EE 679 Speech Processing
Computing Assignment 1: Signal synthesis based on source-filter model
Due: 20/9/2020

Note: You can use Python, Octave or Matlab/Scilab. Make a single zipped folder that includes a pdf (with the solution for each question including method, code fragments, plots and discussion) and sound files. You can also include your code file itself (e.g. .py or .ipynb).

1. Given the following specification for a single-formant resonator, obtain the transfer function of the filter H(z) from the relation between resonance frequency / bandwidth, and the pole angle / radius. Plot filter magnitude response (dB magnitude versus frequency) and impulse response.

   F1 (formant) = 900 Hz
   B1(bandwidth) = 200 Hz
   Fs (sampling freq) = 16 kHz

2. Excite the above resonator ("filter") with a periodic source excitation of F0 = 140 Hz. You can approximate the source signal by narrow-triangular pulse train. Compute the output of the source-filter system over the duration of 0.5 second using the difference equation implementation of the LTI system. Plot the time domain waveform over a few pitch periods so that you can observe waveform characteristics. Play out the 0.5 sec duration sound and comment on the sound quality.

3. Vary the parameters as indicated below and comment on the differences in waveform and sound quality for the different parameter combinations.

(a) F0 = 120 Hz, F1 = 300 Hz, B1 = 100 Hz
(b) F0 = 120 Hz, F1=1200 Hz, B1 = 200 Hz
(c) F0 = 180 Hz, F1 = 300 Hz, B1 = 100 Hz

4. In place of the simple single-resonance signal, synthesize the following more realistic vowel sounds at two distinct pitches (F0 = 120 Hz, F0 = 220 Hz). Keep the bandwidths constant at 100 Hz for all formants. Duration of sound: 0.5 sec

   Vowel F1, F2, F3
   /a/ 730, 1090, 2440
   /i/ 270, 2290, 3010
   /u/ 300, 870, 2240

(Optional: Use glottal pulse shaping and lip radiation filtering. Add a small amount of aspiration noise and pitch jitter.)