ECG Signal Denoising Using Digital Filter and Adaptive Filter

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Abstract - *This paper contains some de-noising method of ECG signal. This paper discussed the efficient method of de-noising the ECG signal and gives better SNR (signal to noise ratio). In this paper we designed some filters which provide good SNR value for QRS plane detection and filters are designed for removing the noise from complete ECG signal. Mainly in ECG signal noise generated by artifacts and cardiac rhythm.*

***Key Words*: Biomedical Signal, LPF, HPF, BPF, LMS ECG, Savitzky-Golay, QRS.**

1. INTRODUCTION

The electrocardiograph (ECG) is an instrument, which records the electrical movement of the heart [1]. ECG signal gives the important information about the heart condition and cardiac disorder. This paper describes filtering of ECG signal which eliminate the noise from the signal which are present in ECG signal like:

1. Power line interferences
2. Baseline drift
3. Motion artifacts
4. Muscle contraction (EMG)

Voltage differences of mV order between the given points of the body. Most of the electrophysiological s flags have frequencies between 0 Hz to 3000 Hz and amplitudes ranging between 10 microvolt's and 10 milli-volts [1]. Here a table given which gives information about noises-

Table -1: Noise and their parameters

Signal	Biological source	Average amplitude	Frequency
Electrocardiogram	Heart	1-5 mV	0.05-100 Hz
Electroencephalogram	Brain	10-50	0-150 Hz
Electromyogram	Muscles	0.1-1 mV	40-3000 Hz
Electrooculogram	Eyeball	0.05-3.5 mV	0-125 Hz

For filtering the ECG signal we use LMS algorithm, band pass filter, low pass filter and wavelet transform.

2. SAVITZKY- GOLAY (SG) FILTER

Savitzky-Golay channels are broadly utilized for smoothing and separation. There were considered smoothing and separation channels approximating polynomials of zero to fourth request with 3 - 25 point lengths. Filter coefficient of the low pass filter and high pass filter are given by the Savitzky Golay algorithm [2][5].

The coefficient of filters determine by the practical values. Depends on the order of filter and cut off frequency, here order is 2 and cut off frequency is 15 for low pass filter [5].

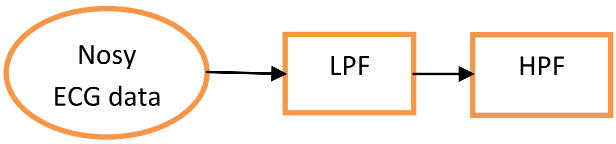

Fig -1: Savitzky – Golay filter (cascade connection of low pass and high pass filter)

In this first noisy ECG data process through Low Pass filter which remove the noise interference and normalized the ECG data. After this LPF output pass through the HPF which gives the high signal to noise ratio for QRS plane detection.

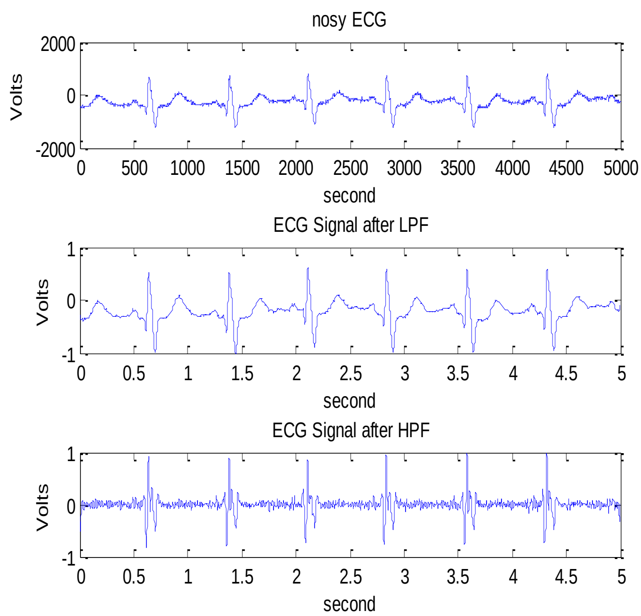

Fig- 2: Output of Savitzky-Golay filter (a) nosy ECG signal (b) output of LPF (c) HPF

This graph showed output of Savitzky-Golay filters where output gives the high SNR values for the QRS plane detection of the ECG signal. Where low pass filter remove the noise frequency which are greater than 50 Hz. High pass filter provides the QRS plan where SNR value is very high.

3. BAND PASS FILTER

For noise cancellation of the ECG, band pass filter is used for QRS detection in algorithm given by the Tompkins [3] and Savitzky- Golay [2].

According to Tompkins algorithm low pass and high pass are connected in cascaded manner for filtering. In Tompkins algorithm digital filter designs which have integer coefficients. LPF is used to remove higher frequency from the ECG data and HPF is used to getting better SNR at QRS plan [3]. Transfer function of low pass and high pass filter are:

$$H(z)= (1-z^{-6})^2 / (1-z^{-1})^2$$

low pass in Z domain

$$x(n)= 2x(n-1)-x(n-2)+y(n-6)+y(n-12)$$

low pass in time domain

where x(n) output of low pass filter, y(n) input of low pass filter nosy ECG signal.

The high pass filter is implemented by the combination of all pass filter and low pass filter with some delay.

$$Hhp(z) = P(z) / Y(z) = z^{-16} – Hlp(z) /32$$

So finally band pass filter equation in time domain:

$$P(n)=y(n-16)-0.03125[x(n-1)+y(n)-y(n-32)]$$

Where P(n) output of band pass filter, y(n) nosy ECG signal, x(n) output of low pass filter.

These equation are depends upon the order of system or filter, here order of filter is 2.

Band pass filter is combination of low pass and high pass filter. Here low pass filter, removes the power line interference and base line effect also. And high pass filter provides the better space for QRS plan which more de-noised by the filter.

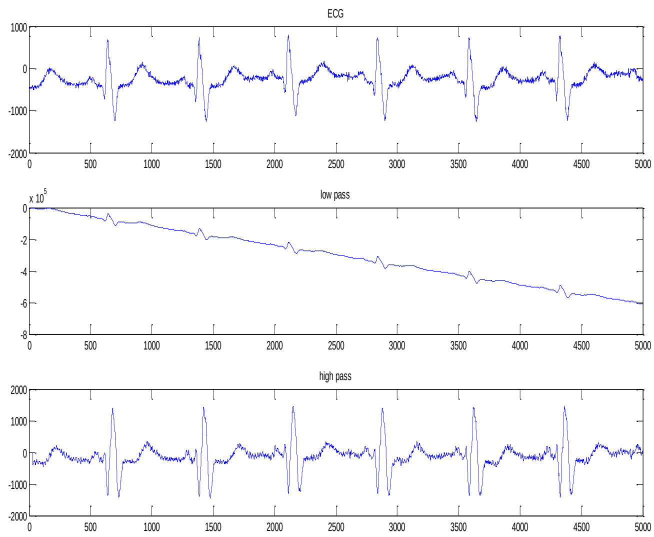

Fig -3: Results of Tompkins filter (a) nosy ECG (b) after Low pass filter (c) output of High pass filter

Another band pass filter can be used for de-noising the ECG signal which is FIR Butterworth filter. Butterworth filter cancels the noise from whole ECG signal and provide good SNR values for every peal and segments.

Using band pass baseline wanders and power line interference noise are removes. For baseline wander cut off frequency is 0.5 Hz (as practically for data set) and cut off frequency for power line interference is 50 Hz. Here both frequencies used as pass band and stop band frequency. This gives very accurate results for noise cancellation.

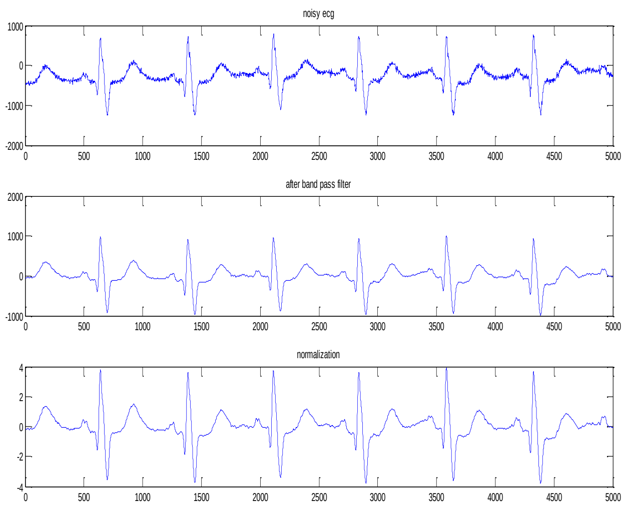

Fig- 4: Results of Butterworth filter (a) nosy ECG signal (b) Output of Butterworth

As the resultant graph showing that band pass filter cancelled the noise from whole signal which is mostly present in signal due to power line, base drift and artifacts.

Lower cut off frequency removes the EEG signal noise and drift effect and higher cut off frequency limited power line and base wander effect. It also limits other biomedical signal which is present in main ECG signal.

4. ADAPTIVE NOISE CANCELLER

Adaptive Noise cancellation is techniques which removes the noise or limit the noise from corrupt signal. A versatile is basically an advanced channel with self-modification attributes. It comprise of two particular section a computerized channel with customizable coefficient, and a versatile calculation, which is utilized to change or alter the coefficient of the channel as appeared in Fig where two information flag d(n) and u(n) are connected at the same time to the versatile separating framework. The flag d(n) is measured flag got utilizing some sensor and it is polluted flag containing both the real flag and clamor and impedance flag, accepted uncorrelated with each other. The flag u(n) is reference input flag and is additionally acquired utilizing a few sensor and it give a measure of the sullying signal which is corresponded somehow with clamor and obstruction in the deliberate flag [4][1].

d(n) – measured signal
s(n) – primary signal
u(n) – reference signal
y(n) – output of Adaptive filter
e(n) – system output signal

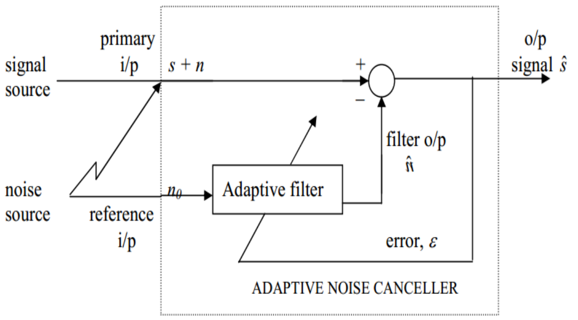

Fig- 5: Block diagram of Adaptive filter

e(n) called to be cost function which going to be minimized by changing the weight of adaptive filter. At the point of adaptive algorithm LMS (Least Mean Square) filter is used in this paper. This process is an adaptive process, which means it cannot require knowledge of signal or noise characteristics [6].

LEAST MEAN SQUARE ALGORITHM: For adaptive filtering LMS algorithm is used mostly. In 1959 Windrow Hoff developed this algorithm for pattern recognition. The LMS calculation is a sort of adaptive filter referred to as stochastic inclination based calculation as it uses the gradient vector of the channel tap weights to focalize on the optimal wiener solution [7]. The objective of designing the FIR wiener channel to produces least mean square gauge of a given flag. let noisy signal is x(n) which is given by:

$$x(n) = d(n) + v(n)$$

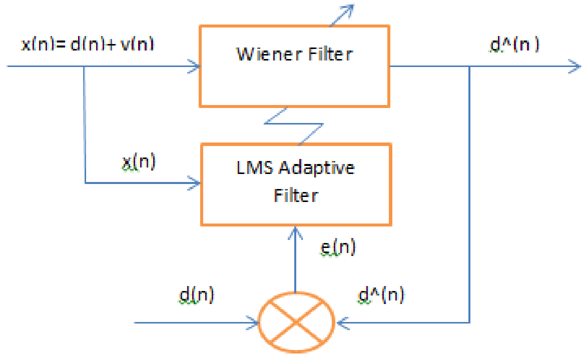

Fig- 6: Adaptive LMS filter block diagram

Here noise signal is v(n) and it is supposed that both signal are WSS (wide sense stationary). Filter coefficient of wiener filter w(n) are known then desired signal obtained by the convolution of x(n) and w(n).

$$d^{\wedge}(n) = \sum_{l=1}^{p-1} w(l)x(n-l)$$

Where (p -1) define the order of filter

Wiener-Hopf condition is utilized to settle the channel coefficient w(n), it is accepted that the v(n) has the zero mean and uncorrelated with the d(n) [6]..

With every emphasis of the LMS calculation the channel tap weights of the adaptive filter are refreshed by the below equation [7].

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

here μ is the step size and it is a positive small constant. w(n) is adaptive filter coefficients at time 'n' and x(n) known as the input vector which contained delayed samples of input signal the progression estimate controls the impact of the updating variable so determination of a reasonable incentive for μ is basic to the execution of the LMS calculation, if the esteem is too little the time the versatile channel takes to merge on the ideal arrangement will be too long; if μ is too vast the versatile channel ends up plainly flimsy and its yield veers [7]

Designing of the LMS algorithm

Mainly 3 steps are present in LMS algorithm

1. The filter output ,d^(n) which is cost function

$$d^{\wedge}(n) = \sum_{l=1}^{p-1} w(l)x(n-l) = w^T(n)x(n)$$

2. Error estimation is given by this equation:

$$e(n)=d(n)-d^{\wedge}(n)$$

3. Filter weights updating equation

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

In the ECG signal power line interference is related to the main ECG signal so noise and signal both are correlated in natural. Both are related so it is difficult to remove noise from signal with any lose from the main ECG signal.

For testing of LMS algorithm first it applied on ideal ECG signal (which is present in MATLAB) and comment upon its working and output results. After that Adaptive LMS algorithm applied on the practical nosy ECG signal.

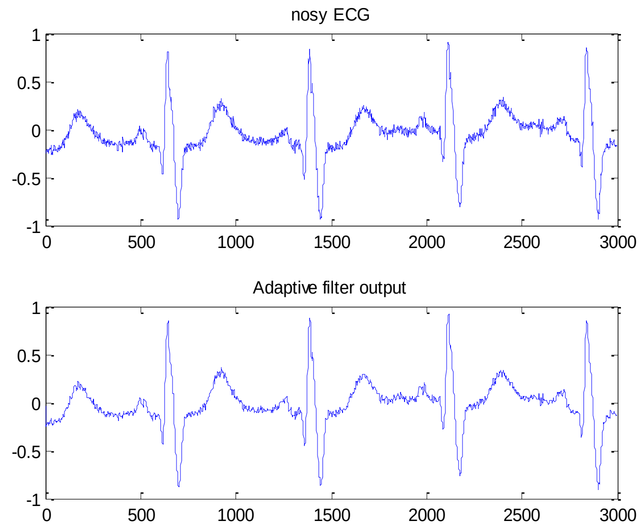

Fig- 7: Adaptive filter (a) nosy ECG signal (b) output of adaptive filter

This graph represents the output of the LMS algorithm which is limiting the noise which is present in the signal. Power line noise (50 Hz) is correlated to the main ECG signal for limiting this noise LPF is used. LPF is IIR Butterwort filter which has order 2 and cut off frequency is 0.05 (practically according data set).

So final output is very smooth and noise very and it is giving good SNA value for whole signal.

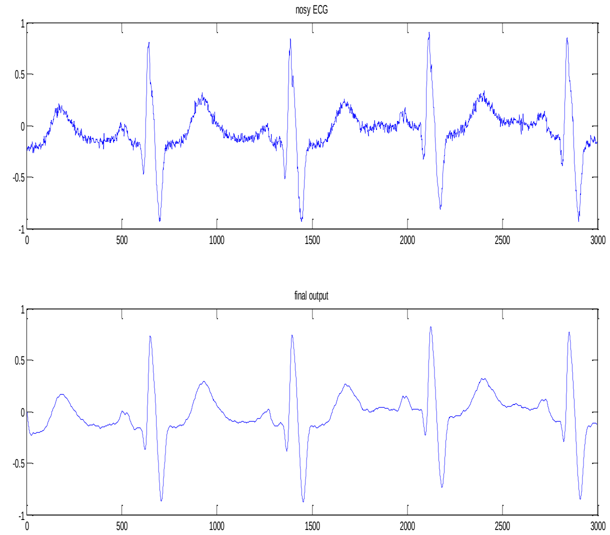

Fig- 8: results of LMS + low pass filter

This is the output of MATLAB code which is made by the combination of adaptive filter and low pass filter.

RESULTS:

This paper contains three methods for noise cancellation or de-noising the ECG signal. In which two methods are used for de-noising the ECG signal for detection QRS plan that are band pass filter (Tompkins) and another is Savitzky-Golay (sg) filter. In Band pass filter, two methods are present one is combination of LPS and HPF and other is Butterworth band pass filter. For QRS plane detection Savitzky-Golay filter provides good SNR value. But Butterworth filter provides good SNR value for whole signal.

Table -2: Comparing PSNR and MSE for BPF filters

ECg signal	Savitzky-Golay		Butterworth	
	PSNR	MSE	PSNR	MSE
S003	63.8864	0.02657	61.8839	0.04213
S005	67.3653	0.01192	63.1549	0.03144
S007	65.5311	0.01819	63.3504	0.03006

In this table Savitzky –golay band pass filter provides the good PSNR values nad less MSE value but it is suitable for only QRS detection.

Adaptive filter is used to remove the problem of find coefficient and frequency for filter. And some time filters are affect the main signal during de-noising process. Adaptive filter is provides good SNR values for signal. It is not that much good but it keep the signal safe and decrease the calculation problem.

Table 3: Comparing PSNR and MSE for Adaptive filter combinations

ECG Signal	Adaptive + LPF		LPF + Adaptive	
	PSNR	MSE	PSNR	MSE
S003	61.7401	0.04355	617903	0.04305
S005	62.3596	0.03776	63.1329	0.03160
S007	63.8078	0.02705	64.0414	0.02564

his table shows that Adaptive filter is removing better noise from the main ECG signal after LPF.

3. CONCLUSIONS

Savitzky-Golay filter is better than Tompkins filtering process for detection of QRS plane. But getting good SNR values for whole signal than Butterworth filter is good but it required cut off frequency and some time it effect the signal so it replaced by the adaptive filtering. Adaptive filtering is provide good SNR values and less calculation for de-noising compared to others.

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