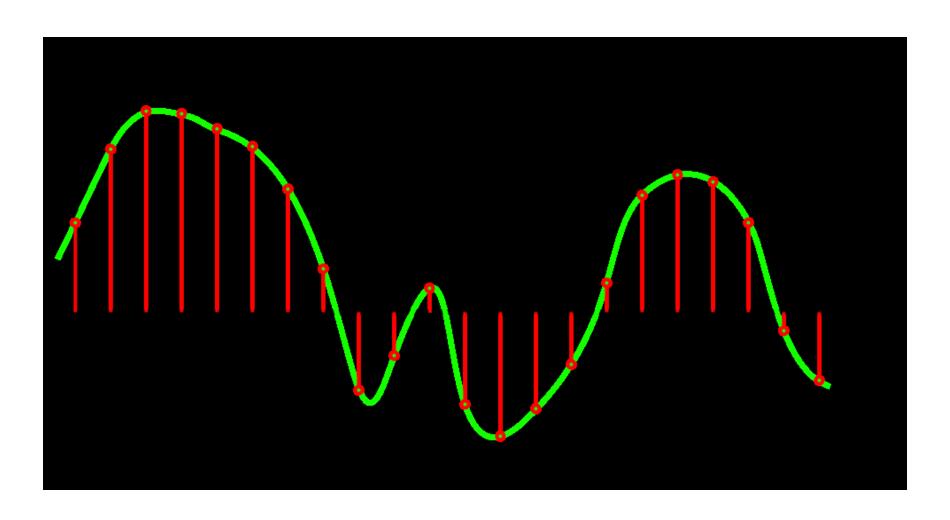
Sampling and Aliasing

Applied DSP practice lab

Sampling

- Process of converting the continuous time signal into a discrete time signal.
- These discrete points are called sample points and we have to choose the points appropriately so that the discrete points when joined together form the original signal reliably.
- The frequency with which these samples are taken from time domain is called sampling frequency denoted as *fs* (*samples/sec*).

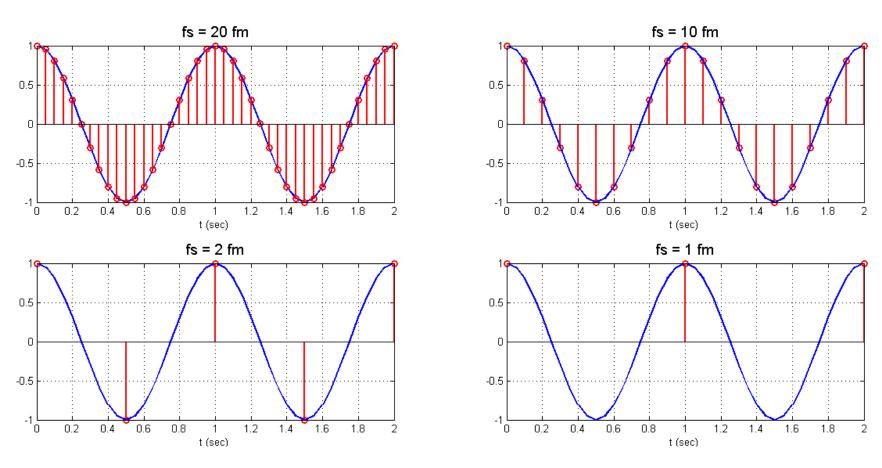
Sampling a continuous time signal



Problem with Sampling

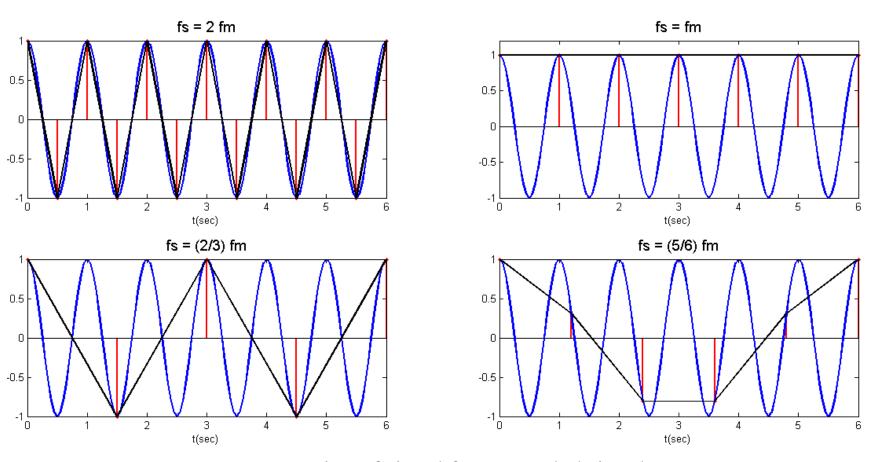
- > One main problem that stems when sampling the signal without appropriate *fs is "aliasing"*.
- ➤ When a sampled signal is interpolated (joining the sample points together), the reconstructed signal may resemble a lower frequency signal if sampling frequency is not properly chosen

Problem with Sampling(contd..)



Sampling a cosine signal of frequency fm=1 Hz with various sample frequency fs.

Problem with Sampling(contd..)



Reconstruction of signal from sampled signal.

To avoid aliasing in sampling composite signals

• Select $fs \ge 2$ fm(Sampling Theorem, fm should be maximum of all available frequencies)

where

fs=sampling frequency

fm=maximum frequency

falias=fm±nfs

1. Decide sampling frequency fs.

2. Find the interval of time domain.

3. Assign values to time variable.

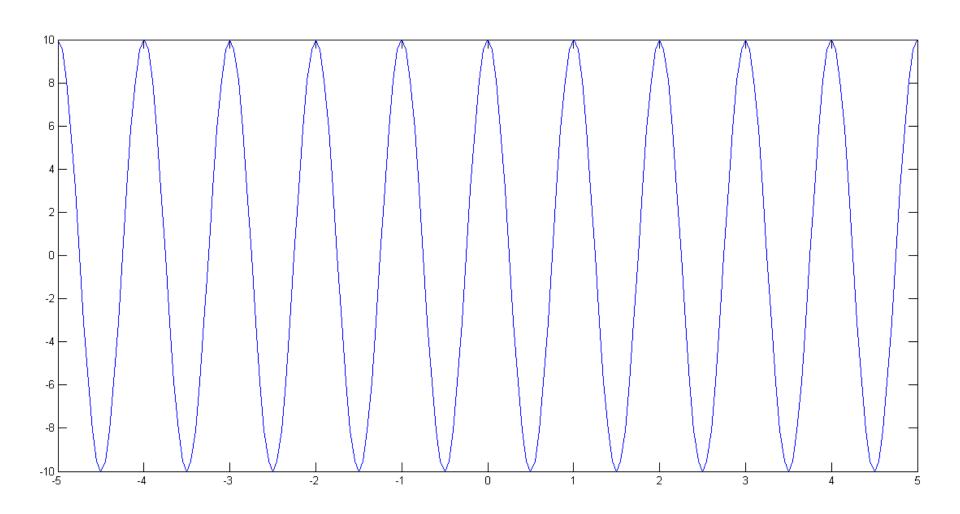
4. Find *f(t)*

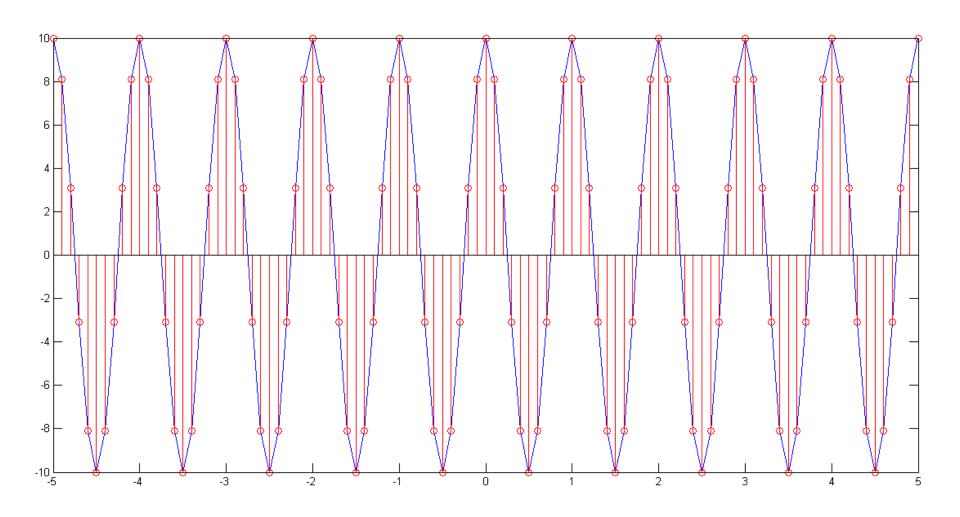
Example

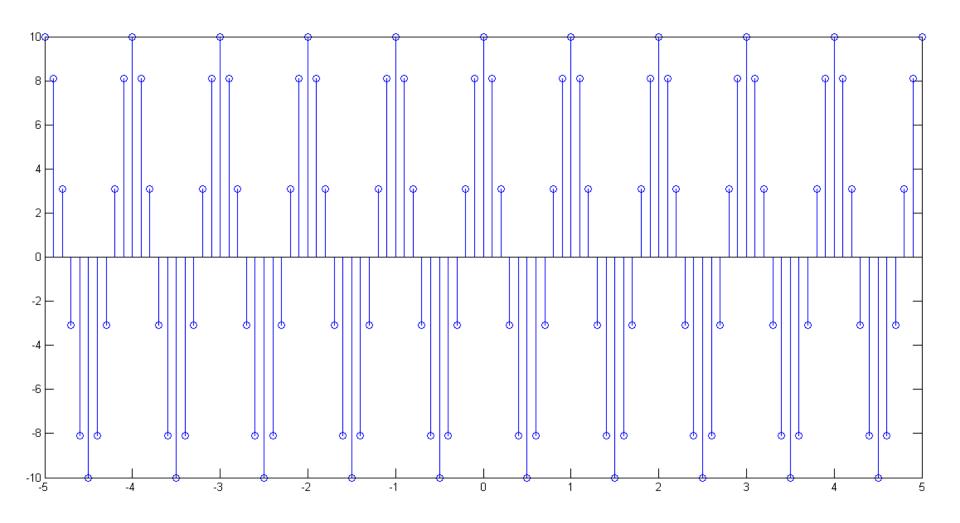
Plot $x(t) = A_m \cos(2\pi f_m t)$ for $\frac{-c}{f_m} \le t \le \frac{c}{f_m}$. Where $A_m = 10$, $f_m = 1$ Hertz, number of cycles needed is 2c = 10.

- 1. $f_s \ge 2f_m = 2$ Samples/sec. Decide $f_s = 10 > 2$.
- 2. c = 5, $T_m = 1/f_m = 1$ sec/cycle. Time starts from $\frac{-c}{f_m}$ till $\frac{c}{f_m}$. The duration is $\frac{2c}{f_m} = 10$ sec.
- 3. Time variable changes by steps of $T_s = 1/f_s$ sec/sample. So, $t = \frac{-c}{f_m}$: $T_s : \frac{c}{f_m}$ in MATLAB.
- 4. Find $x = A_m \cos(2\pi f_m t)$ in MATLAB.
- 5. Plot with stem(t,x) in MATLAB.

Remember the units of variables!







Some formulae

1. Sampling frequency fs (sample/sec)

2. Total duration of signal T (sec)

3. Total number of samples in T (sec) = $fs \times T$

4. fs \geq 2 fm (fm being max. of available freq.)

5. falias= $f_m \pm nf_s$

Further Reading

- Introduction To Signal Processing, Sophocles J. Orfanidis
 - a. Title 1.3 and Examples 1.3.1, 1.3.2
 - b. Title 1.4, Examples 1.4.1 1.4.5