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# Implementation and Performance Estimation of FIR Digital Filters using MATLAB Simulink

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***Abstract— In modern communication systems, filtering is the most widespread and extremely important signal processing technology. FIR filters form the basis of wireless systems in medical devices, industrial control, consumer electronics, and cellular infrastructure. A filter network selectively changes the wave-shape of a signal in a desired approach. The most common filtering purpose is to remove the noise from the signal. The FIR filters are of finite transient duration, more stable, efficiently realisable and providing exact linear phase as compared to IIR filters .FIR filter structure can be used to realize almost any sort of frequency response digitally. FIR filters are invariably used in the situation where linear phase characteristics within the passband of the filter is required. In this paper, implementation and simulation of 1-D and 2-D FIR filters is presented in the MATLAB and Simulink environment. The comparison of the output waveforms show that 2-D FIR filter posses higher speed than 1-D FIR filter.***

***Index Terms— Digital filters, 1-D FIR filters, 2-D FIR filters, Simulink.***

## I. INTRODUCTION

In DSP the major hitch for the developers is the designing of the digital filters for receiver processing and to transmit the various amount of data within desired frequency band according to the filter specifications. Digital filters play vital role in DSP, compared with analog filters, they are preferred in numerous applications like data compression, speech processing and image processing [1].Due to the demanding use of FIR filters in communication and video systems, high performance in speed, area and power consumption is required. Digital filters are essentially used to transform the characteristic of signals in time and frequency domain and have been recognized as primary digital signal processing [2].Filtering is a basic aspect of signal processing which executes direct manipulations on the frequency band of signals, removing undesirable part as noise or extracting useful components. Depending on the type of the unit pulse response of the system, digital filters can be categorised as Finite-duration unit pulse response (FIR) filters or Infinite-duration unit pulse response (IIR) filters. In case of Finite Impulse Response digital filter, impulse response is finite, so it can be used for Fast Fourier Transform algorithm to achieve the filtered signal, which can greatly improve the efficiency of operation [3]. According to Fourier transform theory, the linear convolution of the two spectral sequences in the time domain is the same as multiplication of two corresponding sequences in the frequency domain. A digital filter is simply a discrete-time, discrete-amplitude convolver which performs a convolution of the time domain impulse response and the discrete signal. An FIR filter of length M has the output as a convolution between the input sequence and the impulse response of the filter described by:

$$y(n) = \sum_{k=0}^{M-1} b(k).x(n-k) \tag{1}$$

The response of the FIR filter depends only on the present and past input samples, whereas for the IIR filter, the present response is a function of the present and past M values of the excitation as well as past values of the response [1]. In some applications, the FIR filter circuit must be able to operate at high sample rates, while in other applications, the FIR filter circuit must be a low power circuit operating at moderate sample rates [4]. The FIR filter can be designed by impulse response truncation, windowing design method and by optimal filter designing technique. Practical FIR designs typically consist of filters that meet certain design specifications i.e.,they have a transition width and maximum passband/stopband ripples that do not exceed allowable values [5]. Typical filter applications include signal preconditioning, band selection, and low pass filtering [6]. Digital finite-duration impulse response (FIR) filters are non-recursive filters which are having following advantages compared to infinite-duration impulse response (IIR) filters:
They are always stable.
They can have exactly linear phase.
The design methods are generally linear.
The filter start-up transients have finite duration.
They can be implemented efficiently in hardware.
The particular disadvantage of FIR filters is that they often require a much higher filter order than IIR filters to achieve a desired performance. The delay in case of FIR filters is often much greater than for an equal performance IIR filter. The IIR filter uses 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 [7]. In digital systems, there is the requirement of Multirate digital signal processing when more than one sampling rate is needed. Multirate systems often result in more efficient processing of signals because the sampling rates at various

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internal points can be kept as small as possible [8]. Different sampling rates can be obtained by using up-sampler and down- sampler. Decimation and Interpolation are the two basic operations in multirate digital signal processing. Decimation is used for reducing the sampling rate whereas interpolation is employed for increasing the sampling rate. Communication systems, antenna systems, radar systems, speech and audio processing systems are the various areas where multirate signal processing can be customarily used.
Simulink is a high level simulation software that provides an interactive scientific and engineering environment for system modeling , simulation and anlaysis. For the rapid realization of a digital signal processing application in terms of functional blocks the simulink environment is extremely useful. The following table depicts the simulation environment for the designing of the digital FIR filters. The table includes the structure type, response, filter order, designing method and frequency specification for digital filters.

**Table- I FIR Filter Parameters Specifications**

| **Filter's Parameters** | **Value** |
|---|---|
| Structure | Direct FIR Filter |
| Response | Low Pass Filter |
| Filter Order | 32 |
| Design Method | (Kaiser Window, β = 0.5) |
| Frequency Specification | (0-1) Normalized |
| Wc | 0.5 |

## II. IMPLEMENTATIONS OF FIR FILTERS

The Fig. 1 shows the realization of the 1-D FIR filter using MATLAB Simulink. The FDA tool has been used for obtaining the desired filter length and the windowing technique has been used as the designing method. The different responses of the 1-D FIR filter are shown which are Magnitude versus Phase response, Impulse response, Step response, Pole-zero plot and Round-off noise power spectrum.

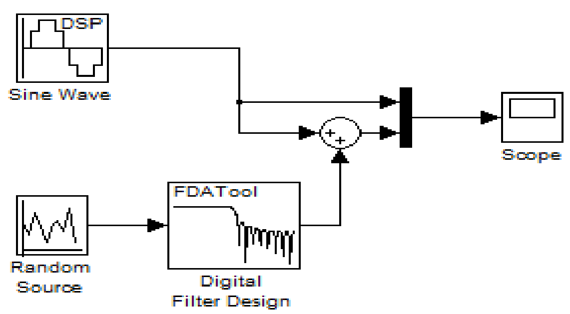


Fig. 1 Implementation of 1-D FIR Filter

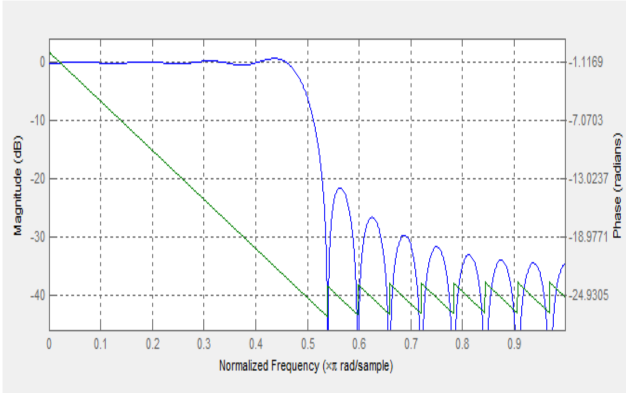


Fig. 2 Magnitude (dB) Versus Phase Response

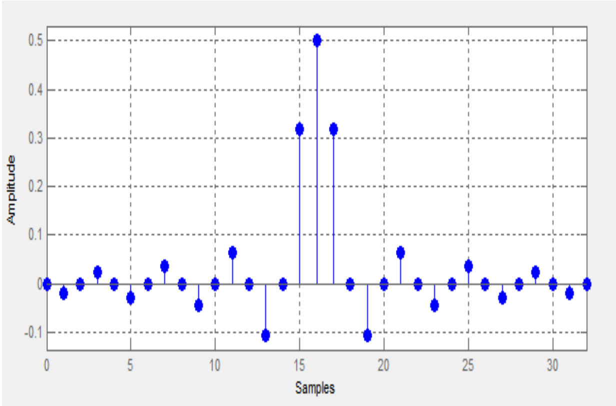


Fig. 3 Impulse Response of 1-D FIR Filter

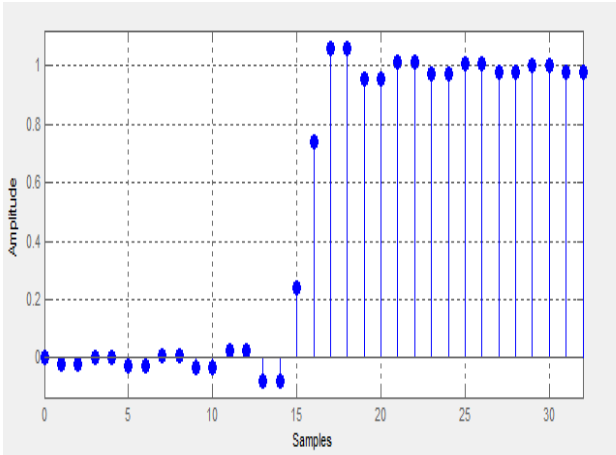


Fig. 4 Step Response of 1-D FIR Filter

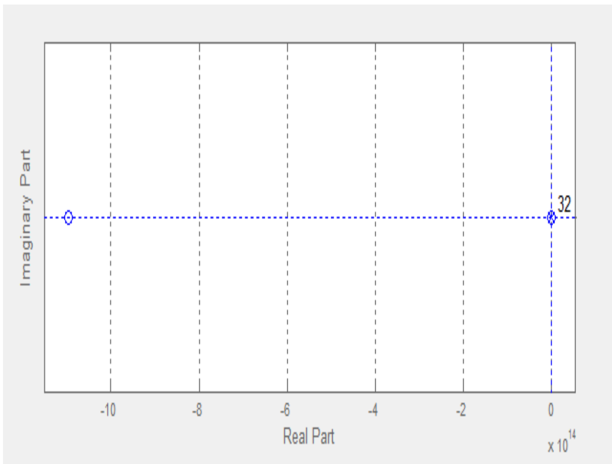


Fig. 5 Pole-Zero Plot of 1-D FIR Filter



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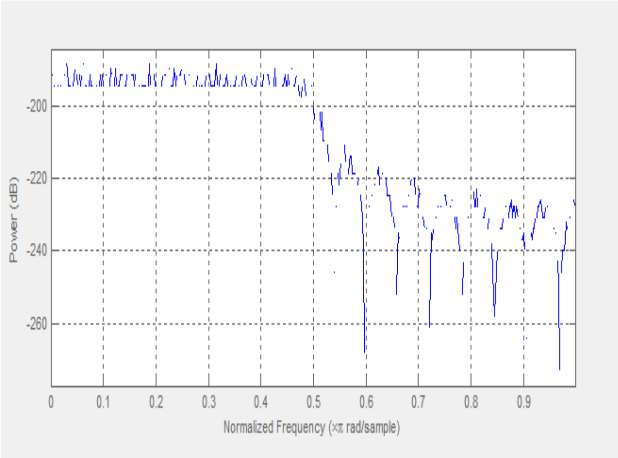


Fig. 6 Round-off Noise Power Spectrum of 1-D FIR Filter

The figures 7 to 9 shows the outputs of the 1-D FIR filter for the simulation times for 10 seconds, 15 seconds and 25 seconds respectively.

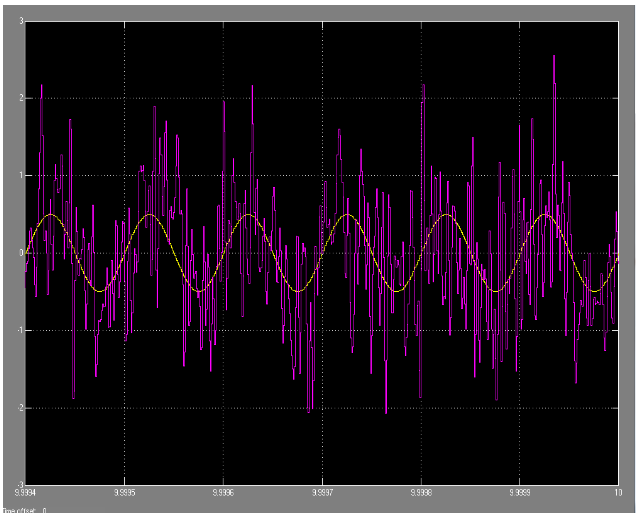

Fig. 7 Simulated Output of 1-D FIR Filter for 10 Seconds

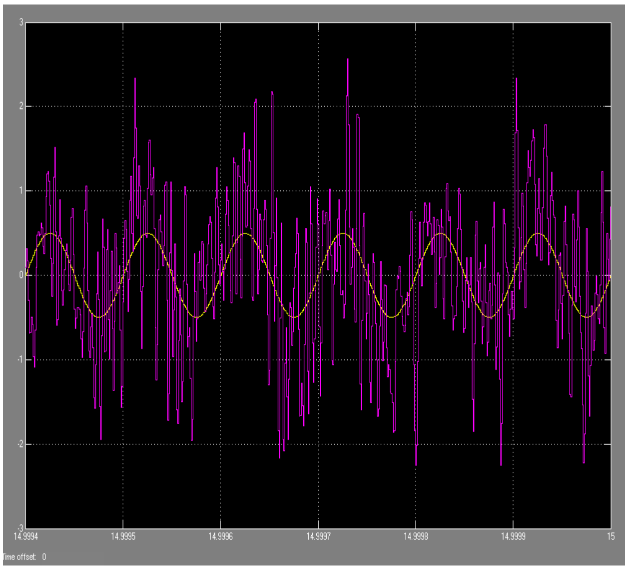

Fig. 8 Simulated Output of 1-D FIR Filter for 15 Seconds

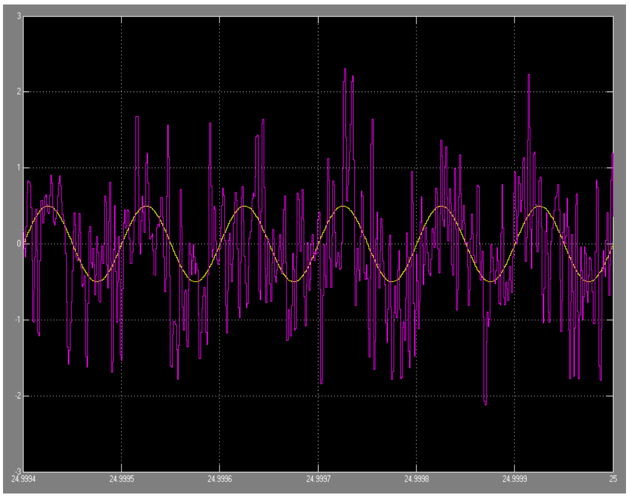

Fig. 9 Simulated Output of 1-D FIR Filter for 25 Seconds

The Fig. 10 shows the realization of the 2-D FIR filter using MATLAB Simulink. The scope is used to acquire the simulated outputs of the 2-D FIR filters. The outputs of 2-D FIR filter are demonstrated in the figures 11 to 13 for the simulation times for 10 seconds, 15 seconds and 25 seconds respectively. The 2-D FIR filter is computationally faster as compared to 1-D FIR filter.

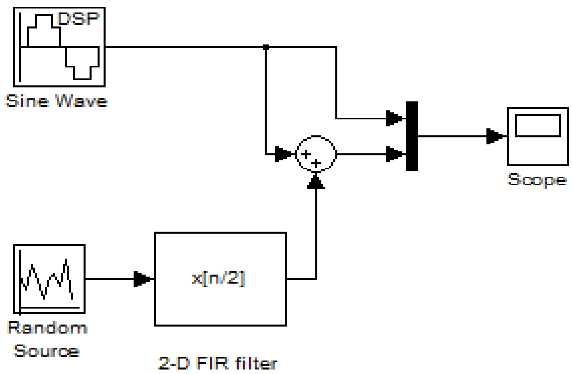


Fig. 10 Implementation of 2-D FIR Filter

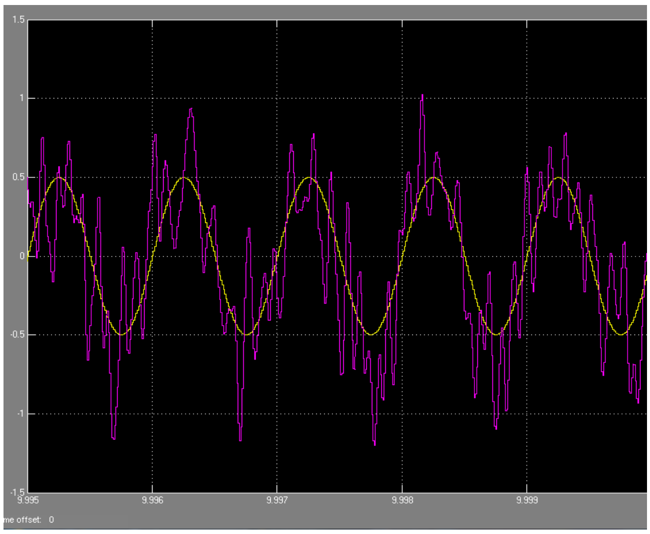

Fig. 11 Simulated Output of 2-D FIR Filter for 10 Seconds



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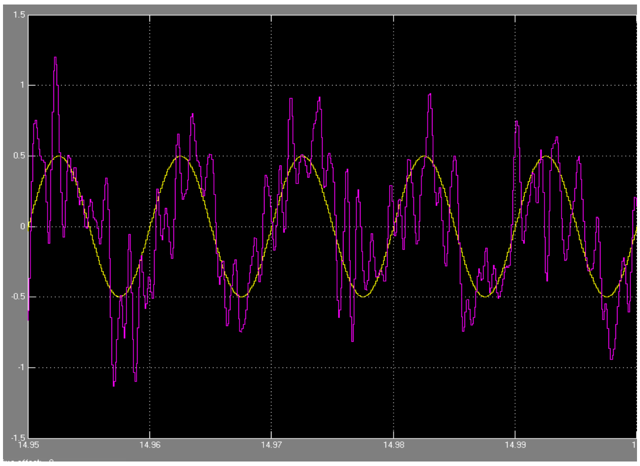

Fig. 12 Simulated Output of 2-D FIR Filter for 15 Seconds

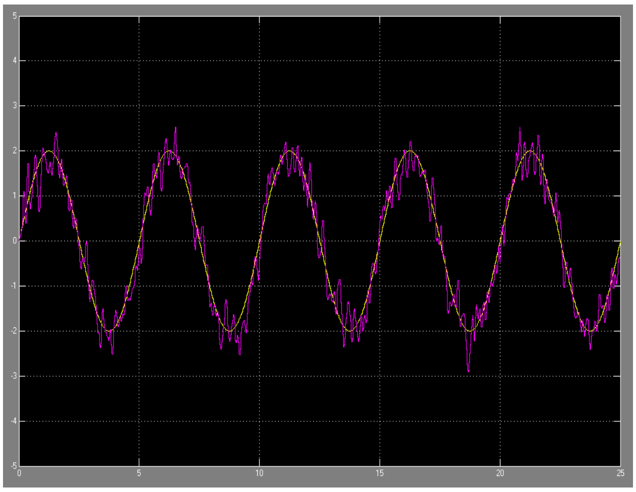

Fig. 13 Simulated Output of 2-D FIR Filter for 25 Seconds

## III. CONCLUSION

The realization of the Direct-type FIR filters are carried out in the MATLAB Simulink environment. The simulated FIR filter embraces the characteristics of linear phase and stability. The numbers of multipliers and adders required are 33 and 32 respectively in the case of 32-order FIR filter. There are 32 states in the simulated FIR filter. The comparison of the output waveforms reveals that the computational speed of 2-D FIR filter is more as compared to 1-D FIR filter. The 2-D FIR filter is more prompt in reducing the noise present in the signal, if we are increasing the phase of the filter, better results can be obtained but in this case if we increase the order of the filter the necessity of the storing the coefficients increases leading to the hardware inconsistency.

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