A Compression study of Antenna Beamforming
Using LMS and NLMS Adaptive Algorithm

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Abstract—Basically beamforming is a technique of directional signal transmission and recitation. Now a days, because of an increasing demand on wireless communication (like RADAR, satellite communication, mobile communication, broadcasting etc.,) the beamforming technique plays a vital role for good transmission and reception purpose. By changing the phase and amplitude of the exciting currents in each of the antenna elements, it is possible to electronically scan the image lobe and/or place nulls in any direction. In beamforming, each users signal is multiplied with complex weights that adjust the magnitude and phase of the signal to and from each antenna. This causes the output from the array of antennas to form a transmit/receive beam in the desired direction and minimizes the output in other direction. In this paper we use different version of LMS adaptive algorithm (basically LMS and NLMS) to steer the antenna beam in particular desired direction. The radiation pattern of for each case plotted in polar plot. Here we steer the beam of of the system in two way (1) Considering the desired direction (position) are in uniformly distributed (2) The desired direction are non uniformly distributed. We campier their simulation results of each case with each other.

Index Terms—LMS, NLMS, steering angle, BWFN, HPBW, Spatial Filtering

I. INTRODUCTION

Beamforming is basically an adaptive signal processing technique. It use to steer the beam to a particular direction without having to mechanical steer the antenna element. The main goal of the the beamforming technique is directional transmission and reception, and which can be achieved by combining the elements of the array in such a fashion that is signal arrives from single and particular direction combat with effective interferences and the signal arrives from other direction compromise with destructive interferences,i,e, We can direct the radiate/receive power towards/from the desired direction (with less interference) and nulling the interfering signal. It improves the signal to noise ratio (SNR) and leads a better signal estimation.

The beamforming can be used for sound and radio wave [3]. It can be used in transmitting and receiving mode. Mainly the beamforming technique take the advantages interference to change the directionality of the array. It used to detect and estimate the desired signal at the out put of the system by means of adaptive spatial filtering. To create an pattern in constructive and destructive interference in the wavefront, the beamformer controls the phase and relative amplitude of the of each element of the array(during transmission) and the information from the different element combined in such a way the desired or expected pattern is preferentially observed. We arrange the paper in following way. In section II we discuss about the LMS algorithm and its different variants. In section III we discuss about the spatial filtering . Simulation work are discussed in section IV, and we conclude our paper with comparing the their convergence issue on beamforming [5].

II. LMS ALGORITHM

Least Mean Square (LMS) algorithm is one of the most popular adaptive signal processing algorithm. LMS is popular because of its simplicity and ease of computation, and because it does not require offline gradient estimation of data. It used in variety of practical application. There is a trade off between the speed of convergence and residual error when adaptation step size is used.It has been reported by Mikhael et al. the fastest speed convergence of LMS algorithm is achieved by choosing the step size properly. Raymond et al. proposed that the square of out put error controls step size adjustment , which gives the good performance in form of fastest convergence and less misadjustment (comparing to the general LMS) [4].

in case of general LMS algorithm the the weight vector coefficient for the finite impulse response (FIR) filter are updated as in following equation

$$w(n+1) = w(n) + \mu e(n)y(n)$$

where $w(n)$ is the weight vector coefficient, and which can be expressed as $w(n) = [w_0(n)w_1(n)..........w_M(n)]$, Where the filter length is $(M + 1)$. $\mu$ is known as step factor (which is convergence parameter of LMS algorithm). $e(n)$ is the output error, which is represents as $e(n) = d(n) - f(n)$. $f(n)$ being the filter out put and $d(n)$ is the reference signal. where we can express the $f(n) = w(n)y^T(n) = \hat{x}(n)$. Where $\hat{x}(n)$ is the original signal. $y(n)$ is the filter input. $y(n) = [y(n)y(n-1)..........y(n-M)]$. step size parameter $\mu$ is an important factor for the system convergence of the LMS algorithm. So choosing of $\mu$ should be done carefully because when $\mu$ is increase the system converge at fast rate and when $\mu$ is small system converge in slow rate. The value of $\mu_{\text{max}}$ (maximum convergence parameter)should be chosen as in following relation. $\mu_{\text{max}} < \frac{\lambda_{\text{max}}}{N}$,where $\lambda_{\text{max}}$ is the largest eigen value correlation matrix of the signal.
Taking the use of any random sequence of bits as the training sequence, and to plot following. (1) Initial beam position (2) Final beam position (3) The error vs. number of iteration (4) Effect of step-size parameter on convergence [1].

A. NLMS Algorithm

The NLMS is normalize least mean square algorithm the weight updating can be represent by the following equation:

\[
w(n + 1) = w(n) + \frac{\mu e(n)y(n)}{\|y(n)\|^2 + \beta}\
\]

where \( w(n) = [w_0(n)w_1(n)\ldots\ldots w_{N-1}(n)]^T \) is the weight vector, \( y(n) = [y(n)^{(0)}y(n-1)^{(0)}\ldots\ldots y(n-M+1)^{(0)}]^T \) is the input, \( \beta \) is constant whose value is very very small (0.001). NLMS converge faster than LMS because the step size is optimize at each iteration. The computational complexity of the more in NLMS than LMS.

B. Leaky LMS Algorithm

In Leaky LMS algorithm to change the penalty for changing the input sequence \( y(n) \) we add a proportionality constant. The weighting update can be represented as following:

\[
w(n + 1) = (1 - \alpha \mu)w(n) + \mu e(n)y(n)\
\]

C. block LMS Algorithm

In Block LMS we accumulate error in a block. The weight updating equation can be represent by the following equation:

\[
w(n + 1) = w^{(B)}(n) + \mu \sum_{j=0}^{i-1} e(n-j)y(n+j)\
\]

III. BEAMFORMING AND SPATIAL FILTERING

Beamforming is one type of signal processing used to form beams to simultaneously receive a signal radiating from a specific location and attenuate signals from other locations [5]. (Spatial filtering is nothing but an filter use to alter the structure of electromagnetic wave. It is commonly use to filter out aberration of beam due to imperfection, dirty du to variation in the gain medium) The spatial correlation is interpreted as correlation between the signal’s spatial direction and the received signal gain. Systems designed to receive spatially propagating signals often encounter the presence of noise signals. If the desired signal and noise signal occupy the same frequency band, unless the signals are uncorrelated, e.g., CDMA signals, then temporal filtering often cannot be used to separate signal from noise [7]. However, the origin of the desired signal and the noise i.e interfering signals tare different spatial location, he desired and interfering signals, the signal can be separate from the noise signal by using the spatial filter at he receiver. spatial separation can be exploited. The spatial filter implementation at the receiver needs processed data collected over a spatial aperture and in same way in case of temporal filter processed data should be collected temporal aperture [6]. An antenna array consists of a set of antenna elements that are spatially distributed known locations with reference to a common fixed point. By changing the phase and amplitude of the exciting currents in each of the antenna elements, it is possible to electronically scan the main beam and/or place nulls in any direction.

A beamformer is a processor used in conjunction with an array of antenna (i.e., antenna elements in an adaptive array) to provide a versatile form of spatial filtering. The spatial samples of propagating wave field are collected by the elements of the array, which are processed by the beamformer [5]. Typically a beamformer linearly combines the spatially sampled time series from each sensor to obtain a scalar output time series in the same manner that an FIR filter linearly combines temporally sampled data. There are two types of beamformers, narrowband beamformer, and wideband beamformer [7]. Basically in our experiment we deals with narrowband beamformer issues only. Because it is easy to implement and less mathematical complexity. Different from a narrowband beamformer, a wideband beamformer samples the propagating wave field in both space and time and is often used when signals of significant frequency extent (broadband) are of interest [3] [2]. For a narrowband beamformer, the output at time \( k \), \( y(k) \), is given by a linear combination of the data at the \( M \) sensors at time \( k \), which can be mathematically expressed by following equation:

\[
y(k) = \sum_{i=1}^{m} w_i^* x_i(k)\]

Where \( * \) denotes complex conjugate. Since we are now using the complex envelope representation of the received signal, both \( x_i(t) \) and \( w_i \) are complex. The weight \( w_i \) is called the complex weight. The spatial discrimination capability of a beamformer depends on the size of the spatial aperture of the array; as the aperture increases, discrimination improves. The absolute aperture size is not important, rather its size in wavelengths is the critical parameter.

IV. EXPERIMENTAL SIMULATION AND RESULTS

Experimental study has been done to compare the performance of two algorithms namely, LMS and NLMS in steering the antenna beam to a desired direction. The antenna is considered to be operating at 500 MHz. The number of adaptive filter coefficients are chosen to be 10. The algorithm is tested iteratively for different desired directions. The step size parameter \( \mu \) is chosen as 0.3 and the initial values of the filter coefficients are taken as 1. The filter is then adapted iteratively with respect to the minimum mean square error criterion. We have taken the number iterations as 30. After 30 iterations the antenna was made to form the beam according to the adaptive filter’s impulse response coefficients. Fig.1 and Fig.2 shows the directions of the beamforming using LMS and NLMS respectively for a desired angle of 45 degrees. Fig. 3 and Fig. 4 compares the minimum mean square error of LMS and NLMS respectively. It can be seen that the the LMS algorithm converges much faster than NLMS, but the initial mean square error is lesser for NLMS when compared to LMS. [6] [3]
V. Conclusion

After discussing about the performance of beam forming using LMS and NLMS adaptive scheme now we will conclude our discussion comparing the result of simulation. After comparing the simulation results of LMS and NLMS in context to beamforming we can conclude our paper in following few lines. If we want to converge the system in less number of iteration with minimal error we should go for NLMS. For fast convergence with large number of iteration LMS is suitable. NLMS converged with minimal error when number of iteration is far greater than that of the LMS for a given error threshold. The above figure are the result of simulation of the beamforming in different desired angle in uniform sequence and non uniform random sequence of desired angle. [7]

REFERENCES


