Speech Signal Processing

Assignment 4 Course Code **ECE448**

Max. points **20**

Note:

- Always cite your sources (be it images, papers or existing libraries). Follow proper citation guidelines
- Unless specifically permitted, collaborations are not allowed.
- Do not copy or plagiarise, if you're caught for plagiarism or copying, penalties are much higher (including an **F** grade in the course) than simply omitting that question.
- Need to mention clearly if any assumptions are being considered.
- No late submissions are accepted.

Syntax to be followed for submission

- A single zip folder has to be uploaded in the moodle, which should contain the snapshots of your Numericals as *ECE448_A4_ < RollNo. > .pdf* and computer based questions (code) should be placed in a folder and named it as *ECE448_A4_cbq*
- For computer based questions you are expected to submit Codes (**Matlab**)

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1. Load a speech signal of your interest and select voiced region then Find pitch using LP analysis? Plot the results (Speech, LP-residual, autocorrelation of LP-residual) **Note: Computer Based Question [5 pts]**

2. For this Question you will use the code in the Matlab file: lpc_try.m. Record a speech signal using your voice in wav format. You may use fs = 16000 as sampling frequency during your recording. **Note: Computer Based Question (MATLAB) [15 pts]**

   More specifically:

   (a) Analysis-Synthesis based on Linear Prediction

   Download the Matlab code lpc_try.m

   At this stage, the lines require your interventions are 44-47.

   - In 44, you are require to compute the autocorrelation function of a given speech frame defined in line 40. You may use the Matlab function **xcorr**.
   - In line 45 you must compute the autocorrelation coefficients using the Levinson recursion. In Matlab, there is a function **lpc** which can be used.
   - In line 46 you must compute the gain of the LP filter, as discussed in lectures.
   - In 47, you must compute the linear prediction error, by inverse filtering the speech signal through the estimated linear prediction filter. You may use the Matlab function **filter**.

Once done, you must be able to load the wav file, perform analysis and synthesis frame-by-frame. At the end you can save your computed speech signal (variable out which is the output from the above Matlab file. To save it, you may use the command **audiowrite**). Listen to the original and the processed speech signal using the Matlab command **soundsc** or **sound**.

Listen to output audio file and briefly write down your observations.

(b) Follow frame-by-frame the analysis procedure

In each analysis frame, compare the magnitude of the frequency response of the estimated linear prediction filter (using Matlab command **freqz**) with the magnitude of the Fourier Transform (Matlab command **fft**) of the speech frame.

- Plot and compare on the same figure (use **hold on** and **hold off** - use different colors) the two magnitude spectra. Do the above in both voiced and unvoiced areas.
- Change the order of the Linear Prediction towards both directions (by increasing and decreasing the initial value (line 24: 24(!)))

Save some interesting - according to you - plots and write down briefly your comments/observations.

Note: *This assignment is strictly based on MATLAB*