Assignment 1

Digital Signals Analysis and Applications - IEC239
Deadline at 11:55pm on 20$^{\text{th}}$ January, ~~2017~~ 2018

- **All questions are compulsory. Follow the instructions carefully.**
- **All coding questions have to be done in MATLAB only.**
- **Ensure that submitted assignment is your original work. Please do not copy any part from any source including your friends, seniors and/or the internet. If any such attempt is caught then serious action will be taken.**
- **The submission - Roll_number.zip - should contain the below files in the directory roll_number with only 7 files : q1.m, q2.m, q4.m, q5.m, q6.m, q7.m, Report.pdf**
- **Report should contain details of algorithm implementation, results and observations and answers to the subjective questions(if any).**
- You are expected to use **vector operations** in all your matlab codes.

Problem 1

Ishan is developing a software for image processing tools. While making it he got into a problem of resizing the image and comes to you for help. You have to make a function **RESIZE** which enlarges an image by a given extent. **RESIZE** takes two arguments **A** (Image matrix) and **X** (times the original image should be resized . X is always a positive integer). It should return an image matrix B of the desired size.
Resizing can be done using interpolation. Try to implement the above using (1) Nearest Neighbour Interpolation and (2) Bilinear Interpolation.
Test your code on the following cases

1. X = 2
2. X = 5

and use the following images for testing

1. Make a 300x300 size image using black and white curves on any software. It should have at least 1 ellipse and 1 polygon.
2. A black and white image of your choice.
3. A colored image of your choice.

Compare the resized images in both the interpolation. Which is better? Can you identify any problems in using bilinear or nearest neighbour interpolations. Can you suggest any other method for it. Write a report on the given points. Also include your observations and results in the report. **(NOTE - Don't use Inbuilt resize method. You have to make your own function. X is a variable)**

Problem 2

Apply convolution on **cameraman.tif** image in matlab using the following filter:

$$M = \begin{bmatrix} -1 & -2 & -1 \\ 0 & 0 & 0 \\ 1 & 2 & 1 \end{bmatrix}$$

(a) What do you think, this filter is doing? Explain.

(b) Perform convolution using M^T. What's the difference now

Problem 3

An image of size (Width, Height, Channels) is convolved with **N** filters of size (**F**,**F**,Channels).The convolution is done with a step size of **S** units, and the input is also padded with a zero padding of **Z**. The convolution always happens in such a way that the filter is always contained in the image.

(a) Predict the dimensions of the output of this convolution.

(b) How many additions and multiplications are involved in this convolution?

Problem 4

(a) Read a wav file in MATLAB. It is originally sampled at 44.1kHz and has digitized at24 bits.Play the sound. Also, record your own voice for this question.

(b) Subsample to 24kHz, 16kHz, 8kHz and 4kHz. Play the sound.

(c) Simulate it in three different environments using the concepts and examples considered in the class. You can download impulse response characterizing the system (environment) from the internet (use freely available resources like http://www.openairlib.net/auralizationdb).

Problem 5

Create a matrix of size 3x3 which when convolved with input.png results in a white line where the white meets the black (sample_out.png). Now do the following and give report on what you see:

(a) Convolve blur.jpg with the matrix you designed previously.

(b) Convolve blur.jpg with the transpose of matrix you designed previously.

(c) Add the images obtained from (a) and (b) after convolution with blur.jpg.

(d) Take any other 3 images instead of blur.jpg and follow steps (a),(b) and (c).

Problem 6

Write a code in Matlab to spot the face given in **F1.jpg** inside the image **Faces.jpg**.

Problem 7

Convolution filters on 1-D audio signals are used to achieve certain sound effects in sound processors for orchestras and musical performances. Figuring out the impulse response of a particular instrument is an experimental problem, since these impulse responses are often infinite in nature and cannot be calculated using traditional deconvolutional methods.

Consider the following example problem where you are given an input signal and an output signal. The output has been generated by convolving the given input signal with a finite impulse response filter. Assuming the filter to be **causal** in nature, find it.

The convolution applied is *valid*. Only values calculated without zero padding are included in the result. Your code should output the filter coefficients rounded to the nearest integer. (*Nearest* integer, neither rounded up nor down).

$$Input = [12, 20, 3, 10, 22, 19, 23, 16, 0, 21, 23, 16, 18]$$
$$Output = [75, 52, 33, 97, 251, 211, 63, 65];$$