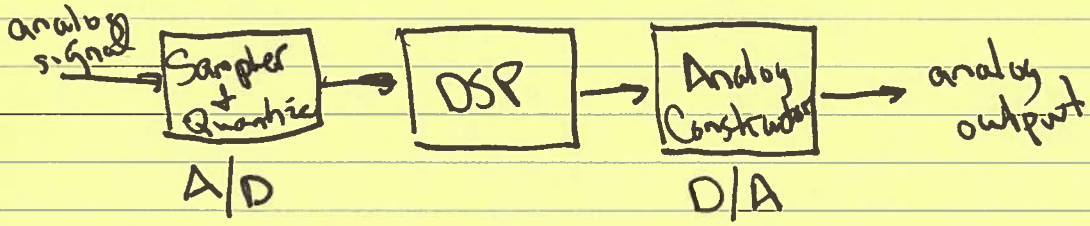
Lecture 1

Basic components of a DSP system:



Three types of signals:

- analog signal: continuous time / amplitude / infinite bandwidth
- discrete-time signal: discrete time / continuous amp
- digital signal: discrete time / discrete amplitude

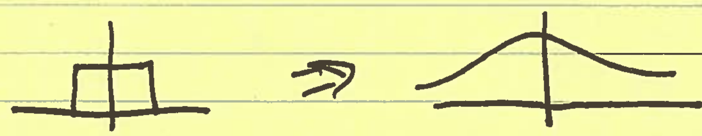
$x(t)$

$x(nT) \equiv x(n) \qquad T = \frac{1}{f_s}$

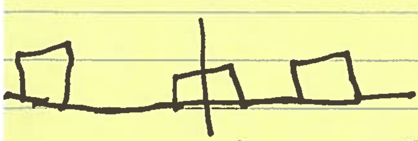
Fourier Transform:

$$X(\Omega) = \int_{-\infty}^{\infty} x(t)\, e^{-j\Omega t}\, dt \qquad \underset{\text{rd/sec}}{\Omega} = 2\pi f \quad Hz = \frac{1}{sec}$$

$$x(t) = \int_{-\infty}^{\infty} X(\Omega)\, e^{+j\Omega t}\, \frac{d\Omega}{2\pi}$$



infinite



Laplace Transform: $X(s) = \int_{-\infty}^{\infty} x(t)\, e^{-st}\, dt$

$$X(\Omega) = X(s)\Big|_{s=j\Omega}$$

$$x(t) \longrightarrow \boxed{h(t)} \longrightarrow y(t)$$

$$y(t) = \int_{-\infty}^{\infty} h(t-t')\,x(t')\,dt'$$

$$Y(\Omega) = H(\Omega)\,X(\Omega)$$

$$H(\Omega) = \int_{-\infty}^{\infty} h(t)\,e^{-j\Omega t}\,dt$$

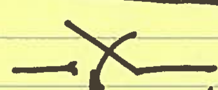
$$x(t) \longrightarrow \boxed{h(t)} \longrightarrow y(t)$$

$x(t) = A\sin\omega t$ $\qquad$ $y(t) = B\sin(\omega t + \phi)$

sinewave in $\Leftrightarrow$ sinewave out

Sampling:

$$x(t) \longrightarrow \boxed{\text{Sampler}} \longrightarrow \cancel{y(t)}\ y(nT)$$



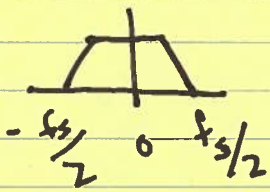
$T = \frac{1}{f_s}$

Sampling Theorem:

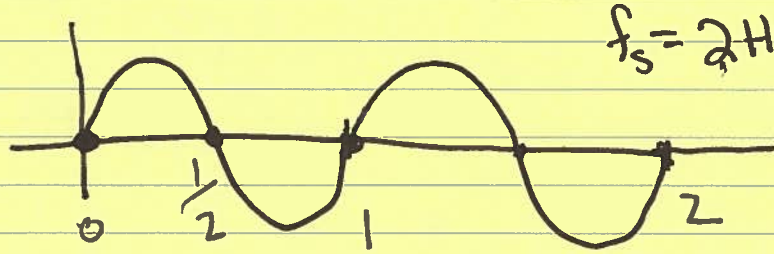
If $x(t)$ is bandlimited to B Hz,
then if $f_s = 2B$, we can recover
$x(t)$ EXACTLY!

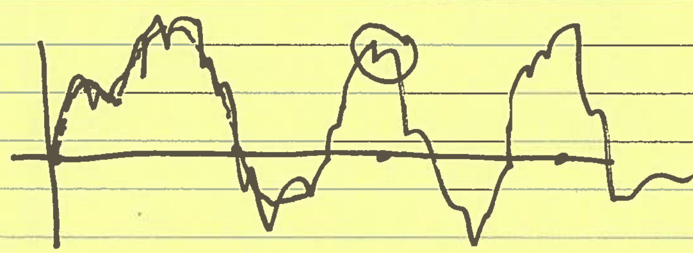
"Pre-filtering"

$$X(t) \rightarrow \boxed{\text{LPF}} \rightarrow \boxed{\text{A/D}} \rightarrow X(nT)$$



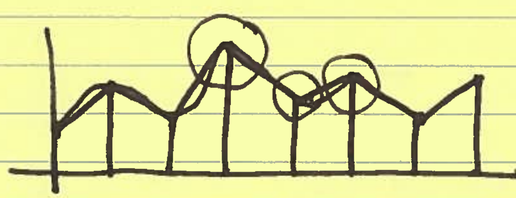
$f_s = 2\,\text{Hz}$





$$X(nT) \rightarrow \boxed{?} \rightarrow X(t)$$

$[0, f_s/2]$



$$X(nT) \rightarrow \boxed{\text{LPF}} \rightarrow X(t)$$