



## ADAPTIVE FILTER DESIGN FOR WAVELET DECOMPOSITION AND RECONSTRUCTION IN IMAGE PROCESSING APPLICATION

M. Santhosh<sup>1</sup>, B. Stephen Charles<sup>1</sup> and M. N. Giri Prasad<sup>2</sup>

<sup>1</sup>Stanley Stephen College of Engineering and Technology, Kurnool, Andhra Pradesh, India

<sup>2</sup>ECE Department, JNTU College of Engineering, Anantapur, Andhra Pradesh, India

E-Mail: [santoshgideon@gmail.com](mailto:santoshgideon@gmail.com)

### ABSTRACT

The design of filtration logic in wavelet transformation has been focused in this paper. The conventional wavelet transformation approaches are observed to be very effective in providing resolution information for each orientation. Though wavelets are used as a prominent approach for image compression, the accuracy of the image processing mainly depends on the filters used and their relative relation for their decomposition and reconstruction operation. In this paper an approach towards designing of filters with adaptive relation is proposed. The simulative observation presents a relatively better observation in image coding for the proposed approach.

**Keyword:** image compression, adaptive filter design, wavelet transform, decomposition, reconstruction.

### 1. INTRODUCTION

With the increase in rapid signal processing the need of accurate data precision during process is increasing. The methods developed in signal processing are basically designed for improving the estimation accuracy, with the improvement in designing the filter architecture. The signal processing accuracy depends on the actual data been received and the mode of operation. To improve the retrieval accuracy the signals are now processed in multi spectral domain, where a signal rather processing directly, is decomposed into multiple spectrum of different frequencies to process called multi-spectral analysis. Multi-spectral analysis is the most effective mode of operation for accurate processing. Multi spectral representation of a signal is developed using banks of filter, where an analysis and a synthesis filter is used for signal representation and reconstruction. The analysis and synthesis filters are effective when used in uniform domain. Under non-uniform environment these filters are not effective as the relation of the synthesis filter is not synchronous to the analysis filter under non uniform condition. To overcome this limitation, in this work a focus is made towards development of design methodology to develop an integrated format of synthesis and analysis filter of uniform filter bank in non-uniform environment.

Filter banks have been of great interest in a number of signal processing applications. A large group of these applications comprise those utilizing subband adaptive filtering. Examples of applications where subband adaptive filtering successfully is applied are acoustic echo cancellation [1, 2, 3, 4, 5], speech enhancement [6], signal separation [7], and beam-forming [8]. To decrease the complexity of filter bank structures, the sampling rate can be reduced in the subbands. These filter banks are referred to as decimated filter banks, and are afflicted with three major types of distortions: amplitude-, phase- and aliasing- distortion. These distortions degrade the performance of the application in which the filter bank is used. Several design methods have

been proposed and evaluated on subband adaptive filtering applications. Prototype filters for modulated filter banks are designed by interpolation of a two-channel Quadrature Mirror Filter (QMF), and evaluated in a real-time acoustic echo cancellation application in [9]. A design method is proposed using an iterative least-squares algorithm where a reconstruction error and the stopband energy are simultaneously minimized in [10]. Other methods have been proposed for the design of perfect reconstruction or para-unitary filter banks, which is a class of perfect-reconstruction filter banks. A method using non-constrained optimization is described in [11], for the design of perfect reconstruction polyphase filter banks with arbitrary delay, aimed at applications in audio coding. Kliewer proposed a method for linear-phase prototype FIR filters with power complementary constraints for cosine modulated filter banks, based on an improved frequency-sampling design [12]. In [13], the design of perfect reconstruction cosine-modulated para-unitary filter banks is discussed. Heller presented a design method for prototype filters for perfect reconstruction cosine-modulated filter banks with arbitrary prototype filter length and delay in [14]. Uniformly modulated filter banks have been of special interest because of their simplicity and efficient implementation [15, 16]. In this report a design method for analysis and synthesis uniformly modulated filter banks is presented, using unconstrained quadratic optimization. The goal of the design method is to minimize magnitude-, phase-, and aliasing- distortion in the reconstructed output signal, caused by the filter bank, as well as to minimize aliasing in the subband signals. Aliasing affects the performance of subband adaptive filters. The goal is to design filter banks, taking into consideration that adaptive filtering in the subbands should cause minimal degradation.

The problem of phase distortion can be compensated by design appropriate phase compensation filters. Least Squares optimization techniques for the design of filter banks have previously been observed, especially, a two stage Least Squares design procedure,



where the analysis filter banks are designed first and the synthesis filter banks subsequently. However the issue of designing in non-uniform filter bank is yet to be addressed. This work presents methods for the design of non-uniform filter banks in multiband signal processing. The filter bank structure are to be developed from a uniformly modulated filter bank by using an all-pass transform which has a lossless frequency function and a nonlinear phase function. The design methods will be modeled based on linear and quadratic frequency domain criteria and linear constraints for multiband adaptive filtering and multiband coding. Analysis filter banks and synthesis filter banks are to be designed in two subsequent stages, with the objectives including minimization of multiband aliasing as well as reconstruction output residual aliasing components on an individual basis. To formulate design objectives methods appropriate for filter banks used in multiband adaptive filtering will be used. Other design objectives are to optimize the overall filter bank response for low amplitude and phase distortion. Designs with phase compensation for linear phase overall response are included.

## 2. WAVELET TRANSFORMATION

To perform the forward DWT the JPEG2000 system uses a one-dimensional (1-D) subband decomposition of a 1-D set of samples into low-pass and high-pass samples. Quantization refers to the process of approximating the continuous set of values in the image data with a finite (preferably small) set of values. After the data has been quantized into a finite set of values, it can be encoded using an Entropy Coder to give additional compression. By entropy, it means the amount of information present in the data, and an entropy coder encodes the given set of symbols with the minimum number of bits required to represent them using Huffman coding. The Huffman decoder block carries out decoding reading the unique code bits passed in place of the data bit. The dequantizer unit dequantizes the decoded data bits. Inverse transformation is the process of retrieving back the image data from the obtained image values. The image data transformed and decomposed under encoding side is rearranged from higher level decomposition to lower level with the highest decomposed level been arranged at the top.

There are several ways wavelet transforms can decompose a signal into various sub bands. The decomposition of the signal into different frequency bands is simply obtained by successive high pass and low pass filtering of the time domain signal. First, the low pass filter is applied for each row of data, thereby getting the low frequency components of the row. But since the low pass filter is a half band filter, the output data contains frequencies only in the first half of the original frequency range. Now, the high pass filter is applied for the same row of data, and similarly the high pass components are separated.

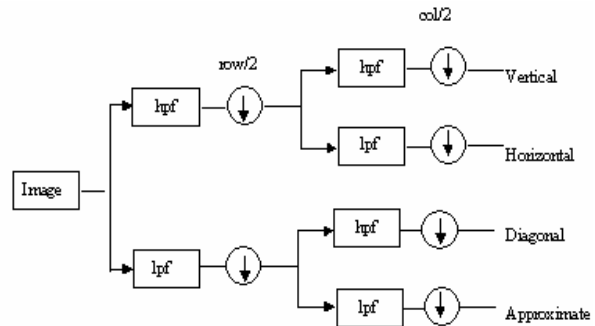


Figure-1. Dyadic decimation of image coefficients.

To perform the forward DWT the JPEG2000 system uses a one-dimensional (1-D) sub band decomposition of a 1-D set of samples into low-pass and high-pass samples. Low-pass samples represent a down-sampled, low-resolution version of the original set and High-pass samples represent a down-sampled residual version of the original set.

## 3. FILTRATION OPERATION

Filter banks with the aliasing cancellation property have been of great interest in numerous applications, and design methods taking aliasing into account have been considered in various approaches. However, aliasing cancellation filter banks are less suitable for multiband adaptive filtering, since this property is not maintained when the multiband signals are modified by individual multiband filtering. This may lead to significant distortion in signal processing, especially when the individual aliasing terms have large magnitude. By appropriate minimization of aliasing terms, filter banks can be used in signaling applications with multiband domain filtering. Examples of such signal application are noise suppression with single channels, for example spectral subtraction, or multiple channels, for example signal arrays with multiband beam-forming. Even echo cancelling is not such an application since the near-end signal is not affected by multiband filters. In various approaches constrained adaption is used in combination with overlap techniques to avoid distortion and maintaining linear convolution properties. Earlier experiments have shown that such techniques are sensitive to fast variations of the adaptive coefficients giving audible echo effects in the output. Design methods for uniform DFT filter banks for multiband adaptive filtering have been used earlier. Non-uniform filter banks have been of interest in signal enhancement, since by appropriate design it is possible to get a model, corresponding to the human observatory system. These are also successfully applied to, signal recognition and signal coding. Non-uniform filter banks have also been proposed for multiband adaptive filtering, e.g. in spectral subtraction for signal enhancement, and beam-forming for multiband signal arrays.



#### 4. ADAPTIVE FILTER DESIGN

Firstly a non-uniform analysis bank will be designed and then a matching synthesis filter bank will be designed, for the given analysis filter bank. Two types of criteria are to be evaluated namely, quadratic criteria and linear criteria, with and without linear constraints. The advantage of using linear criteria and constraints is that each frequency component can be individually controlled. A common aim is to design the analysis and synthesis banks with pre-specified parameters, such as number of multiband, filter lengths, delays and decimation factors. In the first stage the analysis filter bank is required in such way that aliasing terms in the multiband are minimized. In the second stage the synthesis filter bank are to be designed, based on the analysis filter bank, such that the overall response is optimized and the reconstruction aliasing terms are minimized. The filter bank design methods are to be designed for a single prototype for the analysis and synthesis filter banks in order to obtain nearly perfect reconstruction properties. In the two stage design methods the amplitude distortion, phase distortion (delay) and aliasing distortion can be minimized or controlled for the analysis and synthesis filter banks separately. When using the linear criteria, they can be controlled for each frequency component individually, given a certain constraint.

##### a) Encoding filter modeling

The Encoding filter bank design problem reduces to the design of a single prototype Encoding filter,  $H(z)$ , when the Encoding filters are modulated versions of the prototype Encoding filter the purpose of the Encoding filters is to split the original signal into a set of subband signals. Ideal Encoding filters are bandpass filters with normalized center frequencies  $\omega_m = 2\pi \frac{m}{M}, m=0, \dots, M-1$ , and with bandwidth  $2\pi/M$ . The ideal filters have unit magnitude and zero phase in the passband while the stopband magnitude is zero. While zero phase filters require non-causality, the requirements need to be relaxed by using linear phase filters. FIR filters may have exact linear phase but they cannot possess the ideal magnitude requirements. Therefore approximations need to be made.

A straightforward way to design the prototype Encoding filter is to design a lowpass filter, with a passband region centered around  $\omega = 0$ , and a minimum magnitude stopband region using filter design methods such as window techniques or the Parks-McClellan optimal equi-ripple FIR filter design method [17]. The bandwidth is controlled by the passband region and the attenuation of signal components in the stopband region leads to low-energy inband-aliasing terms. These methods cannot control the delay properties of the resulting filter. However, filter design techniques with complex approximation for filters with arbitrary phase exist [18].

##### b) Decoding filter modeling

Similar to the design of Encoding filter banks, the design of Decoding filter banks reduces to the design of a

single Decoding prototype filter. When designing the Decoding filter bank, the focus is on the performance of the Encoding-Decoding filter bank as a whole. This implies that the Decoding filter bank is designed given an Encoding filter bank, i.e., given the prototype Encoding filter. Different applications of filter banks require different strategies. The focus in the proposed method is on applications, which uses filtering operations in the subbands.

In the proposed design of the Decoding filter bank, the goal is to minimize amplitude and phase distortion of the Encoding-Decoding filter bank and to minimize aliasing distortion in the output signal  $Y(z)$ . The objective function is the least square error.

$$\varepsilon_g(h) = \frac{1}{2\pi} \int_{-\pi}^{\pi} |E(e^{j\omega})|^2 d\omega \quad (1)$$

Where

$$E(z) = T(z) - D(z) \quad (2)$$

denotes the total response error. Here,  $T(z)$  is the complex-valued system response. The desired complex-valued Encoding-Decoding filter bank response  $D(z)$  is defined for the special case with  $\zeta_m(z) = 1$ ,

$$D(z) = z^{-\tau_T} \quad (3)$$

Where  $\tau_T$  is the desired total Encoding-Decoding filter bank delay.

The complex error function in Eq. (2) can be written as:

$$E(z) = E_0(z) + \sum_{m=0}^{M-1} \sum_{d=1}^{D-1} E_{m,d}(z). \quad (4)$$

The first term,  $E_0(z)$ , contains the desired spectral components and the summation in the second term contains all the undesired aliasing terms. The error function for the desired spectral content,  $E_0(z)$ , is defined as:

$$E_0(z) = \sum_{m=0}^{M-1} E_{m,0}(z) = A_0(z) - D_0(z) \quad (5)$$

Where the subband filters are unity, i.e.,  $\zeta_m(z) = 1$ . The total response excluding the undesired aliasing terms,  $A_0(z)$ . The desired transfer function for  $d=0$  is:

$$D_0(z) = \sum_{m=0}^{M-1} D_{m,0}(z) = z^{-\tau_T} \quad (6)$$

Note that  $D_0(z) = D(z)$  according to (3). This implies that the desired transfer functions,  $D_{m,d}(z)$ , for  $d > 0$  are zero, i.e.,  $D_{m,d} = 0$ . The error functions for the undesired aliasing terms are defined as:



$$E_{m,d}(z) = A_{m,d}(z) - D_{m,d}(z) = A_{m,d}(z). \quad (7)$$

Aliasing content may cancel out at the final reconstruction summation, so that the residual aliasing in the output signal  $Y(z)$  is zero, even though the individual terms in the error function may have large energy.

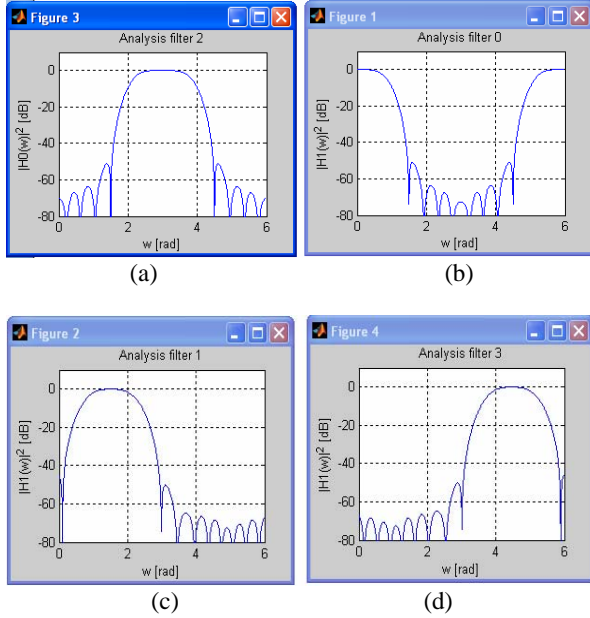
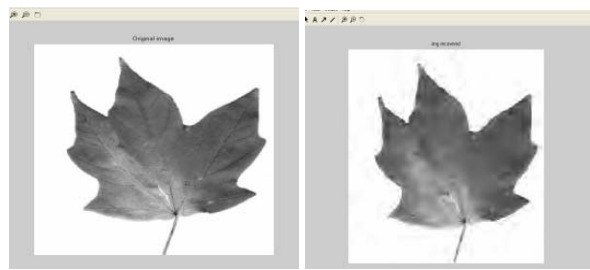


Figure-2. Filter response characteristic for the proposed encoding filters.

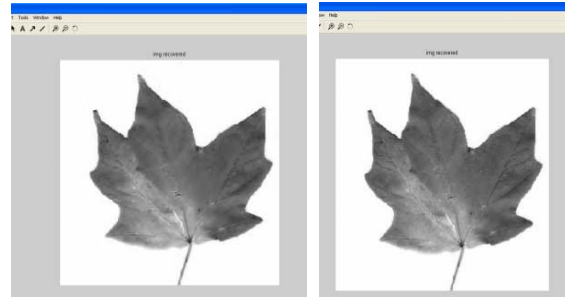
With the incorporation of these filter coefficients on the wavelet filter a comparative improvement in the retrieved quality is observed. The obtained result for the suggested approach is compared with the well known bench mark image compression called ‘JPEG-2000’. The result observations are as outlined below,

**5. RESULT OBSERVATION**

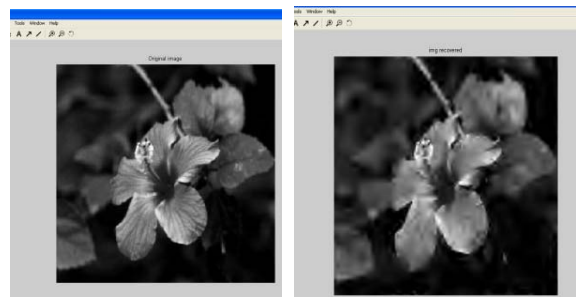
For the evaluation of the suggested approach in this work a comparative analysis is carried out between the conventional and the suggested approach for different format and sizes of image. The obtained result for the developed system is as presented below,



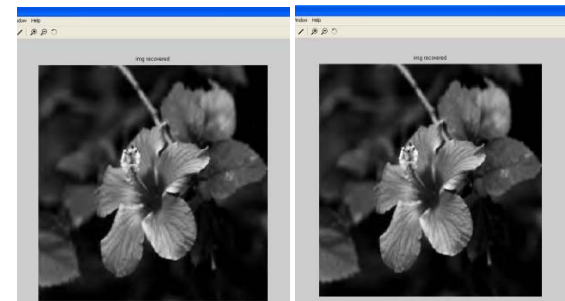
(a) Original leaf image sample, (b) Recovered image at 0.1 bpp.



(c) Recovered image at 0.5 bpp, (d) Recovered image at 0.9 bpp.



(a) Original flower image sample, (b) Recovered flower image sample at 0.1 bpp.



(c) Recovered flower image sample at 0.5 bpp, (d) Recovered flower image sample at 0.9 bpp.

sample	Dimension	Proposed method			Jpeg method		
		CR	PSNR	TIME	CR	PSNR	TIME
S1	64×64	81	46	2.1	78	43	3
	110×110	83.5	49	3.6	72	48	4
	256×256	80	41	2.6	79	35	4.2
	512×512	83.7	43.9	4.2	79.8	40.1	5.3
S2	64×64	90	48	2.2	77	46	3.3
	110×110	89	43	2.9	81	41.5	4.5
	256×256	86.3	46.9	3.8	84	43.7	4.6
	512×512	82	42.1	5.0	76.7	40	5.6
S3	64×64	83	45	4.9	81	42.1	5.3
	110×110	86	40	2.8	86	38	4.6
	256×256	81.8	41.7	5	79.23	40.5	6
	512×512	84	44	4.7	76	42	5.6

A comparative analysis is carried out for the proposed system w.r.t the conventional JPEG 2000 coding system. The obtained comparisons are as outlined in table above.



## 6. CONCLUSIONS

This paper presents a basic design approach to image compression based on the relative filter design approach to encoding and decoding filter banks. The developed approach when introduced to filtration performs better compression than the conventional JPEG-2k approach. The computational effort such a coding system is also observed to be reduced due to relative relational between the filter coefficients. With this a higher processing estimation for image compression approach is been developed. This approach results in effective compression for multiple formats and dimension of images as compared to the conventional compression approach.

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