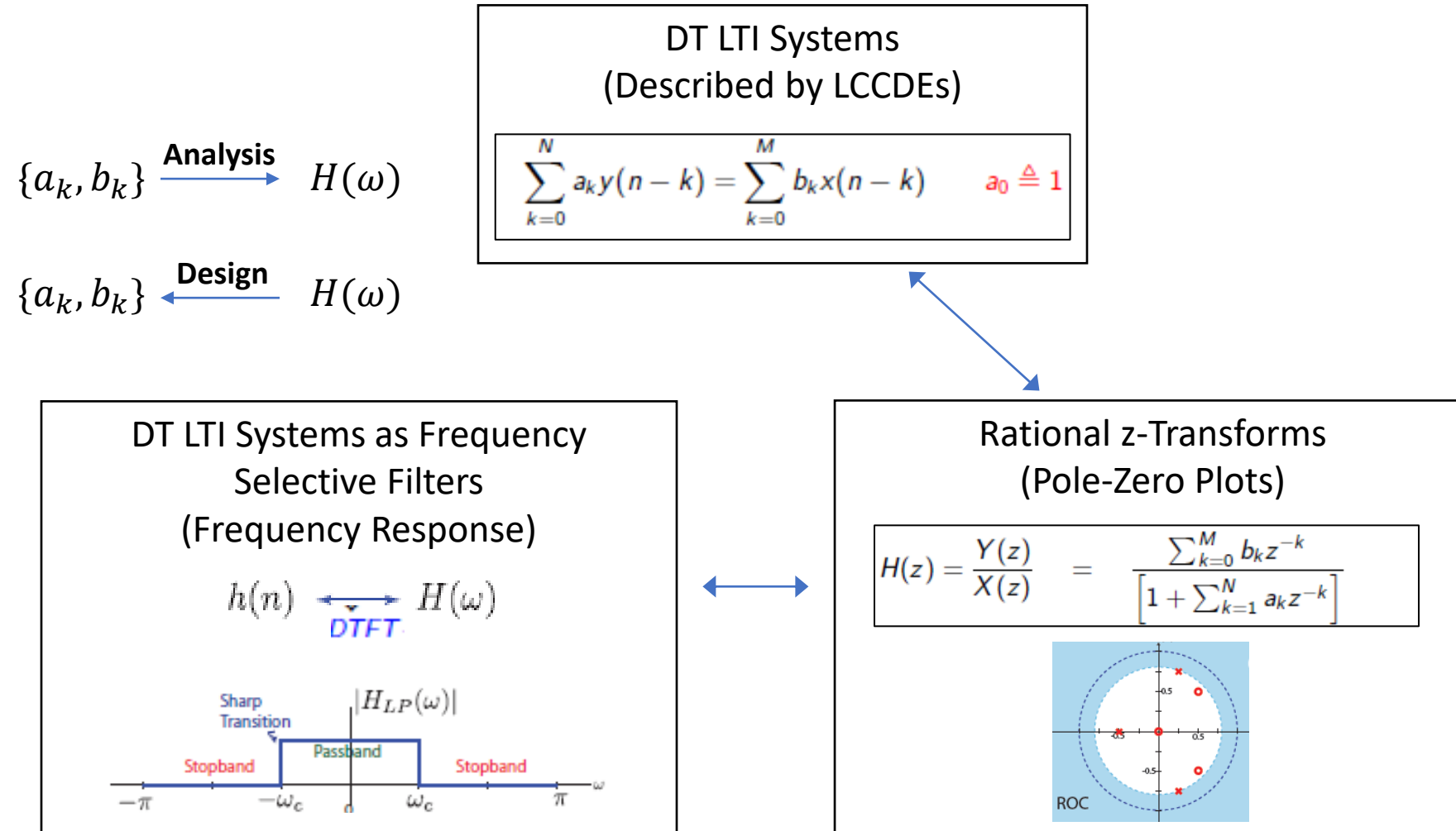




# Introduction to Digital Filters

EE 453 / CE 352  
Saad Baig

# Big Picture Till Now



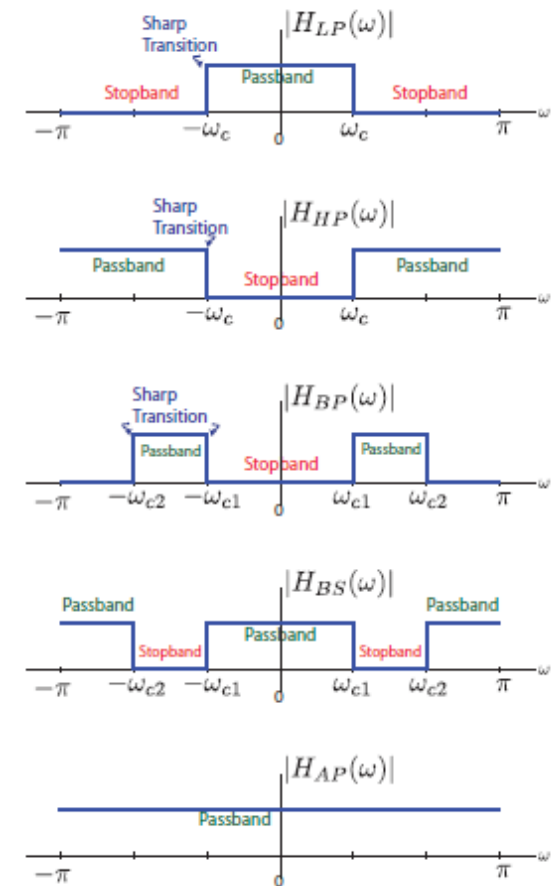
# Big Picture Till Now

$$y[n] = x[n] * h[n] \xleftrightarrow{\mathcal{F}} Y(e^{j\omega}) = X(e^{j\omega}) H(e^{j\omega})$$

- An LTI system can only change the amplitude and phase of a sinusoidal signal. It cannot change the frequency.
- Fourier Transform (frequency components) of the output of an LTI system is the Fourier Transform (frequency components) of the input, multiplied by the frequency response of the system
- This creates the possibility to change the relative amplitude of the frequency components in a signal.
- This process is referred to as **filtering**.

# Big Picture Till Now

- A filter is a device that discriminates, according to some attribute of the input, what passes through it.
- For LTI systems, given  $Y(\omega) = X(\omega)H(\omega)$ ,  $H(\omega)$  acts as a kind of weighting function or **spectral shaping** function of the different frequency components of the signal.



# Let's Design an Ideal Filter

An **ideal lowpass filter** is given by:

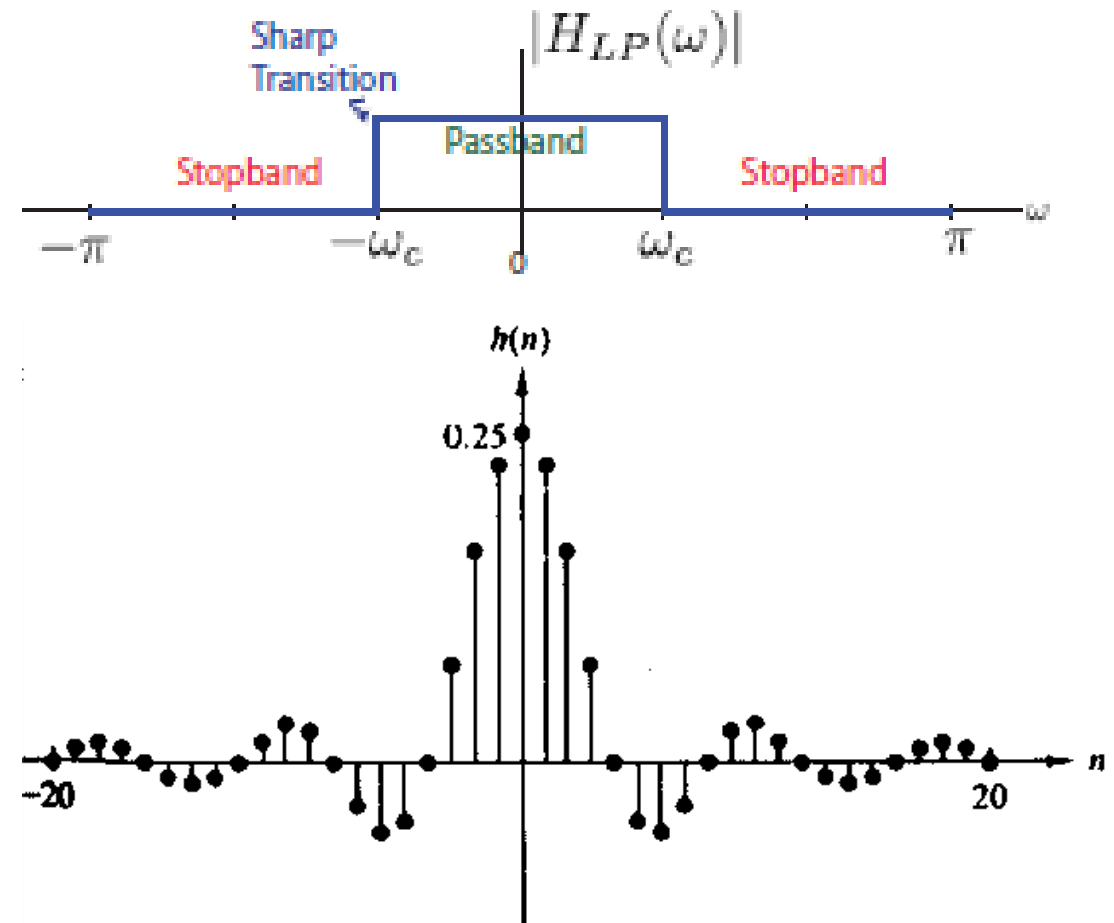
$$H(\omega) = \begin{cases} 1 & |\omega| \leq \omega_c \\ 0 & \omega_c < |\omega| \leq \pi \end{cases}$$

The impulse response is given by:

$$h(n) = \begin{cases} \frac{\omega_c}{\pi} & n = 0 \\ \frac{\omega_c}{\pi} \frac{\sin(\omega_c n)}{\omega_c n} & n \neq 0 \end{cases}$$

$$h(n) \neq 0 \text{ for } n < 0.$$

- Ideal filters are non-causal and hence cannot be realized in practice.

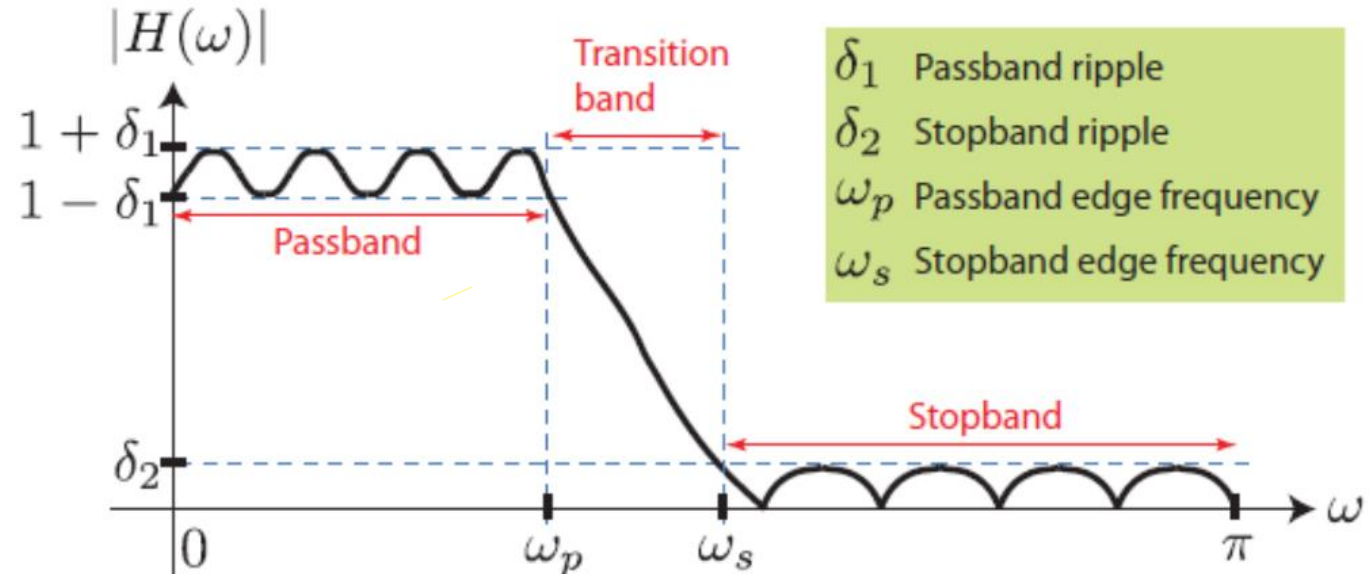


# Practical Filters (Described by LCCDE's)

- Ideal filter characteristics of sharp transitions and flat gains may not be absolutely necessary for most practical applications.
- Relaxing these conditions allows us to realize causal finite parameter filters that can approximate ideal filters.

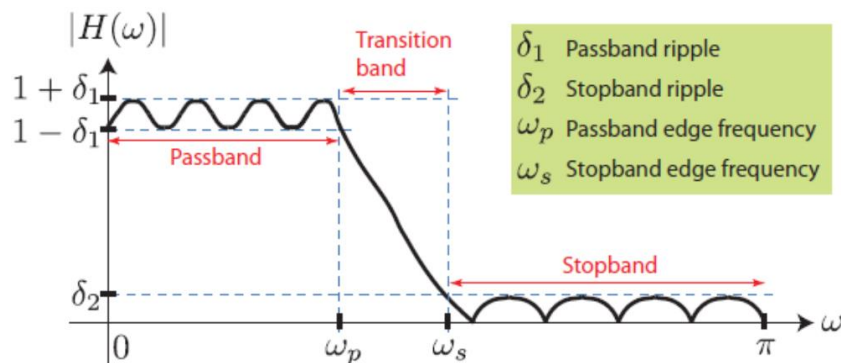
DT LTI Systems  
(Described by LCCDEs)

$$\sum_{k=0}^N a_k y(n-k) = \sum_{k=0}^M b_k x(n-k) \quad a_0 \triangleq 1$$



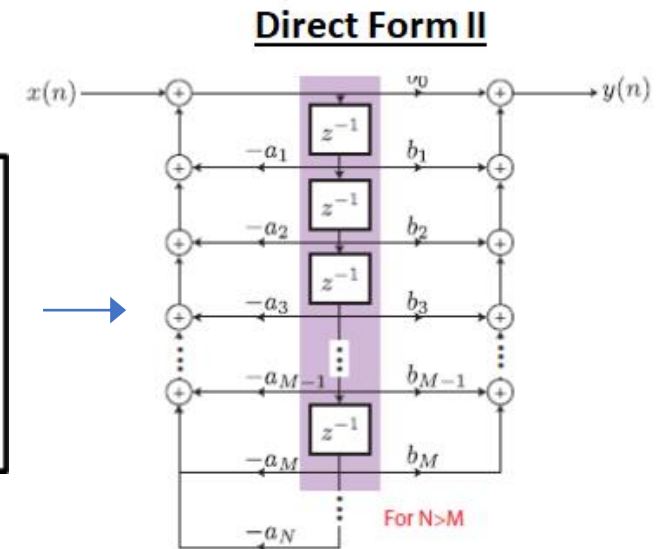
# Three Stages of Filter Design

1. **Specifications** (determined by applications)
2. **Approximation** (come up with a filter description or coefficients to approximate the given specifications)
3. **Implementation** (realize a difference equation as hardware/software)



**DT LTI Systems**  
(Described by LCCDEs)

$$\sum_{k=0}^N a_k y(n-k) = \sum_{k=0}^M b_k x(n-k) \quad a_0 \triangleq 1$$



# Filter Specifications: Magnitude

- $R_p$  is the passband ripple in dB:

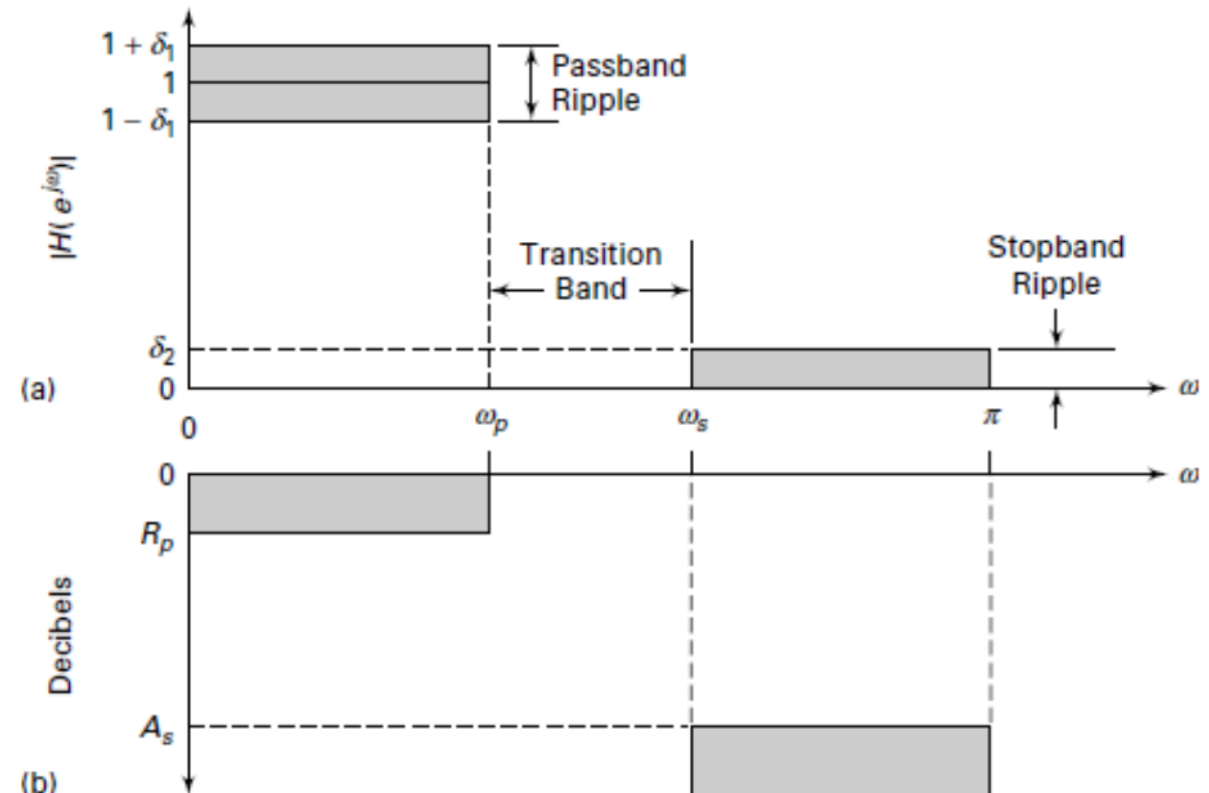
$$R_p = -20 \log_{10} \frac{1 - \delta_1}{1 + \delta_1} > 0 (\approx 0)$$

- $A_s$  is the stopband attenuation in dB:

$$A_s = -20 \log_{10} \frac{\delta_2}{1 + \delta_1} > 0 (>> 1)$$

- Decibel scale:

$$\text{dB scale} = -20 \log_{10} \frac{|H(e^{j\omega})|}{|H(e^{j\omega})|_{\max}} \geq 0$$



filter specifications: (a) absolute (b) relative



# Magnitude Specification: Example

- Given the passband tolerance  $\delta_1 = 0.01$  and the stopband tolerance  $\delta_2 = 0.001$ , determine the passband ripple and the stopband attenuation.

• **Solution:**

$$R_p = -20 \log_{10} \frac{1 - \delta_1}{1 + \delta_1} = -20 \log_{10} \frac{1 - 0.01}{1 + 0.01}$$

$$R_p = 0.1737 \text{ dB}$$

$$A_s = 50 = -20 \log_{10} \frac{\delta_2}{1 + \delta_1} = -20 \log_{10} \frac{0.001}{1 + 0.01}$$

$$\delta_2 = 60 \text{ dB}$$

# Magnitude Specification: Example

□ In a certain filter's specifications, the passband ripple is 0.25 dB and the stopband attenuation is 50 dB. Determine  $\delta_1$  and  $\delta_2$ .

• **Solution:**

$$R_p = 0.25 = -20 \log_{10} \frac{1 - \delta_1}{1 + \delta_1}$$

$$\delta_1 = 0.0144$$

$$A_s = 50 = -20 \log_{10} \frac{\delta_2}{1 + \delta_1} = -20 \log_{10} \frac{\delta_2}{1 + 0.0144}$$

$$\delta_2 = 0.0032$$