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GSU-MUSIC DOA Estimation Algorithm for Mobile Communication

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Abstract-Basically co-located with a base station, a smart antenna system combines an antenna array with a digital signal-processing capability to transmit and receive in an adaptive, spatially sensitive manner. In other words, such a system can automatically change the directionality of its radiation patterns in response to its signal environment. This can dramatically increase the performance characteristics (such as capacity) of a wireless system. In many applications, the desired information to be extracted from an array of sensors is the content of a spatially propagating signal from a certain direction. The content may be a massage content in the signal, such as in communications applications, or merely the existence of the signal, as in the radar and sonar. This paper presents a new way of correctly determining peaks of the MUSIC ("Multiple Signal Classification") spectrum. This method can be used with any smart antenna geometry. The MUSIC algorithm for DOA estimation finds the MUSIC spectrum for various angles and chooses the maxima or peaks as the angles of arrival. The new method GSU-MUSIC gives more accurate results than MUSIC method.

Keywords---DOA (Direction of Arrival), MUSIC (Multiple Signal Classification), GSU (Gold-Section Univariate Minimization), Smart antenna, Beamforming.

I. INTRODUCTION

The smart antenna technology can significantly improve wireless system performance and economics for a range of potential users. It enables operators of PC's cellular and wireless local loop networks to realize significant increase in signal quality, network capacity and coverage. This is a new and promising technology in the field of wireless and mobile communications in which capacity and performance are usually limited by two major impairments multipath and co-channel interference. Multipath is a condition that arises when a transmitted signal undergoes reflection from various obstacles in the environment. This gives rise to multiple signals arriving from different directions at the receiver. Smart antennas (also known as adaptive array antennas and multiple antennas) are antenna arrays with smart signal processing algorithms to identify spatial signal signature such as the **Direction of arrival (DOA)** of the signal and use it to calculate beam forming vectors, to track and locate the antenna beam on the mobile targets. The antenna could optionally be any sensor. Smart antenna techniques are used notably in acoustic signal processing, track and scan Radar, Radio astronomy and Radio Telescopes and mostly in Cellular Systems like W-CDMA and UMTS.

II. LITERATURE REVIEW

In the past decade, there has been a tremendous growth in the area of telecommunications. As cellular phones and the high speed Internet become more and more popular, the demand for faster and more efficient telecommunication systems has been skyrocketing. Both the increase in the number of users and the increase in high data rate transfers from the Internet have produced a huge increase in traffic. In order to handle this heavy traffic, the whole telecommunications infrastructure has been transformed in the past several years. Thousands and thousands of optical fibers were laid underground to allow high speed, wide bandwidth signal transfer. This has solved most of the problems for land-based systems. However, more and more high-speed services are now carried out in a mobile environment where data transfer is done through wireless channels. Compared to the capacity of a fiber, the capacity of a wireless link is the weakest part in the whole infrastructure. Many people have realized this and have tried different methods to increase the bandwidth of the wireless channels. Before going any further, let us explain how wireless communications systems work. In most wireless communications systems, there are two major components - base stations and the mobiles. The base station is located at the center of a coverage area called a 'cell' and the mobiles can be anywhere within the cell. Communication then takes place between the base station and the mobile through the wireless channel. A certain amount of spectrum is assigned to a cell for signal transfer. This spectrum provides the media for the signals to be transmitted. With a wider spectrum, more users can be served within a cell. To serve a large area, one can use a high power base station to cover the whole area. But only a fixed amount of users can be served in this way. So instead of having a single large cell, multiple cells with smaller size are usually used to cover a large area. To avoid severe interference between cells, base stations that are adjacent to each other are allocated different spectrum or channel groups.

III. DOA ESTIMATION ALGORITHMS

MUSIC [1] is a technique based on exploiting the Eigen structure of input covariance matrix. MUSIC [1] stands for Multiple Signal Classification. It is one of the earliest proposed and a very popular method for super-resolution direction finding, which gives the estimation of number of signals arrived, hence their direction of arrival. Eigen vectors are easily obtained by either an Eigen decomposition of sample covariance matrix or a Singular Value Decomposition (SVD) of the data matrix. By MUSIC algorithm the powers and cross correlations between the various input signals can be readily obtained and the DOAs of the multiple incident signals can be estimated by locating the peaks of a MUSIC spatial spectrum. We begin with the signal snapshots Y, which is a K×M matrix. Covariance matrix S is computed via (1). The next step is Eigen analysis, getting the eigenvalues L and the corresponding eigenvectors E. In this paper, we describe a partitioning method in Section IV on the set of the normalized eigenvalues, which gives us an estimate of D. We separate the eigenvectors into M - D noise eigenvectors E_N and the remaining D signal eigenvectors E_S. In MUSIC, the spectrum (8) is evaluated for different angles and the peaks are chosen. Numerically, the estimate is limited by the evaluation interval. Let M be the number of sensors in a smart antenna array. Let there be K snapshots at each sensor. For any k ϵ {1, K} the k'th snapshot at each sensor is for the same instant in time. Let **Y** represent the K×M matrix of snap-shots. This matrix has complex values in general, representing in-phase and quadrature components. The covariance matrix is given by

$$S = Y\,Y^H \tag{1}$$

Where H denotes Hermitian. It has been shown that **L** will contain only positive real values as **S** is positive definite. If there are D < M independent signals, M-D of the eigenvalues will ideally be 0 under no noise condition, but, will be close to 0 depending on the *signal-to-noise ratio* (SNR). After sorting the eigenvalues in **L** in ascending order, the M×(M-D) matrix $\mathbf{E}_N$, the matrix of the M-D eigenvectors corresponding to the M-D lowest eigenvalues is found. $\mathbf{E}_N$ Contain the noise eigenvectors. **E** Can be written as E= $\{E_N | E_S\}$ where $\mathbf{E}_S$ are the signal eigenvectors.

The array manifold for a direction of arrival is the M×1column vector a(θ). It depends on the geometry. For example, for a ULA, it is given by

$$a^t(\theta) = [1, e^{-J\omega}, e^{-2J\omega}, \dots\dots, e^{-J(M-1)\omega}] \tag{2}$$

$$= \left[1, z^{-1}, z^{-2}, \dots\dots, z^{-(M-1)}\right] \tag{3}$$

$z = e^{j\omega}$ And ω is given by

$$\omega = \frac{2\pi d \sin\theta}{\lambda} \tag{4}$$

Where d is the spacing between the ULA elements and λ is the wavelength corresponding to the centre frequency of the narrowband signal. The *filtered* vector $\mathbf{V}_{MUSIC}$ which is analogous to the output of the spatial tuned filter as described in [8] is here given by

$$V_{MUSIC}(\theta) = a^t(\theta)\ E_N \tag{5}$$

Then, the *energy* in the filtered output is

$$E_{MUSIC}(\theta) = V_{MUSIC}(\theta)\ V^H{}_{MUSIC}(\theta) \tag{6}$$

$$= a^t(\theta)\ E_N\ a^*(\theta)\ E_N{}^H \tag{7}$$

For MUSIC, this energy is minimized, i.e., almost close to zero, when the angle corresponds to a DOA. Thus, the objective function for maximization is

$$J_N(\theta) = \frac{1}{E_{MUSIC}(\theta)} \tag{8}$$

It is interesting to note that the evaluation of $J_N(\theta)$ is linear in θ while nonlinear in ω unlike conventional frequency response. The spectrum can be thought of as the inverse of the magnitude squared of the frequency response of the filter given by (5). It is interesting to note that the evaluation of $J_N(\theta)$ is linear in θ while nonlinear in ω unlike conventional frequency response. The spectrum can be thought of as the inverse of the magnitude squared of the frequency response of the filter given by (5). The amplitude of the peaks of the MUSIC pseudo-spectrum is not related quantitatively to that of the corresponding component of the model because resulting peaks only serve to indicate precisely the position of sources. Qualitatively, if the amplitude so the SNR (Signal to Noise Ratio) is more important, the pseudo-spectrum will be less disrupted, resulting in a higher peak value. The amplitude or SNR can be obtained without difficulty by an optimization method of least squares. MUSIC algorithm does not allow obtaining directly the DOA of wave fronts. To know exactly the angles of arrival of the signals, we need to calculate an average over all vectors of an orthonormal basis of the noise space.

We estimate the number of signals, D, as the number of significant eigenvalues in L, i.e., the *cluster* of eigenvalues which carries maximum mean energy. The size of the cluster provides an estimate of D. GSU minimization method [3] finds the extreme values of a univariate function accurately. The amplitude of the peaks of the MUSIC [1] pseudo-spectrum is not related quantitatively to that of the corresponding component of the model because resulting peaks only serve to indicate precisely the position of sources. Qualitatively, if the amplitude so the SNR (Signal to Noise Ratio) is more important, the pseudo-spectrum will be less disrupted, resulting in a higher peak value. The amplitude or SNR can be obtained without difficulty by an optimization method of least squares.

IV. SIMULATION RESULTS

The first simulation shows how two signals are recognized by the MUSIC algorithm. There are two independent narrow band signals, the incident angle is 20° and 600 respectively, those two signals are not correlated, the noise is ideal Gaussian white noise, the SNR is 20dB, the element spacing is half of the input signal wavelength, array element number is 10, the number of snapshots is 200. The simulation results are shown in Figure 1:

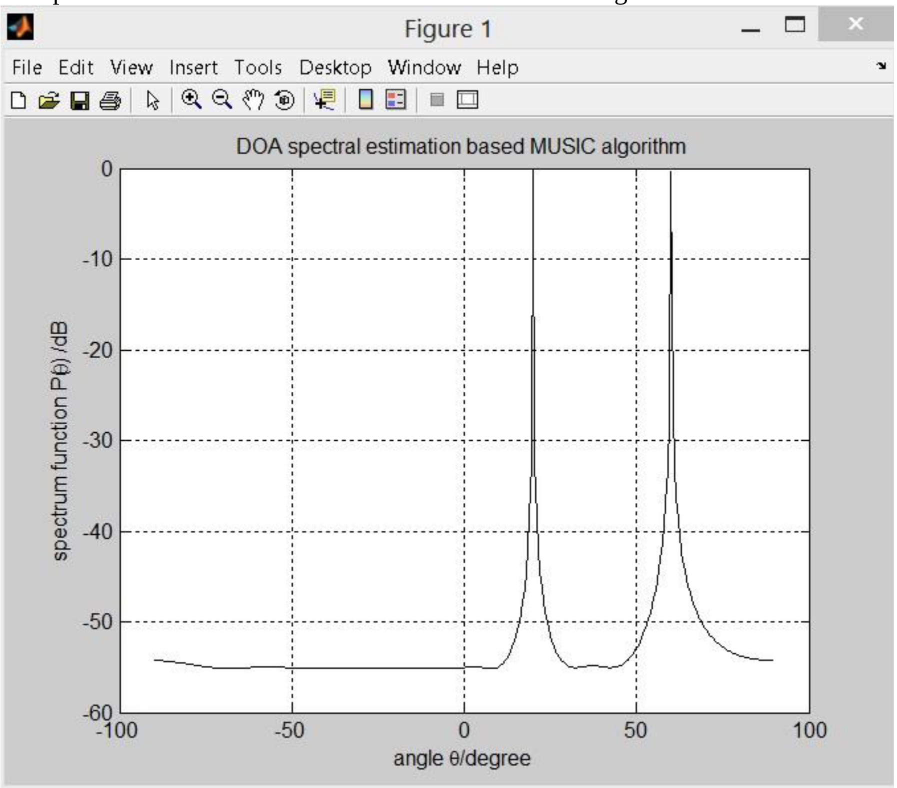

Figure1.Simulation for MUSIC algorithm when signals are non-coherent

When the signals are coherent, let the incident angle is 20° and 600 respectively, those two signals are not correlated, the noise is ideal Gaussian white noise, the SNR is 20dB, the element spacing is half of the input signal wavelength, array element number is 10, and the number of snapshots is 200. The simulation results are shown in Figure 2.

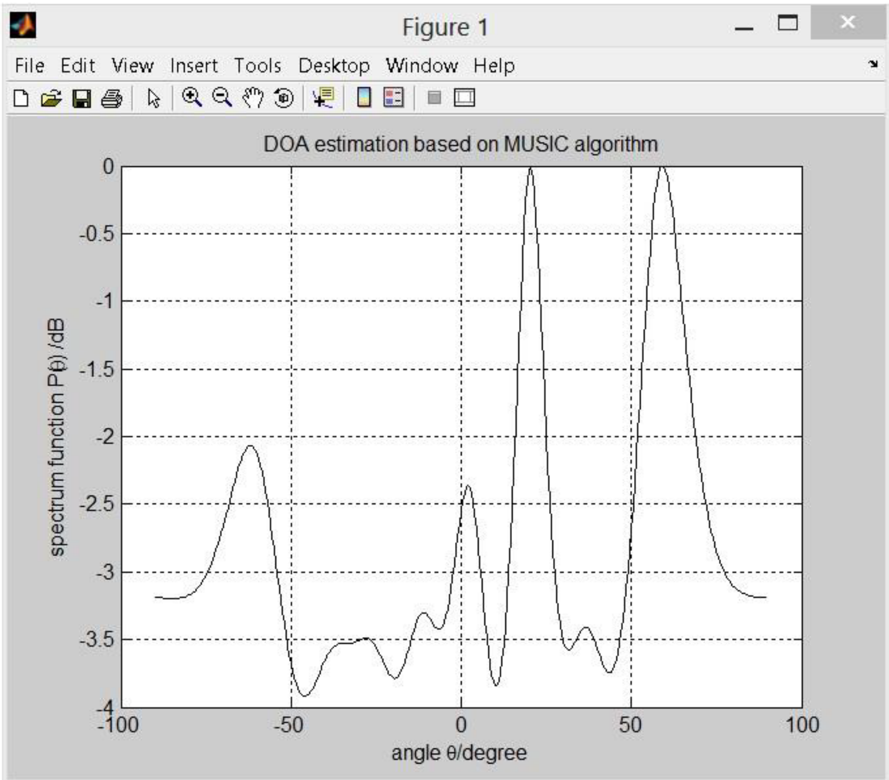

Figure2. Simulation for MUSIC algorithm when the signals are coherent

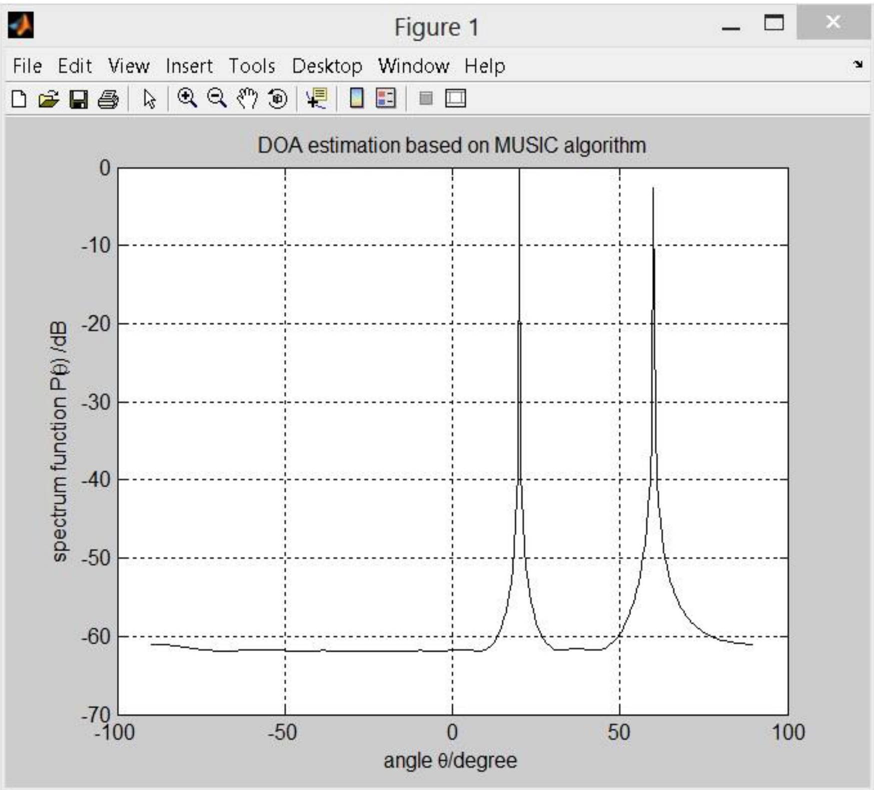

Figure3. Simulation for the GSU-MUSIC algorithm when the signals are coherent

As can be seen from Figure 2 and Figure 3, for coherent signals, classic MUSIC algorithm has lost effectiveness, while improved MUSIC algorithm can be better applied to remove the signal correlation feature, which can distinguish the coherent signals, and estimate the angle of arrival more accurately. Under the right model, using MUSIC algorithm to estimate DOA can get any high resolution. But MUSIC algorithm only focuses on uncorrelated signals; when the signal source is correlation signal, the MUSIC algorithm estimation performance deteriorates or fails completely. This GSU-MUSIC algorithm can make DOA estimation more complete, and have a marked effect both on theoretical and practical study. For signals to stay coherent, there are many jobs needed to be done with the realization of DOA estimation, and thus further research is needed.

V. CONCLUSION

The more the number of array elements, the more the number of snapshots; the more different between the incident angles, the higher resolution the MUSIC algorithm has. When the array element spacing is not more than half the wavelength, the resolution of MUSCI algorithm increases correspondingly with the increase of array element spacing; however, if the array element spacing is greater than half the wavelength, the spatial spectrum causes false peaks in other direction except the direction of signal source. When moving low SNR and small difference of incident angle, the performance of the MUSIC algorithm will decline. Some scholars have proposed some improvements in algorithm, but these problems are still a hot research topic. Hence, the MUSIC algorithm still has much room for development, and it is also worth further study.

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