COMPRESSION OF AUDIO SIGNALS USING HYBRID WAVELET TRANSFORM

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Abstract

The need for transmission of audio signal has been increased over the decades. Keeping this in view, audio compression technology has emerged which facilitates transmission with ease and efficient. Speech Conversion is the process of converting human speech into efficiently encoded format and is decoded to produce a close approximation of the original voice signal. An efficient algorithm named Hybrid Discrete Wavelet Transform (DWT) is used for decomposition of original voice signals into wavelet coefficients at different scales and positions. These coefficients are then truncated to perform further encoding and decoding process. In comparison to the earlier techniques an efficient compression technique is used in this paper. This hybrid wavelet transform decomposes the speech signal into multi resolutions components and discard those with low energy inorder to compress the speech audio signal. Audio Compression technique plays a vital role while processing speech signals in digital communication processes such as satellite communications, internet communications, transmission of biomedical signals, etc. The experimental results are performed on various speech signals using matlab software tool.

Keywords

Audio Compression, Wavelet Transform, Wavelet Decomposition

1. Introduction

Voice communication is a major tool to convey information from one person to other, which is limited to a bandwidth of 4kHz. In present scenario multimedia communication demands for efficient use of transmission bandwidth, storage space and power. In this paper we analysing the compression process of speech signal data to overcome the above consequences. Earlier PCM was majorly used for encoding audio signals, which doesn't help in removal of any kind or redundancies. In past few researches have suggested several efficient algorithms based on transformation such as Fourier Transform (FT), Wavelet Transform and Discrete Cosine Transform (DCT), among which Fast Fourier Transform was been popular which was based on assumption that signals are stationary. Invention of Short time Fourier Transform (STFT) helped to address the issue to some extent but the it was not completely resolved. In present scenario most of the signals are non-stationary, for better results they are to be analysed without any assumptions that signals are linear and stationary as suggested by Fourier analysis.

Usage of wavelets has been a revolutionary step in process of signal analysis which is used to analyse both linear and non-stationary signals. Speech is a non-stationary random process as it is influenced by time varying nature and sudden change in frequencies of human vocal system Non-stationary signals are characterized by their sudden change in their frequency modulations. Wavelets have special feature of localization along with their resolution properties in time- frequency makes them well suitable for coding speech signals. As the multimedia technology has grown, speech compression technique is very necessary due to limited bandwidth.

2. Wavelet Compression

Wavelet is recently developed to overcome the limitations of Fourier Transform of audio signals, which has the ability to examine the audio signal simultaneously in both time and frequency. The original signal called the mother wavelet is divided and scaled by wavelet analysis.Wavelet Transform has emerged as a powerful mathematical tool in various areas of science and technology, majorly in the field of signal processing.

a. Representation of Audio Signal Using Wavelet

Wavelet Transform is a Small Wave that has its energy confined in time and it can be used for analysis of transient, non-stationary,it is always a time varying phenomenon and its not stationary. Its property is like a oscillating wave.The waveform is a limited duration having a average zero value .In a wavelet transform Signal is represented in a time frequency representation.Here ,Decomposition of a signal into set of basic functions known as WAVELETS. Wavelets with large number of vanishing movements is useful for Audio Compression.

b.Wavelets based Compression Techniques:

Wavelet audio signals are concentrated into few neighbouring coefficients. If Wavelet Transform of a signal is considered, many coefficients of the signal become zero or negligible, due to which compression is done by excluding them as small or negligible valued coefficients are insignificant data. Compressing of audio signal is done as per the following steps.

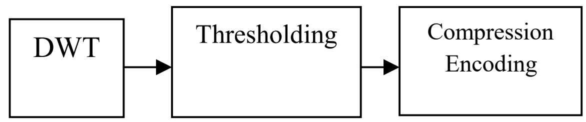

Fig 1. Design model of Audio Compression

i. Thresholding:

Thresholding is done after receiving the coeffecients from different transforms. Coefficients whose values are less than the threshold values are then removed.

ii. Quantization:

Quantization is the process of mapping a set of continuous valued data. It helps in reducing the data found in threshold coefficients ensuring that outcome is with minimum error. Usually uniform quantization is peformed.

iii. Encoding:

While encoding techniques like decomposition with N equal frame is used as it removes the data which is repetitively occurring in signals as a result of which coefficients are reduced. For reducing the bandwidth, Lossless or Lossy encoding techniques are used so as to achieve compression. The original signal is reconstructed from the compressed signal by decoding technique followed by de-quantization.

3. Psychoacoustics Model

Psychoacoustics is the scientific study of how humans responds and perceive to various audio signals. It deals with

physiological responses humans associated with sound like noise, speech and music.
Psychoacoustics deals with area's which include psychology, acoustics, physics, biology, electronic engineering and computer science.

Human ears can normally hear sound frequencies within the range 20Hz (0.02 kHz) to 20,000 Hz (20 kHz) whereas listening ability decreases as the age increases. Most of the adults are unable to hear sound frequencies above 16 kHz. The lowest sound frequency is 12Hz which is identified as a musical tone under ideal laboratory conditions. Humans sense of touch can perceive tones between the range of 4-16 Hz.

Human ears can respond to frequency of 3.6 Hz within the octave of 1000 to 2000 Hz. There are other means of perceiving smaller pitch differences. There is a repetitive variation in volume of the tone due to the interference of two pitches. Beating occure which means amplitude modulation with a frequency equal to the difference in frequencies of the two tones .

4. Design Methodology

The major motive of the algorithm in this paper is to obtain low bit rates by compressing high quality audio signals by maintaining transparent quality. The following steps are considered to achieve this goal:

. Step 1. Designing a wavelet representation for audio signals.

Step 2. Designing a psychoacoustic model to perform perceptual coding and adapt it to the wavelet representation.

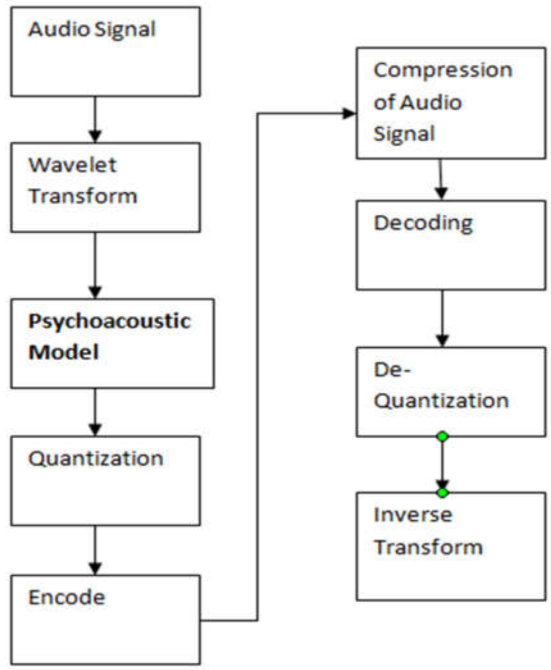

Fig 2. Block Diagram of Proposed system

Step 3. Reducing the number of the non-zero coefficients of the wavelet representation and performing quantization over those coefficients.

Step 4. Perform extra compression to reduce redundancy over that representation

Step 5. Transmit or store the steam of data. Decode and reconstruct. Evaluate the quality of the compressed signal.

5. Results and Discussion

MATLAB is used to implement the audio compression experiment. Using the Matlab command "wavread" the audio signal is loaded.The original signal is also plotted for comparision between original and compressed signal. Here we consider speech based audio signal as input.

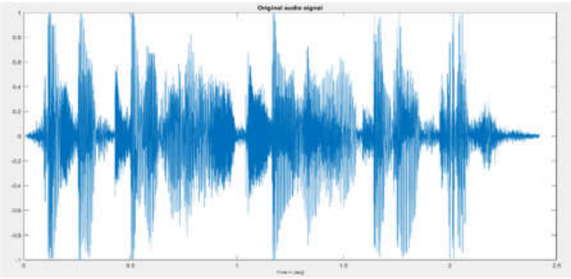

Fig 3. Original Audio Signal

Encoded audio signal is shown in fig 4.

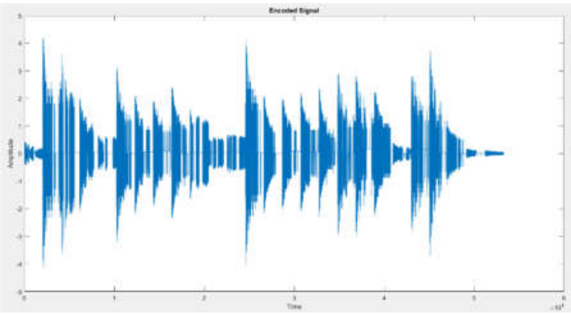

Fig 4. Encoded Audio Signals

After performing the psycho acoustic model the decode and compressed audio signal is shown in fig 6.

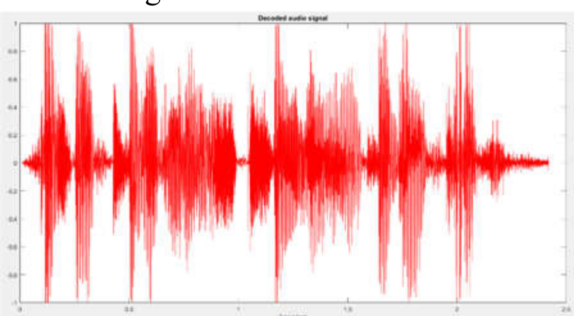

Fig 7. Decoded Audio Signal

6. Conclusion

Using Wavelets Audio Compression was implemented. Error is negligible and reconstruction is good.It is used for transmission and storage. The compression is achieved by representing each sample of digitized data by lesser number of bits and making it occupy lesser space and consequently easy to transmit or store.Good compression ratio is also achieved.

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