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Linear Predictive Coding Algorithm with its Application to Sound Signal Compression

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Abstract:- Recently, with the advances in digital signal processing, compression of audio signals has received great attention for speech processing applications. The perceived loudness of a sound signal by human is spectrally masked by background noises. This effect causes a shift of audible sound pressure level. This paper investigates the application of linear predictive coding (LPC) algorithm as compression of recorded sound signal. LP provides parametric modelling techniques which are used to model the spectrum as an autoregressive process for sound signal compression. Linear prediction and its mathematical derivation will be described. Finally, simulation results show the original recorder signal and the compressed signal sinusoidal with different audio signal. Results indicate that the linear predictive coding compresses the signal to a great extent with small distortion of signal.

Index Terms: **Linear predictive coding (LPC), Autoregressive process.**

I. INTRODUCTION

Uncompressed audio require substantial storage capacity. Data transfer of uncompressed audio data over digital networks requires that very high bandwidth be provided for a single point-to-point communication. To be cost-effective and feasible, multimedia systems must use compressed audio streams. The importance of data compression is not likely to diminish, as a key technology to allow efficient storage and transmission [1]. With recent advances in digital signal processing, compression of sound signals has received great attention in speech signal processing applications [2]–[3]. Different compression methods used for speech signals [4]–[9] were investigated. Here,, either wavelet transforms or predictive coding was used. Other similar application are, electrocardiogram (ECG) [10], and electromyogram (EMG) [11] signals compression. Two families of algorithms that used for the compression of the sound signal namely lossy compression and Lossless compression. Lossy compression [14] means that some data is lost when it is decompressed. Lossy compression bases on the assumption that the current data files save more information than human beings can "perceive”. Thus the irrelevant data can be removed. Lossless compression means that when the data is decompressed, the result is a bit-for-bit perfect match with the original one. The name lossless means "no data is lost", the data is only saved more efficiently in its compressed state, but nothing of it is removed. Here, we are using sub band coding compression technique is used for the compression of the speech signal. Subband coding [12][13] considers the signal only in predefined regions of the spectrum, such as frequency bands. Audio techniques apply Differential Pulse Code Modulation (DPCM) to a sequence of PCM-coded samples. This technique requires a linear characteristic curve for quantization. This paper investigates the application of linear predictive coding (LPC) algorithm as compression of recorded sound signal. LP provides parametric modelling techniques which are used to model the spectrum as an autoregressive process for sound signal compression. Linear prediction and its mathematical derivation will be described. The outline of this paper is as follows. In Section II, we provide the basic of the linear prediction and block diagram is presented for linear predictor. Linear prediction error function is described in Section III. In Section IV levinson-durbin algorithm is explained with mathematical analysis.. Section V present the audio compression techniques based on linear prediction. Simulation results are presented in section VI. Finally, Section VII presents our conclusions and final comments.

II. LINEAR PREDICTION

A linear prediction (LP) model [4] predicts/forecasts the future values of a signal from a linear combination of its past values. A linear predictor model is an all-pole filter that models the resonance (poles) of the spectral envelope of a signal or a system. LP models are used in diverse areas of applications, such as data forecasting, speech coding, video coding, speech recognition, model-based spectral analysis, model-based signal interpolation, signal restoration, noise reduction, impulse detection, and change detection. In the statistical literature, linear prediction models are often referred to as autoregressive (AR) processes. The all-pole LP model shapes the spectrum of the input signal by transforming an uncorrelated excitation signal to correlated output signal whereas the inverse LP predictor transforms a correlated signal back to an uncorrelated flat-spectrum signal. Inverse LP filter is an all-zero filter, with the zero situated at the same position in pole-zero plot as the poles of the all-pole filter and is also known as a spectral whitening, or de-correlation filter . Poles are at denominator of the polynomial and zeros are at numerator of the polynomial.

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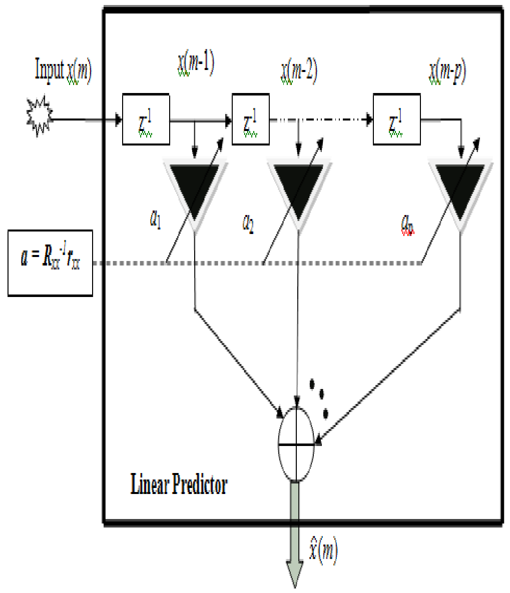

Fig. 1 Diagram of the Linear Predictor

III. LINEAR PREDICTION ERROR

The mean square prediction error becomes zero only if the following three conditions are satisfied: (a) the signal is deterministic, (b) the signal is correctly modelled by a predictor of order *P*, and (c) the signal is noise-free. For example, a mixture of *P*/2 sine waves can be modelled by a predictor of order *P*, with zero prediction error. However, in practice, the prediction error is non-zero because information-bearing signals are random, often only approximately modelled by a linear system, and usually observed in noise. The least mean square prediction error, obtained is

$$E^{(P)} = E[e^2(m)] = r_{xx}(0) - \sum_{k=1}^{P} a_k\, r_{xx}(k) \qquad (1)$$

Where $E^{(P)}$ denotes the prediction error for a predictor of order *P*. The prediction error decreases, initially rapidly and then slowly, with the increasing predictor order up to the correct model order. For the correct model order, the signal *e(m)* is an uncorrelated zero-mean random process with an autocorrelation function defined as

$$E[e(m)\, e(m-k)] = \begin{cases} \sigma_e^2 = G^2 & \text{if } m = k \\ 0 & \text{if } m \neq k \end{cases} \qquad (2)$$

Where σ_e^2 is the variance of *e(m)*.

I.

IV. LEVINSON-DURBIN ALGORITHM

The Durbin algorithm starts with a predictor of order zero for which $E^{(0)}=r_{xx}(0)$. The algorithm then computes the coefficients of a predictor of order *i*, using the coefficients of a predictor of order *i*-1. In the process of solving for the coefficients of a predictor of order *P*, the solutions for the predictor coefficients of all orders less than *P* are also obtained:

$$E^{(0)}=r_{xx}(0) \qquad (3)$$

For $i = 1,\ldots, P$

$$\Delta^{(i-1)}=r_{xx}(i)-\sum_{k=1}^{i-1} a_k^{(i-1)}\, r_{xx}(i-k) \qquad (4)$$

$$k_i = -\frac{\Delta^{(i-1)}}{E^{(i-1)}} \qquad (5)$$

$$a_i^{(i)} = -k_i \qquad (6)$$

$$a_j^{(i)} = a_j^{(i-1)} + k_i\, a_{i-j}^{(i-1)} \qquad \text{where } 1 \le j \le i-1 \qquad (7)$$

$$E^{(i)}=(1-k_i^2)E^{(i-1)} \qquad (8)$$

II. Audio Compression Based on Linear Prediction

Audio waveforms exhibit a high degree of continuity or sample-to-sample correlation, i.e., the tendency of samples to be similar to their neighbours [3]. The reason is that audio samples are digitized from continuous waveforms, and the sampling rate is usually higher than the rate needed at any particular time. To take advantage of this correlation, prior to the encoding process most audio compressors apply a pre-processing component called a predictor [4]. The predictor can reduce the sample magnitude by making a prediction of the current sample based on the knowledge of some given preceding samples, and then subtracting the prediction from the current sample value. As a result, the predictor eliminates the correlation inherent in samples before encoding. An audio file in compacted form is comprised of a header and a sequence of frames. The file header contains properties of the audio signal stream. Each frame consists of its own frame overhead and a sequence of residual codes. The frame overhead provides enough information so that the decoding process can start working without the knowledge of other frames. The frame overhead contains the corresponding block size, the prediction model, the residual coding algorithm, and all relevant parameters. Decoding, the reverse process, retrieves information from the compacted form and reconstructs audio samples.

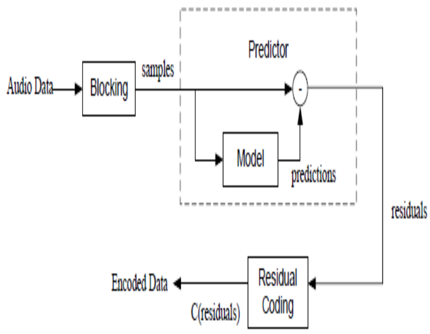

Fig. 2 Encoding Process

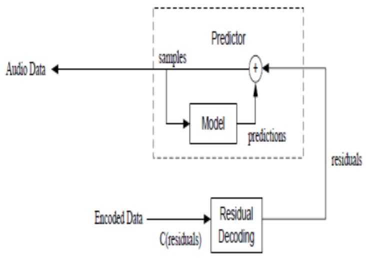

Fig. 3 Decoding Process

V. SIMULATION RESULTS

This section present the audio signal compression is performed using linear predictive coding. Simulation is presented using Matlab.

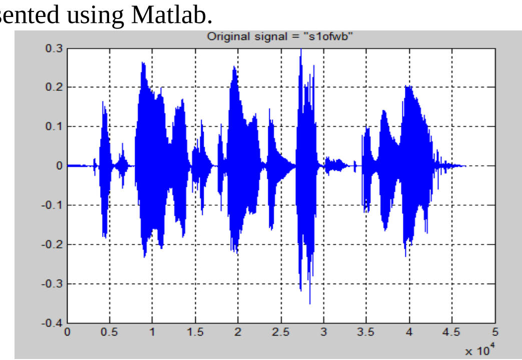

Fig. 4 Original Recorded Speech Signal

We are performing the experiments, with five different recorded voice signals to analysis the level of compression as well as performance variation in sound signal due to compression. The problem of signal compression or source coding is to achieve a low bit rate in the digital representation of an input signal with minimum perceived loss of signal quality. Fig. 4 shows the Original Recorded Speech Signal in different environment condition. Fig. 5 depicts the LPC compressed Speech Signal. Fig. 6 shows the Original Recorded Speech Signal in different environment condition. Fig. 7 depicts the LPC compressed Speech Signal. Fig. 8 shows the Original Recorded Speech Signal in different environment condition. Fig. 9 depicts the LPC compressed Speech Signal.

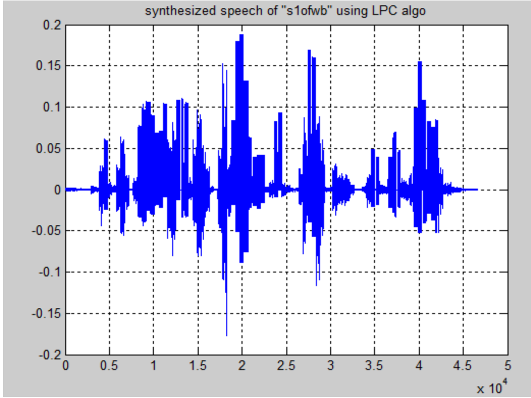

Fig. 5 LPC compressed Speech Signal

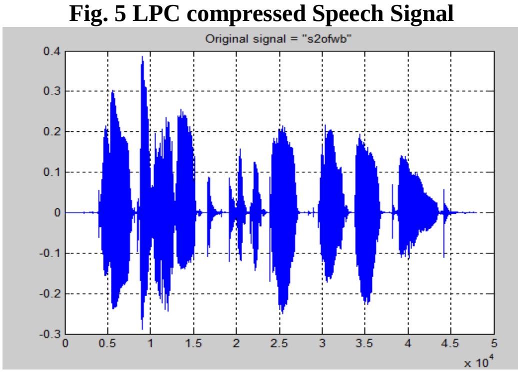

Fig. 6 Original Recorded Speech Signal

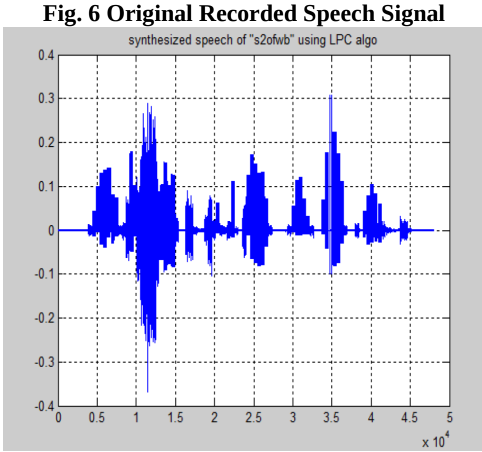

Fig. 7 LPC compressed Speech Signa

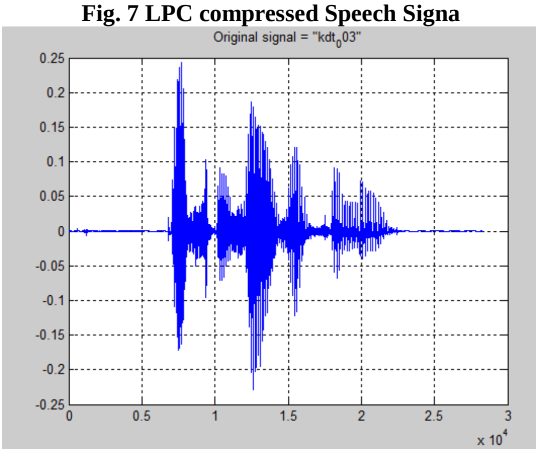

Fig. 8 Original Recorded Speech Signal

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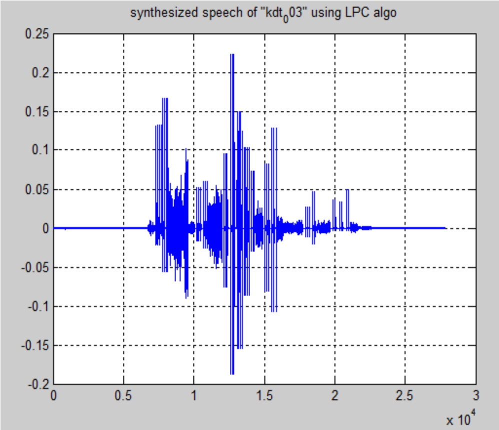

Fig. 9 LPC compressed Speech Signal

VI. CONCLUSIONS

In this paper, a method for compression of sounds signals was investigated. The method is based on linear predictive coding algorithm. The listening of the results of the sound signal and compression signal indicate that the signal is compressed to great extent with less distortion. Results indicate that the linear predictive coding compresses the signal to a great extent with small distortion of signal.

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