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# Denoising of ECG Signal with Adaptive Filtering Algorithm

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**Abstract:** The electrocardiogram is the recording of the electrical potential of heart versus time. The analysis of ECG signal has great importance in the recognition of cardiac abnormalities. The electrocardiographic signals are often contaminated by noise from varied sources. Noises that commonly disturb the basic electrocardiogram are power line interference, instrumentation noise, external electromagnetic field interference, noise due to random body movements and respirational movements. These noises can be classified according to their frequency content. It is essential to reduce these disturbances in ECG signal to improve accuracy and reliability. Adaptive filter system is required to overcome this problem. Simulations are done for random noise pattern in matlab.

**Keywords:** Adaptive filtering, ECG, MATLAB, random noises

## 1. Introduction

Electrocardiogram (ECG) is the record of the electrical potentials produced by the heart. The electrical wave is generated by depolarization and repolarization of certain cells due to movement of Na+ and $k^+$ ions in the blood. The ECG signal is typically in the range of 2 mV and requires a recording bandwidth of 0.1 to 120 Hz . Any disorder in heart rate or rhythm or change in the morphological pattern of ECG signal is an indication of cardiac arrhythmia. It is detected and diagnosed by analysis of the recorded ECG waveform. The amplitude and duration of the P-QRS-T-U wave contains useful information about the nature of disease related to heart. In clinical environment during acquisition, the ECG signal encounters various types of artifacts. Electrocardiography has had a profound influence on the practise of medicine. It has been always advantageous in the diagnosis of the cardiac problems. ECG elucidates the cardiac problems which makes it easier in diagnosis. During the last five decades the analysis of the ECG signal evolved from the simple visual inspection to completely automated diagnosis system. ECG detects changes in cardiac muscles like myocardial infarction, conduction defects and arrhythmia .It recognises the variability's of heart activity, so it is very important to get the ECG signal clean and free from noise.Electrocardiogram is the trans-thoracic interpretation of the electrical activity of heart over a period of time, by electrodes attached to the outer surface of the skin and recorded by a device external to the body known as Electrocardiogram. ECG Signals generated from human body are often very weak so as to be easily covered by background noise. The ones of primary interest are power line interference, external electromagnetic field interference, noise due to random body movements and respirational movements, electrode contact noise, electromyography (EMG) noise, and instrumentation noise. These noises degrade the signal quality, frequency resolution and strongly affect the morphology of ECG signal containing important information. It is essential to reduce disturbances in ECG signal and improve the accuracy and reliability for better diagnosis. Adaptive filters are variable filters whose filter coefficients are adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt the changes in the input signal characteristics. Because of their self-adjusting performance and in-built flexibility, adaptive filters have found use in many diverse applications such as telephone echo cancelling, radar signal processing, and noise cancelling and biomedical signal enhancement. In any communication systems noise is the unwanted signal that mixes up with the desired signal; such noise is removed by using different techniques. Different types of digital filters are used to remove signal components from unwanted frequency ranges. It is difficult to apply filters with fixed coefficients to reduce random noises, because human behaviour is not exact known depending on the time. These difficulties can be solved using an adaptive filter, a system with variable coefficients.

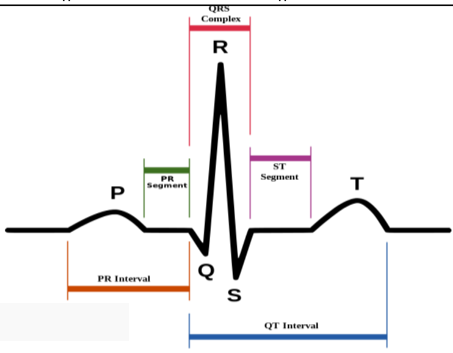


Fig 1.Typical ECG signal

ECG waveform of a normal individual consists of P wave, QRS complex, ST segment, T wave. The labels of Fig. are commonly used in medical ECG terminology.**P wave:** When the electrical impulse is conducted from the SA node towards the AV node and spreads from right to left atrium, the depolarization (contraction) of the atria occurs. The depolarization of atria results the P Wave in the ECG .**QRS complex:** The QRS complex consists of three waves, sequentially known as Q, R and S. The rapid depolarization of both the ventricles results this complex. The muscles of the ventricles have large muscle mass than that of atria, hence its amplitude is much larger than that of P wave.**T wave:** Ventricular repolarisation results the preceding of ST segment and the T wave.

## 2. Literature Survey

During past few years, various contributions have been made in literature regarding noise removal, beat detection and classification of ECG signal. This project investigates the performance of ECG signal using matlab.

Rajvansh Sehamby, Buta Singh[1] published a paper "Noise Cancellation using Adaptive Filtering in ECG Signals: Application to Biotelemetry" in which they proposed a method which helps to reduce the noise interference in the ECG signals and better diagnose results. Presented an implementation of Least Mean Squares (LMS). This paper shows the simplicity of LMS algorithm and ease of implementation, evident from above make this algorithm better in many real time systems to improve the SNR and to reduce the noise of signal. Information of fetal heart rate, derived from the fetal ECG, is valuable in assessing the condition of the baby before or during birth of a baby. Adaptive filter have been used to derive a noise free fetal electrocardiogram signal.

Gaurav Makwana, Lalita Gupta[2] published a paper "De-Noising of Electrocardiogram (ECG) with Adaptive Filter Using MATLAB" in which they told problem associated with biomedical signal like ECG is to extract noise cause by high frequency interference, electromagnetic fields, power line interference and body movement. It is difficult to apply filters with fixed coefficients to reduce random noises. Adaptive filter technique is required to overcome this problem. In this paper adaptive filter is used for noise removal from ECG signal. For this, the original and the desired signals are selected in such a manner that the filter output is the least squared estimate of the original ECG signal. The paper proposed the modifications in the weight update formula for LMS based realizations.

Arpit Sharma, Sandeep Toshniwal, Richa Sharma[5] published a paper " Noise Reduction Technique for ECG Signals Using Adaptive Filters" explained the ECG finds its importance in the detection of cardiac abnormalities. Noise reduction in ECG signal is an important task of biomedical science. ECG signals are very low frequency signals of about 0.5Hz-100Hz and digital filters are very efficient for noise removal of such low frequency signals. In this Paper an adaptive filter for high resolution ECG Signal is presented which estimate the deterministic component of the ECG Signal and remove the noise.

Balwinder Singh, Preeti Singh, Sumit Budhiraja [8] published a paper "Various Approaches to Minimise Noises in ECG Signal: A Survey" discussed about ecg signal is time varying in nature which is most common source used for the diagnosis and analysis of heart diseases present in the patient. Power line interference, baseline wanders and muscle tremors are mostly noticed artifacts or noises. So for accurate delineation of characteristics points of ECG, a good quality of ECG is necessary. It has been found that multi resolution discrete wavelet transform (DWT) is the best choice to remove power line interference while for removal of EMG noise and motion artifact, researchers have preferred discrete Meyer wavelet along with thresholding having characteristics of both hard and soft thresholding. For removal of artifacts and noises, advanced algorithms like empirical mode decomposition (EMD),discrete wavelet domain(DWT) or hybrid approach of both techniques could provide better results.

Akanksha Mittal, Amit Rege[11] published a paper "Design of Digital FIR Filter Implemented with Window Techniques for Reduction of Power line Interference from ECG Signal" in which present paper deals with design and development of digital FIR filters. Heart diseases, which are one of the death reasons, are among the several serious problems in this century and as per the latest survey, 60% of the patients die due to Heart diseases. Electrocardiogram (ECG) signal plays a vital role in the monitoring and diagnosis of the health conditions of human heart In this work a digital FIR filter for reducing 50Hz power line noise in ECG signal is designed and implemented with different windowing methods as of Hamming, Kaiser, and Chebyshev. The work has been done in MATLAB environment. The result obtained for all FIR filters are compared by comparing the waveforms of the original and filtered ECG signals. The filter which gives the best results is the one using Chebyshev Window**.**

Syed Ateequr Rehman, R.Ranjith Kumar[12] published a paper "Performance Comparison of Adaptive Filter Algorithms for ECG Signal Enhancement" in which the goal of the paper is to show the comparison based on signal to noise ratios of all the adaptive filter algorithms used for the analysis of ECG signals with Power line Interference. From the simulation results it is shown that the output SNR values for the algorithms are obtained and compared with each other, with reference to Power-line Interference Noise . Simulation studies shows that the proposed novel algorithms like NLMS and DLMS based adaptive systems present better performances compared to existing realizations LMS, SRLMS and NSRLMS based procedures in terms of signal to noise ratio. In this paper performance comparison of all the adaptive filter algorithms used to remove the Power-line Interference from the ECG signal after its enhancement is presented.

Sibushri .G [13] published a paper "Analysis of ECG Signals for Arrhythmia Using MATLAB" explained filtering of noise in the ECG signals which are very useful in the analysis of the ECG signals. Compare the results of ECG signal filtered by FIR filter with three windows Kaiser, Hamming and Hanning. The average power and signal to noise ratio was carried out to study the effect of noise on ECG signal. Also the adaptive LMS noise cancellation was done Succesfully for 3 different datas from MIT-BIH arrhythmia database.

## 3. System Implementation

**Proposed Work**

In our proposed system ,adaptive filter and hybrid adaptive algorithm (Recursive least squares (RLS) algorithm+ Normalized Least Mean Square(NLMS) adaptive filter algorithm) have been used.

An adaptive filter adjusts or modified its frequency response automatically to improve its performance according to some criterion. Owing to the self- adjusting performance and in-built flexibility, adaptive filters are used in diverse applications. We use adaptive filters where it is must for the filter characteristics to be variable and adapted to changing condition and when there is a spectral overlap between the signal and noise. Adaptive algorithms are used to modify the coefficients of digital filter.

Adaptive filters are used in the situation where the filter coefficients have to be changed simultaneously according to the requirement. Adaptive filters are needed for fast convergence rate and low mean square error. Many algorithms have been proposed and proved that they have better convergence speed and tracking abilities. An adaptive filter may be understood as a self-modifying digital filter that adjusts its coefficients in order to minimize an error function. This error function is also referred to as cost function, is a distance measurement between the reference or desired signal and the output of the adaptive filter. An adaptive algorithm is a procedure of adjusting the parameters of an adaptive filter. The desired signal is represented by d (n).

The error signal is denoted by e (n) which is the difference between the desired response and the output of the filter y (n).

$e$ (n)=d(n)-y(n)

**Adaptive Algorithm**

Adaptive algorithms are used to adjust the coefficients of the digital filter. Such that the error signals is minimized according to some criterion. In adaptive filters, the weight vectors are updated by an adaptive algorithm to minimize the cost function. The algorithms used for noise reduction in ECG is Recursive least squares (RLS) algorithm and Normalized least mean square(NLMS) algorithm.Here we are using hybrid (RLS+NLMS)appraoach.

**Least mean squares( LMS) Algorithm**

In Least Mean Square algorithm, the output is calculated using some initial value of tap-weights.This output is compared with desired signal and error is calculated. The algorithm works in such a manner that it tries to minimize the mean square error.Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error

signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time.
According to this LMS algorithm the updated weight is given by,
$W(n+1)=W(n)+2\ \mu\ e(n).x(n)$
where μ is the step size.

**Recursive least squares (RLS) algorithm**

Recursive Least Square algorithm is based on principle of minimizing the weighted sum of squared error signal related to input signal. RLS gives excellent performance when operating in time varying environments. The enhanced performance is achieved at the cost of increased computational cost and some stability problems.The least mean squares (LMS) that aim to reduce the mean square error. In the derivation of the RLS, the input signals are considered deterministic, while for the LMS and similar algorithm they are considered stochastic.Compared to most of its competitors, the RLS exhibits extremely fast convergence.

**Normalized Least Mean Square(NLMS) adaptive filter algorithm**

Another class of LMS algorithm is NLMS. NLMS is different from LMS in its weight updating rule. In NLMS step size parameter is not constant. The main drawback of the "pure" LMS algorithm is that it is sensitive to the scaling of its input. This makes it very hard to choose a learning rate μ that guarantees stability of the algorithm. The Normalised least mean squares (NLMS) filter is a variant of the LMS algorithm that solves this problem by normalising with the power of the input. One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. This requires an understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In practice this is rarely achievable. Even if we assume the only signal to be input to the adaptive echo cancellation system is speech, there are still many factors such as signal input power and amplitude which will affect its performance.
HYBRID ALGORITHM (RLS+NLMS)
In hybrid approach, we are using two algorithms.i.e RLS and NLMS algorithm.
The filters tap weight updated equations for hybrid algorithm is given by ,
$W(n) = W(n-1) + k(n)\ .\ e^*(n)$
$W(n+1)=W(n)+\ \mu(n).e(n).x(n)$
Where k(n) is gain vector,
e(n ) is estimation error,
μ is the step size.

## 4. Advantages And Application

**Advantages**

1.The RLS gives much better control performance than LMS algorithm. LMS algorithm is three times slower than RLS algorithm.
2. The reason for slow convergence speed of LMS algorithm is that it uses the transient data to minimize the square error while for RLS algorithm use a group of data in same case.
3.As RLS uses more available information under certain restraints, so its convergence speed is much faster than LMS algorithm.
4.NLMS algorithm shows greater stability with unknown signal.
5.NLMS algorithm with good convergence speed, relative computational simplicity and has minimum steady state error.
6.NLMS good for real time applications.
7.NLMS convergence characteristics superior to the LMS.
8.NLMS algorithm performs better for noise reduction in the ECG signal . It results in a more stable ECG signal as compared to LMS algorithm. NLMS has better convergence speed as compared to LMS due to the adaptive step size.

**Applications**

There are many applications in which adaptive filters are used such as room acoustic identification, channel estimation, echo cancellation, equalization in digital communications, blind equalization, adaptive beam forming, biomedical signal processing, and adaptive control systems. Basically there are four fundamental classes of adaptive filter applications as mention below :
1.Linear predictor
The linear prediction estimates the values of a signal at a future time. This model is wide usually in speech processing applications such as speech coding in cellular telephony, speech enhancement, and speech

recognition. In this configuration the desired signal is a forward version of the adaptive filter input signal. When the adaptive algorithm convergences the filter represents a model for the input signal, this model can be used as a prediction model.

**2. Adaptive notch filter**

In certain situations, the primary input is a broadband signal corrupted by undesired narrowband (sinusoidal) interference. The conventional method of eliminating such sinusoidal interference is using a notch filter that is tuned to the frequency of the interference. To design the filter, we need the precise frequency of the interference. The adaptive notch filter has the capability to track the frequency of the interference, and thus is especially useful when the interfering sinusoid drifts in frequency.

**3. Inverse modeling**

The inverse modeling is an application that can be used in the area of channel equalization, for example it is applied in modems to reduce channel distortion resulting from the high speed of data transmission over telephone channels. In order to compensate the channel distortion we need to use an equalizer, which is the inverse of the channel’s transfer function. High-speed data transmission through channels with severe distortion can be achieved in several ways, one way is to design the transmit and receive filters so that the combination of filters and channel results in an acceptable error from the combination of intersymbol interference and noise; and the other way is designing an equalizer in the receiver that
counteracts the channel distortion. The second method is the most commonly used technology for data transmission applications.

**4. Jammer suppression**

Adaptive filtering can be a powerful tool for the rejection of narrowband interference in a direct sequence spread spectrum receiver.

**5. Echo Cancellation**

In telecommunications, echo can severely affect the quality and intelligibility of voice conversation in telephone, teleconference or cabin communication systems. The perceived effect of an echo depends on its amplitude and time delay. In general, echoes with appreciable amplitudes and a delay of more than 1 ms can be noticeable. Echo cancellation is an important aspect of the design of modern telecommunications systems such as conventional wire-line telephones, hands-free phones, cellular mobile (wireless) phones, teleconference systems and in-car cabin communication systems.The traditional solution to this problem prior to the advent of the adaptive filtering solution was to introduce significant loss into the long-distance network so that echoes would decay to an acceptable level before they became perceptible to the callers.

## 5. Conclusion

We have a hybrid technique for  noise reduction in ECG signals. The proposed technique utilizes RLS and NLMS algorithm for noise reduction . The performance of proposed hybrid technique is evaluated using Signal to Noise Ratio (SNR),average power and mean square error(MSE). The conclusion could be drawn that RLS algorithm has faster convergence speed than LMS. LMS algorithm needs external commands for control where as RLS algorithm has higher controlled performance. For optimal output RLS algorithm needs less number of iterations for operation compare to LMS algorithm.  RLS algorithm provides better stability under changing environment and improves further with iterations. NLMS leads to fast convergence and is also stable as compared to LMS.

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