"Acoustic Source Localization"

END SEMESTER PROJECT REPORT
FOR THE DEGREE OF

BACHELOR OF TECHNOLOGY

IN

INFORMATION TECHNOLOGY

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INDIAN INSTITUTE OF INFORMATION TECHNOLOGY, ALLAHABAD

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2nd December, 2016

CANDIDATES' DECLARATION

We hereby declare that the work presented in this project report entitled **"Acoustic Source Localization"** submitted for the evaluation of the end semester exam of the 7th Semester of B.Tech (IT) at Indian Institute of Information Technology, Allahabad is an authenticated record of our original work carried out from August 2016 to December 2016 under the guidance of **Dr.Vrijendra Singh**.

Date: 2nd December, 2016
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CERTIFICATE FROM SUPERVISOR

I do hereby declare that the mini project work prepared under my supervision by B.Tech group titled **"Acoustic Source Localization"** be accepted in the fulfillment of the requirements of the mini project work of Bachelor of Technology in Information Technology, 7th semester. This is to certify that the above statement made by the candidate is correct to the best of my knowledge.

Date: 2nd December, 2016
Place: Allahabad, IIITA

Dr.Vrijendra Singh
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1. Introduction

Sound localization refers to the acoustical engineering (branch of engineering dealing with sound and vibration) technology that is being used to identify the location of a sound source. The sound source localization problem is also a source localization problem. It involves the structure arrangement design of the sensors and signal processing techniques. Usually the location of the source is determined by the direction of the coming sound waves (horizontal and vertical angles) and the distance between the source and the sensors.

The interest in sound localization is widely increasing due to the need for improved solutions in some audio and acoustics fields, such as hearing aids, surveillance and navigation. Existing real-time passive sound localization systems are mainly based on the time-difference-of-arrival (TDOA) approach. Direction of arrival (DOA) estimation of acoustic signals using a set of spatially separated microphones has many practical applications in everyday life. Most practical acoustic source localization schemes are based on time delay of arrival estimation (TDOA) for the following reasons: such systems are conceptually simple. They are reasonably effective in moderately reverberant environments. Moreover, their low computational complexity makes them well-suited to real-time implementation with several sensors.

In this project, the location of the source is found using a two-dimensional-three-element microphone array. A pair of microphones gives the DOA w.r.t. the axis of the microphones. Since the target has two degrees of freedom, the DOA estimated would give only the direction of the source. On coupling a third microphone with previously installed microphones in an ’L’ fashion, the direction of arrival (DOA) w.r.t. the axis containing the third and the center microphone can be obtained. In the second stage, the measured DOA’s are used to obtain the source position by solving a system of equations.

The hardware used for data acquisition, sampling frequency, number of microphones used for data acquisition, and noise present in the signals captured, determine the accuracy of the estimates of the acoustic source localization.

Increase in the number of microphones increases the performance of source location estimation. Many of the conventional microphone array techniques for DOA estimates use large number of microphones typically 10 - 40 microphones. This puts a requirement for large number of data acquisition channels, which implies involvement of huge cost. Especially applications like automatic camera steering uses bulky and huge microphone arrays. This concludes that there is a need for reducing the size and cost involved in time delay estimation of acoustic source.

As we see that in the every day life, we find huge amount of electronic waste which needs to be recycled. So once we implement the acoustic source localization we use the microphones collected from this waste and design the acoustic source localization which reduces the size and cost involved in time delay estimation of acoustic source in the future.

1.1 Motivation

Sound source localization has a wide range of applications from remote sensing to the Global Positioning System. It is used in sound source separation, sound source tracking and speech enhancement. Underwater sonar uses sound source localization techniques to identify the location of a target. It is also used in robots for effective human-robot interaction. With the increasing demand of robotic hearing, some applications such as human-machine interface, handicappers' aid and some military applications are being widely exploited like devices to locate the sources of artillery fire. It is also used in commercial applications like improving speech quality in hands free telephony and video conferencing.

2. Problem Definition

Given a set of M acoustic sensors (microphones) in known locations, we assume that source is present in a defined coordinate system; we know the number of sensors present and single sound source present in the system. The sound source is excited using a broad band signal with defined bandwidth and the signal is captured by each of the acoustic sensors. The Time Difference of Arrival (TDOA) is estimated from the captured audio signals. Our goal is to locate the acoustic source in 2 Dimensional plane using microphone arrays.

2.1 Objectives

Our objective is to implement the acoustic source localization which gives the efficient performance in locating the sound source.

2.2 Assumptions and Limitations

We assume the following conditions under which location of sound source is estimated:

1. Single sound source is taken in the project which is infinitesimally small and it has the property of omni directional source.
2. Reflections from the bottom of the plane and from the surrounding objects and environment are negligible.
3. No disturbing noise sources contribute to the sound field.
4. The sound source to be located is assumed to be stationary during the data acquisition period.

5. The change in sound velocity due to change in pressure and temperature are neglected.
6. The velocity of sound in air is taken as 330 m/sec.
7. Knowledge of positions of acoustic receivers and perfect alignment of the receivers as prescribed by processing techniques.

2.3 Limitations:-

1. In the experimental setup, we have worked for a single beep sound only which was produced by the source while recording different samples.
2. The source is stationary.

2.4 Process of Acoustic Source Localization

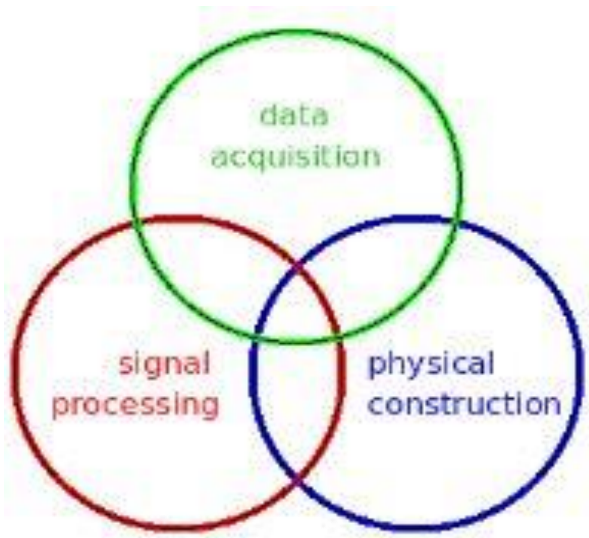


Figure 2.1 Diagramatic representation of the Acoustic Source Localization

Acoustic Source Localization Process includes the following steps:-

1. Data has to be collected using the microphones and a source placed in an accurate coordinate system.
2. The collected data is processed using some signal processing techniques.
3. The final experimental setup has to be made to test and verify the results for the unknown values.

3. Literature Survey

Literature Survey Table

S. No.	Title	Year	Journal/Conference	Objective	Method	Challenge(s) Dealt	Future Scope
1	Acoustic Source Localization Using Time Delay Estimation [1]	2007	Thesis IISC, Bangalore	To localize acoustic sound source in a two-dimensional plane and three-dimensional space using a microphone array.	Dominant Frequency Selection (DFSE) algorithm	The acoustic positioned in a plane and space, have been successfully located. Together with the DFSE for TDOA estimation, the source could be localized.	DFSE for targeting a broad band source has a poor performance. This could further be improvised for accurate TDOA measurements.
2	Acoustic Source Localisation System using Microphone Arrays [2]	2012	Journal International Journal ofAdvanced Research in Computer Science and Software Engineering	To develop an acoustic source localisation system using commonly available sound monitoring and recording devices such as microphones and sound cards.	MMSE estimator based speech enhancement	The proposed method is capable of estimating speech signal even in situations of very high degree of noise or low SNRs.	The estimated speech signals are used to estimate the location of the source.
3	Passive Acoustic Source Localization at a Low Sampling Rate Based on a Five-Element Cross Microphone Array [3]	2015	Journal Open Access Sensors	To develop accurate acoustic source localization at a low sampling rate	Up-Sampling - Generalized Cross Correlation (US-GCC)	Modification of the GCC method based on the US theory to improve the TDOA accuracy and consequently the localization accuracy at a low sampling rate	To perform accurate 3D near-field localization at a low sampling rate, and also, the possibility is given for applying the US-GCC method with an interpolation factor of 15 to a small portable system, especially a multi tasking micro-embedded system.
4	Estimating Time Delay using GCC for Speech Source Localisation [4]	2013	Journal International Journal of Application or Innovation	To locate a source of speech signals using microphone arrays	Generalized Cross-Correlation (GCC)	The paper presented the generalised cross correlation function for speech signals (recorded using microphone arrays) in time	To evaluate the performance of microphone array in locating speech sources and develop a model for the calculation of the

			in Engineering & Management (IJAIEM)			domain and frequency domain to deal with high level of noise (especially the traffic)	cross-correlation function for speech signals (recorded through microphone arrays) in time domain and frequency domain.
5	Design and Implementation of Sound Source Localization System Based on FPGA [5]	2012	Conference International Conference on Systems and Informatics	To promote FPGA's application in the development of sound source localization.	Endpoints with double-threshold	Removing noise interference and and locationg the sound source using EP3C25Q240C8	Based on the obtained results, the range of Φ and θ are limited to ±90°and 0°～90°. So this limitation needs to be further improved for future use.
6	Maximum Likelihood Sound Source Localization and Beam forming for Directional Microphone Arrays in Distributed Meetings [6]	2008	Journal IEEE Transactions on Multimedia	This paper presents a unified maximum likelihood framework of these two techniques, and demonstrates how such a framework can be adapted to create efficient SSL and beamforming algorithms for reverberant rooms and unknown directional patterns of microphones.	ML (Maximum-Likelihood) framework for sound source localization and beamforming with microphone arrays.	The proposed ML-SSL framework can handle reverberant environments and unknown directional patterns of the microphones.	Future work will include extending the current framework to multisource scenarios.
7	A simple and Efficient Estimator for Hyperbolic Location [7]	1994	Journal IEEE Transactions on Signal Processing	An efficient technique in locating a source based on intersections of hyperbolic curves defined by the time differences of arrival of a signal received at a number of sensors.	Hyperbolic position fixing solution	A new approach for localizing a source from a set of hyperbolic curves defined by TDOA measurements.	The new technique offers a computational advantage over the Taylor-series technique and eliminates the convergence problem.
8	Statistical Theory of Passive Location Systems [8]		Journal IEEE Transactions on Aerospace	A derivation of the principal algorithms and an analysis of the performance of the two most important passive location	Estimation methods	Complete derivation of chan's method	Future work will done to minimise the negative aspects of the method.

			and Electronic Systems	systems for stationary transmitters, hyperbolic location systems and direction- finding location systems			
9	Simple Solutions for Hyperbolic and Relative Position fixes [9]	1990	Journal IEEE Transactions on Aerospace and Electronic Systems	To show that if the additional information is the navigator altitude or the range difference to the fourth station , the computation of the navigation fix is reduced to find the roots of a quadratic	Navigation position relative to three stations	The navigation fixes discussed are based on range difference or range sum information, converted from time difference and time sum measurements.	For terrestrial navigation system that rely on ground wave propagation such conversation can be complicated. So the future scope towards this involves in improving.
10	Position Location Solutions by Taylor Series Estimation. [10]	1976	Journal IEEE Transactions on Aerospace and Electronic Systems	This method is useful In solving multimeasurement mixed-mode position-location problems typical of many navigational applications.	Tavlor-Series Method	While convergence Is not proved, examples show that most problems do converge to the correct solution from reasonable initial guesses. The method also provides the statistical spread of the solution errors.	The Taylor-series Method places many Different measurements in a Common mathematical framework. It is hoped that this will encourage the consistent evaluation of the different measurement processes.
11	A Passive Localization Algorithm and Its Accuracy Analysis [11]	1987	IEEE Journal of Oceanic Engineering	A localization technique based on solving a set of linear equations is presented and its accuracy properties are analyzed.	Localization from range differences method	The problem of estimating source location from noisy measurements of range differences (RD 's) is considered.	Using the formulas, the sensor configuration can be optimized for best accuracy for a given surveillance scenario.
12	Hyperbolic Position Location Estimation in the Multipath Propagation Environment [12]	2009	Thesis by the Ministry of Education and Science of the Republic of Poland	calculation of the geographical posi-tion of a mobile station (MS) using the time differences of arrival (TDOA) between several base stations (BS) and MS.	Chan's Foy's Fang's Method	requires a priori knowledge of the approximate location and distance between mobile station and first base station.	Studying the ambiguities in the Fang's and Chan's algorithms

Chan's Method [7]:

$R_i = \sqrt{(X_i - x)^2 + (Y_i - y)^2}$

$= \sqrt{(X_i)^2 + (Y_i)^2 - 2X_i x - 2Y_i y + x^2 + y^2}$ - (i)

R_i = Distance between source and the i^{th} microphone

(X_i, Y_i) = Co-ordinates of i^{th} base station

(x, y) = Co-ordinates of mobile station

$R_{i,1} = v.t_{i,1} = \sqrt{(X_i - x)^2 + (Y_i - y)^2} - \sqrt{(X_1 - x)^2 + (Y_1 - y)^2}$ - (ii)

$R_{i,1}$ = Distance between R_i and R_1

$R_{i,1} = R_i - R_1$ - (iii)

$R_i^2 = ((R_{i,1} + R_1)^2$

$R_{i,1}^2 + 2R_{i,1}R_1 + R_1^2 = X_i^2 + Y_i^2 - 2X_i x - 2Y_i y + x^2 + y^2$ - (iv)

For i = 1 in Equation (i),

$R_1 = \sqrt{X_1^2 + Y_1^2 - 2X_1 x - 2Y_1 y + x^2 + y^2}$ - (v)

Putting equation (v) into equation (iv),

$R_{i,1}^2 + 2R_{i,1}R_1 + X_1^2 + Y_1^2 - 2X_1 x - 2Y_1 y + x^2 + y^2 = X_i^2 + Y_i^2 - 2X_i x - 2Y_i y + x^2 + y^2$

$R_{i,1}^2 + 2R_{i,1}R_1 = X_i^2 + Y_i^2 - X_1^2 - Y_1^2 - 2(X_i - X_1)\,x - 2(Y_i - y_1)\,y$
$= X_i^2 + Y_i^2 - 2X_{i,1}x - 2Y_{i,1}\,y - X_1^2 - Y_1^2$ - (vi)

$\begin{bmatrix} X_{2,1} & Y_{2,1} \\ X_{3,1} & Y_{3,1} \end{bmatrix} \begin{bmatrix} x \\ y \end{bmatrix} = -\frac{1}{2}\begin{bmatrix} R_{2,1}^2 - K_2 + K_1 \\ R_{3,1}^2 - K_3 + K_1 \end{bmatrix} - R_1 \begin{bmatrix} R_{2,1} \\ R_{3,1} \end{bmatrix}$ - (vii)

Where,

$K_1 = X_1^2 + Y_1^2$
$K_2 = X_2^2 + Y_2^2$
$K_3 = X_3^2 + Y_3^2$

K_i = Distance of i^{th} microphone from the origin

Fang's Method [9]:

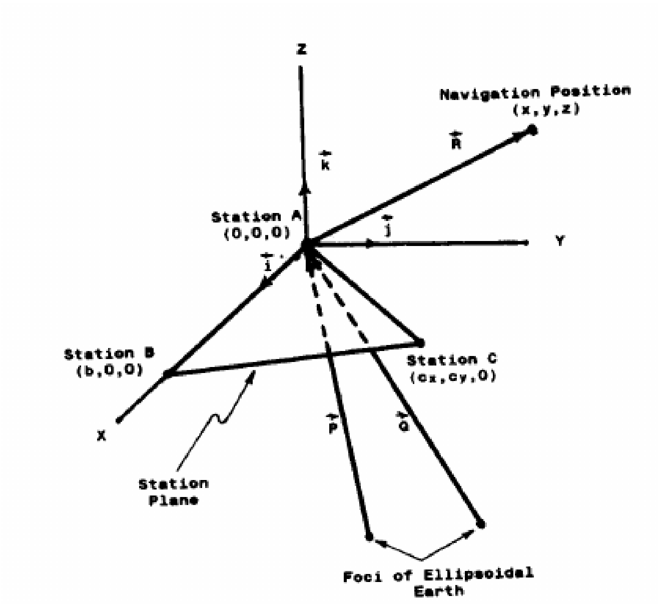


Figure 3.1: Navigation position referenced to local coordinates defined by station baseline [9]

Let V be the Signal Velocity

Differences in time of signal arrival at the station pair A, B and A, C are

$T_{ab} = T_a – T_b$

$T_{ac} = T_a - T_c$

R_{ab} and R_{ac}: Range differences from navigation position to stations, converted from time of arrival difference.

$$(x^2 + y^2 + z^2)^{1/2} - ((x-b)^2 + y^2 + z^2)^{1/2} = V * T_{ab} = R_{ab} \qquad (1)$$
$$(x^2 + y^2 + z^2)^{1/2} - ((x - c_x)^2 + (y - c_y)^2 + z^2)^{1/2} = V * T_{ac} = R_{ac} \qquad (2)$$

$$R^2_{ab} - b^2 + 2b * x = 2 R_{ab} (x^2 + y^2 + z^2)^{1/2} \qquad (3)$$
$$R^2_{ac} - c^2 + 2c_x * x + 2c_y * y = 2 R_{ac} (x^2 + y^2 + z^2)^{1/2} \qquad (4)$$

Where, $c = (c_x^2 + c_y^2)$

Equating (3) & (4) and simplying,

$$y = g * x + h \qquad (5)$$

Where,

$g = \{R_{ac} * (b/ R_{ac}) - c_x\}/ c_y$ (6)
$h = \{c^2 - R^2_{ac} + R_{ac} * R_{ab} (1 - (b/ R_{ab})^2)\}/ 2c_y.$ (7)

Substituting (5) into (3), one obtains

$z = +/- (d*x^2+e*x+f)^{1/2}$ (8)
$z^2 = d*x^2 + e*x + f$ (9)

Where

$d = +/- \{1 - (b/ R_{ab})^2 + g^2\}$ (10)
$e = b*\{1 - (b/ R_{ab})^2\} - 2g * h$ (11)
$f = (R^2_{ab}/ 4) * \{1 - (b/ R_{ab})^2\}^2 - h^2$ (12)

$$\vec{R} = x*\vec{i} + (g*x+h)\vec{j} \pm \sqrt{d*x^2+e*x+f}\,\vec{k}. \quad (13)$$

stations A, B and C′ is, similar to (13),

$$\vec{R} = x*\vec{i} + (g'*x+h')*\vec{j}' \pm \sqrt{d'x^2+e'*x+f'}\,\vec{k}' \quad (14)$$

Taking Scalar product of (13) & (14) with unit vector j and equating, we have:

$$g'*x+h' = (g*x+h)(\vec{j}*\vec{j}') \pm \sqrt{d*x^2+e*x+f}*(\vec{k}*\vec{j}')$$

Or, squaring and simplying

$p * x^2 \; +q*x+r = 0$ (15)

Where

$$p = d*(\vec{k}*\vec{j}')^2 - \{g' - g*(\vec{j}*\vec{j}')\}^2 \quad (16)$$

$$q = e*(\vec{k}*\vec{j}') - 2\{g' - g*(\vec{j}*\vec{j}')\}\{h' - h*(\vec{j}*\vec{j}')\} \quad (17)$$

$$r = f*(\vec{k}*\vec{j}')^2 - \{h' - h*(\vec{j}*\vec{j}')\}^2. \quad (18)$$

Further solving will give the position of navigator.

4. Hardware and Software requirements

a. Hardware requirements :-

i. Microphones/Mobiles

ii. Sound Source

iii. Measuring scale

iv. Measuring tape

b. Software requirements :-

i. MATLAB

ii. Sound recorder app (Schedule Sound Recorder)

iii. Audacity

5. Activity Time Chart

Work done till Mid Semester				Work done during Mid Semester & End Semester			
Phase I	**Phase II**	**Phase III**	**Phase IV**	**Phase V**	**Phase VI**	**Phase VII**	**Phase VIII**
August 20-26	August 26-31	September 1-7	September 8-19	October 4 – 20	October 21 - 31	November 1 – 20	November 21 - 30
		Literature Survey			Obtained the results from the proposed methodolo-gy	Data Collection Improved the algorithm further	Tested and analyzed the results
	Collection of hardware requireme-nts	Experimental Setup	Data Collection and analyzing the format of the data	Data Collection and start Implementation (Coding)			

6. Work Done Till Mid-Sem

Literature Survey: Gone through many research papers and surveyed the various techniques to:

- Different ways of data collection and experimental setups
- reduce noise from the data and process it
- find the location of the acoustic source (geometrically)

Data Collection:

- 3 microphones in a triangular fashion

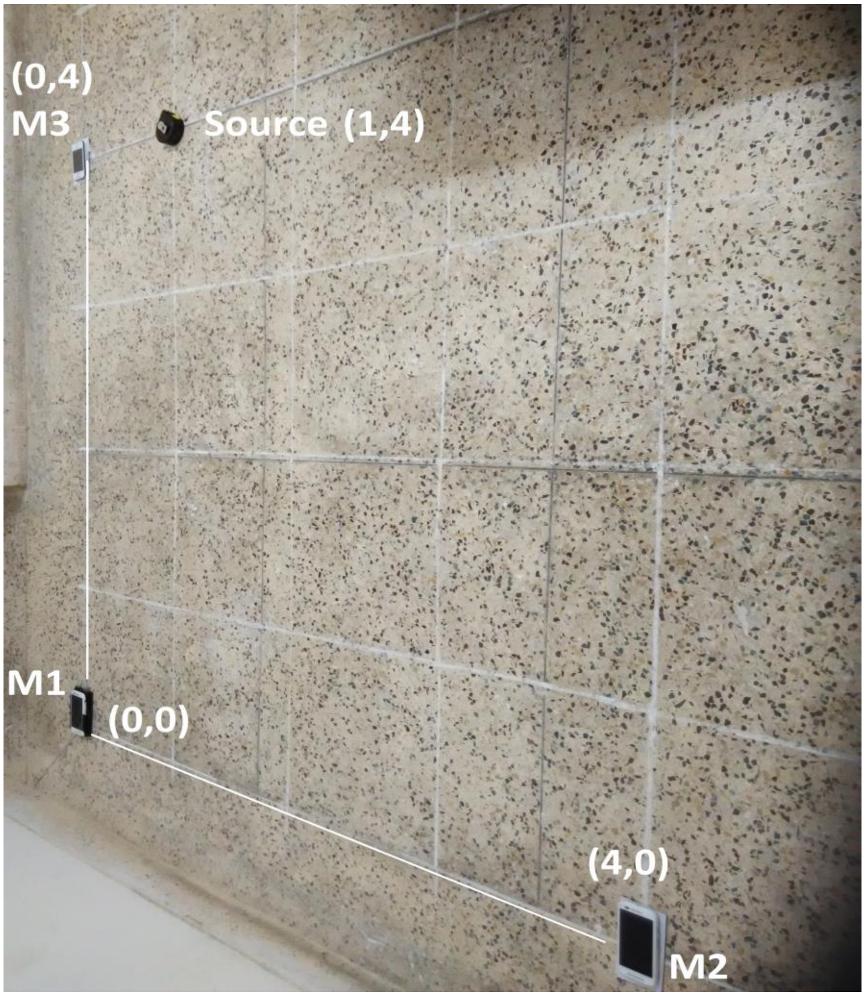


Figure 6.1: Experimental setup with 3 microphones arranged in L-shape and the source have to be located.

- 4 microphones in a square fashion

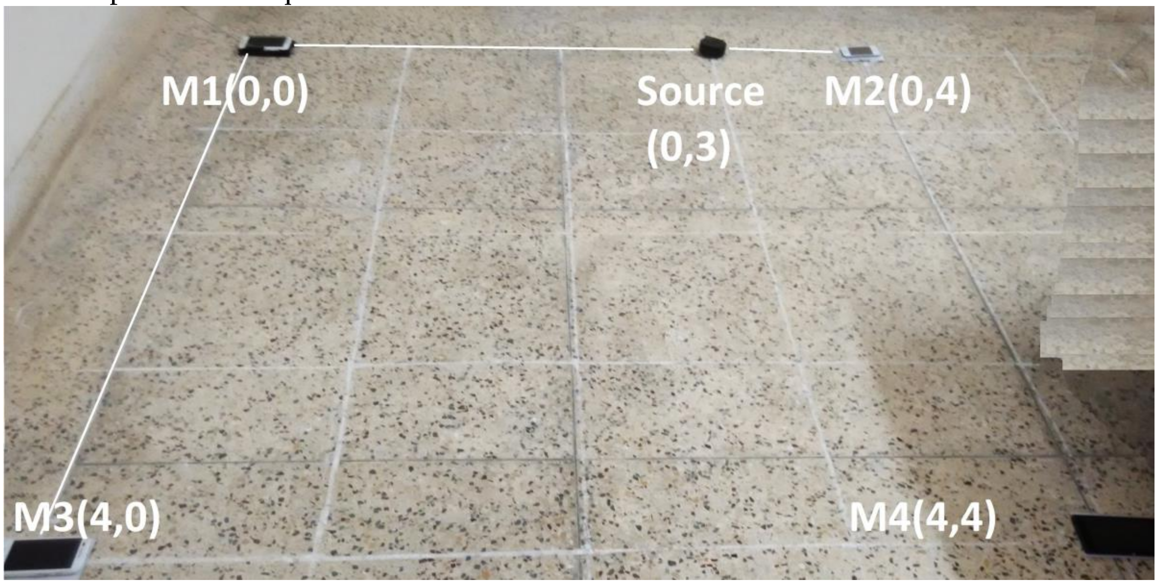


Figure 6.2: Experimental setup with 4 microphones placed in each corner of a square and the source have to be located.

- 4 microphones in a rectangular fashion

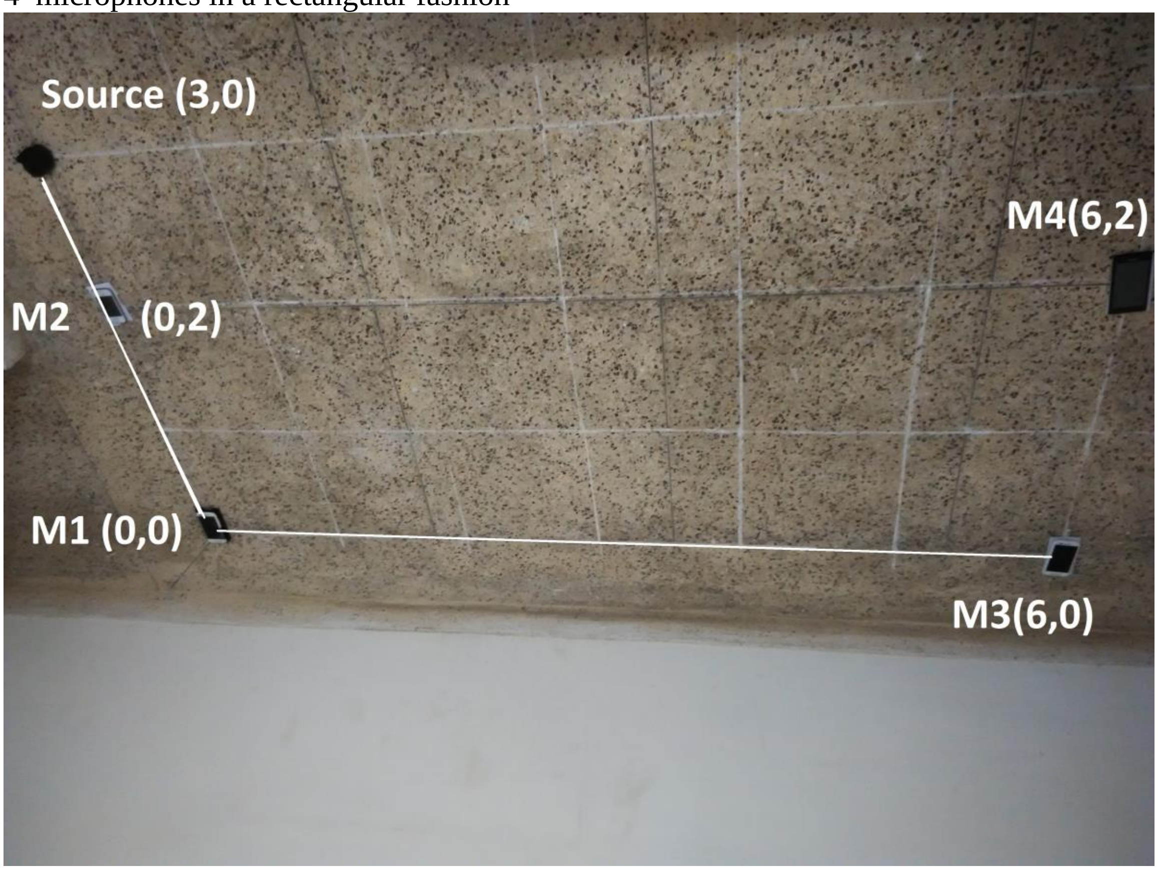


Figure6. 3: Experimental setup with 4 microphones placed in each corner of a rectangle and the source have to be located.

- 4 microphones in a triangular (3 microphones on hypotenous) fashion

Figure6. 4: Experimental setup with 4 microphones placed in each corner of a triangle and one at the middle of the hypotenous and the source have to be located.

- 4 microphones in a trapezium fashion

Figure 6.5: Experimental setup with 4 microphones placed in each corner of a trapezium and the source have to be located.

7. Work done for End-Sem

i. **Literature Survey:** To find TDE techniques.

ii. **Proposed Methodology**: We need to find a methodology based on the Literature Survey done.

iii. **Data Collection and experimental setup**

iv. **Implementation of algorithm for 2-D plane:** The methodology for 2-D will be based on TDE techniques. It is a 2 step process
- First we process the data, reduce the noise and find the TDOA (Time Delay Of Arrival between the pair of microphones).
- Using the obtained TDOA's and geometry, the acoustic source would be located.

v. **Test, Verification and Improvisation** of the algorithm on the basis of experiments.

7.1 Proposed Methodology:

Methodology 1: for 4 microphones
Further from Chan's theorem and equation no. (vii), we have

$$R_{i,1}{}^2 + 2R_{i,1}R_1 = X_i{}^2 + Y_i{}^2 - 2X_{i,1}x - 2Y_{i,1}\,y - X_1{}^2 - Y_1{}^2$$

$$R_{i,1}{}^2 = X_i{}^2 + Y_i{}^2 - 2X_{i,1}x - 2Y_{i,1}\,y - X_1{}^2 - Y_1{}^2 - 2R_{i,1}R_1$$

$$2X_{i,1}x + 2Y_{i,1}y + 2R_{i,1}R_1 = X_i{}^2 + Y_i{}^2 - R_{i,1}{}^2 \qquad \text{- (a)}$$

For four microphones, equation (a) can be written in the form of a matrix, as:

$$\begin{bmatrix} X_{2,1} & Y_{2,1} & R_{2,1} \\ X_{3,1} & Y_{3,1} & R_{3,1} \\ X_{4,1} & Y_{4,1} & R_{4,1} \end{bmatrix} \begin{bmatrix} x \\ y \\ R_1 \end{bmatrix} = \begin{bmatrix} X_2{}^2 + Y_2{}^2 - X_1{}^2 - Y_1{}^2 - c^2 T_{2,1}{}^2 \\ X_3{}^2 + Y_3{}^2 - X_1{}^2 - Y_1{}^2 - c^2 T_{3,1}{}^2 \\ X_4{}^2 + Y_4{}^2 - X_1{}^2 - Y_1{}^2 - c^2 T_{4,1}{}^2 \end{bmatrix} \qquad \text{- (b)}$$

Comparing (b) with A*X = B

We can solve for $X=[x\ y\ R_1]^T$ as

$$X = (A^T A)^{-1} A^T B$$

Methodology 2: for 3 microphones

SX = U - R_1P - (c)

Comparing with Equation (vii),

$$\begin{bmatrix} X_{2,1} & Y_{2,1} \\ X_{3,1} & Y_{3,1} \end{bmatrix} \begin{bmatrix} x \\ y \end{bmatrix} = -\frac{1}{2} \begin{bmatrix} {R_{2,1}}^2 - K_2 + K_1 \\ {R_{3,1}}^2 - K_3 + K_1 \end{bmatrix} - R_1 \begin{bmatrix} R_{2,1} \\ R_{3,1} \end{bmatrix}$$

We have:

$$S = \begin{bmatrix} X_{2,1} & Y_{2,1} \\ X_{3,1} & Y_{3,1} \end{bmatrix}$$

$$U = -\frac{1}{2} \begin{bmatrix} {R_{2,1}}^2 - K_2 + K_1 \\ {R_{3,1}}^2 - K_3 + K_1 \end{bmatrix}$$

$$P = \begin{bmatrix} R_{2,1} \\ R_{3,1} \end{bmatrix}$$

$$X = \begin{bmatrix} x \\ y \end{bmatrix}$$

Let’s take matrix D as,

$$D = [dia\ \{P\}]^{-1}$$

$$= \begin{bmatrix} R_{2,1} & 0 \\ 0 & R_{3,1} \end{bmatrix}^{-1}$$

Now multiplying the equation with matrix D,

$$D*SX = DU - R_1DP$$

Here,

$$D*P = \begin{bmatrix} 1 \\ 1 \end{bmatrix}$$

So, we will neglect the term of R_1

Hence,

$$DSX = DU$$

$$X = (DS)^{-1}DU$$

7.2 Data Collection and Experimental setup for 4 Microphones

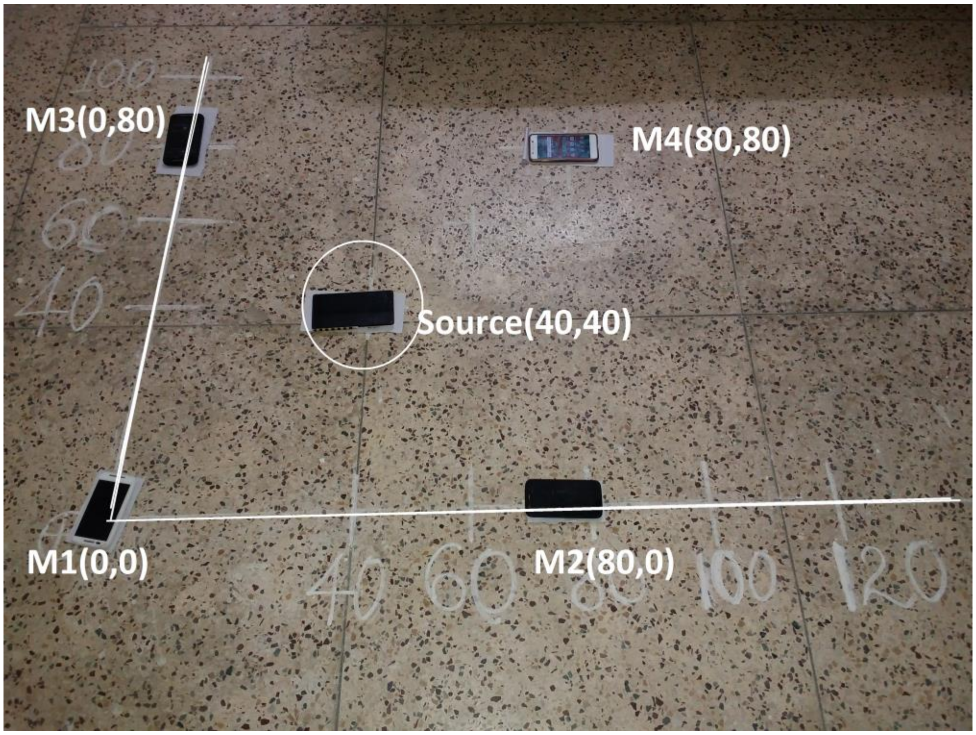
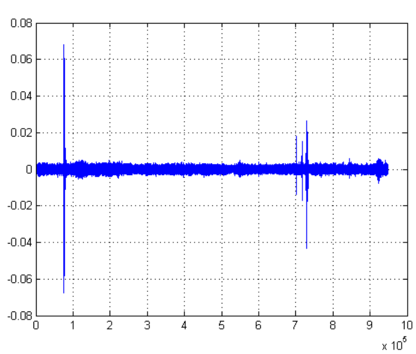


Figure 7.2.1: Experimental setup with 4 microphones where the source is placed at (40, 40) and the microphones at (0, 0), (0, 80), (80, 0) and (80,80)

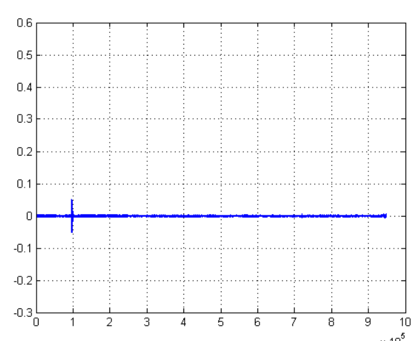
7.3 Results (For 4 microphones)

SAMPLE – 1

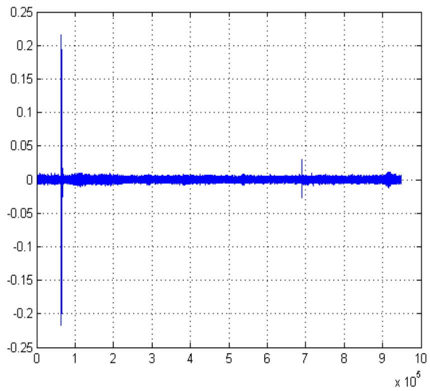
The following are the signal plots for the captured sound from the source where the source is placed at (40, 40) and the microphones at (0, 0), (0, 80) and (80, 0).



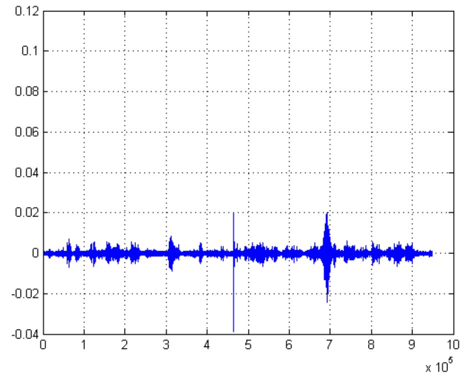
a. Original Signal observed at Microphone1



b. Original Signal observed at Microphone2

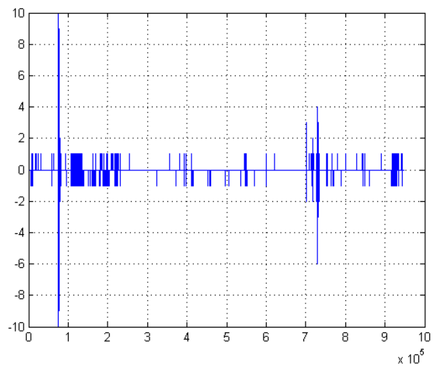


c. Original Signal observed at Microphone3

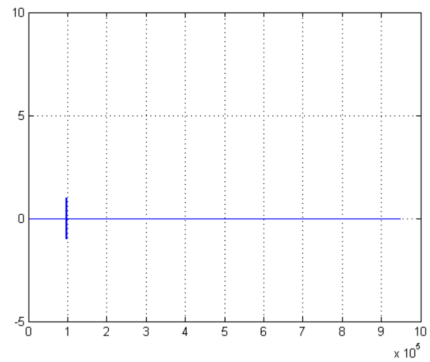


d. Original Signal observed at Microphone4

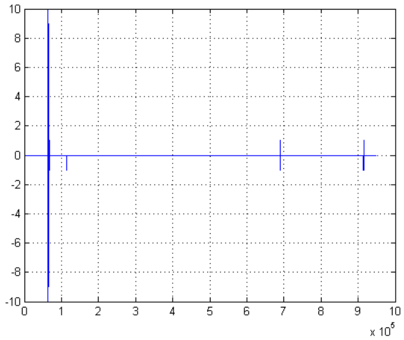
The following are the histogram signal plots for the captured sound from the source where the source is placed at (40, 40) and the microphones at (0, 0), (0, 80) and (80, 0).



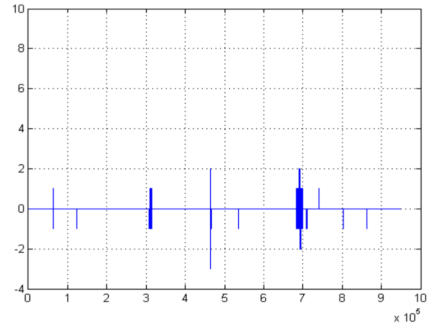
a. Histogram of the signal observed at microphone1



b. Histogram of the signal observed at microphone2



c. Histogram of the signal observed at microphone3



d. Histogram of the signal observed at microphone4

Observations:

The coordinates of the source are (40, 40);
Observed coordinates of the source are (97, 38)

Table 7.3.1: Shows the coordinates of the actual data, observed data and the microphones

SampleNo.	Source(x,y)	Observed(x,y)	M1(x,y)	M2(x,y)	M3(x,y)	M4(x,y)
1	(40,40)	(97,38)	(0,0)	(0,80)	(80,0)	(160,0)
2	(200,40)	(-50,83)	(0,0)	(80,0)	(0,80)	(80,80)
3	(60,40)	(24,82)	(0,0)	(80,0)	(0,80)	(80,80)

Here

M1 = Microphone1
M2 = Microphone2
M3 = Microphone3
M4 = Microphone4

Table 7.3.2: Shows the Time delays of the various microphones for different samples

Sample No.	T1	T2	T3	T4
1	74101	73698	62721	73698
2	55460	64300	56281	52323
3	115021	115423	112250	111121

Here

T1 = Time delay from the source to the microphone1
T2 = Time delay from the source to the microphone2
T3 = Time delay from the source to the microphone3
T4 = Time delay from the source to the microphone4

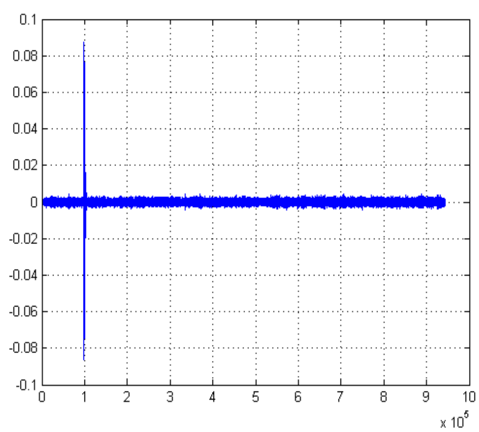
7.4 Data Collection and Experimental setup for 3 Microphones

Figure 7.4.1: Experimental setup with 4 microphones where the source is placed at (40, 40) and the microphones at (0, 0), (0, 80) and (80, 0)

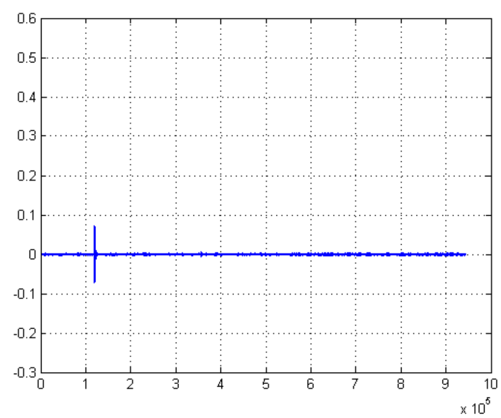
7.5 Results (For 3 microphones)

SAMPLE – 1

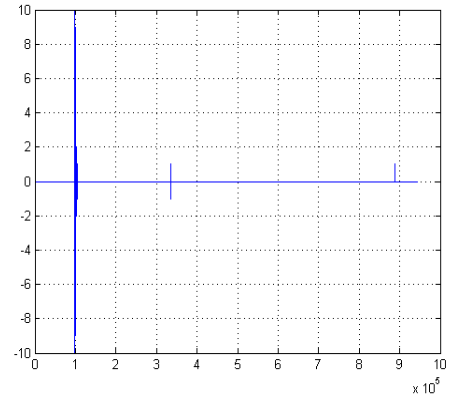
The following are the signal plots for the captured sound from the source where the source is placed at (40, 40) and the microphones at (0, 0), (0, 80) and (80, 0).



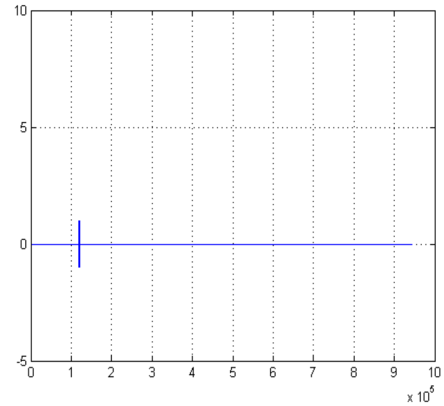
a. Original Signal observed at Microphone1



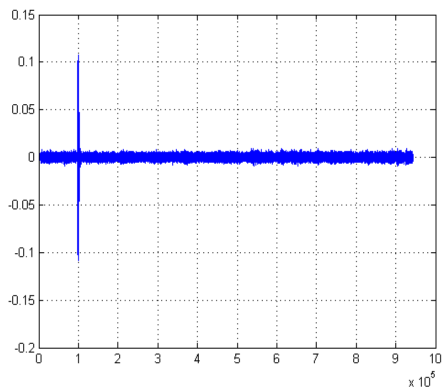
b. Original Signal observed at Microphone2



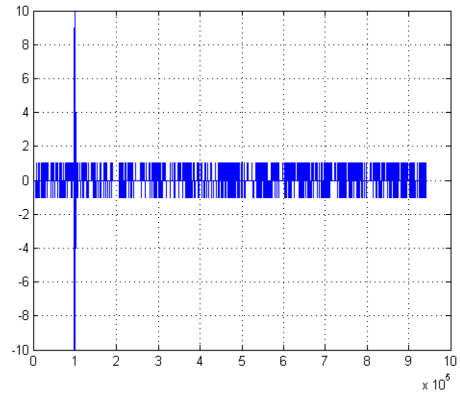
a. Histogram of the signal observed at Microphone1



b. Histogram of the signal observed at Microphone2



c. Original Signal observed at Microphone3



c. Histogram of the signal observed at Microphone3

Observations:

The coordinates of the source are (0, 60);

Observed coordinates of the source are (31, 27)

Table 7.5.1: The coordinates of the actual source position, observed source position and the microphones

SampleNo.	Source(x,y)	Observed(x,y)	M1(x,y)	M2(x,y)	M3(x,y)
1	(60,40)	(30,-27)	(0,0)	(80,0)	(80,80)
2	(0,60)	(31,27)	(0,0)	(0,80)	(80,0)
3	(60,40)	(27,34)	(0,0)	(80,0)	(80,80)

Table 7.5.2: Time at which signal was observed at the various microphones for different samples

Sample No.	T1	T2	T3
1	115021	115423	111121
2	96981	96743	97841
3	234221	235041	233761

8. Conclusion

Hence, we performed two proposed algorithm for sound source localization. The result observed via the methodology is nearby the actual position of the source.

9. Future Work

i. Proposed methodology can be further improved to find the TDOA (Time Delay of Arrival) values for estimation of the exact coordinates of the source.

ii. Proposed methodology can be further extended for continuous signals. In this project, only single sound beep is taken for experiment.

iii. Proposed methodology can be further extended for the moving source.

iv. Proposed methodology can be extended for the localization of the acoustic sound source in a three-dimensional space.

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Suggestions by Board Members