Direction of Arrival Algorithm using GSU-minimization

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Abstract-A smart antenna is a digital wireless communications antenna system that takes advantage of diversity effect at the source (transmitter), the destination (receiver) or both. Diversity effect involves the transmission and/or reception of multiple radio frequency (RF) waves to increase data speed and reduce the error rate. A smart antenna enables a higher capacity in wireless networks by effectively reducing multipath and co-channel interference. This is achieved by focusing the radiation only in the desired direction and adjusting itself to changing traffic conditions or signal environments. Smart antennas employ a set of radiating elements arranged in the form of an array. The GSU-MUSIC algorithm for DOA estimation of smart antenna is similar to MUSIC and it uses iterative approach based on GSU minimization to find accurate values of the peaks. The GSU-MUSIC Algorithm overcomes the problems associated with previous techniques used for DOA estimation of smart antenna.

Keywords- *DOA(Directional of Arrival),Smart antenna,GSU (Gold Section Univariate) minimization,MUSIC (Multiple Signal Classification)*

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I. Introduction

Two of the main types of smart antennas include switched beam smart antennas and adaptive array smart antennas. Switched beam systems have several available fixed beam patterns. A decision is made as to which beam to access, at any given point of time, based upon the requirements of the system. Adaptive arrays allow the antenna to steer the beam to any direction of interest while simultaneously nullifying interfering signals. Beam direction can be estimated using the so-called Direction-of-Arrival (DOA) estimation methods [7]. The smart antenna system finds the direction of arrival of the signal, using Classical methods include Classical Beam former method and Capon's Minimum Variance Distortion less Response (MVDR) while Subspace based techniques are multiple signal classification (MUSIC) and The Minimum Norm Technique. They involve findings of a spatial spectrum of the antenna sensor array and calculating the DOA from the peaks of this spectrum. These calculations are complex. Matrix Pencil is very efficient in case of real time systems and for the correlated sources. Second type of smart antenna is adaptive array antenna. It is the method used to create the radiation pattern of the antenna array by adding constructively the phases of the signals in the direction of the targets/mobiles desired and nullifying the pattern of the targets/mobiles which are undesired/interfering targets.

II. Related Work

The MUSIC algorithm [1] for DOA estimation evaluates the MUSIC spectrum for various angles and chooses the maxima or peaks as the angles of arrival. The values obtained depend on the interval at which the spectrum is evaluated. The coarser the interval, the less accurate are the results in case of MUSIC. To improve accuracy and not depend on the interval, the Root-MUSIC [12] method is used. However, Root-MUSIC [12] is applicable, in its original form, only to uniform linear arrays (ULA). A. J. Weiss and B. Friedlander discussed on Direction finding for diversely polarized signals using polynomial rooting**.** A direction finding algorithm for diversely polarized arrays that is based on polynomial rooting is presented. Using polynomial rooting instead of search reduces significantly the computational requirements of the algorithm and enhances resolution. The basic method is limited to linear uniformly spaced arrays. However, using an interpolation technique, the algorithm is extended to a larger class of arrays. The performance of the proposed algorithm is analyzed. Computer simulations are used to demonstrate the performance of the new technique and to verify the performance analysis.

"ESPRIT method [5] studied by R. Roy and T. Kailath. An approach to the general problem of signal parameter estimation is described. The algorithm differs from its predecessor in that a total least-squares rather than a standard least-squares criterion is used. Although discussed in the context of direction-of-arrival estimation, ESPRIT [5] can be applied to a wide variety of problems including accurate detection and estimation of sinusoids in noise. It exploits an underlying rotational invariance among signal subspaces induced by an array of sensors with a translational invariance structure. The technique, when applicable, manifests significant performance and computational advantages over previous algorithms such as MEM, Capon's MLM, and MUSIC.

III. GSU Minimization method

The GSU-MUSIC algorithm proposed in this paper is a two-stage process. The first stage evaluates the objective function at coarse intervals and determines peaks followed by an iterative approach based on Gold-section univariate minimization [3] to find accurate values of the peaks. Richard Brent investigated the algorithm for minimization without derivatives [3]. This monograph describes and analyzes some practical methods for finding approximate zeros and minima of functions. If the number of peaks found so far is equal to the number of estimated peaks, the algorithm stops with this first stage. The second stage is an iterative step for fine resolution using finer intervals around the peaks found so far for finding peaks that were missing in previous iterations. This project also presents a method, based on a partitioning algorithm for estimating the number of emitters. The performance of GSU-MUSIC is described, including its advantages and comparison of time complexities for MUSIC, Root-MUSIC and GSU-MUSIC. The proposed algorithm gives good results even when the number of snapshots is small. This gives it an additional computational advantage. It does not compromise on the resolving power of MUSIC. In this paper, we present a variation on MUSIC. In this method, which we call GSU-MUSIC, the peaks of the MUSIC spectrum are evaluated quite accurately using Gold-section univariate (GSU) minimization.

IV. FLOWCHART

The steps involved in the MUSIC and GSU based MUSIC algorithm are explained in the flowchart given in figure 1.

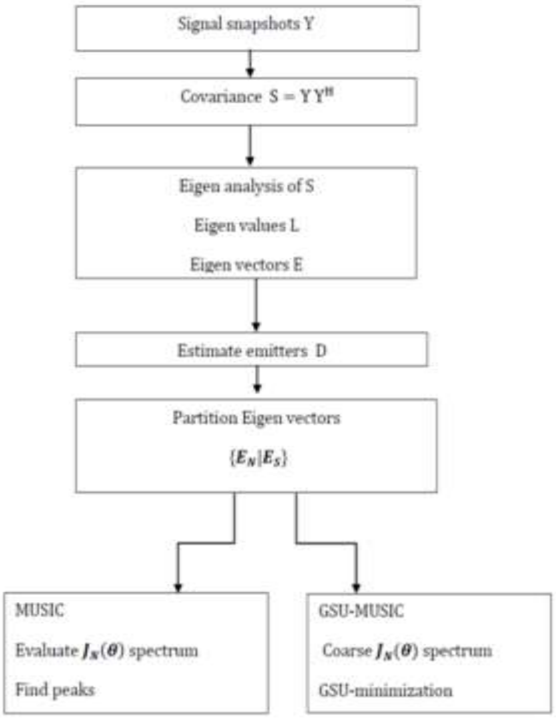

Fig.1 Steps in MUSIC and GSU based MUSIC algorithm

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 $\mathbf{Y}$ 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

$$S = YY^H \tag{1}$$

Where H denotes Hermitian. It has been shown that $\mathbf{L}$ will contain only positive real values as $\mathbf{S}$ 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 $\mathbf{L}$ 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. $\mathbf{E}$ 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

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

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

$z = e^{j\omega}$ 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.

V. MATLAB SIMULATION

The simulation results are as shown in fig .2 and fig.3.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. In other words, we have to calculate the pseudo-spectre on the extent of the parameters space and seek the minima of this function, which limits its performance in terms of speed and computational resources.

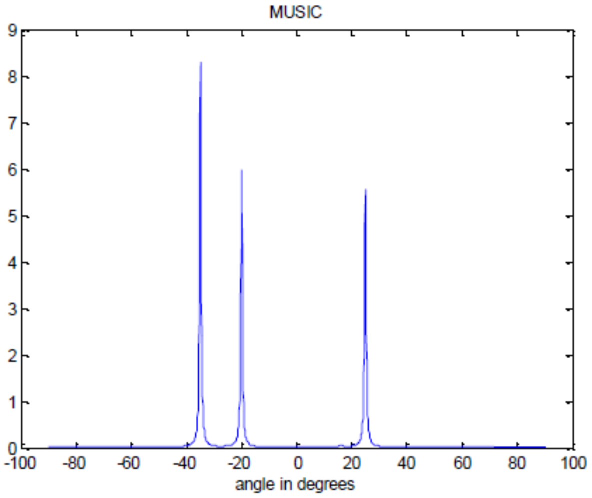

Fig.2 MUSIC algorithm

Several variants of MUSIC method have been proposed to reduce complexity, increase performance and resolution power.

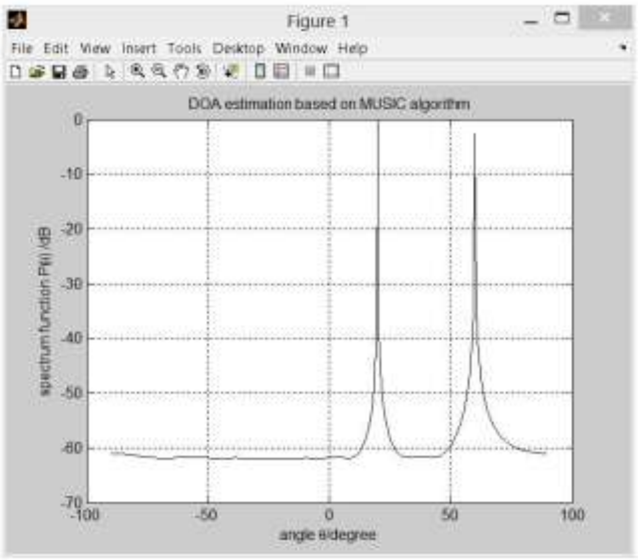

Fig.3 GSU-MUSIC algorithms

VI. CONCLUSION

MUSIC algorithm only focuses on uncorrelated signals; when the signal source is correlation signal, the MUSIC algorithm to stay coherent, there are many jobs needed to be done with the estimation performance deteriorates or fails completely. This improved MUSIC algorithm can make DOA estimation more complete, and have a marked effect both on theoretical and practical study. For signals realization of DOA estimation and thus further research is needed.

ACKNOWLEDGMENT

Completing a task is never a one-man effort. It is often a result of invaluable contribution of a number of individuals in direct or indirect manner. I express with all sincerity and deep sense of gratitude my indebtness to Prof.A.S.Deshpande who have given me many worthwhile suggestions and guided me by encouraging generously throughout the paper.

REFERENCES

[1] R. O. Schmidt, "Multiple emitter location and signal parameter estimation," IEEE Trans, vol. AP-34, no. 3, pp. 276-280, Mar. 1986.

[2] J. Capon, "High resolution frequency-wave number spectrum analysis," Proc. IEEE, vol. 57, no. 8, Aug. 1969.

[3] R. Brent, Algorithms for Minimization without Derivatives. Englewood Cliffs, NJ, USA: Prentice-Hall, 1973.

[4] J. Futz, A. Spanias, M.K. Banavar, Narrowband Direction of Arrival Estimation for Antenna Arrays (Morgan & ClaypoolPublishers, 2008).

[5] L.C. Godara, Application of antenna arrays to mobile communications. part II: beam forming and direction of arrival considerations, Proceedings of the IEEE, Volume 85, August

[6] M.R.Islam, I.A.H. Adam, Performance study of direction of arrival (DOA) estimation algorithms for linear array antenna,IEEE International Conference on Signal Processing Systems, pp.268–271, May 2009.

[7] L.C. Godara, Smart Antennas (CRC press, London, 2004).

[8] Antenna Theory: Analysis Design, Third Edition, by Constantine A. Balanis ISBN 0-471-66782-X Copyright © 2005 John Wiley & Sons, Inc.

[9] Shahriar Shirvani-Moghaddam, Sakineh Almasi-Monfared. A Comprehensive Performance Study of Narrowband DOA Estimation Algorithms. International Journal on Communications Antenna and Propagation (IRECAP). 2011; 1(4).

[10] H. Krim and M. Viberg, "Two decades of array signal processing research," IEEE Signal Processing Magazine, vol. 13, no. 4, pp. 67, July 1996.

[11] K. V. Rangarao and V. Shridhar, "An accurate numerical method for estimating the delay between two omni-directional receiving elements," in Proc. IEEE 30th Digital Avionics Syst. Conf., Seattle,WA, U.S.A., Oct. 2011, pp. 8C3-4–8C3-7.