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# Study for Selection of Optimal Filtering Method for Vibration Signals

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## ABSTRACT

Vibration is the motion of a rigid or elastic body about an equilibrium point, occurring nearly in all physical bodies and generally is unwanted beyond a specified threshold value for different applications. Elimination of such spurious signals, in signal processing, can be done using functional blocks called filters. In the present paper, analysis of the response of acquired vibration signal upon passing through five different types of filters and other virtual instruments (VI's) available on LabVIEW has been discussed. Acquisition process here, has been carried out using PCB Piezotronics accelerometer (Model No: 352C66) and 24-bit dynamic signal acquisition hardware DAQ-CARD (NI PCI 4472)) while filtering has been done in LabVIEW environment by synthesis on a VI which is a prototype of the actual set-up of a sound and vibration analyzer. An appropriate choice among the filters has been made such that it gives an effective trade-off between time and frequency domain responses.

## Keywords

NI LabVIEW; Accelerometer; PCI 4472; Virtual Instruments (VI);

## 1. INTRODUCTION

Vibratory motion accompanying signal of importance generally needs to be eliminated for protection of delicate instruments and hardware of an application. Samuel and Pines [1] described one such application where enhancement in changes in signals that were produced due to damaged components of various flight critical components in helicopters was aimed at. Filters can decimate these undesirable vibration signals and at the same time can prevent any undesired delay in system performance upon sound selection. Since analog filters have more ripples in passband compared to a digital counterpart, hence we have preferably used digital filter here [2]. IIR filtering has been used to achieve desired sharpness in the response [3]. In this experiment that follows, the NI LabVIEW 8.1 software has been used to prepare a prototype of the filter set-up for filtering of vibration signals acquired using an accelerometer. Filtering action is performed on the acquired vibration signals using 5 different low pass filters namely: (i) Butterworth, (ii) Chebyshev (iii) inverse-Chebyshev (iv)Elliptic (v) Bessel's. A trade-off between responses in time and frequency domain was encountered and an appropriate choice was made for filtering signals in this range of frequency.

## 2. THEORY

Filters are functional blocks used for elimination of spurious signals from useful ones. In the experiment done, five basic digital IIR filters have been used for the purpose of filtering. Considerations of the time and frequency domain responses are traded off against filter complexity and hence filter cost too.

The formulae for frequency response of each type of filter used are enlisted below [5, 9, 10]:

(1) Butterworth Filter: $|H(\omega)| = \sqrt{\frac{1}{1+\omega^{2n}}}$ (where, ω is frequency and n is filter order.) (1)

(2) Chebyshev Filter: $|H(\omega)| = 1 + \sqrt{\varepsilon^2 T^2 (\frac{\omega}{\omega 0})}$

(where, ϵ is the ripple factor, $\omega_0$ is the cutoff frequency and T is a Chebyshev polynomial of the $n^{th}$ order.) (2)

(3) Inverse - Chebyshev Filter: $|H(\omega)| = \frac{\prod\left(1-\frac{s}{zk}\right)}{\prod\left(1-\frac{s}{sk}\right)}$

(Where $z_k$ and $s_k$ are zeros and poles respectively.) (3)

(4) Elliptic filter: $|H(\omega)| = \frac{1}{\varepsilon^2 \mathrm{Rn}^2(\zeta,\frac{\omega}{\omega 0})}$ (where, $R_n$ is the *n*th-order elliptic rational function, $\omega_0$ is the cutoff frequency, ϵ is the ripple factor and ζ is the selectivity factor) (4)

(5) Bessel's filter: $|H(\omega)| = \frac{\theta(0)}{\theta(\frac{s}{\omega 0})}$ (where, θ(s)= ∑ ($a_k$*$s_k$) ,k=0 to n) (5)

Predefined functions for changing filter type and parameters are available In LabVIEW [5]. Butterworth filter has no ripples in the passband or stopband [6]. Also they exhibit fairly good amplitude and transient behavior.

Chebyshev filters show response that has ripples in the pass band but a monotonic stop band [7]. This filter improves on the amplitude response at the expense of transient behavior. It has smaller transition region than the same order Butterworth filter.

The inverse Chebyshev has better pass band performance than even the Butterworth filter. In the transition band, it has the steepest roll-off.

Elliptic filters have a shorter transition region than the Chebyshev filter because they allow ripples in both the stop band and pass band. These filters give better frequency discrimination however a degraded transient response.

Bessel's filters are optimized to obtain a better transient response in passband due to linear phase (constant delay) at the expense of frequency response.

Also, odd order filters have an attenuation band that extends from 0 dB to the ripple value while even order filters have a gain equal to the pass band ripple. This fact makes an even order filter a preferable choice over odd order filter.

If gain roll off occurs at higher frequencies it results in excessive phase shift which causes oscillations in the frequency response. This response has been observed in the following experiment too. Referring to Figure 1, increase in oscillation due to increase in cut-off frequency can be noticed.

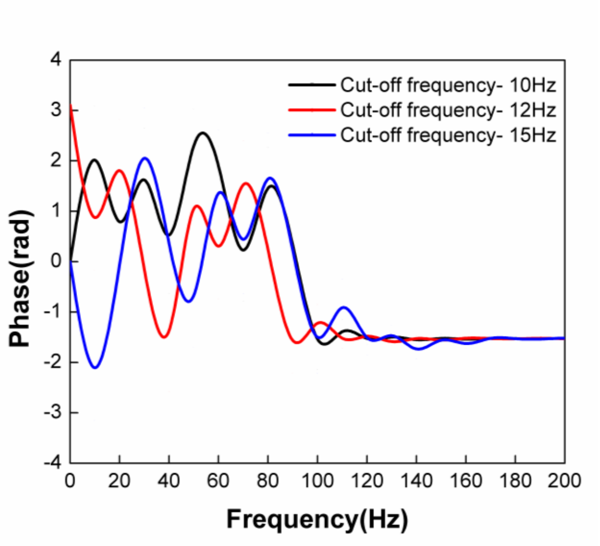


**Fig 1: Superposed graphs of frequency (phase) response of Butterworth filter with increasing cut-off frequency**

## 3. EXPERIMENT

Sample vibration signals, of 25Hz, from a constant frequency vibrating source were acquired using a PCB Piezotronics accelerometer 352C66 SN 100485 [8]. The acquired signal was then converted to a form usable in the LabVIEW Environment, using National Instruments PCI-4472 DAQ which is *a* 24-Bit, 102.4 kS/s, 8- and 4- Channel Dynamic signal acquisition device. Further using the DAQ Assistant Express VI and Write to Measurement File function the signal was written in the '.lvm' format for analysis purpose. A second VI was used for the purpose of filtering, display and writing of data to Excel Sheet for analysis of output waveform. This VI comprised major functions such as Read From Measurement File, Align and Resample, Filter, FFT-based Spectral measurement, Waveform Graphs and Write to Measurement File. Cut-off frequency of the signal is taken as 9 Hz. A mixture of original and noise signal was passed through the Frequency Analyzer VI for this inference. These VI's have been shown in Figures 3 and 4. Filter parameters namely, filter type, filter order and cutoff frequencies were changed and resulting data were stored in an Excel file.

## 4. RESULTS AND DISCUSSION

Figure 2 shows the original vibration signal in time domain. Analysis of filtering of this signal was done in three sections by varying one parameter of the filter in each section.

### 4.1 Keeping order and cut-off frequency constant and varying the filter type:

Figure 5 shows the frequency (magnitude) response of different filters of 3$^{rd}$ order. Response of the Butterworth filter, Chebyshev filter and the Bessel's filter is nearly the same in the stop band. The Elliptic, Chebyshev and inverse-Chebyshev filters have an increasing undesired non-linearity in the stop band. From figure 6, the time response of filters it is seen that the Butterworth filter approaches linear behavior earlier than other filters while the inverse Chebyshev filter approaches linearity in the last-place. Butterworth and the Bessel's filters give best response.

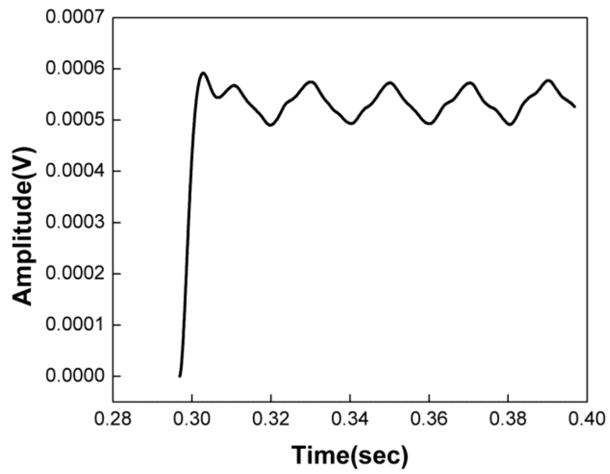


**Fig 2: Original signal from the vibration meter in time domain**

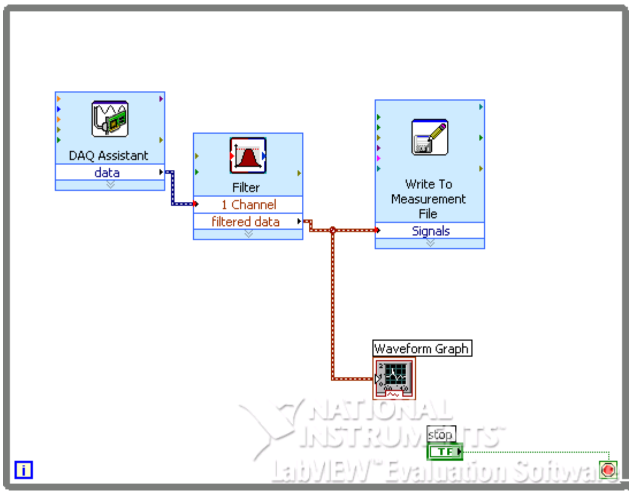


**Fig 3: VI for acquisition of vibration signal**

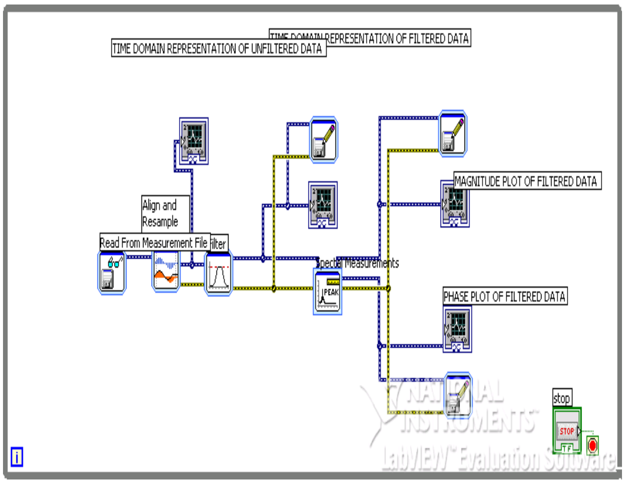


**Fig 4: VI for filtering of acquired vibration signal**

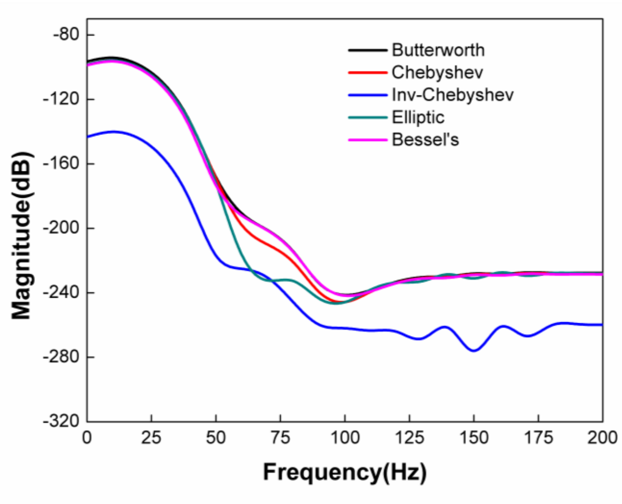


**Fig 5: Superposed graph of the response of 5 filters of 3rd order in frequency domain**

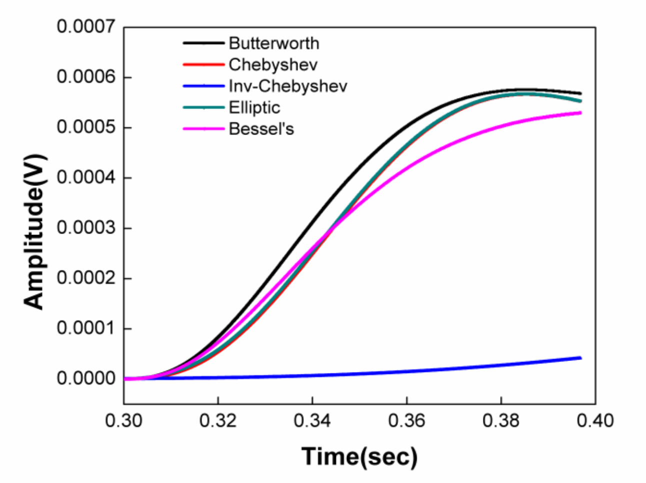


**Fig 6: Superposed graph of the response of 5 filters of 3rd order in time domain**

## 4.2 Keeping filter type and cut-off frequency constant and varying filter order:

Figure 7 shows the frequency response of Butterworth filter on increasing order. Sharper roll-off is noticed with increase of the filter order. Also maximum gain is comprised within pass-bands of higher order filter. Even order filter are seen to give maximum gain in pass band. However, as seen in Figure 8 time response degrades with increase in filter order. A similar inference can also be made from plots with the data from the four other filters, upon increasing their order.

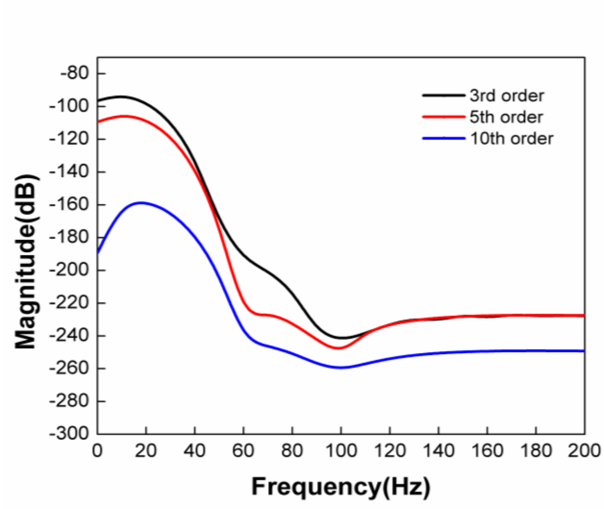


**Fig 7: Superposed graph of frequency response of Butterworth Filter of 3rd, 5th and 10th order**

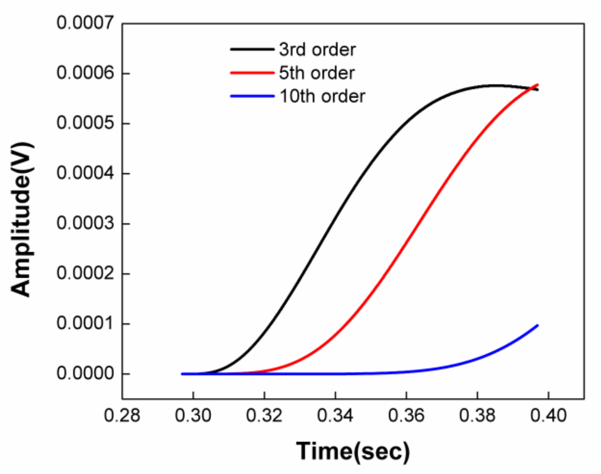


**Fig 8: Superposed graph of time response of Butterworth Filter of 3rd, 5th and 10th order**

## 4.3 Keeping filter type and order constant and varying filter cut-off frequency:

From figure 9 it can be seen that variation in cut-off frequency does not affect the frequency response in a strong manner. Figure 10 shows the effect of increasing cut-off frequency on time response. With increase in cutoff frequency beyond 9Hz, the time response degrades or becomes slower.

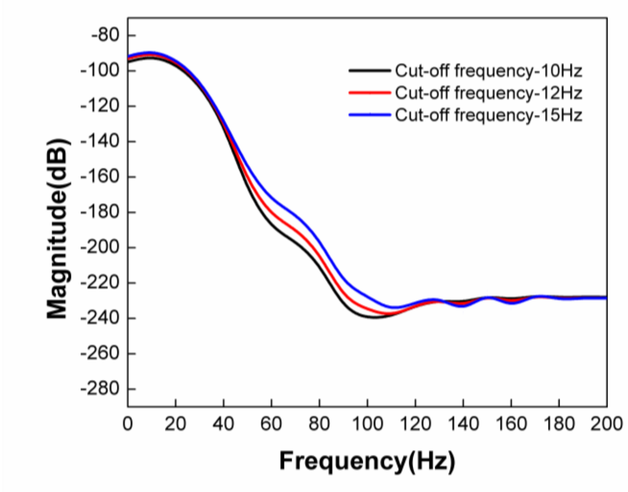


**Fig 9: Superposed graphs of frequency response of Butterworth Filter of 3rd order with increasing cut-off frequency.**

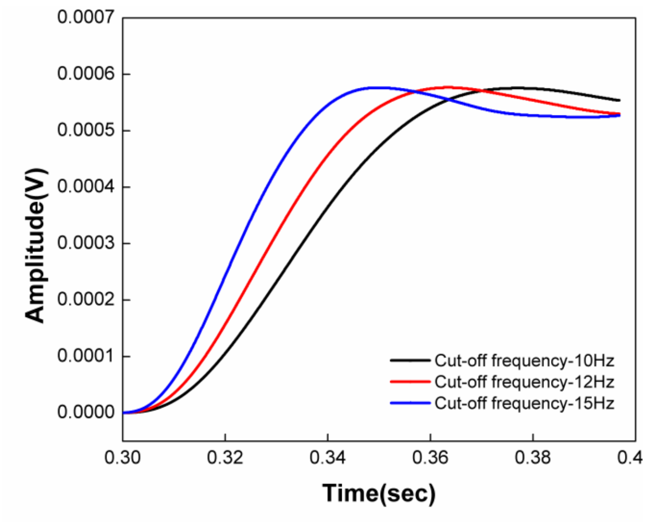


**Fig 10: Superposed graphs of time response of Butterworth Filter of 3rd order with increasing cut-off frequency.**

## 5. CONCLUSION

From the experiment done we infer that Bessel type filters give excellent transient behavior, but less than ideal frequency discrimination. Elliptical filters give better frequency discrimination, but degraded transient response. Upon similar study as 4.2 Elliptic filters give improved stop band attenuation on increasing order, like Butterworth filter. Thus it is also inferred that increasing order degrades time response and enhances frequency response of filters in general.

Also theoretical and practical equivalence can be seen from Figure 4 where odd order filters have an attenuation band that extends from 0 dB to the ripple value however even order filters have a gain equal to the pass band ripple only. Considering the tradeoff between the frequency and time response, we infer that Butterworth filters have fairly good amplitude and transient behavior and they can be a fair choice for filtering vibration signals.

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