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Design of Near Perfect Reconstruction Filter Banks with Real Coefficients

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Abstract- The aim is to design good filter banks, where the filters have good stop band and pass band characteristics. In this paper we present a power full design technique that yields both linear phase and non linear phase filters here we consider all filters to have real coefficients and that all complex conjugate operations disappear. In this paper the main objective is to find the angular Frequency Vs Magnitude.
Keywords-LPF(Low Pass Filter), NPR (near perfect reconstruction), NPRTD(Transfer Domain), FWLE(Finite word Length Effect),Basis Vector.

I.INTRODUCTION

In many applications is need to be decompose a signal into a number of components or to assemble a number of signals into one signal. According to whether the signal is decomposed by the analysis network and then reassembled by synthesis network or vice versa. [2] There are two different applications that is filter banks, and transmultiplexers.

The concept of transmultiplexing at the basis of a number of digital modulation and demodulation problems. In digital modulation, the goal is to combine a number of digital signals into one discrete time signal that is then transmitted over a medium like telephone, cable and wireless.

At the receiver end the signals or then separated by the demodulated and routed to the correct users. Here we decompose a signal into two sets of components like $x[n]=x_a[n]+x_d[n]$

Where $x_a[n]$ represents the low frequency approximation

$x_d[n]$ represents the high frequency approximation

We consider the signal as a vector x and we find a set of basis signals g_k and h_k, indexed by the integer k, such that the signal is given by

$$x=\sum_{k=-\propto}^{+\propto} g_k a[k]+\sum_{k=-\propto}^{+\propto} h_k\, d[k] \qquad [1]$$

Here $x_a=\sum_{k=-\propto}^{+\propto} g_k a[k]$ and $x_d=\sum_{k=-\propto}^{+\propto} h_k\, d[k]$ and the two sequences a[k] and d[k] or expansion coefficients over the new basis signals.

The corresponding dual basis vectors $\tilde{g}_k$ and $\tilde{h}_k$ by an inner product as

$$a[k]=<\tilde{g}_k, x> = \sum_{n=-\propto}^{+\propto} \tilde{g}_k[n]x[n]$$

$$[2]$$

$$d[k]=<\tilde{h}_k, x> = \sum_{n=-\propto}^{+\propto} \tilde{h}_k[n]x[n]$$

The decomposition of the vector (signal) x into two components along the "g" and the "h" is shown in the below eq.

$$g_k[n]= g[n-2k]$$

Where k is an integer and n is number of samples.

$$[3]$$

$$h_k[n]= h[n-2k]$$

At the end, to implement the system using familiar [10] operations, such as filtering, upsampling and downsampling. So considering only particular class of basis signals g_k and h_k of the form shown in the above eq.

Analogously for the corresponding dual basis $\tilde{g}_k$, $\tilde{h}_k$,

$$\tilde{g}_k\ [n]=[n-2k]$$

$$[4]$$

$$\tilde{h}_k[n]=[n-2k]$$

These four signals will be the impulse response of filter that was implemented. From the 1 and 2 eq.'s the equation can be rewritten as

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$$X[n]=\sum_{k=-\propto}^{+\propto} g[n-2k]a[k]+\sum_{k=-\propto}^{+\propto} h[n-2k]\,d[k] \qquad [5]$$

Where a[k] and d[k] are given by explicitly rewriting the equation [2] as

$$\{a[k]=\sum_{k=-\propto}^{+\propto} \tilde{g}^{*}[n-2k]x[n] \qquad [6]$$

$$d[k]=\sum_{k=-\propto}^{+\propto} \tilde{h}^{*}[n-2k]x[n]\}$$

the denominator coefficients of the transfer function are

$$\{a[k]=\sum_{k=-\propto}^{+\propto} \tilde{g}^{*}[-(2k-n)]x[n] \qquad [7]$$

$$d[k]=\sum_{k=-\propto}^{+\propto} \tilde{h}^{*}[-(2k-n)]x[n]\}$$

II. Perfect Reconstruction Conditions in the Time Domain

$$\{<\tilde{g}_{k}, g>=\delta[k-\ell]$$

$$<\tilde{h}_{k}, h>=\delta[k-\ell] \qquad [8]$$

$$<\tilde{g}_{k}, h> = <\tilde{h}_{k}, g> =0\}$$

The dual basis vector $\tilde{g}_k$ and $\tilde{h}_k$ are the orthogonals of basis vectors g_k and h_k .

III.Perfect Reconstruction Condition in the Transform Domain

$$\tilde{G}^{*}(1/z^{*})G(z)+\tilde{G}^{*}(-1/z^{*})G(-z)=2$$

$$\tilde{H}^{*}(1/z^{*})H(z)+\tilde{H}^{*}(-1/z^{*})H(-z)=2$$

$$\tilde{G}^{*}(1/z^{*})H(z)+\tilde{G}^{*}(-1/z^{*})H(-z)=2$$

These four conditions can be written in matrix form as

$$\begin{bmatrix}\tilde{G}(1/z) & \tilde{G}(-1/z)\\ \tilde{H}(1/z) & \tilde{H}(-1/z)\end{bmatrix}\begin{bmatrix}G(z) & H(z)\\ G(-z) & H(-z)\end{bmatrix}=\begin{bmatrix}2 & 0\\ 0 & 2\end{bmatrix}$$

If we consider all vectors in the plane,the inner product can be seen as

$\langle x,y\rangle=\|x\|\,\|y\|\cos\theta$ where θ is the angle between x and y if x and y are orthogonal to each other then $\langle x,y\rangle=0$

Here the main concept is to split the signal in to number of signals (analysis network)and down sample the signal by a factor D and reassemble the signal by interpolate the signal by a factor.

Finally by using finite word length effect (FWEL) method [10] we truncate the coefficients up to required digits.

The low pass filter(LPF)allows the low frequencies of the signal and rejects the high frequencies of the signal.

3.1Basis vector

A **basis** is a set of linearly independent vectors that, in a linear combination, can represent every vector in a given vector space or free module, or, more simply put, which define a "coordinate system" (as long as the basis is given a definite order). In more general terms, a basis is a linearly independent spanning set.

Given a basis of a vector space, every element of the vector space can be expressed uniquely as a finite linear combination of basis vectors. Every vector space has a basis, and all bases of a vector space have the same number of elements, called the dimension of the vector space.

IV.Matlab Results

In this section, simulation results for signal to noise ratio and estimated filter frequency responses. Their reconstruction error measure graphs are also included.

For case 1 NPR COEFFICIENTS generates near NPR filter bank coefficients N,L

Given values L=64
N=256

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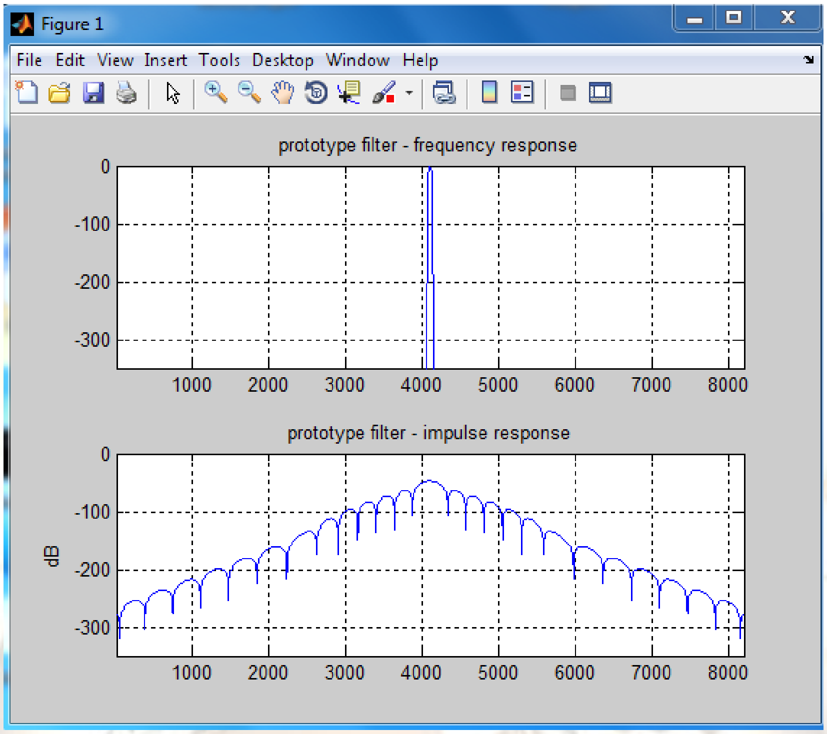

Figure 1:SNR Vs Filter Responses(Frequency and impulse)

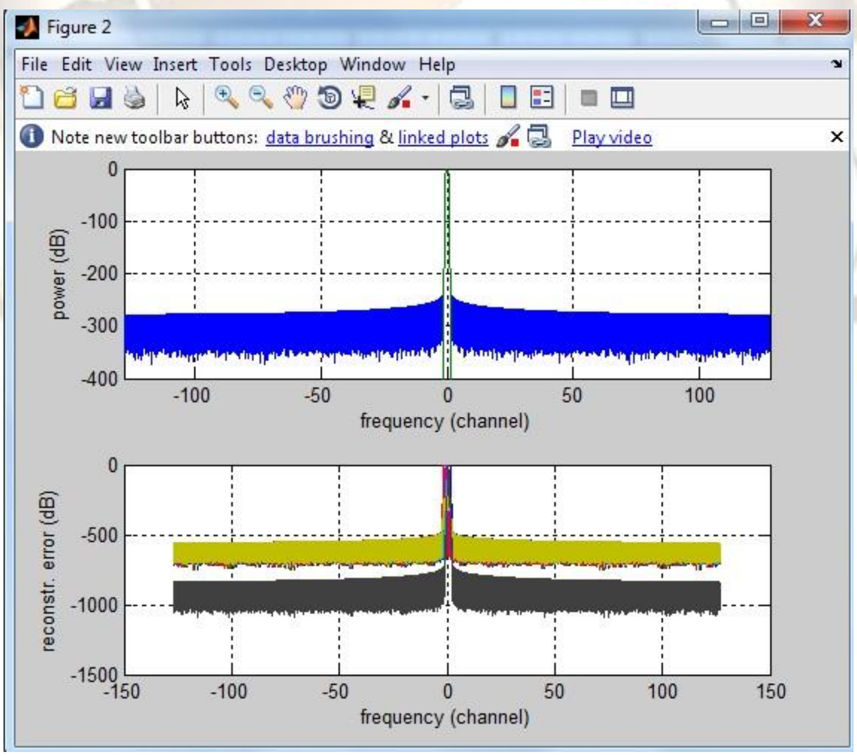

Figure 2:Error(db) Vs Frequency(channel)

For case 2
Given values L=16

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N=64

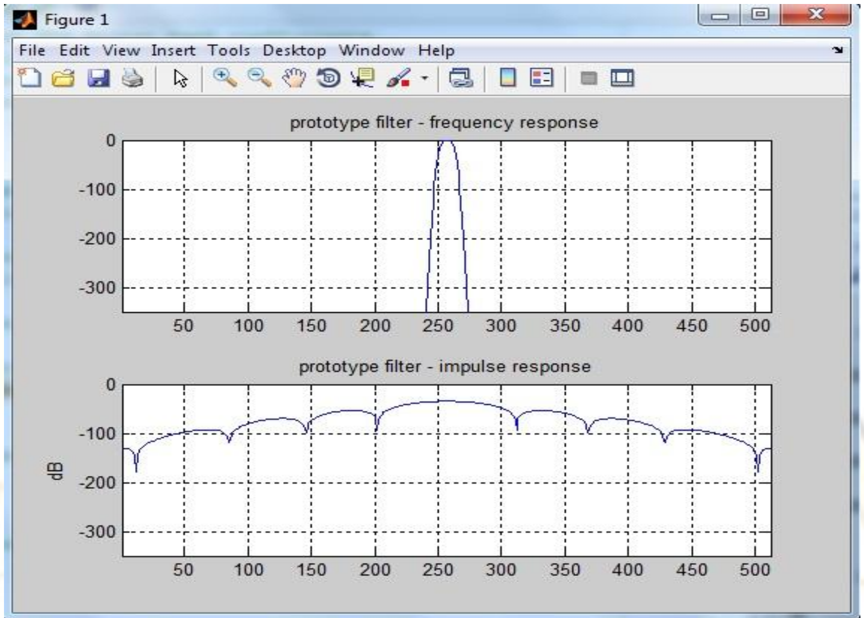

Figure 3:SNR Vs Filter Responses(Frequency and impulse)

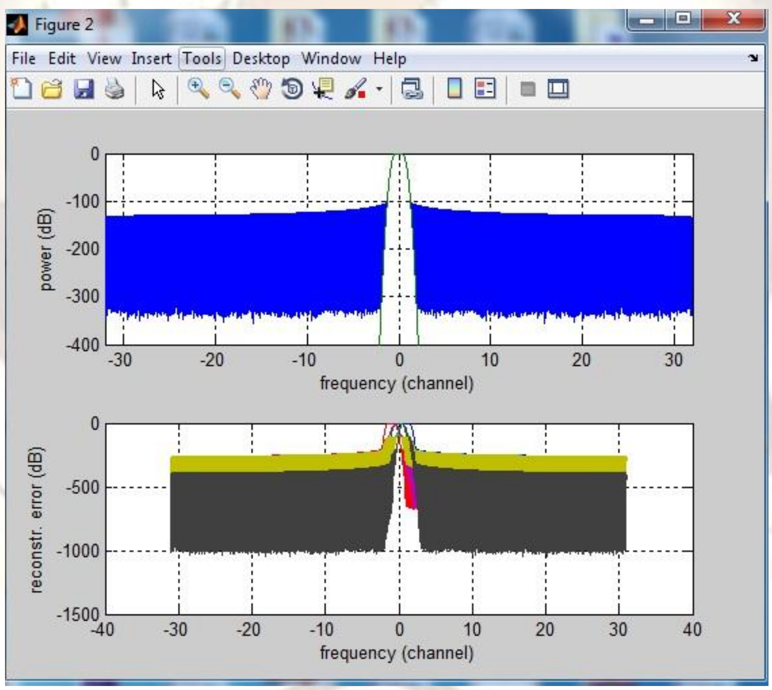

Figure 4:Error(db) Vs Frequency(channel)

Near perfect reconstruction polyphase filter bank graphs here taken coefficients N,L,K

For N=256,L=128,and K=11.4 is simulated graphs given below says the input output vs error value and time

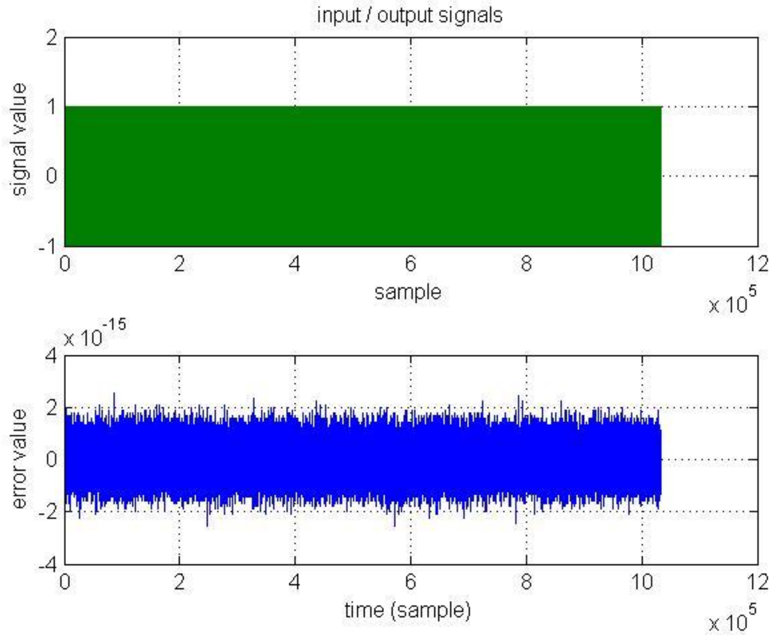

Figure 5:Error(db) Vs Time (samplel)

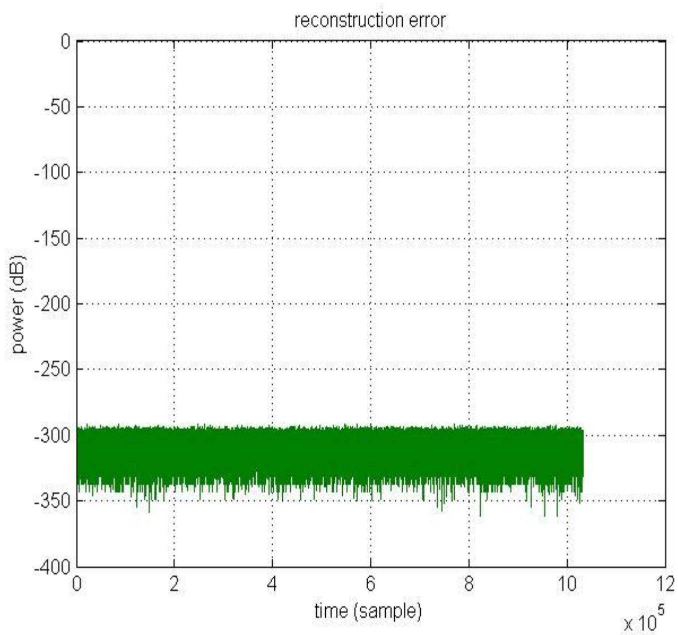

Figure 6:Error(db) Vs Time (samplel)

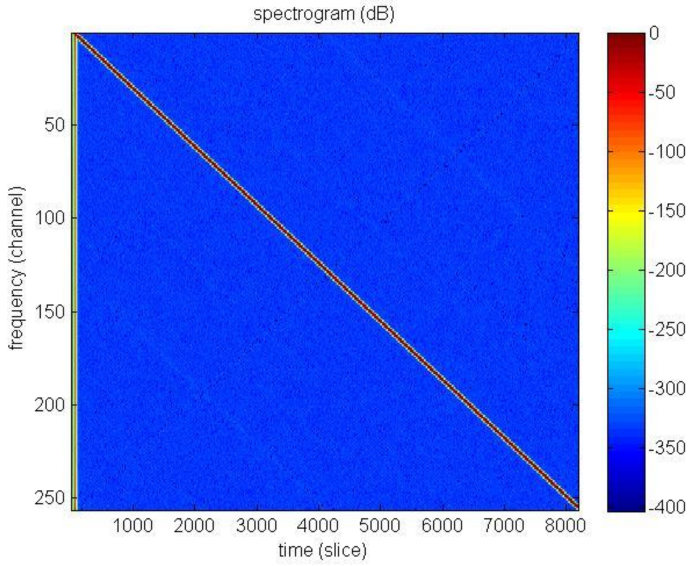

Figure 7:Frequency(channel) Vs Time (slice)

V. CONCLUSION

Our method gives the better performance, compare with existing algorithms. it is applicable for the NPR main objective is to find the filter Frequency Responses and Reconstructing error is achieved.

VI. ACKNOWLEDGEMENT

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