Lab Report-7

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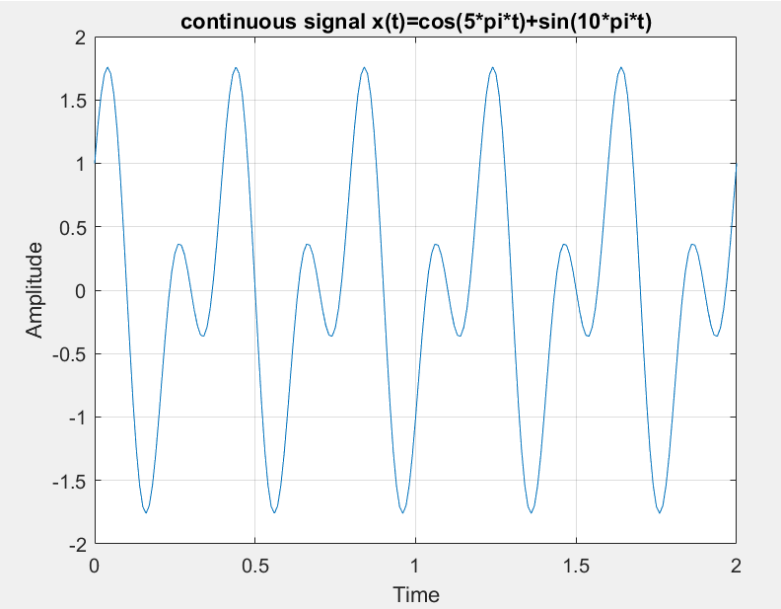
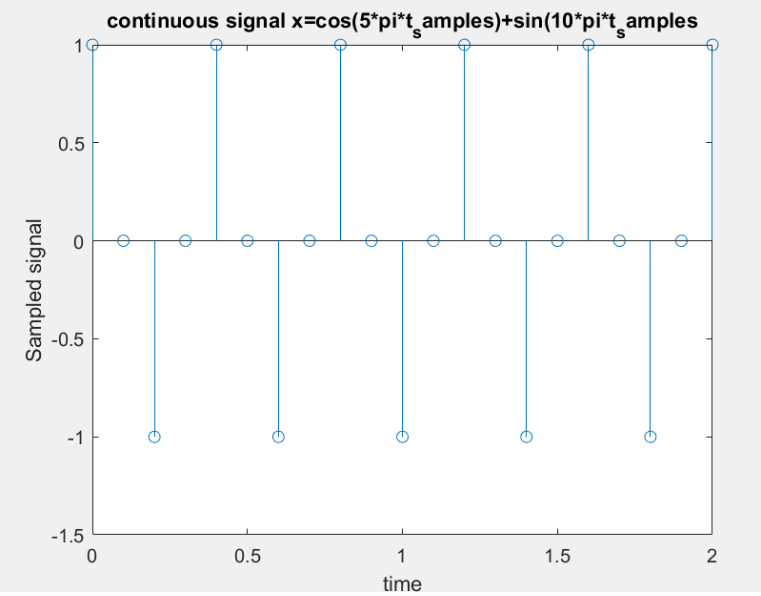
Roll No:2022102030

1)a)

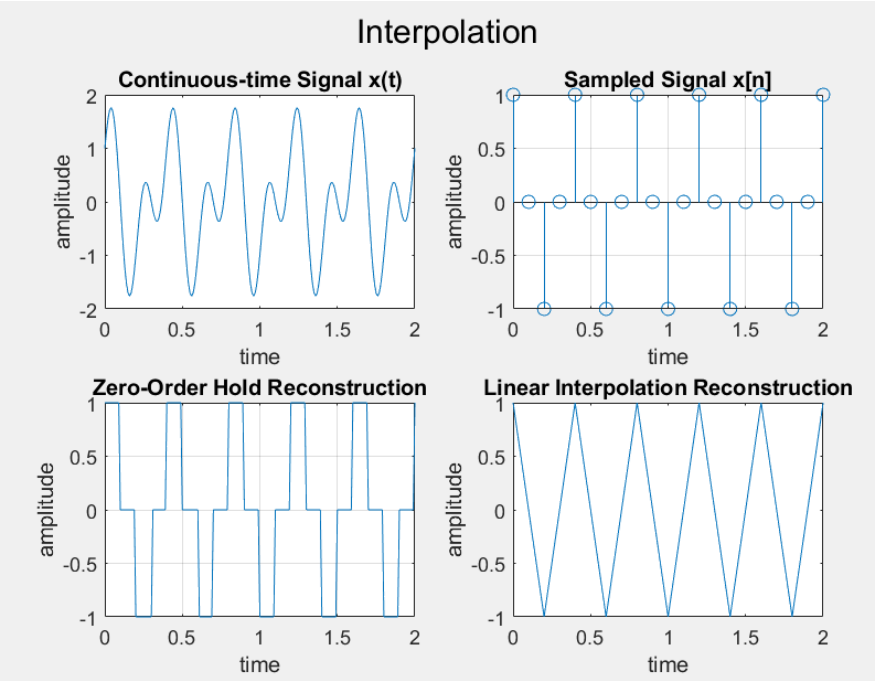


b)

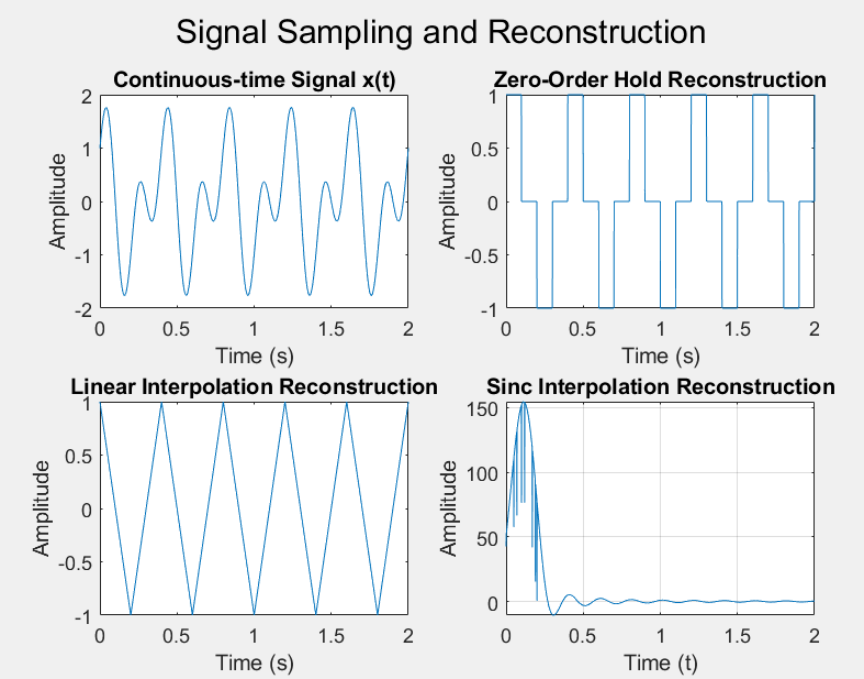


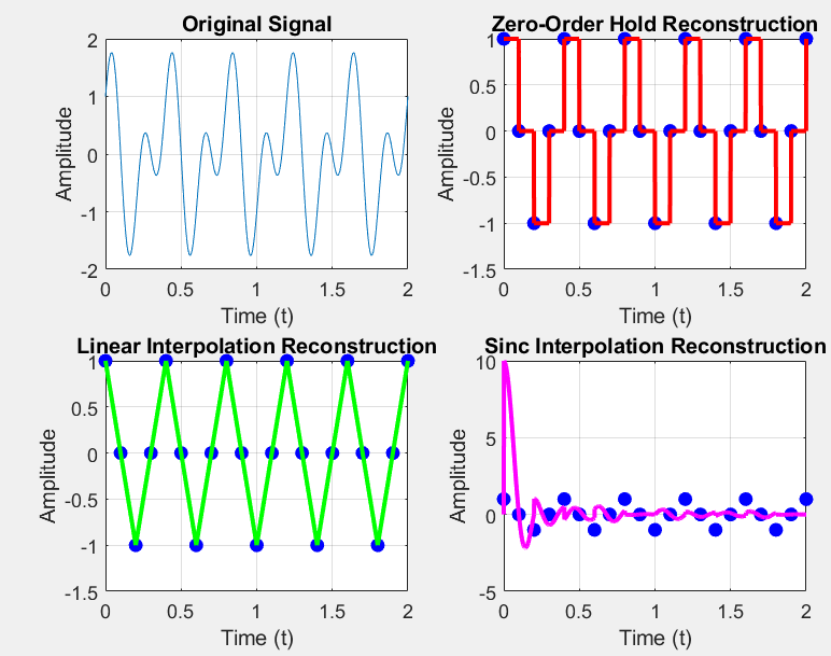
2)

a)

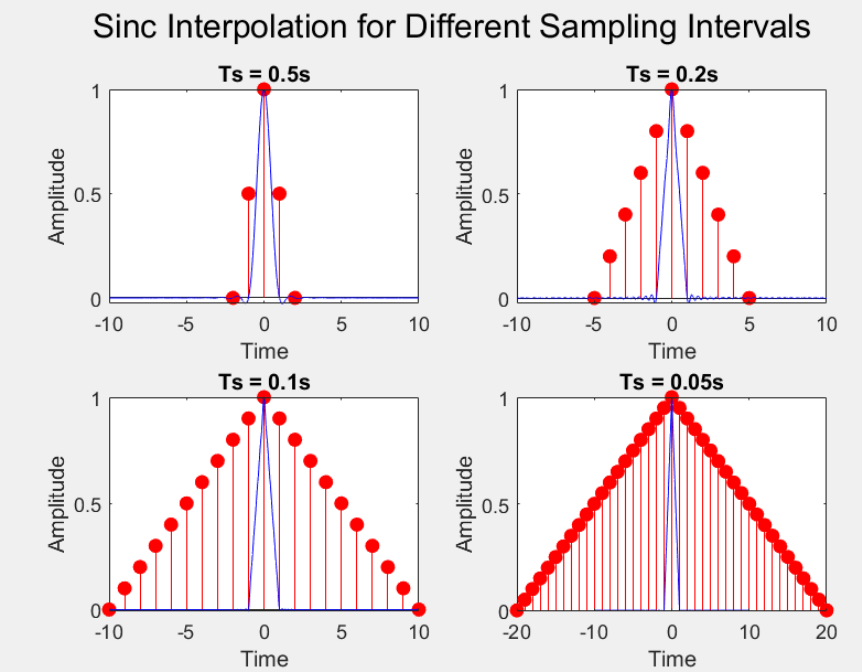

Interpolation
Continuous-time Signal x(t)
Sampled Signal x[n]
Zero-Order Hold Reconstruction
Linear Interpolation Reconstruction
amplitude
time

c)


Signal Sampling and Reconstruction
Continuous-time Signal x(t)
Zero-Order Hold Reconstruction
Linear Interpolation Reconstruction
Sinc Interpolation Reconstruction
Amplitude
Time (s)
Time (t)



3)



As the Ts decreases , the accuracy of the reconstructed signal improves .

Smaller Ts values result in denser sample sets , allowing sinc interpolation to capture the original signal's details with greater fidelity.

Therefore choosing an appropriate and sufficiently small Ts is essential for achieving accurate signal reconstruction when dealing with non-band limited signals like the triangular pulse.

4)

a)The bitrate for all audio files is 1411 bits/sec

b)

```
sampling frequency of audio1=
       44100

sampling frequency of audio2=
       44100

sampling frequency of audio3=
       44100

sampling frequency of audio4=
       44100
```

c)

```
sampling frequency=
       44100

duration Ts1=
    5.9432

sampling frequency=
       44100

duration Ts2=
   11.9291

sampling frequency=
       44100

duration Ts3=
   29.9831

sampling frequency=
       44100

duration Ts4=
   59.7730
```

d)

Bitdepth=bits/samples=no.of bits the ADC used(quantizing)=bitrate/sampling frequency

No. of quantizing levels used to perform ADC is 2^(bitdepth)

```
num_of_levels_1 =

    1.0224

no. of quantization levels for audio_1=
    1.0224

no. of bits the ADC used for quantizing for audio_1=
    0.0320

no. of quantization levels for audio_2=
    1.0224

no. of bits the ADC used for quantizing for audio_2=
    0.0320

no. of quantization levels for audio_3=
    1.0224

no. of bits the ADC used for quantizing for audio_3=
    0.0320

no. of quantization levels for audio_4=
    1.0224

no. of bits the ADC used for quantizing for audio_4=
    0.0320
```

Observation:

=> Audio at Lower sampling Frequencies:

*pitch drop

*Loss of High-frequency content

*Reduced in clarity of sound

*Quantization Noise

=> Audio at Higher sampling frequencies:

*pitch Increase

*Enhanced clarity

*Improved Fidelity(sound become more pronounced)

*No New Information(does not add new information to the audio)

Explanation Based on Fourier Transform:

These observations can be explained by considering the properties of the Fourier transform , particularly the Nyquist Shannon sampling theorem .

The sampling frequency must be atleast twice the highest frequency present in the signal.

*we can say that by changing the sampling frequency it affects the audio signal

*when we listen to audio at lower sampling frequencies ,it is undersampling the signal. Undersampling can lead to aliasing , where high frequency components in the signal fold back into the audible frequency is insufficient to capture the original signal's higher frequency components.

*In the context of the Fourier transform , aliasing occurs because the undersampling violates the Nyquist-shannon sampling theorem. The high frequency components of the signal cannot be accurately represented.

*lowering the sampling frequencies leads to the aliasing where high frequency components fold back into the audible range .

*when we listen to the audio at higher sampling frequencies , it is oversampling the signal.

*oversampling means we are using the sampling frequency significantly higher than the Nyquist frequency required to accurately represent the signal.

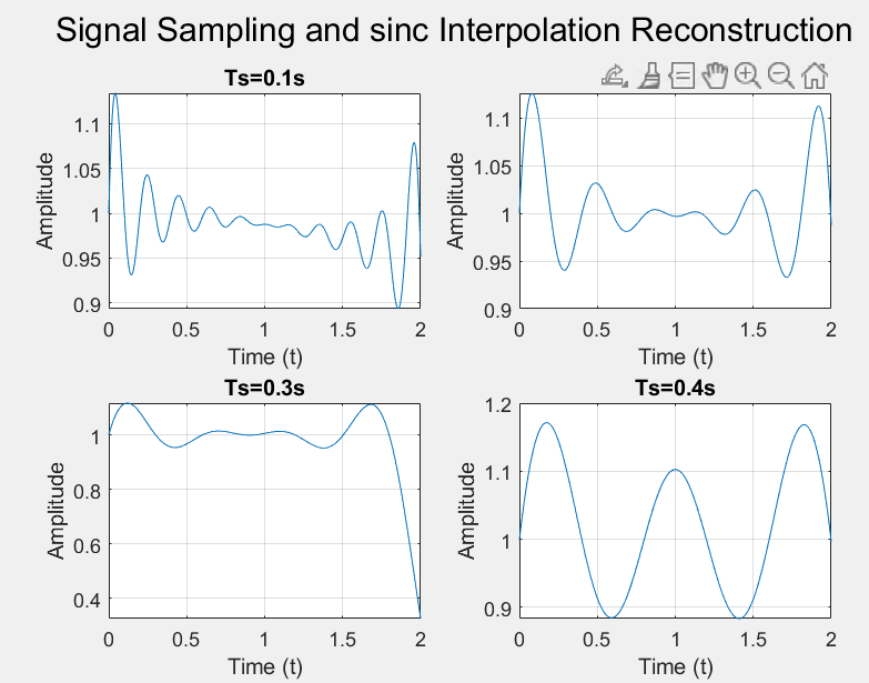
**The fourier transform helps us to understand the importance of sampling frequency in the reconstruction of a continuous signal.when we consider the Nyquist-shannon sampling theorem and sample at or above the Nyquist frequency , you can accurately reconstruct the original signal with minimum loss of information.

- Deviation from the theorem , either by undersamplin or oversampling , affects the ability to reconstruct the signal .

5)

a)The Nyquist rate =2*fs

=>2*5=10Hz



Observations:

As the sampling interval (Ts) increases beyond the Nyquist rate , aliasing becomes more pronounced , and the quality of the reconstructed signal decreases. It is essential to sample at or above the Nyquist rate to avoid aliasing and faithfully reconstruct the original signal.