QMF System Design With Near-perfect Reconstruction By Using Kaiser Window

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Abstract—The researchers nowadays give a lot of attention to the QMF banks because of their wide applications in many signal processing fields, such as image and speech compression sub-band coding of speech and image signals,trans-multiplexers , equalization of wireless communication channels, source coding for audio and video signals , design of wavelet bases, sub-band acoustic echo cancellation, and discrete multi-tone modulation systems. In QMF system the reconstructed signal y(n) is different from the input signal x(n) due to three error which is aliasing distortion,phase distortion ,amplitude distortion. In this paper we are proposing to eliminated the aliasing and phase distortion, all the analysis and synthesis filter must be translated to a single lowpass prototype of even-order symmetric FIR linear –phase filter.Thus the design of the whole filterbank reduces to that of prototype filter. So we are design QMF by using Kaiser window ,pc4 and pc6 so the signal is passed though different windows such and it is found that by this designing we can eliminate the aliasing & phase distortion.The result is compared on MATLAB, and Simulation is done on MODELSIM.

Keywords—QMF,Kaiser Window, Filter bank,Pc4 and Pc6window .

I.   INTRODUCTION

During the last several years there has been substantial progress in multi-rate system research. This includes design of decimation and interpolation filters, analysis/synthesis filter bank also called Quadrature Mirror Filter. Quadrature mirror filter (QMF) bank were the first type of the filter banks used in signal processing application for separating the signals into sub-bands and for reconstructing them from individual sub-bands (Croisier (1976)) [1].These filters find applications where a discrete signal is to be split into a number of consecutive bands in the frequency domain, so that each sub-band signal can be processed in an independent manner. Typical processing includes under-sampling the sub-band signals, encoding them and transmitting over a channel, or merely storing the coded signals. Eventually, at some point in the process, the sub-band signals should be recombined so that the original signal is properly reconstructed. The theory and design of QMF bank was introduced by Johnston [2] in 1980. It used Hanning window to design the low pass prototype FIR filter. Jhonston combined the pass band ripple energy and out of band energies into a single cost function having nonlinear nature. It is minimized by the Hooke and Jeaves optimization algorithm [3]. Creusere and Mitra [4] used a different objective function, which is linear in nature. Park-McClellan algorithm has been used to design the low pass prototype filter. In this technique, pass band frequency is iteratively changed to minimize the reconstruction error using linear optimization
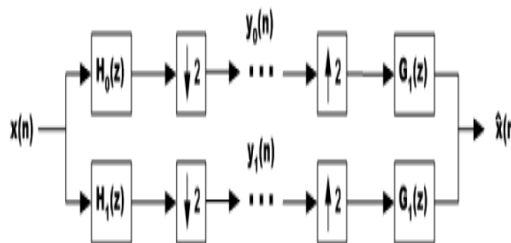

Figure 1:Two-channel filter bank (a) analysis and synthesis filter bank structure

The researchers nowadays give a lot of attention to the QMF banks because of their wide applications in many signal.processing fields, such as sub-band coding of speech and image signals , speech and image compression , trans-multiplexers [8], equalization of wireless communication channels [9], source coding for audio and video signals [10],design of wavelet bases , sub-band acoustic echo cancellation , and discrete multi-tone modulation systems .In several design methods reported aliasing has been cancelled completely by selecting the synthesis filters cleverly in terms of the analysis filters and the PHD has been eliminated using the linear phase FIR filters.The overall transfer function of such an alias and phase distortion free system turns out to be a function of the filter tap coefficients of the low-pass analysis filter only, as the high-pass and lowpass analysis filters are related to each other by the mirror image symmetry condition around the quadrature frequency $\pi/2$. Therefore, the AMD can be minimized by optimizing the filter tap

weights of the low pass analysis filter. If the characteristics of the low-pass analysis filter are assumed to be ideal in its pass-band and stop-band regions, the PR condition of the alias and phase distortion free QMF bank is automatically satisfied in these regions, but not in the transition band. In general, design algorithms of QMF bank can be grouped in two ways: Optimization based and non optimization based. In optimization based approach the design problem is formulated as a multi-objective or single objective nonlinear optimization problem which is solved by various existing methods such as genetic algorithm, penalty methods, Iterative methods of least square (WLS) method. In all such methods, the objective function is constructed as a weighted sum of the objectives, which is further minimized using suitable optimization algorithm. In early stage of research, the design methods developed are based on direct minimization of error function in frequency domain or in time domain. But due to high degree of nonlinearity and complex optimization technique, these are not suitable for filters with larger taps. In order to overcome this problem, Chen and Lee have proposed an iterative technique that results in equi ripple reconstruction error. Similarly other researchers have also developed a new technique for the design of QMF bank with slight modification in algorithm [14]. In all such methods, the design problem has been formulated as linear or nonlinear combination of reconstruction error and passes band residual energy and uses direct form implementation of filter bank.

II. TWO CHANNEL ANALYSIS OF QMF

Before Analysis filter banks followed by synthesis filter bank together form a sub band coding (SBC) filter bank. An analysis filter bank with filter $H_0(Z)$ and $H_1(Z)$ decompose the input signal $X(Z)$ into the sub band signal $X_0(Z)$ and $X_1(Z)$. This is followed by a synthesis filter bank $G_0(Z)$ and $G_1(Z)$. Which reconstructs the output signal $Y(Z)$ from the sub-band signal The analysis filter $H_0(Z)$ and $H_1(Z)$ have typically a low-pass and high-pass frequency responses, respectively, with a cutoff at π.

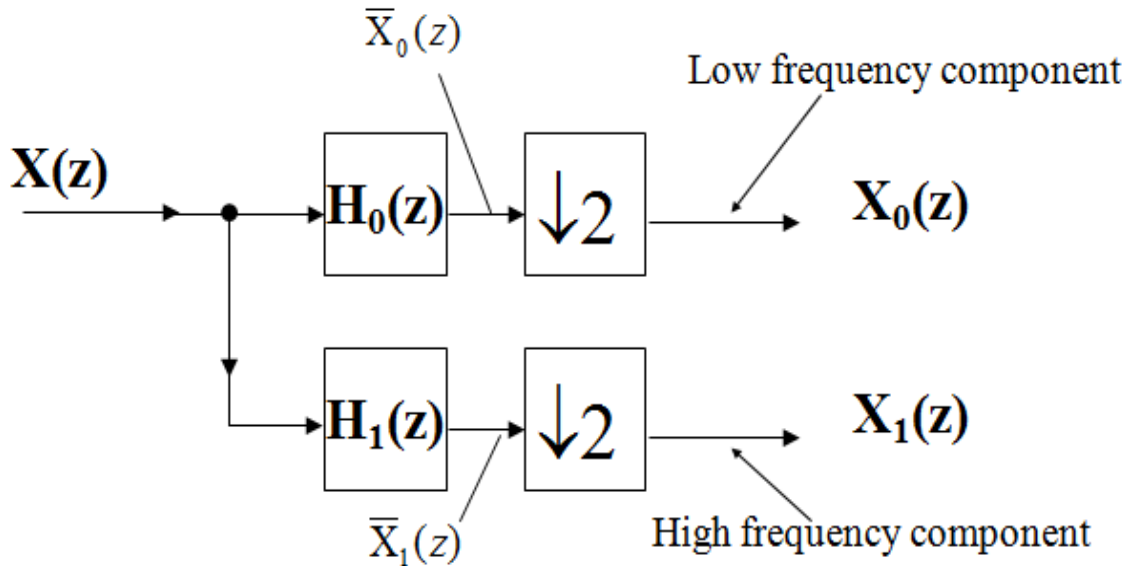Two-Channel Analysis Filter Banks are given bellow.

Figure 2: Two-channel analysis filter bank

The two filtered signals from the fig 2.1

$$\overline{X}_0(z) = X(z) \cdot H_0(z) \tag{2.9}$$

$$\text{(1)}$$

$$\overline{X}_1(z) = X(z) \cdot H_1(z) \tag{2}$$

Down sampling with M=2, using yields the sub band signal

$$X_0(z) = \tfrac{1}{2} X(z^{1/2}) \cdot H_0(z^{1/2}) + \tfrac{1}{2} X(-z^{1/2}) \cdot H_0(-z^{1/2})$$

$$X_1(z) = \tfrac{1}{2} X(z^{1/2}) \cdot H_1(z^{1/2}) + \tfrac{1}{2} X(-z^{1/2}) \cdot H_1(-z^{1/2})$$

These Eq. (3),(4) can be written in matrix form

$$\begin{bmatrix} X_0(z) \\ X_1(z) \end{bmatrix} = \frac{1}{2} \begin{bmatrix} H_0(z^{1/2}) & H_0(-z^{1/2}) \\ H_1(z^{1/2}) & H_1(-z^{1/2}) \end{bmatrix} \cdot \begin{bmatrix} X(z^{1/2}) \\ X(-z^{1/2}) \end{bmatrix}$$
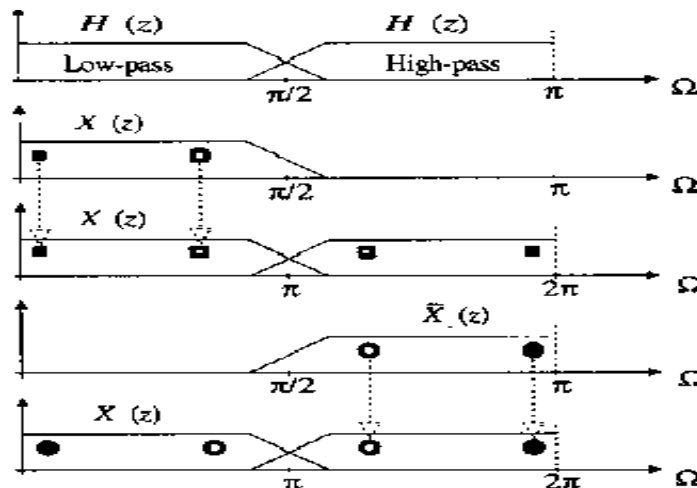
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Figure 3: Transfer function (a) spectra of the filtersignal (b and d) spectra of the down sampled signal(c and e) in the analysis filter bank

III.   FIR FILTER DESIGN USING WINDOW TECHNIQUE

Design Formulation&Proposed work

we have seen that the reconstruction output y(n) is differ form input signal and to eliminate the aliasing and phase distortion we have to design a linear phase prototype FIR filter so to reduce reconstruction error here we will use Kaiser window and two combinational window pc4 and pc6 to design QMF filter

Design steps for Kaiser window

Kaiser window approach has been used to design prototype lowpass filter. Impulse response of lowpass filter *h(n)* of length *(N+1)* designed through Kaiser window function

$$h(n)=h_i(n).w(n)$$

$$h_i(n) = \frac{\sin(\omega_c(n - 0.5N))}{\pi(n - 0.5N)}$$

where hi(n)is the impulse response of the ideal filter with cutoff frequency ω_c.

The Kaiser window function *w(n)* is defined as[15],[23]

$$\omega_i(n) = \frac{I_0\left\{\beta\sqrt{1 - \frac{n}{m}}\right\}}{I_0(\beta)}$$

where *I*o(.) is the zeroth order modified Bessel function, *β* is the   window shape parameter which control the stopband attenuation. The empirical design equation developed by Kaiser is given by-

$$\beta = \begin{cases} 0 & for A_s < 21 \\ 0.5842(A_s - 21) + 0.07886(A_s - 21) & for 21 \le A_s \le 50 \\ 0.1102(A_s - 8.7) & for A_s > 50 \end{cases}$$

The design of prototype filter using Kaiser window can be specified by the three parameter ω_c, *N* and *β*. The order of the window *N* can be estimated by the following formula developed by Kaiser [15], [23]

$$N = \frac{2 \times (A_s - 7.95)}{14.36\Delta\omega}$$

where A_s is the stop band attenuation and $\Delta\omega$ is the normalized transition width.

$$\Delta\omega = \omega_s - \omega_p$$

To determine stopband attenuation we have to choose minimum of passband ripple and stop band attenuation

Here we are showing a flow chart in which to design QMF by using Kaiser window first we need to initialize the parameter of Kaiser window t,s,f ,f_p,f_s .We will fix the passband frequency and stopband frequency and we will vary sampling frequency at every instant first of all we will fix the value of stopband attenuation and vary the sampling frequency and note down the reconstruction error at which the reconstruction error is minimum. Same process will repeat for designing QMF filter by using pc4 and pc6.

Now in second turn we will fix the value of filter order and vary the sampling frequency at different stopband attenuation than we will again note down the sampling frequency at which reconstruction error is minimum we will repate the process for designing QMF filter by using pc4 and pc6 and see the reduced reconstructin error and comapared the result on MATLAB .we will also see the simulation on MODELSIM in which we will calculate the coefficient of Kaiser window for faster communication
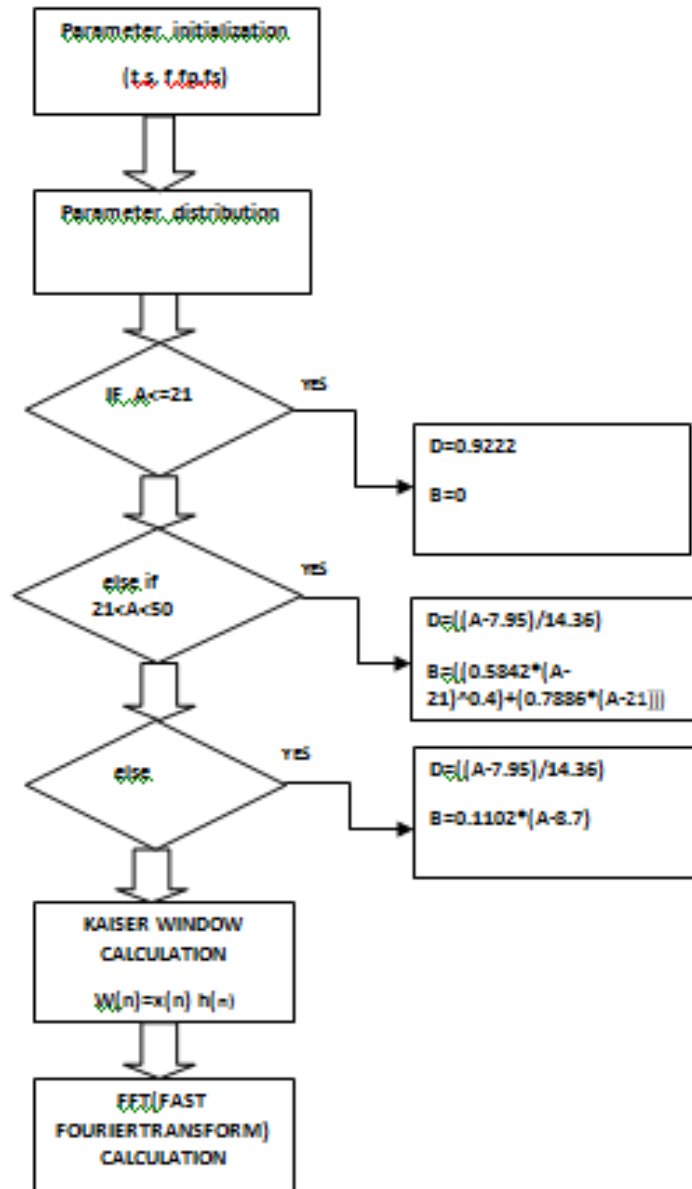

Figure 4: flow chart of designing Kaiser window

IV   RESULT and DISCUSSION

Comparison of proposed method
Results are shown in Tables 1–4. In Table 1, the value of stop band attenuation was selected as 50 dB, resulting in different filter orders for different window functions. In Table 2 results corresponding to filter designs with an order (*N*) of 80 are shown. Same value of filter order resulted in different levels of stop band attenuations. In Table 3 we will see the previous work Finally, in Table 4, a comparison is made of the optimum performance that can be attained with the three window functions of our proposed work.

Table 1-Performance of QMF filter at 50 dB stop band attenuation

Window function	Reconstruction error(dB)	Filter order(N)	Sampling frequency
Kaiser window	0.0090	44	1.20
PC4	0.0018	41	1.30
PC6	0.00805	9	0.15

Table 2 - Performance of QMF filter at $N = 80$

Windows Function	Reconstruction Error(dB)	Stop band attenuation (dB)	Sampling frequency
Kaiser window	0.0083	50	2.20
PC4	3.85×10^{-15}	50	1.14
PC6	0.8707	68	0.89

Table 3 **Previous work** in terms of reconstruction error

Window function	Reconstruction error(dB)	Stop band attenuation(dB)	Filter order (N)	Far end attenuation (dB)
Kaiser window	0.0097	88	90	107
PC4	0.0135	59.50	30	110
PC6	0.0120	55.50	22	72

Table 4 **Proposed work** in terms of reconstruction error with improved result

Window function	Reconstruction error(dB)	Stop band attenuation(dB)	Filter order (N)	Sampling frequency
Kaiser window	0.0083	50	80	2.20
PC4	0.0110	50	52	1.59
PC6	0.0018	50	41	1.30

Apart from the reconstruction error, the far-end attenuation (amplitude of the last ripple in the stop band) is also selected as one of the figures of merits for the comparative study in previous work . This parameter is of significance when the signal to be filtered has great concentration of Spectral energy. In a sub-band coding, the filter is intended to separate out various frequency bands for independent processing. In the case of speech, e.g. the far-end rejection of the energy in the stop band should be more so that the energy leak from one band to another is minimum. From Tables 3 and 4, it is inferred that as the stop band attenuation increases the value of reconstruction error decreases. The result of Kaiser window-designed FIR filter are shown in figure 5(previous work) . The result of Kaiser window-designed FIR filter are shown in figure 6(proposed work) with reduced   reconstructionerr

Figure 5: Variation of The Reconstruction error (kaiser window) by table 3.   (previous work output using MATLAB))
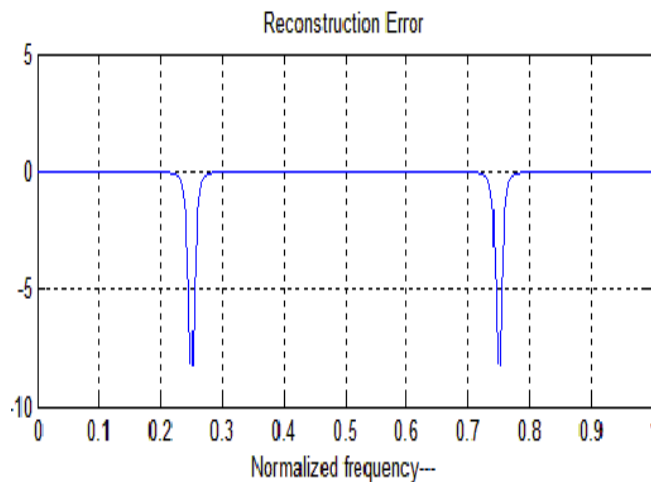

 Figure6:Variation of The Reconstruction error (kaiser window) by table 4.(proposed work output on MATLAB)
These results are outputs of Kaiser window by using MATLAB.

V.   HARDWARE SIMULATION

 First we have been proposed the flow chart of designing a Kaiser window by the last step of that flow chart we took the coefficient of Kaiser window and than fetch the FFT coefficient than by using a serial multiplier we divide the coefficient in product term & sum term because digital filter can't understand real coefficient so for faster calculation we are using MODELSIM to see simulation of Kaiser window .
Figure 8 shows the simulation of Kaiser window coefficient by using MODELSIM.
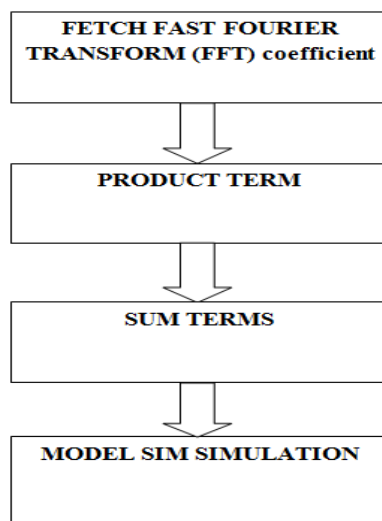

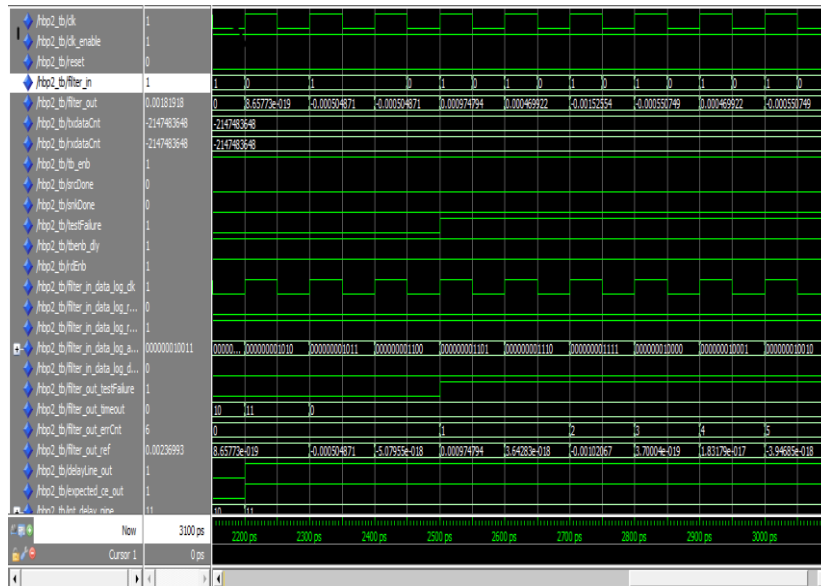Figure 7: flow chart for calculation of kaiser window coefficient

Figure 8: simulation result of calculation of coefficient of Kaiser window by using MODELSIM

VI. Conclusion and Future Scope

In this work a new method to design the two channel quadrature mirror filter (QMF) bank using Kaiser Window , pc4 and pc6 window with aliasing free condition makes the system to implement the method has been developed and simulated with the help of MATLAB 7.1 and simulated on ModelSim.We will find that the reconstruction error has been reduced ans reached near to zero for designing a QMF system with near-perfevt reconstruction .

The research nowadays give a lot of attention to the QMF banks because of their wide applications in many signal processing fields such as sub-band coding of speech image signals ,speech and image compression ,Trans-multiplexers , equalization of wireless communication channels , source coding for audio and video signals, design of wavelet bases ,sub-band acoustic echo cancellation , discrete multi-tone modulation systems .

REFERENCES

[1] G. Eason, B. Noble, and I.N. Sneddon, "On certain integrals of Lipschitz-Hankel type involving products of Bessel functions," Phil. Trans. *Roy. Soc. London, vol. A247, pp. 529-551, April 1955. (references)*

[2] J. Clerk Maxwell, A Treatise on Electricity and Magnetism, *3rd ed., vol. 2. Oxford: Clarendon, 1892, pp.68-73.*

[3] I.S. Jacobs and C.P. Bean, "Fine particles, thin films and exchange anisotropy,*" in Magnetism, vol. III, G.T. Rado and H. Suhl, Eds. New York: Academic, 1963, pp. 271-350.*

[4] K. Elissa, "Title of paper if known," unpublished.

[5] R. Nicole, "Title of paper with only first word capitalized," J. Name Stand. Abbrev., in press.

[6] Y. Yorozu, M. Hirano, K. Oka, and Y. Tagawa, "Electron spectroscopy studies on magneto-optical media and plastic substrate interface*," IEEE Transl. J. Magn. Japan, vol. 2, pp. 740-741, August 1987 [Digests 9th Annual Conf. Magnetics Japan, p. 301, 1982].*