Acoustic Source Localization

(End-Sem Presentation)

Under the guidance of Prof.Vrijendra Singh

Juhi Kumari (IIT2013153)
Pedakam Sneha (IIT2013176)
Vangapalli Tejasree (IIT2013203)

Contents..

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 and the signal is captured by each of the acoustic sensors.
- The Time Difference Of Arrival (TDOA) is estimated from the captured audio signals.
- We now need to locate the coordinates of the source using the data provided.

Motivation

There are many applications of the sound source localization in daily life :-

- ❑ Teleconferencing
- ❑ Location of a dominant speaker in an auditorium
- ❑ Human-Robot interaction
- ❑ Handicappers aid
- ❑ Surveillance

Objective and Targets

- Our goal is to locate the acoustic source in 2-D plane using microphone arrays.

- We want to improve the performance of the source localization using minimum number of microphones which reduces the size and cost involved in time-delay estimation of acoustic source.

Description of Hardware and Software

Hardware requirements :-

- ➔ Microphones/ Mobiles
- ➔ Sound Source
- ➔ Measuring scale and tape

Software requirements :-

- ➔ Matlab
- ➔ Schedule Voice Recorder app
- ➔ Audacity

Work done till date ...

Chan's Method [7]

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

$$= \sqrt{(X_i)^2 + (Y_i)^2 - 2X_i x - 2Y_i y + x^2 + y^2} \qquad \text{- (1)}$$

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

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

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

$$R_{i,1} = R_i - R_1 \qquad \text{- (3)}$$

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

$$R_{i,1}^2 + 2R_{i,1}R_1 + R_1^2 = X_i^2 + Y_i^2 - 2X_i x - 2Y_i y + x^2 + y^2 \qquad \text{- (4)}$$

Eq 1 for i =1

$$R_1 = \sqrt{X_1{}^2 + Y_1{}^2 - 2X_1x - 2Y_1y + x^2 + y^2} \qquad - (5)$$

Putting equation (5) into equation (4)

$$R_{i,1}{}^2 + 2R_{i,1}R_1 + X_1{}^2 + Y_1{}^2 - 2X_1x - 2Y_1y + x^2 + y^2 = X_i{}^2 + Y_i{}^2 - 2X_ix - 2Y_iy + x^2 + y^2$$

$$\begin{aligned} R_{i,1}{}^2 + 2R_{i,1}R_1 &= X_i{}^2 + Y_i{}^2 - X_1{}^2 - Y_1{}^2 - 2(X_i - X_1)x - 2(Y_i - y_1)y \\ &= X_i{}^2 + Y_i{}^2 - 2X_{i,1}\mathrm{x} - 2Y_{i,1}y - X_1{}^2 - Y_1{}^2 \end{aligned} \qquad - (6)$$

$$\begin{bmatrix} X_{2,1} & Y_{2,1} \\ X_{3,1} & Y_{3,1} \end{bmatrix} \begin{bmatrix} x \\ y \end{bmatrix} = -\frac{1}{2} \begin{bmatrix} R_{2,1}{}^2 - K_2 + K_1 \\ R_{3,1}{}^2 - K_3 + K_1 \end{bmatrix} - R_1 \begin{bmatrix} R_{2,1} \\ R_{3,1} \end{bmatrix} \qquad - (7)$$

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

Proposed part

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

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

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

$$\begin{bmatrix} X_{2,1} & Y_{2,1} & R_{2,1} \\ X_{3,1} & Y_{3,1} & R_{3,1} \\ X_{4,1} & Y_{4,1} & R_{4,1} \end{bmatrix} \begin{bmatrix} x \\ y \\ R_1 \end{bmatrix} = \begin{bmatrix} X_2^{\ 2} + Y_2^{\ 2} - X_1^{\ 2} - Y_1^{\ 2} - c^2 T_{2,1}^{\ \ 2} \\ X_3^{\ 2} + Y_3^{\ 2} - X_1^{\ 2} - Y_1^{\ 2} - c^2 T_{3,1}^{\ \ 2} \\ X_4^{\ 2} + Y_4^{\ 2} - X_1^{\ 2} - Y_1^{\ 2} - c^2 T_{4,1}^{\ \ 2} \end{bmatrix} \qquad - (8)$$


M3(0,80)
S(40,40)
M2(80,0)
M1(0,0)

Proposed Methodology

Step1: First filter and normalize the signal via making histogram.

Step 2: Take the

P <- $\log_{10}$ max(signal)

Signal <- signal/(10^(p-1))

Now we are having all the signals with amplitude in the range of [-10, 10]

Step 3: Now take the round of the sample and quantize them.

Step 4: now the signal is in the form of histogram or quantized form

Step 5: Use Sliding Window concept to find the time at which the signal was observed at various microphones.

- ❏ Take a window size, and sliding size.
- ❏ Window size: 500 and sliding size: 20
- ❏ If the amplitude for a particular window is greater than the threshold, then this is the required value which is set accordingly.

Proposed Method

$$SX = U - R_1 P \qquad \text{- (a)}$$

Comparing with Chan's equation

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

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,

$$D^*SX = DU - DR_1P$$

$$D^*S^*X = DU - R_1DP$$

Here,

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

So, we will neglect the term of R_1

Hence,

$$DSX = DU$$

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

Results and Analysis for 4 Microphones

Sample No.	M1 (x,y)	M2 (x,y)	M3 (x,y)	M4 (x,y)	Source (x,y)	Observed (x,y)
1	(0,0)	(80,0)	(0,80)	(80,80)	(60,40)	(24,82)
2	(0,0)	(80,0)	(0,80)	(80,80)	(200,40)	(-50,83)
3	(0,0)	(0,80)	(80,0)	(160,0)	(40,40)	(97,38)

Table 1: Shows the coordinates of the various microphones, source and observed values for different samples.

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

Experimental Setup with 4 Microphones

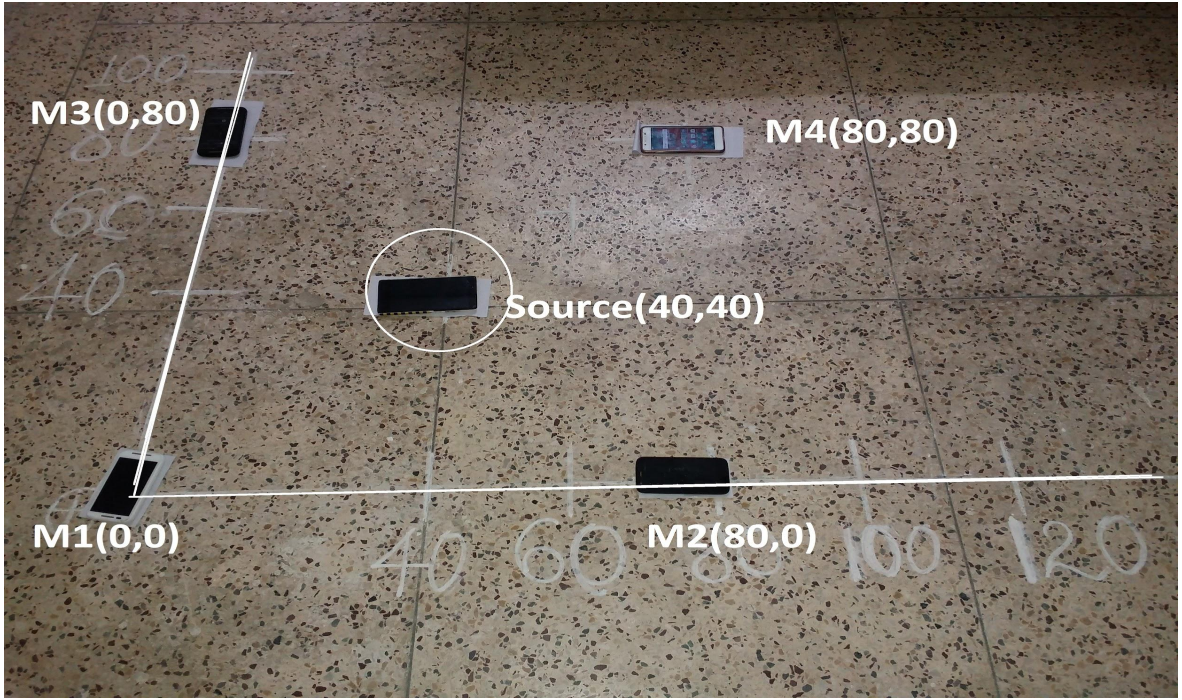
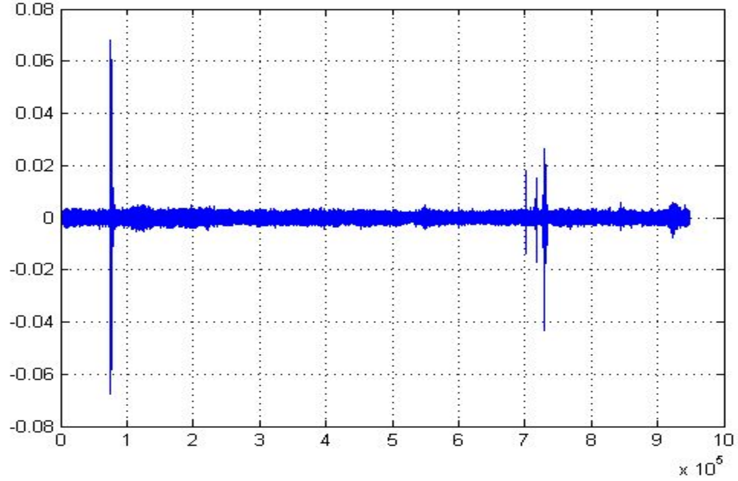
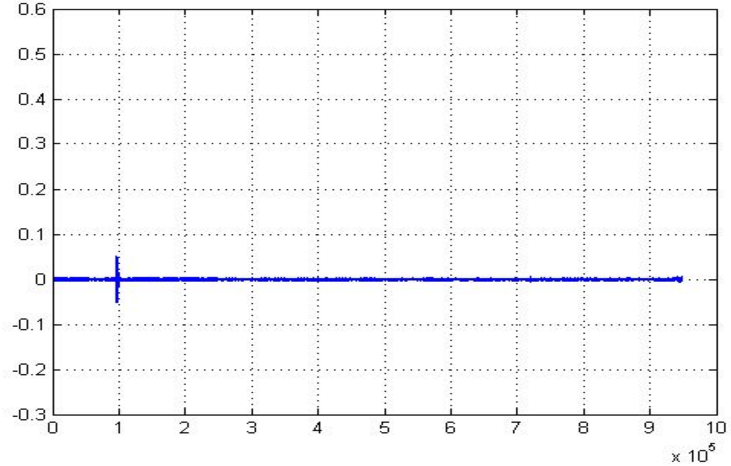


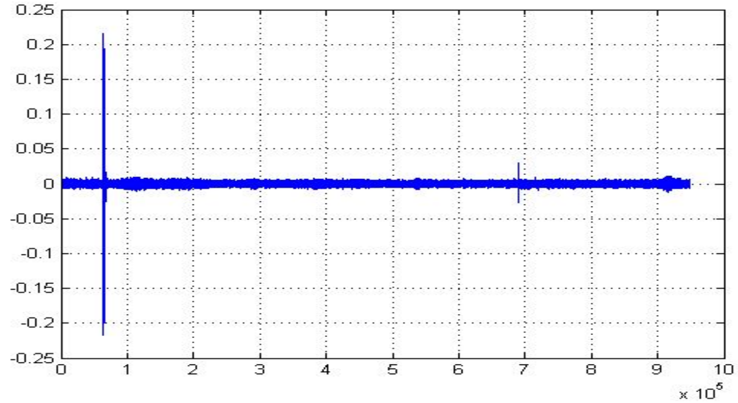
Figure 1 : Experimental setup where microphones are placed in the coordinates and the source have to be located.



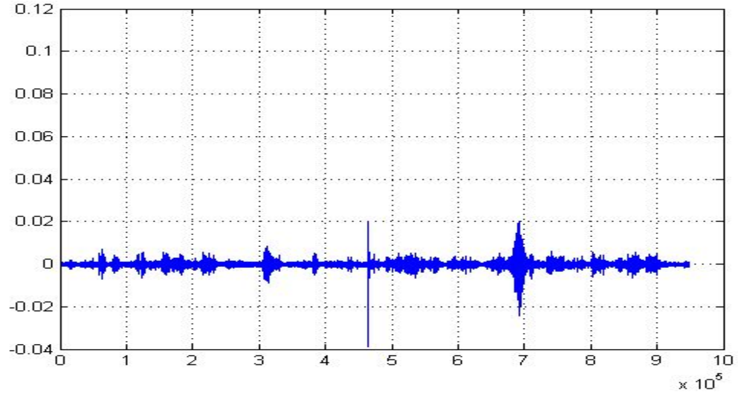
Original signal of microphone 1



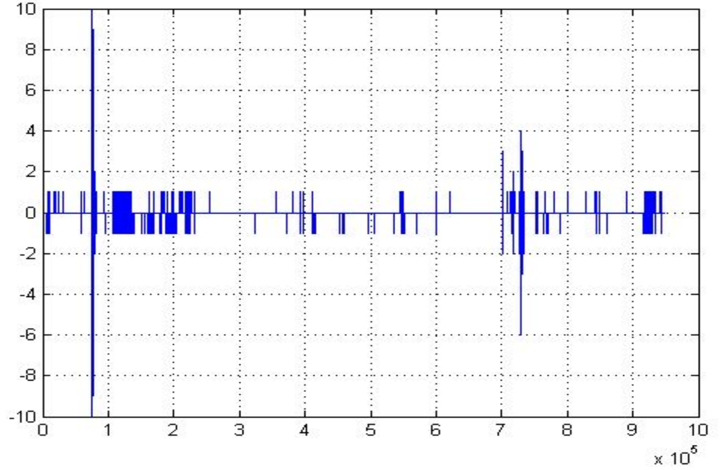
Original signal of microphone 2



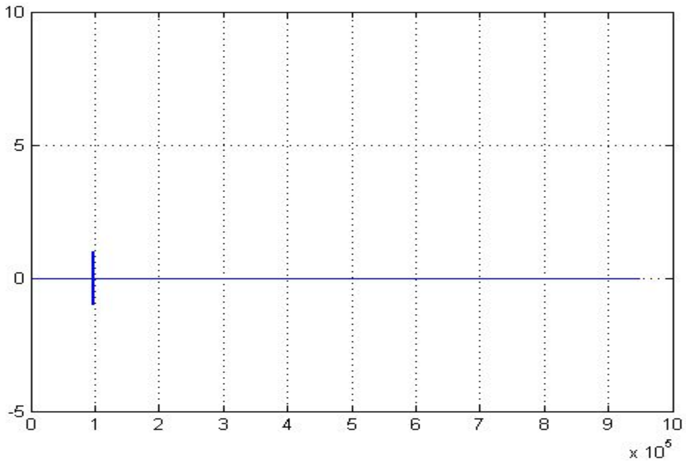
Original signal of microphone 3



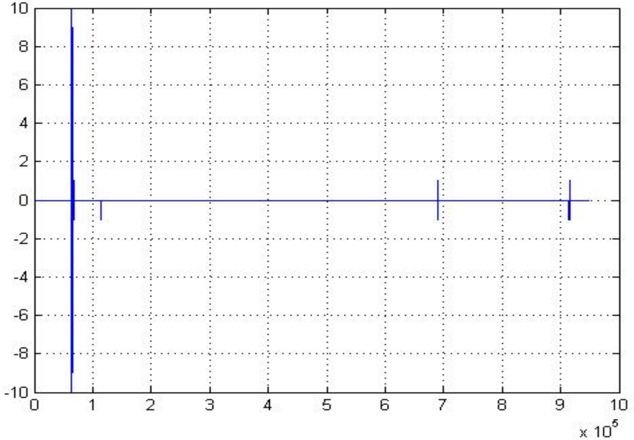
Original signal of microphone 4



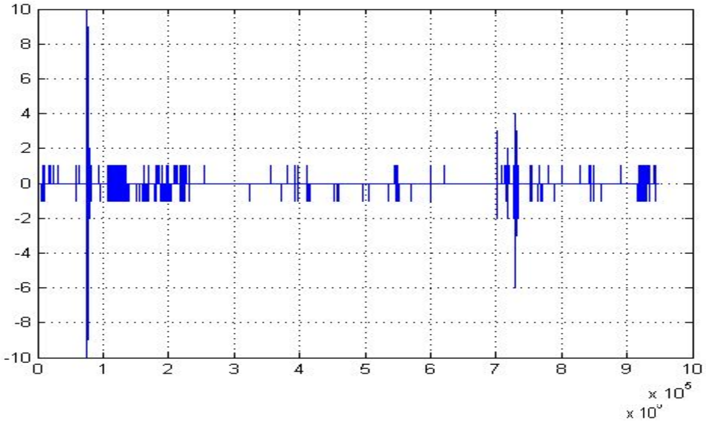
Histogram of microphone 1



Histogram of microphone 2



Histogram of microphone 3



Histogram of microphone 4

Results and Analysis for 3 Microphones

Sample No.	M1(x,y)	M2(x,y)	M3(x,y)	Source(x,y)	Observed(x,y)
1	(0,0)	(80,0)	(80,80)	(60,40)	(30,-27)
2	(0,0)	(0,80)	(80,0)	(0,60)	(31,27)
3	(0,0)	(80,0)	(80,80)	(60,40)	(27,34)

Table 2: Shows the coordinates of the various microphones, source and observed values for different samples.

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

Experimental Setup with 3 Microphones

Data Collection :

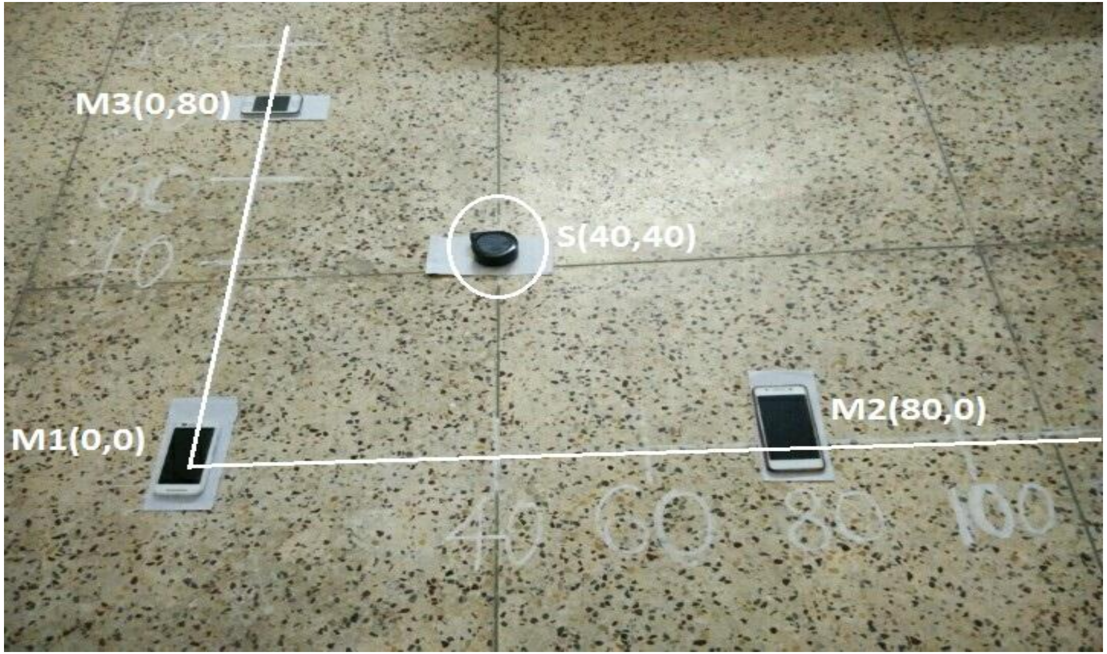
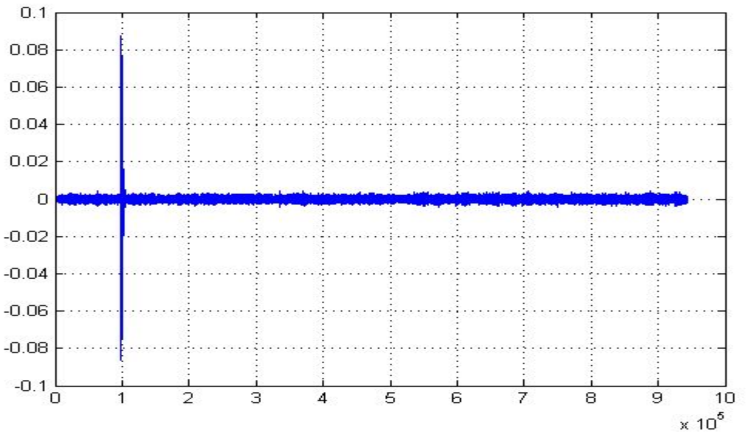
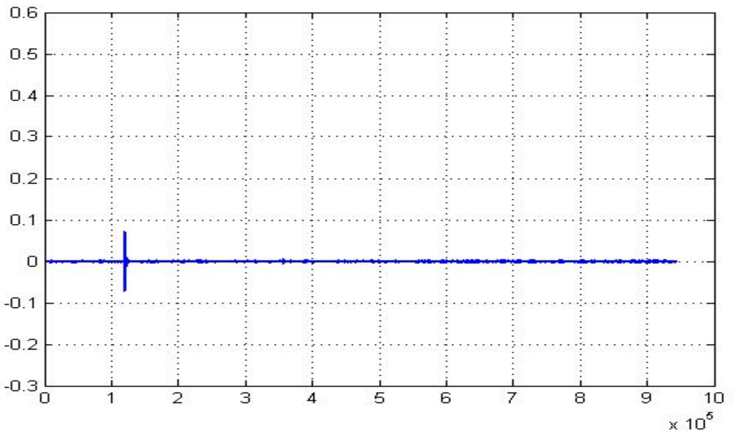


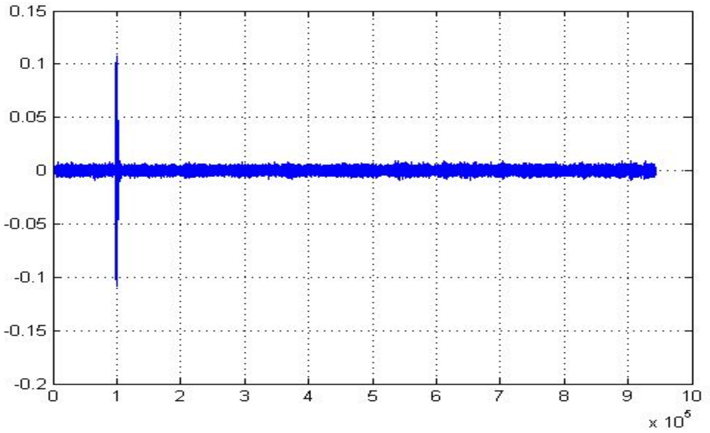
Figure 2 : Experimental setup where microphones are arranged in the coordinates and the source have to be located.



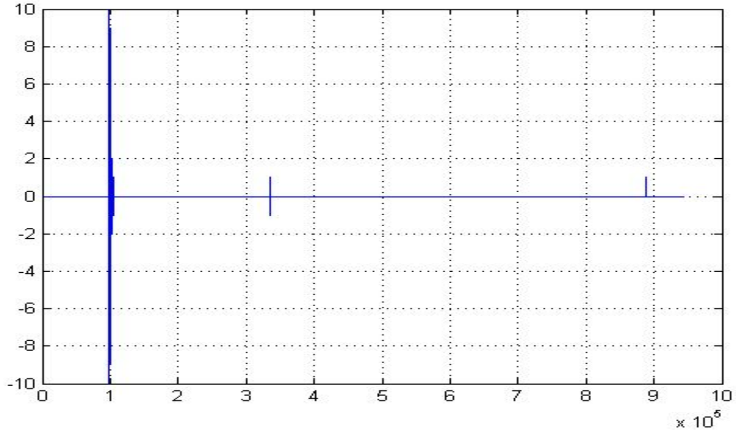
Original signal of microphone 1



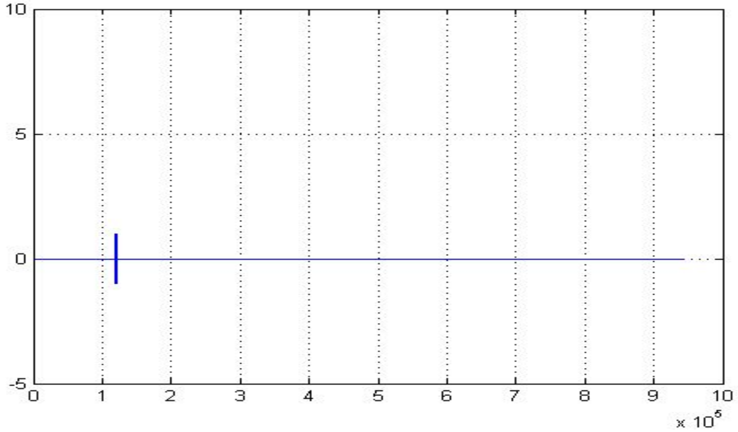
Original signal of microphone 2



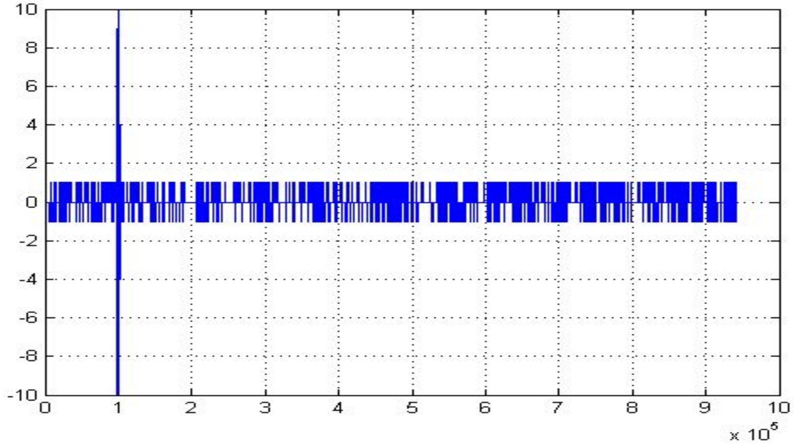
Original signal of microphone 3



Histogram of microphone 1



Histogram of microphone 2



Histogram of microphone 3

References

[1] Ashok Kumar Tellakula, “Acoustic Source Localization Using Time Delay Estimation”, Thesis for Master of Science (IISC Bangalore), 2007.

[2] Ashok Kumar and Maninder Pal, “Acoustic Source Localisation System using Microphone Arrays”, International Journal of Advanced Research in Computer Science and Software Engineering, Volume 2, Issue 11, November 2012.

[3] YueKan, Pengfei Wang, Fusheng Zha, Mantian Li, Wa Gao and Bao yuSong, “Passive Acoustic Source Localization at a Low Sampling Rate Based on a Five-Element Cross Microphone Array”, Journal by Open Access Sensors ISSN 1424-8220, 2015.

[4] Sumit Kumar, Dr. Rajat Gupta, “Estimating Time Delay using GCC for Speech Source Localisation”, International Journal of Application or Innovation in Engineering & Management (IJAIEM), Volume 2, Issue 5, May 2013.

[5] Jinghong Li, Shen Gao and Jianming Li, “Design and Implementation of Sound Source Localization System Based on FPGA”, International Conference on Systems and Informatics (ICSAI), 2012.

[6] Cha Zhang, Dinei Florêncio, Demba E. Ba, and Zhengyou Zhang, “Maximum Likelihood Sound Source Localization and Beamforming for Directional Microphone Arrays in Distributed Meetings”, IEEE TRANSACTIONS ON MULTIMEDIA, VOL. 10, NO. 3, APRIL 2008.

[7] Y.T.Chan, “A simple and Efficient Estimator for Hyperbolic Location”, IEEE Transaction on Signal Processing, vol. 42, no. 8, pp. 1905–1915, August 1994.

[8] Don J Torrieri, “Statistical Theory of Passive Location Systems”, IEEE Transactions on Aerospace and Electronic Systems Vol. AES-20(2):183 - 198 · April 1984.

[9] B. T. Fang, “Simple solutions for hyperbolic and related position fixes,” IEEE Trans. on Aerospace Electron. Systems, vol. 26, no. 5, pp. 748–753, September 1990.

[10] Foy, W.H., “Position Location Solutions by Taylor Series Estimation,” IEEE Trans. on Aerospace Electronics Systems 12(2), 187-194, 1976.

[11] Friedlander B. “A passive localization algorithm and its accuracy analysis” [J]. IEEE Journal Oceanic Engineering, 1987, 12 (1): 234-245.

[12] Jacek Stefanski, “Hyperbolic Position Location Estimation in the Multipath Propagation Environment”, Thesis by Gdansk University of Technology, 2009.