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A Novel Approach for Interference Suppression Using a Improved LMS Based Adaptive Beam forming Algorithm

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ABSTRACT

A novel adaptive beam forming technique is proposed for wireless communication applications based on the minimum bit error rate (MBER) criterion known as LMS algorithm.

LMS (Least Mean Square) algorithm is used for steering the antenna beam electronically. Using the Rectangular, Hamming, Kaiser, Chebyshev windows both the block-data and sample-by-sample adaptive implementations of the MBER solution are developed. By making use of window techniques half power beam width of an antenna is enhanced using Matlab simulation. The gain of the system will definitely improve the performance of CDMA based system, where the number of interferes is quite large and helps to increase the spectral efficiency of wireless communication systems. Any beam former that can depress the large number of interferers will improve the capacity and performance. Such beam formers are called smart antennas. They improve signal to interference ratio (SIR) of the communication system efficiently by forming narrow beam towards desired user and low side towards undesired users. Smart antennas offer a broad range of ways to improve wireless system performance.

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I. INTRODUCTION

Throughout the world, there is significant research and development on smart antennas for wireless systems. Smart antenna systems attract lot attentions now and believably more in the future, as it can increase the capacity of mobile communication systems dramatically [4].This is because smart antennas have tremendous potential to enhance the performance of future generation wireless systems as evidenced by the antennas' recent deployment in many systems. There are two basic types of smart antennas. The first type is the phased array or multi beam antenna, which consists of either a number of fixed beams with one beam turned on towards the desired signal or a single beam (formed by phase adjustment only) that is steered toward the desired signal. The other type is the adaptive antenna array is an array of multiple antenna elements, with the received signals weighted and combined to maximize the desired signal to interference plus noise power ratio [1].

It consists of a uniform linear antenna array for which the current amplitudes are adjusted by a set of complex weights using an adaptive beam forming algorithm. The adaptive beam forming algorithm optimizes the array output beam pattern such that maximum radiated power is produced in the directions of desired mobile users and deep nulls are generated in the directions of undesired signals representing co-channel interference from mobile users in adjacent cells. Prior to adaptive beamforming, the directions of users and interferes must be obtained using a direction-of- arrival (DOA) estimation algorithm.

The goal of direction-of-arrival (DOA) estimation is to use the data received on the downlink at the base-station sensor array to estimate the directions of the signals from the desired mobile users as well as the directions of interference signals [5]. The results of DOA estimation are then used by to adjust the weights of the adaptive beam former so that the radiated power is maximized towards the desired users, and radiation nulls are placed in the directions of interference signals. Hence, a successful design of an adaptive array depends highly on the choice of the DOA estimation algorithm which should be highly Adaptive smart antennas are the array antennas whose radiation pattern is shaped according to some adaptive algorithms [1]. Smart essentially means computer control of the antenna performance. The smart antenna radiation pattern directs the beam towards the users of interest only & nulls toward interference to improve the capacity of cellular system. The adaptive beam forming algorithms takes the fixed beam forming process one step further & allows for the calculation of continuously updated array weights.

According to signal space information smart antenna can form directional beam in space with the adaptive beam forming algorithm, achieving that the main beam aims at the direction of the expected signal while the side lobe and nulls aims at the interference. Now many adaptive algorithms have been proposed on smart antenna.

II. OVERVIEW OF THE PAPER

In section 1 we describe the brief introduction about smart antennas, In section 3 we simply discuss the functions of smart antennas and beam forming for CDMA systems, In section 4 we describe the Proposed Improved LMS algorithm and mathematical analysis, In section 5 we discuss the results and Conclusions.

III. FUNCTIONS OF SMART ANTENNA

Smart antennas have two main functions: They are

- DOA estimation and
- Beamforming.

A. DOA Estimation

The smart antenna system estimates the direction of arrival of the signal, using techniques such as MUSIC (Multiple Signal Classification), estimation of signal parameters via rotational invariance techniques (ESPRIT) algorithms, Matrix Pencil method or one of their derivatives. They involve finding a spatial spectrum of the antenna/sensor array, and calculating the DOA from the peaks of this spectrum. These calculations are computationally intensive.

B. Baemforming

Beamforming is a signal processing technique used in sensor arrays for directional signal transmission or reception. This is achieved by combining elements in the array in a way where signals at particular angles experience constructive interference and while others experience destructive interference. Beamforming can be used at both the transmitting and receiving ends in order to achieve spatial selectivity.

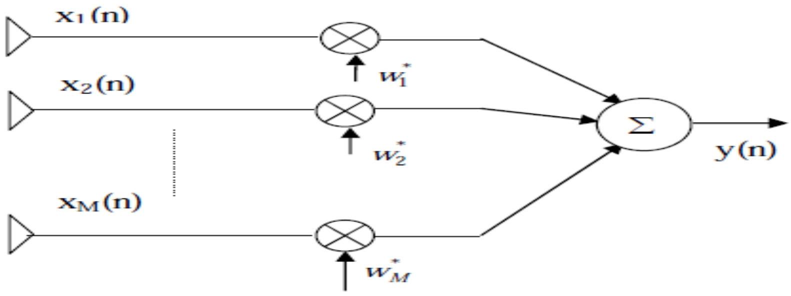


Figure 1-Beamforming Operation

Beam forming is the name given to a wide variety of array processing algorithms that focus array capabilities in a particular direction. A beam forming algorithm points the array spatial filter towards the desired direction algorithmically rather than physically.

Beam forming is also referred to as "electronic" steering since the weights are applied using electronic circuitry following the reception of the signal for the purpose of steering the array in a particular direction. The beam forming operation is shown in Figure 1.

As shown in Figure 3.6, a beam former produces its output by forming a weighted combination of signals from the M elements of the sensor array, that is,

$$y(n) = \sum_{i=1}^{M} W_1^* X_1(n) = W^H X(n) \quad (1)$$

Where, W_1^* is the weight applied to the i^th sensor. ***W*** is commonly referred as beam forming vector given by

$$W = [\, W_1 \cdots\cdots\cdots\cdots W_M]^T \qquad (2)$$

IV. Improved Least Mean Square Algorithm

A simple and novel adaptive algorithm for steering the antenna beam electronically is Least Mean Square algorithm. Generally LMS algorithm is widely used in adaptive filter due to its relatively low computational complexity, good stability properties, and relatively good robustness against implementation errors.

The Least Mean Square (LMS) algorithm, introduced by Widrow and Hoff in 1959 is an adaptive algorithm, which uses a gradient-based method of steepest decent. LMS algorithm uses the estimates of the gradient vector from the available data. LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. Compared to other algorithms LMS algorithm is relatively simple; it does not require correlation function calculation nor does it require matrix inversions.

A. ILMS Algorithm and Adaptive Arrays

Consider a Uniform Linear Array (ULA) with N isotropic elements, which forms the integral part of the adaptive beamforming system as shown in the figure below.

The output of the antenna array is given by,

$$x(t) = s(t)a(\theta_0) + \sum_{i=1}^{N} u_i(t)\, a(\theta_i) + n(t) \qquad (3)$$

S(t) denotes the desired signal arriving at angle θ_0 and $U_i(t)$ denotes interfering signals arriving at angle of incidences θ_i respectively. $a(\theta_0)$ and $a(\theta_i)$ represents the steering vectors for the desired signal and interfering signals respectively. Therefore it is required to construct the desired signal from the received signal amid the interfering signal and additional noise n(t).

As shown above the outputs of the individual sensors are linearly combined after being scaled using corresponding weights such that the antenna array pattern is optimized to have maximum possible gain in the direction of the desired signal and nulls in the direction of the interferers. The weights here will be computed using LMS algorithm based on Minimum Squared Error (MSE) criterion. Therefore the spatial filtering problem involves estimation of signal s(t) from the received signal x(t)(i.e. the array output) by minimizing the error between the reference signal d(t), which closely matches or has some extent of correlation with the desired signal estimate and the beamformer output y(t) (equal to w x(t)). This is a classical Weiner filtering problem for which the solution can be iteratively found using the LMS algorithm.

B. ILMS Algorithm Formulation

From the method of steepest descent, the weight vector equation is given by,

$$w(n+1) = w(n) + \mu[-\nabla(E\{e^2(n)\})] \quad (4)$$

Where μ is the step-size parameter and controls the convergence characteristics of the LMS algorithm; $e^2(n)$ is the mean square error between the beamformer output y(n) and the reference signal which is given by,

$$e^2(n) = [d^*(n) - w^h x(n)]^2 \quad (5)$$

The gradient vector in the above weight update equation can be computed as

$$\nabla_w(E\{e^2(n)\}) = -2r + 2Rw(n) \quad (6)$$

In the method of steepest descent the biggest problem is the computation involved in finding the values r and R matrices in real time. The LMS algorithm on the other hand simplifies this by using the instantaneous values of covariance matrices r and R instead of their actual values i.e.

$$R(n) = x(n)x^h(n)$$

$$r(n) = d^*(n)x(n)$$

Therefore the weight update can be given by the following equation,

$$w(n+1) = w(n) + 2\mu x(n)[d^*(n) - x^h(n)w(n)]$$

$$= w(n) + 2\mu x(n)e^*(n) \quad (7)$$

The LMS algorithm is initiated with an arbitrary value w(0) for the weight vector at n=0. The successive corrections of the weight vector eventually leads to the minimum value of the mean squared error.

Therefore the LMS algorithm can be summarized in following equations;

Output, $y(n) = w^h x(n)$ (8)

Error, $e(n) = d^*(n) - y(n)$ (9)

Weight, $w(n+1) = w(n) + 2\mu x(n)e^*(n)$ (10)

C. Convergence and Stability of the LMS algorithm

The LMS algorithm initiated with some arbitrary value for the weight vector is seen to converge and stay stable for

$$0 < \mu < \frac{1}{\lambda_{max}} \quad (11)$$

Where λ_{max} is the largest eigen value of the correlation matrix R. The convergence of the algorithm is inversely proportional to the eigen value spread of the correlation matrix R. When the eigen values of R are widespread, convergence may be slow. The eigen value spread of the correlation matrix is estimated by computing the ratio of the largest eigen value to the smallest eigen value of the matrix. If μ is chosen to be very small then the algorithm converges very slowly. A large value of μ may lead to a faster convergence but may be less stable around the minimum value. One of the literatures [will provide reference number here] also provides an upper bound for μ based on several approximations as

$$\mu \leq \frac{1}{(3\, trace(R))}$$

D. Steps in LMS algorithm

The LMS algorithm proceeds according to the following steps:

1. The weight vector is initialized by setting all the filter taps to some random values. A common choice is to set all the taps to 0.
2. A constant is picked. While it is possible to determine the theoretical maximum value of that will still guarantee convergence for each particular problem, in practice, is usually picked from experience and decreased/increased if necessary.
3. The vector X_K is formed from the input samples and e[k] is computed by
$e[k] = d[k] - y[k] = d[k] – X^T_K W_K$
4. W_{K+1} is computed using the weight update equation.
5. Set k to k + 1 and go to step 3.

V. MATLAB Simulation Results

In this section the results of low side lobe control techniques are demonstrated using MATLAB. For simulation purposes a 8-element linear array is used with its individual elements spaced at half-wavelength distance.

The interfering signals $u_i(t)$ arriving at angles θ_i are also of the above form. By doing so it can be shown in the simulations how interfering signals of the same frequency as the desired signal can be separated to achieve rejection of co-channel interference. Illustrations are provided to give a better understanding of different aspects of the ILMS algorithm with respect to adaptive beamforming.

Two examples are provided to show the beamforming abilities of the ILMS algorithm. Each example has a normalized array factor plot.

Case 1:

In the first case the desired angle is arriving at 90 degrees and there are three interfering signals arriving at angles 50,100 and 150 degrees respectively. The array factor plot in Figure 5.1 shows that the LMS algorithm is able to iteratively update the weights to force minimum side lobe level at the direction of the interferers and achieve maximum in the direction of the desired signal. In this case the LMS error is almost -0.023.

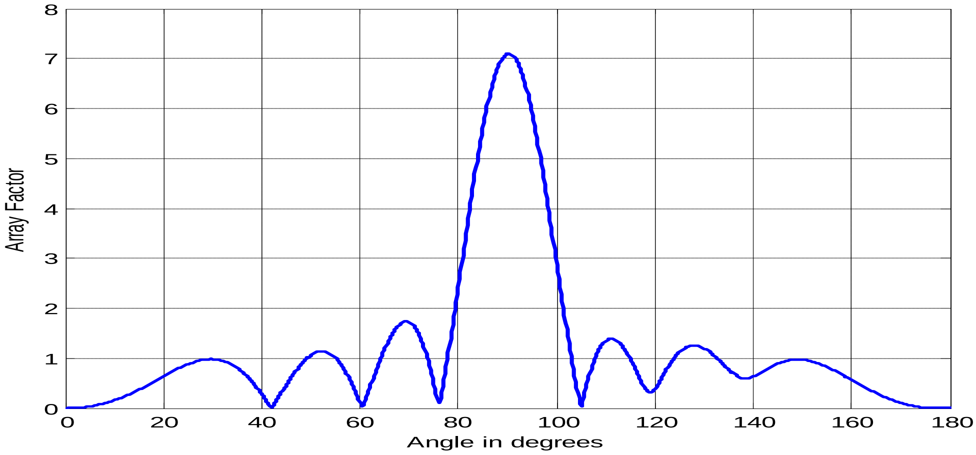


Figure 3. Array factor plot for case 1

Case 2:

In the first case the desired angle is arriving at 45 degrees and there are three interfering signals arriving at angles 10, 60 and 80 degrees respectively. The array factor plot in Figure 5.2 shows that the LMS algorithm is able to iteratively update the weights to force minimum side lobe level at the direction of the interferers and achieve maximum in the direction of the desired signal. In this case the LMS error is almost 0.1847.

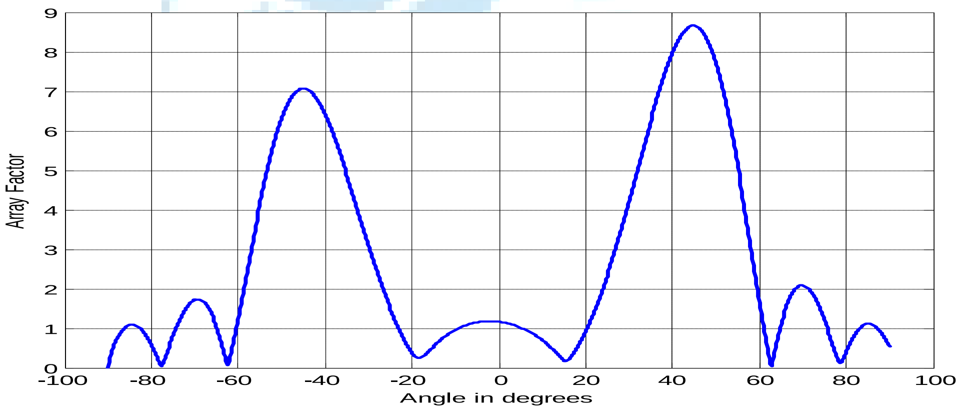
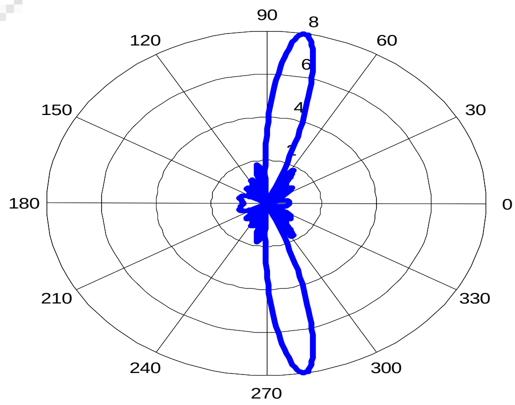


Figure 4- Array factor plot for case 2

The antenna radiation pattern for an 8-element linear array is shown in figure. In this case the desired angle is arriving at 80 degrees and there are three interfering signals arriving at angles 10,40 and 60 degrees respectively. The ILMS algorithm is able to iteratively update the weights and achieve maximum in the direction of the desired signal.



(a) DESIRED SIGNAL

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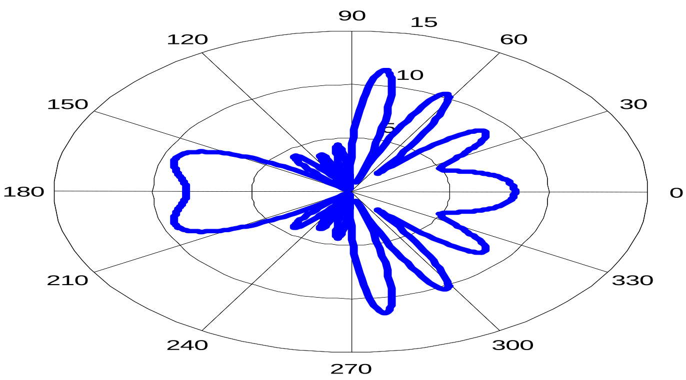
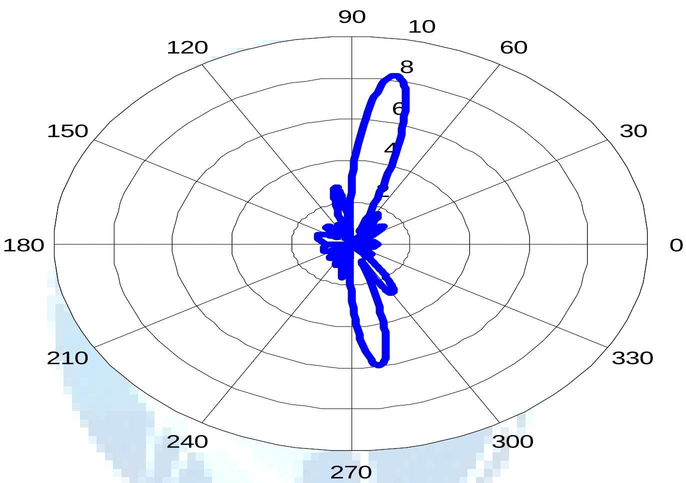


5.

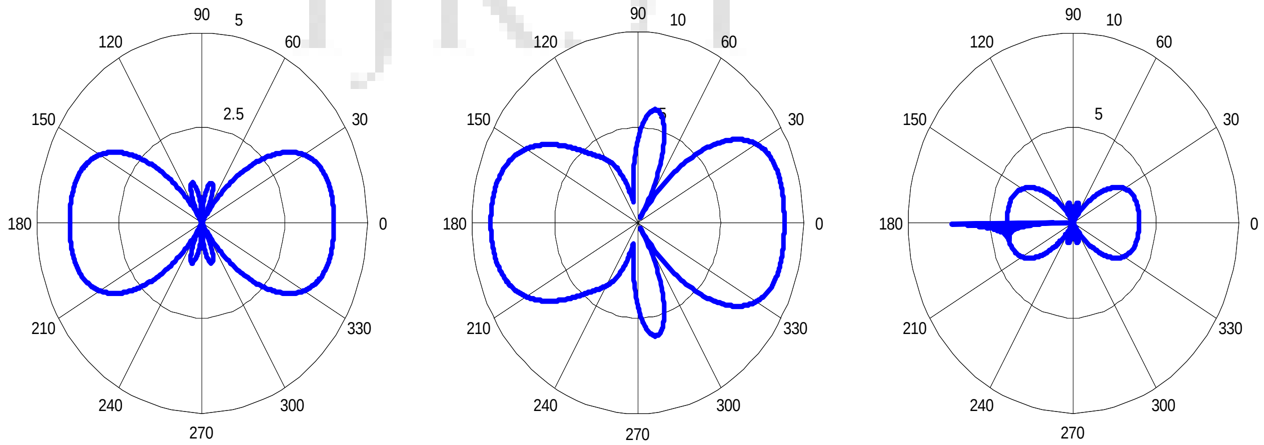
(b) DESIRED AND INTERFERENCE SIGNAL



(c) SIGNAL OBTAINED WITH LMS ALGORITHM

Figure 5. Antenna radiation pattern for (a) Desired signal (b) Desired and interference signal (c) signal obtained with LMS algorithm

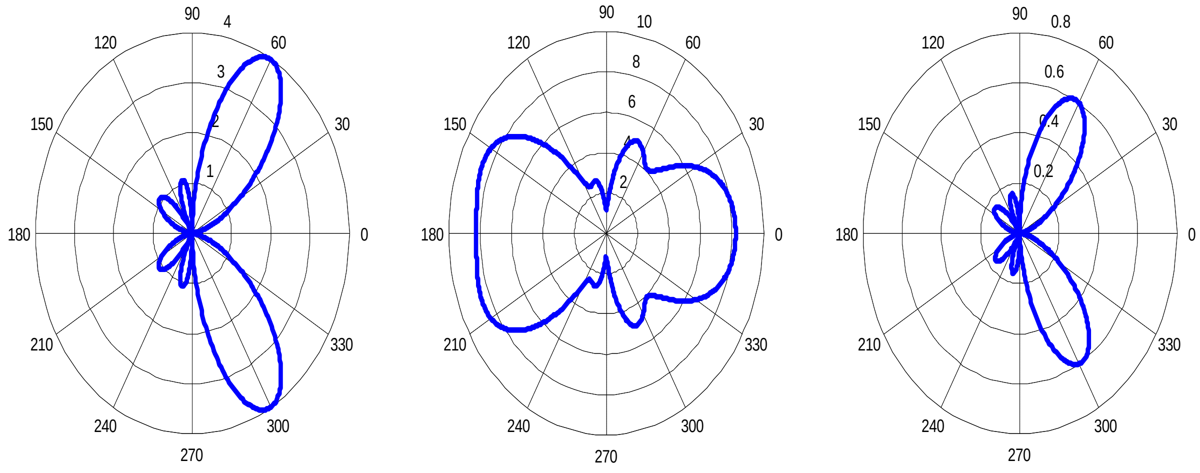
The antenna radiation pattern for a 4 - element linear array with desired angle θ_d and interfering signals arriving at angles θ_{i1}, θ_{i2} $and \theta_{i3}$ is shown below.



(a) DESIRED SIGNAL (b) DESIRED AND INTERFERENCE SIGNAL (c) SIGNAL OBTAINED WITH LMS ALGORITHM

Figure 6 – Polar plot for Rectangular array with N=4, d=λ/2,

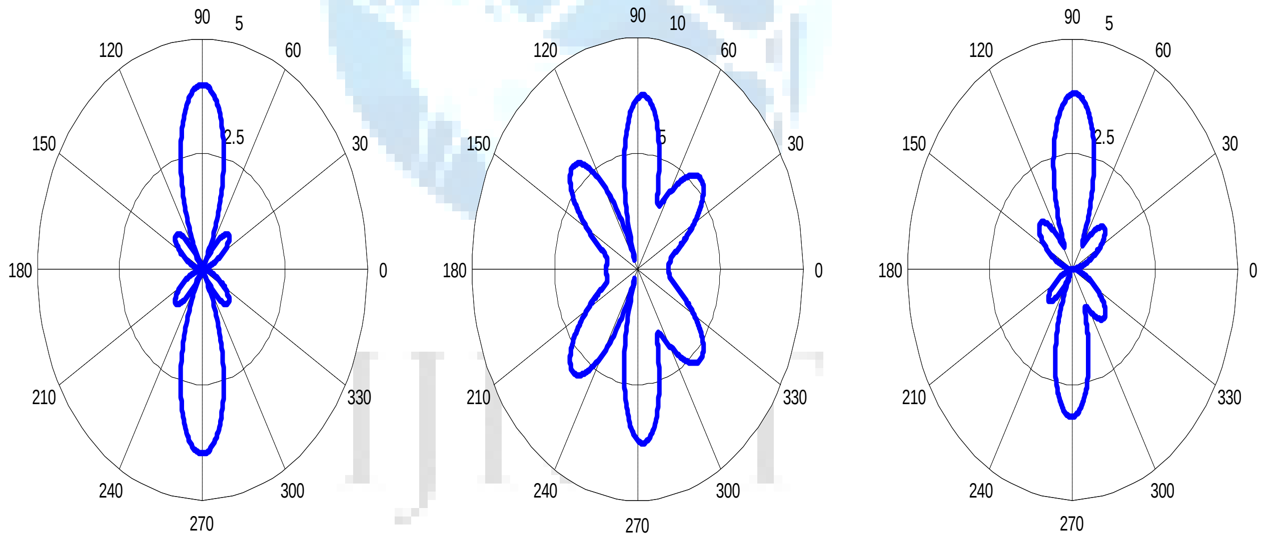
$$\theta_d = 180^\circ, \theta_{i1} = 135^\circ, \theta_{i2} = 80^\circ \ and \ \theta_{i3} = 20^\circ$$



(a) DESIRED SIGNAL (b) DESIRED AND INTERFERENCE SIGNAL (c) SIGNAL OBTAINED WITH LMS ALGORITHM

Figure 7. Polar plot for Hamming array with N=4, d=λ/2,

$$\theta_d = 60^{\circ}, \theta_{i1} = 20^{\circ}, \theta_{i2} = 180^{\circ}\ and\ \theta_{i3} = 135^{\circ}$$



(a) DESIRED SIGNAL (b) DESIRED AND INTERFERENCE SIGNAL (c) SIGNAL OBTAINED WITH LMS ALGORITHM

Figure 8 Polar plot for Chebyshev array with N=4, d=λ/2,

$$\theta_d = 90^{\circ}, \theta_{i1} = 45^{\circ}, \theta_{i2} = 75^{\circ}\ and\ \theta_{i3} = 135^{\circ}$$

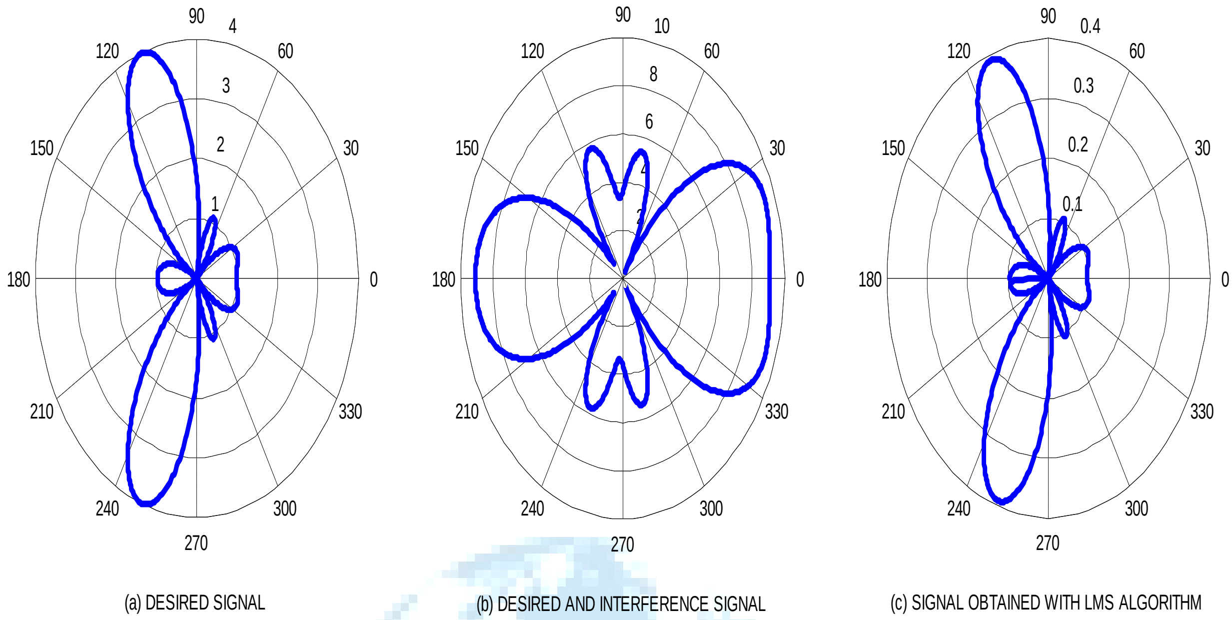


Figure 9 Polar plot for Kaiser array with N=4, d=λ/2,

$$\theta_d = 110^\circ, \theta_{i1} = 20^\circ, \theta_{i2} = 85^\circ \text{ and } \theta_{i3} = 180^\circ$$

Table 1 - Error Value For Different Windows

S.No	Taper	Error
1	Rectangular	-0.0058
2	Hamming	-0.0005
3	Chebyshev	0.0037
4	Kaiser	-0.003

The HPBW of an *N* element array can be estimated This can be validated through MATLAB simulation of the linear array. In this simulation, linear array can be modeled by arranging elements with a uniformly distributed random spacing.

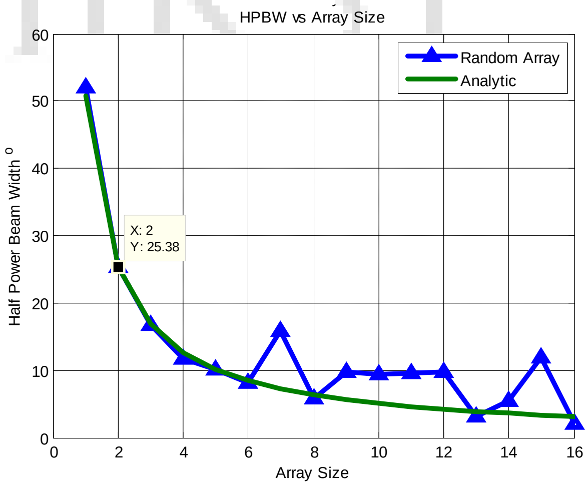


Figure 10 - Half Power Beam Width (HPBW) Calculated for Increasing Array Size.

Table 2: Half Power Beam Width (HPBW)

Array size	Half power beam width(N=16)	Half power beam width(N=20)
2	25.38	25.38
4	12.69	12.69
6	7.714	7.6
8	7.143	5.771
10	8.857	4.171
12	7.057	6.514
14	2.771	2.6
16	5.057	2.429
18	-	2
20	-	2

From the figure 10 it can be analyzed that as the array size increases, the HPBW becomes smaller. The results show good agreement with the theory, indeed the analytical expression seems to form an upper bound. From this result, the array size can be used to vary the beamwidth where we want a high gain narrow beam.

VI. Conclusion

In a CDMA system all transmitted signals turn out to be disturbing factors to all other users in the system in the form of interference. Therefore the system capacity strongly depends on the interference level in the system. Hence In the case of CDMA systems, it is proposed that smart antenna will control interference by forming main lobe towards desired user and low/depressed side lobe towards the undesired users or interferers. Hence narrow beam low side-lobe algorithms, which can taper array weights of any beamformer with suitable taper function, are considered appropriate for CDMA systems. It is found that the role of window function is quite impressive and economical from the point of view of computational complexity and ease associated with its application in controlling side lobes.

It is proposed from this study that Chebyshev window can be flexible to decrease the maximum side lobe level and achieves narrow beam towards desire user when compared to other adjustable and fixed side lobe yielding windows. Therefore using Chebyshev window the gain of the system will be increased and it will definitely improve the performance of CDMA based wireless communication system, where the number of interferes is quite large..

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