Master's Degree in Electrical and Computer Engineering
Computer Audio 2023/2024
Teacher: Diamantino Freitas

5th Laboratory Assignment

LPC coding: signal analysis and re-synthesis. Intonational manipulation of the artificial error signal

Objectives

Using the linear prediction method (LPC) and the Matlab environment, perform sliding coding of the audio signal in order to highlight the sliding LPC model and the error signal, depending on the order of the LPC model and the type of signal segment audio (voiced, unvoiced or mixed speech), also exploring the use of pre-emphasis.

Assess subjectively (timbre and periodicity) and objectively (spectrum and periodicity) the error signal and the degree of its whiteness.

Re-synthesize an excerpt of the signal lasting approximately 1 second in two modalities, the first with the original error signal to produce an exact replica of the original signal and the second with the original error signal replaced by an artificial error signal with identical temporal distribution of energy, consisting of a linear combination of constant frequency pulse train for monotonous intonation and white noise to produce voiced, mixed excitation and unvoiced segments.

Required material/equipment

PC desktop, Matlab, sinal fornecido e auscultadores.

Description of work to be performed

1- Analysis:

1.1 - For the “ASRF20.wav” file provided, write a script in Matlab to calculate the LPC coefficients (p=12), in a sliding manner, with windowed segments of fixed duration (Hamming, 30 ms, 2/3 overlap) and determination of the error signal. The signal must be previously re-sampled to fs=16kHz.

1.2 Appreciate the error signal obtained through critical listening and measuring its characteristics in a sliding manner. Explore the influence of p on the characteristics of the error signal as on slide 34 of the theoretical class notes.

Note 1: - Matlab's LPC function may initially be used, but the LPC algorithm itself (in the AUTO version) must also be developed based on the definition contained in the notes, to compare results; the information may eventually be complemented by Matlab help, available at: https://www.mathworks.com/help/signal/ref/lpc.html, for example.

Note 2: - Then consider a variant of the process with the prior use of a pre-emphasis filter on the signal (for example, 1st order and cutoff frequency of 50Hz, such as y[n]=x[n]-0.98*x[n-1], for 16 kHz) and compare results in terms of the LPC coefficients obtained and the error signal.

2- Re-Synthesis:

2.1 - Re-synthesize the signal in its original form with the original error signal and compare the original and re-synthesized waveforms. Appreciate the differences that may be caused by the use of pre-emphasis, followed by the corresponding de-emphasis at the end.

2.2 - Re-synthesize with a combination of a monotonous fundamental tone and a white noise excitation the parts of a segment approximately 1 second long depending on its type of excitation.

After annotating phonemes in PRAAT, perform LPC coding, as in 1 and, for the segments obtained in the segmentation resulting from the annotation, re-synthesize with a monotonous artificial error signal (constant frequency pulse train or white noise or combination of the two) with appropriate gain.

3 – Report: Make relevant comments, report including algorithms. The result will be a brief report, MATLAB programs with graphics and ".wav" files.

4 - <u>Annex</u>:

The autoregressive model used in the LPC coding/decoding of the signal allows the immediate consideration of these operations in the frequency domain with the corresponding analysis and synthesis filters, as shown below.

The s(n) production regression (FIR) is as follows:

$$s(n) = \sum_{k=1}^{p} a_k \, s(n-k) + e(n) \tag{1}$$

And the z-transform is:

$$S(z) = \left[\sum_{k=1}^{p} a_k \, z^{-k}\right] S(z) + E(z) = \widetilde{S(z)} + E(z) \tag{2}$$

That may be written as:

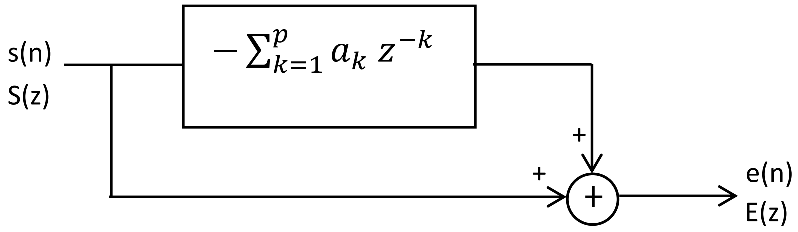
$$S(z)A(z) = E(z) \tag{3}$$

where:

$$A(z) = 1 - \sum_{k=1}^{p} a_k \, z^{-k} \tag{4}$$

and, we will thus have the analysis situation, carried out by the filter A(z), which for this reason is called analysis filter (of the FIR type).

The following diagram illustrates the LPC analysis operation:

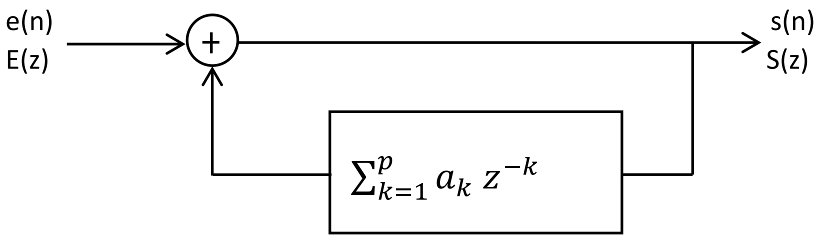


That implements the equation: $E(z) = \left[1 - \sum_{k=1}^{p} a_k\, z^{-k}\right] S(z)$ (5)

Now explaining S(z) as a function of E(z) in eq. (3), the following synthesis equation is obtained:

$$S(z) = \frac{E(z)}{A(z)} = \frac{E(z)}{1-\sum_{k=1}^{p} a_k\, z^{-k}} \quad (6)$$

where we can identify the 1/A(z) synthesis filter (type IIR). The following diagram illustrates the LPC synthesis operation:



where the synthesis filter will be: $\frac{1}{A(z)} = \frac{1}{1-\sum_{k=1}^{p} a_k\, z^{-k}}$ (7)

and LPC encoding/decoding can be viewed as filtering operation.

Coding corresponds to FIR filtering and decoding corresponds to IIR filtering.

If it is desired to use, instead of the original error signal, e(n), another normalized signal u(n) with unitary energy, a gain, G, will be used so that e(n) is replaced by G u(n) and it turns out that G2=Ep, the energy of the prediction error, from which the value of G can be calculated:

$$G^2 = R(0) + \sum_{k=1}^{p} a_k\, R(k) \quad (8)$$

Where R(0) and R(k) are the auto-correlation coefficients with lags 0 and k.