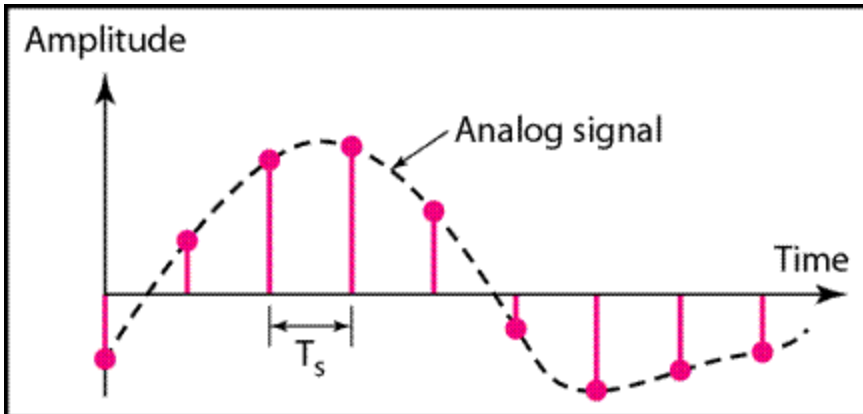


Sampling

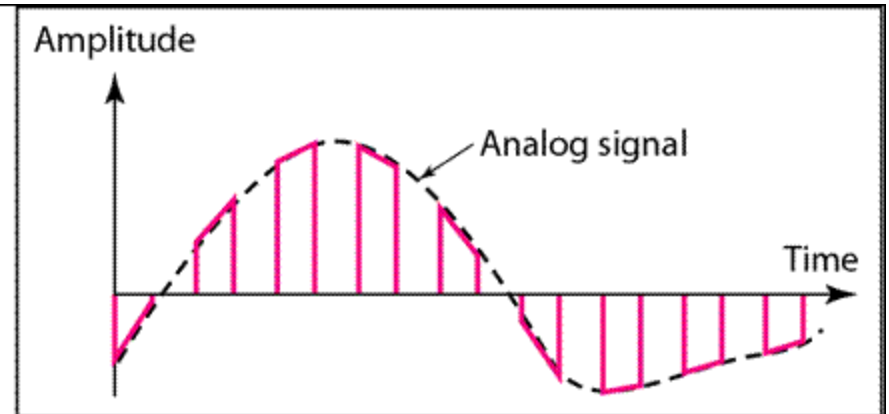
What is Sampling?

- In signal processing, sampling is the reduction of a continuous signal to a discrete signal.
- The sampling frequency or sampling rate, f_s , is the average number of samples obtained in one second (samples per second), thus $f_s = 1/T$.
- In practice, the continuous signal is sampled using an analog-to-digital converter (ADC), a device with various physical limitations.
- In effect, the system commonly referred to as digital is in fact a discrete-time, discrete-level analog of a previous electrical analog.

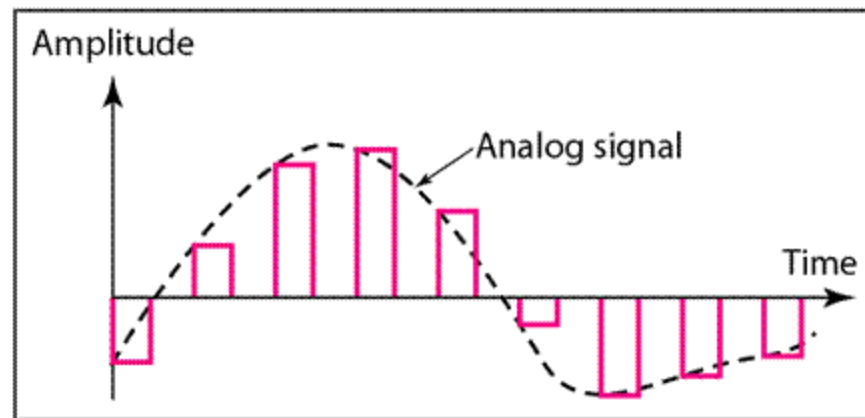
Types of Sampling



a. Ideal sampling



b. Natural sampling



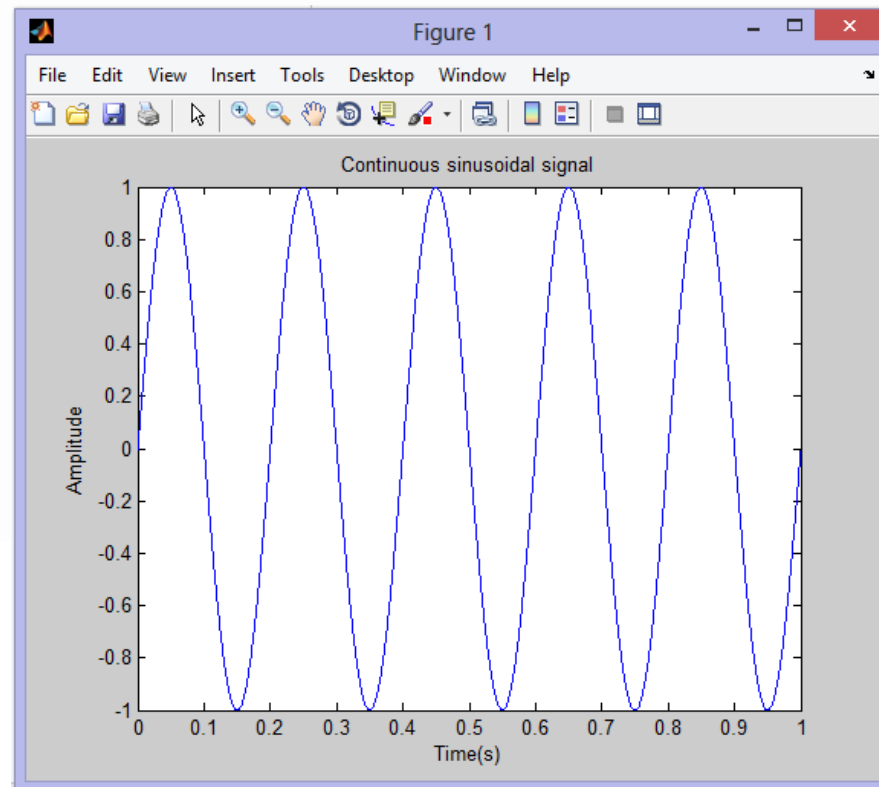
c. Flat-top sampling

- According to **Shannon Theorem**, a signal can be accurately reconstructed from its samples if the samples are taken at a frequency greater than twice that of the highest frequency present in the signal.

$$f_s \geq f_{\max}$$

Generating a Continuous Signal

```
1 fs=5000; %sampling rate 5 kHz
2 f=5; %Frequency of sinusoid
3 t=0:1/fs:1; %time index
4 x=sin(2*pi*f*t);
5 plot(t,x)
6 title('Continuous sinusoidal signal');
7 xlabel('Time(s)');
8 ylabel('Amplitude');
```



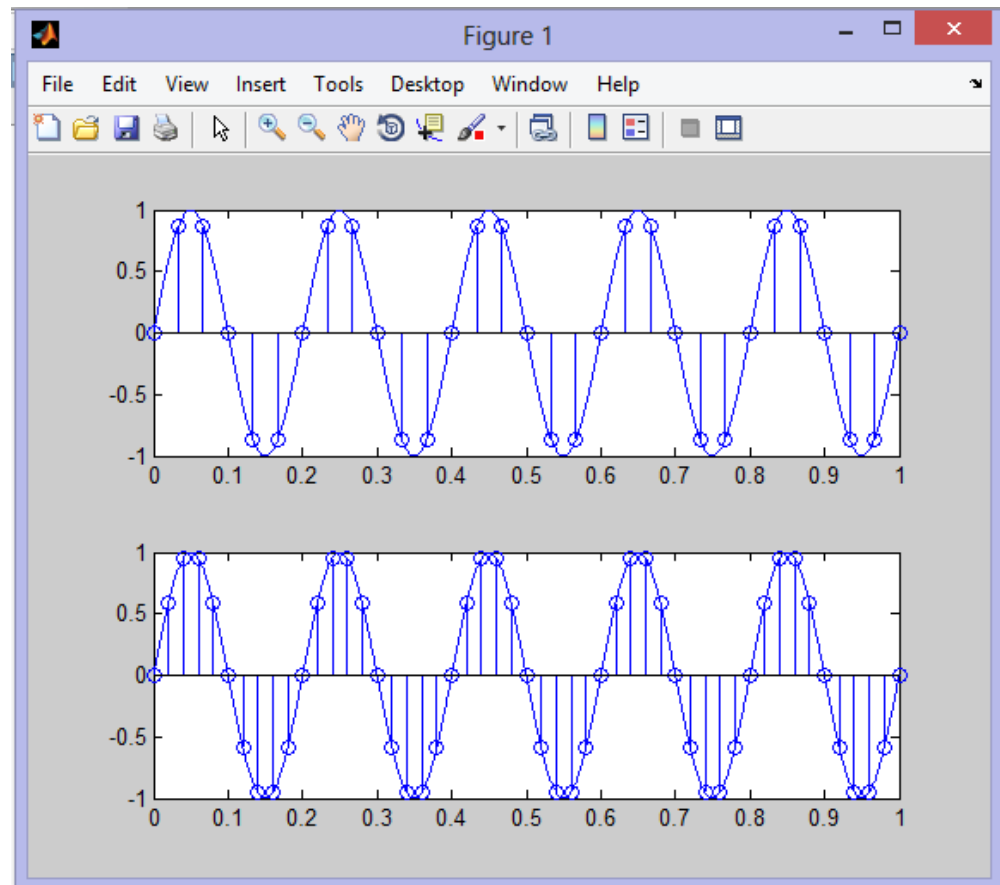
Sampling the Continuous Sinusoid (At the Nyquist Rate)

$F=5$; %Frequency of sinusoid
%The Shannon Rate for this signal is 10 Hz

$f_{s1}=30$; %30Hz sampling rate
 $t_1=0:1/f_{s1}:1$; %time index
 $x_1=\sin(2\pi f t_1)$;

$f_{s2}=50$; %50Hz sampling rate
 $t_2=0:1/f_{s2}:1$; %time index
 $x_2=\sin(2\pi f t_2)$;

`subplot(2,1,1);`
`plot(t1,x1);`
`hold on;`
`stem(t1,x1);`
`subplot(2,1,2);`
`plot(t2,x2);`
`hold on;`
`stem(t2,x2);`



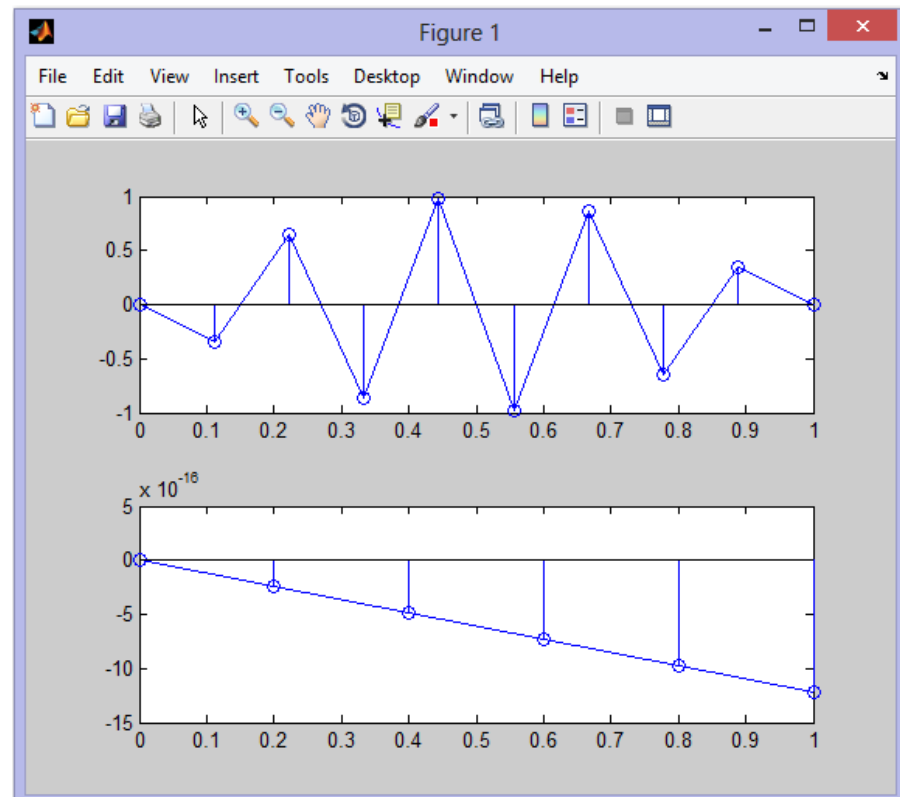
Sampling the Continuous Sinusoid (Below the Nyquist Rate)

$F=5$; %Frequency of sinusoid
%The Shannon Rate for this signal is 10 Hz

$fs_1=30$; %30Hz sampling rate
 $t_1=0:1/fs_1:1$; %time index
 $x_1=\sin(2*\pi*f*t_1)$;

$fs_2=50$; %50Hz sampling rate
 $t_2=0:1/fs_2:1$; %time index
 $x_2=\sin(2*\pi*f*t_2)$;

```
subplot(2,1,1);  
plot(t,x);  
hold on;  
stem(t1,x1);  
subplot(2,1,2);  
plot(t,x);  
hold on;  
stem(t2,x2);
```

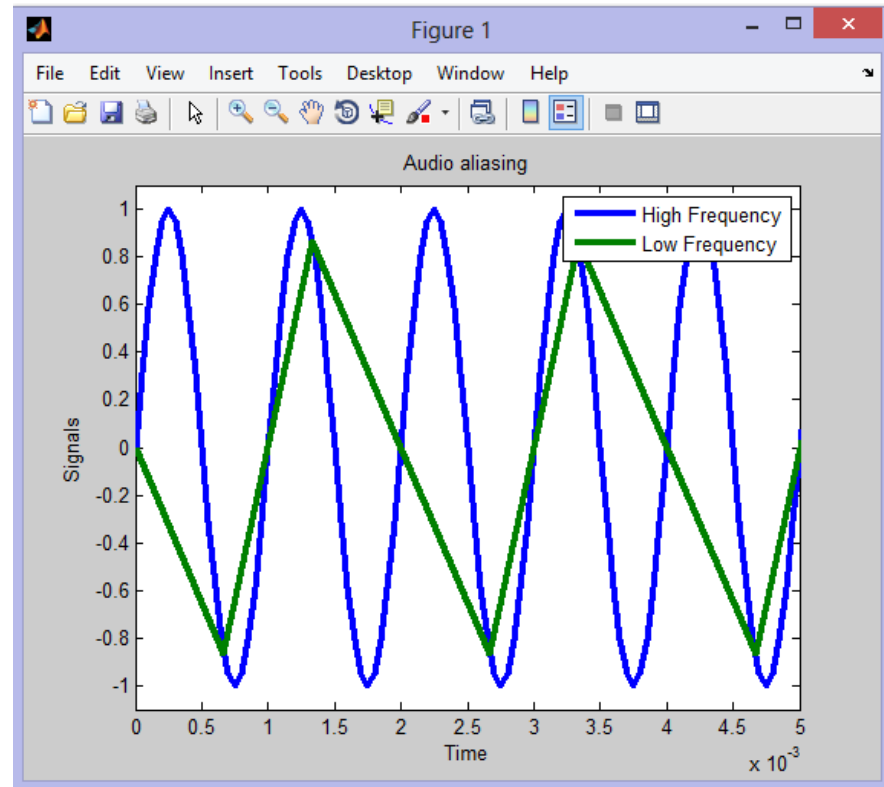


Audio Sampling

```
f=1000; %Frequency of sinusoid
%The Shannon Rate for this signal is 10 Hz
```

```
fs1=20000; %30Hz sampling rate
t1=0:1/fs1:2; %time index
x1=sin(2*pi*f*t1);
```

```
fs2=1500; %50Hz sampling rate
t2=0:1/fs2:2; %time index
x2=sin(2*pi*f*t2);
figure;
plot(t1,x1,t2,x2,'LineWidth',3.0),
axis([0, 0.005, -1.1, 1.1])
legend('High Frequency','Low Frequency')
xlabel('Time')
ylabel('Signals')
title('Audio aliasing');
%%
soundsc(x1,fs1)
%%
pause(3)
soundsc(x2,fs2)
```



Interpretation of this listening?

From sampling theorem, sampling frequency should be twice more than the signal frequency. From the data above, we have two sampling frequencies. For 1500 Hz sampling frequency, the maximum spectrum that can be heard is 750 Hz signal. As for 20000 Hz, the maximum frequency is 10000 Hz. Since the sound's initial frequency is 1000 Hz, it cannot be heard with original frequency for 1500 Hz sampling rate, therefore it is heard as 750 Hz signal. As for 20000 Hz sampling frequency, there is no problem. It is heard as original sound, That's why, there is slightly difference in the sounds produced.

Lab Task

1. Write a MATLAB function **[x, t] = sin_NU(f0, fs, T)** to generate a sine signal. The output parameters **x** and **t** are the signal and time vectors, respectively. The input parameters are **f0** (signal frequency in Hz), **fs** (sampling frequency in Hz), **T** (signal duration in sec.).

2. Using the **sin_NU** function you just made:
 - (a) Generate two 1 kHz sine signals (2 seconds duration), first signal at 20 kHz sample frequency and second signal at 1.5 kHz sample frequency;
 - (b) On the same graph, use the plot function to display the two signals versus t in the range $0 \leq t \leq 5$ msec.;
 - (c) Listen to the two signals one after another using the function `soundsc(x, fs)`; and
 - (d) Give your interpretation of this listening.