PART A

1) Record an audio file with your first name, at a sampling rate of 8000 Hz.
For example, you can use the commands audiorecorder (), recordblocking (), and getaudiodata () in the MATLAB environment in conjunction with your computer microphone. Alternatively, you can use any other program for recording (and sampling at 8000 Hz, eg the Audacity program). Then import the recorded audio file into MATLAB using commands such as wavread (). Try not to make the audio file longer than 2-3 sec, so that the next editing steps do not take much time.

2) Plot the sound file you just recorded in the time field

3) Calculate and graph the signal energy inside a 100msec rolling window and with 50% overlap between adjacent windows.

4) Select a 100msec window from a portion of the signal that is (approximately) periodic. Listen to the recorded signal from your computer speaker using, for example, the audioplayer () and play () commands. Which phoneme does it correspond to?

5) Apply the Discrete Fourier Transform (DFT) calculated on 1024 samples using the fft() command. Draw the frequency content (the measure of this) of the whole signal in normal and logarithmic scale, ie abs (fft (A)) and 20 * log10 (abs (fft (A))). Properly scale the horizontal axis of the diagrams to correspond to the actual signal frequencies (0–8000 Hz).

6) From the diagram you have drawn, calculate (supervising) the fundamental frequency of the signal. Do the same in the field of time (calculate the fundamental period). Confirm the relationship between fundamental period and fundamental frequency.

PART B

1) Consider now a LTI system with input / output ratio:

$$y[n] = x[n] \; + \; a\, x[n - n_o] \tag{1}$$

Plot the step s [n] and its percussion response h [n] (for $0 \leq n \leq 20$) for values a = 0.5 and n_o = 10 in the above difference equation (1), using the commands stepz () and impz () , respectively.

2) Draw the zero / pole diagram of the system for n_o = 10 and a = 0.1, a = 0.01, a = 0.001, using the zplane () command. Compare the zeros and poles of the diagram with the result of the roots () command in the polynomials of the numerator and denominator of the system transfer function (for some of the above values of a).

3) Now consider system (1) again with $a = 0.5$ and $n_o = 2000$, and apply it to the signal you recorded in A.4. Use the filter () command for this purpose or, if you prefer, write your own implementation.

4) Listen to the output signal (corresponding to step A.4) and plot the discrete Fourier transform (DFT) (corresponding to step A.5).

PART C

1) Implement an oscillator (second order system with a double coupled complex pole $z_1 = re^{j\Omega}$, $z_2 = re^{-j\Omega}$) with frequency [see book example (5.3.8) on page 281] with a function of the form y = resonator (x, resonator frequency, r, sampling frequency) where x is the input and y is the output of the system, with the resonator frequency (oscillation frequency) and sampling frequency (sampling frequency) being in Hz.

2) Calculate the impact response and frequency response of the system for your own parameter choices Ω and for $r = 0.95$ (for practical reasons, consider a finite duration of 1sec for the impact). Confirm that the maximum response is at the frequency you selected. What happens to the shock response and frequency response for $r = 0.5$ and $r = 1.2$?

3) Put 3 oscillators in series (with resonant frequencies 500, 1500, 2500 Hz, $r = 0.95$) and calculate the total frequency response of the system.

4) Calculate the discrete Fourier transform (DFT) of the output of the above system to input an impulse train with a period of 100 Hz and a duration of 200msec. Listen to this signal $x[n]$ and the 1st difference of $x[n] - x[n-1]$. Which phoneme does it correspond to?