EE386 Digital Signal Processing Lab Jul-Dec 2021

1: Experiment

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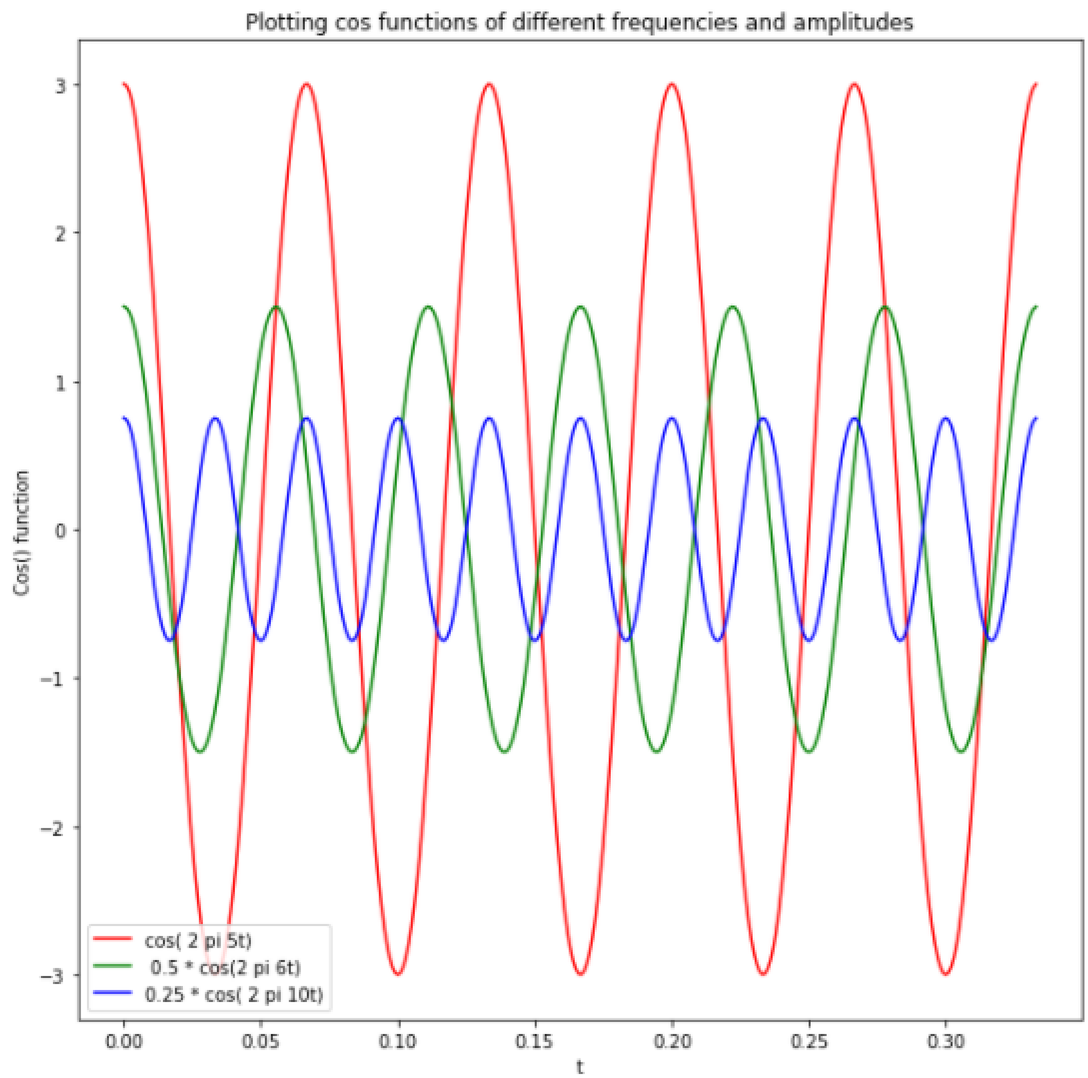
GITHUB LINK FOR CODE: https://github.com/ramyashri1887/DSP-LAB

1) Sampling and frequency-domain aliasing

Problem 1.1



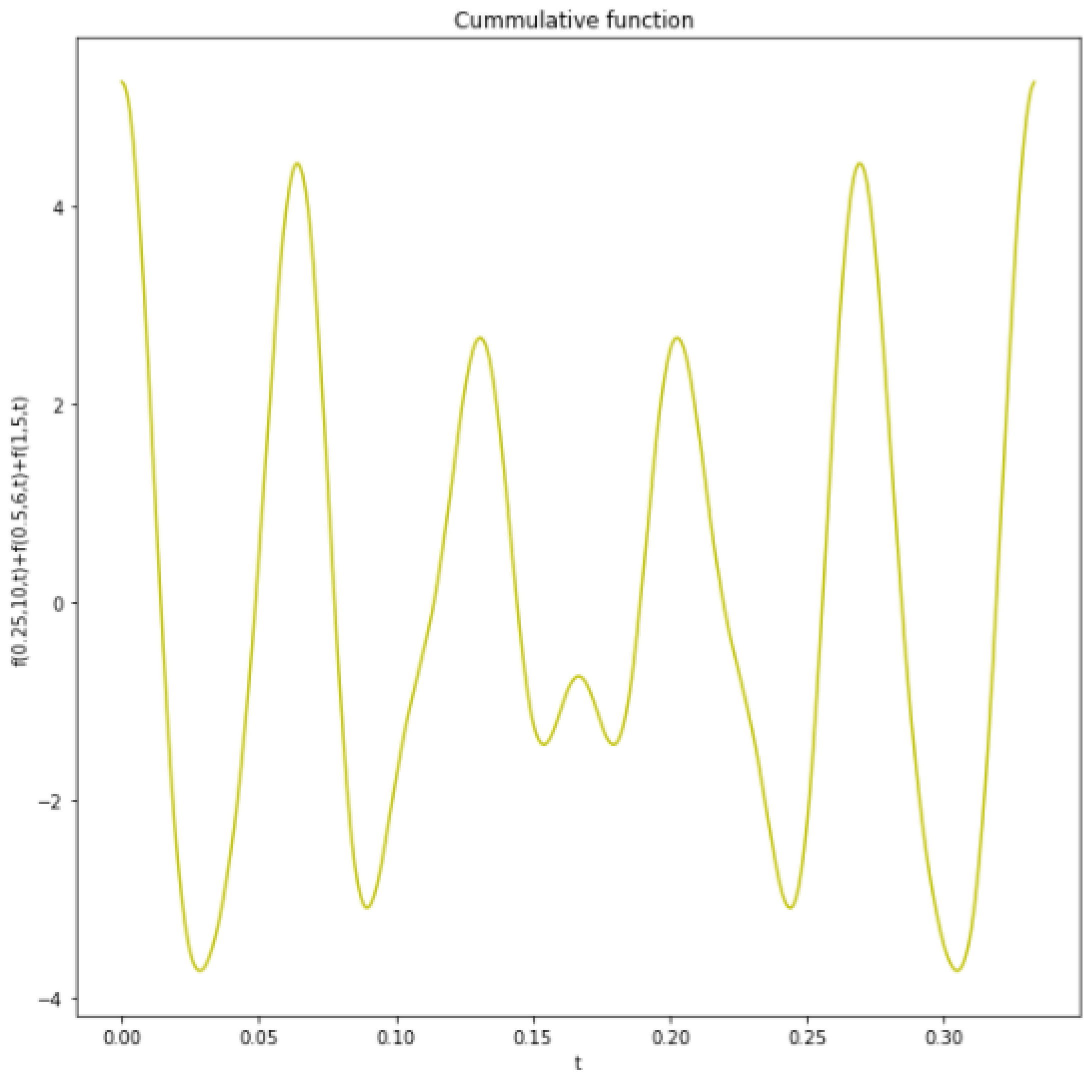
Plot the following waveforms in one single plot for 1/α second duration

Observations

1. Adding three bandlimited signals
   a. 3 * cos(2*pi*(5t)) having a bandwidth of 10f.
   b. 1.5 * cos(2*pi*(6t)) having a bandwidth of 12f.
   c. 0.75 * cos(2*pi*(10t)) having a bandwidth of 20f.

2. Ideally cos signal is bandlimited but the digital representation is buffered hence not bandlimited.

Problem 1.2 Addition of bandlimited signals to form another bandlimited signal



Observations:

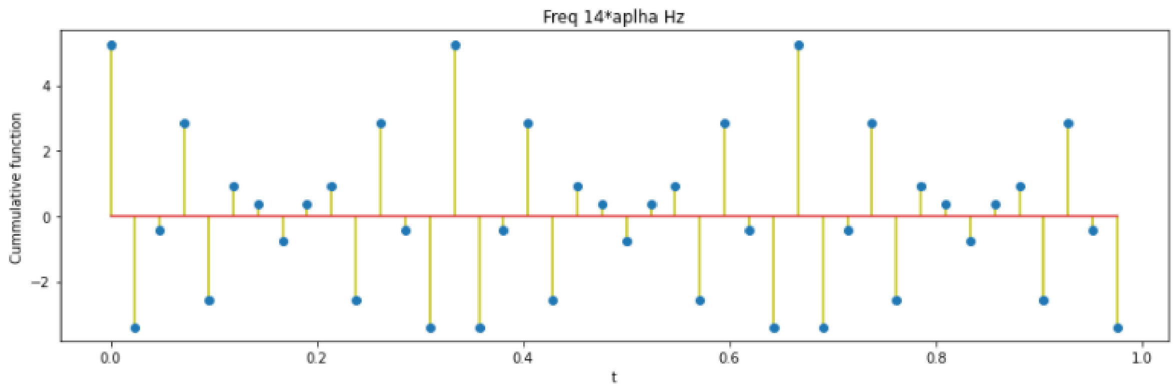
1. Adding n bandlimited signals leads to another bandlimited signal with a bandwidth 2f where f is max(f1, f2, ...fn).

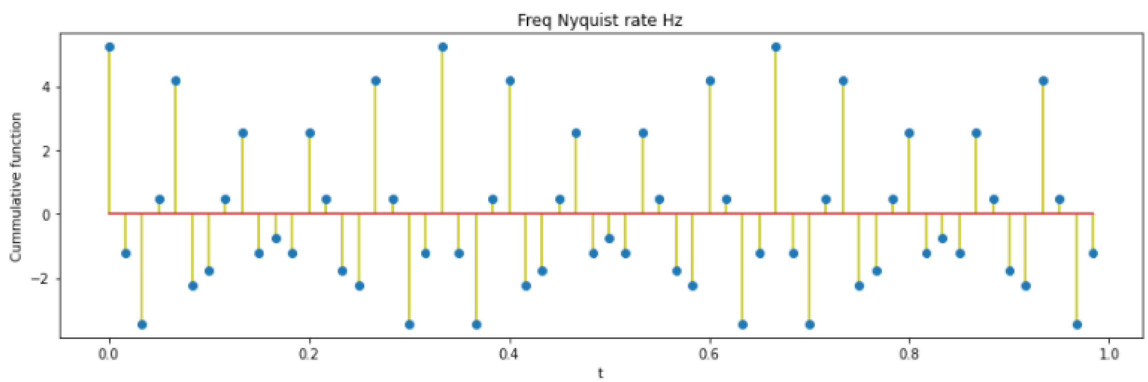
2. **Buffered time-domain signals can not be bandlimited** because a finite length time-domain signal will have an infinite bandwidth whilst an infinite time signal will have a finite bandwidth in the frequency domain.

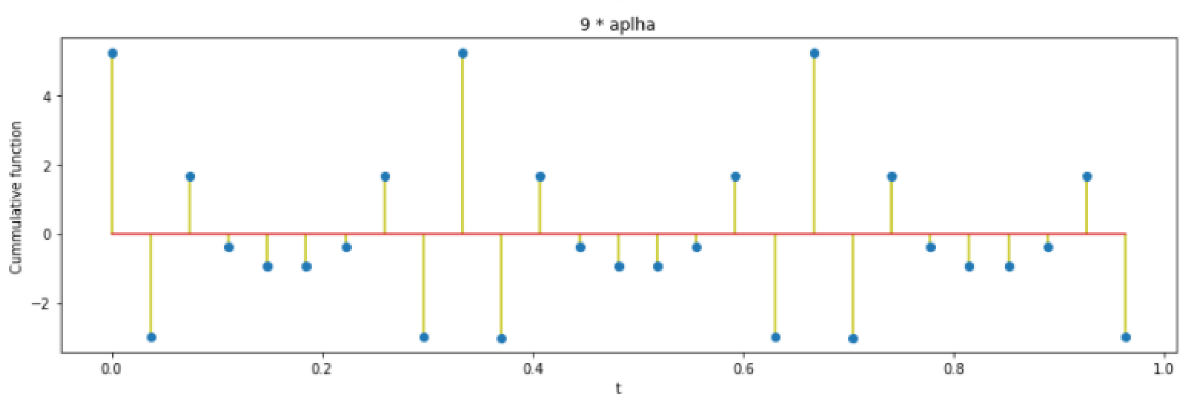
Inference :

1. **Bandlimited space forms a vector space**. Reason: Resultant of the addition of two bandlimited signals is a bandlimited signal. The resultant of a scaled bandlimited signal is also a bandlimited signal. Region satisfies linearity and homogeneity, hence a vector space.
2. Solution for buffered time-domain signal: **Zero paddings** (approximating signal to be zero for the time before and after the buffered region).

Problem 1.3 Sampling the signal at different rates

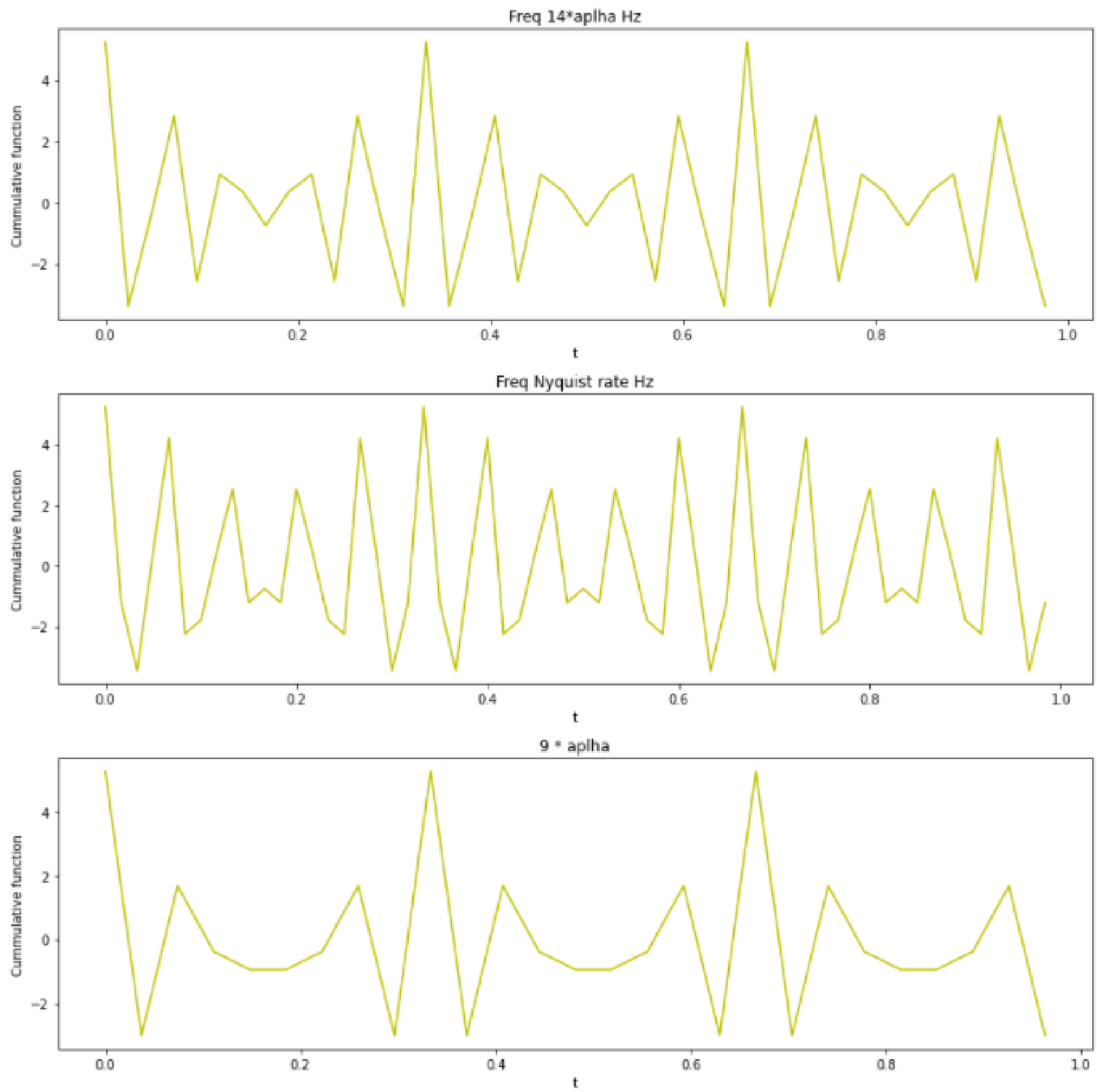






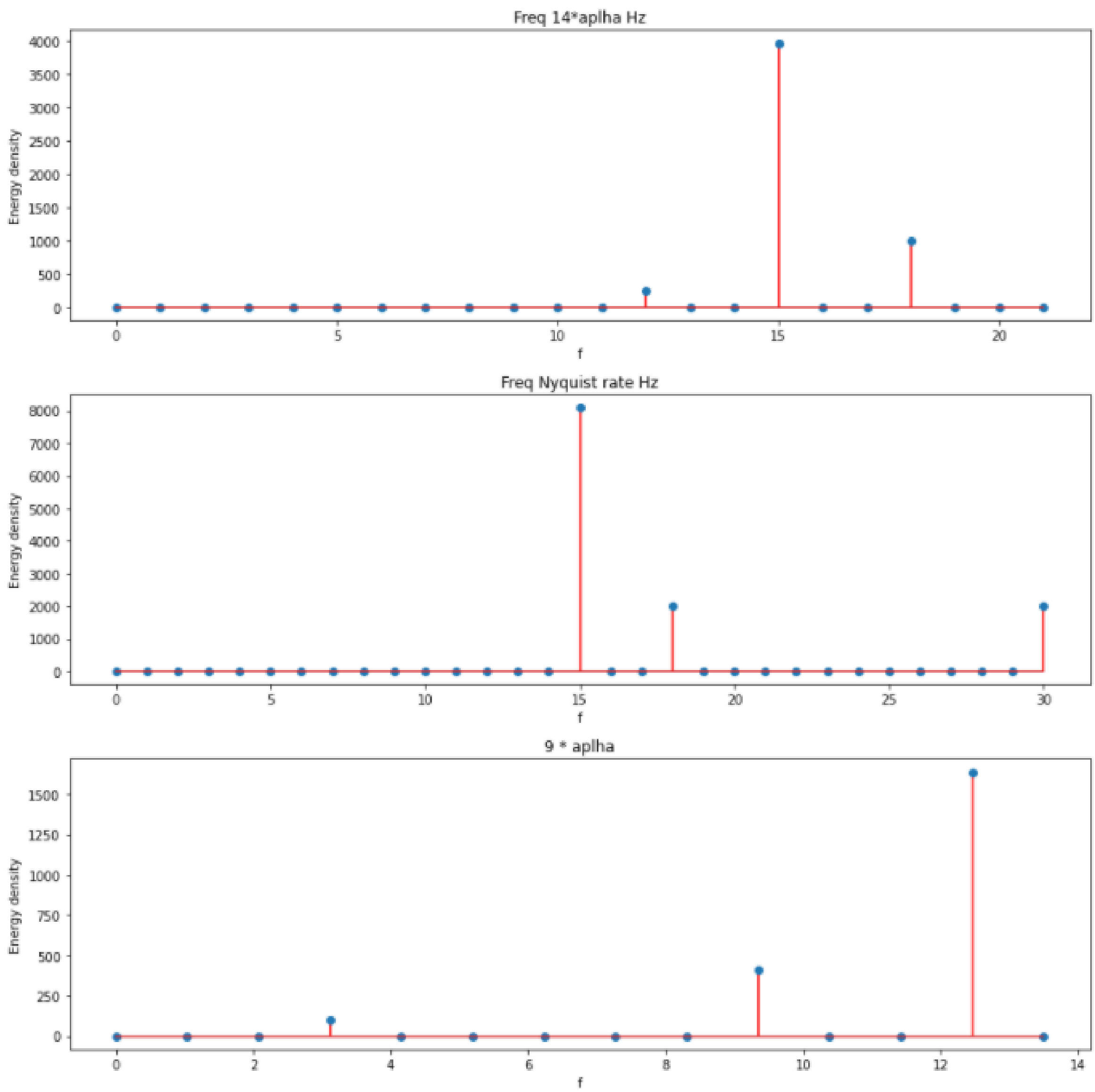
1. Sampling the signal at different rates.
2. Using stem plot to visualize the figure.
3. The sampling rate of 9 alpha should alias 6 alpha hertz to 3 alpha.

Problem 1.4 Signal reconstruction using linear interpolation



1. Signal reconstruction using linear interpolation, to ideally reconstruct a signal using Shannon's formula, a low pass filter must be used.
2. Plot function inherently uses linear interpolation function to plot the discrete values.

Problem 1.5 Energy density plot of the signals



Results:

1. Frequencies 5,6 and 10 were not obtained.
2. Signals can be fully reconstructed by sampling the signal at the Nyquist rate. And using a low pass filter based on Shannon's sampling theorem.

2) Generating digital music

Problem

Generate a sequence containing the tones corresponding to “Do Re Mi Fa So La Ti Do”

Method:

1. Sin(2 pi f t) function with different frequencies for each tone was generated and appended to form the final music. The final note Do is double the frequency of the first note Do, this is called an **octave** in music theory.
2. I have used a different approach, the GitLab approach is also included in the code.
3. The signal was sampled at different rates to study the difference in audio quality for different sampling rates.

Inferences:

1. I couldn’t particularly hear the difference in sample rates of the audio files.
2. Using the ipd function (interactive python display function) and converting sr to sr/2 slows the video down by a factor of two.

3) Resampling

Problem

Load Track00α.wav and generate different .wav files with several values of the sampling rate.

Method

Removed every other sample using the **np.delete( )** method to downsample the signal by a factor of 2, and every 1st and 2nd sample to downsample the signal by a factor of 3.

Inference

1. Upsampling can be done using the following methods

   a. Adding zeroes in between the signal
   Consequences: No change in the frequency content of the signal, but the bandwidth of the signal doubles.

:

   b. Interpolation (Linear and Shanon)
   Consequences: Could change the frequency content of the signal. Shannon’s interpolation yields the right frequencies.
   Reason:

Libraries and Methods used:

1. Numpy
2. Matplotlib
3. Scipy
4. IPython.Display