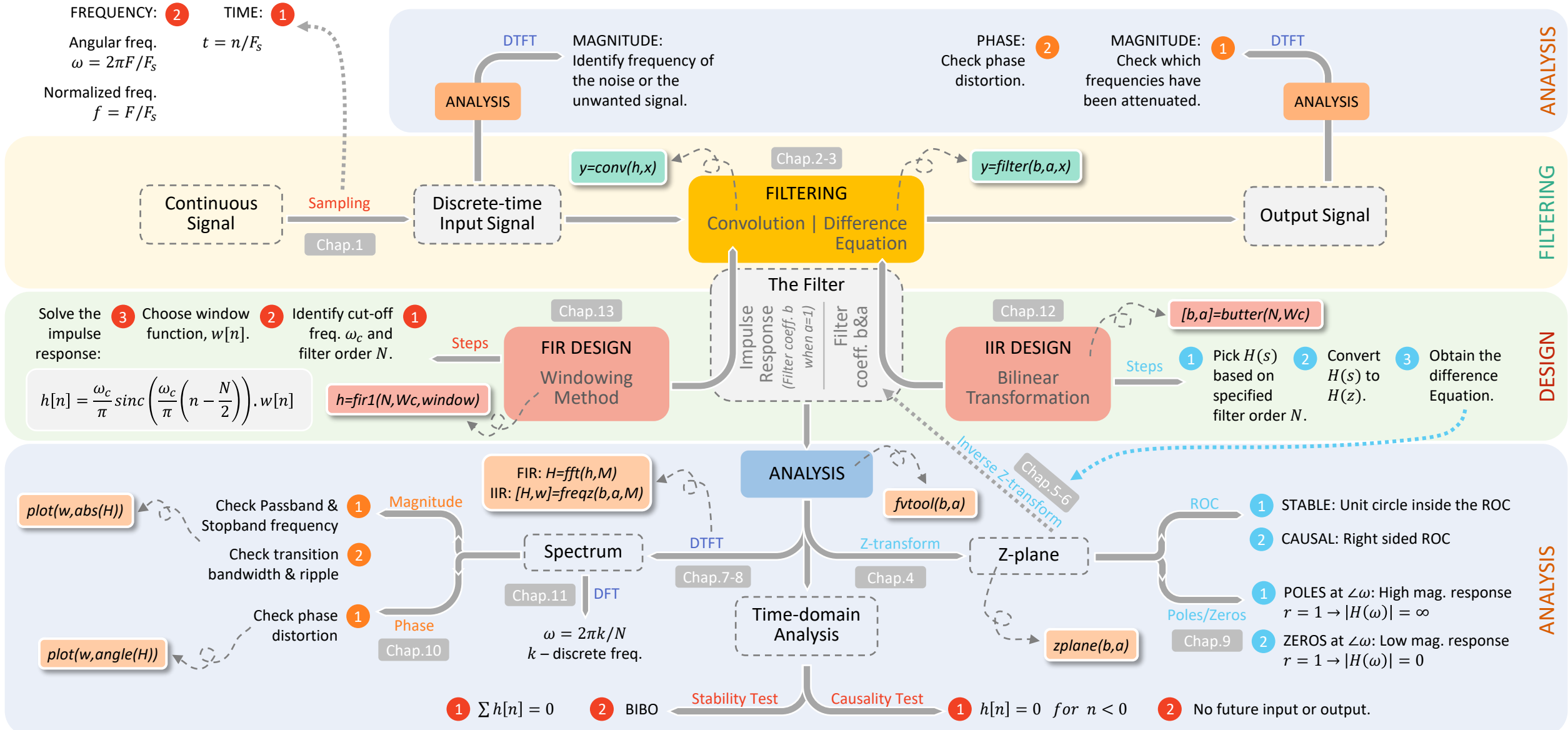


# DSP Concept Mapping





# Discrete-Time Signal

**OPERATION ON SIGNAL VALUES**  
Arithmetic operation is done on each similar sample index.

**SIGNAL LENGTH**  
No of samples counted from the first to the last non-zero samples.

**OPERATION ON INDEX**  
Reordering the samples;  
*Folding, Shifting, Selection.*

Signal Values

$$\mathbf{x}[n] = [ \quad 2.5 \quad 3.1 \quad 4 \quad 2.7 \quad 1.4 \quad ]$$

Math term: *vector*

$$\begin{matrix} n & [ & 0 & 1 & 2 & 3 & 4 & ] \\ & & 0 & T_s & 2T_s & 3T_s & 4T_s & ] t \end{matrix}$$

$$t = nT_s$$

Frequency sampling,  $F_s = 1/T_s$

$F_s \geq 2F_N$ ,  $F_N$  is the signal max freq.

# LTI System

System representation  
based on it's process

DIFFERENCE EQUATION  $\rightarrow$  FIR: Convolution,  $y[n] = x[n]*h[n]$

1. LTI system is represented by *DIFFERENCE EQUATION* restricted to operations:

- i. Delay
- ii. Summation to all samples
- iii. Weight multiplication

2. The weights are the filter, called:

- i. Coefficient a – output samples weights
- ii. Coefficient b – input samples weights

3. LTI system can also be represented by *IMPULSE RESPONSE*,  $h[n]$ :

FIR – equals to coefficient b

IIR – Need z-transform to convert the coefficient a & b to  $h[n]$

$$1y[n] + 5y[n-1] + 3y[n-2] = 2x[n] + 3x[n-1] + 2x[n-2]$$

$$a = [1 \ 5 \ 3] \quad b = [2 \ 3 \ 2]$$

Coefficient a                      Coefficient b

System representation  
based on it's filter

IMPULSE RESPONSE

$h[n]$

FIR  
 $h[n] = b$

IIR  
 $h[n] \xleftrightarrow{\text{z-transform}} a, b$

# Convolution

$$x[n] = [3 \quad 3 \quad 1 \quad 3]$$

$$h[n] = [1 \quad \underline{2} \quad 1]$$

| $n \rightarrow$    |   | -1 | 0        | 1        | 2  | 3 | 4 |   |   |
|--------------------|---|----|----------|----------|----|---|---|---|---|
| $x[n] \rightarrow$ |   | [  | <u>3</u> | 3        | 1  | 3 | ] |   |   |
| $h[n] \rightarrow$ | x | [  | 1        | <u>2</u> | 1  |   | ] |   |   |
|                    |   |    | 3        | 3        | 1  | 3 |   |   |   |
|                    |   |    |          | 6        | 6  | 2 | 6 |   |   |
|                    | + |    |          |          | 3  | 3 | 1 | 3 |   |
| $y[n] \rightarrow$ |   | [  | 3        | <u>9</u> | 10 | 8 | 7 | 3 | ] |

# Stability and Causality Check

**CAUSALITY:** NO FUTURE INPUT/OUTPUT  
**STABILITY:** BIBO

DIFFERENCE EQUATION

$$1y[n] = 2x[n] + 4x[n-1]$$

**CAUSALITY:**  
 $h[n] = 0$  for  $n < 0$   
**STABILITY:**  $\sum |h[n]| < \infty$

IMPULSE RESPONSE

$$h[n] = [2 \ 4]$$

**CAUSALITY:** ROC OUTWARD  
**STABILITY:** ROC INCLUDE UNIT CIRCLE

Z-DOMAIN ROC

$$H(z) = 2 + 4z^{-1}$$

$$\text{ROC: } |z| > 0$$

Z-TRANSFORM  
----->

$$1y[n] - 0.25y[n-2] = 1x[n] + 2x[n-1]$$

$$h[n] = 2.5(0.5)^n u[n] - 1.5(-0.5)^n u[n]$$

INVERSE Z-TRANSFORM  
-----<

$$H(z) = \frac{1z^0 + 2z^{-1}}{1z^0 + 0z^{-1} - 0.25z^{-2}}$$

Poles (Denominator Roots):

$$1z^2 + 0z^1 - 0.25z^0 = 0$$

$$(z - 0.5)(z + 0.5) = 0$$

$$\text{ROC: } |z| > 0.5$$

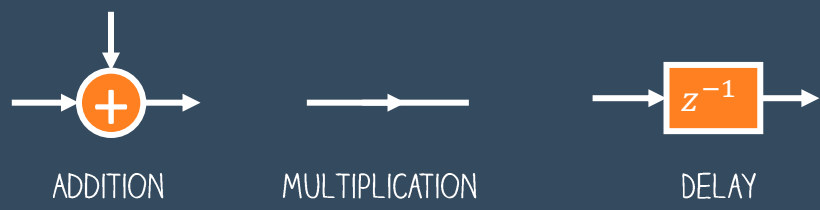
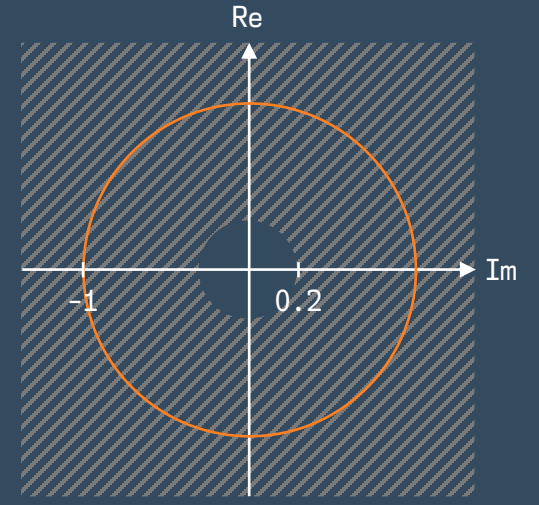
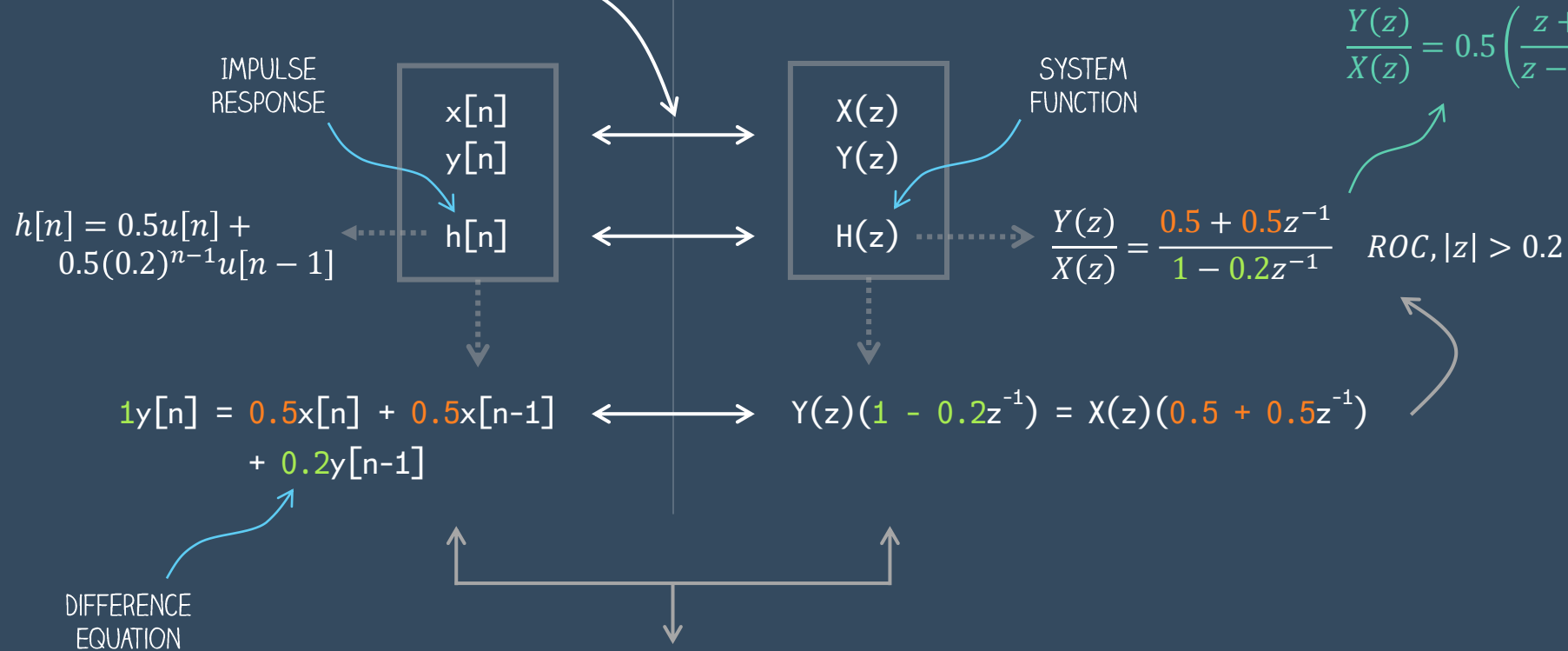
CHECKING BIBO FOR STABILITY IS NOT ALWAYS EASY SINCE WE NEED TO TAKE INTO ACCOUNT ALL POSSIBLE INPUTS. AS AN ALTERNATIVE, IT IS EASIER TO CHECK STABILITY ON THE IMPULSE RESPONSE.

CONVERTING DIFFERENCE EQUATION TO  $h[n]$  IS NOT STRAIGHT FORWARD FOR IIR. THUS, FOR IIR, CHECKING THE Z-DOMAIN ROC WILL BE THE EASIEST WAY TO CHECK STABILITY.

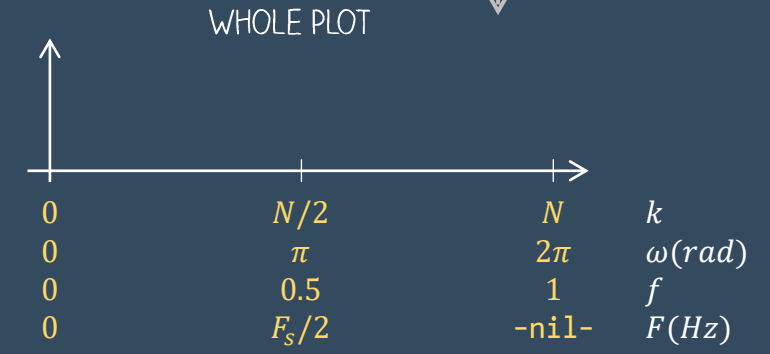
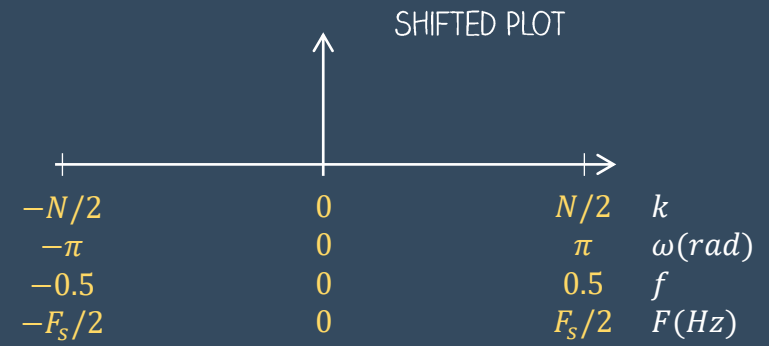
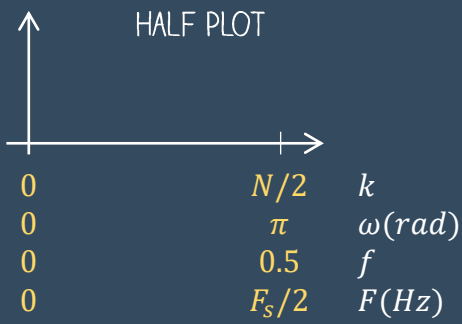
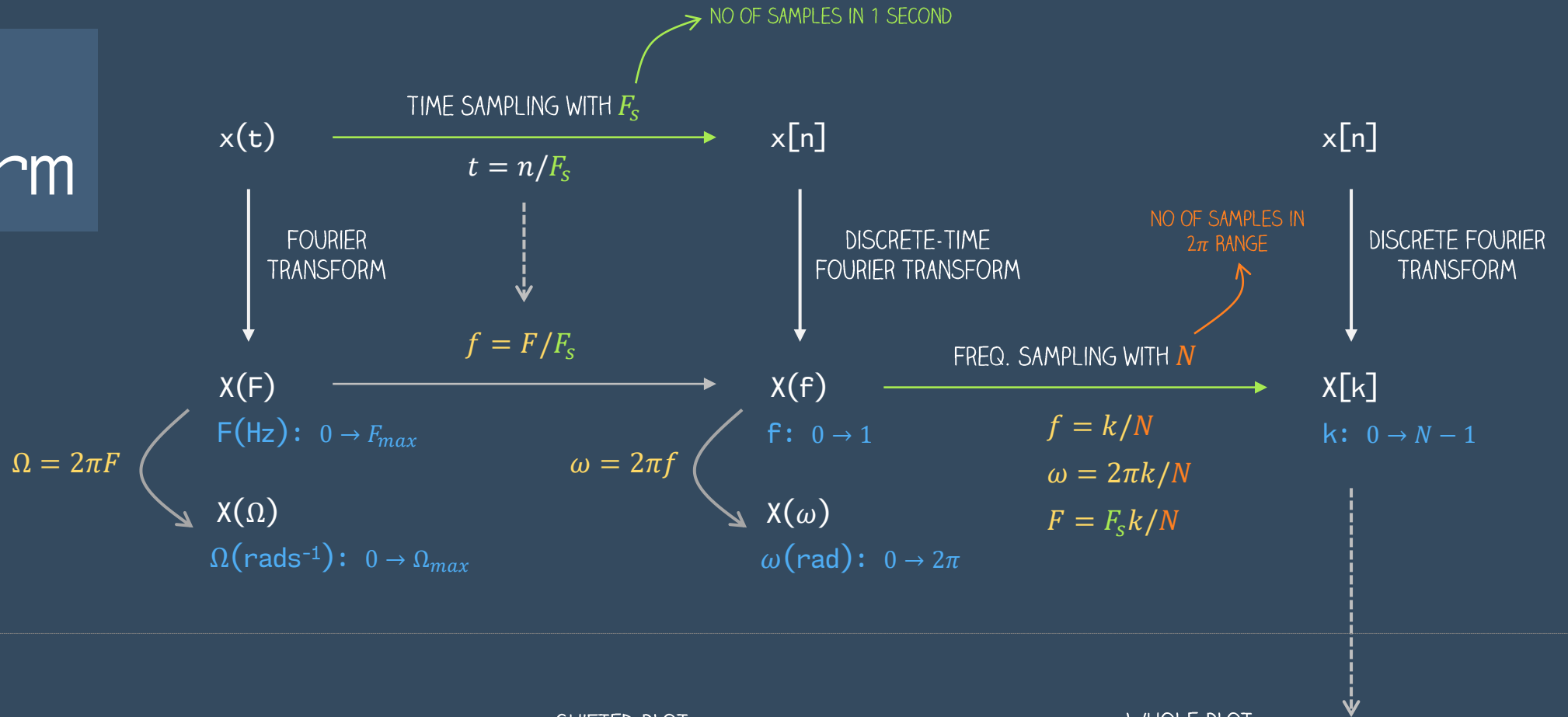
# Z-Transform

TABLE 1 : Z-TRANSFORM PAIR  
TABLE 2 : Z-TRANSFORM PROPERTIES

| TIME-DOMAIN | Z-DOMAIN                 |
|-------------|--------------------------|
| CONVOLUTION | MULTIPLICATION           |
| DELAY $n_d$ | MULTIPLY WITH $z^{-n_d}$ |

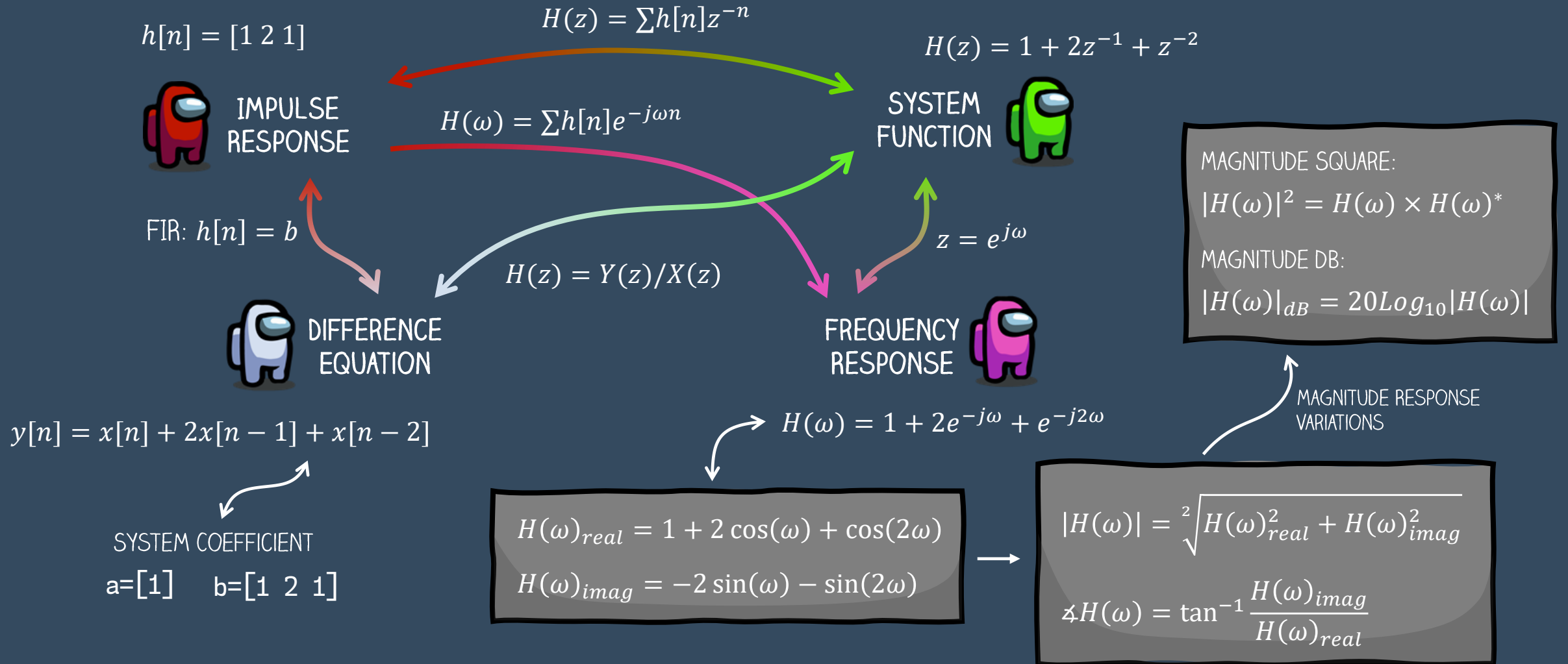


# Fourier Transform





# Frequency Response of a System



# Filtering

## Time Domain

Implementing the filtering process between input  $x[n]$  and impulse response  $h[n]$  or filter coefficient a & b

Convolution → FIR  
 $y[n] = x[n] * h[n]$

Difference Equation → FIR & IIR  
 $y[n] = \sum b_i x[n - i] + \sum a_j y[n - j]$

## Z-Domain

Analysing & Designing the filter in terms of the system function  $H(z)$

ROC → Analyse Stability & Causality.

Poles & Zeros → Analyse and design the frequency response.

## Frequency Domain

Analysing the freq. response of the input  $X(\omega)$ , the output  $Y(\omega)$  and the filter  $H(\omega)$

### MAGNITUDE RESPONSE

Input,  $|X(\omega)|$  - To identify the unwanted frequencies.  
Filter,  $|H(\omega)|$  - To understand which frequency will be attenuated, preserved or amplified.  
Output,  $|Y(\omega)|$  - To check the effect of the filtering on the frequency components.

### PHASE RESPONSE

Filter,  $\angle H(\omega)$  - To understand the type and amount of delay the filter will cause.  
Output,  $\angle Y(\omega)$  - To check the delay effect.

## Time Domain

# Filtering

## Z-Domain

### 1. Solve Difference Equation

$$y[n] = \sum_{k=0}^M b_k x[n-k] + \sum_{k=1}^N a_k x[n-k]$$

### 2. Convolution

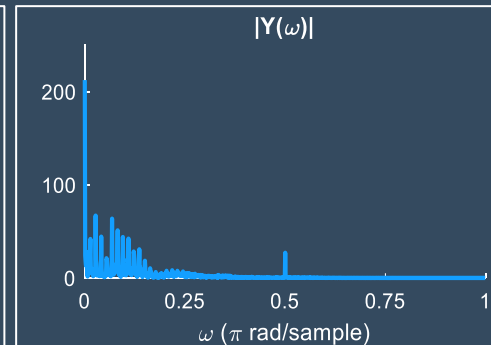
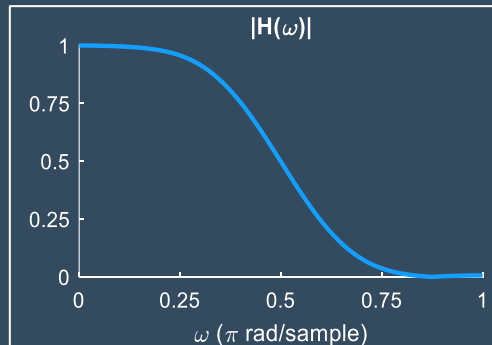
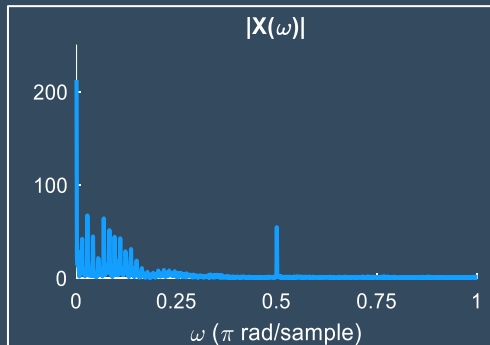
| n →    | -1  | 0 | 1 | 2  | 3 | 4 |   |   |
|--------|-----|---|---|----|---|---|---|---|
| x[n] → | [   | 3 | 3 | 1  | 3 | ] |   |   |
| h[n] → | x [ | 1 | 2 | 1  |   | ] |   |   |
|        |     | 3 | 3 | 1  | 3 |   |   |   |
|        |     |   | 6 | 6  | 2 | 6 |   |   |
|        |     |   |   | 3  | 3 | 1 | 3 |   |
| y[n] → | [   | 3 | 9 | 10 | 8 | 7 | 3 | ] |

## Frequency Domain

$$Y(\omega) = X(\omega) \cdot H(\omega)$$

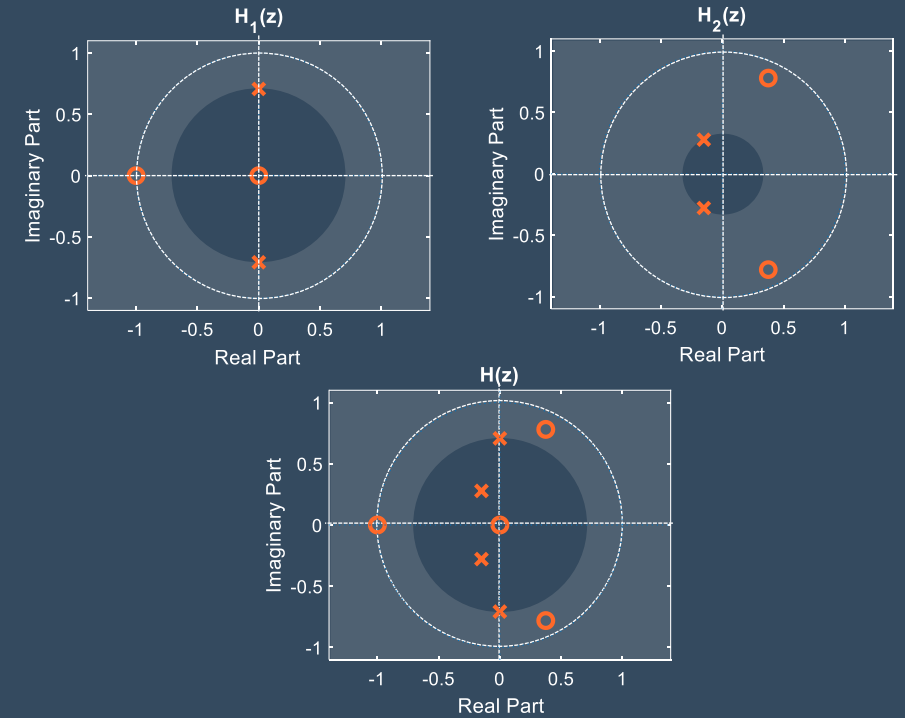
$$|Y(\omega)| = |X(\omega)| \cdot |H(\omega)|$$

$$\angle Y(\omega) = \angle X(\omega) + \angle H(\omega)$$



Represent the filter with system function,  $H(z) = \frac{Y(z)}{X(z)}$

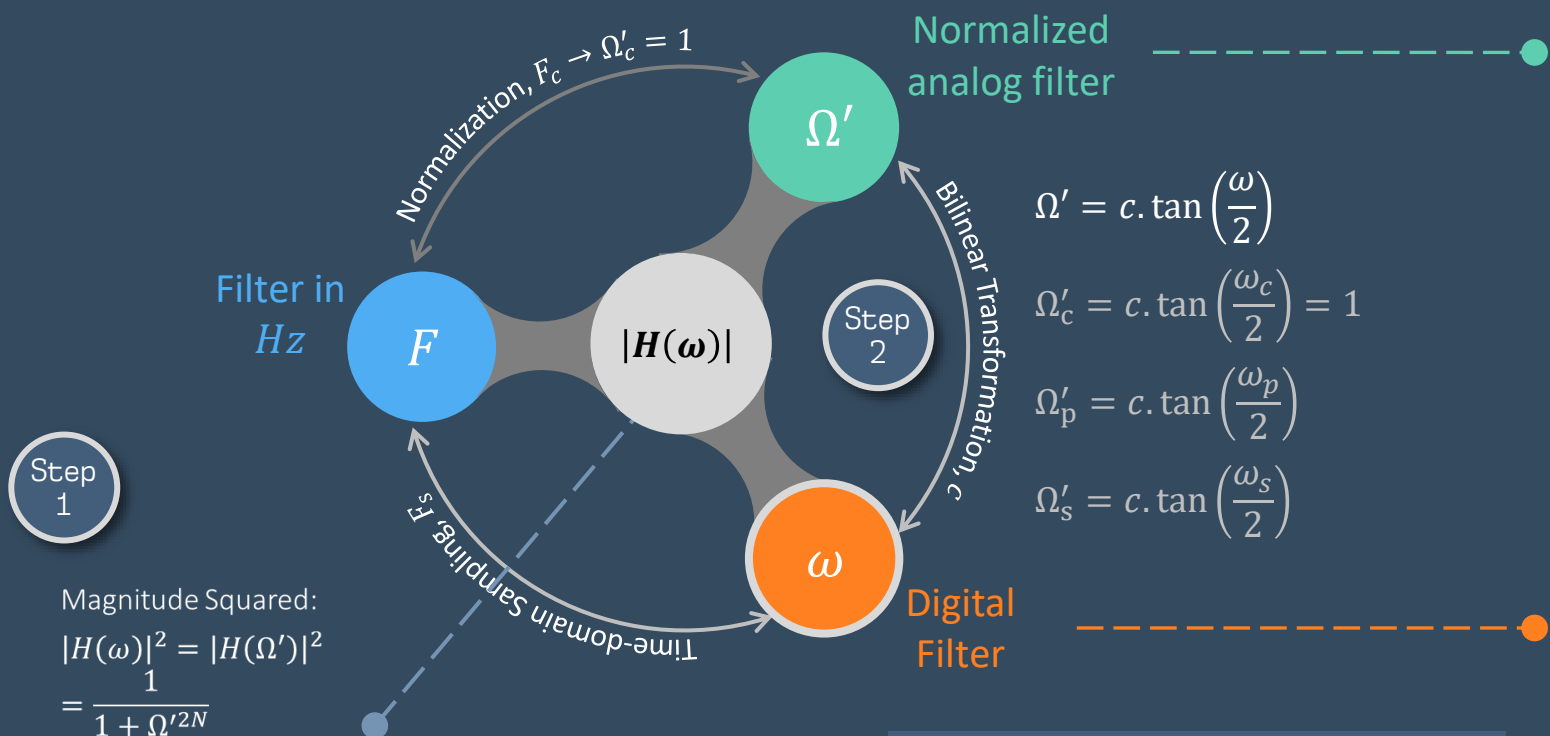
For two cascaded filters,  $H(z) = H_1(z) \cdot H_2(z)$ :



\*Combine both poles and zeros

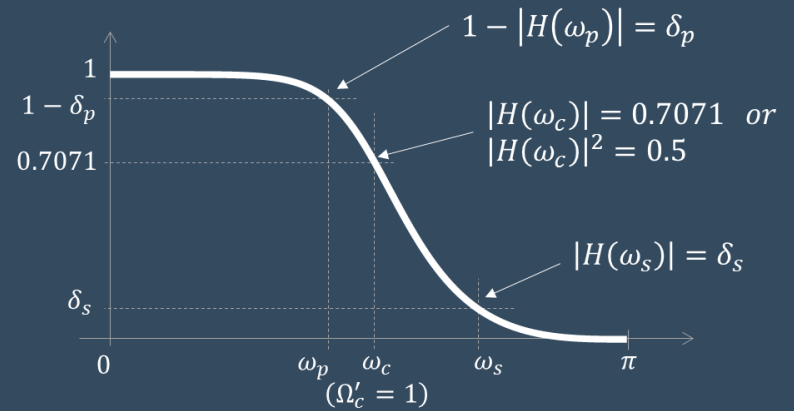
$$*ROC_{H(z)} = ROC_{H_1(z)} \cap ROC_{H_2(z)}$$

# IIR Butterworth Filter Design



Step 1

Magnitude Squared:  
 $|H(\omega)|^2 = |H(\Omega')|^2$   
 $= \frac{1}{1 + \Omega'^{2N}}$



- ### Design Step
1. Specify filter characteristic:  $\omega_c, N, \delta_p, \delta_s, \omega_p, \omega_s$ .
  2. Find filter order  $N$  (if not specified in step 1) and parameter  $c$ .
  3. Based on the  $N$  and  $c$ , convert  $H(s)$  to  $H(z)$ .
  4. Transform  $H(z)$  to the time-domain difference equation

$$\Omega' = c \cdot \tan\left(\frac{\omega}{2}\right)$$

$$\Omega'_c = c \cdot \tan\left(\frac{\omega_c}{2}\right) = 1$$

$$\Omega'_p = c \cdot \tan\left(\frac{\omega_p}{2}\right)$$

$$\Omega'_s = c \cdot \tan\left(\frac{\omega_s}{2}\right)$$

Normalized analog filter system function,  $H(s)$

|                       |                                  |
|-----------------------|----------------------------------|
| 1 <sup>st</sup> Order | $\frac{1}{s + 1}$                |
| 2 <sup>nd</sup> Order | $\frac{1}{s^2 + 1.4142s + 1}$    |
| 3 <sup>rd</sup> Order | $\frac{1}{(s + 1)(s^2 + s + 1)}$ |

$a_i$  – normalized filter coefficient

$$d_1 = 1 + c$$

$$d_2 = 1 - c$$

$$b_{i1} = c^2 + a_i c + 1$$

$$b_{i2} = -2c^2 + 2$$

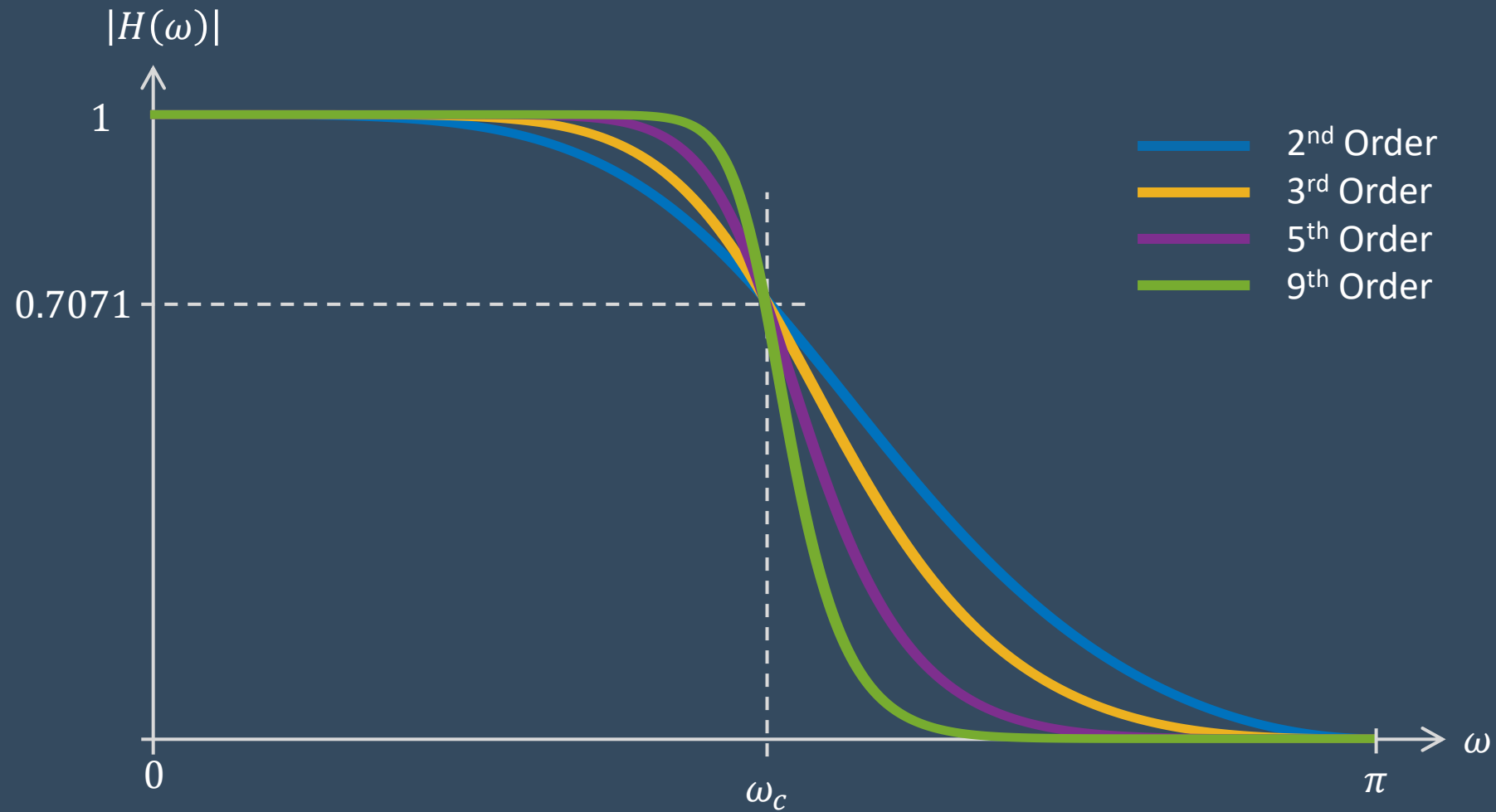
$$b_{i3} = c^2 - a_i c + 1$$

Step 3

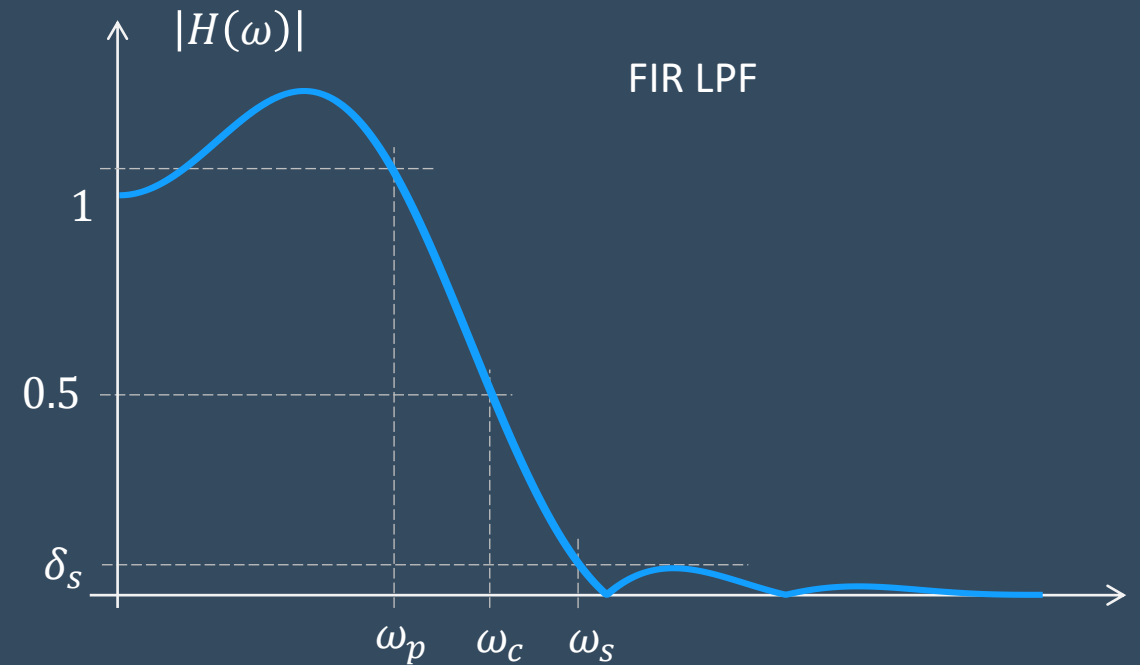
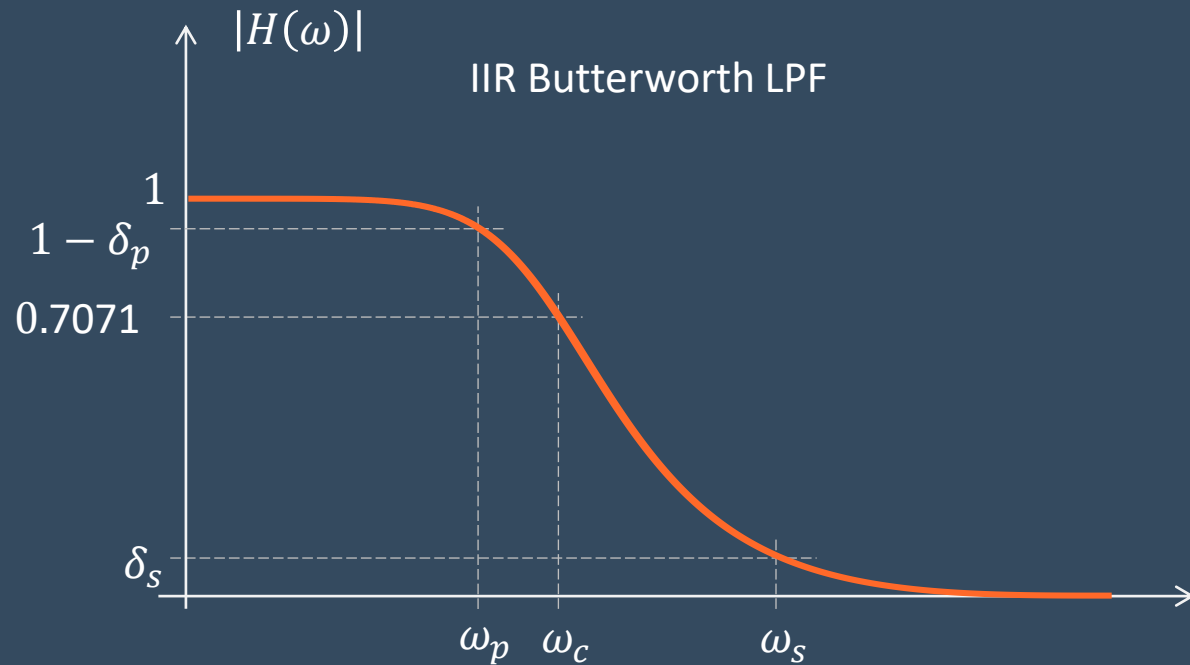
Digital filter system function,  $H(z)$

|                       |   |
|-----------------------|---|
| 1 <sup>st</sup> Order | $\frac{1 + z^{-1}}{d_1 + d_2 z^{-1}}$   |
| 2 <sup>nd</sup> Order | $\frac{(1 + z^{-1})^2}{b_{11} + b_{12} z^{-1} + b_{13} z^{-2}}$                     |
| 3 <sup>rd</sup> Order | $\frac{(1 + z^{-1})^3}{(d_1 + d_2 z^{-1})(b_{11} + b_{12} z^{-1} + b_{13} z^{-2})}$ |

# IIR Butterworth Lowpass Filter

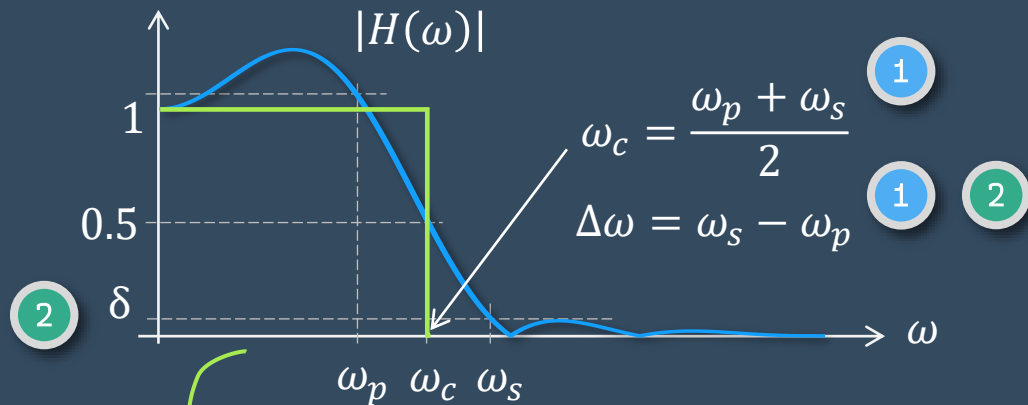


# IIR Butterworth Filter vs FIR Filter

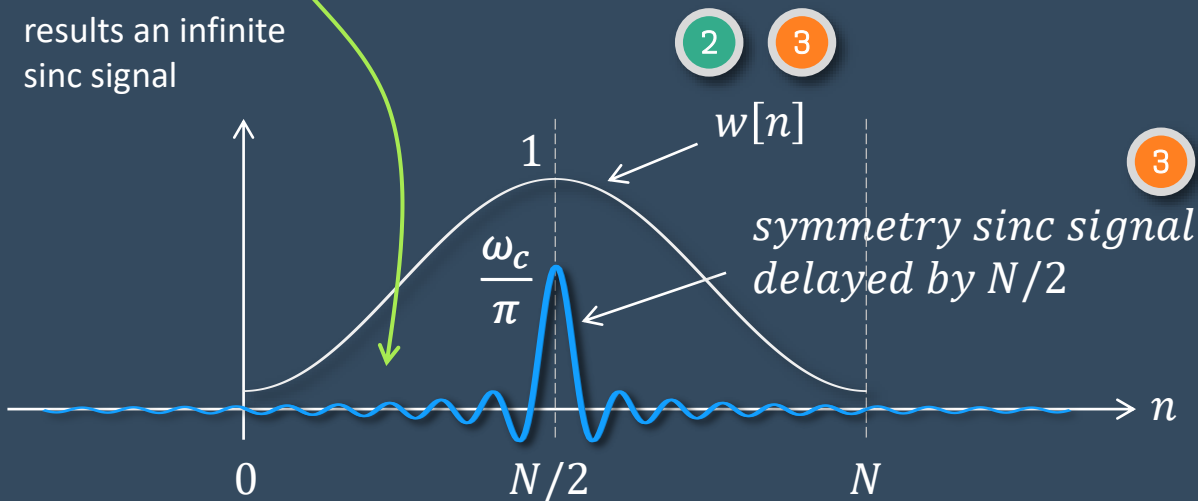


| Characteristic                | IIR Butterworth filter          | FIR Filter                                  |
|-------------------------------|---------------------------------|---|
| Cut-off frequency, $\omega_c$ | Closer to $\omega_p$            | At center between $\omega_p$ and $\omega_s$ |
| $ H(\omega_c) $               | Half power $\rightarrow$ 0.7071 | Half magnitude $\rightarrow$ 0.5            |
| Ripple                        | Gradually decreasing            | Fluctuated                                  |

# FIR Filter Design



Inverse DTFT of the ideal filter results an infinite sinc signal



Step 1

DETERMINE CUT-OFF FREQUENCY,  $\omega_c$

Either empirically specified or computed from the transition bandwidth,  $\omega_p$  and  $\omega_s$  values.

Step 2

DETERMINE FILTER ORDER,  $N$  AND WINDOW FUNCTION,  $w[n]$

Either empirically specified or computed from the transition bandwidth and ripple,  $\delta$  values.

| window      | $\delta$ , dB; (Magnitude) | $\Delta\omega$ |
|-------------|----------------------------|----------------|
| Rectangular | -20.9 dB; (0.0902)         | $1.8\pi/N$     |
| Hanning     | -43.9 dB; (0.0064)         | $6.2\pi/N$     |
| Hamming     | -54.5 dB; (0.0019)         | $6.6\pi/N$     |
| Blackman    | -75.3 dB; (0.0002)         | $11.1\pi/N$    |

Step 3

SOLVE THE IMPULSE RESPONSE EQUATION

$$h[n] = \frac{\omega_c}{\pi} \text{sinc} \left( \frac{\omega_c}{\pi} (n - N/2) \right) \cdot w[n]$$

# Tangent Inverse

Note:

This plot is not the z-domain where the axes are written as real (x-axis) & imag (y-axis). This plot is to illustrate the complex value of  $H(\omega)$

