CCE5223   Speech Processing and Coding

Assignment – June 2012

Linear Prediction Coding – LPC Analysis and Synthesis

Build a Linear Predictive Coding System similar to LPC-10 based on speech sampled at 16 KHz.

Analysis

This requires the following components:
  (i)   separation of the input to frames. Use frames of 20 ms with an overlap of 10 ms
  (ii)   for each frame calculate a value for energy (note tha this can be R0 in the autocorrelation sequence)
  (iii)   build a voiced/unvoiced frame decision based on energy and zero crossing
  (iv)   build a pitch estimation based on autocorrelation techniques
  (v)   use a hamming window and preemphasis, estimate for each frame a 10-bit LPC using the autocorrelation method and the Durbin-Levinson algorithm.

So this should give for each frame a value for pitch (equals 0 for unvoiced),an energy value for R0 and ten LPC coefficients.

Synthesis

Hence build a synthesis system

The model is as Figure 7.24 (b) in Deller et al

 Build a random noise generator whose variance is unity. The output is then multiplied by the gain (energy) measured for that unvoiced frame during the analysis.

For voiced frames a series of unit impulses multiplied by an appropriate gain value for that voiced frame are used. The impulse positions depend on the pitch estimate for that frame, as well as the current pitch from the previous frame if voiced or starting at position zero if previous frame is unvoiced.
The all pole filter has the ten LPC coefficients calculated for that frame.
The duration of a frame, based on 16,000 samples per second rate and a duration of 25ms is (160 x 20) 320 time points. Each output frame must be overlapped with a previous frame depending on the overlap used, before obtaining the continuous output speech waveform.

References   Deller Proakis and Hansen, Chapters 1, 4,5 and 7

The original source is

Markel and Gray , Linear Prediction of Speech 1976

Use as speech source one of the files in the Annotation section of my web page eg go into FMM001.zip and then use fmm1m_2.wav.
You can use MATLAB or C or C++ for your software programmes.