PREDICTIVE CODING BASED ON LEVINSON DURBIN ALGORITHM FOR FEATURE EXTRACTION OF RED PALM WEEVIL

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ABSTRACT

The aim of the project is to develop a system for encoding good quality sound at low bit rate. This could be accomplished by a technique called LPC. This report is a MATLAB implementation of an algorithm. LPC is one of the most powerful sound analysis technique and most useful method for encoding sound. The main objective of this work is to estimate the output sequence from a linear combination of input samples, past output samples or both. The coefficients extracted can further be matched by any of the several methods used for feature matching. This application is often used for sound identification or sound verification. In this article, work was done to extract feature of the weevil which was later used to determine its presence on palms.

Keywords: LPC, Vocoder, Levinson Durbin Algorithm

I. INTRODUCTION

LPC is a popular technique for sound compression and synthesis. Sound compression is usually conducted with the use of voice coder or vocoder. There are two types of vocoder. They are waveform based and model based vocoder [4]. The former will exactly reproduce the original sound signal if no quantization error occurs. The later will never reproduce the original sound, regardless of the quantization error. Because they are based on a model, which involves encoding and transmitting the parameters and not the signal. The LPC vocoder designed here is a model based coder.

All vocoder based on any algorithm they use, depend on four main attributes. They are bit rate, delay, complexity and quality. Bit rate determine the degree of compression that a vocoder can achieve. Any bit rate below 64 Kb/s is considered compression. LPC coder transmits sound at a bit rate of 2.4 Kb/s which is an excellent rate of compression. The next important attribute is the delay involved during transmission of an encoded signal. The general acceptable delay standard for transmission should be less than 300 ms. Next is the complexity of the algorithm used. Complexity surely affects cost and power of vocoder. LPC vocoder when subjected to high compression rate result in greater complexity. The final attribute for any vocoder is its quality. It is a subjective attribute and depends on how the listener accepts it. Since LPC vocoders achieve very low bit rate, they are poor in quality.

The general algorithm for LPC involves analysis or encoding part and synthesis or decoding part. In the encoding section LPC frames the signal and determines the input signal and coefficients of filter that will reproduce the input. This is quantized and transmitted. In the decoding part, LPC rebuilds the filter based on the coefficients. The decoder can determine the input signal that is sent to the filter for synthesis [4]. Figure 1 shows the sound activity of Red Palm Weevil.

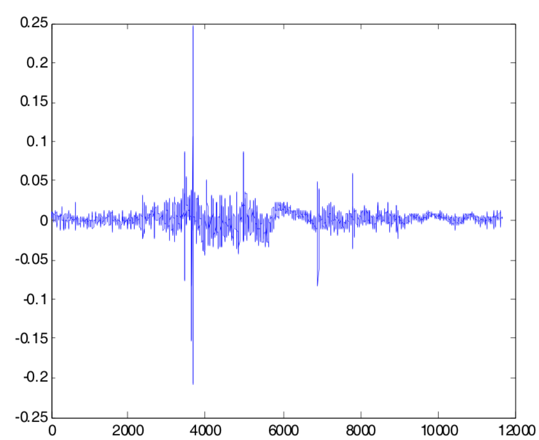

Fig. 1. Signal of Red Palm Weevil

The block diagram of the LPC vocoder is given in Figure 3. it consists of the transmitting and receiving section separated by the channel. All sound recognition system serves two distinguished phases. The first phase is the enrollment or training phase and the second one is the operation session or the testing phase. In the training phase, samples of sound are registered into the system to get trained. During the testing phase, input sound is matched with stored reference model and recognition decision is made. The block diagram explains the process flow of input signal to synthesized output.

The sequence can be described below.

- the frame size and overlapping size are computed.
- window function (Hamming) is applied to each frame as in figure 2.
- LPC filter coefficients, filter gain and error signal are generated.
- The above three signals are transmitted to the receiver.

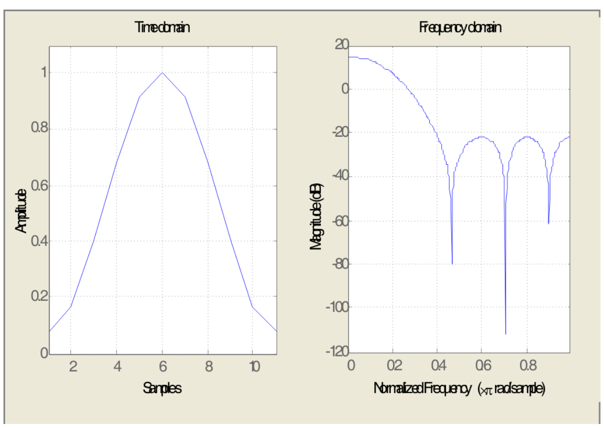

Fig. 2. Hamming window applied to the signal

- An inverse filter with LPC coefficients and filter gain is constructed.
- Apply error signal to inverse filter and synthesize the sound.

LPC Model

The linear predictive coding model has two key components They are Analysis or encoding and synthesis or decoding. The analysis part of LPC involves segmentation and is then examined further for what type of sound, what is the pitch and what parameters are to be considered to design the filter. This is transmitted onto the receiver. The receiver will then perform the synthesis based on the solutions obtained during analysis to perform reproduction of original signal.

II. COMPUTATIONS

The Filter Design

The filter used by the decoder during synthesis is built based on set of coefficients. These coefficients are extracted from original sound signal during encoding and are transmitted to the receiver for use in decoding. The parameters and number of parameters vary from segment to segment. A filter with n parameters is referred as nth order filter.

In order to find filter coefficients that best match the current segment during encoding. The encoder minimizes mean square error. The mean square error is the difference between the original signal and reconstructed signal. The value of mean square error should be as low as possible. The Mean square error is expressed as

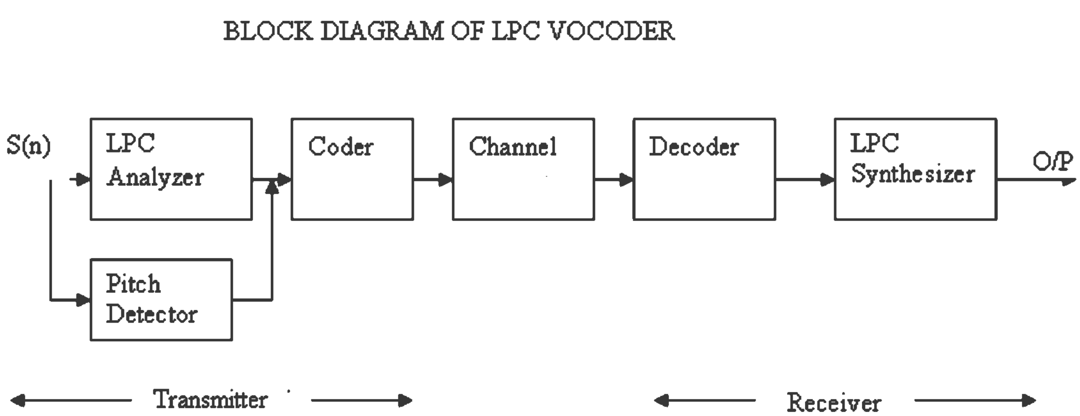

Fig. 3. Block Diagram of LPC Vocoder

$$e_n^2 = \left(y_n \sum_{i=1}^{M} a_1 \; Y_{n-i} + GE_n \right)^2 \qquad \text{...[1]}$$

$Yn =$ sample for current segment

$ai =$ set of coefficients

The analysis is focused in minimizing the average value of en2 for all samples in segment.

The first step in minimizing the average mean square error is to take the derivative.

$$\frac{\partial}{\partial\, aj} E\left[< y_n \sum_{i=1}^{M} a_i \, Y_{n-i} + GE\, n >^2 \right] = 0$$

$$= \Sigma_{i=1}^{M} \; a_i \, E\,[Y_{n-i}\, Y_{n-j}] = E\,[Y_n \, Y_{n-j}] \qquad \text{...[2]}$$

The derivative results in a set of M equations to solve for filter coefficients. $E\,[Yn - i\; Y_n - j]$ has to be estimated. There are two possibilities for this estimation. They are auto correlation and auto covariance.

Auto correlation approach is explained in this article with certain assumptions. Let Yn be current segment and stationery and Yn sequence is zero outside the current segment. In auto correlation, each $E\,[Yn - i\; Yn - j]$ coefficient estimate is converted into an auto correlation function of the form $Ryy\,(|i - j|)$. The estimation of autocorrelation function $Ryy\,(k)$ is now expressed as

$$R_{yy}\,(K) = \sum_{n = n_0 + 1 + k}^{n_o + N} \; Y_n \, Y_{n-k} \qquad \text{...[3]}$$

Using $Ryy\,(k)$ the set of M equation that were acquired from taking the derivative of the mean square error can be written in matrix form as $RA = P$. ...[4]

A contains filter coefficients

$$\begin{bmatrix} R_{YY}(0) & R_{YY}(1) & R_{YY}(2) & & R_{YY}(M-2) & R_{YY}(M-1) \\ R_{YY}(1) & R_{YY}(0) & R_{YY}(1) & \cdots & R_{YY}(M-3) & R_{YY}(M-2) \\ R_{YY}(2) & R_{YY}(1) & R_{YY}(0) & \cdots & R_{YY}(M-6) & R_{YY}(M-3) \\ \vdots & \vdots & \vdots & \cdots & \vdots & \vdots \\ \vdots & \vdots & \vdots & \cdots & \vdots & \vdots \\ R_{YY}(M-1) & R_{YY}(M-2) & R_{YY}(M-3) & \cdots & R_{YY}(1) & R_{77}(0) \end{bmatrix}$$

$$A = \begin{bmatrix} a_1 \\ a_2 \\ a_3 \\ \vdots \\ \vdots \\ a_M \end{bmatrix} \quad P = \begin{bmatrix} R_{77}(1) \\ R_{YY}(2) \\ R_{YY}(3) \\ \vdots \\ \vdots \\ R_{YY}(M) \end{bmatrix} \qquad \text{...[4]}$$

To determine the filter coefficients A, the equation stands as. Now has to be computed. This is an easy task since R is symmetric and interestingly all diagonals consists of same element. This type of matrix is called Toeplitz matrix and can be easily inverted.

The Levinson durbin algorithm is a recursive algorithm and is very efficient, since calculation of R^{-1} has its own advantage when determining filter coefficients. When the algorithm is applied to mth order filter, it computes all filters of order less than M. Hence for n filters, it determines $N = 1, \ldots M - 1/$

The L_D algorithm is given by

1. Set $E_0 = R_{YY}(0),\; r = 0$

While $(i < M)$

2. i++

3. Calculate

$$K_i = \left[\sum_{j=1}^{i=1} a_j\,(i-1)\; R_{YY}(i - j + 1) - R_{YY}(1) \right] / E_{i-1}$$

4. Set $a_j^{(1)} = K_i$

5. Calculate $a_j^{(1)} = a_j^{(i-1)} + k_i\, a_{i-j}^{(i-1)},\; vu = 1, \ldots, i - 1$

6. Calculate $E_i = (1 - k_1^2)\; E_{i-1})$

Levinson-Durbin $(L{-}D)$ Algorithm is used for solving Toeplitz Matrices.The algorithm is named in recognition of its use first by Levinson (1947) and Durbin (1960). This is a direct recursive method for solving the coefficient of prediction filter. It makes particular use of Toeplitz structure of matrix R.

Estimation of Pitch

- The min and max lag for the correlation is specified.
- The short term correlation of error signal with respect to max lag is found out.

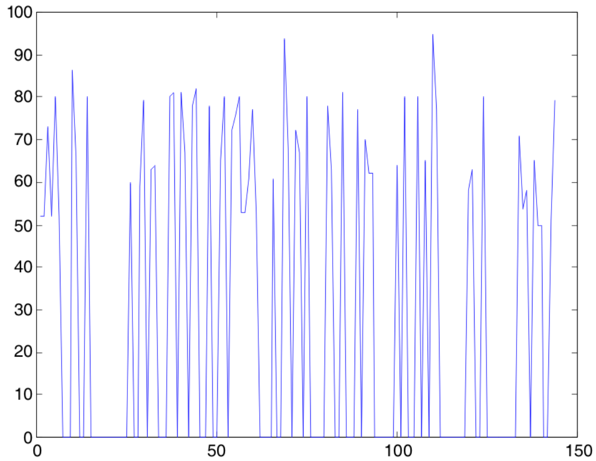

Fig. 4. Pitch of the signal

- Maximum auto correlation is computed.
- Coefficients greater than the threshold is checked.

The computed pitch value is used to calculate the filter gain.

Estimation of Filter Gain

For unvoiced frame, gain is determined by the square root of predictor energies. And for voiced frames it is the square root of the product of pitch and predictor error energies.

III. CONCLUSION

The LPC encoders sends information on each of the different segment to the decoder. The encoder then classifies the sound and the pitch period for every segment will create an excitement signal in decoder. When the excitement signal is given as input, it reproduces the original signal. This is completion of

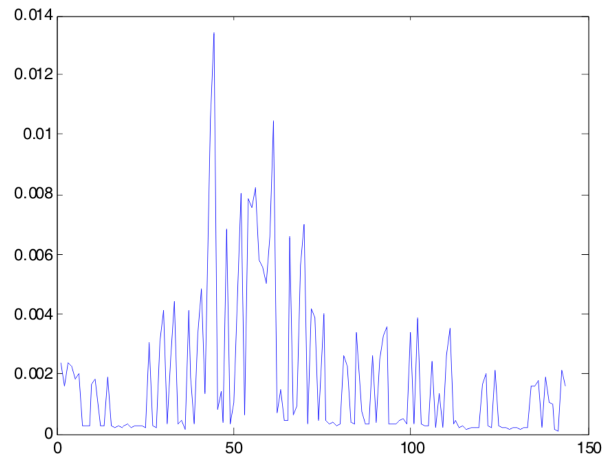

Fig. 5. Gain of the signal

extraction of feature input signal. The feature extracted is trained and then subjected for feature matching by any of the different methods to confirm its data existence in the codebook. The analysis done by simulation can be used to detect existence of any particular species of interest.

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