DESIGN OF DIGITAL IIR FILTER FOR NOISE REDUCTION IN ECG SIGNAL

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***Abstract*— Electrocardiogram (ECG) signal has been widely used in cardiac pathology to detect heart disease. A digital infinite-impulse response (IIR) filter design is proposed in this paper. This includes an implementation and evaluation of butterworth low pass infinite impulse response filter method to remove high frequency noise and for this filter is applied to noisy ECG data sample and original sample are taken as reference signal. The suggested method considers the magnitude response for choosing the cutoff frequency and the FFT spectrum estimate response to find the lowest filter order. The structure and the coefficients of the digital IIR filter are designed using FDA tool in MATLAB. The filter output's average power before and after filtration are calculated using FFT and for simulation of this filter, the hardware is designed using microcontroller At mega 16 A. For hardware designing the samples taken are record no. 108 and record no.119 (taken from MIT-BIH database, ML II signal). Here samples are taken from MIT-BIH arrhythmia database (mitdb) ML II are used.**

***Keywords*— ECG, MIT-BIH database, MLII ECG Data Signal, IIR Filter, FDA tool, FFT spectrum.**

I. INTRODUCTION

First clear electrocardiographic (ECG) signal was obtained over a century ago in 1895 by Willem Einthoven [1]. Since then ECG analysis became a standard and a trusted technique of detecting cardiac defects.

Electrocardiogram (ECG) signal is some index of the functionality of the heart. For example, a physician can detect arrhythmia by studying abnormalities in the ECG signal. Since very fine features present in an ECG signal may convey important information, it is important to have the signal as clean as possible [2]. Figure 1 shows a clean ECG signal. Power line interference is easily recognizable by interfering voltage in the ECG may have frequency 50/60 Hz [3].

In this graph Q is the bottom starting point, R the top point, and S the bottom end point of the QRS Complex. Atrial contraction starts in the P-wave and continues throughout the PR interval. The blood is pumped into the ventricles and the ventricular pressure rises. Ventricular contraction begins at R which corresponds to the peak of the QRS complex, and continues during the ST segment and T-wave [4], Rinky lakhwani, Arpita Singh, Shahanaz Ayub, J.P. Saini paper deals with Comparisan of different digital filters for QRS complex extraction from electrocardiogram [5].

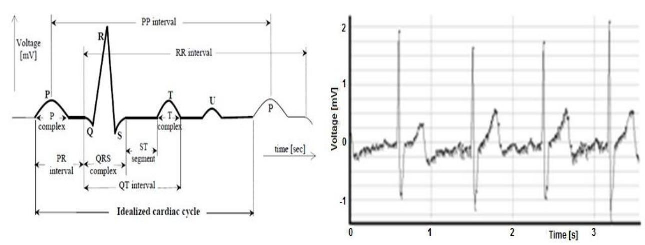

Figure 1: Schematic (left) [3] and actual (right) ECG signals. On the actual signals extreme values are highlighted

The mitral and bicuspid valves close due to increase in ventricular pressure. At R, the closing valves produce the heart sound. Between point R and S ventricular pressure increases greatly since the semi lunar valves are still closed and there is no blood flow. The semi lunar valves open at S when the ventricular pressure equals the aortic pressure. The ventricular contraction forces blood into the aorta and there is an increase in both aortic and ventricular pressure. The blood is pumped from the ventricles and is carried away in the aorta. Ventricular pressure drops. When the pressure drops below aortic pressure, the semi lunar valve shuts. After the T-wave the ventricular

978-0-7695-5069-5/13 $26.00 © 2013 IEEE
DOI 10.1109/CICN.2013.45

pressure falls below atrial pressure and the mitral and bicuspid valves open up [4].

The interference may be due to stray effect of the alternating current fields due to loops in the patient's cables. Other causes are loose contacts on the patient's cable as well as dirty electrodes.

The samples initially have noise so we have to filter this. Here normal and abnormal both type of ECG signal are taken. Normal ecg is defined as constant ECG patterns that are neither linked to corresponding typical symptoms, nor to corresponding clinical and anamnestic findings, and not to drugs. The diagnosis of the normal electrocardiogram is made by excluding any recognized abnormality. It's description is therefore quite lengthy. The normal ECG reveals a regular supraventricular rhythm of60 beats/min e.g. Sample 115 is normal ECG signal as shown in fig. 2.

Anything other than a "normal sinus rhythm" would be considered abnormal. The only "borderline" case would be beating to fast or to slow. Slow is around 50 BPM and fast is around 100 BPM. e.g. Sample 119 is abnormal ECG signal as shown in fig. 3.

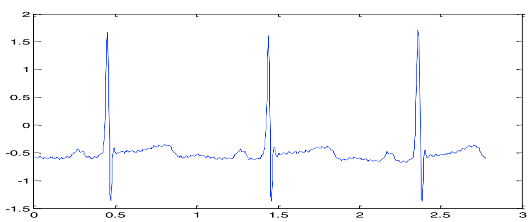

Figure 2: Normal ECG (115) signal plot

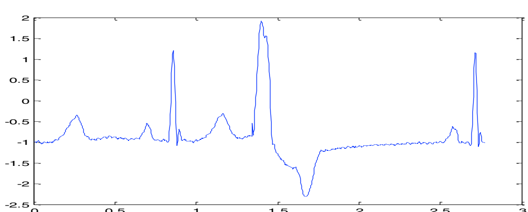

Figure 3: Abnormal ECG (119) signal plot

IIR LOW PASS FILTER-

The typical digital signal processing (DSP) includes z transform, Fourier transform, convolution, correlation, and filtering, etc. The advantages of DSP are programmable, it has high reliance, high precision, powerful anti-interference, easy to maintain, and easy to design for obtaining linear phase, etc. IIR systems have an impulse response function that is non zero over an infinite length of time. IIR Filter may be implemented as either analog or digital filter. In digital filter, the output feedback is immediately apparent in the equation defining the output.

An IIR filter is one whose impulse response theoretically continues forever because the recursive terms feedback energy into the filter and keep it as specified in the following cascade form equation as

$$H(z) = K \prod_{k=1}^{n} \frac{1+b_k z^{-1}}{1+a_k z^{-1}} \prod_{i=1}^{m} \frac{1+d_{i1} z^{-1}+d_{i2}}{1+c_{i1} z^{-1}+c_{i2}} \qquad (1)$$

where K is the gain, a_k and b_k for $k = 1, 2, \ldots, n$ are the first order coefficients, and c_{i1}, c_{i2}, d_{i1}, and d_{i2} for $i = 1, 2, \ldots, m$ are the second-order coefficients.

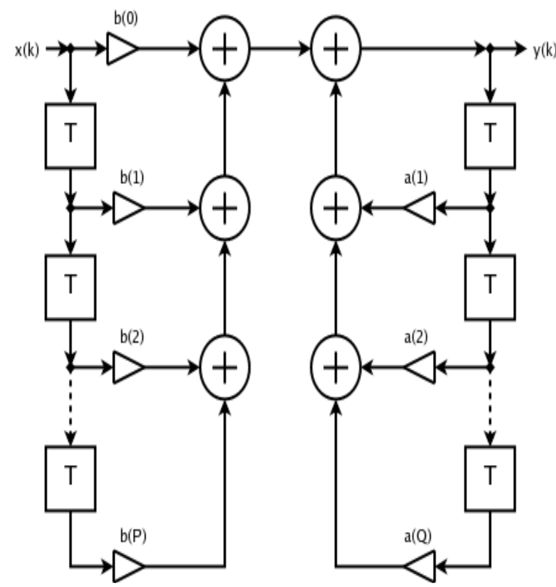

Figure 4: Basic structure of IIR filter

The digital filter design is a process in which a digital hardware or a program is constructed to meet the given specification. The traditional digital IIR filter design involves the analog IIR filter design and the analog-to-digital transformation. When the specification for the digital filter is given, we first change it to the corresponding analog low-pass (LP) filter and use one of the well-known LP filter design methods, such as Butterworth, Chebyshev Type I, and Chebyshev Type II, to fulfill the requirements. Then, the analog LP filter is transferred to the digital filter using the bilinear transformation [6], Mohandas Choudhary, Ravindra Pratap Narwaria "**Suppression of Noise in ECG Signal Using Low pass IIR Filters**" describes a IIR filter design for noise removing[7] it gives various types of IIR filter design but result of Butterworth is best among all. Nalini Singh,Shahanaz Ayub,J.P. Saini "**Removing of high frequency noise in ecg signal using low pass iir filter**"[8] also explain this type of filter for different signal and give good result.

II. METHODOLOGY

In this paper, the processed ECG waveforms were taken from the MIT-BIH Arrhythmias Database and MATLAB help from the math work to design IIR butterworth filter to remove high power noise from ECG. The ECG signal we used is real clinical data and has been sampled at fs = 360Hz for a period of just over 10 seconds. The ECG signal we began with is plotted in the time domain with correct time axis along with the signal amplitude on ML II (in mV) with correct axis as well. The model using digital filter is built in the simulink of the MATLAB. Figure 5 shows the model used in the system.

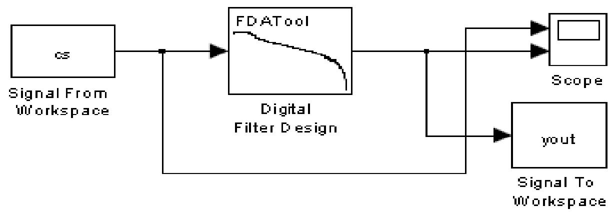

Figure 5: Simulation model used for the ECG processing

In the model, sample value taken from MIT- BIH database are given as signal from workspace (cs).

Signal from workspace is the voltage value of the input signal taken from MIT-BIH database. And the plot of normal ECG signal (Record 108 from MIT-BIH) is shown as figure 6. Here we take the time from 0:00:000 to 0:02:775 second.

The magnitude and phase plot of this filter are shown in Figure 7. An IIR butterworth low pass filter offers the very best of what IIR filters have to offer; very high attenuation with a low order. The FDA tool parameters set up as low pass IIR butterworth filter with sampling frequency 360 Hz and 5^{th} order. Cut off frequency is 20Hz and as we know attenuation (A_P) of filter at cutoff frequencies is fixed at 3db that is half the passband power is set in FDA tool. These properties lend themselves to being a light computational load. This high attention along with a narrow stop band can easily be seen in the magnitude plot of the filter.

As the criteria is to get the original signal without data loss, four different cut-off frequencies, 10 Hz, 15 Hz, 20 Hz and 25Hz are used to observe the effects of the cut-off frequency. At 10Hz cutoff frequency loss is around 62.96%, at 15Hz loss is 38% that means at these frequencies the amplitude of the signal reduced i.e. loss of the signal. The suppression performance of 20 Hz is higher than above frequencies, when time domain result is examined carefully and filtered signal is very closer to the input signal. At 25Hz cut-off frequency extra impulses are added which are not present in original signal. Thus, 20 Hz cut-off frequency is appropriate for high frequency noise removing.

Filter order can be obtained by using Welch power spectrum. Power of noisy signal of record 108 before filtration is -53.3110 db. Table 1 below show the power of sample 108 at different order;

Order	Power after filteration
1	-53.3724
4	-54.0852
5	-56.0660

From table it conclude that 5^{th} order filter reduced more power as compare to other order filter and also this power is less as compare to noisy signal that means the noise is reduces. Hence 5^{th} order is suitable for this purpose and implementation of this order filter is not so tough.

In FDA tool IIR butterworth filter magnitude and phase response shown in figure 6.

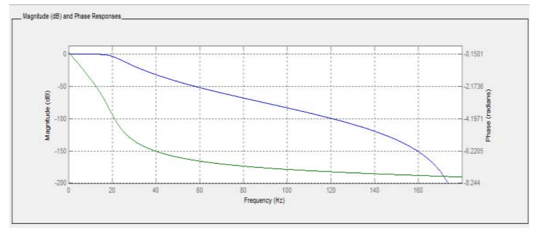

Figure 6: Magnitude and Phase Response of Butterworth Filter

For hardware, the system is designed around the components available in indigenous market and also preferring those which are commonly used in the ECG processing.

The signal is taken from MIT- BIH database with high frequency noise in text form. So for hardware implementation this sample value is feed into the controller.

In this hardware sample 108 and 119 are used and for noisy ecg signal 10 bit DAC used, and for filtered ecg 16 bit DAC used. The following technical specifications were decided and finalized and implemented for the system.

1. Resistance value 1**Ω.**
2. Power Supply ± 12 V, 5V.
3. Crystal oscillator 16MHz.

Basically system has been divided into three blocks

1. Microcontroller
2. 10 bit Digital to Analog converter
3. 16 bit Digital to Analog converter
4. Inverting Operational Amplifier 741U

Figure7 shows basic building block diagram of the designed low pass filter circuit.

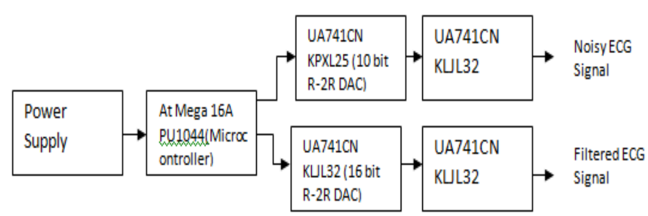

Figure7: Block diagram of the designed low pass filter circuit

Firstly the filter coefficient is initialized into microcontroller then the digital value of noisy ECG signal from MIT-BIH database are given but in microcontroller there's some problem with negative value so for positive input voltage firstly signal level is upped by 3V . Then this value is converted into digital value through ADC within microcontroller and then this noisy input signal is send to the port A and port C. By using 10 bit R-2R DAC this noisy digital ecg signal is converted into analog ecg signal. But here an inverting DAC is used so an inverted signal is obtained. So for original noisy signal here another inverting amplifier is used. In this inverting amplifier a variable resistance is used for maintaining the reference point of the signal because original signal had already shifted for microcontroller, so at this stage original noisy signal is obtained. Using any display instrument (like CRO, DSO) this noisy ECG signal can be seen. In microcontroller the filter coefficient is already feed so within the microcontroller the digital noisy ECG signal is again converted into analog signal and then filtered using its equation. For storing, this signal level is again shifted (because it also have negative value) then this filtered signal is stored on port B and port D. By connecting 16 bit R-2R inverting DAC on port B and port D this signal can be obtained in analog form but here also an inverted filtered signal obtained so again the same process is done and at this stage the original filtered signal is obtained and by using any displaying element this filtered signal can be displayed.

III. RESULT OF FILTERED ECG SIGNAL

Figure 8 shows the ECG signal with high frequency noise. The corresponding FFT spectrum estimate is shown in figure 9. When the signal shown in the figure 8 is filtered with help of designed filter the resulting ECG trace is shown in the figure 10 and the corresponding FFT spectrum estimate is shown in the figure 11.

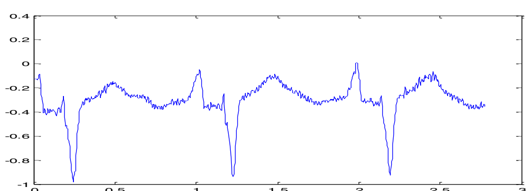

Figure 8: ECG signal before butterworth filter application

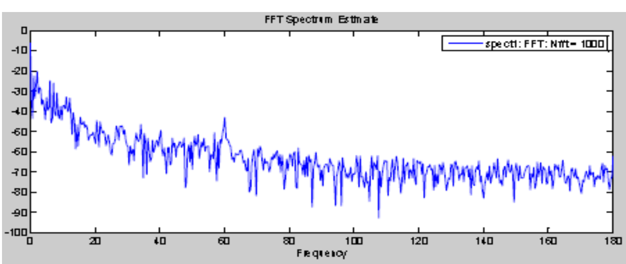

Figure 9: FFT spectrum estimate before butterworth filter application

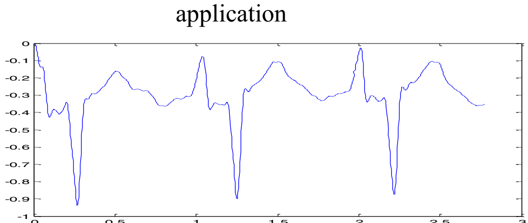

Figure 10: ECG signal after butterworth filter application

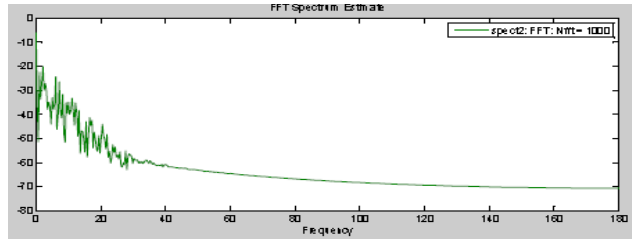

Figure 11: FFT spectrum estimate after butterworth filter application

This filter is used for 10 samples taken from MIT-BIH database. Table 2 shows the power of the filtered signals are less than noisy signal that means the noise is reduced.

Table 2

Sample	Power of Noisy signal(db)	Power of Filtered signal(db)
108	-53.3110	-56.0660
112	-40.6527	-43.3317
113	-28.3356	-31.4285
114	-32.8174	-35.6678
115	-29.0949	-30.4739
119	-33.7735	-34.3959
124	-37.4673	-50.7744
200	-38.6643	-42.3114
201	-30.6989	-33.2008
203	-36.6975	-37.6089

When we give these sample value in filter designed in [7] then the FFT spectrum value of both the filter is given in table below, suppose signal1 stands for filter described in this paper and signal 2 for filter designed in [7]

Table 3

Sample	Power of Noisy signal(db)	Power of Filtered signal1(db)	Power of Filtered signal2(db)
108	-53.3110	-56.0660	-53.5600
112	-40.6527	-43.3317	-42.6016
113	-28.3356	-31.4285	-30.6411
114	-32.8174	-35.6678	-32.8229
115	-29.0949	-30.4739	-28.3668
119	-33.7735	-34.3959	-34.1307
124	-37.4673	-50.7744	-41.0636
200	-38.6643	-42.3114	-40.0625
201	-30.6989	-33.2008	-30.5617
203	-36.6975	-37.6089	-35.0919

From the table it can be see that filter described in this paper remove more noise as compare to filter designed in [7] that means filter described in this paper is best as compare to filter in[7].

From the transfer function equation of the filter the output signal equation (which is given in programming of microcontroller) can be derived in which using filter coefficient the output signal value can be found. Figure 12 and figure 13 shows the filtered output signal versus time,

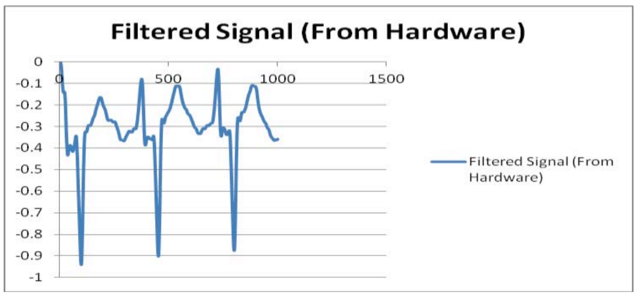

Figure 12: Filtered ECG signal 108 from Hardware

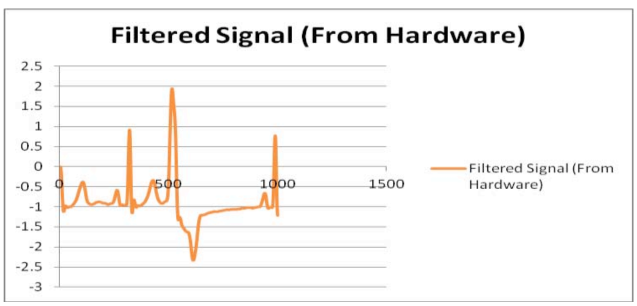

Figure 13: Filtered ECG signal 119 from hardware

When the result of ECG sample 108 and 119 are compare through MATLAB and hardware result is shown in figure 14 and figure 15 respectively.

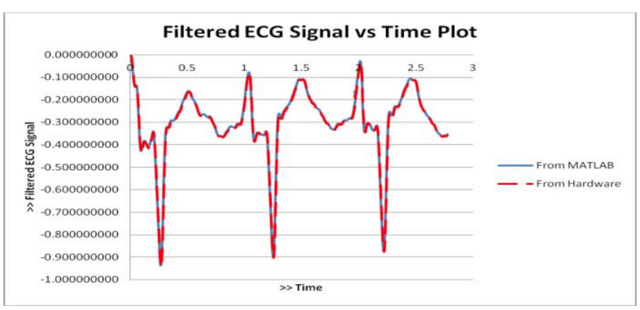

Figure 14: ECG sample 108 through MATLAB and hardware

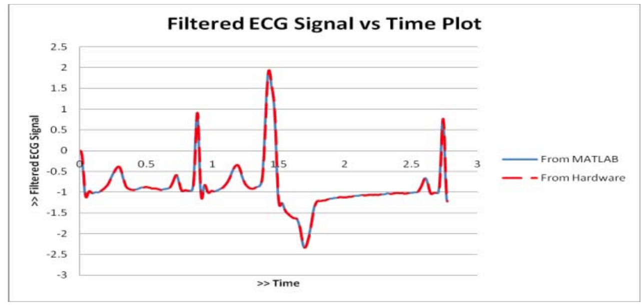

Figure 15: ECG sample 119 through MATLAB and hardware

From figure 14 and figure 15 the simulated data 108 and 119 shows that designing of butterworth filter is verified.

IV. CONCLUSION

The result show that 20 Hz cutoff frequency is appropriate for high frequency noise removing because at this more noise is removed and there is no loss of original sample data. Fifth order filter is appropriate for noise reduction because below this order noise cannot removed properly and above this order implementation becomes tough. From filtered signal result it seems that power of filtered signal is less as compared to power of noisy signal that means noise is removed and this result is also verified through designed hardware result. Table 3 shows a comparison between this designed filter and the filter designed in [7] from which it seems that this designed filter remove more noise as compared to filter designed in [7].

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