ISSN(Online): 2320-9801
ISSN (Print) : 2320-9798

International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 4, Issue 6, June 2016

Performance Analysis of Adaptive Beamforming Techniques

Akshay G. Kathane, D.G. Ganage, S. A. Wagh

M.E. Student, Dept. of E&TC, SCOE Pune University, Maharashtra, India

Assistant Professor, Dept. of E&TC, SCOE, Pune University, Maharashtra, India

Assistant Professor, Dept. of E&TC, SITS, Pune University, Maharashtra, India

ABSTRACT: Antenna Array Beamforming mostly used in wireless communication system for directional transmission or reception of signal. This technique is applied to various areas from medicine to military and communication. The beamforming techniques can be principally divided into two categories first is Conventional Beamformer and another one is an Adaptive Beamformer. Conventional Beamformers use a fixed set of weights and time-delays and an adaptive beamformer is able to automatically update its response to different situations. Some application in wireless communication systems need to design beamformer with specified response like a sidelobe level less than threshold level or forming a null response in interference direction or frequency. This paper aims to design optimal beamforming using Least Mean Square (LMS) and Recursive Least Mean Square (RLS) algorithm. The performance of this algorithm is analyzed by considering parameters like number of array element, Array Factor (AF)), Mean Square Error (MSE) and convergence. The main objective of this paper is to improve wireless communication by minimizing error in signal.

KEYWORDS: Least Mean Square (LMS); Recursive Least Mean Square (RLS); Array Factor (AF); Mean Square Error (MSE).

I. Introduction

Adaptive beamforming signal processing technique was first start to exist in the 1960 for the application of military such as sonar and radar etc. Adaptive beamforming consist array of antenna which is used for quality of signal received from direction of desired user. The received signal has own different frequency component, therefore it is necessary to adjust weight of each antennas according to frequency value. For minimize complexity of weight adjustment used adaptive antenna array [3]. The filter have some benchmark which need to be adjust like reduce the value of noise, MSE etc. and enhance the value of output power, SINR etc. for better performance. The adaptive filter improved slow convergence speed and sensitivity toward noise. The drawback of this filter is it increases number of tap also increases computational cost.

Adaptive beamforming consider linear antenna patteren.Fig.1 shows overview of proposed method. Receivers sensing the signal and equate direction of signal arise to ensure the minimum interference of signal. Beamforming algorithm used for estimation of parameters. Performance evaluation of adaptive filter antenna model by using two most established beamforming techniques LMS and RLS algorithm. The LMS algorithm used for minimize error rate by varying number of input [1]. The most of filter affected by external environment, therefore RLS algorithm consider. This technique doesn't affect by propagation environment change but it quite complex method practically. By comparing these techniques minimize mean error and measure converge rate. The paper organized in following section. Section II gives the information about related work used previously. Section III consists of system model adaptive filter array. Section IV focused on beamforming techniques LMS and RLS. Section V shows simulation result and comparative study of LMS and RLS methods. Conclusion presented in section VI.

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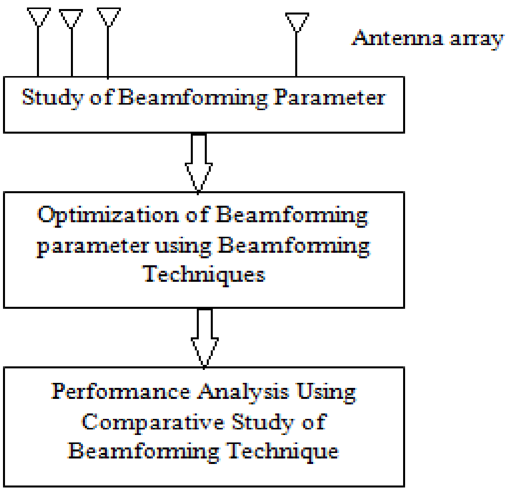

Fig.1 Overview of Proposed Method

II. RELATED WORK

In [1], authors used adaptive line enhancer with beamforming techniques such as LMS and RLS for evaluated low signal to noise ratio. The adaptive filter updates its own parameter according to input signal that make it more efficient with beamforming techniques. In communication system need to understand radiation pattern of signals for better used with antenna array and its effect on antenna array [2]. The [3] and [4] gives the information about the different type of antenna according to its geometry like linear, Planner and Circular and how it can implement with different beamforming techniques an effect of varying its parameter. In [5], authors compare three beamforming techniques LMS, NLMS and RLS using adaptive filter by changing its parameter value estimates the output with less error. To minimize the error in signal need to suppers the steering side lobe level this reduced the interference of signal to main beam. By varying element spacing and number of array element for different techniques evaluate better technique [6]. For better communication wideband beamforming preferred over narrowband [8] by varying number of filer and its value adaptive antenna array used for different type of signal.

III. FUNCTIONAL MODEL OF ADAPTIVE ANTENNA ARRAY

The advantage of adaptive array over conventional array is the parameter of adaptive array adjusts according to characteristics input signal. Antenna array estimate the direction maximum signal arises or main lobe [2]. For filtering used weight coefficients in antenna array also called as filter coefficients. The fig.2 shows the Adaptive filter structure it has input signal with weight coefficient array antenna, instead of one branch delay applied to every branch and it have respective weight coefficient and summation of all branch output [10].

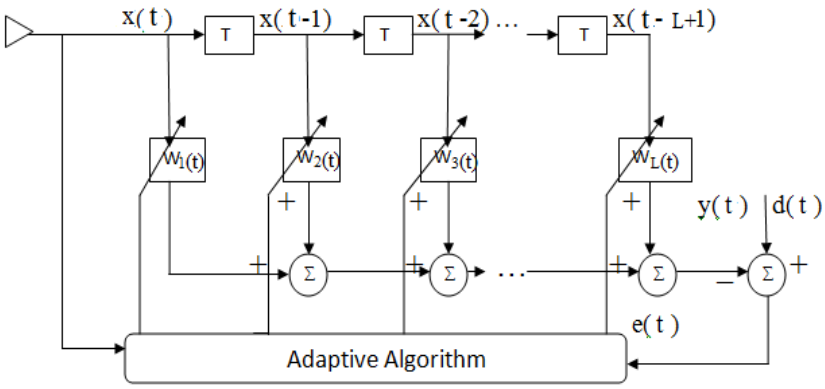

Fig.2 Functional Model of Adaptive filter

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The adaptive beamforming algorithm x(t) is received signal by multiple array antenna with multiply respective weight coefficient. Array antenna structure L is number of filter used; adaptive LMS algorithm used 2L complex iteration and 2L+1 complex multiplication. The adaptive beamforming algorithm minimum error is calculated by difference between desired signal d(t) and output signal y(t). The rate of convergence depends on input signal, but when input signal are correlated convergence rate is very slow. Therefore RLS algorithm is consider in this paper, RLS has improved convergence rate than LMS method.

IV. ADAPTIVE FILTER BEAMFORMING ALGORITHMS

The first design approach to adaptive beam forming is maximizing SNR by Applebaum 1965 [1]. This technique based on the system parameters in order to minimizing noise, while maximize the receive signal power. Another approach by Widrow and Maximum Likelihood Method (MLM) developed Least Mean Squares (LMS) error method in 1969 [2]. The beamforming accomplished by comparative study of LMS and RLS beamforming and estimate convergence rate.

A. *Adaptive filter LMS algorithm:*

An LMS algorithm was developed by Stanford University professor Bernard Widrow and Ted Hoff in 1960 [1]. Adaptive filter used to estimate weight coefficients that used to measure the mean squares of the error signal. The filter used gradient descent method for evaluation of error. In functional model antenna array x(t) indicated the received signal and w(t) is weight function in an array [3]

$$x(t) = [x_1(t), x_2(t), x_3(t), \dots, x_L(t)]^T \qquad \text{eq. (1)}$$

$$w(t) = [w_1(t), w_2(t), w_3(t), \dots, w_L(t)]^T \qquad \text{eq. (2)}$$

Where L is denoted as order of filter and t =0, 1, 2,... compute. The function of error signal denoted by e(t)

$$e(t) = d(t) - y(t) \qquad \text{eq. (3)}$$

The d(t) is desired signal and y(t) is output function denoted as

$$d(t) = w_0^T(t) + v(t) \qquad \text{eq. (4)}$$

$$y(t) = w^T(t)x(t) \qquad \text{eq. (5)}$$

Where $w_0(t)$ is initial weight function and v(t) is function of noise other than input signal. By using this method optimal updated weight function defined as [9]

$$w(t) = w(t-1) + \mu e(t)x(t) \qquad \text{eq. (6)}$$

The μ is converge step size parameter varies between $0 < \mu < \frac{2}{LS_{max}}$, Here S_{max} largest value of density power spectrum and f(w) is cost function of MSE.

$$f(w) = \{|e(t)|^2\} \qquad \text{eq. (7)}$$

B. *Adaptive filter RLS algorithm:*

The RLS is another beamforming algorithm that iteratively works to find weighted coefficient of input signals that reduces cost function [7]. As compare LMS algorithm input signals to RLS input it deterministic. The Gauss first design RLS beamforming technique but Plackett re-developed in 1950 [3]to work on beamforming.Then output function rewrite as [6]

$$y(t) = w^T(t)x(t) \qquad \text{eq. (8)}$$

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Then error function as

$$e(t) = d(t) - y(t) \qquad \text{eq. (9)}$$

The optimal updated weight function [4]

$$w(t) = w(t-1) + k^T(t)[d(t) - w^T(t-1)x(t)] \qquad \text{eq. (10)}$$

Where $k^T(t)$ is gain coefficient. The k(t) is gain vector with L order of filter for initial value it denoted as δ and λ is forgotten factor. Then gain vector defined as [5]

$$k(t) = \frac{L(t-1)x(t)}{\lambda + x^T(t)L(t-1)x(t)} \qquad \text{eq. (11)}$$

The RLS algorithm work more efficiently than LMS algorithm but complexity increases when order of filter more. For better results filter order maintain.

V. SIMULATION RESULTS

In this paper Adaptive Beamforming algorithm LMS and RLS is study using antenna array. These algorithms effectuate by MATLAB. For stimulation result, the delay value is one and numbers of array element t =10, 20 are taken.

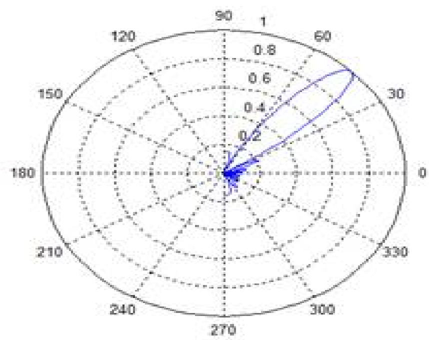

Fig. 3 Polar plot when desired signal is at 45° and interfering signal at -68°

The study of beamforming concept considers the maximum radiation pattern concentred at 45° shown in fig. 3 as main lobe and other side lobes with interfering signal at -68°.

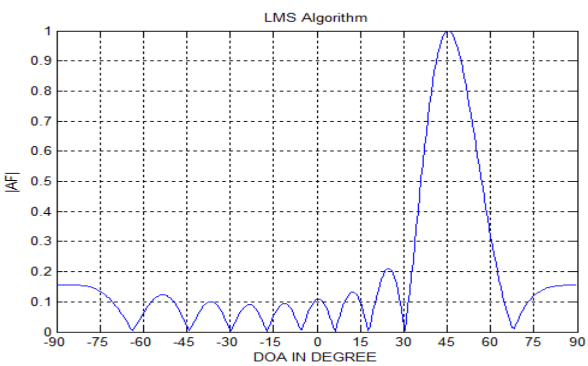

Fig.4. Array Factor when t =10 desired user is 45° degree with interfering signal at -68°

The fig. 4 shows the Array Factor (AF) for LMS algorithm with value of array element t =10, step size μ=0.05 and desired user is 45° degree and interference at -68°.

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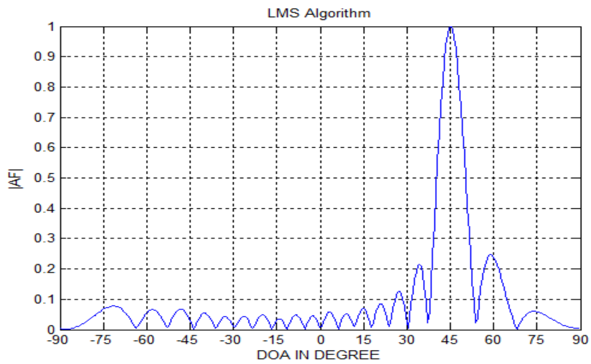

Fig.5.AF when t =20 desired user is 45° degree and interference at -68°.

For same specification with increasing number of array element t=20 found that width of main lobe is decreases and side lobe are suppresses means the value of interfering signal decreases. The accuracy of beam is increase when number of array element increases.

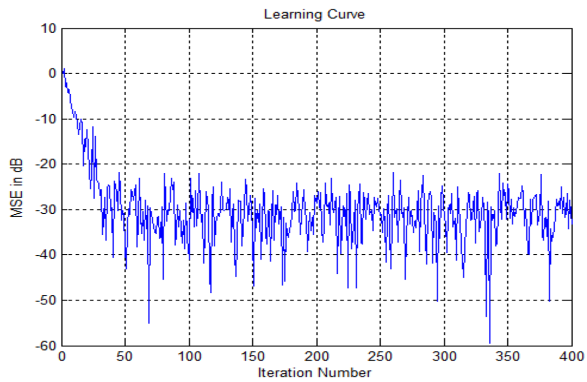

Fig.6. Learning Curve for LMS when number of iteration is 400.

As Fig.6 show learning curve of MSE with number of iteration 400 for LMS algorithm when array element t=10 and step size μ=0.05.

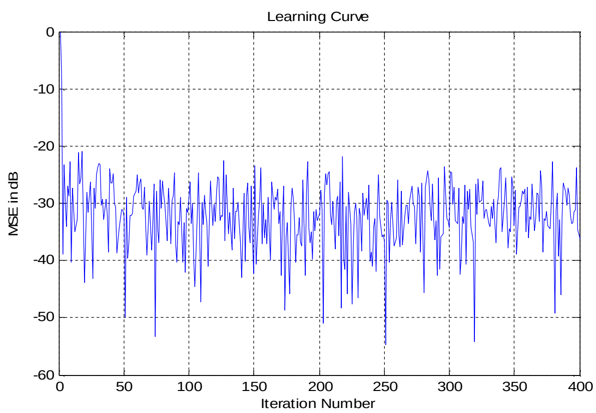

Fig.7. Learning Curve for RLS when number of iteration is 400.

Same specification with forgetting factor λ=0.99 for RLS algorithm fig. 7 shows learning curve is take less number of iteration to stable.

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Table: MSE with No. of Iteration

No. of Iteration	MSE in dB for LMS	MSE in dB for RLS
2	-0.53	-1.29
5	-4.43	-23.56
10	-8.54	-22.46
20	-13.95	-26.98
100	-27.69	-28.23
200	-27.95	-29.58
400	-28.53	-30.11

Table shows value of MSE for LMS and RLS algorithms. For LMS when number of iteration 2 to 20 then value of MSE decreases linearly then after value become stable. But for RLS when number of iteration 2 to 10 then value of MSE decreases more rapidly than LMS after it become stable. As comparing the value of MSE of it conclude that RLS is more stable and efficient than LMS and gives less error value.

VI. CONCLUSION

The comparative study of adaptive beamforming algorithms LMS and RLS used adaptive filter and evaluate the performance parameter like AF, number of array element, MSE etc. When number of array element increases the width of main loop is become narrow and gives more strength beam for desired direction. The number of side loop increases but value of AF decreases. That means interference of signal become less and less. The ability of RLS to minimize the value of MSE earlier than LMS make more efficient and improves convergence rate.

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Vol. 4, Issue 6, June 2016

BIOGRAPHY

Akahay. G. Kathane is a M.E. student in the Electronics & Telecommunication Department, Sinhgad College of Engineering, Pune University, Maharashtra, India. He complete final year project under the guidance of Assistance professor **D. G. Ganage** in the Electronics & Telecommunication Department, Sinhgad College of Engineering, Pune University, Maharashtra, India and **S. A. Wagh** Assistant Professor, in the Electronics & Telecommunication Department., SITS, Pune University, Maharashtra, India