HCTL Open International Journal of Technology Innovations and Research (IJTIR)
http://ijtir.hctl.org
Volume 18, Issue 2, February 2016
e-ISSN: 2321-1814, ISBN (Print): 978-1-944170-16-5

Design and Implementation of Direct Form FIR Filter

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Abstract

The research article presents the design of the direct form of the Finite Impulse Response (FIR) filter using VHDL programming language. Multimedia technology and broadband communication demand the low power and high performance design applications in Digital Signal Processing (DSP). The digital filters are most important element of the communication system and DSP. In the paper 7 tap FIR filter is implemented in Xilinx 14.2 software and functionally simulated in modelsim 10.1 b software. The design of FIR filter of 4 x 4 configurations uses 9 adders and 16 multipliers. The design is also verified on SPARTAN 3E FPGA.

Keywords

Finite Impulse Response (FIR), Field Programmable Gate Array (FPGA), Very High Speed Integrated Circuit Hardware Description language (VHDL)

Introduction

FIR filter is one of the digital filters that give the finite impulse response. The filters are also called as non-recursive digital filters because the filter does not have the feedback or the recursive part of the filter. But the filter can follow the recursive algorithms in their design and realization.

Different methods can be employed for the designing of the FIR filters, but many of them are following the approximation of ideal filter. The main motive of the filter design is not to get ideal characteristics, because it is impossible practically, but it is possible to get good characteristics of a filter in sufficient manner. Filter order can be increased and the transfer function of FIR filter can approach to the ideal behavior with filter order increment. It enhances the complexity of the design and amount of period required for processing input samples of a signal that needed to filter out.

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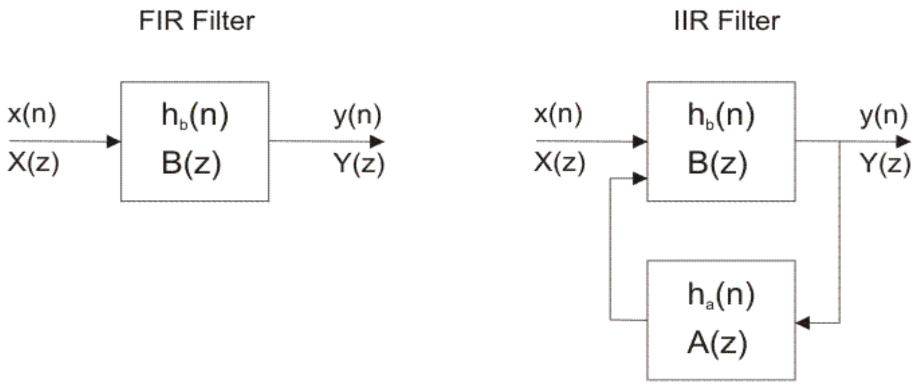

Figure 1: FIR and IIR filters block diagram

The frequency response of the resultant filter can be an oscillatory function or a monotone function that lies within a certain frequency range. The output waveform analysis of frequency response depends on its parameters and on the technique employed in designing process as well. The paper expresses the most popular technique for FIR filter design based on window functions. The general characteristics with respect to the transfer function as well as its derived values from the ideal frequency response depend on window function used and order of the filter designing. There are some advantages and disadvantages in each category of filter design. That's why; it is mandatory and important to choose the category of the filter in designing process. In case of getting linear phase and its characteristics there is one option to choose that is FIR filter not like IIR filter which are not stable. In some cases if the linear phase is not required such as applications on processing of speech signals, the FIR filters are not the good solution. The input and output signals of that present the nonlinear phase systems are shown in fig.2.

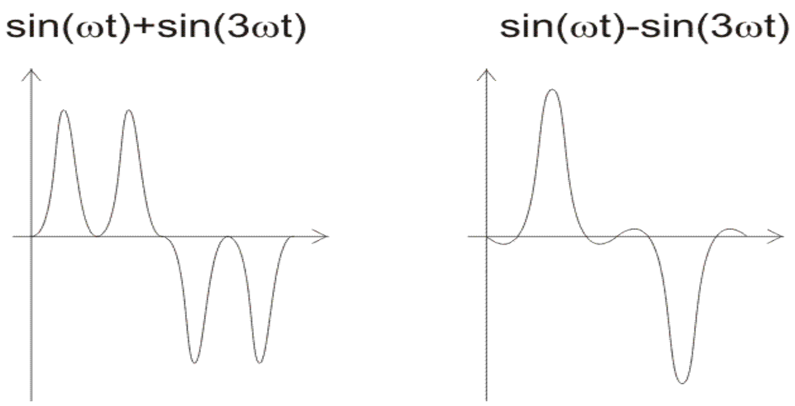

Figure: 2 The effect of non-linear phase characteristic

The phase shift of 0 radians is introduced by the system at the frequency of ω, and the phase shift π radians which is three times of the frequency at that time. Input signal is combination of natural frequency ω and one harmonic present having same amplitude at that frequency three times. In the fig.2, both waves present the two different types of waveforms. The value of the power of these signals is neither changed, nor the amplitudes of harmonic. The phase of such signal at second stage of harmonic is changed. If it is considered that me the input is a speech signal which has the phase

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characteristic are not of the essence, the phase distortion in this case is not so important. All the necessary requirements are satisfied in this case. Moreover, the distortion in the signal is permitted to have if the phase characteristics are considered very important.

The impulse response must be symmetric or anti-symmetric in order to keep the characteristics of FIR filter in linear.
The behavior can be expressed given below

h[n] = h[N-n-1] the property behaves symmetric impulse response
h[n] = -h[N-n-1] the property behaves anti symmetric impulse

The biggest drawback of FIR filter is that it is a high order of designed filter. In comparison to the IIR filter it has remarkably higher order corresponding to the same frequency response. It is one of the good reasons to follow the FIR filter in DSP applications when the linear phase characteristic play very important role. The filter has the coefficients and multiple delay lines are considered in a filter design, or multiple numbers of input samples must be saved for the computation of the output samples, to determine the order of a FIR filter. It can be understood with the help of example, if the filter has the order of 10, it means that it is necessary The FIR filter will save 10 input samples preceding the current sample. The performance of FIR filter is affected by the all eleven samples. The transfer function can be achieved after converting the equation in Z domain.

The transfer function is polynomial that contains the terms if complex variable z^{-1}.

The origin has all the poles of the transfer function. This property of the FIR filter makes it stable whereas the IIR filter has unstable behavior. The wide applications of the FIR filters are found in biomedical, control, and communication because it is easy to implement, stable and has best performance. The simplicity of the filter makes it very attractive for several applications whereas the filters are needed to minimize computational requirements. The important role of the filters is to eliminate the noise and unwanted signal from original input signal by eliminating the desired frequencies coming from the incoming signal. The FIR is much needed for the high end DSP applications and filtering.

Comparison between FIR and IIR Filters

The FIR and IIR filters which are non-recursive and recursive in nature have several characteristics for various applications. The chosen of non-recursive filters is done based on best performance of numerical operations, such as integration and differentiation. The comparison between IIR and FIR filter is listed in table 1.

Table 1: Comparison of FIR and IIR

IIR	FIR
IIR flites are efficient very much	FIR filters are less Efficient
IIR flites have analog equivalent	FIR filters do not have analog equivalent
The filters can be unstable	FIR filters are always stable
The filters has non-linear characteristics in phase	The filters has phase response, which is linear
No efficiency can be gained by decimation in frequency	In FIR filter decimation can increase the efficiency

FIR Filter Realization

The transfer function of the FIR filter can be written by the expression in Z domain

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{n=0}^{N-1} h[n] \cdot z^{-n}$$

The frequency response realization of the filter in time domain is very much interesting for the realization of FIR filter in both software and hardware. The transfer function can be obtained if we do the z-transform of the frequency response of FIR filter. The samples of the FIR filter output are computed using the following relation that presents the convolution property of x(n) and h(n).

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

Where, x[k] = presents the input samples of FIR filter.

h[k] = presents the weight coefficients of FIR filter frequency response; and

y[n] = output samples of the FIR filter.

One of the bestproperties of FIR filters is that the filters are not so sensitive to the accuracy of constants than IIR filters can be in the same order. The FIR filter can be realized in several forms such as direct form realization, direct transposed form of realization and cascade form of realizations. All these form of FIR filter are very play convenient role for the design and hardware implementation of the filter. For the software implementation of the filters, the direct and optimized form of the FIR filter matters. In the paper we have focused directly the direct form realization of the FIR filter, the same realization is depicted in fig.3.

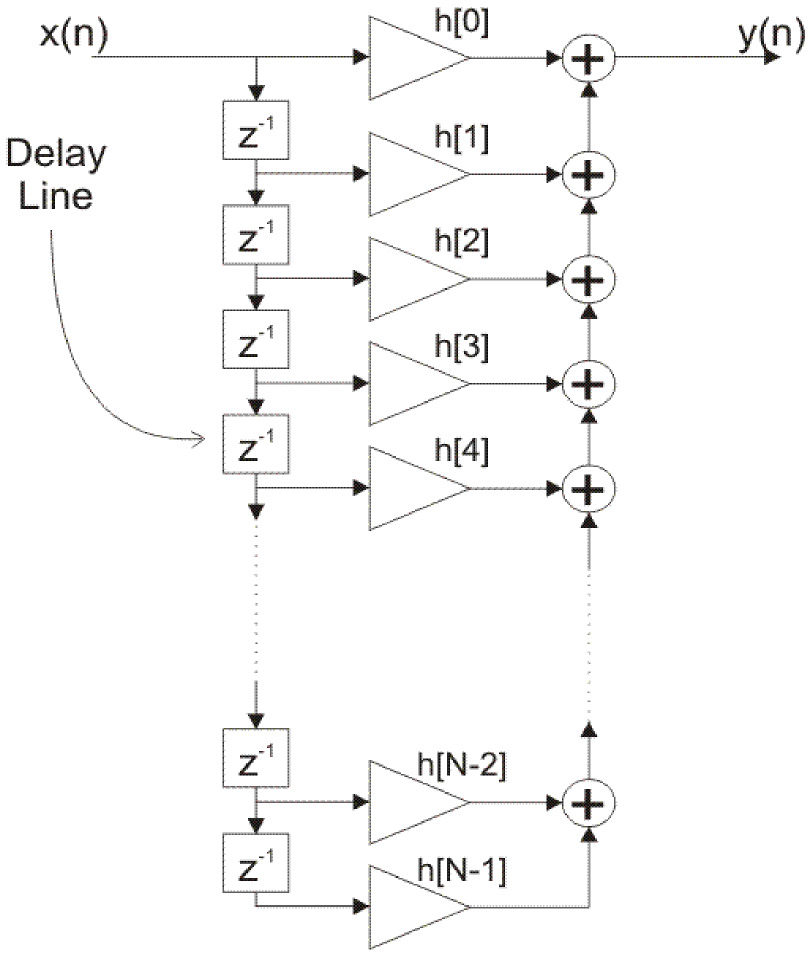

Figure 3: Direct form of FIR filter realization

For the direct realization of such structure, the numbers of multiplication constants are the same as FIR filter frequency response coefficients or the transfer function coefficients. The design of filter and tap depends on the input x(n) and h(n) sequence. Let us assume, that the input sequence x(n) having the length L1= 4 and h(n) also having another sequence length L2. The final output will have the tap coefficients (L1 + L2 -1)

$$x(n) = \{\, x(0), \quad x(1), x(2), x(3)\}$$

$$h(n) = \{\, h(0), \quad h(1), h(2), h(3)\}$$

The convolution sum of the output will be $y(n) = x(n) \otimes h(n)$

The linear convolution can be achieved.

	h(0)	h(1)	h(2)	h(3)
x(n)	x(0)h(0)	x(0)h(1)	x(0)h(2)	x(0)h(3)
x(1)	x(1)h(0)	x(1)h(1)	x(1)h(2)	x(1)h(3)
x(2)	x(2)h(0)	x(2)h(1)	x(2)h(2)	x(2)h(3)
x(3)	x(3)h(0)	x(3)h(1)	x(3)h(2)	x(3)h(3)

Figure 4: Convolution Method

Results and Discussion

The modesim waveform of devoped FIR filter is shown in fig. 5. The input sequnec is x(n), x(n-1), x(n-2) and x(n-3) with tap cofficents h(0), h(1) , h(2) and h(3). The output of the FIR filter is presnetd using the FIR filter output FIR_out1, FIR_out2, FIR_out3, FIR_out4, FIR_out5, FIR_out6 and FIR_out7.

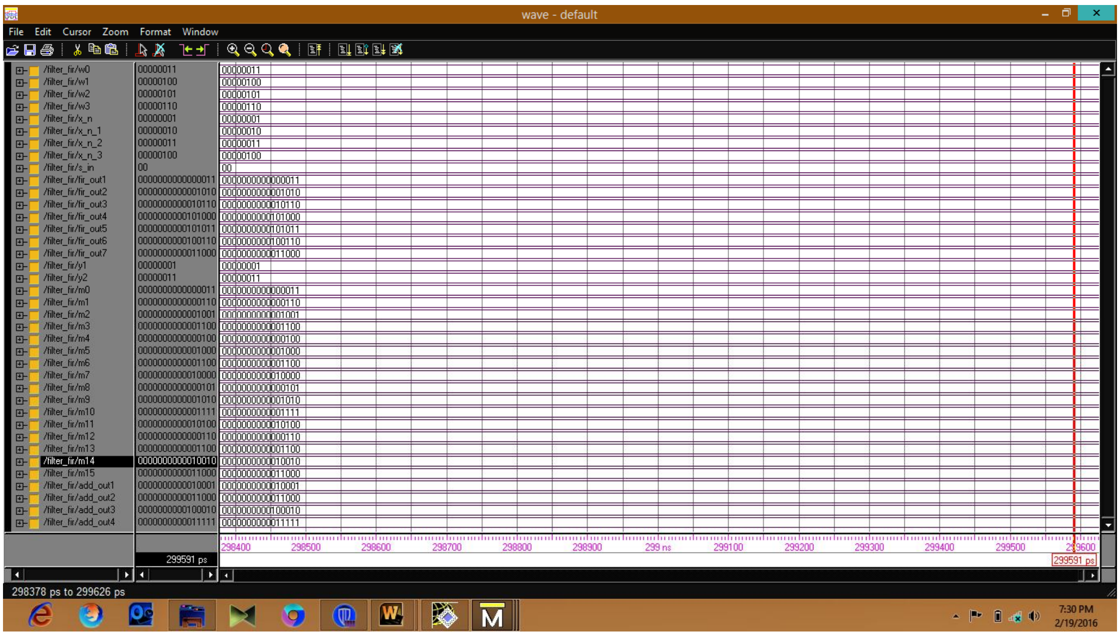

Figure 5: Waveform of FIR Filter in Modelsim

In the simulated waveform the input discrete sequence is x(n) ={1, 2, 3,4} and weight coefficients , h(n) = w(n) = {3, 4, 5, 6} the output achieved is FIR_out1 = 3, FIR_out 2= 10, FIR_out3 =22, FIR_out4 = 40, FIR_out 5= 43, FIR_out6 =38, and FIR_out7 =24. The output doscrete sequnece of the FIR filter is presented as FIR_out = {3, 10, 22, 40, 43, 38, 24}.

Conclusion

The direct form of the FIR filter is realized in VHDL for the input length 4, with weight coefficients 4. The direct form of the FIR filter is applicable in many DSP applications like phased locked loop design and software defined radio. The design is tested on SPARTAN-3E FPGA successfully. The hardware parameters a delay is estimated for the developed design of FIR filter. The combinational delay of the FIR filter is found 20.00 ns. In future the research can be done for the design implementation for larger no. of tap on high end FPGA.

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