

Audio Compression Using Wavelet Transforms

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ABSTRACT: Past decades, demand for the digital information has increased dramatically. To overcome, we compress the information by removing the redundancies present in it. The idea of audio compression is to encode audio data to take up less storage space and less bandwidth for transmission.

The main objective of this project is to study the known audio compressing techniques that use wavelets. To simulate using MATLAB and obtain the compressed audio signal.

KEYWORDS: Discrete wavelet transform, psychoacoustic model, masking, dynamic dictionary.

I. INTRODUCTION

Speech signals has unique properties that differ from a general audio/music signals. First, speech is a signal that is more structured and band-limited around 4kHz. These two facts can be exploited through different models and approaches and at the end, make it easier to compress.

The main goal is to compress high quality audio maintaining transparent quality at low bit rates. In order to do this, the authors explored the usage of wavelets instead of the traditional Modified Discrete Cosine Transform (MDCT).

Compression methods can be classified into three functional categories:

1. Direct Methods
2. Transformation Methods: such as Fourier Transform (FT), Wavelet Transform (WT), and Discrete Cosine Transform (DCT).
3. Parameter Extraction Methods: A pre-processor is employed to extract some features that are later used to reconstruct the signal.

Fast Fourier transforms (FFT) have been among major tools for compression and efficient coding of signals. Particularly, the discrete cosine transform (DCT) based compression algorithms have become industry standard for still and video image compression systems (JPEG, MPEG). The ability of wavelets to outperform traditional orthogonal bases in the mentioned systems is being extensively explored nowadays. Wavelet compression is a form of predictive compression where the amount of noise in the data set can be estimated relative to the predictive function. Most modern compression techniques use a two step process: First, a predictive compression function (such as wavelet transform) is applied. If the choice of the predictive compression function is good, the result will be a new set of data with smaller values and more repetition.

Second, a coding compression step that will represent the data set in its minimal form (Huffman coding, run-length). The compression of speech signals has many practical applications.

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II. RELATED WORK

A number of methods have been proposed for the digital compression of audio signals. Early work on signal compression dates back to the information-theoretic foundation that was laid out by Shannon. Shannon introduced the idea of entropy as a quantity of expressing the information content of a signal. The earliest examples of perceptual coders were developed in the 1970's by Crochiere, Schroeder, Zelinski and Noll. These algorithms utilized a time to frequency transformation stage, e.g. Short-time Fourier Transform and 4-channel non-uniform filter bank that allowed noise shaping in the frequency domain according to some well known psychoacoustic principles. They were followed by the work of several other people in the 1980's who tried to improve on the choice of the transformation stage, accuracy of the psychoacoustic model and use of other coding techniques that further improved coding efficiency. Scientists have found that the audible frequency can be broken up into critical bandwidths, which are non-uniform and dependent on the level of the incoming sound. Signals within one critical bandwidth are hard to separate for a human observer. Another important technique is masking property of auditory system. The presence of a strong audio signal makes the spectral neighbourhood of weaker audio signal imperceptible. If two sounds occur simultaneously and one is masked by the other, this is referred to as simultaneous masking.

The idea of audio compression is to encode audio data to take up less storage space and less bandwidth for transmission. To meet this goal different methods for compression have been designed. It is possible to classify them into two categories: lossless compression and lossy compression. Analysis by synthesis coding is the one where no attempt is made at reproducing the exact speech waveform at the receiver, but to create perceptually equivalent to the signal. One of the most popular techniques is called Linear Predictive Coding (LPC). Some higher quality coders include RELP (Residual Excited Linear Prediction) and CELP (Code Excited Linear Prediction). Accordingly, audio coders are commonly categorized as either parametric coders or waveform coders. All perceptual coders share a similar structure in that they contain a filter bank, psychoacoustic model, and an encoding and quantization stage. Many types of filter banks and time-to-frequency transforms exist where each other has a different set of trade-offs in its design. The wavelet transform or more generally the wavelet filter bank (WFB), is an iterated filter bank that provides a flexible way of analyzing a signal at various resolutions and across various frequency regions. This flexibility is especially appealing in audio compressing. Since the WFB can provide an analysis of the input signal according to the critical band (CB) resolution of the inner ear and, more generally, provide a scheme that can adapt to the time-varying nature of the audio signal.

III. DESIGN METHODOLOGY

The main goal of the algorithm presented in paper is to compress high quality audio maintaining transparent quality at low bit rates. Several steps are considered to achieve this goal:

1. Design a wavelet representation for audio signals.
2. Design a psychoacoustic model to perform perceptual coding and adapt it to the wavelet representation.
3. Reduce the number of the non-zero coefficients of the wavelet representation and perform quantization over those coefficients.
4. Perform extra compression to reduce redundancy over that representation
5. Transmit or store the stream of data. Decode and reconstruct. Evaluate the quality of the compressed signal.

Exemplar based Inpainting technique is used for inpainting of text regions, which takes structure synthesis and texturesynthesis together. The inpainting is done in such a manner, that it fills the damaged region or holes in an image, with surrounding colour and texture. The algorithm is based on patch based filling procedure. First find target region using mask image and then find boundary of target region. For all the boundary points it defined patch and find the priority of these patches. It starts filling the target region from the highest priority patch by finding the best match patch. This procedure is repeated until entire target region is inpainted.

The algorithm automatically generates mask image without user interaction that contains only text regions to be inpainted.

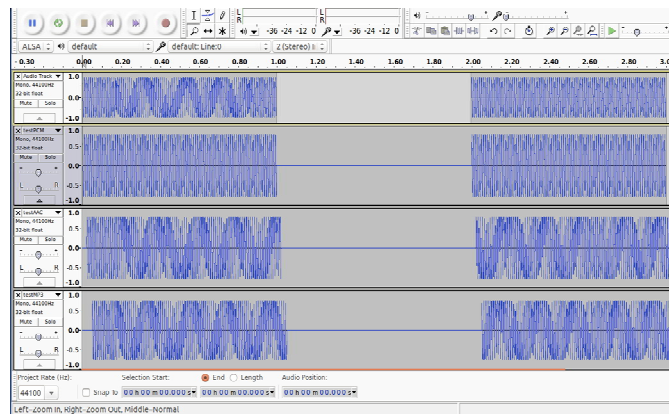
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IV. EXPERIMENTAL RESULTS

1. The audio that is taken under consideration must be compressed
2. If the audio occupies 1Mb, then after the compression technique, it must occupy less than 1Mb.
3. Lower the memory must lesser the time, therefore the time taken to transmit and receive the file must be less than the time taken by the original file



V. CONCLUSION

We have implemented an audio compression procedure using wavelet transform technique. Our algorithm successfully compresses the audio which consists of speech signal. We have applied our algorithm on many audio files and found that it is successfully compressed.

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