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# A Review on Efficient Implementation of Adaptive Filter for Noise Reduction in ECG Signal

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**Abstract:** ***The electrocardiogram (ECG) is a graphical representation of the cardiac activity and it is widely used for the diagnosis of heart diseases. But different noises get contaminated with ECG signal during its acquisition and transmission, which can cause a great deal of hindrance to manual and automatic analysis of ECG signals and they may be interpreted as the abnormal heart conditions. Hence for the proper diagnosis of the heart the ECG signals must be free of noises. 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. The goal for ECG signal enhancement is to separate the valid signal components from the undesired artifact, so as to present an ECG that facilitates easy and accurate interpretation. It is difficult to apply filters with fixed coefficients to reduce random noises. Adaptive filter technique is required to overcome this problem. Comparisons are made for original signal to noisy. Simulations are done for random noise pattern in matlab.***

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

## 1. Introduction

Heart related problems are increasing day by day and hence the Electrocardiogram signals are very important in diagnosis of heart related problems. There are various noise affects or get added in these Electrocardiogram signals at the time of taking signals from the body of the patient and change the original signal. There is a need of removal of this different noise from the original signal. The electrocardiogram (ECG) is a graphical representation of the electrical activity of heart and it is widely used in medical area for the diagnosis of heart diseases by the cardiologist. While taking signals from the body of the patient different signals are added to the ECG. These added signals are nothing but the noise signals, and due to these signals original signal is not given to the cardiologist. There are different types of noise added to the ECG like Power line interference, Muscle contraction noise, Electrode contact noise, Patient movement, Baseline wandering and ECG amplitude due to respiration. Instrumentation noise generated by electronic devices used in signal processing and other less significant noise resource. 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. Electrocardiogram (ECG) is one of the most important parameters for heart activity monitoring. Usually two types of digital filters are used for data processing: frequency-selective filters with fixed coefficients and filters with variable coefficients. Various adaptive and non-adaptive methods are there for ECG signals enhancement. So, a filter with fixed coefficients can't deal with this kind of noise signals and valuable information may be lost. These difficulties can be solved using an adaptive filter, a system with variable coefficients. There are different noises which are added to the Electrocardiogram signals, these are Power line interference: It consists of 50 Hz pickup and harmonics, which can be combination of sinusoid. Typical parameters: Frequency content-50 Hz (fundamental) with harmonics; Amplitude-up to 50 percent of peak-to-peak ECG amplitude. Muscle contraction noise: The baseline electromyogram is usually in the microvolt range and therefore is usually insignificant. Parameters: Standard deviation-10% of peak-to-peak ECG amplitude; Duration-50ms; Frequency-10000 Hz. Electrode contact noise: This kind of noise caused by loss of contact between the electrode and skin, which effect the measurement of signal. Parameters: Duration-1s; frequency-50 HZ time constant-about 1s.Patient movement: Patient movements are transient (but not step) baseline changes caused by variations in the electrode skin impedance with electrode motion. Parameters: Duration-100 to 500ms; amplitude-500% peak-to-peak. Electrosurgical noise: It completely destroys the ECG signal. It can be represented by large amplitude. Parameters: Amplitude-200 % peak-to-peak; Frequency 100 kHz to 1 MHz; Duration-1 to 10s.

## 2. Proposed Work and Objectives

In adaptive filters, the weight vectors are updated by an adaptive algorithm to minimize the cost function. The algorithm used for noise reduction in ECG is least mean square (LMS). 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. Various system performance parameters will be analyzed. All the simulation results using the algorithm will be studied.

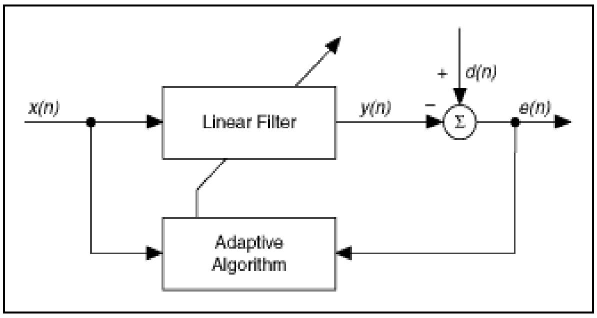


**Figure:** Adaptive filter

The fig. above is given the general adaptive filtering display digital filter carries on filtering on filter coefficient included in the vector w (n), in order to get the error signal e(n) to be the smallest. The vector representation of x (n) is given. This input signal is corrupted with noises. In other words, it is the sum of desired signal d (n) and noise v (n). The input signal vector is x (n) which is given by

$$x(n)=[x(n),x(n-1),x(n-2),\ldots\ldots x(n-N-1)]T \quad (1)$$

$$x\ (n)=d(n)+v(n) \quad (2)$$

The adaptive filter has a Finite Impulse Response (FIR) structure. For such structures, the impulse response is equal to the filter coefficients. The coefficients for a filter of order N are defined as

$$W(n)=[wn(0),wn(1),\ldots\ldots wn(N-1)]T \quad (3)$$

The output of the adaptive filter is y(n) which is given by

$$y(n)=W(n)Tx(n) \quad (4)$$

The error signal or cost function is the difference between the desired and the estimated signal

$$e(n)=d(n)-y(n) \quad (5)$$

Moreover, the variable filter updates the filter coefficients at every time instant

$$W(n+1)=W(n)+\Delta W(n) \quad (6)$$

Where $\Delta W(n)$ is a correction factor for the filter coefficients. The adaptive algorithm generates this correction factor based on the input and error signals.

LMS filter working is based on finding the filter coefficients that leads to minimizing the mean square of the error which is the difference between the desired signal and error signal. In order for LMS filter to approach the optimum filter weights, the algorithm starts by assuming small weights (zero) and at each step, it finds the gradient of the mean square error and then updates the weights. If the mean square error is positive, error is increases positively and if same error is used, filter weights needs to be reduced accordingly. And vice versa, if gradient is negative, filter weights need to be increased. Weight update equation is as below,

$$w(i) = w(i-1) + \mu\, e(i)\, x(i) \ldots\ldots\ldots\ldots\ldots \quad (7)$$

where e(i), x(i), μ and w(i) are the error, the input, step size parameter and the weight function respectively.

If the step size parameter μ is significantly small , LMS algorithm converges in the mean square provided that *u* satisfies the condition given below,

$0 < u < (1 / \lambda max)$ ……..

Where, $\lambda max$ is the largest eigen value of the correlation matrix.

LMS algorithm

If the variable filter has a tapped delay line FIR structure, then the LMS update algorithm is especially simple. Typically, after each sample, the coefficients of the FIR filter are adjusted as follows:

For l = 0……L

$$W_{l,k+1} = w_{lk} + 2\, \mu\, \varepsilon_k\, x_{k-l} \quad (8)$$

μ is called the *convergence factor*.

The LMS algorithm does not require that the X values have any particular relationship; therefor it can be used to adapt a linear combiner as well as an FIR filter. In this case the update formula is written as:

$$W_{l,k+1} = w_{lk} + 2\, \mu\, \varepsilon_k\, x_k \quad (9)$$

Objectives

To study about different filtering technique used in digital signal processing. To develop an algorithm which will filter the ECG signal using adaptive filter technique. To simulate and verify the system response. To analyze system performance and parameter of a system.

## 3. Conclusion

Transfer function of FIR filter will have only zeros, need more memory and/or calculation to achieve a given filter response characteristic. Also, certain responses are not practical to implement with FIR filters. FIR filter need higher order to achieve performance. Delay is more. It has lower sensitivity. FIR filters are less efficient .These are disadvantages of FIR filters. So, to overcome these drawbacks adaptive filter is used. Includes the optimization of algorithms for all kinds of noises and to use the optimized one in the implementation of DSP Microcontroller that estimates the respiratory signal. Implementation of wavelet based denoising for the removal of base line wander. Use of other adaptive methods like FT-RLS, QRD-RLS algorithms for ECG Denoising. Application of blind adaptive filtering for ECG enhancement. Real time application of implemented algorithms. Adaptive filter have been used to derive a noise free electrocardiogram signal.

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