EE386 Digital Signal Processing Lab	**Jul-Dec 2021**

Experiment : 7

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This lab experiment covers various practicality of Digital Signal Processing such as windowing function, different FIR filter designs, spectogram analysis, bode plot analysis. I have used python libraries such as numpy, pandas, scipy etc for demonstration of these techniques and methods. The code to my entire work in this lab experiment is here. And the input files and my output files can be viewed here.

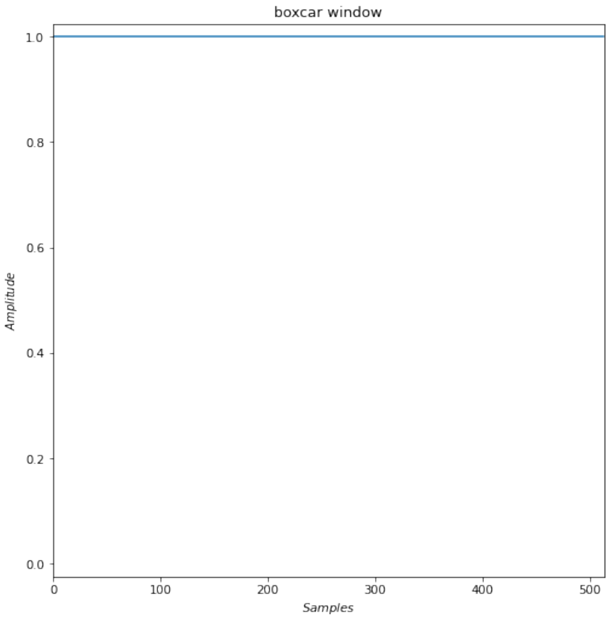
Please Note : I have used $\alpha = 2$ because my registration number is 191910.

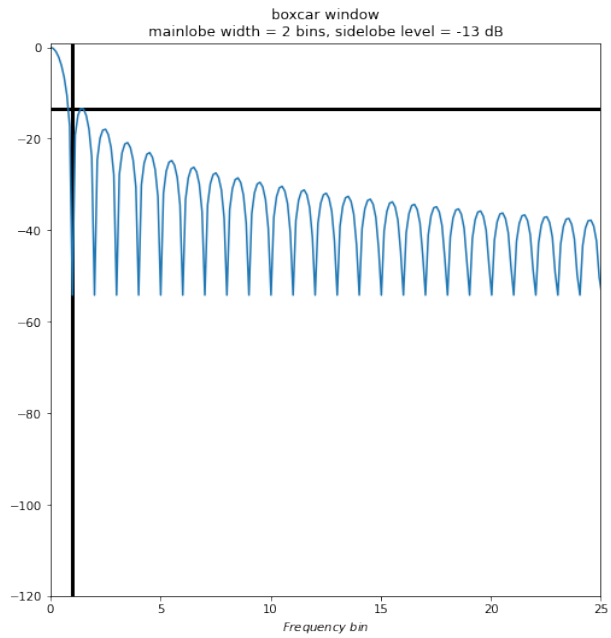
Question 1 - Window Functions

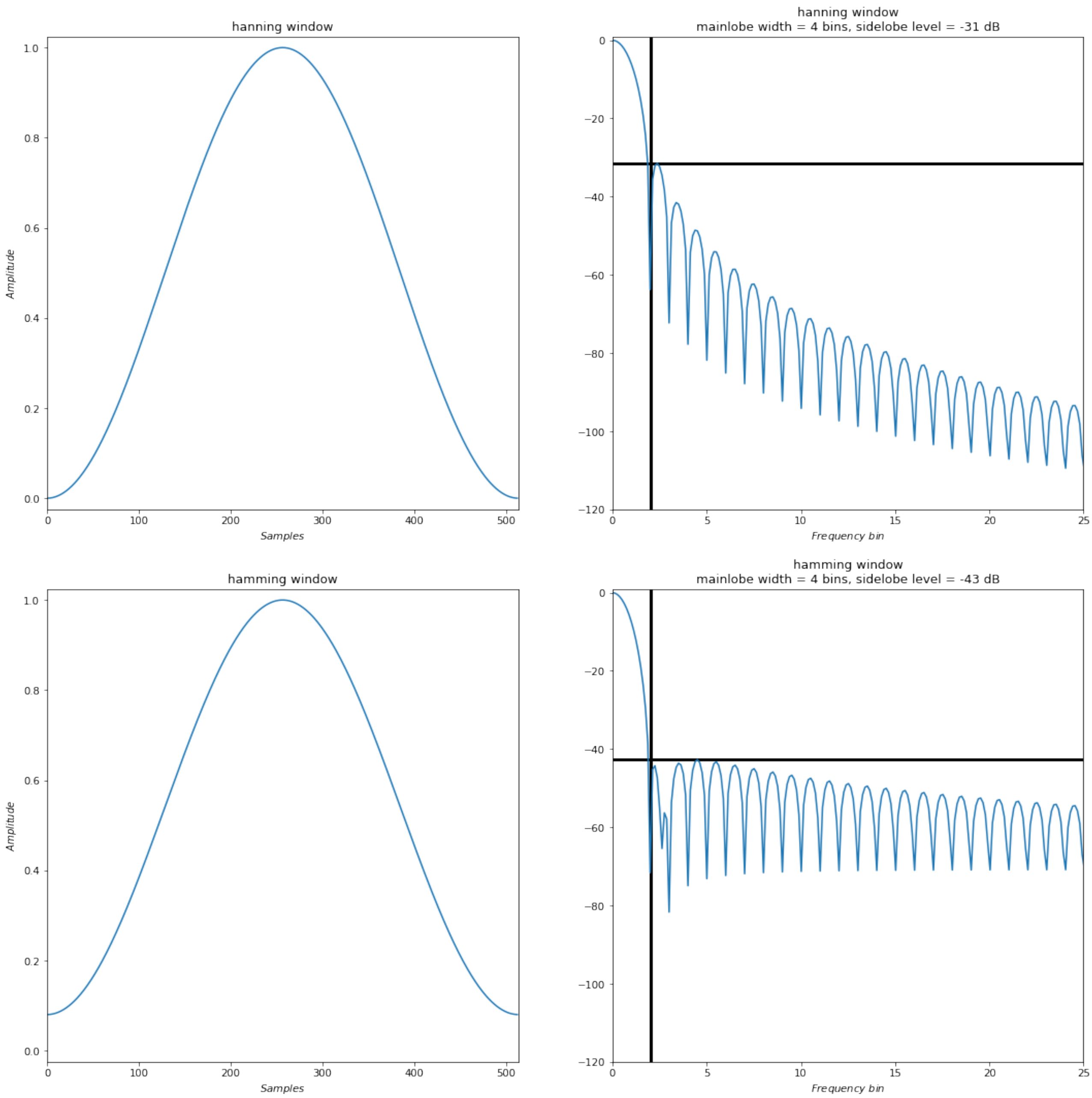
(Subproblem - 1)

In this question we are asked to explore different windowing techniques i.e

- Rectangular Window
- Hamming Window
- Hanning Window

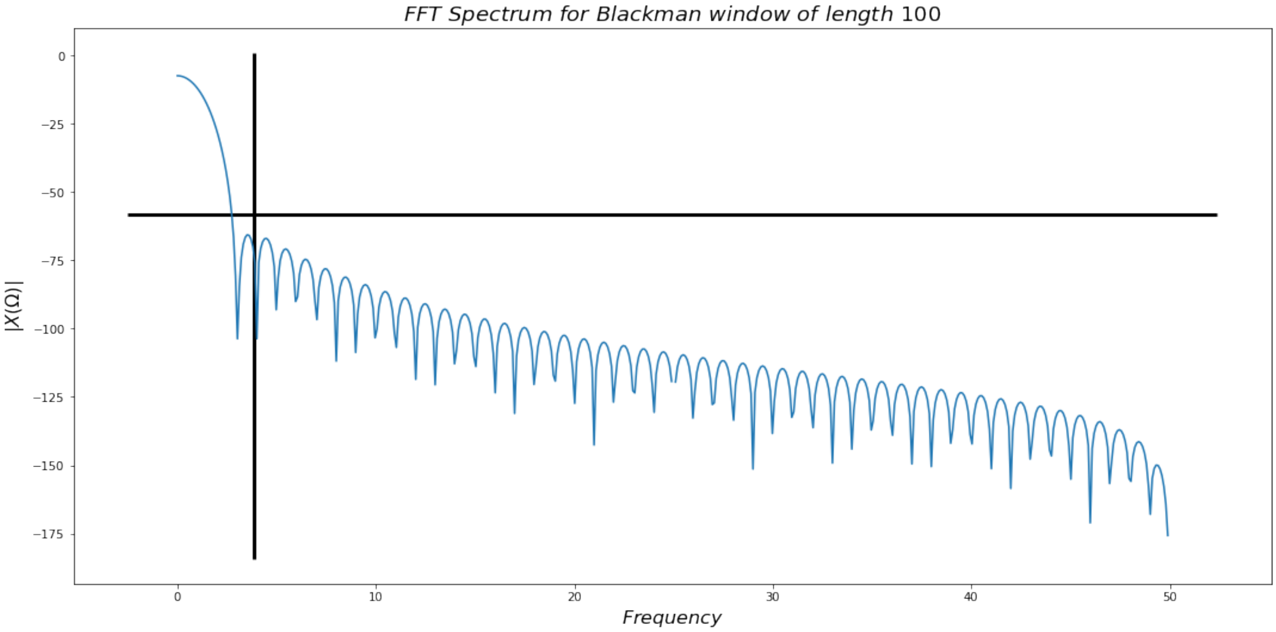


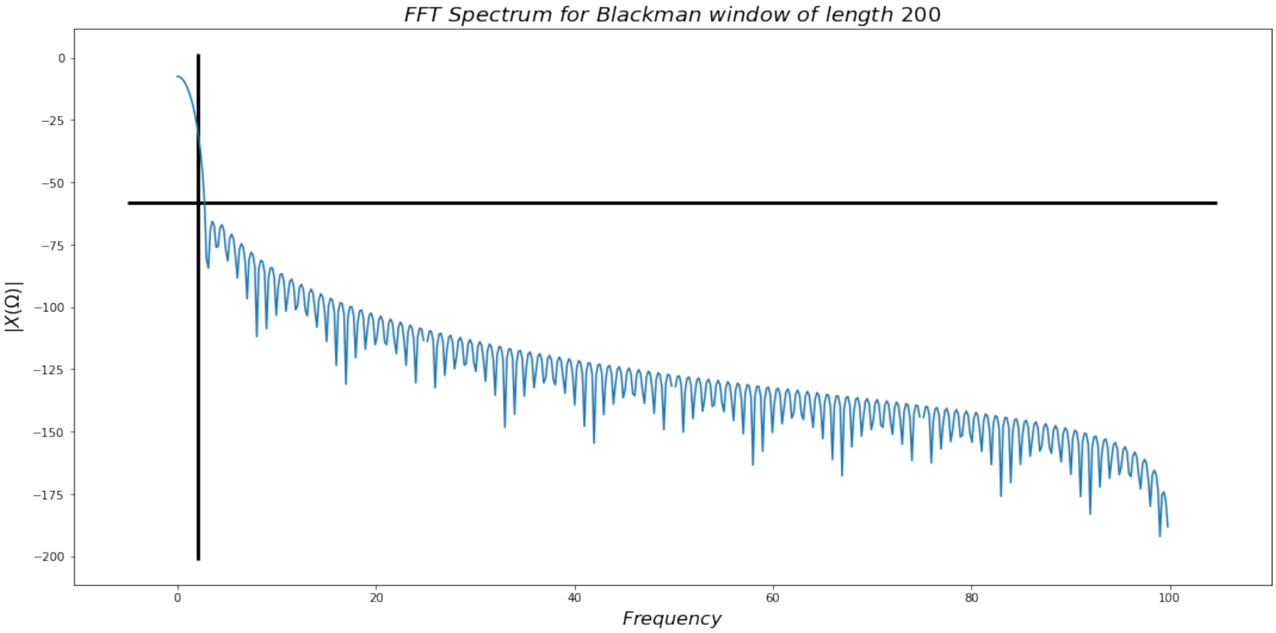


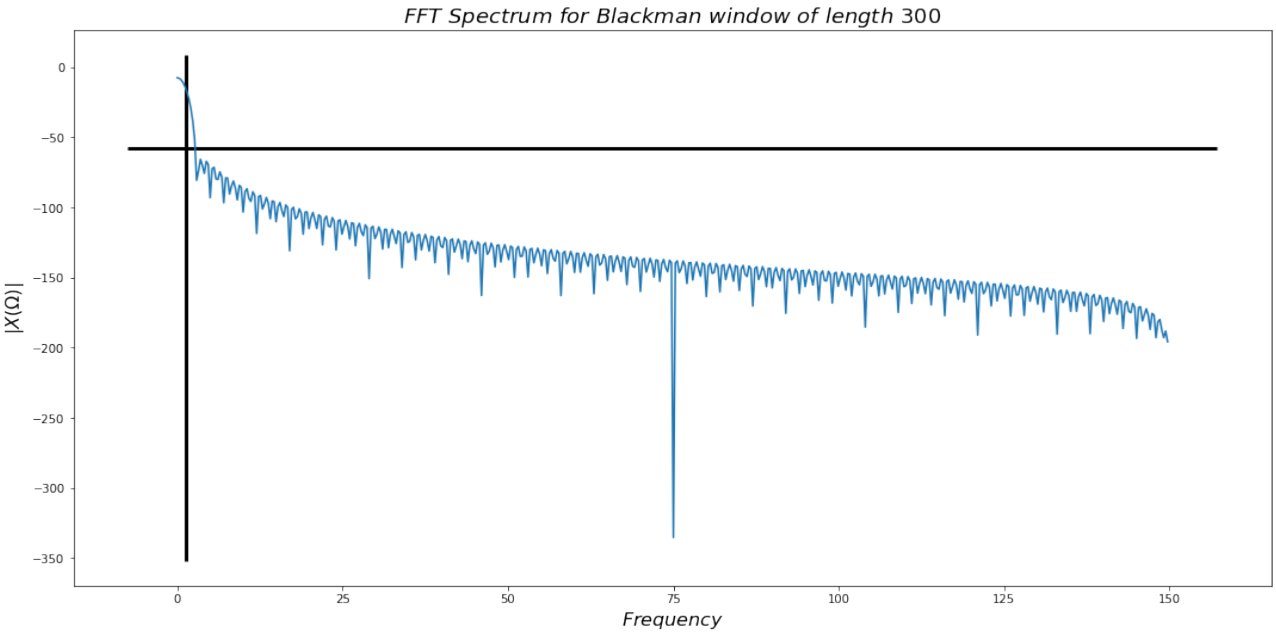


(Subproblem - 2)

In this question,we had to plot the spectrum of the Blackman window (Because = 2 Blackman window) signal for different lengths, using a 1024 point DFT and normalise the magnitude by the actual lengths (Say N = 100, 200, 300).


FFT Spectrum for Blackman window of length 100
|X(Ω)|
Frequency


FFT Spectrum for Blackman window of length 200
|X(Ω)|
Frequency


FFT Spectrum for Blackman window of length 300
|X(Ω)|
Frequency

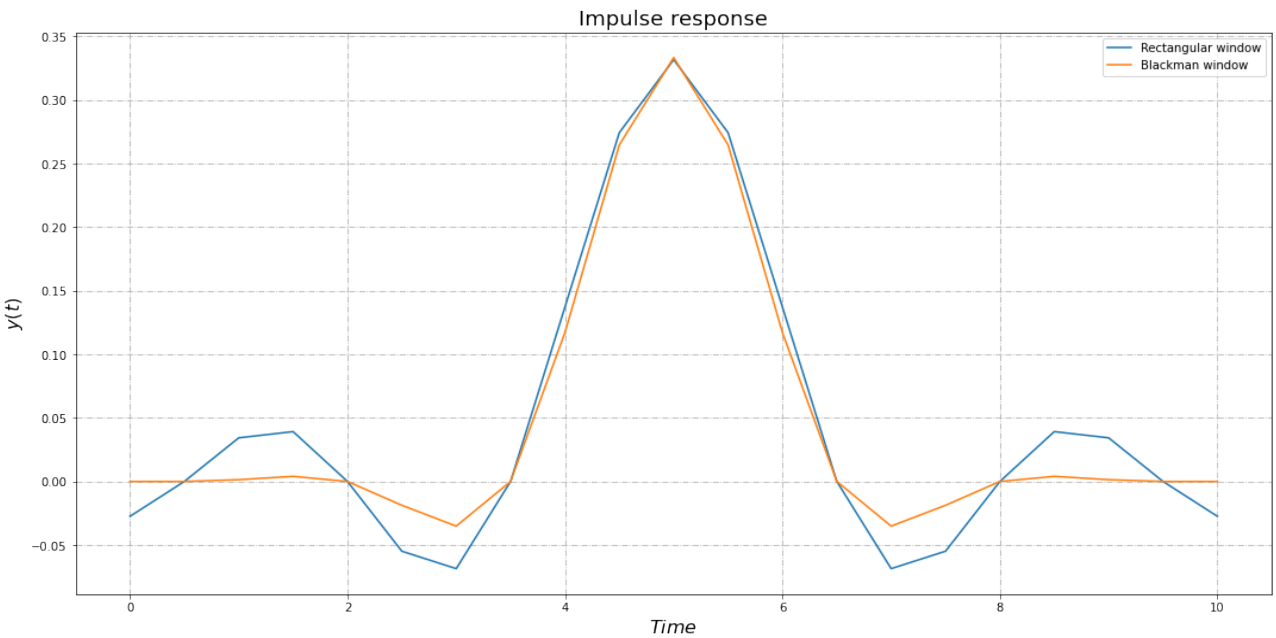
- Rectangular windows have the smallest main lobe size in comparison to the others,helpful in optimizing computational cost and have the highest minimum stopband attenuation.
- The hanning window is used to lessen bad effects on frequency characteristic produced by the final samples of a signal being filtered.It generally has a sharp fall, a higher minimum stopband attenuation and a decently narrow transition region. It is this characteristic of the hanning window which makes it extremely suitable for digital filter design.
- Hamming window has a higher minimum stopband attenuation while also having a wider transition region compared to hanning window.

Question 2 - FIR Filter Design

This question asks us to Design digital low pass FIR filter using rectangular and Blackman window functions for a cut-off frequency of $w_c = \pi/(\alpha + 1)$ rad/sample.

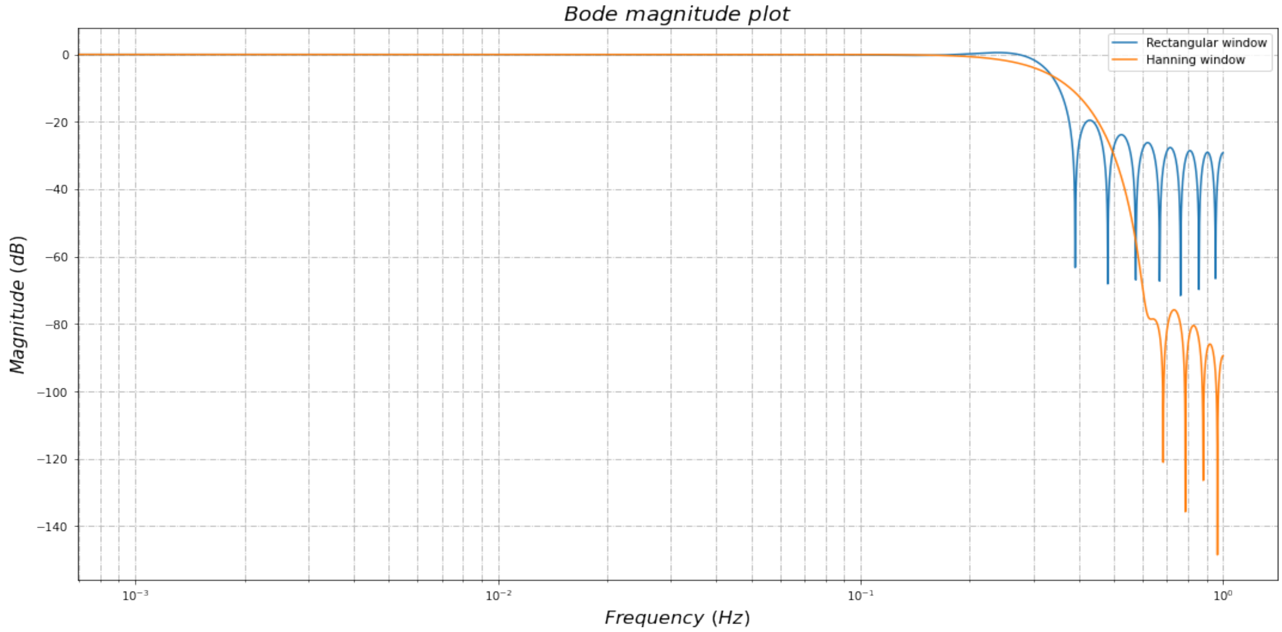
(Subproblem - 1)

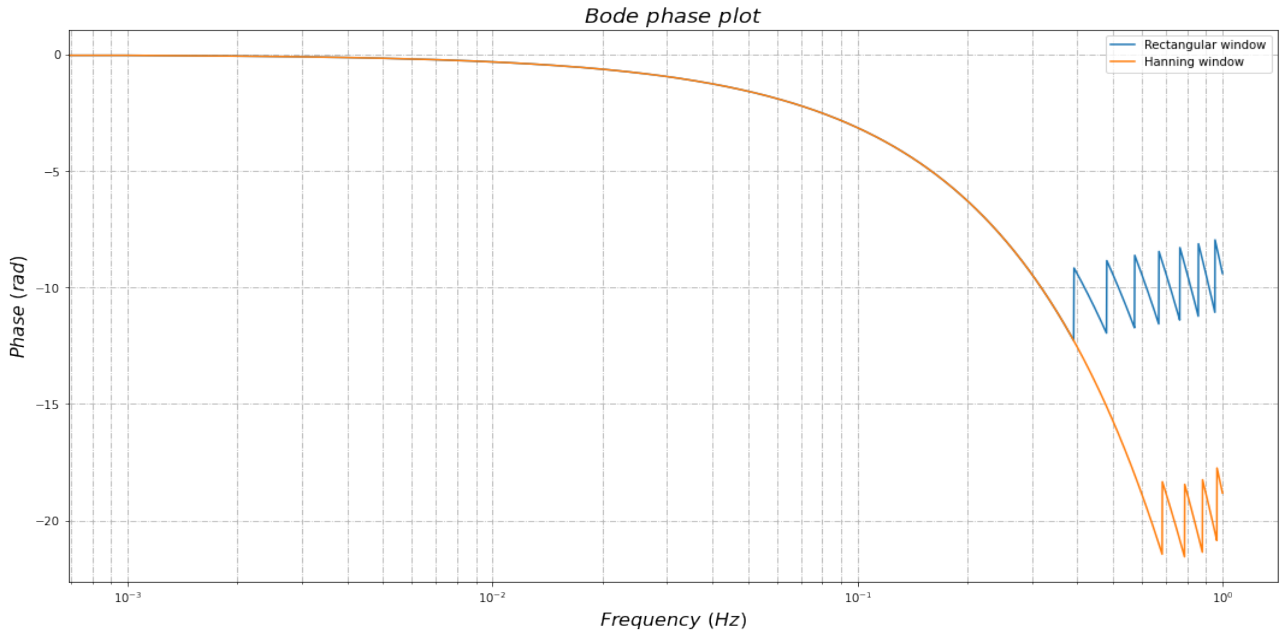
We have to plot the impulse response for this question.



(Subproblem - 2)

We need to perform bode plot analysis for this question.

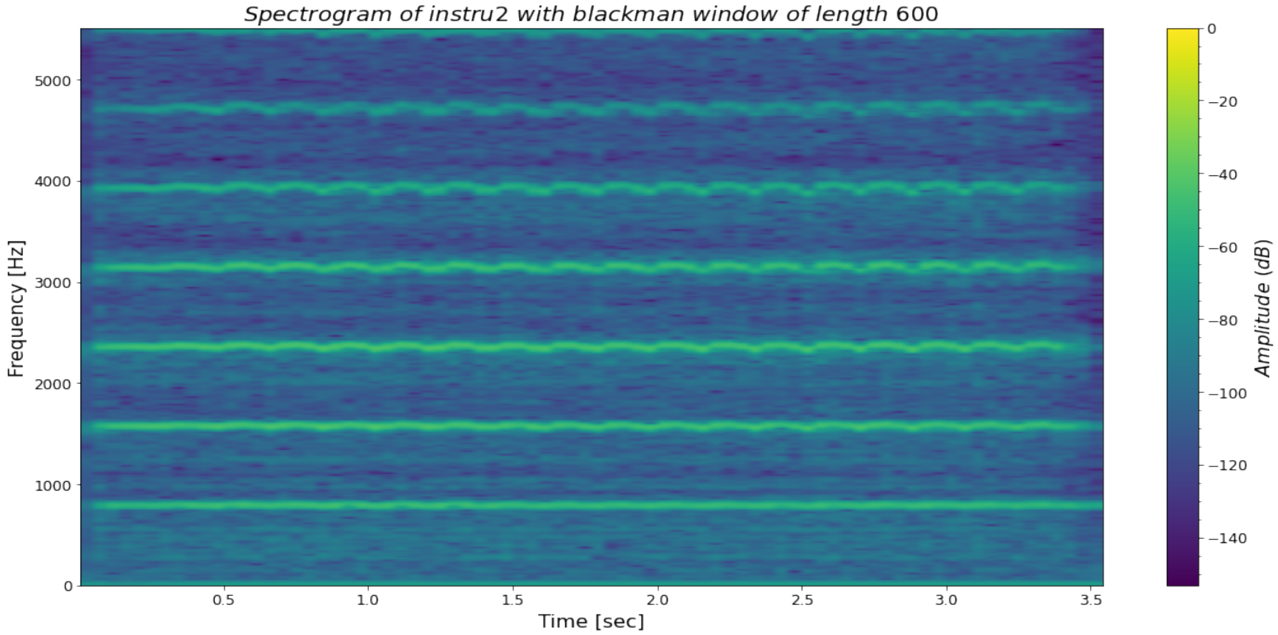


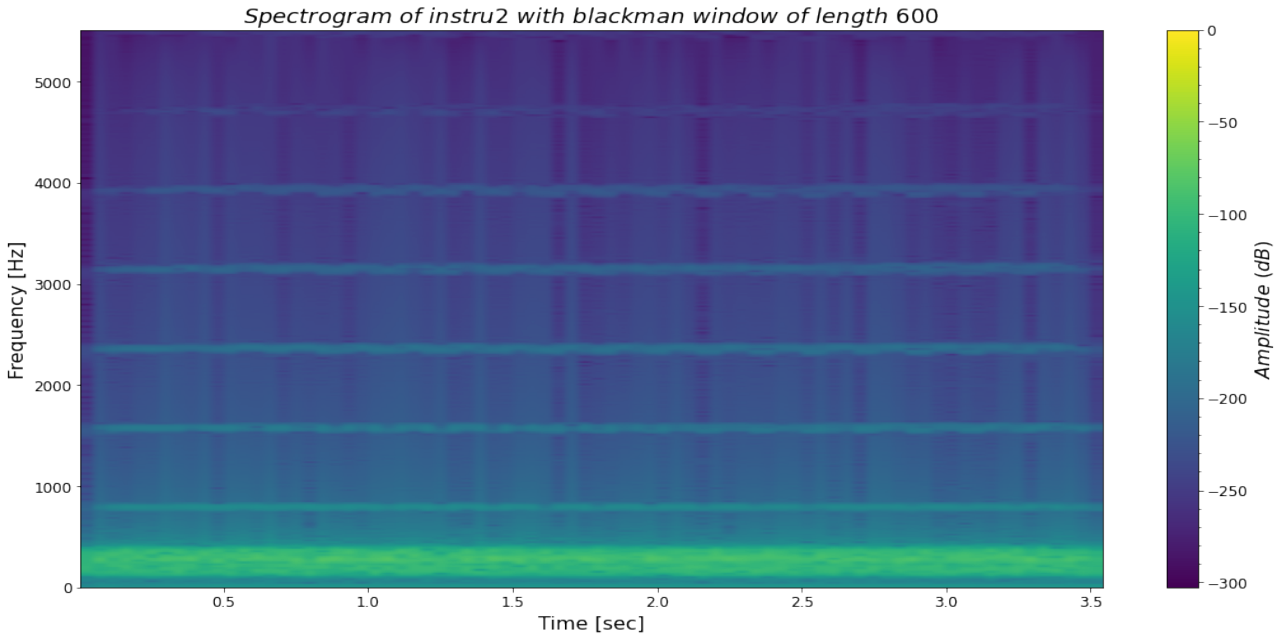


The Bode Plot of the both the filters is plotted using *freqz()* of the scipy library which takes the transfer function as the input and returns the frequency response. The magnitude and phase is plotted against frequency. The scales are logarithmic using *semilogx()*.

Question 3 - Filtering Using FIR Filters

This question asks us to plot the spectrogram of the instru2.wav using any window of our choice and sample duration for the window. And also design a digital FIR filter (using windowing) to only extract the fundamental (the first major peak excluding $\omega = 0$) and remove the rest including $\omega = 0$. Write it into a .wav file and listen. And also plot the spectrogram to ensure that you only have the fundamental.





Question 4

This question asks us to differentiate between the time-domain windowing and the window-based FIR filter design techniques.

The Windowing method for filter design uses samples of the impulse response for the filter coefficients, truncated to the filter length (and then possibly, but not required, tapered by a higher performing window).

The Frequency Sampling method for filter design uses the IDFT (Inverse Discrete Fourier Transform) of the desired frequency response for the filter coefficients, using the number of samples across the frequency response to be equal to the number of coefficients in the filter

he multiplication in the time domain is interpreted as convolution in the frequency domain, i.e., DTFT of window signal is convolved with that of the given signal, $X(\omega) * W(\omega)$. This results in shifted and scaled copies of $W(\omega)$ in the spectrum. Because of the structure of the magnitude spectrum of $W(\omega)$, the position of the mainlobes of the shifted $W(\omega)$ tell the frequency components. Similarly in FIR filter design, a shifted ideal filter impulse response is multiplied with a causal window in the time domain, which results in convolution in the frequency domain.

1 Appendix

- Note : I have used $\alpha = 2$ because my registration number is 191910. Since $\alpha = 1 + \text{mod}(910,3) = 2$
- The link to all the code is here. And the input files and my output files can be viewed
- The link to all input and output files are here.
- The Github repo to all code and previous experiments can be found here