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Discrete Wavelet Transform for Compressing and Decompressing the Speech Signal

Bhavana Pujari
M.E. Student
Department of Electronics,
A.V.C.O.E College,
Savitribai Phule Pune University,
Sangamner, India.

Sheetal S. Gundal
H.O.D.
Department of Electronics,
A.V.C.O.E College,
Savitribai Phule Pune University,
Sangamner, India.

Abstract

In speech compression, the speech coding is done to compress the speech signal to the highest possible compression by maintaining user acceptability i.e. reducing bit rate of signal. In this project, a system proposes a method to Speech Compression and Decompression using Discrete Wavelet Transform (DWT). In DWT, signal is passed through a series of high pass filters and low pass filters for the analysis of high and low frequency component respectively. For many signals, the low-frequency content is the most important part because it gives identity to the signals whereas high-frequency content imparts flavor or nuance. For example, in the human speech removing the high-frequency components would make the voice sound different, but still intelligible whereas removing enough of the low-frequency components would make the voice sound garbage. Discrete Wavelet Transform is useful to remove redundancies and irrelevancies present in the speech signal for the compact representation.

We use the DWT where compression factor is considered as variable. Also it can be varied while most other techniques have fixed compression factor and also DWT significantly improves the reconstruction of the compressed speech signal.

Key words: ***DWT, bit rate, compression factor.***

1. Introduction

The nature of Speech signal is an acoustic and it is the most effective medium for face to face communication and telephony application. Speech coding is the process of obtaining a compact representation of audio signal for efficient transmission over band-limited wired and wireless channels and/or storage.

Compression is a process of converting an input data stream into another data stream that has a smaller size. Compression is possible only because data is normally represented in the computer in a format that is longer than necessary i.e. the input data has some amount of redundancy associated with it. The main objective of compression systems is to eliminate this redundancy so as to reduce the storage of file. When compression is used to reduce storage requirements, overall program execution time may be reduced. This is because reduction in storage will result in the reduction of disc access attempts. The compression algorithms help to reduce the bandwidth requirements and also transmit the data with high security[1].

Data compression is an art used to reduce the size of a particular file. The goal of data compression is to eliminate the redundancy present in a file so as to reduce its size. It is useful in reducing the data storage space and in reducing the time needed to transmit the data. Data compression can either be lossless or lossy. Lossless data compression recreates the exact original data from the compressed data while lossy data compression cannot regenerate the perfect original data from the compressed data. Lossy methods are mainly used for compressing sound, images or video.

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The compression of speech signals has many practical applications.

1. It is used in digital cellular technology where many users share the same frequency bandwidth.
2. Compression allows more users to share the system at a particular time.
3. It can also be used for digital voice storage that are used for digital answering machines and pre-recorded telephone calls that are used for purpose of providing any kind of information to user or advertising. The bit rate can be achieved up to 2.4 kbps to 9.6 kbps[2].
4. For a given memory size, compression allows longer messages to be stored than otherwise.
5. It is required in long distance communication.
6. High quality speech storage, and message encryption.
7. It plays an important role in teleconferencing,
8. Satellite communications and multimedia applications.

Compression techniques can be classified into the two main categories:

1.1Lossless compression

In this, the original file can be perfectly recovered from the compressed file. It is mainly used in cases where it is important that the original signal and the de- compressed signal are almost same or identical. Examples of lossless compression are Entropy Encoding (Shannon-Fano Algorithm, Huffmann coding, Arithmetic Coding,) Run-length, Lempel Ziv Welch (LZW) Algorithm.

1.2 Lossy compression

In this, the original file cannot be perfectly recovered from the compressed file, but it gives its best possible quality for the given technique. Lossy compressions typically attain far better compression than lossless by discarding less-critical data. The aim of this technique is to minimize the amount of data that has to be transmitted. They are mostly used for multimedia data compression.
Ex: FFT, DCT, DWT.

Reasons for compressing the data are as follows:

1) Cost of disk:

The disks are cheap but disks used for high end systems are expensive. Moreover, if replication or mirroring etc. are used all these will ultimately enhance the hardware cost.The manipulation, storage and transmission of speech in their raw form is very expensive, and significantly slows the transmission and make storage costly. Efficient speech compression solutions are becoming critical with the recent growth of data intensive, multimedia based applications.[2]

2) Cost of data management:

It takes very long for backup and recovery when the database is very big. Compressed the database, smaller will be the backup.

3) Memory:

Compressed data will occupy less space thus more data can be placed in the same area (memory). For e.g., If the data is compressed by 50 percentage we can place almost twice the size of the data.

4) Bandwidth and transfer speed:

Compressed data uses lesser bits as compared to uncompressed data resulting in less usage of bandwidth when downloaded. Hence resulting in quicker transfer speed.

2. Literature Survey

In this paper [1] twotechniques for speech compression using Wavelet Transform (WT) has been introduced. The first technique is Zero Wavelet transform (ZWT), eliminates the high frequency coefficients of the wavelet decomposition with energy values below a certain threshold level. The second technique, Average Zero Wavelet Transform (AZWT).It averages the approximate coefficients of the wavelet decomposition. These Coefficients are almost constant at higher decomposition levels of the transform. The wavelet coefficients are, then, quantized using Lloyd's algorithm and coded using the entropy coding technique before being transmitted. At the receiverend, the received signal is decoded and dequantized before being processed.

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Paper [2], describes the application of Discrete Wavelet Transform (DWT) for analysis, processing and compression of multimedia signals like speech and image. More specifically explore the major issues concerning the wavelet based speech and image compression which include choosing optimal wavelet, decomposition levels and thresholding criteria. The simulation results prove the effectiveness of DWT based techniques in attaining an efficient compression ratio of 2.31 for speech.

Paper [3] presents a new adaptive algorithm for speech Compression using Cosine Packet Transform. The proposed algorithm uses packet decomposition, which reduces a computational complexity of a system. In this paper the compression ratio of methods using Wavelet Transform, Cosine Transform, Wavelet Packet Transform and proposed adaptive algorithm using Cosine Packet Transform for different speech signal samples, comparison is presented.
Paper [4] explains FPGA design of speech compression by using different discrete wavelet transform (DWT) schemes including the Daubechies DWT and the Daubechies Lifting Scheme DWT. In this design work, the audio CODEC chip is used to convert analog speech into the digital format. The digital streams can either be stored inside the SDRAM for DWT post processing or compressed in real time by using the Daubechies Lifting Scheme DWT. The low pass filtering part of the DWT result represents the compressed speech. It can be read back from the SDRAM, converted to analog signal and then played clearly in speakers after up-sampling is performed.

In paper [5], a novel algorithm for speech coding utilizing Compressive sensing (CS) principle is developed. The sparsity of speech signals is exploited using gamma tone filter bank and Discrete Cosine Transform (DCT) in which the compressive sensing principle is then applied to the sparse sub-band signals. All parameters will be optimized using informal listening test and Perceptual Evaluation of Speech Quality (PESQ). In order to further reduce the bit requirement, vector quantization using codebook of the training signals will be added to the system. The performance of overall algorithms will be evaluated based on the processing time and speech quality. Finally, to speed up the process, the proposed algorithm will be implemented in a multi-core system, i.e. six cores, using Single Program Multiple Data (SPMD) parallel paradigm.

2. Proposed System

The system proposes a method to Speech Compression and Decompression using Discrete Wavelet Transform (DWT), Speech compression 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 system is to compress the digital speech using discrete wavelet transform. The uncompressed digital speech requires huge amount of memory, the main idea behind the speech compression algorithm is to represent this uncompressed speech with minimum number of bits and optimum speech quality. Discrete Wavelet Transform has been recently proposed for signal analysis. The discrete wavelet transform is useful to remove redundancies and irrelevancies present in the speech signal for the compact representation. 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 encoding operations. Finally the compressed signal is reconstructed. Analysis of the compression process was performed by comparing the compressed-decompressed signal against the original.

Block diagram

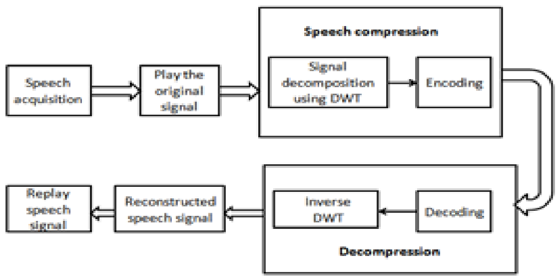

Fig1:Block Diagram of Proposed System

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2. proposed System Description

This project proposes a method to Speech Compression and Decompression using Discrete Wavelet Transform (DWT), Speech compression 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 project is to compress the digital speech using discrete wavelet transform. The uncompressed digital speech requires huge amount of memory, the main idea behind the speech compression algorithm is to represent this uncompressed speech with minimum number of bits and optimum speech quality. Discrete Wavelet Transform has been recently proposed for signal analysis. The discrete wavelet transform is useful to remove redundancies and irrelevancies present in the speech signal for the compact representation. Basically, the operations like transform, encoding is performed. Finally the compressed signal is reconstructed. Analysis of the compression process was performed by comparing the compressed signal against the original.

3. Wavelet decomposition using DWT

In DWT, filters of different cut-off frequencies are used to analyse the signal at different scales. The signal is passed through a series of high pass filters for the analysis of high frequencies, and it is passed through a series of low pass filters for the analysis of the low frequencies. For many signals, the low-frequency content is the most important part. It is what gives the signal its identity. The high-frequency content, on the other hand, imparts flavour or nuance. Take for example the human speech. Removing the high-frequency components would make the voice sound different, but still intelligible. However, removing enough of the low-frequency components would make the voice sound garbage.

The DWT process starts with passing the signal through a half-band digital low-pass filter. This kind of filter removes all kinds of frequencies that are above half of the highest frequency in the signal. Filtering a signal corresponds to the mathematical operation of convolution of the signal with impulse response of the filter:

$$x[n]*h[n] = \sum_{k=-\infty}^{\infty} x[k] \cdot h[n-k]$$

This process produces the DWT coefficients. The actual lengths of the coefficient vectors are slightly more than half the length of the original signal. This has to do with the filtering process, which is implemented by convolving the signal with a filter. The convolution "smears" the signal, introducing several extra samples into the result. This smear is the impulse response of the filter. The low pass filtering removes the high frequency information, but leaves the scale unaffected. Only subsamplingchanges the scale. Subsampling a signal corresponds to reducing the sampling rate, or removing some of the samples of the signal. After passing the signal through a low-pass filter, its scale is now doubled, since the high-frequency half of the signal is removed. As for the resolution of the signal, which is the amount of information in the signal, filtering divides the resolution into two. This procedure can mathematically be expressed as

$$y_{high}[k] = \sum_{n} x[n] \cdot g[2k-n]$$

$$y_{low}[k] = \sum_{n} x[n] \cdot h[2k-n]$$

where yhigh[k] and ylow[k] are the outputs of the low pass and high pass filters, x[n] is the original signal to be decomposed and h[n] and g[n] are low pass and high pass filters, respectively [1]. The decomposition process, also known as sub band coding, can be repeated, with successive approximations being decomposed in turn, so that one signal is broken down into many lower resolution components. For every level, the filtering and subsampling halves the number of samples and halves the frequency band. This is called the waveletdecomposition tree [2]. The decomposition can continue only until the individual details consist of a single sample. In practice, the suitable number of levels is based on the nature of the signal. One important property of DWT is the

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relationship between the impulse responses of the high pass and low pass filters. These filters are dependent of each other and are related by

$$g[L-1-n]=(-1)^n \cdot h[n]$$

The process called reconstruction, or synthesis, is assembling back the processed signal into the original signal without loss of information. The mathematical manipulation that effects synthesis is called the inverse discrete wavelettransform (IDWT). Reconstruction is easy. Where wavelet analysis involves filtering and down sampling, the wavelet reconstruction process consists of upsampling and filtering. If the filters are not ideal, then perfect reconstruction cannot be achieved [2]. The reconstruction formula becomes

$$x[n]=\sum_{k=-\infty}^{\infty}\left(y_{high}[k]\cdot g[-n+2k]\right)+\left(y_{low}[k]\cdot h[-n+2k]\right)$$

Upsampling is the process of lengthening a signal component by inserting zeros between samples and low pass filtering to remove the high frequency images produced by zero-insertion.

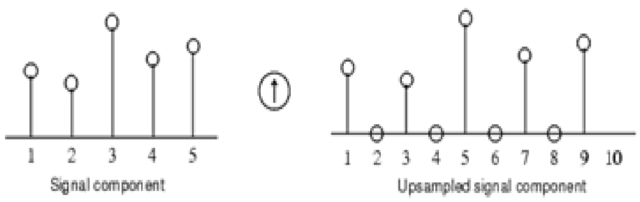

Fig.2 :Upsampling

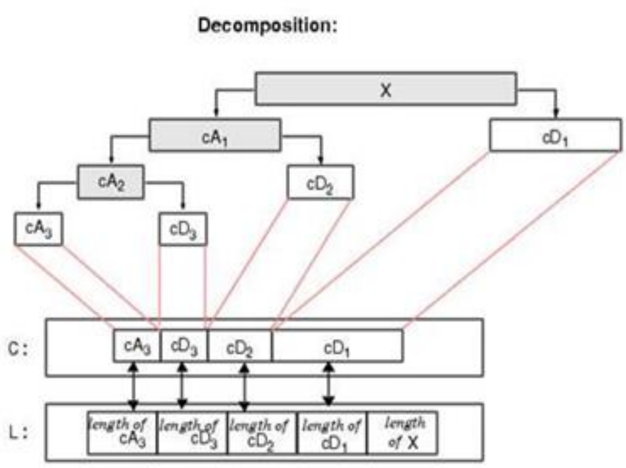

Fig 3: Level 3 Decomposition of a Sampled Signal

4. Usefulness of Discrete Wavelet transforms over continuous Wavelet transforms

i) Multi-resolution representation of the signal by wavelet decomposition that greatly facilitates sub-band coding.

ii) It reaches a good compromise between frequency and time resolution of the signal.

iii) Wavelet transforms are superior to DCT as their basis function over good frequency resolution in the lower frequency range, and at the same time they yield good time resolution at a higher frequency range.

5. Software and Hardware

Proposed system has been designed using dot net language in visual studio 2010.

6. Requirement analysis

Requirement analysis bridges the gap between system engineering and software design.

Requirement analysis is the most important task in software development life cycle.

1) Necessary requirement

- There should be scope for the future expansion.
- Easy updating
- No viruses to be loaded along with the software.
- User Friendly.
- Affordable
- Easy updating and usage of software
- Facility to support all resolutions
- It should be windows based
- Security should be provided

2) System requirement

Hardware System Requirement

- Processor - Intel (R) Core (TM) i3 CPU
- Installed RAM - 1 GB
- Hard Disk - 30 GB
- Keyboard - Standard Windows Keyboard
- Mouse - Two or three button mouse
- Monitor - SVGA

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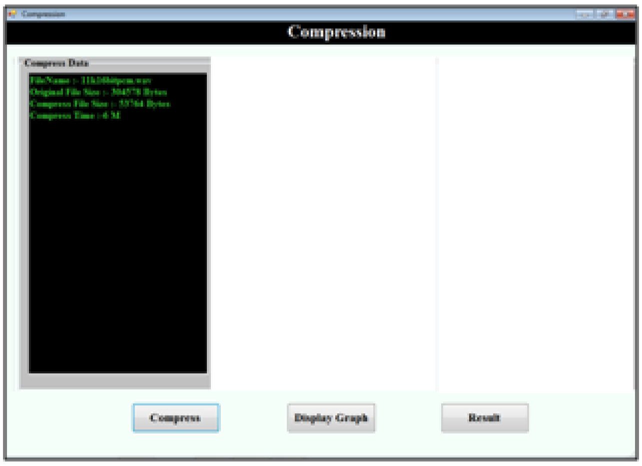
- Software System Requirement
- Operating System - Windows 2000/XP/7/8
- Front end - .net, VB 2010
- Back end - c#

4. Results and Observation

Mathematical Results

Sr.No.	Name of file	Bit rate	Original file size(in bytes)	Compressed file size (in bytes)	Compression ratio	Compression factor	Saving percentage (in %)
1.	8k8bitpcm.wav	64kbps	110532	21491	0.1944	5.1431	80.55
2.	11k8bitpcm.wav	88kbps	152312	29648	0.1946	5.1373	80.53
3.	8k16bitpcm.wav	128kbps	221026	39009	0.1764	5.6660	82.35
4.	11k16bitpcm.wav	176kbps	304578	53764	0.1765	5.6650	82.34

In proposed system, one speech file with variable bit rate was taken as input samples and applied to system .From result table as shown below ,we can see that compression factor is not constant, so overall system that utilized DWT proved to variable compression factor and promising in area of compression. Table shows the comparison between file size of proposed system with original audio file.



5. CONCLUSION

The discrete wavelet transform performs very well in the analysis and processing of non-stationary speech signals. The greatest advantage of wavelet over other techniques is that the compression factor is not constant and it can be varied while most other techniques have fixed compression factors. Discrete wavelet transform significantly improves the reconstruction of the compressed speech signal and also yields higher compression factor as compared to FFT and DCT.

6. REFERENCES

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