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Denoising of ECG Signals Using LMS Adaptive Filters

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ABSTRACT- Electrocardiogram (ECG) signals are the electrical recording of heart activity that reflects the activities and the attributes of the heart and reveals important information. It is a common routine and important cardiac diagnostic tool to know the functional status of heart, but ECG signal can be distorted with noise as, various artifacts corrupt the original ECG signal and reduces it quality. Noise can be any interference which occurs due to motion artifacts or due to power equipment. A typical computer based ECG analysis system includes a signal pre-processing, beats detection and feature extraction stages, followed by classification. Moreover ECG signal processing has become an effective tool for research and clinical practices. The motion artifacts are effectively removed from the ECG signal using various filters. This paper focuses on ECG signal processing using Least Mean Square (LMS) and Notch Filter which has received increasing attention as a signal conditioning and feature extraction technique for biomedical application.

Keywords: *ECG, Signals, LMS Filter, Notch Filter, Noise*

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1. INTRODUCTION

It is difficult to get accurate signals for biomedical recording when patient is diagnosis by medical monitoring equipment such as ECG, EEG 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.

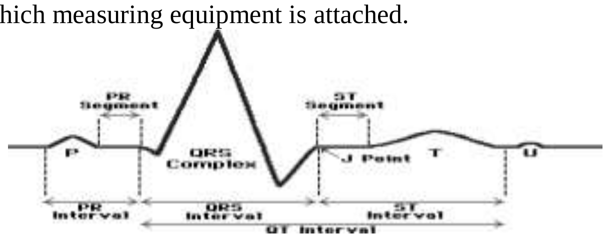

Fig 1.1: ECG Signal

The ECG signal provides following information of the human heart,

- Heart position and its relative chamber size
- Impulse origin and propagation
- Heart rhythm and conduction disturbances
- Drug effects on heart
- Extent and location of myocardial ischemia
- Changes in electrolyte concentrations

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, 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].

Adaptive and Digital 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.

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.

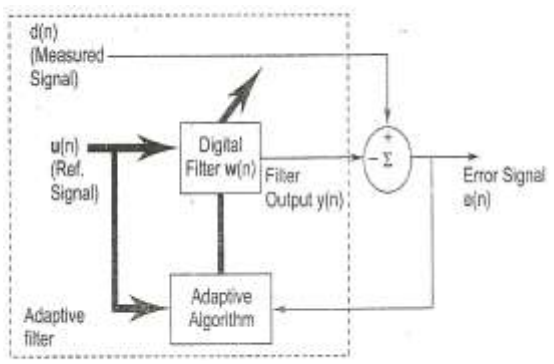

Fig.1.2 Block diagram of an adaptive filter

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

(i) Easier implementation
(ii) Single adjustable parameters μ for controlling the convergence rate but slow convergence

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 analyzing 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.

2. PROPOSED METHOD

In this paper, algorithms are proposed, developed and validated, for removing non-trivial noises in ECGs using the LMS .Firstly, independent component analysis has been studied and found effective to separate out motion induced artifacts in ECG.
Then 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<u<(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) +µ.e (n).u (n)

Table 1.1 Variables use in LMS algorithm

Variable	Description
N	The current time index
u(n)	The vector of buffer input sample at step $n=[u(n).u(n-1)\ldots u(n-M-1)]^T$
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.

Table 1.2 Least Mean Square (LMS) algorithm

S. No.	**LMS algorithm**	
1	Initial Conditions :	0< µ < 1 Length of Adaptive filter=L Input vector: $u[0,0,0\ldots0]^T$ Wight vector: $w[0,0,0\ldots0]^T$
	For each instant of	= 1, 2, 3…, compute:
2	Output signal:	$y(n) = w^T . u(n)$;
3	Estimation Error:	e(n) = d(n)-y(n);
4	Tap-Weight Adaptation:	w (n+1) = w + 2 µ .e(n).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 1.3 given.

Table 1.3Summary Notch Filter

S. No	Description	Parameter
1	Notch frequency	50Hz.

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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

3. SIMULATIONS RESULTS

This paper presents the results of simulation using MATLAB to investigate the performance behaviors 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. All Simulations presented are different parameter such as step size, filter tap, iterations.

3.1 LMS Filter Simulation: In this case μ =0.05, Filter length is 15 and 25 and 1400 iterations Fig. 1.3, 1.4 given below.

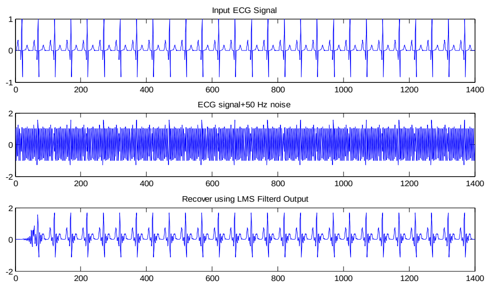

Figure: 1.3 LMS algorithm where μ=0.05, Filter length = 15

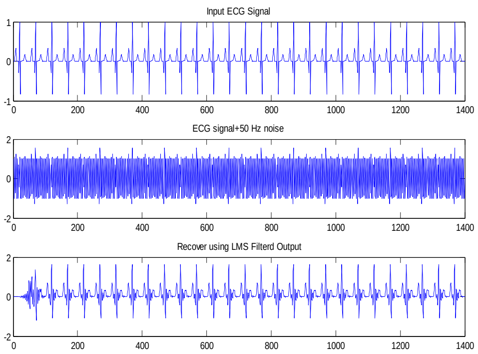

Fig.1.4 LMS algorithm where μ=0.05, Filter length = 25

LMS algorithm: In this case μ =0.09, Filter length is 15, 20 and 35 and 1400 iterations Fig. 1.5, 1.6, 1.7 given below.

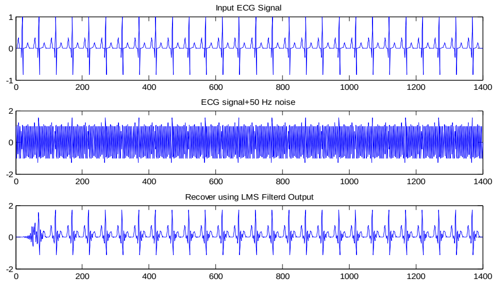

Fig.1.5 LMS algorithm where μ=0.09, Filter length = 15

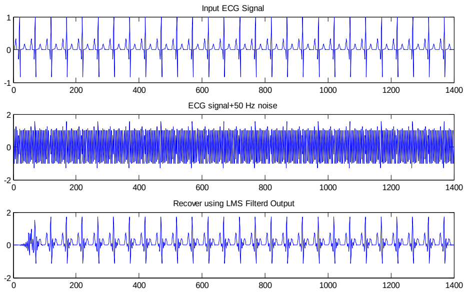

Fig.1.6 LMS algorithm where μ=0.09, Filter length = 20

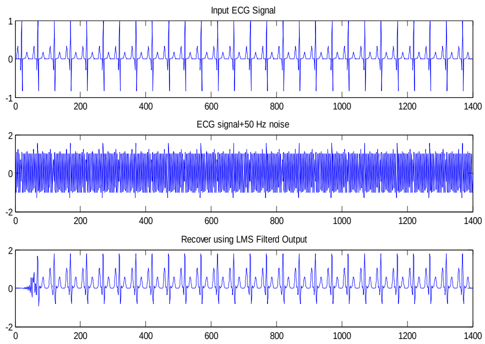

Fig.1.7 LMS algorithm where μ=0.09, Filter length = 35

3.2 Notch Filter Simulation

We can remove the noise in the ECG signal using notch filter at 50Hz.

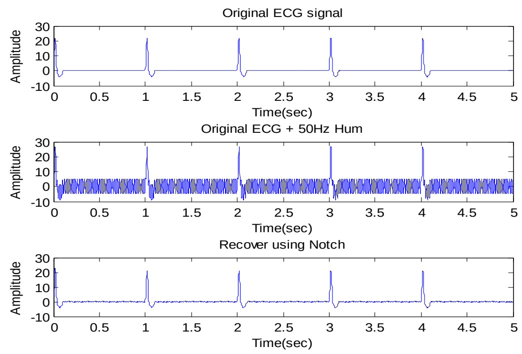

Fig.1.8 Remove the noise is to filter the signal with a notch filter at 50 Hz.

4. CONCLUSION

The objective was to optimize different adaptive filter algorithms so that we can reduce interference. In this paper,

LMS Adaptive algorithm was analyzed 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 µ=0.05 and µ=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.

5. SCOPE

In this work, only the Least-Mean-Squares (LMS) Algorithm has been used. Other adaptive algorithms can be studied and their suitability for application to Adaptive Noise Cancellation can be compared. Other algorithms that can be used include Recursive Least Squares. Moreover, this project does not consider the real time processing of finite-length filters and the causal approximation and also improvement of the thesis can be further implemented with different algorithms such as NLMS and RLS algorithm and wiener filter to achieve the desired results.

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