# Cancellation of Power Line Interference in ECG using Adaptive LMS Algorithm

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**Abstract:** The electrocardiogram (ECG) has the considerable diagnostic significance, and applications of ECG monitoring are diverse and in wide use. Noises that commonly disturb the basic electrocardiogram are power line interference (PLI), instrumentation noise, external electromagnetic field interference, noise due to random body movements and respiration movements. It is essential to reduce these disturbances in ECG signal to improve accuracy and reliability. it is difficult to apply filters with fixed filter co-efficient to reduce these noises. Adaptive filter technique is required to overcome this problem as the filter coefficients can be varied to track the dynamic variations of the signals. Adaptive filter based on the least mean square (LMS) algorithm is applied to noisy ECG to reduce 50 Hz power line noise and motion artifact noise. The removal of power line interference from most sensitive medical monitoring equipments can also be removed by implementing various useful techniques. The power line interference (50/60 Hz) is the main source of noise in most of bio-electric signals. the removal of power line interference and other single frequency tones from ECG signal using the advanced adaptive filtering technique with LMS (least mean square) algorithm. it is based on digital signal processing (DSP) techniques with MATLAB package, with the emphases on design of adaptive LMS algorithm.

**Keywords:** Electrocardiogram (ECG), Adaptive Noise Canceller (ANC), Adaptive filter, least mean square (LMS) algorithm, MATLAB/SIMULINK.

## 1. Introduction

The medical monitoring devices are more sensitive for the biomedical signal recording and need more accurate results for every diagnosis. It is complicated to get accurate result for every biomedical signal's recording while patient is diagnosis by medical monitoring equipments such as ECG, EEG and EMG. The low frequency signal is destroyed by power line interference of 50/60 Hz noise, this noise is also source of interference for biomedical signal recording. The signal can also be corrupted by electromagnetic field (EMF) by the machinery which is placed nearby. The frequency of power line interference 50/60 Hz is nearly equal to the frequency of ECG, so this 50/60 Hz noise can destroyed the output of ECG signal while the patient is diagnosis at hospital or somewhere else. The recording of ECG signal cannot give accurate result due to the power supply or by environment.

There are many reasons for the corruption of ECG signal while recording in hospital or some other place due to the external interference which comes from power transformer or high voltage electric power lines and internal interference comes from the internal power supplies. Other problem occurs by harmonics and high frequency noises. In a noise signal, the signal component holds harmonics with different amplitude and frequency. The harmonics frequency is integral multiple of fundamental frequency such as 50Hz. Due to these interferences the quality of ECG signal cannot be ideal so it is needed to improve the quality of required output of ECG signal.

## 2. Adaptive Filter

Concept of adaptive filter

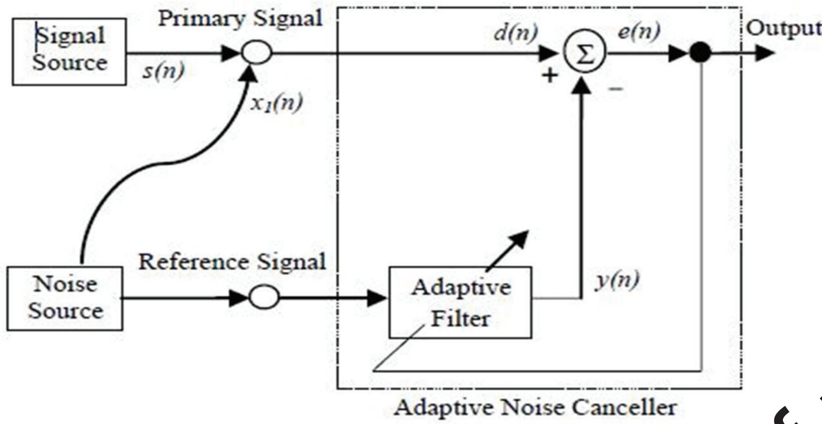


Figure1. Adaptive Noise Canceller

S(n)- source signal d(n)-primary signal $x_1(n)$-noise signal x(n)-noise reference input

y(n)-output of adaptive filter e(n)-system output signal

Fig. 1 shows the adaptive power line setup. In this application, the corrupted signal passes through a filter that tends to suppress the noise while leaving the signal unchanged. This process is an adaptive process, which means it cannot require a priori knowledge of signal or noise characteristics. Adaptive noise cancellation algorithms utilize two signal it can vary in (sensor). One signal is used to measure the ECG + noise signal while the other is used to measure the noise signal alone.

The technique adaptively adjusts a set of filter coefficients so as to remove the noise from the noisy signal. This technique, however, requires that the noise component in the corrupted signal and the noise in the reference channel have high coherence. Unfortunately this is a limiting factor, as the microphones need to be separated in order to prevent the speech being included in the noise reference and thus being removed. With large separations the coherence of the noise is limited and this limits the effectiveness of this technique. In summary, to realize the adaptive noise cancellation, we use two inputs and an adaptive filter. One input is the signal corrupted by noise (Primary Input, which can be expressed as $s(n)+ x_1(n)$). The other input contains noise related in some way to that in the main input but does not contain anything related to the signal (Noise Reference Input, expressed as x(n)). The noise reference input pass through the adaptive filter and output y(n) is produced as close a replica as possible of $x_1(n)$. The filter readjusts itself continuously to minimize the error between $x_1$ (n) and y (n) during this process. Then the output y(n) is subtracted from the primary input to produce the system output $e(n)=s(n)+x_1(n)-y(n)$.

In the system shown in Fig. 1 the reference input is processed by an adaptive filter. An adaptive filter differs from a fixed filter in that it automatically adjusts its own impulse response. Thus with the proper algorithm, the filter can operate under changing conditions and can readjust itself continuously to minimize the error signal. The error signal used in an adaptive process depends on the nature of the application. In noise cancelling systems the practical objective is to produce a system output $e(n)=s(n)+ x_1(n) -y(n)$ that is a best fit in the least squares sense to the signal s. This objective is accomplished by feeding the system output back to the adaptive filter and adjusting the filter through an LMS adaptive algorithm to minimize total system output power. In an adaptive noise cancelling system, in other words, the system output serves as the error signal for the adaptive process. It might seem that some prior knowledge of the signal s or of the noises x1 and x would be necessary before the filter could be designed, or before it could adapt, to produce the noise cancelling s, x1 and x signal y. Assume that s, x1 , x and y are statistically stationary and have zero means. Assume that s is uncorrelated with x1 and x , and suppose that x is correlated with x1 .The output e is

$$e=s+x_1-y \quad (1)$$

Squaring, one obtains $e^2=s^2+(x_1-y)^2+2s(x_1-y)$ (2)

Taking expectations of both sides of (2), and realizing that s is uncorrelated with x1 and with y, yields

$E[e^2]=E[s2]+E[(x_1-y)^2]+2E[s(x_1-y)]=E[s^2]+E[(x_1-y)2]$ (3)

The signal power $E[s^2]$ will be unaffected as the filter is adjusted to minimize E[e2 ]. Accordingly, the minimum output power is mine $[e^2]=E[s^2]$ + mine$[(x_1-y)^2]$ (4)

When the filter is adjusted so that $E[e^2]$ is minimized, $E[(x_1-y)^2]$ is, therefore, also minimized. The filter output y is then a best least squares estimate of the primary noise no. Moreover, when $E[(x_1-y)^2]$ is minimized[(e-s)2] is also minimized, since, from (1), $(e-s)=(x_1-y)$ (5)

Adjusting or adapting the filter to minimize the total output power is thus tantamount to causing the output e to be a best least squares estimate of x1 the signal s for the given structure and adjustability of the adaptive filter and for the given reference input. The output z will contain the signal s plus noise. From (l),the output noise is given by(x1-y). Since minimizing E[e2] minimizes E[( x1-y)2] minimizing the total output power minimizes the output noise power. Since the signal in the output remains constant, minimizing the total output power maximizes the output signal-to-noise ratio.



## 3. LMS Algorithm

The LMS algorithm is a widely used algorithm for adaptive filtering. The algorithm is described by the following equations:

$$y(n)=\sum_{I=0}^{M-1} wi(n)*x(n-i) \quad (1)$$

$$E(n)=d(n)-y(n) \quad (2)$$

$$wi(n+1)=wi(n)+2ue(n)x(n-i) \quad (3)$$

In these equations, the tap inputs x(n),x(n-1),......,x(n-M+1) form the elements of the reference signal x(n), where M-1 is the number of delay elements. d(n) denotes the primary input signal, e(n) denotes the error signal and constitutes the overall system output. wi(n) denotes the tap weight at the nth iteration. In equation (3), the tap weights update in accordance with the estimation error. And the scaling factor u is the step-size parameter u controls the stability and convergence speed of the LMS algorithm. The LMS algorithm is convergent in the mean square if and only if u satisfies the condition: $0 < u < 2$ / tap-input power

$$\text{where tap-input power} = \sum_{K=0}^{M-1} E[|u(n-k)2|]. \quad (4)$$

The condition of the satisfaction can be checked and LMS algorithm's condition must be satisfied if the step size parameter satisfies the condition. The autocorrelation matrix Rx is necessary for the convergence. The condition which is important for the convergence criterion and the convergence factor of LMS algorithm must be chosen in the range is $0< \mu < 1 / \lambda \max$

Where $\lambda$ max is the largest eigen value of the correlation matrix Rx. The speed of the LMS algorithm's convergence is dependent on Eigen value.

## 4. Simulation Results

The ECG signal has been taken and LMS adaptive filter algorithm has been developed. The ECG signal of 50 Hz is displayed in MATLAB environment as ECG Signal and then the noise of 50 Hz is generated and then mixed with the ECG Signal, which is displayed as mixed signal. The adaptive filter is implemented by using LMS algorithm, FIR filter has been designed. The ECG Signal, 50 Hz Noise signal, mixed signal, Error signal and Adaptive LMS filtered output signal have been displayed. The output is nearly same as the ECG inputted signal.

The first input signal to the adaptive filter is white noise. This demo uses the adaptive filter to remove the noise from the signal output. When you run this demo, you hear both noise and ECG signal. Over time, the adaptive filter in the model filters out the noise so you only hear the ECG signal (Original signal).The two signals were added and subsequently fed into the simulation of LMS adaptive filter. The order of the filter was set to M = 32. The parameter μ is varied. Various outputs are obtained for various step size i.e.μ = 0.005, 0.009 system reaches steady state faster when the step size is larger. Fig.2. Original signal, Noisy signal and filter signal for LMS step size i.e.μ= 0.005.

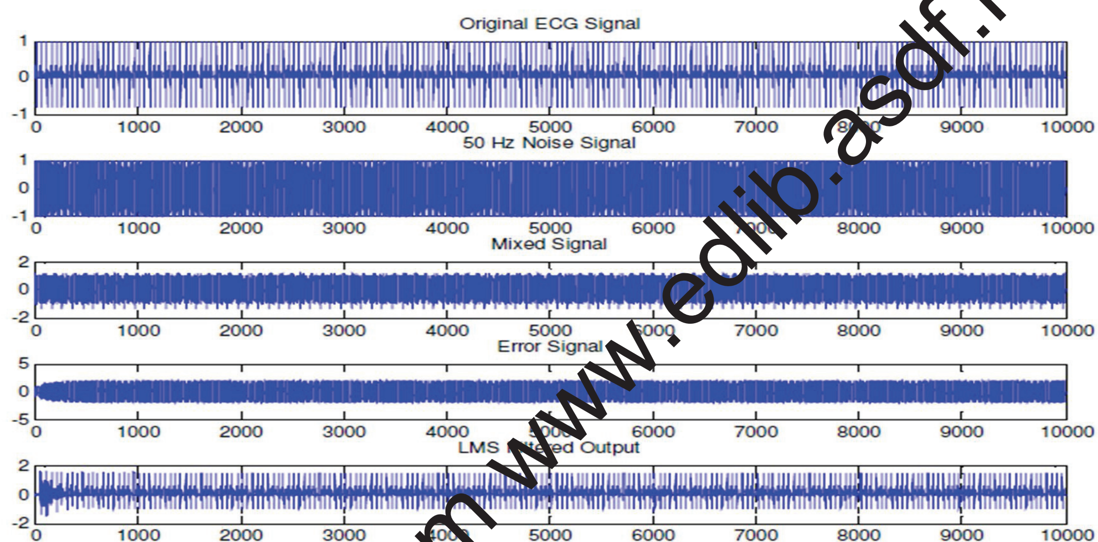


Figure2. Simulation ECG Signal

## Conclusion

It has been proposed a solution for the power line interference its respective harmonics and noise interferences from original ECG signal. The results have been obtained which were required in purpose statement of the report. The value of step size μ play an important role in determining the convergence speed, stability and residual error after convergence. The convergence rate was controlled by LMS step size μ. The ECG signal graphs described in the simulation results verify the adaptation of the LMS adaptive algorithm by changing various parameters like step size, convergence value (μ) and filter taps have various effects on the output graphs. By increasing the filter order it shows a convergence rate but makes the results more precise and by decreasing the step size value it creates the slower convergence but improves the stability and accuracy. The recovered signal closely resembles to the original simulated signal minus the noise. It can be seen that the implementation of the algorithm functions as correctly and efficiently. By comparing the graphs of the input signal of ECG and output signal, it is noticed that the simulation program performs satisfactorily and that noise cancellation from original ECG signal is acquired. Furthermore the general notch rejection filters method also performs the correct operation while filtering the noise from original ECG signal. This technique for the investigation, implemented and analysis of removal of harmonics and high frequency noise from original ECG signal performed satisfactory. It is concluded that the low frequency noise (hum) and high frequency noise can be removed from original ECG signal by the implementation of general notch rejection filters method and the desired result can be achieved accurately.

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