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Beam pattern improvement in CMA using a Prefilter

Swapnil M. Hirikude[1], A.N.Jadhav[2]
[1](Department of E & TC, D.Y.Patil College of Engg. & Tech., Kolhapur, Maharashtra,India).
[2](Department of E & TC, D.Y.Patil College of Engg. & Tech., Kolhapur, Maharashtra,India).

ABSTRACT: *The demand for wireless mobile communications services is growing at an explosive rate; the high demand for 3G services needs increased in system capacity. The most elementary solution would be to increase bandwidth; however, this becomes challenging as the electromagnetic spectrum is becoming increasingly congested. The frequency reuse concept increases capacity however, increasing the number of cells to accommodate growing subscriber needs is neither effective nor an economical option. This has led to development of new technologies that exploit space selectivity using smart-antenna arrays and the associated adaptive beam forming algorithms. In reality, antennas are not smart; it is the digital signal processing, along with the antenna, which makes the system smart. When smart antenna with Adaptive beamforming is deployed in mobile communication using either time division multiple access (TDMA) or code division multiple access (CDMA) environment, it radiates beam towards desired user only. Each beam becomes a channel, thus avoiding interference in a cell. Smart-antenna systems provide opportunities for higher system capacity and improved quality of service. A new beamforming (Hybrid) approach using a pre-filter that decreases noise and interference effects to improve performance of cellular systems is discussed here. This paper also presents the results obtained by applying prefiltering process to the most researched Constant Modulus Algorithm (CMA) blind beam forming algorithm.*

Keywords: *Smart Antenna, DOA, Beamforming, Prefilter, CMA.*

I .INTRODUCTION

Adaptive beamforming can be classified into two categories: Non-blind adaptive algorithms and blind adaptive algorithms [1, 2]. Non-blind adaptive algorithms need statistical knowledge of the transmitted signal to converge to a solution. This is typically accomplished through the use of a pilot training sequence sent over the channel to the receiver to help it identify the desired user. On the other hand, blind adaptive algorithms do not need any training; hence the term „blind" is used. They attempt to restore some characteristics of the transmitted signal in order to separate it from other users in the surrounding environment. In this paper a prefiltering technique [3] which could be used with the blind or non-blind algorithms to enhance their performance in terms of amplitude response (array factor) is presented. This technique acts on the input signal vector **x** (k) as a band pass filter but in spatial domain, so it minimizes the noise and interference effects as a function of the Direction of Arrival (DOA).

II. THE PREFILTERING TECHNIQUE

The proposed prefiltering technique aims to increase the Signal-to-Interference and Noise Ratio (SINR) of the beam forming system by reducing the interference and noise effects on the desired user signal using filtering in spatial domain, or extracting the desired signal from the instantaneous input signal vector **x**(k) of the beamformer, [3] as can be seen in Fig. 1.In this context, it is worth pointing out that in image processing, especially in image compressing techniques, one can find an abundance of techniques that can reconstruct the original image with acceptable performance, without using all transformation components, but rather using only the lower component coefficients of the image transform matrix [4]. This fact is exploited here and employed with some modification in the antenna array processing to obtain a new hybrid beamforming technique. Since the interfering signals are in the same frequency band of the desired signal, they can be analyzed representing them in another domain other than frequency domain so as to distinguish between the mixed signals that form the input signal. Therefore, the technique is based on the idea that the desired and interfering signals arrive at the antenna array from different directions. Thus, these differences between arriving signals can be exploited. The distinction can be obtained by converting the input signal to the spectrum of the spatial domain (this domain is the sine of the direction of arrival, or $\sin\theta$ domain).The desired signal can be extracted from the input system signals simply by making a band-pass filter in the spectrum of the spatial domain, i.e. in the $\sin\theta$ spectrum.

This filtering process is shown in Fig. 2 as mentioned in [3], and can be explained generally as follows:

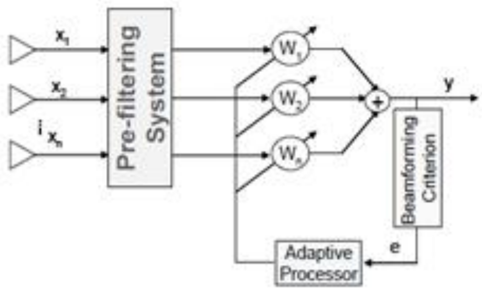

Fig.1. Adaptive beam former with Pre-filtering system

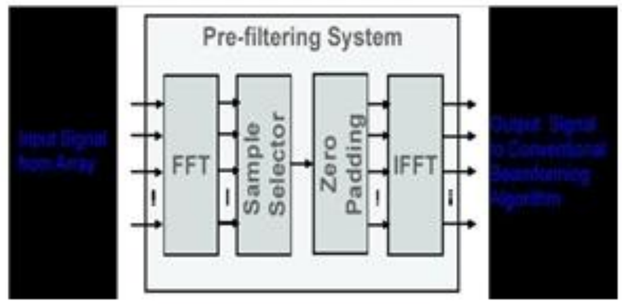

Fig.2. Pre-filtering Process

• The original input signal is passed through a Fast Fourier Transform (FFT) stage to obtain its coefficients in the spectrum domain.

• The Most Significant Coefficient (MSC) of the transformed signal is selected. This is ranked as the largest sample of the transformed desired signal.

• The most significant coefficient is placed at its rank in the M zeros element vector (zero padding).

• The Inverse Fast Fourier Transform (IFFT) is applied to the filtered vector of the previous step to reconstruct an alternative input signal that contains a reduced amount of interference and noise.

• The reconstructed data vector is used as input signal to the conventional adaptive beam forming system.

Mathematically, assuming that the propagation vector for the θ direction of arrival, is given by

$$a(\theta) = e^{-j2\pi\frac{d}{\lambda}\sin\theta(m-1)}, m = 1,2,\ldots\ldots,M \quad (1)$$

Where M is the number of array elements, d is the spacing distance between any two adjacent elements, and λ is the wavelength of the operating carrier frequency. Applying the FFT on the array factor propagation equation (1) gives,

$$A(\Theta) = \mathrm{FFT}\left(a(\theta)\right) = \frac{1}{M}\sum_{m=1}^{M} e^{-j2\pi\frac{d}{\lambda}\sin\theta(m-1)} e^{-j2\pi K(m-1)/M}$$

$$= \frac{1}{M}\sum_{m=1}^{M} e^{-j2\pi\left(\frac{d}{\lambda}\sin\theta(m-1)+\frac{K}{M}(m-1)\right)} \quad (2)$$

Assuming d= λ/2, and solving equation (2) and equating the result to zero, the following formula gives the index K_{MSC} (or the order) of the most significant coefficient as a function of the direction of arrival θ and the number of array elements M.As K_{MSC} must be an integer the equation takes the form of equation (3).

$$K_{MSC} = \mathrm{mod}_M\left\{1 + M\left[-\frac{1}{2}\left(\frac{M(2\pi M\sin\theta-\pi\sin\theta)}{\pi(2M-1)}\right)\right]\mathrm{int}\right\} \quad (3)$$

Where mod $_M$ is the modulus notation performed on M points. Equation (3) can be simplified to

$$K_{MSC} = \mathrm{mod}_M\left(\left[1-\frac{1}{2}\ M\sin\theta\right]\right) \quad (4)$$

This Kmsc is used then to reconstruct the modified input signal which has reduced interference and noise.
Simulation results presented later in this paper show that the prefiltering technique significantly reduces the mean square (MSE). This prefiltered output is then used as input to any conventional beamforming algorithm to enhance its performance. The results obtained by applying this technique to the blind Conventional CMA algorithm are discussed in the next section.

III. CMA ALGORITHM

In this paper, the constant modulus algorithm (CMA) which is presented by Goddard [2] and by Treichler and Agee [6] is used as a typical conventional blind technique. This technique, as all other blind beamforming techniques, does not need previous knowledge about the desired signal. It exploits nearly constant modulus property of most of Telecommunication modulated wireless transmitted signals.CMA recovers the desired signal by forcing the received Signal to have constant amplitude. When the CMA algorithm successfully converges, it converges to the optimal Solution. However, the main problem of the blind adaptation algorithms is their poor convergence property compared with non-blind algorithms which use the training sequences. This algorithm seeks for a signal with a constant magnitude i.e. modulus within the received data vector and is only applicable for modulation scheme which uses symbol of equal power includes phase and frequency modulated signals. The cost function of CMA is not convex, and thus the conversion of CMA is not guaranteed [5]. Another drawback of the CMA algorithm is that it may converge to the undesired signal in cases where the interfering signal is of equal or higher power than the desired signal [7, 8]. The proposed technique overcomes the fore mentioned problems associated with the CMA. Further, it exploits CMA advantages of simplicity and efficiency. A complex envelope signal has an envelope which is generally constant. During transmission process in communication channel, some corruption (or distortion) from the channel (noise, multipath and interference) deform this envelope. The CMA can restore the envelope of the beamformer output signal, that is, restoring the constant envelope of this signal by computing the variation in its modulus and minimizing this variation using the following cost function [2] [5]:

$$J(k) = E||y\,(k)|^p - 1|^q \quad , \quad p = 1, 2, \quad q = 1, 2 \tag{5}$$

Where p and q are coefficients which control the convergence of the algorithm. The cost function of the CMA measures the deviation in the envelope of the output signal of the beamforming system from the unity modulus and tries to minimize this deviation using the effects of selected coefficients p and q. By putting p = 1 and q = 2, we can create a sequential update technique to form the desired weights by exploiting the steepest descent method for the following cost function:

$$J(k) = E\left[\left||y(k) - 1|\right|^2\right] \tag{6}$$

The iterative solution or the steps to implement the CMA to obtain the complex weights can be summarized as Follows:

$$y(k) = w^H(k)x(k) \tag{7}$$

$$e(k) = \; y(k) - \frac{y(k)}{|y(k)|} \tag{8}$$

And

$$w(k+1) = w(k) - \frac{1}{2}\mu x(k)e^*(k) \tag{9}$$

Where y (k) is the filter output, e (k) is the error signal. Equation (9) is the weight w (k+1) update function for the CMA algorithm. μ is rate of adaption also called as a step size, controlled by the processing gain of the antenna.

IV. HYBRID (PREFILTERED) ADAPTIVE BEAM-FORMING PROCESS

The complete hybrid system by applying prefiltering to the conventional adaptive beamforming algorithm discussed in sections II and III respectively is illustrated below with the help of a flowchart in Fig.3. The technique aims at improving the performance of beam forming algorithm by reducing the interference and noise effects on the desired user signal.

V. SIMULATION RESULTS

Simulation of the technique is carried out using MATLAB software. The prefiltered signal is applied to the conventional CMA beam forming algorithm for a Uniform Linear array (ULA) with a distance between the elements d = λ/2. Results of magnitude response of Conventional and Hybrid CMA beamforming algorithms presented here are obtained by varying parameters like number of antenna elements (M) and step size parameter μ, for two or more interferers in random directions and Noise is assumed to be Gaussian. Figure 4 a) Shows the beam pattern gain (magnitude response) of Conventional and Hybrid CMA algorithm for M =8, desired angle at=45 deg and interference angles at 50, 35. µ=0.01 Figure 4 b) & c) show the Polar plot for the same in terms of the Antenna Array factor.

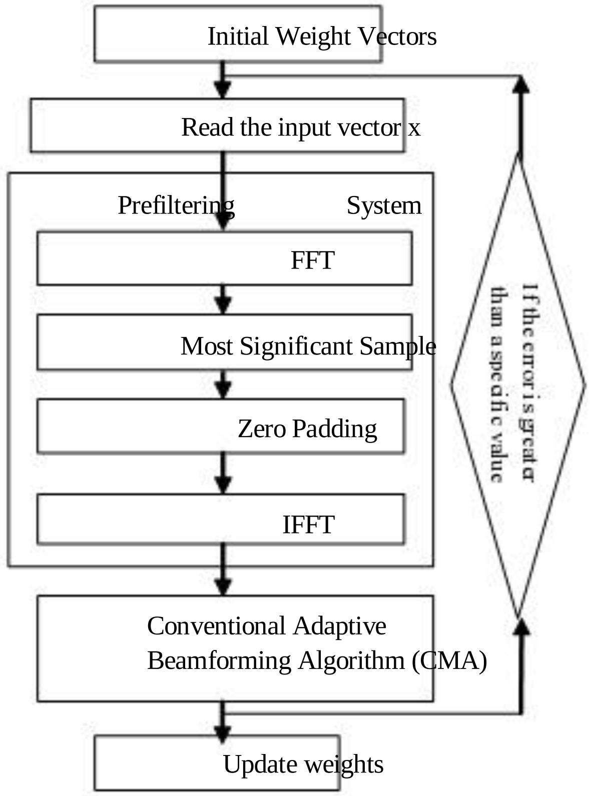

Fig.3. Flowchart of the Hybrid (Prefiltered) adaptive beamforming technique

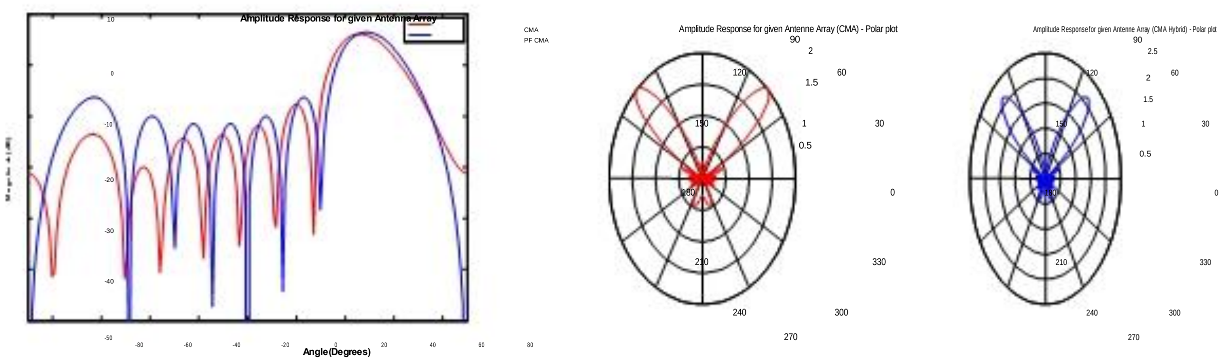

Fig.4.a) Beam pattern gain of CMA and hybrid technique b) Polar plot for CMA c) Polar plot for Hybrid tech.

for M=8, µ=0.01

Figure 5 a) Shows the beam pattern gain (magnitude response) of Conventional and Hybrid CMA algorithm for M =8, desired angle at=45 deg interference angles at 50, 35. µ=0.01 and SNR 10 db .Figure 5 b) & c) show the Polar plot for the same in terms of the Antenna Array factor.

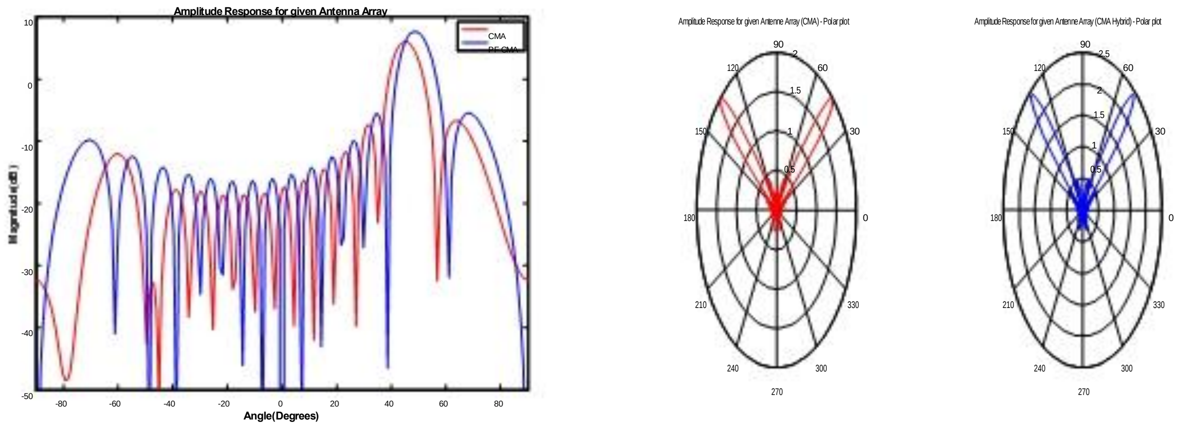

Fig.5. a) Beam pattern gain of CMA and hybrid technique b) Polar plot for CMA c) Polar plot for Hybrid tech. For M=16, and μ=0.01

Plots obtained here do not give exact value of amplitude /gain response G (θ), hence for more accurate estimation we normally refer its computed value. Hence for the further analysis we refer its computed value in MATLAB.

Table no.1 Results obtained for Number of antenna elements M=8 and μ=0.01.

Input DOA (Θ) in deg	Beam pattern Gain for Conventional CMA in (db)	Beam Pattern gain for Prefiltered CMA in (db)	Total improvement in Beam Pattern Gain (db)
10	6	8.62	2.62
45	6.04	6.45	0.41
60	6.02	8.25	2.23
120	6.05	8.09	2.04

Table no.2 Results obtained for Number of antenna elements M= 12 and μ=0.01.

Input DOA (Θ) in deg	Beam pattern Gain for Conventional CMA in (db)	Beam Pattern Gain for Prefiltered CMA in (db)	Total improvement in Beam Pattern Gain (db)
10	6.03	8.83	2.8
45	5.9945	6.6644	0.6699
60	5.99	6.5	0.51
120	6.0398	6.5457	0.5059

Table no.3 Results obtained for Number of antenna elements M=16 and μ=0.01.

Input DOA (Θ) in deg	Beam pattern Gain for Conventional CMA in (db)	Beam Pattern Gain for Prefiltered CMA in (db)	Total improvement in Beam Pattern Gain (db)
10	5.9942	8.3002	2.306
45	6.0322	6.7232	0.691
60	6.0626	6.2931	0.2305
120	6.0065	6.0552	0.1487

After observing Table no.1, 2 & 3 it can be said that the prefiltering improves antenna beam pattern gain than of the Normal CMA algorithm. Also the beam pattern gain improves with the increase in the number of antenna elements.

VI. CONCLUSION

The results obtained indicate that the performance of prefiltered (Hybrid) CMA algorithm is better than the conventional (Normal) beamforming CMA algorithm as far as beam pattern gain/amplitude response of the array is concerned. The Performance of hybrid technique can be enhanced by increasing the number of array elements. The prefiltered technique gives better performance because it uses the reconstructed input signal which has less interference as compared to the direct input signal. The improvement in the beam pattern characteristics can ne used to increase in the quality as well as the system capacity of the system.

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