Implementation and Performance Comparison Study of 1-D and 2-D FIR Filter using MATLAB

Jyotsna Yadav, Arpita Bharti, Rohit Patel, Mukesh Kumar

Abstract- FIR filters have only a finite number of terms in their impulse response. These filters have a number of advantages over the IIR filter types. FIR filter is always stable, realizable, and provides a linear phase response under specific conditions. These characteristics make FIR filters attractive to many filter designers. However, the major disadvantage of FIR filters is that the number of coefficients needed to implement a specific filter is often much larger than for IIR designs. Finite Impulse Response (FIR) filter is a filter structure that can be used to implement almost any sort of frequency response digitally. An FIR filter is usually implemented by using a series of delays, multipliers, and adders to create the filter's output. A Multirate digital signal processing is required in digital system where more than one Sampling Rate is required. This paper brings the performance comparison between the FIR designing methodologies like the 1-D, 2- D FIR Filters. In this paper 1-D, 2-D FIR filters using their operation have been implemented and simulated in the MATLAB and Simulink environment and their response has been studied in the waveforms. Simulation result shows that 2-D filter has increased computation speed as compared to 1-D, and is more efficient in reducing the noise in the signal.

***Keywords-Digital Filters, FIR Filters, 2-D FIR Filters, MATLAB**.*

I. INTRODUCTION

In DSP the major problem for the developers is the designing of the digital filters for receiver processing and to transmit the various amount of data within required frequency band tighter filter specifications is the need. As the name indicates, filter removes undesired components of signal. In other words, it can be for the selection of desired signal components from noisy signal. In general we can say that filter is a signal selection system. There are two types of digital filters Finite impulse response filter and infinite impulse response filters. Speaking from the realization, IIR adopts recursive structure and uses the rational fraction which is equal to the ratio of two polynomials approximate to the frequency character, so it is able to get better frequency selection characters by use of less order [3].

The main objective of this paper to design and performance comparison study of 1-D and 2-D FIR filters using MATLAB. The rest of paper is organized as follows: section-II introduces FIR filter, Section-III introduces 2-D FIR filter, section-IV gives simulation environment, section-V presents designed FIR filters and section-VI presents simulation results. Finally, conclusion is presented in section-VII.

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II. FIR FILTER

Digital filters are classified either as finite duration unit pulse Response (FIR) filter or infinite duration unit pulse response (IIR) filters, depending on the form of the unit pulse response of the system. In FIR system, the impulse response sequence is of finite duration, in other words, it has a finite numbers of non –zero terms. An FIR filter is a filter structure that can be used to implement almost any sort of frequency response digitally. An FIR filter is usually implemented by using a series of delays, multipliers, and adders to create the filter's output.

The difference equation that defines the output of an FIR filter in terms of its input is:

$$Y[n] = b_0X[n] + b_1X[n-1] + \text{----------------} + b_N X[n-N]$$

Where:

- X[n] is the input signal
- Y[n] is the output signal
- bi are the filter coefficients
- N is the filter order

III. 2-D FIR FILTER

Multirate digital signal processing is required in digital system where more than one sampling rate is required. Different sampling rates can be obtained using an up sampler and down sampler. The basic operations in multirate processing to achieve this are decimation and interpolation. Decimation is for reducing the sampling rate and interpolation is for increasing the sampling rate.

There are various areas in which multirate signals processing is used:

(1) Communication Systems
(2) Speech and audio processing system
(3) Antenna system and
(4) Radar system

IV. SIMULATION ENVIRONMENTS

Parameters	**value**
Filter structure	Direct Form
Filter length	32
Filter type	Type 2
Design method	Window (Kaiser, β=0.5)
Frequency specification	Normalized (0-1)
Wc	0.5

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V. DESIGNED FIR FILTERS

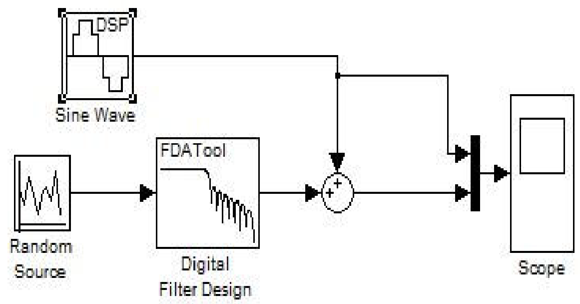

Fig 1: SIMULINK realization of 1-D FIR Filter

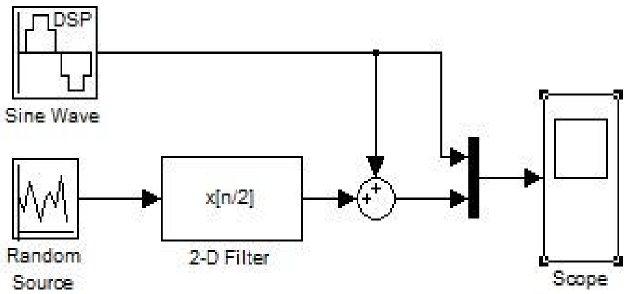

Fig 2: SIMULINK realization of 2-D FIR Filter

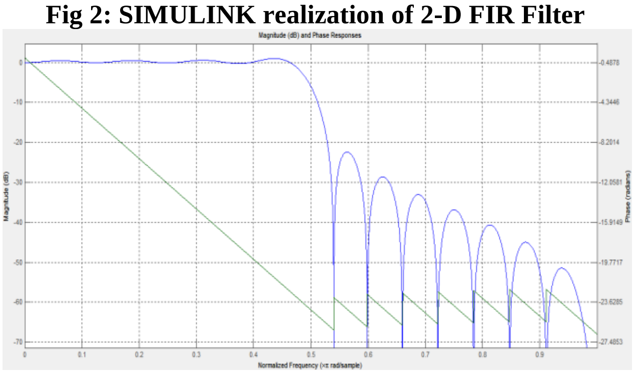

Fig 3: Magnitude and Phase Response of the used FIR Filter

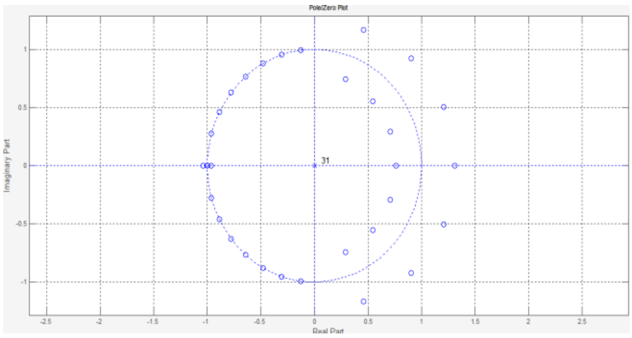

Fig 4: Pole-Zero plot of used FIR Filter

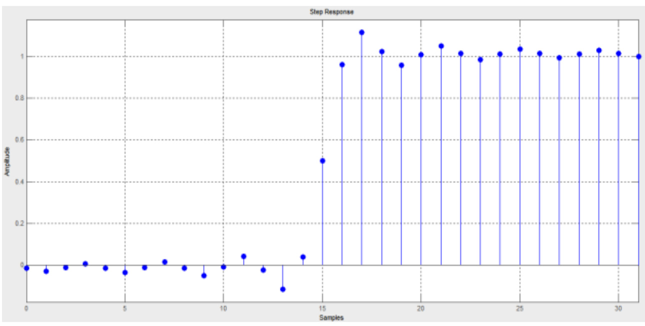

Fig 5: Step Response of the used FIR Filter

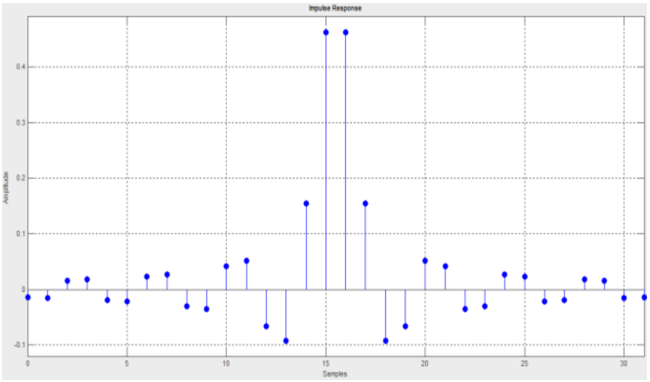

Fig 6: Impulse Response of the used FIR Filter

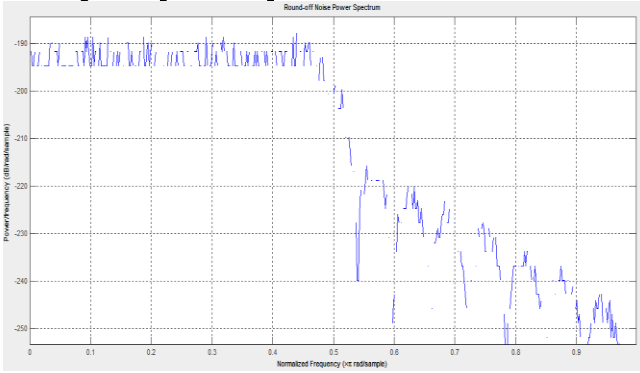

Fig 7: Round-off Noise Power Spectrum of the used FIR Filter

VI. SIMULATION RESULTS

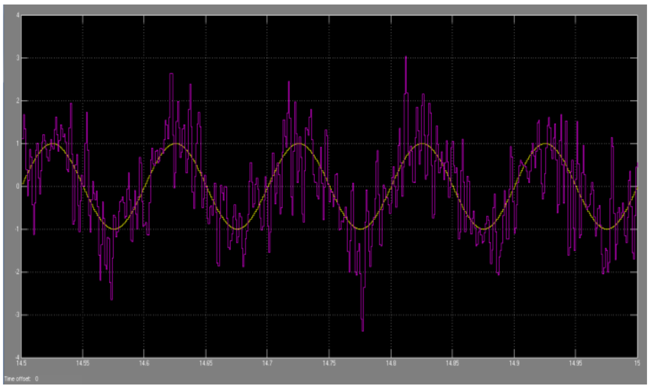

Fig 8: Signal Wave Form of 1D FIR filter for simulation time 15 seconds

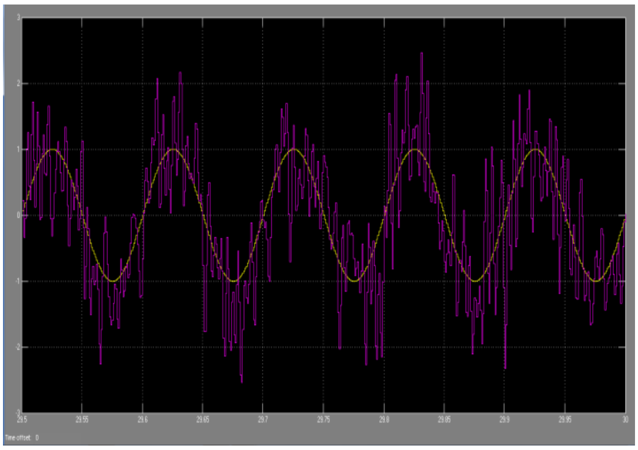

Fig 9: Scope output of 1D FIR filter for simulation time 30 seconds

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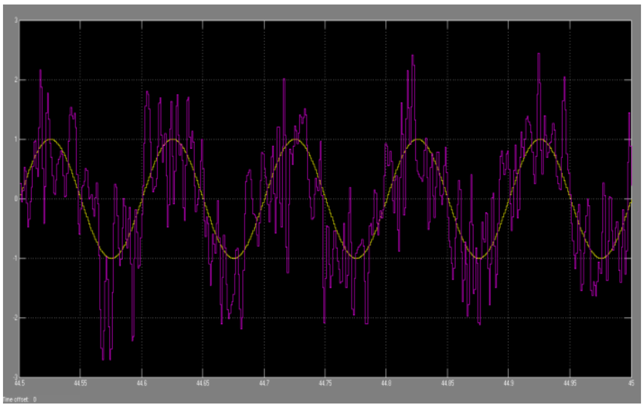

Fig 10: Scope output of 1D FIR filter for simulation time 45 seconds

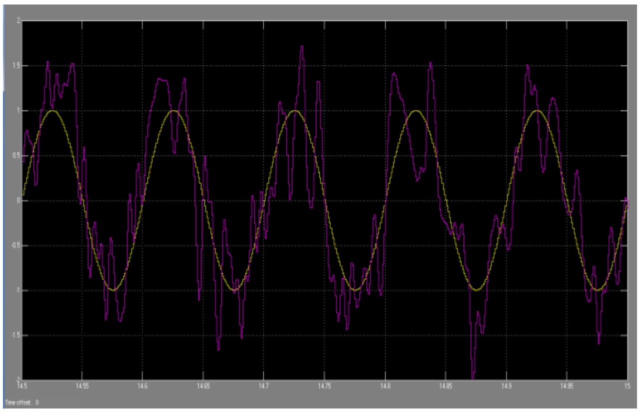

Fig 11: Scope output of 2D FIR filter for simulation time 15 seconds

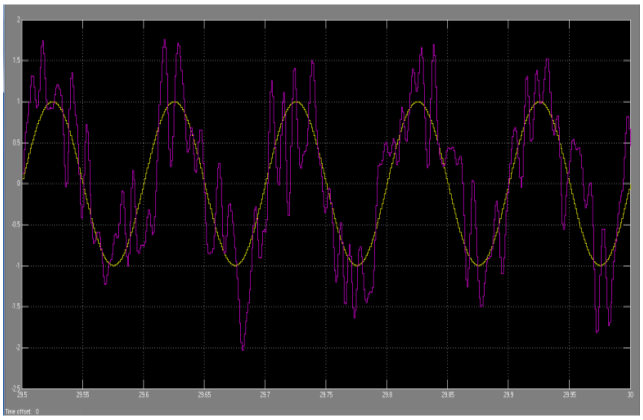

Fig 12: Scope output of 2D FIR filter for simulation time 30 seconds

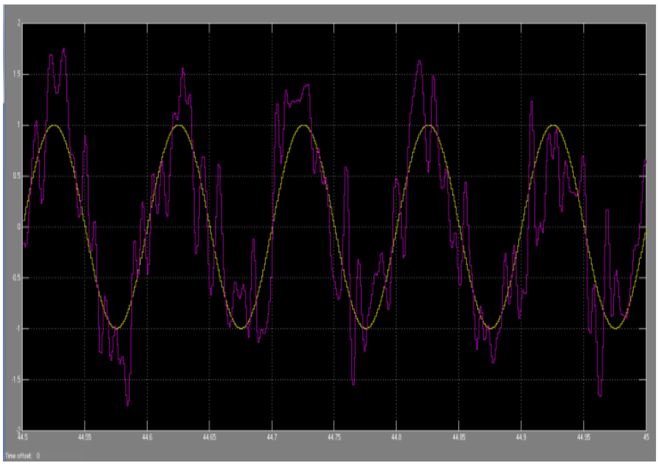

Fig 13: Scope output of 2D FIR filter for simulation time 45 seconds

VII. CONCLUSION

The implementation and simulation of 1D and 2D FIR filter have been done using MATLAB and their responses have been studied and compared in various obtained wave forms as given in the simulation result. After comparing these waveforms we conclude that 2-D filter has increased computation speed as compared to 1-D, and it is more efficient in reducing the noise presented in the signal, if we increasing the phase of the filter better results can be obtained but in this case if we increase the order of the filter the need of the storing the coefficients increases leading to hardware incompatibility.

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