A NOVEL ADAPTIVE ALGORITHM FOR REMOVAL OF POWER LINE INTERFERENCE FROM ECG SIGNAL

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***ABSTRACT*: *During data acquisition and transmission of biomedical signals like electrocardiography (ECG), different types of artifacts are embedded in the signal. Since an ECG is a low amplitude signal these artifacts greatly degrade the signal quality and the signal becomes noisy. The sources of artifacts are power line interference (PLI), high frequency interference electromyography (EMG) and base line wanders (BLW). Different digital filters are used in order to reduce these artifacts. ECG signal is a non-stationary signal, it is difficult to find fixed filters for the removal of interference from the ECG signal. In order to overcome these problems adaptive filters are used as they are well suited for the non-stationary environment. In this paper a new algorithm "Modified Normalized Least Mean Square" has been proposed. A comparison is made among the new algorithm and the existing algorithms like LMS, NLMS, Sign data LMS and Log LMS in terms of SNR, convergence rate and time complexity. It has been observed that the performance of new algorithm is superior to the existing ones in terms of SNR and convergence rate however it is more complex than the other algorithms. Results of simulations in MATLAB are presented and a critical analysis is made on the basis of convergence rate, signal to noise ratio (SNR), and computational time among the filtering techniques.*

Keywords: *Electrocardiography, Power Line Interference, Least Mean Square, Normalized Least Mean Square, Signal to Noise Ratio*

1. INTRODUCTION

Electrocardiogram (ECG) is a very significant physiological signal of the human body. It contains information about the performance of the human heart. A doctor can detect many types of irregularities /diseases by analyzing waveform of the ECG signal. In short ECG to a human body is the same as a trouble shoot message to a digital system since a trouble shoot message detects hardware malfunction. Similarly ECG waveform can trigger alarms in case of danger or prior heart attacks.

One of the major issues in biomedical signal processing is the presence of outliers/artifacts. The major artifacts arise in ECG signal may be due to technical fluctuation like Power line interference, Electromagnetic fields of different electronic devices, Instrumentation noise or it may have biological resources such as random body movements, muscular reaction and respiratory movements[8].These artifacts affect the signal quality and lead to wrong clinical diagnosis. The artifacts need to be suppressed or removed in order to achieve reliable and improved signal quality and thus providing a higher level of health caring activity to the subject. One of the major artifacts in ECG signal is the Power line interference (PLI).The PLI is 50/60 Hz depending upon the supply. The frequency content of ECG signal is also nearly the same. Many techniques exists in the literature to remove the PLI such as low pass filter, Notch filter [1]. Since these are fixed filters and most of the information in the ECG signal has the same frequency as PLI. So these filters eliminate some useful information from the ECG signal. Adaptive filter are a useful technique for the removal of artifacts from biomedical signal. In 2010, Dr K.L Yadav and S.Singh have make use of LMS and RLS adaptive algorithm in the paper titled "Performance evaluation of different adaptive filter for ECG signal processing"[2]. In another paper "Efficient sign based normalized adaptive filtering techniques for cancellation of artifact in ECG signal; Application to wireless biotelemetry" by M.Z Rahman, R.A Shaik and D.V Rami Koti Reddy have make use of various simple and effective sign based normalized adaptive filters for cancellation of artifacts in ECG ,which are found computationally superior and efficient having multiplier free for update loop[3].For the removal of artifacts from ECG signal an unbiased and normalized adaptive filtering techniques were also used in the paper title "Filtering of Noise in Electrocardiographic Signals Using An Unbiased and Normalized Adaptive Artifact Cancellation System" by Yunfeng Wu, Rangaraj M. Rangayyan Ye Wu, and Sin-Chun Ng [9] In 2012, performance comparison is made between Modified LMS and RLS algorithm in the paper title "Performance comparison of modified LMS and RLS algorithms in Denoising of ECG Signal"[4]. An ECG signal consists of sudden large peaks. In existing adaptive algorithms steady state is destroyed when a sudden large peak comes and hence the primary convergence is destroyed. We have proposed a novel approach to hold the primary convergence. Our proposed approach produces better results than the existing ones and hence can be applied to automatic ECG analysis. In E-health or automatic ECG analysis PLI should be more accurately filtered because it overlaps the frequency spectrum of the ECG signal which may lead to wrong ECG analysis or diagnosis. For removal of PLI, Our proposed approach produces better results in term of convergence rate and SNR. So it can be applied in automatic ECG analysis applications.

2. Adaptive Algorithm

FIR based adaptive filters are preferred due to their stability. In adaptive noise cancellation system a refrence signal i-e corelated with the noise is fed input to the adaptive filter as shown in fig 1.Desired signal d(n) is the combination of the primary signal that is informtion signal x(n) and noise .

$$d(n) = x(n) + noise \quad (1)$$

$y(n)$ is the filter output and is calculated using Equation (2).

$$y(n)=w^T(n) * x(n) \quad (2)$$

Filter output vector $y(n)$ can be expressed by Equation (3)

$$y(n)=[y(0)y(1)y(2)\ldots\ldots y(n-1)]^T \quad (3)$$

$x(n)$ and $w^T(n)$ are input signal and tapped weight vector and are expressed in vector form by Equation 4 and Equation 5 respectively.

$$x(n)=[x(0)x(1)x(2)\ldots\ldots x(n-1)] \quad (4)$$

$$w(n)=[w(0)w(1)w(2)\ldots\ldots w(n-1)]^T \quad (5)$$

Error signal is obtained by subtracting the filter output $y(n)$ with the received signal $d(n)$ and given by the Equation (6).

$$e(n)=d(n) - y(n) \quad (6)$$

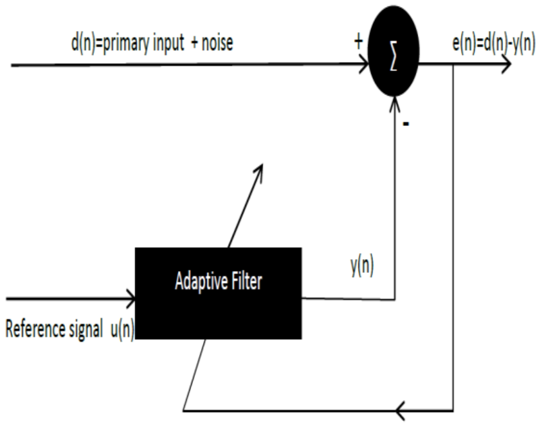

Figure 1. Adaptive Noise Cancellation System

2.1 Least Mean Square Algorithm

The LMS algorithm was first proposed by Widrow and Hoff in 1959 through their studies of pattern recognition [5]. It is based upon stochastic gradient algorithm which finds optimal weights using the gradient vector to converge on an estimated optimal solution. It is simple, less complex and probably the most widely used algorithm.

The weight vectors in each iteration are updated in the LMS algorithm according to the following equation.

$$w(n+1)=w(n) + \mu * e(n) * x(n) \quad (7)$$

Here $w(n+1)$ is the adapted weight and w(n) is the previous weight. μ is the step size that determines how much of the error to be adjusted. It directly affects the convergence rate. A large step size converges faster but may not lead to the optimal solution whereas optimal solution is achieved by selecting the smaller step size but it converges slowly. $e(n)$ is the error signal and $x(n)$ is the input signal.

2.2 Normalized Least Mean Square Algorithm (NLMS)

In simple LMS algorithm it is difficult to choose the step size. This problem is solved by the NLMS algorithm**.** In the NLMS algorithm step size becomes adaptive. Due to the adaptivity of step size this algorithm found stable and have fast convergence rate as compared to the LMS algorithm [10].

The coefficient weight vector update relation for the NLMS algorithm is given by the following equation.

$$w(n+1)=w(n) + \mu/[x(n)*x'(n)] * e(n) * x(n) \quad (8)$$

Here the step size is normalized using the above relation.

2.3 Sign Data LMS Algorithm (SDLMS)

The Sign Data Least Mean Square algorithm is obtained by the well-known LMS algorithm by replacing the input signal vector $x(n)$ with the vector $sgn\{x(n)\}$ [8].The advantage of using the Sign Data LMS algorithm is the less computational complexity as sign {.} of every function is calculated very fast and it can be used in high speed bio telemetry applications. The weight updating formula for the Sign Data LMS algorithm is same as the simple LMS except that the input signal is quantized by using the well-known sign function.

$$w(n+1)=w(n) + \mu * e(n) * sgn\{x(n)\} \quad (9)$$

$$sgn(x) = \begin{cases} 1 & x > 0 \\ -1 & x < 0 \\ 0 & x = 0 \end{cases}$$

The computational complexity of the Sign Data LMS algorithm is also less than the simple LMS because of the multiplication operation is replaced with the shifting operation.

2.4 Log Least Mean Square (LLMS) Algorithm

In Log LMS algorithm, instead of quantizing the input signal the error signal is quantized using the log function.

$$w(n+1)=w(n) + \mu * Q\{ e(n) \} * x(n) \quad (10)$$

$Q(\cdot)$ is the logarithmic quantizing function. It will simply convert input into the power-of-two value [4] .The quantizing function is defined by the following relation.

$$Q(z)=2^{l \log_2 (|z|)} * sgn(z) \quad (11)$$

2.5 Proposed Modified Normalized Least Mean Square (MNLMS) Algorithm

In above discussed algorithms the output of the system $e(n)$ is used to update the filter coefficients. Steady state is destroyed when large peak come into the periodic life signal. It results in signal distortion. The iteration function for the modified NLMS is given by the following formula:

$$w(n+1)=w(n) + \mu/[x(n)*x'(n)] * Q\{e(n)\} * x(n) \quad (12)$$

Where $Q[.]$ represents the logarithmic quantization function and is defined by equation 13.

$$Q(z)=\alpha * 2^{\theta(z)} * sgn(z) \quad (13)$$

In this algorithm we have divided the processing of ECG signal in two different stages, Convergence stage and Extracting stage. In first stage, the proposed algorithm iterates with the rapid convergence rate to achieve a primary convergence. When primary convergence is achieved the algorithm enters into the extracting stage.

$$\Theta(z)= \log_2 |z/\alpha|$$

At convergence stage α represents the power of two values and is always less than 1.The term z/α is used to amplify the ECG signal to integer type by shifting operation. When convergence state is achieved primarily the algorithm iterates for the extraction of information from the signal using the following relation.

$$\Theta(z)= - \log_2 |z/\alpha|$$

At extracting stage Switching between the above two stages, a threshold value € is introduced. When output of the system becomes less than or equal to € for large number of consecutive iterations that is

$$|e(n)- e(n-1)| < €$$

Than we can conclude that the convergence state is achieved and algorithm enters in the second stage

3. Implementation of Algorithms

Adaptive algorithms discussed in this paper are implemented in MATLAB R2012b. Five records of the original biomedical ECG signal were extracted from the record MIT-DB Arrhythmia database. However the plots are shown only for the record number 100 of 10 seconds recording of ECG signal containing 3600 samples as the sampling rate of the signal is 360.The Power line interference (PLI) was generated in MATLAB by using a sine wave of 60 HZ and is then added to the original signal to make it noisy. The step size for linear and nonlinear algorithms was chosen to be 0.005. The reference signal was generated in MATLAB by using the knowledge about the frequency content of the artifact. The all discussed adaptive algorithms were implemented on this noisy ECG signal and their performance was evaluated in terms of SNR, Convergence rate and Time complexity.

4. SIMULATION RESULTS

In simulation input to adaptive filter is a signal correlated with the PLI type noise. The filter order was chosen to be two and step size chosen to be 0.005. Figure 2 & 3 shows the original ECG signal and ECG signal corrupted by PLI type noise.

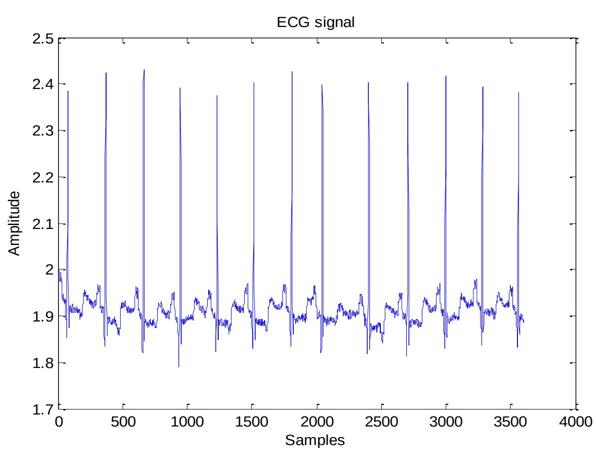

Fig.ure 2. Original ECG signal

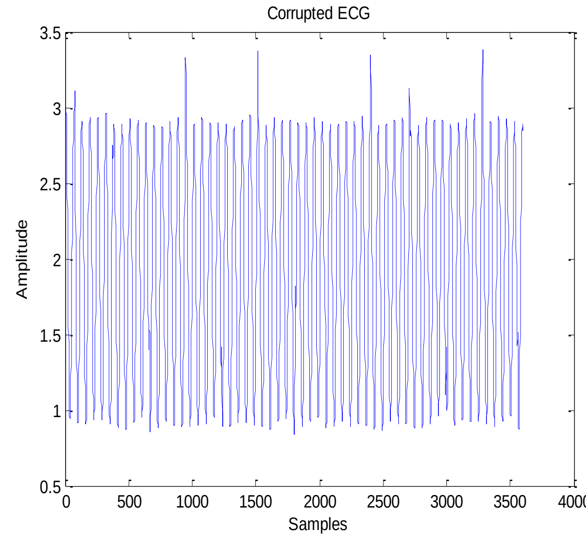

Figure 3. Corrupted ECG signal by PLI type noise

4.1 RESULTS OF REMOVING ARTIFACT FROM ECG SIGNAL

Step size for each algorithm was taken to be 0.005.

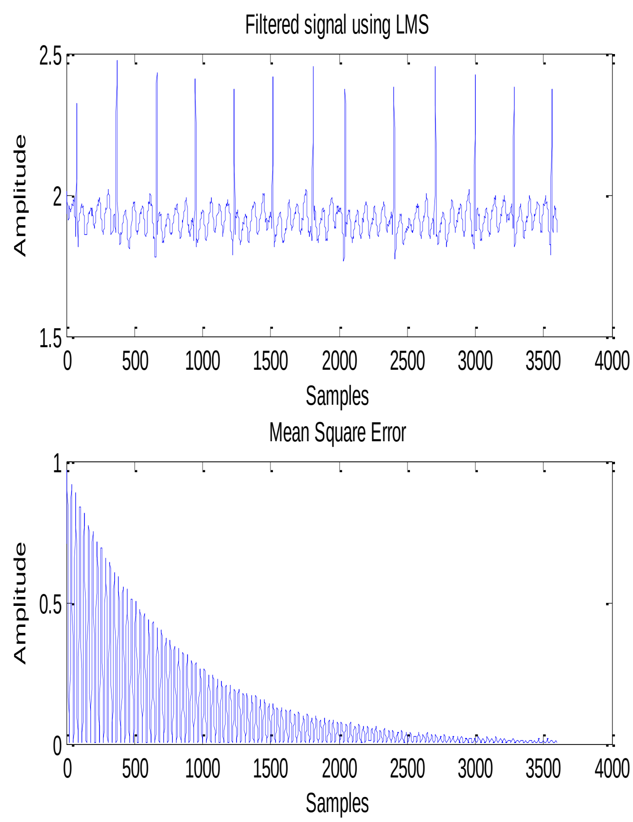

Figure 4. Filtered signal & MSE plot using LMS

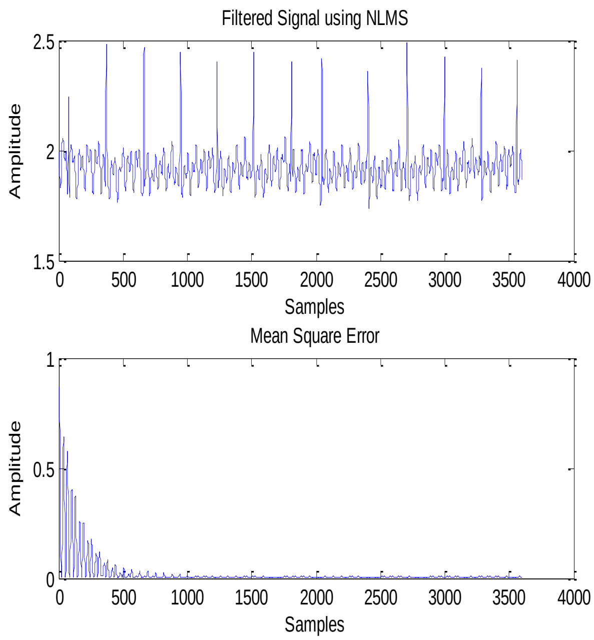

Figure 5. Filtered signal & MSE plot using NLMS

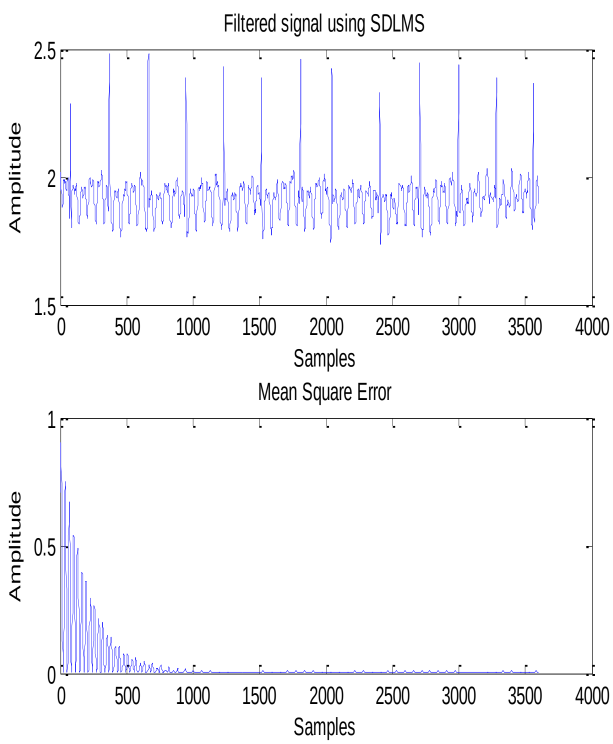

Figure 6. Filtered signal & MSE plot using SDLMS

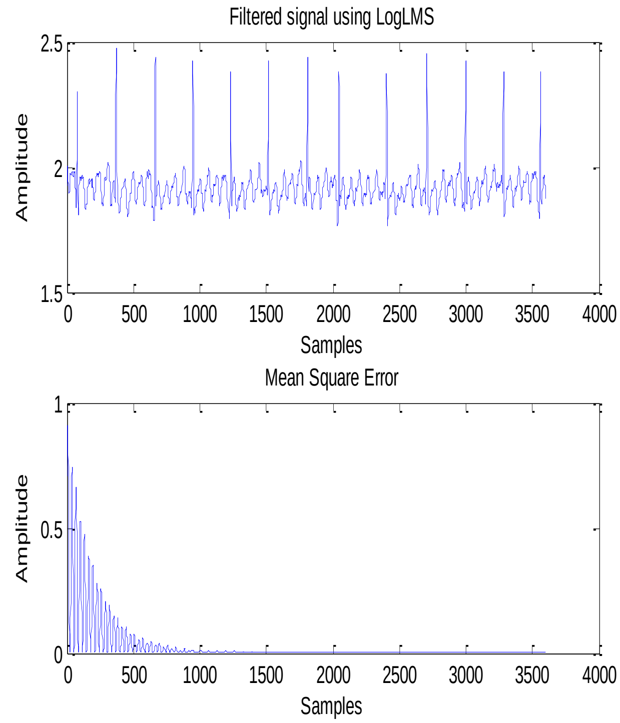

Fig.ure 7. Filtered signal & MSE plot using Log LMS

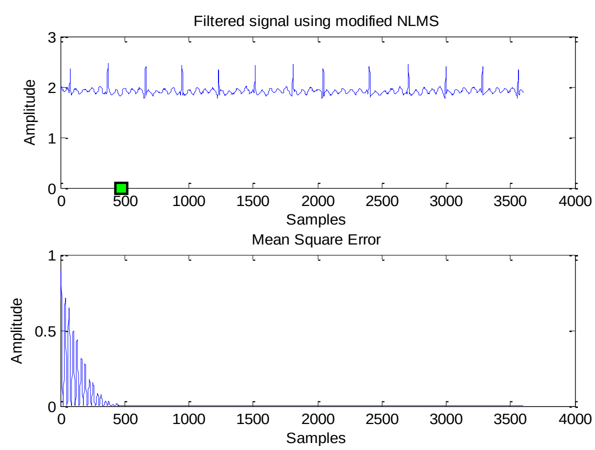

Fig.ure 8. Filtered signal & MSE plot using proposed modified NLMS

Table 1. Analysis of the algorithm in terms of SNR, Convergence Rate & Time complexity.

ECG	Noise	LMS	NLMS	SDLMS	LLMS	MNLMS
SNR	**PLI**	21.67	19.04	19.76	21.86	22.17
Convergence		3000	1000	750	800	500
Time complexity (secs)		0.148	0.170	0.114	1.136	1.506

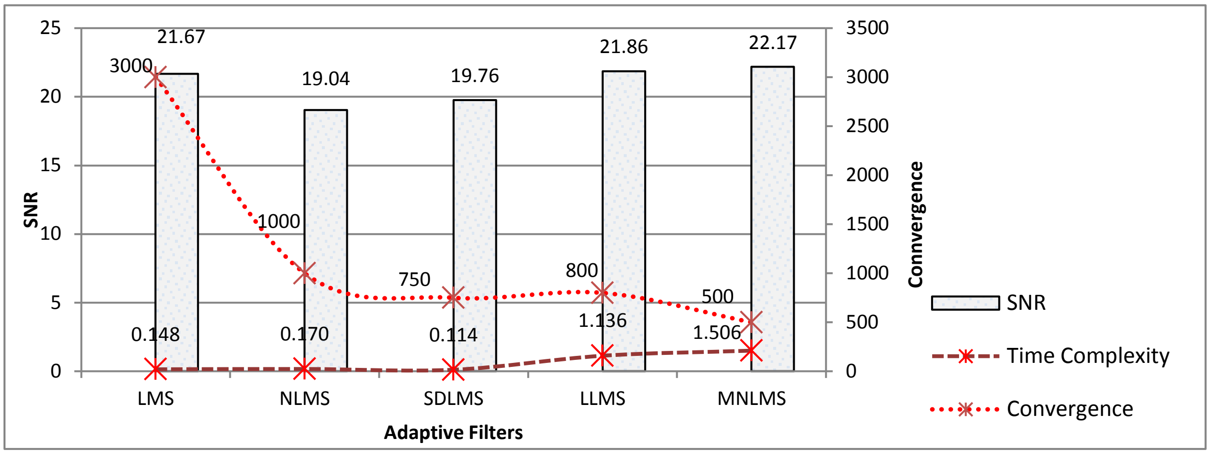

Fig.ure 9. Histogram of PLI type noise

5. FUTURE WORK AND DISCUSSION

We have analyzed the response of different adaptive algorithms in this paper for the removal of PLI type noise and we found that for PLI type noise the MNLMS gave the best SNR and convergence speed at the cost of more time complexity. Since an ECG signal consists of many artifacts so in future one may design a multistage filter having multiple reference inputs whose one filter should be MNLMS type. ECG stills remain an open & extensive area for research for signal processing .Wavelet transform can be used for the extraction of ECG signal and distance between two consecutive R-R peaks can be calculated to determine the heartbeat. In automatic ECG analysis the decision has to be made by the machine so if the machine get more purified signal it can analyze the ECG signal in a better way and the chances of error are minimized. Furthermore, our proposed approach can be implemented in e-health system for the better analysis of biomedical signal.

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