# Acoustic source Localization

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***Abstract*—In this paper I will be discussing about my project which is implementation of Acoustic source Localization or Estimation of direction of arrival using two techniques Correlation technique and Capon's technique where I have considered a sine signal as input sound or acoustic signal from the source and have found the angle of arrival with both the techniques.**

## I. Introduction

Acoustic source localization refers to the process of retrieving the direction information of several electromagnetic waves/sources from the outputs of a number of receiving antennas that form a sensor array.Direction of arrival estimation is a major problem in array signal processing and has wide applications in radar, sonar, wireless communications, etc.High resolution direction of arrival can resolve closely spaced frequencies.Used for locating and tracking signal sources in both military and civilian communications. With the development of sparse representation and compressed sensing, the last decade has witnessed a tremendous advance in this research topic.
Many other engineering applications require direction of arrival estimation, including wireless communications, radar, radio astronomy, sonar, navigation, object tracking, and rescue and other emergency aid devices.Direction of arrival estimation is usually researched as part of the more wide area of array processing in its contemporary form. Much of the early work in this subject was devoted to radio direction finding, or calculating the direction of electromagnetic waves impinging on one or more antennas.
The development of efficient methods for direction of arrival estimates and adaptive beam shaping has been a focus of signal processing elements of smart antenna systems. The development of digital beam forming systems is fueled by recent adaptive beam forming developments.
An array antenna system with innovative signal processing can improve the resolution of direction of arrival estimation instead of using a single antenna. Multiple sensors are placed across a space in an array sensor system. The received waveform is spatially sampled using this array arrangement. In terms of signal reception and parameter estimation, a sensor array outperforms a single sensor.

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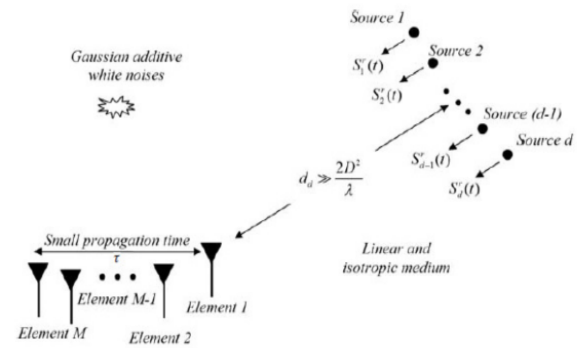



## II. Array of Sensors

Consider an array of 'r' sensors with random locations and directional characteristics that receive signals from 'q' narrowband sources with known centre frequency and locations. Because the signals are narrowband, the propagation delay across the array is significantly smaller than the reciprocal of the signal bandwidth, hence the array output can be written as, using a complex envelope representation.

$$u(t) = \sum_{k=1}^{q} a(\phi_k)s_k(t) + n(t)$$

- where,$x(t) = [x_1(t)..........)]^T$,is the vector of signals received by the array antenna.
- $s_k(t)$,is the signal emitted by the $k^{th}$ source as received at the reference sensor 1 of the array.
- $a(\phi_k) = [1, e^{-j\omega\tau_1(\phi_k)}], .....]^T$,is the steering vector of the array towards the direction $\phi_k$.
- $\tau_i(\phi_k)$ is the propagation delay between the first and the $i^{th}$ sensor for a waveform coming from the direction .
- $n(t) = [n_1(t), n_2(t), ........]^T$ is the noise vector.

## III. Techniques:

1. Correlation Technique
2. Capons Technique
3.MUSIC
4.Root MUSIC
5.ESPRIT

## IV. Factors affecting signal source localization:

The direction of arrival estimation is affected by many factors such as signal to noise ratio, number of array elements, and number of snapshots.

### A. *Signal to noise ratio*

Signal to noise ratio measures the difference between the received signal and the background noise level. In wireless systems, it is always important to optimize the performance of the transmitting and receiving antennas. Thus, the signal to noise ratio directly affects the performance of super-resolution DOA estimation algorithms.

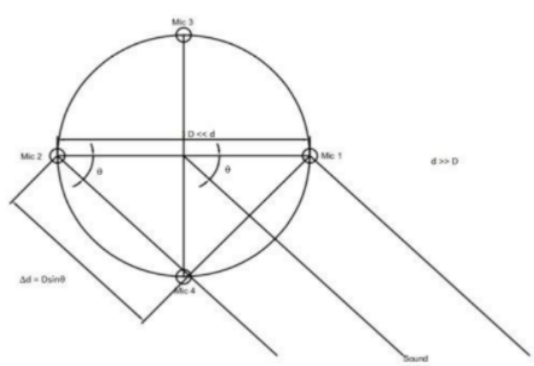
### B. *Number of array elements*

Array antennas are a set of two or more individual sensors used for receiving electromagnetic waves. The desired signals from antennas are combined and processed in order to achieve an improved performance over that of a single antenna.

### C. *Number of snapshots*

In the time domain, the number of snapshots is defined as the number of samples. In the frequency domain, the number of snapshots is defined as the number of time sub-segments of discrete Fourier transform due to the direction of arrival the more the number of snapshots.

## V. Correlation Technique



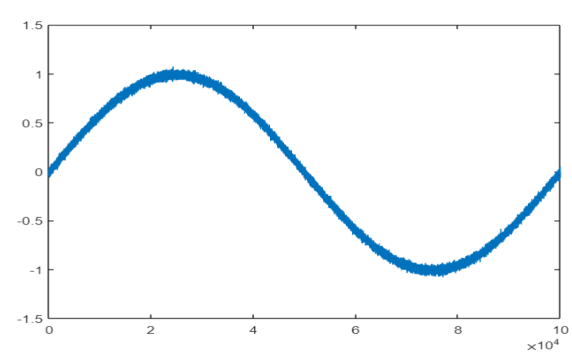

- Let the distance between the microphones be 'D' and the distance between the center of array and sound source be 'd'.
- Since we place the mics very close, we can assume that d ¿¿ D.
- Let the angle of arrival be q.
- Then the difference between distance can be approximated as △d = Dsinq.
- Let the speed of sound be v = 340m/s.
- △ =△d/v
- Then q = acos[v△/D].

```
% Defining Params
Distance = 4e-2;         %In Meters
SoundSpeed = 340;        %In Meters per second
%AoA = 40;                 % Angle of attack in degrees
maxError = 0.1;
Estimated_angle = zeros(1, 180);
AoA = (0:179);
Fsa = [100, 200, 500, 1000]*1e3;             % Sampling frequency% Sampling frequency
for j = 1:4
Fs = Fsa(j);
tend = 0.1;
t = 0:1/Fs:tend*(1-1/Fs);
for k = 1:180
% Generating the two Microphone arrays using angle data.
Distance_Difference =  Distance*cos(AoA(k)*pi/180);
Actual_Time_delay = Distance_Difference/SoundSpeed;
MicA = sin(2*pi*10*(t));
MicB = sin(2*pi*10*(t+Actual_Time_delay));
% Plotting
% subplot(2,1,1);
% plot(MicA);
% subplot(2,1,2);
% plot(MicB);
% Adding noise
maxError = 0.01;
Error_signal_MicA = 2*(randn(size(MicA))-0.5)*maxError;
Error_signal_MicB = 2*(rand(size(MicA))-0.5)*maxError;
%plot(Error_signal_MicA);
MicA = MicA+Error_signal_MicA;
MicB = MicB+Error_signal_MicB;
```

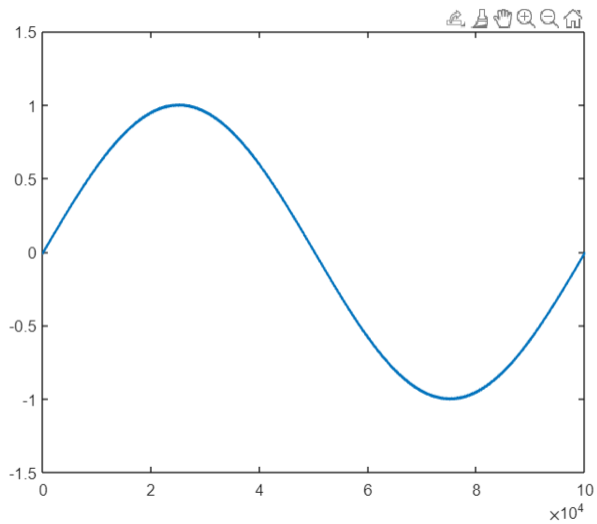
```
% Decoding Time Delay
c = xcorr(MicA, MicB);
L = length(t);
[maxnum, id] = max(c);
diff = (id - L)/L * tend;
error = (Actual_Time_delay - diff)/Actual_Time_delay;

% Getting angle Data
Estimated_angle(k) = 180*abs(acos(SoundSpeed*diff/Distance))/pi;


end
subplot(2,2,j);
plot(AoA);
hold on
plot(Estimated_angle);
ylabel('Angle');
title(strcat("Estimation of angle with sampling frequncy of ", mat2str(Fs), " Hertz"));
legend('AoA', 'Est Angle');
hold off
end
display(strcat("With maximum error of ",  mat2str(maxError*100) , "%"));
```

### A. *Output*

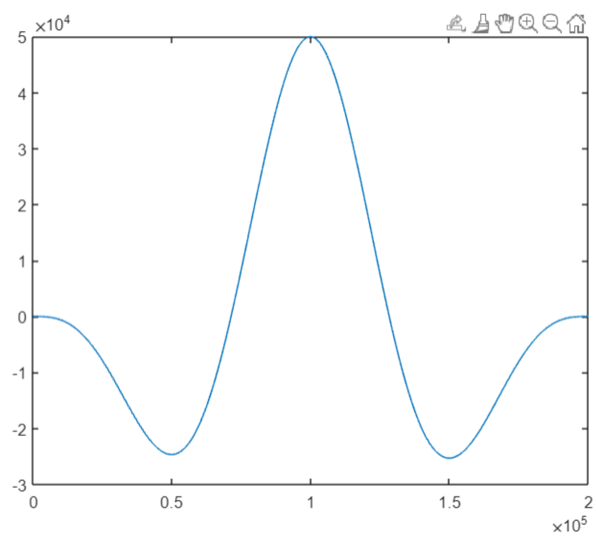
Signals in both the Mic's
Mic-A:



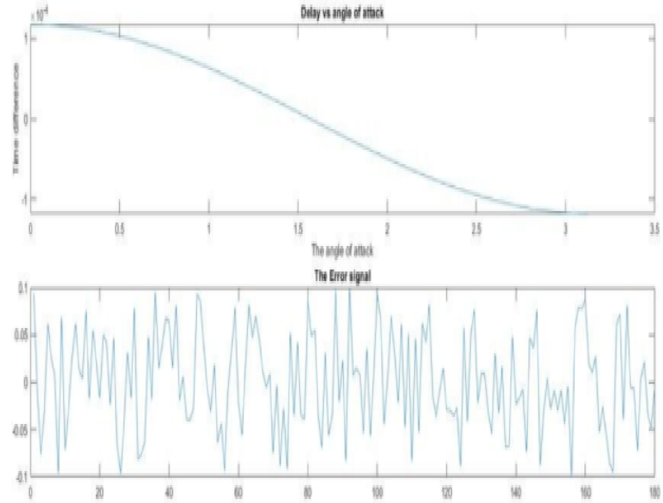
Mic-B:



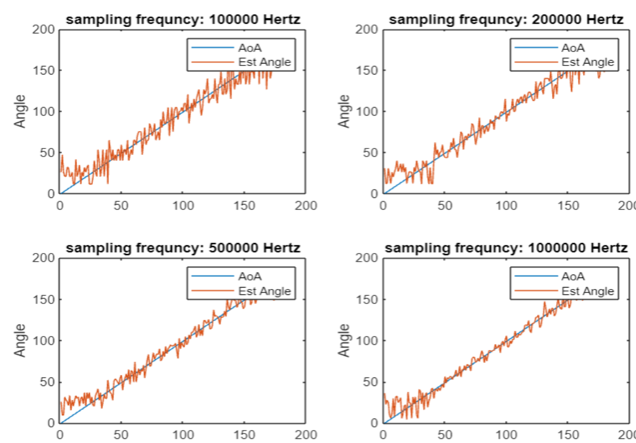
Correlation:



Delay Vs Angle of Arrival:



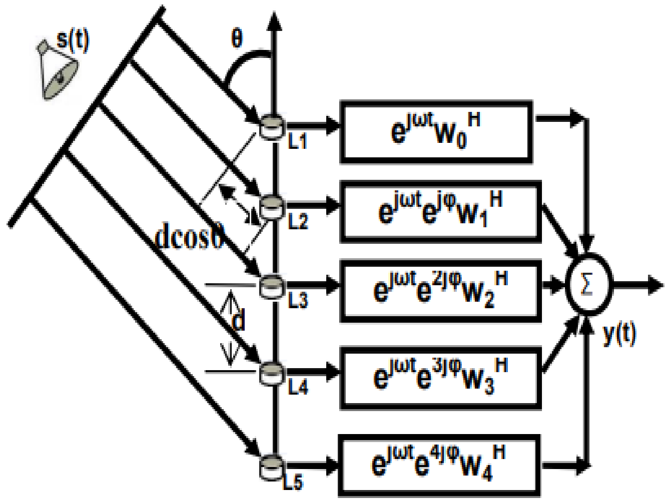

### 1) *DOA Estimation for different sampling frequency:*




## VI. Beamforming

Beamforming is the technique of merging sounds or electromagnetic signals that emanate from a single direction and impinge on different receiver sensors. The resultant signal has a higher intensity due to coherent combining after adequate phase adjustment for each sensor. As a result, the sensor's gain would resemble a huge dumbbell-shaped lobe directed in the desired direction. This crucial notion is employed in a variety of applications including communication, voice, and sonar.

## A. System Model for Estimating DOA




$$x(t) = a(\theta)s(t),$$

where x(t) is the array output, s(t) = exp(j$\omega$t) is the signal coming from the source and steering vector, $a(\theta) = [1, exp(j\phi), ....., exp(j(L-1)\phi)]$,assuming the propagation delay between the source and the first sensor is normalized to unity and the phase delay between the sensors, $\phi = -\omega dcos\theta/c$.
A single signal at the DOA $\theta$ , thus results in a scalar multiple of the steering vector. If M signals impinge on an L-dimensional array from distinct DOAs $\theta_1, \theta_2, ....., \theta_M$, the output vector will be:

$$x(t) = \sum_{m=1}^{M} a(\theta_m)s_m(t)$$

where $s_m(t)$ denotes the baseband signal waveforms from $m_{th}$ source. The output equation can be put in a more compact form by defining a steering matrix and a vector of signal waveforms as
$A(\theta) = [a(\theta_1), a(\theta_2), ....., a(\theta_M)]$ and
$s(t) = [s_1(t), s_2(t), ....., s_M(t)]^T$
In the presence of an additive noise n(t)

$$x(t) = A(\theta)s(t) + n(t)$$

¿Main Constraint in these methods is $M < L$
To find DOA the idea is to steer the array in one direction at a time and measure the output power. The steering locations which result in maximum power yield the DOA estimates. The array response is steered by forming a linear combination of the sensor outputs

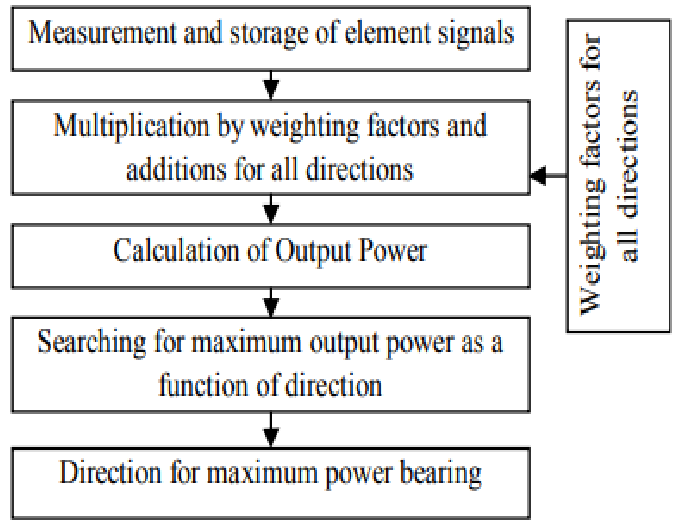
$$y(t) = \sum_{l=1}^{L} w_l^* x_l(t) = w^H x(t)$$

where w is the weighting vector used for cancelling the phase delay between the sensors. N-samples of y(t) are taken with time interval T between the samples and t = kT , where k = 1,2,...,N.

The output power is:

$$P(w) = \frac{1}{N}\sum_{t=1}^{N} |y(t)|^2$$
$$P(w) = \frac{1}{N}\sum_{t=1}^{N} w^H x(t)x^H(t)w$$
$$P(w) = w^H Rw$$

## B. Block Diagram




For an array of arbitrary geometry, this algorithm maximizes the power of the beamforming output for a given input signal. Let, we wish to maximize the output power from a certain direction . The problem of maximizing the output power is then formulated as

$$maxE[w^H x(t)x^H(t)w] = max(w^H E[x(t)x^H]w)$$

when carrying out the above maximization. The resulting solution for w is then,when carrying out the above maximization. The resulting solution for w is then,

$$w_{BF} = \frac{a(\theta)}{\sqrt{a^H(\theta)a(\theta)}}$$
$$P_{BF} = \frac{a^H(\theta)Ra(\theta)}{a^H(\theta)a(\theta)}$$

The standard beamwidth for a ULA is $\phi_B$ = 2/L , and sources whose electrical angles are closer than $\phi_B$ will not be resolved by the Conventional Beamformer,regardless of the available data quality.
In an attempt to alleviate the limitation of the conventional beamformer, such as its resolving power of two sources spaced closer than the beamwidth, proposed modifications is given by Capon which is also known as the Minimum Variance Distorsionless Response Filter.
The optimization problem is proposed as:

$$\text{Min P(w) subject to } w^H a(\theta) = 1$$

This beamformer attempts to minimize the power contributed by noise and any signals coming from other directions than $\theta$ , while maintaining a fixed gain in the look direction $\theta$

like as a sharp spatial bandpass filter.The optimal w can be found using the technique of Lagrange multipliers, resulting in

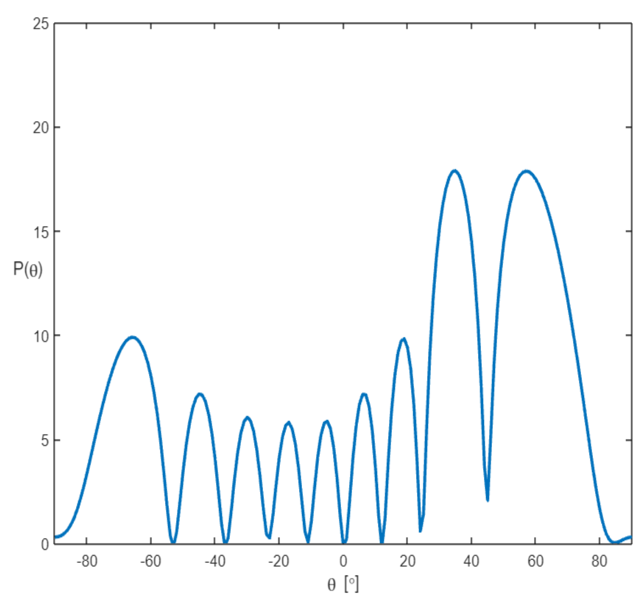
$$w_{CAP} = \frac{R^{-1}a(\theta)}{a^{H}(\theta)R^{-1}a(\theta)}$$
$$P_{CAP} = \frac{1}{a^{H}(\theta)R^{-1}a(\theta)}$$

### C. MATLAB CODE

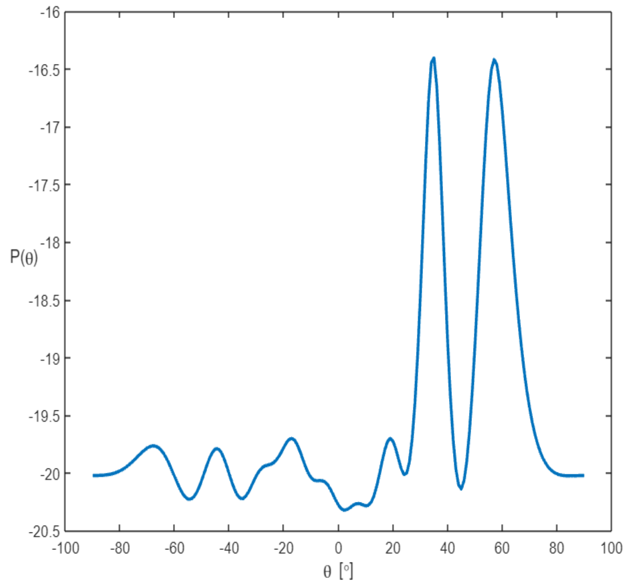
```
clear all;
close all;
fs = 10e6;
ts = 1/fs;
T = 0.0001;
t = 0:ts:T-ts;
Ns = length(t); %number of samples
fc = 1e6;
lambda = 3*10^8/fc;
M = 10;%number of antenna elements
d = 0.5*lambda; %distance between neigbouring antennas
s1 = exp(1i*2*pi*fc*t);

%teta = [40]/180*pi; %direction of arrival in degrees
teta = [0, 40]/180*pi; %directions of arrivals in degrees
amp = [1 1];
%relative phase delay between signals received through neogbouring elements
delta_fi = -2*pi*d/lambda*sin(teta);
for m=1:M
    aU(:,m) = exp(1i*((m-1)*delta_fi)); %steering vector
end
x = zeros(M,Ns);
for k=1:length(teta)
    x = x + amp(k)*aU(k,:).'*s1*exp(1i*randn()); %data at the outputs of ante
end
SNR = 10;
sz = (2^(-0.5))*sqrt(10^(-SNR/10))*(randn(M,Ns)+1i*randn(M,Ns));
Px = var(aU(k,1).'*s1*exp(1i*randn())); %power of signal
x = x + sz; %adding noise
Psz = var(sz(1,:)); %noise variance
Rxx = x*x'/Ns; %data covariance matrix
SNR_true = 10*log10(Px/Psz); %true SNR in dB
iRxx = inv(Rxx); %inverse of covariance matrix
teta_v = -pi/2:pi/180:pi/2; %range of scanned directions in radians
teta_v_deg = teta_v/pi*180; %same as above but in degrees
Nteta = length(teta_v); %number of scanned directions
for k=1:Nteta
    teta = teta_v(k); %current scannig direction
    delta_fi = -2*pi*d/lambda*sin(teta); %relateive phase delay
    for m=1:M
        aT(m) = exp(1i*((m-1)*delta_fi)); %steering vector
    end
     a=aT.'; %transpose of steering vector
     wbf = a; %weight vector for conventional beamforming
     Pbf(k) = wbf'*Rxx*wbf;
     wMVDR = (iRxx*a)/(a'*iRxx*a); %weight vector for Caponbeamforming
     PMVDR(k) = wMVDR'*Rxx*wMVDR/1; %spatial power spectrum of Capon beamformer
 end
Pbf_dB = 10*log10(Pbf);
PMVDR_dB = 10*log10(PMVDR);
figure
plot(teta_v_deg, Pbf_dB,'k','LineWidth',2);
axis([-90 90 0 20*log10(M)+5])
xlabel('\theta [\circ]');
ylabel('P(\theta)','rotation',0);
figure
plot(teta_v_deg, PMVDR_dB,'k','LineWidth',2);
xlabel('\theta [\circ]');
ylabel('P(\theta)','rotation',0);
```

### D. Outputs:

#### 1.Conventional Beamforming



*E. Outputs:*
*2.Capons Beamforming*




## VII. Inference

From the above outputs we can see that the power of the noise as well as the power of the signal in other direction is less in capons beamforming method compared to the conventional beamforming method and also if the two sources are at a angular distance less than $2\pi/L$ where L is no.of sensors then the conventional beamforming can't resolve them better than capons beamforming.

## VIII. Conclusion

In this project i have implemented the Acoustic source Localization or Estimating the direction of arrival using two techniques Correlation technique and Capon's technique where i have considered a sine signal as input sound or acoustic signal from the source and have found the angle of arrival with both the techniques.

## IX. REFERENCES

1) Direction Of Arrival (DOA) Estimation Using Array Signal Processin by Dhabale, Ashmika. UC Riverside Electronic Theses and Dissertations
2)PERFORMANCE EVALUATION OF DIRECTION OF ARRIVAL ESTIMATION USING UNIFORM AND NON-UNIFORM LINEAR ARRAYS FOR SIGNAL SOURCE LOCALIZATION by KWIZERA Eva.