Interpolated Finite Impulse Response (IFIR) Filter Approach: A Case Study

B. N Jagadale
Dept. of Electronics, Kuvempu University
Shankaragatta-577451, INDIA

ABSTRACT
The finite impulse response filters are inherently stable and have linear phase response property. The main drawback of these filters is lies in requirement of higher orders for similar magnitude response compared to Infinite Impulse response filters. Also the amount of computational complexity needed for the implementation of the filter, especially for the filters with narrow transition band is much higher. In order reduce the computational complexity of narrowband FIR filters, the interpolated FIR (IFIR) filter technique is used.

Keywords
Linear phase, Narrowband filter, FIR filter, Equiripple, Pass band.

1. INTRODUCTION
Finite impulse response (FIR) filters are considered to be more useful as compare to the infinite impulse response (IIR) filters due its desirable characteristics like better stability and linear phase [1-5]. However, the computational requirements of a FIR filter along with order of the filter for a given specifications are usually greater than that of an IIR filter for the same requirements. This is has very significance in design of narrowband FIR filter, where filter order can become very large. It has been proposed by Neuvo et al. [6] that an alternative approach to narrowband FIR filter design uses two filters in a cascade arrangement as shown in Fig. 1.

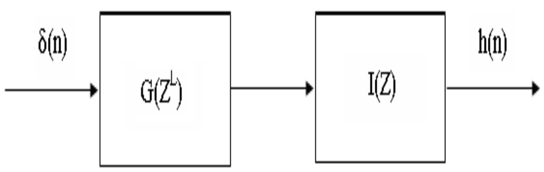

Fig 1: Interpolated FIR filter

This method helps in reducing the number of multipliers needed for the design and the implementation, also it still meets the the narrowband filter specifications. The advantage of this approach is reduction in the order of the filter. i.e. The combined order of the two filters is less than that of the single [7-10] narrowband filter.

This paper is organized as follows. Section 2, gives brief review of narrowband filters design. The design procedure for IFIR filter is presented in section 3. Section 4 gives Results and discussion and conclusions are given in Section 5.

2. NARROWBAND FILTER DESIGN
In this section principle of narrowband low pass filter, whose cut-off frequencies are considerably lower than the sampling rate is discussed. Figure 1 shows two stages of IFIR filter. Both filter stages are processed at the same sampling rate, that means the sampling rate of output is same as that of input [11-14].

The transfer function $G(z^L)$ is a function of z^L in the first stage. This filter can be implemented with a transversal filter with the transfer function $G(z)$. Here each delay element z^{-1} is replaced by a series of L delay elements as shown in figure 2. This is equivalent to inserting L-1 zeroes between the original coefficients of. $G(z)$

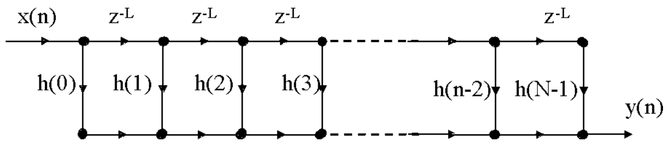

Fig 2: Transversal (L- delays)

Here, from $G(e^{j\Omega})$ to $G(e^{jL\Omega})$, the frequency response is compressed by a factor of L and L-1, image frequency responses are produced as shown in Figure 3.

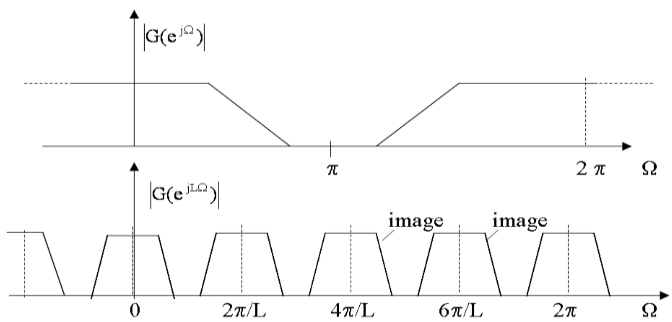

Fig 3: Amplitude frequency responses $|G(e^{j\Omega})|$ and $|G(e^{jL\Omega})|$

When same number of coefficients are used, the frequency response of $G(e^{jL\Omega})$ is L times narrower than the frequency response $G(e^{j\Omega})$ of the prototype. And the filter $I(z)$ is used to eliminate the image transfer functions. The overall filter

$$H(z) = G(Z^L).I(Z) \qquad (1)$$

consists of the components of $G(z^L)$.

The computational complexity is reduced by a factor of L due to compression of the spectrum. However, there is addition of the complexity due to anti-imaging filter $I(z)$. With an approximate choice of the factor L, the filter slope of $I(z)$ can be, made less steep, so that in general $I(z)$ has only small number of coefficients.

Note that the filter $G(z^L)$ in Figure.1 cannot be operated at a sampling rate that has been reduced by a factor of L. In each delay chain Z^{-L}, L different input values stored are processed at the original sampling rate. All of these contribute to the final output signal $y(n)$. All filter operations are performed at the original sampling rate with the coefficients $h(0), h(1), h(2).............h(N-1)$. This will reduce the number of coefficients.

3. DESIGN PROCEDURE FOR IFIR FILTER

Various methods for designing procedures for IFIR filters have been developed [15-18]. For the given the cut-off frequencies Ω_P and Ω_S, the steps for synthesizing the filter $H(z)$ are as follows.

Step 1: Determine a suitable interpolation factor L so that first image frequency response can be separated from the base band frequency using a filter $I(z)$.

Step 2: Design of the prototype $G(z)$ with the passband cut-off frequencies $\Omega_{P,G} = L.\Omega_P$ and the stopband cut-off frequencies $\Omega_{S,G} = L.\Omega_S$

Step 3: Choose passband cutoff frequency $\Omega_{P,I}$ of the anti-imaging filter $I(z)$ as Ω_P
$\Omega_{P,I} = .\Omega_P$

Step 4: The Stopband cutoff frequencies $\Omega_{S,I}$ of $I(z)$ must be $\Omega_{S,I} = 2\pi / L - \Omega_S$.

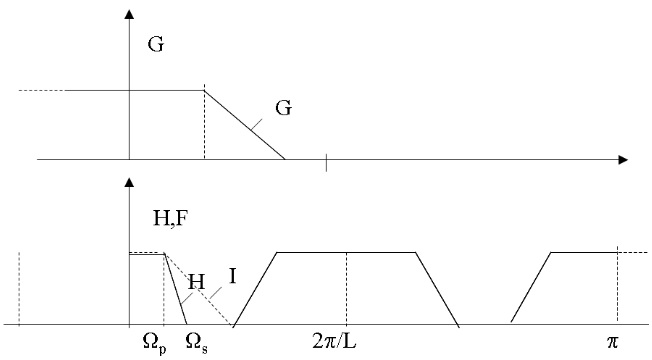

Fig 4: Determination of cut-off frequency for the frequency $G(e^{j\Omega})$ and $I(e^{j\Omega})$

The required passband ripple δ_P of the filter $H(z)$ must be distributed among the passband ripple of the filter $G(z)$ and $I(z)$, and is given by

$$(1+\delta_{P,G}).(1+\delta_{P.I}) = 1+\delta_P \qquad (2)$$

For small value of the ripple δ_P can use the approximation.

$$(\delta_{P,G}) + (\delta_{P.I}) \cong \delta_P \qquad (3)$$

If the stopband ripple δ_S of $H(z)$ is known then stopband ripple of the filter $G(z)$ and $I(z)$ are given by

$$\delta_{S,G}.(1+\delta_{P,I}) = \delta_S \,, \qquad {}_S\Omega_S \le \Omega \le \Omega_{S,I}$$

and

$$\delta_{S,I}.(1+\delta_{P,G}) = \delta_S \,, \qquad {}_S\Omega_{S,F} \le \Omega \le \pi \qquad (4)$$

Provided δ_P is small. And also it is possible to make further approximation

$$\delta_{S,G} = \delta_{S,I} \approx \delta_S \,.$$

4. RESULTS

Here IFIR filter is designed with specifications as given in table 1. The conventional filter design using the Parks McClellan algorithm [5] would require a higher order. The IFIR design for interpolated factor $L = 2$ with single-stage of $I(z)$ uses N=182. Notice that the system $G(z^2)I(z)$ has linear phase property since $G(z\ )$ and $I(z)$ have this property.

Table 1. Specifications for the design.

1	Sampling frequency	2000Hz
2	Passband Edge Frequency	0.09π
3	Stopband Edge Frequency	0.11π
4	Passband Ripple	0.02
6	Stopband Attenuation	0.001

Figure 5, shows the frequency responses for the IFIR LPF filters with single-stage $I(z)$ and $L = 2$ as well as the frequency responses for the subfilters, $I(z)$ and $G(z\ )$

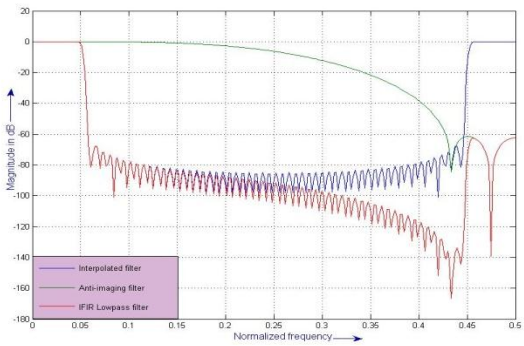

Fig 5: Frequency responses for the IFIR LPF and Subfilters

5. CONCLUSIONS

The design methodology of interpolated finite impulse response filter decomposes the sharp narrow pass band filter into an image suppressor and a periodic model filter. The Interpolated FIR design efficiently reduces the computational complexity and also this approach offers advantages during the filter design phase like reduction in numerical inaccuracies and convergence difficulties. The IFIR technique allows us to design two filters $G(z)$ and $I(z)$ of much lower order. The technique can be used in IFIR Highpass and Bandpass Filter. Also multi stage IFIR filters further reduce the computational complexity.

6. REFERENCES

[1] Oppenheim A. V., and Schafer, R. W, 1989. Discrete-Time Signal Processing. Prentice-Hall, Englewood Cliffs, NJ

[2] Proakis, J. G, Rader, C. M, Ling, F. and Nikias. C.L, 1992. Advanced Signal Processing. Macmillan, New York.

[3] Tan, L. 2000. Digital signal processing, Academic press, UK.

[4] Mitra, S. K, 2001. Digital Signal Processing, Tata-McGraw Hill, New Delhi.

[5] Rabiner, L. R. and Gold, B 1975.. Theory and Applications of Digital Processing. Prentice- Hall, Englewood Cliffs, NJ.

[6] Neuvo, Y., Dong C. Y, and Mitra, S. K. 1984. Interpolated Finite impulse response filters. IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-32,pp.563-570.

[7] Lyons, R. 2003. Interpolated narrow and lowpass FIR filters. IEEE Signal Process. Mag., pp.50-57..

[8] Haykin. S, 1991 Adaptive Filter Theory. Prentice - Hall, Englewood Cliffs, NJ.

[9] Vaidyanathan, P. P . 1993. Multirate Systems and Filter Banks Prentice - Hall Signal Processing Series

[10] Mitra,S. K. 2006. Digital Signal Processing: A Computer-Based Approach, New York, NY: McGraw Hill pp. 427-578.

[11] McClellan, J. H., Parks, T. W. and Rabiner, L. R. 1973. A Computer program for designing optimum FIR linear phase digital filters. IEEE Trans. Audio Electroacoust., Vol. AU 21, pp. 506-52.

[12] Bartolo, A. Clymer, B. D. Burges, R. C. and Turnbull, J. P. 1998 An efficient method for FIR filtering based on impulse Response rounding. IEEE Trans. On Signal Processing Vol. 46 No. 8, pp. 2243-2248.

[13] Kaiser, J. Hamming, R. 1977. Sharpening the response of a Symmetric nonrecursive filter by multiple use of the same filter IEEE Trans. Acoust., Speech, Signal Process., vol. ASSP-25 no. 5, pp. 415-422,

[14] Mann , D. A. 2008. Interpolated FIR (IFIR) Filters: A case Study. IEEE Transactions.

[15] Saramaki, T. Neuvo, Y. and Mitra, S. K. 1988 "Design of Computationally efficient interpolated FIR filters," IEEE Trans. Circuits Syst., vol. CAS-35, pp. 70-88.

[16]Webb, J. L., Munson. D. 1996. "A new approach to designing Computationally efficient interpolated FIR filters" IEEE Trans. Signal Processing, vol. 44, pp. 1923-1931.

[17] Mehrnia, A. Willson, J. A. 2004. On optimal IFIR filter design IEEE Proceedings of the 2004 International Symposium on Circuits and Systems, vol. 3, pp. 133-13.

[18] Jovanovic-Dolecek, G. Mitra, S. K 2005. Multiplier-free FIR filter design based on IFIR structure and rounding," *IEEE 48th Midwest Symposium on Circuits and Systems*., vol. 1, pp. 559-562.