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# Design of Digital Filter and Filter Bank using IFIR

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**Abstract** – *A filter is an essential tool of digital signal processing. Since, finite impulse response (FIR) filters have property such as: linearity and stability, hence FIR filters are more favorable than infinite impulse response (IIR) filters. However, large filter order makes computational complexity high. For low computational complexity with linear phase property, interpolated infinite impulse response (IFIR) filters are used. This research paper explores the performance of IFIR filters in terms of computational cost reduction (CR) and overall filter response. Two stages IFIR filter has been designed to improve the computational complexity implementation of Uniform M-channel filter bank has been done by using single stage and two stages IFIR prototype filter.*

***Key Words***: Use of FIR, IIR, IFIR, CR, M-channel.

## 1.INTRODUCTION

This paper includes the mathematical analysis of two-channel and M-channel filter bank with brief introduction. The designing of uniform M-channel filter bank has been done by single stage and two stage IFIR prototype filter to obtain greater value of CR. The basic steps to design single stage and two stage IFIR filter are given in section 1.3. For achieving PR or NPR condition, a suitable optimization algorithm is used by calculating filters coefficients. After designing filter bank with single stage IFIR filter, further improvement in computational cost by using two stages IFIR prototype filter. Different window functions have been used to design digital filter and the results are evaluated in terms of aliasing error, amplitude error, reconstruction error, computational cost reduction, number of iterations and computational time.

### 1.1 M-channel Filter Bank

In M-channel filter bank, input signal decomposes into M number of frequency bands and then these signals are processed separately. M-channel filter bank is further characterized into two types: uniform M-channel filter bank and non-uniform M-channel filter bank. In M-channel uniform filter bank, input signal is divided into M number of frequency bands which have same bandwidth. The block diagram of M-channel filter bank, where $H_K(z)$ denotes analysis filter and $G_K(z)$ denotes synthesis filter. The mathematical analysis is given below. Expression of the reconstructed signal is obtained by ignoring the coding and quantization errors. Expression of $v_K(n)$ is achieved by convolving input signal to corresponding filter response, which is given as:

$$V_K(z) = (z)H_K(z)$$

The output signal can be expressed as:

$$\mathbf{Y(z)} = \frac{1}{M}\sum_{k=0}^{n} A_l(z)X(zW^l), 0 \le l \le M-1 \ldots\ldots\ldots \mathbf{(1)}$$

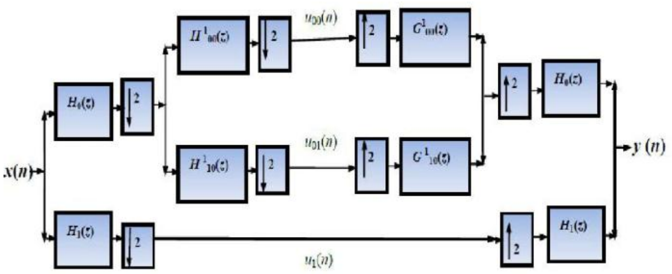


**Figure:** Non-uniform tree structured M-channel filter bank

### 1.2 Optimization of Prototype Filter

After designing prototype filter, a suitable optimization technique has been used to achieve perfect or nearly perfect reconstruction condition. In this Chapter, optimization of cutoff frequency of model filter has been done by means of calculating the coefficients IFIR prototype filter. The steps of optimization of IFIR prototype filter have been presented by block diagram, depicted in Fig.1. After optimization of prototype filter, all the analysis and synthesis filters are derived from a single prototype filter. In M-band cosine modulated filter bank, the PR condition can be achieved, if Eq. 1 is satisfied. By solving this equation, at the value of $\omega$ is equal to $\pi/2M$, and then it leads to Eq. 2.

$$|H_0(e^{jW})|^2 + |H_0(e^{j(W-\pi/M)})|^2 = 1, \text{ for } 0<W<\pi/M$$

$$|H_0(e^{j\pi/2M})| = 0.707 \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(2)$$

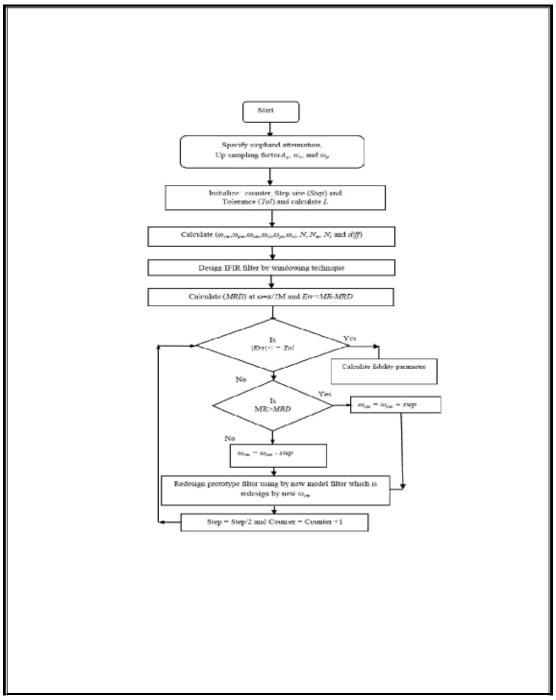


**Figure 1:** Block diagram of proposed algorithm

## 2. Results and discussions

This Section presents comparative analysis of proposed algorithm with other existing algorithms with the help of examples and tabular results. The performance of proposed algorithm is evaluated in terms of significant parameters listed below:

Computational cost reduction *(%CR)*

- Computational cost reduction (%CR)

  $$\%(CR) = \frac{Multi(FIR) - Multi(IFIR)}{Multi(FIR)}$$

- Amplitude distortion($e_{am}$)

  $e_{am}=\max(1-|T_0(e^{j\omega})|)$

- The worst case aliasing distortion($e_a$)

  $e_a=\max(T_1(e^{j\omega}))$ for $\epsilon[0,\pi]$, $\omega 1\leq l\leq M-1$

***Example:*** In this example, 32-band filter bank has been designed by using single stage IFIR prototype filter. The designing of model and interpolator filter has been done using Blackman window function for a given stop band attenuation (As) = 85dB, pass band frequency ($\omega$p) = $\pi/4M$ with stop band edge at $\pi/M$.

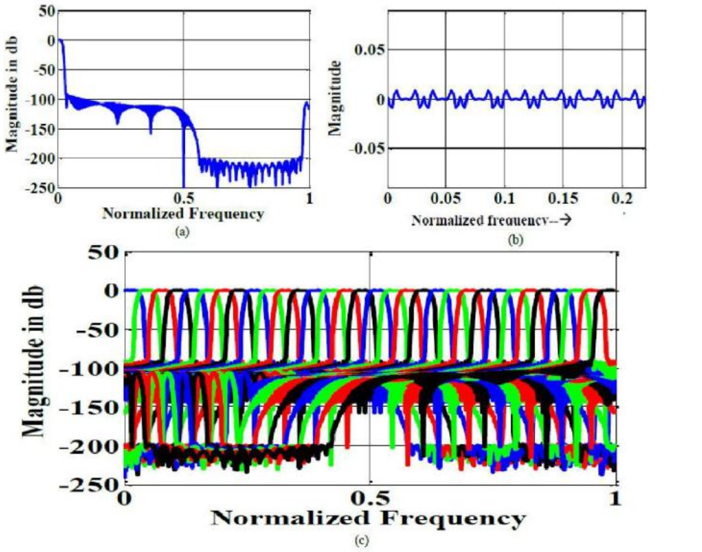


**Figure** : Response of (a) IFIR filter with Kaiser window,(b) Reconstruction error in dB and(c) magnitude response of 32-channel filter bank with by single stage IFIR filter bank

**Kaiser *window***: Aliasing error = 3.077×105, reconstruction error = 9.30×103, % computational cost reduction =43.50

***Blackman window***: Aliasing error = 5.213×105, reconstruction error = 6.700×103, % computational cost reduction = 43.50

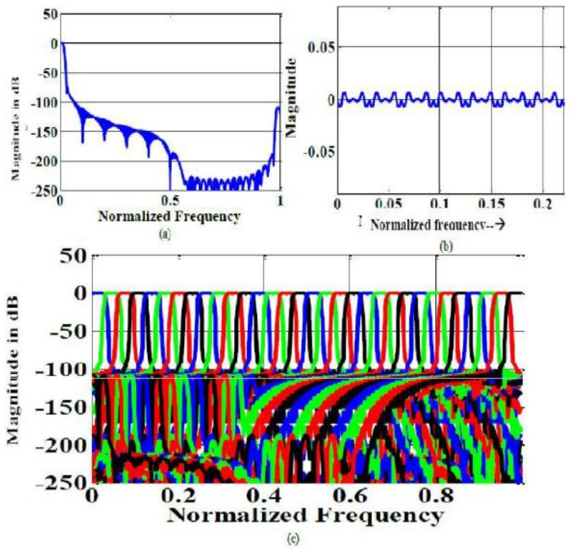


**Figure** *:* Response of (a) IFIR filter with Blackman window,(b) Reconstruction error in dB and (c) Magnitude response of 32-channel filter bank with by single stage IFIR filter bank.

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Simulated results of 32-channel filter bank have been presented in Figs. 16 and 17 by using Kaiser and Blackman Kaiser Window function, respectively. These figures show response of single stage IFIR filter, reconstruction error and analysis filter bank designed with single stage IFIR filter bank. Various examples have been taken to examine the response of IFIR filter for designing filter and filter bank .From these fidelity parameters and simulated results; it has been found that the prototype filter design by IFIR filter gives significant value of computational cost reduction with excellent values of reconstruction and aliasing error. For greater value of *L,* the computational cost reduction is also higher.

## 2.1 Two Stages IFIR Filter

Single stage IFIR filter has been discussed in Chapter 1. The main aim of IFIR filter is to reduce the computational complexity, by using up-sampling process. The structure of single stage IFIR filter is has only one up-sampling factor. If the value of up sampling factor is more, than large ordered interpolator filter is required, which introduce high computational complexity, thus to reduce computational complexity two stages IFIR is used. In two stages IFIR filter instead of single up-sampling factor, two up sampling factors are used.

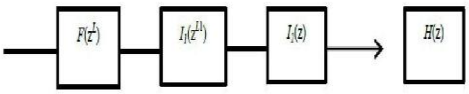


Figure : Two stages IFIR structure

The designing steps of two stages IFIR filter has been written below:

Step 1: Specify $A_s$ , $w_p$ and $w_i$ .

Step 2: Calculate $N$, $L$, $w_p$ $w_{pi}$ $w_{si}$ $N_m$ *and* $N_i$ .

Step 3: Design model filter.

Step 4: Up sample model filter by $L$.

Step 5: Select second up sample factor ($L1$)

Step 6: Calculate specifications of first and second interpolator filter ( $w_{pi1}$ ,$w_{si1}$ ,$w_{pi2}$ , $w_{si2}$ , $N_{i1}$ ,$N_{i2}$ ) with $L2$, $w_{pi}$ $w_{si}$

Step 7: Convolve first interpolator filter with second interpolator filter

Step 8: Convolve model filter with resultant interpolator filter.

## 2.2 Iterative Algorithm

After designing two stages IFIR filter, optimization of prototype filter has been done by calculating the coefficients of prototype filter. After optimization of prototype filter, all analysis and synthesis filters are derived by applying cosine modulated technique. The steps of optimization of two stages IFIR prototype filter are:

1. Specify normalized pass band edge frequency ( $w_p$ ) stop band edge frequency ( $w_s$ ), and stop band attenuation ( $A_s$ ), step size (*step*), tolerance (*To l*).

2. Select, and ideal magnitude response (*MR*=0.707).

3. Calculate the order of filter all specifications of model and interpolator filter ( $w_{pm}$ , $w_{sm}$ , $w_{pi}$ , $w_{si}$ , $N_m$ , $N_i$ ) using given specifications and $L$.

4. Design two stages IFIR filter by using above steps 1 to 7.

5. Calculate the magnitude response of designed IFIR filter (*MRD*) at 3dB cut of frequency (*w* )=*p* /2M and value of error (*Err=MR-MRD*).

6. Check if error is within the *to l* level or not.

(a)If No, Then compare the value of *MRD* with *MR* and accordingly cutoff frequency is varied using the *step*

i. If *MR* >*MRD*, then increase $w_{cm}$ = $w_{cm}$ + *step* .

ii. Otherwise $w_{cm}$ = $w_{cm}$ - *step*.

(b) If yes, then, design the other filters by applying cosine modulation technique.

7. Redesign the IFIR prototype filter by redesigning model filter using new $w_{cm}$ on same order. Calculate *MRD* and also *Error*.

8. Increment the counter by 1 and *Step=step /2*

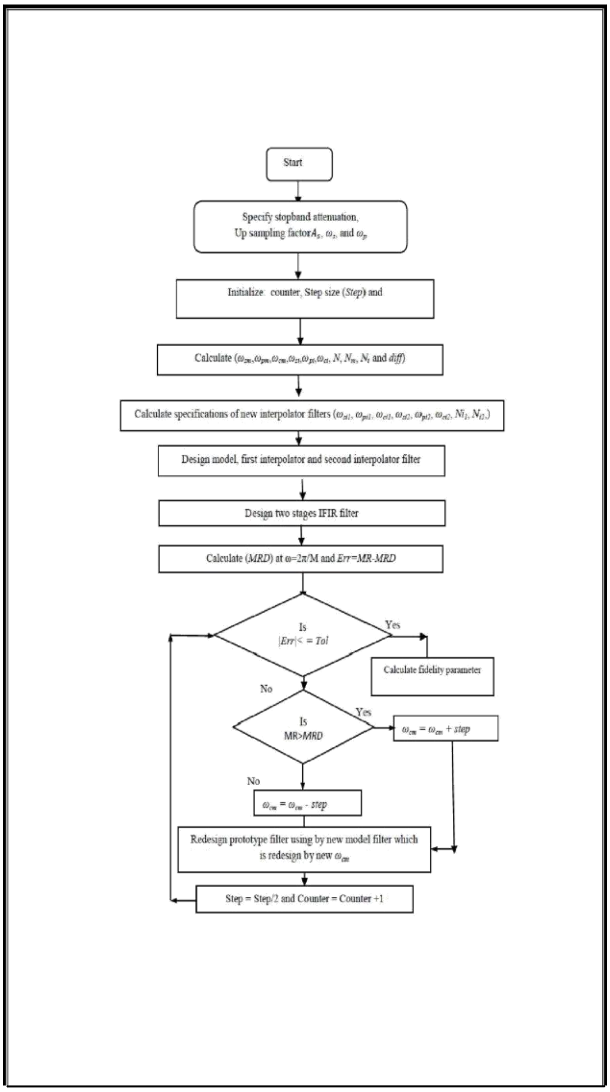


**Figure 2:** Block diagram of proposed algorithm

## 3.CONCLUSIONS

In this Paper, designing of digital filter and filter bank has been done using IFIR prototype filter. With the help of above examples, tabular results and above discussion on simulated results, this can be concluded that, IFIR filter is an efficient filter, which can be used as prototype filter for nearly perfect reconstruction cosine modulated filter bank, which provides significant reduction of computational cost. In this Chapter, more significant improvement in computational cost and *SLFOR* of prototype filter has been achieved by introducing two stages IFIR technique for designing prototype filter. The simulated responses and fidelity parameters of table show that two stages IFIR filter gives up to 82 % of computational cost reduction. Single stage and two stages IFIR filter can be used to design non uniform filter bank or Trans multiplexer.

## REFERENCES

1. P. J. Proakis, and G. Dimitris G. Manolakis., "Introduction to digital signal processing," Prentice Hall Professional Technical Reference, 1988.

2. A. Antoniou, "Digital signal processing," signal, systems, and filters, New York: McGraw-Hill, 2005.

3. S. K. Mitra, and J. F. Kaiser, "Handbook for digital signal processing ".John Wiley & Sons, Inc., 1993.

4. T. W. Parks, and C. Sidney Burrus, "Digital filter design," Wiley-Interscience, 1987.

5 .S. Mitra, and K. Yonghong, "Digital signal processing: a computer-based approach," Vol. 2. New York: McGraw-Hill, 2006

6. Y. Neuvo, D. Cheng-Yu, S. K. Mitra, "Interpolated finite impulse response filters," IEEE Transaction on Acoustic Speech Signal Process Vol.32, No.3, pp. 563-570, 1984.

7. T. Saramaki, T. Neuvo, and S. K. Mitra, "Design of computationally efficient interpolated FIR filters," IEEE Transactions on Circuits and Systems, Vol. 35, No.1, pp. 70-88, 1988.

8. P. P. Vaidyanathan, "Optimal design of linear phase FIR digital filters with very flat pass band and equation ripple stop band," IEEE Transactions on Circuits and Systems, Vol. 32, No.9, pp. 904-917, 1985.

9. T. Karp, N. J. Fliege, "Modified DFT filter banks with perfect reconstruction," IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing, Vol. 46, No.11, pp. 1404-1414, 1999.

10. S. C. Powell and P. M. Chau, "Efficient narrowband FIR and IFIR filters based on powers-of-two sigma delta coefficient truncation," IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing, Vol. 41, No.8, pp. 497-505, 1994.

11. Y. Lian, L. Zhang, and C. C. Ko, "An improved frequency response masking approach for designing sharp FIR filters," Signal processing, Vol. 81, No.12, pp. 2573-2581, 2001

12. G. Jovanovic-Dolecek, and A. Fernandez-Vazquez, "Novel droop-compensated comb decimation filter with improved alias rejections," AEU-International Journal of Electronics and Communications Vol. 67, No.5, pp. 387-396, 2013.

13. Y. C. Lim, and L. Yong, "Frequency-response masking approach for digital filter design: complexity reduction via masking filters factorization," IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing, Vol. 41, No.8, pp. 518-525, 1994.

14. Y. Lain, "The optimum design of half-band filter using multi-stage frequency-response masking technique." ELSEVIER on Signal Processing, Vol. 44, No.3, pp. 369-372, 1995.

15. R. Yang, L. Bede, and C. L. Yong, "A new structure of sharp transition FIR filters using frequency response masking," IEEE Transactions on Circuits and Systems, Vol. 35, No.8, pp. 955-966, 1988.

16. I. SINGH and P Mehta, "Joint Optimization of Interpolated FIR Filter, International Journal of Engineering," Vol. 2, No.1 2013.