ECE 537 **HW #5** **Fall 2006**

Univ. of Illinois **Due Mon., Oct. 9, 2007** **Prof. Allen**

Topics of this homework: History, Linear prediction of speech; Cepstral Analysis; STFT
Questions and corrections to: `jontalle @ uiuc.edu`.
The login and password to get the papers on the website is
login: students
password: study

To Do:

1. I have provided a short speech sound on the website (wav file). Write a program to read in the file, and do LPC analysis on it. I hope to provide this program, so you can concentrate on the analysis. Check the website for such a program (`readwav.m`).

   After carefully reading these instructions, if you are unsure of how to proceed (i.e., you don't program in Matlab well enough to proceed), then meet me for office hours, and I will give you a walk-through on how to proceed. I would prefer to meet in groups. I am also willing to set up a special lecture to discuss the Matlab aspects of this small project. I have a few people in mind, base on the first home works. I want to help you.

   (a) loop through the speech 5 [ms] and a time, taking 20 [ms] segments of speech. For each 20 [ms] frame, at each of the 5 [ms] *frame boundaries*, find the LPC coef. vector $A(n) \equiv a_k(n)$ where n is the frame index. Make the order of the analysis 12. Use Matlab's `lpc()` command. Note that the A vector has the form $[1, a_1 a_2 \cdots a_K]$. Use $A(n)$ based on frames $[n, n-1, n-2, n-3]$ to process speech in frame n.

   Plot all the roots in of A as single points, in [kHz], as a function of time, in [ms] (i.e., the frame index n, with $t = nD$, where D is the number of samples corresponding to 5 [ms]). D does not need to be exactly 5 ms, but should be rounded to the nearest sample of the sampling period. That is, D is within $\pm 1/2F_s$ of 0.005 [s].

   (b) filter the speech through the LPC filter. To do this you will need to swap the $A(k)$ vector of coefficients every 5 [ms]. Do this with the command `filter(A,1,sk)`, where sk is a D sample speech vector. Be sure to save the state of the filter for each block. Plot the output of the time-varying filter operation. This should look like the error signal described in the Atal paper that I asked you to read Atal and Hanauer (1971).

2. History. Identify the following, with a sentence as to what they did and *when* (the year, approximately) they did it:

   (a) Homer Dudley

   (b) Harvey Fletcher

   (c) Claude Shannon

   (d) Nyquist

   (e) Bode

   (f) William Strutt

   (g) A.G. Bell

   (h) George Campbell

   (i) d'Alembert

   (j) Newton

   (k) Daniel Bernoulli

   (l) Bishnu Atal

References

Atal, B. and Hanauer, S. (**apr 1971**), "Speech Analysis and Synthesis by Linear Prediction of the Speech Wave," J. Acoust. Soc. Am. **50**(2(2)), 637–655.