Analysis of the reconstructed signal using a two-channel reconstruction FIR filter bank.

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Abstract - This paper discusses about the application of two-channel reconstruction FIR filter banks which constructs the uses the analysis bank and synthesis bank that have been designed for Quadrature Mirror Filter (QMF) Banks. The paper focuses on the improvement of performance over the conventional designs as QMF banks have been used. The improved reconstructed signal at QMF bank differs from those usually used for filter design in a way which makes the usual filter design techniques difficult to apply. The analysis bank and synthesis bank are designed on the emphasis of the: stop band rejection, transition band width, and pass and transition band. Unlike most the other filter designs problems regarding the transition of different bands.

Keywords – Quazi Newton optimization, filter banks, Analysis bank & synthesis bank, Quadrature Mirror Filter, gradient.

I. INTRODUCTION

Filter banks are used in signal processing applications such as speech coding, TDM-FDM, trans multiplexing, image coding and the reconstruction of the signal from the individual bands [6][7]. In this paper, QMF Banks represents two new approaches to the design of two-channel perfect-reconstruction linear-phase FIR filter banks.

The analyze and the design filter on the impulse responses of the analysis filter bank is direct.

The synthesis filter bank is formulated by changing the signs of odd-order coefficients in the analysis of the technique and it provides with two band division and permits an alias-free reconstruction of the signal. Both bands are divided by a factor of two, the reconstruction of this method is referred to as a Quadrature Mirror Filter (QMF). [8][9]. The QMF technique provides a reduction in filter order with the Nyquist frequencies of the bands and therefore a more efficient design is come to play.

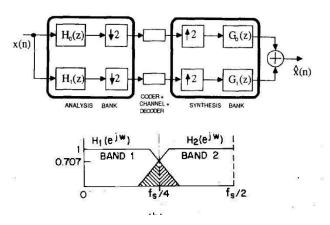


Fig. 1 Two-channel analysis and synthesis filter bank.

Figure 1 shows a generic two-channel FIR filter bank, where $H_0(z)$ and $H_1(z)$ represent the low-pass and high pass filters in the analysis bank, respectively, and $G_0(z)$ and $G_1(z)$ are the synthesis filters. With the assumption that channels and codecs are perfect we can relate the reconstructed signal as [1][2].

$$\begin{split} X(z) &= \frac{1}{2} [H_0(z) G_0(z) + H_1(z) G_1(z)] x(z) \\ &+ \frac{1}{2} [H_0(-z) G_0(z) \\ &+ H_1(-z) G_1(z)] x(-z)] \end{split}$$

The remainder of this paper elaborates on the above discussion and is structured as follows. Section II describes the problem formulation. Section III then focuses on results and the paper finally concludes in section IV.

II. PROBLEM FORMULATION

Quasi-Newton Optimization algorithm is used to simulate the Quadrature Mirror Filter. The algorithm requires the formulation of the error function and the gradient which is represented with "E" and "gra" respectively. The time that a pulse is present to the total transmission time [2, 3].

$$E = E_r + \alpha E_s$$

Here, Er is the reconstructed error due to channel amplitude response, Es is the stop band error and α represents weight constant [3]

$$\begin{split} E_r &= 2 \sum [|H(e^{j\omega})|^2 + |H(e^{j(\omega + \pi)})|^2 - 1]^2 \\ E_s &= \sum [|H(e^{j\omega})| - 0]^2 = \sum [|H(e^{j\omega})|]^2 \end{split}$$

Where the summation limit for E_r is $\omega = 0$ to $\pi/2$ and summation limit for E_s is ω_s to $\pi/3$.

$$|H(e^{j\omega})| = \sum b(k)\cos[\omega(k-0.5)]$$

Where the sum is from N/2 to k.

$$b(k) = 2h\left(\frac{N}{2} - k\right), k = 1, 2, \dots \frac{N}{2}$$

The coefficient vector is,

$$\bar{x} = \left[x(1)x(2) \dots \dots x\left(\frac{N}{2}\right) \right]$$
and
$$\bar{c} = \left[\cos(\omega(k - 0.5))^T \right]$$

For each frequency point ωi we are calculating $|H(e^{jwi})|$ and $|H(e^{j(wi+\pi)})|$ in the band $\theta \rightarrow \pi/2$ and the desired channel magnitude vector,

$$\bar{u} = [1, 1, \dots \dots 1],$$

with length of w_1

$$\begin{aligned} \left| H(e^{j\omega i}) \right| &= \bar{x}^T \bar{c} \\ \text{and} \\ \left| H(e^{j\omega i}) \right|^2 &= \bar{x}^T Q_1 \bar{x} \end{aligned}$$

Similarly,

$$\begin{aligned} \left|H\left(e^{j(\omega i+\pi)}\right)\right|^2 &= \bar{x}^T.Q_2\bar{x} \\ \left|H\left(e^{j\omega i}\right)\right|^2 + \left|H\left(e^{j(\omega i+\pi)}\right)\right|^2 &= \bar{x}^T.Q\bar{x} \\ E_{ri} &= \left[\bar{x}^TQ\bar{x} - 1\right]^2 \end{aligned}$$

for each $\omega i \in w_1$, $zI_i = \overline{x}.T.Q.\overline{x}$, calculated for every ωi .

$$el_i = zl_i - ul_i$$

and for every frequency point let,

$$sl_i = \frac{dzl_i}{d\bar{x}}$$

Let $errf_i$ be the error function,

$$errf_i = e_1^T e_1$$

And, the gradient,

$$gra_1 = 2(\bar{s}l_i)(e_{1_i})$$

It is similar for the stop band [3]. The total error and gradient are given by:

$$\begin{split} &Errf(total) = errf_1 + ALFA.* \, errf_2 \\ &gra(total) = gra_1 + ALFA.* \, gra_2 \end{split}$$

III. RESULTS

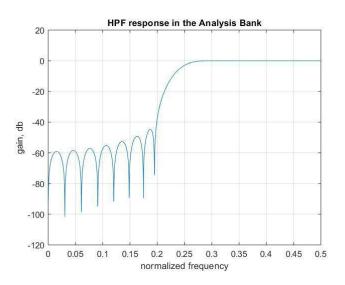


Fig. 2 High pass filter response in the analysis bank.

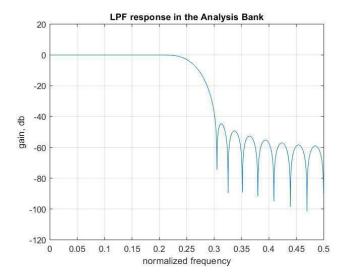


Fig. 3 Low pass filter response in the analysis bank.

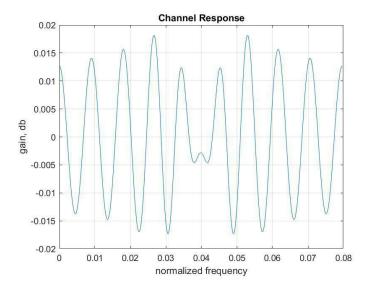


Fig.4 Reconstructed channel response in the analysis bank.

The above figures illustrate the analysis bank responses and the reconstructed channel responses respectively. And the figure 5 shows the output response of the synthesis bank after applying the 100-points ramp function to the analysis bank.

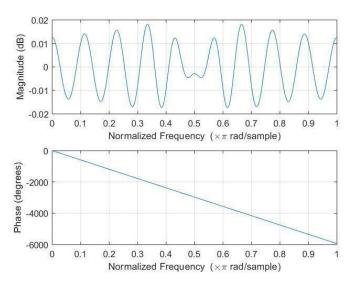


Fig.5 Reconstructed channel response in the synthesis bank.

After the simulation the reconstructed signal to noise ratio of the of the MSE has been found is 56.631345 db.

IV. CONCLUSION

To conclude this paper, it is necessary to mention that, despite of having an unsophisticated algorithm, optimization of QMF is very useful. Use of QMF enables to avoid aliasing effects due to dissemination. Consequently, band splitting can be performed up to large number of sub-bands without using sophisticated filters [5]. This derived filter provides a good family of

designs that allow the designer to choose characteristics that are appropriate to signal being filtered [3].

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