DESIGN AND SIMULATION OF TWO CHANNEL QMF FILTER BANK FOR ALMOST PERFECT RECONSTRUCTION

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ABSTRACT

In this paper a two channel FIR QMF bank for perfect reconstruction has been presented. The main problem of filter bank design is to develop the prototype filter H0 such that it has minimum error in stopband, passband and transition band. To design QMF filter bank, all other filters are derived from the prototype filter H0. The designed and simulated proposed QMF filter bank has been developed using Remez algorithm with MATLAB. The developed filter bank performance has been compared with existing Window based filter bank in terms of Peak Reconstruction Error. It can be observed from the simulated results that proposed method has reduced PRE up to 50% as compared to Kaiser Window based filter bank.

KEYWORDS: *Equiripple Technique, FIR Filters, Kaiser Window, Perfect Reconstruction, QMF Filter bank.*

I. INTRODUCTION

The substantial progress in multirate digital filters and QMF filter banks has been made because of wide applications in many processing fields such as subband coding of speech, image signal processing, antenna system and transmultiplexer [1]. The advance development in QMF filter is done for applications such as biomedical signal processing and design of wavelet bases [2]. In QMF bank the input signal x[n] splits into two subband signals having equal bandwidth. The subband signals are then processed and finally combined by a synthesis filter bank resulting in an output signal y[n]. The subband signal are band limited to frequency ranges much smaller than that of the original signal, so before processing they are down sampled. After processing these signals are up sampled before being combined by the synthesis band into a higher rate signal. The combined structure is called a Quadrature Mirror Filter.

The reconstructed signal is different from input signal due to three errors: Aliasing distortion, phase distortion and magnitude distortion. The aliasing distortion and phase distortion is removed with use of suitable design of the synthesis filters and linear phase FIR filter respectively. Amplitude distortion is not possible to eliminate completely, but can be minimized using computer aided techniques or using cascading filters [3]. The transition bandwidths of the multichannel filter banks are usually shorter than those of two channel QMF filter bank, the lengths of the two channel filters are usually shorter than those in multichannel filter bank. Moreover, as only a single prototype filter is required for the design of QMF bank and all other filters are derived from the prototype filter. FIR filters guarantees the stability and no phase distortion of the filter bank. A two channel QMF bank could not achieve the exact perfect reconstruction with the prototype filters having very good frequency selectivity [4]. The optimal Algorithm is required that it minimizes the maximum error between the desired frequency response and actual frequency response of digital filter is of linear phase with minimum possible order.

This paper is organized as follows. In Section II, the principle of two channel QMF bank is described. Section III is devoted to the design and simulation of QMF filter bank. Several design examples and comparisons with other conventional methods are given in Section IV. Finally, a conclusion is drawn in Section V.

II. TWO CHANNEL QMF BANK

The multirate digital structure that employs two decimator in the signal analysis section and two interpolators in the signal synthesis section is shown in fig. 1. The input signal x[n] is first passed through a two band analysis filter bank $H_0(z)$ and $H_1(z)$, which typically have low pass and high pass frequency responses respectively ,with cutoff frequency at $\pi/2$, as indicated in fig. 2. The subband signals are then down sampled by factor of two, coders are inserted after down sampler to encode by exploiting the special spectral properties of the signal such as energy levels. At receiving end, decoders are used to produce approximations of the original down sampled signals. The decoded signals are then down sampled and passed through the synthesis filter bank $G_0(z)$ and $G_1(z)$. If the down-sampling and up-sampling factors are equal to or greater than the number of bands of the filter bank, then the output can be made to retain all of the characteristics of the input signal using proper filters structure.

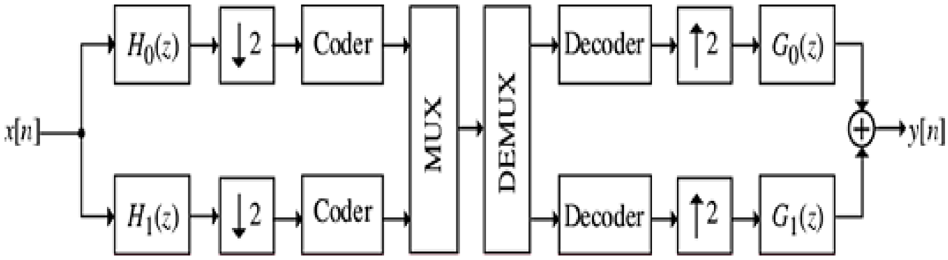

Fig. 1. General structure of a two channel QMF bank

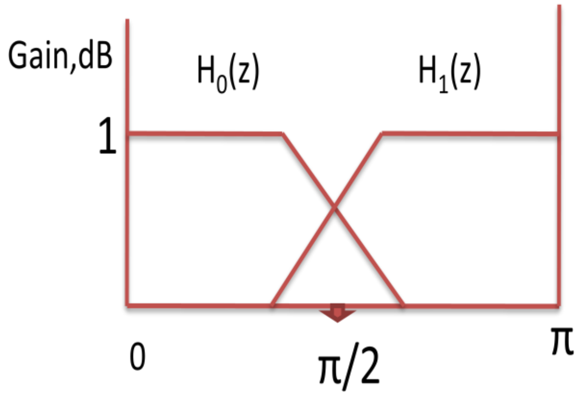

Fig.2. Magnitude response of overlapping analysis filters.

The z-Transforms of the input signal $X(z)$ are

$$X_0(z) = X(z)\, H_0(z) \quad (1)$$

$$X_1(z) = X(z)\, H_1(z) \quad (2)$$

The output signals $Y_0[n]$ and $Y_1[n]$ are added to obtain the single output y[n].The z-transform of y[n] is given as

$$Y(z) = Y_0(z) + Y_1(z) \quad (3)$$

$$Y(z) = \frac{1}{2}[H_0(z)G_0(z) + H_1(z)G_1(z)]X(z) + \frac{1}{2}[H_0(-z)G_0(z) + H_1(-z)G_1(z)]X(-z) \quad (4)$$

$$Y(z) = T(z)X(z) + A(z)X(-z) \quad (5)$$

Where, $T(z)$ is the distortion transfer function and $A(z)$ is the aliasing distortion. The first term is a desired signal and the second term is because of effect of aliasing which is to be eliminated with

$$A(z) = 0 \quad (6)$$

There fore,

$H_0(-z)G_0(z) + H_1(-z)G_1(z)$=0 (7)

This condition is simply satisfied by selecting [5], the equations for perfect reconstruction conditions are.

$H_0(z) = H(z)$ (8)

$H_1(z) = H(-z)$ (9)

$G_0(z) = H(z)$ (10)

$G_1(z) = -H(z)$ (11)

Now consider the condition for which the output of QMF bank is identical to the input except for an arbitrary delay, for all possible inputs. When this condition is satisfied, the filter bank is called a Perfect Reconstruction QMF bank.

$T(z) = \frac{1}{2}[H_0(z)G_0(z) + H_1(z)G_1(z)] = Z^{-k}$ (12)

And

$H^2(z) - H^2(-z) = 2Z^{-k}$ (13)

The alaising and phase distortion has been eliminated completely [6,7] .The distortion transfer function of the two-channel analysis/synthesis filter bank satisfying the Perfect Reconstruction property is a pure delayed function.

$T(z) = Z^{-n0}$ (9)

The FIR two–channel filter bank with linear–phase analysis and synthesis filters will be perfect reconstruction type if,

$\left|H_0\left(e^{jw}\right)\right|^2 + \left|H_1\left(e^{jw}\right)\right|^2 = 1$ (10)

The equation (7) cannot satisfy exactly due to finite length of filter so it always exhibits some amplitude distortion unless it is a constant for all value of ω.

$\Phi = \max\left\{\left|H_0\left(e^{jw}\right)\right|^2 + \left|H_1\left(e^{jw}\right)\right|^2 - 1\right\}$ (11)

The two methods, the window method and Remez exchange algorithm are used generally to design prototype FIR filter, $H(z)$. The objective function can be minimized using iteration procedure.

The peak reconstruction error [8] is given as

$PRE = \max 20\log_{10}\left\{\left|H_0\left(e^{jw}\right)\right|^2 + \left|H_1\left(e^{jw}\right)\right|^2\right\}$ (12)

Window method is closed form method and is applied in paper [9]. The design entails a relatively insignificant amont of computation. Disadvantage of window method is that it needs a higher-order filter to satisfy the requird specifications. A higher-order filter means more computations per sample, which implies that these filters are slower and less efficient in real-time applications. Design of Remez exchange Design of Remez exchange algorithm is optimal. Minimum filter order implies a more efficient and faster filter for real-time applications.

III. SIMULATED RESULTS

In QMF filter bank only a single prototype filter is required for the design and all other filters are derived from that prototype filter. The simulation is done using matlab tool firpr2chfb. For the design of low pass filter design the specification are length of filter (N+1) =62, passband frequency w_p = 0.37π .

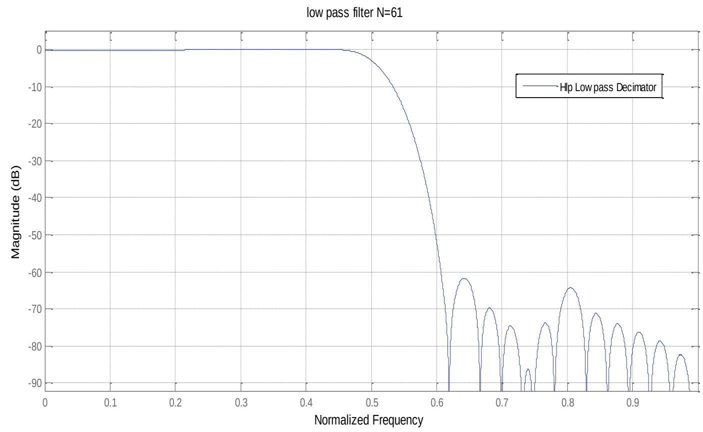

Fig. 3. Magnitude responce of lowpass analysis filter.

The analysis highpass filter is designed using lowpass prototype filter and is decimated by factor of two. In paper [10], proposes an efficient method for design of two channel QMF bank for subband image coding. The choice of the filter bank is important as it effect the system design complexity. The design problem is formulated as weighted sum of reconstructed error and passband and stopband energy of lowpass filter.

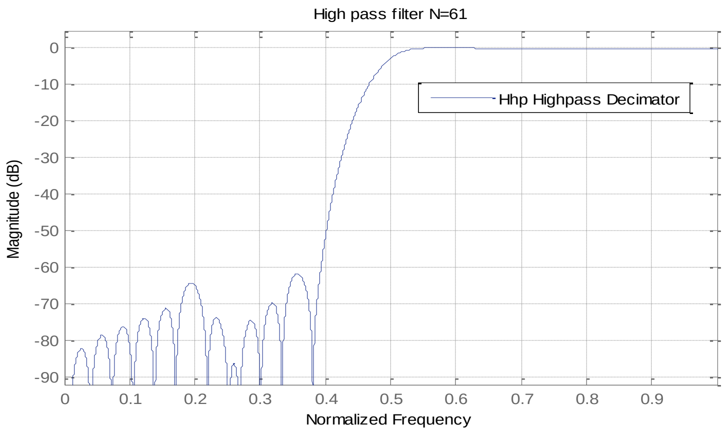

Fig. 4 . Magnitude responce of lowpass analysis filter.

The synthesis lowpass and highpass filter is derived from lowpass prototype filter and is interpolated by factor of two.The fig. 5, and fig. 6 shows magnitude responses of synthesis lowpass and highpass filter respectively.

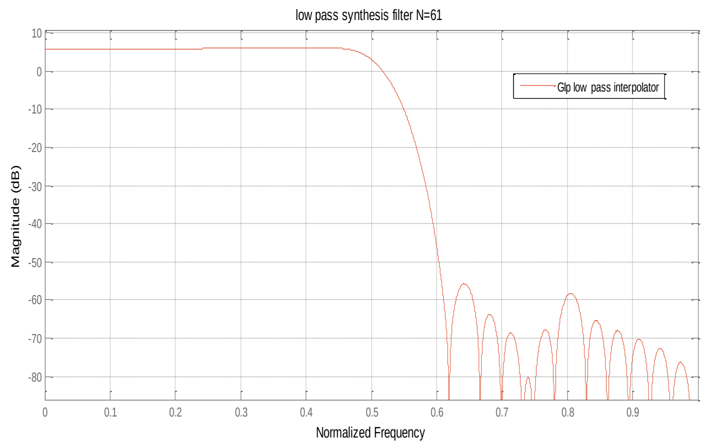

Fig. 5 . Magnitude responce of lowpass analysis filter.

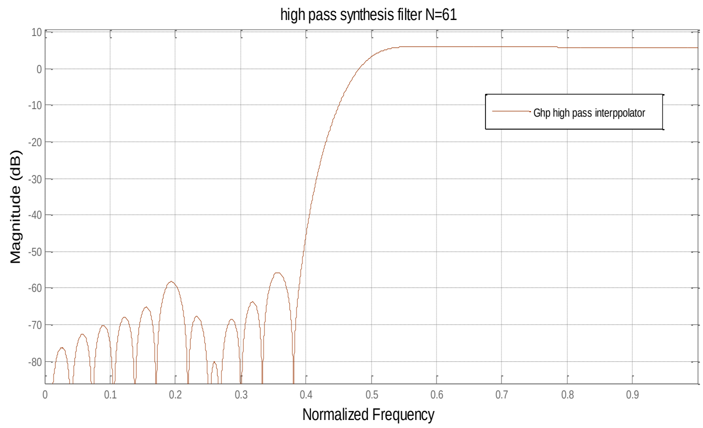

Fig. 6. Magnitude responce of lowpass analysis filter.

The combined magnitude response of all the four filter ,lowpass and highpass analysis and synthesis is shown in fig. 7

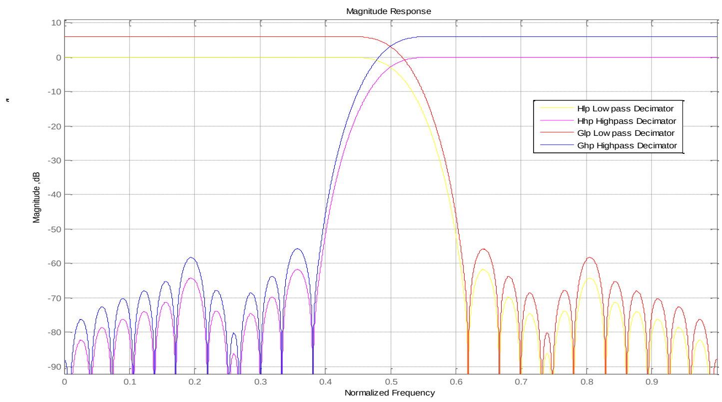

Fig. 7 . Combined magnitude responce of analysis and synthesis filters.

IV. RESULTS ANALYSIS

The simulation using matlab tool firpr2chfb for prototype filter of length (N+1)=62 and passband frequency $w_p = 0.37\pi$ as shown in table 1, number of multipliers per input sample and number of adders per input sample of synthesis filters is 62 and 60 respectively and number of multipliers per input sample and number of adders per input sample of analysis filters is 31 and 30.50 respectively.

Table 1. Comparison of Analysis and Synthesis filters.

Filter s	H0(z)	H1(z)	G0(z)	G1(z)
No. of Multipliers	62	62	62	62
No. of Adders	61	61	60	60
No. of States	60	60	30	30
MPIS	31	31	62	62
APIS	30.5	30.5	60	60

Table 2. Shows the comparision of proposed work with the existing work. Proposed work for design specification $\omega_p = 0.37\pi$, As (Stopband attenuation) = 80dB, and length of filter (N+1) = 61. The performance is evaluated in terms of Peak Reconstruction error and the result shows that, the reconstruction error in dB is 0.0043, which is very less as compared to the existing methods.

Table 2. Result Comparison.

Method	As(dB)	Reconstruction error	Phase Response
Paper [6]	33	0.015	Linear
Paper[7]	34	0.016	Linear
paper[5]	49.5	0.009	Non-Linear
Paper[9]	80	0.0086	Linear
Proposed method	80	0.0043	Linears

V. CONCLUSION

The filters designed with the remez function have smaller maximum deviation and performance is evaluated in terms of Peak Reconstruction Error. The MATLAB based results show that when compared with Kaiser Window based filters the proposed method provide better reconstruction error =0.0043. From the simulated result of QMF filter bank using matlab it is observed that the number of adders and multipliers per input sample of lowpass and highpass filters are equal as highpass filter is derived from low pass filter but, the number of adders and multipliers per input sample of the synthesis filters is almost double as compared to the analysis filters.

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