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Efficient Adaptive Beamforming Algorithms for Smart Antennas

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Abstract: In this paper we introduc the concept of adaptive beam forming for smart antennas. In previous days the beams can be steered based upon the omni directional antennas. Due to this power consumption should be more and also power can be waste because the antennas are propagating in all directions. So to avoid this we go for the new technology that is smart antenna, by using smart antennas we can steer the beams only in the desired direction and placing the null values in other directions. The beams can be steered based upon the adaptive algorithms and direction of arrival algorithms (DOA). By using of this smart antenna we can required low power consumption because of placing the nulls in the unwanted directions. Why we are preferring the smart antennas because it having high capacity, interference rejection capability, handover, easily access to the technology, efficient beam steering. In this paper we use LMS algorithm which is proposed and compare it with the remaining adaptive algorithms like normalized, varaiabe step size algorithms. By using normalization and variable step size strategy the convergence rate could be improved. Simulation results conform that the proposed variable step size based adaptive algorithms performs better than conventional LMS based beam former.

Keywords: adaptive beam forming, convergence rate, least mean square, normalized algorithms, variable step size algorithms.

1. INTRODUCTION

Adaptive beam forming is one of the promising techniques used in modern wireless communications. In [1] Weinchen wang et. al., proposed a secure communication by using two users interference network. In this by using two transmitters the information can be received and the artificial noise can be simultaneously to the two users and energy receivers. In [2] yishinyeh et. al., proposed a special 5^{th} generation cellular application that is 28GHZ four channel phased array receiver in 130nm SiGe BICMOS technology. For 5^{th} generation cellular applications multiple beams are required for multiple users. This can be obtained only by using digital beam forming multi input multi output (MIMO) systems. In [3] Ben peiffer et. al., proposed a retro directive beam forming for a distributed arrays. The distributed array consists of network of wireless transceivers and transmitting the signals by using the multi antenna to the external receivers. In [4] Yavuzyaman et. al., proposed beam forming protocols, they are one is IEEE 802.11ad and other one is IEEE 802.15.3c and these protocols can be obtained by using two beam forming (BF) one is binary search beam forming and other one is linear search beam forming to improve BF setup time in adaptive algorithms. In [5] weinchen wang et. al., In the robust beam forming uses the multi-pair one-way relay netwok. In this the communication can be done by via decode-and-forward multi-antenna relay. In [6] Rouhancao et. al., tells that to maximize the secrecy in the source side while get the good quality of service (QOS) in the destination side by considering the power. This optimization requires a joint beam forming algorithms. In [7] Yong wang et. al., presents a new beam forming that is super directive frequency-invariant beam forming for circular sensor networks. In this method the weight vector is in the closest form because of the minimizing the minimum mean square error between the known applied by homogenizing the electric field and give the power density to the mobile users. In [8] Thierry gilles et. al., how the beam forming can be applied to the homogeneous electric field, by distributing the power density to the mobile users. In [9] William c. Barrot et. al., tells that the beam forming is nothing but adding of the all the output voltages that is coming from the many individual sources by using array of antennas instead of single antennas there by decreasing the signal-noise ratio. In [10] Yun cao et. al., studies the cognitive network consisting of a two-way amplify-and-forward(AF) relay by using two antenna terminals (SU1 and SU2) and it provides high SINR ratio. In [11] Nehasinghal et. al., discuss about the binary error ratio[BER] and Signal-to-Noise ratio[SNR] for different channels by using the beam forming techniques at the transmitting side, receiving side or at both sides. In [12] M.S. Hussain et. al., introduced robust broadband beam forming for circular microphone arrays. The microphone arrays are useful for the cancelling the co-channel interference and increasing the signal strength in wanted directions. In [13] Henrikklessig proposed a spatial traffic model to generate the data traffic maps to calculate the traffic adaptive beam forming algorithm is proposed. In [14] W. Orozco-Tupacyupanqui et. al., proposed an adaptive hybrid beam forming (HB) that depends upon the convex combination of two independent transversal adaptive filters. It can be uses the recursive mean square (RLS) and dual uni variety filtering approach. The main of this beam forming is to produce good radiation pattern, even one of the filters are failed. In [15] Slim Zaidi proposed a collaborative beam forming (CB) means that to establish a energy efficient and reliable communications about the longest distances. In[16] M. Papez et. al., proposed a beam forming is a method of processing the microphone array to produce the images that represents the length of the image. In [17] M. M. Rashid et. al., proposed a robust beam forming uses the adaptive rectangular array beam forming using fixed diagonal loading (FDL) and optimal diagonal loading (ODL) are used. Due to this it improves the rectangular array performance, signal-to-noise ratio plus interference ratio of the beam former. In [18] Spandana Vemulapalli proposed the performance of an ultra wideband *v*-slot, *l*-probe fed microchip atch antenna for earlier detection of the breast cancer. For this we are using microwave imaging space time (MIST) beam forming algorithm in the frequency domain. In [19] JingweiXu a robust adaptive beam forming is proposed for the frequency diverse array-spacetime signal adative processing to indicate the fast moving target detection. In [20] Paolo Baracca et. al., compare the two beam forming configurations where the signal-to-leakage plus noise ratio (SLNR) is applied by either all the base stations or by the macro BSs. This can be accessed by using the two schemes one is achievable rates and other one is feedback overhead. In [21] Dingjiexu proposed a variable tap-length LMS algorithm based on adaptive parameters for tapped-delay-line (TDL).the main objective of this is to increase the convergence rate and high sensitivity to the noise. In [22] Yun tan et. al., the modified least mean squares (MLMS) algorithm proposed to generalize the fractional order α ($0 < \alpha <= 1$). This α is step size and if it is smaller it gives smaller weight noise and bigger α gives faster convergence speed. In [23] Neilj. Bershad et. al., proposed two algorithms LMS and NLMS algorithms. These models can be applied to the non-stationary systems and the mathematical models are derived for the mean and mean square deviation methods (MSD). In

[24] Jorge plata-chaves et. al., a distributed adaptive algorithm to solve the nose specification parameter estimation problem where the nodes are used to estimate the parameters of the local interest, global interest. So the LMS algorithm are used for estimating the set of local and common parameters. In [25] Pramodkumarmeher et. al., proposed a delayed least mean square adaptive filter. For obtaining the lower adaption delay and area-delay-power efficient implementation, by using of the partial product generator and propose a strategy for optimized balanced pipelining. By using this we can obtaining 17% area-delay and 14% less energy-delay product. In [26] Long shi et. al., propose a new variable step-size diffusion least mean square algorithm that adjusts the step size for the every iteration at each node. This algorithm achieves fast convergence speed and it is used in the non-stationary environment places. In [27] Dingjiexu et. al., proposed a variable tap-length algorithm based on adaptive parameters for tapped-delay-line (TDL) structure. The main purpose of this is to overcome the some drawbacks in the previous variable tap-length adaptive filter algorithm such as slow convergence rate and high sensitivity to the noise. In [28] Solomon nunoo et. al., propose the step size $l0$-norm normalized LMS (NLMS) algorithm. The step size can varies in accordance with the changes in the men square error (MSE), for allowing the small changes in the system and produce the smaller steady state errors. In [29] Vasundhara et. al., proposes a sparseness controlled SAMIPAPA (VSS-SC-SAMIPAPA) algorithm are used for cancelling the feedback in the hearing aids. The performance of this algorithm is in terms of misalignment and added stable gain (ASG) for both white noise. In [30] Yeguixiaoet.al. proposed a variable step-size LMS (VSS-LMS) algorithm for Fourier analysis of noise sinusoidal signals, to confirm the improved performance and tracking ability of proposed algorithm. Based on these contributuins, we implement an adaptive beam former using NLMS and VSSLMS algorithms.

2. ADAPTIVE BEAM FORMING

In the particular applications the gain of the single antenna may not be efficient. So in that situations the group of antennas are used that is array of antennas are used. The adaptive beam former is a system that performs the spatial signal processing with an array of transmitters or receivers. The main purpose of this is directing the beam towards the desired antenna while successfully rejecting all other targets in another direction. So the antennas can be steered in two ways. One is mechanical steering and other is electronic steering. The adaptive beam forming can be performed by using adaptive algorithms. Most of the adaptive algorithms are based on the convergence rate. The functional block diagram of adaptive array systems are as shown in Figure 1.

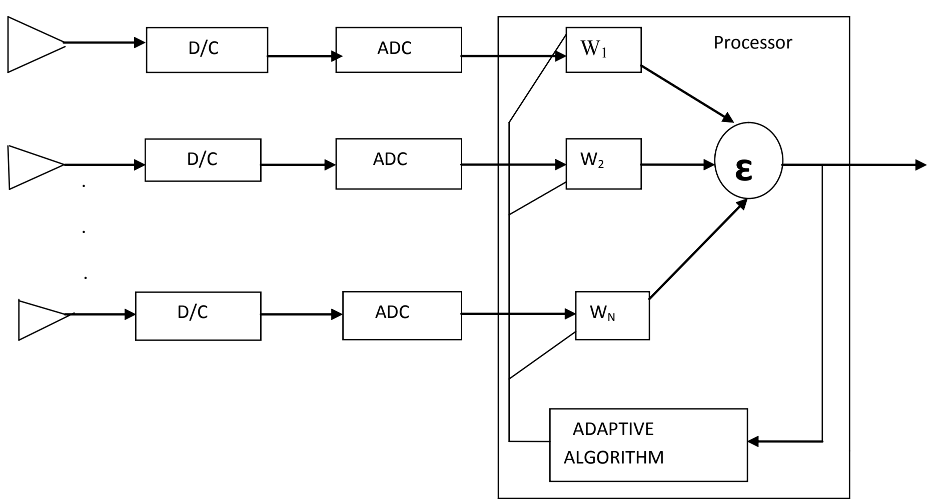

Figure 1: Functional block diagram of adaptive beamforming system

As shown in the above Figure 1 it contains number of antennas and number of complex weights. The output of this can be can be obtained by summing up all the weights and the wanted signals that can be received by the antennas. Then the output can be given to the adaptive beam forming algorithms. The adaptive array systems can locate and track the users and as well as interferences. Then the antenna arranges the antenna in a desired direction to get perfect communication by nullifying the interference signals. After this process can be taken then the system converts it into baseband signal and digitalized them, the signal of interest (SOI) can be located by using DOA algorithms, and it continuously track the SOI and signal of non interest signals by updating the complex weights each time. The direction of arrival means that the direction in which a propagating wave arrives at a point. In the adaptive beam former two main important points are differentiated. First one is that the part of the desired signal can be known, then estimate the mean square error (MSE) signal by taking the difference between the received signal and the known signal. The weights can be updated continuously by nullifying the MSE value. Due to this the signal-to-interference ratio will be reduced, because of the weights are updated in accordance with the incoming signal but not based on the interference signals. The second one is the identify the DOA from all sources of transmitting signals by using the array of antenna systems, and the weights are updated in accordance with the desired signals and nullifying the interference signals in other directions. This approach may turn deficient in practical use because of there are so many DOA's due to the multi paths, and the algorithms are also not work properly to adjust the beam towards the desired signals. The adaptive array system is also used the frequency reused concepts, because of the multi users share the same frequency over the conventional channel. Due to this the capacity of the adaptive systems are also increases.

3. ADAPTIVE ALGORITHMS FOR BEAM FORMING

Adaptive algorithm trains the adaptive filter which is also known as the transversal filter. The weights at every iteration are updated by this algorithm through the quadratic MSE surface will be estimated and then moved towards the negative direction by a minute amount of the gradient. The step size can be defined as that which determines the constant and then it allows the optimal weights from the estimated weights. These weights will have a transient behavior and convergence, the LMS algorithm will be covariance characterized will be effected greatly affected and also will have the practical importance. Various algorithms used in this work is summarized as followes,

An adaptive filter is considered in which the length is considered as L and weights considered to be W and the input signal which is arrived from the antenna array is considered to be $X(n)$ and there is a presence of error $e(n)$ between the desired signal $d(n)$ and the array output $y(n)$ it will be reduced by using the adaptive processor. It is useful for the adaptive processing and it is also very simple and the necessary for the beam forming. The linear array which is uniform has its output response as:

$$y(n) = h^{\wedge H}(n)x(n)$$

we consider the adaptive filter where input signal $x(n)$ is convolved by an unknown filter $h(n)$ filter which has an additive interference signal $v(n)$ before being observed as $d(n)$. The value of error signal estimation is $e(n) = d(n) - y(n)$. From $d(n)$ the convolved signal $y(n)$ is subtracted, it results the output signal $e(n)$ which

will contain both interference $v(n)$ and as well as residual signal $r(n) = y(n) - y^{\wedge}(n)$. The signal of interest in the system can be defined as the interference $v(n)$ which arrives for the LMS adaptive algorithm near the recursion for the updating the step as

$$h(n) = h(n - 1) + \mu e(n)x(n)$$

where, μ is constant step and filter can be updates by using the recursive relation.

Variable Step Size Least Mean Square Algorithm

In [30] weight update equation for the familiar LMS algorithm is written as,

$$h(n + 1) = h(n) + \mu\ e(n)h(n)$$

$$h(n) = [a\hat{}(n), b\hat{}(n)]\mathrm{T}$$

where, μ is the step size with a small positive value, $e(n)$ is an error signal output by the adaptive analyzer.

$$\mathrm{J}(n) = (1/2)(e2(n)) + (1/2)\ \alpha\mu 2(n - 1)$$

The first term can be represented as same as the convention LMS algorithm. For controlling the magnitudes of step size the second term can be introduced. From this resultant algorithm has good stability. Here weighting factors are served by small positive constant i.e. α

$$\mu(n) = \mu(n - 1) - \eta\Delta_{\mu}\ \mathrm{J}(n)$$

Adaptive behavior of step size sequence ($\mu(n)$) is controlled by the small positive constant (η).

The improved equation for the variable step size based on the LMS algorithm, it removes the unwanted noise and provides the good anti-interference stability. The equation for the new variable step size is as,

$$\mu(n) = \alpha \log (1 + (1/2)\ (e(n)\ e(n - 1))/\delta^2)$$

to make this algorithm more convergence, the step size $\mu(n)$ must be in the range between $0 < \mu(n) < 1/\lambda_{max}$, and it is also a positive value based on the δ and α values.

The error signal $e(n)$ or $\mathrm{N}(n)$ can be obtained by

$$e(n) = \mathrm{N}(n) = \mathrm{X}^{\mathrm{T}}(n)\Delta h(n)$$

where, $\Delta h(n) = h(n) - h^*(n)$

$$e^2(n) = \mathrm{N}^2(n) - \mathrm{N}(n\mathrm{E})\mathrm{X}^{\mathrm{T}}(n)\Delta h(n) - \mathrm{X}^{\mathrm{T}}(n)\Delta h(n)\mathrm{N}(n) + \mathrm{X}^{\mathrm{T}}(n)\Delta h(n)\mathrm{X}^{\mathrm{T}}(n)\Delta h((n)$$

the error is independent of the input signal $\mathrm{X}(n)$ and also its mean value is zero. So by taking the expectations on both sides of the above two equations we obtain the reduced formula for the new LMS algorithm.

$$\mathrm{E}[e^2(n)] = \mathrm{E}[\mathrm{N}^2(n)] + \mathrm{E}[\mathrm{X}^{\mathrm{T}}(n)\Delta h(n)\mathrm{X}^{\mathrm{T}}(n)\Delta h(n)]$$

$$\mathrm{E}[e(n)e(n - 1)] = \mathrm{E}[\mathrm{X}^{\mathrm{T}}(n)\Delta h(n)\mathrm{X}^{\mathrm{T}}(n - 1)\Delta h(n - 1)]$$

In the above equation $e(n)e(n - 1)$ relates to only on the input signal $\mathrm{X}(n)$ but not on the interference signal $\mathrm{N}(n)$, so because of this the this algorithm has stronger anti-interference ability. This algorithm has minimum SNR ratio and smaller steady-state error.

4. SIMULATION RESULTS

This section illustrates the waveforms for the different types of adaptive beam forming algorithms like LMS, NLMS, VSSLMS algorithms. So these algorithms are done in the MATLAB code and results are obtained as follows. The Figure 2 shows the convergence rate of the each algorithms. The convergence rate will be increased when move from NLMS to VSSLMS algorithms. The convergence curves can be obtained by taking the number of iterations on x-axis and error signal on y-axis. Among the algorithms considered the VSSLMS based beam former performs better than conventional NLMS and GNGDLMS algorithm.

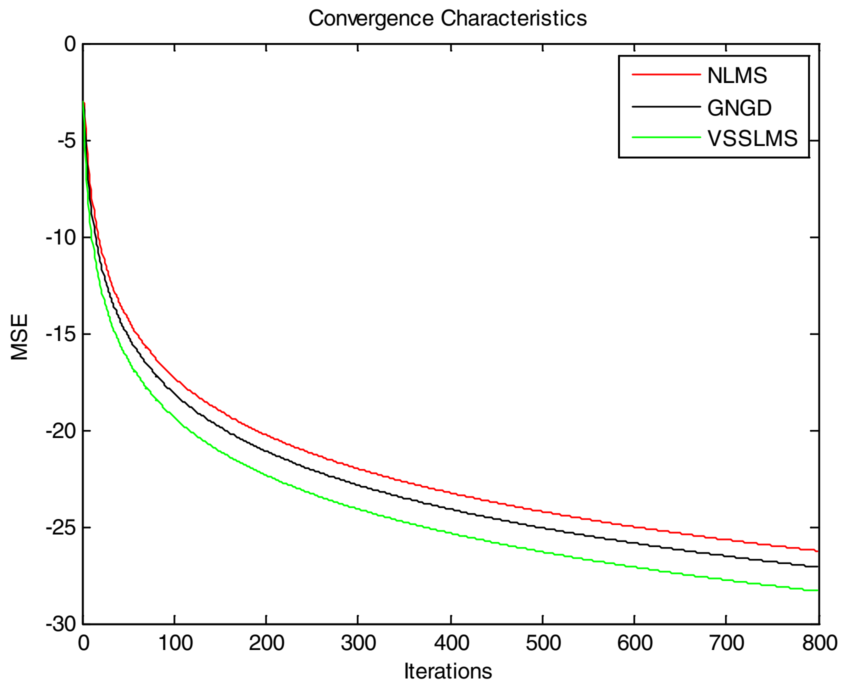

Figure 2: Convergence curves for NLMS, GNGD and VSSLMS algorithms

The mean square errors for various algorithms is as shown in Figure 3 by keen observation the error amplitude is decreases from LMS to VSSLMS. To observe the waveforms of the below Figure it is clear that in the LMS the convergence starts from after 70th sample and in GNGD it starts from after 30th Sample, in VSSLMS the convergence starts from 15th sample.

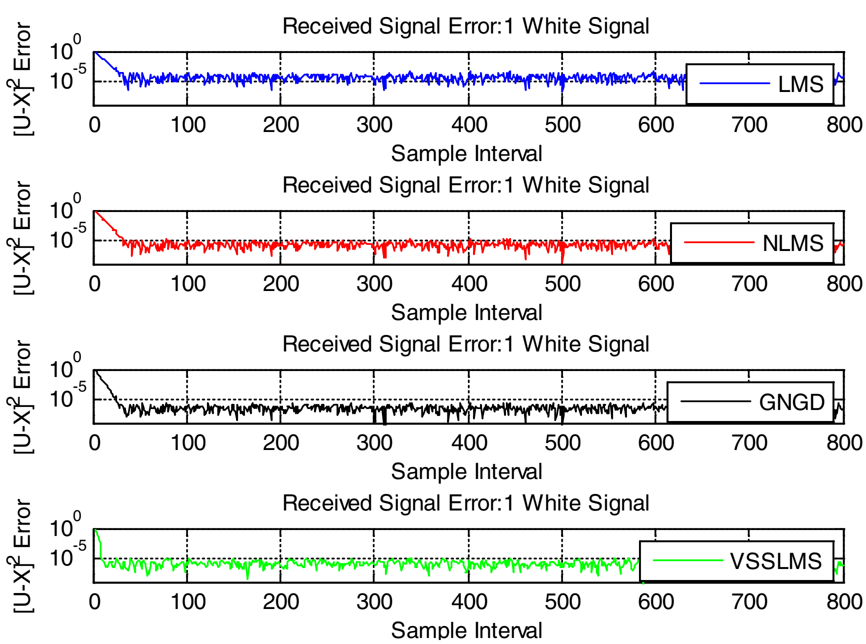

Figure 3: Mean square error signals for (a) LMS (b) NLMS (c) GNGDLMS (d) VSSLMS algorithms

In our simulation consists of two received signals with three 3 DOA's. The simulation results for various algorithms are presented below.

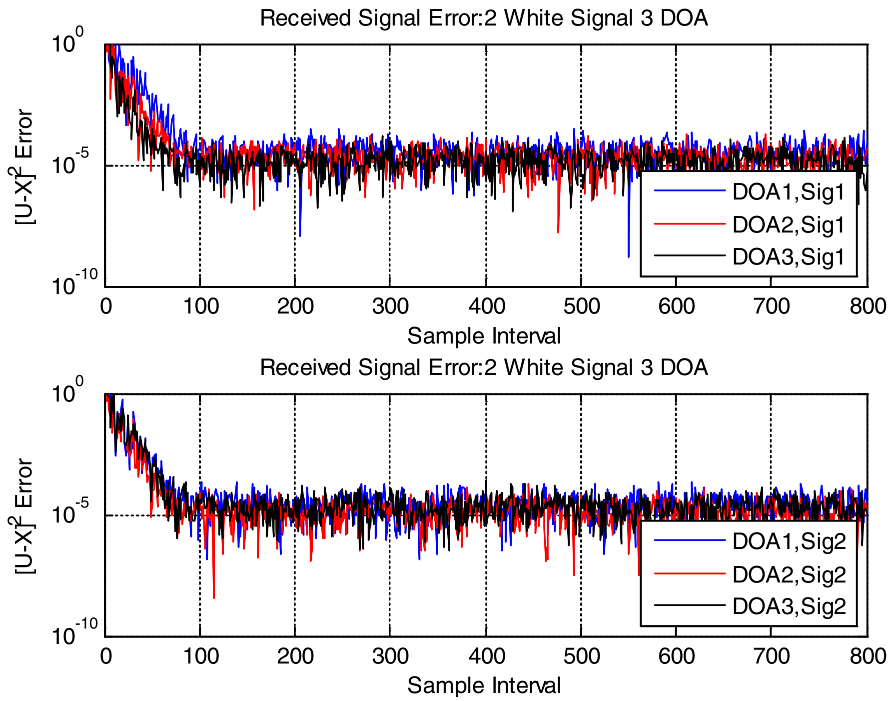

Figure 4: Received signal error with 2 signal and 3DOAS each for LMS algorithm

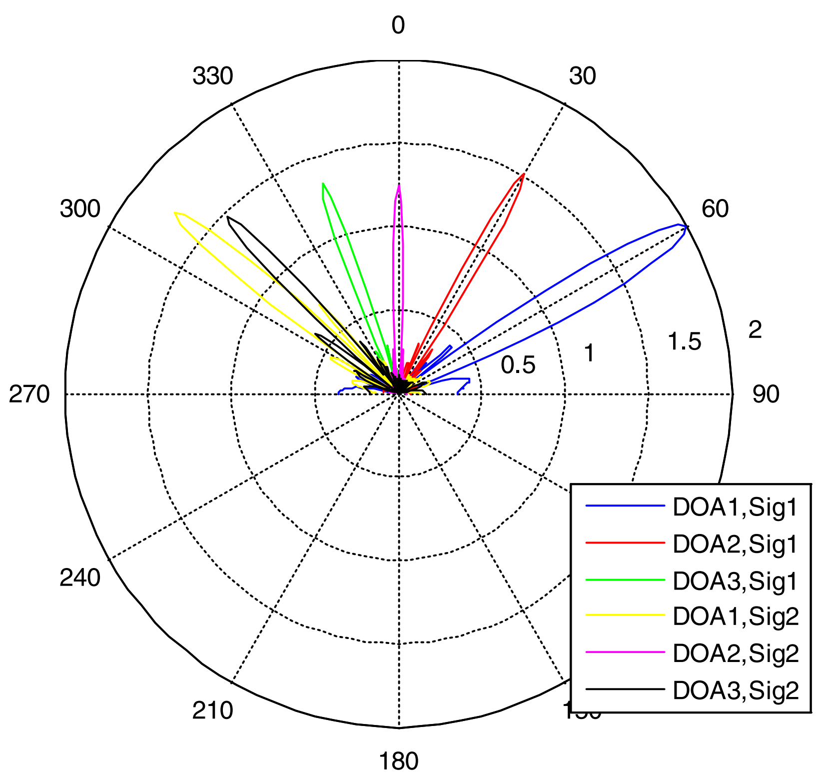

Figure 5: Polar plots for VSSLMS based beamformer

The Figure shows that the major lobe represents the with highest amplitude which is the desired signal and the minor lobes represents the interference signals.

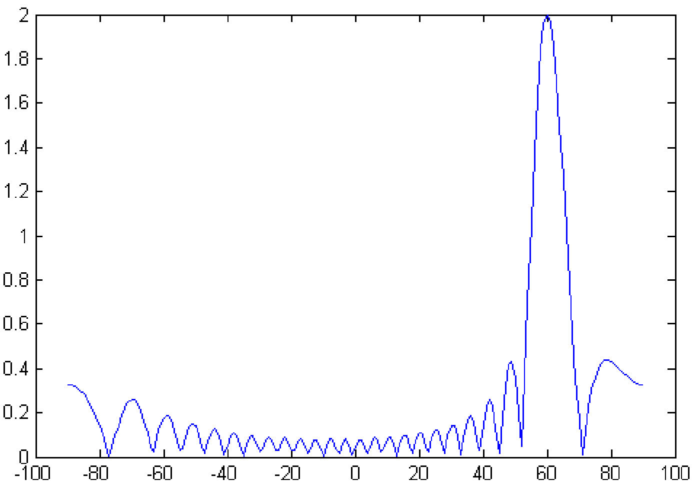

Figure 6: Rectangular plot for one DOA signal in VASSLMS based beamformer

5. CONCLUSION

In this paper we proposed several beam forming algorithms with some simulation results. By comparing with LMS algorithms we can get 44% to 55% increase in overall convergence rate for VSSLMS algorithm and as well as minimum mean square error. By using VSSLMS algorithm we can steer the beam towards the desired target and place null the interference signals. This proposed implementation achieves fast convergence due to normalization, minimum mean square error. Hence, this technique is well suited for practical applications like mobile communications, medical telemetry, radar communications etc.

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