

# A comparative study of digital FIR and IIR band- pass filter

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**Abstract:** Digital filters are an integral part of signal processing which help us to remove the frequencies which are of less importance or considered unwanted, also termed as noise sometimes. Generally, these filters are categorized into two types, finite and infinite impulse response filters. In this paper we try to analyze the parameters such as order, computational flexibility and precision control as parameters to compare between FIR and IIR Band pass filters. Infinite impulse response have been said to work efficiently even with fewer design parameters, less memory requirements, and lower computational complexity compared to FIR filters. This paper thereby tries to draw a fine line by comparing the spectrums and the frequency responses available for the given specifications.

**Key Words:** Digital Filter, FIR, IIR, Band-pass filter, Order, precision, Spectrum, Frequency response

## 1. INTRODUCTION

A digital filter is considered to be a system that can perform mathematical operations upon a sampled signal otherwise a discrete signal. This is to enhance or reduce certain characteristics of that signal. The digital filters are used for two important functions that is the separation of the signals that have been a result of combination of two or more signals, or restore a signal that has been distorted. Though analog filters can be designed for the same purpose, digital filters provide superior results. In the practical implementation generally, the filter's input and output signals are in the time domain.

Digital filters are classified into, Finite Impulse Response (FIR) Filters and Infinite Impulse Response (IIR) filters. Again these filters can be implemented as low pass, high pass, band-pass, and band-stop filters, using windowing and equiripple methods. Considering these linear filters, we mainly consider these responses, impulse response, step response and a frequency response.

These responses, have in them the complete information about the filter, but each are showcased differently. They are considered to be important, because they will help in visualizing about the working of filters under different

instances. In case any one of the response is specified, which will make the other two responses fixed then, they are directly calculated. The analysis can be easily compared using the various responses that we can obtain by implementing these filters. The step response depicts how the information is represented in the time domain, when the information is modified by a system whereas a frequency response shows how this information is being depicted when the domain is in terms of frequency. The main reason why it is of importance, is because of its criticality in the design of filter, where it is not possible for a filter to optimize for both applications. The performance in either of the domains has to be compromised. A good performance in time domain requires a compromise in the parameters regarding frequency domain, and vice versa.

## 1.1 Finite Impulse Response (FIR) filter

FIR filter generally has an impulse response of finite period. This makes it easy to settle at zero in a given period. Generally FIR filters are used to generate the any frequency response. These filters are made of multipliers, adders and set of delays to achieve required results. For a filter of length L working on delays applied on the input, we can generate coefficients  $h(k)$ , used in the multiplication. Thereby the result contains total sum of delayed samples which are multiplied by the generated coefficients. FIR filter can be represented by this difference equation

$$y[n] = \sum_{k=0}^{L-1} h[k].x[n - k]$$

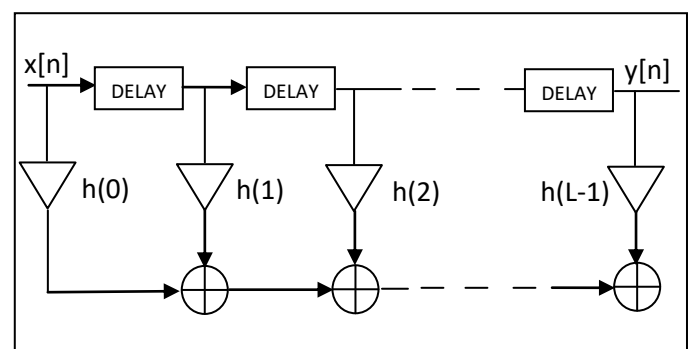


Fig -1: FIR filter structure

## 1.2 Infinite Impulse Response (IIR) filter

They are called Infinite response filters due to the presence of the feedback mechanism. Though the filter has been designed in such a mechanism wherein it shows better performance than the existing FIR filter of the same specification, it cannot be used in the filters needing linearity in phase. IIR filters depict the property of linearity and time invariance. Considering M and N as feed forward and feedback order,  $b_p$  and  $a_q$  are the feed forward and feedback coefficients, then its difference equation is given by

$$y[n] = \frac{1}{a_0} \left( \sum_{p=0}^M b_p x[n-p] - \sum_{q=0}^N a_q y[n-q] \right)$$

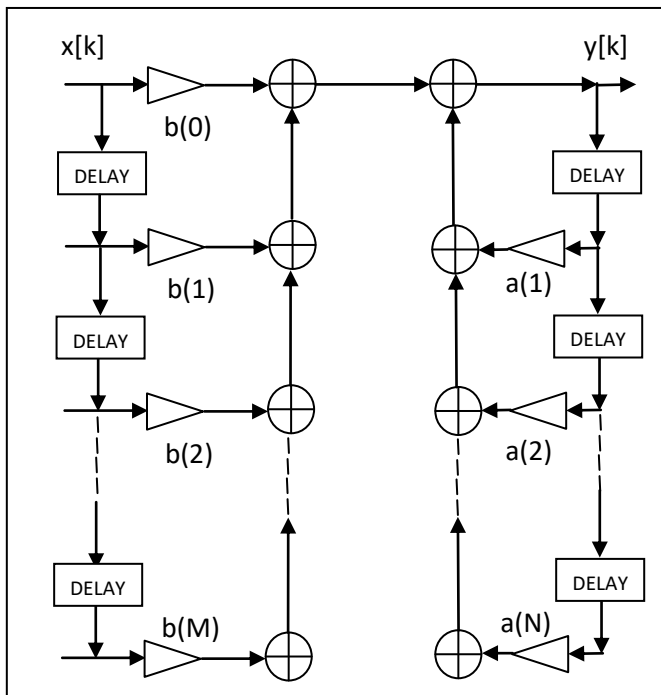


Fig -2: IIR filter structure

## 2. OBJECTIVE

- To obtain the spectrum of a signal obtained by mixing two signals of different frequencies
- Filtering the signal using IIR and FIR Band pass filter, and analysis of few filter parameters
- Obtain frequency spectrum of the filter signals and analyze the results

## 3. BAND PASS FILTER

Band pass filters are generally used in the applications where filtering out a particular frequency is required, while allowing the other frequency to be transmitted. If  $f_1$  and  $f_2$  are the cut off frequencies, then the center frequency is considered at  $f_0$ . The difference of the two cut off frequencies, gives us the filter bandwidth, and the range of frequencies existing between these two cut offs, are called at the filter pass band. The choice of the filter bandwidth is important to optimize the number of frequencies that can be transmitted in the filter pass band range. Since this filter allows a particular frequency to pass, it is also called as band select filter.

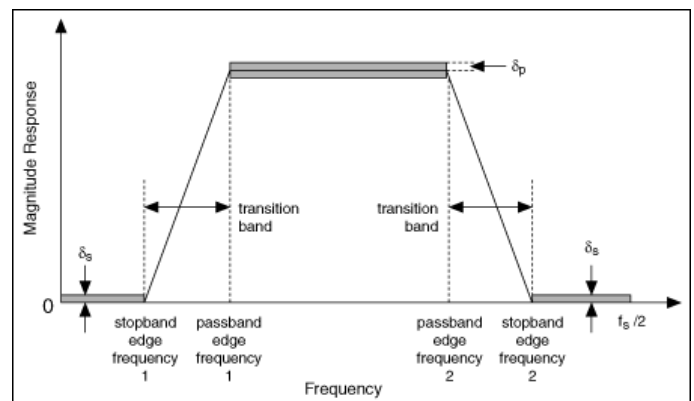


Fig -3: Magnitude v/s frequency response of a Band-pass filter

### 3.1. FIXED POINT APPROXIMATION

Fixed point arithmetic is a peculiar format that is used in the implementation of digital filters pertaining to the applications in the areas of DSP processors and in VLSI domain. One of the methods of usage is obtaining the two's complement of the given fixed point format. Considering N bit number, with its MSB representing the sign bit, the lower N-1 bits will then represent magnitude. This N-bit number is used to represent signed numbers from the range from  $-2^{N-1}$  to  $2^{N-1}-1$ . While considering two's complement logic, the negative representation of a binary number is formed by finding the one's complement of the number and then add one to the LSB. Generally in IIR filter due to the feedback mechanism, we come across instability certain time. Main advantage of controlling the fixed point approximation is

- Size and power consumption when we need to implement in hardware
- Memory usage and speed
- Cost of implementation

### 4. METHODOLOGY

Initially two sinusoidal signals with a carrier frequency of  $f_{c1} = 20$  KHz and  $f_{c2} = 40$  KHz is considered, sampled at a frequency of 80MHz. The two signals are multiplied with each other, via a product modulator. In the spectrum, we notably observe peaks at  $(f_{c2} + f_{c1})$  and  $(f_{c2} - f_{c1})$ . One of the frequency component is removed by passing the obtained signal via a FIR and IIR band pass filter, for comparison of the parameters.

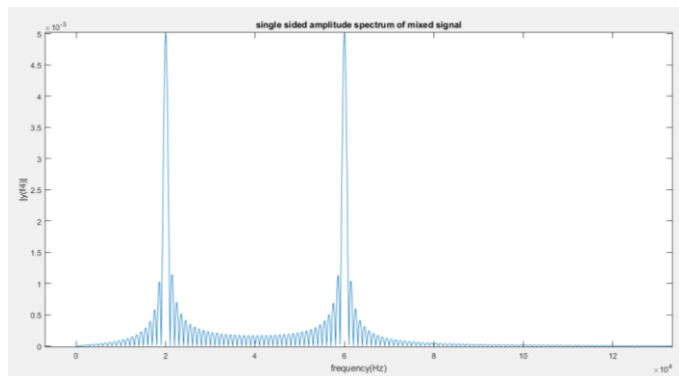


Fig -4: Spectrum of mixed signal

### 5. COMPARISON OF FIR AND IIR FILTERS

On the basis of the number of coefficients required, the order of the filter and the sampling frequency at which the filter works, for a given IIR and FIR band pass filter following comparison can be made

- The requisite for an IIR filter is a choice of lower order compared to the FIR specifications for the same parameters
- IIR can attain the same filtering characteristic easily by less memory consumption and computations than a similar FIR filter
- The Necessity of the side lobes required are very less in the stop band of IIR filter

Table -1: Band pass filter specifications

Filter Specifications Initial			
FIR		IIR	
ORDER	1972	ORDER	10
METHOD	Equiripple	METHOD	Butterworth
CUT OFF FREQUENCIES	59KHz,63KHz	CUT OFF FREQUENCIES	59KHz,63KHz
SAMPLING FREQUENCY	40MHz	SAMPLING FREQUENCY	40MHz
ATTENUATION	60dB	ATTENUATION	60dB

Table -2: Band pass filter specifications

Filter Specifications compromised			
FIR		IIR	
ORDER	18	ORDER	7
METHOD	Equiripple	METHOD	Butterworth
CUT OFF FREQUENCIES	59KHz,63KHz	CUT OFF FREQUENCIES	59KHz,63KHz
SAMPLING FREQUENCY	40MHz	SAMPLING FREQUENCY	40MHz
ATTENUATION	60dB	ATTENUATION	60dB

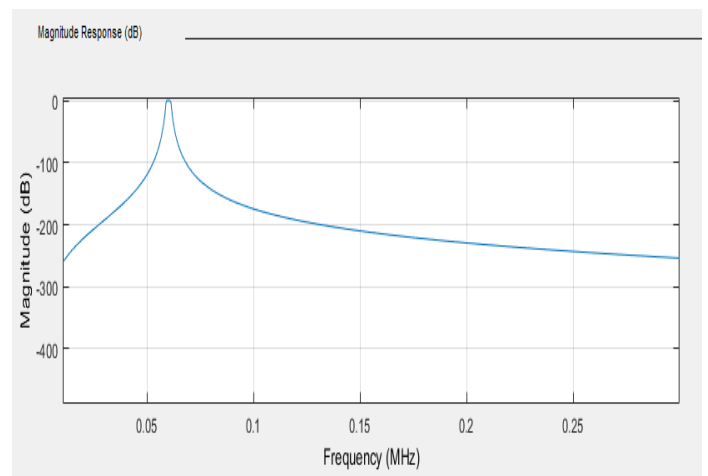


Fig -5: Frequency response of IIR Band pass Filter

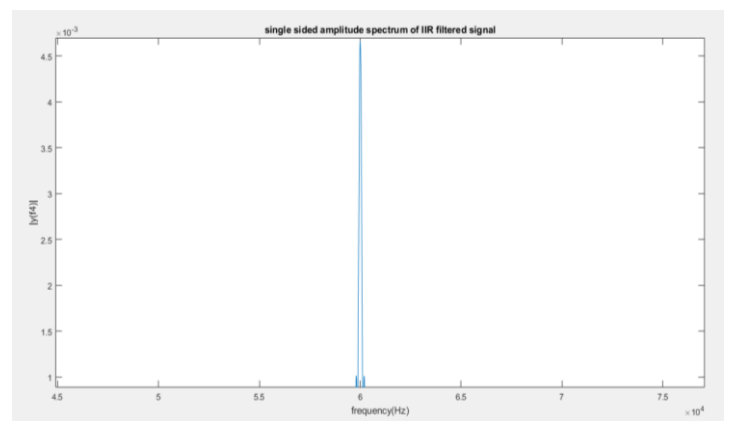


Fig -6: Spectrum of IIR filtered Band-pass signal

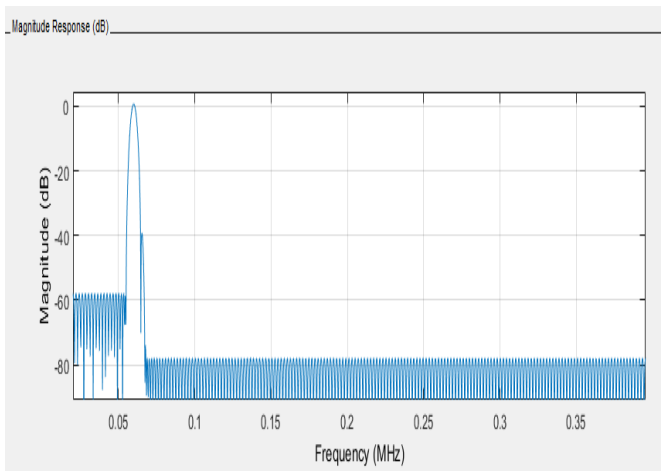


Fig -7: Frequency response of FIR Band pass Filter

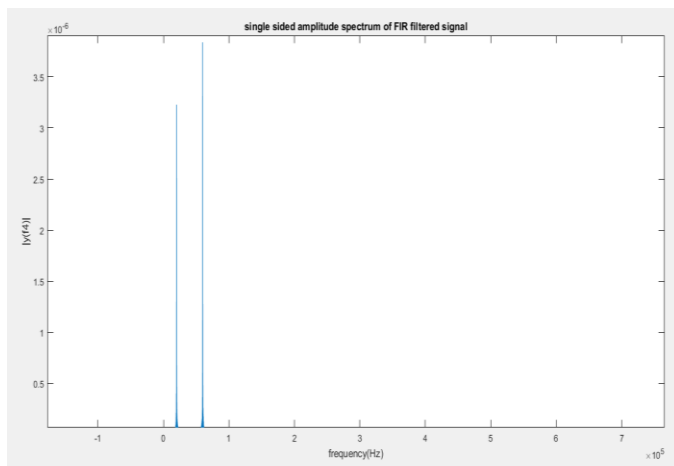


Fig -8: Spectrum of FIR filtered band pass signal

## 6. CONCLUSION

This paper mainly deals with the analysis of IIR And FIR Digital Band pass Filter parameters such as order, computational complexity and flat pass band in frequency response . Each technique comes with its own benefits and flaws. It is observed that while filtering based on IIR method, an improved signal with minimal noise can be obtained. We also notice that fine precision in the finite bit has its effect on the filter. Also a keen difference can be seen that the IIR filters are susceptible to errors with the precision being considered, thereby they must be used as a cascaded structure. We also observe that at lower orders, FIR Band-pass filter cannot completely filter out one of the frequency component, the actual order that needs to be chosen is practically not feasible to meet design considerations. For the same specifications, we observe IIR filters handle the response very well keeping the orders as low as possible.

## REFERENCES

- [1] Ifeachor, E., and Jervis, B., Digital Signal Processing: A Practical Approach, Addison-Wesley, and 1995 M. Young, The Technical Writer's Handbook. Mill Valley, CA: University Science, 1989.
- [2] Lyons, R., Understanding Digital Signal Processing, Addison-Wesley, '97.K. Elissa, "Title of paper if known," unpublished.
- [3] Gulati K., Gupta M., and Rajni, Noise Detection In IIR Digital Filter Using MATLAB, 2012 Second International Conference on Advanced Computing & Communication Technologies.
- [4] "Survey on IIR and FIR Digital Filter " ,Tej Singh , Prof. Anshuj Jain , Prof. Bharti Chourasia
- [5] Parameshwaran R1, HariHaran K1, Dr. Vaithiyanathan R3 and Aiswariya R2 Prakash B2 "Pipelined Approach for Second Order Recursive Filter" International Journal of Applied Engineering Research, ISSN 0973-4562, Vol. 8, No. 20 (2013)