FACULTY OF ENGINEERING, UNIVERSITY OF JAFFNA

DIGITAL SIGNAL PROCESSING – EC5010

LABORATORY SESSION 1
DIGITAL SIGNAL PROCESSING THEORY AND APPLICATION

PART1: SAMPLING, TIME DOMAIN & FREQUENCY DOMAIN REPRESENTATION

Refer PART 1 in the prelab report,

Generate the digital signal, x[n], obtained in your pre lab.

$$y[n]=0.2+x(n) \qquad (1)$$

Plot y[n] scaling the x-axis to ms.

1. Write a Matlab program to plot the frequency spectrum of y(n) using fft command. The plot should display only the spectrum up to half of the sampling frequency. Scale the x-axis to (a) Hz, (b) to radiant and (c) normalized scale.

2. What is the DC value of the signal?

3. Identify, if any aliased frequency components.

4. Plot the frequency spectrum of y(n) up to the sampling frequency. Could you see the frequency component, 2000Hz? Explain the reason. Describe how you could make that frequency component appeared in the signal (demonstrate your idea and show the spectrum with 2000Hz).

5. Generate another sinusoidal signal, p[n], with frequencies 900Hz and 1200Hz sampled at 4000Hz.

A digital signal q[n] is defined as

$$q[n] = \begin{cases} x[n] & 0 < n < 400 \\ p[n] & 400 \le n < 4000 \end{cases} \qquad (3)$$

Plot the signal q[n] and its spectrum with appropriate scaling on x-axis. Identify what is the important information which is missing in the magnitude spectrum of the signal.

Hint: you could plot another signal r[n] which is given by,

$$r[n] = p[n] + x[n] \quad 0 < n < 4000 \qquad (4)$$

How do you think you could incorporate that missing information (you will study this in a later chapter)

PART2: TIME DOMAIN PROCESSING OF AUDIO SIGNALS

Load the audio wave (.wav) file given in the site.

1. Identify the sampling frequency of the recording.
2. Write a matlab program to plot the time domain signal with ms in the -axis frequency spectrum of audio signal given using fft command. The plot should display only the spectrum up to half of the sampling frequency. Scale the x-axis to (a) Hz, (b) to radiant and (c) normalized scale.
3. Play the audio signal and find the samples separately for words (Hint: you can use the signal plot to identify the separation where).
4. Then segment the words from the sentence recorded in the previous step, play them one by one separately. (You must use Matlab software to process the acoustic signal).
5. Save them separately in .wav format.

PART3: CONVOLUTION AND FILTERING IN LTI SYSTEM

Refer PART3 in the prelab report

1. If a signal x(n) is passed through this filter calculate the output of the filter.
2. Using inbuilt function *'conv'* from Matlab to find the convolution output.
3. Then use the '*filter'* command to get the output and compare both of these outputs. Explain the reason difference?
4. Explore the 'initial' and 'final' conditions in the 'filter' command. What is the meaning or interpretation of them? Then combine your output with the 'final' condition.

5. Plot using subplot

   (i) The signal

   (ii) Output of convolution

   (iii) Output of applying filter command

   (iv) Output in part c.

6. Explain the differences in the various plots in previous figure.

7. Without using any inbuilt commands from Matlab (implement the convolution using for or while loop). You can use the impulse response obtained in your paper work.

   Here plot the outputs separately for each convolution steps.