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Design and Simulation of Smart Antenna Array Using Adaptive Beam forming Method

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Abstract— ***This paper presents design of a smart antenna system based on direction-of-arrival estimation and adaptive beam forming. In order to obtain the antenna pattern synthesis for a ULA (uniform linear array) we use Dolph Chebyshev method. For the estimation of Direction-of-arrival (DOA) MUSIC algorithm (Multiple User Signal Classification) is used in order to identify the directions of the source signals incident on the antenna array comprising the smart antenna system. LMS algorithm is used for adaptive beam forming in order to direct the main beam towards the desired source signals and to adaptively move the nulls towards the unwanted interference in the radiation pattern. Thereby increasing the channel capacity to accommodate a higher number of users. The implementation of LMS algorithm helps in improving the following parameters in terms of signal to noise ration, convergence rate, increased channel capacity, average bit error rate. In this paper, we will discuss and analyze about the DolphChebyshev method for antenna pattern synthesis, MUSIC algorithm for DOA (Direction of Arrival) estimation and least mean squares algorithm for Beam forming.***

Index Terms—**Uniform Linear Array (ULA), Direction of Arrival (DOA), Multiple User Signal Classification (MUSIC), Least Mean Square (LMS).**

I. INTRODUCTION

A smart antenna has the ability to diminish noise, increase signal to noise ratio and enhance system competence [1]. The diversity effect in smart antenna refers to the transmission and/or reception of manifold RF-waves to increase the data speed as well as to diminish the error rate [2,3].

Smart antennas are considered as an effectual counter measures to boost channel bandwidth and capacity as well as to reduce the channel interference since they provide wide bandwidth less electromagnetic interruption, flexibility, less weight, phase control independent of frequency and low propagation loss [4]. The smart antenna concept is applied to different types of antenna arrays. The smart antenna system measures the direction of arrival of the signal. It focuses on identifying a spatial spectrum of the antenna array, and estimation the DOA from the peaks of the spectrum [5]. The smart antenna technique allows main lobe positioning towards preferred direction while manipulating the nulls resulting in an increased signal to noise ratio. Using beam forming technique at the receiver, two or more transmitters can share the same traffic channel to keep in contact with base station at the same time. An adaptive antenna array is employed at the base station to create numerous antenna beams simultaneously. Each beam captures one transmitter by automatically pointing its pattern towards that transmitter while nulling other co-channel transmitters and multipath signals [6].

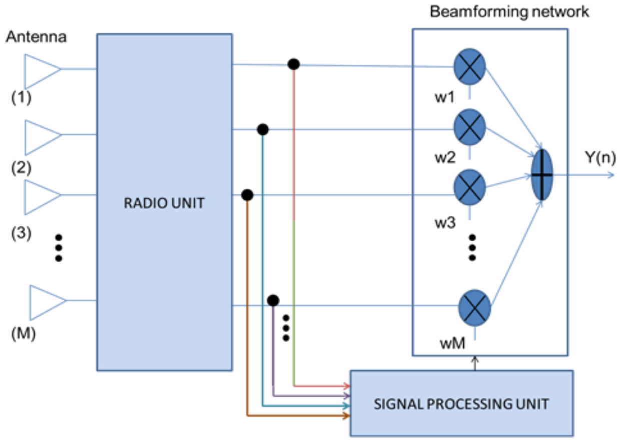

Fig 1: Functional Block Diagram of Smart Antenna System

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antenna array shown in fig(1) consists of a set of antenna sensors which are combined together in a particular geometry which may be linear, circular, planar, and conformal arrays commonly [7]. ULA is the most common geometry for smart antennas because of its simplicity, excellent directivity and production of the narrowest main lobe in a given direction in comparison to the other array geometries [8]. In a ULA, the elements are aligned along a straight line and with a uniform inter-element spacing usually $d=\lambda/2$, where λ denotes the wavelength of the received signal. If $d < \lambda/2$ mutual coupling effects cannot be ignored and the AOA estimation algorithm cannot generate desired peaks in the angular spectrum. On the other hand, if $d > \lambda/2$, then the spatial aliasing leads to misplaced or unwanted peaks in the spectrum. As so, $d = \lambda/2$ is the optimum inter element spacing in the ULA configuration. In this section a simple ULA is proposed to improve AOA estimation accuracy at the broadside angle.

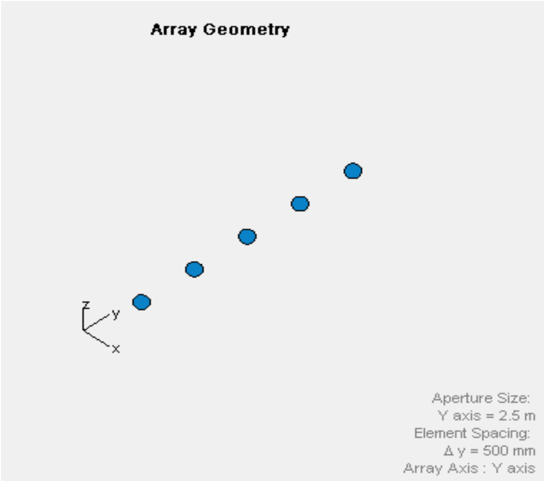

Fig 2: Uniform Linear Array Geometry

Here in fig (2), the ULA geometry shows 5 antenna element array geometry with a spacing of 0.5m between each element arranged along the Yaxis.

II. DOLPH CHEBYSHEV ARRAY PATTERN SYNTHESIS

In order to obtain a pencil beam pattern, we need the narrowest possible main beam for a given maximum side lobe level. Usually the antenna array factors for arrays having uniform weights have unequal side lobe levels. For an optimal pattern, it is desirable that all the side lobes have equal magnitude. This can be done by increasing the lower side lobe slightly and reducing the larger side lobes which decreases the overall maximum side lobe level. The DolphChebyshev method is a popular weighting method used for obtaining the weights and current excitations for uniformly spaced linear arrays steered to broadside. The main advantage of using this method for pattern synthesis is that the side lobe levels can be specified and the minimum null-null beam width can be obtained.

A class of polynomials called chebyshev polynomials (with known coefficients) are used to match them to the array factor (the unknown coefficients being the weights). These polynomials all have equal ripples with a peak magnitude of 1.0 in the range [-1,1]. Mathematically the chebyshev polynomials are defined by the equation shown below:

$$T_n(x) = \cos[n\cos^{-1}(x)] \tag{1}$$

The first few Chebyshev polynomials are as follows:

$T_0(x)=1$

$T_1(x)=x$

$T_2(x)=2x^2-1$

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$T_3(x)=4x^3-3x$

$T_4(x)=8x^4-8x^2+1$

$T_5(x)=16x^5-20x^3+5x$

In order to use the Chebyshev polynomials, as array factor, the change has to be done from the far field angle of the pattern to that of the argument x of the polynomials.
For an N array element, the array factor can be of the form :

$$AF=\sum_{n=1}^{M} a_n \cos[(2n-1)u] \quad \text{(even array)} \qquad (2)$$

$$AF=\sum_{n=1}^{M} a_n \cos(2nu) \quad \text{(odd array)} \qquad (3)$$

Where $u = kd\cos\theta/2$, $k=2\pi/\lambda$ and a_n represents the excitation coefficients. In order to obtain the radiation pattern, we determine the array element excitations that produces an array factor given by the equation (2) and (3).

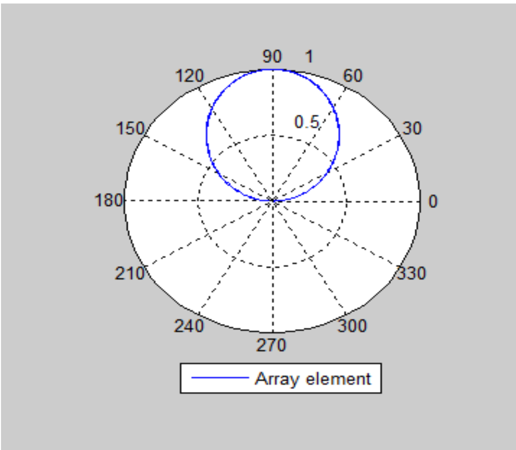

Fig 3: Array element for a 5 element ULA with a spacing of 0.5 m

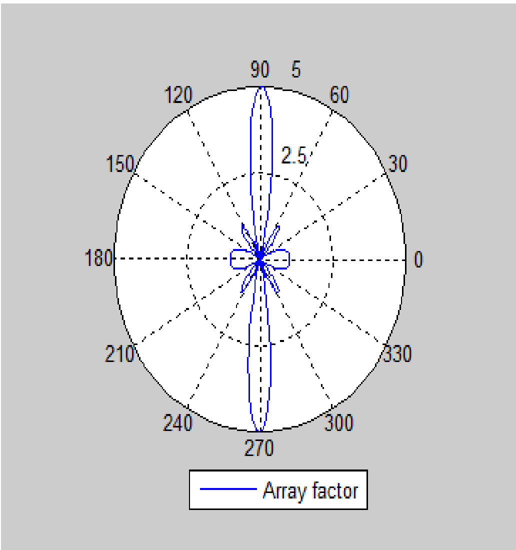

Fig 4: Array factor for a 5 element ULA with a spacing of 0.5 m

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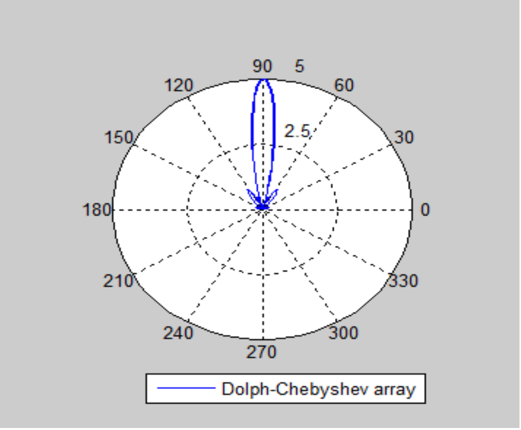

Fig 5: Dolph Chebyshev array pattern synthesis for a 5 element ULA with a spacing of 0.5 m

III. SIGNAL MODEL FOR ULA CONFIGURATION

After finding the radiation pattern, we consider the uniform linear array consisting of N number of antenna elements and let it receive M narrow band source signals $S_m(t)$ from the desired users arriving at direction $\theta_1,\theta_2,....\theta_M$. The array also receives I source signal $S_i(t)$ from undesired users arriving at directions $\theta_1,\theta_2,\ldots\theta_I$. At a particular instant of time t=1,2,…K, where K is the total number of snapshots taken. The desired users signal vector $X_s(t)$ can be defined as

$$X_M(t)=\sum_{m=1}^{M} a(\theta_m) s_m(t) \tag{4}$$

Where $a(\theta_m)$ is the N X 1 array sterrng vector which represents the array response at direction θ_m and is given by

$$a(\theta_m) = [\exp[j(n-1)\Psi_m]]^T; \qquad 1 \le n \le N \tag{5}$$

Where $[\ ]^T$ transposition operator, and Ψ_m represents phase shift from element to element along the array. This can be defined by

$$\Psi_m = 2\Pi(\frac{d}{\lambda})\sin\theta_m \tag{6}$$

Where d is the inter element spacing and λ is the wavelength of the received signal.
The desired users signal vector $X_M(t)$ can be written as

$$X_M(t) = A_M\, s(t) \tag{7}$$

Where $A_M(t)$ is N X M matrix of the desired user signal direction vectors and is given by

$$A_M = [a(\theta_1), a(\theta_2),\ldots, a(\theta_M)] \tag{8}$$

and s(t) is the M X 1 desired user signal source waveform vector defined as

$$s(t) = [\, s_1(t), s_2(t),\ldots,s_M(t)]^T \tag{9}$$

The undesired users signal vector $X_I(t)$ is given by

$$X_I(t) = A_I\, i(t) \tag{10}$$

Where A_I is the N X I matrix of undesired users signal direction vectors defined as

$$I(t) = [\, i_1(t), i_2(t),\ldots,i_I(t)]^T \tag{11}$$

Hence the overall received signal vector X(t) can be written as

$$X(t) = X_M(t) + n(t) + X_1(t) \tag{12}$$

where n(t) which represents white Gaussian noise.
Thus, the spatial correlation matrix of the received signals R_{xx} is defined as

$$R_{xx}= E\{\, X(t)\, X^H(t)\} \tag{13}$$

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where E{.} represents ensemble average and $(.)^H$ is the Hermitian transposition operator.
Substituting for X(t) from (12) in (13) gives

$$R_{xx}= A_M R_{ss} A_M^H + n(k)^H + A_I R_{ii} A_I^H \quad (14)$$

where $R_{ss}= E\{s(t)\, S^H(t)\}$ is a M X M desired users source waveform covariance matrix; $R_{ii} = E\{i(t)\, i^H(t)\}$ is an I X I undesired users source waveform covariance matrix.

IV. MUSIC ALGORITHM

DOA estimations are used to calculate the direction of the desired user signal. They are classified as conventional beam forming techniques, subspace-based techniques, and maximum likelihood techniques. A common DOA estimation algorithm is MUSIC (Multiple Signal Classification) [9]. The assumption that MUSIC makes is that the noise in the channel is uncorrelated. The Eigen values and vectors of array correlation matrix R_{xx}, are of prime importance as the pseudo spectrum function of the MUSIC algorithm mainly depends on them. The first equation used in this algorithm is the covariance matrix R_{xx}, obtained from previous equations , has *M* signal Eigen values ([10],[11]) with its corresponding eigenvectors given as:

$$V_S = [v_1 , v_2 ,\ldots\ldots,v_M] \quad (15)$$

The covariance matrix R_{xx}has $N - M$ remaining eigenvalues which further represent noise eigenvalues with corresponding eigenvectors as follows:

$$V_n = [v_{M+1} ,v_{M+2} ,\ldots\ldots,v_N] \quad (16)$$

The normalized MUSIC angular spectrum is defined as ([12],[13]):

$$P(\theta) =\frac{A^H A}{A^H V_n V_n^H A} \quad (17)$$

In equation (17), in the MUSIC angular spectrum, it is evident that the peaks occur at angles θ and accordingly the array manifold matrix A is orthogonal to E_n, which is the noise subspace matrix. The desired directions-of arrival is defined by the angles θ, formed by the signals impinging on the sensor array.

The number of elements in the sensor array restricts the number of signals that have the probability of being detected. In [10] and [11], upon verification, it was found that an N element sensor array can detect up to N-1 uncorrelated signals. The graphical representation of the DOA analysis using MUSIC algorithm is obtained and displayed in fig (6).

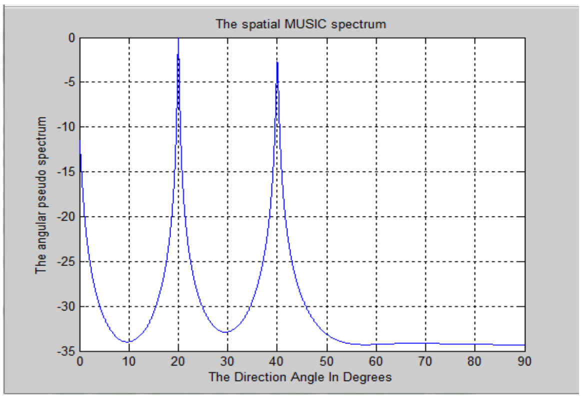

Fig 6: MUSIC pseudo spectrum for two desired users placed at 20^0 and 40^0

From the above graphical representation it can be concluded that when there is an increase in the angular form between two desired signal sources, the results obtained are much better.

V. ADAPTIVE BEAMFORMING USING LMS ALGORITHM

After the DOA estimation the next step of signal processing algorithms is beam forming. Beam forming can be referred as spatial filtering because it filters out some of the incoming signals from different spatial direction, while it amplifies the other signals. Upon the estimation of the angles of arrival for desired and undesired users

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the next step is to adaptively move the main beam towards the desired user and place the nulls towards the interferers.

The first step for beam forming is to find the complex weights vector W which can be obtained using adaptive beam forming algorithm. LMS (Least Mean Square) is an adaptive weighted beam forming radiant weights method. The output from the array antenna Y(n) is compared with the reference signal d(n) which is identical to the desired signal. The main objective is to minimize the error between the desired signal and the reference signal. The output of the uniform linear antenna array is given by

$$Y(n)=W^H X(n) \qquad (18)$$

Where W is the complex weights vector and X is the received signal vector given in (12).
The error signal is given as

$$e(n) = d(n) - W^H X(n) \qquad (19)$$

This error signal e(n) is used by the beam former to adaptively adjust the complex weights vector W so that the MSE(mean squared error) is minimized. The LMS algorithm which is based on the steepest descent method recursively computes and updates weights vector W. By doing so successive correction to the weights vector W results in minimum mean square error. The weights vector W is initialized arbitrarily and updated using the following LMS equation.

$$W(n+1) = W(n) + \mu X(n)\, e^*(n) \qquad (20)$$

Where W(n+1) denotes the weights vector to be computed at iteration n+1 and μ is the LMS step size which is related to the convergence rate. The convergence rate describes how fast the LMS reaches the steady state. The adaptive step size should be within the range specified as

$$0 < \mu < (1/\lambda_{max}) \qquad (21)$$

Where λmax is the maximum Eigen values of the input covariance matrix R_{xx}obtained in equation (13).

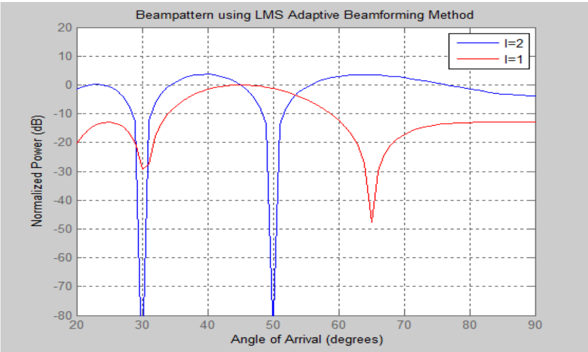

Fig 7: Beam forming using LMS Adaptive Bam forming Method

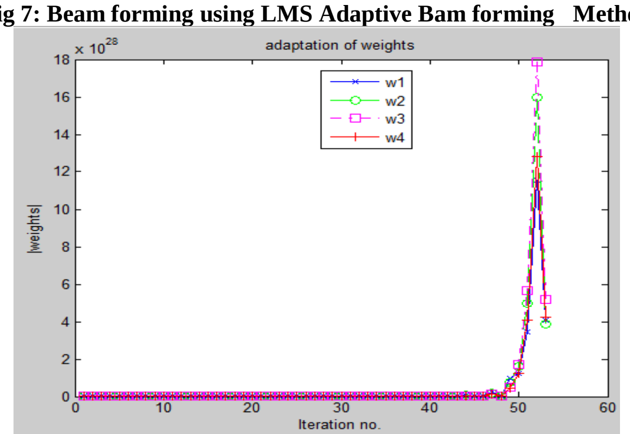

Fig 8: Plot displaying the adaptation of weights using LMS algorithm

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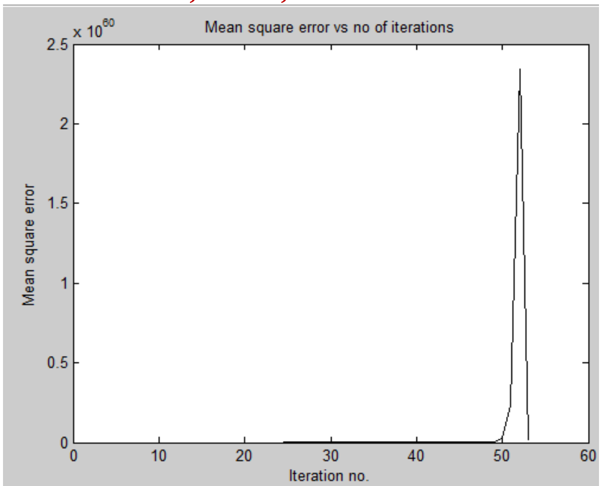

Fig 9: Plot between the mean square error and the number of iteration the algorithm takes to obtain the desired signal

Some of the important features of LMS algorithm is its simplicity in implementation, stable and robust performance against different signal conditions, slow convergence (due to eigen value spread), MSE behavior of the LMS algorithm.

VI. PERFORMANCE METRICS

A. Bit Error Rate

Bit error rate is defined as the number of bit errors divided by the total number of bits transferred during a particular time interval. It is a unit less performance measure, often expressed in percentages. Due to co-channel interference and Raleigh scattering there is a consequence of reduction in signal power and multipath fading of signals can occur. This causes the bit error rate to increase gradually. In order to decrease the bit error rate we use various equalizers in the receiver section of the smart antenna system.

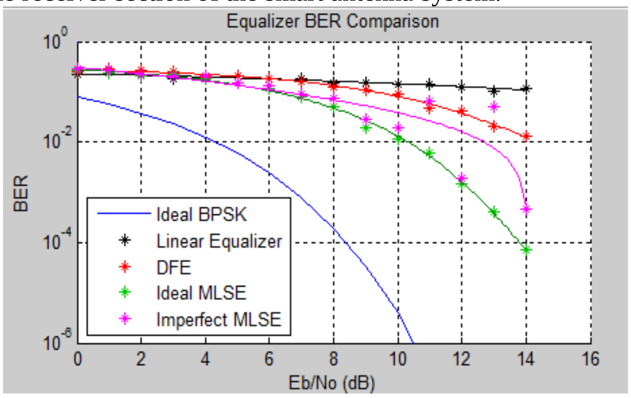

Fig 10: Plot showing the equalizer BER comparison

B. Signal to noise ratio

Signal to Noise ratio can be defined as ratio of the signal power to the noise power. The standard unit for signal to noise ratio is measured in decibels. The Signal to Noise ratio is improved using LMS algorithm.

VII. CONCLUSION

This paper presents a setup of a smart antenna system for the performance analysis of the MUSIC DOA estimation and LMS adaptive beam forming algorithm. The performance analysis process is done in two steps. Firstly, the collection of real data measurements of the signals impinging on the smart antenna array. Secondly, the measured data is processed using Matlab8.1(R2013a) in order to predict the performance of the smart antenna algorithms being examined. Instead of using ULA geometry, an UCA (uniform circular array) can be

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used for antenna pattern synthesis. The various antenna pattern synthesis other than DolpChebyshev method which can be used includes Taylor method, Orchard's method or Fourier series method. For DOA estimation MVDR (minimum variance distortion less response) algorithm can be also used. Also instead of using the standard LMS beam forming algorithm, the algorithms like NLMS (normalized least mean square) or RLS (recursive least square) can be implemented for the adaptation of weights to minimize the MSE and to obtain a radiation pattern that has a beam in the direction ofthe source that is transmitting the reference signal, while the nulls in the radiation pattern is directed towards the interference. An extended work has to be done on increasing the channel capacity of the smart antenna system by designing a channel model thereby increasing the performance of the system antenna system.

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