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Redundancy Remove Technique for Real Time Voice Compression

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***Abstract*:** The aim of this paper is to develop a code for encoding and compressing speech in real time . A powerful speech analysis technique called “redundancy remove technique” is implemented. The paper justifies the implementation of the redundancy remove technique. This technique is used because it provides extremely accurate estimation of speech parameters, and is relatively efficient for computation. The speech quality and complexity were analyzed.

Keywords: *Matlab , voice compression , redundancy , speech parameters.*

I. INTRODUCTION

Voice compression is the conversion of analog voice signals to digital signals using minimum band width. It is a pretty old technology and dates back to early 90's, which witnessed the coming of voice compression technology being used by satellite and International long distance operators. Voice compression is becoming increasingly popular and with good reason, it saves enormous amounts of money and in some cases allows for communications that would not otherwise be possible. It can be also used in situations where the amount of bandwidth available is limited..

Due to the increasing demand for speech communication, speech coding technology has received augmenting levels of interest from the research, standardization, and business communities. Advances in microelectronics and the vast availability of low-cost programmable processors and dedicated chips have enabled rapid technology transfer from research to product development . This encourages the research community to investigate alternative schemes for speech coding, with the objectives of overcoming deficiencies and limitations.

The standardization community pursues the establishment of standard speech coding methods for various applications that will be widely accepted and implemented by the industry.

II. METHODOLOGY

There are a variety of compression techniques commonly used in the Internet and other systems. **T**he components of a system are capturing , transforming , coding and transmitting as shown in Figure (1).

Figure (1) components of simple compression system

An input signal is converted from some continuously varying physical value (e.g. voice wave) into a continuously electrical signal by some electro-mechanical device. This continuously varying electrical signal can then be converted to a sequence of digital values, called samples, by some analog to digital conversion circuit.
Two factors determine the accuracy of the sample with the original continuous signal:

1. The sampling rate is based on Nyquist theorem, the digital sampling rate must be twice of the highest frequency in the continuous signal.
2 The number of bits used in each sample. (Known as the quantization level.).

The original data can be transformed in a number of ways to make it easier to apply certain compression techniques. Generally the coding system components are illustrated in details on the following diagram (Figure (2)):

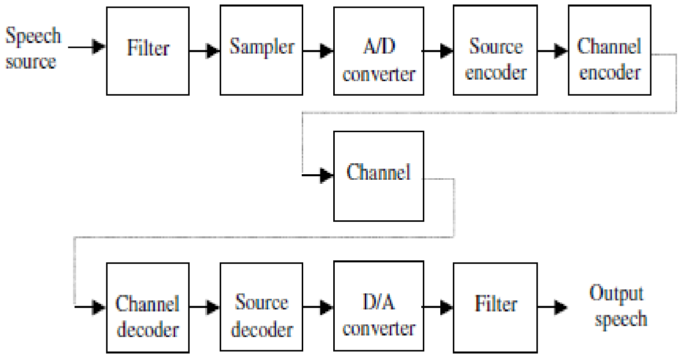

Figure (2) Block diagram of a speech coding system

III. REDUNDANCY REDUCTION TECHNIQUE

Redundancy reduction is a technique for the reduction of data from audio sources. These techniques are applied to reduce the quantity of information to be stored or transmitted, but are independent of the end-application, medium or transmission channel, i.e. do only exploit the properties of the source signal itself or the final receiver exposed to this signal (the listener).Redundancy reduction processes have proven highly effective in compressing the bandwidth of voice signal.

Audio compression algorithms are often referred to as “audio encoders”. Applications reduces required storage space as well as reduces required transmission bandwidth. Lossy and lossless algorithms are the audio compression algorithms that are implemented as audio codec.

Lossless Audio Compression removes redundant data, resulting signal is same as original – perfect reconstruction. Lossy Audio Encoding removes irrelevant data, resulting signal is similar to original.
Information redundancy is reduced in both forms of compression with the use of methods like coding, pattern recognition and linear prediction. However, lossy algorithms provide greater compression rates and are therefore used in mainstream consumer audio devices. Lossy audio compression algorithms can be defined as the ones that do not retain every bit of data and only reproduce a signal that sounds more or less like the original.

IV. PROGRAMMING

The algorithm (flow Chart) is shown in figure (3) below. The processing starts with recording audio signal (. WAVE format file). A sampling processing based on the principles of the Nyquist theorem is conducted on the recorded file .The redundant samples are then eliminated in order to reduce the size of the file. Therefore the data in the file get compressed and send over the LPT port.

Matrix Laboratory (MATLAB) is used for coding the compression procedure. Matlab is a numerical computing environment and fourth-generation programming language. Developed by Math Works, MATLAB allows matrix manipulations, plotting of functions and data, implementation of algorithms and creation of user interfaces.

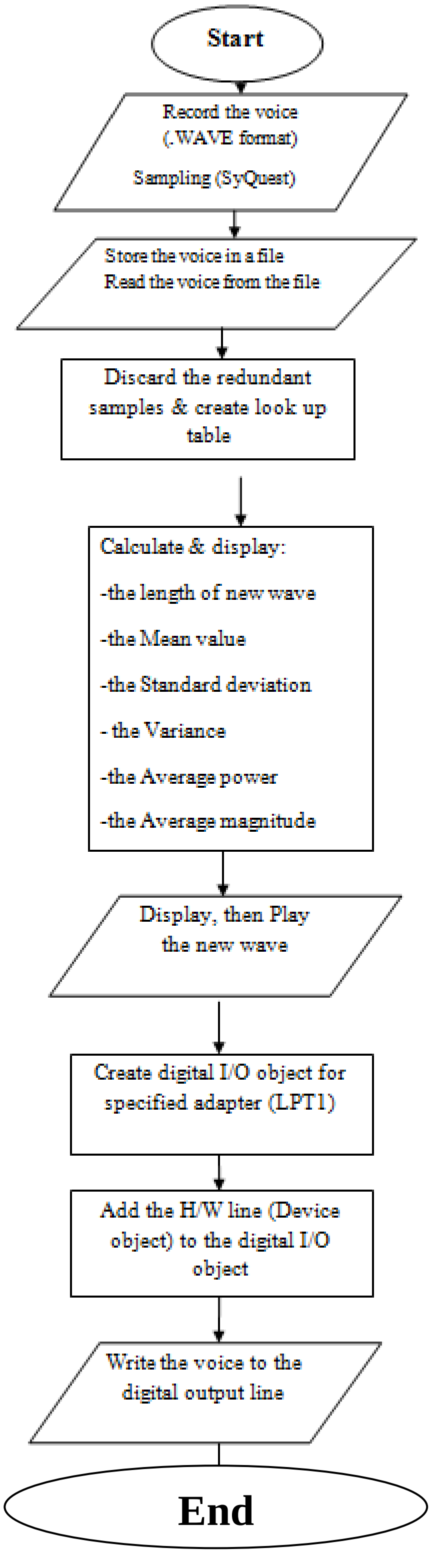

Figure (3) flow Chart for data compression and sending

A reverse procedure can be conducted for capturing and decompressing the file in order to return it to its original form.

V. RESULTS

It is seen that:
The data rates associated with uncompressed digital audio are substantial.
Digital audio compression enables more efficient storage and transmission of audio data..
The redundancy remove technique is simple, Powerful, high-compression and high audio quality algorithm.

Figure (4) . a shows the original voice signal before compression and figure (4).b shows the original voice signal after compression.

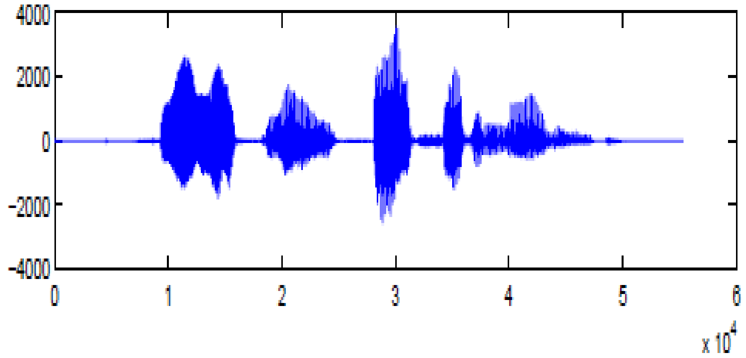

Figure (4).a the original voice signal

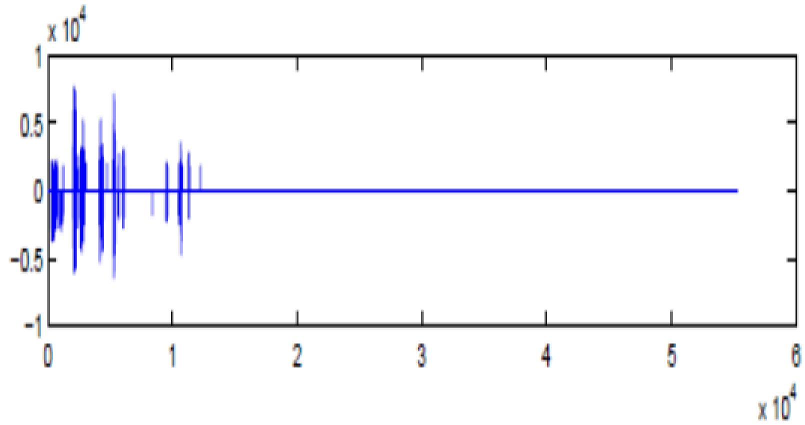

Figure (4).b The compressed voice signal

VI. CONCLUSION

The design of a particular coding algorithm is often dictated by the target application. Therefore, during the design of an algorithm the relative weighting of the influencing factors requires careful consideration in order to obtain a balanced compromise between the often conflicting objectives .The redundancy remove method reduces the band width needed for voice traffic and thus provides high voice quality , better processing time and excellent performance. Therefore ,it is favored to be implemented for real-time applications.

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