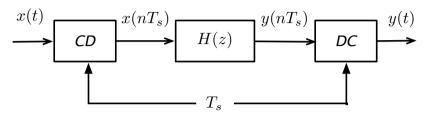
# SIGNALS AND SYSTEMS USING MATLAB Chapter 12 — Introduction to the Design of Discrete Filters

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## Frequency selective discrete filters

• Filtering of continuous and discrete-time signals



Discrete filtering of analog signals using ideal continuous to discrete (C/D), or a sampler, and discrete to continuous (D/C) converter, or a reconstruction filter

• Filtering of periodic or aperiodic signals

H(z) transfer function of discrete-time LTI system periodic input

$$x[n] = \sum_{k} A_k \cos(\omega_k n + \phi_k)$$

steady-state response

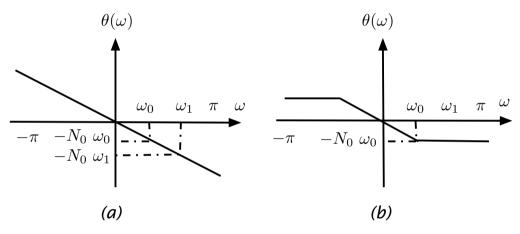
$$y_{ss}[n] = \sum_{k} A_{k} |H(e^{j\omega_{k}})| \cos(\omega_{k}n + \phi_{k} + \theta(\omega_{k}))$$

x[n] aperiodic signal with X(z)

$$Y(z)=H(z)X(z)$$
 or on the unit circle when  $z=e^{j\omega}$   $Y(e^{j\omega})=H(e^{j\omega})X(e^{j\omega})$ 

Filtering of

$$x[n] = 1 + \cos(\omega_0 n) + \cos(\omega_1 n)$$
  $\omega_1 = 2\omega_0$   $n \ge 0$  all-pass filter transfer function:  $H(z) = \alpha z^{-N_0}$ 



(a) Linear, (b) nonlinear phase

Steady-state response for linear phase

$$y_{ss}[n] = \alpha \left[ 1 + \cos(\omega_o(n - N_0)) + \cos(\omega_1(n - N_0)) \right] = \alpha x[n - N_0]$$

Steady-state response for non-linear phase

$$y_{ss}[n] = \alpha[1 + \cos(\omega_0(n - N_0)) + \cos(\omega_1(n - 0.5N_0))] \neq \alpha x[n - N_0].$$

## Group delay

$$au(\omega) = -rac{d heta(\omega)}{d\omega}.$$

General expression of linear phase: If  $\tau(\omega) = \tau$  (constant)

$$\theta(\omega) = \begin{cases} -\tau\omega - \theta_0 & -\pi \le \omega < 0 \\ 0 & \omega = 0 \\ -\tau\omega + \theta_0 & 0 < \omega \le \pi \end{cases}$$

line through origin if  $\theta_0 = 0$ 

Example: IIR and FIR filters with input x[n] and output y[n]

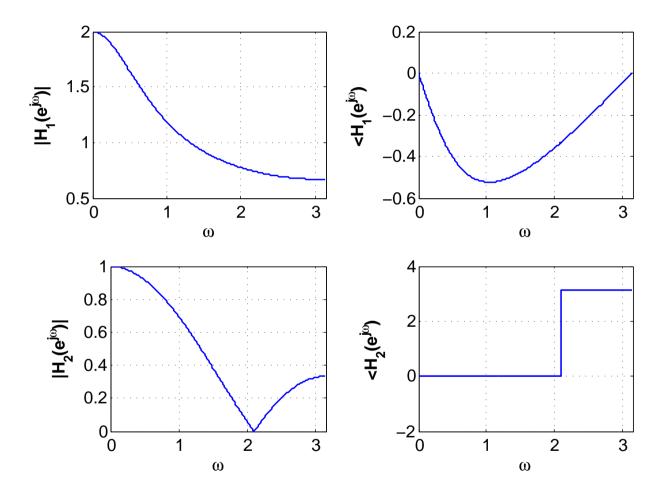
(i) 
$$y[n] = 0.5y[n-1] + x[n]$$
, (ii)  $y[n] = \frac{1}{3}(x[n-1] + x[n] + x[n+1])$ 

(i) 
$$H_1(z) = \frac{1}{1 - 0.5z^{-1}}$$

(ii) 
$$H_2(z) = \frac{1}{3}[z^{-1} + 1 + z] = \frac{1 + z + z^2}{3z} = \frac{(z - 1e^{j2.09})(z - 1e^{-j2.09})}{3z}$$

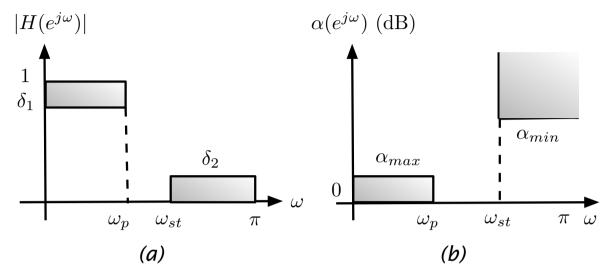
$$\angle H_1(e^{j\omega}) = - \ an^{-1}\left(rac{0.5\sin(\omega)}{1-0.5\cos(\omega)}
ight)$$
 non-linear

$$\angle H_2(e^{j\omega}) = \begin{cases} 0 & \text{when } 1 + 2\cos(\omega) \ge 0 \\ -\pi & \text{when } 1 + 2\cos(\omega) < 0 \end{cases}$$
 non-linear



Magnitude and phase responses of  $H_1(z)$  (top), and  $H_2(z)$  (bottom)

# **Filter specifications**



Low-pass magnitude specifications for IIR filter: (a) linear scale, (b) loss scale

Passband:  $\delta_1 \leq |H(e^{j\omega})| \leq 1$   $0 \leq \omega \leq \omega_p$ Stopband:  $0 < |H(e^{j\omega})| \leq \delta_2$   $\omega_{st} \leq \omega \leq \pi$ 

Loss function  $\alpha(e^{j\omega}) = -10\log_{10}|H(e^{j\omega})|^2 = -20\log_{10}|H(e^{j\omega})|$  dBs

Passband:  $0 \le \alpha(e^{j\omega}) \le \alpha_{max}$   $0 \le \omega \le \omega_p$ 

Stopband:  $\alpha_{min} \leq \alpha(e^{j\omega}) < \infty$   $\omega_{st} \leq \omega \leq \pi$ 

$$\alpha_{max} = -20\log_{10}\delta_1, \quad \alpha_{min} = -20\log_{10}\delta_2$$

Example: Not-normalized loss specifications

Passband 
$$10 \le \hat{\alpha}(e^{j\omega}) \le 11$$
  $0 \le \omega \le \frac{\pi}{2}$  Stopband  $\hat{\alpha}(e^{j\omega}) \ge 50$   $\frac{3\pi}{4} \le \omega \le \pi$ 

Let 
$$\hat{\alpha}(e^{j\omega}) = 10 + \alpha(e^{j\omega}) \Rightarrow$$

$$0 \le \alpha(e^{j\omega}) \le 1 \qquad 0 \le \omega \le \frac{\pi}{2}$$

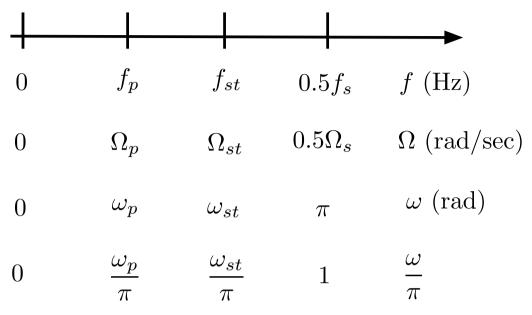
$$\alpha(e^{j\omega}) \ge 40 \qquad \frac{3\pi}{4} \le \omega \le \pi$$

$$\hat{\alpha}(e^{j0}) = 10, \ \alpha_{max} = 11 - 10 = 1, \ \alpha_{min} = 50 - 10 = 40$$

$$H(z)$$
 normalized filter  $\Rightarrow \hat{H}(z) = KH(z)$ 

dc frequency: 
$$-20\log_{10}|\hat{H}(e^{j0})| = -20\log_{10}K - 20\log_{10}|H(e^{j0})|$$
 or  $10 = -20\log_{10}K + 0 \ \Rightarrow \ K = 10^{-0.5} = \frac{1}{\sqrt{10}}$ 

## Frequency scales, time specifications



Frequency scales in discrete filter design

• Time-domain specifications: desired impulse response  $h_d[n]$ 

low-pass filter desired frequency response

$$H_d(e^{j\omega}) = \left\{ egin{array}{ll} 1e^{-j\omega N} & 0 \leq \omega \leq \omega_c \ 0 & \omega_c < \omega \leq \pi \end{array} 
ight.$$

desired impulse response

$$h_d[n] = rac{1}{2\pi} \int_{-\pi}^{\pi} H_d(e^{j\omega}) e^{j\omega n} d\omega = rac{1}{2\pi} \int_{-\omega_c}^{\omega_c} 1 e^{-j\omega N} e^{j\omega n} d\omega$$

## IIR filter design

## **Approaches**

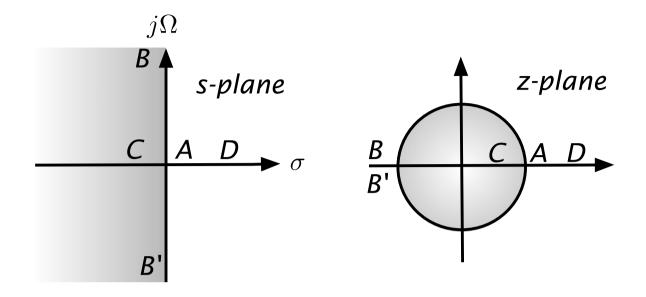
- Optimization techniques
- Analog filter design and transformations
  - Sampling transform  $z = e^{sT_s}$  (impulse invariant method)— possible aliasing
  - Bilinear transformation— results from trapezoidal rule approximation

Let 
$$H(s) = \frac{Y(s)}{X(s)} = \frac{1}{s}$$

$$y(nT_s) = \int_{(n-1)T_s}^{nT_s} x(\tau)d\tau + y((n-1)T_s)$$

$$y(nT_s) \approx \frac{[x(nT_s) + x((n-1)T_s)]}{2} + y((n-1)T_s)$$

$$Y(z) = \frac{T_s(1+z^{-1})}{2(1-z^{-1})}X(z) \Rightarrow H(z) = \frac{Y(z)}{X(z)} = \frac{T_s}{2} \frac{1+z^{-1}}{1-z^{-1}}$$
Directly from  $H(s)$  by letting  $s = \frac{2}{T_s} \frac{1-z^{-1}}{1+z^{-1}}$ 
inverse transformation  $z = \frac{1+(T_s/2)s}{1-(T_s/2)s}$ 

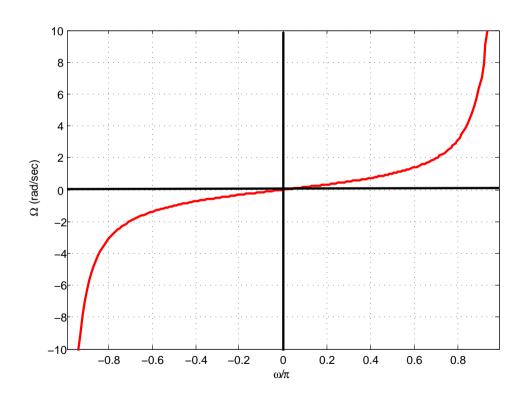


Mapping of s-plane into Z-plane by bilinear transformation

$$r = \sqrt{\frac{(1 + \sigma/K)^2 + (\Omega/K)^2}{(1 - \sigma/K)^2 + (\Omega/K)^2}}$$
$$\omega = \tan^{-1}\left(\frac{\Omega/K}{1 + \sigma/K}\right) + \tan^{-1}\left(\frac{\Omega/K}{1 - \sigma/K}\right)$$

Warping effects of BT 
$$\Omega = K \tan(\omega/2) = K \left[ \frac{\omega}{2} + \frac{\omega^3}{24} + \cdots \right]$$

$$\Omega pprox rac{\omega}{T_s}$$
 for small  $\omega$ 



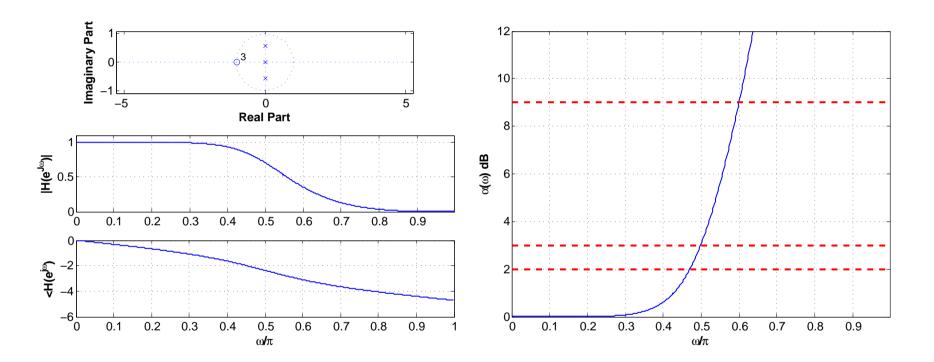
Linear relation between  $\Omega$  and  $\omega$  (K=1) for small  $\omega$ , warping as  $\omega \ o \ \pm \pi$ 

## Design of Butterworth low-pass discrete filters

Applying 
$$\Omega = K \tan(\omega/2)$$
 to analog Butterworth equation  $|H_N(j\Omega')|^2 = \frac{1}{1+(\Omega')^{2N}} \Omega' = \frac{\Omega}{\Omega_{hp}}$  gives  $|H_N(e^{j\omega})|^2 = \frac{1}{1+\left[\frac{\tan(0.5\omega)}{\tan(0.5\omega_{hp})}\right]^{2N}}$  Using frequency transformation  $\frac{\Omega_{st}}{\Omega_p} = \frac{\tan(\omega_{st}/2)}{\tan(\omega_p/2)} \Rightarrow$   $N \geq \frac{\log_{10}[(10^{0.1\alpha_{min}}-1)/(10^{0.1\alpha_{max}}-1)]}{2\log_{10}\left[\frac{\tan(\omega_{st}/2)}{\tan(\omega_p/2)}\right]}$   $2\tan^{-1}\left[\frac{\tan(\omega_p/2)}{(10^{0.1\alpha_{max}}-1)^{1/2N}}\right] \leq \omega_{hp} \leq 2\tan^{-1}\left[\frac{\tan(\omega_{st}/2)}{(10^{0.1\alpha_{min}}-1)^{1/2N}}\right]$   $\Omega'_{hp} = 1 \rightarrow \omega_{hp} \Rightarrow K_b = \frac{\Omega'}{\tan(0.5\omega)}\left|_{\Omega'=1,\omega=\omega_{hp}} = \frac{1}{\tan(0.5\omega_{hp})}\right|$   $H_N(s) \rightarrow H_N(z) = H_N(s)\left|_{s=K_h(1-z^{-1})/(1+z^{-1})}\right|$ 

Example: Low-pass filter specifications

$$\omega_p = 0.47\pi \text{ (rad)}$$
  $\alpha_{max} = 2 \text{ } dB$   $\omega_{st} = 0.6\pi \text{ (rad)}$   $\alpha_{min} = 9 \text{ } dB$   $\alpha(e^{j0}) = 0 \text{ } dB$ 

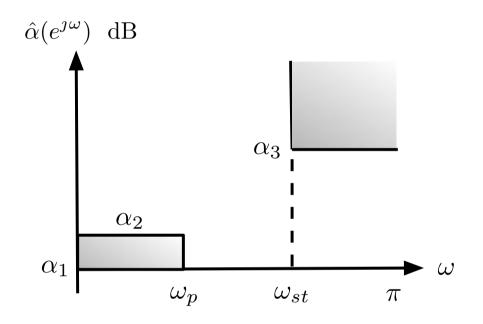


Design of low-pass Butterworth filter using MATLAB: poles and zeros, magnitude and phase response (left); verification of specifications using loss function  $\alpha(\omega)$ 

$$H(z) = \frac{0.166 + 0.497z^{-1} + 0.497z^{-2} + 0.166z^{-3}}{1 - 0.006z^{-1} + 0.333z^{-2} - 0.001z^{-3}}$$

Example: Design of Butterworth LPF for processing analog signal

 $f_p=2,250$  Hz passband frequency,  $\alpha_1=-18$  dB dc loss  $f_{st}=2,700$  Hz stopband frequency,  $\alpha_2=-15$  dB loss in passband  $f_s=9,000$  Hz sampling frequency,  $\alpha_3=-9$  dB loss in stopband

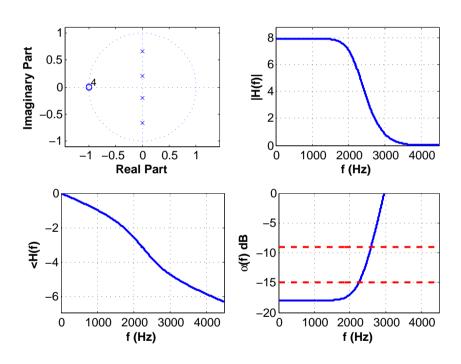


Normalized specs 
$$\hat{\alpha}(e^{j0}) = -18 \text{ dB, dc gain}$$
 
$$\alpha_{max} = \alpha_2 - \alpha_1 = 3 \text{ dB}$$
 
$$\alpha_{min} = \alpha_3 - \alpha_1 = 9 \text{ dB,}$$
 
$$\omega_p = \frac{2\pi f_{hp}}{f_s} = 0.5\pi \text{ (half-power freq.)}$$
 
$$\omega_{st} = \frac{2\pi f_{st}}{f_s} = 0.6\pi$$

$$T_s = 1/f_s = (1/9) \times 10^{-3} \; \operatorname{sec/sample} \; \Rightarrow \; K_b = \cot(\pi f_{hp} T_s) = 1$$

$$0.5\omega_{hp}=\pi/4 \ \Rightarrow \ lpha(e^{j\omega})=10\log_{10}(1+( an(0.5\omega))^{2N})$$

at 
$$\omega = \omega_{st} \ \alpha(e^{j\omega_{st}}) \geq \alpha_{min} \ \Rightarrow \ N = \left\lceil \frac{\log_{10}(10^{0.1\alpha_{min}} - 1)}{2\log_{10}(\tan(0.5\omega_{st}))} \right\rceil = 4$$



$$H'(z) = GH(z)$$
 with dc loss of  $-18dB$ 
 $\alpha(e^{j0}) = -18 = -20 \log_{10} G \Rightarrow G = 7.94$ 
 $H'(z) = GH(z) = \frac{(z+1)^4}{(z^2+0.45)(z^2+0.04)}$ 

# Design of Chebyshev Butterworth low-pass discrete filters

Mapping 
$$\Omega_p' = 1 \rightarrow \omega_p \Rightarrow K_c = \frac{1}{\tan(0.5\omega_p)}$$

$$\frac{\Omega}{\Omega_p} = \frac{\tan(0.5\omega)}{\tan(0.5\omega_p)} \Rightarrow |H_N(e^{j\omega})|^2 = \frac{1}{1 + \varepsilon^2 C_N^2(\tan(0.5\omega)/\tan(0.5\omega_p))}$$

C(.) Chebyshev polynomials

$$\varepsilon = (10^{0.1\alpha \max} - 1)^{1/2}$$

$$N \ge \frac{\cosh^{-1} \left( [(10^{0.1\alpha \min} - 1)/(10^{0.1\alpha \max} - 1)]^{1/2} \right)}{\cosh^{-1} [\tan(0.5\omega_{st})/\tan(0.5\omega_{p})]}$$

$$\omega_{hp} = 2 \tan^{-1} \left[ \tan(0.5\omega_{p}) \cosh\left(\frac{1}{N} \cosh^{-1}\left(\frac{1}{\varepsilon}\right)\right) \right]$$

$$H_N(z) = H_N(s)|_{s=K_c(1-z^{-1})/(1+z^{-1})}$$

Example: Difference between even and odd order lowpass Chebyshev filters

Specs for first filter 
$$\alpha(e^{j0})=0$$
 dB  $\omega_p=0.47\pi$  rad,  $\alpha_{max}=2$  dBs  $\omega_{st}=0.6\pi$  rad,  $\alpha_{min}=6$  dBs

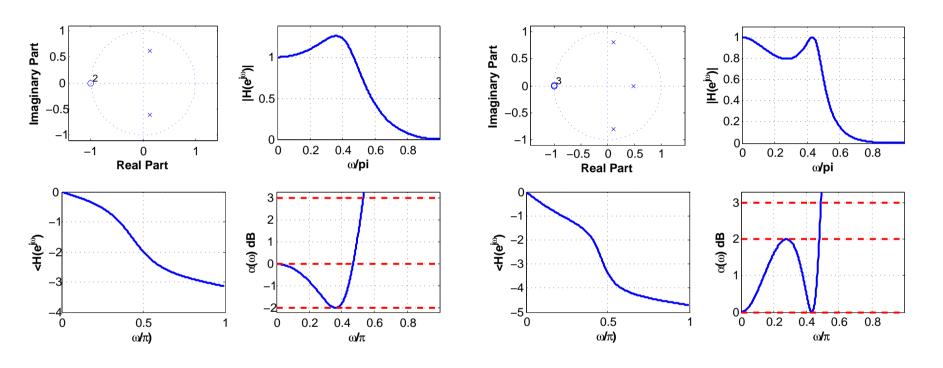
Specs for second filter

$$\omega_p = 0.48 \ rad$$
, others the same

Transfer functions

$$H_1(z) = \frac{0.224 + 0.449z^{-1} + 0.224z^{-2}}{1 - 0.264z^{-1} + 0.394z^{-2}}, \quad \omega_{hp} = 0.493\pi \ rad.$$

$$H_2(z) = \frac{0.094 + 0.283z^{-1} + 0.283z^{-2} + 0.094z^{-3}}{1 - 0.691z^{-1} + 0.774z^{-2} - 0.327z^{-3}}$$



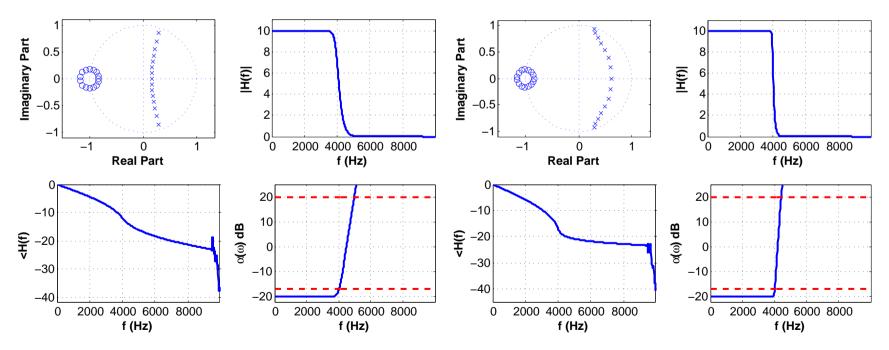
Two Chebyshev filters with different transition bands: even-order filter for  $\omega_p=0.47\pi$  on the left, and odd-order filter for  $\omega_p=0.48\pi$  (narrower transition band) on the right.

Example: Butterworth and Chebyshev filter designs to filter an acoustic signal

dc gain = 10, half-power frequency  $f_{hp} = 4 \ KHz$ band-stop frequency  $f_{st} = 5 \ Khz$   $\alpha_{min} = 60 \ dBs$ sampling frequency  $f_s = 20 \ KHz$ 

#### Normalized specs

dc gain =  $10 \Rightarrow \alpha(e^{j0}) = -20$  dBs half-power frequency:  $\omega_{hp} = 2\pi f_{hp}(1/f_s) = 0.4\pi$  rad band-stop frequency:  $\omega_{st} = 2\pi f_{st}(1/f_s) = 0.5\pi$  rad



Equal-order (N = 15) Butterworth (left) and Chebyshev filters acoustic signal

## **Rational frequency transformations**

Given prototype LPF we wish to transform it to desired filter using

$$G(z^{-1}) = Z^{-1}$$
 that

- is rational
- maps inside of the unit circle in the Z-plane into the inside of the unit circle in the z-plane
- maps |Z| = 1 into |z| = 1

If 
$$Z=Re^{j\theta},\ z=re^{j\omega}$$
  $G(e^{-j\omega})=|G(e^{-j\omega})|e^{j\angle(G(e^{-j\omega}))}=1\ e^{-j\theta}$  all-pass characteristics

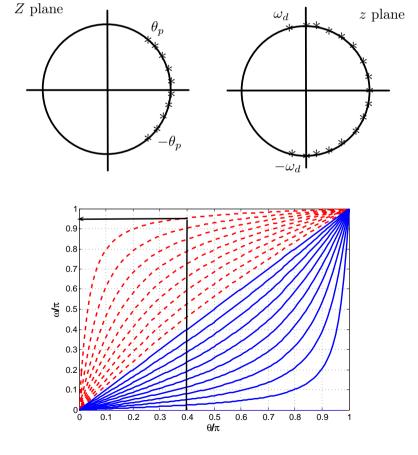
$$Z^{-1} = G(z^{-1}) = A \prod_{k} \frac{z^{-1} - \alpha_{k}}{1 - \alpha_{k}^{*} z^{-1}}$$

#### LP to LP transformation

$$Z^{-1} = A \frac{z^{-1} - \alpha}{1 - \alpha z^{-1}}$$

$$A = 1, \quad \alpha = \frac{\sin((\theta_p - \omega_d)/2)}{\sin((\theta_p + \omega_d)/2)}$$

- Zero frequencies in two planes are mapped into each other, then A
- ullet Letting  $Z=1e^{j heta}$  and  $z=1e^{j\omega}$  in transformation we get lpha
- $\bullet$  Non-linear relation between frequencies  $\theta$  and  $\omega$



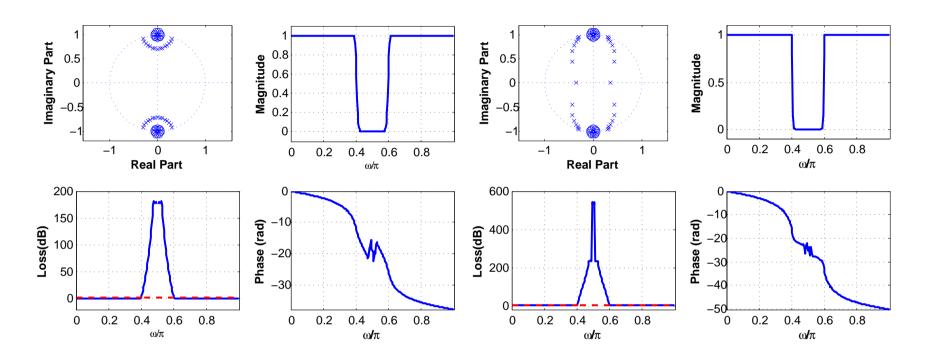
## LP to HP/BP/BE transformation

• Low-pass to high-pass transformation

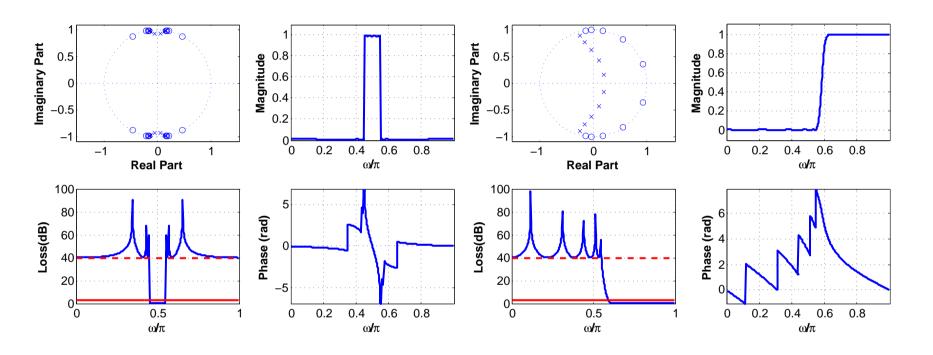
LP-HP 
$$Z^{-1} = -\left(\frac{z^{-1} - \alpha}{1 - \alpha z^{-1}}\right)$$
 (linear) 
$$\alpha = \frac{\cos((\theta_p + \omega_d)/2)}{\cos((\theta_p - \omega_d)/2)}$$

• Low-pass to band-pass and band-eliminating transformation

LP-BP 
$$Z^{-1} = -\left(\frac{z^{-2} - bz^{-1} + c}{cz^{-2} - bz^{-1} + 1}\right)$$
 (quadratic)  
LP-BE  $Z^{-1} = \frac{z^{-2} - (b/k)z^{-1} - c}{-cz^{-2} - (b/k)z^{-1} + 1}$  (quadratic)  
 $b = 2\alpha k/(k+1)$   
 $c = (k-1)/(k+1)$   
 $\alpha = (\cos((\omega_{du} + \omega_{d\ell})/2))/(\cos((\omega_{du} - \omega_{d\ell})/2))$   
 $k = \cot((\omega_{du} - \omega_{d\ell})/2))\tan(\theta_p/2)$ 



Bandstop Butterworth (left) and Chebyshev (right) filters



Elliptic band-pass filter (left) and high-pass using cheby2:(clockwise for each side from top left) poles/zeros, magnitude, phase frequency responses, loss.

## FIR filter design

#### Window Method

Desired low-pass frequency response

$$|H_d(e^{j\omega})| = \begin{cases} 1 & -\omega_c \le \omega \le \omega_c \\ 0 & \text{otherwise} \end{cases}$$
  
linear phase  $\theta(\omega) = -\omega M/2$ 

impulse response

$$h_d[n] = \begin{cases} \sin(\omega_c(n - M/2)/(\pi(n - M/2)) & n \neq M/2 \\ \omega_c/\pi & n = M/2 \end{cases}$$

Window w[n] of length M and centered at M/2

windowed impulse response 
$$h[n] = h_d[n]w[n]$$
 designed FIR filter  $H(z) = \sum_{n=0}^{M-1} h[n]z^{-n}$ 

- Window method is a trial-and-error procedure
- Symmetry of h[n] with respect to M/2, independent of whether this is an integer or not, guarantees the linear phase of the filter

#### Window functions

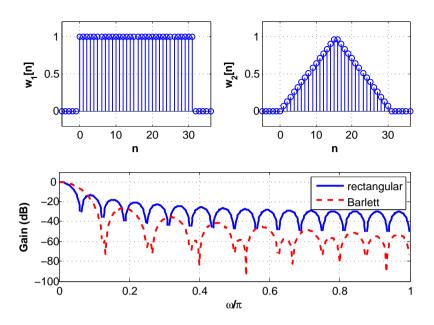
Rectangular window of length 
$$N$$

$$w[n] = \begin{cases} 1 & -(N-1)/2 \le n \le (N-1)/2 \\ 0 & \text{otherwise} \end{cases}$$

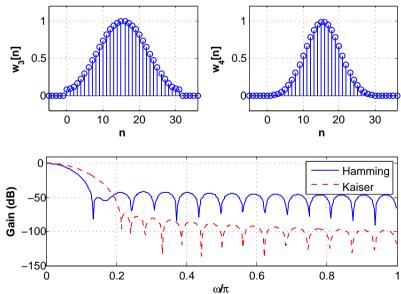
- ullet  $H_{\mathsf{W}}(e^{j\omega})=H_d(e^{j\omega})$ , requires rectangular window of infinite length
- $h_W[n] = h_d[n]w[n]$ , in frequency

$$H_{\mathsf{W}}(e^{j\omega}) = \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(e^{j\theta}) W(e^{j(\omega-\theta)}) d\theta$$
  
=  $\int_{-\pi}^{\pi} H_d(e^{j\theta}) \delta(\omega-\theta) d\theta = H_d(e^{j\omega})$ 

- If N is finite, convolution gives distorted  $H_d(e^{j\omega})$ .
- Good approximation of  $H_d(e^{j\omega})$  using a finite window w[n] requires the window must have
  - spectrum approximating an impulse in frequency
  - most of the energy concentrated in the low frequencies



Rectangular and Bartlett causal windows and their spectra



Hamming and Kaiser causal windows and their spectra

Example: Design low-pass FIR filter of length M=21 for filtering analog signals

$$H_d(e^{jf}) = \begin{cases} 1 & -125 \le f \le 125 \text{ Hz} \\ 0 & \text{elsewhere in } -f_s/2 < f \le f_s/2 \end{cases}$$
  
 $f_s = 1000 \text{ Hz}$ 

Use rectangular Hamming windows

Using  $\omega = 2\pi f/f_s$ , the discrete frequency response is given by

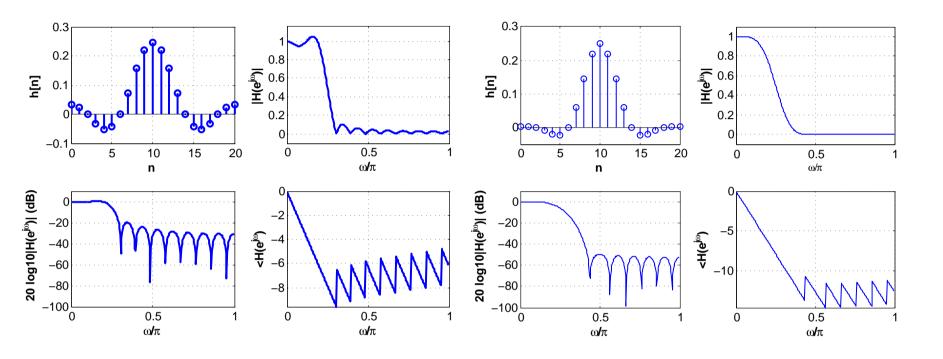
$$H_d(e^{j\omega}) = \left\{ egin{array}{ll} 1 & -\pi/4 \leq \omega \leq \pi/4 \ 0 & ext{elsewhere in } -\pi < \omega \leq \pi \end{array} 
ight.$$

The desired impulse response:

$$h_d[n] = \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(e^{j\omega}) e^{j\omega n} d\omega = \begin{cases} \sin(\pi n/4)/(\pi n) & n \neq 0 \\ 0.25 & n = 0 \end{cases}$$

Rectangular window:

$$\hat{H}(z) = H_{\mathsf{W}}(z)z^{-10} = 0.25z^{-10} + \sum_{n=0, n \neq 10}^{20} \frac{\sin(\pi(n-10)/4)}{\pi(n-10)}z^{-n}$$



Low-pass FIR filters using rectangular (left) and Hamming windows.

Design high-pass FIR filter of order M-1=14, and cut-off frequency  $0.2\pi$  using the Kaiser window.

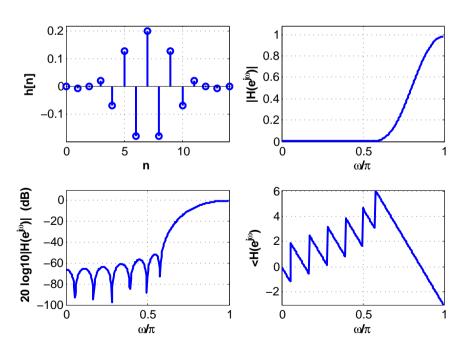
 $h_{lp}[n]$  impulse response of ideal low-pass filter

$$H_{lp}(e^{j\omega}) = \left\{ egin{array}{ll} 1 & -\omega_c \leq \omega \leq \omega_c \ 0 & ext{otherwise in } [-\pi, \ \pi) \end{array} 
ight.$$

Modulation property of the DTFT:

$$2h_{lp}[n]\cos(\omega_0 n) \Leftrightarrow H_{lp}(e^{j(\omega+\omega_0)}) + H_{lp}(e^{j(\omega-\omega_0)})$$

If  $\omega_0 = \pi$ ,  $h_{hp}[n] = 2h_{lp}[n]\cos(\pi n) = 2(-1)^n h_{lp}[n]$  is the desired impulse response of the high-pass filter



High-pass FIR filter design using Kaiser window.

#### Realization of discrete filters

- hardware or software realization
- issues to consider
  - computational complexity, minimal realizations
  - quantization
- common realizations
  - Direct Form
  - Cascade
  - Parallel
- Direct minimal realization
  - Constant numerator

transfer function 
$$H(z) = \frac{Y(z)}{X(z)} = \frac{B(z)}{A(z)} \frac{b_0}{1 + \sum_{k=1}^{N-1} a_k z^{-k}}$$
  
input/output relation  $y[n] = -\sum_{k=1}^{N-1} a_k y[n-k] + b_0 x[n]$ 

requires N-1 delays for the output, and none for the input. This is a minimum realization of H(z)

Polynomial denominator

numerator 
$$B(z) = \sum_{k=0}^{M-1} b_k z^{-k}$$

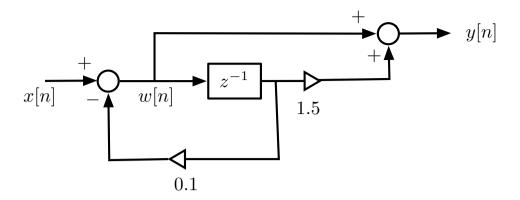
$$Y(z) = H(z)X(z) = B(z) \left[ \frac{X(z)}{A(z)} \right]$$

$$\frac{W(z)}{X(z)} = \frac{1}{A(z)} \implies w[n] = -\sum_{k=1}^{N-1} a_k w[n-k] + b_0 x[n]$$

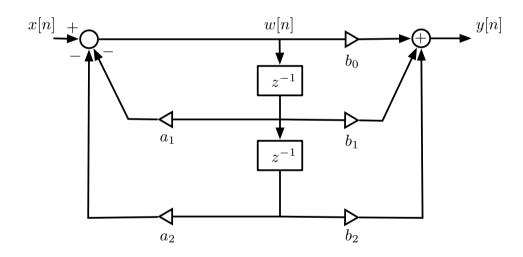
Output y[n] obtained from

$$Y(z) = B(z)W(z) \Rightarrow y[n] = \sum_{k=0}^{M-1} b_k w[n-k]$$

an input–output equation which uses the delayed signals  $\{w[n-k]\}$  from above. Number of delays used corresponds to the order of the denominator A(z) which is the order of the filter, minimal realization



Minimal direct realization of  $H(z)=(1+1.5z^{-1})/(1+0.1z^{-1})$ 

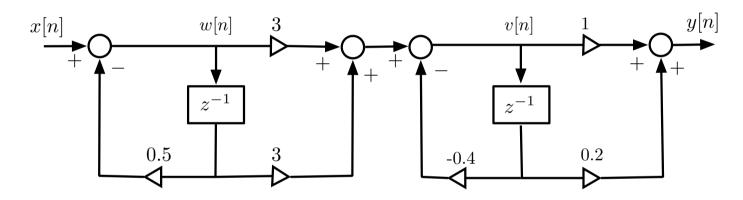


Minimal direct realization of first- and second-order filters

## • Cascade realization

$$H(z) = \frac{B(z)}{A(z)} = \prod_i H_i(z)$$

# $H_i(z)$ realized using minimal direct form

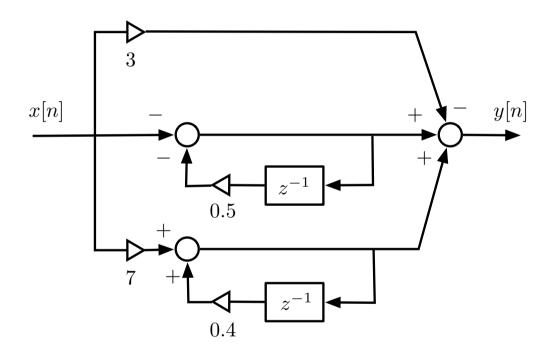


Cascade realization of 
$$H(z) = \left[3(1+z^{-1})/(1+0.5z^{-1})\right]\left[(1+0.2z^{-1})/(1-0.4z^{-1})\right]$$

## • Parallel realization

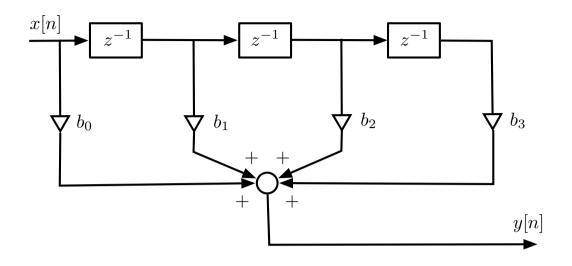
$$H(z) = \frac{B(z)}{A(z)} = C + \sum_{i=1}^{r} H_i(z)$$

C constant,  $H_i(z)$  first- or second-order filters with real coefficients

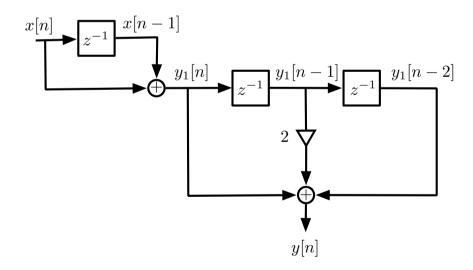


Parallel realization for  $H(z) = (3 + 3.6z^{-1} + 0.6z^{-2})/(1 + 0.1z^{-1} - 0.2z^{-2})$ 

## Realization of FIR filters



Direct form realization of FIR filter of order M = 3.



Cascade realization of FIR filter.