

## Audio Compression Using Daubechie Wavelet

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**Abstract:** Compression of audio signal is one of the basic technologies in multimedia. This paper describes about Daubechie wavelet based audio compression. This method uses optimal wavelet selection and wavelet coefficients quantization procedure together with different Companding methods like A-law and U-law Compandings. This paper presents a technique to incorporate psychoacoustic models into an adaptive wavelet packet scheme to achieve perceptually transparent compression of high-quality audio signals [1].

**Keywords:** A-law companding, U-law companding, Daubechie wavelet, Bior Wavelet, Symlet Wavelet.

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### I. Introduction

Compression is process of converting an input data stream into another data stream that has smaller size. Compression provides the reduction in redundancy also used to reduce storage requirements overall program execution time may be reduced. This is because reduction in storage will result in reduction of disc access attempts. The compression algorithm help to reduce the bandwidth requirements and also provide a level of security for the data being transmitted. The wavelets consist of banks of low pass filters, high pass filters and down sampling units. Half of the filter convolution results are discarded because of the down sampling at each wavelet decomposition stage. Only the approximation part of the Daubechie wavelet results is kept so that the number of samples is reduced by half[6].

#### Different Compression Techniques

Different types of compression techniques are available they are lossless compression and lossy compression techniques. Redundancy information present in audio signal will removed in loss less compression. It is disadvantages such as it doesn't give the constant output data rate and very small compression ratio, and advantage is it can be applied

To any data stream. In loss compression the information is irrelevant in that the receiver will not able to recognize the missing.

#### Wavelet Transforms:

The wavelet theory allows a very general and flexible description to transform signal from time domain to time-frequency domain, so called time-scale domain. Wavelet transform uses short window for high. Wavelet Transform uses short window for high frequencies, leading to a good time resolution and larger windows for low frequencies leading to a good frequency resolution.

### II. Wavelet Compression

Wavelet compression is a form of data compression well suited for audio compression, video compression, image compression. Wavelet compression methods are adequate for representing transients. Such as percussion sounds in audio, high-frequency components in 2-D images. This means transient elements of data signal can be represented by a smaller amount of information that would be the case of some other transforms Like Symlets filters, Coiflets filters, Biorthogonal filters, Reverse Biorthogonal Filters, Discrete Meyer, Harr[5]

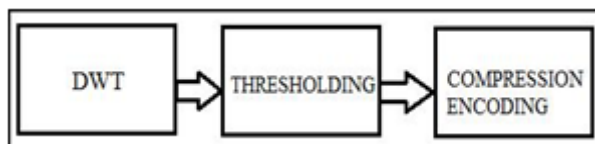
#### Wavelet Representation For Audio Signal

A wavelet transform can be defined as a "small wave" that has its energy concentrated in time, and it provides a tool for the analysis of transient, non-stationary or time varying phenomenon. It has oscillating wave like property. Wavelet is a waveform of limited duration having an average value zero. They are localized in space. Wavelet transform provides a time-frequency representation of the signal. In wavelet transform, the signal is decomposed into set of basic functions also known as WAVELETS. Wavelets with large number of vanishing moments are useful for this audio compression method, because if a wavelet with a large number of vanishing moments is used, a precise specification of the pass bands of each sub band in the wavelet

decomposition is possible. Thus, we can approximate the critical band division given by the auditory system with this structure and quantization noise power could be integrated [3][4].

**Wavelet Based Compression Techniques:**

Wavelets concentrate speech signals into a few neighbouring coefficients. By taking the wavelet transform of a signal, many of its coefficients will either be zero or have negligible magnitudes. Data compression can then be done by treating the small valued coefficients as insignificant data and discarding them. Compressing a speech signal using wavelets involves the following stages[2].



**Figure 1:** Compression System Design

**a) Thresholding:**

After the coefficients are received from different transforms, thresholding is done. Very few DCT coefficients represent 99% of signal energy; hence thresholding is calculated and applied to the coefficients. Coefficients having values less than threshold values are removed.

**b) Quantization:**

It is a process of mapping a set of continuous valued data to a set of discrete valued data. The aim of quantization is to reduce the information found in threshold coefficients. This process makes sure that it produces minimum errors. We basically perform uniform quantization process.

**c) Encoding:**

We use different encoding techniques like decomposition using N equal frames. Encoding method is used to remove data that are repetitively occurring. In encoding we can also reduce the number of coefficients by removing the redundant data. Encoding can use any of the two compression techniques, lossless or lossy. This helps in reducing the bandwidth of the signal hence compression can be achieved. The compressed speech signal can be reconstructed to form the original signal by DECODING followed by DE-QUANTIZATION. This would reproduce the original signal.

**Psychoacoustic Model  
Simplified masking model**

Auditory masking depends on time and frequency of both the masking signal and the masked signal. It is assumed in this paper that masking is additive, so they estimate the total masked power at any frequency by adding the masked power due to the components of the signal at each frequency. Thus, minimum masked power within each band can be calculated. From this, they get the final estimate of masked noise power. Then the idea is that a listener will tolerate an additive noise power in the reproduced audio signal as long as the power spectrum is less than the masked noise power at each frequency[1][2]. The authors noted that even though this masking model has several drawbacks, it yields reasonable coding gains. The main problems that this psychoacoustic model has are:

- The shape of the masking property used is valid for masking by tonal signals, which is not the same for masking by noise.
- The model is based on psychoacoustic studies for the masking of a single tone like signal (quantization error could happen if it contains several components).
- Masking is assumed to be additive (a power law rule of addition should be used instead).

**Masking constraint in the Wavelet Domain**

This masking model is incorporated within the framework of the wavelet transform based coder. The idea is to convert the perceptual threshold of each sub-band into a wavelet constrain. To do that the authors defined  $e$ , an  $N \times 1$  error vector consisting of the value of the Discrete Fourier Transform of the error in reconstructing the signal from a sequence of approximate wavelet coefficients ( $N$  is the length of the audio frame). Also  $RD$  is defined as a diagonal matrix with entries equal to discretized value of one over the masked noise power. The psychoacoustic model implies that the reconstruction error due to the quantization or approximation of the wavelet coefficients corresponding to the given audio signal may be made inaudible

**Reducing The Number Of Non-Zero Coefficients Optimization Criterion:**

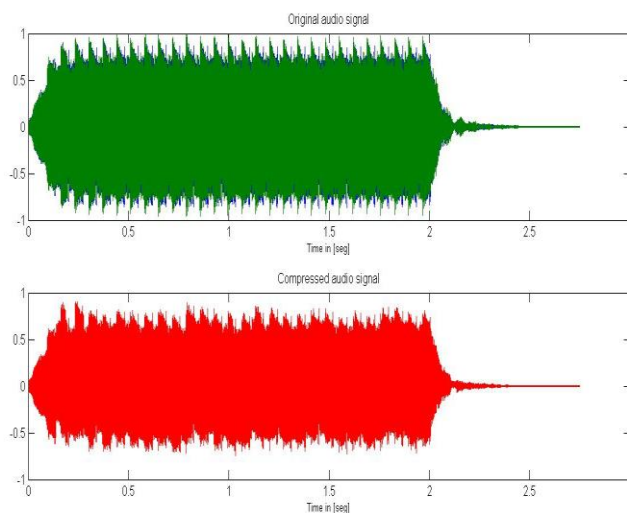
For each frame, an optimum wavelet representation is selected to minimize the number of bits required to represent the frame while keeping any distortion inaudible. This wavelet selection is the strongest compression technique, because it highly reduces the number of non-zero wavelet coefficients. In addition to that, those coefficients may be encoded using a small number of bits. Therefore, this technique choosing an analysis wavelet and allocating bits to each coefficient in the resulting wavelet representation. Figure explains how this technique works. It shows a signal vector representation by a particular choice of a basis. The radius of the sphere shown is equal of the norm of the time domain signal, and the error ellipse corresponds to the perceptual semi-norm calculated by the psychoacoustic model. The audio segment can be represented using any vector whose tip lies inside the error ellipse with no perceptual distortion. Hence, the projection of the error ellipsoid along each coordinate axis specifies the coarsest quantization that can be used along the axis without producing any perceptual degradation. Therefore, a large projection along a particular coordinate axis implies that only a small number of bits to quantize that coordinate need to be used. Exploiting this fact, a low bit rate representation of the signal can be achieved by the rotation of the vector representation of the signal via a unitary wavelet transformation.

To apply this technique, let  $R_k(\theta)$  be the number of bits assigned to the quantization of the  $k$ th transform coefficient  $x_k^q(\theta)$  when the wavelet identified by the vector  $O$  is used to decompose frame  $x$ . The goal is to minimize.

$$R(\theta) = \sum_{k=1}^N R_k(\theta)$$

by properly choosing  $O$  and the number of bits  $R_k(\theta)$  assigned to the quantization of each transform coefficient  $x_k^q(\theta)$ , The minimization must be done under the constraint on the perceptual encoding error. It is proven in the paper that, for a particular choice of a wavelet, the bit rate requirement may be computed using the following formula directly from the transform coefficients.

**III. Simulation Result**



**Figure: Fordb10wavelet**

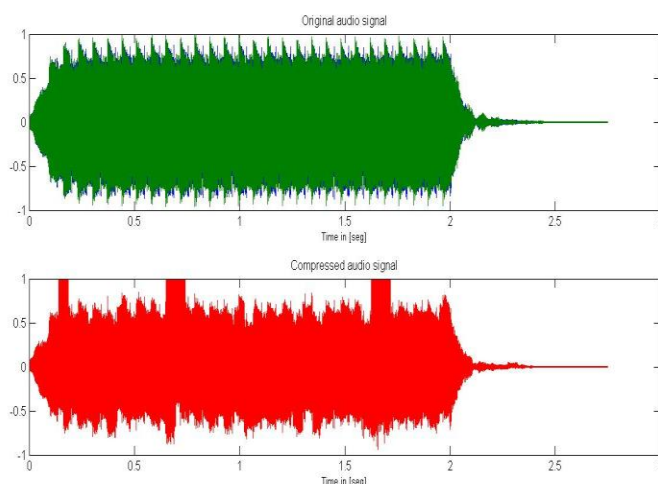


Figure: For BIOR 1.5 wavelet

#### IV. Conclusion

A brief summary of the audio compression techniques using Daubechie Wavelets and the main considerations that people do when evaluating their results has been presented. A MATLAB simulation of the selected paper was successfully implemented, simplifying some of its features, but keeping its main structure and contributions. The quality of the compressed signal obtained with the MATLAB implementation is lower than any current standard compressing schemes, but considerable better than one obtained just by “blindly compressing” the signal.

#### References

- [1]. High-Quality Audio Compression Using an Adaptive Wavelet Packet Decomposition and Psychoacoustic Modeling. IEEE TRANSACTIONS ON SIGNAL PROCESSING, VOL. 46, NO. 4, APRIL 1998.
- [2]. Deepen Sinha. Low Bit Rate Transparent Audio Compression using Adapted Wavelets. IEEE TRANSACTIONS ON SIGNAL PROCESSING. VOL. 41. NO. 12. DECEMBER 1993.
- [3]. Johnson IhyehAgbinya. discrete wavelet transform techniques in speech processing 1996 IEEE TENCON - Digital Signal Processing Applications
- [4]. M.V.Patil Audio and Speech Compression Using DCT and DWT Techniques. International Journal of Innovative Research in Science, Engineering and Technology Vol. 2, Issue 5, May 2013
- [5]. Ciprian Doru Giurcaneanu, Ioan Tabus, and Jaakko Astola. Integer wavelet Transform Based Lossless Audio Compression.
- [6]. C.D. Giurcaneanu, I. Tabus, and J. Astola. Adaptive context based sequential prediction for lossless audio compression. In *Proc. Eusipco-98, IX European Signal Processing Conference*, volume 4, pages 2349–2352, Rhodes, Greece, Sept. 1998