

homework\_chapter7.pdf

Homework: 4.33, 4.37, 4.50 of the attachment

Tutorial Problems: 7.41, 7.44

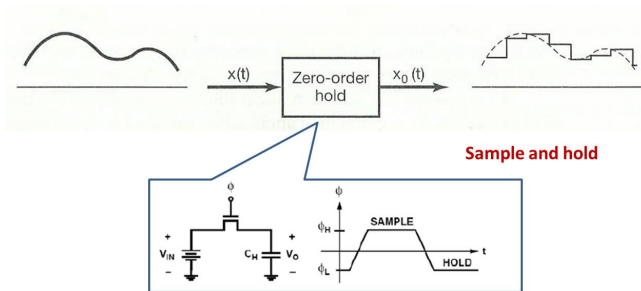


# Zero-Order Hold

- It's difficult to generate ideal impulse chain in practical implementation.

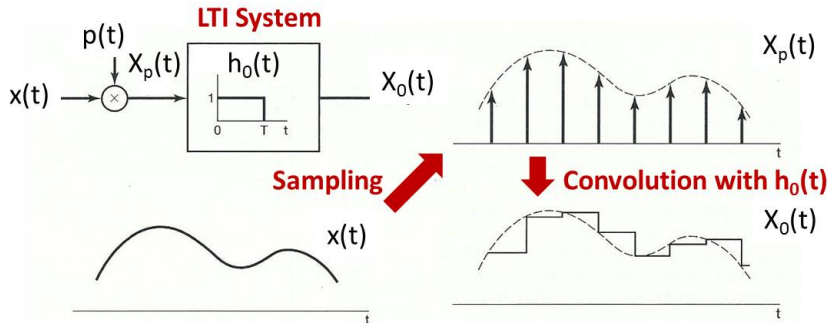
$$x_p(t) = \sum_{n=-\infty}^{\infty} x(nT)\delta(t - nT)$$

- Alternative approach: zero-order hold



- How to interpret the system of "zero-order hold" mathematically?

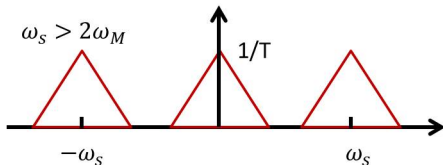
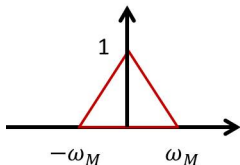
# Interpretation of Zero-Order Hold



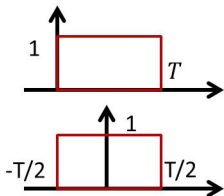
- Zero-order hold: sampling + interpolation with rectangular impulse response
- An approximation of the signal to be sampled.

# Frequency Analysis (1/2)

- Step 1: Impulse-train sampling



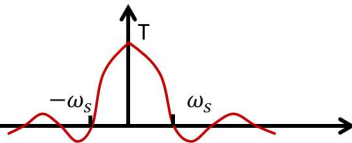
- Step 2: Frequency response of  $h_0(t)$



Fourier Transform



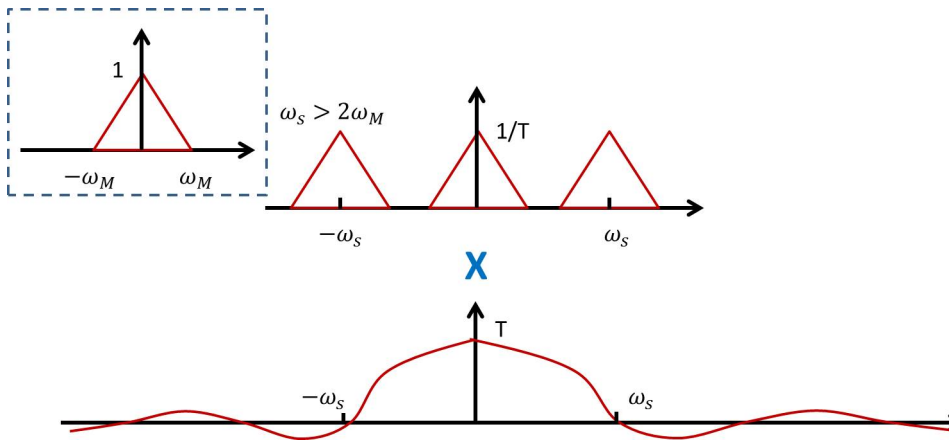
Same Amplitude



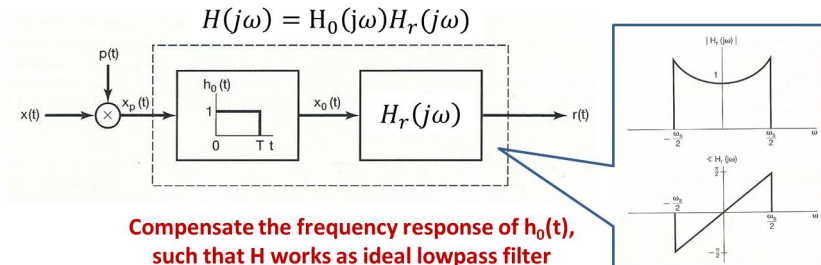
$$H(j\omega) = 2e^{-j\omega T/2} \frac{\sin \omega T/2}{\omega}$$



## Frequency Analysis (2/2)



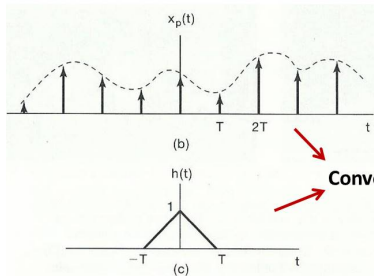
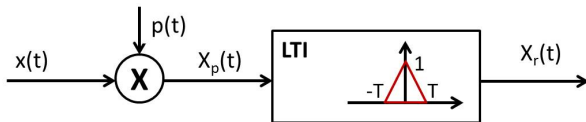
# Reconstruction



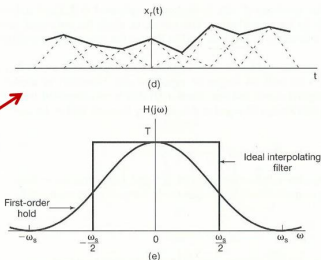
$$H_r(j\omega) = e^{j\omega T/2} \frac{\omega T}{2 \sin \omega T/2} \quad -\frac{\omega_s}{2} \leq \omega \leq \frac{\omega_s}{2}$$

- $H(j\omega)$  should be an ideal low-pass filter from  $-\omega_s/2$  to  $\omega_s/2$

# First-Order Hold

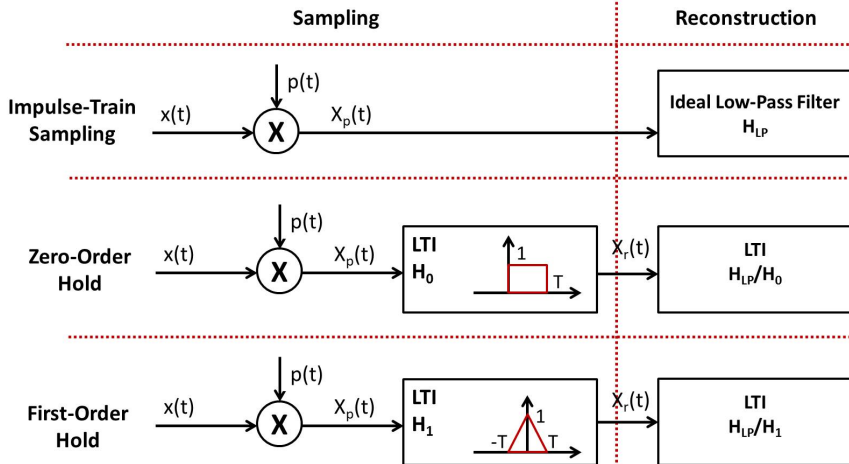


**Convolution**



- First-order hold: sampling + interpolation with triangular wave
- How to reconstruct?

# Summary: Sampling Approaches





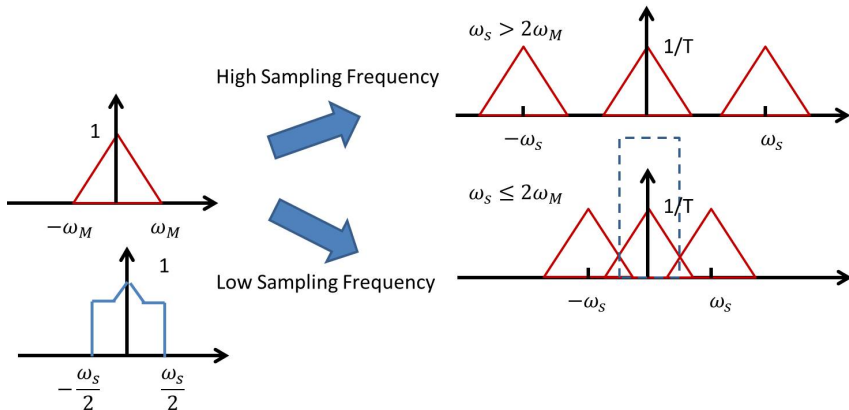
## Problem 2

### Problem (7.7)

A signal  $x(t)$  undergoes a zero-order hold operation with an effective sampling period  $T$  to produce a signal  $x_0(t)$ . Let  $x_1(t)$  denote the result of a first-order hold operation on  $x(t)$ . Specify the frequency response of a filter that produces  $x_1(t)$  as its output when  $x_0(t)$  is the input.

# Undersampling & Aliasing

- **Undersampling:** insufficient sampling frequency  $\omega_s < 2\omega_M$
- Perfect reconstruction is impossible with undersampling.
- **Aliasing:** distortion due to undersampling

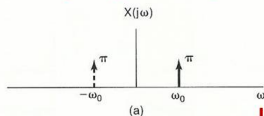


# Aliasing: Example

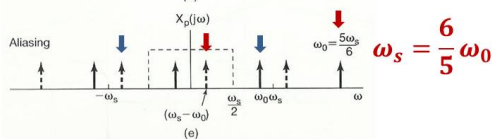
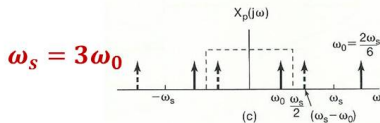
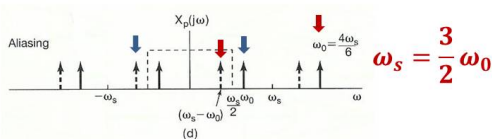
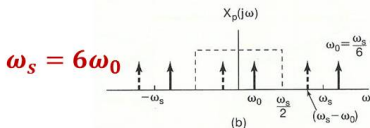
Signal before sampling:  $\cos \omega_0 t$

Sampling rate:  $\omega_s$

Lowpass Filter:  $-\frac{\omega_s}{2} \sim \frac{\omega_s}{2}$



Undersampling



$\cos \omega_0 t$

Aliasing:  $\cos(\omega_s - \omega_0)t$

**Low-pass filtering: Interpret the samples by cosine function with frequency lower than  $\omega_s/2$**

Original:

$$\omega_s = \frac{3}{2} \omega_0$$

Reconstructed:

$$\cos\left(\frac{1}{2} \omega_0\right)t$$

(c)

Original:

$$\omega_s = \frac{6}{5} \omega_0$$

Reconstructed:

$$\cos\left(\frac{1}{5} \omega_0\right)t$$

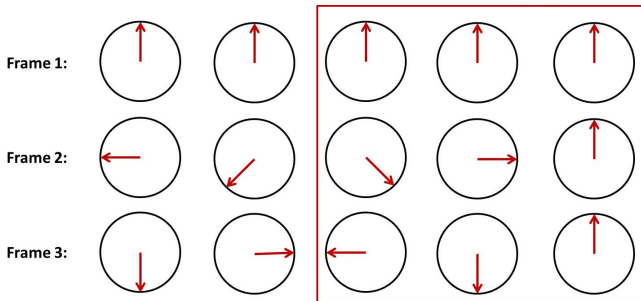
(d)

# Aliasing in Movies

- Wheel's rotation in movies



Aliasing

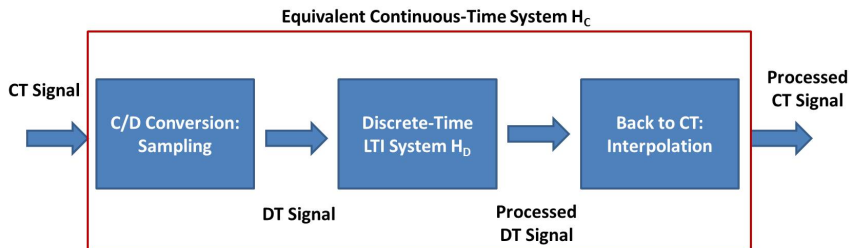


# Process Continuous-Time Signals Discretely



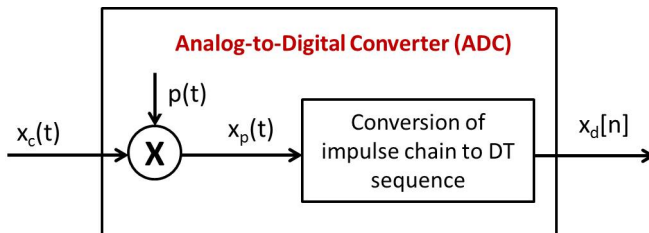
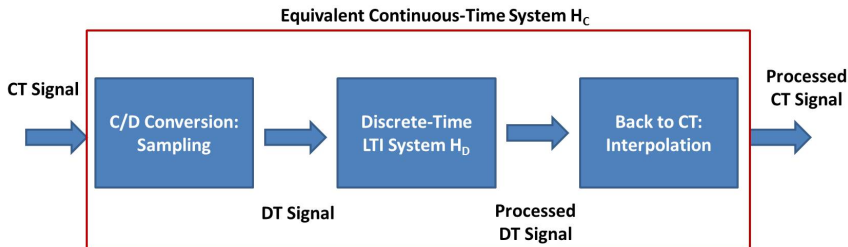
- People would like to process continuous-time signal in discrete-time (digital) domain

# Block Diagram



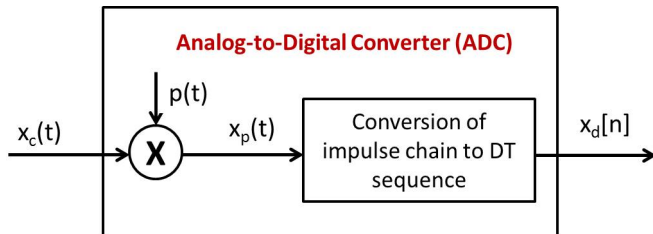
- It is much easier to design DT system.
- What's the relation between  $H_C$  and  $H_D$ ?

# Discretization: C/D Conversion





# Discretization: C/D Conversion



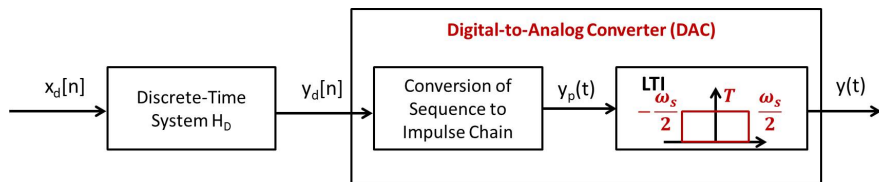
- Mathematical Interpretation (Fourier Transform)

$$x_c(t) \longleftrightarrow X_c(j\omega)$$

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

$$x_d[n] = x_c(nT) \longleftrightarrow X_d(e^{j\omega}) = X_p(j\omega/T)$$

# DT Processing and Conversion



- Mathematical Interpretation (Fourier Transform)

$$y_d[n] = x_d[n] * h_D[n] \quad \longleftrightarrow \quad Y_d(e^{j\omega}) = X_d(e^{j\omega})H_D(e^{j\omega})$$

$$y_p(t) = \sum_{n=-\infty}^{\infty} y_d[n]\delta(t - nT) \quad \longleftrightarrow \quad Y_p(j\omega) = Y_d(e^{j\omega T})$$

$$y(t) = y_p(t) * h_{LP}(t) \quad \longleftrightarrow \quad Y(j\omega) = Y_p(j\omega)H_{LP}(j\omega)$$

# Discussion

- From  $x_c(t)$  to  $x_d[n]$ , from  $y_d[n]$  to  $y(t)$ , what happen in time or frequency domain? Can you imagine it?
- What's the relation between  $x_c(t)$  and  $y(t)$ ?



# Input vs. Output

$$\begin{aligned}Y(j\omega) &= Y_p(j\omega)H_{LP}(j\omega) = Y_d(e^{j\omega T})H_{LP}(j\omega) \\&= X_d(e^{j\omega T})H_D(e^{j\omega T})H_{LP}(j\omega) \\&= X_p(j\omega)H_D(e^{j\omega T})H_{LP}(j\omega) \\&= \left[ \frac{1}{T} \sum_{k=-\infty}^{+\infty} X_c(j(\omega - k\omega_s)) \right] H_D(e^{j\omega T})H_{LP}(j\omega) \\&= X_c(j\omega)H_D(e^{j\omega T}) \\&= X_c(j\omega)\tilde{H}_D(e^{j\omega T})\end{aligned}\tag{1}$$

where

$$\tilde{H}_D(e^{j\omega T}) = \begin{cases} H_D(e^{j\omega T}) & |\omega| < \omega_s/2 \\ 0 & \text{otherwise} \end{cases}\tag{2}$$

- It is equivalent to a continuous-time LTI system  $H_C(j\omega) = \tilde{H}_D(e^{j\omega T})$
- $H_D(e^{j\omega T})$  is a periodic extension of  $\tilde{H}_D(e^{j\omega T})$  with period  $\omega_s = 2\pi/T$



# System Design

$$H_C(j\omega) = \tilde{H}_D(e^{j\omega T})$$



- How can we design a CT LTI system with frequency response  $H_C$  via DT LTI system?

- Step 1: Sampling frequency  $\omega_s$  or  $2\pi/T$  should be larger than Nyquist rate

- Step 2:  $\tilde{H}_D(e^{j\omega T}) = H_C(j\omega)$

- Step 3: Frequency response of DT LTI system

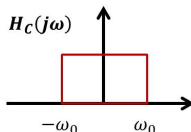
$$H_D(e^{j\omega T}) = \sum_{k=-\infty}^{\infty} \tilde{H}_D(e^{j(\omega - k\omega_s)T}) \text{ or}$$

$$H_D(e^{j\omega}) = \sum_{k=-\infty}^{\infty} \tilde{H}_D(e^{j(\omega - k\omega_s T)}) = \sum_{k=-\infty}^{\infty} H_C(j\frac{\omega - 2k\pi}{T})$$

# System Design Example

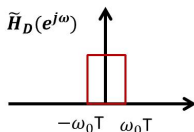
- How to implement an ideal CT lowpass filter?

Objective of Design:



$$H_C(j\omega) = \tilde{H}_D(e^{j\omega T})$$

Scale by T



Repetition

