

Performance Analysis of ECG Signals Using LMS, NLMS and NOTCH Filters

Chhavi Saxena¹, Dr. P.D. Murarka², Dr. Hemant Gupta³

Assistant Professor, Department of Computer Science, Arya College of Engineering and I.T., Jaipur, Rajasthan, India¹

Professor, Department of Electronics and Communication, Arya College of Engineering and I.T., Jaipur, Rajasthan, India^{2,3}

ABSTRACT: Electrocardiogram (ECG) signal is the electrical recording of heart activity. It reflects the activities and the attributes of the human heart and reveals very important hidden information. The information extracted from the signal is used to analysis and identifying various pathological conditions, but these ECG signal can be distorted with noise. Noise can be any interference due to motion artifacts or due to power equipment that are present where ECG had been taken. A typical computer based ECG analysis system includes a signal pre-processing, beats detection and feature extraction stages, followed by classification. Automatic identification of arrhythmias from the ECG is one important biomedical application of pattern recognition. Moreover ECG signal processing has become a prevalent and effective tool for research and clinical practices. The motion artifacts are effectively removed from the ECG signal which is shown by beat detection on noisy and cleaned ECG signals after LMS and NLMS processing. This paper focuses on ECG signal processing using Least Mean Square (LMS) Normalized Least Mean Square (NLMS) and Notch Filter which has received increasing attention as a signal conditioning and feature extraction technique for biomedical application.

KEYWORDS: ECG, Signal Pre-processing, Pattern recognition, Noise

I. INTRODUCTION

It is complicated to get accurate result for biomedical signal's recording while patient is diagnosis by medical monitoring equipment such as ECG, EEG, EOG and EMG. The signal can be corrupted by electromagnetic field (EMF) by the machinery which is placed nearby [1] or by the fluctuation in electric supply to which measuring equipment is attached.

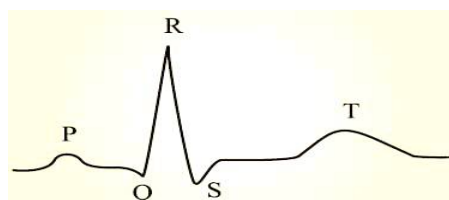


Fig 1.1: ECG Signal

The noise from electric power system is a main source of noise during the recording or monitoring of ECG. Different noises have different frequencies; the noise with low frequency creates problem with ECG signal and some time high frequency noises also interfere ECG i.e. mobile phone. The frequency is measured in cycle/second or in "Hertz". For example the electric power used in daily life is 50 Hz in India [3].

Interference cancellation is widely used in a number of application such as acoustic and speech signal processing, data communication, biological signal acquisition etc. There is different method for interference cancellation in ECG signal,

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the outline of the methods used for interference cancellation is shown in Fig.1.2 The drawback of non-adaptive techniques is that they are time invariant in nature. The above mentioned problem has been overcome by the adaptive methods [6].

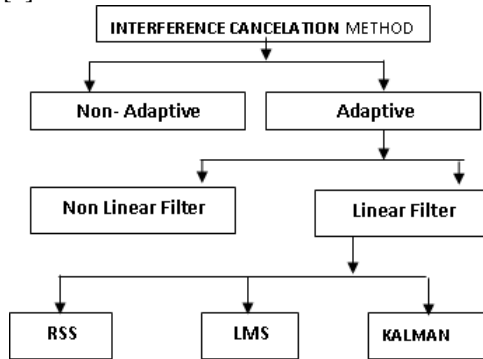


Fig.1.2 Different Methods of interference Cancellation

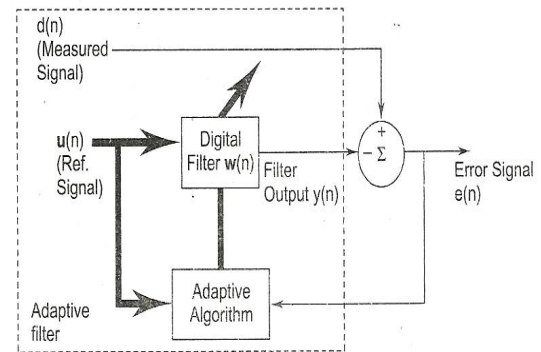


Fig.1.3 Block diagram of an adaptive filter

Adaptive Filter-An adaptive filter self-adjusts its transfer function according to an optimization algorithm driven by an error signal. Because of the complexity of the optimization algorithms, most adaptive filters are digital filter. Block Diagram of adaptive filter is shown in Fig 1.3.

Digital Filter Digital filter is a mathematical algorithm implemented in hard ware and software. Digital input signal produces a digital output signal for purpose of achieving a filter objective. A digital filter is used for two purposes: (i) Separation of signal that have been combined and (ii) restorations of signal that have been distorted in some way signal separations needed when signal has been contaminated with interference, noise, or other signal.

Least Mean Square (LMS) Algorithm-The LMS is an approximation of the steepest descent algorithm, which uses an instantaneous estimate of the gradient vector. The estimate of the gradient is based on sample values of the tap input vector and an error signal. The algorithm iterates over each tap weight in the filter, moving it in the direction of the approximated gradient. The idea behind LMS filters is to use the method of steepest descent to find a coefficient vector $n w$ which minimizes a cost function [4][6]. In practical application of adaptive filtering, a fixed step size Algorithm is required for (a) Easier implementation (b) Single adjustable parameters μ for controlling the convergence rate but slow convergence.

Normalized LMS (NLMS) Algorithm-Normalized Least Mean Square (NLMS) is actually derived from Least Mean Square (LMS) algorithm. The need to derive this NLMS algorithm is that the input signal power changes in time and due to this change the step-size between two adjacent coefficients of the filter will also change and also affect the convergence rate. Due to small signals this convergence rate will slow down and due to loud signals this convergence rate will increase and give an error. So to overcome this problem, try to adjust the step-size parameter with respect to the input signal power. Therefore the step-size parameter is said to be normalized. When design the LMS adaptive filter, one difficulty we meet is the selection of the step-size parameter μ . The main drawback of the pure LMS algorithm is that it is sensitive to the scaling of its input $u(n)$. This makes it very hard to choose a learning rate μ that guarantees stability of the algorithm [7].

When the convergence factor μ is large, the algorithm experiences a gradient noise amplification problem. In order to solve this difficulty, we can use the NLMS (Normalized Least Mean Square) algorithm.

Normalization algorithm results in smaller step size values than the conventional LMS algorithm. The normalized algorithm usually converges faster than the LMS algorithm, since it utilizes a variable convergence factor aiming at the minimization of the instantaneous output error.

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Comparison between LMS & NLMS-ECG signals which are generated contain noise signal. To reduce these noise signals LMS algorithm and NLMS algorithm are used. Here two algorithms are studied and output results for noise cancelling are shown in Fig. 1.4 & Table 1.1 It is clear that these algorithms remove the noise efficiently present in the ECG signal and the performance of NLMS is better than LMS.

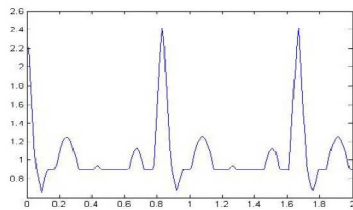


Fig.1.4 (a) Original ECG signal

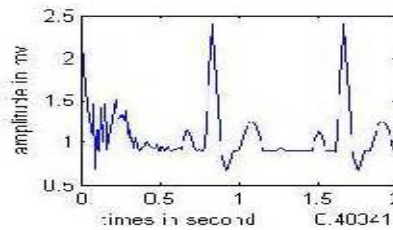


Fig.1.4 (b) Signal using LMS algorithm

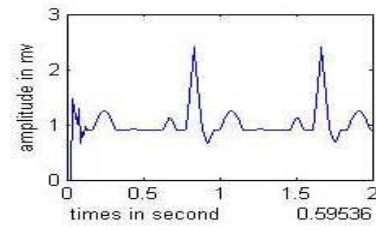


Fig.1.4 (c) Signal using NLMS algorithm

Table 1.1 Summary of ECG noise cancellation

S. No.	Algorithm	Percentage of noise cancellation
1	LMS	48.34%
2	NLMS	59.54%

Notch Filter-Notch filter is used in many applications where specific frequency component is eliminated. For example instrumentation and recording system signals are interfered by power line frequency 50Hz and these interferences are eliminated by notch filter [8].

The Notch filter removes the noise by attenuating the entire signal content at 50 Hz. This result in a loss of the frequency components of the desired signal range around 50 Hz. Above mentioned proposed work is implemented using MATLAB as it is commercial software product, of Math works. This programming language is very powerful allows matrix manipulation, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with other programming languages (C, C++, FORTRAN and Java).One of the most beneficial features is graphical visualization which helps us have confidence in results by monitoring and analysing resultant plots. MATLAB has some advantages compared with conventional computer languages for technical problem solving. Among them are following [2].

- (1) Ease of use.
- (2) Platform independence.
- (3) Predefined functions.
- (4) Device-independent plotting.
- (5) Graphical user interface.

II. RELATED WORK

Electrocardiogram (ECG) has become one of the important tools in the diagnosis of heart disease. Electrocardiography has an important role in cardiology since it consists of effective, simple, on-invasive, low cost way for the diagnosis of cardiac disorders which have a high rate of occurrence and are very relevant to their impact on patient life and social costs (Linh & Osowski, 2003).

Jiashu Zhang et.al stated that the LMS adaptive noise cancellers are often used to recover signal corrupted by additive noise. A major drawback of conventional LMS algorithms is that the excess mean-square errors increase linearly with the desired signal power. This results in degraded performance when the desired signal exhibits large power fluctuations whereas a normalized difference LMS algorithm is proposed to deal with the situation when the desired

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signal is strong, e.g., speech signals. Results demonstrate the superiority of the proposed NLMS algorithm over conventional LMS algorithms in achieving much smaller steady-state excess mean square errors [5].

Jafar Ramadhan Mohammed has given the general idea that the background noise or wideband noise and sinusoidal noise can be significantly suppressed using the proposed adaptive noise cancellation scheme based on adaptive line enhancer and Normalized Least Mean Square filters. The proposed scheme is comprised of two stages. The first stage uses an ALE filters, which are used to reduce sinusoidal noise from input signals, whereas the wideband noise is reduced using Normalized LMS adaptive filter in the second stage. To demonstrate the effectiveness of the proposed scheme, it is compared to the traditional adaptive noise cancellation scheme. The good performance of the proposed scheme has been verified via computer simulations in noisy and reverberant environment [9].

Lin Bai et.al has proposed that the adaptive noise cancellation being used widely to reduce noise from a noisy speech sound. However the least-mean-square algorithm and its variants, such as the normalized NLMS, do not perform well in ANC since the desired speech signal has a bad effect on the convergence rate and steady state miss adjustment of these algorithms. Thus, we propose a new adaptive algorithm that further relaxes the constraint in the CS-LMS algorithm. The new algorithm attempts to minimize the estimation error of the a posteriori error and the estimation is obtained using the concept of Taylor's expansion. The analysis and simulation results show that the proposed new algorithm outperforms the NLMS and CS-LMS algorithms [10].

Different filter structures have been proposed by Thakor et al (1991) to eliminate the base line wander, 60 Hz power line interference, muscle noise and motion artifact[11].

III. METHODOLOGY

In this paper, two new algorithms are proposed, developed and validated, one for removing non-trivial noises in ECGs using the LMS and the other deploys the NLMS extracted features to improve recognition of arrhythmias. Firstly, independent component analysis has been studied and found effective to separate out motion induced artifacts in ECG.

The second algorithm has been developed for ECG feature extraction, in which the independent component analysis has been used to obtain a set of features, or basis functions of the ECG signals generated hypothetically. The selection of the appropriate feature set for classifier has been found important for better performance and quicker response. Above said two algorithms are than compared with Notch Filter

Implementations of LMS Algorithm:

- Initialization: If prior knowledge of the tap weight vector $w(n)$ is available, use it to select an appropriate value for $w(n)$, otherwise, set $w(0)=0$. Take, $0 < \mu < (1/M, S_{max})$ Where: S_{max} = the maximum value of PSD of the tap input $u(n)$
- Data:
 - Give $u(n) = M$ by 1 tap input vector at time n ,
 - $d(n) =$ desired response at time n
 - To be computed: $w(n) =$ estimate of tap-weight vector at time n
- Computation: $y(n) = w(n).u(n)$
 $e(n) = d(n) - y(n)$
 $w(n+1) = w(n) + \mu.e(n).u(n)$

Table 1.2 Variables use in LMS algorithm

Variable	Description
N	The current time index
$u(n)$	The vector of buffer input sample at step $n=[u(n).u(n-1)...u(n-M-1)]^T$
$u^*(n)$	Complex conjugate of the vector of buffered input sample at step n
$w(n)$	The vector of filter weight estimates at step n .
$y(n)$	The filtered out put at step n .
$e(n)$	The error signal at step n .
$d(n)$	The desired input signal at step n .
M	The filter length.
μ	The adaption step size.

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Table 1.3 Least Mean Square (LMS) algorithm

S. No.	LMS algorithm	
1	Initial Conditions :	$0 < \mu < 1$ Length of Adaptive filter=L Input vector: $u[0,0,0\dots 0]^T$ Wight vector: $w[0,0,0\dots 0]^T$
	For each instant of time, $n = 1, 2, 3, \dots$, compute:	
2	Output signal:	$y(n) = w^T \cdot u(n);$
3	Estimation Error:	$e(n) = d(n)-y(n);$
4	Tap-Weight Adaptation:	$w(n+1) = w + 2 \mu \cdot e(n) \cdot u(n);$

Implementations of NLMS Algorithm

In standard form of least mean square (LMS) filter in adjustment applied to the tap weight vector of the filter at iteration n consist of product of three terms:

- (i) The step size parameter μ , which is under the designer's control.
- (ii) The tap input vector $u(n)$, which is supplied by a source of information.
- (iii) The estimation error $e(n)$, which is calculated at iteration n.

The adjustment is directly proportional to the tap input vector $u(n)$. Therefore, when $u(n)$ is large, the LMS filter suffer from gradient noise amplification problem. Also, the maximum step size μ guarantee stability signal, $u(n)$. One important technique for optimizing speed of convergence while maintaining the adjustment applied to tap weight vector at iteration $n+1$ is 'normalized' with respect to the squared Euclidean norm of the tap -input vector $u(n)$ at iteration n -hence the term "normalized". It can be briefly described as follows:

- Initialization: If prior knowledge of the tap weight vector $w(n)$ is available, use it to select an appropriate value for $w(0)$, otherwise set $w(0)=0$. Take, $0 < \mu < 2(E[|u(n)|^2]D(n)/E[|e(n)|^2])$; where, $E[|u(n)|^2]$ = input signal power, $E[|e(n)|^2]$ = error signal power, $D(n)$ mean square deviation of weight vector.
- Data: Given $u(n) = M$ by 1 tap input vector at time n.
 $d(n)$ = desired response at time n.
To be computed $w(n)$ = estimate of tap-weight vector at time n.
- Computation: $y(n) = w(n)^T \cdot u(n)$
 $e(n) = d(n) - y(n)$
 $w(n+1) = w(n) + \mu \cdot e(n) \cdot u(n)$

The algorithm required $2 \cdot M + 1$ addition and $2 \cdot M + 1$ multiplication at any iteration n where M is the tap length or filter order. The computation complexity depends on the order of the filter and it must be carefully chosen. The following figure shown the flow chart to implement the algorithm

Table 1.4 Variables used in NLMS algorithm

Variable	Description
N	The current time index
$u(n)$	The vector of buffer input sample at step $n = [u(n), u(n-1), \dots, u(n-M+1)]^T$
$u^*(n)$	Complex conjugate of the vector of buffered input sample at step n
$w(n)$	The vector of filter weight estimates at step n
$y(n)$	The filtered out put at step n
$e(n)$	The error signal at step n
$d(n)$	The desired input signal at step n
M	The filter length
μ	The adaption step size
C	Small constant
A	NLMS adaption constant

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Table 1.5 Normalized Least Mean Square (NLMS) algorithm

S.NO	NLMS ALGORITHM	
1	Initial Conditions:	$0 < \alpha < 2$ and c is small constant. Length of Adaptive Filter= L Input vector: $u[0,0,0...0]^T$ Weight vector $[0,0,0..0]^T$
For each instant of Time, $n=1,2,...$,compute:		
2	Output signal:	$y(n) = w^T \cdot u(n)$;
3	Estimation Error:	$e(n) = d(n)-y(n)$;
4	adaption step size:	$w(n)+[2 \alpha/c+u(n)^T \cdot u(n)]e(n)u(n)$ $w(n+1) = w + 2 \mu \cdot e(n) \cdot u(n)$;

Implementation of the Notch filter

A narrow band, notch filter was designed using simulation in MATLAB software, to cancelled 50 Hz Hum cancellation using notch filter from the ECG signal. The filter was designed to meet the following specifications in table 1.6 given.

Table 1.6 Summary Notch Filter

S. No	Description	Parameter
1	Notch frequency	50Hz.
2	3 dB Width	10 Hz.
3	Attenuation at notch frequency	3 dB.
4	Sampling frequency	600Hz.
5	R	0.947
6	b_0	0.9463

IV. EXPERIMENTAL RESULTS

This paper presents the results of simulation using MATLAB to investigate the performance of various adaptive filter algorithms in non stationary environment with different step sizes. The principle means of comparison is the error cancellation capability of the algorithms which depends on the parameters such as step size, filter length and number of iterations.

Results for LMS algorithm: In this case $\mu = 0.05$, Filter length is 15 and 25 and 1400 iterations Fig. 1.5, 1.6 given below.

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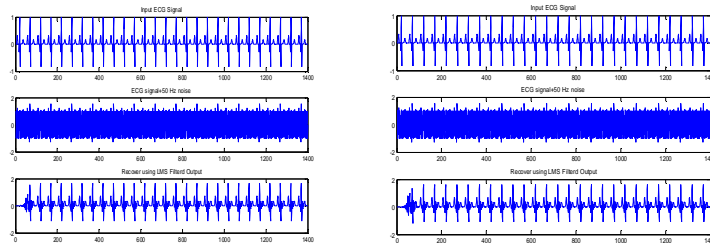


Figure: 1.5 LMS algorithm where $\mu=0.05$, Filter length = 15 Fig.1.6 LMS algorithm where $\mu=0.05$, Filter length = 25

LMS algorithm: In this case $\mu = 0.09$, Filter length is 15, 20 and 35 and 1400 iterations Fig. 1.7, 1.8, 1.9 given below .

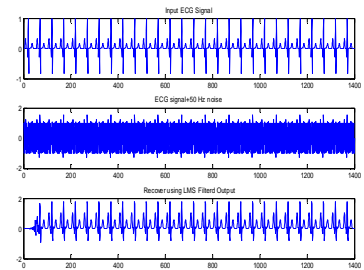
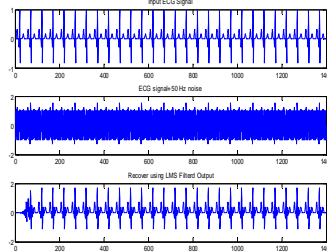
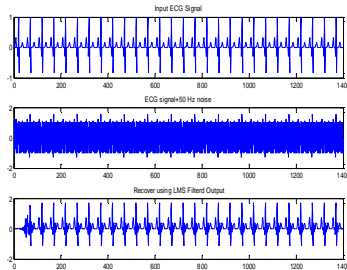


Fig.1.7 LMS algorithm where $\mu=0.09$, Filter length = 15 Fig.1.8 LMS algorithm where $\mu=0.09$, Filter length = 20
Fig.1.9 LMS algorithm where $\mu=0.09$, Filter length = 35

Results for NLMS algorithm:

In this case $c = 0.019$, $\alpha = 0.102$, Filter length is 15 and 30 at 1400 iterations Fig.1.10, 1.11, 1.12 shown in given below.

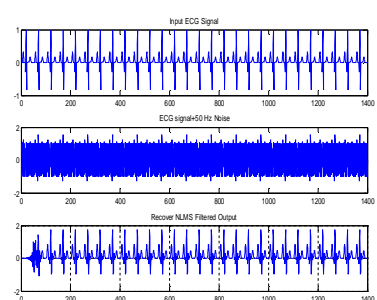
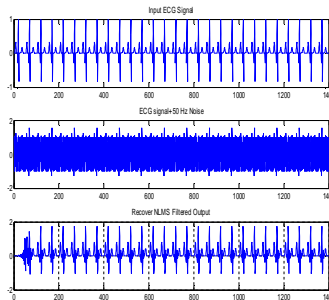
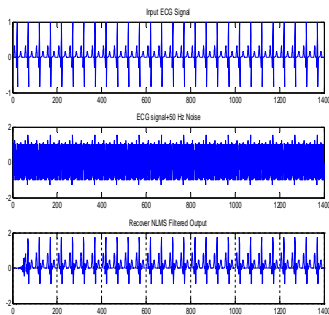


Fig.1.10 NLMS algorithm for Filter length = 15 Fig.1.11 NLMS algorithm for Filter length = 25
Fig.1.12 NLMS algorithm where $c=0.019$, $\alpha=0.102$, Filter length = 30

NLMS algorithm: In this case $c = 0.019$, $\alpha=0.102$, Filter length is 40 and 1400 iterations in fig 1.13

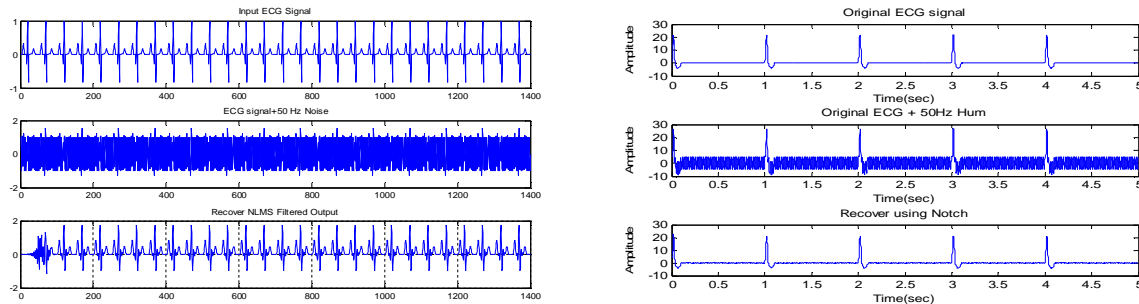


Fig.1.13 NLMS algorithm where $c=0.019$, $\alpha=0.102$, Filter length. = 40 **Fig.1.14 Remove noise with a notch filter at 50 Hz.**

Results of Notch Filter

We can remove the noise in the ECG signal using notch filter at 50Hz. Fig 1.14 shows the results.

V. CONCLUSION

The objective was to optimize different adaptive filter algorithms so that we can reduce interference. In this paper, two Adaptive algorithms were analysed and compared with Notch Filter.

The parameter, LMS step size μ play important role in determine the convergence speed and stability. Convergence speed can be controlled by parameter step size μ . I have plot LMS response for step size $\mu=0.05$ and $\mu=0.09$ with different filter length 15, 20, 25, 30. I observed that with increased filter order, accuracy increased and with increased step size convergence rate took place fast.

These results shows that the LMS algorithm has slow convergence but simple to implement and gives good results if step size is chosen correctly and is suitable for stationary environment. The merits of LMS algorithm is less consumption of memory and amount of calculation. The NLMS algorithm changes the step-size according to the energy of input signals hence it is suitable for both stationary as well as non-stationary environment. The implementation of algorithms was successfully achieved for $c = 0.019$, $\alpha = 0.102$, filter length is 15, 20, 30, 40 and 1400 iterations.

The noise cancellation performance of NLMS Filter was observed consistently better when compared with LMS Filter and Notch Filter.

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