Signals and Systems Lab

EL-223

Semester Project

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Instructions

1- This is a group project and maximum 2 Students/group are allowed. Students are not allowed to make group with other section students.
2- Choose one topic and submit topic name along with group member details on SLATE till **30th March, 2019 11:55pm.**
3- **Total Marks = 10 absolutes** (Mid Demo marks=4, Final Demo marks=6)
4- There will be two demos for this project. Mid demo will be conducted during **3rd Week of April.** You must need to complete code (without GUI) till mid demo.
5- It requires SLATE submission as well as printed copy of the report. Project hard copy should be submitted to your lab instructor at the time of final demo.
6- **Final demos** will be conducted during **First week of May.**
7- All groups need to submit a project report covering below mentioned points:
   - Project Description
   - Complete Code
   - GUI and results screenshots

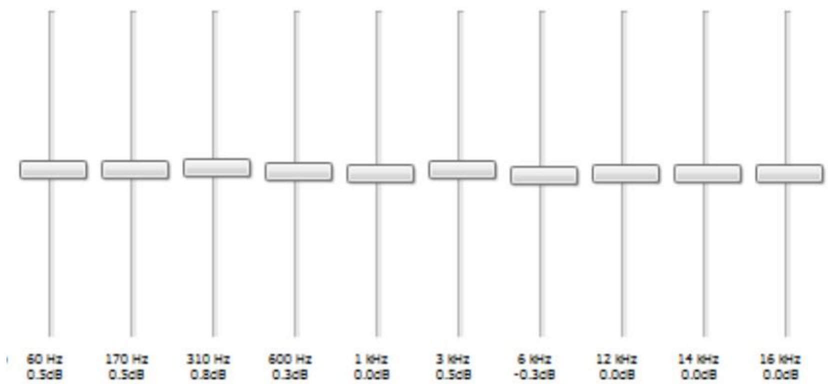
Note: Students who are willing to present their own project idea other than given projects, can submit project proposal till 30th March, 2019 11:55pm. Please note that you still have to choose one project from the given projects just in case your project proposal got rejected, the chosen project will be assigned to you.

Project 1 - Audio Equalizer

Audio equalizer is a software/hardware tool to realize a multiband frequency selective filter to manipulate the audio at certain frequencies. Common practice is to vary the gains of the filter bands to yield different audio in terms of perception.

Your task is to make a 10-band software audio equalizer with the following frequency bands: [0 – 60 Hz], [61 Hz – 170 Hz], [171 Hz – 310 Hz], [311 Hz – 600Hz], [601 Hz – 1 kHz], [1 kHz – 3 kHz], [3 kHz – 6 kHz], [6 kHz – 12 kHz], [12 kHz – 14 kHz], [14 kHz – 16 kHz].

The range of gains that should be available for all bands is [-20dB – 20dB]. A snapshot of an equalizer is shown below.



Your software should be able to read audio file (preferably music file) stored in computer and equalization should be applied in real-time.

1- You should make a GUI, in MATLAB, to play an audio file and apply equalizer.
2- The GUI should contain all necessary options/buttons/selections to play file and sliders to change filter band gains.
3- A separate plot in GUI showing the frequency response of a multiband equalizer based on the values of sliders should be the part of GUI also.
4- The upper cutoff frequency of each slider (band) and the range of gain should be visible on the horizontal and vertical axis respectively.

Deliverables

1- A software with GUI of an audio equalizer.
2- A hardcopy report containing problem, significance and the methodology used. Explain all steps in methodology. Paste your source code, well commented, in the appendix of the report.

Project 2 - DTMF (Dual Tone Multi Frequency) Telephone Encoder/Decoder

he DTMF stands for 'Dual Tone Multi-frequency' which is one of the techniques for converting the analogue signal to digital using DTMF decoder. The DTMF decoder circuit mostly used in mobile communications system which recognizes the sequence of DTMF tones from the standard keypad of the mobile phone. When you press the buttons on the keypad, a connection is made that generates two tones at the same time. A "Row" tone and a "Column" tone. These two tones identify the key you pressed to any equipment you are controlling. If the keypad is on your phone, the telephone company's "Central Office" equipment knows what numbers you are dialing by these tones, and will switch your call accordingly. If you are using a DTMF keypad to remotely control equipment, the tones can identify what unit you want to control, as well as which unique function you want it to perform.

When you press the digit 1 on the keypad, you generate the tones 1209 Hz and 697 Hz. Pressing the digit 2 will generate the tones 1336 Hz and 697 Hz. Sure, the tone 697 is the same for both digits, but it takes two tones to make a digit and the decoding equipment knows the difference between the 1209 Hz that would complete the digit 1, and a 1336 Hz that completes a digit 2.

1	2	3	697 Hz
4	5	6	770 Hz
7	8	9	852 Hz
*	0	#	941 Hz
1209 Hz	1336 Hz	1477 Hz	

Figure 1

DTMF Using MATLAB:

Your GUI May look like as shown in Fig. 2.

1- For each pressed key, you need to generate DTMF signal. Frequencies for each key pressed are given in Figure 1.
2- Identify low and high frequencies signals from DTMF Input, and plot in Filtered Low and High freq. signals plot.
3- You need to compute FFT for both low and high freq. signals and plot in you GUI.
4- The key detected should be displayed on your GUI.
5- Reset button will clear all the inputs/graphs.
6- Close button will close the GUI.

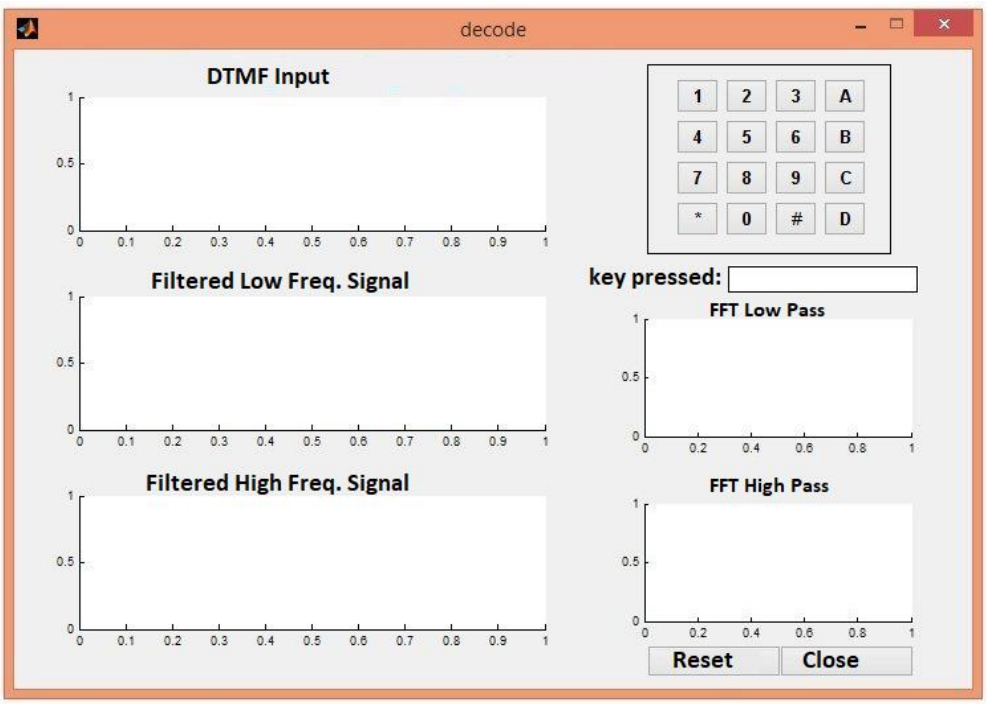


Figure 2

Project 3- Images manipulation:

In this project, you are supposed to transmit an MRI image of a patient suspecting a tumor living in rural area who wants to communicate with his doctor in a Big City. At the transmitter end, the data must be compressed (resize the image by discarding few samples of image such that the overall quality of image remains same). At the receiver end, the image is expanded again in original size (Interpolate the image such that the quality of image remains unaffected)

At Doctor’s end, the image enhancement is needed so that MRI scan can be further evaluated for types of tumor detection. Sharp well defined tumors are supposed to be benign and smooth tumors with irregular boundaries are supposed to be malignant tumors.

For this you first have to convert images to frequency domain by applying Fourier transform and then have to apply high pass filters for edge detection and low pass filters for smooth shapes detection.

You need to design GUI in MATLAB for showing your final results.

Note: You can use any MRI image available on internet. You are not supposed to classify tumors, you just have to show the output images.

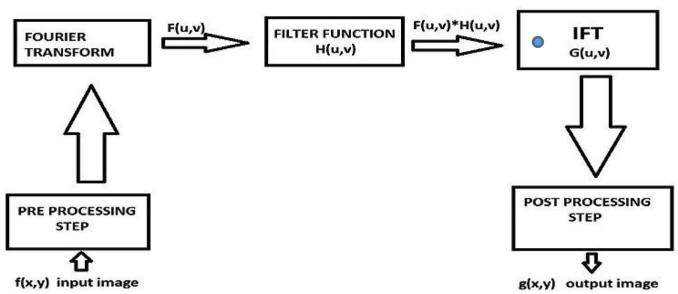
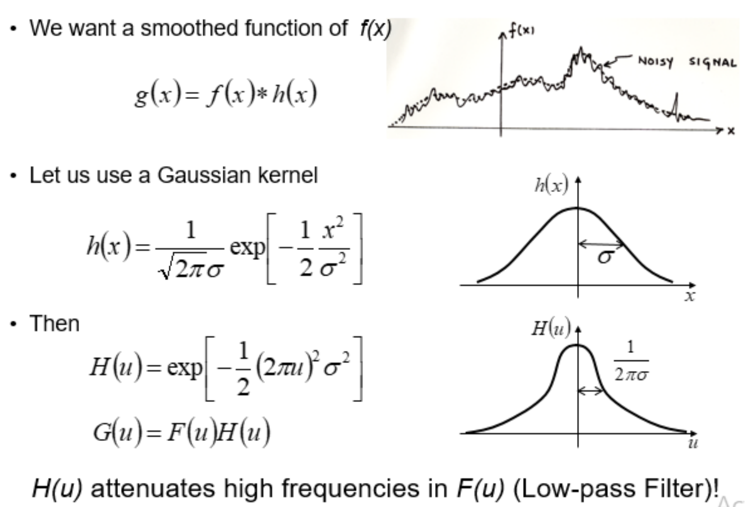


Figure 1: basic steps for filtering in frequency domain

For Smoothing or sharpening of a signal we can either use convolution in time domain or multiplication in frequency domain:

For example:



Please note that there will be a ringing effect while smoothing, show that as well.

Useful Material for studying:

http://www.giassa.net/?page_id=174 Downsampling /Upsampling of images

http://www.di.univr.it/documenti/OccorrenzaIns/matdid/matdid374175.pdf Image Enhancement in frequency domain

http://www.inf.tu-dresden.de/content/institutes/ki/is/VORLESUNGEN/BV_0708/ImageProcessing02.pdf for interpolation concept

http://biomedpharmajournal.org/vol7no2/image-sharpening-by-gaussian-and-butterworth-high-pass-filter/ Image sharpering

http://northstar-www.dartmouth.edu/doc/idl/html_6.2/Filtering_an_Imagehvr.html Image Filtering in spatial domain

Useful Commands for Image Processing:

```
I = imread('trees.jpg','jpg');
%Convert image to grayscale (intensity) values for simplicity (for now)
I = rgb2gray(I);

%Determine the dimensions of the source image
%Note that we will have three values - width, height, and the number
%of color vectors, 3
[j k] = size(I)
imshow(I)
```