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

Experiment : 2

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This lab experiment covers some basic practicality of Digital Signal Processing such as signal generation, various tone generation, convolution of signals and amplitude modulation. Along with Python, I have used libraries such as numpy, pandas, scipy etc. The code to my entire work in this lab experiment is here. And the input files and my output files can be viewed here.

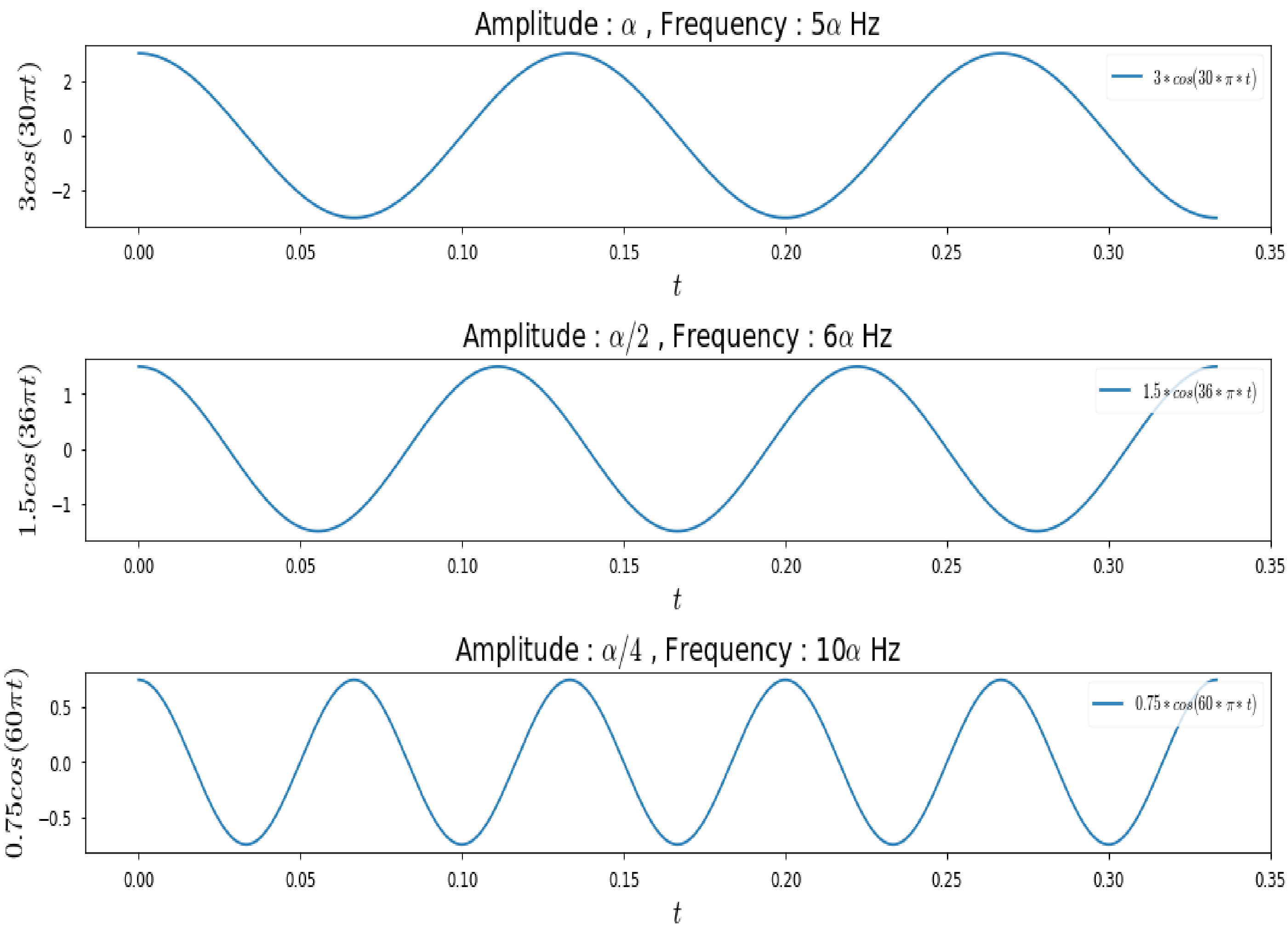
Please Note : I have used $\alpha = 3$ because my registration number is 191910.

Question 1 - Sampling and frequency-domain aliasing

(Subproblem - 1)

The question asks to create 3 different Cosine waveforms with different amplitudes and frequencies and to plot all the functions in the same plot for $\frac{1}{\alpha}$ duration. The different Amplitudes are α, $\frac{\alpha}{2}$, $\frac{\alpha}{4}$ and differnt Frequencies are 5α, 6α, 10α.

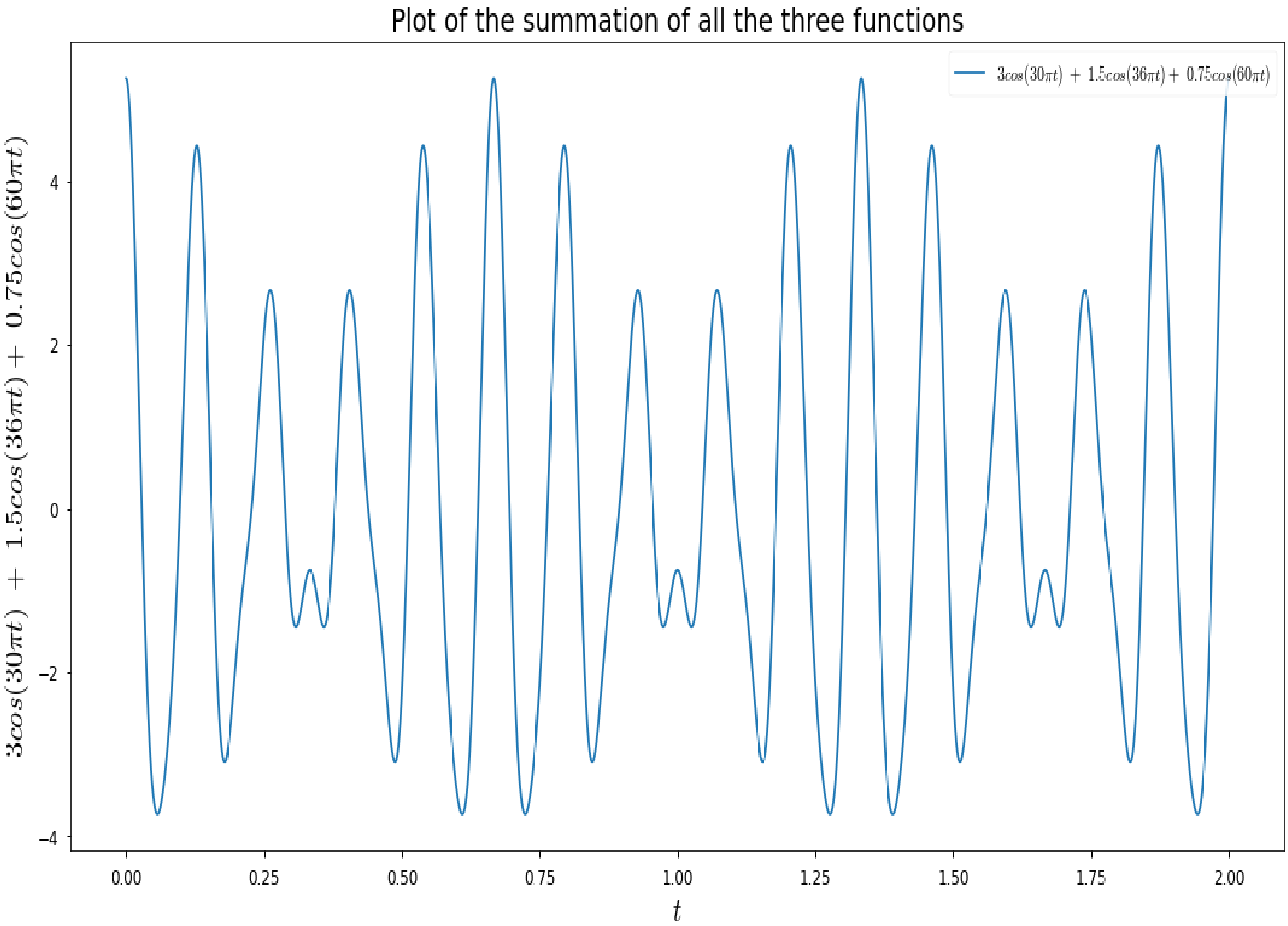
- So the three Cosine Functions are :
  - $3cos(30\pi t)$
  - $1.5cos(36\pi t)$
  - $0.75cos(60\pi t)$



(Subproblem - 2)

In this subproblem, the question asked us to sum all the three Cosine functions which were earlier plotted in subproblem 1 and to plot the new summation function i.e

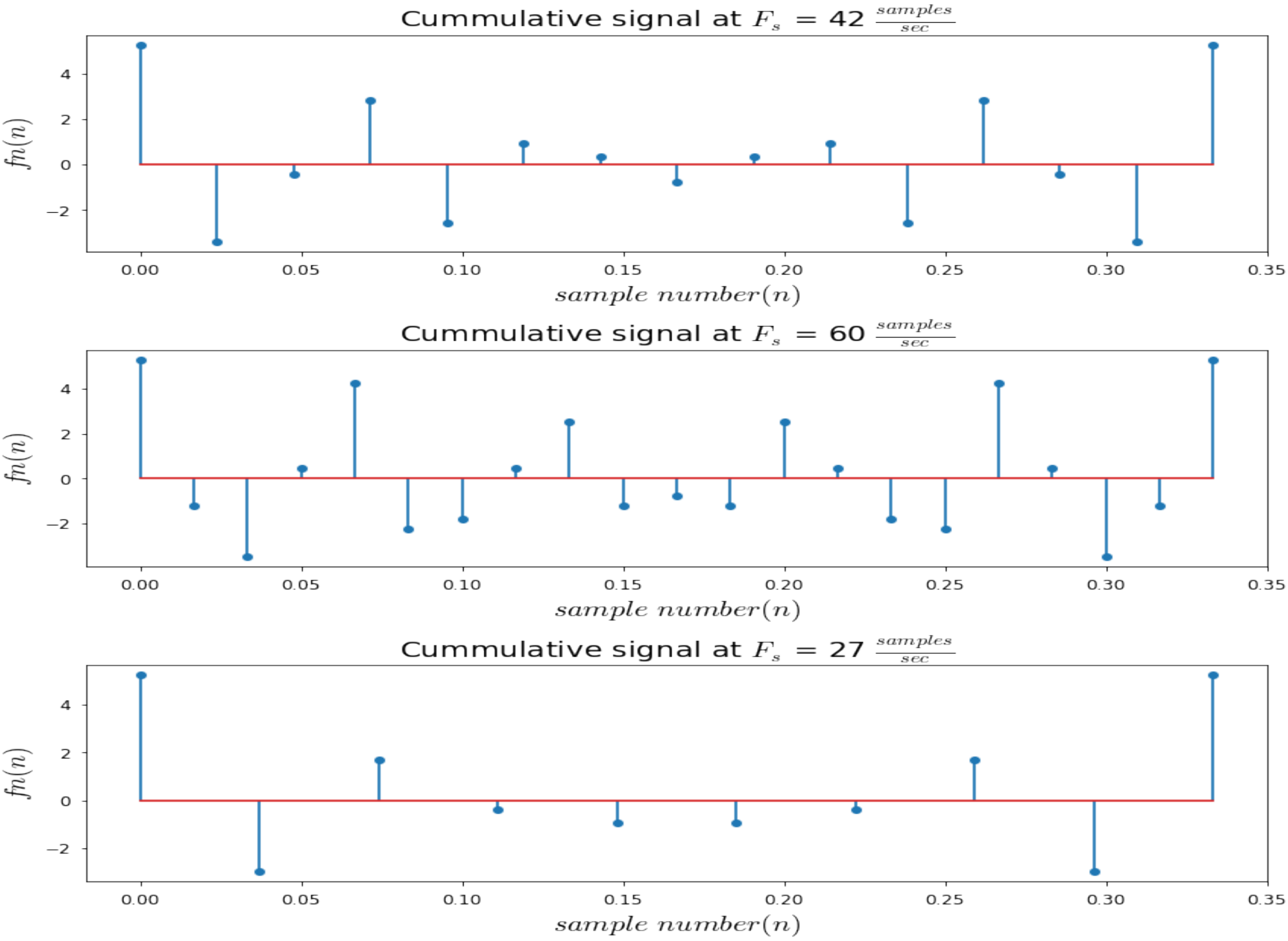
$3cos(30\pi t)\ +\ 1.5cos(36\pi t) +\ 0.75cos(60\pi t)$



(Subproblem - 3)

This question asks us to sample the cumulative signal i.e the summation signal ($3cos(30\pi t) + 1.5cos(36\pi t) + 0.75cos(60\pi t)$) at three different sampling frequencies, 14α, 20α, 9α

- Sampling frequency $= 14\alpha = 42\ \frac{samples}{sec}$
- Nyquist rate sampling frequency of signal = 2 times the max frequency $= 2*10\alpha = 20\alpha = 60\ samples/sec$
- 6α frequency is aliased to 3α, so Folding Frequency $= \frac{6\alpha+3\alpha}{2} = 4.5\alpha = \frac{F_s}{2}$ is 4.5α. From this we can say that Sampling frequency $= F_s = 9\alpha = 27\ samples/sec$



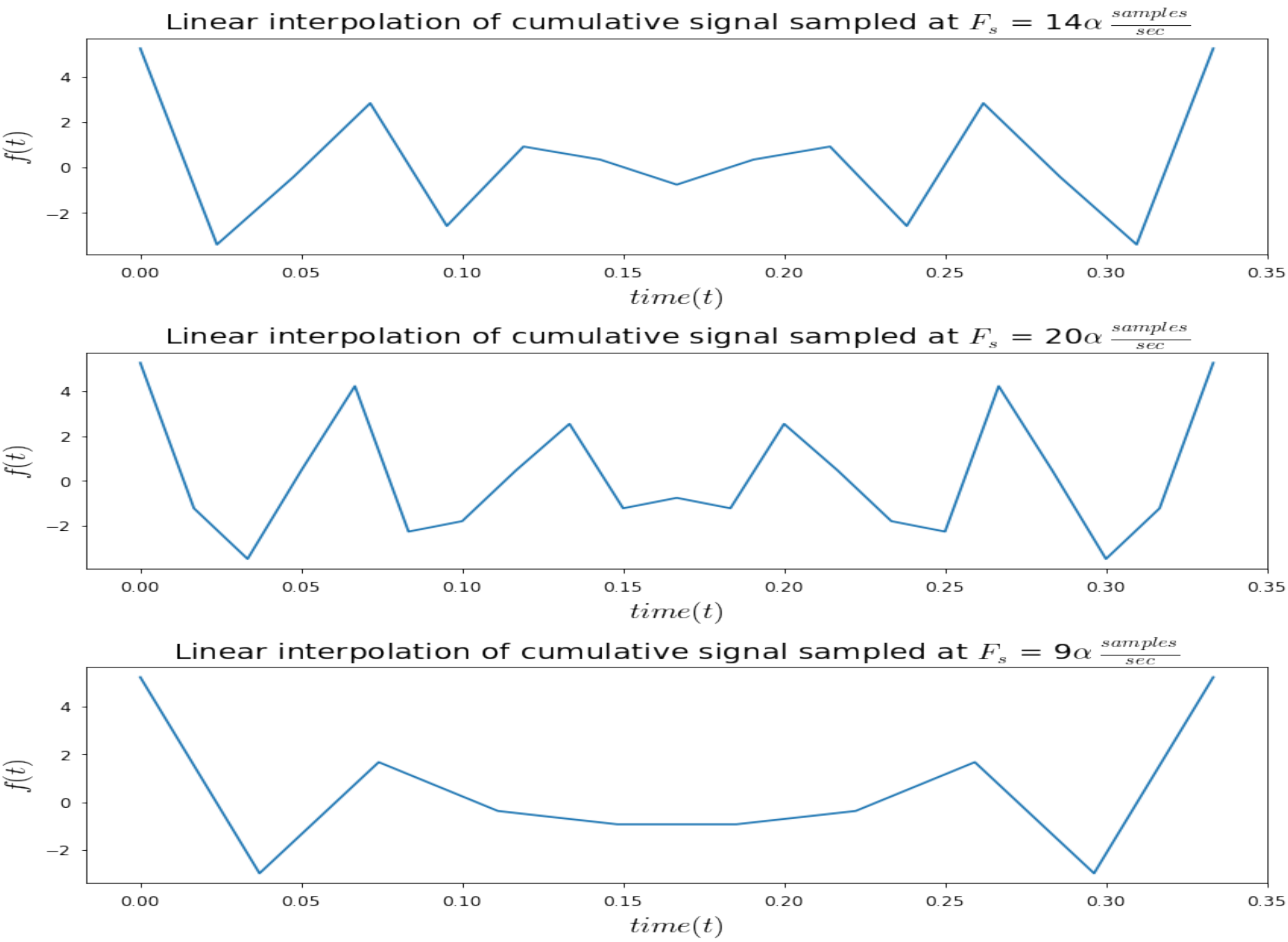
(Subproblem - 4)

This question asked us to preform linear interpolation on the sampled responses for the different sampling rates.

- We have sampled at 3 frequencies : 9α, 14α, 20α
- When it's sampled at 14α,
  - This sampling frequency is greater than max frequency 10α, but less than Nyquist sampling frequency 20α i.e, $20\alpha \leq\ 14\alpha \leq\ 20\alpha$
  - At this frequency the signal is underdamped as we can see from the plots below because it doesn't contain all the characteristic features of the original cumulative signal. We can clearly see that some dips and peaks are not present in this frequency sampling, which were originally present in the cumulative signal.
  - Hence we cannot we cannot use the plot sampled at this frequency to analyse the original cumulative signal's behaviour or characteristics.
- When we sample at Nyquist Sampling rate

  - According to Nyquist Sampling Theorem, we can use the plot from Nyquist rate of sampling frequency to learn about the behaviour of the original cumulative signal. And also we can observe this from the plot that most of the peaks or dips are present from the original cumulative signal

- At maximum sampling frequency,
  - This frequency sampling is definitely highly undersampled as we have previously discussed that 14α is itself very undersampled.
  - So even this sampling should not be used to draw conclusions about the original signal.

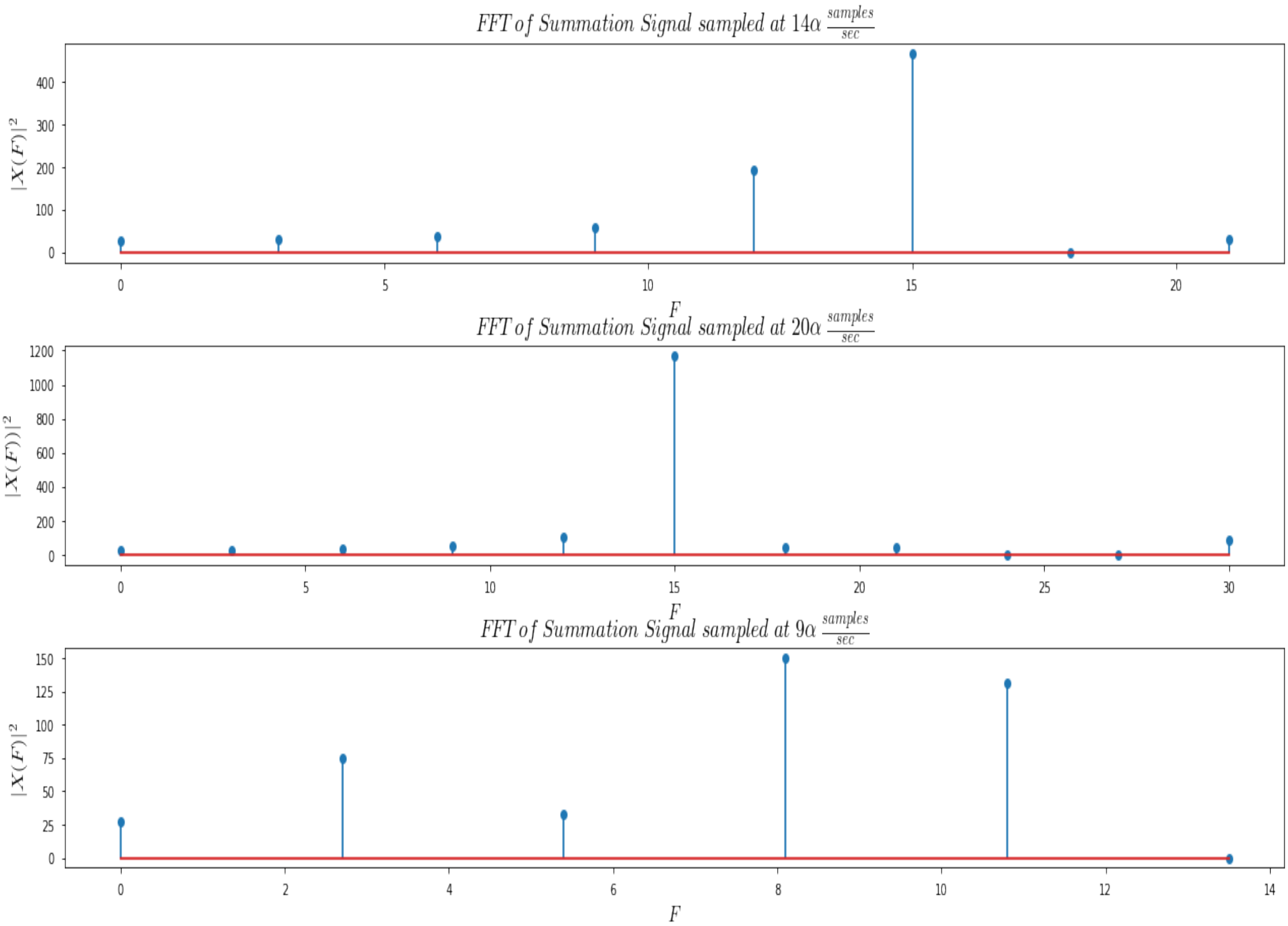


(Subproblem - 5)

The question asks us to draw the energy plots for the 3 different sampling frequencied plots from the third subproblem

- Aliased Frequencies :
  - For 14α :

    - $*$ 30 Hz frequency would be aliased to 12 Hz
  - For 9α :
    - $*$ 15 Hz frequency would be aliased to 12 Hz
    - $*$ 18 Hz frequency would be aliased to 9 Hz
    - $*$ 30 Hz frequency would be aliased to 3 Hz



Question 2 - Generating Digital Music

A sequence had to be generated containing the tones corresponding to "Do Re Mi Fa So La Ti Do", the signals had to be appended together and these signals had to be generated using different sampling rates. Once the generation is done, they had to be stored/written into .wav files.

We can generate the signal of the type $y(t) = sin(2 * \pi * f * t)$,where f is the frequency that corresponds to each note. The link to the outputs is here.

- If the signal is undersampled, then there is a high chance of it being aliased and it can also lead to crossing over of frequencies. We can also observe that few frequencies are folded i.e aliased when it's undersampled.

- The required digital music is produced by horizontally stacking the sinusoidal waveforms.

Question 3 - Resampling

The question asked us to load the respective track i.e Track003.wav and generate different .wav files with several values of the sampling rate (for example, half the original sampling rate, 1/3rd of the original sampling rate etc.) and see the effect of this different sampling rate on the audio.

- So, I have downsampled for the sampling rates of $\frac{1}{2}$, $\frac{1}{3}$, $\frac{1}{4}$
- I could observe that the output files sound as if the speakers were busted or if there is too much noise added, this gave me an intuition to think that it's aliased.
- My output files for this downsampling can be found here

1 Appendix

- Note : I have used $\alpha = 3$ because my registration number is 191910. Since $\alpha = 1 + \text{mod}(910{,}4) = 1 + 3$
- The link to all the code is here
- The link to all input and output files are here
- The Github repo to all code and previous experiments can be found here