PRESERVING QUALITY WHILE COMPRESSING THE VOLUME OF AUDIO AND VIDEO DATA

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Abstract *The proposed technique protects the data quality by applying Linear Predictive Coding-10 and achieves the low bit rate by applying Quadrature Mirror Filter. A comprehensive analysis is on the basis of given parameters as size, compression time, Signal to Noise Ratio, power, energy, power in air, energy in air, mean, standard deviation and intensity.*

Keywords: *Modulation, Energy, LPC, Compression, Quality, Bandwidth.*

INTRODUCTION Speech coding is a powerful tool for data compression ofdigital audio files which contains speech. Speech coding has two main applications as mobile telephony and voice over IP. The main objective ofthe speech coding is to characterize speech in digital form with as few bits as possible while maintaining the simplicity and quality required for the particular application. There is always a trade off between lower bit rate and voice quality. Next section of this paper describes an introduction of the existing speech coding techniques and problems ofthe existing techniques. To achieve the better results an enhanced speech coding technique is proposed that is based on the Linear Predictive Coding-l0 and then merged with Quadrature Mirror Filter in the next section. The results ofthe existing techniques and the proposed technique is then presented and compared in tabular and graphical form. The last section concludes the paper.

Usually speech coding standards deal with narrowband speech. Narrowband speeches are digitizing speeches with sampling frequency of8 kHz. Narrow-band speech coding has following standards

- Waveform Coders
- Parametric Coders
- Hybrid Coders

Waveform coders contain the following coders

- PCM (Pulse Code Modulation)
- DPCM (Differential pulse code modulation)
- ADPCM (Adaptive Differential Pulse Code Modulation)

Waveform coders were widely used in early digital communication systems. Waveform coders have higher bit rate to maintain quality of data. The elementary coding scheme for waveform coding is Pulse Code Modulation(PCM). Waveform coders have very low complexity and delay, but the main disadvantage of the waveform coders is that they require large number of bits to maintain good data quality. Parametric coders contain the following coders

- LPC (Linear Predictive Coder)
- MELP (Mixed Excitation Linear Prediction)

The parametric coders use multiple parametric models to generate speech signals. Parametric codes make no attempt to preserve quality of the synthetic speech. In Linear Predictive Coder a filter is used that is known as time-varying

filter. The coefficients of the filter are derived by an LP analysis procedure. Hybrid coders contain the following coder

CELP (Code-Excited Linear Prediction) a variations of CELP is used. Hybrid coders combine features of waveform coders and parametric coders. The main objective of the hybrid coders is to capture the dynamics of the signal and match the synthetic signal to original signal in the time domain. CELP use long-term and short term linear predictive models for speech synthesis. The complexity of the Hybrid coders some times becomes questionable. Existing techniques make a wide trade off between the data quality, performance bit rate parameters. The main objective of data quality is badly affected in this parametric trade off.

LINEAR PREDICTIVE CODING-l0

Linear predictive coding (LPC) is used in speech processing for representing the phantom packet of a digital signal in compressed form, using the information of a linear.

thedata.The basic working principleofthe first sectionisthesameasthatofLPC-IO. The major enhancement that is doneon the LPC-IOisthat insteadofthe10 coefficient thistechnique uses the first18 Coefficients

*H (z)= G / [I-II ak*z-k]*

The summation is starting from k = 1 up to p, which is 18 for proposed technique. For proposed technique the coefficients are computed by auto-correlation formulation. This method is superior to the covariance method. A process of decision making is done in LPC analysis to determine that either frame is voiced or unvoiced. A complete process is done to determine the pitch of the voice signal. For the decoder of the proposed technique only the first few low frequencies of the signal are required. For more compression as of the LPC-10 the proposed technique uses discrete Cosine Transform (DCT). The main reason to utilize this technique is that DCT concentrates most of the energy of the signal in the first few coefficients

The second section of the proposed technique is based on the Quadrature Mirror Filter to transmit the audio data at low bit rate. As the coefficients could be quantized using only 4 bits in the DCT due to this reason QMF make the four sub-band of the audio file that is compressed by the LPC-18. After this division the files are ready to transmit. If at the receiving end we lost Ist sub-band then it is not possible to reconstruct the original file. However if from the remaining three sub-bands, any of the band is lost, a voice can be constructed with low quality but is understandable. Here is the architecture of the proposed technique.

COMPARISON AND RESULTS The comparison and results are in graphical and tabular fonn. In the implementation of the project the results are analyzed in two main categories listed as follow: concentrates most of the energy of the signal in the first few coefficients.

Figure 3: D Model

- Subjective Mode
- Objective Mode

The following table shows the values of the file that is being compressed by LPC-I0 compressor. In subjective mode the results are presented into audio fonn, and then the comparison is made between existing and he proposed technique. While in the objective mode results are presented in graphical fonn on the basis of following given parameters:

- Signal-to-Noise Ratio
- Compression Time
- Standard Deviation
- Power of files
- Sampling Period of file

All these values are calculated by the software named PRAAT.

PRAAT is software used for phonetic analysis and sound manipulations. By Using PRAAT, a detailed analysis of the existing technique (linear predictive coding LPC-IO) with the proposed technique is perfonned. An audio file is

passed through this software and the software show the values of the following parameters. In the tables these parameters are simplified as following

Period
Frequency
Power
Energy
Power in Air
Energy in Air
Mean
Standard Deviation

The last form of the data on the transmission end is the four sub-bands of the voice. We have to transmit these four sub-bands on some medium or a channel. The Proposed system leaves this option to the user that what kind of transmission they will use. Data is presented on the transmission end in such a flexible and manageable form that it can be transmitted on both type of mediums e.g., Serial transmission or parallel transmission. The given results signify the serial transmission. For parallel transmission intelligent devices are required on the base control station. These devices have the control and ecision making procedure that decides which sub-band is to be transmitted on which channel.

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