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DIGITAL GENERIC DATA COMPRESSION ALGORITHMS USING MATLAB

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Abstract

Generic data compression algorithms is an area of digital processing that is focusing on reducing bit rate of the speech signal for transmission or storage without significant loss of quality. The focus of this paper is to compress the digital speech using wavelet transform. The main idea behind the generic data compression algorithms is to represent uncompressed speech with minimum number of bits and optimum speech quality.

Wavelet transform has been recently proposed for signal analysis. The wavelet transform is useful to remove redundancies and irrelevancies present in the speech signal for the compact representation. The classical subband coding method also known as filter bank method is used in which signal is decomposed into basis of wavelet functions using high pass and low pass filters. Speech coding is a lossy scheme and is implemented here to compress one-dimensional speech signal.

Basically, this scheme consists of three operations which are the transform, threshold techniques (by level and global threshold), and run-length encoding operations. Finally the compressed signal is reconstructed. The generic data compression algorithms using filter bank method is implemented and simulated by using wavelet transform algorithm for different wavelets such as Haar, Daubechies series (Daub-4, Daub-6, Daub-8, Daub-12, Daub-16 Daub-32 Daub-64) and bi-orthogonal wavelet.

Keywords: Wavelet transform, Filter bank method, Lossy scheme, Haar, Daubechies series and bi-orthogonal wavelet

Introduction

Speech coding has been and still is a major issue in the area of digital speech processing. Speech coding is the act of transforming the signal speech at hand, to a more compact form, which can then be transmitted with a considerably smaller size.

The motivation behind this is the fact that access to unlimited amount of bandwidth is not possible. Therefore, there is a need to code and compress speech signals. Speech compression is required in long-distance communication, high-quality speech storage, and message encryption.

Wavelets can be used for the speech compression as it can remove redundancy and irrelevancy present in the digital speech. We have used wavelet transform for the purpose of speech compression in this paper. Output of high pass filter represents detail components and output of low pass filter represents approximate components.

In order to compress the speech signal, small wavelets are removed by thresholding i.e. the details below some threshold value are replaced by zero. Speech signal compression using wavelet transform is given a considerable attention in this paper.

The wavelet transform is a valuable tool for many engineering applications. We have selected wavelet analysis for the speech compression, as it is useful from a psychoacoustic point of view.

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Materials and methods

Uncompressed multimedia (graphics, audio and video) data requires considerable storage capacity and transmission bandwidth. Despite rapid progress in mass-storage density, processor speeds and digital communication system performance demand for data storage capacity and data-transmission bandwidth continues to outstrip the capabilities of available technologies.

The recent growths of data intensive multimedia-based web applications have not only sustained the need for more efficient ways to encode signals and audios but have made compression of such signals central to storage and communication technology. To enable Modern High Bandwidth required in wireless data services such as mobile multimedia, email, mobile, internet access, mobile commerce, mobile data sensing in sensor networks, Home and Medical Monitoring Services and Mobile Conferencing, there is a growing demand for rich Content Cellular Data Communication, including Voice, Text, Audio and Video.

One of the major challenges in enabling mobile multimedia data services will be the need to process and wirelessly transmit very large volume of this rich content data. This will impose severe demands on the battery resources of multimedia mobile appliances as well as the bandwidth of the wireless network. While significant improvements in achievable bandwidth are expected with future wireless access technology, improvements in battery technology will lag the rapidly growing energy requirements of the future wireless data services.

One approach to mitigate this problem is to reduce the volume of multimedia data transmitted over the wireless channel via data compression technique such as AUDIO, AUDIO2000 and MPEG. These approaches concentrate on achieving higher compression ratio without sacrificing the quality of the Audio. However these Multimedia data Compression Technique ignore the energy consumption during the compression and RF transmission. Here one more factor, which is not considered, is the processing power requirement at both the ends i.e. at the Server/Mobile to Mobile/Server.

Thus in this paper we have considered all of these parameters like the processing power required in the mobile handset which is limited and also the processing time considerations at the server/mobile ends which will handle all the loads. Since audios will constitute a large part of future wireless data, we focus in this paper on developing energy efficient, computing efficient and adaptive audio compression and communication techniques. Based on a popular audio compression algorithm, namely, wavelet audio compression, we present an Implementation of Advanced Audio Compression Algorithm Using Wavelet Transform.

AUDIO COMPRESSION METHODOLOGY

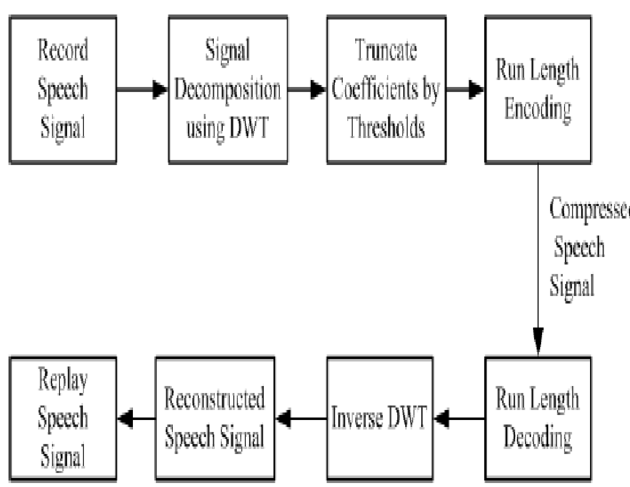
The storage requirements for the video of a typical angiogram procedure is of the order of several hundred Mbytes

- Transmission of this data over a low bandwidth network results in very high latency
- Lossless compression methods can achieve compression ratios of ~2:1
- We consider lossy techniques operating at much higher compression ratios (~10:1)

Key issues:

- High quality reconstruction required
- Angiogram data contains considerable high-frequency spatial texture
- Proposed method applies a texture-modeling scheme to the high-frequency texture of some regions of the audio. This allows more bandwidth allocation to important areas of the audio

Figure:



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Different types of Transforms used for coding are:

1. FT (Fourier Transform)
2. DCT (Discrete Cosine Transform)
3. DWT (Discrete Wavelet Transform)

The Discrete Cosine Transform (DCT):
The discrete cosine transform (DCT) helps separate the audio into parts (or spectral sub-bands) of differing importance (with respect to the audio's visual quality). The DCT is similar to the discrete Fourier transform: it transforms a signal or audio from the spatial domain to the frequency domain.

Discrete Wavelet Transform (DWT):
The discrete wavelet transform (DWT) refers to wavelet transforms for which the wavelets are discretely sampled. A transform which localizes a function both in space and scaling and has some desirable properties compared to the Fourier transform. The transform is based on a wavelet matrix, which can be computed more quickly than the analogous Fourier matrix.

Most notably, the discrete wavelet transform is used for signal coding, where the properties of the transform are exploited to represent a discrete signal in a more redundant form, often as a preconditioning for data compression. The discrete wavelet transform has a huge number of applications in Science, Engineering, Mathematics and Computer Science.

Wavelet compression is a form of data compression well suited for audio compression (sometimes also video compression and audio compression). The goal is to store audio data in as little space as possible in a file. A certain loss of quality is accepted (lossy compression).

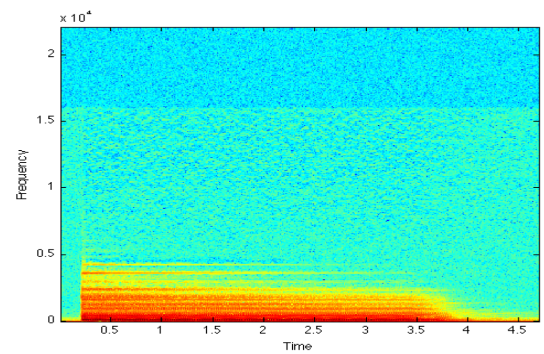
Using a wavelet transform, the wavelet compression methods are better at representing transients, such as percussion sounds in audio, or high-frequency components in two-dimensional audios, for example an audio of stars on a night sky.

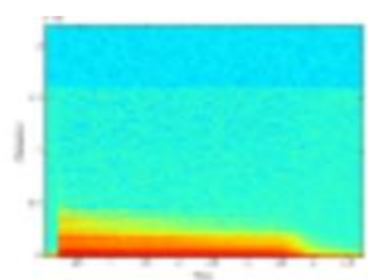
This means that the transient elements of a data signal can be represented by a smaller amount of information than would be the case if some other transform, such as the more widespread discrete cosine transform, had been used. First a wavelet transform is applied.

This produces as many coefficients as there are signals in the audio (i.e.: there is no compression yet since it is only a transform). These coefficients can then be compressed more easily because the information is statistically concentrated in just a few coefficients. This principle is called transform coding. After that, the coefficients are quantized and the quantized values are entropy encoded and/or run length encoded.

Results and discussion

The audio on the top is the original audio and the audio on the bottom is the compressed one (The point is that the audio on the top is uncompressed and the bottom is compressed using Haar wavelet method and the loss of quality is not visible. Of course, audio compression using Haar Wavelet is one of the simplest ways.)





Conclusion

Haar wavelet transform for audio compression is simple and crudest algorithm.as compared to other algorithms it is more effective. The quality of compressed audio is also maintained.

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