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A Study of Music and LMS Algorithms for Smart Antenna System

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***Abstract*: This paper presents practical design of a smart antenna system based on direction-of-arrival estimation and adaptive beam forming. Direction-of-arrival (DOA) estimation is based on the MUSIC algorithm for identifying the directions of the source signals incident on the sensor array comprising the smart antenna system. Adaptive beam forming is achieved using the LMS algorithm for directing the main beam towards the desired source signals and generating deep nulls in the directions of interfering signals. The smart antenna system designed involves a hardware part which provides real data measurements of the incident signals received by the sensor array. Results obtained verify the improved performance of the smart antenna system when the practical measurements of the signal environment surrounding the sensor array are used. This takes the form of sharper peaks in the MUSIC angular spectrum and deep nulls in the LMS array beam pattern.**

Keywords — Smart antennas, DOA estimation, adaptive beam forming, least mean squares.

I. INTRODUCTION

Smart antennas (also known as adaptive array antennas, multiple antennas and recently MIMO) are antenna arrays with smart signal processing algorithms used 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/target. The antenna could optionally be any sensor.[1,2]

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.[3,4]

The definition of a *Smart antenna* is an antenna array system that is aided by a processing system that processes the signals received by the array or transmitted by the array using suitable array algorithms to improve wireless system performance. An antenna array consists of a set of distributed antenna elements (dipoles, monopoles or directional antenna elements) arranged in certain geometry (e.g., linear, circular or rectangular grid) where the spacing between the elements can vary.

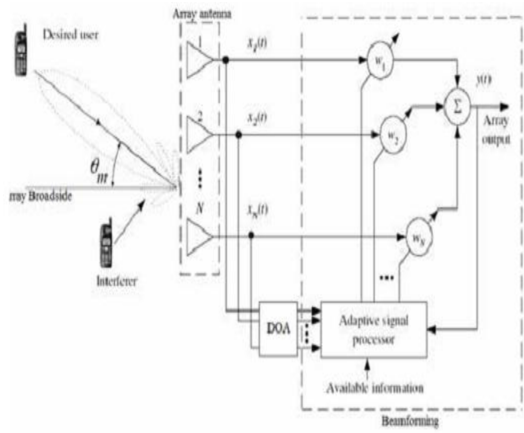


Fig. 1. A functional block diagram of a smart antenna system.

the desired signal strength and reduces the interference from other signals. Hence a smart antenna can be viewed as a combination of "regular or conventional" antenna elements whose transmit or received signals are processed using "smart" algorithms.Generic implementation of Smart Antenna System, shows a generic implementation smart antenna system.[5]

The antenna arrays have input or output as RF signals in the analog domain. These signals are passed to/from the RF analog front end which usually consists of low noise amplifiers, mixers and analog filters. In the receive mode, the RF signals are converted to digital domain by analog to digital converters (ADCs) and in transmit mode, the base band digital signals are converted to RF using digital to analog converters (DACs). The down-conversion from RF to base band or up-conversion from base band to RF can involve the use of IF signals. The base band signals received from each antenna is then combined using the "smart" algorithms in a digital processing section. Each antenna element hence has a RF chain going from the antenna element to RF front end to digital conversion for receiver and vice-versa for transmitter.

II. TYPES OF SMART ANTENNAS

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 in time, based upon the requirements of the system. Adaptive arrays allow the antenna to steer the beam to any direction of interest while simultaneously nulling interfering signals. Beamdirection can be estimated using the so-called direction-of-arrival (DOA) estimation methods.

In 2008, the United States NTIA began a major effort to assist consumers in the purchase of digital television converter boxes.[6] Through this effort, many people have been exposed to the concept of smart antennas for the first time. In the context of consumer electronics, a "smart antenna" is one that conforms to the EIA/CEA-909 Standard Interface.

III. DOA ESTIMATION ALGORITHMS

The DOA algorithms are classified as quadratic type and subspace type. The Barltett and Capon (Minimum Variance Distortion less Response) are quadratic type algorithms.. The both methods are highly dependent on physical size of array aperture, which results in poor resolution and accuracy, Subspace based DOA estimation method is based on the Eigen decomposition. The subspace based DOA estimation algorithms MUSIC and ESPRIT provide high resolution; they are more accurate and not limited to physical size of array aperture. The various DOA algorithm performance is analyzed based on number of snapshots, number of users, user space distribution, number of array elements, SNR and MSE.

IV. IMPLEMENTATION OF ALGORITHM

The high demand on the usage of the wireless communication system calls for higher system capacities. The system capacity can be improved either enlarging its frequency bandwidth or allocating new portion of frequency spectrum to wireless services. But since the electromagnetic spectrum is a limited resource, it is not easy to get new spectrum allocation without the international coordination on the global level. One of the approaches is to use existing spectrum more efficiently, which is a challenging task. Efficient source and channel coding as well as reduction in transmission power or transmission bandwidth or both are possible solutions to the challenging issue. With the advances in digital techniques, the frequency efficiency can be improved by multiple access technique (MAT), which gives mobile users access to scarce resource (base station) and hence improves the system's capacity. Family of existing Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA) and Code Division Multiple Access (CDMA) can be enlarged by adding a new parameter, space or angle [7], which results in MAT known as „Space Division Multiple Access‟ (SDMA). At the receiver's side, the transmitted signal is received with its multipath components plus interferers‟ signal, as well as with present noise. Thus, detection of the desired signal is a challenging task. The Smart Antenna System (SAS) employs the antenna elements and the digital signal processing which enables it to form a beam to a desired direction taking into account the multipath signal components. In this way, Signal-to-Interference-and-Noise Ratio (SINR) improves by producing nulls towards the interferers Signal-Of–No-Interest (SONI).The performance of SAS greatly depends on the performance on DOA estimation.

In the project we are investigating the performance of simulated MUSIC,MUSIC algorithms with MATLAB as a tool. The performance of these algorithms is analyzed by considering parameters like number of array elements, user space distribution, number of snapshots, signal to noise ratio, Mean Square Error (MSE), which results in optimum array design.

V. MUSIC ALGORITHMS INTRODUCTION

MUSIC stands for Multiple Signal Classification, one of the high resolution subspace DOA algorithms, which gives the estimation of number of signals arrived, hence their direction of arrival. MUSIC deals with the decomposition of covariance matrix into two orthogonal matrices, i.e., signal-subspace and noise-subspace. Estimation of DOA is performed from one of these subspaces, assuming that noise in each channel is highly uncorrelated. This makes the covariance matrix diagonal. The covariance matrix is given by:

$$Sx = F(\theta)S_s F(\theta)^H + \sigma^2_w I$$

Where $F(\theta) = [F(\theta_1): F(\theta_2):\ldots\ldots\ldots.: F(\theta_D)]$

is a MxD array steering matrix. is noise variance and I is an identity matrix of size MxM.

Writing the spatial covariance matrix in terms of Eigen values and eigenvectors gives:

$$Sx = \sum_{i=1}^{M} Pi\ \Phi i\ \Phi^H_i$$

The noise subspace eigen values and eigenvectors are

$Pi = \; i = D+1, D+2, \ldots\ldots M$

$\Phi i = \; i = D+1, D+2, \ldots\ldots M$

The noise subspaces can be written in the form of MX (M-D):

$U_N = [\Phi_{D+1}, D+2, \ldots\ldots\ldots \Phi_M]$

Above Equation indicates that we can find out the desired value DOA of teta1, teta2,......, D by finding a set of vectors that span and projecting array manifold matrix onto for all values of and evaluating the D values of , where the projection is zero .

$\|fi^H U_N\|2 = 0 \qquad i = 0,1,2,\ldots\ldots D$

The MUSIC Pseudospectrum is given as,

$Pmu(\theta) = 1 / abs[F(\theta)^H U_N U_N^H F(\theta)]$

VI. MUSIC ALGORITHM

The multiple signal classification (MUSIC) method [Sch86] is a relatively simple and efficient Eigen structure variant of DOA estimation methods. It is perhaps the most studied method in its class and has many variations. Some of these are discussed in this section.

VII. SPECTRAL MUSIC

In its standard form, also known as spectral MUSIC, the method estimates the noise subspace from available samples. This can be done either by Eigen value decomposition of the estimated array correlation matrix or singular value decomposition of the data matrix with its N columns being the N array signal vector samples, also known as snapshots.

Once the noise subspace has been estimated, a search for M directions is made by looking For steering vectors that are as orthogonal to the noise subspace as possible. This is normally accomplished by searching for peaks in the MUSIC spectrum given by

$P_{MU}(\theta) = 1 / |S_\theta^H \; U_N|^2$

where U_N denotes an L by L – M dimensional matrix, with L – M columns being the eigenvectors corresponding to the L – M smallest eigenvalues of the array correlation matrix and $\mathbf{S}_\theta$ denoting the steering vector that corresponds to direction θ.It should be noted that instead of using the noise subspace and searching for directions with steering vectors orthogonal to this subspace, one could also use the signal subspace and search for directions with steering vectors contained in this. This amounts to searching for peaks in

$P_{MU}(\theta) = |U^H_S \; S_\theta|^2$

Where , U_S denotes an L × M dimensional matrix with its M columns being the eigenvectors corresponding to the M largest eigenvalues of the array correlation matrix.

It is advantageous to use the one with smaller dimensions. For the case of a single source, the DOA estimate made by the MUSIC method asymptotically approaches the Cramer–Rao lower bound, that is, where the number of snapshots increases infinitely, the best possible estimate is made. For multiple sources, the same holds for large SNR cases, that is, when the SNR approaches infinity [Fri90, Por88]. The Cramer–Rao lower bound (CRLB) gives the theoretical lowest value of the covariance for an unbiased estimator.

In, an application of the MUSIC algorithm to cellular mobile communications was investigated to locate land mobiles, and it is shown that when multipath arrivals are grouped in clusters the algorithm is able to locate the mean of each cluster arriving at a mobile. This information then may be used to locate line of sight.

VIII. ROOT-MUSIC

For a uniformly spaced linear array (ULA), the MUSIC spectra can be expressed such that the search for DOA can be made by finding the roots of a polynomial. In this case, the method is known as root-MUSIC [Bar83]. Thus, root-MUSIC is applicable when a ULA is used and solves the polynomial rooting problem in contrast to spectral MUSIC's identification and localization of spectral peaks. Root-MUSIC has better performance than spectral MUSIC.

IX. CONSTRAINED MUSIC

This method incorporates the known source to improve estimates of the unknown source direction. The situation arises when some of the source directions are already known. The method removes signal components induced by these known sources from the data matrix and then uses the modified data matrix for DOA estimation. Estimation is achieved by projecting the data matrix onto a space orthogonal complement to a space spanned by the steering vectors associated with known source directions. A matrix operation, the process reduces the signal subspace dimension by a number equal to the known sources and improves estimate quality, particularly when known sources are strong or correlated with unknown sources.

X. Beam Space Music

The MUSIC algorithms discussed so far process the snapshots received from sensor elements without any preprocessing, such as forming beams, and thus may be thought of as element space algorithms, which contrasts with the beamspace MUSIC algorithm in which the array data are passed through a beamforming processor before applying MUSIC or any other DOA estimation algorithms. The beamforming processor output may be thought of as a set of beams; thus, the processing using these data is normally referred to as beamspace processing. A number of DOA estimation schemes are discussed in where data are obtained by forming multiple beams using an array.

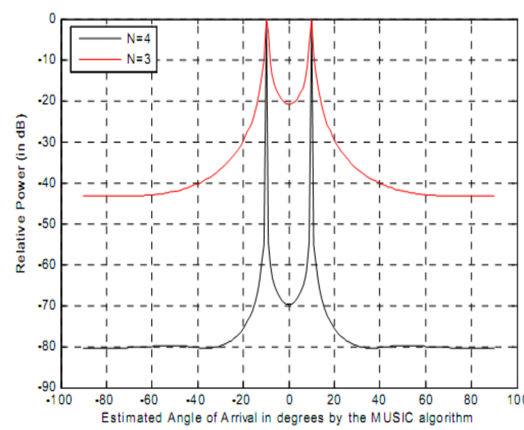
The DOA estimation in beam space has a number of advantages such as reduced computation, improved resolution, reduced sensitivity to system errors, reduced resolution threshold, reduced bias in the estimate, These advantages arise from the fact that a beamformer is used to form a number of beams that are less than the number of elements in the array; consequently, less data to process a DOA estimation are necessary.This process may be understood in terms of array degrees of freedom. Element space methods have degrees of freedom equal to the number of elements in the array, whereas the degrees of freedom of beamspace methods are equal to the number of beams formed by the beamforming filter. Thus, the process reduces the array's degrees of freedom. Normally, only M + 1 degree of freedom to resolve M sources are needed. The root-MUSIC algorithm discussed for the element space case may also be applied to this case, giving rise to beamspace root-MUSIC. Computational savings for this method are the same as for beamspace methods compared to element space methods in general.

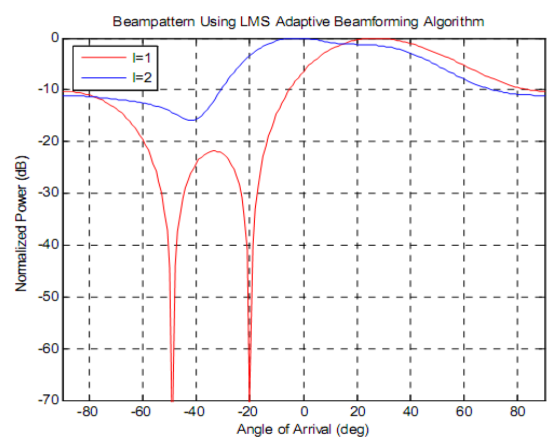
XI. Least Mean Squares Algorithm

Application of least mean squares (LMS) algorithm to estimate optimal weights of an array is widespread and its study has been of considerable interest for some time. The algorithm is referred to as the constrained LMS algorithm when the weights are subjected

to constraints at each iteration, whereas it is referred to as the unconstrained LMS algorithm when weights are not constrained at each iteration. The latter is applicable mainly when weights are updated using reference signals and no knowledge of the direction of the signal is utilized, as is the case for the constrained case. The algorithm updates the weights at each iteration by estimating the gradient of the quadratic MSE surface, and then moving the weights in the negative direction of the gradient by a small amount. The constant that determines this amount is referred to as the step size. When this step size is small enough, the process leads these estimated weights to the optimal weights. The convergence and transient behavior of these weights along with their covariance characterize the LMS algorithm, and the way the step size and the process of gradient estimation affect these parameters are of great practical importance. LMS algorithm performance using the gradient estimate by this method can be analyzed using an approach similar to that used in previous sections. However, the results on the mean and covariance of the gradient, and the covariance of the weights and misadjustments are stated in this section. The method described in this section is for updating weights of the constrained optimal beamformer. The methods applicable to other processors can easily be developed using a similar approach. The method uses orthogonal sequences to perturb the weights of the processor, and then measures the output power of the processor to estimate the required gradient. The LMS algorithm using the gradient estimated by this method is referred to as perturbation Algorithm

XII. Simulation





CONCLUSION

This paper presented a practical setup of a smart antenna system for the performance evaluation of the MUSIC DOA estimation and LMS adaptive beam forming algorithms. The performance evaluation process is based on two steps. First, a hardware part is implemented used to collect real data measurements of the signals incident on the smart antenna sensor array. Second, the measured data is processed in Mat lab which is used to predict the performance of the smart antenna algorithms being investigated. Results obtained verify the improved performance of the smart antenna system when the real data measurements of the signal environment surrounding the sensor array are used. This takes the form of sharper peaks in the MUSIC angular spectrum indicating locations of desired users and deep nulls in the LMS array beam pattern indicating the location of the undesired interference signals.

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