International Journal of Applied Engineering Research
ISSN 0973-4562 Volume 10, Number 10 (2015) pp. 25547-25562

http://www.ripublication.com

Desired Sound Source Localisation Using Advanced Signal Processing Techniques in Multiple Sources

D. Lakshmi Srinivas[1], P. Satyanarayana [2], V.S. Bhagavan[3]

[1, 2]Department of E.C.E
[3]Department of Mathematics
K L University
(Koneru Lakshmaiah Education Foundation), AP, India.

Abstract

This paper provides a basic method of finding the direction of arrival of sound signals coming from different sound sources. The Sound source localisation is done in two steps; one is of Time Difference of Arrival (TDOA) and second is of Direction of Arrival (DOA). TDOA is obtained using generalised cross correlation method where, the correlated signal peaks at the delay between the sensor signals. From the obtained TDOA the near field and far field condition of the sources is done and corresponding favourable algorithms like Triangulation, Beamforming, MUSIC algorithms are used to find the DOA of particular sound source signals. The comparison techniques are done for Beamforming and MUSIC algorithm and various factors effecting TDOA and DOA are also discussed.

Keywords:- Coherence, GCC, PHAT, TDOA, DOA, Triangulation, Beamforming, MUSIC.

1. INTRODUCTION

As the microphone array processing took light in modern days, source localization has become one of the most important issues in various applications. The Source localisation helps in various applications like video conferencing to steer the cameras, in industries to find faults in working of machinery and in aero acoustics for suppressing the huge amount of sound levels during lift off. In order to localise various sound sources which are acting simultaneously in medium one has to know the spectral characteristics of the incidental acoustical energy on the sensors also. This provides another insight for finding exact sound sources. As the microphone array signal processing is expanding its growth in various areas, many signal processing

algorithms like Generalised Cross Correlation (GCC), Phase Transformation method (PHAT), Smoothed coherence Transform are used for finding TDOA and Beamforming, MUSIC, Root MUSIC and ESPIRIT are used in finding DOA for sound source localisation.

From the basic definitions of Sound near field region is nearer to the source of about wavelength of the sound source signal or three times the largest dimension of the source and inverse square law is not applicable[1], [2]. Far field region is the region end of near field region to infinity where inverse square law is applicable where 6dB reduction for twice the distance and sound source acts as point source[3]. For near field simulation, source is assumed to lie near to the sensor array where the distance is less than the maximum wavelength of source signal. The output of each microphone is time delayed version to the reference microphone. Consider *d* as the distance between two microphones, θ_n and r_n indicating the DOA of sound wave and radial distance from the source to sensor respectively. Let us consider a linear array of three sensors, source is near to the sensor 1.In this near field condition, Sound waves fall on the sensors as spherical wave front and have directivity. The microphones are separated by distance *d* from each other as shown in the fig.1.

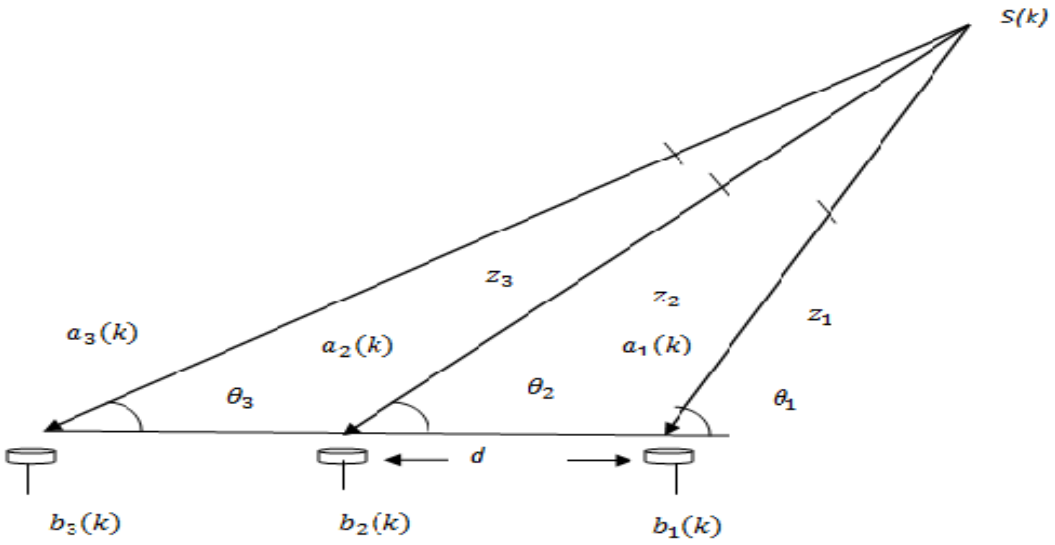


Fig.1 Sound Source in Near Field Region.

In case of far field condition, the sound waves are received as plane wave front as shown in the fig.2

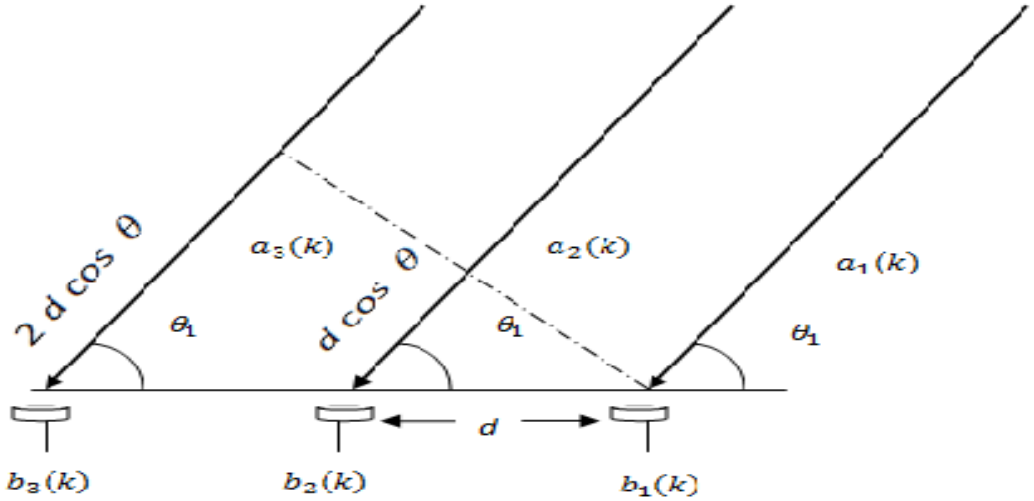


Fig.2 Sound Source in Far field Region.

2.0 Computation of Time Difference of Arrival (TDOA)

Among various signal processing methods like generalised cross correlation (GCC), phase transform (PHAT), Smoothed Coherence Transform, we use generalised cross correlation method to find the Time Difference of Arrival of sound wave among various sensors[4]. Cross correlation method peaks at the time difference of arrival between the two sensor signals[5]. Cross correlation method is employed because of less computation time and ease of implementation. In aero acoustics also cross correlation method yields good results than PHAT, Smoothed coherence transform. Considering the microphone array, two sensors output signals $b_1(k)$ and $b_2(k)$ from the fig.1, 2, the cross correlation expression is given as

$$R_{b_1b_2} = \mathrm{E}\,[b_1(k)\; b_2(\mathrm{k} + \mathrm{p})] \qquad - \qquad (1)$$

The function $\xi_{b_1b_2}(f)$ denotes the generalised cross spectrum of the sensor signals where, it is equal to the product of cross spectrum of sensor signal $\chi_{b_1b_2}(f)$ and the frequency domain weighting function Δ (f). The expression for generalised cross spectrum is given by

$$\xi_{b_1b_2}(f) = \Delta\,(\mathrm{f})\; \chi_{b_1b_2}(f) \qquad - \qquad (2)$$

If the frequency domain weighting function Δ (f) = 1, it is equivalent to the cross spectrum method, if Δ (f) $= \frac{1}{\sqrt{E[Y1(f)|^2 E[Y2\,(f)|^2]}}$, it results in smoothed coherence transform and if Δ(f) $= \frac{1}{|\theta_{y_1y_2}(f)|}$, it is called PHAT method where the output of the PHAT peaks at the TDOA of sound wave.

For near field source as in fig.1, the TDOA of sound wave between the sensors

$b_1$and b_2 is given as

$$\tau_{b_1 b_2} = \frac{z_2 - z_1}{c} \quad - \quad (3)$$

And the TDOA of sound wave between the sensors b_1 and b_3 is given as

$$\tau_{b_1 b_3} = \frac{z_3 - z_1}{c} \quad - \quad (4)$$

C indicates the speed of sound in air which is usually 340m/sec at 20^{o}c and 1 atmosphere. As the temperature of the medium increases speed of sound wave increases and is given as

$$C = 332 + 0.6\ T_c \quad - \quad (5)$$

T_c refers to temperature in Centigrade's.

For far field condition as in fig.2, where the sensors are arranged in linear fashion, the delay between the reference sensors (generally first sensor near to source is taken) and the n^{th} sensor is given as

$$\tau_{b_1 b_n} = (n-1)\ \tau \quad - \quad (6)$$

Where $\tau = \frac{d \cos\theta}{c}$. τ is the delay of sound wave between the reference and the adjacent sensors. From the fig.2, we can clearly see that the extra distance travelled by the sound wave to reach $b_2$sensor compared to $b_1$sensor is of d cos θ. The extra distance travelled by the sound wave to reach $b_3$sensor compared with $b_1$sensor is 2 d cos θ. The extra distance travelled for the n^{th} sensor compared to $b_1$sensor is (n-1) d cos θ. As the extra distance required to travel is of integer multiple, the corresponding TDOA's are also integer multiple which is given in (6).

2.1 Computation of Direction of Arrival (DOA) in Near Field

The DOA calculation in near field can be estimated using the triangulation method which is usually applied in GPS positioning. The triangulation [6] method is used to localise the sound sources in near field. It gives both the Direction of arrival (DOA) and the radial distance of source from the sensors which are given in the fig.1. From the fig.1, applying cosine rule, it is easy to derive two equations which are given as

$$z_2^2 = z_1^2 + d^2 + 2\ z_1\ d\ \text{Cos}\ (\theta_1) \quad - \quad (7)$$

$$z_3^2 = z_1^2 + 4d^2 + 4\ z_1\ d\ \text{Cos}\ (\theta_1) \quad - \quad (8)$$

By solving the four equations (4), (5), (7), (8) in terms of known variables d, $\tau_{b_1 b_2}$, $\tau_{b_1 b_3}$, θ_1 can be derived easily.

2.2 Computation of Direction of Arrival (DOA) in Far Field

Direction of Arrival in far field is calculated using Delay and Sum Beam former, MVDR Beam former, MUSIC, Root MUSIC and ESPIRIT algorithms usually[6], [7], [8]. In this paper, Delay and Sum Beam former and MUSIC algorithms are used to find DOA when source is in far field. Beam forming is usually done in time domain and frequency domain[8]. In time domain, the respective delayed sensor signals w.r.t reference sensor are shifted back, weighted and summed up. The output of the summation peaks at the Direction of Arrival of sound wave. This provides high signal to noise ratio also. In case of frequency domain beam forming, the time difference of arrival in time domain matches exactly to the difference in phase of sensors received signals. The phase part is calculated and it is weighted by $w_1=w_2\ldots. = w_n =\frac{1}{n}$ where n is the number of sensors and the output of each sensor is summed up to find the DOA of sound wave.

Let us assume the sound wave from the source reaching the 1st sensor is $a_1(k)$ and the output of the 1st sensor is$b_1(k)$. The relation between $a_1(k)$ and $b_1(k)$ is given as

$$b_1(k)= a_1(k) + \eta\,(k) \quad - \quad (9)$$

Where η (k) is the noise added to the received acoustic signal. The corresponding delay between the reference (1st) and the n^{th} sensor is given in the equation (6). After computing the delay, the frequency domain delay and sum beam former is computed and the equation of spatial domain filter is given as

$$S_{dS}(\theta,\Psi)=\frac{1}{N}\sum_{n=1}^{N} e^{j2\pi(n-1)fd\frac{\cos\theta}{c}} e^{-j2\pi(n-1)fd\frac{\cos\Psi}{c}} \quad - \quad (10)$$

Where *f* denotes the frequency of incoming signal and *d* indicates the distance between the two sensors. The main concept of grating lobes or spatial aliasing occurs when the distance between the sensors *d* satisfies the condition $d>\frac{\lambda}{2}$, where $\lambda=\frac{c}{f}$.

DOA can be calculated more effectively using MUSIC algorithm, which is defined as MUltiple Signal Classification. It plots the pseudospectrum having the peak in the direction of arrival of sound waves from various sources. If there are n sensors and s number of sources, MUSIC results in s eigen values and eigen vectors corresponding to sources and n-s eigen values and eigen vectors corresponding to noise. The output, pseudospectrum equation is given as

$$P_{MU}(\theta) = \frac{1}{|a(\theta)^H E_N E_N{}^H a(\theta)|} \quad - \quad (11)$$

Where, $a(\theta)$ is the steering vector matrix and E_N is the diagonal matrix containing eigenvalues of output correlation matrix. In this paper sound source signal is not assumed and localisation is done from correlation matrix for each source separately by filtering from coherence function the desired frequency source.

2.3 Multiple Sources and Coherence

The above mentioned procedure works well in case of single sound source in the medium. In case of multiple sound sources in the medium, the TDOA and DOA which are calculated from the received sensor signal results in not exact estimation or localisation of sound source. To avoid this problem, coherence between reference and the corresponding sensor signals is calculated. The coherence function peaks at frequencies of different sound sources which are present in the medium. Coherence is defined as measure of degree of linear association between the two signals. The coherence of various sensors w.r.t the reference sensor is taken to find the predominant frequency sound sources in the medium [9]. The coherence function is given as

$$\gamma_{b_1 b_2} = \frac{\xi_{b_1 b_2}(f)}{\xi_{b_1 b_1}(f)\xi_{b_2 b_2}(f)} \quad - \quad (12)$$

Where $\xi_{b_1 b_2}(f)$the generalised is cross power spectral density and the functions $\xi_{b_1 b_1}(f), \xi_{b_2 b_2}(f)$ are the generalised auto power spectral densities. The coherence value always lies in between 0<=$\gamma_{b_1 b_n}$<=1.

3. Assumptions and Simulations

Let us assume there are two sound sources in near field and two sound sources in far field. The DOA of two near field sound sources sound are 80^o and 68^o respectively and DOA of two far field sound sources are 52^o and 45^o respectively on the 1st sensor of linear array. The output of linear array sensors signals is assumed to be noisy with SNR for each sample of 3dB.

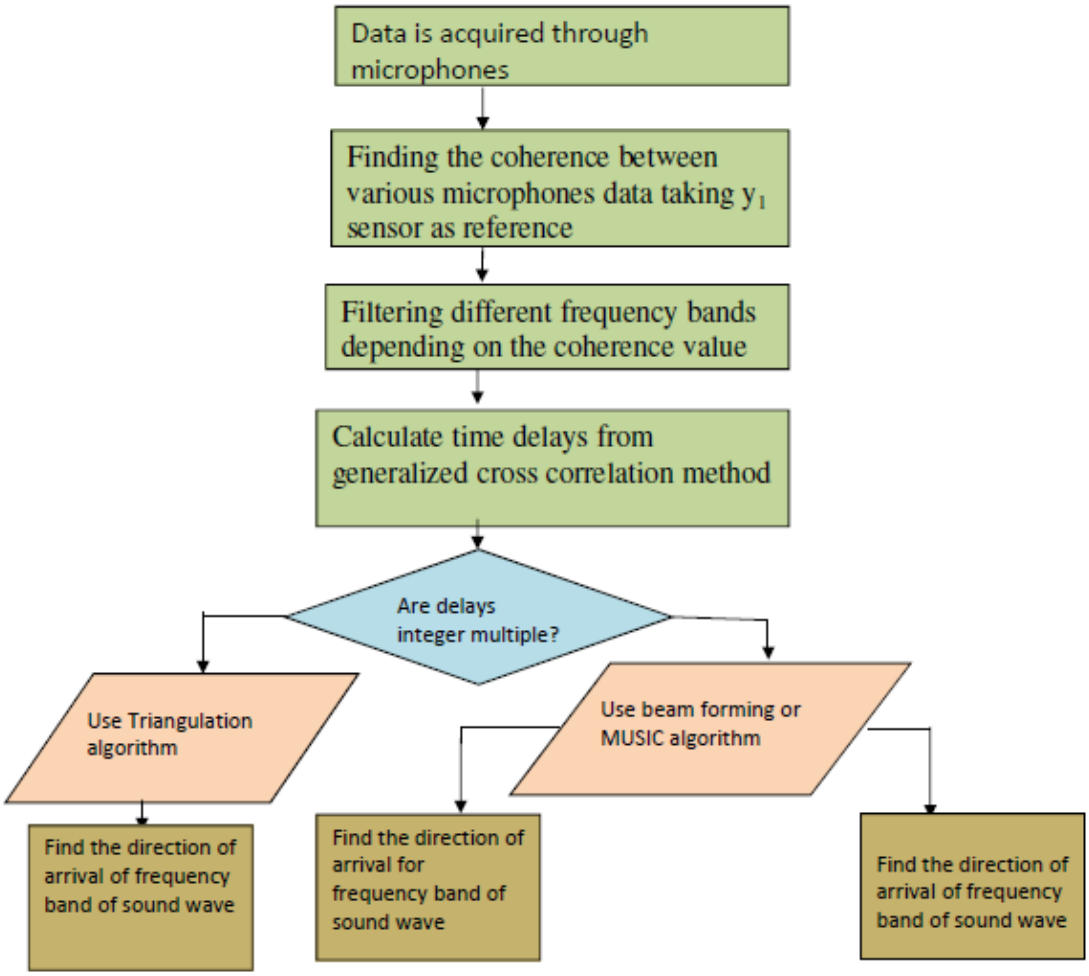
The two near field sound sources have frequency of 250Hz and 350 Hz respectively. The two far field sound sources are having the frequency of 1200Hz and 2000Hz. The table shows the assumed sound sources and their direction of arrival on the 1st sensor.

Tab.1 DOA and Frequency of assumed different sound sources.

Direction of Arrival	Frequency
80^o source in Near Field	250 Hz
68^o source in Near Field	350 Hz
52^o source in Far Field	1200 Hz
45^o Source in Far Field	2000 Hz

These are the assumptions that are made for four different frequency sound sources which are acting in the medium.

The overall algorithm is given in the flowchart



4. Factors affecting TDOA and DOA

The beam width of the output spatial signal depends on the number of sensors used, and it is directly proportional to the number of sensors used in the array. As the number of sensors increases the beam width decreases and the beam is sharper pointing to the desired DOA. The Signal strength increases by summing the output of each sensor signal and it peaks at the DOA of the particular Sound wave in spatial domain. The TDOA and DOA are more dependent on the sampling frequency f_s, as the sampling frequency increases; TDOA and DOA are more accurate. So in this paper f_s is assumed as 100 KHz and minimum number of sensors required is n=5.

5.0 Results and Discussions

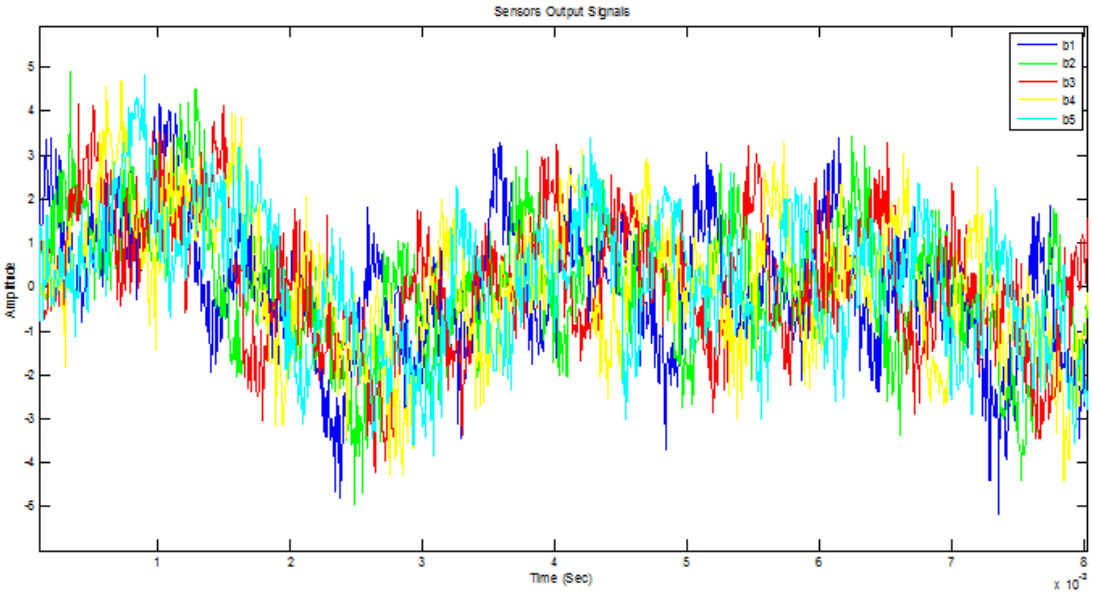


Fig.3 Assumed sensors signal for various sources.

The coherence value from the equation (12) using b_1 sensor as the reference sensor is given in the fig.4

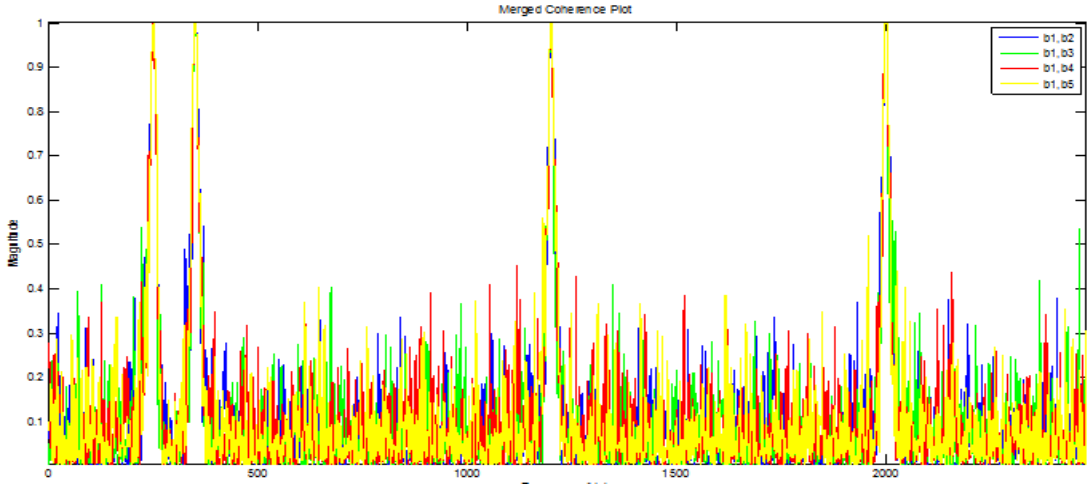


Fig.4 Merged Coherence plot for assumed source data and for *d* =0.085m.

From the above plot we can see the coherence plot peaking at four different frequency bands with centre frequency as 250Hz, 350Hz, 1200Hz, and 2000Hz

respectively which are different sound source frequencies in the medium. The respective frequency bands are filtered and the corresponding TDOA, DOA are found from the equations (1), (3), (4), (7), (8) in near field regions as well as the far field regions depending on the obtained delays from the cross correlation function.

For the simulated 250HZ and 350Hz near field sound sources the radial distances and angle on to the first three sensors is given in the table.2, 3 respectively.

Tab.2 DOA and Radial Distance for 250Hz Sound Source.

	Exact values		Results Obtained	
Sensors	Theta	Radial Distance	Theta	Radial Distance
b_1	79.9^o	0.91m	79.1^o	0.92m
b_2	74.7^o	0.93m	74.2^o	0.94m
b_3	69.8^o	0.95m	69.1^0	0.97m

Tab.3 DOA and Radial Distance for 350Hz Sound Source.

	Exact values		Results Obtained	
Sensors	Theta	Radial Distance	Theta	Radial Distance
b_1	69.05^o	1.01m	69.1^o	0.99m
b_2	64.72^o	1.05m	64.2^o	1.04m
b_3	60.01^o	1.08m	59.1^0	1.09m

The percentage of error in calculating DOA and Radial Distance using triangulation algorithm are 0.0072%, 1.98% respectively. In this paper, the main aim is to improve DOA more accurately and the error in DOA calculations is very less.

The far field sources of frequency 1200Hz and 2000Hz, DOA is calculated using beamforming and MUSIC algorithms. The beamforming plot for 52^o sound source is given in fig.5

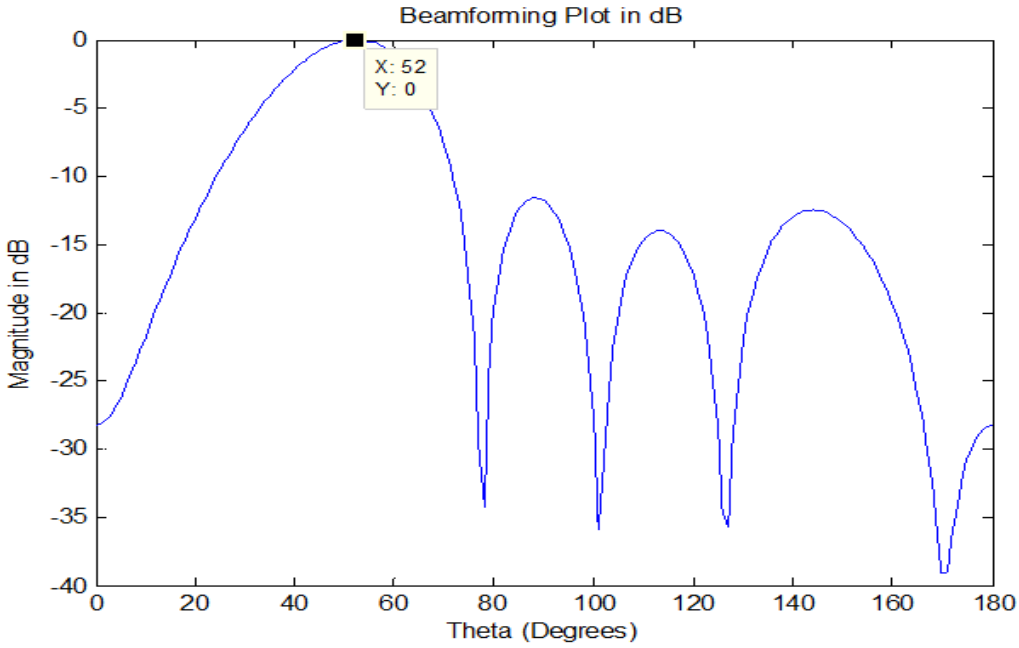


Fig.5 Beamforming plot for *f*=1200Hz, n=5, *d*=0.085m.

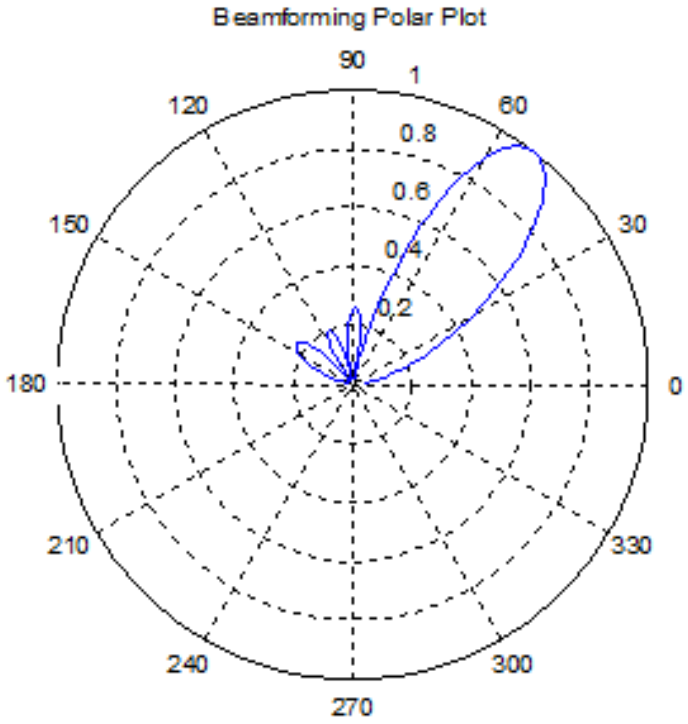


Fig.6 Beamforming Polar plot for fig.5

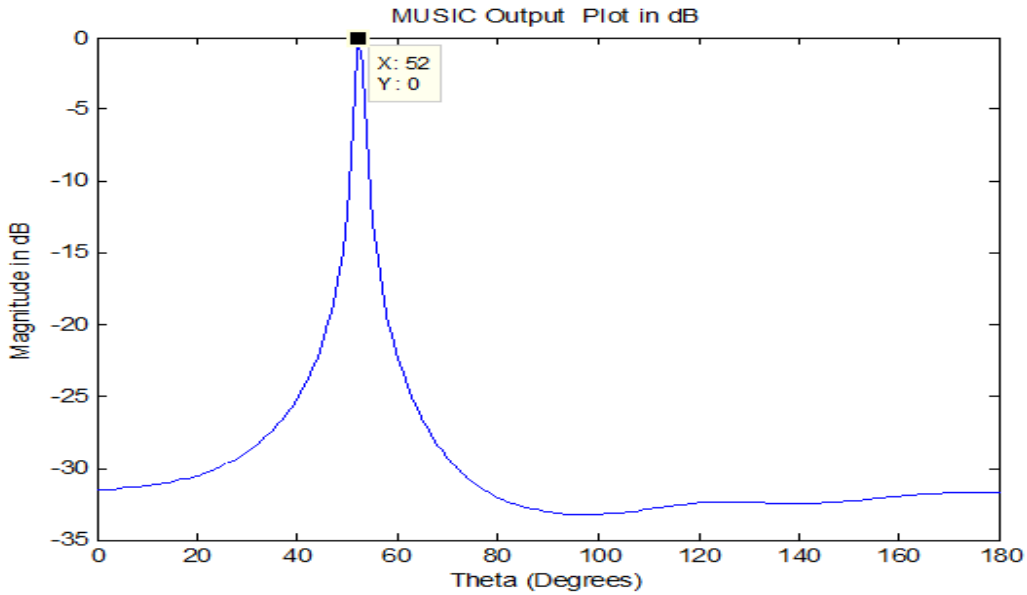


Fig.7 MUSIC output plot for *f*=1200Hz, n=5, *d*=0.085m.

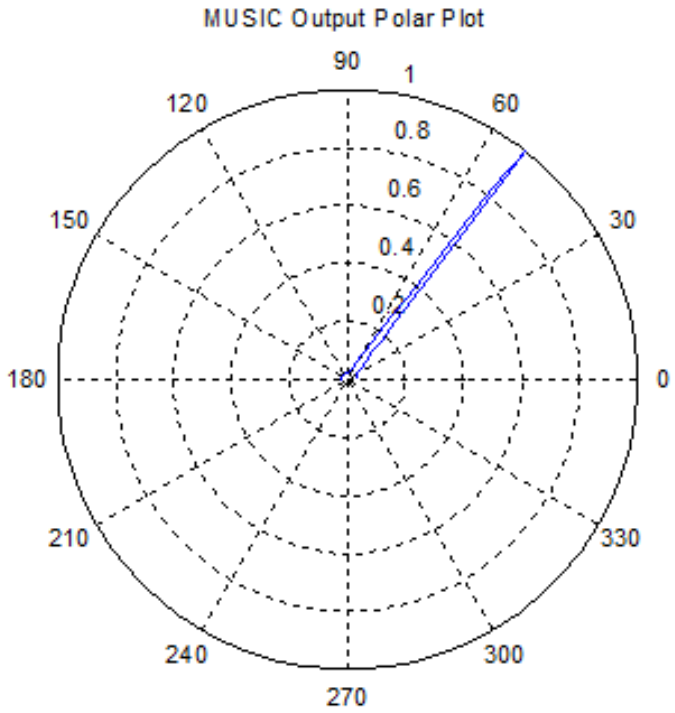


Fig.8 MUSIC Output Polar plot for fig.7

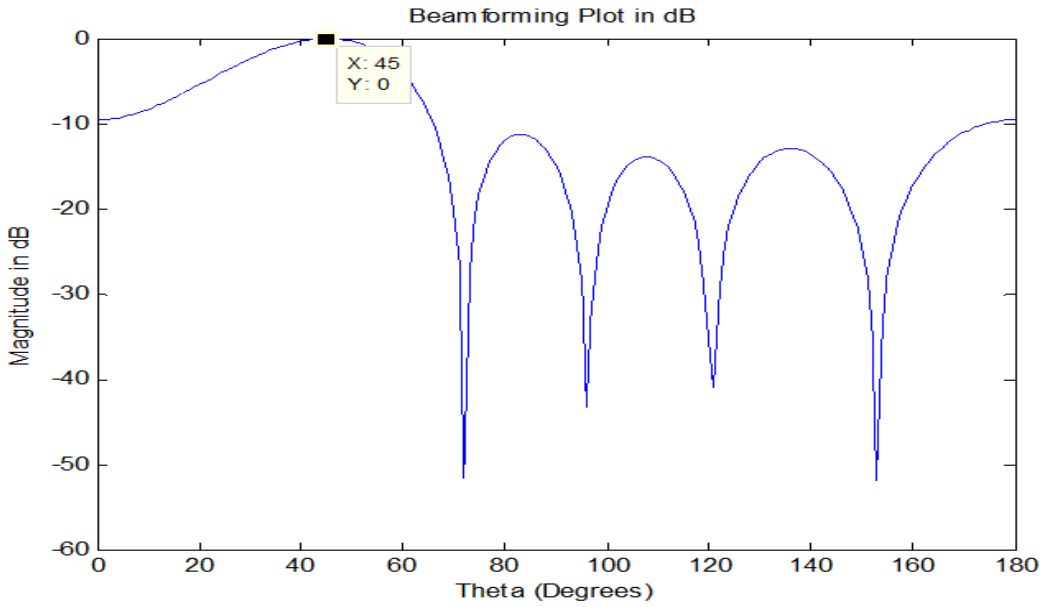


Fig.9 Beamforming plot for *f*=2000Hz, n=5, *d*=0.085m.

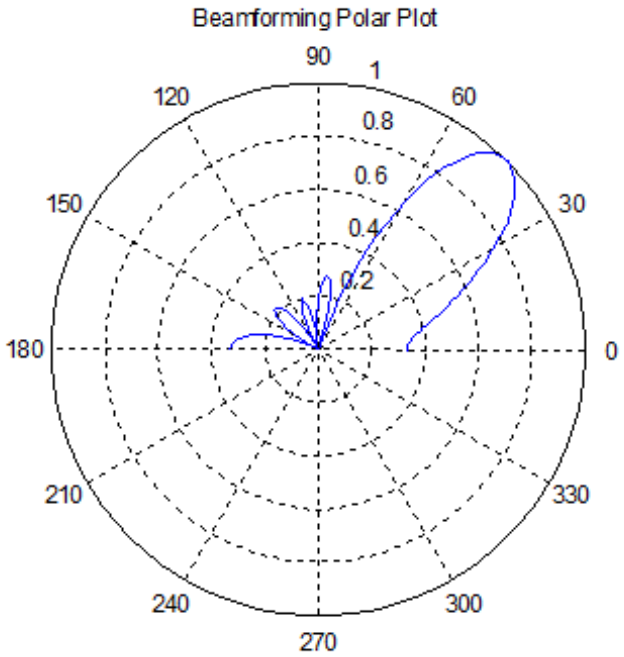


Fig.10 Beamforming Polar plot for fig.9

The power of MUSIC output gives a sharp peak than power of beamforming output for *f*=2000Hz, *d*=-0.085m and with the same number of sensors in both n=5. The corresponding power plots for beamforming and MUSIC algorithms are shown in fig.13, 14 respectively.

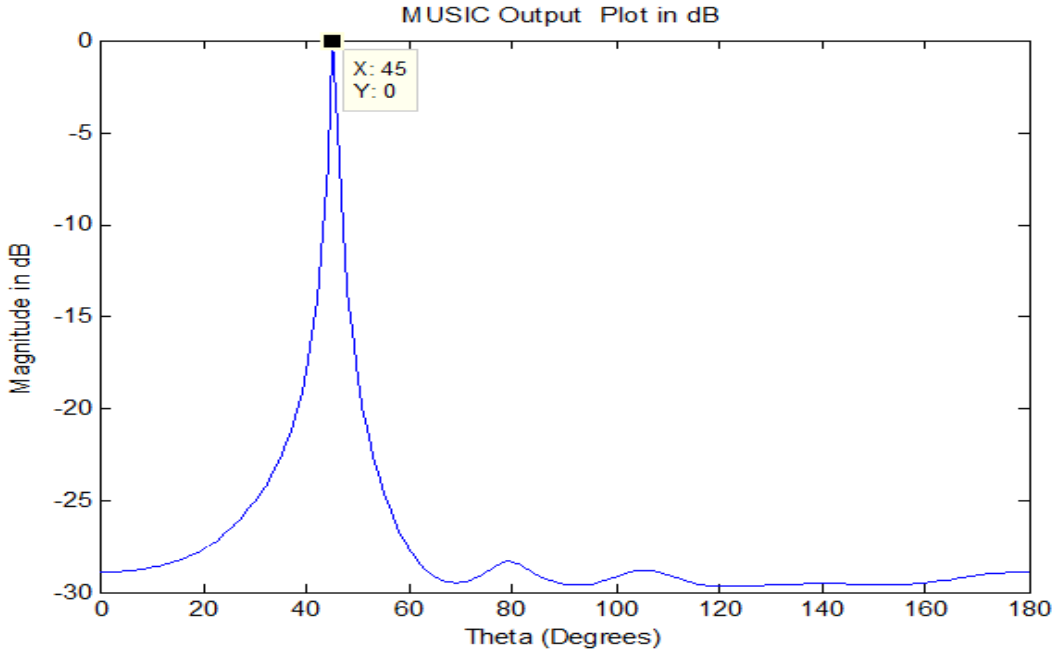


Fig. 11 MUSIC Output plot for *f*=2000Hz, n=5, d=0.085m.

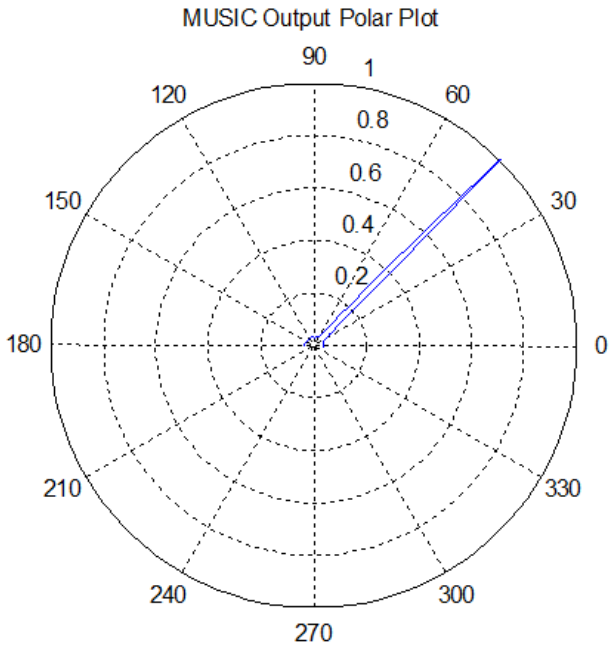


Fig.12 MUSIC Output Polar plot for fig.11

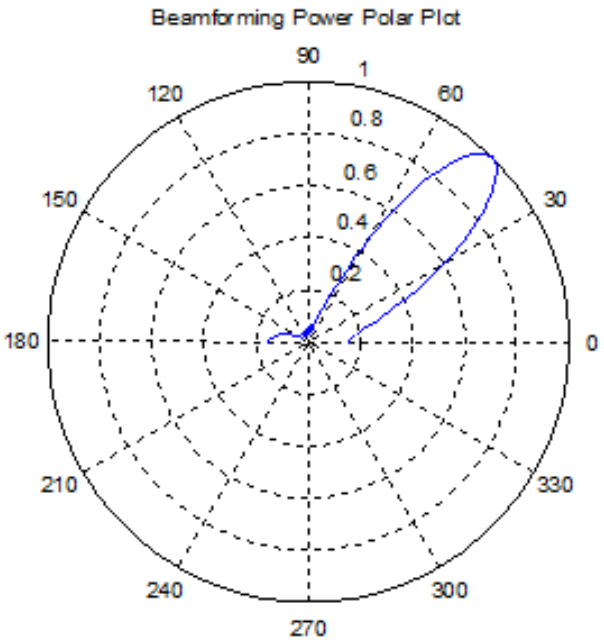


Fig.13 Beamforming Output power Polar plot for *f*=2000Hz, n=5, *d*=0.085m.

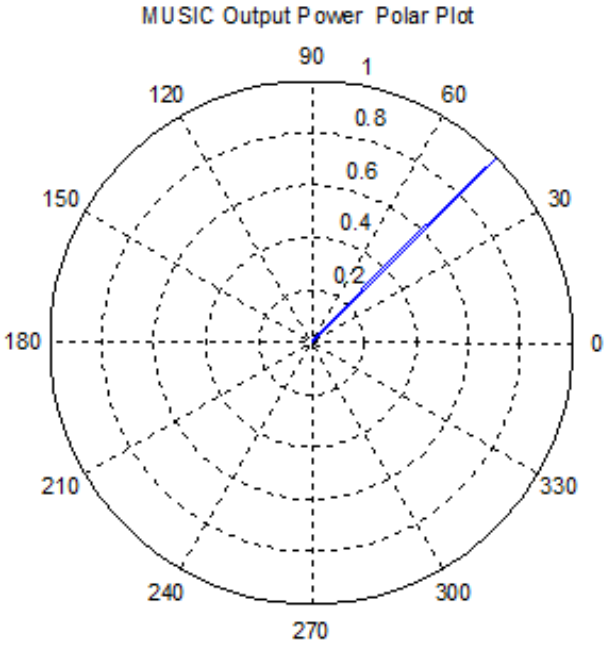


Fig.14 MUSIC Output Power Polar plot for *f*=2000Hz, n=5, d=0.085m.

The approximate view of DOA of different sound sources on the sensor b_1 is given in the fig.15

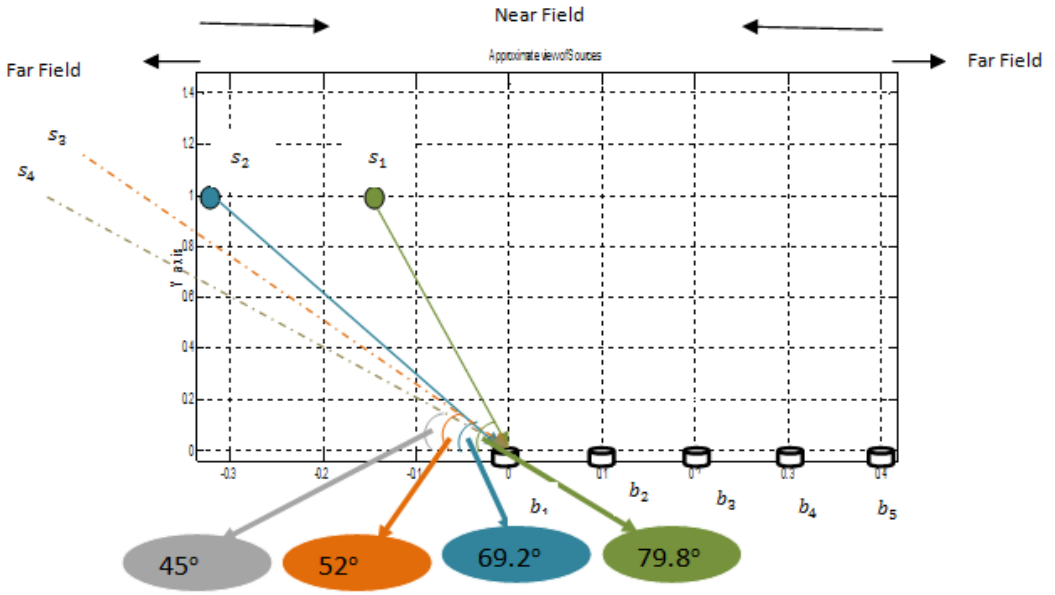


Fig.15 Approximate view of near field and far field Sound Source.

Tab. 4 DOA of various frequency sound sources.

Frequency	Direction of Arrival
250Hz	79.8^o
350Hz	69.2^o
1200Hz	52^o
2000Hz	45^o

Source s_1 and s_2 are of near field sources with frequency 250 Hz, 350 Hz and s_3 and s_4 are of far field sources with 1200Hz, 2000Hz respectively. The two near field sources are located in the grid and DOA on b_1 sensor for four different sound sources is plotted.

5. Conclusion

As the basic environment contains different sound sources in medium, based on the TDOA, it is easy to classify near field and far field sources. The dominant sources can be found from the coherence function, they are filtered, and TDOA, DOA are computed respectively in the localisation process of the desired dominant sound source. This is the basic step using coherence function in finding desired sound source in multiple sources. The application of the above mentioned algorithm is implemented in real jet acoustics which will be discussed in future paper.

6. References

[1] Colin H Hansen, "*Fundamentals of Acoustics"*, Department of Mechanical Engineering, University of Adelaide, Australia, chap.1.

[2] G.J.J Ruijgrok, *"Elements of Aviation Acoustics"*,Yes dee publishing, India, chap.1.

[3] J. Paul Guyer,"*Fundamentals of acoustics*", Continuing Education and Development, Inc, 9 Greyridge Farm Court, Stony Point, Newyork, 10980.

[4] Ali Pour Mohammad, Syed Mohammad Ahadi, "*Real Time High Accuracy 3-D Based Sound Source Localization Using a Simple 4-Microphone Arrangement"*,IEEE systems journal, vol .6, No.3, 2012.

[5] Brent c Kirkwood, 2003,"*Acoustic Source Localization Using Time Delay Estimation"* M.sc. thesis, Technical University of Denmark.

[6] Jacob Benesty, Jingdong Chen, and Yiteng Huang, 2008,"*Microphone Array Signal Processing"*, Springer Publications, USA, chap. 3, 6.

[7] Mahmoud, Mohamed L. Rabeh, Emad M. Zieur, Sherif Hekala, "*Optimization of MUSIC algorithm for angle of arrival estimation in wireless communications",* NRIAG Journal of Astronomy and Geophysics,(2013) 2, pp. 116- 124.

[8] Tshilidzi Mukwevho, Jaco Jordaan, Guillaume Noel,"*Advanced beamforming techniques for Acoustic Source localization*" IEEE Africon 2009 September 23-25, Nairobi, Kenya.

[9] Ki Hong Shin and Joseph Hammond, 2008,"*Fundamentals of signal processing for Sound and Vibration Engineers*", John Wiely & Sons Ltd, England, chap.10.

[10] Jopaul K Ignatius, Dr S. Sankaran, Satyanarayana R. Chakravarthy, "*Hot flow simulation of Acoustics and suppression by injecting water during Rocket lift off",* AIAA vol- 53,No-1, pp. 235-245, 2015.