Removing The Power Line Interference from Ecg Signal using Adaptive Filters

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Abstract
Adaptive filters are primary methods to remove the power line interference from the ECG signal. The frequency range of ECG signal is generally 0.05 Hz to 100 Hz, and that of the power line interference is 50 Hz which lies in the ECG signal band. So, it has become very crucial to remove the power line interference from the ECG signal. In this paper Kalman based least mean square (KLMS) filter has been proposed. The Kalman based Least mean square filter essentially minimize the mean square error and remove the 50Hz power line interferences. The experimental results shows that the Kalman based LMS filter is more effective compare to other filter techniques.

Index
power line interference; ECG signal; Kalman LMS ; SNR

I. INTRODUCTION

There are various biomedical signals present in the human body, by examining these biomedical signal one can check the health condition whether that person is clinically fit or not. Electrocardiogram is one of them. ECG signal is electric representation of the activity of human's heart. Various cardiac diseases can be recognized with the help of ECG signal. While recording process of ECG signal, several types of noises may encounter in it. The common type of noises are power line interference (PLI), electrode motion noise (EM), muscle artifacts, baseline wander etc. It is essential to remove or minimize these interferences prior to further diagnosis for any medical application. The QRS segment is very important and it is predominantly used for clinical observation. So if the noise changes the amplitude or time duration of the segment then recognizing the true condition of patient is very difficult task. Therefore the primary concern is to preprocess the ECG signal. The objective is to separate the valid signal component from the undesired noises so that the accurate interpretation of ECG could be done.

With the latest advancements in electronics, several techniques are used for removal of unwanted entities from signals especially that are implied in the most sophisticated applications. 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 thesis report presents the removal of power line interference from ECG signal using the different adaptive techniques. Different types of digital filters (FIR and IIR) have been used to solve the problem [I]-[4]. However, it is difficult to apply these filters with fixed coefficients to reduce the power line interference, because the ECG signal is known as a non-stationary signal. Recently, Adaptive filtering has become one of the effective and popular methods for the processing and analysis of the ECG signal [5]. It is well known that adaptive filters with LMS algorithm show good performance for processing and analysis of the most of the biomedical signals which are non-stationary [5]. And in this study, we have used adaptive filters to remove the power line interference from the ECG signal. We have used different adaptive filter algorithms, such as, BLMS, DLMS, Filtered XLMS, and Kalman based LMS algorithm. We have used the Signal Processing Toolbox of the mentioned algorithms built in MATLAB.

II. ADAPTIVE FILTER BASIC

The so-called adaptive filter, is the use of the result of the filter parameters a moment ago, automatically adjust the filter Parameters of the present moment, to adapt to the unknown Signal and noise or over time changing statistical properties in order to achieve optimal filtering. Adaptive filter has "self regulation" and "tracking" capacities. Filter out an increase noise usually means that the contaminated signal through the filter aimed to curb noise and signal relatively unchanged.. For the purpose of the filter can be fixed, and can also be adaptive. Fixed filter designers assume that the signal characteristics of the statistical computing environment fully known, it must be based on the prior knowledge of the signal and noise [2]. However, in most cases it is very difficult to meet the conditions; most of the practical issues must be resolved using adaptive filter. Adaptive filter is through the observation of the existing signal to understand statistical properties,

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which in the normal operation to adjust parameters automatically, to change their performance, so its design does not require of the prior knowledge of signal and noise characteristics.

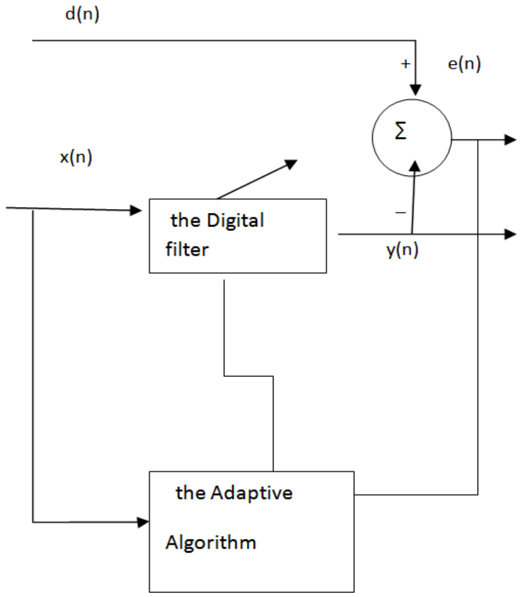

Fig.1 Configuration for adaptive noise cancellation

The figure above is given the general adaptive filtering display digital filter carries on filtering on the input signal x(n), produce output signal y(n). Adaptive algorithm adjusts the filter coefficient included in the vector w (n), in order to let the error signal e(n) to be the smallest. Error signal is the difference of useful signal d(n) and the filter output y(n). Therefore, adaptive filter automatically carry on a design based on the characteristic of the input signal x(n)and the useful signal d(n). Using this method, adaptive filter can be adapted to the environment set by these signals. When the environment changes, filter through a new set of factors, adjusts for new features. The most important property of adaptive filter is that it can work effective in unknown environment, and to track the input signal of time-varying characteristics. Adaptive filter is through the observation of the existing signal to understand statistical properties, which in the normal operation to adjust parameters automatically, to change their performance, so its design does not require of the prior knowledge of signal and noise characteristics.

III. BLOCK LMS (BLMS)

The BLMS algorithm is a modified version of the LMS algorithm. This algorithm provides significant improvements in decreasing mean-squared error (MSE) and consequently minimizing signal distortion [6].

The weight update relation for BLMS algorithm is as follows,

$$w(k+1)=w(k)+\mu \sum_{i=0}^{L-1} x(kl+i)e(kL+i)$$

This gives the filter output of BLMS algorithm in its final form as follows,

$$y(k)=w^T(k)x(k)$$

The output error computation of BLMS as shown below

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

IV. DELAY LMS (DLMS)

The DLMS algorithm is extensively used in different application of adaptive filtering due to low computational complexity and stability. The D-LMS algorithm is introduced to minimize the error between a given preferred signal and output of the linear filter by adjusting recursively the parameters of a linear filter.

The weight update relation for DLMS algorithm is as follows

$$W(n + 1) = w(n) + \mu(n) \text{ x delay}(n)\, e(n)$$

In this paper, if number samples < 20 take μ=0.32 otherwise μ=0.8;

V .FILTERED-X (XLMS)

In ANC applications the adaptive filter is the controller. In this case W (z) is updated by the XLMS algorithm. The secondary transfer function is denoted by A_2(Z) . $\hat{A}_2$(Z) is a model of the secondary transfer function which is identified off-line. The system is described as follows:

$$e_n = d_n - A_2(Z) y_n$$

The weight update relation for XLMS algorithm is as follows

$$w_{n+1} = w_n + \mu e_n \bar{r}_n$$

Where r_n is the vector formed from the actual and delayed samples of the filtered reference signal

$$r_n = \hat{A}_2 (Z)\, x_n$$

$\hat{A}_2$(Z) Can be either infinite or finite impulse response (IIR or FIR) filter, but it is usually an FIR filter. The identification of A_2(Z) can be done by the system . It is a utilization of the simple LMS adaptive filter.

The algorithm of XLMS filter as shown below

S=Read ECG signal

Initialization of F,T, length filter and N values

I = sin (2×Π×F×T/Fs);

n= S+I;

h = adaptfilt.nlms (μ);

[yS,eS] = filter (h, dS);

VII. KALMAN BASED LMS

The proposed kalman based LMS algorithm consists of two basic process. The first is a filter process that involves computing the output of the kalman filter produced by a set of tag inputs and also generating an error estimation by comparing this output to a known desire signal. The second is an adaptive process involves the automatic adjustment of tap weights of the filter according to the error estimation computed in the first process. The algorithm of LMS algorithm as shown below

Define the of k,φS, φI

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vS=exp(1j*(i1)*2*pi*d*sin(øS));
vI=exp(1j*(i-1)*2*pi*d*sin(øI));
I=rand(N.k)                    %interference signal
for n = 1:k)
   x = S(n)*vS + I(n)*vI/kalman equations;
   %y = w*x.';
   y=w'*x;                     %output signal
   e = conj(S(n)) - y;         %error signal
   w=w+mu*conj(e)*x;           %weights update
end
 End
```

Arrayfactor=w* e-j(k) πsinφ

The equation for kalman filter is given by

p(n+1)+qv/σ(n+1

This algorithm uses a steepest decent method [4] and computes the weight vector recursively using the equation.

$$W(n+1) = W(n) + \mu X(n)[d^*(n) - X^H(n)W(n)]$$

Then the reconstructed signal's signal-to-noise ratio(SNR),percentagemeansquareerror (%MSE)and error standard deviation(ESD) are calculated to measure the performance of different adaptive filters.

Moreover, a pragmatic step size (μ) is used for algorithm updating and determining both how quickly the adaptive filter adapts to the filter solution.

VIII. PERFOMANCE MEASUREMENTS

To assess the performance of the proposed filters for removal of noise and to evaluate their comparative performance, different standard performance indices have been used in the thesis. These are defined as follows:

A) Signal to Noise Ratio Improvement (SNR):

SNRI in dB is defined as the difference between the Signal to Noise Ratio (SNR) of the restored image in dB and SNR of noisy image in dB i.e.

SNRI (dB) = SNR of restored image in dB- SNR of noisy image in dB

Where,

SNR of restored image dB=

$$10\log_{10}\left[\frac{\sum_i\sum_j s_{i,j}^2}{\sum_i\sum_j (s_{i,j}-\hat{s}_{i,j})^2}\right]$$

The higher value of SNR reflects the better visual and restoration performance.

B) Mean square error (MSE):

A small minimum MSE is an indication that the small portion PLI signal is occur at the output ECG signal system.

$$\text{MSE} = \left(\frac{1}{M \times N}\right)\sum_{i=0}^{M-1}\sum_{j=0}^{N-1}(\alpha_{i,j} - \beta_{i,j})^2.$$

Here, αi,j is the input ECG signal coefficients and βi,j is the desired ECG signal coefficients . M and N represent the size of the ECG signal.

IX. SIMULATION RESULTS

The 4-beat original ECG signal is generated by using MATLAB whose sampling frequency is 500 Hz for each beat and amplitude is 1mv. The 50 Hz power line interference is also generated with sampling frequency of 2000 Hz. The power line interference is then added to the original ECG signal to get the mixed signal. Finally, the power line interference is removed using different adaptive filters based on different algorithms, such as, BLMS, DLMS, XLMS and Kalman based LMS algorithm.

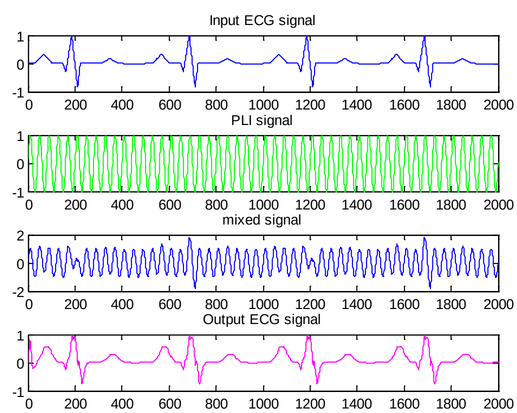

Figure2.Power line interference Removing using BLMS filter

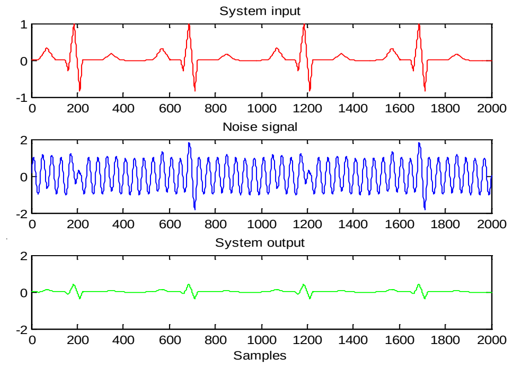

Figure 3 Power line interference Removing using DLMS filter

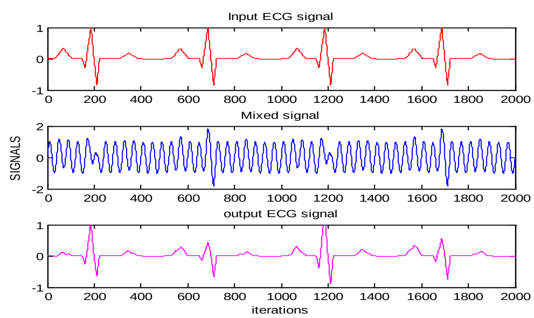

Figure 4 Power line interference Removing using XLMS Filter

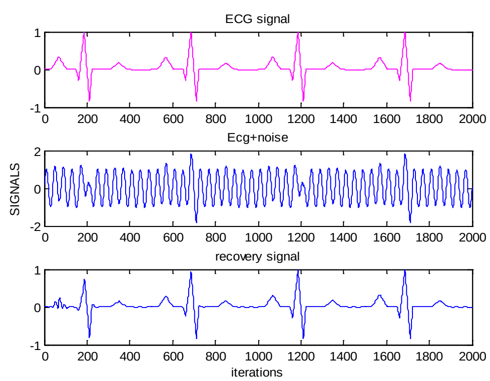

Figure 5 Power line interference removing using kalman LMS filter

Table.1. VALUES OF PERFORMANCE PARAMETERS OF DIFFERENT ADAPTIVE FILTERS

Algorithm	Signal To Noise Ratio	Percentage Mean Square Error	Error Standard Deviation
BLMS	12.6704	0.2463	0.1530
DLMS	12.8945	0.2039	0.1522
XLMS	13.3731	0.1917	0.1492
Kalman-LMS	15.1820	0.1813	0.1306

X. CONCLUSION

In this paper, the Kalman based LMS technique has better performance compare to other adaptive techniques. It is more efficient, low computatation complexity; minimize the error and less power line interference. The SNR of reconstructed ECG signal of Kalman based LMS filter is higher than that of the other filters. So, the Kalman based LMS filter is more appreciable for removing the power line interference from the ECG signal.

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