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 impulse response sequence is of finite duration, 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:

\[ Y[n] = b_0X[n] + b_1X[n-1] + \cdots + b_N X[n-N] \]

Where:
\[ \cdot X[n] \text{ is the input signal} \]
\[ \cdot Y[n] \text{ is the output signal} \]
\[ \cdot b_i \text{ are the filter coefficients} \]
\[ \cdot N \text{ is the filter order} \]

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.


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:

- Communication Systems
- Speech and audio processing system
- Antenna system and
- Radar system

IV. SIMULATION ENVIRONMENTS

<table>
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<td>Filter type</td>
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<td>Frequency specifcation</td>
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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

VI. SIMULATION RESULTS

Fig 6: Impulse Response of the used FIR Filter

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

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
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.

REFERENCES


