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Performance Analysis of Music Algorithm for Smart Antenna

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Abstract: In wireless communication, smart antenna technology are growing exponentially and. Performance of smart antenna depends on efficiency of digital signal processing algorithms. This can adjust its own beam pattern in order to emphasize on the signal of interest and minimize the interference signal. Various Direction of Arrival (DOA) algorithms are used to estimate the number of incidents plane waves on the antenna array and their angle of incidence of the smart antenna arrays. This paper analyzes the Multiple Signal Classification (MUSIC) as DOA algorithms developed for smart antenna applications. The performance of the MUSIC algorithm is analyzed using MATLAB software. The simulation results show that the resolution of the DOA techniques improves as number of snapshots, number of array elements and signal-to-noise ratio increases.
Keywords: Smart Antenna, Direction of Arrival, Beam forming, MUSIC

I. INTRODUCTION

Wireless communication has always been one of fast growing technology its behavior is dynamic, hazardous and turbulent. One of the developments in wireless telecom is the smart antenna system. New wireless applications with smart antenna technology are growing exponentially. In addition the largest algorithms those control smart antenna have been effective in dynamic environments. Smart antennas are now becoming a critical adjusts for increasing the performance of wireless application.

The Smart Antenna term generally refers to any antenna array, its radiate power desire direction and minimizes undesired direction. Its array consists of asset of distributed antenna elements (dipoles, monopoles or directional antenna elements) arranged in certain geometry of desired signal strength and reduce the interference from other signals. The direction of interest is also known as Direction of Arrival (DOA) of the incident signal. This plays significant role in smart antenna systems.

Instead of using a single antenna, an array antenna system with digital signal processin0g can enhance the resolution of a signal DOA. An array sensor system has multiple sensors distributed in space.

This array sensor system has multiple sensors distributed in space.this array configuration provide spatial samplings of the received waveform.A sensor array has better performance than the single sensor in signal reception and parameter estimation. [Instead of using a single antenna, an array antenna system
with innovative signal processing can enhance the resolution

An array sensor ssystem has multiple sensors distributed in space. There are many different super resolution algorithm includeing spectal estimation,based,and eigen-analysis to name a few [5,6,7]. In this paper ,we discuss the estimating the DOA of multiple signals using uniform linearbarray (ULA) antenna with a class of Multiple Signal Classification.Detailed simulation results for the algorithm to demonstrate the performance are presented in this paper

II.PRINCIPLE OF DIRECTION OF ARRIVAL

DOA is for the direction of array antenna of the radio wave.

A sensor array has multiple sensors distributed in space; here we use an array antenna with an *M* element.

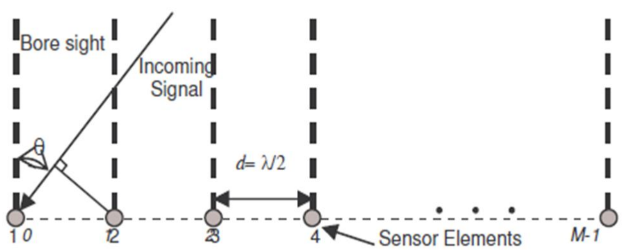

Fig. 1. Uniform linear array antenna

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As shown the Fig.1general configuration for a ULA antenna having M elements arranged along a straight line with the distance between adjacent elements, be $d=\lambda/2$, where λ is wavelength of incoming signal, θ, is measured relative to the antenna bore sight. Suppose a plane target signal waveform comes from the direction of $k=\sin\theta$. The difference of the propagation path $\Delta d_i = d_i sin\theta_i$. The propagation time delay τ_i is

$$\tau_i = \frac{\Delta d_i}{c} = \frac{d_i sin\theta_i}{c}$$

C= speed of light

For narrow band signal the relative phase shift of the i_{th} element is

$$\beta_i = -\frac{2\pi}{\lambda} d_i sin\theta_i$$

The distance between the two neighboring sensors has to be no more than one half of the signal wavelength. Consider there are *L* independent signal sources impinging on the antenna array we want to use it for find their direction of arrival (DOA).The input signal to each individual sensor is the combination of *L* independent signals and environmental noise. For i_{th} sensor element the array output can be expressed as

$$x_i(t) = \sum_{k=1}^{L} S_{k,i}(t) + n_i(t) \qquad i= 0,1,\ldots,M-1 \qquad (3)$$

Where $s_{k,i} = s_{k,0}(t - \tau_{k,i})$ and $s_{k,0}$ is signal emitted by the k^{th} sources as received at the references sensor *i* of the array, $n_i(t)$ is the noise at the i^{th} sensor and there vector is represented as $n(t) = [n_0(t), n_1(t), \ldots, n_{M-1}(t)]^T$, $\tau_{k,i}$ is the propagation delay between first and the i^{th} sensor for a waveform coming from that direction. for antenna array element M, we can define array input vector $x(t) = [x_0(t), x_1(t), \ldots, x_{M-1}(t)]^T$

For the narrowband input signals, signal $s_{k,i}(t)$ is related to the signal $s_{k,0}(t)$ and phase shift factor between them is $\beta_{k,i}$. For wide band input signal, the delay time of signal i^{th} sensor from reference signal at the origin may not be an integer multiple of the sampling time; interpolation filtering is required to emulate their delay. Space Time Adaptive Processor is used to estimate the DOA of wideband signals. In this paper we consider only narrowband signal

A. Music algorithm

MUSIC (Multiple **S**ignal **C**lassification) is a high resolution techniques that based on subspace eigen structure of covariance matrix, which gives the estimation of number of signals arrived and their direction of arrival i.e. signal-subspace, noise subspace. Estimation of DOA is performed from one of these subspace, for that assuming noise in each channel is highly uncorrelated, this makes covariance matrix diagonal.

In uniform linear array (ULA) as shown in fig().The eleventvant angle θ is measured from the bore sight of the reference element to the direction of the incoming signal *S(t)*.Thus the range of θ for ULA is from -90° to 90° with respect to the bore sight. let M be the number of array element in a smart antenna array. Let there be N snapshots at each array element for any N Є {1,….., n}, the nth snapshots at each sensor is for same instant in time .Let A represent the N×M matrix of snapshots, it has complex values , representing in-phase and quadrature components. The covariance matrix is given by

$$R = A^H A$$

Where ^{H} denotes complex conjugate transpose. The analysis of *R* yields the eigenvalues *L* and corresponding eigenvectors *E*. Suppose a narrowband signal *S(t)* impinging a ULA with *N* snapshots from the direction of elevation angle θ; the array output is written as

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

Where *n(t)* is additive white noise, $a(\theta)$ is steering vector of ULA represent the response from all elements in the array,

$$a(\theta) = [1, e^{-jw}, e^{-j2w}, \ldots, e^{-j(M-1)w}] \qquad (6)$$
$$= [1, z^{-1}, z^{-2}, \ldots, z^{-(M-1)}] \qquad (7)$$

$z = e^{jw}$ w is given by

$$w = 2\pi \frac{d}{\lambda} sin\theta$$

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Where λ is the wavelength corresponding to the center frequency of the narrowband signal and *d* is spacing between the ULA elements.

The signal auto covariance matrix can be written as the average of *N* array output samples:

$$\hat{R} = \frac{1}{N}\sum_{t=1}^{N} x(t)\, x^{H}(t)$$

MUSIC algorithm required subspace estimation from signal autocovariance matrix and it can be obtained from eigen decomposition process. The subspace estimation can be given by:

$$\hat{R} = A_S E_S A_S^H + \sigma_n A_n A_n^H$$

Output power of MUSIC algorithm is given by:

$$P_{MUSIC} = \frac{1}{[a^H(\theta) E_N E_n^H a(\theta)]}$$

Since the directional antenna is not uniform, therefore both power spectrum and steering vector need to be modified in order to include the array gain.

$$a_d(\theta) = G(\theta) a(\theta)$$

$$P_{MUSIC} = \frac{1}{[a_d^H(\theta) E_N E_n^H a_d(\theta)]}$$

Where $G(\theta)$ is the gain of array at specific direction of elevation angle.

III. EVALUTION AND SIMULATION RESULT

The simulation is carried out on MATLAB platform where the Uniform Linear Array with different M number of element is used. The simulation has been run for two signals coming from different angles 10°, 50° for 200 snapshots, SNR of 20dB and 10 array elements. Performance of the algorithm has been analyzed by considering number of snapshots, number of array element and SNR.

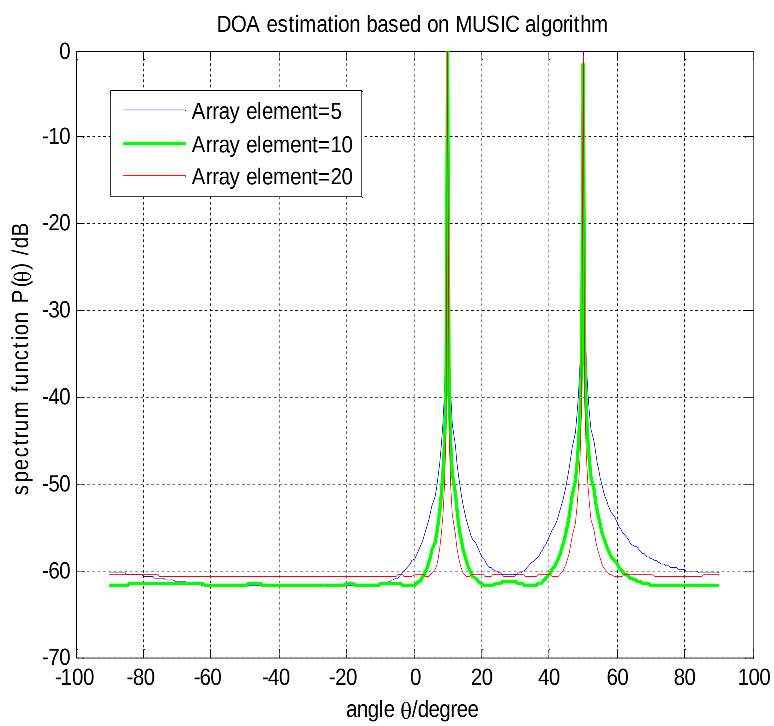

Fig. 2: MUSIC spectrum for varying number of array elements

The graph shows array element increases from 5 to 20, with increase in the number of array element, peaks in the spectrum become sharper and the directivity of the array become good and hence resolution capacity of MUSIC increases.

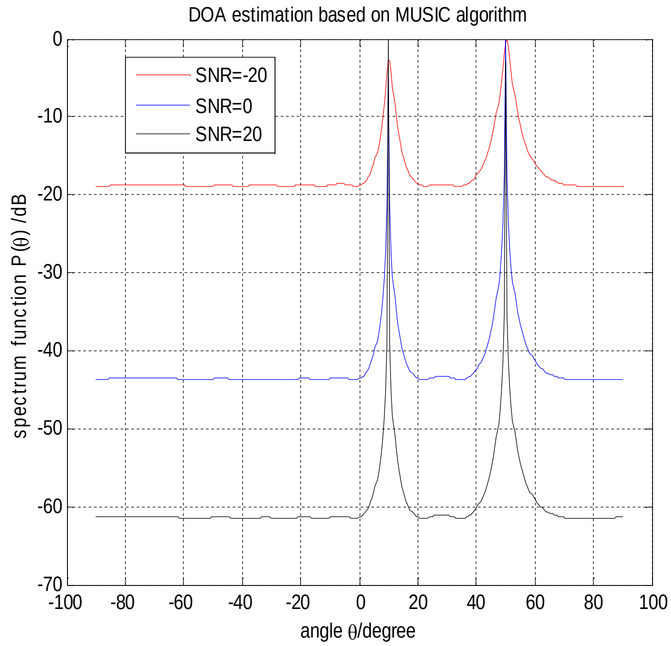

Fig. 3: MUSIC spectrum for varying number of SNR

Fig.3 indicate that the increase in the number of SNR, the beam width of MUSIC spectrum becomes narrow, the direction of the signal become clearer and accuracy of algorithm is also increased.

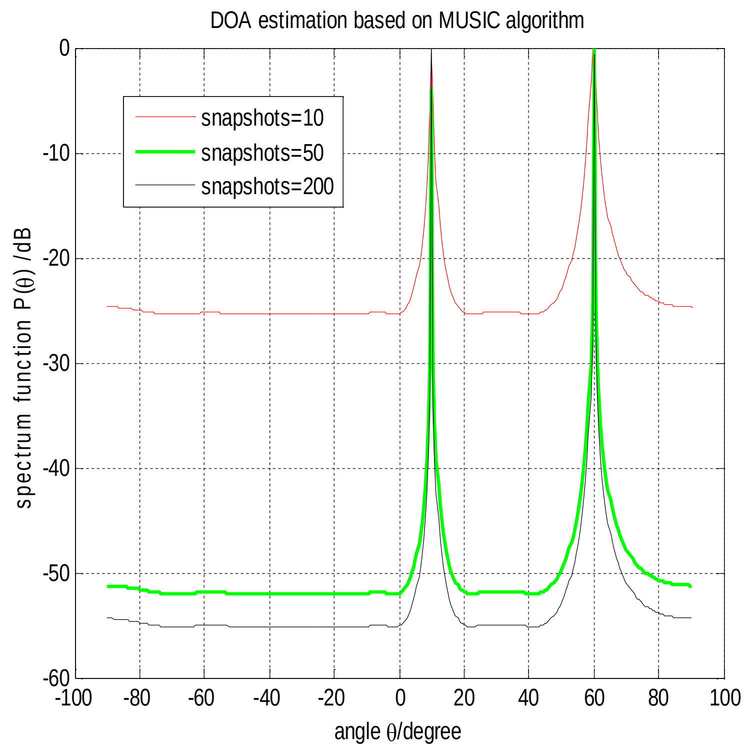

Fig. 4: MUSIC spectrum for varying number of snapshots

Fig 4 indicates that as snapshots increase from 10 to 200, the direction of the array element becomes good and the accuracy and resolution capacity of MUSIC increases, peaks in the spectrum becomes further sharper for snapshots 100,200.

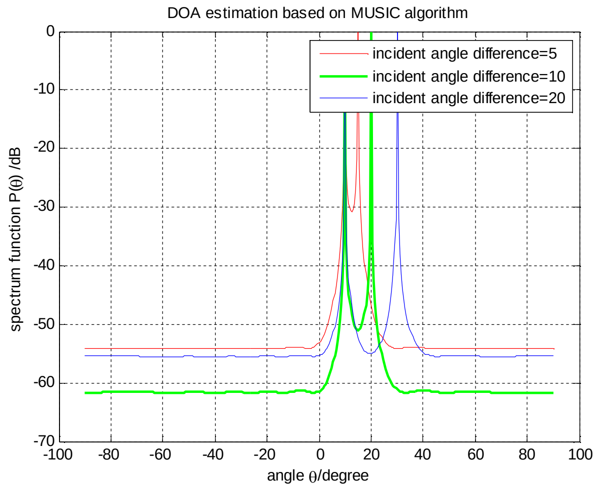

Fig. 5: MUSIC spectrum for varying incident angle difference.

Fig. 5 shows the incident angle difference is 5°, 10°, 20° respectively with the increase in incident angle difference, the beam width of the MUSIC spectrum become narrow, direction of the signal becomes clear. When the signal wave angle space is very small, the algorithm cannot estimate the number of signal sources.

IV. CONCLUSION

This paper presents results of direction of arrival estimation using MUSIC algorithm. The simulation result shows that performance of MUSIC improves with higher number of snapshots of signals, with more elements in the array and greater angular separation between the signals. These improvements are analyzed in the form of sharper peaks in MUSIC spectrum and smaller errors in angle detection. Simulation results indicate that as number of snapshots increases, the beam is narrow and more directive, which results in accurate detection of closely spaced signals. For MUSIC, the ideal value of number of snapshots is 200, which gives MSE is zero.

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