ECE 537 **HW #5** **Fall Oct 17, 2018**

Univ. of Illinois **Due 1 week** **Prof. Allen**

Topics of this homework: History, Linear prediction of speech; Cepstral Analysis; STFT
Questions and corrections to: `jontalle @ illinois.edu`.

To Do:

1. Write a program to read in the file **WhenAllElse.wav** (use Matlab's wavread.m function), and do LPC analysis (matlab's lpc.m program) on it. You will find wav files at: `http://hear.ai.uiuc.edu/ECE537/Assignments/files/`

   (a) Loop through the speech in frames of 5 [ms], taking a total of a 20 [ms] segment. That is, form $A(n)$ based on the four 5 [ms] previous frames $[n, n-1, n-2, n-3]$, thus process 20 [ms] of speech for frame index n. For each 20 [ms] frame, at each of the 5 [ms] *frame boundaries*, find the LPC "coefs vector" $A(n) \equiv [a_k(n)]$ where n is the frame index. Make the order of the analysis $K = 12$. Use Matlab's `lpc()` command. Note that the A vector has the form $[1, a_1, a_2, \cdots a_K]$.

   Plot all the roots 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]). You will need to use Matlab's `root()` command to find these roots, and then remove the negative (i.e., redundant lower half z plane) roots. Note the sample period 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).

   Graded based on the quality of the your report.

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.