

# **Department of Electrical Engineering**

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Course/Section: BEE-6B

Semester: 4<sup>th</sup> semester

## **EE-232 Signals and Systems**

### Lab #9: Sampling & Reconstruction

Name	Reg. no.	Report Marks / 10	Viva Marks / 5	Total/15
Saad Iqbal	32903			

### **Objectives:**

To introduce the students to the concepts of Sampling and Reconstruction of Continuous Time Signals.

- Introduction to Sampling and Reconstruction Theory
- Sampling of Continuous Time Signals in Matlab
- Reconstruction of Continuous Time Signals from Sampled Signals
- Demo of sampling and aliasing

## Pre-Lab:

You should perform the following steps with the con2dis GUI:

1. Set the input to  $x(t) = \cos(20\pi t)$

Done.

2. Set the sampling rate to  $f_s = 50$  samples/sec.

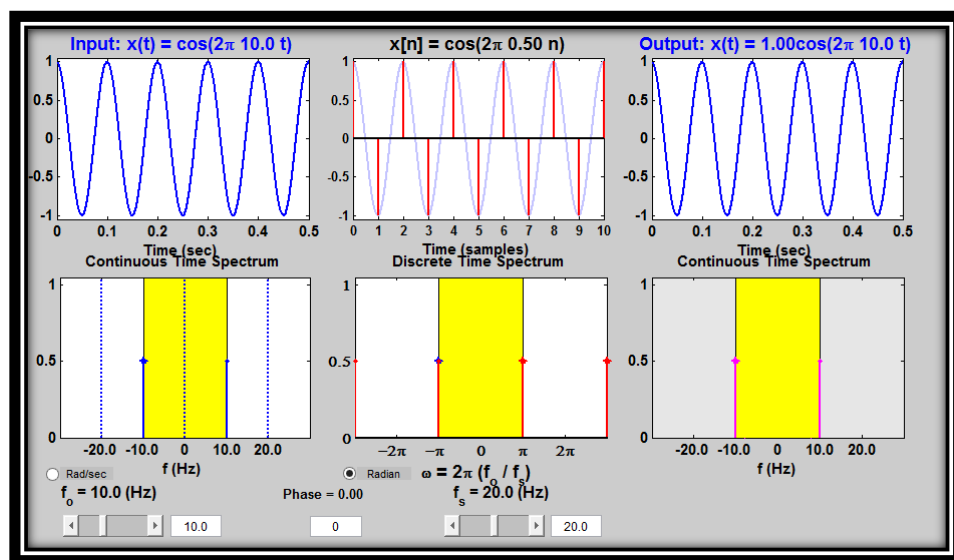
Can't possible because, maximum allowance sampling frequency is 30Hz.

3. Determine the locations of the spectrum lines for the discrete-time signal,  $x[n]$ , found in the middle panels. Click the Radian button to change the axis to from  $f$  to  $\omega$ .

Done.

4. Determine the formula for the output signal,  $y(t)$  shown in the rightmost panels. What is the output frequency in Hz?

$$\begin{aligned} Y(t) &= 1.00 \cos(\pi f_s - f) & \text{For, } f_s < 2f \\ Y(t) &= 1.00 \cos(2\pi f) & \text{For, } f_s > 2f \end{aligned}$$

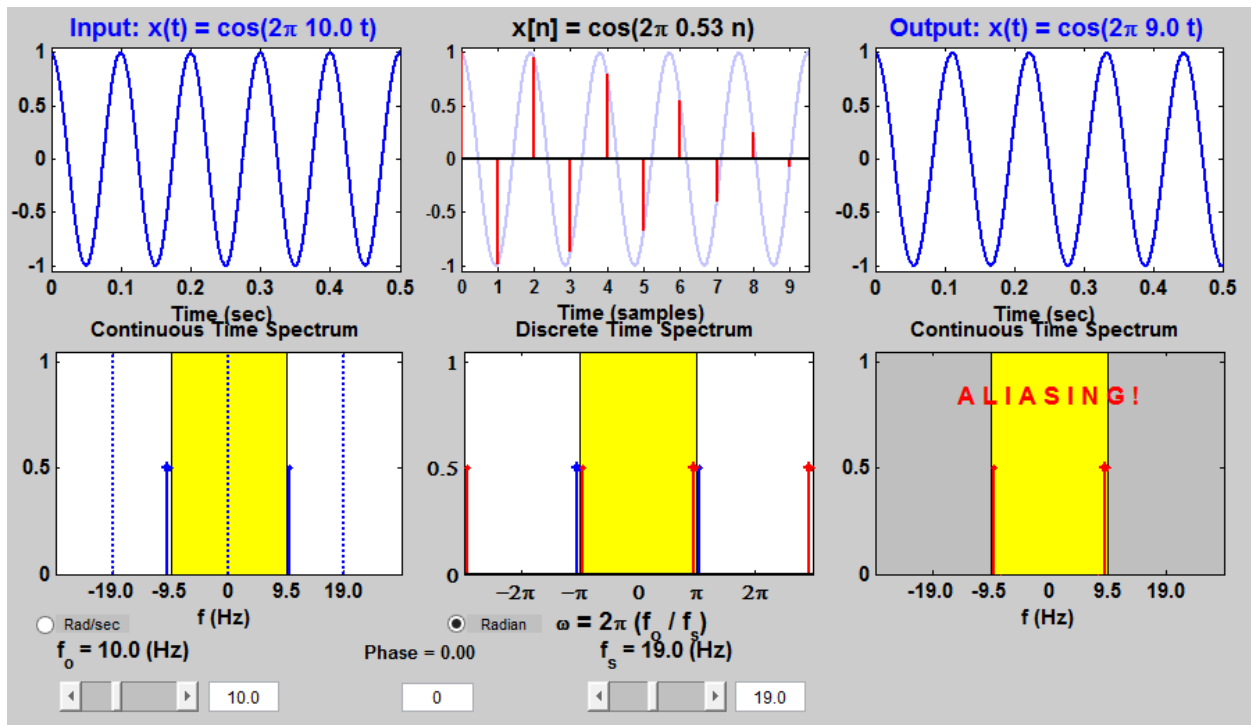


Demo of Con2dis

5. How to find the sampling frequency such that we will not be having an aliased output?

In this example,  $f_s > 20$   $f_s > 2f$

6. Show the aliased output by changing sampling frequency?



Aliased output

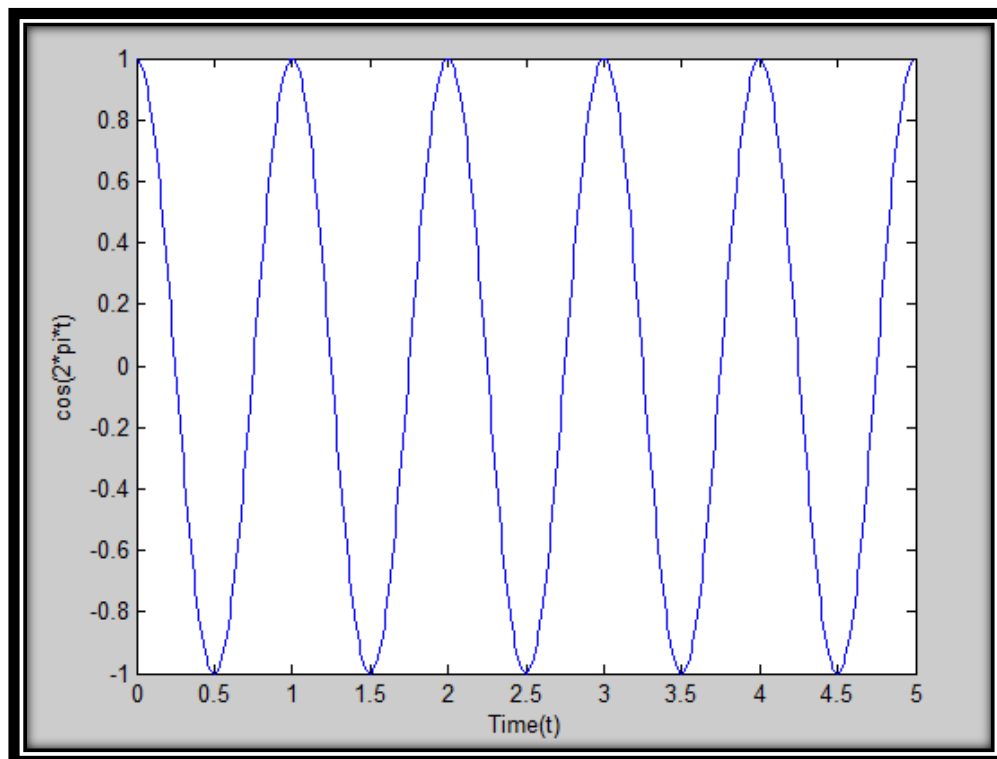
## Lab Task:

1. Assume a continuous time sinusoidal signal with frequency 1 Hz. Since Matlab can only handle discrete sample signals we will assume that the signal having 100 samples per cycle is a continuous time signal.

## Matlab code:

```
signal_frequency=1;  
Data_frequency=100;  
Increment=1/100;  
time=0:Increment:5/signal_frequency-Increment;  
x=cos((2*pi*signal_frequency*time));  
plot(time,x)  
xlabel('Time(t)')  
ylabel('cos(2*pi*t)')
```

## Matlab graph:



**Labtask1**

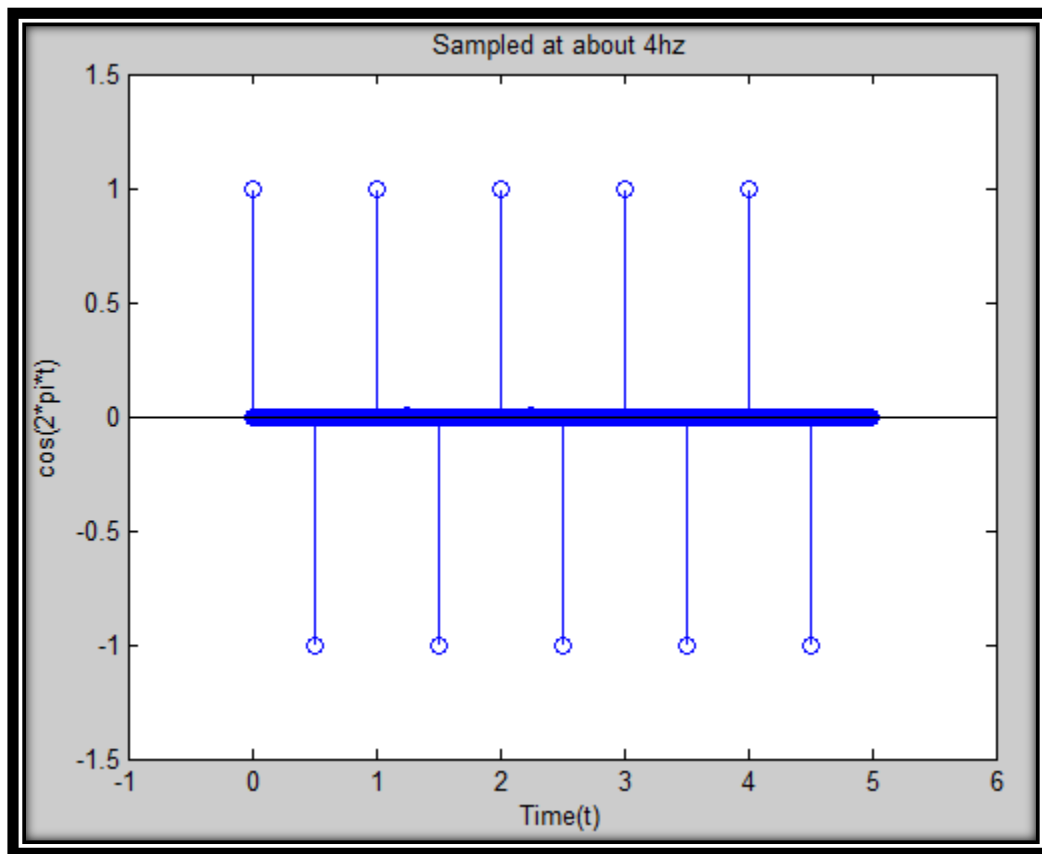
2. Assume that sampling frequency is 4 times the highest frequency in the above generated signal. Only pick appropriate evenly spaced samples, starting from the first sample. The code given below may help you in identifying which samples to pick.

### Matlab code:

#### Previous code

```
.  
.   
.   
D_Sample_frequency=4;  
Sample_increment=floor((1/D_Sample_frequency)*(Data_fre  
quency));  
Time_length=length(time);  
samples_data=zeros(1,Time_length);  
for i=1:Sample_increment:Time_length  
samples_data(i)=x(i);  
end  
stem(time,samples_data)  
axis([-1 6 -.5 1.5])  
title('Sampled at about 4hz')  
xlabel('Time(t)')  
ylabel('cos(2*pi*t)')
```

Matlab graph:



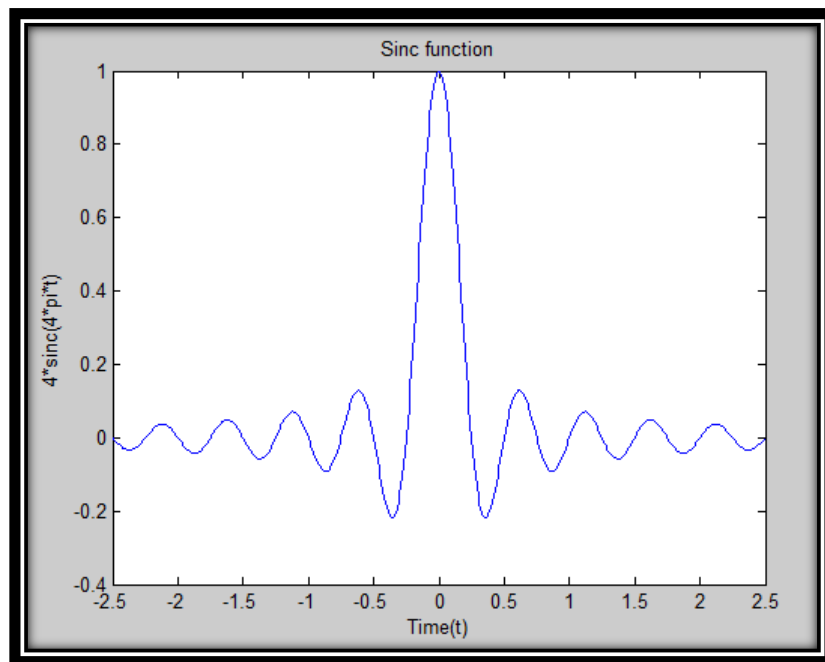
**Labtask2**

3. Since filtering with a low pass filter in frequency domain means convolution with a sinc in time domain construct a sinc function using the code given below:

Matlab graph:

```
.  
.   
.   
sinc_scale= -2.5:Increment:2.5;  
plot(sinc_scale,sinc(4*sinc_scale))  
title('Sinc function')  
xlabel('Time(t)')  
ylabel('4*sinc(4*pi*t)')
```

Matlab code:



**Labtask3**

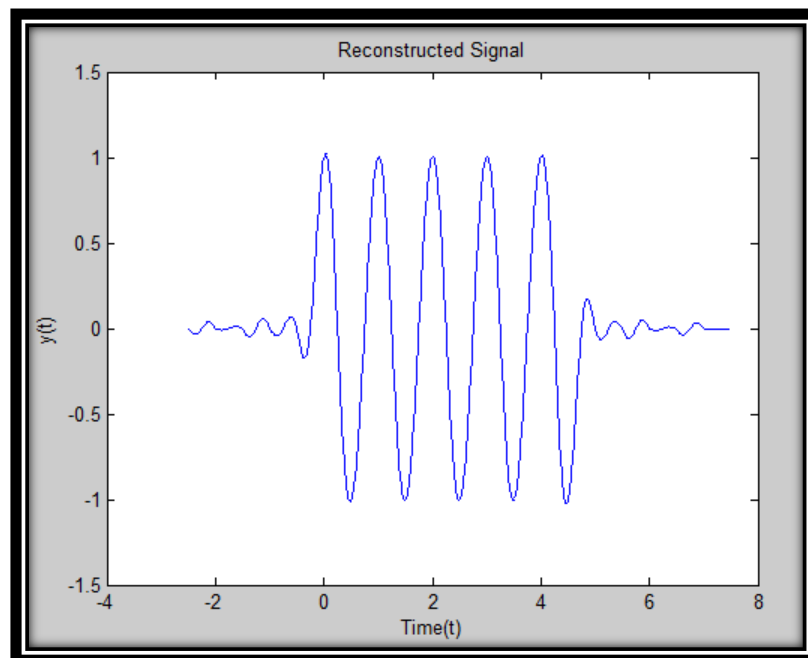
Because of sample frequency =4Hz and amplitude 4 is just a scaling factor.

4. Convolve sinc with sampled signal. Plot the resultant continuous time reconstructed signal.

#### Matlab code:

```
.  
.br/>reconstruct=conv(sinc(4*sinc_scale),samples_data);  
n_scale=-2.5:Increment:signal_frequency*5+2.5-  
Increment;  
plot(n_scale,reconstruct)  
title('Reconstructed Signal')  
xlabel('Time(t) ')  
ylabel('y(t) ')
```

#### Matlab graph:



**Labtask4**

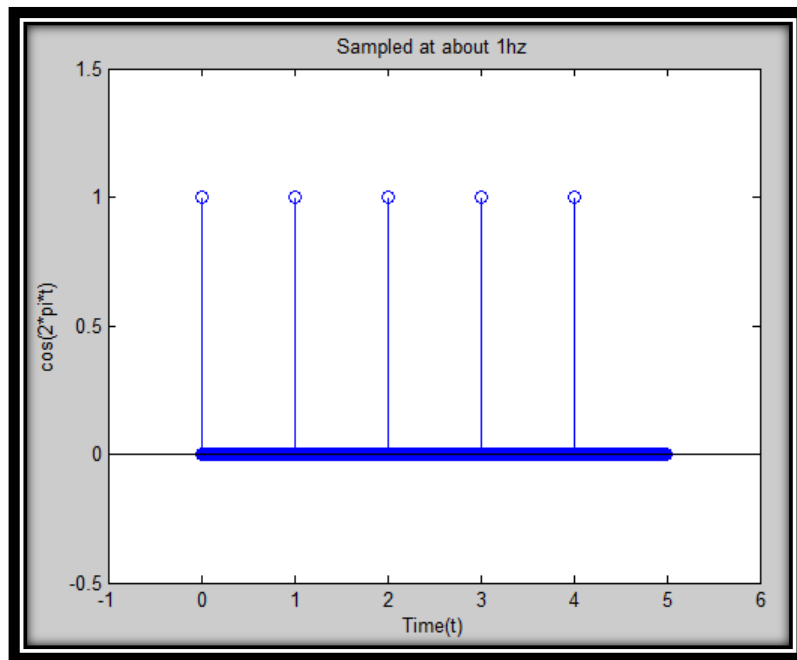
Reconstructed signal start from -2.5 because of sinc lower time bound. In reality, reconstructed signal is delayed by half of sinc time period used in filtering.



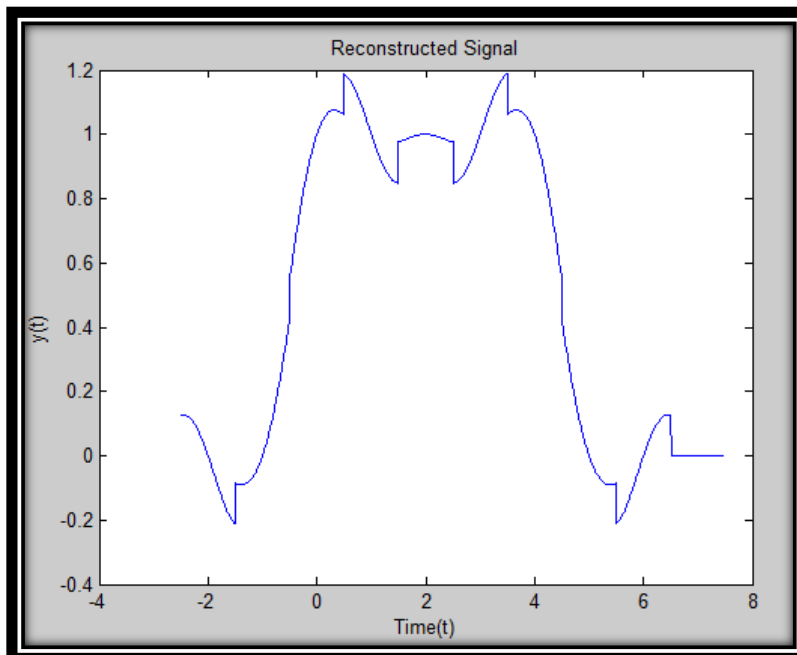
5. What happens when sampling frequency is set equal to:

Using same procedure steps, but change sample frequency and sinc frequency to demanding frequency:

1) Signal Frequency

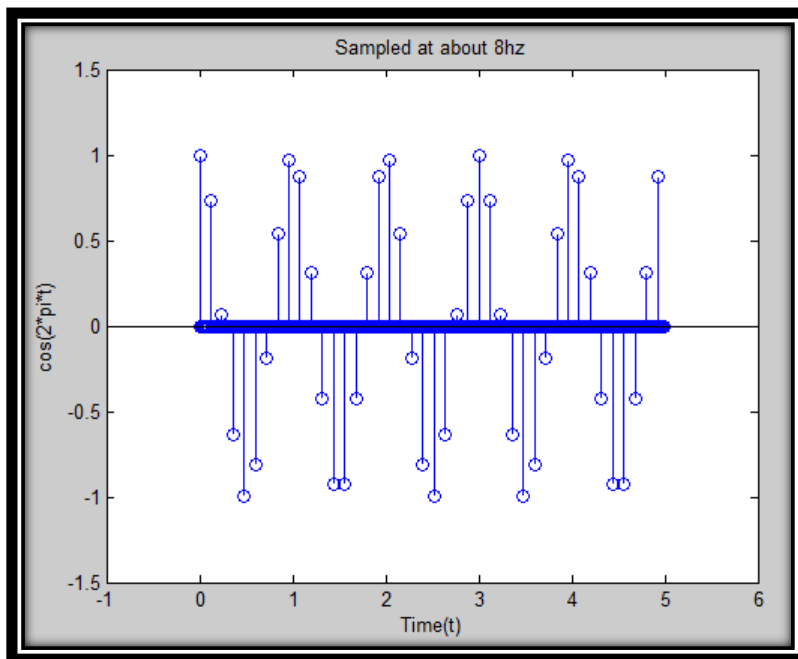


**After sample**

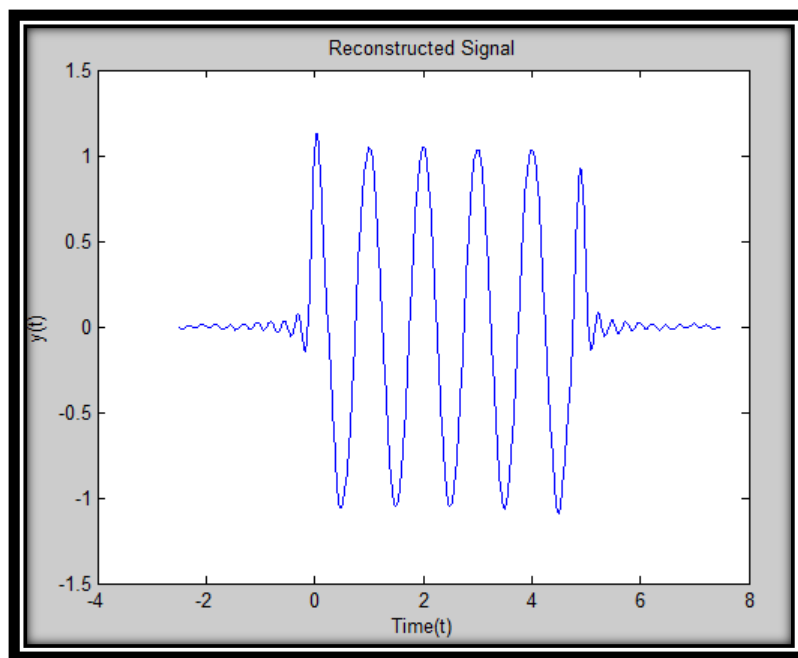


## Reconstructed signal

2) 8 times the signal frequency

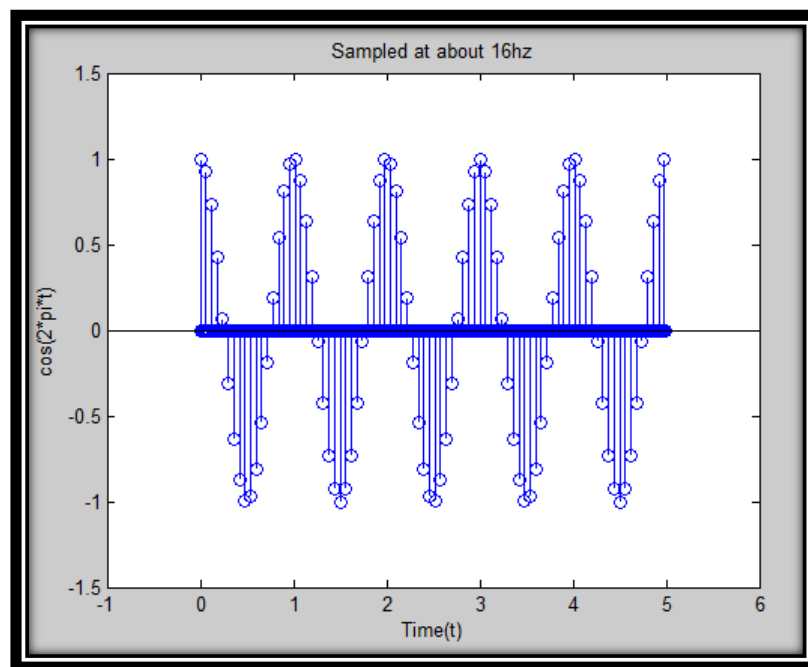


After sample

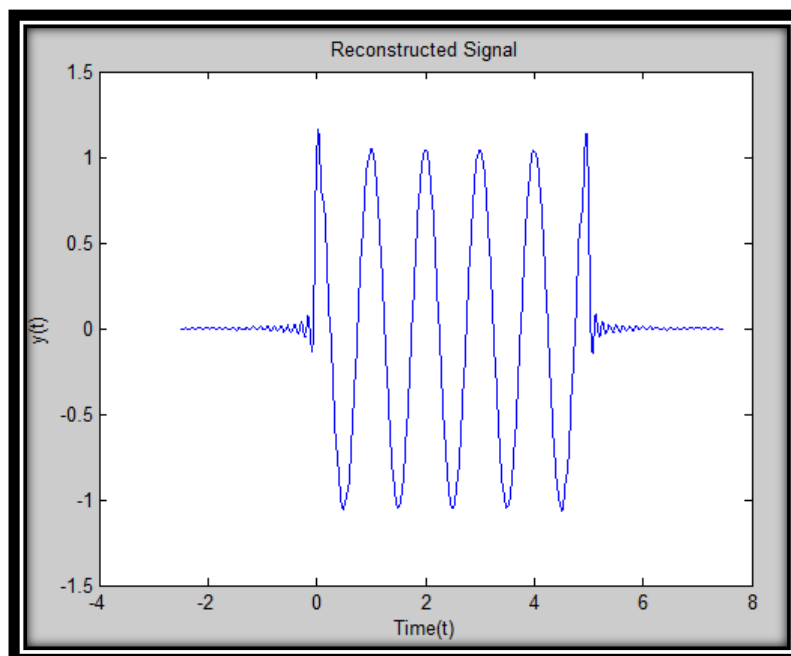


Reconstructed signal

3) 16 times the signal frequency.



**After sample**



**Reconstructed signal**

**Conclusion:**

In this lab, we learnt sampling and its significance in signal processing.