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Optimal Interpolated FIR (IFIR) Digital Filter Design with Spectral Estimation for Radar and Sonar System

Sandeep Kaur[1], Mandeep Singh Saini[2], Palvee[3]
[1] M.Tech (E.C.E), IITT Engineering College, Pojewal, P. T. U. (Punjab)
[2, 3] M.Tech (E.C.E), Guru Nanak Dev Engineering College, Ludhiana, P. T. U. Regional Centre, (Punjab)

ABSTRACT:-

Finite impulse response (FIR) filters are highly desirable in digital filter design because of their inherent stability and linear phase. However, when narrow transition band characteristics are required, they typically have a much higher filter order than their infinite impulse response counterparts with equivalent magnitude spectrums.

Widowing methods and the frequency response sampling method were presented with examples, but they were improved upon with the IFIR design technique.

The results shown that the computational cost of a Interpolated FIR (IFIR) is less with comparing the computational cost of a signal stage FIR filter and Multistage FIR filter which can be used both at the receiver and the transmitter.

Keywords: **Bandwidth (BW), Inter Symbol Interference (ISI), Interpolated filtering (IFIR), Finite impulse response filter (FIR)**

I. INTRODUCTION

Finite Impulse Response (FIR) filters are often used in phase-sensitive applications because they can always be designed to have linear phase. They are also inherently stable because all of the poles lie at the origin.

FIR filters have the major drawback of having a much higher filter order than their IIR counterparts with equivalent magnitude spectrums [6].

In order to reduce the extra computational complexity that accompanies high filter orders, implementations such as the Interpolated Finite Impulse Response (IFIR) method were created. The IFIR technique has been shown to significantly reduce the computational complexity of practical FIR filters.

The processes of sampling rate reduction (often called decimation) and sampling rate increase (or interpolation).

Sampling rate increase and sampling rate reduction are basically interpolation processes and can be efficiently implemented using finite impulse response (FIR) digital filters.

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The IFIR approach results in a Two-stage decimator/interpolator. For the multistage approach, the number of stages can be either automatically optimized or manually controlled. But multirate/multistage design introduces the most delay as compare with IFIR Design.

II. FILTER ORDER ESTIMATION

Kaiser J. (1976) developed formula to determine the filter order. But they do not always provide the correct filter order. The smallest integer value that lies above the estimation should be checked for accuracy after the implementation. The parameters given include normalized pass band edge angular frequency ω_p and normalized stop band edge angular frequency ω_s, peak passband ripple $\boldsymbol{\delta}_p$ and peak stop band ripple δ_s [4].

$$\boldsymbol{N} \cong \frac{-20\,\boldsymbol{log}\sqrt{\boldsymbol{\delta_p}\,\boldsymbol{\delta_s}}-\mathbf{13}}{\mathbf{14.6}\left(\dfrac{\boldsymbol{\omega_s}-\boldsymbol{\omega_p}}{\boldsymbol{2\pi}}\right)} \qquad (1)$$

Where N is filter order.

III. PROBLEM FORMATION

This filter can be designed using the window method. Hamming window or a Dolph-Chebyshev window can be used to design the specified filter shown (MATLAB, 2007) in Fig.1.

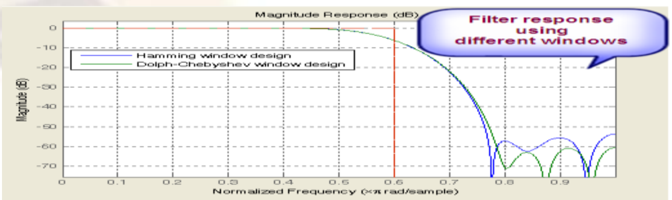

Fig. 1. Filter response using different windows

The Hamming window is defined as follows:

$$W\,[n] = \begin{cases} 0.54 - 0.46\,cos\,\dfrac{2\pi n}{M-1}, 0 \le n \le M-1 \\ 0 \end{cases} \quad (2)$$

To see what effects these windows have on the magnitude response of a filter, Kaiser Window design are determine a suitable filter order shown (MATLAB, 2007) in fig. 2.

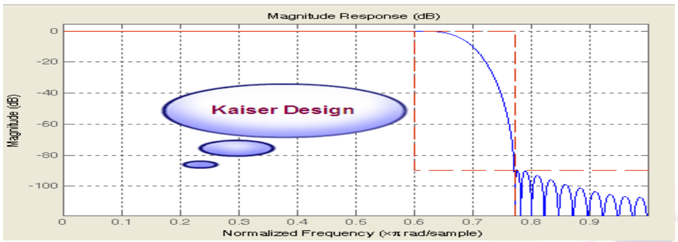

Fig. 2. Kaiser Design

The Kaiser Window design is not an optimal design. Because Filter order of Kaiser Window design is (68) as compare with Park McClellan Algorithm design result in the filter with the smallest possible order is (53).

So it can be concluded that McClellan based Equiripple design is optimal in terms of Filter Order (53).

We can still use equiripple designs to decrease filter order (20) but we loose control over the transition width which will increase shown (MATLAB, 2007) in Fig. 3.

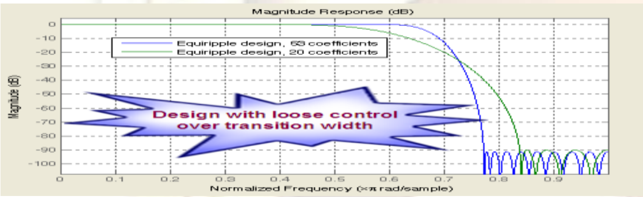

Fig. 3. Design with loose control over transition width

Another option when the number of coefficients is set is to maintain the transition width at the expense of control over the pass band ripple/stop band attenuation shown (MATLAB, 2007) in Fig. 4.

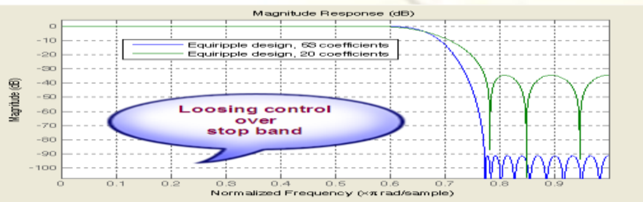

Fig. 4. Loosing control over stop band

Stop-band Attenuation Control is possible to increase the attenuation in the stop-band while keeping the same filter order and transition width by shown in Fig. 5.

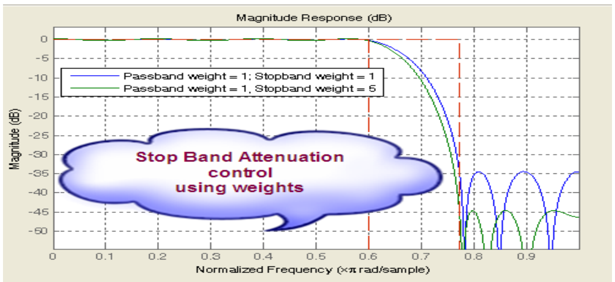

Fig. 5. Stop band attenuation control using weights

Another possibility is to specify the exact stop band attenuation desired and Loose control over the pass-band ripple shown (MATLAB, 2007) in Fig. 6.

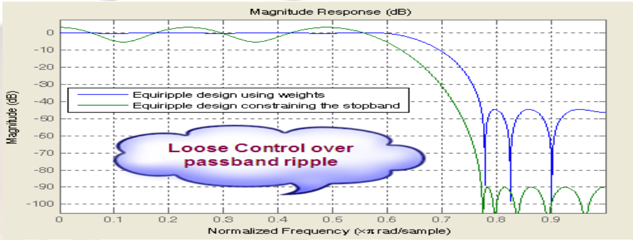

Fig. 6 Loose control over pass band Ripple

So we conclude that selection of parameter is import for communication filter design optimal equiripple linear phase method is best for FIR filter design.

IV. SOLUTION OF PROBLEM

A. Equiripple FIR Filter

When an equiripple filter is desired, a computer-aided iterative approach is usually employed to reach the specifications within a certain error (ε). The resultant FIR filters of different algorithmic approaches are called equiripple because their minimized weighted error function ε(ω) exhibits an equiripple behavior. The most common approach is to use the Parks-McClellan algorithm.

B. Equiripple IFIR Filter

In it's simplest form, the IFIR design can be thought of as a cascade of two filters. This is depicted in Fig. 7 and expressed in equation 3.

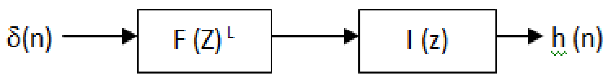

Fig. 7 IFIR filters structure.

$$H(z) = F(z^L)\, G(z) \qquad (3)$$

F (z) is called the shaping filter because it determines the shape of the resulting filter. F (z) ^{L} is an upsampled version of this shaping filter. I (z) is known as the imaging filter or interpolator because it reconstructs the sparse impulse response given by F (z) ^{L} and suppresses the undesired pass band images that result from the up sampling. This technique greatly reduces the number of multipliers needed to meet given specifications. Where

ω_p = Pass band edge angular frequency

ω_s = Normalized stop band edge angular frequency

$\boldsymbol{\delta_p}$ =Peak pass band ripple

δ_s =.Peak stop band ripple

When the IFIR design is used, first an up sampling factor, L, must be found. From [1], the largest value of L is given by

$$L_{max} = \left[\frac{\pi}{\omega_s}\right] \tag{4}$$

Improvement of IFIR Filter design equation given as

$$\boldsymbol{L_{max}} = \left[\frac{\boldsymbol{2\pi}}{\boldsymbol{\omega_p + \omega_s + \sqrt{2\pi(\omega_s - \omega_p})}}\right] \tag{5}$$

C. Spectrum Estimation

Spectrum estimation is useful in a variety of disciplines i. e communication Engineering, it is helpful in detecting the signal component (carrier) which has the noise component in it. In Radar and Sonar, it is useful in detecting the Target.

The estimate for Power Density Spectrum is called the Periodogram.

The Periodogram for a sequence [x_1, ... , , x_N] is given by the following formula:

$$S\ (e^{jw}\ ) = \frac{1}{2\pi N}\left|\sum_{n=1}^{N} x_n\ e^{-jwn}\right|^2 \tag{6}$$

Where ω is in units of radians/sample.If we define the frequency variable in Hz, the periodogram is defined as:

$$S(f) = \frac{1}{F_s N}\left|\sum_{n=1}^{N} x_n\ e^{-j\left(2\pi f/F_s\right)}\right|^2 \tag{7}$$

Where Fs is the sampling frequency. The periodogram is an estimate of the PSD of the signal defined by the sequence [x_1, ... , x_N].

If you weight your signal sequence by a window [w_1, ... , w_N], then the weighted or modified periodogram is defined as

$$S(f) = \frac{1}{2\pi N} \left|\sum_{n=1}^{N} x_n \omega_n\ e^{-j(2\pi n)}\right|^2 \Big/ \frac{1}{N}\sum_{n=1}^{N} |\omega_n|^2 \tag{8}$$

Cancelling the common factors and denoting the squared l^2 norm of the window sequence by $\|\omega\|^2$ the modified periodogram can be simplified as:

$$S\ (e^{jw}\ ) = \frac{1}{2\pi}\ \frac{\left|\sum_{n=1}^{N} x_n\ e^{-jwn}\ \right|^2}{\|\omega\|^2} \tag{9}$$

V. Results and Discussion

THE DISADVANTAGE OF FINITE IMPULSE RESPONSE (FIR) FILTERS IS THAT THE FILTER ORDER TENDS TO GROW INVERSELY PROPORTIONAL TO THE TRANSITION BANDWIDTH OF THE FILTER.

In thesis paper, FIR filter Fp = 0.01, Fs = 0.105, δp = 0.001, δs = 0.001 and with up sampling factor is 3 performance shown in Fig. 7.

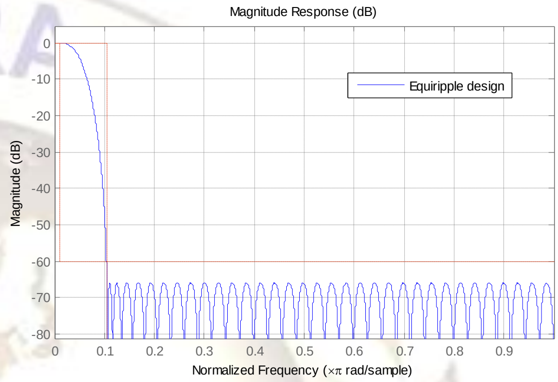

Fig. 7. Equiripple FIR Design

The IFIR design algorithm achieves an efficient design for the above specifications in the sense that it reduces the total number of multipliers required. To do this, the design problem is broken into two stages, a filter which is upsampled to achieve the stringent specifications without using many multipliers. The IFIR Filter design is shown in Fig. 8.

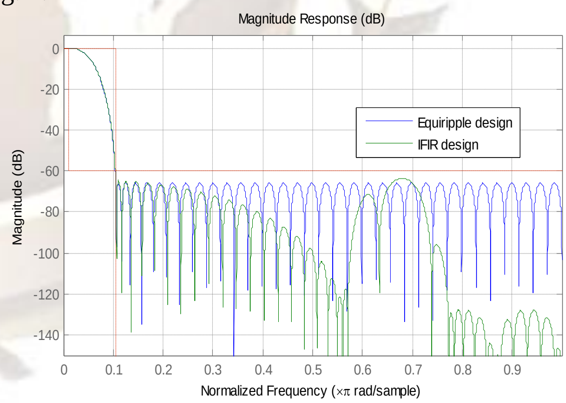

Fig. 8. Equiripple and IFIR Design with up sampling factor 3

We can see, that we can control the up sampling factor. if we wanted to up sample by 8 rather than 3 then performance of all design shown in Fig. 9.

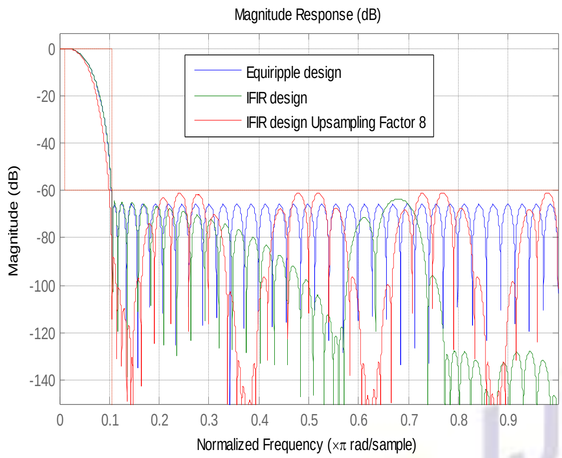

Fig. 9. IFIR Filter with up sampling factor 8

It is possible to design the two filters used in IFIR conjunctly. By doing so, we can save a significant number of multipliers at the expense of a longer design time (due to the nature of the algorithm, the design may also not converge altogether in some cases) automatically determine the best factor shown in Fig.10.
For this design, the best up sampling factor found was 5. The number of non-zero multipliers is now only 27.

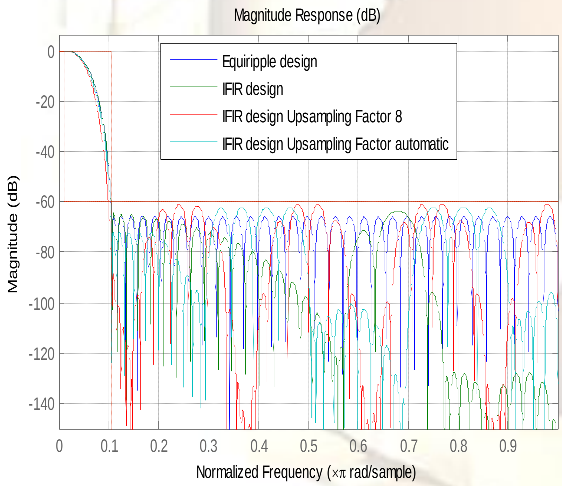

Fig .10. IFIR Automatically determine the best sampling factor 5

By using multirate/multistage techniques which combine decimation and interpolation we can also obtain efficient designs with a low number of MPIS. For decimators, the number of multiplications required per input sample (on average) is given by the number of multipliers divided by the decimation factor.

But Notice that the stop band attenuation for the multistage design is about double that of the other designs and Also notice the pass band gain for this design is no longer 0 dB. This is due to the use of interpolators as part of the design. Each interpolator has a nominal gain equal to its interpolation factor. The total interpolation factor for the 3 interpolators is 6, which is the gain (in linear units) of the overall filter shown in Fig. 11.

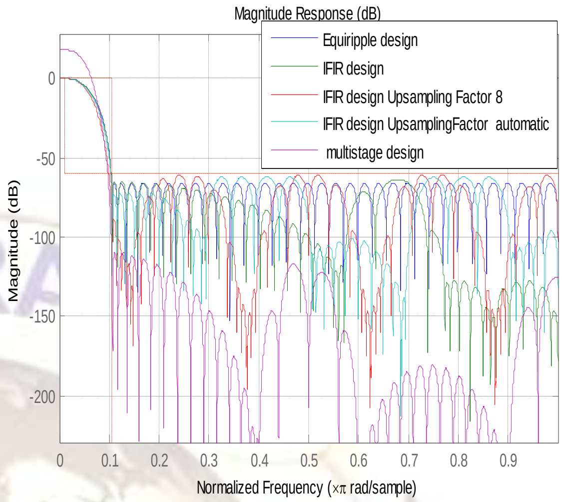

Fig. 11.Mulitstage Design

As comparisons all Filters we can compute the group delay for each design. Notice that the multirate/multistage design introduces the most delay. The IFIR design introduces more delay than the single-stage equiripple design, but less so than the multirate/multistage design.

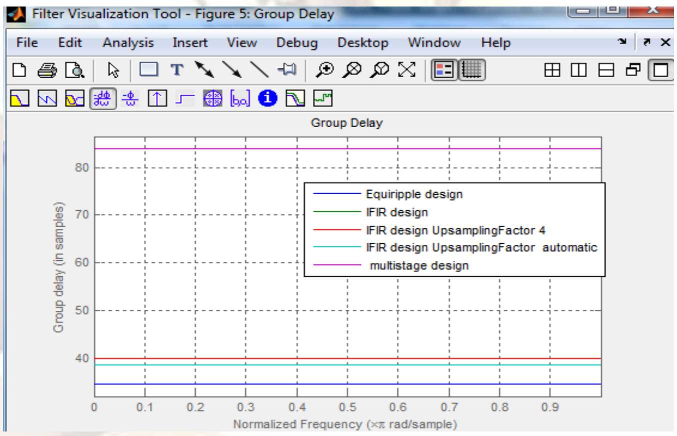

Fig. 11.Group Delay

Then we filtering a Signal the IFIR and multistage/multirate design perform comparably to the single-stage equiripple design while requiring much less computation shown in Fig. 12.

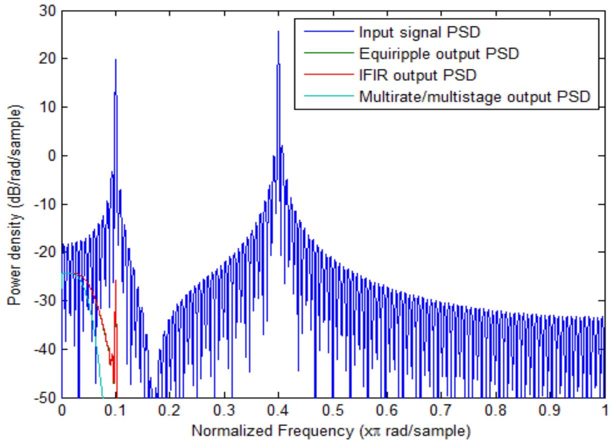

Fig.12. Power Density Spectrum

. Signal stage FIR filters Length or number of multiplier is 70 and density factor 16 with stop band attenuation 60 db shown as Table 1

TABLE: 1
SIGNAL STAGE FIR DESIGN

```
Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure  : Direct-Form FIR
Filter Length     : 70
Stable            : Yes
Linear Phase      : Yes (Type 2)

Design Method Information
Design Algorithm : equiripple

Design Options
Density Factor : 16
Maximum Phase  : false
Minimum Order  : any
Minimum Phase  : false
Stopband Decay : 0
Stopband Shape : flat
Uniform Grid   : true
```

```
Stopband Atten.    : 60 dB

Measurements
Sampling Frequency : N/A (normalized frequency)
Passband Edge      : 0.01
3-dB Point         : 0.045602
6-dB Point         : 0.056041
Stopband Edge      : 0.105
Passband Ripple    : 0.0088111 dB
Stopband Atten.    : 65.8569 dB
Transition Width   : 0.095

Implementation Cost
Number of Multipliers            : 70
Number of Adders                 : 69
Number of States                 : 69
Multiplications per Input Sample : 70
Additions per Input Sample       : 69
```

If we compare Table 1 no of multipliers (70) with Table 2 no of multipliers is only (35) with up sampling factor is 3. So IFIR design is good as compare with FIR Design.

TABLE: 2
IFIR design upsampling factor 3

```
% -------------------------------
% IFIR design
% -------------------------------

Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure  : Cascade
Number of Stages  : 2
Stable            : Yes
Linear Phase      : Yes (Type 2)

Design Options
JointOptimization : false
UpsamplingFactor  : 3

Design Specifications
Sampling Frequency : N/A (normalized frequency)
Response           : Lowpass
Specification      : Fp,Fst,Ap,Ast
```

```
Stopband Atten.    : 60 dB

Measurements
Sampling Frequency : N/A (normalized frequency)
Passband Edge      : 0.01
3-dB Point         : 0.04573
6-dB Point         : 0.05624
Stopband Edge      : 0.105
Passband Ripple    : 0.010607 dB
Stopband Atten.    : 63.6182 dB
Transition Width   : 0.095

Implementation Cost
Number of Multipliers            : 35
Number of Adders                 : 33
Number of States                 : 77
Multiplications per Input Sample : 35
Additions per Input Sample       : 33
```

If we increase up samlping factor (8) as shown in Table 3 then no of multipliers is (56) too large as compare with Table 2.

TABLE: 3
IFIR design upsamlping factor 8

```
% -------------------------------
% IFIR design Upsampling Factor 8
% -------------------------------

Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure  : Cascade
Number of Stages  : 2
Stable            : Yes
Linear Phase      : Yes (Type 2)

Design Options
JointOptimization : false
UpsamplingFactor  : 8

Design Specifications
Sampling Frequency : N/A (normalized frequency)
Response           : Lowpass
Specification      : Fp,Fst,Ap,Ast
```

```
Stopband Atten.    : 60 dB

Measurements
Sampling Frequency : N/A (normalized frequency)
Passband Edge      : 0.01
3-dB Point         : 0.042746
6-dB Point         : 0.052133
Stopband Edge      : 0.105
Passband Ripple    : 0.0082216 dB
Stopband Atten.    : 60.9808 dB
Transition Width   : 0.095

Implementation Cost
Number of Multipliers            : 56
Number of Adders                 : 54
Number of States                 : 103
Multiplications per Input Sample : 56
Additions per Input Sample       : 54
```

If we select up sampling factor automatic it is (5) and no of multipliers is only (27) is less if compare other design.

So Table 4 Optimal Design is good with interpolation factor is 5 as compare to other Design.

TABLE: 4
IFIR design upsamlping factor automatic is 5

```
% -------------------------------
% IFIR design UpsamplingFactor  automatic
% -------------------------------

Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure  : Cascade
Number of Stages  : 2
Stable            : Yes
Linear Phase      : Yes (Type 2)

Design Options
JointOptimization : true
UpsamplingFactor  : 5

Design Specifications
Sampling Frequency : N/A (normalized frequency)
Response           : Lowpass
Specification      : Fp,Fst,Ap,Ast
```

```
Stopband Atten.    : 60 dB

Measurements
Sampling Frequency : N/A (normalized frequency)
Passband Edge      : 0.01
3-dB Point         : 0.044285
6-dB Point         : 0.054335
Stopband Edge      : 0.105
Passband Ripple    : 0.004204 dB
Stopband Atten.    : 62.1757 dB
Transition Width   : 0.095

Implementation Cost
Number of Multipliers            : 27
Number of Adders                 : 25
Number of States                 : 77
Multiplications per Input Sample : 27
Additions per Input Sample       : 25
```

Multistage FIR filter no of multipliers is (64), So The Table 5 design is not suitable Design.

TABLE 5
Multistage design

```
% -------------------------------
%  multistage design
% -------------------------------

Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure  : Cascade
Number of Stages  : 6
Stable            : Yes
Linear Phase      : Yes (Type 1)

Design Options
HalfbandDesignMethod : equiripple
NStages              : auto
UseHalfbands         : false

Design Specifications
Sampling Frequency : N/A (normalized frequency)
Response           : Lowpass
```

```
Stopband Atten.    : 60 dB

Measurements
Sampling Frequency : N/A (normalized frequency)
Passband Edge      : 0.01
3-dB Point         : 0.03813
6-dB Point         : 0.046395
Stopband Edge      : 0.105
Passband Ripple    : 0.009095 dB
Stopband Atten.    : 127.368 dB
Transition Width   : 0.095

Implementation Cost
Number of Multipliers            : 64
Number of Adders                 : 55
Number of States                 : 42
Multiplications per Input Sample : 15.25
Additions per Input Sample       : 12.625
```

TABLE: 6
IMPLEMENTATION COST COMPARISION

Implementation cost comparison					
Technique	Number of multipliers MUL	Number of Adders ADD	Number of states STATE	Multiplication per input sample MULT/ SAMPLE	Additions per Input Sample ADD/ SAMPLER
Single stage FIR	**70**	**69**	**69**	**70**	**69**
IFIR Filter L=3	**35**	**33**	**77**	**35**	**33**
IFIR Filter L=8	**34**	**32**	**80**	**34**	**32**
IFIR Filter L=5	**27**	**25**	**77**	**27**	**25**
Multi stage FIR	**64**	**55**	**42**	**15.25**	**12.625**

VI. CONCLUSIONS

This paper has investigated many different design techniques for linear-phase finite impulse response FIR filters. FIR filters have much greater computational complexity and large cost than Interpolated (IFIR) filter.

In this paper, we have Investigated Widowing methods and the frequency response sampling method was presented with examples, but they were improved with the IFIR design technique. We has also compare Multirate FIR Filter with IFIR Filter but result shows IFIR filter cost is less as compare to other design.

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Authors:

MANDEEP SINGH SAINI Mr. Saini received his B.TECH From Nagpur University during 2001. They obtain M.Tech from GNDEC, Ludhiana (Punjab Technical University) during Apr. 2012. His field of interest is Digital Signal Processing and VLSI Architecture Design. He has published 20 Papers in International journal and 01 paper in Int. conference.

SANDEEP KAUR

Miss Sandeep Kaur has passed Diploma in ECE in the year 2007 from Govt. Polytech. College for Girls, Ludhiana. She has passed B.Tech in ECE in the year 2010 from Bhutta college of Engg. & Tech. Ludhiana. Now, she is doing M.Tech in ECE from IITT College, Pojewal.