

SPRING 2010

# 即時控制系統設計 Design of Real-Time Control Systems

## Lecture 22 Sampling

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NTU-EE

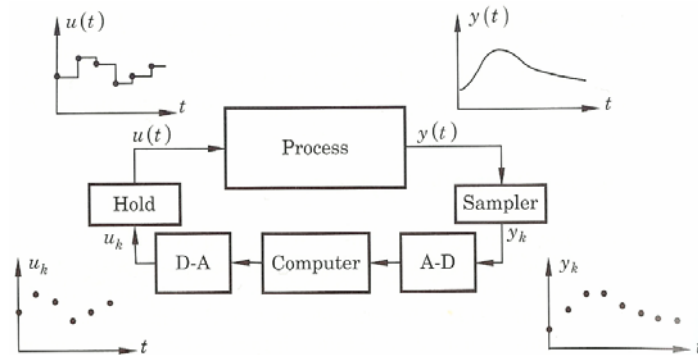
Feb10 – Jun10

Figures and images used in these lecture notes are adopted from  
"Signals & Systems" by Alan V. Oppenheim and Alan S. Willsky, 1997  
"Computer-Controlled Systems: Theory & Design" 3rd Ed. by KJ Astrom & B. Wittenmark, 1997

### Outline

- Representation of a CT Signal by Its Samples: The Sampling Theorem
- Reconstruction of a Signal from Its Samples Using Interpolation
- The Effect of Under-sampling: Aliasing
- Discrete-Time Processing of Continuous-Time Signals

### Computer-Controlled System (7.2)

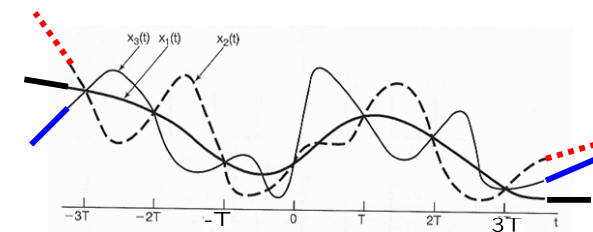


1. Wait for a clock pulse
2. Perform A/D conversion
3. Compute control variable
4. Perform D/A conversion
5. Update regulator state
6. Go to step 1

03/29/03

### The Sampling Theorem

- Representation of CT Signals by its Samples

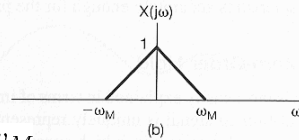


$$x_1(kT) = x_2(kT) = x_3(kT)$$

**The Sampling Theorem:**

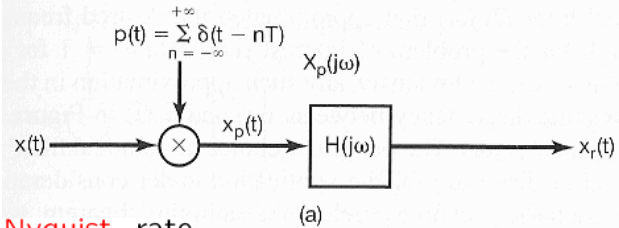
$x(t)$  : a band-limited signal

with  $X(j\omega) = 0$  for  $|\omega| > \omega_M$



if  $\omega_s > 2\omega_M$  where  $\omega_s = \frac{2\pi}{T}$

$\Rightarrow x(t)$  is uniquely determined by  $x(nT), n = 0, \pm 1, \pm 2, \dots$



$\Rightarrow 2\omega_M$  : Nyquist rate  
 $\omega_M$  : Nyquist frequency

**Impulse-Train Sampling:**

$p(t)$  : sampling function

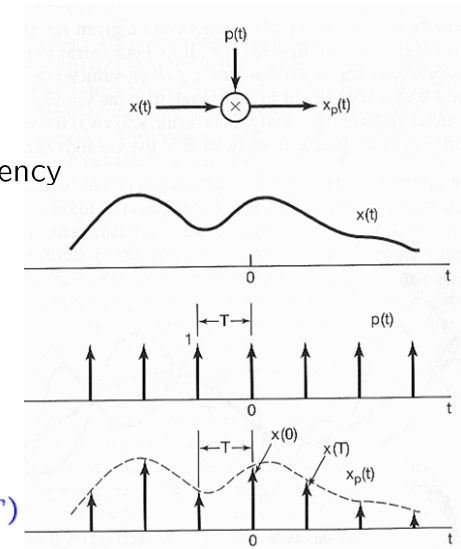
$T$  : sampling period

$\omega_s = \frac{2\pi}{T}$  : sampling frequency

$\Rightarrow x_p(t) = x(t) p(t)$

$$p(t) = \sum_{n=-\infty}^{+\infty} \delta(t - nT)$$

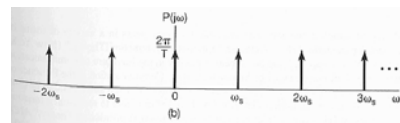
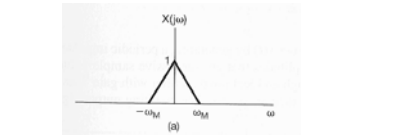
$$x_p(t) = \sum_{n=-\infty}^{+\infty} x(nT)\delta(t - nT)$$



**Impulse-Train Sampling:**

$$x(t) \xleftrightarrow{\mathcal{F}} X(j\omega)$$

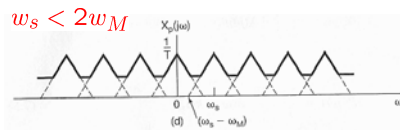
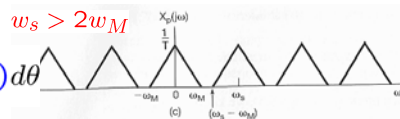
$$P(j\omega) = \frac{2\pi}{T} \sum_{k=-\infty}^{+\infty} \delta(\omega - k\omega_s)$$



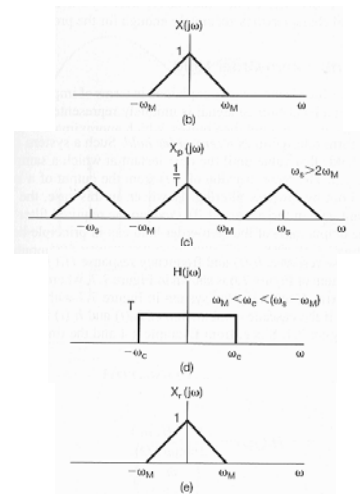
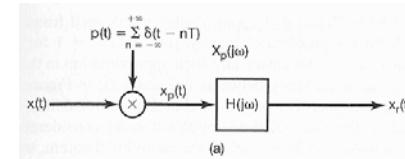
From multiplication property,

$$X_p(j\omega) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} X(j\theta) P(j(\omega - \theta)) d\theta$$

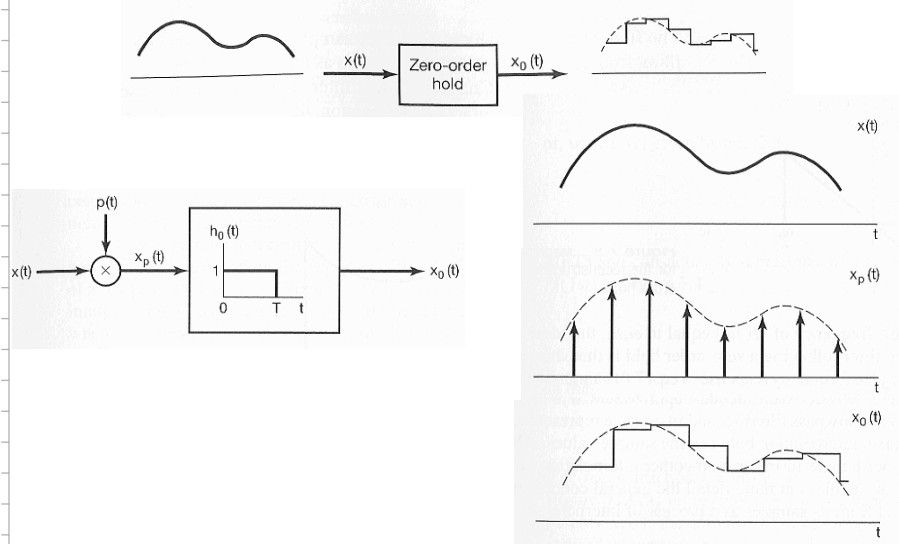
$$= \frac{1}{T} \sum_{k=-\infty}^{+\infty} X(j(\omega - k\omega_s))$$



**Exact Recovery by an Ideal Lowpass Filter:**



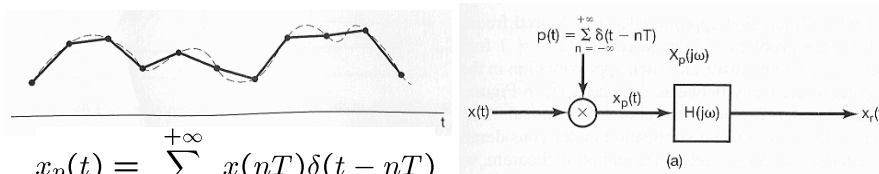
Sampling with Zero-Order Hold:



- Representation of a CT Signal by Its Samples: The Sampling Theorem
- Reconstruction of a Signal from Its Samples Using Interpolation
- The Effect of Under-sampling: Aliasing
- Discrete-Time Processing of Continuous-Time Signals

Reconstruction of a Signal from its Samples Using Interpolation

Exact Interpolation:



$$x_p(t) = \sum_{n=-\infty}^{+\infty} x(nT)\delta(t - nT)$$

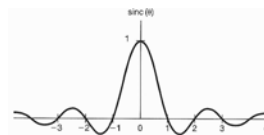
$$x_r(t) = x_p(t) * h(t)$$

$$x_r(t) = \sum_{n=-\infty}^{+\infty} x(nT)h(t - nT)$$

$$x_r(t) = \sum_{n=-\infty}^{+\infty} x(nT) \frac{w_c T}{\pi} \frac{\sin(w_c(t - nT))}{w_c(t - nT)}$$

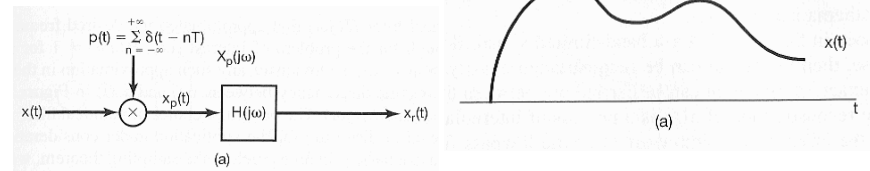
ideal lowpass filter

$$h(t) = \frac{w_c T \sin(w_c t)}{\pi w_c t}$$

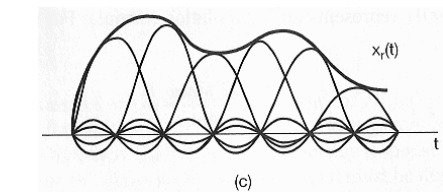
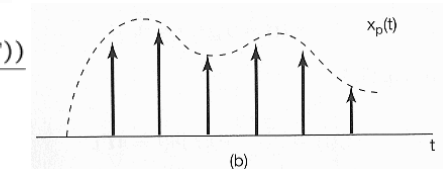
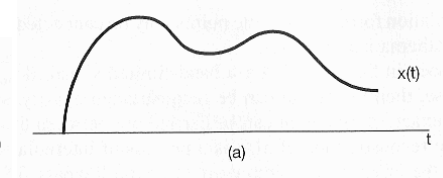
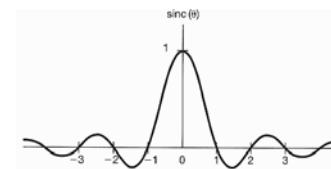


Reconstruction of a Signal from its Samples Using Interpolation

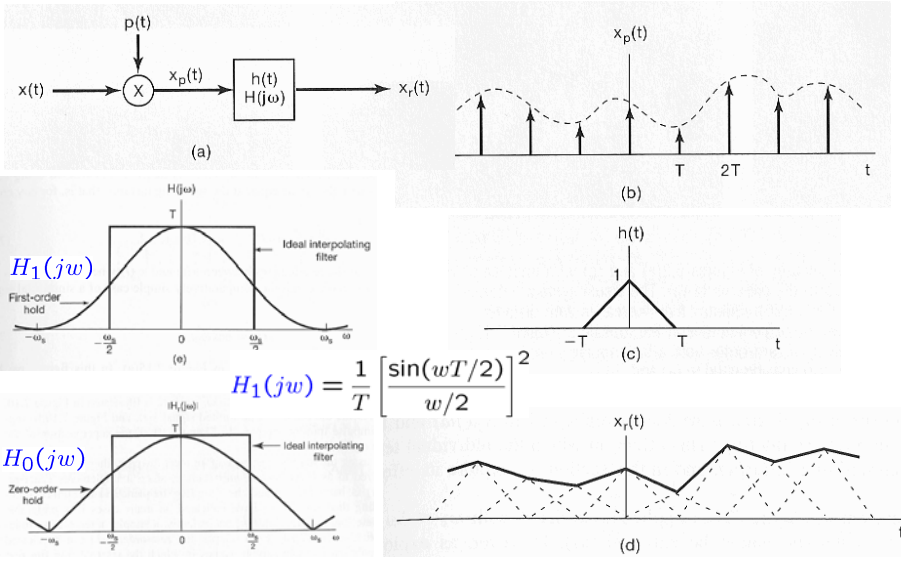
Exact Interpolation:



$$x_r(t) = \sum_{n=-\infty}^{+\infty} x(nT) \frac{w_c T}{\pi} \frac{\sin(w_c(t - nT))}{w_c(t - nT)}$$



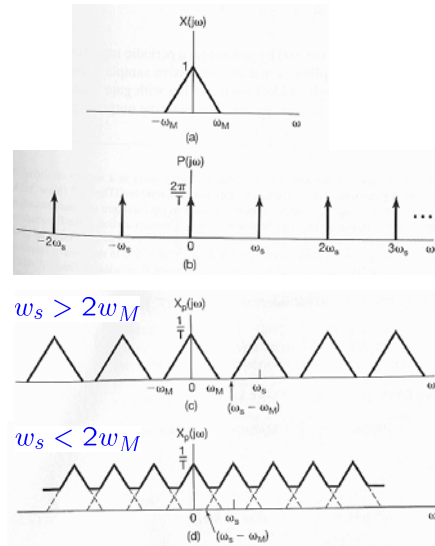
Higher-Order Holds:



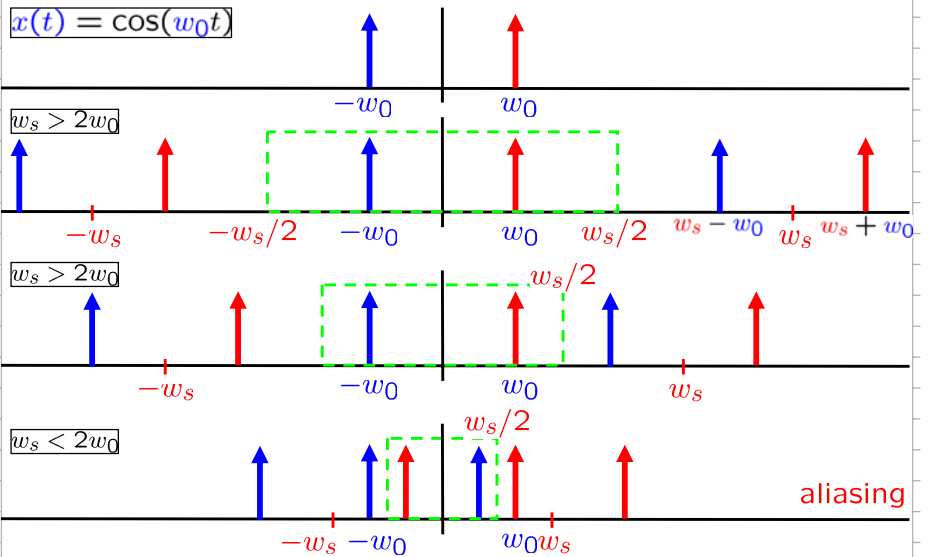
- Representation of a CT Signal by Its Samples: The Sampling Theorem
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Effect of Under-sampling: Aliasing

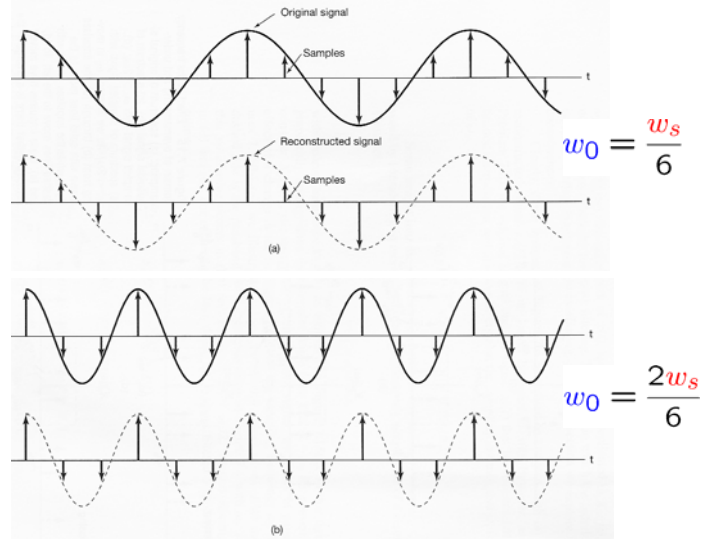
Overlapping in Frequency-Domain: Aliasing



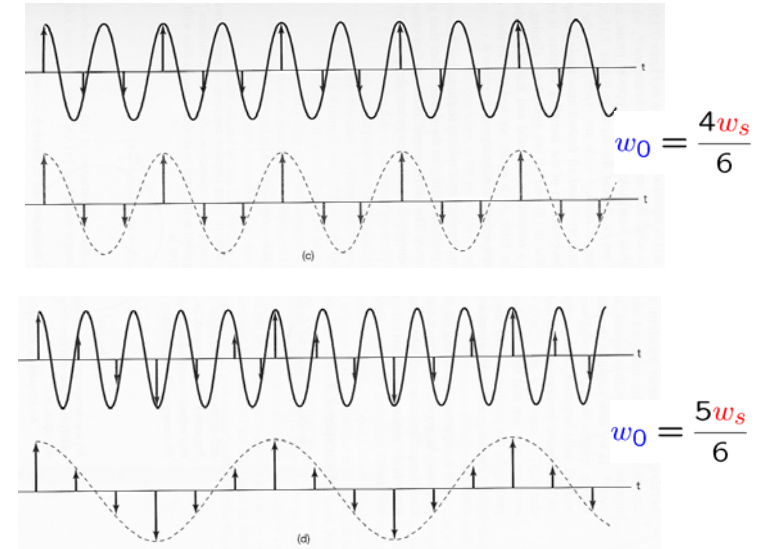
Overlapping in Frequency-Domain: Aliasing



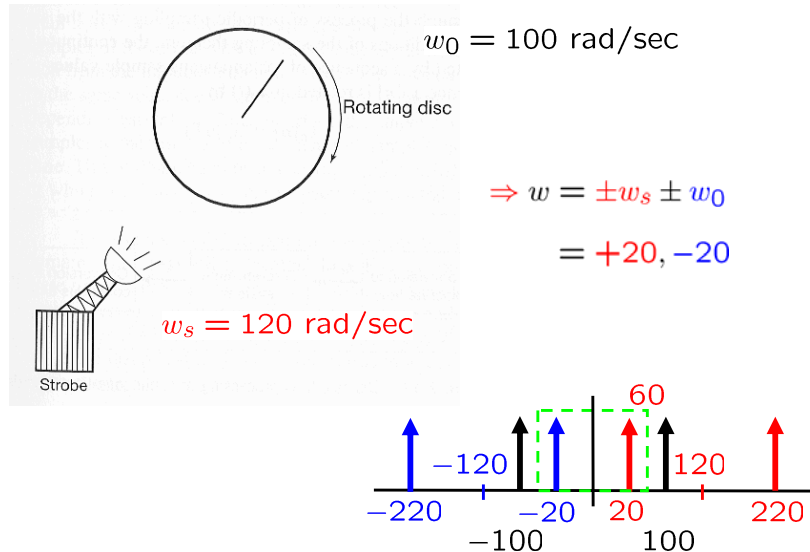
Overlapping in Frequency-Domain: Aliasing



Overlapping in Frequency-Domain: Aliasing

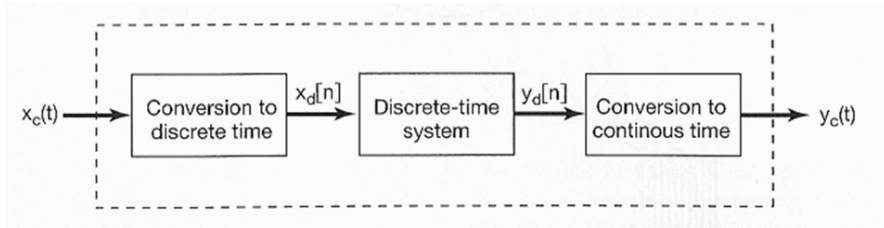


Strobe Effect:

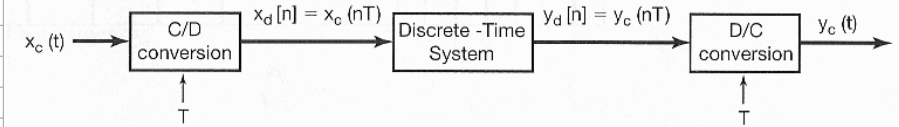


- Representation of a CT Signal by Its Samples: The Sampling Theorem
- Reconstruction of a Signal from Its Samples Using Interpolation
- The Effect of Under-sampling: Aliasing
- Discrete-Time Processing of Continuous-Time Signals

Discrete-Time Processing of CT Signals:



C/D or A-to-D (ADC) and D/C or D-to-A (DAC):



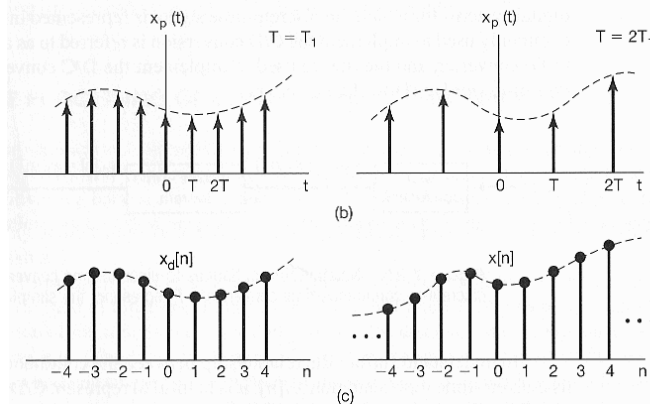
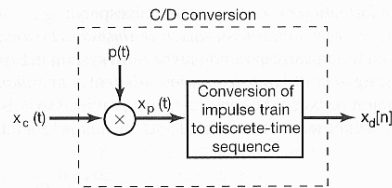
C/D: continuous-to-discrete-time conversion

A-to-D: analog-to-digital converter

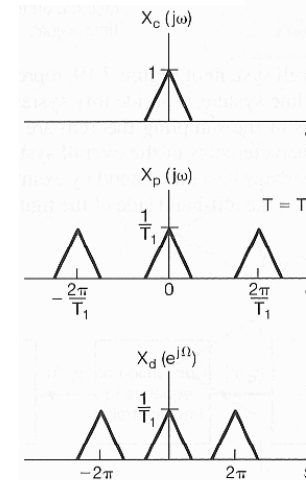
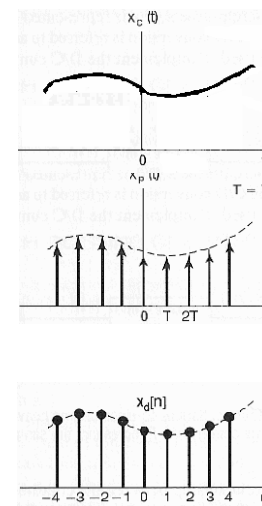
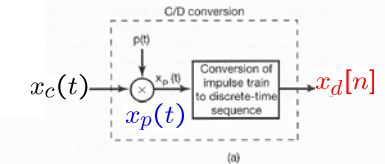
D/C: discrete-to-continuous-time conversion

D-to-A: digital-to-analog converter

C/D Conversion:



C/D Conversion:



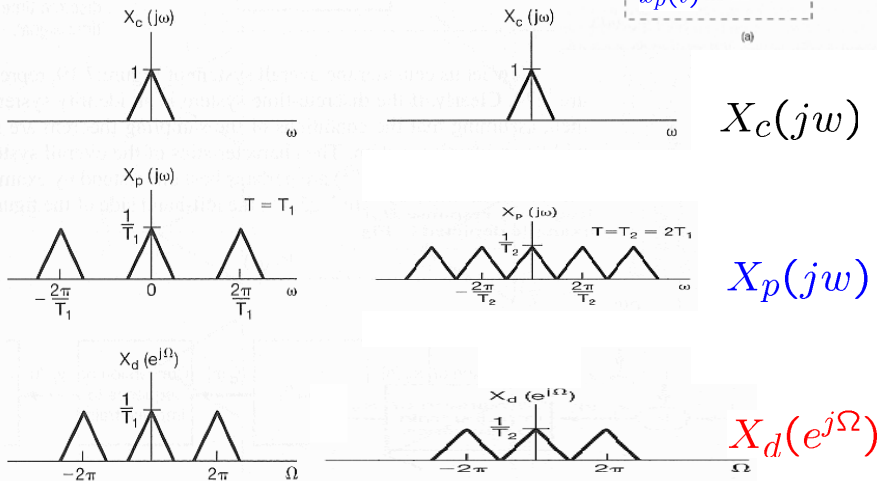
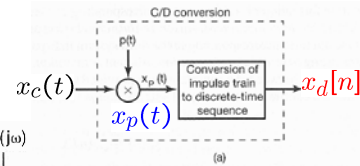
$$X_c(j\omega)$$

$$X_p(j\omega)$$

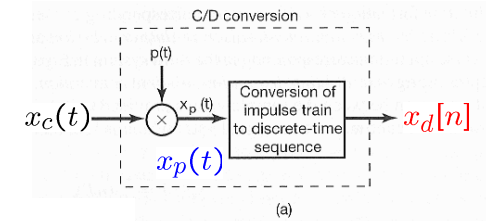
$$X_d(e^{j\Omega})$$



**C/D Conversion:**



**C/D Conversion:**



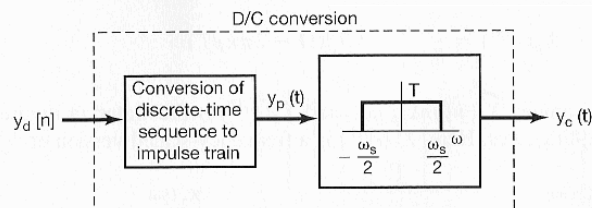
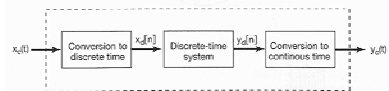
$$x_p(t) = \sum_{n=-\infty}^{+\infty} x_c(nT)\delta(t - nT)$$

$$X_p(j\omega) = \sum_{n=-\infty}^{+\infty} x_c(nT)e^{-j\omega nT} = \frac{1}{T} \sum_{K=-\infty}^{+\infty} X_c(j(\omega - k\omega_s))$$

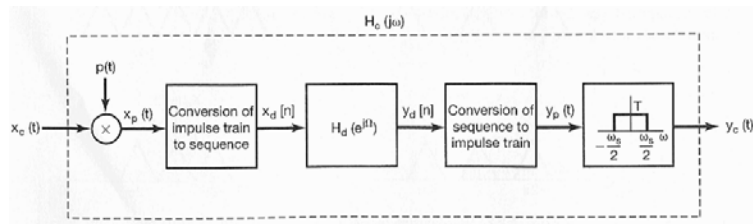
$$X_d(e^{j\Omega}) = \sum_{n=-\infty}^{+\infty} x_d[n]e^{-j\Omega n} = \sum_{n=-\infty}^{+\infty} x_c(nT)e^{-j\Omega n}$$

$$\Rightarrow X_d(e^{j\Omega}) = X_p\left(j\frac{\Omega}{T}\right) = \frac{1}{T} \sum_{K=-\infty}^{+\infty} X_c\left(j\left(\frac{\Omega}{T} - k\frac{2\pi}{T}\right)\right)$$

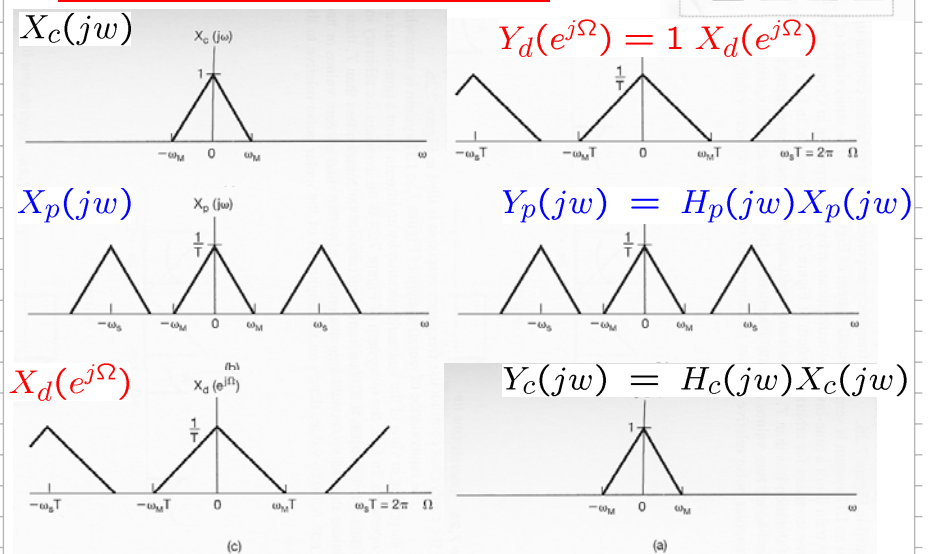
**D/C Conversion:**



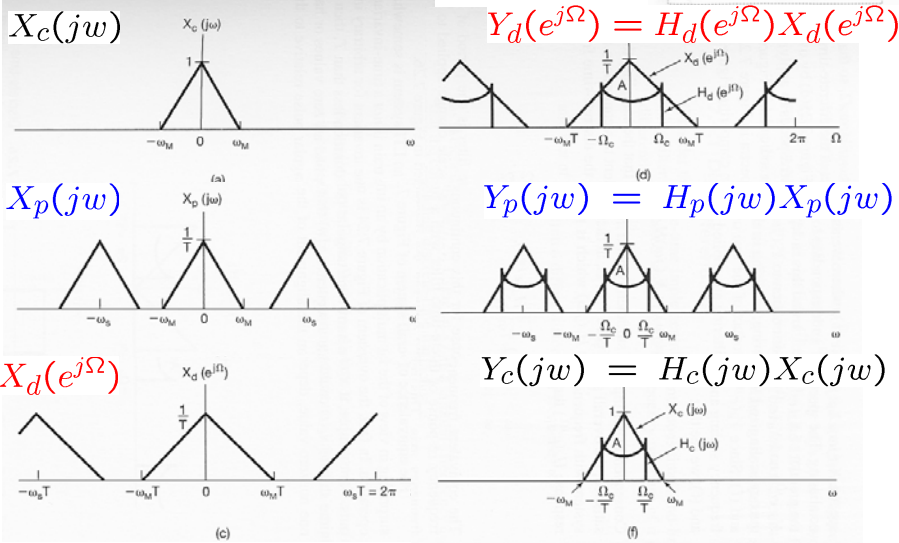
**Overall**



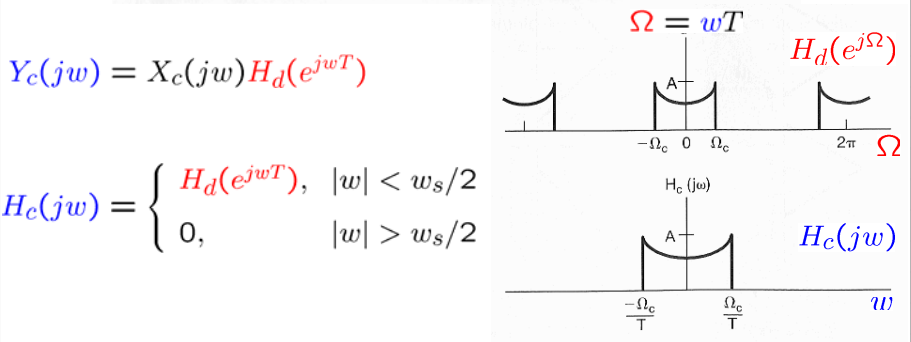
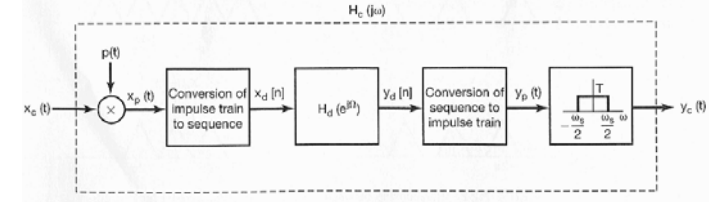
**Frequency-Domain Illustration:**



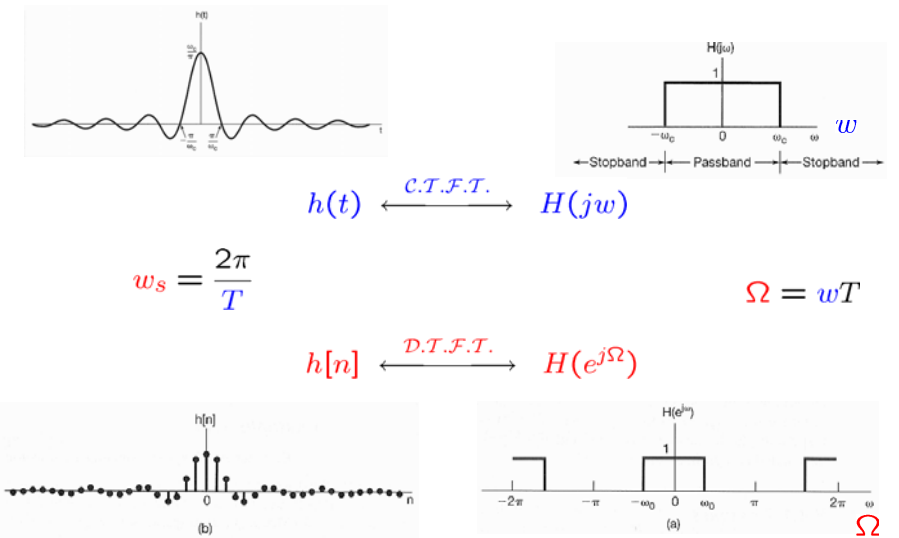
Frequency-Domain Illustration:



CT & DT Frequency Responses:



In Summary



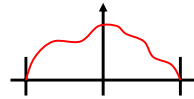
Sampling & Reconstruction (7.3)

The Sampling Theorem:

- If the sampling instants are sufficiently close, very little is lost by sampling a CT signal
- If the sampling points are too far apart, much of the information about a signal can be lost
- So, when a CT signal can be uniquely given by its sampled version?



▪ Theorem 7.1: (Shannon, 1949)



- $f(t)$ : a continuous-time signal
- $F(w)$ : the Fourier transform of  $f(t)$   
 $\rightarrow F(w) = 0$  outside  $(-w_0, w_0)$
- $w_s$ : sampling frequency
- $\Rightarrow$  If  $w_s > 2w_0$   
 Then  $f(t)$  can be computed by:

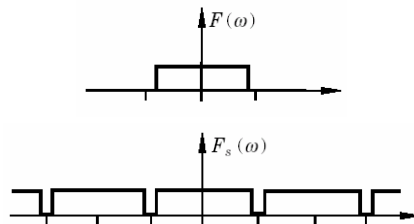
$$f(t) = \sum_{k=-\infty}^{\infty} f(kh) \frac{\sin(w_s(t - kh)/2)}{w_s(t - kh)/2} \quad \text{sinc} \frac{w_s(t - kh)}{2}$$

▪ Note that:

- $w_N = w_s/2$ : Nyquist frequency
- Reconstruction of signals:  
 $F(w) = 0$  when  $w > w_N$

▪ Reconstruction:

- $F(w) = \int_{-\infty}^{\infty} e^{-iwt} f(t) dt$
- $f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} e^{iwt} F(w) dw$
- $F_s(w) = \frac{1}{h} \sum_{k=-\infty}^{\infty} F(w + kw_s)$   
 $= \sum_{k=-\infty}^{\infty} C_k e^{-ikhw}$   
 $= \sum_{k=-\infty}^{\infty} f(kh) e^{-ikhw}$
- $F(w) = \begin{cases} hF_s(w) & |w| \leq \frac{w_s}{2} \\ 0 & |w| > \frac{w_s}{2} \end{cases}$



$$C_k = \frac{1}{w_s} \int_0^{w_s} e^{ikhw} F_s(w) dw$$

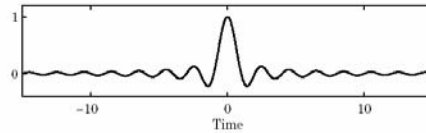
▪ Shannon Reconstruction:

- For periodic sampling of band-limited signals
- $$f(t) = \sum_{k=-\infty}^{\infty} f(kh) \frac{\sin(w_s(t - kh)/2)}{w_s(t - kh)/2}$$
- However, it is NOT a causal operator

▪ Shannon Reconstruction:

- Let's look at the **impulse response**:

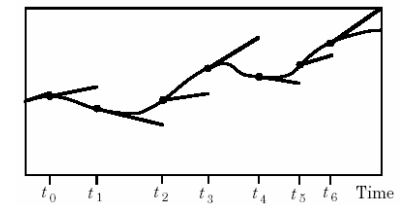
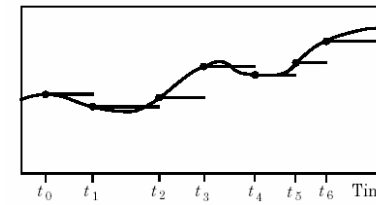
$$h(t) = \frac{\sin(\omega_s t/2)}{\omega_s t/2}$$



- The weights are **10%** after **3 samples**  
**< 5%** after **6 samples**
- This construction has a **delay**  
⇒ **Not good for control**
- Only applied to **periodic sampling**

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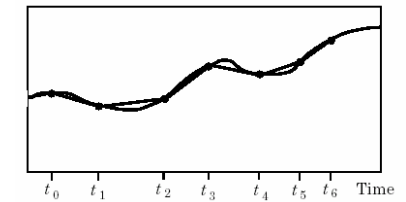
▪ Zero-Order Hold (ZOH) & First-Order Hold (FOH)



- They are **causal** operators

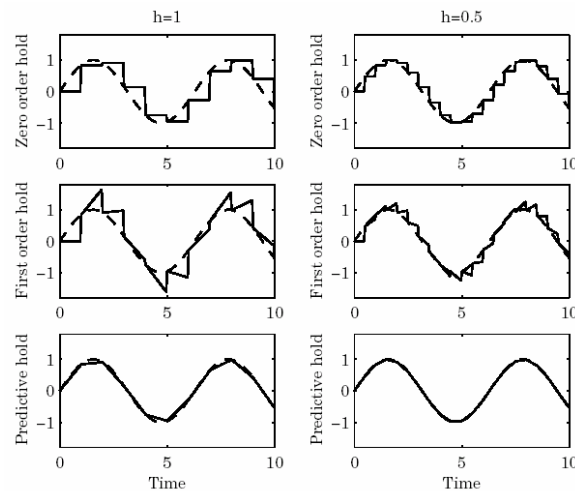
▪ Predictive FOH:

- It is **NOT causal**  
But, can be replaced  
by **model prediction**



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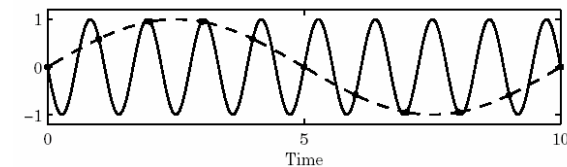
▪ Sinusoidal signal with  $h = 1$  and  $h = 0.5$



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▪ Aliasing:

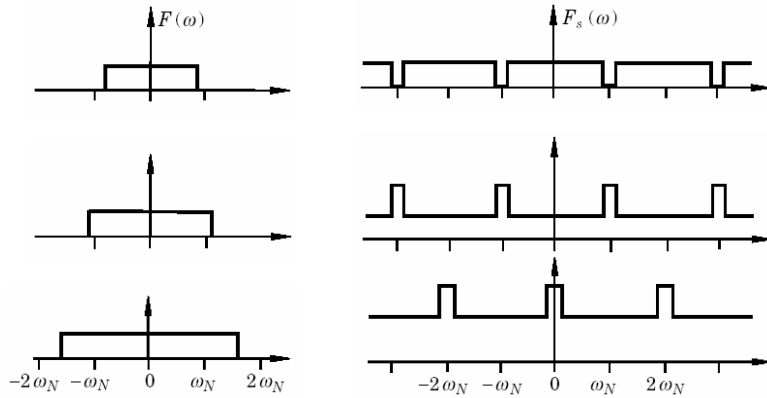
- Two signals with frequency, **0.1 Hz** and **0.9 Hz**
- They have the **same values** at all **sampling instants**



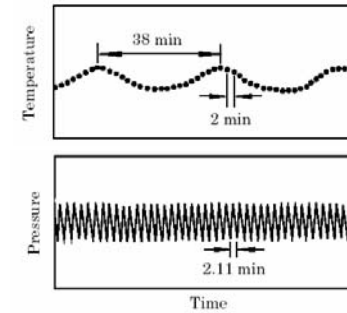
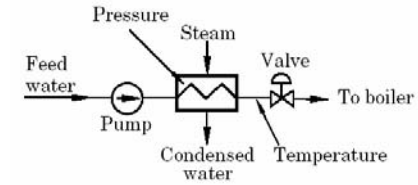
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Fourier transform of sampled signal:

$F(w) = \int_{-\infty}^{\infty} e^{-iwt} f(t) dt$       •  $F_s(w) = \sum_{k=-\infty}^{\infty} f(kh) e^{-ikhw}$

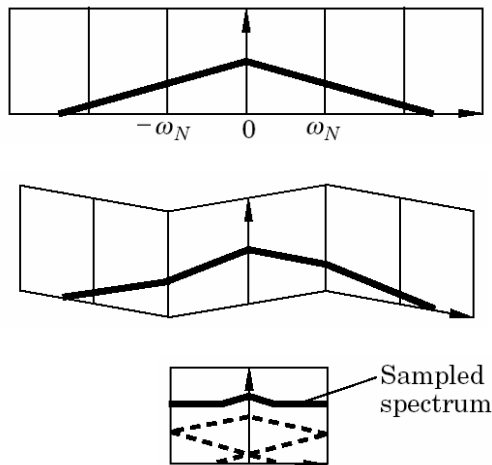


Example 7.1: Feed-water heater in a ship boiler

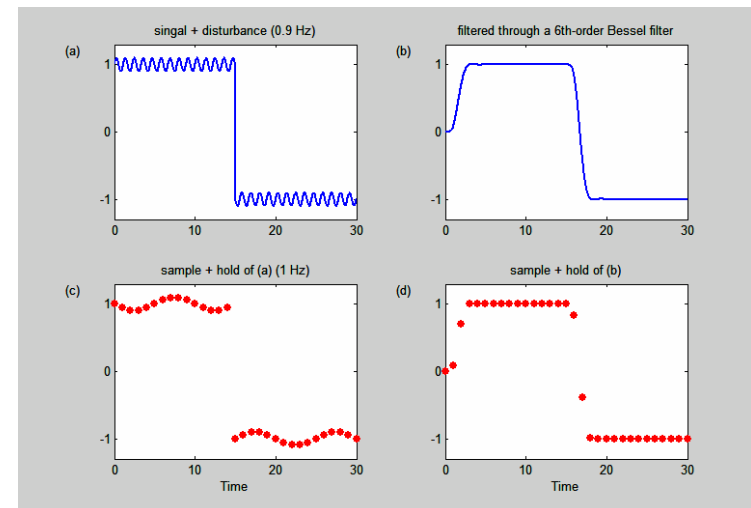


$w_s = \frac{2\pi}{2} = 3.142 \text{ rad/min}$   
 $w_0 = \frac{2\pi}{2.11} = 2.978 \text{ rad/min}$   
 $w_s - w_0 = 0.1638 \text{ rad/min}$   
 $\Rightarrow T_s = 38 \text{ min}$

Frequency Folding

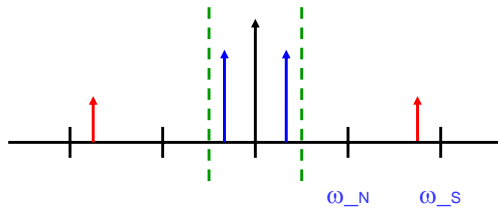


Pre-Sampling Filter in Example 7.2:



Pre-Sampling Filter in Example 7.2:

- With a sinusoidal perturbation (0.9Hz)
- Sampling frequency = 1 Hz



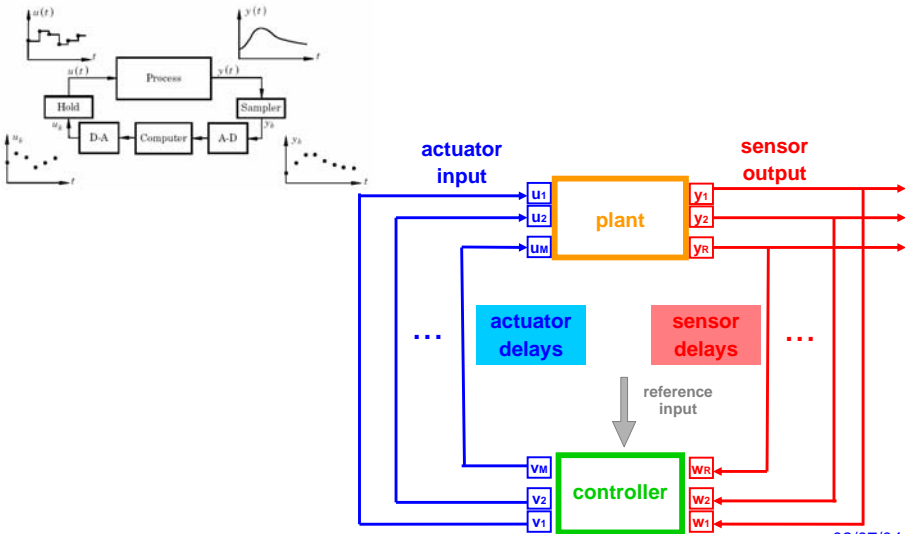
Post-Sampling Filter:

- Because signal from D/A is piecewise constant
  - May excite some oscillatory modes
  - So, use higher-order hold! such as piecewise linear signal

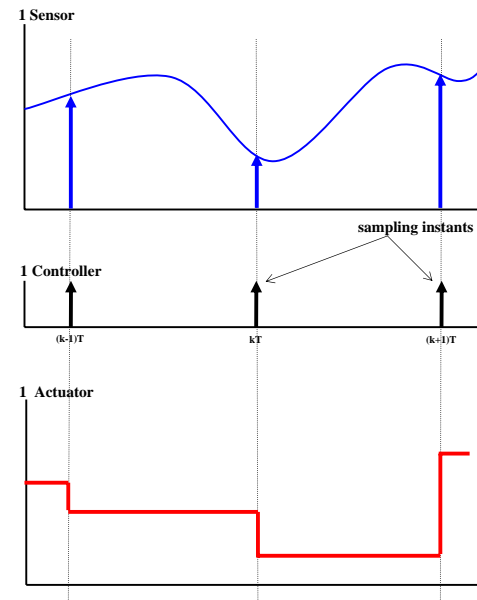
Timing Analysis

Real-Time Control Systems:

- Computing, Communication, and Control

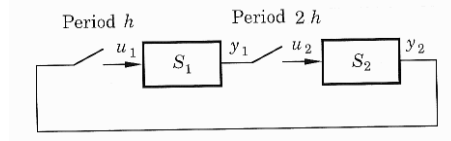


Timing Analysis

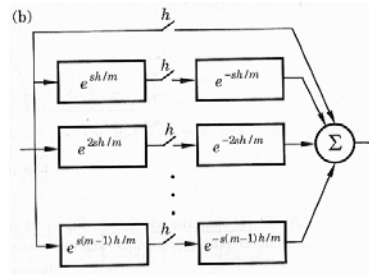
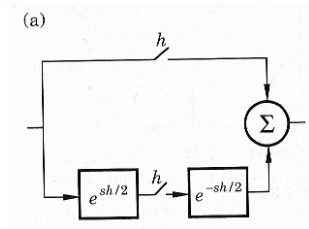




Multi-rate System:



Switch Decomposition:



03/29/03

Multi-rate systems

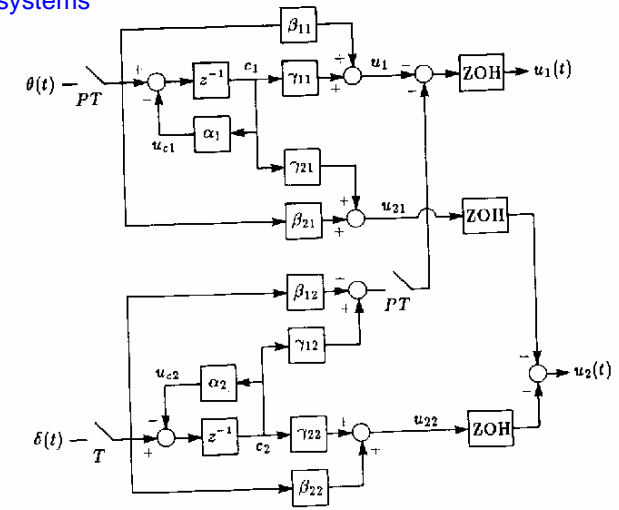


Fig. 7. TLA compensator structure.

M.C. Berg, N. Amit, J.D. Powell, "Multirate digital control system design", IEEE-TAC 33(12): 1139-1150, Dec 1988