LTI Discrete-Time Systems in Transform Domain Ideal Filters Zero Phase Transfer Functions Linear Phase Transfer Functions

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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 functions
 - Infinite impulse response (IIR) transfer functions

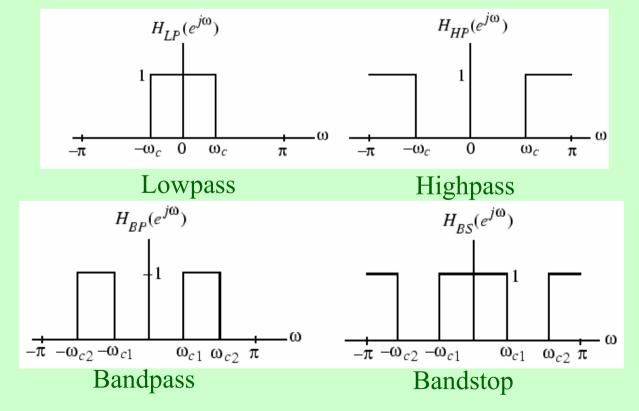
Types of Transfer Functions

- Several other classifications are also used
- In the case of digital transfer functions with frequency-selective frequency responses (filters), one classification is based on the shape of the magnitude function $|H(e^{j\omega})|$ or the form of the phase function $\theta(\omega)$
- Based on the above, four types of ideal filters are usually defined

- Frequency response equal to one at frequencies we wish to keep
- Frequency response equal to zero at all other frequencies

- The range of frequencies where the frequency response takes the value of **one** is called the **passband**
- The range of frequencies where the frequency response takes the value of **zero** is called the **stopband**

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



• The inverse DTFT of the frequency response of the ideal lowpass filter is (see previous chapter)

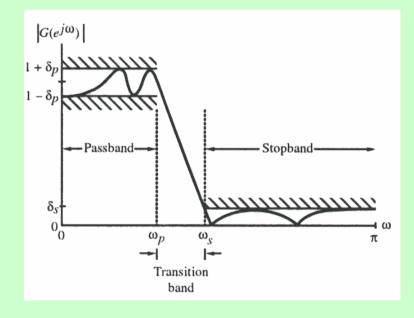
$$h_{LP}[n] = \frac{\sin \omega_c n}{\pi n}, \quad -\infty < n < \infty$$

• The above impulse response is non-causal and is of doubly infinite length. Moreover, is not absolutely summable.

- The remaining three ideal filters are also characterized by doubly infinite, non-causal 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 filters

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



- 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**
- A zero phase cannot be causal

- 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

$$x[n] \longrightarrow H(z) \longrightarrow v[n] \qquad u[n] \longrightarrow H(z) \longrightarrow w[n]$$
$$u[n] = v[-n], \qquad y[n] = w[-n]$$
$$V(e^{j\omega}) = H(e^{j\omega})X(e^{j\omega}), \qquad W(e^{j\omega}) = H(e^{j\omega})U(e^{j\omega})$$
$$U(e^{j\omega}) = V^*(e^{j\omega}), \qquad Y(e^{j\omega}) = W^*(e^{j\omega})$$

• Combining the above equations we get $Y(e^{j\omega}) = W^*(e^{j\omega}) = H^*(e^{j\omega})U^*(e^{j\omega})$ $= H^*(e^{j\omega})V(e^{j\omega}) = H^*(e^{j\omega})H(e^{j\omega})X(e^{j\omega})$ $= |H(e^{j\omega})|^2 X(e^{j\omega})$

- The function fftfilt implements the above zero-phase filtering scheme
- In the case of a causal transfer function with a non-zero phase response, the phase distortion can be avoided by ensuring that the transfer function has a unity magnitude and a **linear-phase** 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 $\omega = 2\pi$

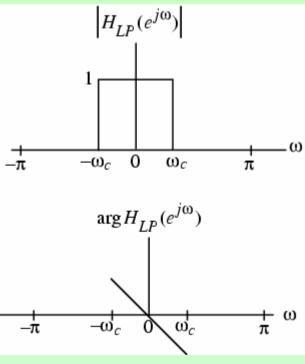
• Note also $|H(e^{j\omega})| = 1$ $\tau(\omega) = D$

- The output y[n] of this filter to an input $x[n] = Ae^{j\omega n}$ is then given by $y[n] = Ae^{-j\omega D}e^{j\omega n} = Ae^{j\omega(n-D)}$
- If x_a(t) and y_a(t) represent the continuoustime 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
- 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

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

- Figure below shows the frequency response of 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



• 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 $\sin \omega_c (n - n_o)$

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

• As before, the above filter is non-causal and of doubly infinite length, and hence, unrealizable

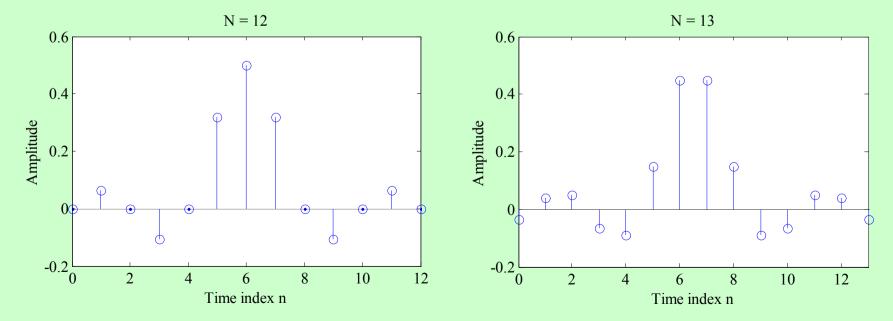
- By truncating the impulse response to a finite number of terms, a realizable FIR approximation to the ideal lowpass filter can be developed
- The truncated approximation may or may not exhibit linear phase, depending on the value of n_o chosen

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

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

will be a length *N*+1 causal linear-phase FIR filter (symmetric filter: see analysis later)

• 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:

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

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

- It is 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
- Consider a causal FIR transfer function *H*(*z*) of length *N*+1, i.e., of order *N*:

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

The above transfer function has a linear phase if its impulse response h[n] is either symmetric, i.e.,

 $h[n] = h[N-n], \ 0 \le n \le N$

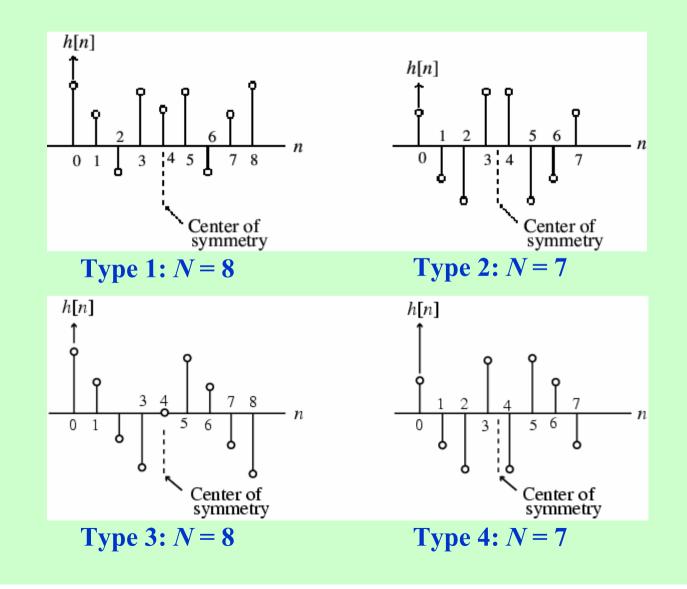
or is antisymmetric, i.e.,

$$h[n] = -h[N-n], \quad 0 \le n \le N$$

- 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 each of the 4 cases



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 brackets 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^{jω}) = e^{-jNω/2} H̃(ω)
 where the amplitude response H̃(ω), also
 called the zero-phase response, is of the
 form

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

Example 2: A Type 1 Linear Phase Filter

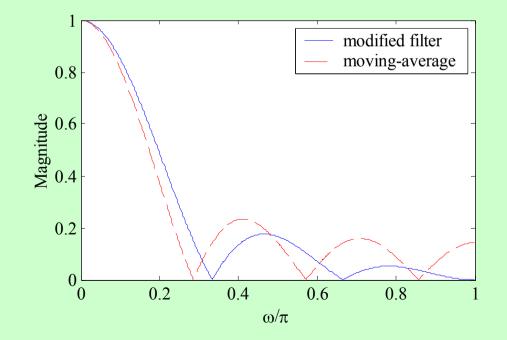
• Consider

 $H_0(z) = \frac{1}{6} \left[\frac{1}{2} + z^{-1} + z^{-2} + z^{-3} + z^{-4} + z^{-5} + \frac{1}{2} z^{-6} \right]$ which is seen to be a slightly modified version of a length-7 moving-average FIR filter

• The above transfer function has a symmetric impulse response and therefore a linear phase response

Example 2: A Type 1 Linear Phase Filter

A plot of the magnitude response of H₀(z) along with that of the 7-point moving-average filter is shown below



Example 2: A Type 1 Linear Phase Filter

- Note that the improved magnitude response obtained by simply changing the first and the last impulse response coefficients of a moving-average (MA) filter
- It can be shown that we can express $H_0(z) = \frac{1}{2}(1+z^{-1}) \cdot \frac{1}{6}(1+z^{-1}+z^{-2}+z^{-3}+z^{-4}+z^{-5})$ which a cascade of a 2-point MA filter with a 6-point MA filter
- Thus, $H_0(z)$ has a double zero at z = -1, i.e., $\omega = \pi$

Type 2: Symmetric Impulse Response with Even Length

- In this case, the degree N is odd
- Assume N = 7 for simplicity
- The transfer function is of the form $H(z) = h[0] + h[1]z^{-1} + h[2]z^{-2} + h[3]z^{-3} + h[4]z^{-4} + h[5]z^{-5} + h[6]z^{-6} + h[7]z^{-7}$

• Making use of the symmetry of the impulse response coefficients, the transfer function can be written as

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

• The corresponding frequency response is given by

 $H(e^{j\omega}) = e^{-j7\omega/2} \{2h[0]\cos(\frac{7\omega}{2}) + 2h[1]\cos(\frac{5\omega}{2}) + 2h[2]\cos(\frac{3\omega}{2}) + 2h[3]\cos(\frac{\omega}{2})\}$

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

- Here the phase function is given by $\theta(\omega) = -\frac{7}{2}\omega + \beta$ where again β is either 0 or π
- As a result, the phase is also a linear function of ω in the generalized sense
- The corresponding group delay is $\tau(\omega) = \frac{7}{2}$ indicating a group delay of $\frac{7}{2}$ samples

• The expression for the frequency response in the general case for Type 2 FIR filters is of the form

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

where the amplitude response is given by

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

Type 3: Antiymmetric Impulse Response with Odd Length

- In this case, the degree N is even
- Assume N = 8 for simplicity
- Applying the symmetry condition we get $H(z) = z^{-4} \{h[0](z^4 - z^{-4}) + h[1](z^3 - z^{-3}) + h[2](z^2 - z^{-2}) + h[3](z - z^{-1})\}$

- The corresponding frequency response is given by $H(e^{j\omega}) = e^{-j4\omega}e^{-j\pi/2} \{2h[0]\sin(4\omega) + 2h[1]\sin(3\omega) + 2h[2]\sin(2\omega) + 2h[3]\sin(\omega)\}$
- It also exhibits a generalized phase response given by

$$\theta(\omega) = -4\omega + \frac{\pi}{2} + \beta$$

where β is either 0 or π

• The group delay here is $\tau(\omega) = 4$

indicating a constant group delay of 4 samples

• In the general case

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

where the amplitude response is of the form

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

Type 4: Antiymmetric Impulse Response with Even Length

- In this case, the degree N is even
- Assume N = 7 for simplicity
- Applying the symmetry condition we get $H(z) = z^{-7/2} \{h[0](z^{7/2} - z^{-7/2}) + h[1](z^{5/2} - z^{-5/2}) + h[2](z^{3/2} - z^{-3/2}) + h[3](z^{1/2} - z^{-1/2})\}$

- The corresponding frequency response is given by
 H(e^{jω}) = e^{-j7ω/2}e^{-jπ/2}{2h[0]sin(^{7ω}/₂) + 2h[1]sin(^{5ω}/₂)
 +2h[2]sin(^{3ω}/₂) + 2h[3]sin(^ω/₂)}

 It again exhibits a generalized phase
- response given by $\theta(\omega) = -\frac{7}{2}\omega + \frac{\pi}{2} + \beta$

where β is either 0 or π

- The group delay is constant and is given by $\tau(\omega) = \frac{7}{2}$
- In the general case we have $H(e^{j\omega}) = je^{-jN\omega/2}\tilde{H}(\omega)$

where now the amplitude response is of the form

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

- Consider first an FIR filter with a symmetric impulse response: h[n] = h[N n]
- Its transfer function can be written as

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

• By making a change of variable m = N - n, we can write $\sum_{n=0}^{N} h[N-n]z^{-n} = \sum_{m=0}^{N} h[m]z^{-N+m} = z^{-N} \sum_{m=0}^{N} h[m]z^{m}$

- But, $\sum_{m=0}^{N} h[m] z^m = H(z^{-1})$
- Hence for an FIR filter with a symmetric impulse response of length N+1 we have $H(z) = z^{-N}H(z^{-1})$
- A real-coefficient polynomial H(z) satisfying the above condition is called a mirror-image polynomial (MIP)

• Now consider first an FIR filter with an antisymmetric impulse response:

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

• Its transfer function can be written as

we get

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

By making a change of variable $m = N - n$

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

 Hence, the transfer function H(z) of an FIR filter with an antisymmetric impulse response satisfies the condition

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

 A real-coefficient polynomial H(z) satisfying the above condition is called a antimirror-image polynomial (AIP)

- It follows from the relation $H(z) = \pm z^{-N} H(z^{-1})$ that if $z = \xi_o$ is a zero of H(z), so is $z = 1/\xi_o$
- Moreover, for an FIR filter with a real impulse response, the zeros of H(z) occur in complex conjugate pairs
- Hence, a zero at $z = \xi_o$ is associated with a zero at $z = \xi_o^*$

• Thus, a complex zero that is not on the unit circle is associated with a set of 4 zeros given by

$$z = re^{\pm j\phi}, \quad z = \frac{1}{r}e^{\pm j\phi}$$

• A zero on the unit circle appears as a pair $z = e^{\pm j\phi}$

as its reciprocal is also its complex conjugate

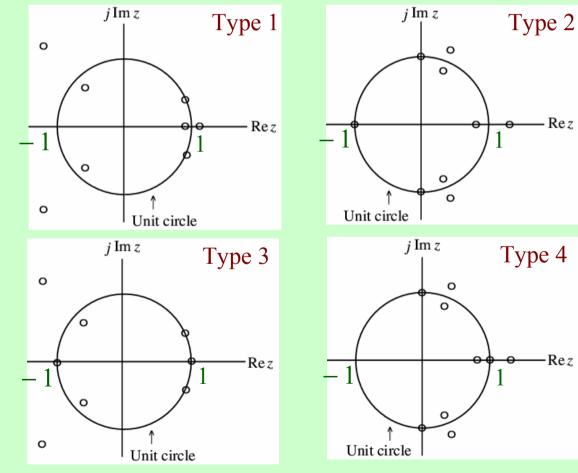
- Since a zero at z = ±1 is its own reciprocal, it can appear only singly
- Now a Type 2 FIR filter satisfies $H(z) = z^{-N}H(z^{-1})$

with degree N odd

• Hence $H(-1) = (-1)^{-N} H(-1) = -H(-1)$ implying H(-1) = 0, i.e., H(z) must have a zero at z = -1

- Likewise, a Type 3 or 4 FIR filter satisfies $H(z) = -z^{-N}H(z^{-1})$
- Thus $H(1) = -(1)^{-N} H(1) = -H(1)$ implying that H(z) must have a zero at z = 1
- On the other hand, only the Type 3 FIR filter is restricted to have a zero at z = -1since here the degree N is even and hence, $H(-1) = -(-1)^{-N} H(-1) = -H(-1)$

• Typical zero locations shown below



• Summarizing

(1) Type 1 FIR filter: Either an even number or no zeros at z = 1 and z = -1

(2) Type 2 FIR filter: Either an even number or no zeros at z = 1, and an odd number of zeros at z = -1

(3) Type 3 FIR filter: An odd number of zeros at z = 1 and z = -1

- (4) Type 4 FIR filter: An odd number of zeros at z = 1, and either an even number or no zeros at z = -1
- The presence of zeros at z = ±1 leads to the following limitations on the use of these linear-phase transfer functions for designing frequency-selective filters

- A Type 2 FIR filter cannot be used to design a highpass filter since it always has a zero z = -1
- A Type 3 FIR filter has zeros at both z = 1 and z = -1, and hence cannot be used to design either a lowpass or a highpass or a bandstop filter

- A Type 4 FIR filter is not appropriate to design a lowpass filter due to the presence of a zero at z = 1
- Type 1 FIR filter has no such restrictions and can be used to design almost any type of filter

 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^{jω}) and Y(e^{jω}) denoting their DTFTs

- Then the condition $|H(e^{j\omega})| \le 1$ implies that $|Y(e^{j\omega})|^2 \le |X(e^{j\omega})|^2$
- 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 **lossless system**

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