Lecture 8 LTI Discrete-Time Systems in the Transform Domain

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Types of Transfer Functions

LTI
$$\begin{array}{c}
x[n] \\
X(z)
\end{array}$$

$$h[n] \\
Y(z)$$

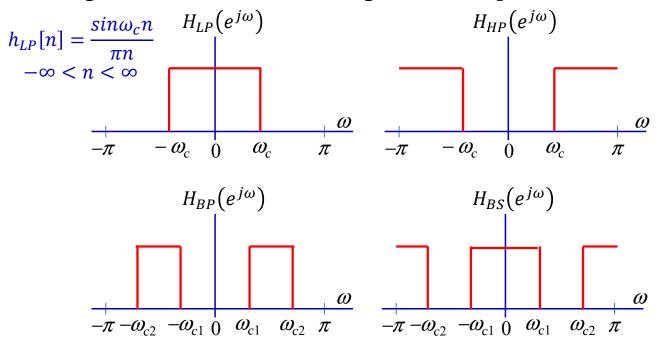
$$Y(z)$$

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

- 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 → 知知 7 和 2

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 $\omega = 2\pi$.

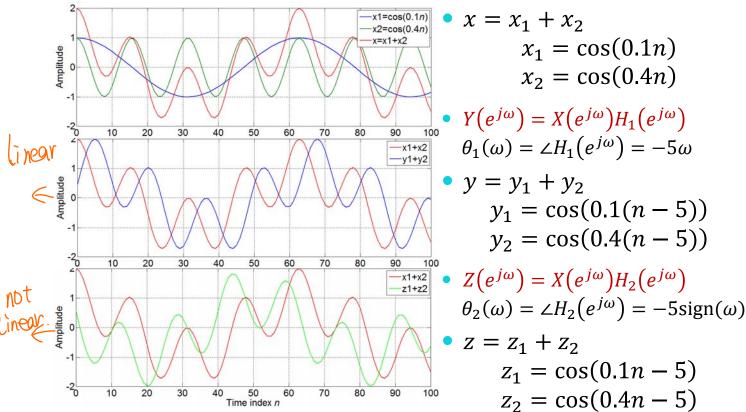
• Note: $\theta(\omega) = -\omega D$.

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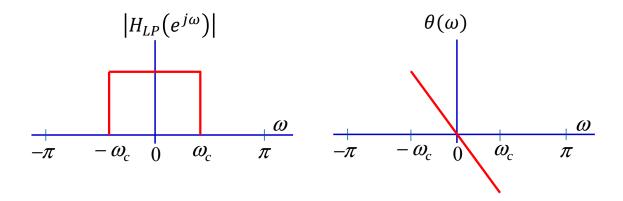
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Phase Distortion



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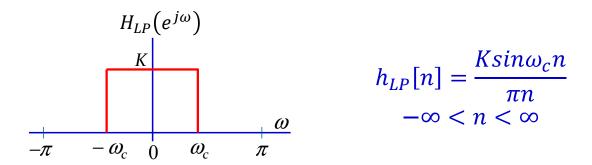
- It is desired to pass input signal components in a certain frequency range undistorted in both magnitude and phase
- The transfer function should exhibit a unity magnitude response and a linear phase response in the band of interest.



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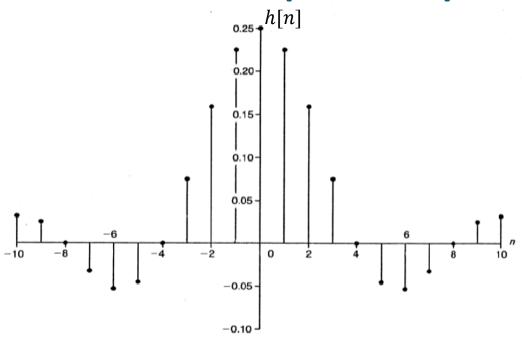
Design of Idea Lowpass FIR filter



Let
$$K = 1$$
, and $\omega_c = \pi/4$, $n = 0, \pm 1, ..., \pm 10$,

$$h[0] = 0.25,$$
 $h[\pm 4] = 0,$ $h[\pm 8] = 0,$ $h[\pm 1] = 0.225,$ $h[\pm 5] = -0.043,$ $h[\pm 9] = 0.025,$ $h[\pm 2] = 0.159,$ $h[\pm 6] = -0.053,$ $h[\pm 10] = 0.032$ $h[\pm 3] = 0.075$ $h[\pm 7] = -0.032$

Non-causal FIR Impulse Response



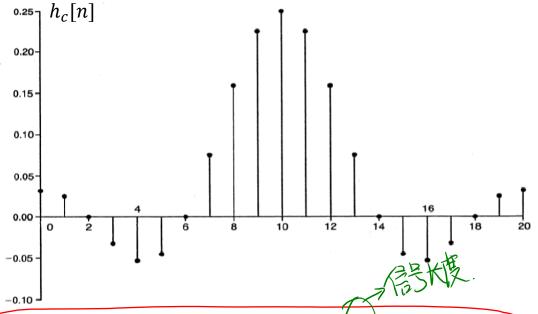
We can make it causal if we shift $h_{LP}[n]$ by 10 units to the right:

$$h_c[n] = \frac{K}{\pi(n-10)} sin\omega_c(n-10), \qquad n = 0, 1, ..., 20$$

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Causal FIR (N=21) Impulse Response

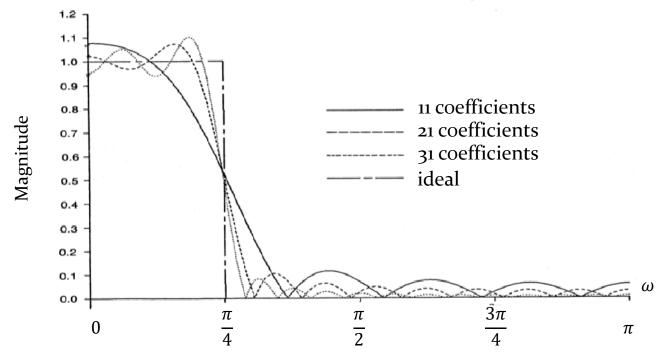


Notice the symmetry: $h_c[n] = h_c[N-1-n]$ which satisfies the linear phase condition.

Frequency response: $H_c(e^{j\omega}) = \sum_{n=0}^{20} h_c[n]e^{-j\omega n}$

Magnitude of filter frequency response

for filter length 11, 21, and 31

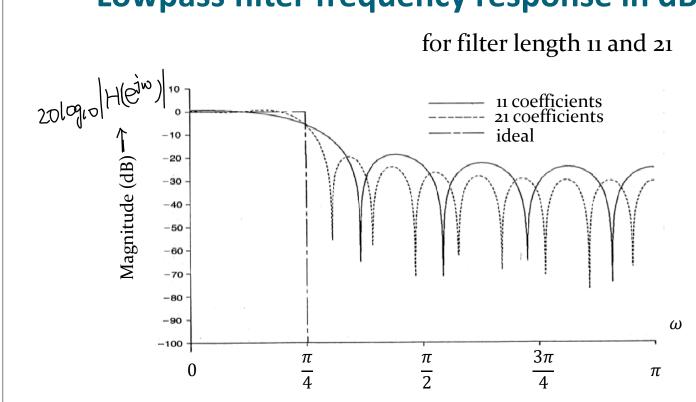


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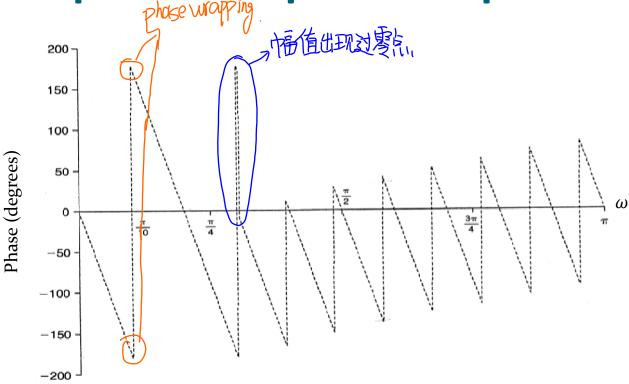
Lowpass filter frequency response in dB

for filter length 11 and 21



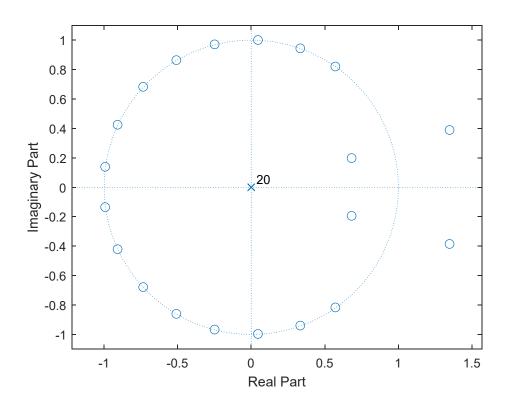
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Lowpass filter phase response



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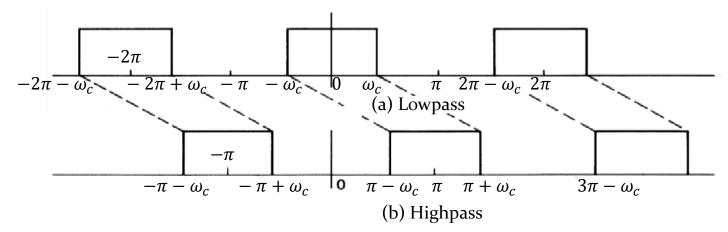
Pole-zero plot for H(z)



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Magnitude Characteristics of Idea Lowpass and Highpass Filters



The above figure implies

$$H_{HP}(e^{j\omega}) = H_{LP}(e^{j(\omega-\pi)})$$

$$h_{HP}[n] = h_{LP}[n] \cdot e^{j\pi n} = h_{LP}[n] \cdot (-1)^n / (-1)^n$$

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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

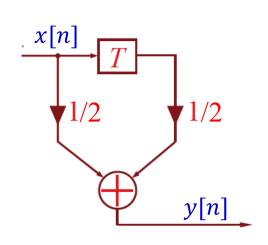
•
$$H(z) = \frac{1+z^{-1}}{2} \leftrightarrow \{h[n]\} = \left\{\frac{1}{2}, \frac{1}{2}\right\}$$

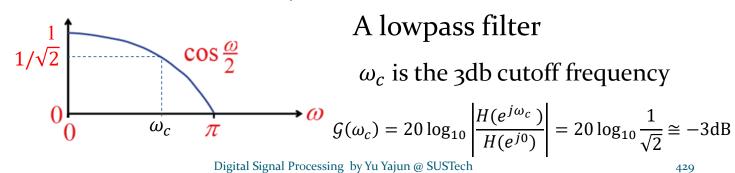
$$\uparrow$$

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

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

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





A lowpass filter

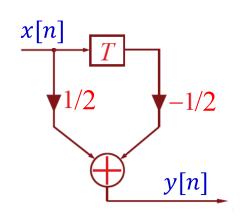
$$\mathcal{G}(\omega_c) = 20 \log_{10} \left| \frac{H(e^{j\omega_c})}{H(e^{j0})} \right| = 20 \log_{10} \frac{1}{\sqrt{2}} \cong -3 \text{dB}$$

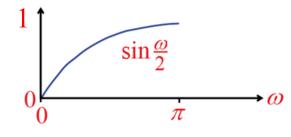
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Simple Examples

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

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





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 $-d\{\theta(\omega)\} = \alpha$.

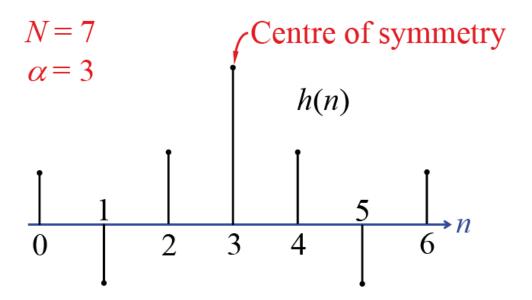




- Its impulse response is either symmetrical or antisymmetrical.
- If its impulse response is symmetrical, its phase response is $-\alpha\omega$.
- If its impulse response is anti-symmetrical, its phase response is $\pm 0.5\pi \alpha\omega$.

Example: Symmetrical impulse response,

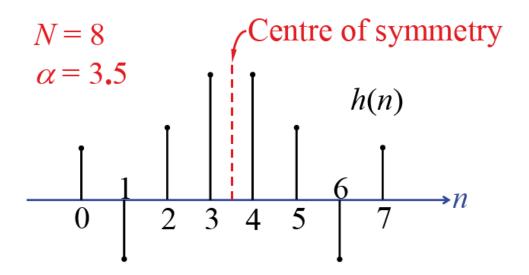
N odd, where *N* is the length of the impulse response



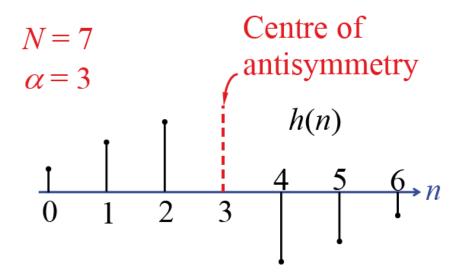
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Example: Symmetrical impulse response, *N* even.



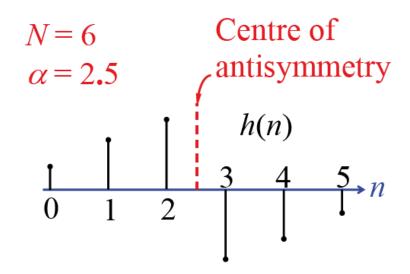
Example: Anti-symmetrical impulse response, *N* odd.



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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. 换礼
- Type I: Symmetrical impulse response, *N* odd.

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

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

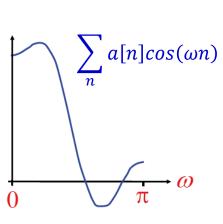
$$= e^{-j\omega \frac{N-1}{2}} \left[\sum_{n=0}^{(N-3)/2} 2h[n] \cos\left(\omega \left(\frac{N-1}{2} - n\right)\right) + h\left[\frac{N-1}{2}\right] \right]$$

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

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

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

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

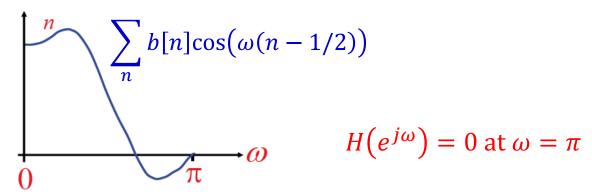


Type II:

• Symmetrical impulse response, N even

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

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



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Type III:

Anti-symmetrical impulse response, N odd

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

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

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

$$c[0] = 0$$

$$H(e^{j\omega}) = 0 \text{ at } \omega = 0 \text{ and } \pi$$

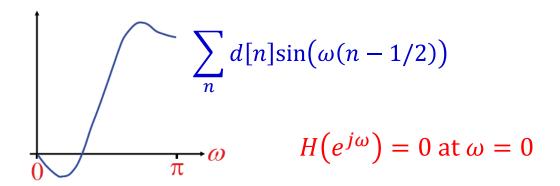
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Type IV:

Anti-symmetrical impulse response, N even

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

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



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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

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

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

$$\Rightarrow z = -1 \text{ or } z = 1 \Rightarrow (z + 1) \text{ or } (z - 1)$$

Complex zeros

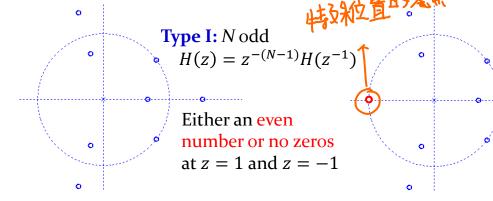
$$z = re^{\pm j\varphi} \text{ and } z = \frac{1}{r}e^{\pm j\varphi}, r \neq \pm 1$$

$$\Rightarrow 1 + az^{-1} + cz^{-2} + az^{-3} + z^{-4}$$

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

o j Imz
9 Re z
Unit Circle

Zero-Locations of EIR Filters

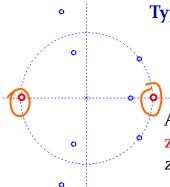


H(-1) = -H(-1)Either an even number or no zeros at z = 1 and an odd number of zeros

 $H(z) = z^{-(N-1)}H(z^{-1})$

at z = -1

Type II: *N* even



Type III: N odd $H(z) = -z^{-(N-1)}H(z^{-1})$ H(-1) = -H(-1) H(1) = -H(1)

An odd number of zeros at z = 1 and z = -1

Type IV: *N* even $H(z) = -z^{-(N-1)}H(z^{-1})$ H(1) = -H(1)

An odd number of zeros at z = 1 and either an even number or no zeros at z = -1

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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 <u>not causal</u>, 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

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

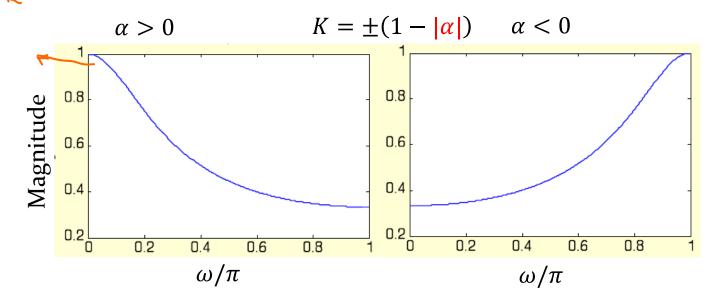
• Its squared-magnitude function is given by

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

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

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

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



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Improved Lowpass IIR Filters

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

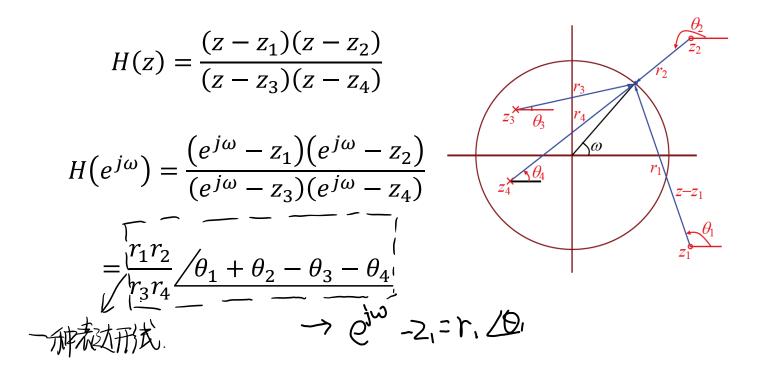
• A factor $1 + z^{-1}$ added to the numerator to force the magnitude function to have a zero at $\omega = \pi$

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

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

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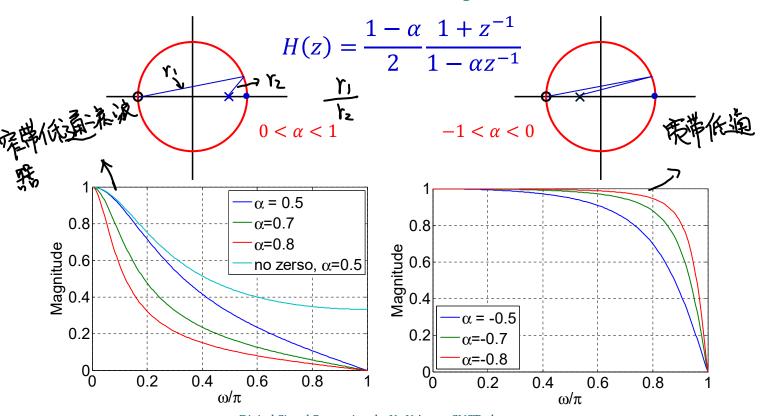
Zeros, Poles and Geometric Interpolation of Frequency Response



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Pole, Zero, and Response



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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 $\omega = 0$

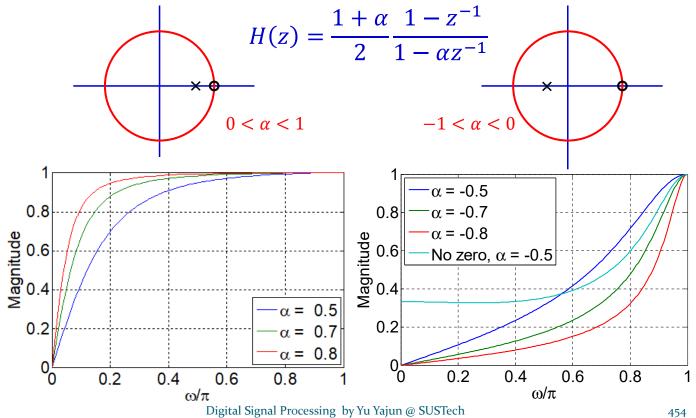
$$|H_{LP}(e^{j\omega})|^2 = \frac{(1+\alpha)^2(1-\cos\omega)}{2(1+\alpha^2-2\alpha\cos\omega)}$$

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

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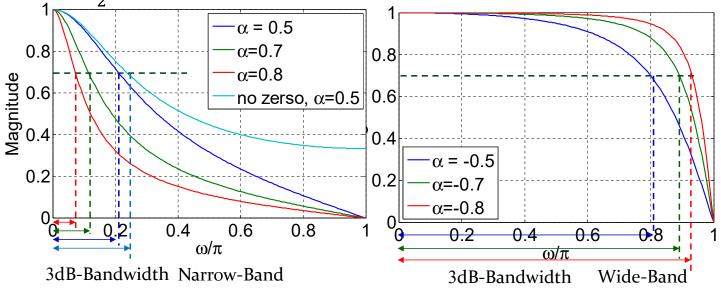
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Pole, Zero, and Response



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

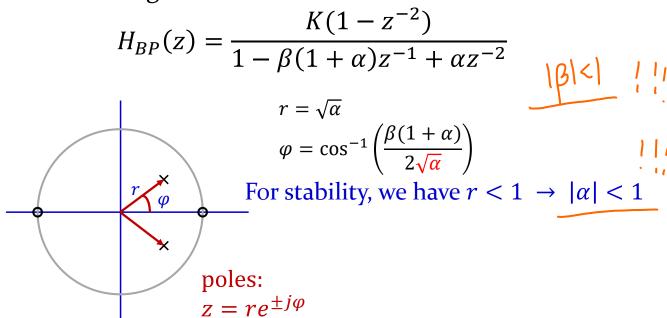
$$\cos \omega_c = \frac{2\alpha}{1 + \alpha^2} \Longrightarrow \omega_c = \cos^{-1} \frac{2\alpha}{1 + \alpha^2}$$

• 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



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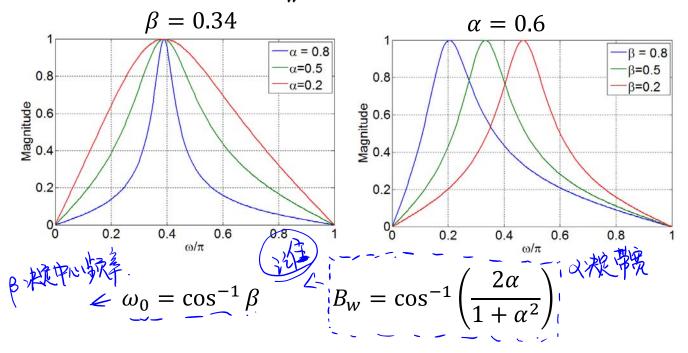
• The squared magnitude function:

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

- $\left|H_{BP}(e^{j\omega})\right|^2 = 0$ at $\omega = 0$ and $\omega = \pi$
- $|H_{BP}(e^{j\omega})|^2 = \frac{2K}{1-\alpha}$, 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

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.,

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

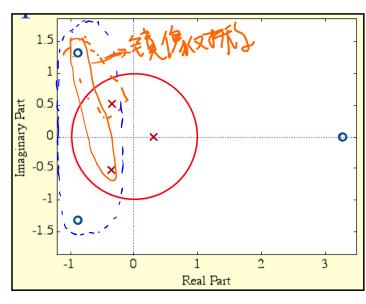
An *M*-th order causal **real-coefficient** allpass transfer function is of form

$$A_{M}(z) = \frac{d_{M} \pm d_{M-1}z^{-1} + \dots + d_{1}z^{-M+1} + z^{-M}}{1 + d_{1}z^{-1} + \dots + d_{M-1}z^{-M+1} + d_{M}z^{-M}}$$
$$= \frac{z^{-M}D_{M}(z^{-1})}{D_{M}(z)}$$

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

$$A_3(z) = \frac{-0.2 + 0.18z^{-1} + 0.4z^{-2} + z^{-3}}{1 + 04z^{-1} + 0.18z^{-2} - 0.2z^{-3}}$$

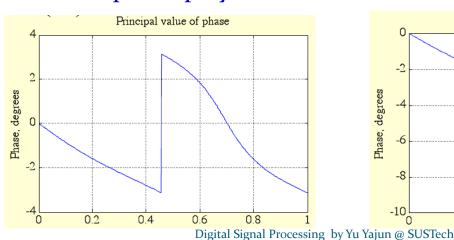


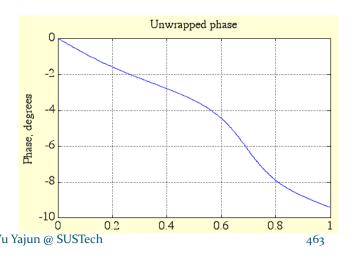
- The poles of a causal stable transfer function must lie inside the unit circle in z-plane
- Pairs of conjugated poles and zeros for real coefficient allpass filter, unless real pole or zeros.

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





茅亚种混波器有点不太特

Factorized form of <u>real</u> (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: $z_{pi} = \lambda_i = r_i e^{j\theta_i}$

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

 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

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

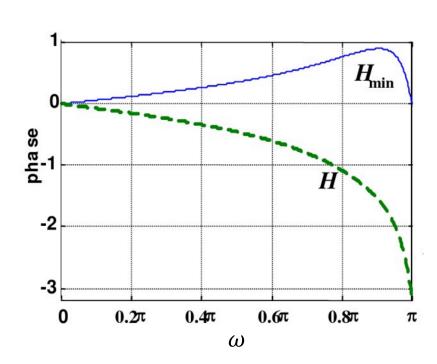
Find the minimum phase system having the same magnitude response as that of H(z)

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

$$H(z) = \frac{b + z^{-1}}{1 + az^{-1}} \frac{1 + bz^{-1}}{1 + bz^{-1}} = \frac{1 + bz^{-1}}{1 + az^{-1}} \underbrace{\begin{vmatrix} b + z^{-1} \\ 1 + bz^{-1} \end{vmatrix}}_{1 + az^{-1}} \underbrace{\begin{vmatrix} b + z^{-1} \\ 1 + bz^{-1} \end{vmatrix}}_{1 + az^{-1}}$$

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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 <u>zeros</u> 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 $h_1[n] \otimes h_2[n] = \delta[n]$
- Then, in z-Domain

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

• If we have $H_1(z)$, then

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