- The output y[n] of a frequency-selective LTI discrete-time system with a frequency response  $H(e^{j\omega})$  exhibits some delay relative to the input x[n] caused by the nonzero phase response  $\theta(\omega) = \arg\{H(e^{j\omega})\}$ of the system
- For an input

1

 $x[n] = A\cos(\omega_o n + \phi), \quad -\infty < n < \infty$ 

#### the output is $y[n] = A |H(e^{j\omega_o})| \cos(\omega_o n + \theta(\omega_o) + \phi)$

- Thus, the output lags in phase by  $\theta(\omega_o)$  radians
- Rewriting the above equation we get  $y[n] = A \Big| H(e^{j\omega_o}) \Big| \cos \left( \omega_o \left( n + \frac{\theta(\omega_o)}{\omega_o} \right) + \phi \right)$

• This expression indicates a time delay, known as **phase delay**, at  $\omega = \omega_o$  given by

$$\tau_p(\omega_o) = -\frac{\theta(\omega_o)}{\omega_o}$$

 Now consider the case when the input signal contains many sinusoidal components with different frequencies that are not harmonically related

- In this case, each component of the input will go through different phase delays when processed by a frequency-selective LTI discrete-time system
- Then, the output signal, in general, will not look like the input signal
- The signal delay now is defined using a different parameter

 To develop the necessary expression, consider a discrete-time signal x[n] obtained by a double-sideband suppressed carrier (DSB-SC) modulation with a carrier frequency ω<sub>c</sub> of a low-frequency sinusoidal signal of frequency ω<sub>o</sub>:

 $x[n] = A\cos(\omega_o n)\cos(\omega_c n)$ 

- The input can be rewritten as  $x[n] = \frac{A}{2}\cos(\omega_{\ell}n) + \frac{A}{2}\cos(\omega_{u}n)$ where  $\omega_{\ell} = \omega_{c} - \omega_{o}$  and  $\omega_{u} = \omega_{c} + \omega_{o}$
- Let the above input be processed by an LTI discrete-time system with a frequency response  $H(e^{j\omega})$  satisfying the condition  $H(e^{j\omega}) \cong 1$  for  $\omega_{\ell} \le \omega \le \omega_{u}$

• The output *y*[*n*] is then given by

 $y[n] = \frac{A}{2} \cos(\omega_{\ell} n + \theta(\omega_{\ell})) + \frac{A}{2} \cos(\omega_{u} n + \theta(\omega_{u}))$  $= A \cos\left(\omega_{c} n + \frac{\theta(\omega_{u}) + \theta(\omega_{\ell})}{2}\right) \cos\left(\omega_{o} n + \frac{\theta(\omega_{u}) - \theta(\omega_{\ell})}{2}\right)$ 

Note: The output is also in the form of a modulated carrier signal with the same carrier frequency ω<sub>c</sub> and the same modulation frequency ω<sub>o</sub> as the input

- However, the two components have different phase lags relative to their corresponding components in the input
- Now consider the case when the modulated input is a **narrowband signal** with the frequencies  $\omega_{\ell}$  and  $\omega_{u}$  very close to the carrier frequency  $\omega_{c}$ , i.e.  $\omega_{o}$  is very small

• In the neighborhood of  $\omega_c$  we can express the unwrapped phase response  $\theta_c(\omega)$  as  $\theta_c(\omega) \cong \theta_c(\omega_c) + \frac{d \theta_c(\omega)}{d\omega} \Big|_{\omega = \omega_c} \cdot (\omega - \omega_c)$ 

by making a Taylor's series expansion and keeping only the first two terms

• Using the above formula we now evaluate the time delays of the carrier and the modulating components

• In the case of the carrier signal we have  $-\frac{\theta_c(\omega_u) + \theta_c(\omega_\ell)}{2\omega_c} \cong -\frac{\theta_c(\omega_c)}{\omega_c}$ 

which is seen to be the same as the phase delay if only the carrier signal is passed through the system

• In the case of the modulating component we have

$$-\frac{\theta_c(\omega_u) - \theta_c(\omega_\ell)}{\omega_u - \omega_\ell} \cong -\frac{d\theta_c(\omega)}{d\omega}\Big|_{\omega = \omega_c}$$

• The parameter

$$\tau_g(\omega_c) = -\frac{d\theta_c(\omega)}{d\omega}\Big|_{\omega=\omega}$$

is called the group delay or envelope delay caused by the system at  $\omega = \omega_c$ 

- The group delay is a measure of the linearity of the phase function as a function of the frequency
- It is the time delay between the waveforms of underlying continuous-time signals whose sampled versions, sampled at *t* = *nT*, are precisely the input and the output discrete-time signals

- If the phase function and the angular frequency ω are in radians per second, then the group delay is in seconds
- Figure below illustrates the evaluation of the phase delay and the group delay



• Figure below shows the waveform of an amplitude-modulated input and the output generated by an LTI system



- <u>Note</u>: The carrier component at the output is delayed by the phase delay and the envelope of the output is delayed by the group delay relative to the waveform of the underlying continuous-time input signal
- The waveform of the underlying continuoustime output shows distortion when the group delay of the LTI system is not constant over the bandwidth of the modulated signal

- If the distortion is unacceptable, a delay equalizer is usually cascaded with the LTI system so that the overall group delay of the cascade is approximately linear over the band of interest
- To keep the magnitude response of the parent LTI system unchanged the equalizer must have a constant magnitude response at all frequencies

- <u>Example</u> The phase function of the FIR filter  $y[n] = \alpha x[n] + \beta x[n-1] + \alpha x[n-2]$ is  $\theta(\omega) = -\omega$
- Hence its group delay is given by  $\tau_g(\omega) = 1$ verifying the result obtained earlier by simulation

• <u>Example</u> - For the *M*-point moving-average filter  $h[n] = \begin{cases} 1/M, & 0 \le n \le M - 1\\ 0, & \text{otherwise} \end{cases}$ 

the phase function is

$$\theta(\omega) = -\frac{(M-1)\omega}{2} + \pi \sum_{k=0}^{\lfloor M/2 \rfloor} \mu \left(\omega - \frac{2\pi k}{M}\right)$$

• Hence its group delay is

$$\tau_g(\omega) = \frac{M-1}{2}$$

• The convolution sum description of the LTI discrete-time system is given by

$$y[n] = \sum_{k=-\infty}^{\infty} h[k] x[n-k]$$

• Taking the DTFT of both sides we obtain

$$Y(e^{j\omega}) = \sum_{n=-\infty}^{\infty} y[n]e^{-j\omega n}$$
$$= \sum_{n=-\infty}^{\infty} \left(\sum_{k=-\infty}^{\infty} h[k]x[n-k]\right)e^{-j\omega n}$$

19

• Or,

$$Y(e^{j\omega}) = \sum_{k=-\infty}^{\infty} h[k] \left( \sum_{\ell=-\infty}^{\infty} x[\ell] e^{-j\omega(\ell+k)} \right)$$
$$= \sum_{k=-\infty}^{\infty} h[k] \left( \sum_{\ell=-\infty}^{\infty} x[\ell] e^{-j\omega\ell} \right) e^{-j\omega k}$$
$$\underbrace{X(e^{j\omega})}$$

• Hence, we can write

$$Y(e^{j\omega}) = \left(\sum_{k=-\infty}^{\infty} h[k]e^{-j\omega k}\right) X(e^{j\omega}) = H(e^{j\omega}) X(e^{j\omega})$$

- In the above  $H(e^{j\omega})$  is the frequency response of the LTI system
- The above equation relates the input and the output of an LTI system in the frequency domain

- It follows from the previous equation  $H(e^{j\omega}) = Y(e^{j\omega}) / X(e^{j\omega})$
- For an LTI system described by a linear constant coefficient difference equation of the form we have

$$H(e^{j\omega}) = \frac{\sum_{k=0}^{M} p_k e^{-j\omega k}}{\sum_{k=0}^{N} d_k e^{-j\omega k}}$$

- A generalization of the frequency response function
- The convolution sum description of an LTI discrete-time system with an impulse response h[n] is given by

$$y[n] = \sum_{k=-\infty}^{\infty} h[k] x[n-k]$$

• Taking the *z*-transforms of both sides we get

$$Y(z) = \sum_{n=-\infty}^{\infty} y[n] z^{-n} = \sum_{n=-\infty}^{\infty} \left( \sum_{k=-\infty}^{\infty} h[k] x[n-k] \right) z^{-n}$$
$$= \sum_{k=-\infty}^{\infty} h[k] \left( \sum_{n=-\infty}^{\infty} x[n-k] z^{-n} \right)$$
$$= \sum_{k=-\infty}^{\infty} h[k] \left( \sum_{\ell=-\infty}^{\infty} x[\ell] z^{-(\ell+k)} \right)$$

The Transfer Function  
• Or, 
$$Y(z) = \sum_{k=-\infty}^{\infty} h[k] \left( \sum_{\ell=-\infty}^{\infty} x[\ell] z^{-\ell} \right) z^{-k}$$
  
• Therefore,  $Y(z) = \left( \sum_{\substack{k=-\infty \\ k=-\infty \\ H(z)}}^{\infty} h[k] z^{-k} \right) X(z)$   
• Thus,  $Y(z) = H(z)X(z)$ 

• Hence,

H(z) = Y(z) / X(z)

- The function H(z), which is the z-transform of the impulse response h[n] of the LTI system, is called the transfer function or the system function
- The inverse *z*-transform of the transfer function *H*(*z*) yields the impulse response *h*[*n*]

• Consider an LTI discrete-time system characterized by a difference equation

$$\sum_{k=0}^{N} d_k y[n-k] = \sum_{k=0}^{M} p_k x[n-k]$$

• Its transfer function is obtained by taking the *z*-transform of both sides of the above equation  $\sum_{k=0}^{M} -k$ 

$$H(z) = \frac{\sum_{k=0}^{M} p_k z^{-k}}{\sum_{k=0}^{N} d_k z^{-k}}$$

• Thus

• Or, equivalently as

$$H(z) = z^{(N-M)} \frac{\sum_{k=0}^{M} p_k z^{M-k}}{\sum_{k=0}^{N} d_k z^{N-k}}$$

• An alternate form of the transfer function is given by

$$H(z) = \frac{p_0}{d_0} \cdot \frac{\prod_{k=1}^M (1 - \xi_k z^{-1})}{\prod_{k=1}^N (1 - \lambda_k z^{-1})}$$

• Or, equivalently as

$$H(z) = \frac{p_0}{d_0} z^{(N-M)} \frac{\prod_{k=1}^{M} (z - \xi_k)}{\prod_{k=1}^{N} (z - \lambda_k)}$$

- $\xi_1, \xi_2, ..., \xi_M$  are the finite **zeros**, and  $\lambda_1, \lambda_2, ..., \lambda_N$  are the finite **poles** of H(z)
- If N > M, there are additional (N M) zeros at z = 0
- If N < M, there are additional (M N) poles at z = 0

- For a causal IIR digital filter, the impulse response is a causal sequence
- The ROC of the causal transfer function

$$H(z) = \frac{p_0}{d_0} z^{(N-M)} \frac{\prod_{k=1}^{M} (z - \xi_k)}{\prod_{k=1}^{N} (z - \lambda_k)}$$

is thus exterior to a circle going through the pole furthest from the origin

• Thus the ROC is given by  $|z| > \max_k |\lambda_k|$ 

- <u>Example</u> Consider the *M*-point movingaverage FIR filter with an impulse response  $h[n] = \begin{cases} 1/M, & 0 \le n \le M - 1\\ 0, & \text{otherwise} \end{cases}$
- Its transfer function is then given by

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

- The transfer function has *M* zeros on the unit circle at  $z = e^{j2\pi k/M}$ ,  $0 \le k \le M 1$
- There are M 1 poles at z = 0 and a single pole at z = 1 M = 8
- The pole at *z* = 1 exactly cancels the zero at z = 1
- The ROC is the entire z-plane except z = 0



- Example A causal LTI IIR digital filter is described by a constant coefficient difference equation given by
   y[n] = x[n-1]-1.2 x[n-2] + x[n-3]+1.3 y[n-1] -1.04 y[n-2]+0.222 y[n-3]
  - Its transfer function is therefore given by

$$H(z) = \frac{z^{-1} - 1.2z^{-2} + z^{-3}}{1 - 1.3z^{-1} + 1.04z^{-2} - 0.222z^{-3}}$$

• Alternate forms:

$$H(z) = \frac{z^2 - 1.2z + 1}{z^3 - 1.3z^2 + 1.04z - 0.222}$$
$$= \frac{(z - 0.6 + j0.8)(z - 0.6 - j0.8)}{(z - 0.3)(z - 0.5 + j0.7)(z - 0.5 - j0.7)}$$

• Note: Poles farthest from z = 0 have a magnitude  $\sqrt{0.74}$ 

• **ROC:** 
$$|z| > \sqrt{0.74}$$



- If the ROC of the transfer function H(z)includes the unit circle, then the frequency response  $H(e^{j\omega})$  of the LTI digital filter can be obtained simply as follows:  $H(e^{j\omega}) = H(z)|_{z=e^{j\omega}}$
- For a real coefficient transfer function H(z)it can be shown that  $|H(e^{j\omega})|^2 = H(e^{j\omega})H^*(e^{j\omega})$  $= H(e^{j\omega})H(e^{-j\omega}) = H(z)H(z^{-1})|_{z=e^{j\omega}}$

• For a stable rational transfer function in the form

$$H(z) = \frac{p_0}{d_0} z^{(N-M)} \frac{\prod_{k=1}^{M} (z - \xi_k)}{\prod_{k=1}^{N} (z - \lambda_k)}$$

the factored form of the frequency response is given by

$$H(e^{j\omega}) = \frac{p_0}{d_0} e^{j\omega(N-M)} \frac{\prod_{k=1}^M (e^{j\omega} - \xi_k)}{\prod_{k=1}^N (e^{j\omega} - \lambda_k)}$$

- It is convenient to visualize the contributions of the zero factor (z ξ<sub>k</sub>) and the pole factor (z λ<sub>k</sub>) from the factored form of the frequency response
- The magnitude function is given by

$$\left|H(e^{j\omega})\right| = \left|\frac{p_0}{d_0}\right| e^{j\omega(N-M)} \left|\frac{\prod_{k=1}^M \left|e^{j\omega} - \xi_k\right|}{\prod_{k=1}^N \left|e^{j\omega} - \lambda_k\right|}\right|$$

### which reduces to $|H(e^{j\omega})| = \left| \frac{p_0}{d_0} \right| \frac{\prod_{k=1}^{M} |e^{j\omega} - \xi_k|}{\prod_{k=1}^{N} |e^{j\omega} - \lambda_k|}$

• The phase response for a rational transfer function is of the form  $\arg H(e^{j\omega}) = \arg(p_0/d_0) + \omega(N-M)$  $+ \sum_{k=1}^{M} \arg(e^{j\omega} - \xi_k) - \sum_{k=1}^{N} \arg(e^{j\omega} - \lambda_k)$ 

• The magnitude-squared function of a realcoefficient transfer function can be computed using

$$|H(e^{j\omega})|^{2} = \left|\frac{p_{0}}{d_{0}}\right|^{2} \frac{\prod_{k=1}^{M} (e^{j\omega} - \xi_{k})(e^{-j\omega} - \xi_{k}^{*})}{\prod_{k=1}^{N} (e^{j\omega} - \lambda_{k})(e^{-j\omega} - \lambda_{k}^{*})}$$

• The factored form of the frequency response

$$H(e^{j\omega}) = \frac{p_0}{d_0} e^{j\omega(N-M)} \frac{\prod_{k=1}^M (e^{j\omega} - \xi_k)}{\prod_{k=1}^N (e^{j\omega} - \lambda_k)}$$

is convenient to develop a geometric interpretation of the frequency response computation from the pole-zero plot as  $\omega$ varies from 0 to  $2\pi$  on the unit circle

- The geometric interpretation can be used to obtain a sketch of the response as a function of the frequency
- A typical factor in the factored form of the frequency response is given by

$$(e^{j\omega} - \rho e^{j\phi})$$

where  $\rho e^{j\phi}$  is a zero if it is zero factor or is a pole if it is a pole factor

• As shown below in the *z*-plane the factor  $(e^{j\omega} - \rho e^{j\phi})$  represents a vector starting at the point  $z = \rho e^{j\phi}$  and ending on the unit circle at  $z = e^{j\omega}$ 



As ω is varied from 0 to 2π, the tip of the vector moves counterclockise from the point z = 1 tracing the unit circle and back to the point z = 1

• As indicated by  $|H(e^{j\omega})| = \left|\frac{p_0}{d_0}\right| \frac{\prod_{k=1}^{M} |e^{j\omega} - \xi_k|}{\prod_{k=1}^{N} |e^{j\omega} - \lambda_k|}$ 

the magnitude response  $|H(e^{j\omega})|$  at a specific value of  $\omega$  is given by the product of the magnitudes of all zero vectors divided by the product of the magnitudes of all pole vectors

• Likewise, from

 $\arg H(e^{j\omega}) = \arg(p_0/d_0) + \omega(N-M)$  $+ \sum_{k=1}^{M} \arg(e^{j\omega} - \xi_k) - \sum_{k=1}^{N} \arg(e^{j\omega} - \lambda_k)$ 

we observe that the phase response at a specific value of  $\omega$  is obtained by adding the phase of the term  $p_0/d_0$  and the linear-phase term  $\omega(N - M)$  to the sum of the angles of the zero vectors minus the angles of the pole vectors

- Thus, an approximate plot of the magnitude and phase responses of the transfer function of an LTI digital filter can be developed by examining the pole and zero locations
- Now, a zero (pole) vector has the smallest magnitude when  $\omega = \phi$

- To highly attenuate signal components in a specified frequency range, we need to place zeros very close to or on the unit circle in this range
- Likewise, to highly emphasize signal components in a specified frequency range, we need to place poles very close to or on the unit circle in this range