Lecture 8 LTI Discrete-Time Systems in the Transform Domain

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Types of Transfer Functions

LTI
$$\frac{x[n]}{X(z)} \xrightarrow{h[n]} y[n]$$

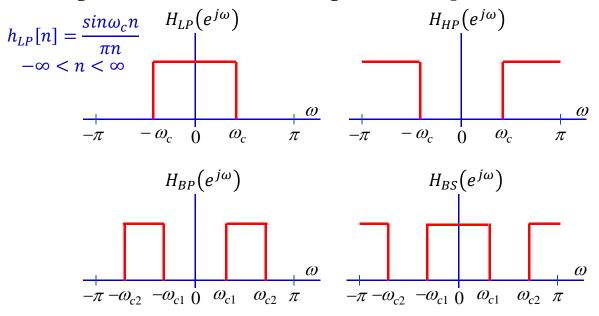
$$y[n] \xrightarrow{Y(z)} Y(z)$$

$$y[n] = \sum_{k=-\infty}^{\infty} x[k]h[n-k], \quad Y(z) = X(z)H(z)$$

- For digital transfer function with frequency selective frequency response, there are two types of classifications.
 - Based on the shape of magnitude function $|H(e^{j\omega})|$
 - Based on the form of phase function $\theta(\omega)$

Magnitude Characteristics

Digital Filter with Ideal Magnitude Responses:



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- The range of frequencies where the magnitude response takes the value of one is called the passband
- The range of frequencies where the magnitude response takes the value of zero is called the stopband
- The frequencies , ω_c , ω_{c1} and ω_{c2} are called the **cutoff** frequencies
- An ideal filter has a magnitude response equal to one in the passband and zero in the stopband, and has a zero phase everywhere

Phase Characteristics

- Linear Phase Transfer Function
 - In many application, it is necessary that the digital filter designed does not distort the phase of the input signal components with frequencies in passband.
- The most general type of a filter with a linear phase has a frequency response given by

$$H(e^{j\omega}) = Ae^{-j\omega D}$$

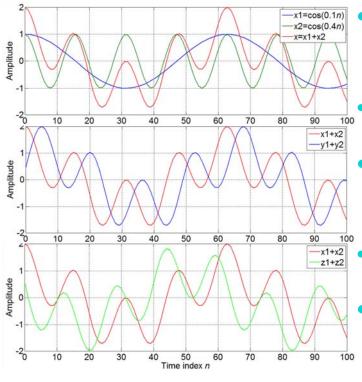
which has a linear phase from $\omega = 0$ to $\omega = 2\pi$.

• Note: $\theta(\omega) = -\omega D$.

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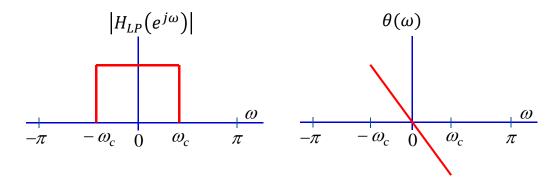
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Phase Distortion



- $x = x_1 + x_2$ $x_1 = \cos(0.1n)$ $x_2 = \cos(0.4n)$
- $Y(e^{j\omega}) = X(e^{j\omega})H_1(e^{j\omega})$ $\theta_1(\omega) = \angle H_1(e^{j\omega}) = -5\omega$
- $y = y_1 + y_2$ $y_1 = \cos(0.1(n-5))$ $y_2 = \cos(0.4(n-5))$
 - $Z(e^{j\omega}) = X(e^{j\omega})H_2(e^{j\omega})$ $\theta_2(\omega) = \angle H_2(e^{j\omega}) = -5\operatorname{sign}(\omega)$
- $z = z_1 + z_2$ $z_1 = \cos(0.1n - 5)$ $z_2 = \cos(0.4n - 5)$

- It is desired to pass input signal components in a certain frequency range undistorted in both magnitude and phase
- The transfer function should exhibit a unity magnitude response and a linear phase response in the band of interest.



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Linear-Phase FIR Transfer Functions

- It is impossible to design an IIR transfer function with an exact linear-phase
- It is always possible to design an FIR transfer function with an exact linear-phase response
- We consider real impulse response h[n]

Simple Examples

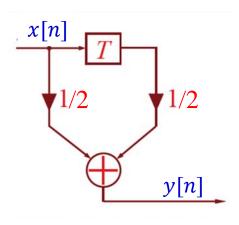
•
$$H(z) = \frac{1+z^{-1}}{2} \leftrightarrow \{h[n]\} = \left\{\frac{1}{2}, \frac{1}{2}\right\}$$

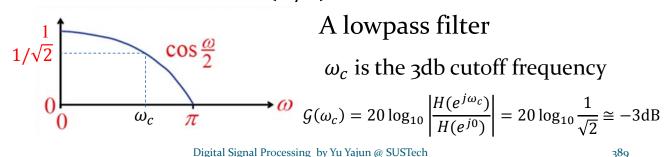
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$$H(e^{j\omega}) = \left(1 + e^{-j\omega}\right)/2$$

$$= e^{-j\omega/2} \frac{e^{j\omega/2} + e^{-j\omega/2}}{2}$$

$$= e^{-j\omega/2} \cos(\omega/2)$$





A lowpass filter

$$G(\omega_c) = 20 \log_{10} \left| \frac{H(e^{j\omega_c})}{H(e^{j0})} \right| = 20 \log_{10} \frac{1}{\sqrt{2}} \approx -3 \text{dB}$$

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Simple Examples

•
$$H(z) = \frac{1-z^{-1}}{2} \leftrightarrow \{h[n]\} = \{\frac{1}{2}, -\frac{1}{2}\}$$

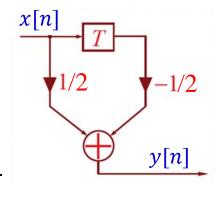
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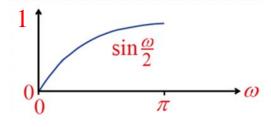
$$H(z_i\omega) = (1 - z_i\omega)/2$$

$$H(e^{j\omega}) = (1 - e^{-j\omega})/2$$

$$= e^{j\pi/2} e^{-j\omega/2} \frac{e^{j\omega/2} - e^{-j\omega/2}}{2j}$$

$$= e^{j(\pi/2 - \omega/2)} \sin(\omega/2)$$





A highpass filter

Linear Phase FIR Filter

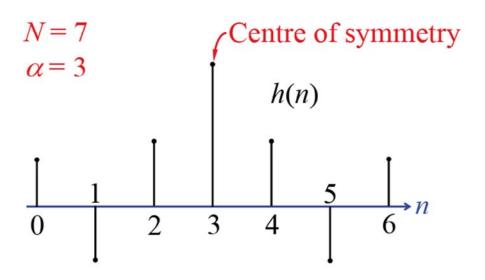
- An FIR filter may be designed to have linear phase characteristics. The phase response, $\theta(\omega)$, of a linear phase FIR filter is $\beta \alpha \omega$, where $\alpha = \frac{N-1}{2}$, ω is the frequency, $\beta = 0$ or $\pm 0.5\pi$ and N is the filter length.
- Its frequency response is given by $e^{-j\left(\frac{N-1}{2}\omega-\beta\right)}R(\omega)$, where $R(\omega)$ is a real function.
- The group delay is $-d\{\theta(\omega)\} = \alpha$.

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- Its impulse response is either symmetrical or antisymmetrical.
- If its impulse response is symmetrical, its phase response is $-\alpha\omega$.
- If its impulse response is anti-symmetrical, its phase response is $\pm 0.5\pi \alpha\omega$.

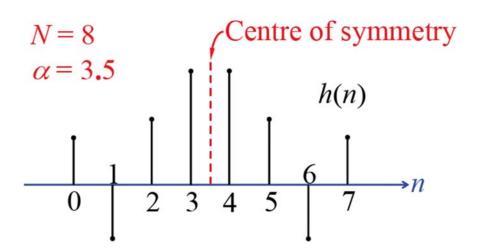
Example: Symmetrical impulse response, N odd, where N is the length of the impulse response



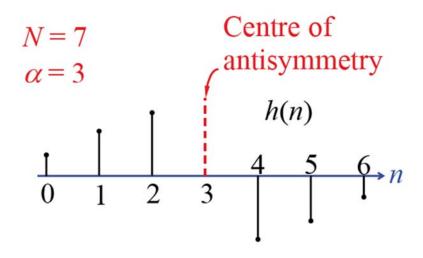
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Example: Symmetrical impulse response, N even.



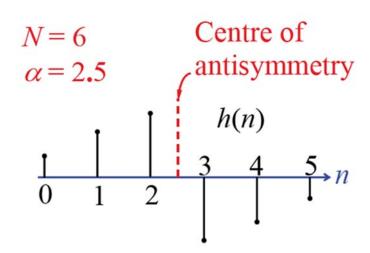
Example: Anti-symmetrical impulse response, N odd.



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Example: Anti-symmetrical impulse response, N even.



Frequency response of linear phase FIR filter

- Four cases, depending on whether *N* is odd or even and whether the impulse response is symmetrical or antisymmetrical.
- Type I: Symmetrical impulse response, *N* odd.

$$H(e^{j\omega}) = \sum_{n=0}^{\frac{N-3}{2}} h[n]e^{-j\omega n} + h\left[\frac{N-1}{2}\right]e^{-j\omega\frac{N-1}{2}} + \sum_{n=\frac{N+1}{2}}^{N-1} h[n]e^{-j\omega n}$$

$$= e^{-j\omega\frac{N-1}{2}} \left[\sum_{n=0}^{\frac{N-3}{2}} h[n]\left(e^{j\omega\left(\frac{N-1}{2}-n\right)} + e^{-j\omega\left(\frac{N-1}{2}-n\right)}\right) + h\left[\frac{N-1}{2}\right]\right]$$

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$$= e^{-j\omega \frac{N-1}{2}} \left[\sum_{n=0}^{(N-3)/2} 2h[n] \cos\left(\omega \left(\frac{N-1}{2}-n\right)\right) + h\left[\frac{N-1}{2}\right] \right]$$

$$\xrightarrow{m=\frac{N-1}{2}-n} e^{-j\omega \frac{N-1}{2}} \left[\sum_{m=1}^{(N-1)/2} 2h\left[\frac{N-1}{2}-m\right] \cos(\omega m) + h\left[\frac{N-1}{2}\right] \right]$$

$$\therefore H(e^{j\omega}) = e^{-j\omega \frac{N-1}{2}} \sum_{n=0}^{(N-1)/2} a[n] \cos(\omega n)$$

$$a[0] = h\left[\frac{N-1}{2}\right]$$

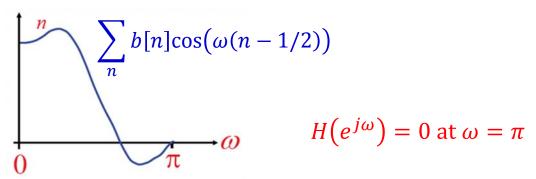
$$a[n] = 2h\left[\frac{N-1}{2}-n\right],$$
for $n = 1, 2, ..., \frac{N-1}{2}$

Type II:

• Symmetrical impulse response, *N* even

$$H(e^{j\omega}) = e^{-j\omega \frac{N-1}{2}} \sum_{n=1}^{N/2} b[n] \cos(\omega(n-1/2))$$

$$b[n] = 2h[N/2 - n], \text{ for } n = 1, 2, ..., N/2$$



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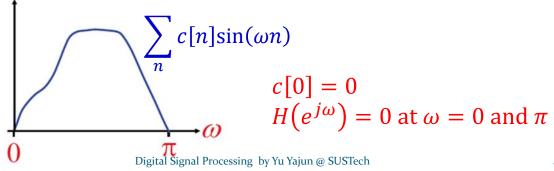
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Type III:

Anti-symmetrical impulse response, N odd

$$H(e^{j\omega}) = e^{-j\omega \frac{N-1}{2}} e^{j\frac{\pi}{2}} \sum_{n=1}^{(N-1)/2} c[n] \sin(\omega n)$$

$$c[n] = 2h \left[\frac{N-1}{2} - n \right], \text{ for } n = 1, 2, ..., \frac{N-1}{2}$$



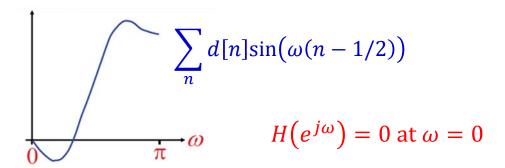
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Type IV:

Anti-symmetrical impulse response, N even

$$H(e^{j\omega}) = e^{-j\omega \frac{N-1}{2}} e^{j\frac{\pi}{2}} \sum_{n=1}^{N/2} d[n] \sin(\omega(n-1/2))$$

$$d[n] = 2h[N/2 - n], \text{ for } n = 1, 2, ..., N/2$$



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Mirror Image Polynomial

• For an FIR filter with a symmetric impulse response, its transfer function H(z) can be written as:

$$H(z) = \sum_{n=0}^{N-1} h[n]z^{-n} = \sum_{n=0}^{N-1} h[N-1-n]z^{-n}$$

$$\xrightarrow{m=N-1-n} H(z) = \sum_{m=0}^{N-1} h[m]z^{-N+1+m} = z^{-(N-1)} \sum_{m=0}^{N-1} h[m]z^{m} = z^{-(N-1)}H(z^{-1})$$

• A real-coefficient polynomial H(z) satisfying the above condition is called a **mirror-image polynomial**.

Example:
$$H_1(z) = -1 + 2z^{-1} - 3z^{-2} + 6z^{-3} - 3z^{-4} + 2z^{-5} - z^{-6}$$

Amplitude response: $\check{H}_1(\omega) = 6 - 6\cos(\omega) + 4\cos(2\omega) - 2\cos(3\omega)$
Phase response: $\theta_1(\omega) = -3\omega$

• If $H_2(z) = -H_1(z)$, then $\check{H}_2(\omega) = \check{H}_1(\omega)$, and $\theta_2(\omega) = -3\omega + \pi$

Antimirror-Image Polynomial

• Similarly, for an FIR filter with an anti-symmetric impulse response, its transfer function H(z) can be written as:

$$H(z) = \sum_{n=0}^{N-1} h[n]z^{-n} = \sum_{n=0}^{N-1} -h[N-1-n]z^{-n} = -z^{-(N-1)}H(z^{-1})$$

• A real-coefficient polynomial H(z) satisfying the above condition is called an antimirror-image polynomial.

Example:
$$H_3(z) = 1 - 2z^{-1} + 3z^{-2} - 3z^{-4} + 2z^{-5} - z^{-6}$$

Amplitude response: $\check{H}_3(\omega) = 6\sin(\omega) - 4\sin(2\omega) + 2\sin(3\omega)$

Phase response: $\theta_3(\omega) = -3\omega + \pi/2$

• If $H_4(z) = -H_3(z)$, then $\check{H}_4(\omega) = \check{H}_3(\omega)$, and $\theta_4(\omega) = -3\omega - \pi/2$

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Mirror Image Symmetry of Zeros

- The zeros of linear phase FIR filter with real coefficients exhibit mirror image symmetry with respect to unit circle.
- $H(z) = z^{-(N-1)}H(z^{-1})$ or $H(z) = -z^{-(N-1)}H(z^{-1})$
- Zeros are in forms of:
 - Real zeros

•
$$z = r$$
 and $z = \frac{1}{r}$, $r \neq \pm 1$

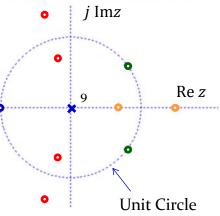
$$\Rightarrow (z - r)(z - r^{-1}) \Rightarrow 1 + az^{-1} + z^{-2}$$

•
$$z = -1$$
 or $z = 1 \implies (z + 1)$ or $(z - 1)$

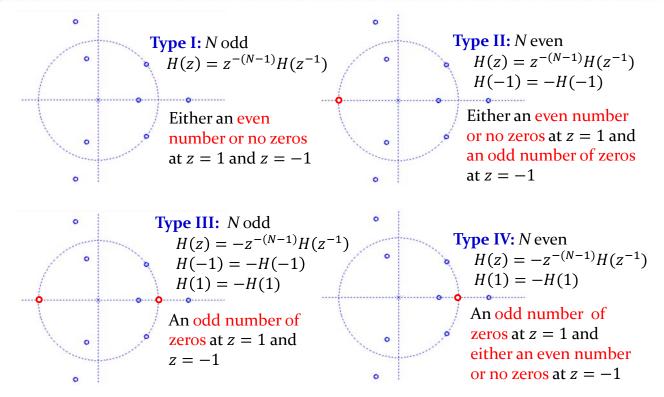
Complex zeros

•
$$z = re^{\pm j\varphi}$$
 and $z = \frac{1}{r}e^{\pm j\varphi}$, $r \neq \pm 1$
 $\Rightarrow 1 + az^{-1} + cz^{-2} + az^{-3} + z^{-4}$

•
$$z = e^{\pm j\varphi} \implies 1 + az^{-1} + z^{-2}$$



Zero-Locations of FIR Filters



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Zero-Phase FIR Filters

- The impulse response h[n] is symmetric around h[0], i. e., h[n] = h[-n].
- Thus, the frequency response $H(e^{j\omega})$ is real, but not necessary positive (unlike $|H(e^{j\omega})|$)
- For example, the type I FIR filter:

$$H(e^{j\omega}) = \sum_{n=-\frac{N-1}{2}}^{-1} h[n]e^{-j\omega n} + h[0] + \sum_{n=1}^{\frac{N-1}{2}} h[n]e^{-j\omega n}$$
$$= h[0] + 2\sum_{n=1}^{\frac{N-1}{2}} h[n]\cos(\omega n)$$

 Zero-phase FIR filter is not causal, but the causality may be recovered by delaying the impulse response

Simple IIR Filters

General Form

$$y[n] = \sum_{m=0}^{M} a_m x[n-m] - \sum_{m=1}^{N} b_m y[n-m]$$

$$\frac{Y(z)}{X(z)} = H(z) = \frac{\sum_{m=0}^{M} a_m z^{-m}}{1 + \sum_{m=1}^{N} b_m z^{-m}}$$

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Lowpass & Highpass IIR Filter

$$H(z) = \frac{K}{1 - \alpha z^{-1}}, \qquad 0 < |\alpha| < 1, \qquad \alpha, K \text{ are real}$$

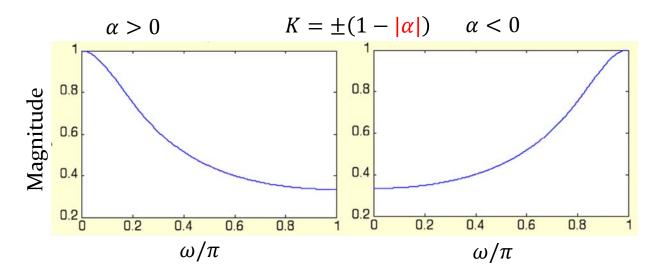
Its squared-magnitude function is given by

$$\left|H(e^{j\omega})\right|^2 = H(z)H(z^{-1})\Big|_{z=e^{j\omega}} = \frac{K^2}{(1+\alpha^2) - 2\alpha\cos\omega}$$

• When
$$\alpha > 0$$
, $|H(e^{j\omega})|^2_{\max} = \frac{K^2}{(1+\alpha^2)-2\alpha}$, at $\omega = 0$,

$$\left|H(e^{j\omega})\right|^2_{\min} = \frac{K^2}{(1+\alpha^2)+2\alpha}$$
, at $\omega = \pi$.

• When $\alpha < 0$, $\left| H(e^{j\omega}) \right|^2_{\max} = \frac{K^2}{(1+\alpha^2)+2\alpha}$, at $\omega = \pi$, $\left| H(e^{j\omega}) \right|^2_{\min} = \frac{K^2}{(1+\alpha^2)-2\alpha}$, at $\omega = 0$.



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Improved Lowpass IIR Filters

$$H(z) = \frac{1 - \alpha}{2} \frac{1 + z^{-1}}{1 - \alpha z^{-1}}, \qquad 0 < |\alpha| < 1$$

• A factor $1+z^{-1}$ added to the numerator to force the magnitude function to have a zero at $\omega=\pi$

$$|H_{LP}(e^{j\omega})|^2 = \frac{(1-\alpha)^2(1+\cos\omega)}{2(1+\alpha^2-2\alpha\cos\omega)}$$

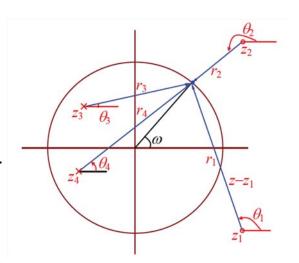
 $|H_{LP}(e^{j0})| = 1$, and $|H_{LP}(e^{j\pi})| = 0$

Zeros, Poles and Geometric Interpolation of Frequency Response

$$H(z) = \frac{(z - z_1)(z - z_2)}{(z - z_3)(z - z_4)}$$

$$H(e^{j\omega}) = \frac{(e^{j\omega} - z_1)(e^{j\omega} - z_2)}{(e^{j\omega} - z_3)(e^{j\omega} - z_4)}$$

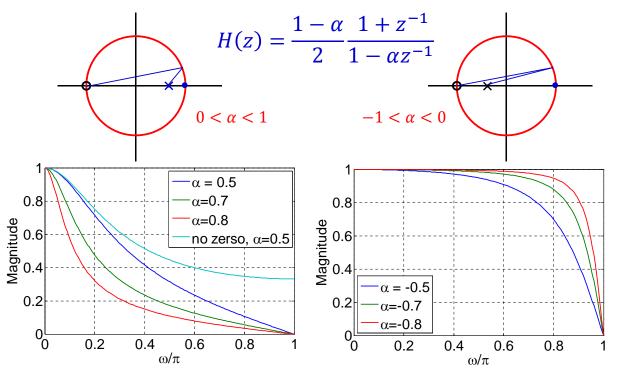
$$= \frac{r_1 r_2}{r_3 r_4} / \theta_1 + \theta_2 - \theta_3 - \theta_4$$



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Pole, Zero, and Response



Improved Highpass IIR Filters

$$H(z) = \frac{1+\alpha}{2} \frac{1-z^{-1}}{1-\alpha z^{-1}}, \qquad 0 < |\alpha| < 1$$

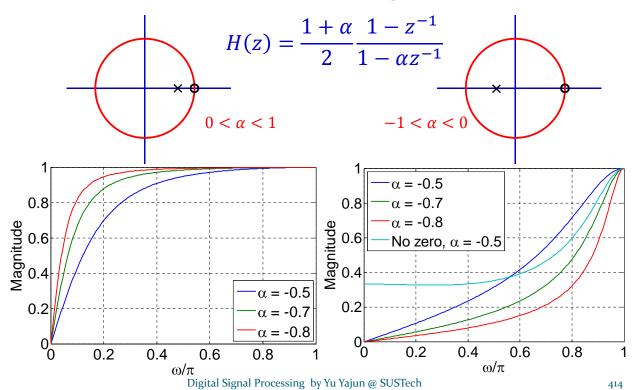
• A factor $1 - z^{-1}$ added to the numerator to force the magnitude function to have a zero at $\omega = 0$

$$\begin{aligned} \left| H_{LP}(e^{j\omega}) \right|^2 &= \frac{(1+\alpha)^2 (1-\cos\omega)}{2(1+\alpha^2-2\alpha\cos\omega)} \\ \left| H_{LP}(e^{j0}) \right| &= 0, \text{ and } \left| H_{LP}(e^{j\pi}) \right| = 1 \end{aligned}$$

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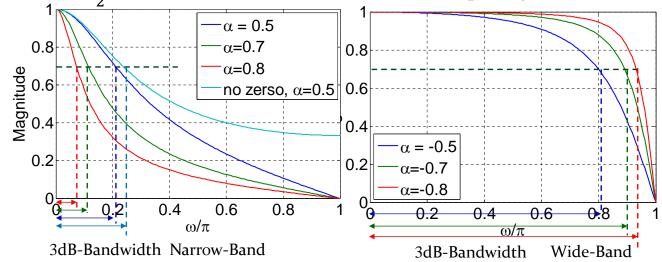
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Pole, Zero, and Response



3-dB Cutoff Frequency

• The frequency where the magnitude is reduced to $\frac{1}{\sqrt{2}}$ of the ideal passband gain, or the squared magnitude is reduced to $\frac{1}{2}$ of the ideal one, is the 3-dB cutoff frequency.



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Compute the 3-dB cutoff frequency

$$H(z) = \frac{1 - \alpha}{2} \frac{1 + z^{-1}}{1 - \alpha z^{-1}}, \qquad 0 < |\alpha| < 1$$

• Given α , compute ω_c . Squared magnitude function

$$|H_{LP}(e^{j\omega})|^2 = \frac{(1-\alpha)^2(1+\cos\omega)}{2(1+\alpha^2-2\alpha\cos\omega)} = \frac{1}{2}$$

which, when solved, yields

$$\cos \omega_c = \frac{2\alpha}{1 + \alpha^2} \Longrightarrow \omega_c = \cos^{-1} \frac{2\alpha}{1 + \alpha^2}$$

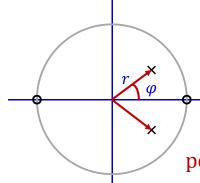
• Given ω_c , determine α . A solution resulting in a stable transfer function is

$$\alpha = \frac{1 - \sin \omega_c}{\cos \omega_c}$$

Bandpass IIR Digital Filter

A 2nd-order general form

$$H_{BP}(z) = \frac{K(1-z^{-2})}{1-\beta(1+\alpha)z^{-1}+\alpha z^{-2}}$$



$$r = \sqrt{\alpha}$$

$$\varphi = \cos^{-1}\left(\frac{\beta(1+\alpha)}{2}\sqrt{\alpha}\right)$$

For stability, we have $r < 1 \rightarrow |\alpha| < 1$

poles: $z = re^{\pm j\varphi}$

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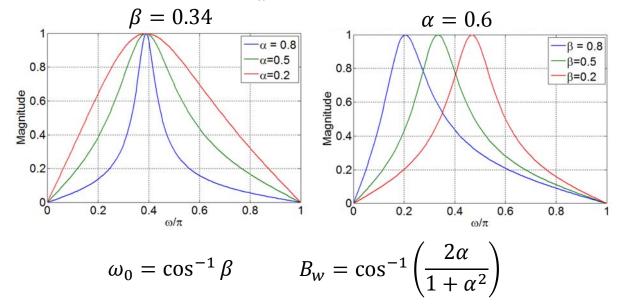
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The squared magnitude function:

$$\left| H_{BP}(e^{j\omega}) \right|^2 = \frac{4K^2 \sin^2 \omega}{(1+\alpha)^2 (\beta - \cos \omega)^2 + (1-\alpha)^2 \sin^2 \omega}$$

- $\left|H_{BP}(e^{j\omega})\right|^2 = 0$ at $\omega = 0$ and $\omega = \pi$
- $\left|H_{BP}(e^{j\omega})\right|^2 = \frac{2K}{1-\alpha}$, the maximum, at $\omega_0 = \cos^{-1}\beta$
 - ω_0 is called the **center frequency** of the bandpass filter
 - Choose $K = \frac{1-\alpha}{2}$ to make the maximum magnitude to be 1.
- The frequencies, ω_{c1} and ω_{c2} , where $\left|H_{BP}(e^{j\omega})\right|^2 = \frac{1}{2}$ are the 3-dB cutoff frequencies.

- 3-dB Bandwidth $B_w = \omega_{c2} \omega_{c1} = \cos^{-1}\left(\frac{2\alpha}{1+\alpha^2}\right)$
- Quality factor $Q = \frac{\omega_0}{B_W}$



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Example

- Design a second-order bandpass digital filter with center frequency at 0.4π and a 3dB bandwidth of 0.1π .
- A: $\beta = \cos \omega_0 = \cos(0.4\pi) = 0.309016994$ $\frac{2\alpha}{1+\alpha^2} = \cos B_w = \cos(0.1\pi) = 0.951056516$

 $\Rightarrow \alpha = 1.37638192$ (not stable) or $\alpha = 0.726542528$

So, the transfer function of the second-order bandpass filter is:

$$H_{BP}(e^{j\omega}) = \frac{0.136728736(1-z^{-2})}{1 - 0.53353098z^{-1} + 0.726542528z^{-2}}$$

Allpass Filter

Allpass Transfer Function: an IIR transfer function
 A(z) with unity magnitude response for all frequencies,
 i.e.,

$$\left|A(e^{j\omega})\right|^2 = 1$$
, for all ω

 An *M*-th order causal **real-coefficient** allpass transfer function is of form

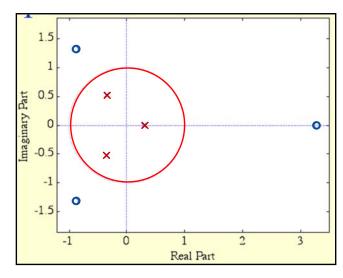
$$A_M(z) = \frac{d_M + d_{M-1}z^{-1} + \dots + d_1z^{-M+1} + z^{-M}}{1 + d_1z^{-1} + \dots + d_{M-1}z^{-M+1} + d_Mz^{-M}}$$
$$= \frac{z^{-M}D_M(z^{-1})}{D_M(z)}$$

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- Implying that the poles and zeros exhibits mirrorimage symmetry in z-plane.
 - Example:

$$A_3(z) = \frac{-0.2 + 0.18z^{-1} + 0.4z^{-2} + z^{-3}}{1 + 0.4z^{-1} + 0.18z^{-2} - 0.2z^{-3}}$$

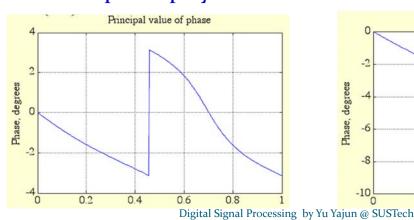


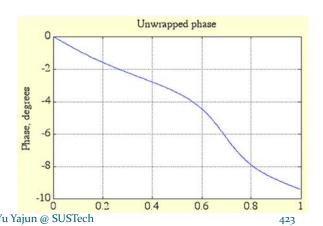
- The poles of a causal stable transfer function must lie inside the unit circle in z-plane
- Pairs of conjugated poles and zeros for real coefficient allpass filter, unless real pole or zeros.

• It can be shown that

$$\begin{aligned} \left| A_{M} \left(e^{j\omega} \right) \right|^{2} &= A_{M}(z) A_{M}(z^{-1}) \Big|_{z=e^{j\omega}} \\ &= \frac{z^{-M} D_{M}(z^{-1})}{D_{M}(z)} \cdot \frac{z^{M} D_{M}(z)}{D_{M}(z^{-1})} = 1 \end{aligned}$$

- Q: what's the use of allpass filters?
- A: Its phase plays the role





 Factorized form of real (or complex) coefficient allpass filter

$$A_M(z) = \pm \prod_{i=1}^{M} \left(\frac{-\lambda_i^* + z^{-1}}{1 - \lambda_i z^{-1}} \right)$$

Pole:
$$z_i = \lambda_i = r_i e^{j\theta_i}$$

Zero:
$$z = \frac{1}{\lambda_i^*} = \frac{1}{r_i} e^{j\theta_i}$$

 Hence, all zeros of a causal stable allpass transfer function must lie outside the unit circle in a mirrorimage symmetry with its poles suited inside the unit circle.

A First-Order Allpass Filter

A first-order complex coefficient allpass transfer function

$$A(z) = \frac{-\lambda^* + z^{-1}}{1 - \lambda z^{-1}}, |\lambda| < 1, \lambda = re^{j\varphi}$$

Its frequency response is given by

$$A(e^{j\omega}) = \frac{-\lambda^* + e^{-j\omega}}{1 - \lambda e^{-j\omega}} = e^{-j\omega} \frac{1 - re^{j(\omega - \varphi)}}{1 - re^{-j(\omega - \varphi)}}$$

So the phase function is

$$\theta(\omega) = -\omega - 2\tan^{-1}\frac{r\sin(\omega - \varphi)}{1 - r\cos(\omega - \varphi)}$$
$$\frac{d\theta(\omega)}{d\omega} = \frac{-(1 - r^2)}{(1 - r\cos(\omega - \varphi))^2 + r^2\sin^2(\omega - \varphi)} < 0$$

The phase of the first-order allpass filter decreases monotonically.

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Minimum Phase System

- **Definition:** A causal system with all zeros located inside the unit circles in z-plane is a minimum phase system, denoted as $H_{\min}(z)$.
- Property: Any real coefficient causal system can be represented as

$$H(z) = H_{\min}(z)A_m(z)$$

where, $H_{\min}(z)$ has the same magnitude response as H(z), and $A_m(z)$ is an allpass system.

Minimum Phase System Property

• **Proof:** Assume that H(z) has only one zero $z = \frac{1}{a^*}$, |a| < 1, located outside the unit circle.

Thus, H(z) can be represented as

$$H(z) = H_1(z)(z^{-1} - a^*).$$

- According to definition, $H_1(z)$ is a minimum system.
- And H(z) can be equivalent to

$$H(z) = H_1(z)(z^{-1} - a^*) \frac{1 - az^{-1}}{1 - az^{-1}}$$
$$= H_1(z)(1 - az^{-1}) \frac{z^{-1} - a^*}{1 - az^{-1}}$$

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• i.e.,
$$H(z) = H_{\min}(z)A_m(z)$$
, where $H_{\min}(z) = H_1(z)(1 - az^{-1})$

and

$$A_m(z) = \frac{z^{-1} - a^*}{1 - az^{-1}}$$

- Why it is called a minimum-phase system?
 - The minimum phase-lag property: the phase delay of the minimum phase system is always less than the phase delays of the other systems with the same magnitude response.

Example

 Q: Given the transfer function of a real coefficient causal stable LTI system

$$H(z) = \frac{b+z^{-1}}{1+az^{-1}}, |a| < 1 \text{ and } |b| < 1$$

Find the minimum phase system having the same magnitude response as that of H(z)

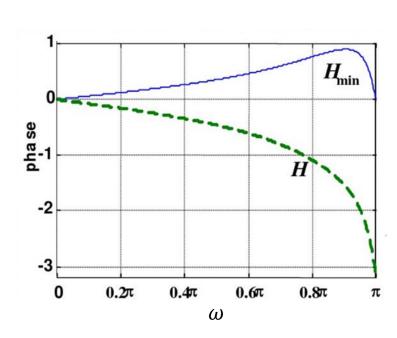
• A: Since the zero of H(z) is $-\frac{1}{b}$ and |b| < 1, it is not a minimum phase system.

$$H(z) = \frac{b + z^{-1}}{1 + az^{-1}} \frac{1 + bz^{-1}}{1 + bz^{-1}} = \frac{1 + bz^{-1}}{1 + az^{-1}} \frac{b + z^{-1}}{1 + bz^{-1}}$$

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• So the minimum phase system is



$$H_{\min}(z) = \frac{1 + bz^{-1}}{1 + az^{-1}}$$

The phase response of $H(e^{j\omega})$ and $H_{\min}(e^{j\omega})$ when a=0.9, and b=0.4

Maximum Phase System

- A causal system with all zeros located outside the unit circles in z-plane is a maximum phase system, denoted as $H_{\text{max}}(z)$.
- Example: The transfer function of a real coefficient causal stable LTI system is give by

$$H(z) = \frac{b+z^{-1}}{1+az^{-1}}, |a| < 1 \text{ and } |b| < 1$$

Obviously, this is a maximum phase system.

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Inverse System

- Inverse system has $h_1[n] \oplus h_2[n] = \delta[n]$
- Then, in z-Domain

$$H_1(z)H_2(z) = 1$$

• If we have $H_1(z)$, then

$$H_2(z) = \frac{1}{H_1(z)}$$