SIGNAL PROCESSING

LAB – 6 REPORT

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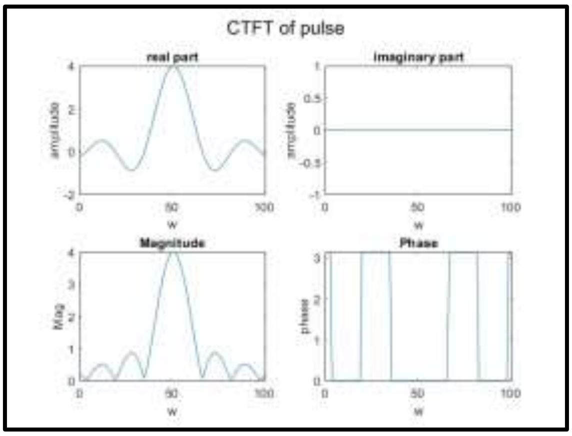
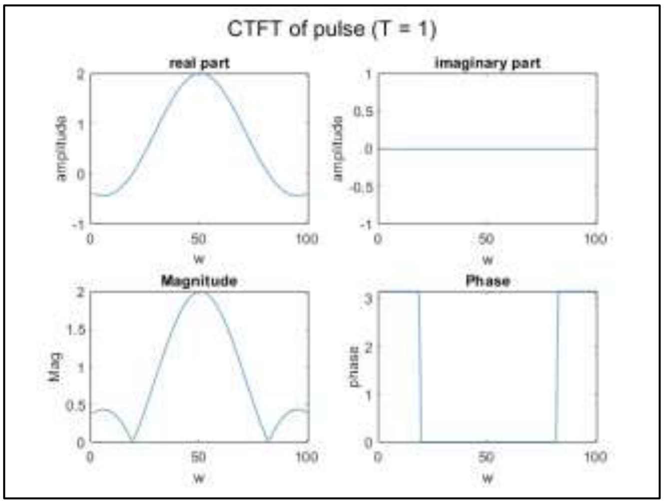
Question – 1

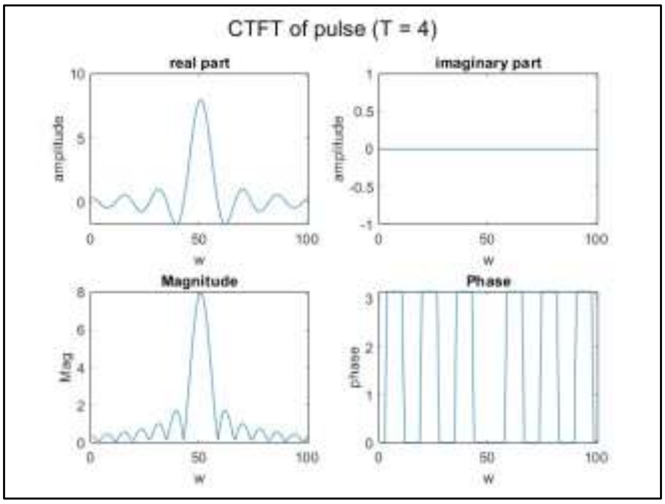
(b) Obtained plot for the CTFT of rectangular pulse :



This supports the claim that a rectangular pulse in the time domain is a sinc function in the frequency domain and vice versa.

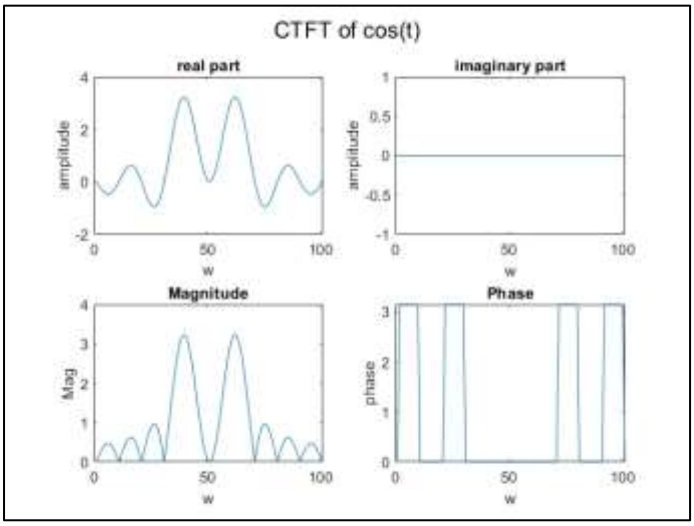
(c) when the value of T is changed :

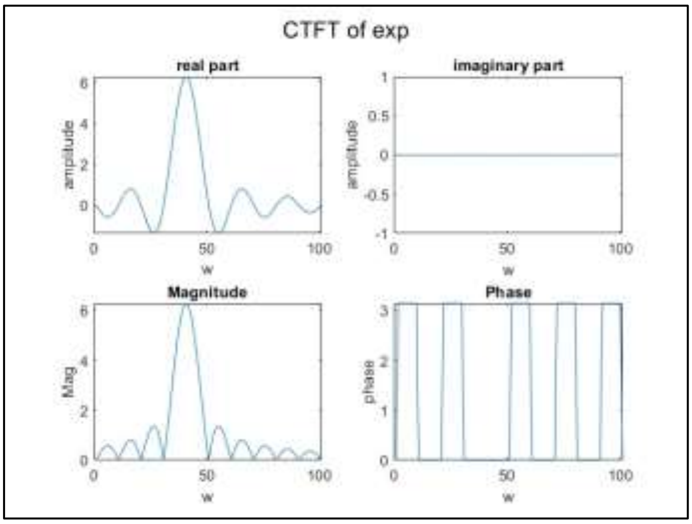




We can observe the property of time scaling from these two plots while changing the value of T.

(d) Obtained plots for x(t) = e^{jt} and x(t) = cos(t)





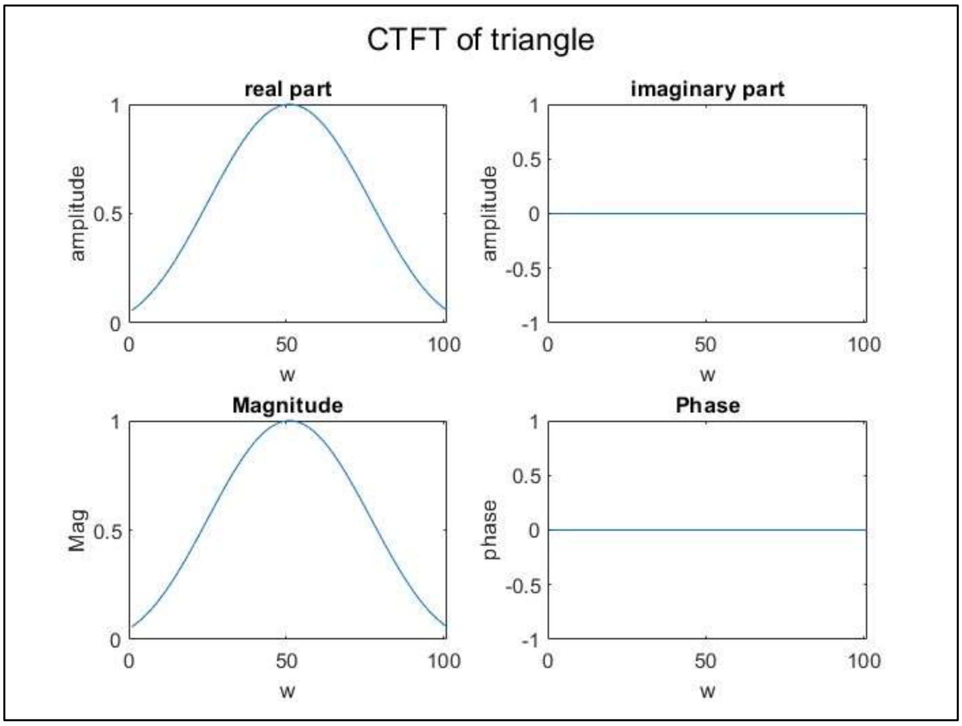
e^{jt} has a similar CTFT to that of a recatangular pulse, here it is just shifted and the CTFT of the cosine signal is the addition of two shifted sinc functions.

(e)

xt is expressed in matlab as follows :

```
xt = heaviside(t+1/2) - heaviside(t-1/2);
```

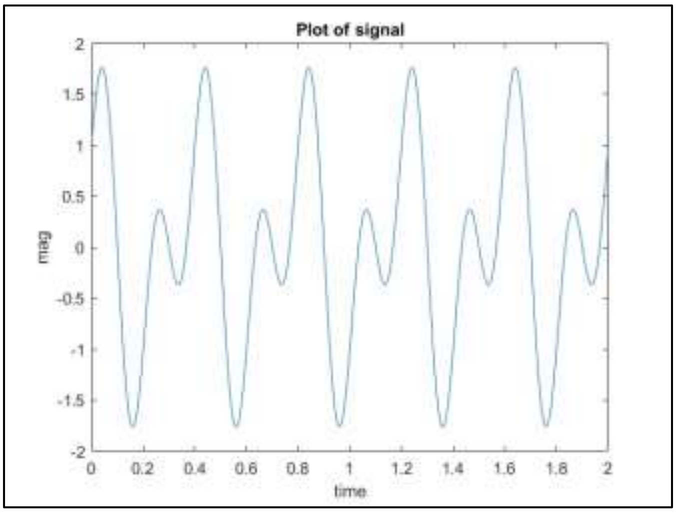
And the following plot is the obtained CTFT for a triangular pulse:

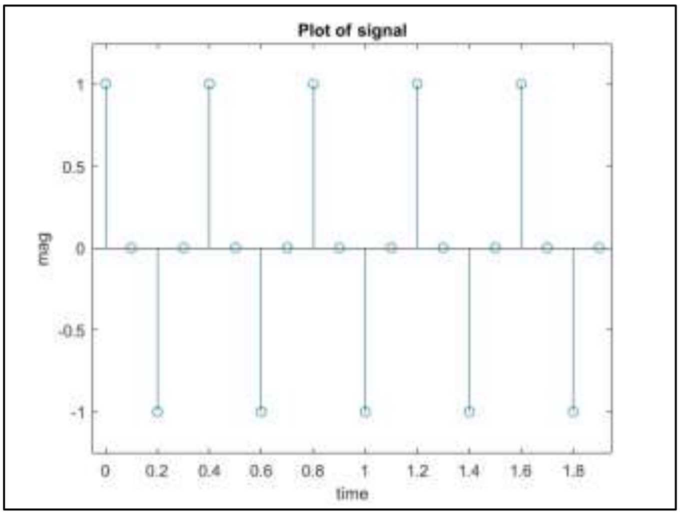


We can obtain the triangular pulse as a result of self-convolution of a box function ranging from -1/2 to 1/2. The observed CTFT has 0 phase and is a square of the sinc function.

Question 2

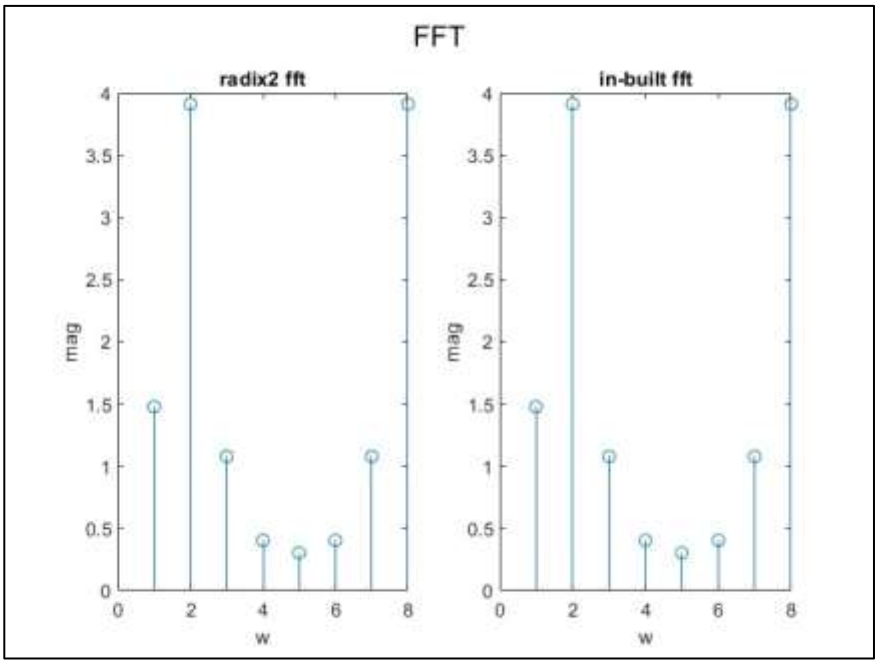
Sampling x(t) = cos(5pit) + sin(10pit) and plotting we get :





Question 3

Obtained plots using our radix2fft function and the inbuilt fft function:



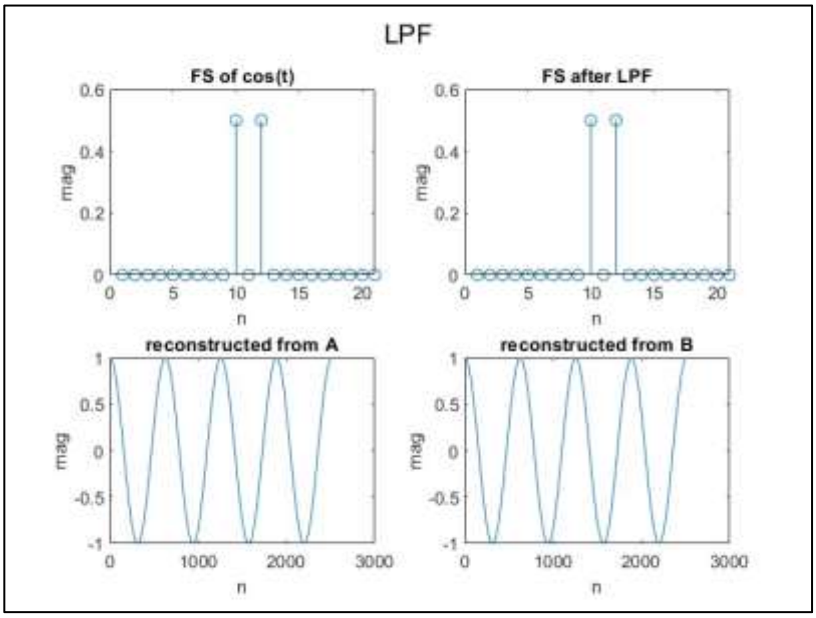
As we can see they are identical, this means that our radix2fft function is working well and as expected.

Question 4

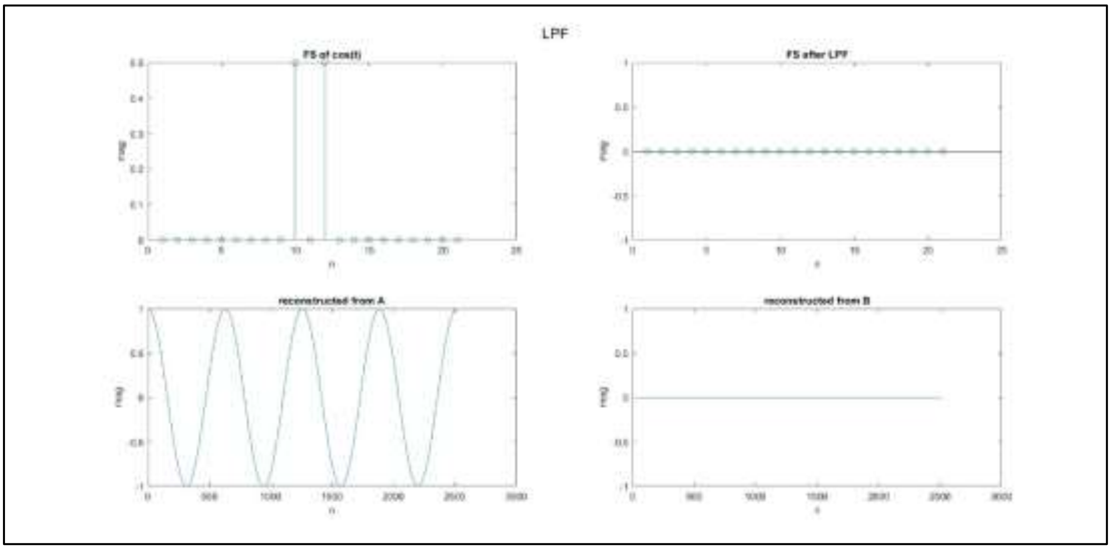
(B)

When we change the cutoff from 2 to 0.5, the frequencies that actually contribute to the ffts of the signal are filtered out, hence we get a zero signal or empty signal.

Wc = 2 :

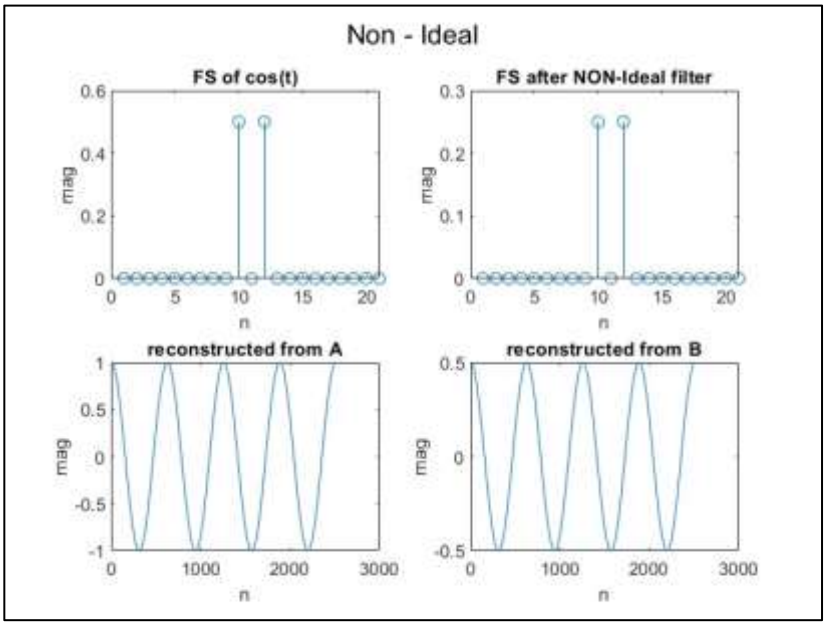


Wc = 0.5 :



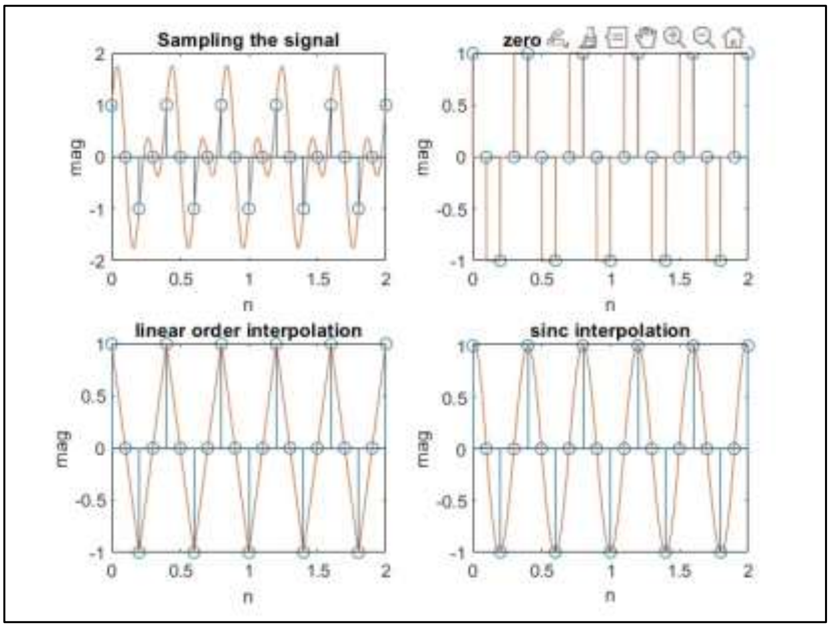
(D)

The filter given is a low pass filter. The magnitude of the transfer function is the amplification factor, that is how the complex nature is manifested.



Question 5

Results of the three different types of interpolations :



We can observe that sinc interpolation gives us the best results which are more accurate to that of the original signal compared to linear or zero-order.

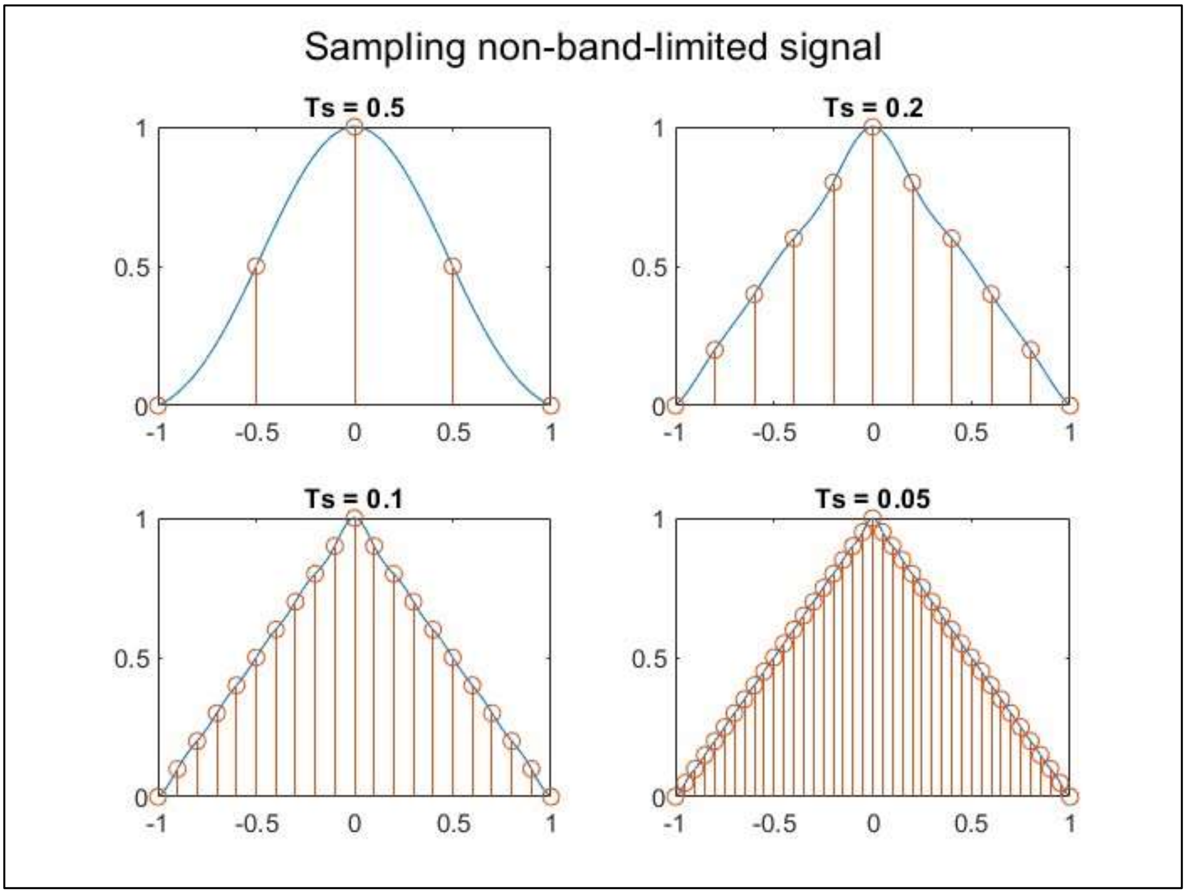
```
>> q5
MAE for 0-order interpolation =
    1.7601

MAE for linear interpolation =
    1.2076

MAE for sinc interpolation =
    1.0341
```

We can see that the error decreases as we go from zero-order to linear to sinc interpolation.

Question 6



As sampling frequency increases the number of samples obtained are more and facilitate for a much more accurate reconstruction.

Question 7

Following is observed :

File name	Bit-rate	fs	Time	Bits used
Wav1	256 kbps	8000	33.5296	32
Wav2	512 kbps	16000	33.5296	32
Wav3	1411 kbps	44100	29.6287	32 (31.99 but approx.)
Wav4	1411 kbps	44100	58.9936	32

**Time = length / fs, **Bits used = bit-rate / fs*

No. of quantizations that can be performed is 2^{32} for all the audio files.

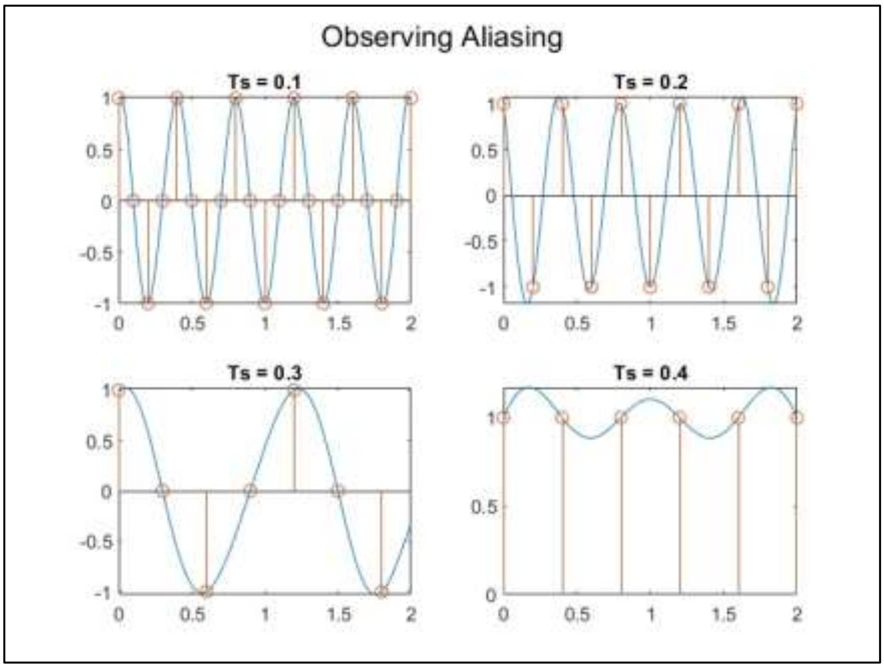
Changing the sampling frequency of the audio file and listening to it, you can observe that the audio files length has changed and the pitch is affected (pitch and frequency are related).

Question 8

Nyquist rate = 2 * w_m, where w_m is the max frequency of a signal.

Therefore the, Nyquist rate for x(t) = cos(5*pi*t) is 2 * 2.5 Hz which is 5Hz.

Obtained plot:



As Ts is increased to 0.4 we can observe that aliasing is taking place and the reconstructed signal is overlapping. This happens because the sampling frequency is less than the Nyquist rate. As Ts increases the plot also gets less accurate.