Digital Signal Processing Lab
Experiment 1
Sampling

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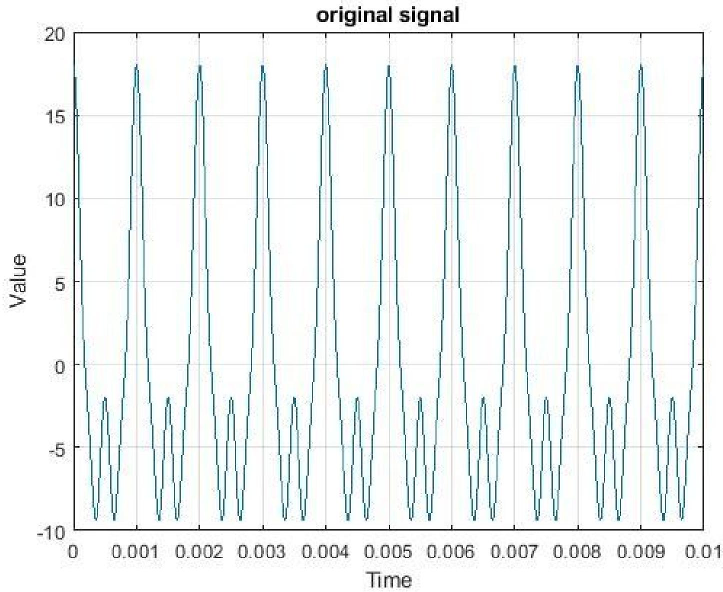
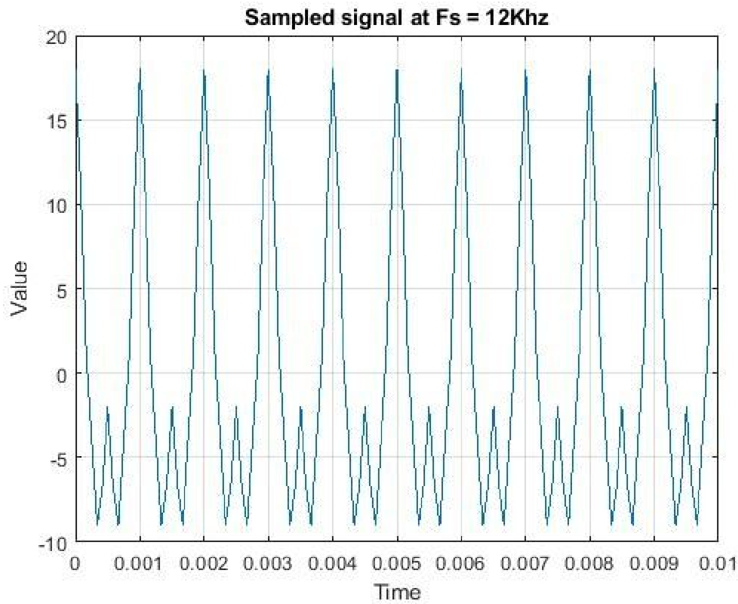
a) Sampling of Sinusoidal Waveform

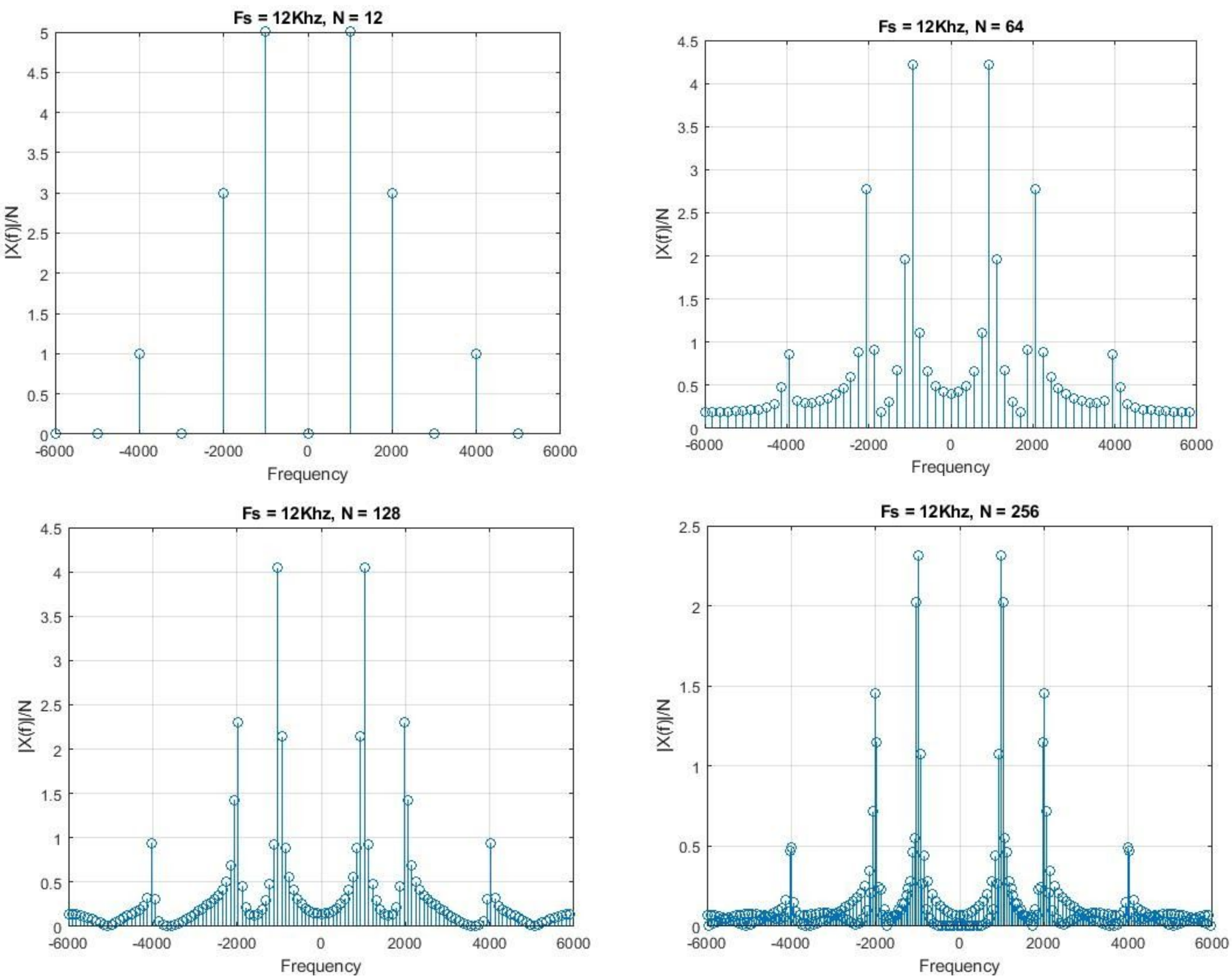


This is the original signal plot, I have taken signal to be
x(t) = 10cos(2*pi*1000*t) + 6cos(2*pi*2*1000*t) + 2cos(2*pi*4*1000*t)
To plot the signal in its original form I have sampled it in much higher rate then its highest frequency component, the resultant graph can be seen above, which is smooth as expected.



The above Figure shows the sampled signal output at 12Khz sampling rate, since sampling rate is much higher than the maximum frequency component of the signal i.e(4Khz) the signal looks approximately the same as the original one, however corner can be seen at peeks which is smooth in the original signal, this is due to the fact that with decrease in sampling frequency, we lost information of the signal start becoming less smooth.

Results of DFT with Different N.



Observations:

N here corresponds to the total number of frequency points that will result in the DFT output. To understand the difference in the output with increasing N, it’s important to understand the difference of DTFT and DFT.

If we are interested in the frequency response, then we want a continuous function of frequency, which would be the Discrete Time Fourier Transform, as opposed to the Discrete Fourier Transform (which the FFT computes) which is a discrete function of frequency. freqz returns samples of the DTFT, while fft returns samples of the DFT.

Since in the above experiment fft function is used so with increase in N it increases the sample points taken in frequency domain to calculate DFT and thus with increase in N DFT starts becoming more and more smooth.

b) Sampling at below Nyquist rate and effect of aliasing.

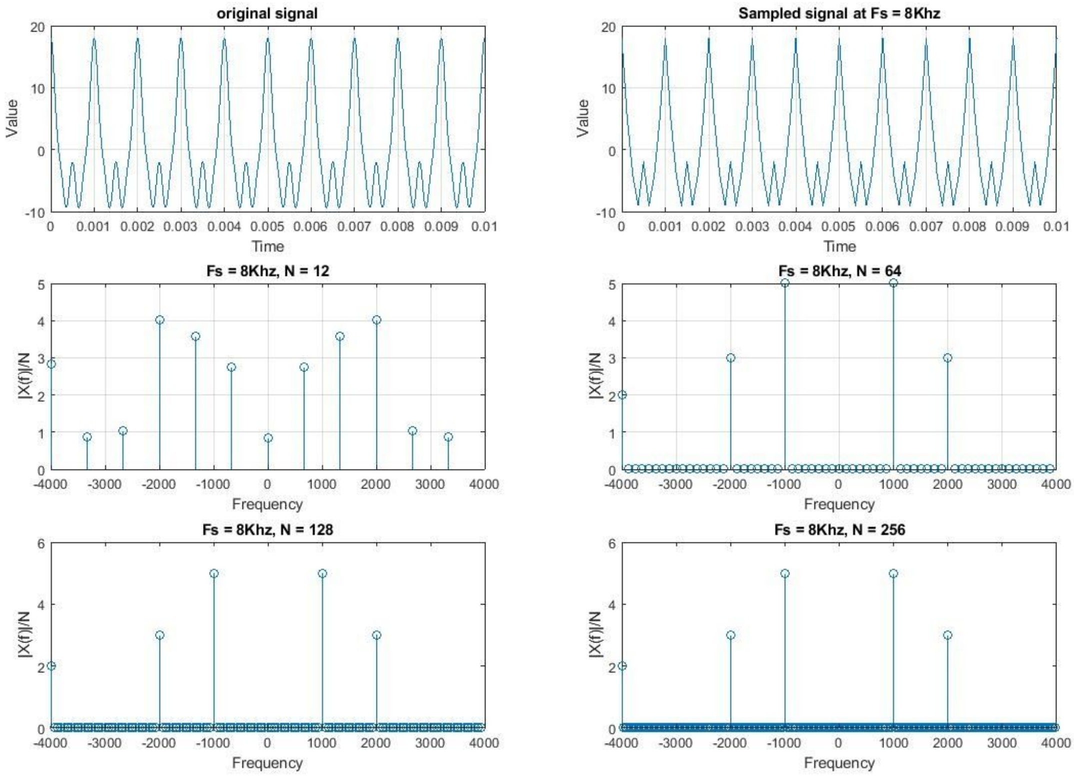
The above experiment is repeated with 3 more sampling frequency to observe the effect of aliasing.

According toNyquist Theorem minimum sampling rate should be 2Fmax, i.e 2*4KhZ = 8Khz.

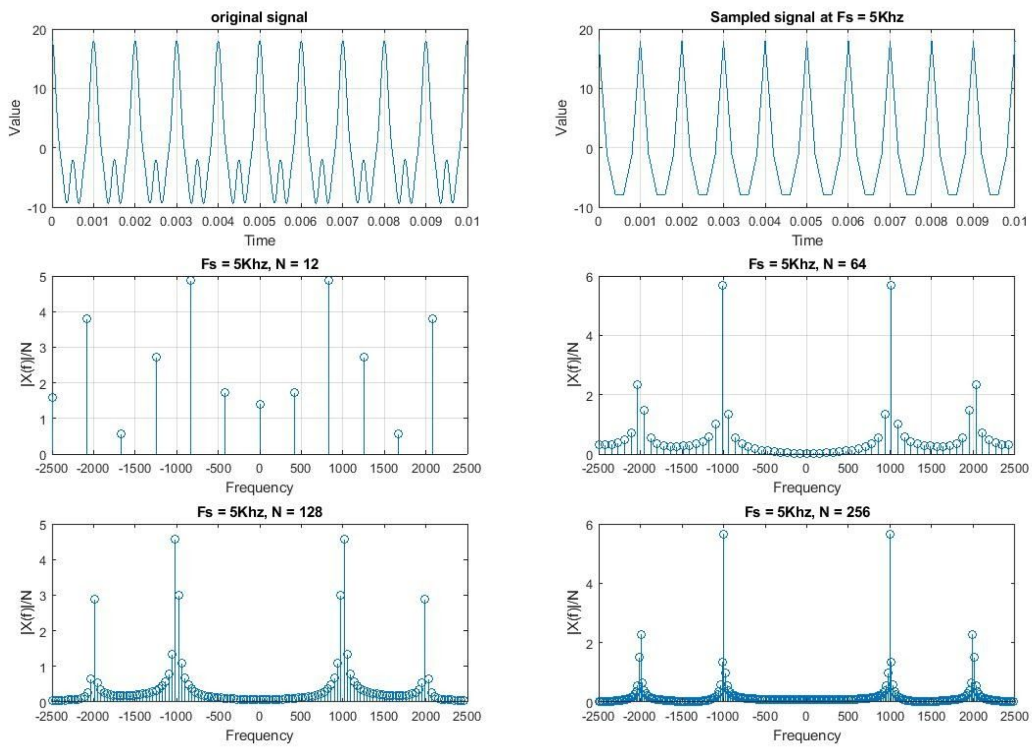
Frequency used to sample are: 8Khz, 5Khz, and 4Khz.

Results:

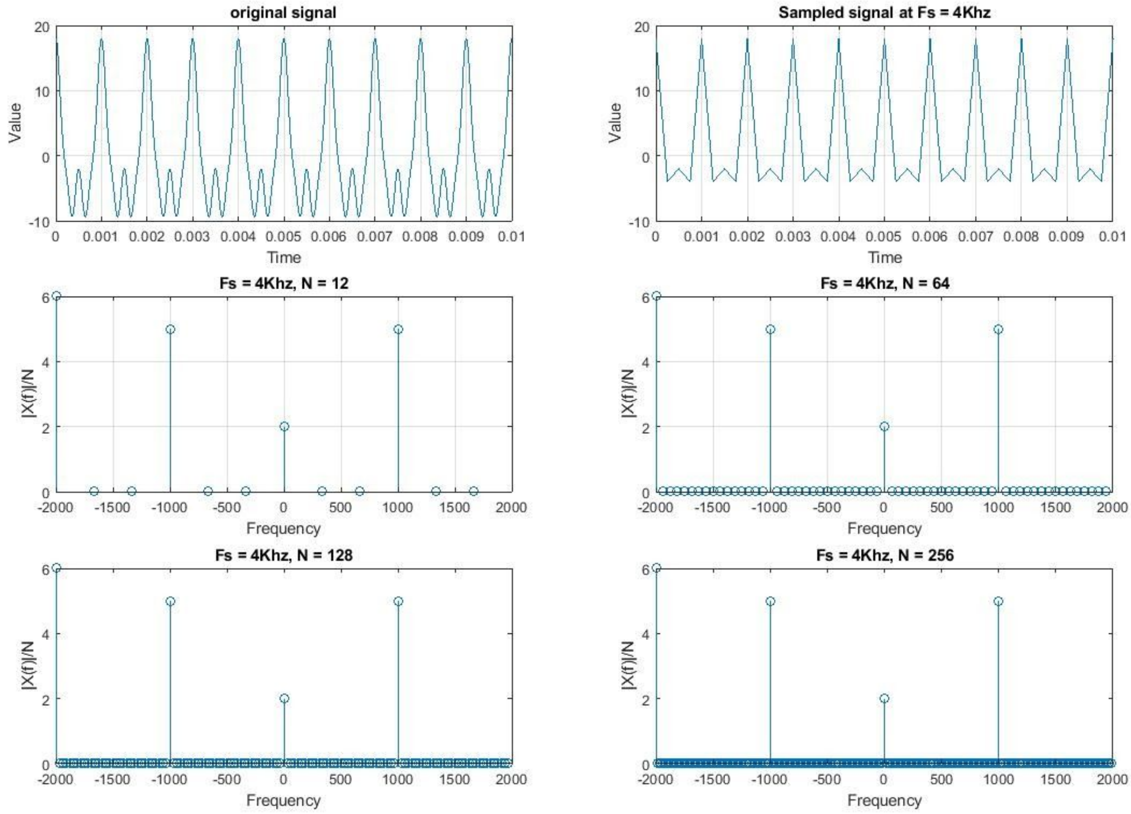
For Frequency = 8Khz



For Frequency = 5Khz



For Frequency = 4Khz



Observations:

1) As mentioned above, the minimum sampling frequency to avoid aliasing is 8Khz, so any frequency below that will result in aliasing due to which sampled signal will be distorted w.r.t original signal.
   Clearly from the above results, the sampled signal at 8Khz looks the same as the original signal but with decrease in sampling frequency the sampled signal becomes more and more distorted w.r.t original signal due to aliasing. This results validates the above theory.

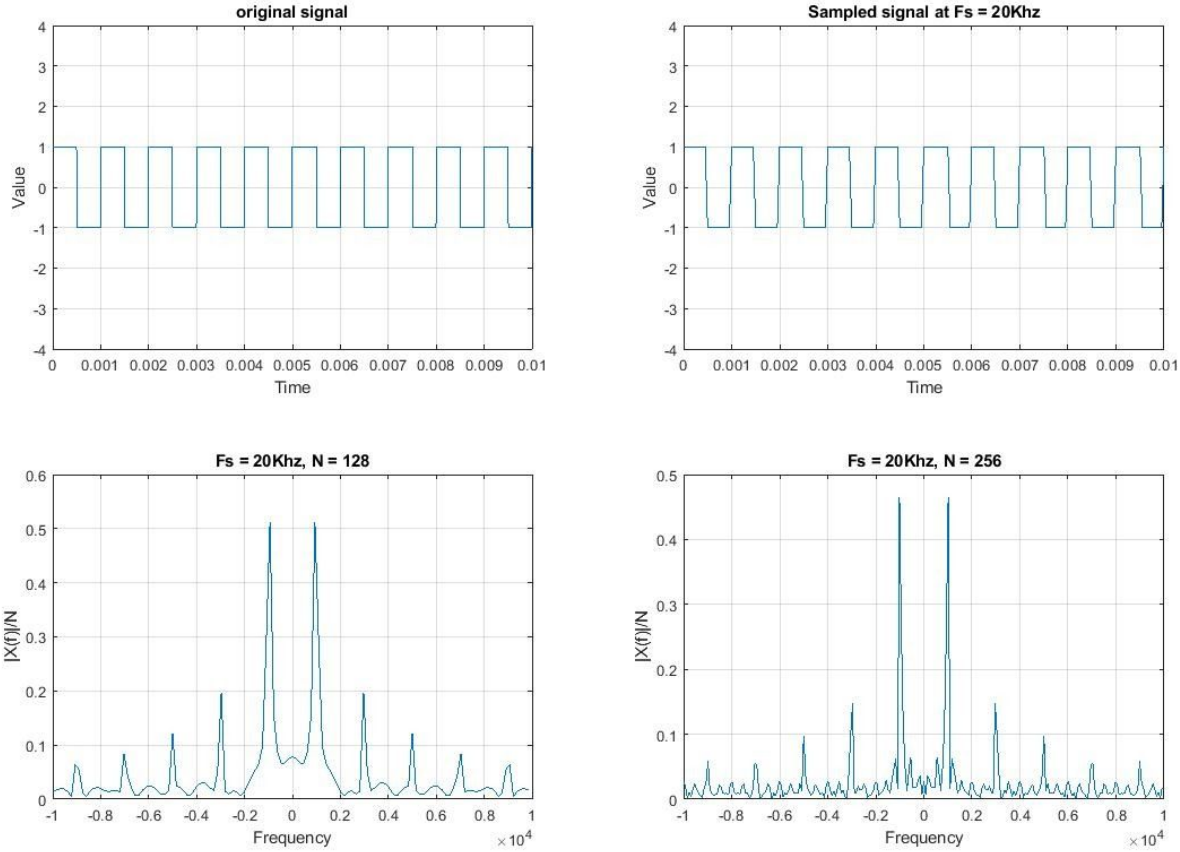
2) Reason for change in signal due to change in N is same as discussed in part a.
   It can be seen from the plot that as N increases number of points sampled in the frequency domain increases for all cases as expected from the theory.

c) Spectrum of Square Wave.

In this experiment I have plotted the square wave of frequency 1Khz, and sampled the signal at Fs = 20Khz.

Then DFT of signal is calculated, for N= 128 and N=256.

Results:



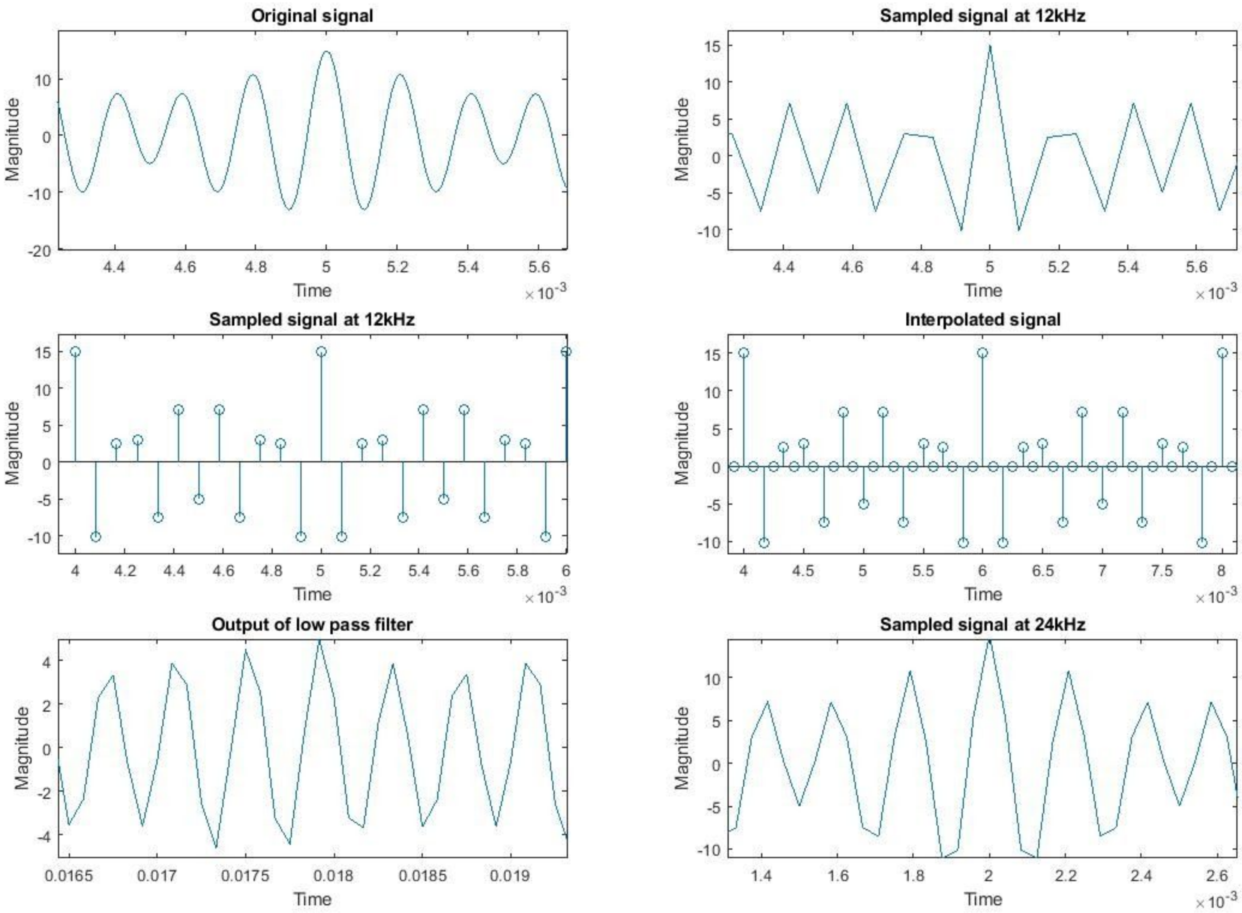
Observations:
Since ideal square waves contain infinite frequency components, so theoretically ideal sampling rate should be infinite to avoid aliasing but practically we can't do that so in this experiment we have taken Fs = 20Khz which is sufficiently large to avoid aliasing. However certain effect of aliasing can still be seen due to above reason.

Reason for change in DFT with increasing N is same as discussed in part a.
Since Square wave have all odd harmonics present of fundamental frequency, i.e 1Khz so in DFT peaks can be seen at 1, 3, 5, 7 Khz respectively, verifying the theory.

d) Interpolation or Upsampling.

1) Low pass signal with f = 5Khz used was :
   x(t) = 10*cos(2*pi*5*1000*t) + 4*cos(2*pi*4*1000*t) + 1*cos(2*pi*2*1000*t)
2) To avoid aliasing, Fs used was 12Khz.
3) Chebyshev low pass filter was used as a low pass filter with cutoff 6Khz, ripple 10dB, and order 20 to insure high roll-off rate.

Results



Observations:
Fig 1 shows the actual low pass signal, and fig 2 shows the sampled output, it is similar to the original signal as Fs was greater than 2*Fmax, however it shows the peaky nature due to the reasons discussed in part a.

Fig 3 shows the sampled signal at Fs = 12Khz and Fig 4 shows the interpolated output, it is clearly seen that there is 1 zero between every 2 peaks of sampled output which was expected.

Fig 5 shows the output of interpolated signal when passed from low pass filter and fig 6 shows the original signal sampled at 24Khz. As expected from the theory both signal was similar in nature however magnitude of low pass output was much lower due to loss of power as many frequency components are filtered out from the filter and shift in signal is also observed due to transient response.

This link contains the original Matlab code for the experiment:
https://drive.google.com/drive/folders/1991oUyEI_mlqwBgSeDIaHZq1H2arPRZW?usp=sharing