The Designing of a FIR Low Pass Filter and Amplifier for Small Voltage Signals using Kaiser Window

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ABSTRACT

Digital signal processing (**DSP**) is the study of signals in a digital representation and the processing methods of these signals. A **digital filter** uses a digital processor to perform numerical calculations on sampled values of the signal. The analog input signal must first be sampled and digitized**.** The resulting binary numbers, representing successive sampled values of the input signal, are transferred to the processor, which carries out numerical calculations on them. These calculations typically involve multiplying the input values by constants and adding the products together. If necessary, the results of these calculations, which now represent sampled values of the filtered signal, are output to convert the signal back to analog form. To achieve the desired filtering effect, Digital filters can achieve virtually any filtering effect that can be expressed as a mathematical function or algorithm.

1. INTRODUCTION

In the design of frequency selective filters, desired filter characteristics are specified in the frequency domain in terms of the desired magnitude & phase response of the filter. In the filter design process, the coefficients of a causal FIR are determined that closely approximates the desired frequency response specifications.

Although the frequency response characteristics possess by ideal filter may be desirable, they are not absolutely necessary in most practical applications if these conditions are relaxed, it is possible to realize causal filters that approximate ideal filters as closely as desired.

In this paper the EMG signals are considered which have the voltage levels in the range of 10mv and frequency up to 120 Hz which is considered as the low frequency small voltage signal. Now a low pass FIR filter for the small signal having the cutoff frequency of 120 Hz is designed.

2. SYSTEM MODEL

A desirable property of the window function is that the function is of finite duration in the time domain and that the Fourier transform has maximum energy in the main lobe or a given peak side lobe amplitude. In a Kaiser window the side lobe level can be controlled with respect to the main lobepeak by varying a parameter,α. The Kaiser Window function is given by

$$w_{k(n)} = \begin{cases} \dfrac{I_0(\beta)}{I_0(\alpha)}, & for\ |n| \le \dfrac{M-1}{2} \\ 0, & else\ where \end{cases}$$

Where

$$\beta = \alpha\left[1 - \left(\frac{2n}{M-1}\right)^2\right]^{0.5}$$

$I_0(\beta)$ & $I_0(\alpha)$ are the Bessel functions

The actual passband ripple A_p and minimum stop band attenuation A_s are given by

$$A_p = 20\log_{10}\frac{1+\delta p}{1-\delta p} \quad \text{dB}$$

$$As = -20\log_{10}\delta_s \quad \text{dB}$$

$\delta = \min(\delta_p, \delta_s)$

From the Kaiser design equation

$$\alpha = \begin{cases} 0, & for\ A_s \le 21 \\ 0.5842(A_s-21)^{0.4} + 0.07886(A_s-21), & for\ 21 < A_s \le 50 \\ 0.1102(A_s - 8.7) & for\ A_s > 50 \end{cases}$$

Order of the filter can be find from the equation

$$M \ge \frac{FD}{\Delta F} + 1$$

Where the parameter D is

$$f(x) = \begin{cases} 0.9222, & for A_s \le 21 \\ \dfrac{A_s - 7.95}{14.36}, & for A_s > 21 \end{cases}$$

The impulse response is computed from

$$\mathrm{h(n)} = \mathrm{w_k(n)h_d(n)}, \text{for } |n| \le \frac{M-1}{2}$$

for Low Pass FIR Filter

$$f(x) = \begin{cases} \left(2\dfrac{f_c}{F}\right)\dfrac{\sin 2\pi n f_c/F}{2\pi n f_c/F}, & for\ n > 0 \\ 2\dfrac{f_c}{F}, & for\ n = 0 \end{cases}$$

Where $f_c = 0.5(f_p + f_s)$, and $\Delta F = (f_s - f_p)$

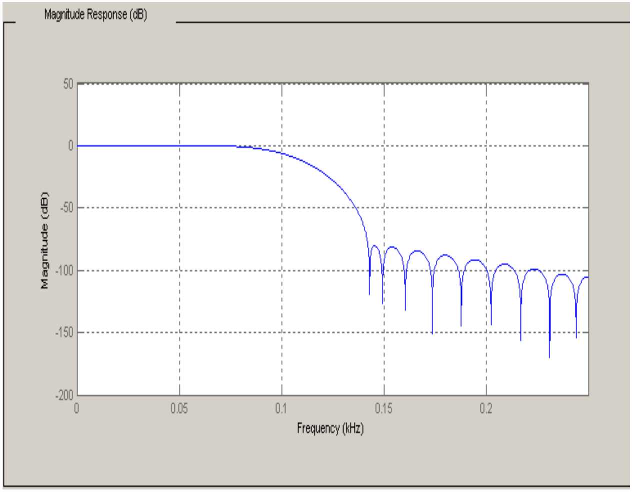
Algorithms 1

1. Start from initial point $\delta = \min(\delta_p, \delta_s)$
2. Calculate the minimum stopband attenuation (A_s).
3. Compute the parameter α and parameter D
4. Now we calculate the order of filter M which generate the coefficients of low pass FIR filter.
5. These coefficients can also be find out in MATLAB by using filter designing and analysis.

Coefficient:

h(0)=2.823182709039e-005,h(1)=-0.0001281807058205, h(2)=-0.0005583885039392,h(3)=7.814605953656e-019, h(4)=0.002324543544181, h(5)=0.002522685109819, h(6)=-0.004154983325905, h(7)=-0.01054471968294, h(8)=6.154784193546e-018,h(9)=0.0234258008186, h(10)=0.02093083214918, h(11)=-0.03013912511589, h(12)=-0.07130955821932,h(13)=1.413841573615e-017, h(14)=0.1948586436646,h(15)=0.3727442184403, h(16)= 0.3727442184403, h(17)=0.1948586436646, h(18)=1.413841573615e-017,h(19)= -0.07130955821932, h(20)= -0.03013912511589, h(21)=0.02093083214918, h(22)=0.0234258008186, h(23)=6.154784193546e-018, h(24)=-0.01054471968294, h(25)=-0.004154983325905, h(26)=0.002522685109819, h(27)=0.002324543544181, h(28)=7.81460595365e-019, h(29)=0.0005583885039392, h(30)=0.0001281807058205, h(31)=2.823182709039e-005

Simulated frequency response



Algorithm2: Real Time Implementation

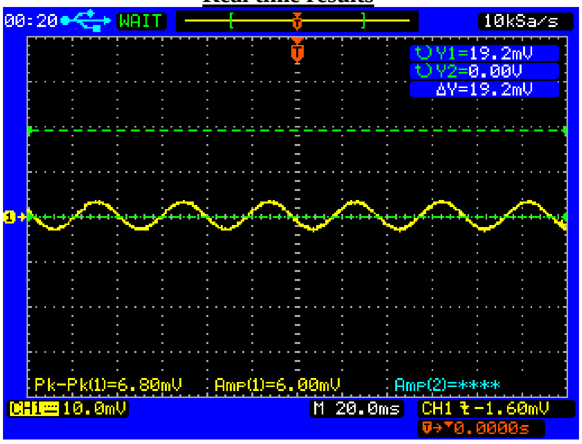
1. Initialize the header files such filter files, kit operating files & input-output line files to include it in the program.
2. Store the analog signal into the buffer registers.
3. Store the filter coefficients into a memory location.
4. Generate the codec for programming at buffering level.
5. Shuffle the buffer according to order of filter.
6. Multiply the filter coefficients with discrete signal values and add it up to the order of the filter.
7. Store the resultant discrete samples and fetch it to the output section.

Note: since used order of the filter as 31 and the input signal is having the voltage level 6.8 mV hence the output after filtering is being attenuated with a factor 1/3 and the signal will disappear in real time application so the need of amplification occurs.

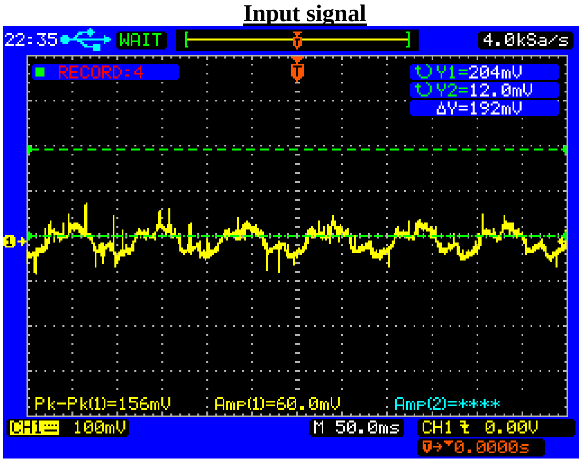
Algorithm3: Real Time Implementation

1. The applied analog signal is digitized through DSP processor.
2. These digitized samples are passed through the multiplication factor of gain of amplifier.
3. The gained signal is factorized with the coefficient of the order of the filter.
4. The amplification factor depends on the requirements and be so chosen that it should not disturbed the shape of output means the noise level should be minimum.
5. These amplified samples stored into the memory location of the processor.
6. Stored samples are multiplied and add with the coefficients of the filter number of times according to the order of filter.

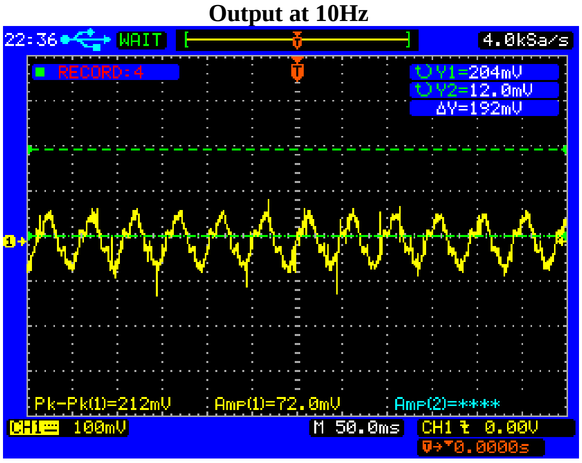
Real time results



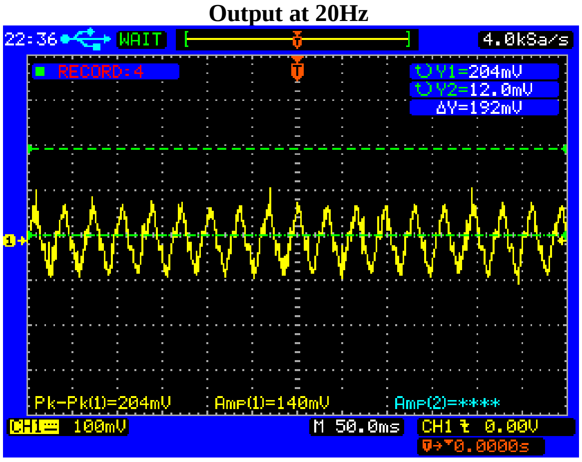
Input signal



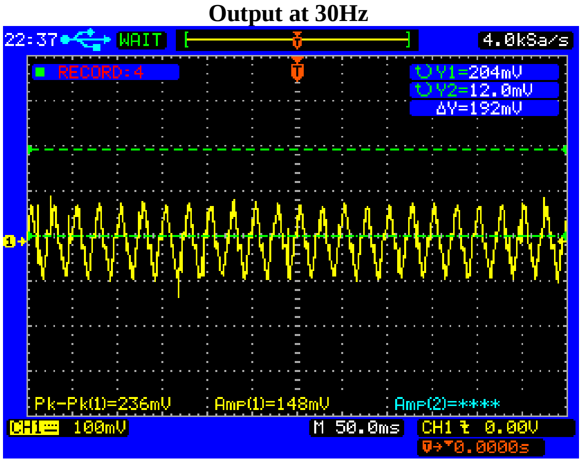
Output at 10Hz



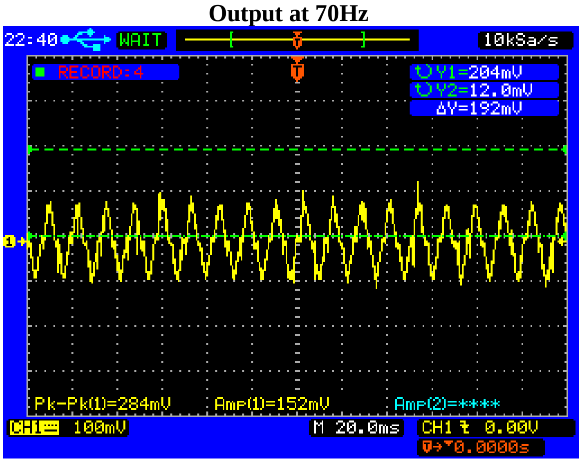
Output at 20Hz



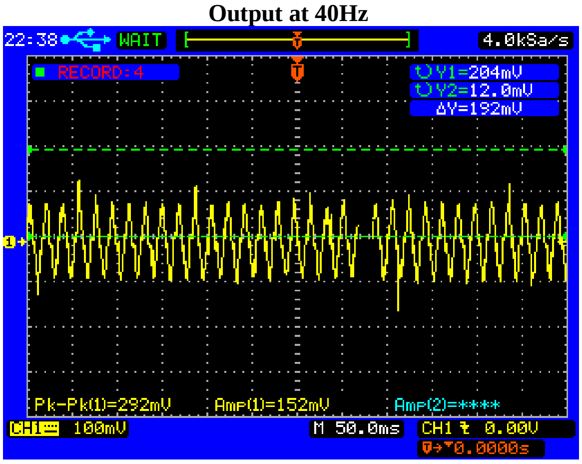
Output at 30Hz



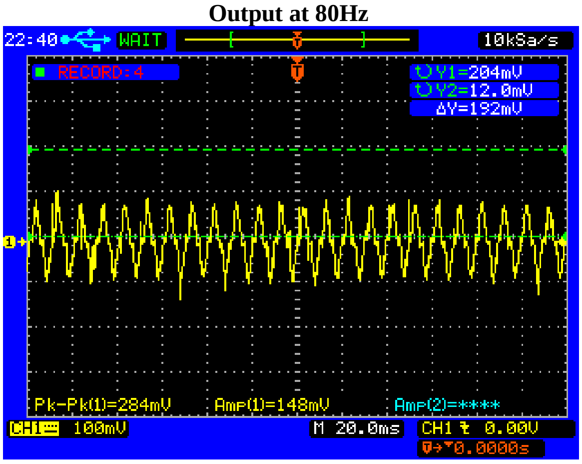
Output at 70Hz



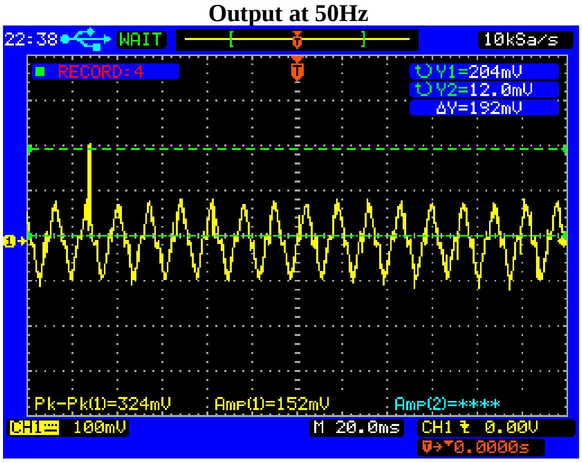
Output at 40Hz



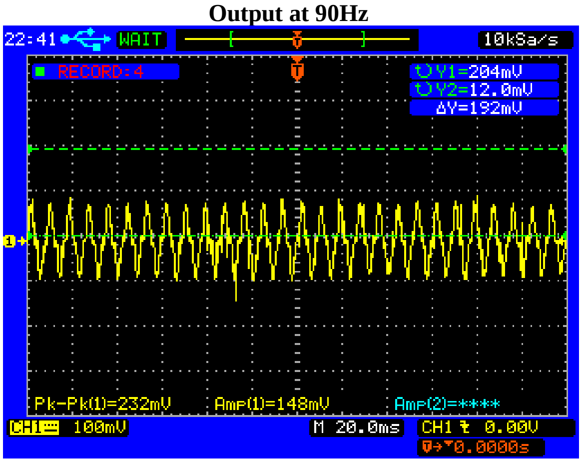
Output at 80Hz



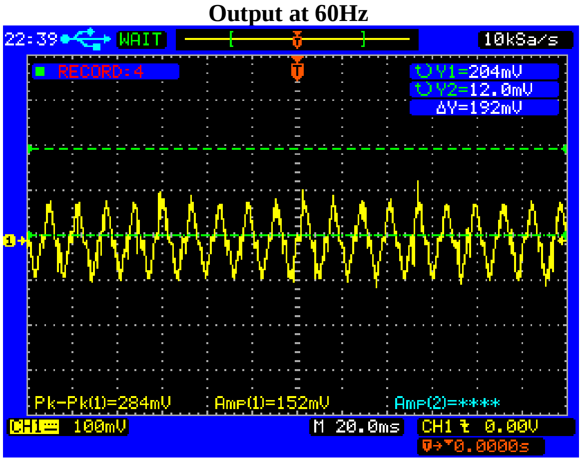
Output at 50Hz



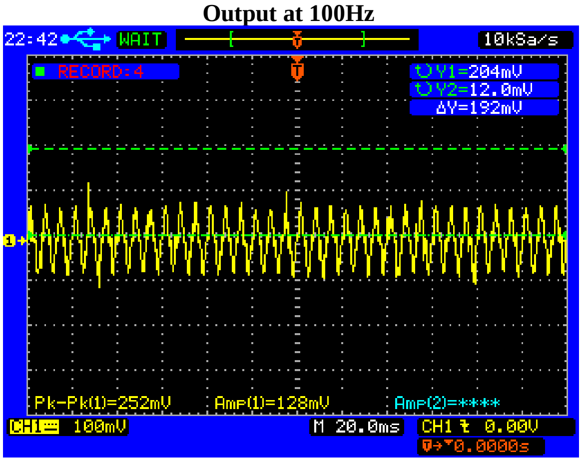
Output at 90Hz



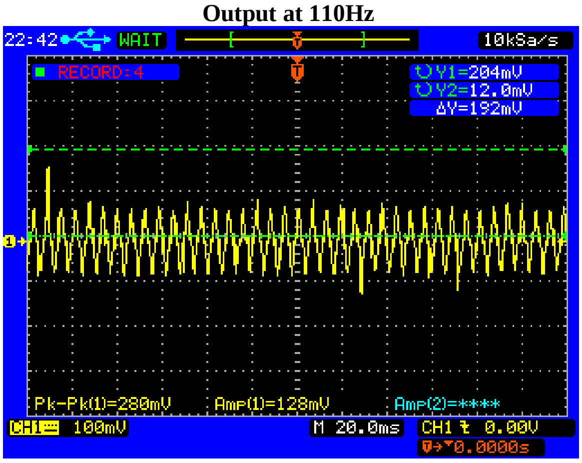
Output at 60Hz



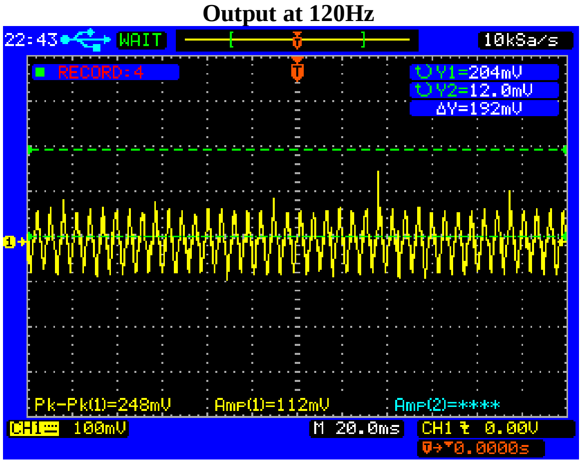
Output at 100Hz



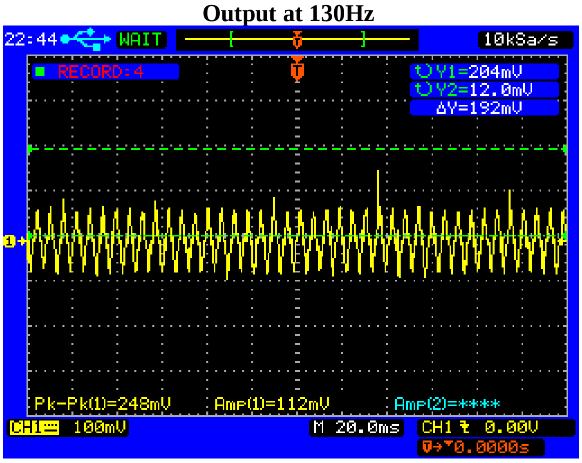
Output at 110Hz



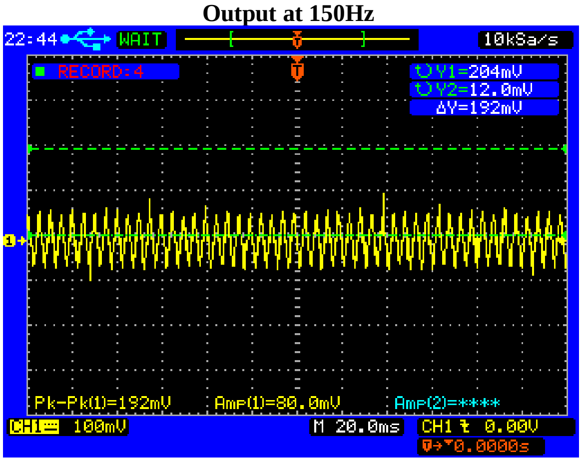
Output at 120Hz



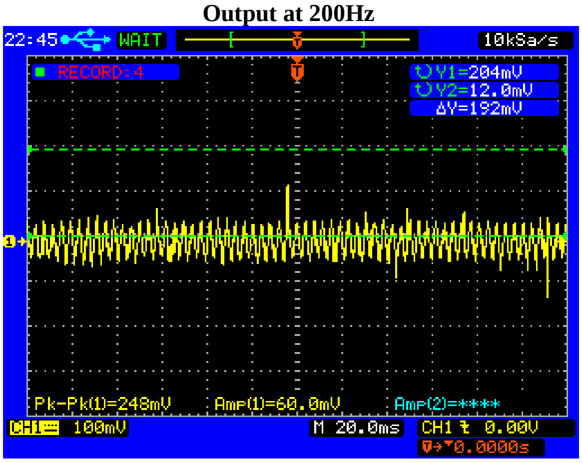
Output at 130Hz



Output at 150Hz

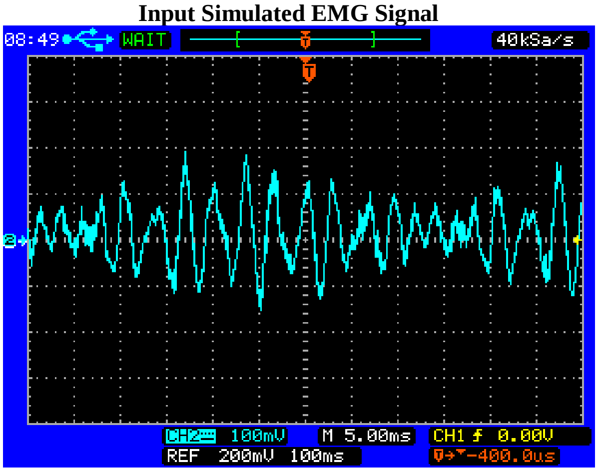


Output at 200Hz

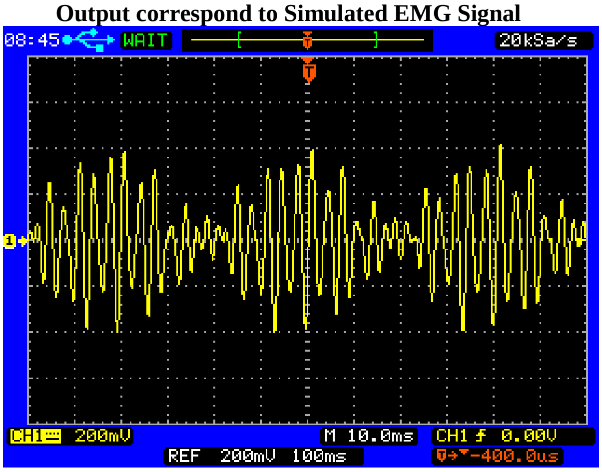


Output at 400Hz

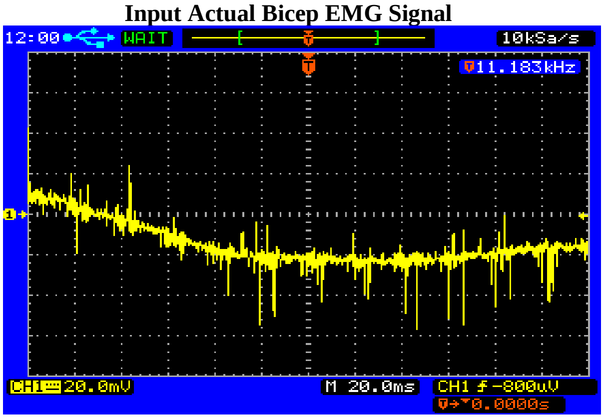
Input Simulated EMG Signal



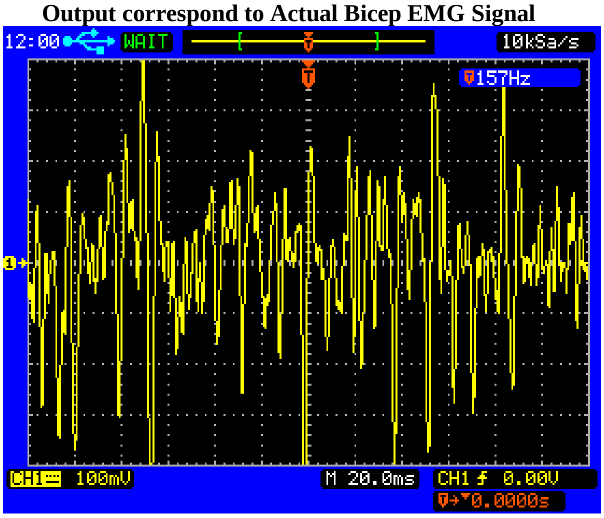
Output correspond to Simulated EMG Signal



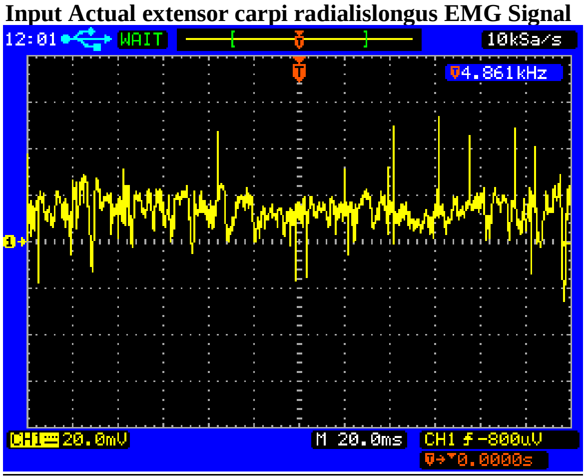
Input Actual Bicep EMG Signal



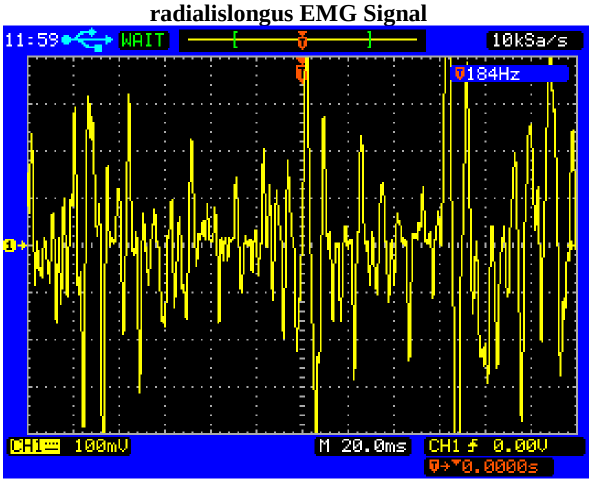
Output correspond to Actual Bicep EMG Signal



Input Actual extensor carpi radialislongus EMG Signal



Output correspond to Actual extensor carpi radialislongus EMG Signal



3. ANALYSIS

Ffrom the comparison of simulated result & real time results, the frequency response is nearly identical & efficient. The proposed digital FIR low pass filter and amplifier can be extensively used in bio-medical engineering field. This can also be used in précised equipment manufacturing for low signal analysis. The proposed method is highly effective in analyzing the EMG signals. It is easy to use & apply in real time processing of any random signal obtained from the bio-medical sensors.

4. FUTURE SCOPE:

Digital signal processing is a rapid growing field; the most of work in signal processing is being digitized for the accessibility and the reliability of digital signal processing. It is good era to work in this field for the research scholars. The key factor in digital signal processing is the filter designing to meet the requirements in various applications.

5. CONCLUSION

In this paper, a digital FIR low pass filter having cutoff frequency 120Hz applicable on very small voltage signal like EMG is designed. The faithful amplification is also been achieved through the digital amplification process. From result studies, the proposed method is highly effective & efficient.

6. REFERENCES:

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