COMPENG 4TL4: Digital Signal Processing
Lab #1

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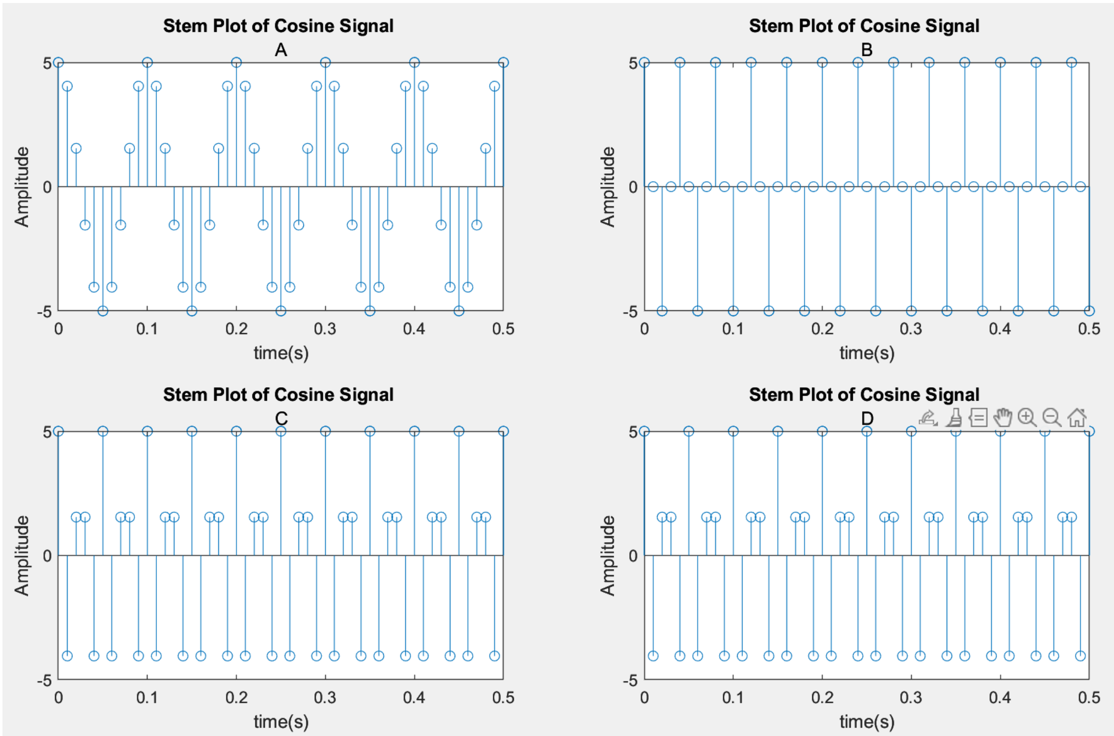
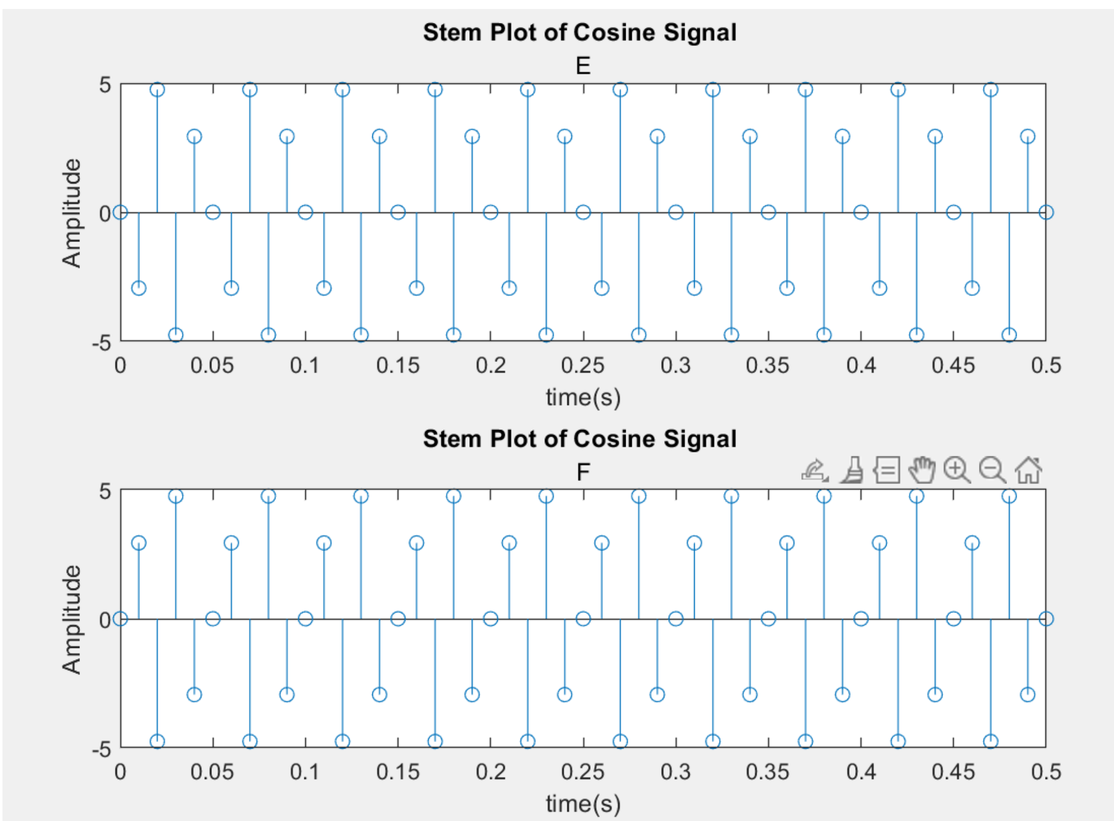
As a future member of the engineering profession, the student is responsible for performing the required work in an honest manner, without plagiarism and cheating. Submitting this work with my name and student number is a statement and understanding that this work is my own and adheres to the Academic Integrity Policy of McMaster University and the Code of Conduct of the Professional Engineers of Ontario. Submitted by **[Zhengda Li, li939, 400324486]**

Experiment 1: Point sampling of a sinusoid

(a) All signals are shown below:





(b)

Period of signal A: 0.1s

Period of signal B: 0.04s

Period of signal C: 0.05s

Period of signal D: 0.05s

Period of signal E: 0.05s

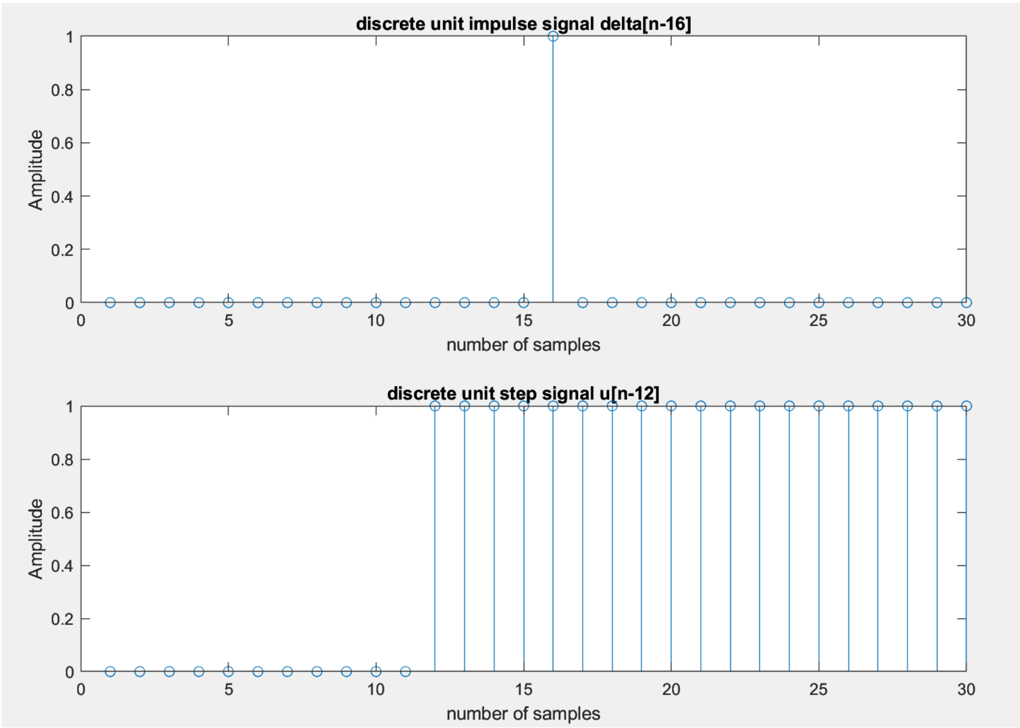
Period of signal F: 0.05s

According to Nyquist theorem the sampling frequency should be at least two times the signal frequency to be able to restore the original signal without distortion. In this experiment the sampling frequency is 100Hz, which means it can restore the signal with frequency of less than 50Hz without distortion. As a result, we can observe that signal A (10Hz) and signal B (25Hz) have the same period as the original signal. Whereas signal D and signal F, which have frequency of 60 Hz, are distorted because there is aliasing happening. Although signal C and E have frequency of 40 Hz, which is within the range of 50 Hz, but because of the sampling frequency is not the integer times signal frequency so the sampling points did not locate at signal's maximum and minimum and result in incorrect variation and period. In another words, you need to perform convolution to match the discrete signal to the continuous signal.

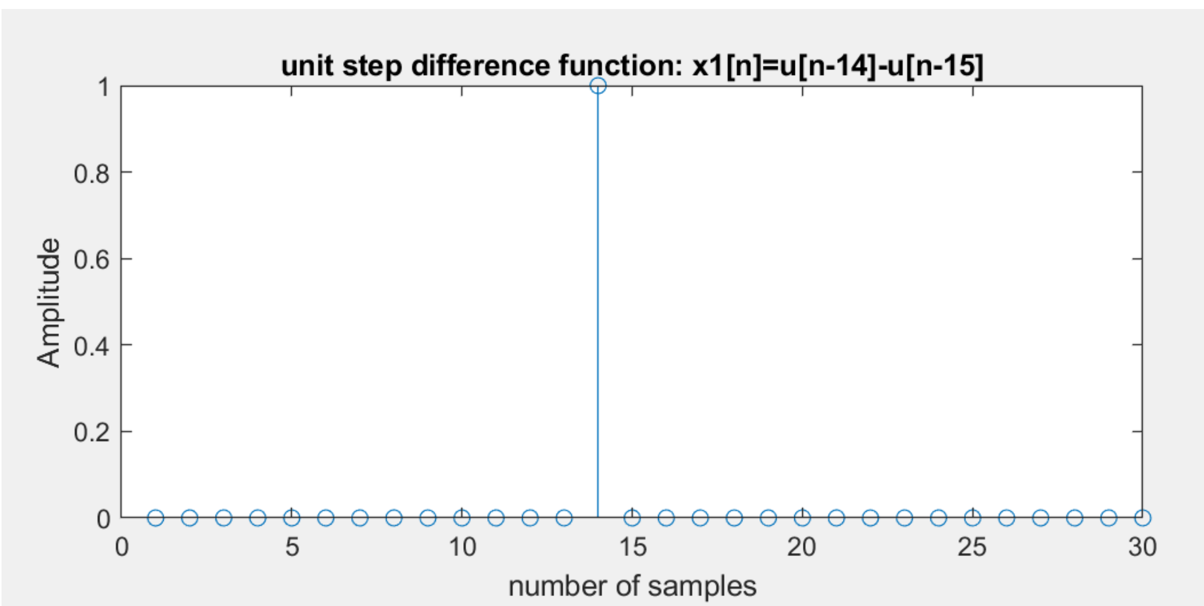
(c) We can observe that sampled signals are very different if comparing C and E or D and F. This difference is due to the phase shift and also aliasing, but the period is the same as phase shift does not impact period.

Experiment 2: Working with unit impulse and unit steps

(a)

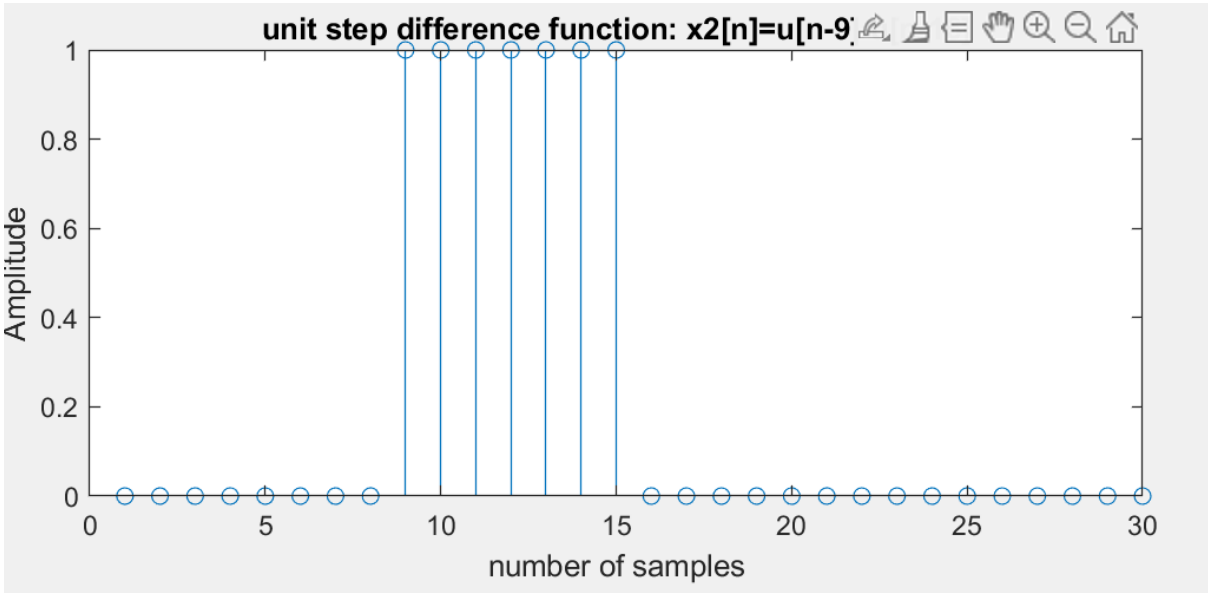


(b)



$u[n-14]-u[n-15]=\delta[n-14]$

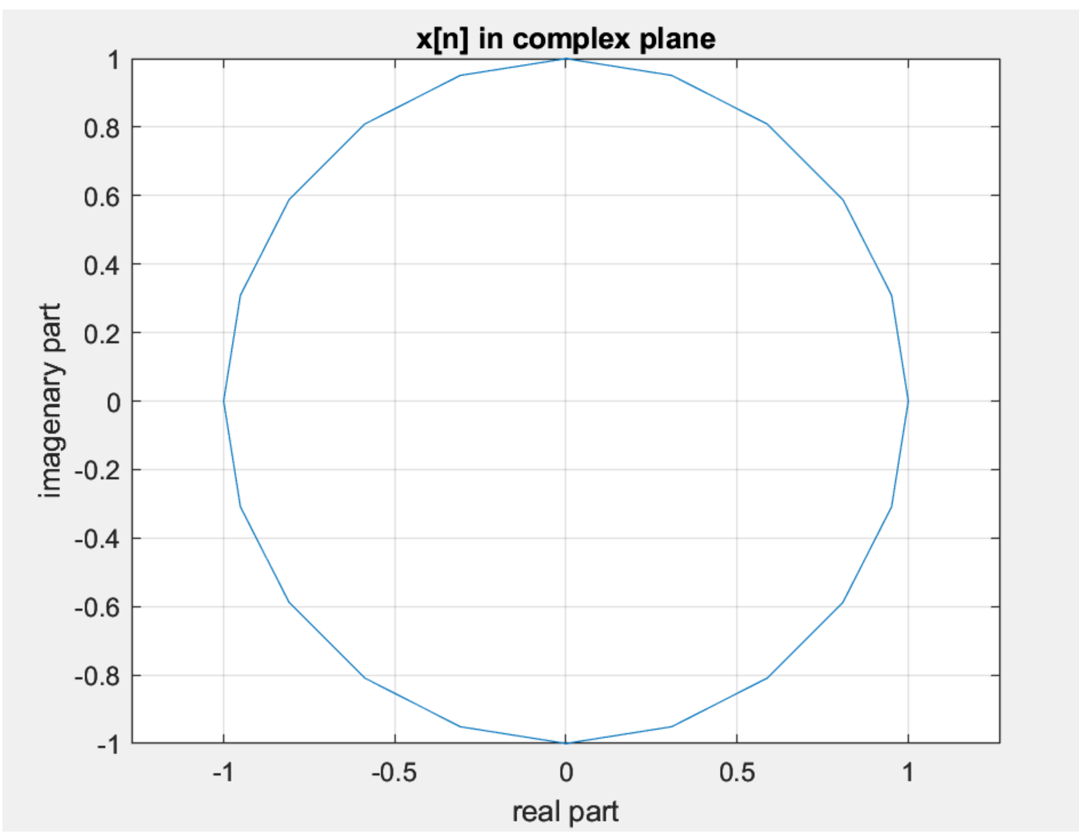
(c)



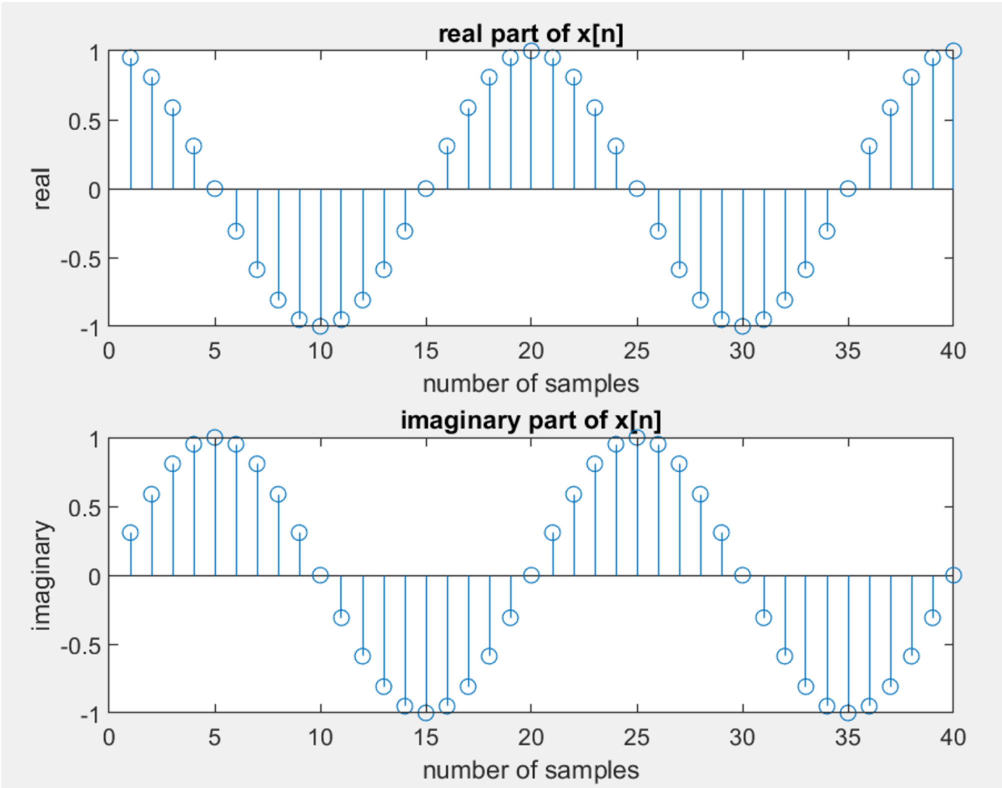
$u[n-9] - u[n-16] = \sum_{k=9}^{15} \delta[n-k]$

Experiment 3: Complex signals

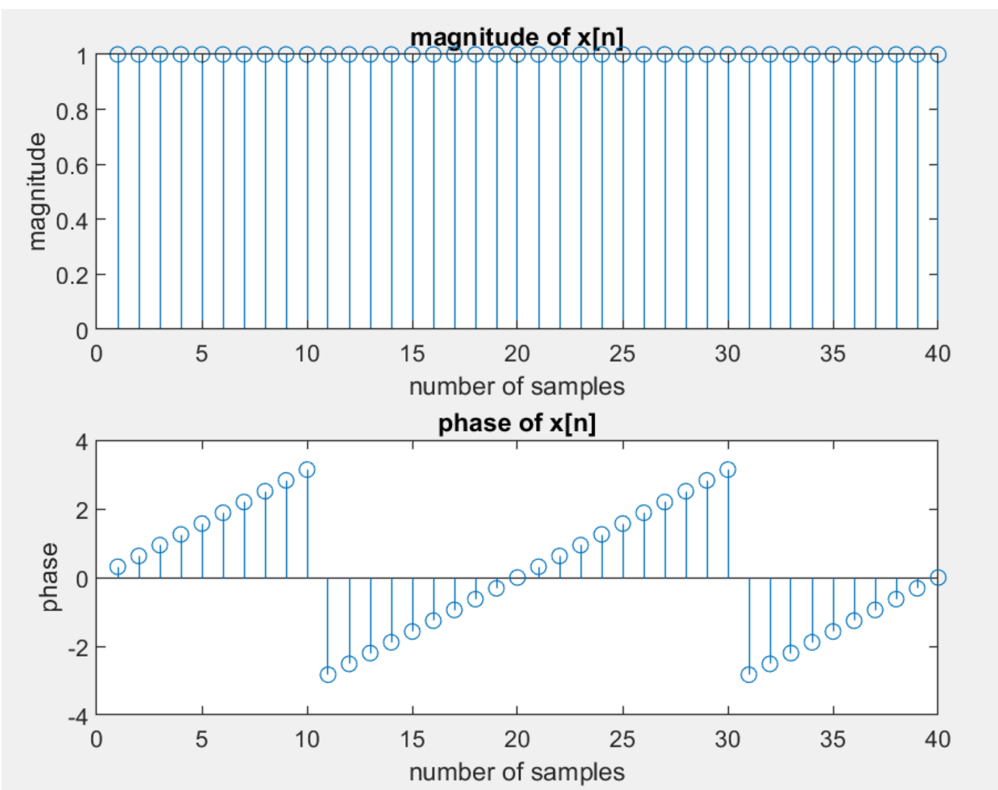
(a)



(b)


real part of x[n]
real
number of samples
imaginary part of x[n]
imaginary
number of samples

(c)

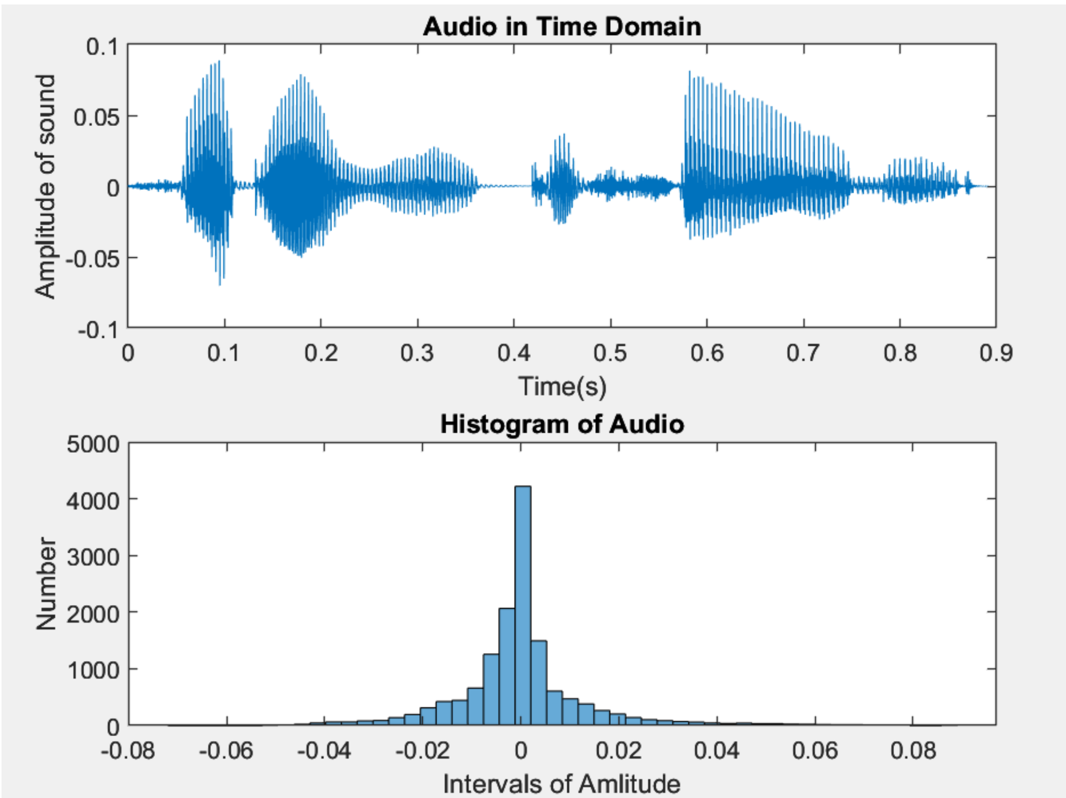

magnitude of x[n]
magnitude
number of samples
phase of x[n]
phase
number of samples

The magnitude plot shows that the energy of the signal is the same at all the time.

The slope of the phase plot is 0.31. This slope is related to the rate of change of x[n] for angular in complex plane. This tells us the period of the signal is 20.

Experiment 4: Quantization of a speech signal

(ab)



We can see the number of bits for this wav file is 16 bits by running the info command in MATLAB, and the amplitude of the signal distributed in the range of (-0.08,0.08]. we can also observe that the histogram is a Gaussian distribution.

(c)

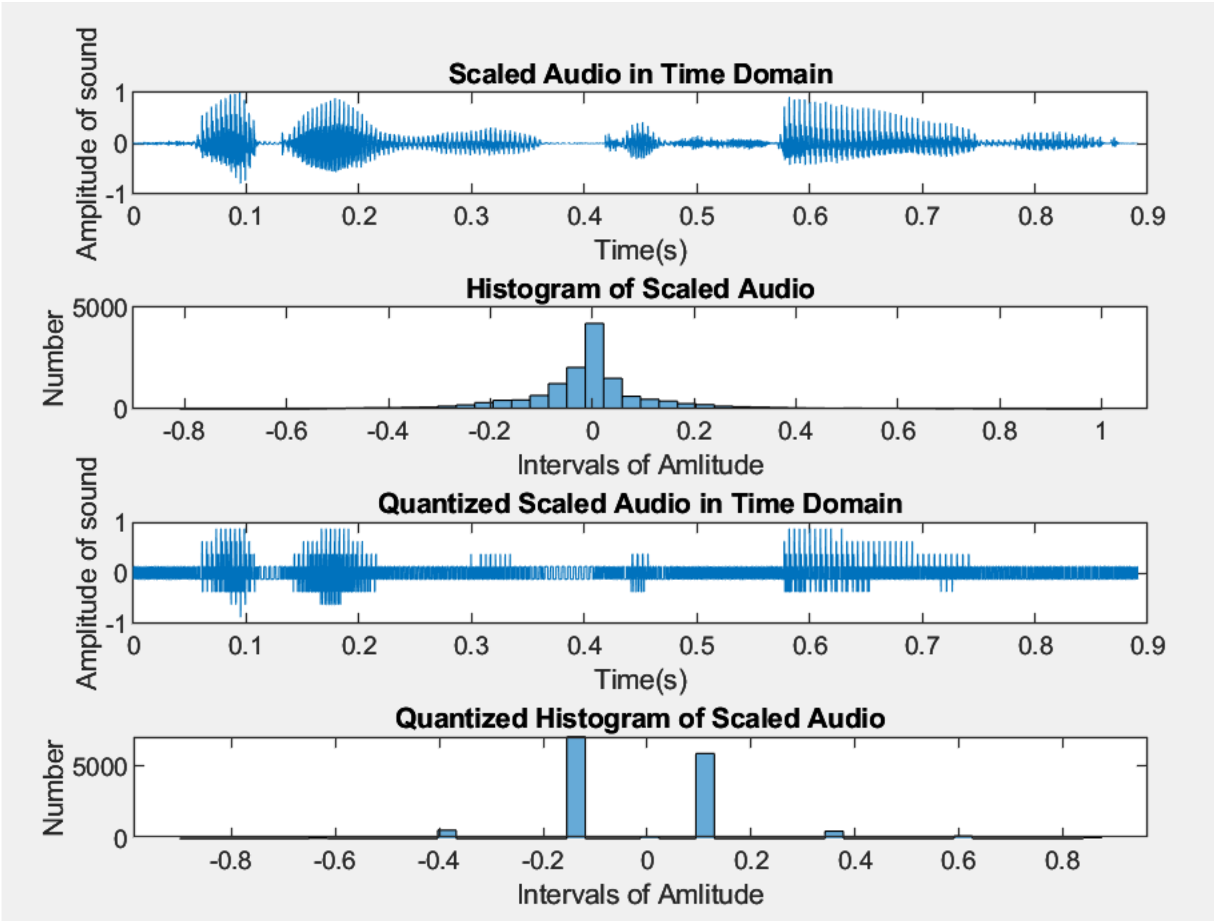
We could hear the sound very clear.

(d)

The script for quantizer is shown below:

```
function quant=quantization(y,b)
step=2/2^b;
quant=sign(y).*(2*floor(abs(y)/step)+1)*step/2;
quant(find(quant>1)) = 1-step/2;
quant(find(quant<-1)) = -1+step/2;
```

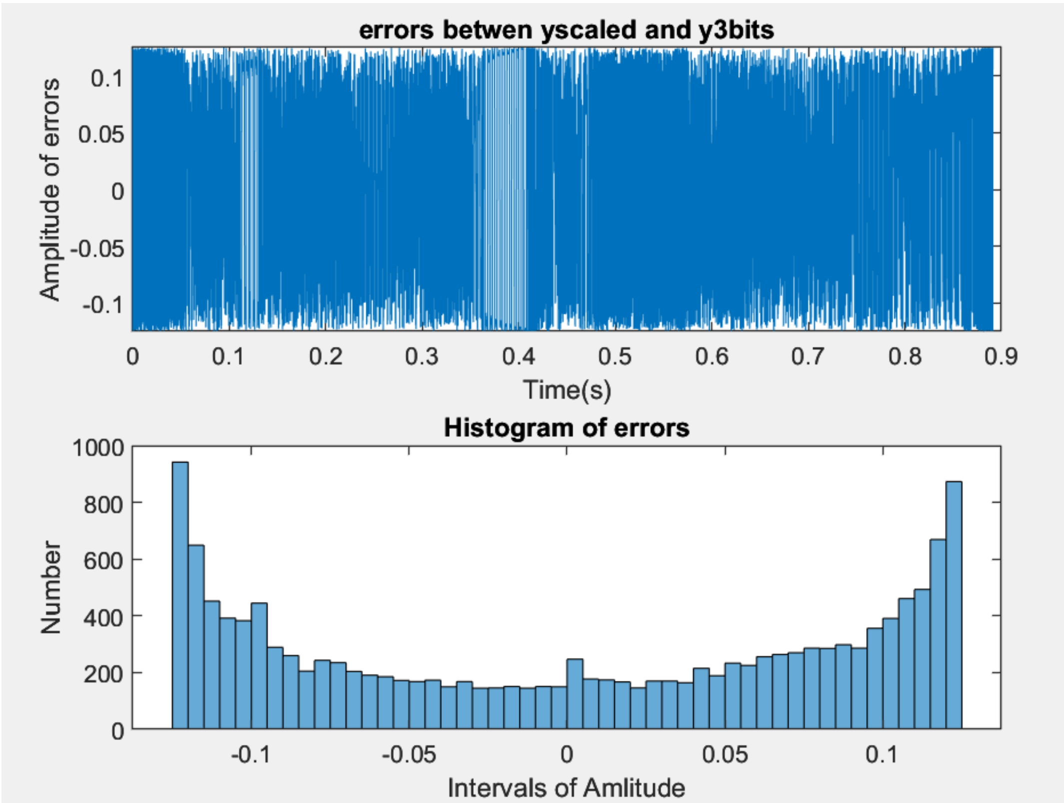
(ef)



The magnitude of the signal is within the range of (-1,1) after the scaling. The audio is quantized by 3 bits and theoretically it should have only 8 values. However, it only has 5 values in the histogram chart which is likely due to quantization algorithm quantizing all values into 5 numbers and no number goes into the rest 3 values. It is not the most accurate quantization algorithm which has been point out in the demo.

The audio after quantization does not sound as clear as before and there is noise in the audio. This is due to the quantization has only 8 intervals from -0.875 to 0.875 and most points distributed around 0.

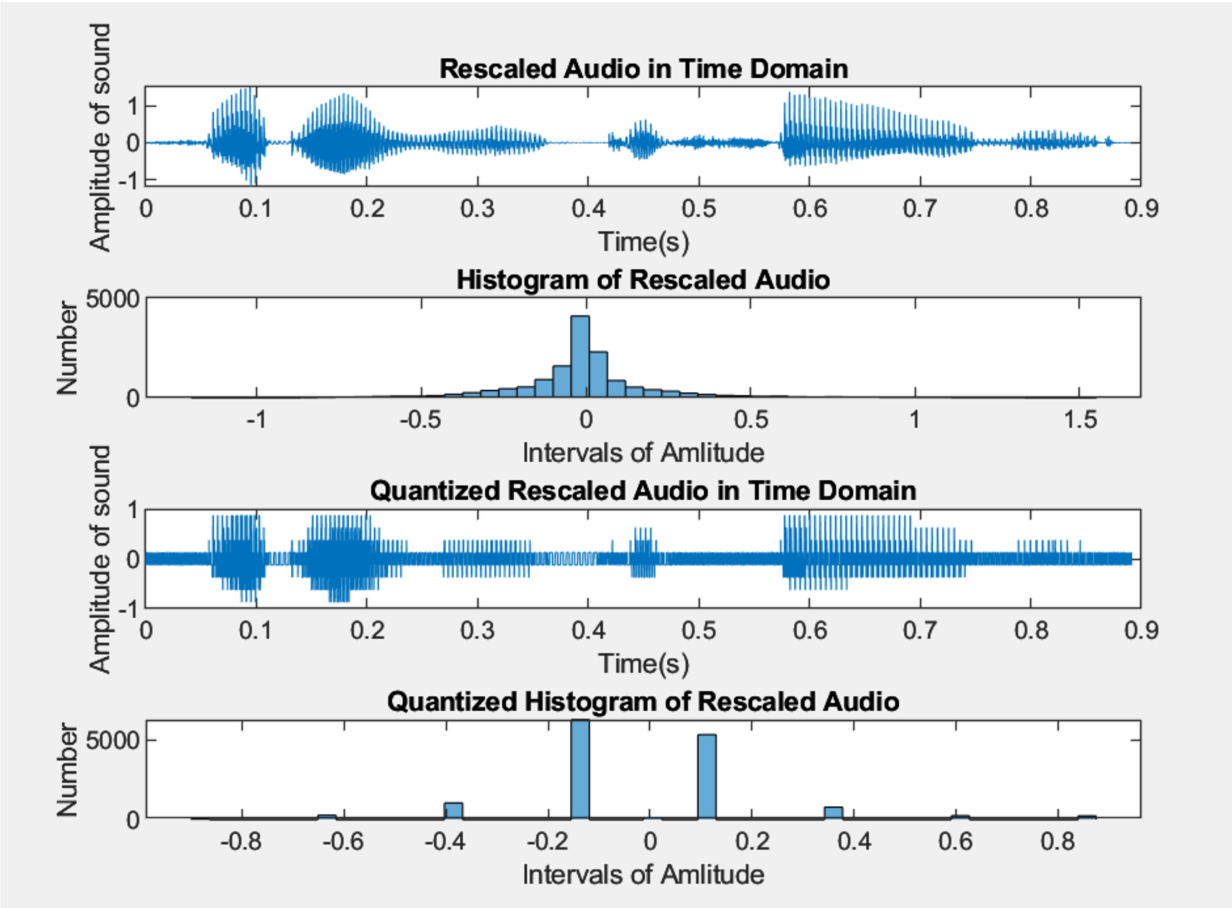
And then we look at the errors of quantization:



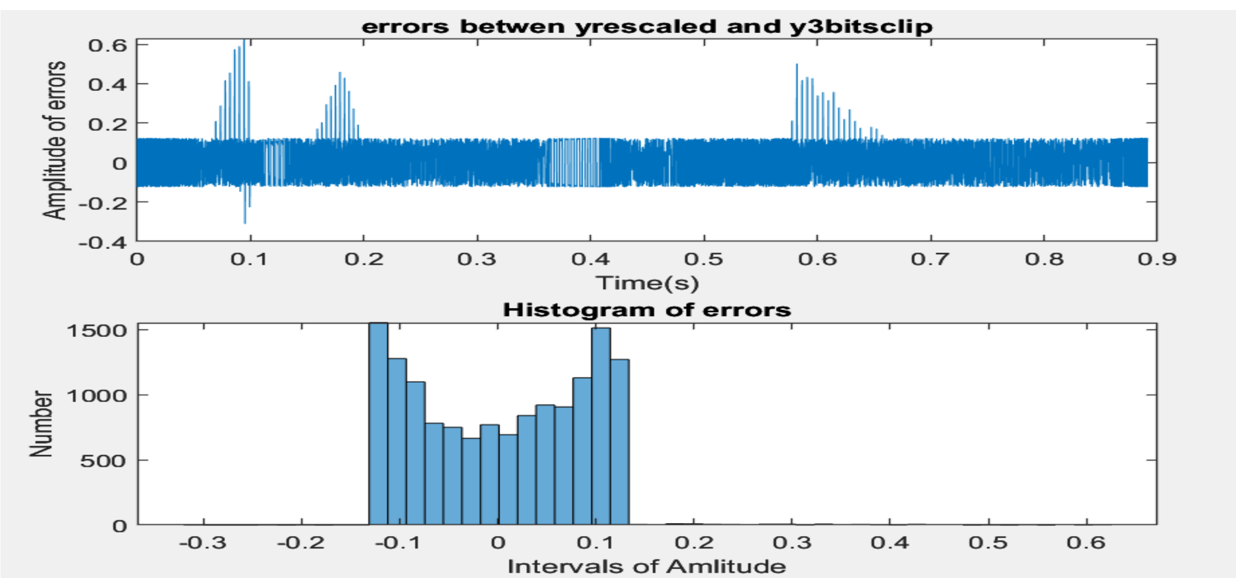
We can observe from the graph of error that the error is uniformly distributed on (-0.25, 0.25), which is obtained by $\frac{1-(-1)}{2^3} = 0.25$.

Then we can find the mean and variance of the error by using mean and var command in the MATLAB. The mean of the error is 0.0025, which is close to 0. And the variance of the error is 0.0076 which is close to $\frac{0.25^2}{12} = 0.0052$ thus fit standard statistical model of quantization noise.

(g)



We rescale the signal and make the amplitude to exceeds the range of (-1,1). After the quantization we could see the signal is clipped but the histogram is similar to the previous quantized signal. The audio sounds unclear like in the last part, but there is more noise compared to the last part.



Due to the clipping, it can be seen that some errors are not uniformly distributed like before. However, the histogram is similar to the previous one, it looks not uniformly distributed because of the larger range of x-axis. It is still uniformly distributed given the same x-axis range.

The mean of the error increase to 0.0037 and the variance is 0.0076 which is the same as before given the similar histogram.