2020-2021 Fall

EE-301 Term Project

Introduction:

- In this project, you are asked write a MATLAB script in order to analyze spectral content of a tone generated by a musical instrument and apply simple signal processing operations on that signal.

- In this project, you will analyze frequency components of discrete time signals in MATLAB via "fft" function. Therefore, you may refer to lecture on discrete Fourier transform (DFT), which is uploaded as project supplementary.

- Projects will be done by groups that contain at most 2 students. Although it is not recommended, you may do your project alone.

- You should submit your project report along with your MATLAB codes. You will not get any point from project, if you do not submit your MATLAB codes.

Project Description:

In this project, you are required to record single tone sound generated by a musical instrument. You may use either an actual instrument such as guitar, piano etc., or use virtual instruments via internet. You may use the following link to use an online piano,
URL : https://www.onlinepianist.com/virtual-piano

In order to record sounds, you should use a microphone and built in MATLAB functions, which you can find in;
https://www.mathworks.com/help/matlab/import_export/record-and-play-audio.html

As the first part of the project, you will record same tone generated from the instrument by using different sampling rates by using the "*record*" object of MATLAB. The recording duration should be 3 seconds. The sampling rates you should use for MATLAB recorder are $F_{s_1} = 44100\ Hz$, $F_{s_2} = 11025\ Hz$, $F_{s_3} = 4900\ Hz$ and $F_{s_4} = 2756\ Hz$, where the corresponding signals are denoted by $x_1[n], x_2[n], x_3[n]$, and $x_4[n]$, respectively.

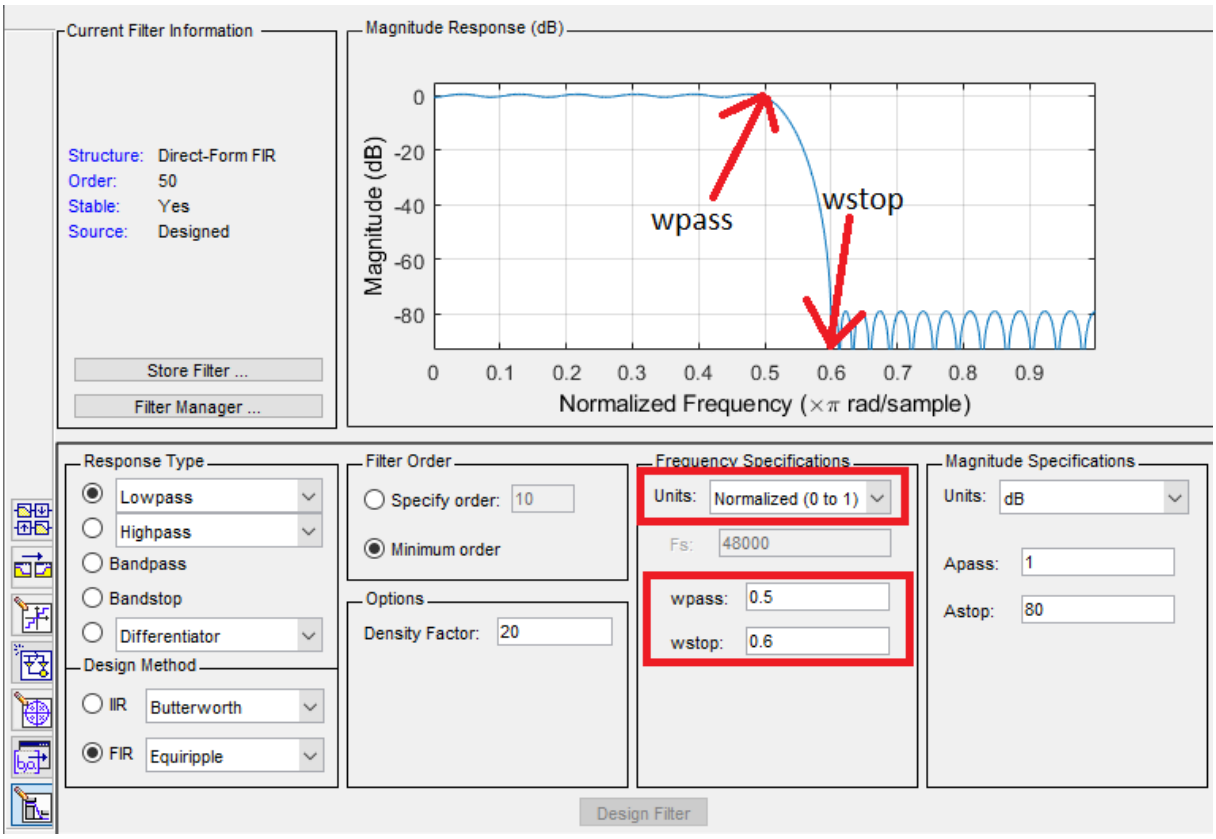
1) After making the recording, play the recorded signals in MATLAB using "*play*" function. Observe the effects of the sampling rate and provide comments.

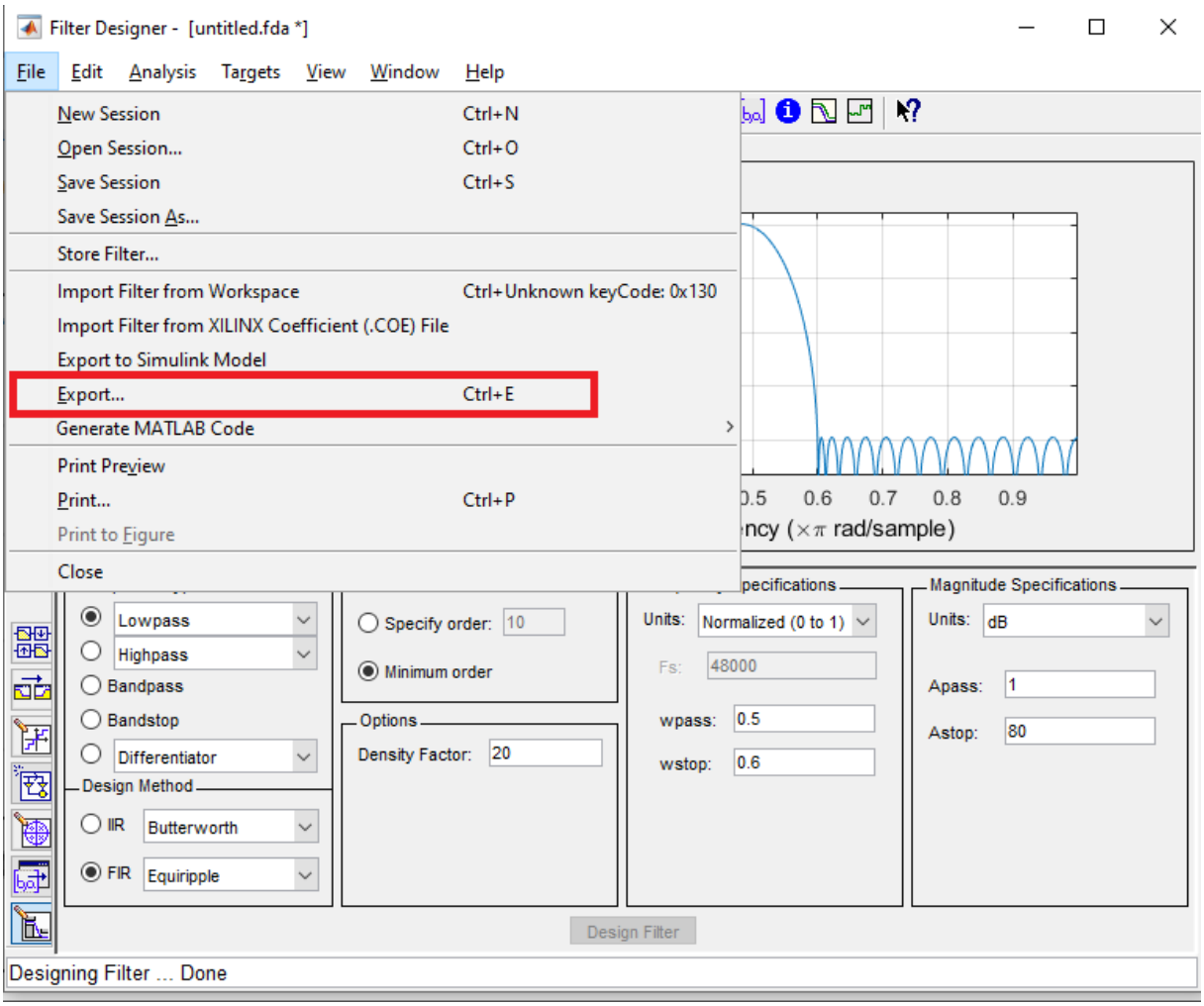
2) Plot the spectral contents of $x_1[n], x_2[n], x_3[n]$, and $x_4[n]$ on separate plots (4 plots are required) via taking the "DFT"s of those signals and comment on results. Note that you should use *"getaudiodata"* function to extract the recorded sound signal to your workspace. Make sure that you normalize each DFT by the length of the respective signals before plotting. Make also sure that you use *"fftshift"* function to center the spectrum around zero frequency. The horizontal axis of your plots should be in Hz.

As the second part of the project, you are going to investigate aliasing effects and apply anti-aliasing filtering to avoid aliasing. For this purpose, you will work on the signal with the highest sampling rate (with $F_{s_1} = 44100\,Hz$). Firstly, you will decrease the sampling rate by just discarding a portion signal, which is referred as downsampling. By using this approach, you will decrease sampling rate of the signal $x_1[n]$, which is captured with $F_{s_1} = 44100\,Hz$, to $F_{s_4} = 2756\,Hz$ where $\frac{F_{s_1}}{F_{s_4}} = 16$. This can be done simply as,

$$x_5[n] = x_1[16n].$$

In addition, you will obtain another signal $x_6[n]$, whose sampling rate is also F_{s_4}, but it is obtained from $x_1[n]$ through a different process. This time you will filter $x_1[n]$ with an anti-aliasing filter before downsampling. You can use "filterDesigner" filter design tool to design anti-aliasing filter. It sufficient to define normalized cut-off frequency (w_{pass} and w_{stop}) for your anti-aliasing filter. Then, you can export the filter coefficients, $h[n]$, and use it as the discrete time impulse response of the filter to convolve with $x_1[n]$.





3) Calculate the maximum possible values of w_{pass} and w_{stop} for a proper filtering that prevents aliasing. Take $w_{pass} = w_{stop} * 0.8$.

After convolving $x_1[n]$ with $h[n]$, perform downsampling operation,

$$x_6[n] = \hat{x}_1[16n],$$

where $\hat{x}_1[n]$ is the anti-aliasing filter output.

4) Plot and compare frequency responses of $x_4[n], x_5[n]$ and $x_6[n]$. Then by using built-in "sound" function of MATLAB, listen $x_4[n], x_5[n]$ and $x_6[n]$ signals for $F_{s_4} = 2756\ Hz$ and comment on the results. The horizontal axis of your plots should be in Hz.