Comparative Performance Analysis of LMS and Notch Filter to Remove the Noise from ECG Signal

Salma Masuda Lisa, Tushar Kanti Roy, Feroz Ahmmed Bhuyan, Dr. Md. Zahurul Islam Sarkar

Abstract— The electrocardiogram (ECG) is an important diagnosis event for emergency cardiac patient. But the problem is, the ECG signal is easily corrupted by the different noise signals such as, power line interference (PLI), the external electromagnetic fields, random body movements, etc. It is observed that the patient does not get proper treatment due to the deficiency of proper cardiac detection. Thus, the aim of this paper is to eliminate noise from the ECG signal to recover the original one. Here two methods are used for de-noising the noisy ECG signal such as, least mean square (LMS) algorithm and Notch filter in terms of mean square error (MSE). Finally, the simulation is performed and the performances of these two methods are compared based on the MSE.

Index Terms—Adaptive noise canceller, ECG, LMS algorithm, mean square error, Notch filtering.

I. INTRODUCTION

The ECG is a diagnostic tool that measures and records the electrical activity of the heart in exquisite detail [1]. Interpretation of these details allows diagnosis of a wide range of heart conditions. These conditions can vary from minor to life threatening. Unfortunately, the artificial large amplitude signals of similar frequency often reach to the skin surface and mix with the ECG signals [2] which arise from several internal and external sources. Thus, the ECG may be corrupted by various kinds of noise such as, power line interference, electrode contact noise, motion artifacts, muscle contraction, baseline drift, instrumentation noise generated by electronic devices and electrosurgical noise [3]. From various artifacts contaminate ECG recording, the most common are the power line interference and baseline drift. Between these the power line interference is easily recognizable since the interfering voltage in the ECG signal has frequency 50 Hz. The extraction of high-resolution ECG signal from recordings contaminated with background noise is an important issue to investigate the proper diagnostic of a patient [4]. Recently,

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Salma Masuda Lisa, Department of Electronics & Telecommunication Engineering, Rajshahi University of Engineering & Technology, Rajshahi, Bangladesh, +8801534817581.

Tushar Kanti Roy, Department of Electronics & Telecommunication Engineering, Rajshahi University of Engineering & Technology, Rajshahi, Bangladesh.

Feroz Ahmmed Bhuyan, Department of Electrical and Electronic Engineering, Rajshahi University of Engineering & Technology, Rajshahi, Bangladesh, +8801728809437.

Dr. Md. Zahurul Islam Sarkar, Department of Electrical and Electronic Engineering, Rajshahi University of Engineering & Technology, Rajshahi, Bangladesh, +8801789538712.

adaptive and conventional filters have been developed as one of the most common and effective tools for processing and analysis of biomedical signals such as ECG. The goal of the adaptive and conventional filters is to enhancement the ECG signal which is done by separating the valid signal components from the undesired artifacts to present an accurate interpretation of ECG signal [5]. 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 biomedical signal noises, because human behavior is not exactly known depending on the time. To overcome this problem an appropriate solution is to use an adaptive filter technique. There are several approaches exist in the literature to enhance the ECG signal using adaptive filters, which permit to detect time varying potentials and to track the dynamic variations of the signals. PLI is a significant source of noise during bio-potential measurements [6] which degrades the signal quality of ECG signal and overwhelms tiny features that may be critical for proper clinical monitoring and diagnosis. Throughout this paper, the PLI is added with the input ECG signal and then PLI suppression is done using LMS algorithm and Notch filter. In general, these methods can be categorized in two non adaptive and adaptive filtering. The non adaptive filtering approaches mainly include IIR filter, FIR filter and notch filter [7]. Adaptive filtering techniques are the popular approaches for the processing and analysis of the ECG signals. Adaptive filters permit to detect time varying potentials and to track the dynamic variations of the signals.

The rest of the paper is organized as follows. Section II briefly introduces the theory of adaptive filter. The performances of the LMS algorithm and Notch filter are analyzed in Section III. Finally, the paper is concluded in Section IV.

II. THEORY OF ADAPTIVE FILTER

The generalized adaptive filtering diagram is shown in fig. 1. The error mixed signal is removed by using the adaptive filtering technique. An adaptive filter has the property of self-modifying its frequency response to change the behavior in time, allowing the filter to adapt the response to the input signal characteristics change. Due to this property of an adaptive filter, the overall performance and construction are flexible. Beside this the adaptive filters have been employed in different applications such as, telephonic echo cancellation, radar signal processing, navigation systems, communications channel equalization and biomedical signals processing [8]. The most common adaptive filters, which are used during the adaption process, are the finite impulse response (FIR) types. These are preferable because they are stable, and no special adjustments are needed for their implementation. The adaptive filter has two inputs the primary input d(n), which represents the desired signal corrupted with undesired noise, and the reference signal x(n), which is the undesired noise to be filtered out of the system. The goal of adaptive filtering systems is to reduce the noise portion to obtain the uncorrupted desired signal. In order to achieve this task, a reference of the noise signal is needed. That reference is fed to the system, which is called a reference signal x(n). However, the reference signal is typically not the same signal as the noise portion of the primary signal: it can vary in amplitude. phase or time delay. Therefore, the reference signal cannot be simply subtracted from the primary signal to obtain the desired portion at the output. . Therefore, the basic idea for the adaptive filter is to predict the noise signal in the primary signal, and then subtract that noise from it. The prediction is done based on the filtering reference signal x(n), which contains a solid reference of the noise present in the primary signal. The resulting signal is called error signal e(n) which presents in the output of the system.



Fig. 1: General adaptive filter configuration

Fig. 2 shows the adaptive noise cancellation setup. In this application, the corrupted signal passes through a filter that tends to suppress the noise while leaving the original signal unchanged. This process is an adaptive process, which means that it cannot require a priori knowledge of signal or noise characteristics [9]. The technique adaptively adjusts a set of filter coefficients so as to remove the noise from the noisy signal.



The noise reference input pass through the adaptive filter and output y(n) is produced as close a replica as possible by $x_1(n)$. The filters readjust their impulse response 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 which can be written as follows

$$e(n) = s(n) + x_1(n) - y(n)$$
(1)

This is the de-noised signal. 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 by readjust itself continuously to minimize the error signal. 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 [10].

A. LMS Algorithm

The LMS algorithm is a widely used algorithm for adaptive filtering. The algorithm is described by the following equations [11]

$$w(n+1) = w(n) + 2\mu e(n)x(n)$$
 (2)

Here x(n) is the input vector of time delayed input values,

$$x(n) = [x(n)x(n-1)x(n-2)...x(n-N+1)]^{T}$$
(3)
The vector $w(n) = [w_{0}(n)w_{1}(n)w_{2}(n)...w_{N-1}(n)]^{T}$

represents the coefficients of the adaptive tap weight vector at a time n The parameter μ is known as the step size parameter which is a small positive constant. This step size parameter controls the influence of the updating factor. Selection of a suitable value for μ is imperative to the performance of the LMS algorithm. If the value of μ is too small then the convergence time of the adaptive filter will be too long to get an optimal solution but if μ is too large the adaptive filter becomes unstable and its output diverges [12].

B. Notch Filter

Notch filters are designed to reject the signal content at a specific frequency by sharply attenuating the gain at that frequency [13]. Its aim is to remove one or a few frequencies from a broader spectrum. To design a digital notch filter, there are many methods exist in the literature. The major things to design a digital notch filter are given below

- (1) Analog filter transformation.
- (2) All pass filter implementation.
- (3) Pole-zero placement technique.

The multiple notch filters can be used for the removal of multiple narrowband or multiple frequency interferences such as, the removal of single –tone interference and its harmonics. Notch filter cannot achieve a desired sharpness of response with many fewer coefficients, the notch filter can implement unusual characteristics such as, being an all pass. Secondly, having less number of side lobes in the stop band and implementation required few parameters, less memory requirements and low computational complexity. The transfer function of the IIR single notch filter is based on that of the all-pass filter, except that the gain of notch frequency is exactly zero. The following subsections will explain each of the design techniques and their revolution. To create a null in the frequency response of a filter at a frequency ω_0 a pair of

complex conjugate zeros on the unit circle at an angle ω_0 is

introduced [14] i.e.,
$$z_{1,2} = e^{\pm j\omega_0}$$
 (4)
Thus the transfer function for a Notch filter is simply

$$H(z) = b_0 (1 - e^{j\omega_0} z^{-1}) (1 - e^{-j\omega_0} z^{-1})$$

= $b_0 (1 - 2\cos\omega_0 z^{-1} + z^{-2})$ (5)

C. Mean Square Error (MSE)

The mean squared error (MSE) of an estimator measures the average of the squares of the "errors", that is, the difference between the estimator and what is estimated. MSE is a risk function, corresponding to the expected value of the squared error loss or quadratic loss. The difference occurs because of randomness or the estimator doesn't account for information that could produce a more accurate estimate. It is estimated between the de-noised ECG signal and the original ECG signal given by the equation [15]

$$MSE = \frac{1}{N} \sum_{n=1}^{N} (x(n) - \overline{x(l)})^2$$

where x(n) is the original ECG signal, $\overline{x(l)}$ is de-noised ECG signal and N is the length of ECG signal [15].

III. SIMULATION RESULTS

In this section, the simulation results are presented to analyze the performances of LMS and Notch algorithms under the PLI with ECG signal. Fig. 3 shows the original ECG signal. The 50 Hz noise signal is shown in Fig. 4 which is added with the original ECG signal. The resultant mixed signal is different from original ECG signal which is shown in Fig. 5. After applying the LMS filtering, the filtered output is shown in Fig. 6.





This paper also uses the Notch filter to remove the noise signal. The output of the Notch filtering is shown in Fig. 7. From Figs. 6 and 7, it is clear that they are able to remove the noise from the noisy ECG signal. The convergence speeds of these two filtering techniques are compared based on the MSE. Fig. 8 shows the MSE comparison result of these two methods.



IV. CONCLUSION

In this work, the LMS and Notch filtering techniques are used to remove noise signal from the ECG signal. The performances of the LMS and Notch filtering are analyzed in terms of MSE. From the MSE analysis, it is found that the adaptive LMS algorithm gives the faster noise cancellation than the Notch filter. However, after a certain time the MSE for the Notch filtering is much less than the LMS filtering technique. Future work will deal with the application of the designed approach for large-scale applications.

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