An Efficient Algorithm to Design Nearly Perfect-Reconstruction Two-Channel Quadrature Mirror Filter Banks

S. K. Agrawal\(^{(C.A.)}\) and O. P. Sahu\(^{*}\)

Abstract: In this paper, a novel technique for the design of two-channel Quadrature Mirror Filter (QMF) banks with linear phase in frequency domain is presented. To satisfy the exact reconstruction condition of the filter bank, low-pass prototype filter response in pass-band, transition band and stop band is optimized using unconstrained indirect update optimization method. The objective function is formulated as a weighted sum of pass-band error and stop-band residual energy of low-pass prototype filter, and the square error of the distortion transfer function of the QMF bank at the quadrature frequency. The performance of the proposed algorithm is evaluated in terms of Peak Reconstruction Error (PRE), mean square error in pass-band and stop-band regions and stop-band edge attenuation. Design examples are included to illustrate the performance of the proposed algorithm and the quality of the filter banks that can be designed.

Keywords: Nearly Perfect-Reconstruction, Optimization, Phase Distortion, Two-Channel Filter Banks.
signal $\hat{x}(n)$ suffers from Aliasing Distortion (ALD), Phase Distortion (PHD), and Amplitude Distortion (AMD) due to the fact that the analysis and synthesis filters are not ideal [28]. Therefore, the main emphasis of most of the researchers while designing the prototype filter for two-channel QMF bank has been on the elimination or minimization of these three errors to obtain a Perfect Reconstruction (PR) or NPR system [12-17, 28].

By setting the synthesis filters cleverly in terms of the analysis filters, aliasing can be cancelled completely and PHD has been eliminated by using linear phase FIR filters [12-17]. The overall transfer function of such an ALD and PHD free system turns out to be a function of the filter tap weights of the low-pass prototype filter only that is known as low-pass prototype filter. In QMF banks, the high-pass and low-pass analysis filters are related to each other by the mirror-image symmetry condition: $H_2(z) = H_1(-z)$, around the quadrature frequency $\pi/2$. Due to this symmetry constraint, the AMD can only be minimized in this case by optimizing the filter tap weights of the low-pass prototype filter [28].

This paper proposes a novel algorithm to design the two-channel QMF bank without using any matrix inversion which generally affects the performance and effectiveness of some of optimization methods. The error measure to be minimized is formulated as a weighted sum of pass-band error and stop-band residual energy of low-pass prototype filter and the square error of the distortion transfer function of the QMF bank at the quadrature frequency $\omega = \pi/2$. Unconstrained variable metric method [29] is used to minimize the error measure by optimizing the filter tap weights of the low-pass prototype filter.

The paper is organized as follows. Section 2 briefly describes the principle of two-channel QMF bank and then formulation of the design problem in frequency domain. Section 3 presents proposed algorithm for minimization of the objective function. In section 4, we discuss the design results of the proposed QMF bank and comparison with different methods. Finally, some concluding remarks are drawn in section 5.

2 The Design Problem

The expression for the overall system function, or distortion transfer function of the alias free two-channel QMF bank can be written as [15-23]

$$T(z) = \frac{1}{2}[H_1^2(z) - H_1^2(-z)]$$

(1)

where, for alias cancellation, synthesis filters are defined as given below:

$$F_1(z) = H_2(-z) \text{ and } F_2(z) = -H_1(-z)$$

(2)

To obtain the perfect reconstruction QMF bank, the overall transfer function $T(z)$ must be a pure delay, i.e.,

$$T(z) = \frac{1}{2}[H_1^2(z) - H_1^2(-z)] = cz^{-n_0} \text{ or}$$

$$\hat{x}(n) = cx(n - n_0)$$

(3)

Equation (3) clearly shows that if the prototype filter $H_1(z)$ is selected to be a linear phase FIR, then $T(z)$ also become linear phase FIR and phase distortion is eliminated. To assure the linear phase FIR constraint, impulse response $h[n]$ of the low pass prototype filter $H_1(z)$ should be symmetric $h[n] = h[N-1-n], 0 \leq n \leq N-1$. $N$ is the filter length [1]. With this selection, the corresponding frequency response is given by [30].

$$H_1(e^{j\omega}) = A(\omega)e^{-j\omega(N-1)/2}$$

(4)

where $A(\omega) = \pm |H_1(e^{j\omega})|$ is the amplitude function. For real impulse response, the magnitude response $|H_1(e^{j\omega})|$ is an even function of $\omega$, hence, by substituting Eq. (4) into Eq. (1), the overall frequency response of the two-channel QMF bank becomes

$$T(e^{j\omega}) = \frac{1}{2}(e^{-j\omega(N-1)}|H_1(e^{j\omega})|^2 -$$

$$(-1)^{(N-1)}|H_1(e^{j(\pi-\omega)})|^2]$$

(5)

For odd filter length, above equation gives $T(e^{j\omega}) = 0$ at $\omega = \pi/2$, this entails severe amplitude distortion around quadrature frequency. Therefore, $N$ must be chosen even to avoid this distortion and from Eq. (5), Exact Reconstruction (ER) condition can be written as

$$T(\omega) = |A(\omega)|^2 + |A(\pi-\omega)|^2 = c \ , \text{ for all } \omega.$$  

(6)

In this case after eliminating ALD and PHD, we can only minimize amplitude distortion rather than

**Fig. 1** The 2-channel quadrature mirror filter bank.
completely eliminated due to mirror image symmetry constraint [30]. If the prototype filter \( H(z) \) characteristics are assumed ideal in pass band and stop band regions then the exact reconstruction condition is automatically satisfied in the range of frequencies 
\[ 0 < \omega < \omega_p \text{ and } \omega_p < \omega < \pi \] [1], where, \( \omega_p \) and \( \omega \) are pass band and stop band edge frequencies, respectively. The main difficulty comes in transition band region 
\[ (\omega_p < \omega < \omega_s) \], therefore the AMD must be controlled in this region. Hence, the aim is to optimize the coefficients of \( H(z) \), such that the exact reconstruction condition nearly satisfied.

To design the low-pass analysis filter \( H(z) \), we propose to minimize the following error function '\( \phi \)' based on the method presented in [19].

\[
\phi = \alpha_1 \phi_p + \alpha_2 \phi_s + \alpha_3 \phi_t
\] (7)

where \( \phi_p \) and \( \phi_s \) are the measure of pass-band error and stop-band residual energy of filter \( H(z) \), \( \phi_t \) is the square error of \( T(z) \) at \( \pi/2 \), in transition band, and \( \alpha_1, \alpha_2 \) and \( \alpha_3 \) are real constants.

The ER condition of Eq. (6) at \( \omega = \pi/2 \), can be reduced to [19].

\[
A \left( \frac{\omega}{2} \right) = (1/2)^{1/2} A(0)
\] (8)

where \( A(0) \) and \( A(\pi/2) \) are the amplitude responses of prototype filter at zero frequency and quadrature frequency, respectively. Consequently, the square error \( \phi_t \) is given by:

\[
\phi_t = \left[ A \left( \frac{\omega}{2} \right) - (1/2)^{1/2} A(0) \right]^2
\] (9)

\( \phi_p \) and \( \phi_s \) may be taken as:

\[
\phi_p = \frac{1}{\pi} \int_0^{\omega_p} \left[ A(0) - A(\omega) \right]^2 d\omega
\] (10)

\[
\phi_s = \frac{1}{\pi} \int_{\omega_s}^{\pi} \left[ A(\omega) \right]^2 d\omega
\] (11)

To design the low-pass analysis filter \( H(z) \), we propose to minimize the following error function ‘\( \phi \)’ based on the method presented in [19].

\[
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\[
\phi_s = \frac{1}{\pi} \int_{\omega_s}^{\pi} \left[ A(\omega) \right]^2 d\omega
\] (11)

3 Computation of Optimum Filter Tap Weights

For real symmetric impulse response with even \( N \), the corresponding frequency response of low-pass prototype filter \( H_1(e^{j\omega}) \) is given by [1].

\[
H_1(e^{j\omega}) = \left[ \sum_{n=0}^{(N/2)-1} 2h(n) \cos \omega \left( \frac{(N-1)}{2} + n \right) \right] e^{-j\omega (N-1)/2}
\] (12)

\[
H_1(e^{j\omega}) = A(\omega) e^{-j\omega (N-1)/2}
\] (13)

where \( A(\omega) \) is the amplitude function and given by:

\[
A(\omega) = 2h_1^Tc(\omega) = b^Tc(\omega)
\] (14)

The vector \( c(\omega) \) and vector \( b^T = 2h_1^T \) are given by:

\[
c(\omega) = [\cos \omega ((N-1)/2), \cos \omega ((N-1)/2) - 1, \ldots, \cos (\omega/2)]^T, \ b = [b_0 b_1 b_2 \ldots b_{(N/2)-1}]^T
\] (15)

At \( \omega = 0 \), the amplitude function is calculated as

\[
A(0) = b^Tc(0) = b^T1,
\] (16)

where \( 1 \) is the vector of all 1’s.

3.1 Expressions for \( \phi_p, \phi_s \) and \( \phi_t \)

By using Eqs. (9), (14) and (16), \( \phi_t \) can be expressed as:

\[
\phi_t = \left[ b^Tc(\pi/2) - (1/2)^{1/2} b^T1 \right]^2 = [b^Tq - A_1]^2
\] (17)

where vector \( q \) is equal to vector \( e(\omega) \), when it is evaluated at \( \omega = \pi/2 \) and \( A_1 = 0.707 A(0) \).

Similarly pass band error \( \phi_p \) can be realized.

\[
\phi_p = b^TWb
\] (18)

where \( W \) is a real, symmetric and positive definite matrix, given by

\[
W = \int_0^{\omega_p} \left[ c(\omega) - c(0) \right] \left[ c(\omega) - c(0) \right]^T d\omega
\] (19)

3.2 Minimization of the Objective Function

The error function or objective function ‘\( \phi \)’can be realized by substituting Eqs. (17), (18) and (20) into Eq. (7).

\[
\phi = \alpha_1 b^T W b + \alpha_2 b^T Z b + \alpha_3 [b^T q - A_1]^2
\] (22)

\[
2A_1 b^T q + A_1^2
\] (23)

where matrix \( U \) and \( Y \) are given by:

\[
U = \alpha_1 W + \alpha_2 Z + \alpha_3 Y \quad \text{and} \quad Y = qq^T
\] (24)

The error function given by Eq. (22) is a quadratic function and matrix \( U \) is a symmetric and positive definite matrix, therefore, \( \phi \) can be minimized by unconstrained variable metric method [29]. In this method the inverse of Hessian matrix is approximated using Broyden-Fletcher-Goldfarb-Shanno formula. This method has self-correcting properties and exhibits super-linear convergences near the optimal point therefore, it is suitable for QMF design problem. If \( b_i \) is the approximation of the minimum point at the \( i \)th stage of iteration and \( \lambda_i \) is the optimal step length in the search
direction, then the improved approximation can be calculated as:

$$b_{i+1} = b_i - \lambda_i [H] \nabla \phi_i = b_i + \lambda_i s_i$$

(25)

where $\nabla \phi_i$ is the gradient of the objective function $\phi$ and $s_i$ is the search direction, when evaluated at the design vector $b_i$, both are given by:

$$\nabla \phi_i = 2Ub + \alpha_j [-2A_i q]$$

(26)

and

$$s_i = -[H] \nabla \phi_i ,$$

(27)

and matrix $[H]$ is the estimate of inverse of Hessain matrix. Initially, the matrix $[H]$ is taken as the identity matrix $[I]$ and up-dation of this matrix is done using Broyden-Fletcher-Goldfarb-Shanno formula [29].

$$[H_1] = [H] + \frac{1}{\lambda_i} \left( \begin{array}{c|c} g_i & d_i \\ \hline d_i^T & d_i^T - \frac{d_i g_i}{d_i^T g_i} \end{array} \right)^T g_i$$

(28)

where

$$g_i = \nabla \phi_{i+1} - \nabla \phi_i$$

(29)

and

$$d_i = b_{i+1} - b_i = -\lambda_i [H] \nabla \phi_i$$

(30)

In the direction of $s_i$, the optimum step length $\lambda_i$ can be obtained by equating the derivative of error function ($b_i + \lambda_i s_i$) with respect to $\lambda_i$ to zero. The derivative $\partial \phi / \partial \lambda$ is given, follows giving term:

$$\lambda_i = \left( \begin{array}{c|c} g_i^T s_i - \alpha_3 A_i q^T s_i \end{array} \right) / \left( \begin{array}{c|c} s_i^T g_i \end{array} \right)$$

(31)

Initial values of filter coefficients $h_i(n)$, are chosen as the same which was taken in [14], [23] to satisfy unit energy constraint on the filter coefficients. The step-by-step description of the design algorithm that minimizes the error function is as shown in Table 1.

4 Case Study

A MATLAB program has been written which implements the design procedure for prototype low-pass filter described in the previous section and tested on a desktop computer equipped with an Intel Dual core CPU @ 2.10 GHz with 1 GB RAM. This section presents two design examples to examine the effectiveness of the proposed algorithm. The performance of the algorithm is evaluated in terms of five significant quantities:

(i) Mean square error in the pass band ($\phi_p$),

(ii) Stop band error ($\phi_s$),

(iii) Stop-band first lobe attenuation ($A_s$),

(iv) Stop-band edge attenuation ($A_e$)=

$$-20 \log_{10}(H(\omega_s))$$

(v) Measure of reconstruction error ($e$) in dB =

$$\max_{\omega} \left[ 10 \log_{10} |T (e^{j \omega})| - \min_{\omega} \left| 10 \log_{10} |T (e^{j \omega})| \right. \right]$$

In both the examples stop band first lobe attenuation ($A_s$) has been obtained from respective zoomed

Table 1 Algorithm for designing the prototype filter.

<table>
<thead>
<tr>
<th>Steps of Design Algorithm</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: Select filter length $N$, $\omega_s$ and $\omega_s$.</td>
</tr>
<tr>
<td>2: Start with an initial design vector $b_i = [h(0) h(1) h(2) \ldots h(N/2) - 1]), b_i is zero except $h((N/2) - 1) = \frac{1}{\sqrt{N}}$, and assume initial values of $\alpha_1$, $\alpha_2$ and $\alpha_3$.</td>
</tr>
<tr>
<td>3: Compute $b_0 = 2b_i$ and set the iteration number, $i = 1$.</td>
</tr>
<tr>
<td>4: Find the objective function $\phi_i$ by using Eq. (22), at the design vector $b_i$.</td>
</tr>
<tr>
<td>5: Compute $\nabla \phi_i$ and the search direction $s_i$, by using Eqs. (26) and (27), at $b_i$. Determine the optimum step length, $\lambda_i$, by using Eq. (31).</td>
</tr>
<tr>
<td>6: Compute the new approximation $b_{i+1} = b_i - \lambda_i [H_i] \ nabla \phi_i = b_1 + \lambda_i s_i$.</td>
</tr>
<tr>
<td>7: Find $\phi_{i+1}$ and $\nabla \phi_{i+1}$ at the design vector $b_{i+1}$. Also compute matrix $[H_{i+1}]$ using Eq. (28). If $\phi_{i+1} \geq \phi_i$, choose the optimum point as $b_i$; stop the procedure and go to step (9). If $\phi_{i+1} &lt; \phi_i$, set $b_i = \phi_{i+1} = b_{i+1}$.</td>
</tr>
<tr>
<td>8: Set the new iteration number as $i = i + 1$, and go to step (4).</td>
</tr>
<tr>
<td>9: Compute $b_i = (1/2) b_i$, as the optimum solution and stop the procedure.</td>
</tr>
</tbody>
</table>

Table 2 Optimized filter tap weights in example 1.

<table>
<thead>
<tr>
<th>Filter Taps ($N = 42$)</th>
</tr>
</thead>
<tbody>
<tr>
<td>$h_0(0)= 0.0003002780$</td>
</tr>
<tr>
<td>$h_0(1)= 0.0005656455$</td>
</tr>
<tr>
<td>$h_0(2)= -0.0008973255$</td>
</tr>
<tr>
<td>$h_0(3)= -0.0010666203$</td>
</tr>
<tr>
<td>$h_0(4)= 0.0026311284$</td>
</tr>
<tr>
<td>$h_0(5)= 0.00143944439$</td>
</tr>
<tr>
<td>$h_0(6)= -0.0057610160$</td>
</tr>
<tr>
<td>$h_0(7)= -0.00129397138$</td>
</tr>
<tr>
<td>$h_0(8)= 0.01075131240$</td>
</tr>
<tr>
<td>$h_0(9)= -0.0003089173$</td>
</tr>
<tr>
<td>$h_0(10)= -0.0181847013$</td>
</tr>
<tr>
<td>$h_0(11)= -0.0349951307$</td>
</tr>
<tr>
<td>$h_0(12)= 0.02886533479$</td>
</tr>
<tr>
<td>$h_0(13)= -0.01076104979$</td>
</tr>
<tr>
<td>$h_0(14)= -0.0445102234$</td>
</tr>
<tr>
<td>$h_0(15)= 0.02518183956$</td>
</tr>
<tr>
<td>$h_0(16)= 0.07003271098$</td>
</tr>
<tr>
<td>$h_0(17)= -0.05705282394$</td>
</tr>
<tr>
<td>$h_0(18)= -0.12723852401$</td>
</tr>
<tr>
<td>$h_0(19)= 0.1702847829$</td>
</tr>
<tr>
<td>$h_0(20)= 0.59424041885$</td>
</tr>
</tbody>
</table>
Example 2: For filter length \( N = 24 \), \( \omega_s = 0.6 \pi \), \( \omega_p = 0.4 \pi \), \( \alpha_1 = 0.7 \), \( \alpha_2 = 0.1 \) and \( \alpha_3 = 1 \), the filter coefficients obtained (for \( 0 \leq n \leq N/2 - 1 \)) are listed in Table 3.

Normalized magnitude plots of analysis filters \( H_1(z) \), \( H_2(z) \) and distortion function \( T(z) \) are shown in Fig. 3(a). The reconstruction error (in dB) of QMF bank is plotted in Fig. 3(b). The significant quantities obtained are \( \text{PRE} (e) \) in dB = 0.0139, \( (\mathcal{E}_p) = 1.162 \times 10^{-8} \), \( (\mathcal{E}_s) = 7.48 \times 10^{-5} \), \( (A_s) = 25.06 \) dB and \( (A_L) = 34.85 \) dB.

Table 3 Optimized filter tap weights in example 2.

<table>
<thead>
<tr>
<th>Filter Taps ( (N = 24) )</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>( h_1(0) ) = 0.00447423300</td>
<td>( h_1(1) ) = -0.00914936838</td>
</tr>
<tr>
<td>( h_1(2) ) = -0.00294793823</td>
<td>( h_1(3) ) = 0.02070572773</td>
</tr>
<tr>
<td>( h_1(4) ) = -0.00236718296</td>
<td>( h_1(5) ) = -0.03861209528</td>
</tr>
<tr>
<td>( h_1(6) ) = 0.01581765598</td>
<td>( h_1(7) ) = 0.06758584810</td>
</tr>
<tr>
<td>( h_1(8) ) = -0.04819461467</td>
<td>( h_1(9) ) = -0.12906666302</td>
</tr>
<tr>
<td>( h_1(10) ) = 0.16402285083</td>
<td>( h_1(11) ) = 0.60339846987</td>
</tr>
</tbody>
</table>

4.1 Discussion of Results

Table 4 presents a comparison of the proposed method results with other existing methods [19, 20, 21, 25, 26] results (with similar design specifications of two-channel QMF bank i.e., \( \omega_s = 0.6\pi \) and \( \omega_p = 0.4\pi \)) for \( N = 24 \). The proposed method gives improved performance than all the other methods in terms of \( \text{PRE} \) and \( \phi_p \).

The percentage reduction in peak reconstruction error with respect to other existing methods [25, 19, 20, 26, 21] is 53.66%, 44.62%, 31.81%, 26.84%, and 18.71%, respectively, calculated using Eq. (32).

\[
\text{Percentage Reduction in PRE} = \frac{\text{PRE}_{\text{other method}} - \text{PRE}_{\text{proposed method}}}{\text{PRE}_{\text{other method}}} \times 100 \% \tag{32}
\]

The proposed method also gives improved performance than the methods of [19], [25] and [26] in terms of better stop-band error \( \phi_p \) and stop-band edge attenuation \( A_s \). However algorithm presented in [21] obtains best results for \( \phi_p \) and \( A_s \). Table 5 shows that the proposed method is better than considered existing method in terms of computational complexity because, it does not involve any matrix inversion and determination of eigenvectors in each iteration. The proposed method requires less computation time to design the low-pass prototype filter in comparison of other existing methods. Consequently, the overall performance of the proposed method is better than other existing methods and filter bank designed by this method can be used for real time applications.
Table 4 Comparison of the proposed algorithm with other existing optimization algorithms based on the significant quantities for filter length $N = 24$.

<table>
<thead>
<tr>
<th>Methods</th>
<th>$\phi_s$ (dB)</th>
<th>$\phi_p$ (dB)</th>
<th>PRE (dB)</th>
<th>$A_s$ (dB)</th>
<th>Phase Response</th>
</tr>
</thead>
<tbody>
<tr>
<td>Upender et al. [26]</td>
<td>7.99×10^{-5}</td>
<td>1.845×10^{-7}</td>
<td>0.019</td>
<td>22.78</td>
<td>Linear</td>
</tr>
<tr>
<td>Kumar et al. [21]</td>
<td>7.49×10^{-5}</td>
<td>1.61×10^{-7}</td>
<td>0.0202</td>
<td>25.31</td>
<td>Linear</td>
</tr>
<tr>
<td>Algorithm in [20]</td>
<td>8.49×10^{-5}</td>
<td>9.23×10^{-8}</td>
<td>0.0251</td>
<td>23.03</td>
<td>Linear</td>
</tr>
<tr>
<td>Sahu et al. [19]</td>
<td>7.49×10^{-5}</td>
<td>1.61×10^{-7}</td>
<td>0.0202</td>
<td>26.15</td>
<td>Linear</td>
</tr>
<tr>
<td>Ghosh et al. [25] (MJADE pBX algorithm)</td>
<td>1.30×10^{-4}</td>
<td>9.14×10^{-7}</td>
<td>0.030</td>
<td>20.11</td>
<td>Linear</td>
</tr>
<tr>
<td>Ghosh et al. [25] (JADE algorithm)</td>
<td>3.19×10^{-4}</td>
<td>4.10×10^{-6}</td>
<td>0.050</td>
<td>20.25</td>
<td>Linear</td>
</tr>
<tr>
<td>Proposed method</td>
<td>7.48×10^{-5}</td>
<td>1.16×10^{-8}</td>
<td>0.0139</td>
<td>25.06</td>
<td>Linear</td>
</tr>
</tbody>
</table>

Table 5 Comparison of the proposed algorithm with some other existing optimization algorithms based on computational complexities.

<table>
<thead>
<tr>
<th>Technique</th>
<th>Any matrix inversion in each iteration</th>
<th>Selection of initial $h(n)$</th>
<th>Evaluation of eigenvectors in each iteration</th>
<th>CPU time (sec.)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Proposed method</td>
<td>No</td>
<td>Assumed</td>
<td>No</td>
<td>0.094</td>
</tr>
<tr>
<td>Kumar et al. [21]</td>
<td>Yes</td>
<td>Assumed</td>
<td>No</td>
<td>0.11</td>
</tr>
<tr>
<td>Upender et al. [26]</td>
<td>No</td>
<td>Assumed</td>
<td>No</td>
<td>4.48</td>
</tr>
<tr>
<td>Sahu et al. [19]</td>
<td>Yes</td>
<td>Assumed</td>
<td>No</td>
<td>0.66</td>
</tr>
<tr>
<td>Chen &amp; Lee [15]</td>
<td>Yes</td>
<td>To be determined</td>
<td>No</td>
<td>----</td>
</tr>
<tr>
<td>Ghosh et al. [25]</td>
<td>No</td>
<td>Assumed</td>
<td>No</td>
<td>----</td>
</tr>
<tr>
<td>Jain &amp; Crochiere [14]</td>
<td>Yes</td>
<td>Assumed</td>
<td>Yes</td>
<td>1.68</td>
</tr>
</tbody>
</table>

5 Conclusion
This paper has presented an efficient algorithm for designing two-channel QMF banks with linear phase. The QMF design problem is basically a nonlinear and multidimensional optimization problem. The proposed algorithm exhibits superlinear convergence near the optimal point therefore, it is suitable for QMF design problem. Error function for the design problem is minimized by optimizing the prototype filter coefficients. The proposed algorithm has been tested on two design examples by writing a MATLAB program. The Design results clearly show that this technique gives improved performance in terms of peak reconstruction error as compared with several state-of-art existing techniques and it is also efficient for QMF banks with larger filter taps.

References
areas include signals and systems, digital signal processing, communication systems and fuzzy systems.

Surendra Kr. Agrawal was born in Rajasthan, India in 1975. He received B.E. and M.E. degrees in 1998 and 2007, respectively. He is working as Assist. Professor at Department of Electronics and Communication Engineering, Government Women Engineering College, Ajmer, India. He is pursuing his Ph.D. degree from National Institute of Technology (NIT), Kurukshetra, India. His research interests are in the areas of multirate signal processing and digital communication.

Om Prakash Sahu is Professor at Department of Electronics and Communication Engineering, National Institute of Technology, Kurukshetra, India. He has more than 75 papers in his credit in various national and international conferences and journals. His research interests and specialization areas include signals and systems, digital signal processing, communication systems and fuzzy systems.