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Fpga Implementation Of High Speed Fir Low Pass Filter For Emg Removal From ECG

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Abstract

The main objective of this project is to implement efficient hardware architecture for high speed FIR filter for Electro-myo-gram (EMG) removal from Electro-cardio-gram (ECG) signals. For removing EMG signal from ECG signal, I proposed a symmetry coefficient with branched tree architecture in FIR low pass filter. By using this architecture, I can reduce the delay and increase the performance of the filter. Here I designed the filter by using the coefficient quantization technique. By using this quantization technique I can also reduce the area of the filter. The final result will be expected to be a better performance regarding logic occupation and speed of the filter. This architecture is implemented in FPGA by using Xilinx ISE Design Suite 13.1 and matlab R2010a and it will be targeted into the device of Virtex-5. This is the referred device in modern digital signal processing.

Index Terms: *ECG, EMG…….*

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1. INTRODUCTION

The electrocardiogram (ECG) signal is an electrical signal generated by the heart's beating, which can be used as a diagnostic tool for examining the function of the heart. A typical scalar electrocardiographic lead is shown in Fig: 1.1, where the significant features of the waveform are the P, Q, R, S and T waves, the duration of each wave, and certain time intervals such as the P-R, S-T, and Q-T intervals. Accurate determination of the QRS complex is reliable on the interval of R-R wave peak. The largest amplitude portion of the ECG, caused by currents generated when the ventricles depolarize prior to their contraction. This QRS complex has greater amplitude and atrial repolarisation.

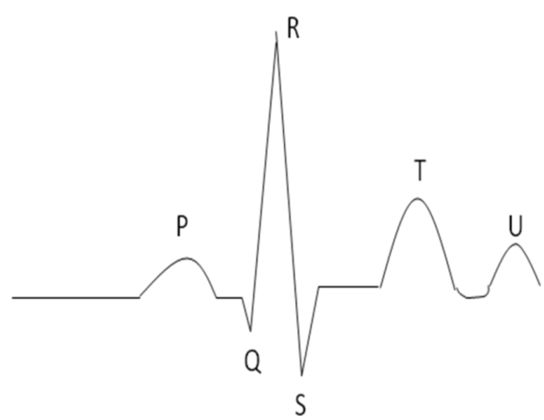

Fig 1.1: Typical ECG signal

The largest amplitude portion of the ECG, caused by currents generated when the ventricles depolarize prior to their contraction. This QRS complex has greater amplitude and atrial repolarisation. Typically an ECG has five deflections, arbitrarily named "P" to "T" wave. The Q, R, and S wave occur in rapid succession, do not appear in all leads, and reflect a single event, and thus are usually considered together. A Q wave is any downward deflection after the P wave. An R wave follows as an upward deflection, and the S wave is any downward deflection after the R wave. The T wave follows the S wave, and in some cases an additional U wave follows the T wave.

There have been several investigations dealing with the QRS complex detection for ECG signals. For instance, Pan and Tompkins proposed an algorithm to recognize QRS complex in which they analyzed the positions and magnitudes of sharp waves and used a special digital low pass filter to reduce the false detection of ECG signals.

1.2 Noises in ECG Signal:

There are several noises in ECG signal some of them are given below

1. EMG signal
2. Power line interference
3. Electrode contact noise
4. Motion artifacts
5. Baseline drift and ECG amplitude modulation with respiration
6. Instrumentation noise generated by electronic devices used in signal processing, and
7. Electrosurgical noise

1.2.1 EMG (Electromyogram) Signal:

Muscle contraction cause art factual millivolt-level potential to be generated. This muscle contraction can be assumed to be transient bursts of zero-mean band limited Gaussian noise. The variance of the distribution may be estimated from the variance and duration of the bursts. Typical parameters of the muscle contraction has a standard deviation of 10 percent of peak-to-peak ECG amplitude and duration of 50ms and the frequency content of dc to 10000 Hz.

1.2.2 Power Line Interference:

Power line interference consists of 60Hz pickup and harmonics, which can be modelled as sinusoids and combination of sinusoids. These power line interference varied in a power line noise include the amplitude and frequency content of the signal. These characteristics are generally consistent for a given measurement situation and, once set will not change during detector evaluation. Typically, the amplitude up to 50 percent of peak to peak ECG amplitude

1.2.3 Electrode Contact Noise:

Electrode contact noise is transient interference caused by loss of contact between the electrode and skin, which effectively disconnects the measurement system from the subject. The loss of contact can be permanent, or can be intermittent, as would be the case when a loose electrode is brought in and out of contact with the skin as a result of movements and vibration. This switching action at the measurement system input can result in large artifacts since the ECG signal is usually capacitively coupled to the system. With the amplifier input disconnected, 60 Hz interference may be significant.

Electrode contact noise can be modelled as a randomly-occurring rapid baseline transition which decays exponentially to the baseline value and has a superimposed 60 Hz component. This transition may occur only once or may rapidly occur several times in succession. Characteristics of this noise signal include the amplitude of the initial transition, the amplitude of the 60 Hz component, and the time constant of the decay.

1.2.4 Motion Artifacts:

Motion artifacts are transient baseline changes caused by changes in the electrode-skin impedance with electrode motion. As the impedance changes, the ECG signal amplifier sees a different source impedance, which forms a voltage divider with the amplifier input impedance. Therefore, the amplifier input voltage depends on the source impedance, which changes as the electrode position changes. The usual cause of motion artifacts will be assumed to be vibrations or movement of the subject. The shape of the baseline disturbance caused by motion artifacts will be assumed to be a biphasic signal resembling one cycle of a sine wave. The peak amplitude and duration of the artifact are variables.

1.2.5 Baseline Drift with Respiration

The drift of the baseline with respiration can be represented as a sinusoidal component at a frequency of respiration added to the ECG signal. The amplitude and frequency of the sinusoidal component should be variables. The amplitude of the ECG signal also varies by about 15 percent with respiration. The variation could be reproduced by amplitude modulation of the ECG by the sinusoidal component which is added to the baseline. Typically, the amplitude variation takes 15 percent of peak to peak ECG amplitude and also baseline variation takes 15 percent of peak to peak ECG amplitude variation at 0.15 to 0.3 Hz.

1.2.6 Noise Generated by Electronic Devices

Artifacts generated by electronic devices in the instrumentation systems cannot correct by QRS detection algorithm. The input amplifier has saturated and no information about the ECG can reach the detector. In this case an alarm must sound to alert the ECG technician to corrective action.

1.2.7 Electrosurgical Noise

The Electrosurgical noise completely destroys the ECG signal and can be represented by a large amplitude sinusoidal with frequencies approximately between 100 kHz and 1MHz. Since the sampling rate of an ECG signal is 250 to 1000 Hz, an aliased version of this signal would be added to the ECG signal. The amplitude, duration, and possibly the aliased frequency should be variable. Typically the amplitude has 200 percent of peak to peak to ECG amplitude.

The Digital FIR filter works under quantized analog input and produced digital output. The filter is designed by using filter coefficients. The input signal is convoluted with these filter coefficients and produced the desired output signal.

Here we proposed a symmetric branched tree architecture connection for FIR filter to remove the EMG signal from ECG signal. For this FIR filter, the coefficients are generated by using FDA tool to increase speed and for reducing the hardware consumption we used quantized coefficients in Q(16,14) format where 16 bits represent filter coefficient and 14 bit represent the fraction and MSB bit represent the sign bit.

MSB (15)	Bit (14)	Bit (13)	Bit (12)	Bit (11)	Bit (10)	Bit (9)	Bit (8)	Bit (7)	Bit (6)	Bit (5)	Bit (4)	Bit (3)	Bit (2)	Bit (1)	LSB (0)
S	M	M_F	M_F	M_F	M_F	M_F	M_F	M_F	M_F	M_F	M_F	M_F	M_F	M_F	M_F

Table 1.1: Numeric representation format Q (16, 14) bit distribution

Here, "S" denotes the sign bit, "m" denotes the non fractional magnitude and "MF" denotes the fractional magnitude.

2. LOW PASS FIR FILTERS

2.1 Ideal Low Pass Filter:

An Ideal Low Pass Filter allows all the frequency components below the designed cut-off frequency and rejects all the frequency components above the designed cut-off frequency.

Its frequency response satisfies

$$H_{LP}(e^{j\omega}) = \begin{cases} 1, & 0 \le \omega \le \omega_c \\ 0, & \omega_c < \omega \le \pi \end{cases} \qquad (1)$$

The impulse response of the ideal low pass filter (1) can easily be found to

$$h_{LP}(n) = \frac{\sin(\omega_c n)}{\pi n}, \quad -\infty < n < \infty \qquad (2)$$

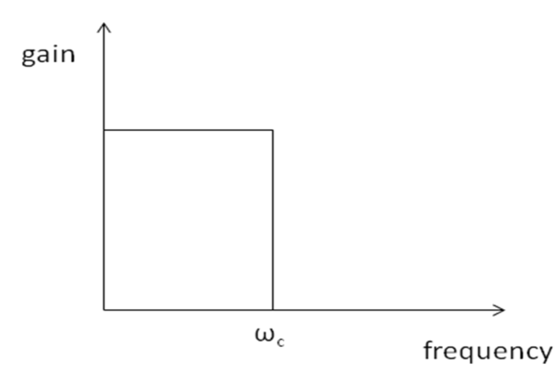

Fig 2.1: Ideal low pass filter

But when it comes to practical oriented we cannot design the exact ideal filter. So some distortion will occurred at the sampling frequency.

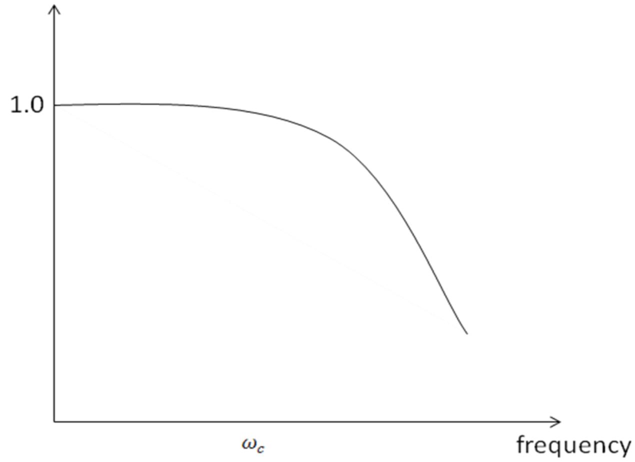

Fig 2.2: Practical low pass filter

2.2 Digital FIR Filter:

A filter is used to modify an input signal in order to facilitate further processing. A digital filter works on a digital input and produces a digital output. According to Dr. U. Meyer –Baese, "the most common digital filter is the Linear Time Invariant (LTI) filter". Designing an LTI involves arriving at the filter coefficients which, in turn, represents the impulse response of the proposed filter design. These coefficients, in linear convolution with the input sequence will result in the desired output. The linear convolution process with the input sequence with the input sequence will result in the desired output. The linear convolution process can be represented as

$$y(n) = x(n) * h(n) \tag{3}$$

Here, $y(n)$ represent the output of the filter and $x(n)$ is the digital input of the filter. The impulse response of the filter is given by $h(n)$ and the operator $*$ denotes the convolution operation.

2.3 Optimal filter design method:

Here optimal filter in the sense that a filter that is the best that can be achieved for the given number of impulse response coefficients. The design of optimal linear phase FIR filters can be achieved by considering the filter design problem as weighted chebyshev approximation problem. Thus it is possible to derive a set of conditions for which the designed filter is optimal and unique.

In this method, following terms are defined to formulate the FIR filter design problem as chebyshev approximation problem.

$H_d(\omega) =$ The desired frequency response of the filter
$H(\omega) =$ The frequency response of the designed filter
$w(\omega) =$ The frequency response of the weighted function

The weighing function enables the designer to choose the relative size of the error in different types of linear phase FIR filters (i.e, whether length N of the filter is odd or even and filter is symmetric or anti-symmetric) can be written as

$$H(\omega) = e^{-j\omega(N-1)/2} e^{j\pi L/2} |H(\omega)| \tag{4}$$

TABLE-2.1: Values of L and $|H(\omega)|$ for four different types of FIR filters

	L	$\lvert H(\omega)\rvert$
Case 1: N odd Symmetrical impulse response	0	$\sum_{n=0}^{(N-1)/2} a(n)\cos(\omega n)$
Case 2: N even Symmetrical impulse response	0	$\sum_{n=1}^{N/2} b(n)\cos[\omega(n-0.5)]$
Case 3: N odd Anti-Symmetrical impulse response	1	$\sum_{n=1}^{(N-1)/2} c(n)\sin(\omega n)$
Case 4: N even Anti-Symmetrical impulse response	1	$\sum_{n=1}^{N/2} d(n)\sin[\omega(n-0.5)]$

Each of the expressions for $|H(\omega)|$ in the above table can be written as a product of fixed function of ω [call this as $Q(\omega)$] and a term that is a sum of cosines[*call this as* $P(\omega)$]. Thus the expressions for $|H(\omega)|$ in TABLE 2.1

become as

TABLE-2.2: Expressions for $P(\omega)$ and $Q(\omega)$ for different types of filters

	$Q(\omega)$	$P(\omega)$
Case 1	1	$\sum_{n=0}^{(N-1)/2} p(n)\cos(\omega n)$
Case 2	$\cos(\omega/2)$	$\sum_{n=0}^{N/2-1} q(n)\cos(\omega n)$
Case 3	$\sin\omega$	$\sum_{n=0}^{(N-3)/2} r(n)\cos(wn)$
Case 4	$\sin(\omega/2)$	$\sum_{n=0}^{N/2-1} s(n)\cos(\omega n)$

The weighted error of approximation $E(\omega)$ is, by definition

$$H(\omega) = W(\omega)[H_d(\omega) - |H(\omega)|]$$

$$= W(\omega)[H_d(\omega) - P(\omega)Q(\omega)] \quad (5)$$

Since $Q(\omega)$is a fixed function of frequency it can be factored out and gives

$$E(\omega) = \widehat{W}(\omega)[\widehat{H}_d(\omega) - P(\omega)] \quad (6)$$

Where $\widehat{W}(\omega) = W(\omega)Q(\omega)$

$$\widehat{H}_d(\omega) = H_d(\omega)/Q(\omega)$$

The Chebyshev approximation problem may now be stated as finding the set of coefficients $[p(n), q(n), r(n), \text{or } s(n)]$ to minimize the maximum absolute value of $E(\omega)$ over the frequency bands in which the approximation is being performed. The Chebyshev approximation problem may be stated mathematically as follows:

$$\text{Min}_{coefficients}[max|E(\omega)|] \quad (7)$$

3. LOW PASS FIR FILTER ARCHITECTURES

3.1 Low Pass Serial FIR Filter architecture:

For these FIR Filter it require one adder, one multiplier and one delaying unit. So with respect to hardware efficiency it is a good option but FIR filter made using this architecture is very slow and the through put of the device is very low.

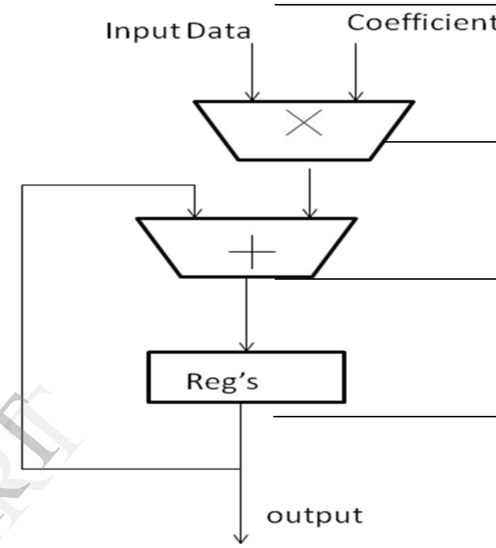

Fig 3.1: Serial FIR filter architecture

3.2 Conventional FIR Filter:

The n^{th} – order LTI FIR filter can be seen to consist of a collection of a "tapped delay line," adders and multiplier is an FIR coefficient, often referred to as a "tap weight" for obvious reasons. Historically, the FIR filter is also known as the "transversal filter," "tapped delay Line" structure

$$y(n) = \sum_{k=0}^{L-1} x[n-k]\,a[k] \quad (9)$$

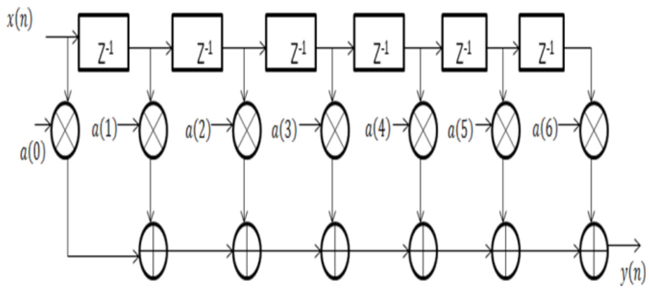

Fig 3.2: Conventional 5-Tap FIR filter

Here I am show for simple 5 tap FIR filter. The main advantage of the conventional Tap FIR Filter is more accurate and the main disadvantages of this filter is it takes so much time to execute the filter and also required more hardware design that mean it occupies large area and also increased the cost.

3.3 Branched Tree Architecture

In the conventional Tapped FIR Filter architecture, the adders are connected in branch tree form they add data parallely instead of serial addition as it was connected in serial adder FIR design.

So the advantage of this design compared to previous design this that it reduces the critical delay of the FIR filter architecture. But it also increases the area of the filter so whenever the area of the filter is increased the cost of the filter is also increased.

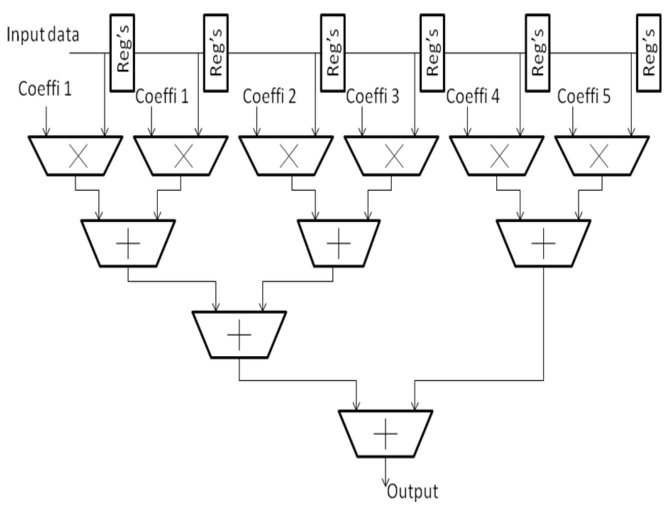

Fig 3.3: Branched tree architecture

3.4 Symmetric Coefficient FIR Filter

3.4.1 Odd Symmetric Coefficient FIR Filter

The impulse response for many filters possesses significant symmetry. This symmetry can generally be exploited to minimize the arithmetic requirements and produce area- efficient filter realizations.

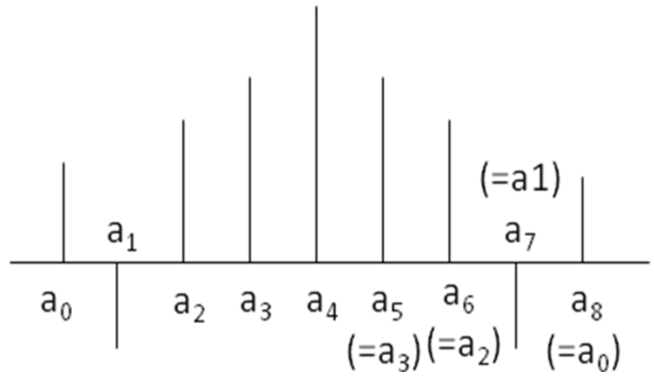

Fig 3.4: Symmetric FIR-odd number of terms

Instead of implementing this filter using the conventional tapped FIR filter architecture, we can reduce half of the multiplications. This significant reduction in the computation workload can be exploited to generated efficient filter hardware implementations.

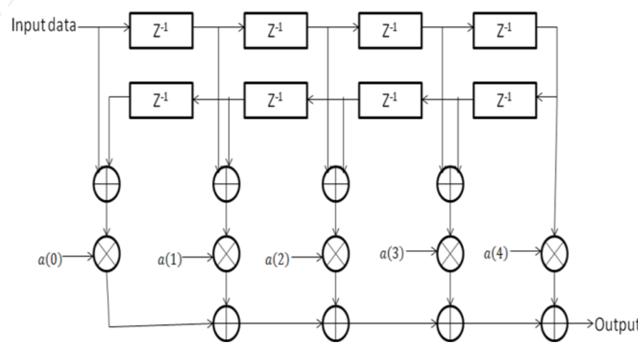

Fig 3.5: Exploiting coefficient symmetry – odd number of filter tap

3.4.2 Even Symmetric Coefficient FIR Filter

Coefficient symmetry for an even number of terms can be exploited

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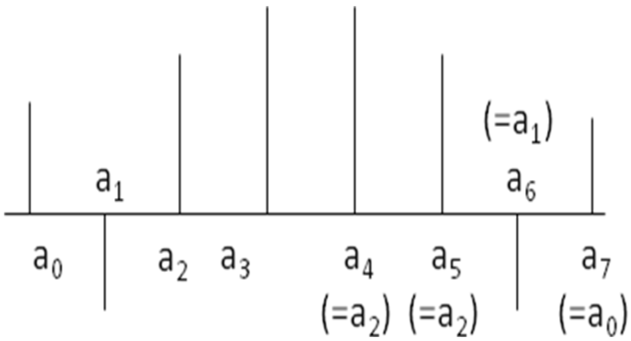

Fig 3.6: Symmetric FIR-odd number of terms

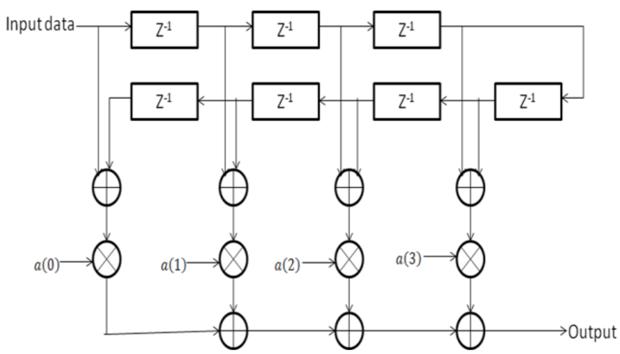

Fig 3.7: Exploiting coefficient symmetry – even number of filter tap

Instead of implementing this filter using the conventional tapped FIR filter architecture, we can reduce half of the multiplications. This significant reduction in the computation workload can be exploited to generated efficient filter hardware implementations.

3.4.3 Symmetric Coefficient FIR Filter with Branched Tree Architecture

By using this architecture, I can reduce the multiplier so it increased the performance of the filter and execute the operation very fast and also we changed in the adder section. In this adder section, I implemented the branch tree architecture so that it increased the speed of the filter and at the same time we reduce the cost of the filter.

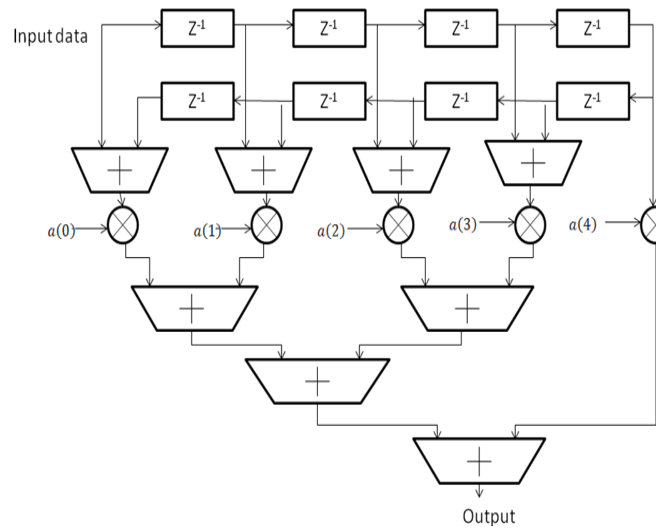

Fig 3.8: Symmetry FIR filter with branched tree

In these FIR filter architecture that process data using symmetry coefficient are added parallel instead of serial. It gives the valid output after 38 clock cycle or in other words we can say that it having the latency of 38 clock cycles and here we are designed 37 order FIR filter with 256Hz sampling frequency. For this FIR filter, the coefficients are generated by matlab fdatool kit. These coefficients are generated in the form of $Q(16, 14)$. Where total bit length of coefficient is 16 bit among them 14 bits are fraction and MSB represent the sign bit.

4. SIMULATIONS AND RESULTS

4.1 Input Signal:

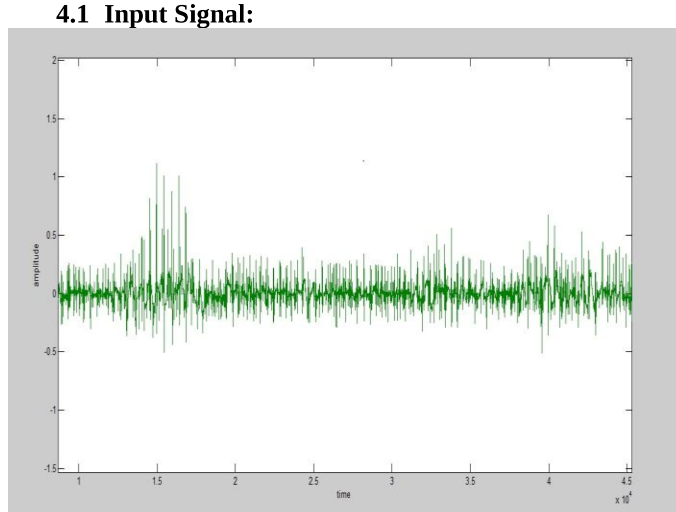

Fig 4.1: ECG signal with EMG

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4.2 Magnitude Response of the Filter:

Order =37
Sampling Frequency=256Hz
Pass Band Frequency=35Hz
Stop Band Frequency=45Hz
Ripple Pass Band=48.25dB
Ripple Stop Band=1.925dB

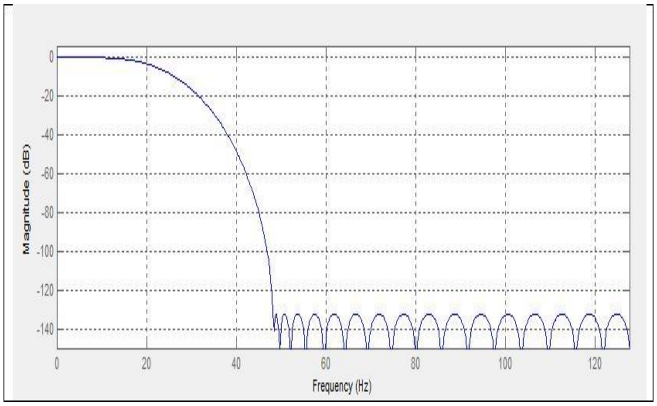

Fig 4.2: Magnitude response of the filter n=37

4.3 Synthesis Report:

Table 4.1: Device utility summary

Device Utility Summary			
Logic Utilization	Used	Avialable	Utilization
Number of Slice Registers	608	93120	1%
Number of Slice LUTs	868	46560	1%
Number of fully used LUT-FF pairs	308	908	33%
Number of bonded IOBs	49	240	20%

4.4 Output Signal:

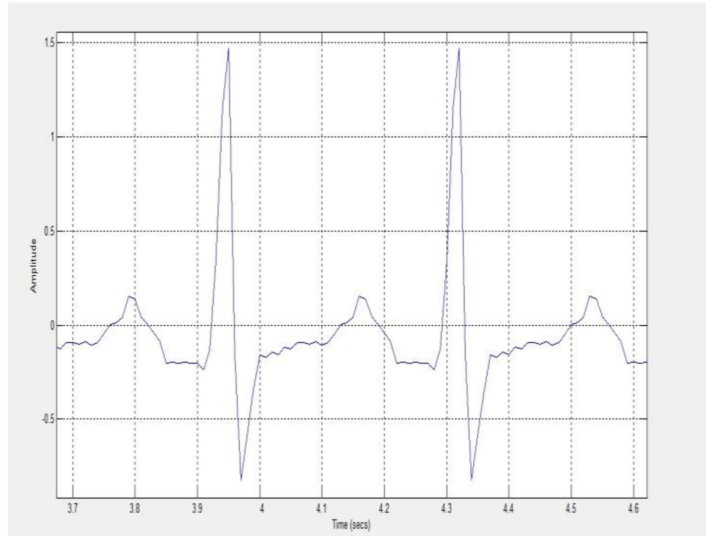

Fig 4.3: ECG signal

CONCLUSION

Everyone can produce the ECG signals more accurately in the powerful software tools but when coming to the hardware design it is the biggest challenge to get that much of accurate signal and another biggest challenge to reduce the cost of the device and also hardware consumption at the same time to produce better ECG signal. In this paper we proposed the new design approach to design the high speed FIR low pass filter for filtering EMG from ECG signal.

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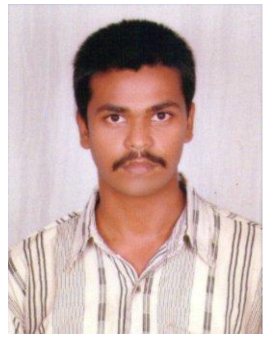	LEELA KRISHNA M RESEARCH SCHOLAR ROBOTICS SRM UNIVERSITY TAMILNADU INDIA
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