# Design and Implementation Low Pass FIR Digital Filter Using Windowing Techniques

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**ABSTRACT**

A filter can need to have a certain frequency response, or a particular reaction to an impulse, step, or ramp, or mimic an analogue system. Digital filters can be categorised into Finite Impulse Response (FIR) filters & Infinite Impulse Response (IIR) filters depending on the system response. The thesis addresses FPGA low pass FIR filter design. Theoretical and experimental findings conducted FIR low pass filter point to the window design approach as very straightforward and user-friendly due to the presence of well-defined equation. Comparison indicated that the Direct-Form structure technique is simpler and performs better than other typical filter structures whereas Kaiser window gives the minimal main-lobe width and a sharp cut-off indicating smaller transition width. Experimental research of coefficient quantisation reveals a link between the frequency response, number of coefficients, and bit count.

**Key words**: Digital Signal Processing (DSP),Finite Impulse Response (FIR) Filter, Low- Pass Filter, Verilog HDL, Windowing Techniques, Hamming Window, Hanning Window.

## 1. INTRODUCTION

Filtering is the fundamental procedure in digital signal processing. Many electronic devices employ this function to cancel out superfluous or signal-damaging portions of the signal. Filters serve two purposes: signal separation and signal restoration. Signals tainted by noise and interference need separation methods. Breath signal and cardiac signal of the mother will contaminate a gadget measuring the electrical activity of a baby's heart inside the mother's womb. Filters are applied at such periods to isolate the signals and examine them separately. Signal restoration is the method utilised when signal gets distorted. Audio recording done with subpar equipment is filtered to produce better sound signal output than the original it formerly generated. Analogue or digital filters can either solve them. Analogue filters, less expensive, quicker, and with great dynamic range in both amplitude and frequency. By comparison, digital filters show far better performance level.

Quality is greater than analogue filters to digital filters can reach performance unique. The filtering issue is approached makes a significant difference. Analogue filters have limits when controls stress accuracy and stability, such as with resistors and capacitors in electronics. By contrast, many people overlook digital filters to improve filtering performance. The emphasis changes signal limits and the processing of theoretical problems. It is a filter in the time domain of the input and output signals telling the DSP. Since it is often produced by the signals of the time pattern at consistent intervals. But this model is not the only way to start. The second most frequent method by which the space matches the sample period. Many more domains are conceivable; nonetheless, time and space are by far the most prevalent. Every linear filter impulse response is a step response

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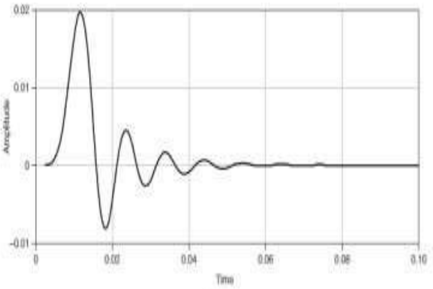
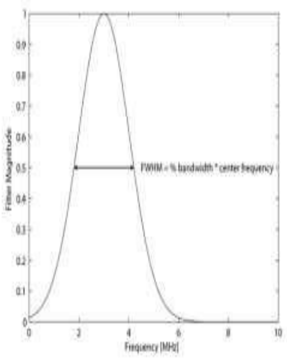
and frequency response. The filter is full information about every one of these responses, but in a different form. Of the three listed, the other two are set 2 and may be computed straight away. Their reactions vary in various contexts to characterise the filter since the three representations, the most essential.



Impulse Response



Frequency Response

Convolution of the digital impulse response with the discrete time signal input is the simplest method to realise a digital filter response. Every linear filter can be created in this way. Filter kernel: impulse response employed in this manner; filter designers provide a unique name. The digital filter is another method as well. Applying a filter weights each input sample in the convolution product and computes it by summing them all. Using the previously computed values, recursive filters and the input points produce these representatives. A collection of coefficients based on the idea of recursive filters replaces a filter kernel. A recursive filter provides motivation by feeding the output of the filter. With exponentially decreasing amplitude, impulse responses are sinusoidal. So, they have endless long impulse reactions. So- called recursive filters, or infinite impulse response filters, are IIR filters. Convolution filters, Finite Impulse Response or FIR filters, are seen throughout the course of. When the system's output is motivated by the input impulse response. Input, sometimes known as an edge, is a step in the same way; the step reaction is the output. Comprehensive induction process, so the impulse response is a significant stage in the reaction. A step in the food clears the filter; notice the waveform; or (1) to combine the impulse response: This stage offers two approaches to identify a reaction. While the discrete codes are utilised, mathematically valid codes to be used for the integration of continuous, discrete integration, i.e., a running total. Impulse response frequency response (FFT algorithm employing the technique) DFT appears to be taking

## 2. LITERATURE SURVEY

A notable change in optimisation theory has been seen since the start of the nineteenth century. Until this century, classical linear programming and conventional non-linear optimisation methods like Lagrange's Multiplier, Bellman's principle and Pontyagrin's principle were common. Regrettably, the optima on uneven non-linear surfaces cannot be determined using these derivative-based optimisation methods any more. The evolutionary algorithms research community has already suggested one approach to this issue. Genetic algorithm (GA), enunciated by John Holland in the year 1975, is one such popular algorithm which is based on the concept of "survival of the fittest" by Charles Darwin [2]. Holland and his colleagues, particularly Goldberg and Dejong, popularised the GA theory and showed how biological crossings and chromosome mutations might be realised in GA to enhance the solution quality over repeated iterations. Design methods based on Genetic Algorithm (GA) are quite popular for FIR filter synthesis. GA-based minimum phase digital FIR filter design has also been effectively detailed in [3]. The mean squared error (MSE) function is applied to get the optimal-pass band and stop-band responses; the mean absolute error (MAE) is used to maximise the transition band response. According to [4], GA has been used to optimise the design of infinite impulse response (IIR) filters.

Proposed in [5], the IIR filter design under the mixed criterion of H2 norm and ∞ norm is realised using GA. Studies have demonstrated that the filter created by GA outperforms traditional Butterworth filter in either design technique optimisation capacity or designed filter performance. These methods increase the signal to noise ratio (SNR) and bring the frequency domain performance closer to the theoretical one.

Published in [6], a novel approach for creating nonrecursive and recursive frequency sampling filter Investigated is the use of a hybrid real-coded GA to maximise stop band attenuation by means of optimisation of transition sample value. A change lets the coefficient word length be optimised simultaneously, hence lowering total number of design steps and streamlining the design process. The methods can reliably optimise filter with as many as six transition samples. The methods proposed in this article might provide the foundation for combining several of the optimisations. Research on enhancing this integration by means of a binary coded GA to optimise nonlinear phase, quantised coefficient FIR filter are presented together with a GA viewpoint problem analysis [7]. For high speed low complexity filter design, it is common practice to constrain the filters‟ coefficient to be power of two or a sum of power of two terms (p2), avoiding the full multiplication [8]. Sometimes, tapped connectivity of several sub filters is employed to improve ripple and stop band attenuation performance. Polynomial sharpening methods are an extension of the straightforward cascade structures appropriate for hardware implementation. The design of p2 sharpening filter based on a specific genetic algorithm has been proposed in the above article. The suggested method maximises the sharpening polynomial coefficient stated as p2 terms as well as the FIR sub-filter. This allows getting better performances than the classical p2 design techniques when FIR filters with long impulse response are involved. Using this specific genetic algorithm with a particular free parameters encoding around a set of suitable leading values, allows obtaining a very high reduction of the computational cost. It has been shown in [9] that optimizing both the polynomial and the filter coefficient allow obtaining very good performances; sometimes better that the simple infinite precision sharpening techniques.

The algorithm Particle Swarm Optimization (PSO) was inspired by biological and sociological motivations and can take care of optimality on rough, discontinuous and multimodal surfaces. The aim of this optimization technique is to determine the best-suited solution to a problem under a given set of constraints. Several researchers over the decades have come up with different solutions to linear and nonlinear optimization problems. The optimization problem, now-a-days, is represented as an intelligent search problem, where one or more agents are employed to determine the optima on a search landscape, representing the constrained surface for the optimization problem [8]. In mid 1990s, Eberhart and Kennedy enunciated an alternative solution to the complex non-linear optimization problem by emulating the collective behaviour of bird flocks, particles and called their brainchild Particle Swarm Optimization (PSO) [10]. Particle swarm optimization techniques can be used to design infinite impulse response (IIR) filter [11]. It is observed that the particle swarm algorithms are able to converge very rapidly when the error surface is relatively constant. This is the fundamental advantage of particle swarm algorithm for online adaptive filtering. Though standard PSO exhibits a fast convergence initially, it fails to improve further because swarm quickly becomes stagnant, converging to a suboptimal solution. However, with the same set of algorithm parameters, the Modified Particle Swarm Optimization (MPSO) particles do not stagnate, allowing it to reach the noise floor along with the GA. Particle Swarm Optimization techniques has also been used to design frequency sampling FIR digital filter. By applying PSO to optimize transition sample values, the maximum stop band attenuation in FIR filter is obtained. It has been experimented that to design FIR filter, PSO is more superior to GA not only in the convergence speed but also in the performance of filter [12]. Stable IIR digital filters with non-standard amplitude characteristics can be designed using particle swarm optimization techniques [14]. PSO has been the newly elaborated techniques for optimization of multi-modal functions. Design of IIR digital filters with non-standard amplitude characteristics which considerably differ from typical Butterworth, Chebyshev, and Cauer approximations, is possible using presented method. The IIR digital filter with linearly falling amplitude characteristics is designed with the use of proposed method. The filter is stable and fulfils all prescribed design assumptions. Differential Evolution (DE) algorithm has been emerged as a very competitive form of evolutionary computing more than one decade ago. The first written article on DE appeared as a technical report by R. Storn and K. V. Price in 1995. One year later, the success of DE was demonstrated at the First International Contest on Evolutionary Optimization in May 1996, which was held in conjunction with the 1996 IEEE International Conference on Evolutionary Computation (CEC). DE finished third at the 1st ICEO, which was held in Nagoya, Japan. In quick times, DE has turned out to be the best evolutionary optimization algorithm for solving real valued test function. Price presented DE at the second International contest on Evolutionary Optimization in 1997 and it turned out as one of the best among the competing algorithms. In 2005 CEC competition on real parameter optimization, on 10- D problems classical DE secured 2nd rank and a Self adaptive DE variant called SaDE secured 3rd rank although they performed poorly over 30-D problems. Later depending upon many improved DE variants some optimization techniques has been proposed in the period 2006-2009 such as , improved SaDE, jaDE ,Opposition Based DE (ODE) , DE with global and local neighbourhoods (DEGL) and so on A linear phase finite impulse response (FIR) filter has been designed using Differential Evolution Particle Swarm Optimization (DEPSO) technique as mentioned in. Here two different fitness functions have been studied and experimented, each having its own significance. The first study considers a fitness function based on the pass-band and stop-band ripple, while the second study considers fitness function based on the mean squared error between the actual and the ideal filter response. DEPSO seems to be one of the promising tools for FIR filter design especially in a dynamic environment where filter coefficient have to be adapted and fast convergence is of paramount significance. In recent times, it has been observed that the DE algorithm based on reverse gene can also be applied for the design of digital filter [38]. It can produce new chromosomes in every generation by combining reversed gene of special chromosome into a single entity.

## 3.PROPOSED SYSTEM

Finite impulse response filters, or FIR filters, are non-recursive type filters with finite period impulse response. IIR, or Infinite Impulse response, are recursive filters with infinite period impulse response. A non-recursive type filter is one whose output sample relies on both the current input sample and the prior input sample. Recursive type filters use one or more of their outputs as an input. FIR filter's impulse response is of finite period since it settles to zero in finite time. FIR filter is significant since it can have no phase distortion. FIR filters are preferred over IIR in digital processing applications. With precise linear phase, FIR filters are always stable and simple to construct. FIR filters are also less susceptible to quantisation mistakes in the filter.

**FIR Filter**

In digital signal processing systems, finite impulse response (FIR) filter is quite significant. Its output sample relies on the current input sample and past input sample. Thus, it is a non-recursive filter. It is steady and may simply be constructed with precisely linear phase.

The difference equation for the FIR filter is given as

y(n) = b0 x(n)+b1x(n-1)
+......................................... + bN x(n-N)

It defines the relation of the input signal to the output signal.

It can also be expressed as = 0 otherwise

Where wα(n) is the frequency response

The equation for Hamming window can be obtained by substituting α = 0.54 in equation 1.

wHα(n) = 0.54 + 0.46

for – (N -1)/2 ≤ n ≤ (N-1)/2.

= 0 otherwise

**C. Kaiser Window**

Kaiser Window generates a sharp central peak. It reduces side lobes and transition band is narrowed.

The Kaiser window is given by

$$y(n) = \sum_{i=0}^{N} b_i\, x(n-i)$$

$$\frac{Io[\alpha\sqrt{1-(2n/N-1)^2}]}{Io(\alpha)}$$

Where x(n) is the input signal, y(n) is the output signal, bi is the filter coefficient and N is the filter order

wk(n) =

for $|n| \leq \frac{N-1}{2}$

**B. Hamming Window**

Hamming window is a type of raised cosine window. The window sequence of cosine window is of the form

$$w\alpha(n) = \alpha + (1-\alpha)\cos\frac{2\pi n}{N-1}$$

for – (N -1)/2 ≤ n ≤ (N-1)/2. = 0 otherwise

Where α is the adjustable parameters and Io(x) is the modified zeroth- order Bessel function.[13]

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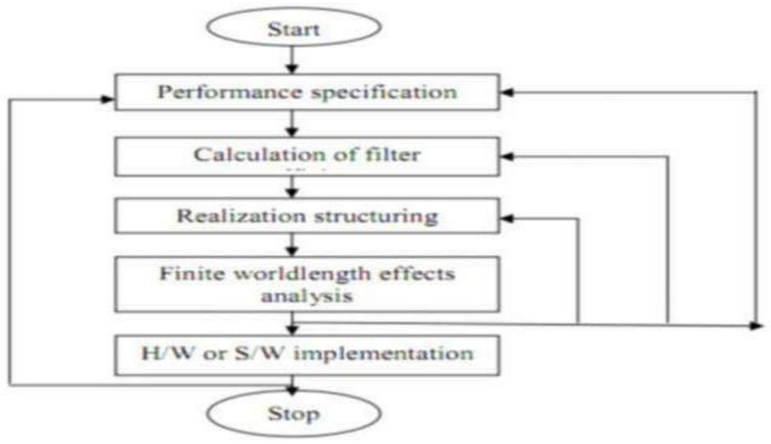
FIR Filter Design




Summary of design stage for digital filter

Fig 1: Proposed System

accurate signed multiplier. The delay of the 8-bit multiplier can be lowered, though, by cutting the multiplier

## 3. RESULTS AND DISCUSSION

Magnitude and phase responses of a 20th order digital low pass filter using Hamming, Rectangular, and Kaiser window functions are observed and investigated as shown in Figures 5.1, 5.2 and 5.3. The input signal frequency is below the cut off frequency i.e., 2 MHz, so we get to observe the output signal using Hamming window. The input signal frequency is below the cut off frequency horizontally in the centre and separating it into two smaller 8×4 groups.

i.e., 2 MHz, so we get to observe the output signal using rectangular window. The input signal frequency is below the cut off frequency i.e., 2 MHz, so we get to observe the output signal using Kaiser Window. The input signal frequency is above the cut off frequency i.e., 2 MHz, so we cannot observe the output signal using Hamming window. The input signal frequency is above the cut off frequency i.e., 2 MHz, so we cannot observe the output signal using rectangular window.The input signal frequency is above the cut off frequency i.e., 2 MHz, so we cannot observe the output signal using Kaiser Window.

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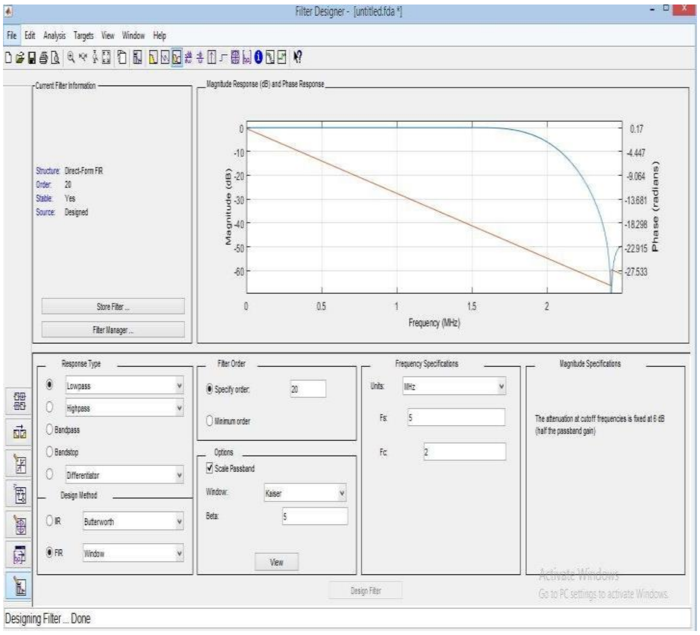


Figure 2: FIR LPF using Hamming window

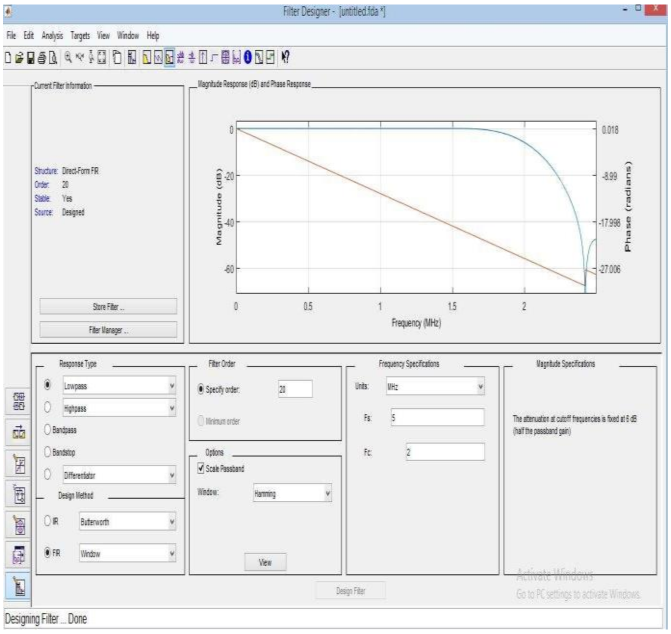


Figure 4: FIR LPF using Kaiser Window

Table 1: Filter Specifications

| Parameters | Values |
|---|---|
| Filter type | Low pass |
| Design method | FIR Window (β =5 for Kaiser) |
| Filter order | 20 |
| Cut off frequency | 2 MHZ |
| Sampling frequency | 5 MHZ |

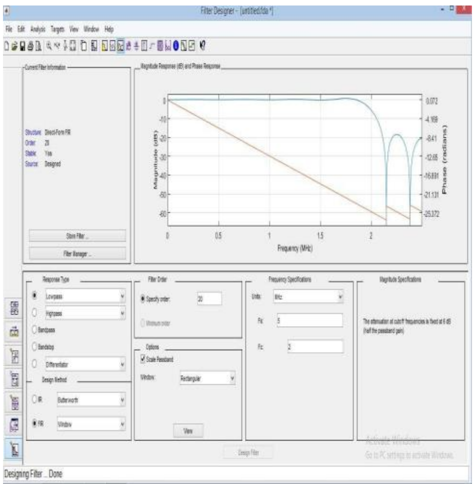

Figure 3: FIR LPF using rectangular window

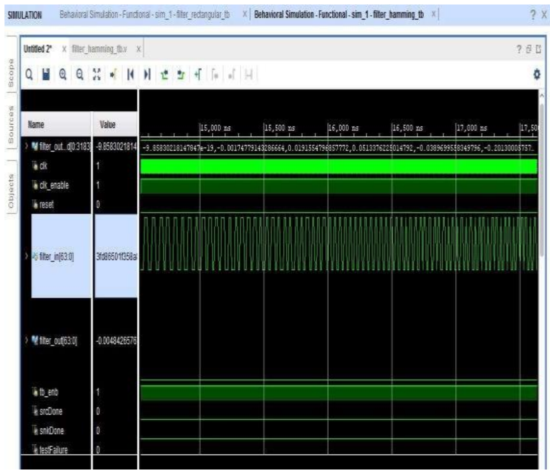

Figure 5: Simulation result of the FIR LPF using Hamming window
Figure 6: Simulation result of the FIR LPF Using rectangular window

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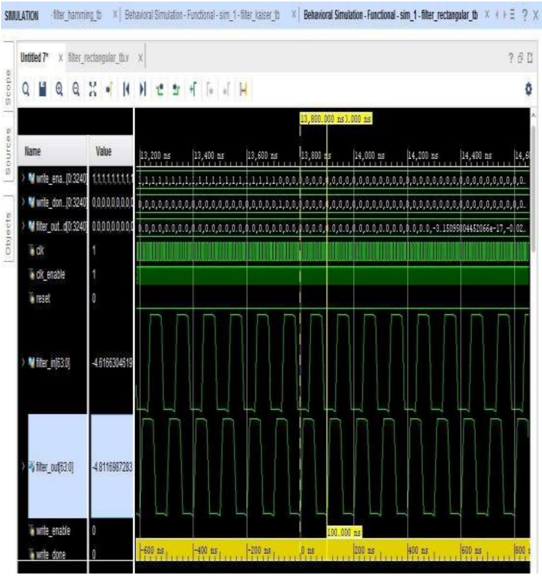
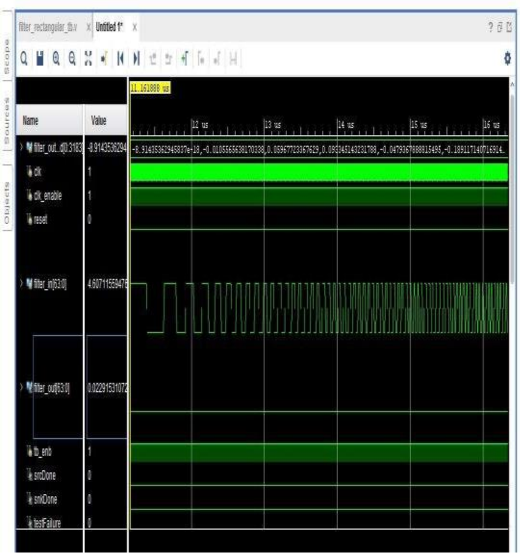


Figure 7: Simulation result of the FIR LPF using Kaiser Window
Figure 8: Simulation result of the FIR LPF at cut off frequency using Hamming Window

Table 2: Comparison of 20th-order fir LPF using different window function

| **Window technique** | **Filter order** | **No. of side lobes** | **Transition width** |
|---|---|---|---|
| Hamming window | 20 | 1 | Moderate |
| Rectangular window | 20 | 2 | Narrowest |
| Kaiser Window | 20 | 1 | Variable |



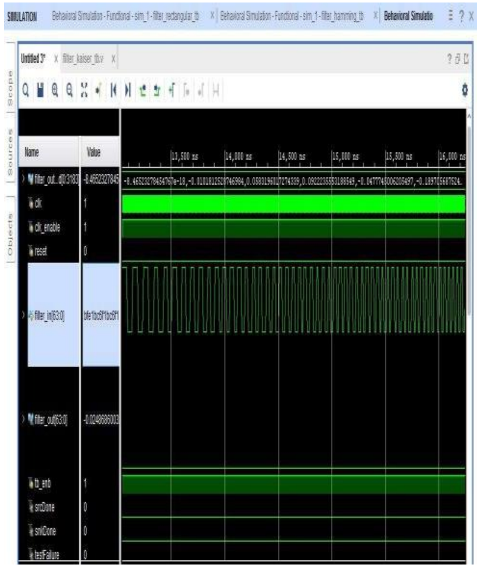


Figure 9: Simulation result of the FIR LPF at cut off frequency using Rectangular Window

Figure 10: Simulation result of the FIR LPF at cut off frequency using Kaiser Window

## 4. CONCLUSION

FIR filters, which are stable and have linear phase properties, are extensively used in wired and wireless communications, video, audio processing, and mobile devices. From software level to the hardware level, this work presents a new design strategy for an efficient FIR digital filter. The main goal is to include all the areas used in the efficient hardware realisation of filters i.e. design approach, selection of structure and the algorithm to lower the arithmetic cost of FIR filtering.

Using feed forward pipelining retiming techniques, low pass FIR filter is built. Broadcast and non-broadcast low pass digital FIR filter architectures are used to verify the rise in area and speed using the present broadcast, non broadcast FIR filter. The results of synthesis and simulation are discussed in full. When compared to all other constructed designs, feed forward non-broadcast FIR filter (Kaiser, where
\u03b2=5) structure has quickest speed

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