Design of Adaptive Filters for Satellite Communication

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Abstract: In the last few years, many ruthless digital platforms have been developed for secure satellite communication. These platforms finds wide application in military oriented programs and are largely used for command, tracking and telemetry. Adaptive digital filters are used to enhance the performance of the standard direct sequence spread spectrum by suppressing the interference. To increase the robustness of on-board transponder for jammer attack, the SST equipment for next programs will be featured with adaptive digital rejection filter. Even in DSSS systems, it make use of much larger bandwidth for transmission than that of which is required for the transmission of the information. By using this excess bandwidth the system becomes advantageous as it will be less sensitive to multiple interferences, which may include jammers too. Adaptive Interference Rejection (AIR) algorithms are based on Lest Mean Square (LMS), Normalized Least Mean Square (NLMS) technique. Adaptive Filter is a type of filter which has the quality of self-designing in a non-stationary environment by operating on a recursive algorithm. Structural basis for implementation of adaptive filter is by using transversal filter. Tap weights are used to define the finite impulse response. Optimum filter is obtained by minimizing the mean square error of the error signal which is the difference between desired response and the transversal filter output. The dependence of the mean-squared error on the unknown tap weights may be viewed to be in the form of a multidimensional paraboloid with the unique minimum point, where the paraboloid is referred as error performance surface.

Keywords: **TTC transponder, Adaptive Interface Rejection (AIR), LMS algorithm, NLMS algorithm, MMSE, Gradient Vector.**

I. Introduction

Communication is the process of transmitting messages or information from one point to another. The performance of the communication system gets break down due to the additive noise. It's an unwanted signal that gets added up during the process of communication. Handling of noise is the major challenge in communication system. In general, communication receiver uses loop filter and data filters to remove the noise. Implementation of these filters decides the performance of communication receiver. In digital domain FIR filter and IIR filters can be used. But these filters are pre-designed by considering the channel noise. In deep space missions and multi-access environment like CDMA systems it is difficult to characterize the channel noise. Because of that filter performance will not be up to the mark which in turn affects the performance of the receiver. To solve the above stated problem, adaptive filters are used where the filter co-efficient are calculated depending on the input signal. Many ruthless digital platforms have been developed for satellite secure communication. Adaptive digital filters are used to enhance the performance of the standard direct sequence spread spectrum by suppressing the interference. A well-known, DS/SS systems uses a larger bandwidth for transmission than that which is required to transmit the information. Adaptive Interference Rejection (AIR) algorithms based on Least Mean Square (LMS), Normalized Least Mean Square (NLMS). In many areas of digital signal processing applications, adaptive filters are widely used for echo cancellation, noise cancellation, channel equalizer and industrial applications.

To design a linear filter with a noisy data as input, we can minimize the effects of noise at the filter output according to some statistical criterion. The filter-optimization problem can be minimized by using the approach known as mean-square value of the error signal that is defined as the difference between some desired response and actual filter output. When the statistical characteristics of the input data match the prior information, the filter is said to be optimum based on its design. When the information is completely unknown, it may not be possible to design the optimum wiener filter. So estimate and plug procedure is an efficient approach that we may use in above mentioned situations. This is a two-stage processes whereby the filter first "estimates" the statistical parameters of the relevant signals and then "Plugs" the result so obtained into a non-recursive formula for computing the filter parameters. In real-time operation, the disadvantage of this procedure is it requires excessively elaborate and costly hardware.

Adaptive filter will be the more efficient method to use. These devices will be capable of self-designing in the environment where the signal characteristics is unknown by using the recursive algorithm. The algorithm starts with some initial conditions which are predetermined and it completely ignores the environment. A wide variety of recursive algorithm have been developed for the operation of adaptive algorithm. The choice of one algorithm over another is determined by various factors and they are rate of convergence, Misadjustment, robustness, computational requirements, structure and numerical properties.

There is no unique solution for the problem of adaptive filtering. Rather, we have "kits of tools" represented by a variety of recursive algorithm, each of which offers a desirable features of its own. There are three distinct methods for deriving recursive algorithm for the operation of adaptive filter and they are A) Approach based on wiener filter theory B) Approach based on Kalman filter theory C) Methods of least squares. Least mean square (LMS) algorithm is one of the most widely used and popular adaptive filter algorithms. It is simple for implementation and has satisfactory convergence behavior.

II. Related work

L.Simone, D.Gelfusa, S.Ciarcia, G.Fittipaldi,[1] has considered On-board TTC Transponder is the major element of the communication link between satellite and ground station for secure communication. Adaptive digital filters are used to enhance the performance of the standard direct sequence spread spectrum by suppressing the interference. Gary J. Saulnier, Pankaj K.Das, Laurence B. Milstein,[2] has implemented the widrow-hoff LMS algorithm, where adaptive system is used to suppress the narrow band interference in DSSS. Several narrow-band interference are considered for the experiment. Laurence B. Milstein, Ronald A.Iltis [3] has explained the advantage of using larger bandwidth than required bandwidth to transmit the information. By this the system becomes less sensitive to many types of interference. S. Haykin [4][5] in 1996 has published a book called "Adaptive Filter Theory". In digital signal processing adaptive filters finds application in many fields like industrial application, noise cancellation, echo cancellation and channel equalizer. Widrow –hoff proposed the least mean square algorithm for the problem of adaptive filtering and it is the standard algorithm for many DSP applications.

An adaptive filter based on least mean square is composed of finite impulse response filter where its co-efficient are updated by the LMS algorithm. A.H.Sayed in 2003[6] wrote a book "Fundamentals of Adaptive filtering", which explained about basic concepts of adaptive filter. Derivation of wiener-hopf equation, modifying that equation using "Method of steepest decent" which was the well-known optimization theory. Estimation of gradient vector which results to the famous algorithm called Least Mean-Square algorithm. Mrinal Rahul Bachute analysed the of various LMS algorithm [7] that is carried out. The implementation of existing LMS and RLS is done. The analysis says that the LMS has fast convergence and also it is less complex in computation when compared with RLS algorithm. Certain modification in LMS algorithm leads to the advancement of Leaky LMS, CLMS, SLMS and SLMS. P.Priya, Dr.P.Babu constructed a productive architecture to obtain optimized balance pipeline and novel partial product generator from the delayed LMS adaptive filter [8] [9] is proposed in this paper. The filter was designed fixed point implementation but can also be implemented for various filter lengths. Less area and power can be achieved. L. Bharani, P. Radhika considered the normalized least mean square error algorithm [10] which is most popular due to its simplicity. The disagreement of fast convergence rate and minimum mean square error associated with a fixed step size NLMS are sorted using an optimal step size NLMS algorithm. The proposed algorithm has superior performance in convergence and final error i.e fast convergence and low error.

III. On-Board TTC Transponder

The communication link between the satellite and ground station is the On-board Telemetry Tracking and Command (TTC) Transponder. It helps in transmission of Telemetry data (TM) data for downlink, Telecommand (TC) data as the reception of uplink and satellite tracking through range and range rate measurement. It increases the performance like demodulation, modulation, flexibility for data rate and has high anti-jamming capacity by converting analogues technology to digital technology. Advantage of using transponders are high reliability, robustness, easy re-configurability, high modularity and high anti-jamming performance.

a) Adaptive Interference Rejection Filtering

On-board transponder's robustness against jamming attack is increased by using adaptive digital rejection filter in SST equipment. A well-known, DS/SS systems uses a larger bandwidth for transmission than that which is required to transmit the information. The advantage of large bandwidth is that the system becomes less sensitive to many type of interference, including jammer. To further aid the spread spectrum receiver in suppressing Continuous Wave (CW) or narrow-band jammer, Adaptive Interference Rejection (AIR) algorithms based on Lest Mean Square (LMS) and Normalized Least Mean Square (NLMS) technique can be employed.

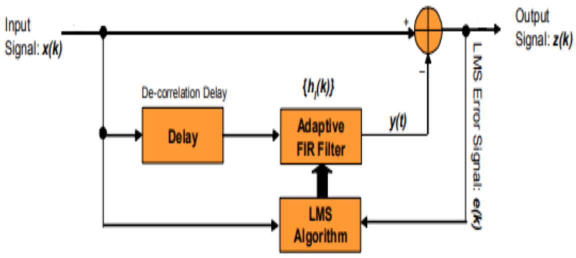

Figure 1. Adaptive Interference Rejection (AIR) based on LMS algorithm

Adaptive filtering algorithms are used to separate the periodic component from the broadband component of the input signal by exploiting the difference between the correlation times of the CW interfering and the useful spread spectrum signal: • The delayed version of the input signal is adaptively filtered and then subtracted from the real-time input signal to form a difference signal that serves as the error term for the adaptive algorithm. • The adaptive algorithm controls the weights of the adaptive filter in order to minimize the power of the difference signal.

IV. ADAPTIVE ALGORITHMS

Adaptive algorithms are used to adjust the weights of the digital filter. Such that the error signals e_k is minimized according to some criterion. The most commonly used adaptive algorithms are least mean square (LMS), normalized least mean square (NLMS) because of their robustness and simplicity.

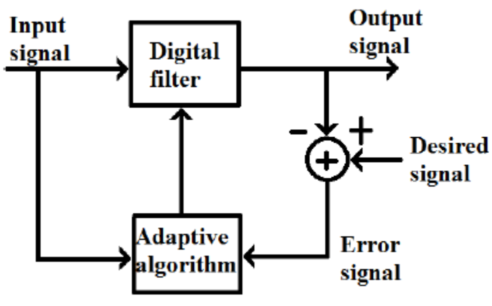

Figure 2. Basic adaptive filter

a) Least mean square (LMS) algorithm

LMS algorithm is one of the most successful and most commonly used adaptive algorithms for suppression of noise. Instead of computing W_{opt} in one go as suggested by wiener–Hopf equation, in the LMS the coefficients are adjusted from sample to sample in such or way as to minimize the MSE.

The LMS algorithm is based on the steepest descent algorithm where the weight vector is updated from sample to sample as follows

$$\mathbf{w}_{k+1} = \mathbf{w}(n) - \mu\nabla(n) \tag{1}$$

Where **w** (n) and **∇**(n) are the weight and the true gradient vectors respectively at the n^{th} sampling instant. M controls the stability and rate of convergence.

The steepest descent algorithm in the above equation still requires knowledge of R and P. since ∇_n is obtained by evaluating the equation.

$$\nabla = \frac{d\mathcal{E}}{dw} = 0 - 2P + 2RW \tag{2}$$

The LMS algorithm is a practical method of obtaining **w**(n) estimates of the filter weights in real time without matrix inversion in the equation $W_{opt} = R^{-1}P$ or the direct computation of the auto correlation and cross correlation.

$$\nabla = -2P + 2RW$$

In the LMS algorithm, immediate estimates are used for the calculation of gradients. Thus n

$$\nabla(n) = -2P + 2RW \tag{3}$$

$$\nabla(n) = -2x(n)\, y(n) + 2\, x(n)\, x^T(n)\, W(n) \tag{4}$$

$$\nabla(n) = -2x(n)\,(y(n) - x^T(n)\, W(n)) \tag{5}$$

$$\nabla(n) = -2x(n)e(n) \tag{6}$$

Where, $e(n) = y(n) - x^T(n)\, w(n)$

Replace, the value of ∇_n in steepest descent algorithm yields

$$w(n+1) = w(n) - \mu\nabla(n) \tag{7}$$

by substituting (6) in (7), we get

$$w(n+1) = w(n) - \mu(-2x(n)e(n)) \tag{8}$$

$$w(n+1) = w(n) + (2\mu x(n)e(n)) \tag{9}$$

Clearly, the LMS algorithm above doesn't require prior knowledge of the correlations R and P, but instead uses their immediate estimates as shown above.

The weights obtained by the LMS algorithm are only estimates, but these estimates improve gradually with time as weights are adjusted and the filter learns the characteristics of the signals. Eventually, the weights converge.

The condition of convergence is

$$0 < \mu < 1/\lambda_{max} \tag{10}$$

Where, λ_{max} is the maximum Eigen value of the input data covariance matrix.

The simplicity of the LMS algorithm and ease of implementation, make it the algorithm of first choice in many real-time systems. The LMS algorithm requires approximately 2N+1 multiplications and 2N+1 addition for each new set of input and output samples.

Most signal processors are suited to the mainly multiply accumulate arithmetic operations involved, making a direct implementation of the LMS algorithm attractive.

Table 1: Inputs to the LMS filter

LMS Algorithm	
Signal to noise ratio	**1**
Number of taps	**10**
Step size mu	**0.05,0.005**

Step Size (mu) = 0.05

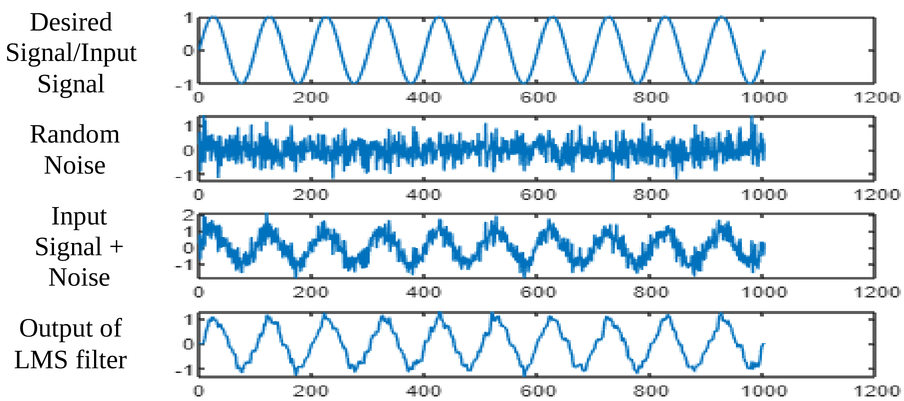

Figure 3. Output of LMS Filter

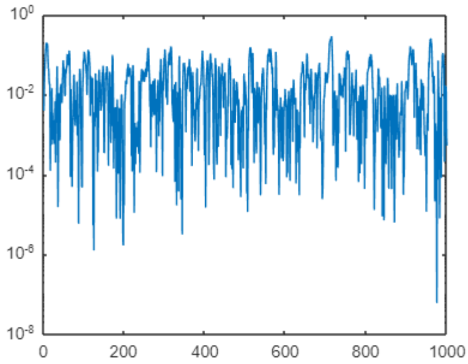

Figure 4. Minimum mean square error

Step size (mu) =0.005

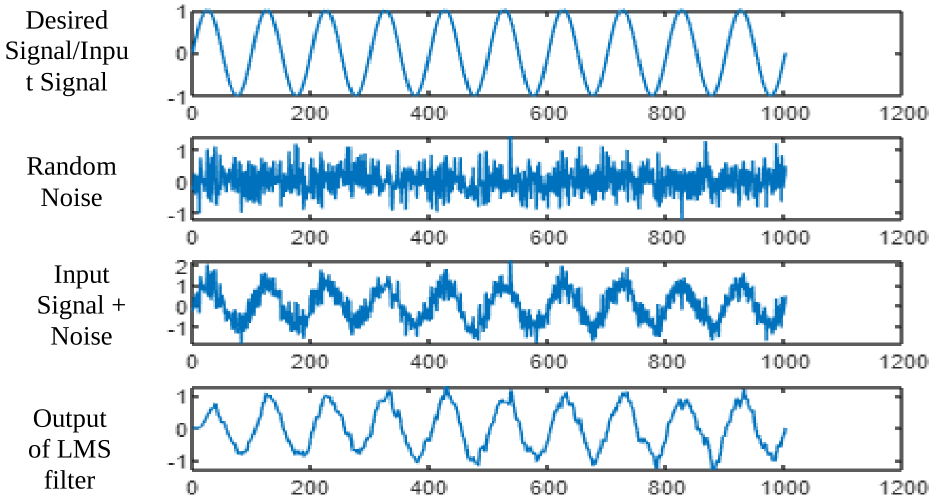

Figure 5. Output of LMS Filter

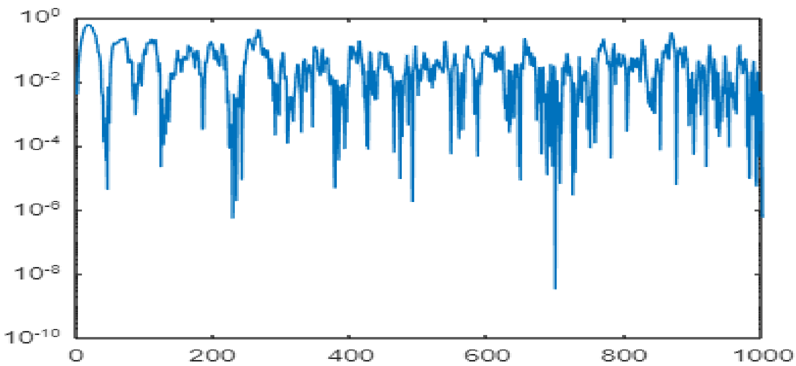

Figure 6. Minimum mean square error

By considering the above graphs we can observe that,

1. The step size with μ= 0.05, has the faster rate of convergence and also the number of iterations will be less and result will not be accurate. Minimum mean square error will be more.
2. The step size with μ= 0.005, has slow convergence and there is no much fluctuation in weights. Minimum mean square error will be less.

b) Normalized least mean square (NLMS) algorithm

The scaling of input is the drawback of the pure LMS algorithm because it is sensitive to it. As a result choosing of the step size becomes difficult to get a stability of the algorithm.

The normalized least mean square filter is a variant of LMS algorithm that solves this problem by normalizing with the power of input.

$$w(n+1)= w(n)+ \mu x(n)e(n) \qquad \text{where } \mu(n) = \frac{\mu}{x(n)\, xT(n)}\, x(n)e(n) \qquad (11)$$

In the previous equation, the NLMS algorithm becomes the same as the standard LMS algorithm except that the NLMS algorithm has a time varying step size μ_n. The convergence speed of adaptive filter is improved by the step size.

In practice, at some time x_n can be very small. The necessary modification to have a robust NLMS algorithm is as follows,

$$w(n+1) = w(n) + \frac{\mu}{\delta + x(n)\, x^T(n)} x(n)e(n) \qquad (12)$$

So, that the gain constant can't go to infinity.

Table 2. Inputs to the NLMS filter

NLMS Algorithm	
Signal to Noise ration	**1**
No. of taps	**10**

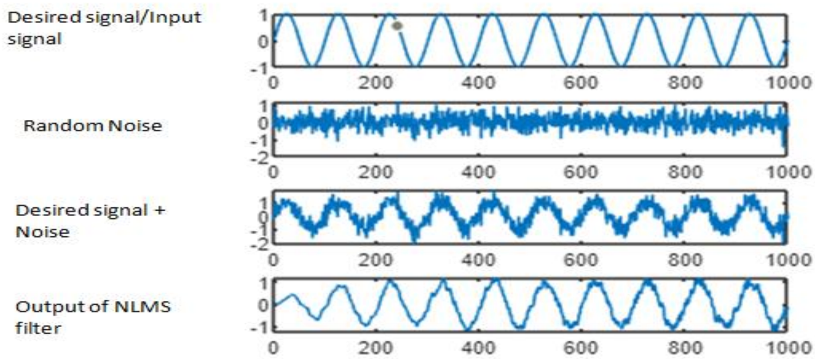

Figure 7. Output of NLMS Filter

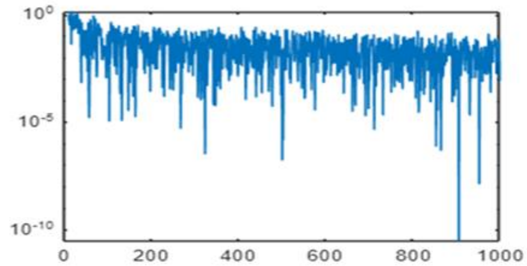

Figure 8. Minimum mean square error

V. Comparison b/w LMS and NLMS algorithm

Here two algorithms Mean Square (LMS) and Normalized Least Mean Square (NLMS) are described on comparative level. For both algorithms if we perform application of adaptive filter for error reduction where NLMS uses normalization of step size can optimize speed of convergence. , these algorithms were analyzed and compared by investigating the effects of different system parameters such as input SNR, no. of iterations and filter length on the performance.

As LMS has a fixed step-size, it is not suitable to operate in non- stationary environment. But for NLMS the step–size changes according to the energy of the input signals. Hence, it is suitable to operate in non-stationary environment as well as in stationary environment. NLMS algorithm has the better noise reduction capability and faster convergence speed than LMS algorithm due to its upgradeable step-size. At high noise environment, NLMS has the lower mean squared error among the two but at very low noise environment the MSE for LMS is better than NLMS. This is mainly because at low noise environment, the noise power is too low to upgrade the step-size parameter of NLMS algorithm.

Table3 .Inputs to the LMS and NLMS filter

Adaptive Algorithms	
Signal to Noise ration	1
No. of taps	5
mu	0.02

LMS Algorithm

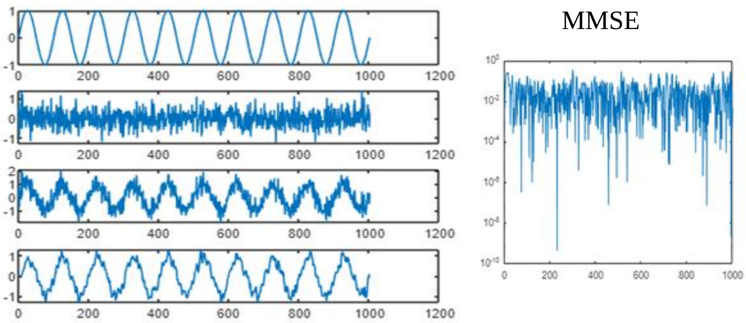

Figure 9. Output of LMS filter and Minimum mean square error.

NLMS Algorithm

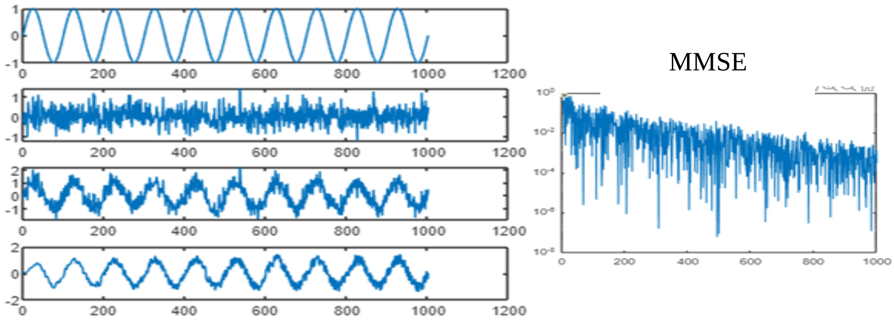

Figure 10. Output of NLMS filter and Minimum mean square error

By considering the above graphs we can observe that,

1. The performance of NLMS filter is better than LMS filter when the output is considered for same input signal.
2. The convergence is fast and the minimum mean square error is low, when NLMS filter is considered.

VI. Conclusion

When the algorithm are analyzed and compared their design, the effects of different system parameters such as input signal to noise ratio, number of iteration, filter length on the performance of the adaptive noise cancellation system. LMS algorithm has fixed step-size, it is not suitable in non-stationary environment. But if NLMS algorithm is considered, the step-size changes according to the energy of the input signal. Hence it is suitable to operate in non-stationary environment as well as in stationary environment. NLMS algorithm has better noise reduction capability and faster convergence speed than LMS algorithm due to its upgradeable step-size.

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