PERFORMANCE COMPARISON FOR EFFICIENT DIGITAL DECIMATOR FILTER STRUCTURES

Pankaj R. Ambilduke[1], Manish Kumar Gurjar[2]

[1]*M-Tech Final year Student, Department of Electronics and Communication Engineering, TIT College Bhopal, MP, India*

[2]*Assistance Professor, Department of Electronics and Communication Engineering, TIT College Bhopal, MP, India*

Abstract

In many practical applications of DSP, there is a problem of changing the sampling rate of a signal, either increasing it or decreasing it by some amount. For example, Telecommunication system transmits and receives different types of signals (e.g. fax, speech, video, etc), so there is a requirement to process the various signals at the different rates with corresponding bandwidth of the signals. The process of converting a signal from a given rate to a different rate is called as "sampling rate conversion" and the systems that employ multiple sampling rates in the processing of digital signals are called as "Multirate DSP systems". ". Digital audio engineering is an area that has benefited significantly from Multirate techniques. For example, they are used in the compact disc player to simplify the D/A conversion processes, while at the same time maintaining the quality of the reproduced sound. This project is based on to design and development of highly efficient Multirate digital filter structures for filtration of noisy signal in which the high sampling rate is decreased to the desired lower sampling rate. Digital filters such as IIR filter, FIR filter and CIC filter is designed and taking their performance in case of magnitude response, step response, impulse response, pole-zero plot, filter coefficients, storage requirements, hardware requirements, number of stages and simulated waveforms for same input specifications. Finally compare them and discuss the advantages and limitations of these filters. These filter structures designed in the Simulink model in Matlab 2012a environment.

Keywords: *Digital Filter, CIC Filter, FIR Filter, IIR Filter, Sampling Rate, Interpolation Filter, Impulse Response, Frequency Response, Decimator Filter.*

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1. INTRODUCTION

1.1 Overview

In Digital Signal Processing (DSP) system digital filter is vital part. Filter is the basic component of all signal processing and telecommunication systems. The primary functions of a filter are one or more of the followings:
(a) To confine a signal into a prescribed frequency band for example as in anti-aliasing filter or a TV channel selector,
(b) To divide a signal into multiple sub-band signals for sub-band signal processing, for example in music coding,
(c) To modify frequencies in channel of signal in voice graphic equalizers
(d) To construct I/p and o/p connection of a system such as a mobile communication spectrum, audio production, musical instruments and telephone line echo.

1.2 FIR Filter

The filter having impulse response of finite duration is called as FIR filter in digital signal processing. This is inverse to infinite impulse response (IIR) filter, whose feedback path may continue to respond indefinitely. The impulse response of N^{th} order isolated time FIR filter remain as $(N+1)^{th}$ (all element will be non zero) before it goes to zero. FIR filters can be distinct time or the continuous time, and digital or analog.

1.2.1 What is "FIR"?

"Finite Impulse Response" stands for "FIR". If we put inclination in which a single sample "1" will be followed by many "0" means, starting will be from "1" and tail will be "0".

1.2.2 Why Is The Impulse Response "Finite"?

In the same case, because of no feedback the impulse response will remain as it is. No feedback confirms that the inclination response will finite. Therefore no feedback can be assumed as finite impulse response.

The FIR filter will remain as it is although after giving finite impulse response even after applying feedback. Consider the example of the moving average filter, in which the N^{th} sample is subtracted (fed back) from new incoming samples each time. After N^{th} sample of impulse the o/p will be zero as this filter has a finite impulse response even if it uses feedback.

1.2.3 How do I Pronounce "FIR"?

Somebody utter the letters F-I-R; where people may enunciate as a kind of tree, use favor the tree. Pronounce as a type of tree. We prefer the tree. The discrepancy is come whether you say about *an* F-I-R or *a* FIR filter.

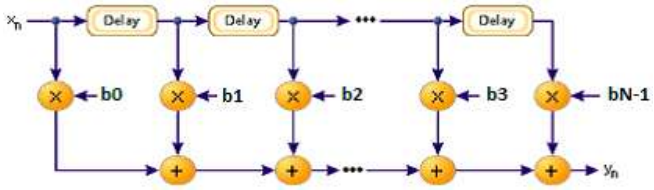

Fig 1.2.1: Basic formation of FIR filter

1.3 IIR Filter

The digital filter which provides infinite impulse response is called as IIR filter. The filter which is having feedback and also known as recursive digital filter such kind of filter is called as FIR filter. This is the reason for the betterment of IIR filter than FIR filter. Also frequency response can be takes in consideration.

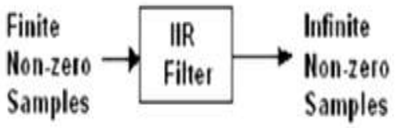

Fig 1.3.1: Simple block diagram of IIR filter

IIR filter uses current input sample value, past input value and output sample values to obtain current output sample value.

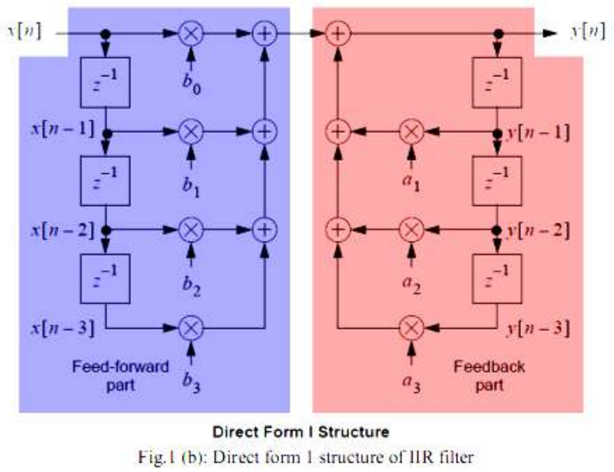

Fig 1.3.2: Direct Form 1 IIR Filter structure

Transfer function of FIR filter will have only zeros, need more memory, while transfer function of IIR filter will have both zeros and poles and will require less memory than FIR. FIR filters are preferred due to non-recursive and its linear phase response. Feedback is not involved in FIR, therefore they are stable. IIR filters are not stable because they are recursive in nature and feedback is also involved in the process of calculating output sample values.

1.4 Objectives

- To design and implement the IIR and FIR filter using MAC units and FDA tool
- To generate the filter coefficients in Matlab workspace and calculate total gain
- To study the CIC filter and its advantages over the FIR filter
- To design and implement the CIC filter for decimation process
- To compare the storage requirement for filter coefficients, frequency responses, step responses, impulse responses, pole/zero plots/ rounded of noise, hardware requirements and simulated waveforms of IIR, FIR and CIC filters
- To find out the most reliable structure among from these designed filters

1.5 Problem Statement

In a many applications of digital signal processing, there is always need of change of sampling rates from lower sampling rate to higher sampling rate or higher sampling rate to lower sampling rate or both at the same time. In this case, single rate DSP system fails. Thus most of the practical applications, Multirate signal processing systems are used. Decimation and Interpolation are two main functions of Multirate signal processing. In case of digital filters, to implement IIR filter, more numbers of hardware are required while in FIR filter, less hardwares are required as compare to IIR filter. But by using CIC filter, very less hardwares are required as compared to both IIR and FIR filter.

1.6 Motivation

Why would we want to change the sample rate in a filter? There are two reasons. The first is performance. The second is cost. Multirate systems generally perform a processing task with improved performance characteristics while simultaneously offering that performance at significantly lower cost than traditional approaches. *Multirate filters* are digital filters that operate with one more sample rate change embedded in the signal processing architecture. Occasionally, the use of changing of sample rate in filtering is the natural process of the signal processing chain. In other cases, the sample rate change is imposed to access the cost advantages related to multirate processing.

1.7 CIC Filter

This filter is used to convert data faster. The application of narrowband is extraction of narrow band from wideband sources, and narrowband construction of the wideband signal is very important. For this function we require two basic signal procedures 1.Decimation and 2. Interpolation. With digital hardware is becoming faster, there is still the need for efficient solutions.

Effective way of performing the decimation and interpolation is introduced by Hogenauer E.B. in the year 1981. The filter which handles arbiter and great rate changes are called as cascaded integrator comb filter, which are suitable for hardware execution. Integrator and comb are the two fundamental building block of CIC filter. Single pole IIR filter with a union response coefficient are simply integrator.

2. LITERATURE SURVEY

The literature survey of this project is taking from various papers which are based on design of digital filters. They all used various methods, techniques and different softwares for designing and development of FIR, IIR and CIC filter structures. The major aim behind design these techniques were to eliminate the noise from input signal.

In this paper all the work is discussed on performance of FIR band pass filter on FPGAs in which they remove the high frequency noise from respective audio communication signal. Although they also provided the Noise Extraction System which is very useful for noise elimination and this system is designed in Xilinx System Generator. [1]

During this paper author presented the some methods for reduction and controlling the dynamic power utilization of digital FIR filter. These various methods are included with low power series multiplier and also serial adder, breakdown transformation in linear phase architecture, combinational booth multiplier, shift/add multipliers and provide to FIR filters to reduced the power consumption and therefore there is reduction in power consumption and due to this glitches are also reduced. [2]

This paper explained the FIR filter which is designed using Matlab Simulink in Xilinx System generator environment and also they performed hardware co-simulation on Xilinx Virtex-4 FPGA kit. At the ending of paper, they gave comparison between the results originate from the software simulations and FPGA via hardware co-simulation. [3]

In this paper they designed and implemented and also simulate 1-D and 2-D FIR filters which is presented in the MATLAB and Simulink tool. The overall evaluation of the obtained output waveforms of 2-D FIR filter and 1-D FIR filter is given. [4]

This paper they explained the different orders of FIR filters design structures with the rectangular window with specified cut off frequency and then compared them to find out the efficient order of the FIR filter using dissimilar order in Boxcar windows process which has been confirmed through the MATLAB simulation results. [5]

This paper they are presented the Matlab program which is used to implement a low pass FIR filter structure using changeable window function system and which is based on Hamming window method. Finally the comparisons of low pass FIR filters with different values of alpha are given. [6]

3. PROPOSED METHODOLOGY

Digital filter can be realized by using simulink in matlab environment for this matlab 20012a has been used. Any sort of digital frequency response is obtained by implementing FIR filter structure. FIR filter can be implemented by combination of sequence of delays, multipliers and adders for getting filter output. IIR filter is implemented by using delay elements, multipliers, adders and constant elements. This filter uses feedback system and due to this it provides more clear and clean signal at output side.

Generally multipliers and adder are consumes more power and thus to overcome this problems CIC filter is used.

This CIC filter is designed by using delay elements and adders only. Thus it is also called as multiplier-less filter and hence it consumes very less power. But output of CIC filter is not so accurate and hence it required extra reconstruction filter.

3.1 Filter Design Steps

- **Filter specification:** it will include stating the type of filter ex. Low pass filter, required amplitude or phase response and tolerance, we have set to allow, the sampling frequency and the length to input data.
- **Coefficient calculation:** we find out the coefficient of the transfer function H (z) which will assure the specification given in (1). In critical requirement, in step (1) several factors can influence coefficient calculation method.
- **Realization:** Realization process involves the conversion of transfer functions obtained in (1) into a suitable filter structure or network
- **Analysis of finite word length effects:** Analysis of the result of quantization of the filter coefficient and the input data as well as filtering effect is passed out. operation using fixed word lengths on the filter performance.
- **Implementation:** implementation involves the production of the software code, hardware and performing the actual filtering operation.

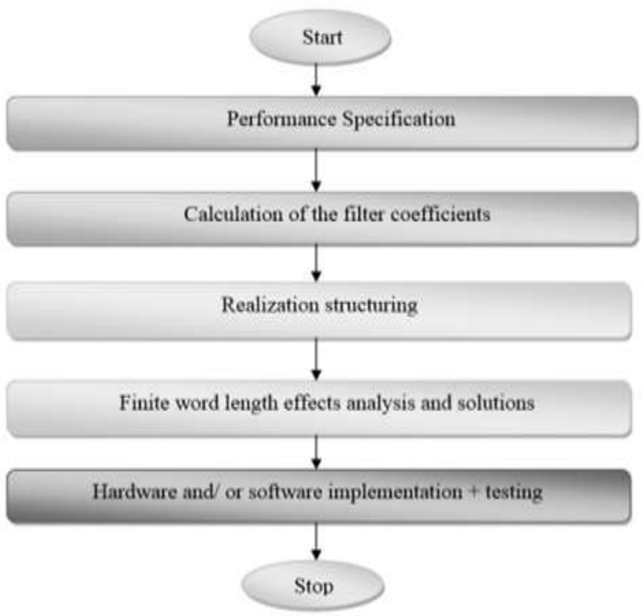

Fig 4.2.1: Flowchart of filter design steps

4. IMPLEMENATION AND RESULT ANALYSYS

The design and implementation of digital filters done by using Matlab Simulink 2012a. In this chapter, design and implementation of FIR filter IIR filter and CIC filters done. The frequency responses, impulse responses, step responses, pole-zero plots, number of filter coefficients, orders of filter and simulated waveforms taken. According to these all and storage requirements and hardware requirements, comparison done.

4.1 Design of FIR Filter

In this design of FIR filter, two Sine wave input signals are used. First input signal is used for as low frequency information signal i.e. clean signal with frequency 10 rad/sec i.e. 1.59 Hz and second is noisy input signal with frequency of 1000 rad/sec i.e. 159.15 Hz. These two input signals are added together by the sum block and thus total input frequency is of 160.74 Hz and this added signal given to the FIR filter block. The sampling frequency is 312.48 Hz. Pass band frequency is 1.59 Hz and stop band frequency is 3 Hz with decimation factor of 4. This FIR filter block is designed with delay element, multipliers, constants and adders. Multiplier can multiply the given signal and constant i.e. filter coefficient.

The output of multiplier is given to the adder. Now, Output of delay signal is given to the next delay and another multiplier at the same time. This multiplier multiply third delayed signal with the constant and the output of this multiplier is given to the adder. Now, the output of first multiplier and second multiplier is added together and then output of adder is given to the next adder. Following figure shows the designed FIR filter structure with order 15.

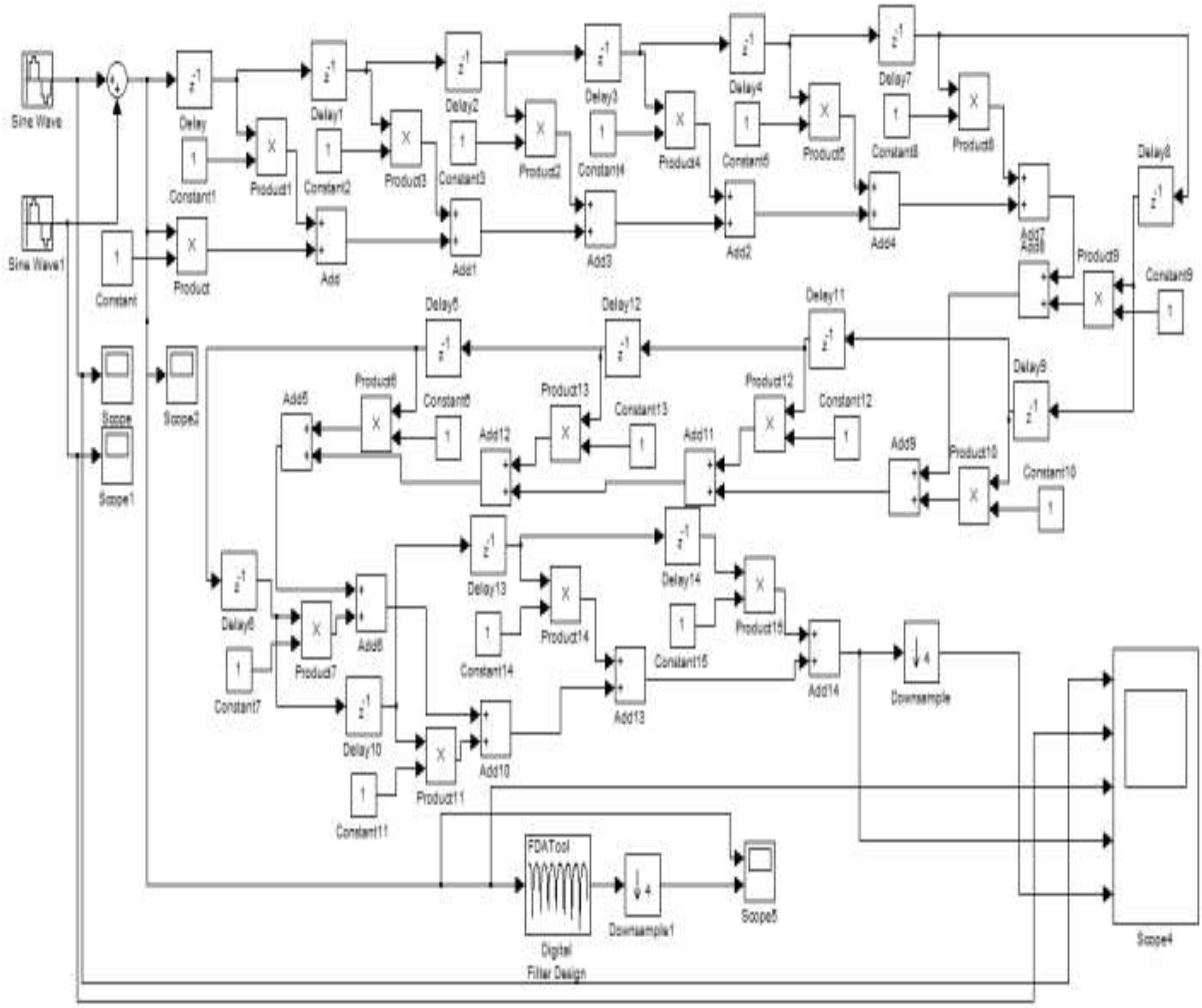

Fig 4.1.1: Simulink model of FIR filter with order 15

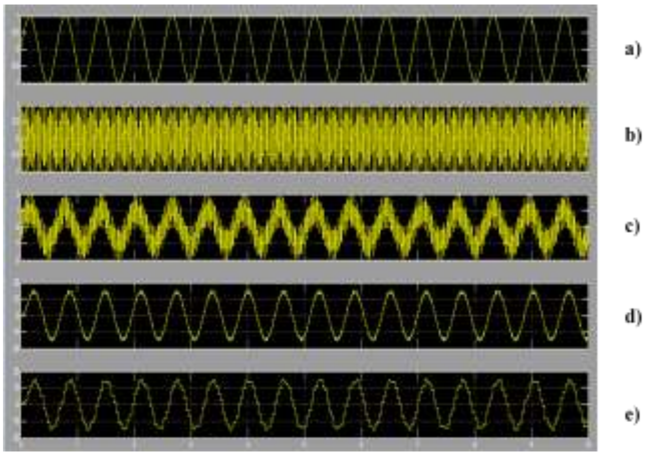

Fig 4.1.2: a) Clean input signal b) Noisy input signal c) input signal to FIR filter (Clean + Noisy) d) Filtered output signal from FIR filter e) Decimated signal with factor 4

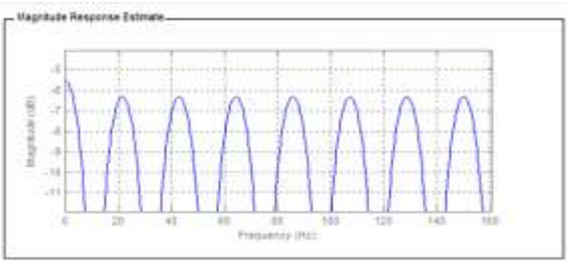

Fig 4.1.3: Magnitude response of FIR filter

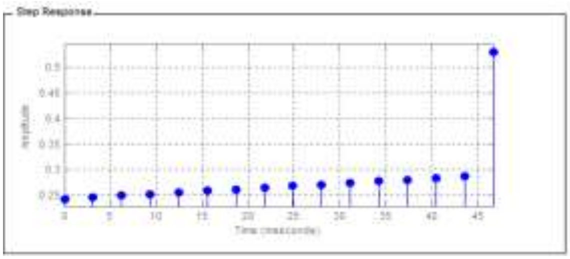

Fig 4.1.4: Step response of FIR filter

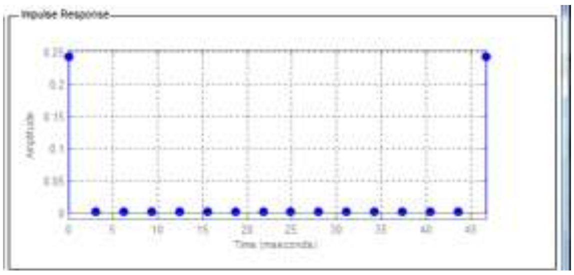

Fig 4.1.5: Impulse response of FIR filter

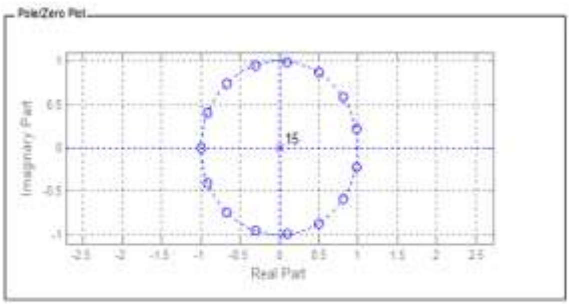

Fig 4.1.6: Pole zero plot of FIR filter

4.2 Design of IIR Filter

In this design of IIR filter, two Sine wave input signals are used. The specification of signals is same as previous. First input signal is used for as low frequency information signal i.e. clean signal and second is noisy input signal. These two input signals are added together by the sum block and this added signal given to the IIR filter block. This IIR filter block is designed with delay element, multipliers, constants and adders. Multiplier can multiply the given signal and constant i.e. filter coefficient.

The output of multiplier is given to the adder. Now, Output of delay signal is given to the next delay and another multiplier at the same time. This multiplier multiply third delayed signal with the constant and the output of this multiplier is given to the adder. Now, the output of first multiplier and second multiplier is added together and then output of adder is given to the next adder. The figure 4.2.1 shows IIR filter structure for direct form 1 structure. The figure 4.2.2 shows IIR filter structure by using digital filter design i.e. FDA tool.

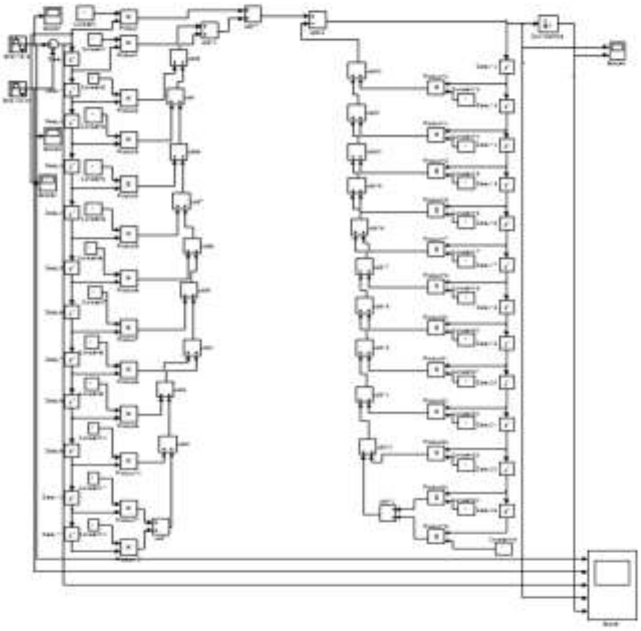

Fig 4.2.1: Simulink model of Direct Form -1 IIR filter

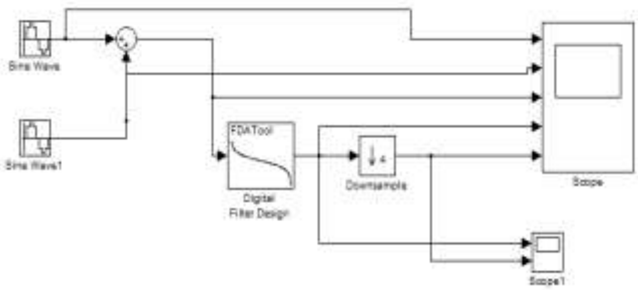

Fig 4.2.2: Simulink model of IIR filter using FDA tool

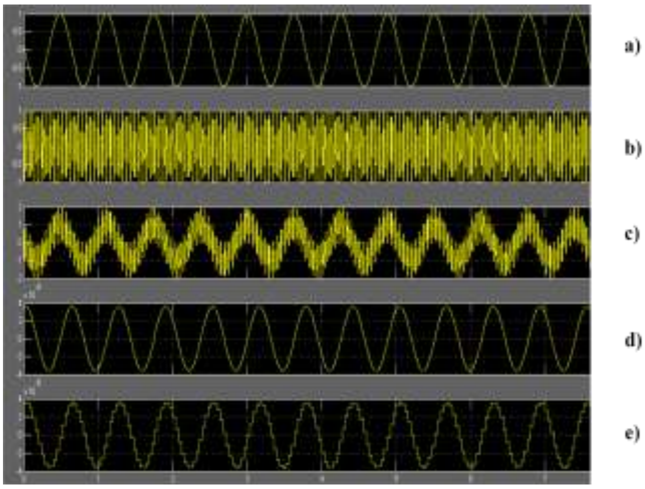

Fig 4.2.3: a) Clean input signal b) Noisy input signal c) input signal to IIR filter (Clean + Noisy) d) Filtered output signal from IIR filter e) Decimated signal with factor 4

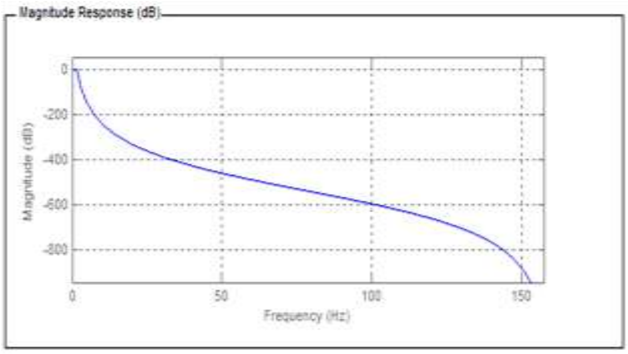

Fig 4.2.4: Magnitude response of IIR filter

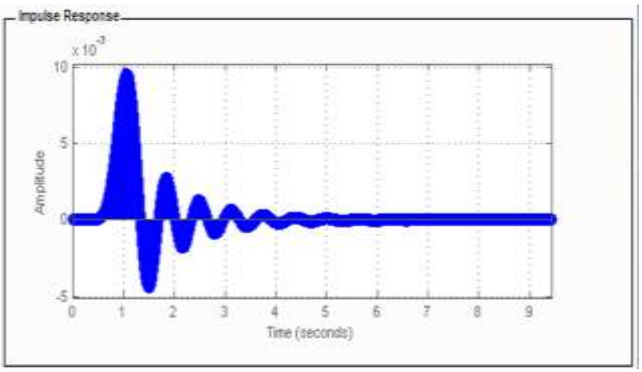

Fig 4.2.5: Impulse response of IIR filter

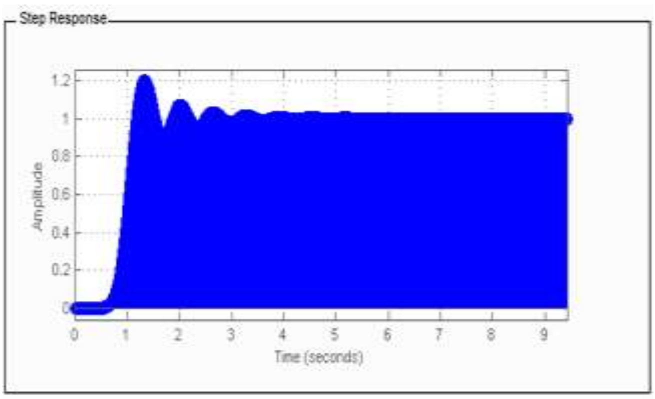

Fig 4.2.6: Step response of IIR filter

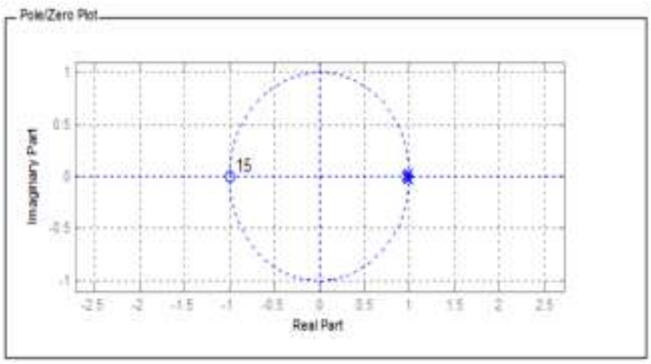

Figure 4.2.7: Pole-zero plot of IIR filter

4.3 Design of CIC Filter

In this design of CIC filter, two Sine wave input signals are used. The specifications of signals are same as previous. First input signal is used for as low frequency information signal i.e. clean signal and second is noisy input signal. These two input signals are added together by the sum block and this added signal given to the CIC filter block. This CIC filter block is designed with unit delay element and adders.

This is five stage CIC filter. Designed model consist of five integrator and five comb filter stages and they are separated by down sampler with decimation factor of 4. The figure 5.4.1 shows CIC filter structure with five stages.

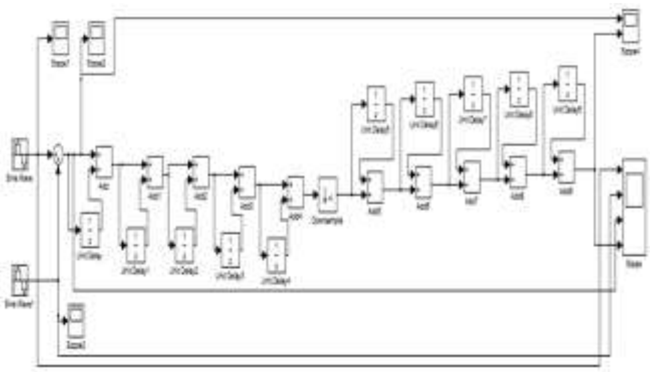

Fig 4.3.1: Simulink model of five stage CIC filter

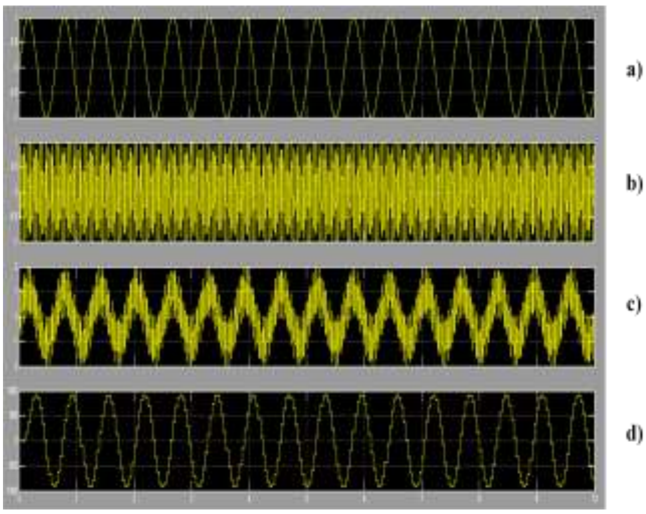

Fig 4.3.2: a) Clean input signal b) Noisy input signal c) input signal to CIC filter (Clean + Noisy) d) Decimated signal with factor 4

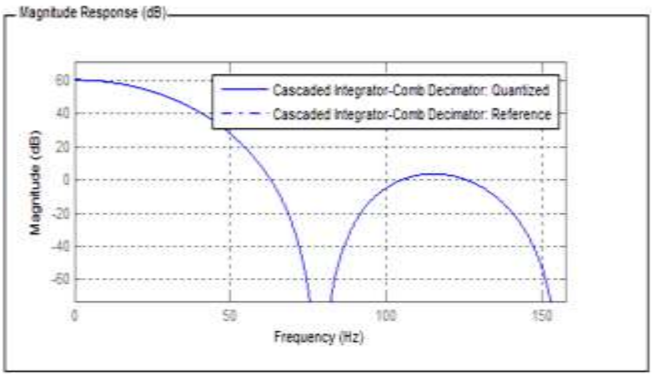

Figure 4.3.3: Magnitude response of CIC filter

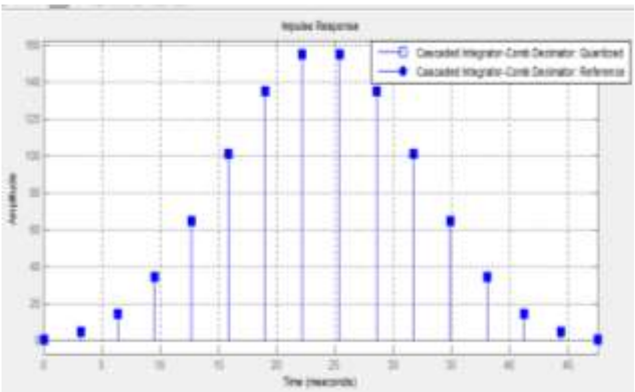

Figure 4.3.4: Impulse response of CIC filter

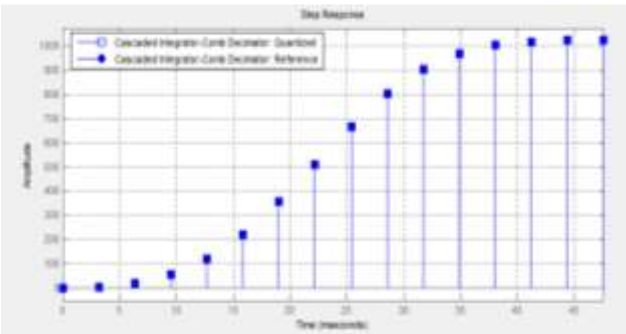

Figure 4.3.5: Step response of CIC filter

4.4 Comparison of IIR, FIR and CIC Filters

This section contain the comparison of IIR filter, FIR filter and CIC filter according to the storage requirement, hardware requirement, Number of stages, filter gain and simulated waveforms.

Table 4.4.1: Comparison of IIR, FIR and CIC filter

Parameters	FIR filter	IIR filter	CIC filter
Total number of filter coefficients	16	30	0
Total number of States	16	15	10
Total number of Multipliers	16	30	0
Total number of Adders	15	30	10
Filter output gain	1	1	1
Decimation Factor	4	4	4

5. CONCLUSION

The design of FIR filter, IIR filter and CIC filter are implemented using Matlab Simulink model 2012a. These filters are used for decimation process with the decimation factor 4. Finally these filters compared on the basis of storage requirements, hardware requirements and filtered simulated waveforms. IIR filter provided highly filtered output signal as compare with the FIR filter but it required more number of hardware and filter coefficient storage. Thus it has more power consumption. FIR filter used less filter coefficient storage as well as hardware and thus power consumption is less as compare with IIR filter.

CIC filter is one of the best options for Multirate filtering and it is multiplier-free filter and also it has no filter coefficients. Thus no storage required and also used less hardware and thus power consumption is very low. But its output response is not accurate and it required one extra reconstruction filter at output side to get desired output signal. _ Figure 6.1: Comparison of FIR, IIR and CIC filter

6. FUTURE WORK

By implementing low-complexity algorithm on reconfigurable FPGA the performance and efficiency of the system can be further improved. Low power VLSI technique can be used to achieve very low power consumption system. So in future, the processors can attain high efficiency by selectively applying reconfigurable and committed hardware to the tasks for which they are best matched.

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BIOGRAPHIES

Pankaj R. Ambilduke Obtained his Bachelor’s degree in Electronics from Kolhapur University and Master’s degree in Electronics and Communication from TIT College the University of R.G.P.V. Bhopal He currently lectures at the STC, SPRT Khamgaon, Maharashtra.

Manish Kumar Gurjar Obtained his Master’s degree in Electronics and Communication from the University of R.G.P.V. Bhopal He currently assistance Professor at the TIT College Bhopal, Madhya Pradesh (MP)