity measured using triangulation method. The location of the source is then displayed on the plan position indicator. This report will be discussed about the background study, problem statement, objectives, and scope of study. In Chapter 2 we derive the key formula for the coordinate of the source of acoustic object. We then design the circuit based on the underlying concept. In Chapter 4, we apply programming in the system and show how the coordinate of the source will be displayed on the plan position indicator.

ACKNOWLEDGEMENTS

First of all I thank the almighty GOD, whom I believe his blessings this project will not be at this stage. I have not forgetting also my parents who despite the distance always cheer me and encourage me to work more.

Special thanks to my supervisor, Dr Lee Yen Cheong who were always there whenever I need any helps, guidance or clarifications. Your advice and comments really help me all the way through understanding and finishing this project. I will not forget also to thank my fellow friends especially, Faris Ahmad, Mr Yasir Salih who support and help me in this project. Also to all my friends and everyone that has been involved in helping my Final Year Project.

Lastly, I will not forget to thank Universiti Teknologi PETRONAS for providing me the environment to develop such project and pursue knowledge for my future betterment.

THANK YOU.

TABLE OF CONTENTS

LIST OF FIGURES viii

LIST OF ABBREVIATIONS x

CHAPTER 1 INTRODUCTION 1

1.1 Background of Study 1

1.2 Problem Statement 2

1.3 Objectives and Scope of Study 2

CHAPTER 2 LITERATURE REVIEW AND THEORY 3

2.1 Literature Review 3

2.1.1 CCTV System 3

2.2 Theory 5

2.2.1 Triangulation Method 5

CHAPTER 3 METHODOLOGY 11

3.1 Procedure Identification 11

3.2 Prototype-Based Methodology 12

3.3 Hardware and Software Required 13

3.4 Process Flow 14

3.5 Design using Simulink 16

3.5.1 Setting up Input in Audio Processing 17

3.5.2 Filtration of an Acoustic Signal 19

3.5.3 Measuring the Sound intensity 22

CHAPTER 4 RESULTS AND DISCUSSION 25

4.1 Calculation in Simulink 25

4.2 Results Comparison 26

CHAPTER 5 CONCLUSION AND RECOMMENDATION 35

5.1 Conclusion 35

5.2 Recommendation 36

APPENDICES 40

Appendix A 41

Appendix B 42

Appendix C 45

Appendix D 48

Appendix E 51

Appendix F 52

LIST OF FIGURES

FIGURE 2.1 Triangulation Method... 5

FIGURE 2.2 Effective Range of Microphone................................. 10

FIGURE 3.1 Flow Chart of the Project...................................... 11

FIGURE 3.2 Prototyped Based Methodology................................. 12

FIGURE 3.3 Process Flow.. 14

FIGURE 3.4 System Layout... 15

FIGURE 3.5 Embedded MATLAB Function Block........................ 17

FIGURE 3.6 Input Setting in Simulink...................................... 18

FIGURE 3.7 Signal Before Alarm... 18

FIGURE 3.8 Signal After Alarm... 18

FIGURE 3.9 Frequency Analysis 20

FIGURE 3.10 Fast Fourier Transform....................................... 20

FIGURE 3.11 FIR Filter... 21

FIGURE 3.12 Signal Before and After Bandpass Filter..................... 21

FIGURE 3.13 Design in Simulink.. 23

FIGURE 4.1 Finalized Design for Calculation............................... 28

FIGURE 4.2 Results in MATLAB 28

FIGURE 4.3 Alarm located in polar coordinate............................... 29

FIGURE 4.4 Results at 30°... 30

FIGURE 4.5 Results at 60°... 30

FIGURE 4.6 Results at 90°... 30

FIGURE 4.7 Results at 120°.. 31

FIGURE 4.8 Results at 150°.. 31

FIGURE 4.9 Results at 180° 31

FIGURE 4.10 Results at 0.5 meters distance.............................. 32

FIGURE 4.11 Results at 1.0 meters distance.................................... 32

FIGURE 4.12 Results at 1.5 meters distance.................................. 33

FIGURE 4.13 Results at 2.0 meters distance.................................. 33

FIGURE 4.14 Results at 2.5 meters distance.................................. 33

FIGURE 4.15 Results at 3.0 meters distance.................................. 34

FIGURE 4.16 Error Chart.. 34

LIST OF ABBREVIATIONS

FFT Fast Fourier Transform

FIR Finite Impulse Response

PIC Peripheral Interface Controller

USB Universal Serial Bus

CHAPTER 1
INTRODUCTION

1.1 Background of Study

In ancient time, people used triangulation method to measure the far distance between two points. For instance, the distance between the Earth and the Moon is measured using parallax. The Moon forms a triangle with two points located on Earth. The angle between the line connecting the Moon and the fix point on Earth and the horizontal line connecting the fix points is measured. Since the distance of the horizontal line can be measured easily, the distance to the Moon can be calculated using trigonometry. Nowadays, the GPS system using mobile phone is based on the triangulation technique where the distance between the phone and receivers which are mounted on the tower can be calculated provided the speed of radio signal is known. The location of the phone is determined by the interception of the circles centred at the tower.

In this project, we design a system to pinpoint the location of the source of an acoustic object coming from an open space in front of the system device. This system, for example, can be used to prevent the vehicle break-in and theft of motor vehicle in the car parking area.

1.2 Problem Statement

The crime rate related to vehicle break in and theft of motor vehicle are increasing [3], at an alarming rate due to numerous limitation of the security system such as insufficient man power for security and lack of CCTV coverage. The CCTV camera also pushes crime into blind spot where camera does not get down those "hidden" corner. In most of the cases, the security action only be taken as late as after the incident was reported. So, this project is carried out to improve the security system in car parking area so that the security can take a prompt action to locate and arrive at a crime scene by using the acoustic locating system. Therefore they can take immediate action on any suspicious activities that triggered the car alarm in car parking area.

1.3 Objectives and Scope of Study

The objectives of this project are as follows:

- Using triangulation method to locate the acoustic object
- Only a specific acoustic object is identified and captured by the system
- Design and prepare the prototype of the system

This project will be focused on the open car parking area. The sound intensity is measured and the location of the alarm will be calculated using MATLAB. The calculation is performed by using the formula that will be carried out in the programming and the location of the sound alarm will be display on the screen. This project covers some knowledge in Digital Signal Processing studies for sounds filtration which is basically used to remove unwanted signal and noise.

CHAPTER 2

LITERATURE REVIEW AND THEORY

This chapter describes some related theory and the main reading materials that inspires in developing this project; the materials were referred technical papers and websites relating to the surveys on current technology; CCTV and also sound intensity concept.

2.1 Literature Review

2.1.1 CCTV System

The current technology used in reducing the crime rate at car parking area is using closed circuit television also known as CCTV. At first sight, one might think that the introduction of CCTV produced the decrease in thefts. In *Crime Reduction Effects And Attitudes Towards Its Use* article by Coretta Philips has evaluated the effectiveness of CCTV in reducing crime [4]. Based on the research by Tiliey, it stated that CCTV has reducing the crime rate. It also claimed that CCTV provides public reassurance and reduces fear of crime as people discouraged from visiting CCTV-covered areas.

In Hartlepool parking area, the CCTV has been installed and monitored by security officers in April 1990 and after two years evaluated, they find out the number of crime rate has declined[5]. Unfortunately in Coventry, the car theft cases has been fluctuated after 5 years evaluation (1987 to 1992) and this installation involved in high cost. To reduce the cost impact, Tilley mentioned the successful parking faclility in Lewicham where the CCTV installation only consisted of three fixed-lens and one dummy. Although it is cost effective, but the it still can not reduce and eliminate the crimes.

The existing security system for car parking area is using the CCTV to monitor the area. The problem is, they have to installed so many CCTV at the overy corner and it might cost higher and sometimes the observation is being made by the security officer after the incidence. The disadvantage of the system is that they might not capture the location because of the limitation of recorded area; blind-spot. In the article *CCTV System* by Lazaro J Hester mentioned about the some irresponsible and professional theft that tend to change the angle of the camera so that the criminal target area will not be displayed to the security room [6]. Moreover, CCTV need high capasity of memory to record 24 hours activities.

This Acoustic Object Locating System has an advantage over the CCTV System in term of accuracy and effectiveness in reducing the crime rate. The CCTV is only able to capture the certain range of location but Acoustic Object locating system, the system covers all over area as long as it is still in effective range of microphone in. It also can give the exact location of the crimes immediately after the sound alarm is triggered and cheaper.

2.2 Theory

Locating system for an acoustic object can be achieved by using triangulation method. This project is focus on the sound intensity coming from the acoustic object where the location can be calculated by using the intensity approach.

2.2.1 Triangulation Method

The central technology of the design of the acoustic location system is using the triangulation method. The formula is derived in order to calculate the location of the acoustic object using the polar coordinate system represented by the radius distance connecting the source to origin and the azimuthal angle between the line connecting the source to the origin and the reference line connecting along the microphones.

In this study, there will be 3 microphones placed along a straight line with equal distance apart, and sound intensities are measured at these 3 points and the discrepancies in the measured values between the microphones provide the information to calculate the location of the source of sound.

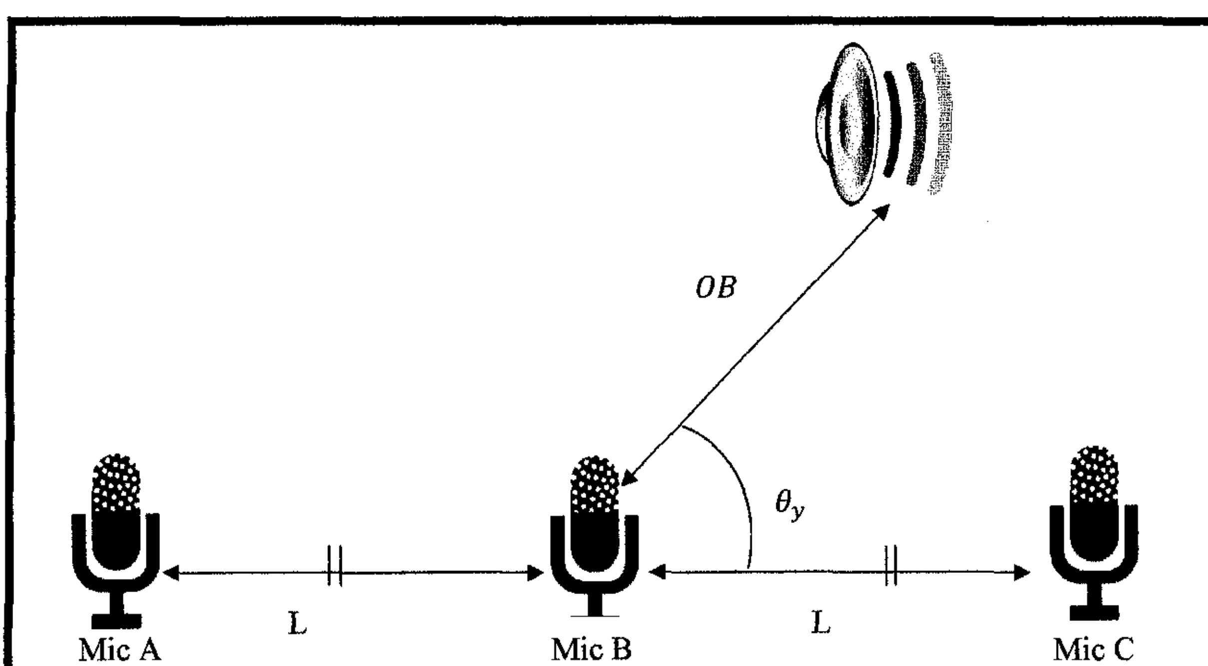


Figure 2.1: Triangulation Method

The figure above shows the geometry of the triangulation method. The speed of sound is 343 m/s at 20°C or 1125.33 ft/s [7]. The concept behind this project is using the discrepancy of sound intensity captured at different position where the sound intensity is defined by the power per unit area [2] .The sound intensity level formula is given by:

$$I_A = \frac{P}{4\pi R^2} \tag{1}$$

where P is the source power and R is the distance between the acoustic point source and the observation point. From this equation the sound intensity varies with the distance according to $\frac{1}{R^2}$ .The intensity of the sound received at different points will be different as the distance between the microphones (listener's location) and the source point vary. By the definition of the unit in decibel, the intensity received by each microphone is given by [9]]:

$$S_A = 10\log\frac{I_A}{I_o} \qquad S_B = 10\log\frac{I_B}{I_o} \qquad S_C = 10\log\frac{I_C}{I_o} \tag{2}$$

The locating system is designed to locate the exact location of the source of sound. A formula based on the concept of triangulation is derived for these purposes. The input data such as the relative sound intensity; ΔS_{AB} and ΔS_{BC} also the distance between the microphones; L are gathered as the variables to the formula.

The location of the acoustic object is calculated using the formula and the radius of the acoustic object to the origin and the azimuthal angle are then found. After some algebra, we obtain the formula for OB, the distance between the source and microphone B and θ_y , the azimuthal angle which are given below. The full derivation of the formula as per below:

The difference between Intensity of Microphone A and B

$$\Delta S_{AB} = S_A - S_B$$

$$= 10\ log_{10}(\frac{I_A}{I_o}) - 10\ log_{10}(\frac{I_B}{I_o})$$

$$= 10\ log_{10}(\frac{I_B}{I_A})$$

$$= 10\ log_{10}(\frac{P}{4\pi OA^2}) - 10\ log_{10}(\frac{P}{4\pi OB^2})$$

$$= 10\ log_{10}(\frac{OB}{OA})^2 \tag{3}$$

The difference between Intensity of Microphone B and C

$$\Delta S_{BC} = S_B - S_C$$

$$= 10\ log_{10}(\frac{I_B}{I_o}) - 10\ log_{10}(\frac{I_C}{I_o})$$

$$= 10\ log_{10}(\frac{I_B}{I_C})$$

$$= 10\ log_{10}(\frac{P}{4\pi OB^2}) - 10\ log_{10}(\frac{P}{4\pi OC^2})$$

$$= 10\ log_{10}(\frac{OC}{OB})^2 \tag{4}$$

By using Trigonometry Properties,

$$OA^2 = OB^2 + L^2 - 2LOB\ cos\theta_y \tag{5}$$

$$OC^2 = OB^2 + L^2 - 2LOB\ cos\theta_r$$

$$= OB^2 + L^2 + 2LOB\ cos\theta_y \tag{6}$$

Rearrange the value of ΔS_{AB} in equation (3)

$$\Delta S_{AB} = 10\ log_{10}(\frac{OB}{OA})^2$$

$$\frac{OB^2}{OA^2} = 10^{\ 0.1\Delta S_{AB}}$$

$$OA^2 = \frac{OB^2}{10^{0.1\Delta S_{AB}}} \tag{7}$$

Then, rearrange the value of ΔS_{BC}

$\Delta S_{BC} = 10\ log_{10}(\frac{OC}{OB})^2$

$\frac{OC^2}{OB^2} = 10^{\ 0.1\Delta S_{BC}}$

$OC^2 = 10^{\ 0.1\Delta S_{BC}}\ (OB^2\ )$ (8)

Substitution equation (7) into the equation (5)

$OA^2 = OB^2 + L^2 - 2LOB\ cos\theta_y$

$\frac{OB^2}{10^{0.1\Delta S_{AB}}} = OB^2 + L^2 - 2LOB\ cos\theta_y$

$(10^{-0.1\Delta S_{AB}})OB^2 = OB^2 + L^2 - 2LOB\ cos\theta_y$

$OB^2(10^{-0.1\Delta S_{AB}} - 1) - L^2 - 2LOB\ cos\theta_y = 0$ (9)

Substitution of equation (8) into equation (6)

$OC^2 = OB^2 + L^2 + 2LOB\ cos\theta_y$

$(10^{0.1\Delta S_{BC}})OB^2 = OB^2 + L^2 + 2LOB\ cos\theta_y$

$OB^2\ (10^{0.1\Delta S_{BC}}) - 1 - L^2 - 2LOB\ cos\theta_y$ (10)

To find OB, the equation (9) is added to equation (10), then we will get

$OB^2\ (10^{-0.1\Delta S_{AB}} + 10^{0.1\Delta S_{BC}} - 2) - 2L^2 = 0$

$OB^2 = \frac{2L^2}{(10^{-0.1\Delta S_{AB}}+10^{0.1\Delta S_{BC}}-2)}$

$OB = \frac{\sqrt{2}\,L}{\sqrt{(10^{-0.1\Delta S_{AB}}+10^{0.1\Delta S_{BC}}-2)}}$ (11)

To find , the equation (9) is compare to equation (10)

$OB^2\ (10^{-0.1\Delta S_{AB}} - 10^{0.1\Delta S_{BC}}) + 4\ LOB\cos\theta_y = 0$

$$\cos\theta_y = \frac{OB^2(10^{0.1\Delta S_{BC}} - 10^{-0.1\Delta S_{AB}})}{4LOB}$$

$$\cos\theta_y = \frac{\sqrt{2}L}{\sqrt{(10^{-0.1\Delta S_{AB}} + 10^{0.1\Delta S_{BC}} - 2)}} \cdot \frac{(10^{0.1\Delta S_{BC}} - 10^{-0.1\Delta S_{AB}})}{4L}$$

$$\theta_y = \cos^{-1}\left(\frac{\sqrt{2}(10^{0.1\Delta S_{BC}} - 10^{-0.1\Delta S_{AB}}}{4\sqrt{(10^{-0.1\Delta S_{AB}} + 10^{0.1\Delta S_{BC}} - 2)}}\right) \tag{12}$$

where S_A: Intensity measure at Microphone A

S_B : Intensity measure at Microphone B

S_c : Intensity measure at Microphone C

L : The distance between the microphones

OB: The distance between the source and Microphone B

θy :The azimuthal angle between the source and Microphone B

Referring to the article, The Case Against Car Alarm, A.Friedman (2003) the car alarm sound can reach about 125 decibels (dB) [10]. Theoretically, the shortest distance between the microphone and the source will give the highest sound intensity. In this project, the experiment will be conducted to achieve the desired objectives.

2.2.2 Effective Range of Microphone

The effective range of microphone is defined by the maximum distance at which the targeted sound is captured. Figure 3 shows the effective range of the microphones and the derivation is carried out to get the range of ℓ_E.

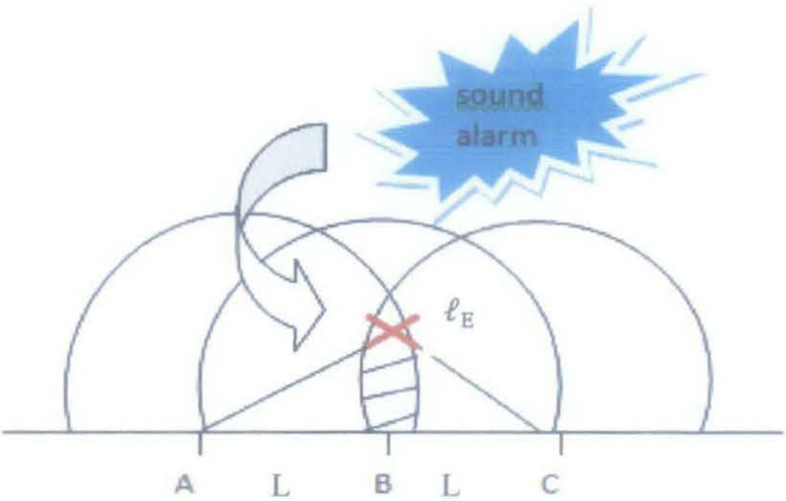


Figure 2.2 : Effective range of Microphone

The effective range, ℓ_E of the microphone must obey,

$$\ell_E > L \tag{5}$$

If the distance between the microphones is too short, it will reduce the performance of the system. Therefore, to optimize the performance of the system with high accuracy output, the distance between the microphones and their effective radius has to be taken into account.

CHAPTER 3

METHODOLOGY

This chapter describes the work related to this project in details. There are three major parts in this project; project layout, hardware and software part.

3.1 Procedure Identification

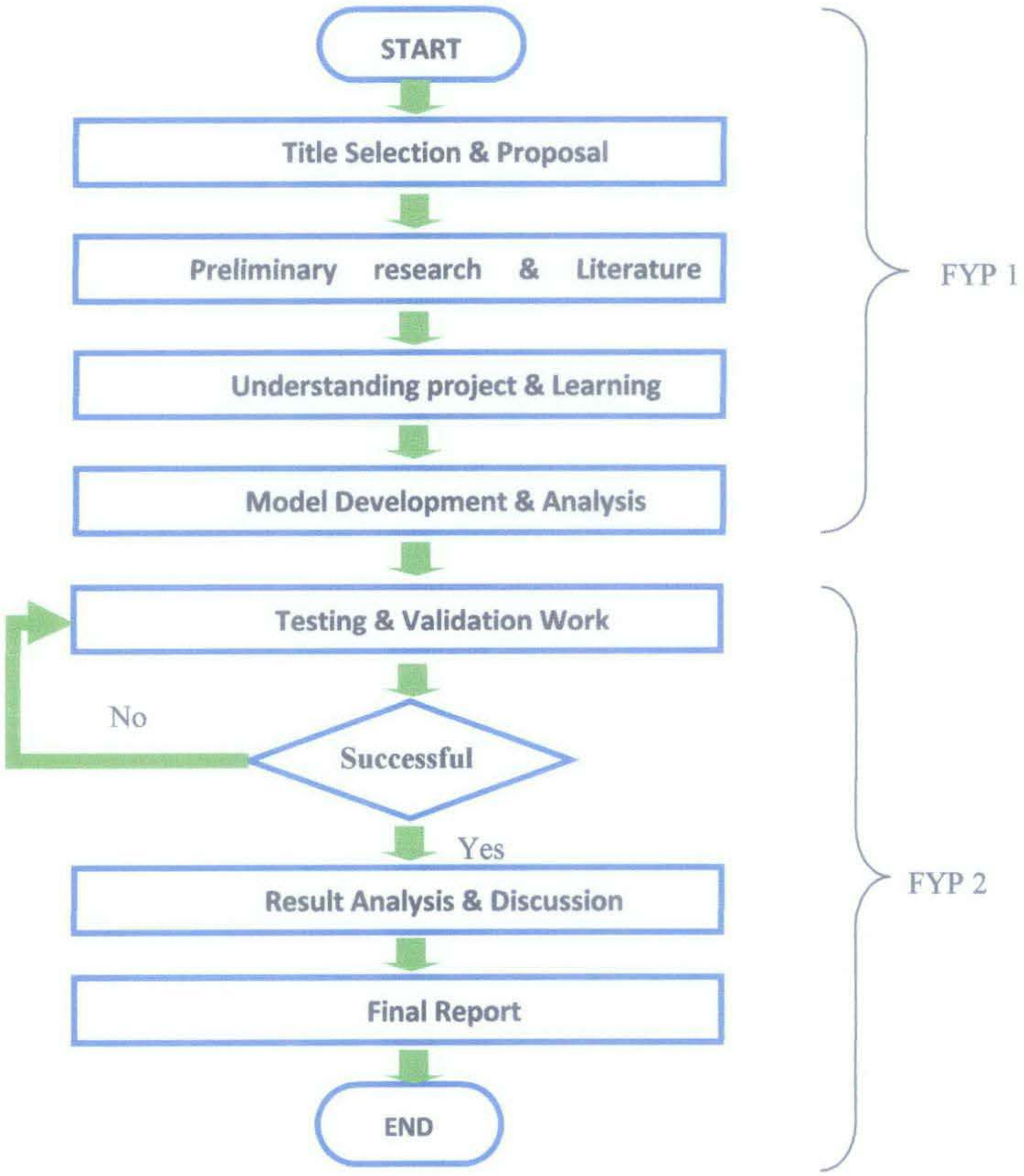


Figure 3.1 : Flow Chart of the Project

Figure 3.1 shows the activities Flow Chart while of the project planning for FYP 1 and the continuation in FYP 2.

Procedure Identification is very important to keep the project always on the track which will help the project finish on time. Those procedures for the implementation had to pass several systematic steps. Each step had to finish before move on to another steps to make sure the project success.

3.2 Prototype-Based Methodology

The prototyping-based methodology performs the analysis, design and implementation phases concurrently and all the three phases are performed repeatedly in a cycle until the system is completed.

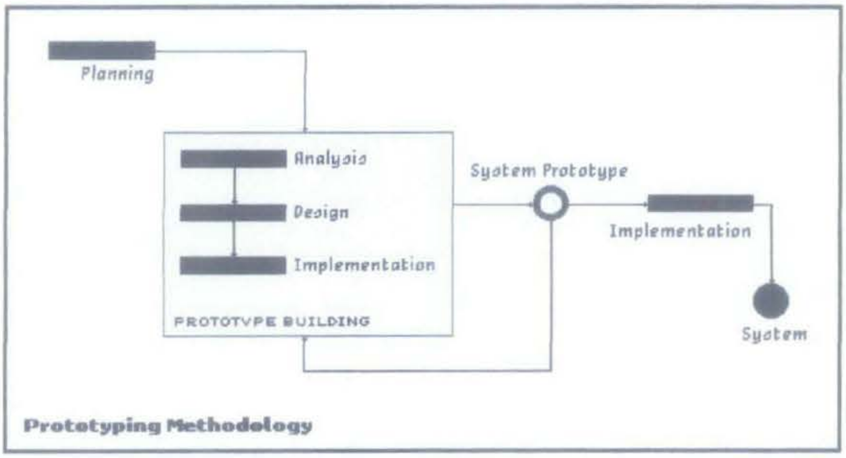


Figure 3.2: Prototyped Based Methodology

During preparation of the project prototype, there are several steps to be implemented in order to get the desired results. For instance, if there are problems occurs, faulty during the execution, it will be debugging and re-do until it is successful.

3.3 Hardware and Software Required

Hardware Requirement

- 3 microphones
- 3 USB Sound Cards
- Personal Computer
- USB hub

Software Requirement

- MATLAB Program
- Simulink Program

3.4 Process Flow

Figure 3.3 shows the process flow the project. MATLAB uses an interactive system for doing numerical computation and visualization in this project.

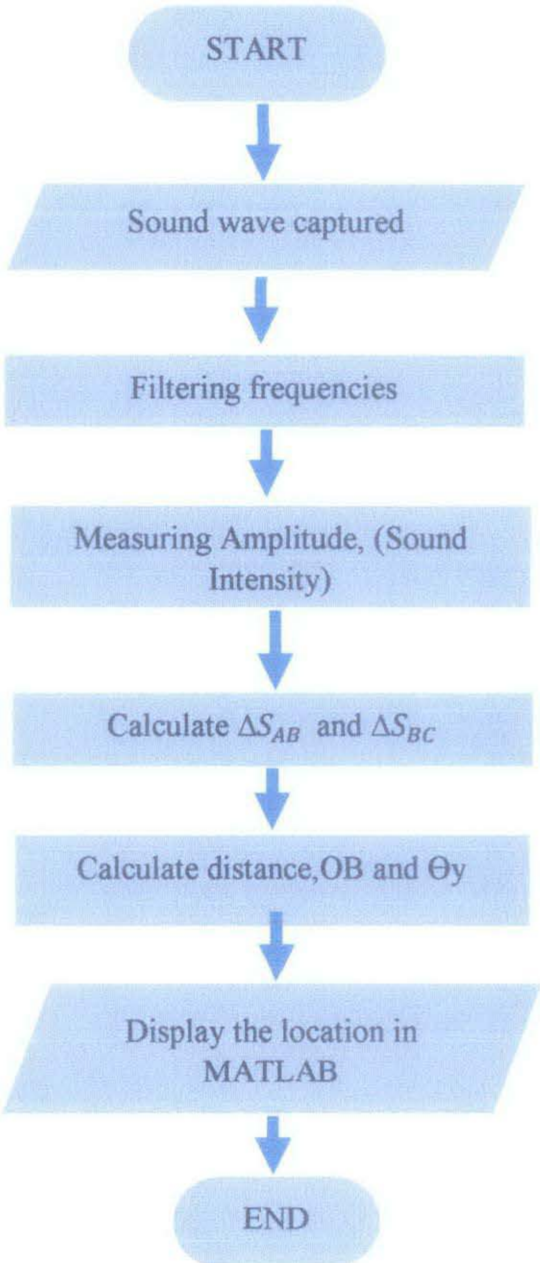


Figure 3.3 : Process Flow

The system consists of:

- Input interface which consists of the 3 microphones and sound card [Appendix B]
- Sound filtration
- Sound intensity measurement
- System interfaces consists of Simulink and MATLAB. The location of the sound alarm will be display on the plan position indicator.

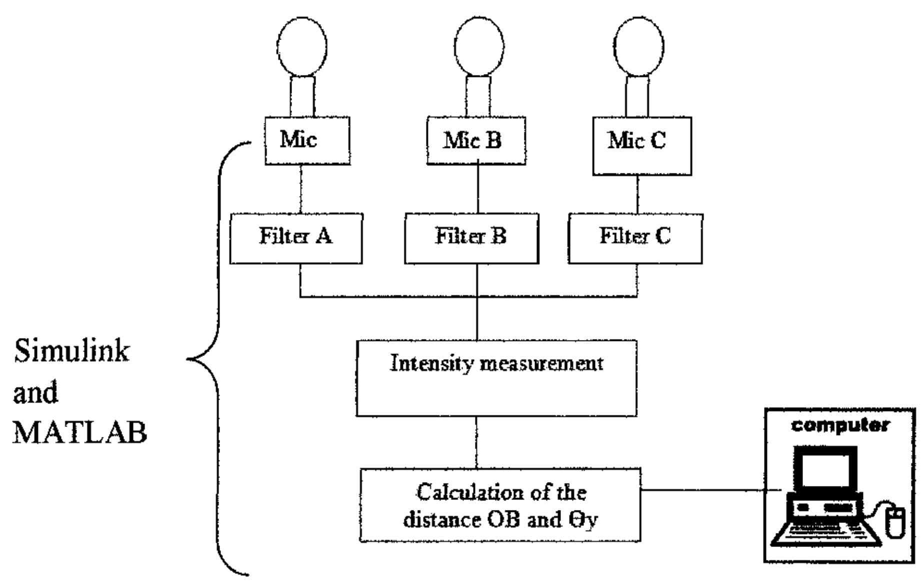


Figure 3.4: System Layout

Basically, this system calculates and displays the location of the sound alarm. When the signals enter the system, the microphones will received the signals and send them to the processing system for filtering process. This filter is used to eliminate the unwanted signal from triggering the system [11].

When the car alarm is triggered, system will automatically calculate the distance between the source and the alarm including the azimuthal angle then the location will be display in MATLAB.

3.5 Design using Simulink

Simulink is a graphical extension to MATLAB for modeling and simulation of systems [12]. The system are drawn by using the block diagrams.There are a few elements of block diagram including virtual input and output devices is used. Simulink is integrated with MATLAB and data can be easily transferred between the programs. It is used to model the systems, build controllers, and simulate the systems. This program can be started from the MATLAB command prompt by entering command : >> simulink

The Embedded MATLAB Function block allows MATLAB functions to be added into Simulink models for deployment to embedded processors.This function is useful for coding algorithms that are better stated in the textual language of the MATLAB software than in the graphical language of the Simulink. Basically,this block works with a subset of the MATLAB language called the Embedded MATLAB subset, which provides optimizations for generating efficient, production-quality C code for embedded applications [12].

Here is an example of a Simulink model that contains an Embedded MATLAB Function block:

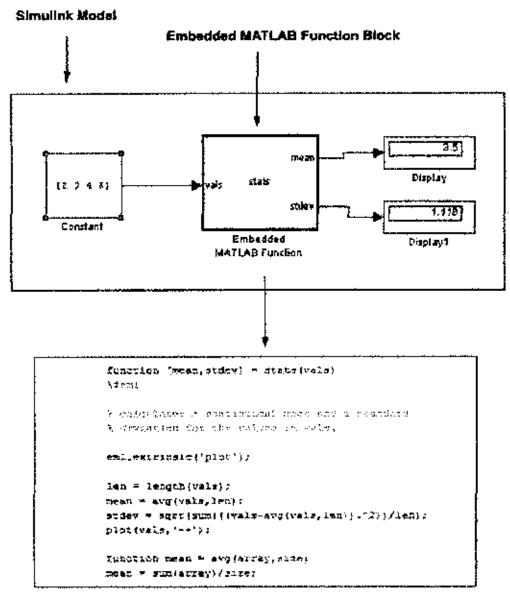


Figure 3.5 : Embedded MATLAB Function Block [13]

In Simulink, the processing part is conducted. After done all the design, the system will be testing and the results appear in MATLAB windows.

3.5.1 Setting up Input in Audio Processing

To capture the entering signal of the sound in the system, the omnidirectional microphones are chosen **[Appendix A].** Basically, the computer has only one sound card and it is not possible to connect with 3 microphones using the same sound card [14]. There are two methods to be considered in choosing the connection between the microphones with the computer; connect them together by using the multiplexer or extra sound card. The microphone we used has wide frequency response (20 Hz – 20 kHz), and lower cost [4].

Throughout this project, the USB Sound Card is chosen to connect the microphones with the processing system. Since we use 3 microphones, 3 extra USB sound cards are needed in this project **[Appendix B]** One USB sound cards will be connected to each microphone.

The experiment is carried out to investigate the behaviour of the signal and the filtering system. The figure shown below is the initial design of the system. The circuit design and programming is done in the block diagram itself. The microphones are aligned in a straight line and measured the values; distance between the microphones and the source.

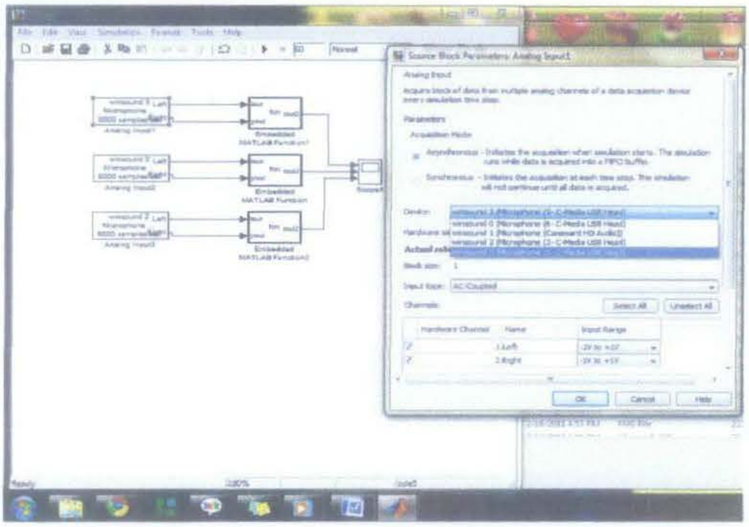
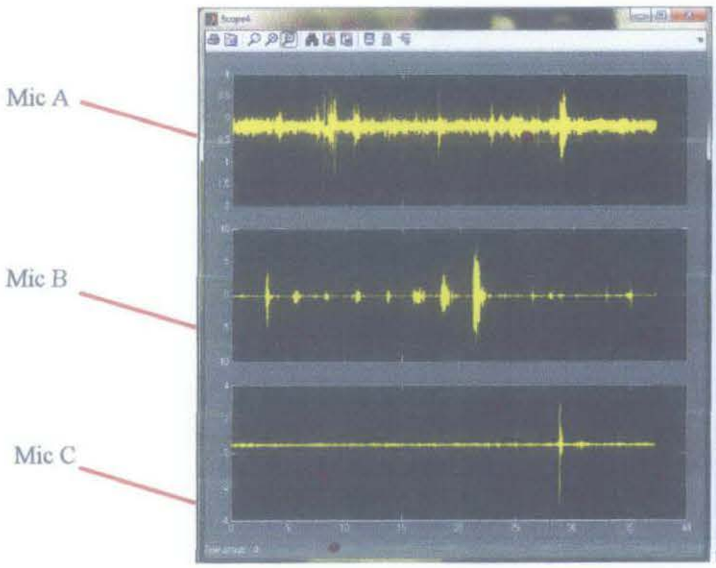


Figure 3.6: Input setting in Simulink

The 3 microphones are connected together into the same computer. The figures below show the signal received from microphone A, Microphone B and from microphone C before and after the alarm are activated in 60 seconds time.



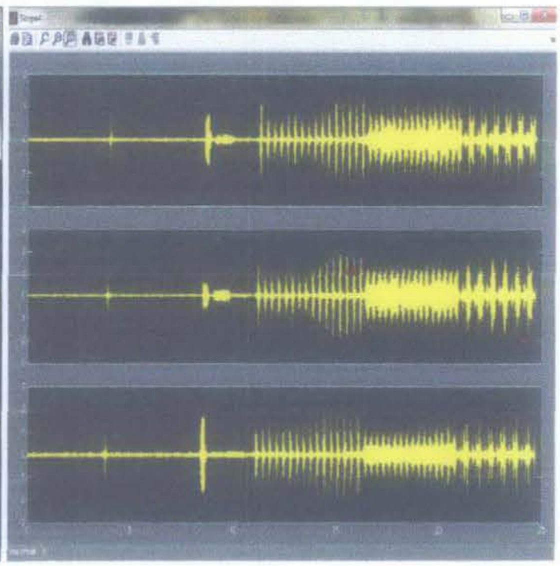


Figure 3.7: Before alarm

Figure 3.8: After alarm

In a normal condition with the existence of alarm, the maximum amplitude receives in the experiment is around -21.30 dB. From these figures we can conclude that the shorter the distance between microphones and the source, the intensity value will be bigger which has been explained in the Chapter 2 before.

The signal is transmitted to the processing system and analysed using Simulink and MATLAB programming.

3.5.2 Filtration of an Acoustic Signal

After the signals get into the system, they will be distinguished by using FFT in MATLAB. The sound coming from 3 different microphones are transformed from time domain to frequency domain. FFT algorithms generally fall into two classes: decimation in time, and decimation in frequency. FFT is used to change the signal from time domain to frequency domain [23]. This to ensure we filtered the desired frequency and we also can easily get the value of sound intensity.

From this FFT, the signal is simplified from time domain to frequency domain. It is used to make the data and the frequency of the unwanted signal can easily be observed, determined and removed. The signal converted from the time domain to frequency domain will make frequency sampling and intensity calculation easier. The maximum value of intensity is shown at 2143 Hz frequency.

From Figure 3.10, the normalized value of the frequency domain is compared to the normalized value in is equals to the maximum decibel, -21.3 dB as shown in Figure 3.9.

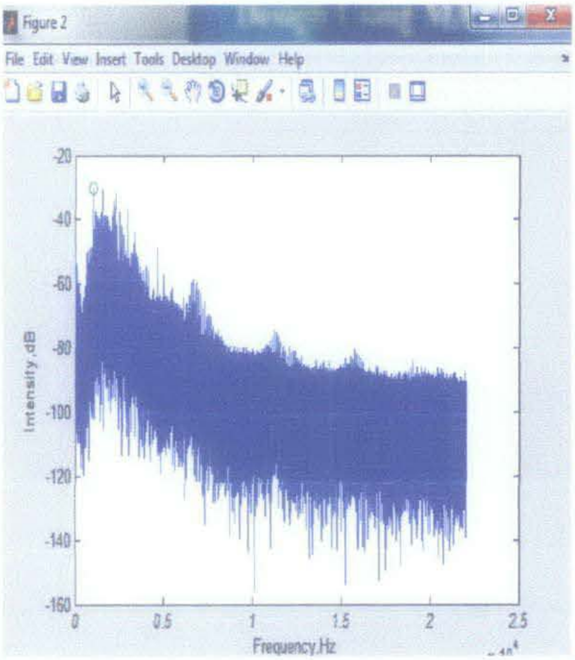


Figure 3.9: Frequency Analysis

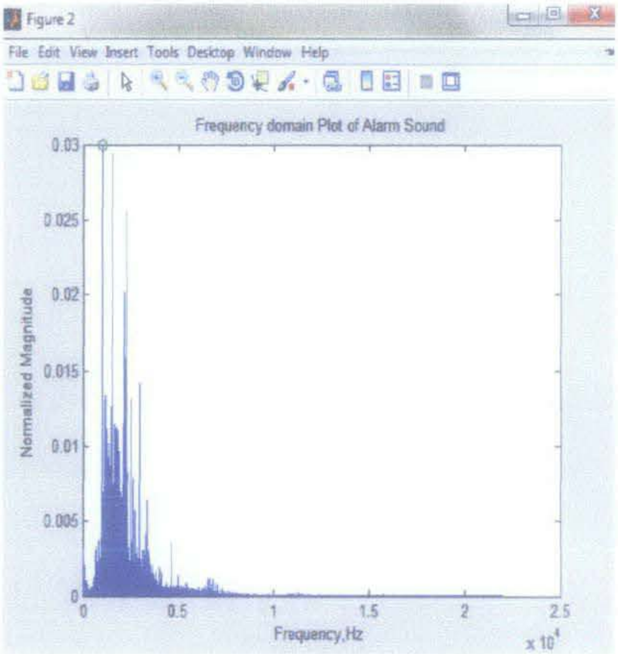


Figure 3.10 : Fast Fourier Transfor

Next, Finite Impulse Response (FIR) Bandpass Filter method is used to remove the unwanted signal such as dog barking, wind and car movement sound. In this project, only certain frequency is allowed to go through to the sound intensity devices before the calculation is executed and the point of sources is determined.

FIR filter is one of the primary types of filters used in Digital Signal Processing and it is finite because it does not have any feedback. Most of the time filter specifications refer to the frequency response of the filter. There are different methods to find the coefficients from frequency specifications [15].

- Window design method
- Frequency sampling method
- Weighted least squares design
- Parks-McClellan method

FIR can be implemented in Simulink. From the time domain signal, this filter will sample them and the values will be used to give an impulse response; time-reverse of the predicted input signal.

Figure below is the MATLAB interface for the FIR filter. The FIR filter can be implemented using FDAtool in MATLAB which is basically used to design and edit the filters. From the FDAtool, we can get the filter designation which is band pass filter. This filter removes the frequency of the unwanted sound.

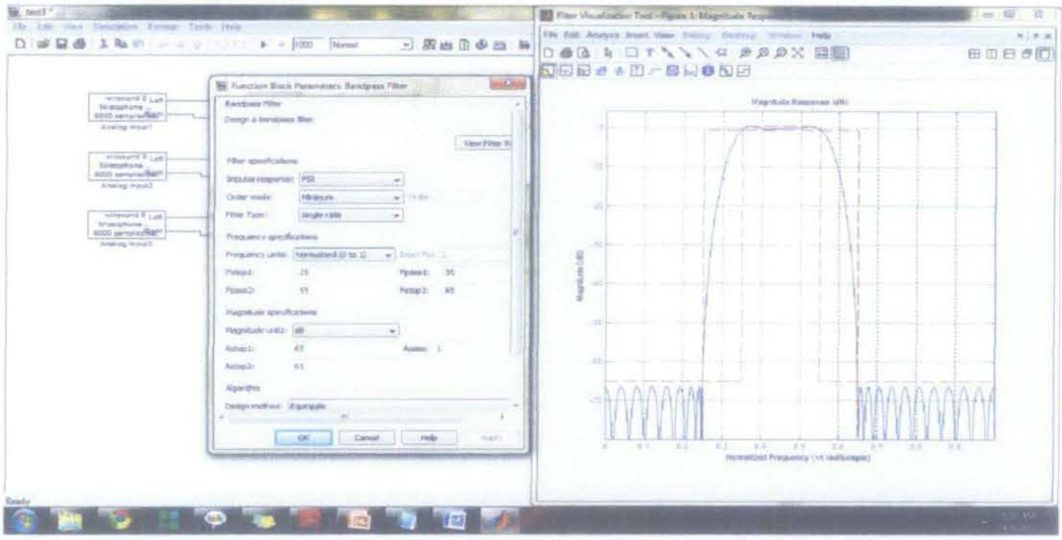


Figure 3.11: FIR Filter

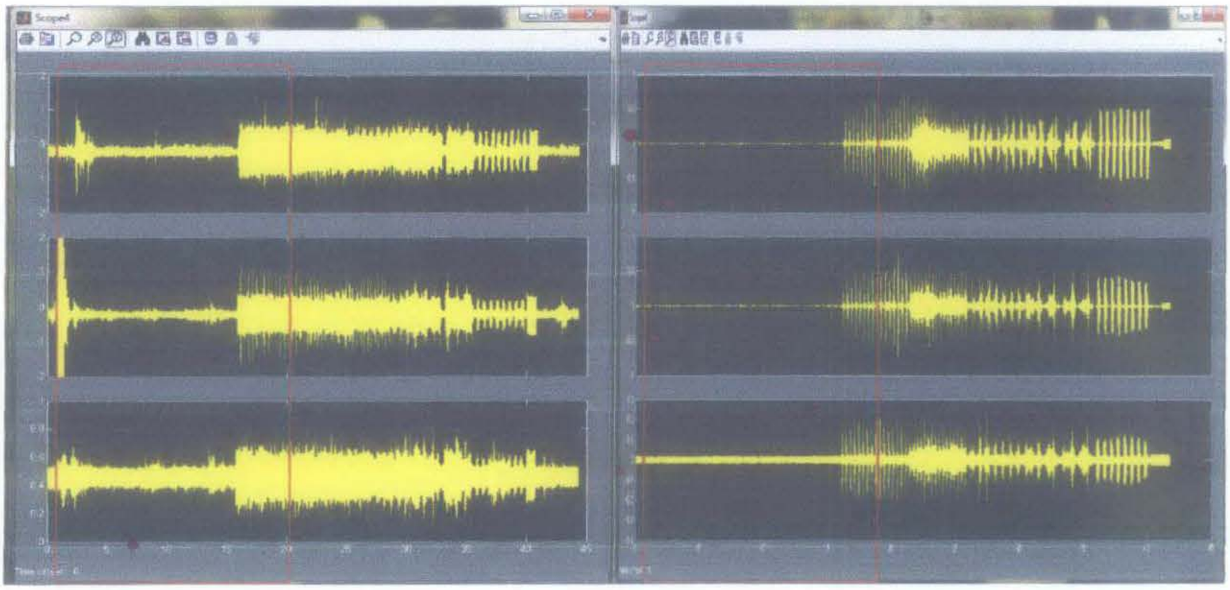


Figure 3.12 : Signal before and after Band pass Filter

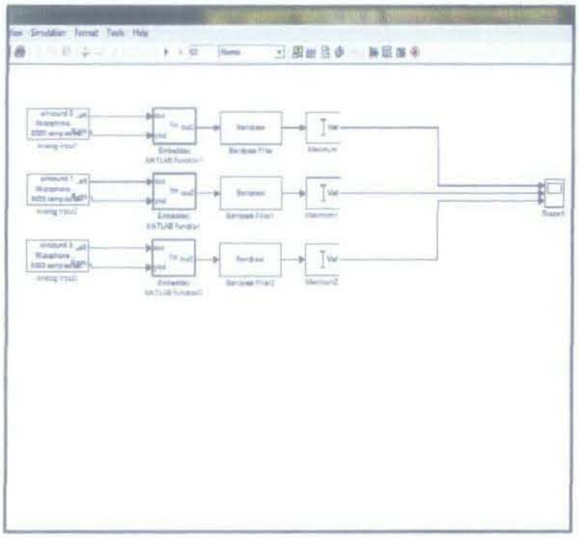
3.5.3 Measuring the Sound intensity

When the car alarm is triggered, the system automatically receives the signal from the sound. Then after the filtration, the source of alarm is determined using the calculation in MATLAB and Simulink. The calculation of the source will be taken place once the difference of intensities is measured.

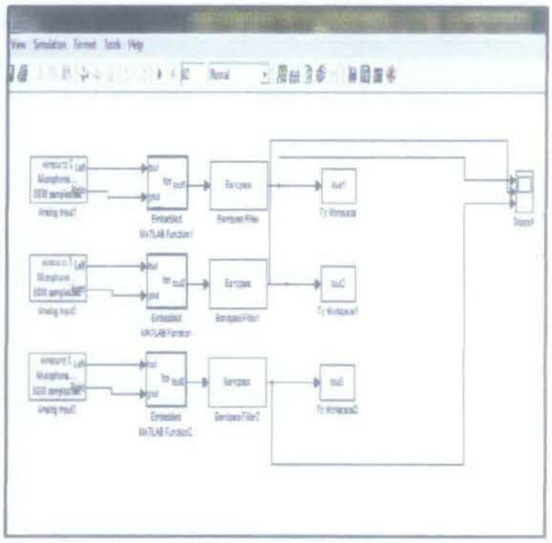
After executing the system using Design A, there are some problems occurred. The 'Maximum Value' block design in Simulink returns the value of the maximum elements of the input signal over time and maximizes the amplitude in approximately 10 seconds sample time. The output can be the maximum of the entire input, of each row, of each column, or over the dimension of the input signal specified in the 'Dimension' parameter. Software limitation in finding the maximum peak of the data has led us to use other method where the value is taken manually from the graph and the new model is applied separately. As an alternative, the design has been revised and changes and Design B is used instead.

The intensity is measured when alarm is detected, then the measuring data are processed in the processing system set by the formulas Eqs. (3) and (4) which require the input of the intensity difference, ΔS_{AB} between the microphone A and B and ΔS_{AB} the intensity difference between microphone B and C.

\



Design A



Design B

Figure 3.13: Design in Simulink

From Simulink, the signal is then sent to MATLAB for amplitude measurement. The value OB and $\emptyset y$are calculated in the system.

```
>> L=0.8;
[ymax,maxindex]=max(tout1.signals.values);
[ymax1,maxindex1]=max(tout2.signals.values);
[ymax2,maxindex2]=max(tout3.signals.values);
peakval = ymax;
peakval1 = ymax1;
peakval2 = ymax2;
dSab = peakval-peakval1;
dSbc = peakval1-peakval2;
OB = abs(sqrt(2)*L/(sqrt(10^(-0.1*dSab))+10^(0.1*dSbc)-2))
theta_y = abs(acos((sqrt(2)*(10^(0.1*dSbc)-10^(-0.1*dSab)))/(4*sqrt(10^(-0.1*dS
theta_y_deg=theta_y*180/pi;
polar (theta_y_deg,OB,'O')

OB =

    0.9746


theta_y =

    1.5774

>> theta_y_deg=theta_y*180/pi

theta_y_deg =

   90.3764
```

Figure 3.14 : Coding in MATLAB

Next, the location of the source is shown on the plan position indicator. This is executed using MATLAB. Lastly, to verify the accuracy of the data, the Percent Error Formula is used. This is to calculate how far we have gone from our experimental value to the theoretical value. The smaller the errors, the project is more accurate and succeeds.

CHAPTER 4

RESULTS AND DISCUSSION

4.1 Calculation in Simulink

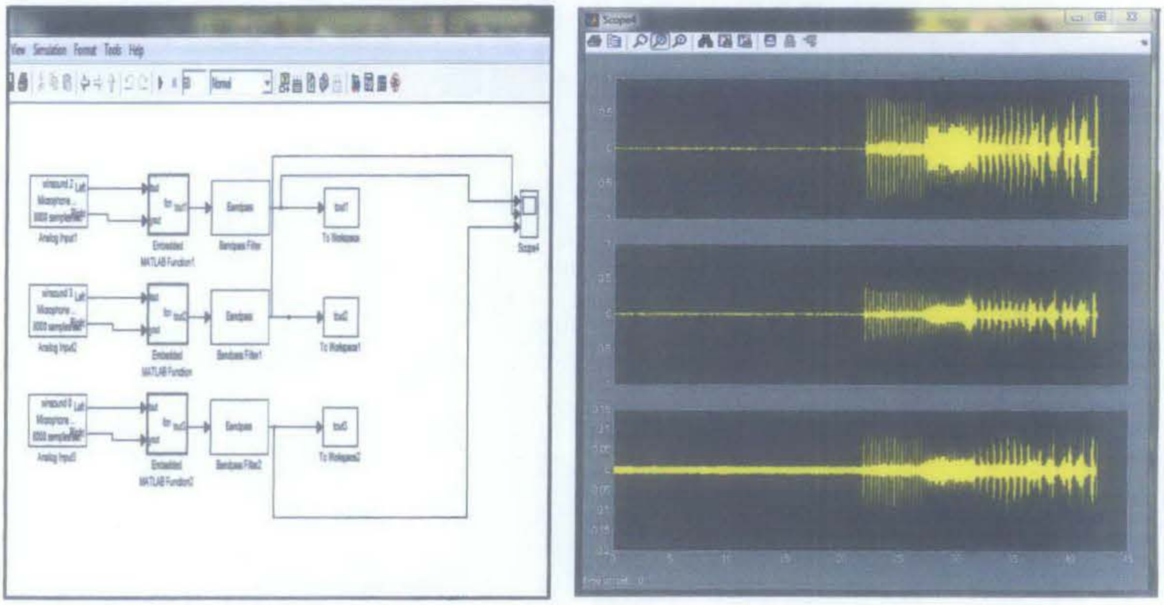


Figure 4.1: Finalized Design for calculation

The signal is sent to MATLAB from Simulink for amplitude measurement and also to calculate OB and angle,$\emptyset y$ by using Design B.

```
OB =

    0.9746


theta_y =

    1.5774

>> theta_y_deg=theta_y*180/pi

theta_y_deg =

   90.3764
```

Figure 4.2: Results in MATLAB

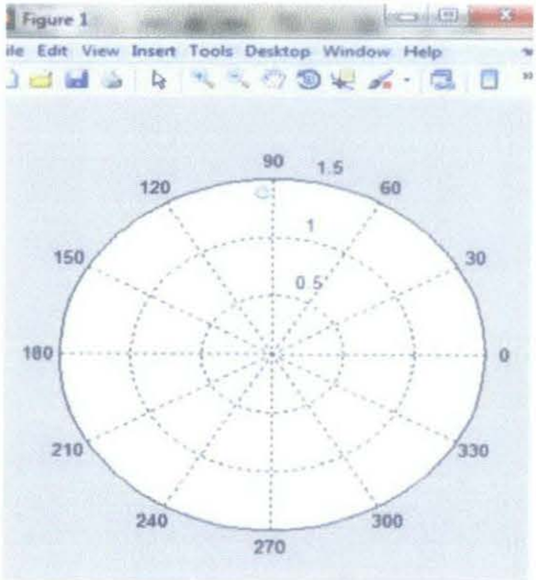


Figure 4.3: Alarm located in polar coordinate

4.2 Results Comparison

A few experiments with different spots have been conducted and below are the results from a few experiments conducted in different angles and distances. The experiments were done in the effective range of microphone which was below 3 metres and at the open area.

The figure below shows the results of the system taken at different angles and distance. Each graph represents the results from different angles and with distance from 0.5 to 3.0 meters.

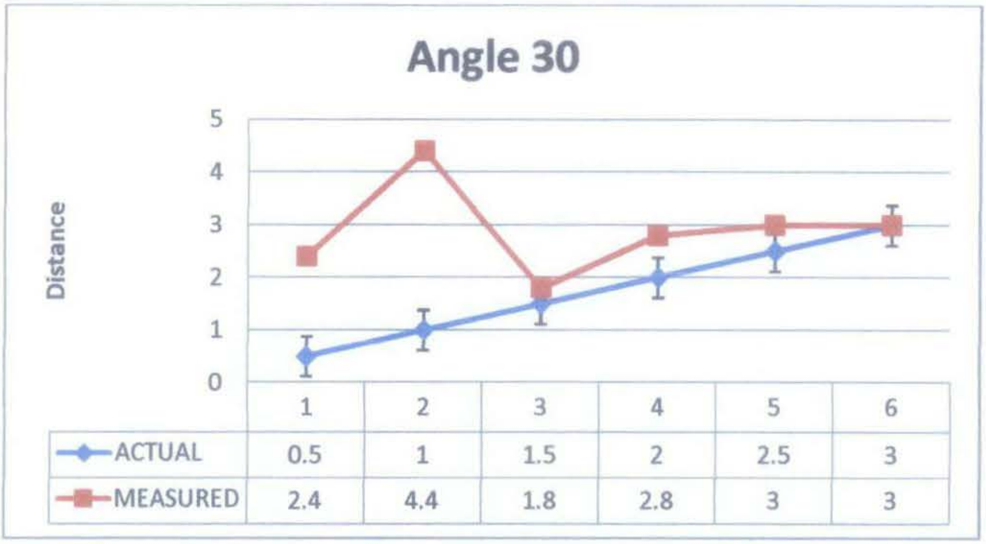


Figure 4.4: Results at 30°

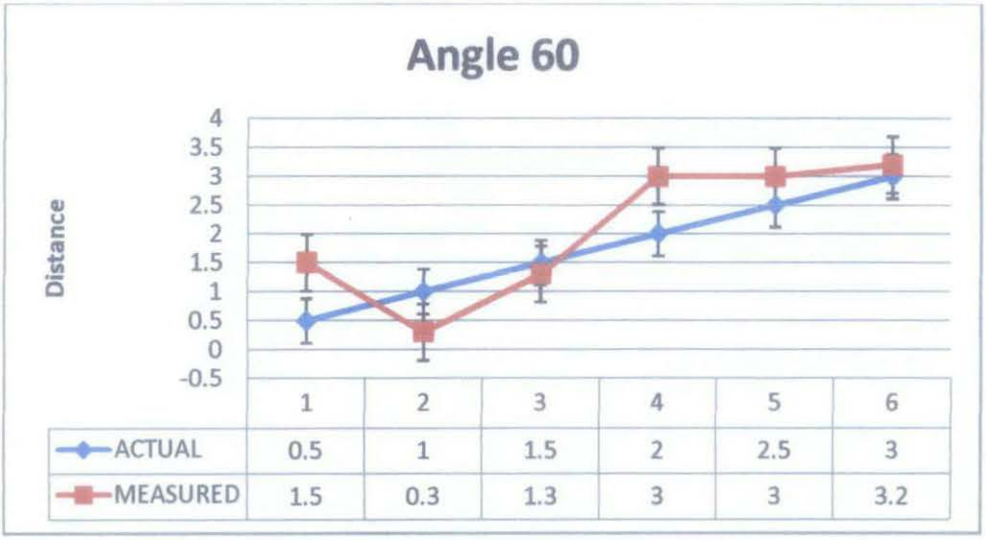


Figure 4.5: Results at 60°

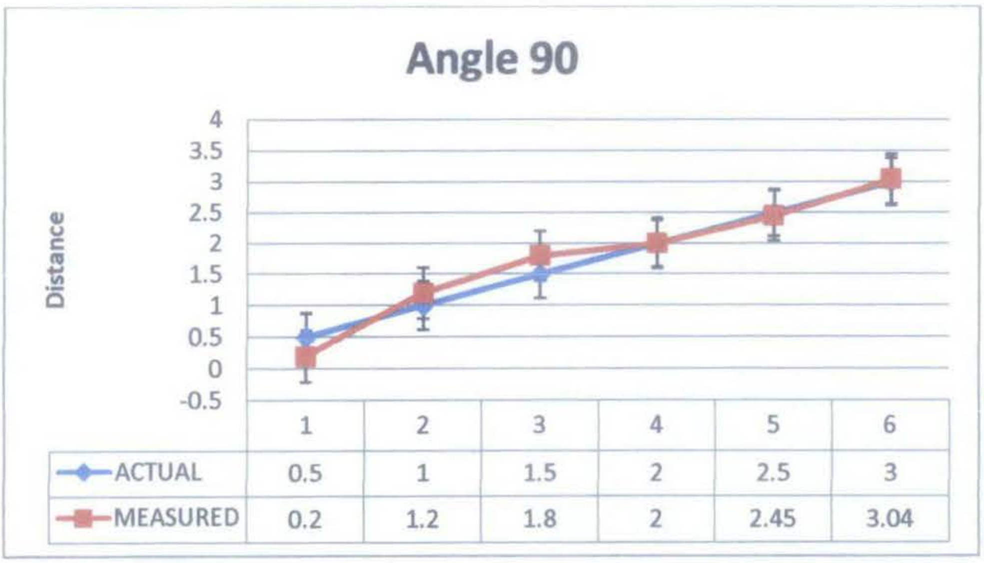


Figure 4.6 :Results at 90°

Figure 4.7: Results at 120°

Figure 4.8: Results at 150°

Figure 4.9: Results at 180°

The figure below shows the results of the system taken at different distances and angles. Each graph represents the results from different distance and each graph shows the angles from 30 to 180 degrees with an interval of 30 degrees.

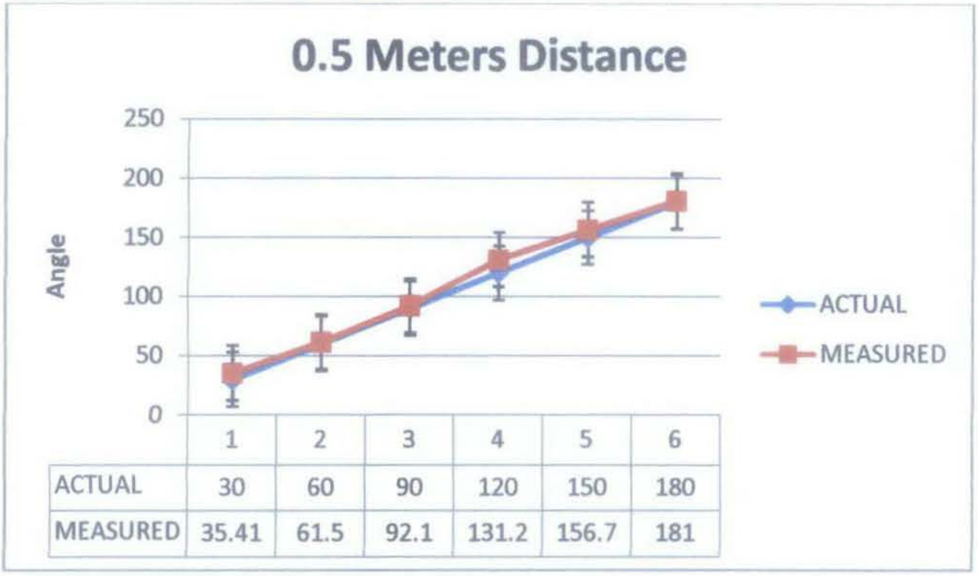


Figure 4.10: Results at 0.5 meters distance

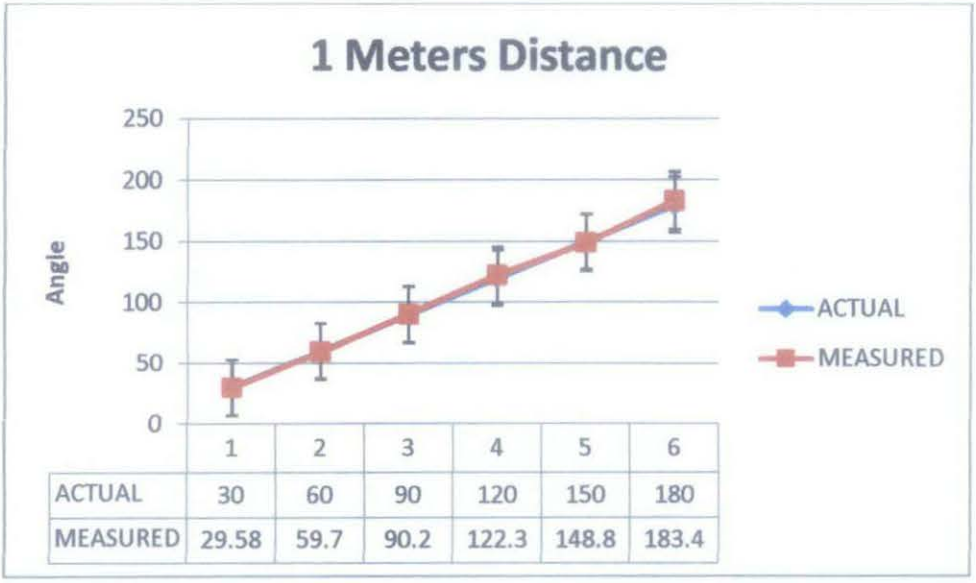
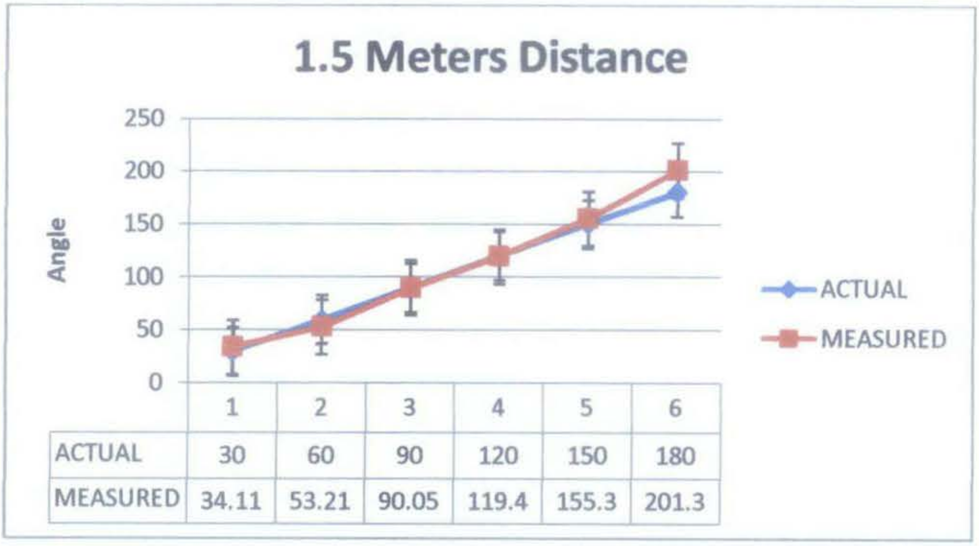
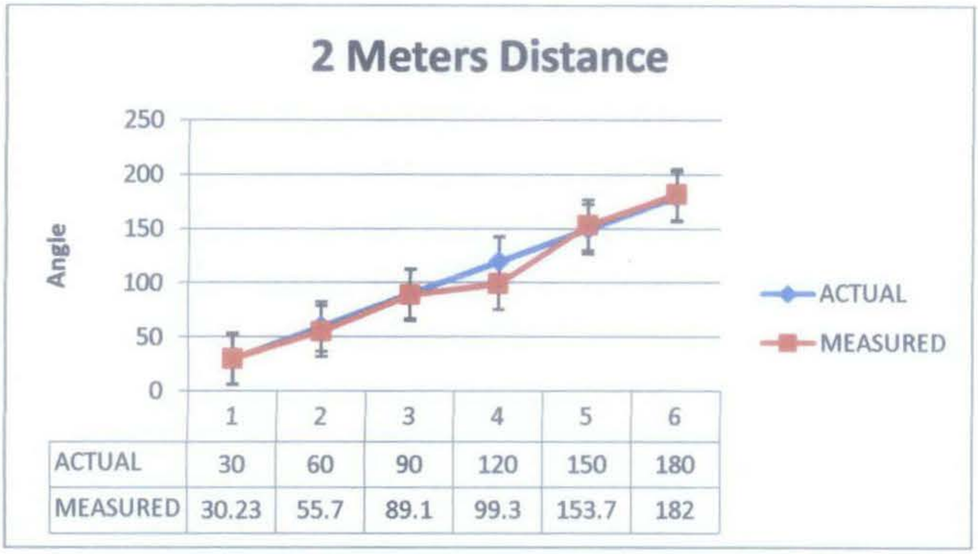


Figure 4.11: Results at 1.0 meter distance



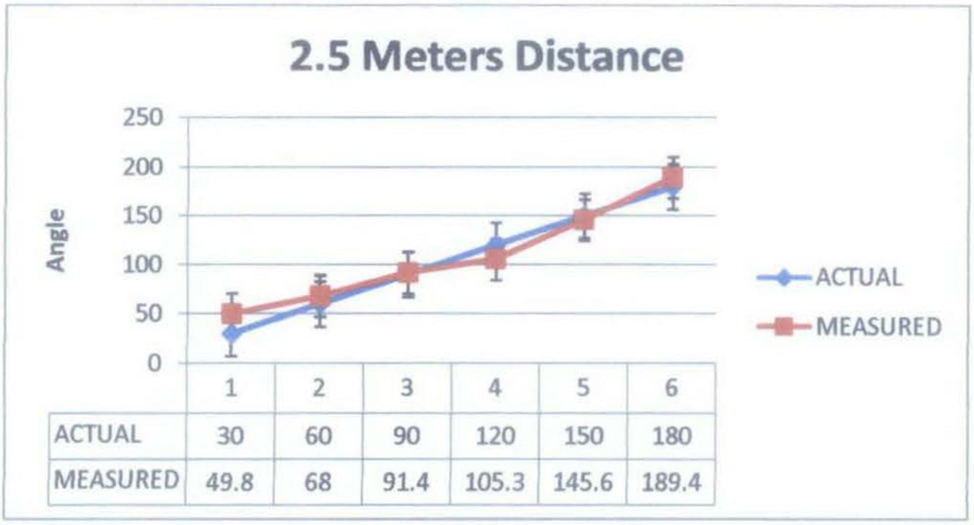
	1	2	3	4	5	6
ACTUAL	30	60	90	120	150	180
MEASURED	34.11	53.21	90.05	119.4	155.3	201.3

Figure 4.12: Results at 1.5 meters distance



	1	2	3	4	5	6
ACTUAL	30	60	90	120	150	180
MEASURED	30.23	55.7	89.1	99.3	153.7	182

Figure 4.13: Results at 2.0 meters distance



	1	2	3	4	5	6
ACTUAL	30	60	90	120	150	180
MEASURED	49.8	68	91.4	105.3	145.6	189.4

Figure 4.14: Results at 2.5 meters distance

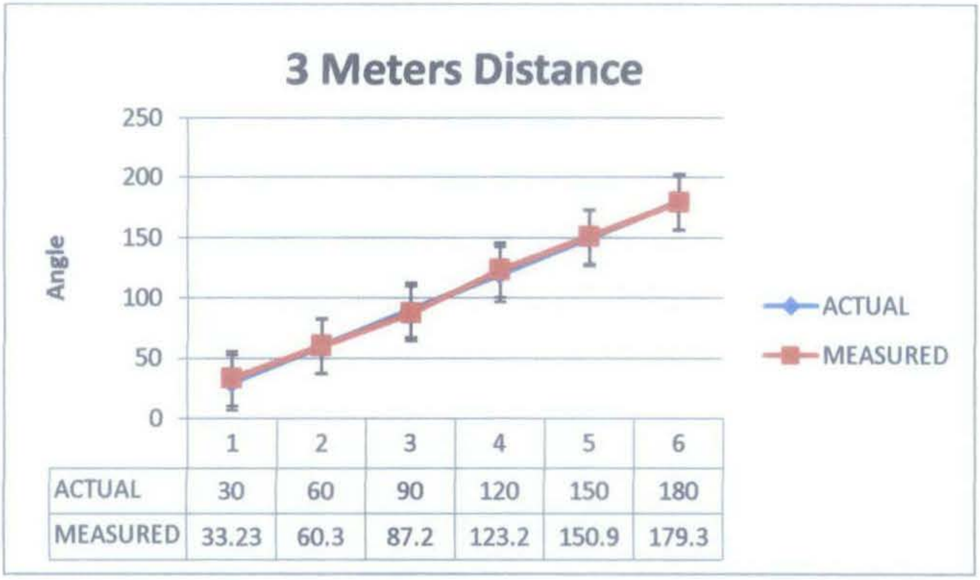


Figure 4.15: Results at 3.0 meters distance

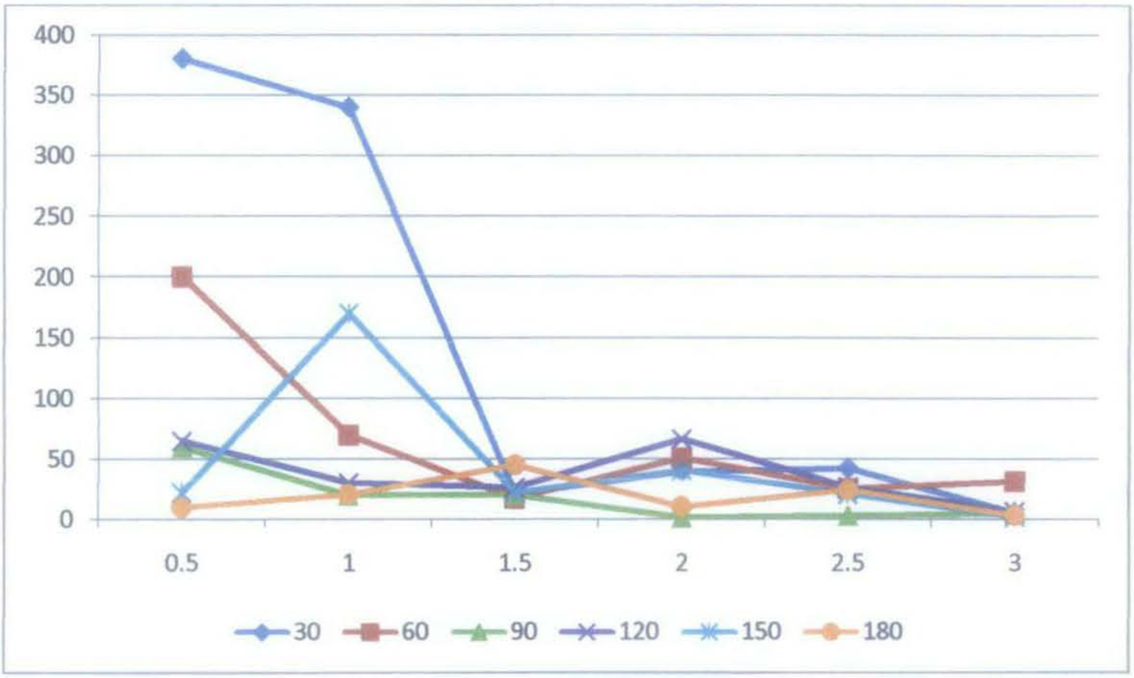


Figure 4.16: Error Chart

In this section we discuss the outcome performance measures of the Acoustic Object Locating System and the problems encountered by the system.

According to figures 4.4 – 4.15 above, we see that the blue line represents the actual value and the brown line represents the measured value. The tables the actual and measured values, the discrepancy distance, L and the percentage error are given in the Appendix F while figure 4.16 shows the error chart. The discrepancy distance L is determined based on the triangle law of cosines,

$$L = \sqrt{{R_A}^2 + {R_M}^2 - 2R_A R_M \cos(\theta_A - \theta_M)}$$

where R_A is the actual radius, R_M is the measured radius, θ_A is the actual angle and θ_M is the measured angle. The percentage error is calculated by

$$\text{Error} = \frac{L}{R_A} \times 100\%$$

we see that the performance of the system at 90° and 180°are nearly agree with the actual location. However in the range of 30°-60° and 150°, the system is comparatively less accurate. The values at 90° are gives them, most satisfied results compared to others with the percentage error within 3% at distance 2-3 meters apart.

In the error chart, the percentage of error reduces with distance at 30° and 60°. At 30°, we see that when the distance is 0.5 meters, the percentage of error is huge however it decreases substantially with distance beyond this point.

At 90° the percentage error varying along the distance, however the percentage error tends to decreasing at distance 2-3 meters. The graph shows the system performed well at this point and the percentage error is relatively low. At 120°, the outcome behaved similarly with the one along 90°. At 150°, the average error is almost constant except a single spike at 1 meter. Last but not least, 180°.the graph shows the system performed especially at 3 meters apart.

According to Appendix F the lowest percentage error occurs at 90°, 2 meters away from the origin. The distance of the measure value from the actual value is just 0.09 meters which accounts to 1.57% of error. The highest percentage error occurred at 30°with the distance 0.5 meter. This account to the error of 380.56% with the discrepancy 1.9 meters.

By comparing the angles with the distance show that the accuracy of the system increases with distance. This is shown in the graphs in which the distance at 0.5 meters suffers with high error. The error is reduced from 0.5 meter through 1 meter and lastly 1.5 to 3 meters give the least error. The accuracy is determined by the percentage error.

Referring to the 6 experiments taken, we can see that the results are varies as the angle between the source and microphone B is differ. From this observation, we can see that in 90° degree, the system gives better results. The omnidirectional microphones might be more sensitive in capturing the sound in 90° angle. Similarly, the results for 120° and 180° are quite good as well. Moreover, from the experiment, we observed that the results are more accurate as the distance between microphone B and the sound source are long enough and but still in the effective range of microphone.

As mentioned in Chapter 2, the microphone has its own effective range of accuracy. The 3-to-1rule [16], stated that for multiple microphones, the distance between 3 microphones should be at least three times the distance from each microphone to its intended sound source to eliminate the phase cancellation. From the experiments, it shows that the percentage error is getting smaller as the distance between microphone is long enough apart.

From percentage error, we can see that the error is acceptable. However, the result is slightly inaccurate due to noise of the environment and also from the devices itself. Moreover, the distance of the microphones with the source might affect the reading. It is because when the source of sound and the microphones is located near to each other, the microphones might be more sensitive and not able to differentiate the signals well. The discrepancies of the sound intensity between each microphone are too small for the system to differentiate. This might contributed to the huge error as shown in Figure 4.10.

There is limitation to the software where Simulink is unable to do some part of the programming and unable to measure the maximum amplitude directly but is taken up by MATLAB for the subsequent task. More troubleshooting will be conducted to tackle this problem.

CHAPTER 5

CONCLUSION AND RECOMMENDATION

5.1 Conclusion

The acoustic object locating system is a feasible project which can benefit to the community in reducing car crime in car parks and their existence can be of valuable assistance to the security. The system consists of the acoustic sensors and a processing unit to produce the accurate acoustic location using the triangulation method. The theoretical calculation and filter design part which is used to filter away the unwanted signal other than the alarm signal are already done. The distance of the location is successfully determined and plotted in map. The error is significant due software problems, noises, sensitivity and the quality of the equipment. More experiments will be conducted to test and debug the system.

5.2 Recommendation

To achieve better performance of the system, it is recommended to use high quality hardware such as microphones and sound cards. But due to cost constraint, it was not chosen. It is recommended that a more sensitive microphone is used so that the area of receiving signal will be more abroad and wide. The scope of the study was focused only on the open car parking area, due to the sound reflection, so it is recommended to widen the scope to close parking area.

The chosen of filter might affect the effectiveness of the system. Therefore, it is recommended to improve the filtration system. The cut-off frequencies of the signal have has to be precise so that the unwanted signal would not be triggered. Next, we find that the Simulink program has several limitations and several part of this project cannot be executed using the program .It is also recommended to use other software which can support the system well.

Next, it is recommended if this project can be extended to the security system by adding the CCTV system so that the location can be detected and at the same time the camera will show the real places. The warning alarm should be warned the control room once the acoustic object is triggered.

REFERENCES

[1] Trevor J.Cox And Peter D'antonio,2003, 'Engineering art: *The Science Of Concert Hall Acoutic'*, Maney of the Institute of Materials, Minerals And Mining.

[2] *The Speed of Sound,* Retrieved March 2011from <http://www.physicsclassroom.com/class/sound/u1112c.cfm/>

[3] Chistopher Teh, 30 September, *Private car theft statisic in Malaysia: one stolen every 24 minutes.* Retrived April 2011 <http://christopherteh.com/blog/2010/09/30/private-car-theft-statistics-in-malaysia-one-stolen-every-24-minutes/>

[4]Coretta Phillip, *Home Office Policing and Reducing Crime Unit, Retrieved April 2011 from* < http://www.popcenter.org/library/crimeprevention/>

[5] Tiliey, N. (1993a). *Understanding Car Parks, Crime and CCTV: Evaluation Lessons from Safer Cities.* (Crime Prevention Unit Series Paper, #42.) London, UK: Home Office

[6] Lazaro J Hester, *CCTV System- The Advantages And Disadvantages*, Retrieved April 2011 from , <http://ezinearticles.com/?CCTV-System---The-Advantages-and-the-Disadvantages&id=3998759>.

[7] Zitzawitz, Neff & Mark Davids. *Physics: Principles and Problems*, New York: Glencoe, 1995: 308.

[8]Wikipedia.*Sound Intensity* Last View: 29 September 2010 at 12:03,Retrieved from <http://en.wikipedia.org/wiki/Sound_intensity >

[9]*The Decibel,* NDT Resource Center Retrieved April 2011 from <,http://www.ndt-ed.org/GeneralResources/decibel/decibel.htm>

[10] Aaron Friedman (July 2003), *The Case Against Car Alarm* , Retreived November 2010, from <http://www.noisefree.org/caralarms/banalarms.html>

[11] Paul Bourke, June 1933, *Fast Fourier Transform,* Retrieved April 2011 from <http://paulbourke.net/miscellaneous/dft/>

[12] Yaser Rahmati, *Simulink Basic Tutorial,* Retreived March 2011 from, http://pecworld.zxq.net/Software/Matlab/Part_1/Details/04/18_Simulink_Basics_Tutorial.htm

[13] Introduction to Embedded MATLAB Function Blocks, Retreived March 2011 from,< http://www.mathworks.com/help/toolbox/simulink/ug/f6-106261.html>

[14] Michael Pollick, L. S. Wynn, *What Are Omnidirectional Microphones?* http://www.wisegeek.com/what-are-omnidirectional-microphones.htm, August 2010

[15] Finite Impulse response (FIR) Filters, Retrieved April 2011 from ,<http://www.scribd.com/doc/38745616/Fir-Filter>, last visit 28/10/2010.

[16] Shure, Microphones Techniques for Live Sound, *Article Acoustic Characteristic,*2007

[17] Yasir Salih Osman Ali, *Video Localization Using Array Microphones,* Universiti Teknologi PETRONAS, December 2009

[18] J. Strobel,E.Wigley and N.Evans, *BUZZ- Acoustical Engineering Methodology to Measure Student Engagement,* 2009, Purdue University

[19]Microphones, HyperPhysic, http://hyperphysics.phy-astr.gsu.edu/hbase/audio/mic.html, last visit 28/10/2010

[20] S. Knight, USB Sound Card Creative, 27/3/2009

[21] : *Microphone Mixer: AKG DMM 4/2/4,* June 2010

[22] Jeff Morton, 28 Jan 2008, *Sound Activated Recorder with Spectogram in C#,* Retrieved February 2011 from <http://www.codeproject.com/KB/audio-video/SoundCatcher.aspx,>

APPENDIX A

TYPES OF MICROPHONES

The simplest and most used input to detect the alarm sound is the microphone. A microphone is a transducer which converts acoustical energy (sound signals) into electrical energy (an audio) [19].There are two different styles of microphones which are unidirectional and omnidirectional [14]. Both microphones have their own performance where unidirectional mainly pick up sounds and aimed directly into center while for omnidirectional, it pick up the sound from any direction because the electronic pick-up is placed in the center of a mesh-covered dome.

Referring to the Article Microphones Techniques for Live Sound, it stated that the omnidirectional microphone has equal output or sensitivity at all angles which is a full 360 degrees.[16] The advantage of omnidirectional is that it is more sensitive and we can get more accurate data. It also picks up the maximum amount of ambient sound

APPENDIX B

INTERFACES BETWEEN PC AND MICROPHONE

There are several ways to establish the interface between PC and Microphone such as by using the USB Sound Card, Multiplexer and Mixer.

USB Sound Card

The USB Sound Card consists of the microphone and also the headphone jacks. The cards can be connected to the USB port in the computer and powered up from the motherboard which connected through the USB port [20]. It also has a minimal voltage draw from the USB port and can work well on a multiple sound card from the PC.

Figure : USB Sound Card [8]

Referring to Sound Devices (2007), full speed of USB devices signal is 12 Mb/s, while low speed devices use a 1.5 Mb/s sub channel. In order to connect 3 microphones at the same time, we might use 3 USB Sound Cards.

The advantages of the sound card are that it is simple and user friendly compared to the PIC microcontroller. If we consider the cost consume and the budgetary, we may use only 2 USB sound cards instead of 3 because one more sound card is already built-in in our PC. The microphone can be connected directly into the microphone jack on the computer.

Multiplexer

Multiplexer allows the microphones data to be fed sequentially to the computer based on predetermined sequence or via instruction given by the computer. At the computer side the received signal is de-multiplexed to reconstruct the original signals; hence it is very essential to synchronize the multiplexing and the de-multiplexing process so the computer can be able to reconstruct the correct signals.

This method is used in Video Localization Project, Yasir (2009) where PIC microcontroller is used for the execution and interfaces which might involved time lagging due to the multiplexer hold time [17]. The advantage of using this method is the signals of 3 microphones can be transmitted synchronously by using only sound card and we can reduce the budgetary for the project.

Sound Mixer

Figure: Microphone Mixer [21]

The sound mixer is designed with a mixing algorithm of multiple microphones. The mixer comes with four microphone and two stereo aux inputs, mixing algorithms with a noise-sensitive threshold to prevent accidental activation of input channels, and a digital signal processor with a dbx compressor [21]. There is peak hold display in this device which can be used for the signal observation. Microphone Mixer also include the headphone output and PC interface.

APPENDIX C

THE TRIANGULATION FORMULA METHOD

The difference between Intensity of Microphone A and B

$$\Delta S_{AB} = S_A - S_B$$

$$= 10\ log_{10}(\frac{I_A}{I_o}) - 10\ log_{10}(\frac{I_B}{I_o})$$

$$= 10\ log_{10}(\frac{I_B}{I_A})$$

$$= 10\ log_{10}(\frac{P}{4\pi OA^2}) - 10\ log_{10}(\frac{P}{4\pi OB^2})$$

$$= 10\ log_{10}(\frac{OB}{OA})^2 \quad (3)$$

The difference between Intensity of Microphone B and C

$$\Delta S_{BC} = S_B - S_C$$

$$= 10\ log_{10}(\frac{I_B}{I_o}) - 10\ log_{10}(\frac{I_C}{I_o})$$

$$= 10\ log_{10}(\frac{I_B}{I_C})$$

$$= 10\ log_{10}(\frac{P}{4\pi OB^2}) - 10\ log_{10}(\frac{P}{4\pi OC^2})$$

$$= 10\ log_{10}(\frac{OC}{OB})^2 \quad (4)$$

By using Trigonometry Properties,

$$OA^2 = OB^2 + L^2 - 2LOB\ cos\theta_y \quad (5)$$

$$OC^2 = OB^2 + L^2 - 2LOB\ cos\theta_r$$

$$= OB^2 + L^2 + 2LOB\ cos\theta_y \quad (6)$$

Rearrange the value of ΔS_{AB} in equation (3)

$$\Delta S_{AB} = 10\ log_{10}(\frac{OB}{OA})^2$$

$$\frac{OB^2}{OA^2} = 10^{\ 0.1\Delta S_{AB}}$$

$$OA^2 = \frac{OB^2}{10^{0.1\Delta S_{AB}}} \qquad (7)$$

Then, rearrange the value of ΔS_{BC}

$$\Delta S_{BC} = 10\ log_{10}(\frac{OC}{OB})^2$$

$$\frac{OC^2}{OB^2} = 10^{\ 0.1\Delta S_{BC}}$$

$$OC^2 = 10^{\ 0.1\Delta S_{BC}}\ (OB^2\ ) \qquad (8)$$

Substitution equation (7) into the equation (5)

$$OA^2 = OB^2\ +\ L^2 -\ 2LOB\ cos\theta_y$$

$$\frac{OB^2}{10^{0.1\Delta S_{AB}}} = OB^2\ +\ L^2 -\ 2LOB\ cos\theta_y$$

$$(10^{-0.1\Delta S_{AB}})OB^2 = OB^2\ +\ L^2 -\ 2LOB\ cos\theta_y$$

$$OB^2(10^{-0.1\Delta S_{AB}} -\ 1) - L^2 -\ 2LOB\ cos\theta_y = 0 \qquad (9)$$

Substitution of equation (8) into equation (6)

$$OC^2 = OB^2\ +\ L^2 +\ 2LOB\ cos\theta_y$$

$$(10^{0.1\Delta S_{BC}})OB^2 = OB^2\ +\ L^2 +\ 2LOB\ cos\theta_y$$

$$OB^2\ (10^{0.1\Delta S_{BC}}) - 1 - L^2 -\ 2LOB\ cos\theta_y \qquad (10)$$

To find OB, the equation (9) is added to equation (10), then we will get

$$OB^2\ (10^{-0.1\Delta S_{AB}} + 10^{0.1\Delta S_{BC}} - 2) - 2L^2 = 0$$

$$OB^2 = \frac{2L^2}{(10^{-0.1\Delta S_{AB}+10^{0.1\Delta S_{BC}-2})}}$$

$$OB = \frac{\sqrt{2}\,L}{\sqrt{(10^{-0.1\Delta S_{AB}} + 10^{0.1\Delta S_{BC}} - 2)}} \tag{11}$$

To find , the equation (9) is compare to equation (10)

$$OB^2\,(10^{-0.1\Delta S_{AB}} - 10^{0.1\Delta S_{BC}}) + 4\,LOB\cos\theta_y = 0$$

$$\cos\theta_y = \frac{OB^2(10^{0.1\Delta S_{BC}} - 10^{-0.1\Delta S_{AB}})}{4LOB}$$

$$\cos\theta_y = \frac{\sqrt{2}L}{\sqrt{(10^{-0.1\Delta S_{AB}} + 10^{0.1\Delta S_{BC}} - 2)}} \cdot \frac{(10^{0.1\Delta S_{BC}} - 10^{-0.1\Delta S_{AB}})}{4L}$$

$$\theta_y = \cos^{-1}\left(\frac{\sqrt{2}(10^{0.1\Delta S_{BC}} - 10^{-0.1\Delta S_{AB}}}{4\sqrt{(10^{-0.1\Delta S_{AB}} + 10^{0.1\Delta S_{BC}} - 2)}}\right) \tag{12}$$

APPENDIX D

CODING USING MATLAB

Part 1 -Using Command Windows

```
>> [data fs]=wavread('D:\Notes\Final Year 1 Roxx\FYP1\Sound System\ALARM1.wav');

>> [tone,Fs,nbits]=wavread('D:\Notes\Final Year 1 Roxx\FYP1\Sound System\ALARM1.wav');

>> x=tone;

>> n=length(x);

>> t=(0:n-1)'/Fs;

>> soundsc(x,Fs)

>> plot(t,x)
```

Part 2 - Calculation with mfile

```
function [OB,L]= length(L,dSab,dSbc);

%COmpute the length of microphone and sound (OB)

%sides have intensity L,dSab,dSbc

%Inputs:

%L,dSab,dSbc:Intensity

%Output:

%OB:Length of Sounrce and Microphone

%USage:

%Written by HAzilah,October 21,2010.

 L=assign;

 dSab = Sa-Sb;

 dSbc = Sb-Sc;

OB = sqrt(2)*L/(sqrt(10^(-0.1*dSab))+10^(0.1*dSbc)-2);


command: Length=(L,dSab,dSbc)
```

Part 3-Calculation using command windows (assign value L,dSab,dSbc)

1. Length of OB

```
dSab=5;

dSbc=6;

L=7;

OB = (sqrt(2)*L)/(sqrt(10^(-0.1*dSab)+10^(0.1*dSbc)-2))
```

Part 4- Fast Fourier Transform

```
tone,Fs,nbits]=wavread('C:\Users\Hedgie\Desktop\Alarm1_001.wav');

>> [data Fs]=wavread('C:\Users\Hedgie\Desktop\Alarm1_001.wav');

>> x=tone;

>> n=length(x);

>> t=(0:n-1)'/Fs;

>> plot(t,x)

>> soundsc(x,Fs)

>> y=2*abs(fft(data))/length(data);

>> y=y(1:end/2);

>> f_nyquist=Fs/2;

>> x=linspace(0,f_nyquist,length(y));

>> [y_max index]=max (y);

>> f_principle=x(index);

>> figure

>> plot(x,y,f_principle,y_max, )

>> plot(x,y,f_principle,y_max,'o' )

>> xlabel('Frequency,Hz');

>> ylabel('Normalized Magnitude');

>> title('Frequency domain Plot of Alarm Sound');

>> [tone,Fs,nbits]=wavread('C:\Users\Hedgie\Desktop\test\Alarm3.wav');
```

Part 5-Maximum Value and Calculation

```
L=0.65;

[ymax,maxindex]=max(tout1.signals.values);

[ymax1,maxindex1]=max(tout2.signals.values);

[ymax2,maxindex2]=max(tout3.signals.values);

peakval = ymax;

peakval1 = ymax1;

peakval2 = ymax2;

dSab = peakval-peakval1;

dSbc = peakval1-peakval2;

OB = sqrt(2)*L/(sqrt(10^(-0.1*dSab))+10^(0.1*dSbc)-2)

theta_y = acos((sqrt(2)*(10^(0.1*dSbc)-10^(-0.1*dSab)))/(4*sqrt(10^(-0.1*dSab)+10^(0.1*dSbc)-2)))

theta_y_deg=theta_y*180/pi

polar (theta_y_deg,OB,'O')
```

APPENDIX E

ACOUSTIC DETECTION AND TRACKING SYSTEM

This application has been developed by Lemuel P.Mathew. The system of the invention identified target to enable the tracking by generating the information of elevation and azimuth of the signals. This acoustical coordinates system consists of at least one pair of acoustical transducer (microphone) which located separatedly to receive the signals from the target. After converting the signals from analog to digital, the processing unit generated the phase differences signals between each of frequency bin received by the microphones and get the phase different slopes.The processing unit will determine and identify the target from this value.

Theoretically, the bearing of sound signals from the target can be determined by a comparison between time of arrival of the signals from that target at various microphones. Their system was using two pairs of microphones which mutually in orthogonal array. By the phase difference between signals received of each of paired microphones, the direction of the target is computed in both azimuth and elevation.

After received the signals, the outputs of the microphones are sequentially sampled and multiplexed before they are fed into a fast Fourier transform (FFT). In this FFT, the signals are transform into real and imaginary components so that the value of frequency bin can be preserved [22].The output of FFT is then fed to the digital processor to compute the power and phase angle for each sound frequency bin coming from each microphones.The azimuth and elevation of each target can then be computed from the phase difference slopes for the paired azimuth and elevation microphones respectively. Signals having common phase difference slopes are then grouped together, these common slopes indicating signals originating at common targets

APPENDIX F

RESULTS

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	0.5	2.4	30	35.41	1.90	380.56
2	1	4.4	30	29.58	3.40	340.00
3	1.5	1.8	30	34.11	0.32	21.49
4	2	2.8	30	30.23	0.80	40.00
5	2.5	3	30	49.8	1.07	42.65
6	3	3	30	33.23	0.17	5.64

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	0.5	1.5	60	61.5	1.00	200.05
2	1	0.3	60	59.7	0.70	70.00
3	1.5	1.3	60	53.21	0.26	17.30
4	2	3	60	55.7	1.02	50.84
5	2.5	3	60	68	0.63	25.17
6	3	3.2	60	60.3	0.95	31.58

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	0.5	0.2	90	92.1	0.30	60.04
2	1	1.2	90	90.2	0.20	20.00
3	1.5	1.8	90	90.05	0.30	20.00
4	2	2	90	89.1	0.03	1.57
5	2.5	2.45	90	91.4	0.08	3.14
6	3	3.24	90	87.2	0.28	9.48

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	0.5	0.8	120	131.2	0.32	64.88
2	1	1.3	120	122.3	0.30	30.35
3	1.5	1.9	120	119.4	0.40	26.69
4	2	3	120	99.3	1.33	66.61
5	2.5	2.2	120	105.3	0.67	26.83
6	3	3.1	120	123.22	0.20	6.61

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	0.5	0.4	150	156.7	0.11	22.57
2	1	2.7	150	148.89	1.70	170.03
3	1.5	1.8	150	155.33	0.34	22.44
4	2	2.8	150	153.7	0.81	40.72
5	2.5	2	150	145.6	0.53	21.15
6	3	3	150	150.2	0.01	0.35

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	0.5	0.45	180	181	0.05	10.14
2	1	1.2	180	183.4	0.21	21.03
3	1.5	1.8	180	201.3	0.68	45.16
4	2	1.8	180	182	0.21	10.53
5	2.5	2	180	189.45	0.62	24.84
6	3	3.1	180	179.3	0.11	3.56

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	0.5	2.4	30	35.41	1.90	380.56
2	0.5	1.5	60	61.5	1.00	200.05
3	0.5	0.2	90	92.1	0.30	60.04
4	0.5	0.8	120	131.2	0.32	64.88
5	0.5	0.4	150	156.7	0.11	22.57
6	0.5	0.45	180	181	0.05	10.14

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	1	4.4	30	29.58	3.40	340.00
2	1	0.3	60	59.7	0.70	70.00
3	1	1.2	90	90.2	0.20	20.00
4	1	1.3	120	122.3	0.30	30.35
5	1	2.7	150	148.89	1.70	170.03
6	1	1.2	180	183.4	0.21	21.03

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	1.5	1.8	30	34.11	0.32	21.49
2	1.5	1.3	60	53.21	0.26	17.30
3	1.5	1.8	90	90.05	0.30	20.00
4	1.5	1.9	120	119.4	0.40	26.69
5	1.5	1.8	150	155.33	0.34	22.44
6	1.5	1.8	180	201.3	0.68	45.16

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	2	2.8	30	30.23	0.80	40.00
2	2	3	60	55.7	1.02	50.84
3	2	2	90	89.1	0.03	1.57
4	2	3	120	99.3	1.33	66.61
5	2	2.8	150	153.7	0.81	40.72
6	2	1.8	180	182	0.21	10.53

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	2.5	3	30	49.8	1.07	42.65
2	2.5	3	60	68	0.63	25.17
3	2.5	2.45	90	91.4	0.08	3.14
4	2.5	2.2	120	105.3	0.67	26.83
5	2.5	2	150	145.6	0.53	21.15
6	2.5	2	180	189.45	0.62	24.84

	DISTANCE (METERS)		ANGLE (DEGREES)			
NO	ACTUAL	MEASURED	ACTUAL	MEASURED	L	% ERROR
1	3	3	30	33.23	0.17	5.64
2	3	3.2	60	60.3	0.20	6.69
3	3	3.24	90	87.2	0.28	9.48
4	3	3.1	120	123.22	0.20	6.61
5	3	3	150	150.2	0.01	0.35
6	3	3.1	180	179.3	0.11	3.56