Design and Simulation of Low Pass IIR Filter for ECG Interference Reduction

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***Abstract*- In this paper, a low pass IIR filter has been designed and simulated to reduce interference in ECG signals. ECG signal is of prime importance because many of people suffer from the heart diseases. So, there should be more accuracy in recording the ECG signal because it contains very low frequency and amplitude signals. Thus, it can beeasily contaminated with different types of noises. The main focus of this paper is to reduce the noise created by electronic gadgets and instruments in vicinity of the ECG recording machine. The filter has been designed using two IIR filters i.e. Butterworth and Chebyshev. The performance of both the filters have been compared and analysed in terms of PSD. The filters have been designed and simulated using MATLAB. It can be observed from the simulated result that performance of Butterworth filter is better.**

Keywords- ECG, IIR filter, Butterworth, Chebyshev

I. INTRODUCTION

Electrocardiography or ECG machine is a device that measures and records the electrical activity of the heart from Electrodes placed on the skin in specific locations.ECG signals are formed of P wave, QRS complex, T wave.

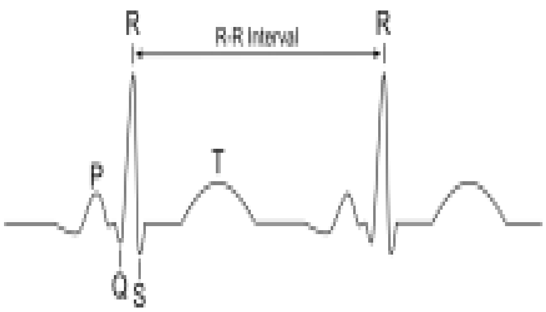

Fig.1 A standard scalar ECG signal

ECG graphically gives useful information that relates to the heart functioning by means of a base line and waves representing the heart voltage changes during a period of time, usually a short period [1]-[3]. Putting leads on specific part of the human body, it is possible to get changes of the bioelectrical heart signal where one of the most basic forms of organizing them is known as Einthoven lead system.

In spite of the special value, the ECG is considered only a laboratory test. It is not an absolute truth concerning the cardiac pathologies diagnosis. There are examples of patients presenting string heart diseases which present a normal ECG, and also perfectly normal patients getting an abnormal ECG [4]. Therefore, an ECG must always be interpreted with the patient clinical information. According to [5] a signal can be analysed and processed in two domains, time and frequency. ECG signal is one of the human body signals which can be analysed and worked in time domain or frequency domain. Figure 1 presents typical waves in an ECG signal. P, Q, R, S, T and U are specific wave forms identified in the time domain of an ECG signal. The QRS complex, formed by Q, R and S waves, represents a relevant wave form because the heart rate can be identified locating two successive QRS complex.
During recording process different types of noises are inserted in the ECG signal such as noise generated from the biological and enviromental resources .Major noises including base line wandering ,motion antifacts ,EMG interface are the noises generated by biological resources. Power line interferenc and instrumentation noise are related with enviromental noise. Among the above mentioned noises, power line interference is a significant source of noise. The ECG signal can be wrongly manipulated if the noise is added to it. To remove interference from the ECG signal, filtering process is used. Both analog filters and digital filters are appropriate for the task of filtering but vast advantage of digital filters over analog filters makes digital filters a better choice these days.[6]-[7]

II. IIR FILTER

IIR filters are characterized by the following recursive equation -

$$y(n) = \sum_{k=0}^{\infty} h(k)x(n-k) = \sum_{k=0}^{N} b_k x(n-k) - \sum_{k=1}^{M} a_k \ y(n-k) \quad \dots (1)$$

where h(k), is the impulse response of the filter, b_k and a_k are the coefficients of the filter, and x(n) and y(n) are the input and output to the filter. The transfer function for the IIR filter is given by

$$H(z) = \frac{b_0 + b_1 z^{-1} + \dots + b_N z^{-N}}{1 + a_1 z^{-1} + \dots + a_M z^{-M}} = \frac{\sum_{k=0}^{N} b_k z^{-k}}{1 + \sum_{k=1}^{M} a_k z^{-k}} \quad ..(2)$$

Equations (1) and (2) are the characteristics equations for IIR filters.

For designing IIR filter, the filter specification is required by the designer such as the stop band attenuation, pass band attenuation, ripples allowed in pass band and stop band etc. The next step is to calculate the coefficient of the required filter and select an appropriate method for designing the filter such as pole-zero placement, impulse invariant, matched z- transform or bilinear z-transformation. Now a suitable filter structure is employed for realizing the designed filter.

III. IIR FILTER DESIGN SIMULATION

Two filters are used i.e. Butterworth and Chebyshev to remove high frequency noise, low pass filter is used. Butterworth filter has a magnitude response given by the following equation:

$$|H(j\Omega)| = \frac{A}{[1 + (\frac{\Omega}{\Omega_c})^{2N}]^{0.5}} \quad \dots (3)$$

where A is the filter gain and Ω_c is the 3 db cut –off frequency and N is the order of the filter.

The Chebyshev low pass filter has a magnitude response given by the following equation:

$$|H(j\Omega)| = \frac{A}{[\, 1 + \varepsilon^2 C_N^2 \left(\frac{\Omega}{\Omega_c}\right)]^{0.5}} \quad \dots (4)$$

where A is the filter gain, ε is a constant and Ω_c is the 3 dB cut-off frequency.[8]

IV. REMOVAL OF HIGH FREQUENCY NOISE: BUTTERWORTH LOW PASS FILTER

Now, aim is to remove interference above 100 Hz. Here, a design to implement the Butterworth low pass filter is presented. The sampling frequency of 1000Hz and filter order 4 is used.Fig.2 shows magnitude response is maximally flat and monotonic in the stop band. Fig.3 shows phase response is linear in pass band. Fig.4 shows all the poles in the unit circle i.e. designed filter are stable.

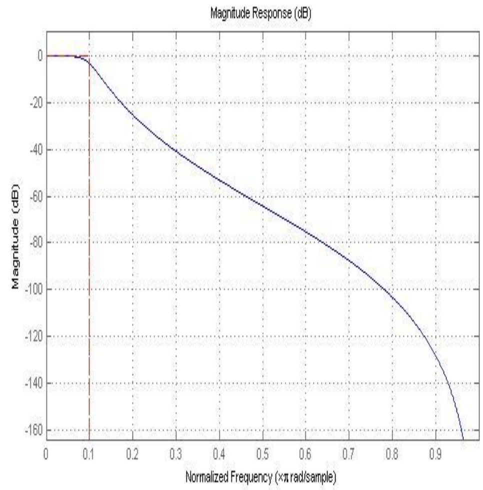

Figure 2. Magnitude response for low pass Butterworth filter

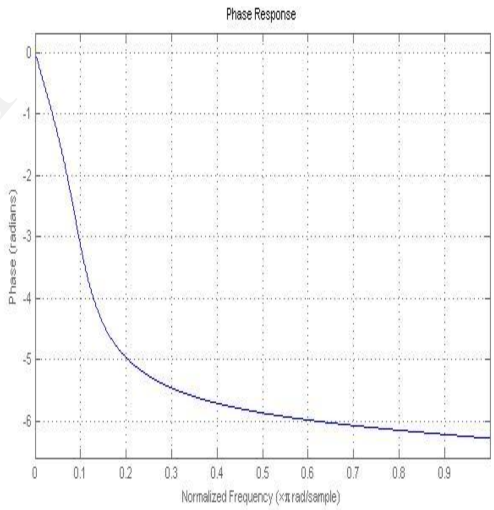

Figure 3. Phase response for low pass Butterworth filter

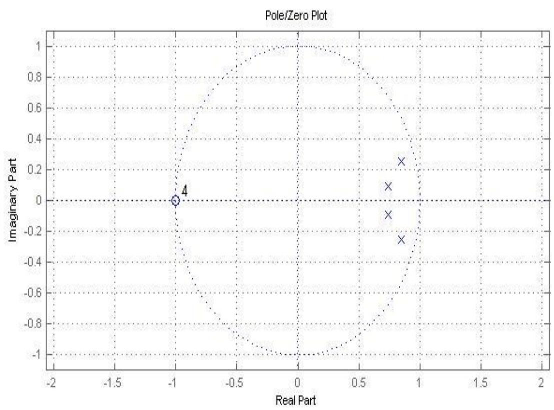

Figure 4. Pole zero plots for low pass Butterworth filter

V. REMOVAL OF HIGH FREQUENCY NOISE: CHEBYSHEV TYPE 1

Now, aim is to remove interference above 100 Hz. Here, a design to implement the Chevyshev type1 low pass filter is presented. The sampling frequency of 1000Hz and filter order 4 is used.Fig.5 shows magnitude response is maximally flat and monotonic in the stop band. Fig.6 shows phase response is approximately linear in pass band. Fig.7 shows all the poles in the unit circle i.e. designed filter are stable.

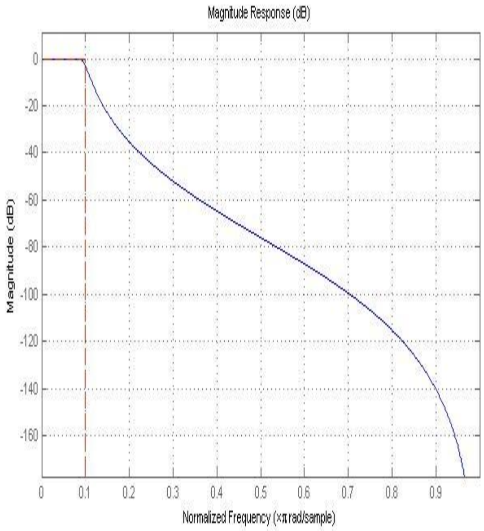

Figure 5. Magnitude response for low pass Chebyshev type1 filter

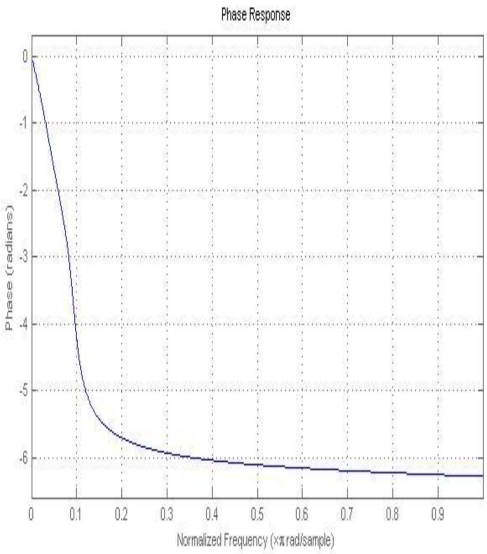

Figure 6. Phase response for low pass Chebyshev type1 filter

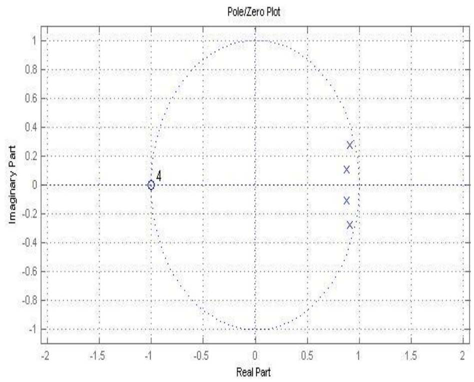

Figure 7. Pole zero plots for low pass Chebyshev type1 filter

VI. RESULTS AND DISCUSSION

Implementation of Butterworth low pass filter

After implementation, fig.8 shows ECG signal before and fig.9 shows signal after low pass filtration respectively.

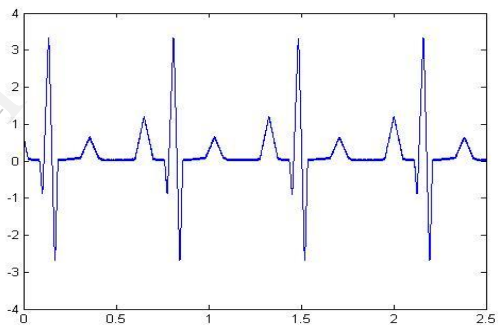

Figure.8 ECG signal before low pass filtration

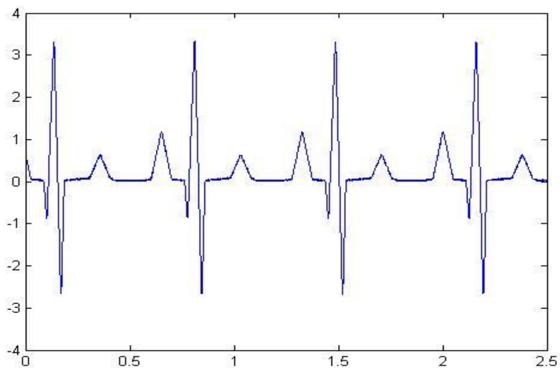

Figure.9 ECG signal after low pass filtration

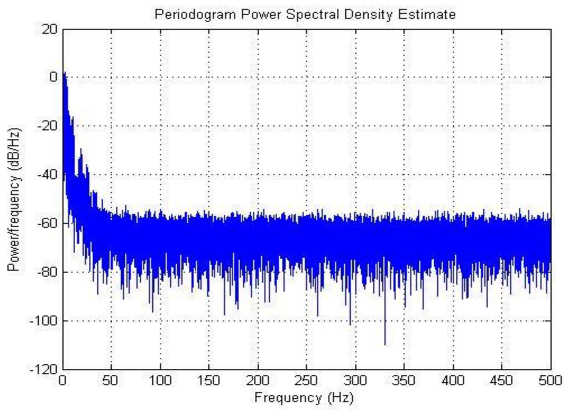

Figure.10 Frequency spectrum before low pass filtration

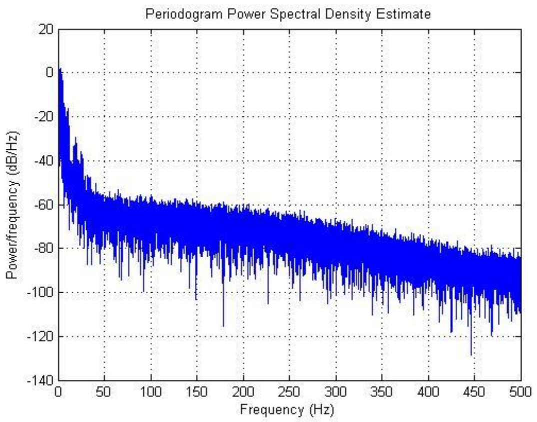

Figure.11 Frequency spectrum after low pass filtration

At a frequency of 100 Hz before filtration spectrum exhibits signal power of-84 dB. After application of filter, power dropped to -81 dB shows removal of high frequency noise.

Implementation of Chebyshev type 1 low pass filter

After implementation, fig.12 shows ECG signal before and fig.13 shows signal after low pass filtration respectively.

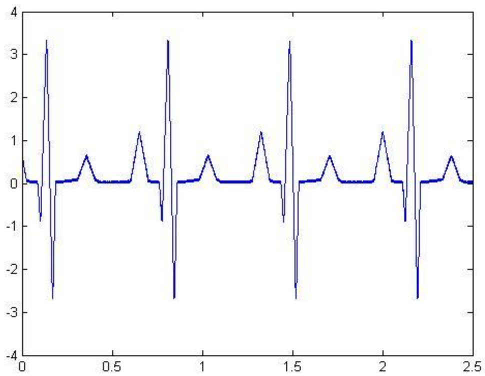

Figure.12 ECG signal before low pass filtration

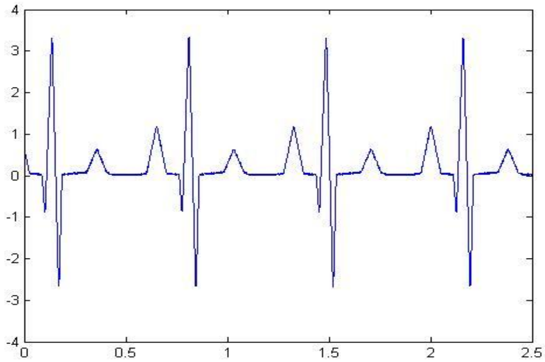

Figure.13 ECG signal after low pass filtration

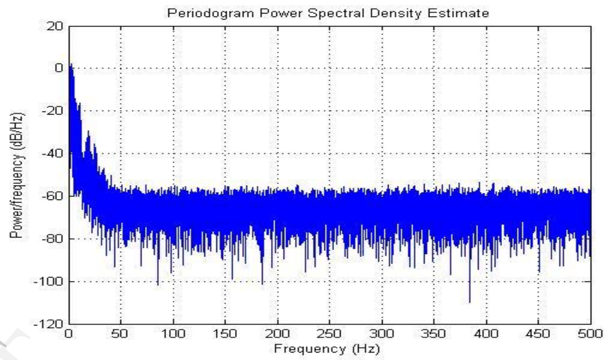

Figure.14 Frequency spectrum before low pass filtration

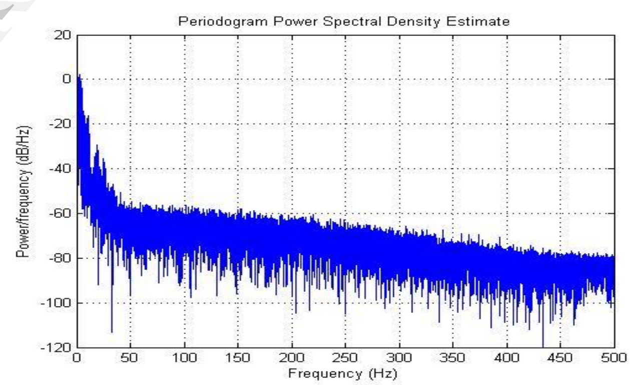

Figure.15 Frequency spectrum after low pass filtration

At a frequency of 100 Hz, before filtration spectrum exhibits signal power of-83 db. After application of filter, power dropped to -82 dB shows removal of high frequency noise.

Table 1 Comparative analysis of the two proposed IIR filter

Filter Type	Before Filtration Power at 100 Hz(dB)	After Filtration Power at 100 Hz(dB)
Butterworth	-84	-81
Chebyshev type 1	-83	-82

VII. CONCLUSIONS

The proposed method presents a powerful and simple tool for investigation of ECG analysis in frequency domain. The ECG is the record of time varying bio-electric potential generated by electrical activity of heart. Various filters have been used to achieve spike free signal.In this paper we have compare the results of ECG signal filtered by two IIR filter: Butterworth and Chebyshev. The comparative analysis (table-1) shows that theButterworth is better choice to reduce interference.

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