DIFFERENT SPEECH COMPRESSION TYPES IN SIMULTANEOUS INTERPRETING

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This publication paper describes the different types of speech compression techniques. First of all we are eager to give the definition of the notion “speech compression”. The aim of speech compression is to reduce the number of bits required to represent speech signals by removing the redundant bits so-that the less bandwidth is required for transmission. Before discussing the speech compression coding techniques, it is important to understand the digitization process. The speech signal is represented in its digital form, that is, the process of speech signal digitization. There are basically two the key features of speech signal are voiced and unvoiced speech and their characteristics. In broader terms, speech compression techniques are mainly focused on removing short-term correlation (in the order of 1ms) among speech samples and long-term correlation (in the order of 5 to 10 ms) among repeated pitch patterns. In this section, we will start with speech signal digitization and then discuss speech signal features and speech compression techniques.

Speech compression is use in the encoding system. The bit rate reduction is use in the encoding system. By the use of bit rate reduction algorithm, the minimum bits are used to compare the original information.

There are different speech compression techniques are present. Basically is divided in two types:

- Lossy
- Lossless.

Lossy compression means a class of data compression algorithm that allows the exact original data to be reconstructed from the exact original data to be reconstructed from the compressed data but bit rate and is better than lossless. It is compression ratio is higher than lossless compression. While lossless means output signal and input signals sounds undistinguished. Speech coder analysed using subjective and objective analysis. Subjective is making judgments by listening output and original signal. Playing back signal and checking quality. Objective includes technical assess. Such as computing segmental signal to noise ratio (SEGSNR) between original and output signal. Speech compression motivation is to remove redundancy in speech representation to reduce transmission bandwidth and storage space or memory (and apart to reduce cost). The purpose of speech compression is to reduce the number of bits required to rep- resent Speech signals (by reducing redundancy) in order to minimize the requirement for transmission bandwidth (e.g., for voice transmission over mobile channels with limited capacity) or to reduce the storage costs (e.g., for speech recording). Before we start describing speech compression coding techniques, it is important to understand how speech signal is represented in its digital form, that is, the process of speech signal digitization. There are in general three basic speech compression techniques, which are waveform-based, parametric based and hybrid coding techniques.

➢ WAVEFORM BASED SPEECH CODING

Waveform-based codecs are intended to remove waveform correlation between speech samples to achieve speech compression. It aims to minimize the error between the re- constructed and the original speech waveforms. It is classified as time domain and frequency domain

• Time domain: such as A. PCM (Pulse code modulation) B. ADPCM (Adaptive Differential PCM)

• Frequency domain or Transform coding: such as A. Fast Fourier Transform (FFT) B. Discrete Cosine Transform (DCT) C. Continuous Wavelet Transform (CWT) D. Discrete Wavelet Transform (DWT) Waveform coders are able to produce original signal at decoder (Lossless). Bit rate range – 64 kb/s to 16 kb/s. At bit rate lower than 16 kb/s, the quantization error for waveform based speech compression coding is too high, and this results in lower speech quality.

➢ PARAMETRIC-BASED SPEECH CODING

Parametric-based compression methods are based on how speech is produced. Instead of transmitting speech waveform samples, parametric compression only sends relevant parameters related with speech production to the receiver side and reconstructs the speech from the speech production model. Thus, high compression ratio can be achieved. Bit rate range – 1.2 kb/s to 4.8kb/s

Many different techniques are explored to represent waveform-based excitation signals such as multi-pulse excitation, codebook excitation and vector quantization. The most well known one, so called Codebook Excitation Linear Prediction (CELP)‖ has created a huge success for hybrid speech codec in the range of 4.8 kb/s to 16 kb/s for mobile/wireless/satellite communications. Types of Hybrid speech compression: A. Codebook Excitation Linear Prediction (CELP) B. Vector Sum Excited Linear Predictive Coder (VSELP)

Today, many compression techniques are developed and some techniques are in process. But this paper only discusses the general idea about the Waveform-based speech compression, Parametric-based speech compression and Hybrid based speech compression. Parametric based codec is higher in implementation complexity but can achieve better compression ratio. This paper has been written to understand Parametric-based speech compression in the better manner and relate it to the future work.

As the data grows day by day, the short and compression Communication is required, Speech compression techniques plays a vital role in it. In this paper all the possible speech compression techniques are discussed that can be used to compress the data before transmission of data so that it can consume less bandwidth.

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