Chapter 7 LTI Discrete-Time Systems in the Transform Domain

Zhiliang Liu
Zhiliang_Liu@uestc.edu.cn
4/15/2019

Types of Transfer Functions

The analysis methods of discrete LTI systems in the transform domain (ZT and DTFT).

$$y(n) = x[n] \circledast h[n] \Leftrightarrow Y(z) = X(z)H(z)$$
$$[Y(e^{j\omega}) = X(e^{j\omega})H(e^{j\omega})]$$

H(z): transfer function

 $H(e^{j\omega})$: frequency response

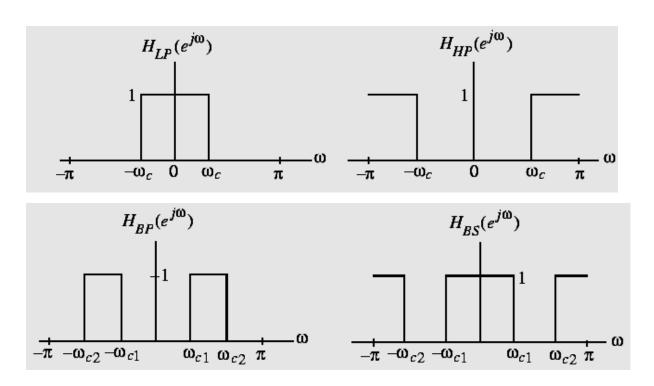
Types of Transfer Functions

- The time-domain classification of an LTI digital transfer function sequence is based on the *length of its impulse response*:
 - Finite impulse response (FIR) transfer function
 - Infinite impulse response (IIR) transfer function

Types of Transfer Functions

- In the case of digital transfer functions with frequency-selective frequency responses, there are two types of classifications
 - (1) Classification based on the shape of the magnitude function $|H(e^{j\omega})|$
 - (2) Classification based on the form of the phase function $\theta(\omega)$

• Frequency responses of the four popular types of ideal digital filters with real impulse response coefficients are shown below:



• Lowpass filter: <u>Passband</u>: $0 \le \omega \le \omega_c$

Stopband: $\omega_c \leq \omega \leq \pi$

• Highpass filter: <u>Passband</u>: $\omega_c \le \omega \le \pi$

<u>Stopband</u>: $0 \le \omega \le \omega_c$

Bandpass filter:

Passband: $\omega_{c1} \leq \omega \leq \omega_{c2}$

Stopband: $0 \le \omega < \omega_{c1}$ and $\omega_{c2} < \omega < \pi$

Bandstop filter:

<u>Stopband</u>: $\omega_{c1} < \omega < \omega_{c2}$

Passband: $0 \le \omega \le \omega_{c1}$ and $\omega_{c2} \le \omega \le \pi$

- The frequencies ω_c , ω_{c1} , and ω_{c2} are called the <u>cut-off frequencies</u>
- An ideal filter has a magnitude response equal to 1 in the passband and 0 in the stopband, and <u>has a 0</u> phase everywhere

• Earlier in the course we derived the inverse DTFT of the frequency response $H_{LP}(e^{j\omega})$ of the ideal lowpass filter:

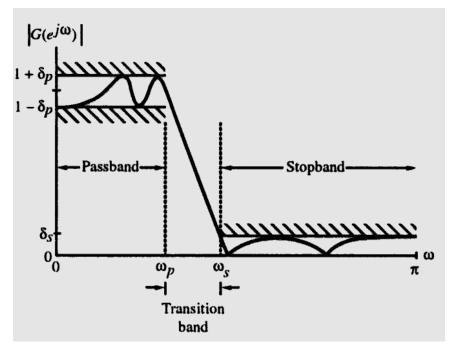
$$h_{LP}[n] = \sin \omega_c n / n \pi, -\infty < n < \infty$$

 We have also shown that the above impulse response is not absolutely summable, and hence, the corresponding transfer function is not BIBO stable

- Also, $h_{LP}[n]$ is not causal and is of infinite length
- The remaining three ideal filters are also characterized by infinite, noncausal impulse responses and are not absolutely summable
- Thus, the ideal filters with the ideal "brick wall" frequency responses cannot be realized with finite dimensional LTI filter

- To develop <u>stable and realizable transfer</u> <u>functions</u>, the ideal frequency response specifications are relaxed by including a <u>transition band</u> between the passband and the stopband
- This permits the magnitude response to decay slowly from its maximum value in the passband to the 0-value in the stopband

 Moreover, the magnitude response is allowed to vary by a small amount both in the passband and the stopband



Typical magnitude response specifications of a lowpass filter 11

Magnitude Characteristics

- One common classification is based on an ideal magnitude response
- A digital filter designed to pass signal components of certain frequencies without distortion should have a frequency response equal to 1 at these frequencies, and should have a frequency response equal to 0 at all other frequencies

- The range of frequencies where the frequency response takes the value of 1 is called the passband
- The range of frequencies where the frequency response takes the value of 0 is called the stopband

A causal stable real-coefficient transfer function H(z) is defined as a bounded real (BR) transfer function if

 $|H(e^{j\omega})| \le 1$ for all values of ω

Let x[n] and y[n] denote, respectively, the input and output of a digital filter characterized by a BR transfer function H(z) with $X(e^{j\omega})$ and $Y(e^{j\omega})$ denoting their DTFTs

•Then the condition $|H(e^{j\omega})| \le 1$ implies that

$$\left|Y\left(e^{j\omega}\right)\right|^{2} \leq \left|X\left(e^{j\omega}\right)\right|^{2}$$

Integrating the above from $-\pi$ to π , and applying Parseval's relation we get

$$\sum_{n=-\infty}^{\infty} |y[n]|^2 \le \sum_{n=-\infty}^{\infty} |x[n]|^2$$

• Thus, for all finite-energy inputs, the output energy is less than or equal to the input energy implying that a digital filter characterized by a BR transfer function can be viewed as a *passive structure*

If $|H(e^{j\omega})| = 1$, then the output energy is equal to the input energy, and such a digital filter is therefore a <u>lossless system</u>

- •A causal stable real-coefficient transfer function H(z) with $|H(e^{j\omega})|=1$ is thus called a <u>lossless bounded real (LBR)</u> <u>transfer function</u>
- The BR and LBR transfer functions are the keys to the realization of digital filters with low coefficient sensitivity

• <u>Example</u>: Consider the causal stable IIR transfer function

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

where K is a real constant

Its square-magnitude function is given by

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

- Thus, for $\alpha>0$, the maximum value of $|H(e^{j\omega})|^2$ is equal to $K^2/(1-\alpha)^2$ at $\omega=0$ and the minimum value is equal to $K^2/(1+\alpha)^2$ at $\omega=\pi$
- On the other hand, for $\alpha < 0$, the maximum value of $2\alpha\cos\omega$ is equal to -2α at $\omega = \pi$ and the minimum value is equal to 2α at $\omega = 0$

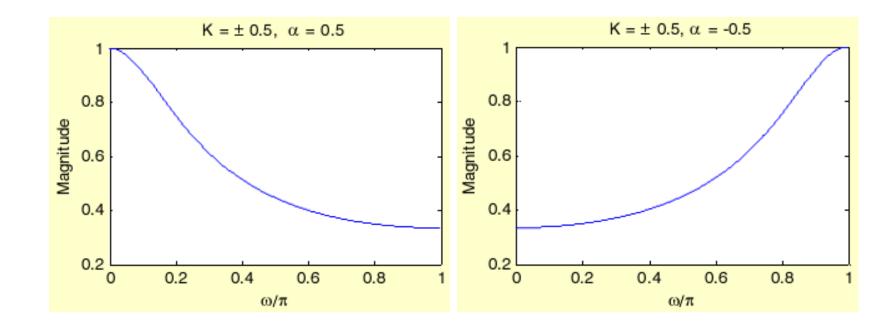
• Here, the maximum value of $|H(e^{j\omega})|^2$ is equal to $K^2/(1+\alpha)^2$ at $\omega=\pi$ and the minimum value is equal to $K^2/(1-\alpha)^2$ at $\omega=0$

Hence,

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

is a BR function for $K=\pm (1-|\alpha|)$,

Plots of the magnitude function for $\alpha=\pm 0.5$ with values of K chosen to make H(z) a BR function are shown on the next slide



Lowpass filter

Highpass filter

Definition

• 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

is called an allpass transfer function

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

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

• If we denote the denominator polynomials of $A_M(z)$ as $D_M(z)$:

$$D_M(z) = 1 + d_1 z^{-1} + \dots + d_{M-1} z^{-M+1} + d_M z^{-M}$$

then it follows that $A_M(z)$ can be written

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

as:

• Note from the above that if $z=re^{j\varphi}$ is a pole of a real coefficient allpass transfer function, then it has a zero at $z=(1/r)e^{-j\varphi}$

- The numerator of a real-coefficient allpass transfer function is said to be the mirror-image polynomial of the denominator, and vice versa
 - We shall use the notation $D_M(z)$ to denote the mirror-image polynomial of a degree-M polynomial $D_M(z)$, i.e.,

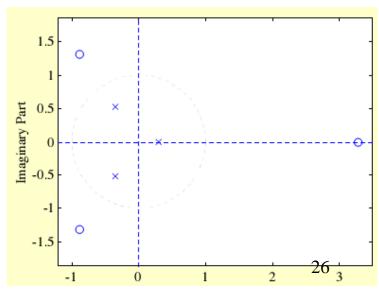
$$\tilde{D}_{M}(z) = z^{-M} D_{M}(z^{-1})$$

The expression

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

implies that the poles and zeros of a realcoefficient allpass function exhibit mirror-image symmetry in the z-plane

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



• To show that $|A_M(e^{j\omega})| = 1$ we observe that

$$A_M(z^{-1}) = \pm \frac{z^M D_M(z)}{D_M(z^{-1})}$$

Therefore

$$A_{M}(z)A_{M}(z^{-1}) = \frac{z^{-M}D_{M}(z^{-1})}{D_{M}(z)} \frac{z^{M}D_{M}(z)}{D_{M}(z^{-1})}$$

Hence

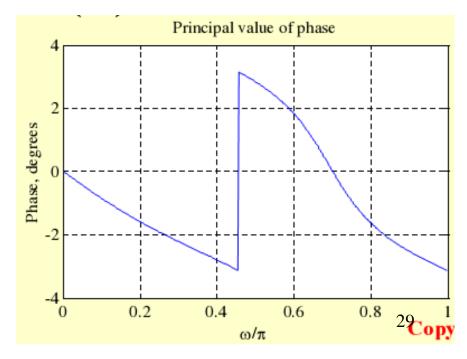
$$|A_{M}(e^{j\omega})|^{2} = A_{M}(z)A_{M}(z^{-1})\Big|_{z=e^{j\omega}} = 1$$

- Now, the poles of a causal stable transfer function must lie inside the unit circle in the z-plane
- Hence, all zeros of a causal stable allpass transfer function must lie outside the unit circle in a mirror-image symmetry with its poles situated inside the unit circle

 Figure below shows the principal value of the phase of the 3rd-order allpass

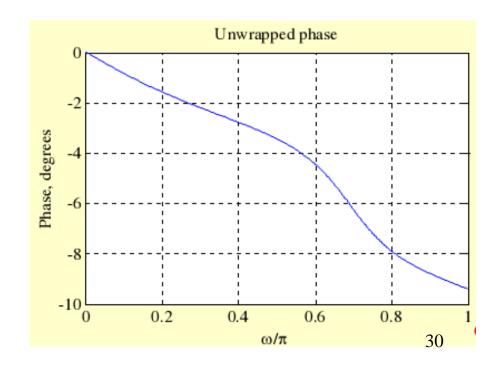
function
$$A_3(z) = \frac{-0.2 + 0.18 z^{-1} + 0.4 z^{-2} + z^{-3}}{1 + 0.4 z^{-1} + 0.18 z^{-2} - 0.2 z^{-3}}$$

•Note the discontinuity by the amount of 2π in the phase $\theta(\omega)$



• If we unwrap the phase by removing the discontinuity, we arrive at the unwrapped phase function $\theta_c(\omega)$ indicated below

Note: The unwrapped phase function is a continuous function of ω



• The unwrapped phase function of any arbitrary causal stable allpass function is a continuous function of ω

Properties

• (1) A causal stable real-coefficient allpass transfer function is a lossless bounded real (LBR) function or, equivalently, a causal stable allpass filter is a <u>lossless</u>

(2) The magnitude function of a stable allpass function A(z) satisfies:

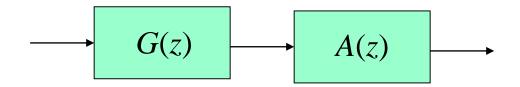
$$\begin{vmatrix} \langle 1, & \text{for } | z | > 1 \\ | A(z) | \begin{cases} <1, & \text{for } |z| > 1 \\ | >1, & \text{for } |z| < 1 \end{cases}$$
 (Problem 7.2)

(3) Let $\tau(\omega)$ denote the group delay function of an allpass filter A(z), i.e.,

$$\tau_{g}(\omega) = -\frac{d}{d\omega}[\theta_{c}(\omega)]$$
 (Problem 7.3)

A Simple Application

- A simple but often used application of an allpass filter is as a delay equalizer
 - Let G(z) be the transfer function of a digital filter designed to meet a prescribed magnitude response
 - The nonlinear phase response of G(z) can be corrected by cascading it with an allpass filter A(z) so that the overall cascade has a constant group delay in the band of interest



• Since $|A(e^{j\omega})|=1$, we have

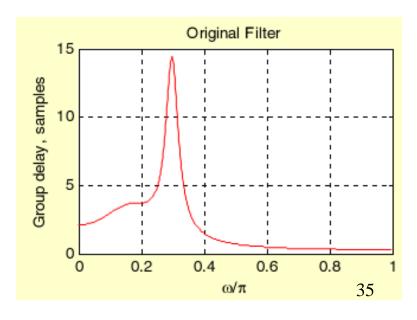
$$|G(e^{j\omega})A(e^{j\omega})| = |G(e^{j\omega})|$$

Overall group delay is the given by the sum of the group delays of G(z) and A(z)

• Example: Figure below shows the group delay of a 4th order elliptic filter with the following specifications: ω_p =0.3 π ,

 $\delta_p = 1 dB$, $\delta_s = 35 dB$

The nonlinear phase response



• Figure below shows the group delay of the original elliptic filter cascaded with an 8th order allpass section designed to equalize the group delay in the passband



Classification Based on Phase Characteristics

- A second classification of a transfer function is with respect to its phase characteristics
- In many applications, it is necessary that the digital filter designed does not distort the phase of the input signal components with frequencies in the passband

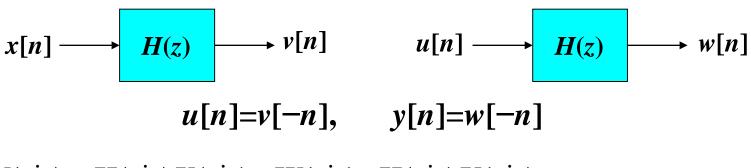
- One way to avoid any phase distortion is to make the frequency response of the filter real and nonnegative, i.e., to design the filter with a zero phase characteristic
- However, it is impossible to design a causal digital filter with a zero phase

- For non-real-time processing of real-valued input signals of finite length, zero-phase filtering can be very simply implemented by relaxing the causality requirement
- One zero-phase filtering scheme is sketched below

$$x[n] \longrightarrow H(z) \longrightarrow v[n] \qquad u[n] \longrightarrow H(z) \longrightarrow w[n]$$

$$u[n]=v[-n], \qquad y[n]=w[-n]$$

- It is easy to verify the above scheme in the frequency domain
- Let $X(e^{j\omega})$, $V(e^{j\omega})$, $U(e^{j\omega})$, $W(e^{j\omega})$, and $Y(e^{j\omega})$ denote the DTFTs of x[n], v[n], u[n], w[n], and y[n], respectively
- From the figure shown earlier and making use of the symmetry relations we arrive at the relations between various DTFTs as given on the next slide



$$V(e^{j\omega}) = H(e^{j\omega})X(e^{j\omega}), W(e^{j\omega}) = H(e^{j\omega})U(e^{j\omega})$$
 $U(e^{j\omega}) = V^*(e^{j\omega}), Y(e^{j\omega}) = W^*(e^{j\omega})$

Combining the above equations we get

$$Y(e^{j\omega}) = W^*(e^{j\omega}) = H^*(e^{j\omega})U^*(e^{j\omega})$$
$$= H^*(e^{j\omega})V(e^{j\omega}) = H^*(e^{j\omega})H(e^{j\omega})X(e^{j\omega})$$
$$= |H(e^{j\omega})|^2 X(e^{j\omega})$$

• This is a zero-phase filter with a frequency response $|H(e^{j\omega})|^2$

- The function fftfilt implements the above zerophase filtering scheme
- In the case of a causal transfer function with a nonzero phase response, the phase distortion can be avoided by ensuring that the transfer function has a unity magnitude and a linearphase characteristic in the frequency band of interest

• The most general type of a filter with a linear phase has a frequency response given by

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

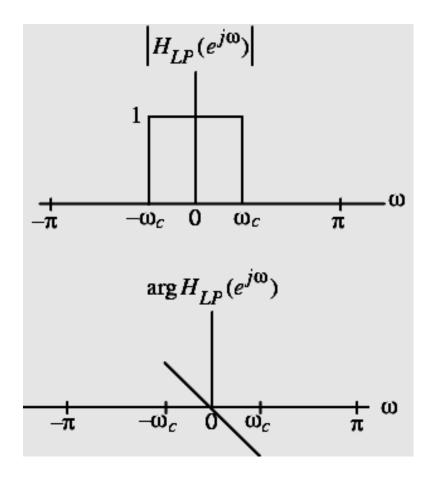
which has a linear phase from $\omega = 0$ to $\omega = 2\pi$

• Note also $|H(e^{j\omega})|=1$ $\tau_g(\omega)=D$

- The output y[n] of this filter to an input $x[n]=Ae^{j\omega n}$ (eigenfunction) is then given by $y[n]=e^{-j\omega D}Ae^{j\omega n}=Ae^{j\omega(n-D)}$
- If $x_a(t)$ and $y_a(t)$ represent the continuous-time signals whose sampled versions, sampled at t = nT, are x[n] and y[n] given above, then the delay between $x_a(t)$ and $y_a(t)$ is precisely the group delay of amount D

- If D is an integer, then y[n] is identical to x[n], but delayed by D samples (y[n] = x[n-D])
- If D is not an integer, y[n], being delayed by a fractional part, is not identical to x[n]
- In the latter case, the waveform of the underlying continuous-time output is identical to the waveform of the underlying continuous-time input and delayed *D* units of time

• Figure right shows the frequency response if a lowpass filter with a linear-phase characteristic in the passband



- Since the signal components in the stopband are blocked, the phase response in the stopband can be of any shape
- <u>Example</u> Determine the impulse response of an ideal lowpass filter with a linear phase response:

$$H_{LP}(e^{j\omega}) = \begin{cases} e^{-j\omega n_o}, & 0 < |\omega| < \omega_c \\ 0, & \omega_c \le |\omega| \le \pi \end{cases}$$

 Applying the frequency-shifting property of the DTFT to the impulse response of an ideal zero-phase lowpass filter we arrive at

$$h_{LP}[n] = \frac{\sin \omega_c (n - n_o)}{\pi (n - n_o)}, -\infty < n < \infty$$

• As before, the above filter is noncausal and of doubly infinite length, and hence, unrealizable

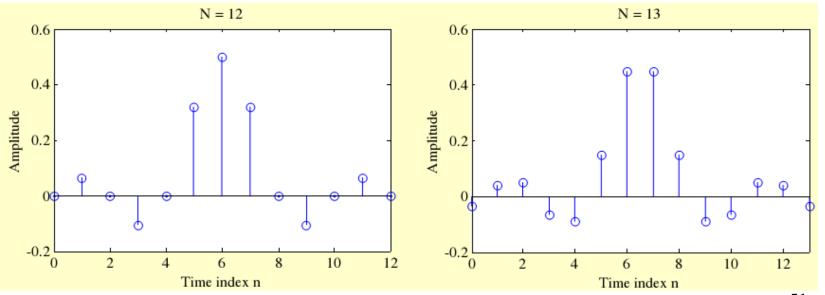
- By truncating the impulse response to a finite number of terms, a realizable FIR approximation to the ideal lowpass filter can be developed
- The truncated approximation may or may not exhibit linear phase, depending on the value of n_0 chosen

• If we choose $n_0 = N/2$ with N a positive integer, the truncated and shifted approximation

$$h_{LP}[n] = \frac{\sin \omega_c (n - N/2)}{\pi (n - N/2)}, \quad 0 \le n \le N$$

will be a length N+1 causal linearphase FIR filter

• Figure below shows the filter coefficients obtained using the function sinc for two different values of N



• Because of the symmetry of the impulse response coefficients as indicated in the two figures, the frequency response of the truncated approximation can be expressed as:

$$\stackrel{\wedge}{H}_{LP}(e^{j\omega}) = \sum_{n=0}^{N} \stackrel{\wedge}{h}_{LP}[n]e^{-j\omega n} = e^{-j\omega N/2} \stackrel{\sim}{H}_{LP}(\omega)$$

where $\tilde{H}_{LP}(\omega)$, called the zero-phase response or amplitude response, is a real function of ω

- It is nearly impossible to design a linear-phase IIR transfer function
- It is always possible to design an FIR transfer function with an exact linear-phase response
- We now develop the forms of the linear-phase FIR transfer function H(z) with real impulse response h[n]
- Consider a causal FIR transfer function H(z) of length N+1, i.e., of order N:

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

• If H(z) is to have a linear-phase, its frequency response must be of the form

$$H(e^{j\omega}) = e^{j(c\omega + \beta)}H(\omega)$$

where c and β are constants, and $_{\mathrm{H}(\omega)}$, called the amplitude response, also called the zero-phase response, is a real function of ω

- For a real impulse response, the magnitude response $|H(e^{j\omega})|$ is an even function, i.e., $|H(e^{j\omega})| = |H(e^{-j\omega})|$
 - Since $|H(e^{j\omega})| = |H(\omega)|$, the amplitude response is then either an even function or an odd function of ω , i.e.

$$H(-\omega) = \pm H(\omega)$$

• The frequency response satisfies the relation $H(e^{j\omega})=H^*(e^{-j\omega})$, or equivalently, the relation

$$e^{j(c\omega+\beta)}H(\omega)=e^{-j(-c\omega+\beta)}H(-\omega)$$

If $H(\omega)$ is an even function, then the above relation leads to $e^{j\beta}=e^{-j\beta}$ implying that either $\beta=0$ or $\beta=\pi$

• From

$$H(e^{j\omega}) = e^{j(c\omega + \beta)}H(\omega)$$

We have

$$H(\omega) = e^{-j(c\omega + \beta)}H(e^{j\omega})$$

Substituting the value of β in the above we get

$$H(\omega) = \pm e^{-jc\omega} H(e^{j\omega}) = \pm \sum_{n=0}^{N} h[n]e^{-j\omega(c+n)}$$

• Replacing ω with $-\omega$ in the previous equation we get

$$H(-\omega) = \pm \sum_{l=0}^{N} h[l] e^{j\omega(c+l)}$$

Making a change of variable l=N-n, we rewrite the above equation as

$$H(-\omega) = \pm \sum_{l=0}^{N} h[N-n]e^{j\omega(c+N-n)}$$

As
$$H(\omega) = H(-\omega)$$
, we have
$$h[n]e^{-j\omega(c+n)} = h[N-n]e^{j\omega(c+N-n)}$$

The above leads to the condition

$$h[n] = h[N-n], \qquad 0 \le n \le N$$
with $c = -N/2$

Thus, the FIR filter with an even amplitude response will have a linear phase if it has a symmetric impulse response

If $H(\omega)$ is an odd function of ω , then from

$$e^{j(c\omega+\beta)}H(\omega)=e^{-j(-c\omega+\beta)}H(-\omega)$$

We get
$$e^{j\beta} = -e^{-j\beta}$$
 as $H(-\omega) = -H(\omega)$

The above is satisfied if $\beta = \pi/2$ or $\beta = -\pi/2$

Then

$$H(e^{j\omega}) = e^{j(c\omega + \beta)}H(\omega)$$

Reduces to

$$H\left(e^{j\omega}\right) = je^{jc\omega}H\left(\omega\right)$$

The last equation can be rewritten as

$$H(\omega) = -je^{-jc\omega}H(e^{j\omega}) = -j\sum_{n=0}^{N}h[n]e^{-j\omega(c+n)}$$

As $-H(-\omega) = H(\omega)$, from the above we get

$$-H(-\omega) = j\sum_{l=0}^{N} h[l]e^{j\omega(c+l)}$$

• Making a change of variable l=N-n, we rewrite the last equation as

$$-H(-\omega) = j\sum_{l=0}^{N} h[l]e^{j\omega(c+l)}$$

Equating the above with

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

We arrive at the condition for linear phase as

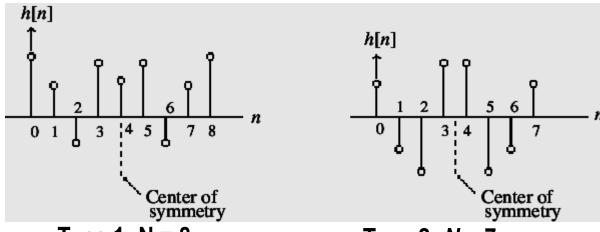
$$h[n] = -h[N-n], \quad 0 \le n \le N$$
with $c = -N/2$

Therefore a FIR filter with an odd amplitude response will have linear-phase response if it has an antisymmetric impulse response

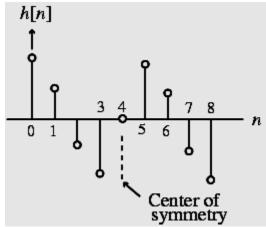
- Since the length of the impulse response can be either even or odd, we can define four types of linear-phase FIR transfer functions
- For an antisymmetric FIR filter of odd length, i.e., N even

$$h[N/2] = 0$$

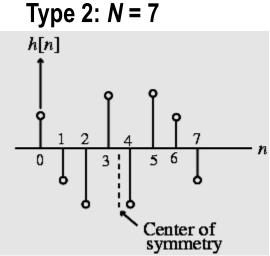
We examine next the each of the 4 cases



Type 1: N = 8



Type 3: N = 8



Type 4: N = 7

If the impulse response of FIR filter h[n] is real causal (N+1)-point (n:[0,N]) sequence, and satisfied with :

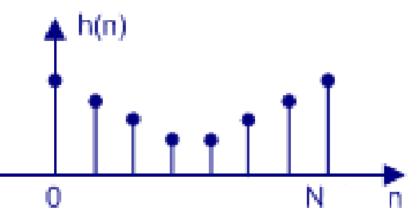
$$h[n] = h[N - n]$$
 (symmetric) or $h[n] = -h[N - n]$ (antisymmetric), then

$$h[n] \Leftrightarrow H(e^{jw}) = \overset{\circ}{H}(\omega)e^{-jN\omega/2} = |H(\omega)|e^{-jN\omega/2}e^{j\beta}$$

or

$$h[n] \Leftrightarrow H(e^{jw}) = jH(\omega)e^{-jN\omega/2} = |H(\omega)|e^{-jN\omega/2+\pi/2}e^{j\beta}$$

where the amplitude response $H(\omega)$ can become negative,



$$\beta = 0, \pi$$
 for $H(\omega) \ge 0, or < 0$

The magnitude and phase of FIR filter are given by: N_{∞}

given by:
$$\varphi(\omega) = \begin{cases} -\frac{N\omega}{2}, & \text{for } H(\omega) \ge 0 \\ -\frac{N\omega}{2} - \pi, & \text{for } H(\omega) < 0 \end{cases}$$

$$|H(e^{j\omega})| = |H(\omega)|$$

$$\mathbf{Or} \quad \varphi(\omega) = \begin{cases} -\frac{N\omega}{2} + \frac{\pi}{2}, & \text{for } H(\omega) \ge 0 \\ -\frac{N\omega}{2} - \frac{\pi}{2}, & \text{for } H(\omega) < 0 \end{cases}$$

The group delay is $\tau(\omega) = N/2$.

Proof: When h[n] is symmetric,

$$h[n] = h[N - n]$$

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

$$= z^{-\frac{N}{2}} \sum_{n=0}^{N} h[n] \left[\frac{1}{2} Z^{-(n-\frac{N}{2})} + \frac{1}{2} Z^{(n-\frac{N}{2})} \right]$$

So,

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

$$= e^{-jN\omega/2} \stackrel{\cup}{H} (\omega)$$

In similar way, we can get:

$$H(e^{j\omega}) = -je^{-jN\omega/2}\sum_{n=0}^{N}h[n]\sin[(n-N/2)\omega]$$

$$= je^{-jN\omega/2} \overset{\circ}{H} (\omega) \qquad \qquad \qquad (2)$$

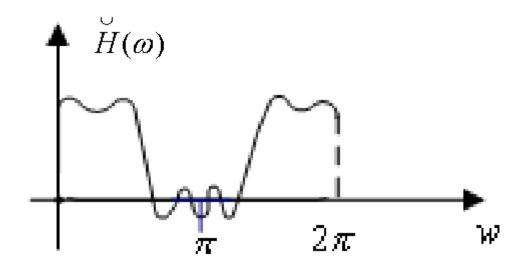
when h[n] is antisymmetric.

Type 1: h[n] = h[N-n] with Odd Length From (1),

$$\stackrel{\cup}{H}(\omega) = \sum_{n=0}^{N} h[n] \cos \left[\omega \left(n - \frac{N}{2}\right)\right]$$

$$H(\omega) = h[\frac{N}{2}] + \sum_{n=1}^{N/2} 2h[\frac{N}{2} - n] \cos [\omega n]_{69}$$

Because $\cos n\omega$ is symmetric at $\omega = 0, \pi, 2\pi$, therefore, $H(\omega)$ is symmetric at $\omega = 0, \pi, 2\pi$.



Type 2: h[n] = h[N - n] with Even Length From (1),

$$\stackrel{\circ}{H}(\omega) = \sum_{n=0}^{N} h[n] \cos \left[\omega \left(n - \frac{N}{2}\right)\right]$$

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

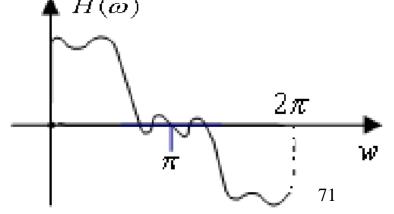
$$\overset{\cup}{H}(\omega) = \sum_{n=1}^{(N+1)/2} 2h \left[\frac{N+1}{2} - n \right] \cos \left[\omega \left(n - \frac{1}{2} \right) \right]$$

Because $\cos \left[\omega \left(n - \frac{1}{2} \right) \right]$ is antisymmetric at

$$\omega = \pi$$
, therefore, $\overset{\circ}{H}(\omega)$

is antisymmetric at

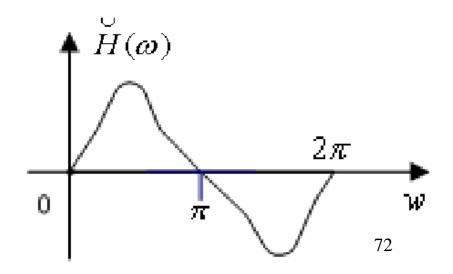
$$\omega = \pi \quad (H(\pi) = 0)$$



Type 3: h[n] = -h[N-n] with Odd Length

From (2),
$$H(\omega) = 2\sum_{n=1}^{N/2} h[\frac{N}{2} - n]\sin(n\omega)$$

Because $\sin n\omega$ is zero and antisymmetric at $\omega = 0, \pi, 2\pi$, therefore, $H(\omega)$ is antisymmetric at $\omega = 0, \pi, 2\pi$ ($H(\omega) = 0$ at $\omega = 0, \pi, 2\pi$).

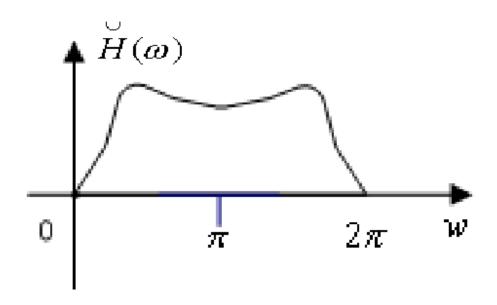


Type 4: h[n] = -h[N-n] with EvenLength

From (2),
$$U(\omega) = 2\sum_{n=1}^{(N+1)/2} h\left[\frac{N+1}{2} - n\right] \sin(n-\frac{1}{2})\omega$$

Because
$$\sin \left[\omega \left(n-\frac{1}{2}\right)\right]=0$$
 at $\omega=0,2\pi$.

Therefore, $H(\omega)$ is antisymmetric at $\omega = 0.2\pi$.



Zero Locations of Linear-Phase FIR Transfer Functions

An Linear-Phase FIR Transfer Functions

satisfies

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

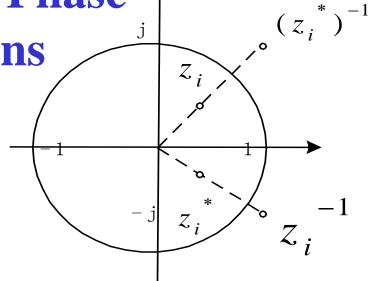
$$H(z) = -z^{-N} H(z^{-1})$$

If h[n] is real, z_i is a zero of H(z), then, z_i^{-1} is also a zero of H(z).

That means the zeros of H(z) are symmetric on the unit circle of Z-plane.

74

Zero Locations of Linear-Phase FIR Transfer Functions



Discussion:

(a) Type 2 FIR has zero at z = -1 ($\omega = \pi$), it can not be used to design a highpass filter.

(b) Type 3 FIR has zeros z=-1 ($\omega=\pi$), z=1 ($\omega=0$), it can not be used to design a highpass, lowpass or bandstop filter.

Zero Locations of Linear-Phase FIR Transfer Functions

(c) Type 4 FIR has zeros at z = 1 ($\omega = 0.2\pi$), it can not be used to design a lowpass filter. (d) Type 1 FIR has no such restrictions and can be used to design almost any type of filter.

Simple Digital Filters

Example 1. Lowpass and Highpass

$$H_{0}(z) = \frac{1}{2}(1+z^{-1})$$
 $H_{1}(z) = \frac{1}{2}(1-z^{-1})$ $H_{0}(e^{j\omega}) = e^{-j\omega/2}\cos(\omega/2)$ Linear Phase? $H_{1}(e^{j\omega}) = je^{-j\omega/2}\sin(\omega/2)$ Type?

Example 2. The M-order FIR Comb Filter

$$y[n] = \frac{1}{M} \{x[n] + x[n-1] + x[n-2] + \dots + x[n-M+1] \}$$

The impulse response

Simple Digital Filters

$$h[n] = \frac{1}{M} [u[n] - u[n - M]]$$

$$H(z) = \frac{1}{M} \frac{1 - z^{-M}}{1 - z^{-1}}$$

$$H(e^{j\omega}) = \frac{1}{M} \frac{1 - e^{-jM\omega}}{1 - e^{-j\omega}}$$

$$\left|H\left(e^{j\omega}\right)\right| = \frac{1}{M} \left|\frac{\sin(\frac{M\omega}{2})}{\sin(\frac{\omega}{2})}\right|$$
 Linear-phase?

Type 1: Symmetric Impulse Response with Odd Length

- In this case, the degree N is even
- Assume N = 8 for simplicity
- The transfer function H(z) is given by

$$H(z) = h[0] + h[1]z^{-1} + h[2]z^{-2} + h[3]z^{-3}$$
$$+ h[4]z^{-4} + h[5]z^{-5} + h[6]z^{-6} + h[7]z^{-7} + h[8]z^{-8}$$

- Because of symmetry, we have h[0]=h[8], h[1]=h[7], h[2]=h[6], and h[3]=h[5]
- Thus, we can write

$$H(z) = h[0](1+z^{-8}) + h[1](z^{-1}+z^{-7})$$

$$+ h[2](z^{-2}+z^{-6}) + h[3](z^{-3}+z^{-5}) + h[4]z^{-4}$$

$$= z^{-4}\{h[0](z^{4}+z^{-4}) + h[1](z^{3}+z^{-3})$$

$$+ h[2](z^{2}+z^{-2}) + h[3](z+z^{-1}) + h[4]\}$$

• The corresponding frequency response is then given by

$$H(e^{j\omega}) = e^{-j4\omega} \{2h[0]\cos(4\omega) + 2h[1]\cos(3\omega) + 2h[2]\cos(2\omega) + 2h[3]\cos(\omega) + h[4]\}$$

• The quantity inside the braces is a real function of ω , and can assume positive or negative values in the range $0 \le |\omega| \le \pi$

The phase function here is given by

$$\theta(\omega) = -4\omega + \beta$$

where β is either 0 or π , and hence, it is a linear function of ω in the generalized sense

The group delay is given by

$$\tau(\omega) = -\frac{d\theta(\omega)}{d\omega} = 4$$

indicating a constant group delay of 4 samples

• In the general case for Type 1 FIR filters, the frequency response is of the form

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

where the amplitude response $H(\omega)$, also called the zero-phase response, is of the form

$$\stackrel{\circ}{H}(\omega) = h\left[\frac{N}{2}\right] + 2\sum_{n=1}^{N/2} h\left[\frac{N}{2} - n\right] \cos(\omega n)$$

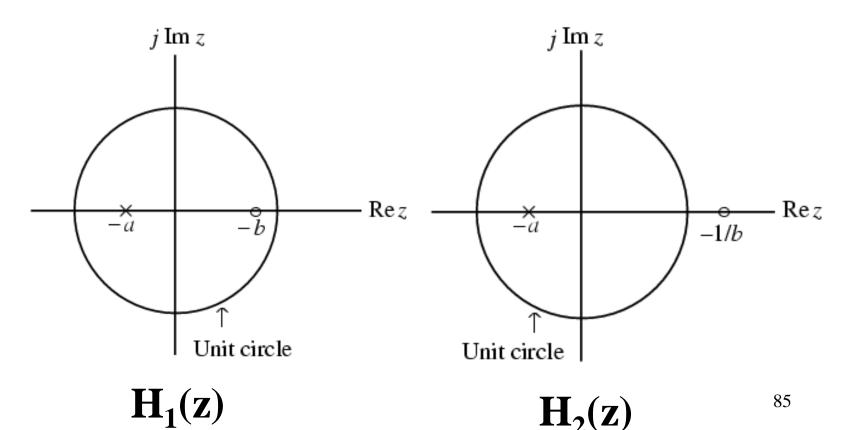
83

Consider the two 1st-order transfer functions:

$$H_1(z) = \frac{z+b}{z+a}, \quad H_2(z) = \frac{bz+1}{z+a}, \quad |a| < 1, \quad |b| < 1$$

- Both transfer functions have a pole inside the unit circle at the same location z = -a and are stable
- But the zero of $H_1(z)$ is inside the unit circle at z = -b, whereas, the zero of $H_2(z)$ is at z = -1/b situated in a mirror-image symmetry

 Figure below shows the pole-zero plots of the two transfer functions



However, both transfer functions have an identical magnitude function as

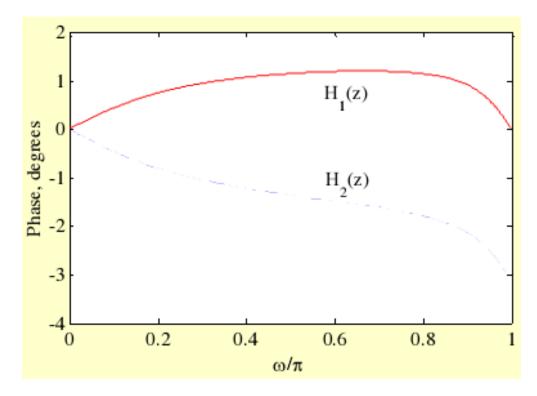
$$H_1(z)H_1(z^{-1}) = H_2(z)H_2(z^{-1})$$

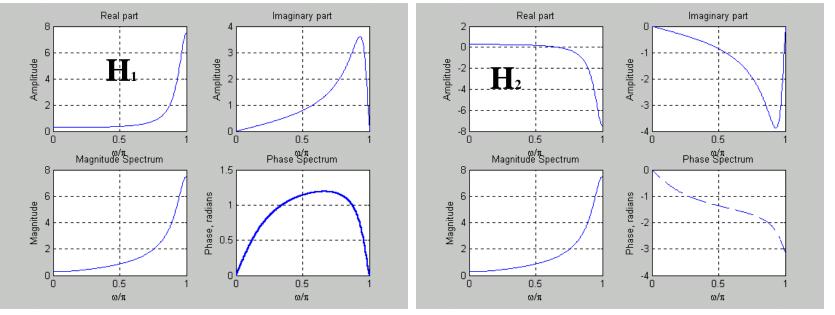
The corresponding phase functions are

$$\arg[H_1(e^{j\omega})] = \tan^{-1} \frac{\sin \omega}{b + \cos \omega} - \tan^{-1} \frac{\sin \omega}{a + \cos \omega}$$

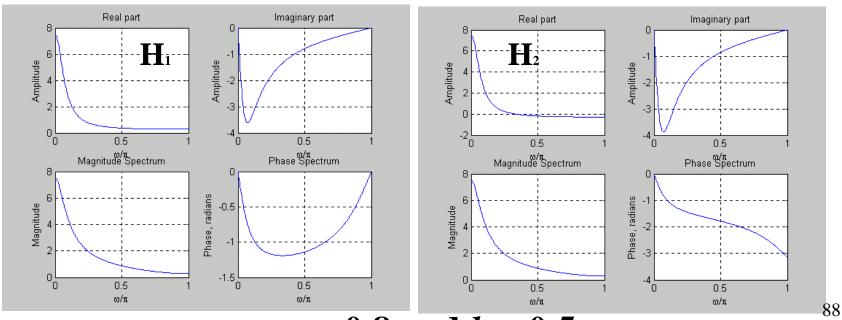
$$\arg[H_2(e^{j\omega})] = \tan^{-1} \frac{b \sin \omega}{1 + b \cos \omega} - \tan^{-1} \frac{\sin \omega}{a + \cos \omega}$$

• Figure below shows the unwrapped phase responses of the two transfer functions for a = 0.8 and b = -0.5





a = 0.8 and b = -0.5



a = -0.8 and b = 0.5

- From this figure it follows that $H_2(z)$ has an excess phase lag with respect to $H_1(z)$
- The excess phase lag property of $H_2(z)$ with respect to $H_1(z)$ can also be explained by observing that we can write

$$H_{2}(z) = \frac{bz+1}{z+a} = \left(\frac{z+b}{z+a}\right) \left(\frac{bz+1}{z+b}\right)$$

$$H_{1}(z) \qquad A(z)$$

- Where A(z) = (bz+1)/(z+b) is a stable allpass function
- The phase functions of $H_1(z)$ and $H_2(z)$ are thus related through

$$arg[H_2(e^{j\omega})] = arg[H_1(e^{j\omega})] + arg[A(e^{j\omega})]$$

• As the unwrapped phase function of a stable first-order allpass function is a negative function of ω , it follows from the above that $H_2(z)$ has indeed an excess phase lag with respect to $H_1(z)$

- Generalizing the above result, let $H_m(z)$ be a <u>causal</u> <u>stable</u> transfer function with all zeros inside the unit circle and let H(z) be another causal stable transfer function satisfying $|H(e^{j\omega})| = |H_m(e^{j\omega})|$
- These two transfer functions are then related through $H(z) = H_m(z)A(z)$ where A(z) is a <u>causal</u> <u>stable</u> allpass function

• The unwrapped phase functions of $H_m(z)$ and H(z) are thus related through

$$\arg[H(e^{j\omega})] = \arg[H_m(e^{j\omega})] + \arg[A(e^{j\omega})]$$

- H(z) has an excess phase lag with respect to $H_m(z)$
- A causal stable transfer function with all zeros inside the unit circle is called a <u>minimum-phase</u> <u>transfer function</u>

- A causal stable transfer function with all zeros outside the unit circle is called a <u>maximum</u>-<u>phase transfer function</u>
- A causal stable transfer function with zeros inside and outside the unit circle is called a <u>mixed-phase transfer function</u>

• <u>Example</u>: consider the mixed-phase transfer function

$$H(z) = \frac{2(1+0.3z^{-1})(0.4-z^{-1})}{(1-0.2z^{-1})(1+0.5z^{-1})}$$

• We can rewrite H(z) as

$$H(z) = \left[\frac{2(1+0.3z^{-1})(1-0.4z^{-1})}{(1-0.2z^{-1})(1+0.5z^{-1})} \right] \left(\frac{0.4-z^{-1}}{1-0.4z^{-1}} \right)$$



Thanks!

Any questions?