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PRESERVING QUALITY WHILE COMPRESSING THE VOLUME OF AUDIO AND VIDEO DATA

Abstract: *High data quality at lowbit rateis an essential goal that people want to achieve. It isnecessary to transfer data atlow bit rate so that the bandwidth of the medium can be utilized efficiently. In most of the speech coding techniques the goal of low bit rate transfer is achieved but the data quality is affected badly. The proposed technique isan attempt to improve the data quality at low bit rate as well as fast transmission of data. The proposed technique protects the data quality by applying Linear Predictive Coding-10 and achieves the lowbit 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

Of digital audio files which contains speech. Speech coding has two main applications as mobile telephony and voice over IP. The main objective

Of the 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 of the existing techniques. To achieve the better results an enhanced speech coding technique is proposed that is based on the Linear Predictive Coding-10 and then merged with Quadrature Mirror Filter in the next section. The results of the existing techniques and the proposed technique is then presented and compared in tabular and graphical form. The last section concludes the paper.

Speech coding is categories into two broader categorizes as

- Wide-band speech coding
- Narrow-band speech coding

Usually speech coding standards deal with narrowband speech. Narrowband speeches are digitizing speeches with sampling frequency of 8 kHz. Narrow-band speech coding has following standards

- Waveform Coders
- Parametric Coders
- Hybrid Code

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.

CONCLUSION

The comparison of the existing and proposed technique shows that the proposed technique has better results and it improves the data quality. The proposed technique has the flexibility to implement it on the serial and parallel data transmission environment. Using limited modifications, this technique can be implemented for parallel environment. 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.

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