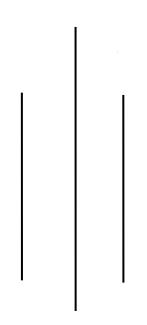
# KATHMANDU UNIVERSITY

# DHULIKHEL, KAVRE



Subject: COMP-407: Digital Signal Processing

Lab no: 3

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Group: CE 4<sup>th</sup> year 1<sup>st</sup> sem

Level: UNG

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#### **Sampling:**

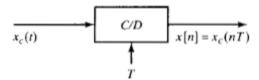
In signal processing, sampling is the reduction of a continuous-time signal to a discrete-time signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal).

A sample is a value or set of values at a point in time and/or space.

The typical method of obtaining a discrete-time representation of a continuous-time signal is through periodic sampling, wherein a sequence of samples, x[n], is obtained from a continuous-time signal  $x_c(t)$  according to the relation

$$x[n] = x_c(nT) \qquad \quad -\infty < n < \infty.$$

T is the sampling period, and its reciprocal, fs, = 1/T. is the sampling frequency The sampling frequency or sampling rate, fs, is the average number of samples obtained in one second (samples per second), thus fs = 1/T.



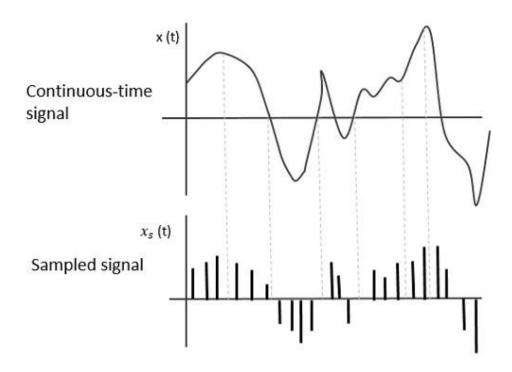


Fig: Sampling of continuous signal

#### **Nyquist Sampling Theorem:**

Let  $x_c(t)$  be bandlimited signal with  $X_c(J\Omega)=0$  for  $|\Omega| \ge \Omega_N$ , then  $x_c(t)$  is uniquely determined by its samples  $x[n] = x_c(nT)$  if  $\Omega_s \ge 2\Omega_N$ ;  $n=0,\pm 1,\pm 2,\pm 3,...$ 

Here,  $\Omega_s$  is sampling frequency in radians per seconds

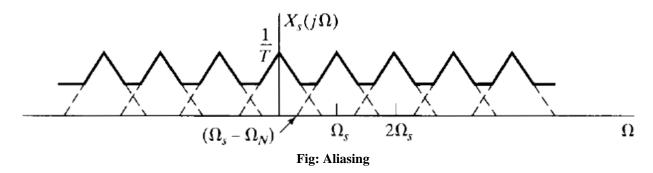
 $\Omega_{\mbox{\scriptsize N}}$  is the maximum component frequency of the signal to be sampled

 $X_c(J\Omega)$  is the Fourier transform of  $x_c(t)$ 

Nyquist rate is twice the maximum component frequency of the function being sampled.

#### **Aliasing**

If the inequality  $\Omega_s \ge 2\Omega_N$  does not hold, the copies of  $X_c(J\Omega)$  overlap, so that when they are added together,  $X_c(J\Omega)$  is no longer recoverable by lowpass filtering. In this case, the reconstructed output  $x_r(t)$  is related to the original continuous-time input through a distortion referred to as aliasing distortion, or simply, aliasing.



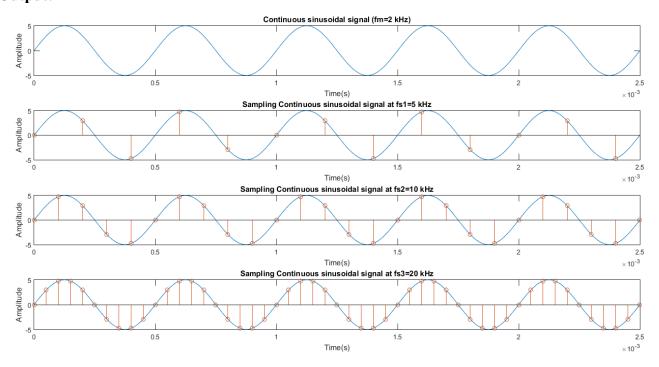
**Question 1**: Generate the signal  $x = 5\sin(2 \text{ pi f t})$  with 5 cycles, where f = 2 kHz signal and sample the signal with frequency 5 KHz, 10 Khz, 20 KHz. (Title and label each figure)

#### Code:

```
f=2e3; %Frequency of sinusoid cycles=5; % generate five cycles of sinusoid t=0:1/500e3:cycles*1/f; %time index x = 5*sin(2*pi*f*t); subplot(4,1,1); plot(t,x); title('Continuous sinusoidal signal (fm=2 kHz)'); xlabel('Time(s)'); ylabel('Amplitude'); fs1=5e3; %5kHz sampling rate t1=0:1/fs1:cycles*1/f; %time index x1 = 5*sin(2*pi*f*t1); fs2=10e3; %10kHz sampling rate t2=0:1/fs2:cycles*1/f; %time index x2=5*sin(2*pi*f*t2);
```

```
fs3=20e3; %20kHz sampling rate
t3=0:1/fs3:cycles*1/f; %time index
x3=5*sin(2*pi*f*t3);
subplot(4,1,2);
plot(t,x);
hold on;
stem(t1,x1);
title('Sampling Continuous sinusoidal signal at fs1=5 kHz');
xlabel('Time(s)');
ylabel('Amplitude');
subplot(4,1,3);
plot(t,x);
hold on;
stem(t2,x2);
title('Sampling Continuous sinusoidal signal at fs2=10 kHz');
xlabel('Time(s)');
ylabel('Amplitude');
subplot(4,1,4);
plot(t,x);
hold on;
stem(t3,x3);
title('Sampling Continuous sinusoidal signal at fs3=20 kHz');
xlabel('Time(s)');
ylabel('Amplitude');
```

#### **Output:**

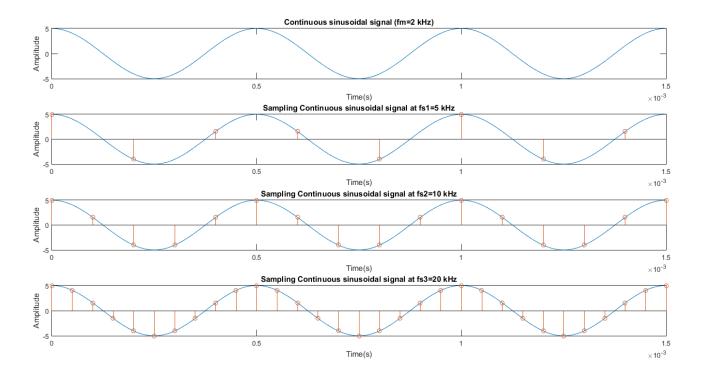


**Question 2**: Generate the signal  $x = 5\cos(2 \text{ pi f t})$  with 3 cycles, where f = 2 kHz signal and sample the signal with frequency 5 KHz, 10 Khz, 20 KHz. (Title and label each figure)

#### Code:

```
%ques 2
f=2e3; %Frequency of sinusoid
cycles=3; % generate five cycles of sinusoid
t=0:1/500e3:cycles*1/f; %time index
x = 5*\cos(2*pi*f*t);
figure;
subplot(4,1,1);
plot(t,x);
title('Continuous sinusoidal signal (fm=2 kHz)');
xlabel('Time(s)');
ylabel('Amplitude');
fs1=5e3; %5kHz sampling rate
t1=0:1/fs1:cycles*1/f; % time index
x1 = 5*\cos(2*pi*f*t1);
fs2=10e3; %10kHz sampling rate
t2=0:1/fs2:cycles*1/f; %time index
x2=5*cos(2*pi*f*t2);
fs3=20e3; %20kHz sampling rate
t3=0:1/fs3:cycles*1/f; %time index
x3=5*\cos(2*pi*f*t3);
subplot(4,1,2);
plot(t,x);
hold on;
stem(t1,x1);
title('Sampling Continuous sinusoidal signal at fs1=5 kHz');
xlabel('Time(s)');
ylabel('Amplitude');
subplot(4,1,3);
plot(t,x);
hold on;
stem(t2,x2);
title('Sampling Continuous sinusoidal signal at fs2=10 kHz');
xlabel('Time(s)');
ylabel('Amplitude');
subplot(4,1,4);
plot(t,x);
hold on;
stem(t3,x3);
title('Sampling Continuous sinusoidal signal at fs3=20 kHz');
xlabel('Time(s)');
ylabel('Amplitude');
```

### **Output:**



### **Conclusion:**

Thus, we performed sampling on continuous sinusoidal waves at different sampling frequencies and viewed the results in MATLAB