

# TTT4120 Digital Signal Processing Fall 2016

**Design of Digital Filters: IIR** 

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### Lecture in course book\*

- Proakis, Manolakis Digital Signal Processing, 4th Ed.
  - 10.3.3 IIR filter design by the bilinear transformation
  - 10.3.4 Characteristics of commonly used analog filters
- A compressed overview of topics treated in the lecture, see "Design av digitale filtre" on Blackboard

\*Level of detail is defined by lectures and problem sets

# **Contents and learning outcomes**

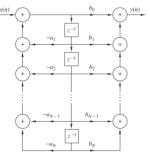
- IIR filter
- Bilinear transformation
- Examples

3

# **IIR filters**

- Moving and recursive averages
- Filter has both poles and zeros

$$H(z) = \frac{\sum_{k=0}^{M} b_k z^{-k}}{1 + \sum_{k=1}^{N} a_k z^{-k}}$$



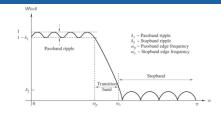
- IIR filters designed,  $\{a_k\}$  and  $\{b_k\}$ , by specifying poles and zeros in the z-plane
- In general IIR filters can, for a given filter order, satisfy a tighter specification than FIR filters (lower computational complexity)

#### IIR filters...

- In contrast to FIR filter design, IIR filters are typically designed by utilizing known analog filter design
  - Take an analog design and transform it to the digital domain
  - Nice thing: closed-form solutions exist
  - How to transform the analog solutions to discrete-time?
- Three ways of describing an analog filter
  - System function:  $H_a(s) = \frac{B(s)}{A(s)} = \frac{\sum_{k=0}^{M} \beta_k s^k}{\sum_{n=0}^{N} \alpha_k s^k}$
  - Impulse response:  $H_a(s) = \int_{-\infty}^{\infty} h_a(t)e^{-st}dt$
  - Differential equations:  $\sum_{k=0}^{N} \alpha_k \frac{d^k y(t)}{dt^k} = \sum_{k=0}^{M} \beta_k \frac{d^k x(t)}{dt^k}$
- We will be using the system function

5

#### **IIR filters...**



- IIR filter design quite different from the FIR design
  - Step 1: State filter specs of digital filter,  $\{\omega_p, \omega_s, \delta_1, \delta_2\}$
  - Step 2: Map the specs to analog domain,  $\omega_p \to \Omega_p, \, \omega_s \to \Omega_s$
  - Step 3: Design an analog filter (for resistors, capacitors, and inductors) using the Laplace transform H(s)
  - Step 4: The design in digital domain is obtained using mapping s = f(z), or  $H(z) = H(s)|_{s=f(z)}$

# Transformation between s- and z-planes

- We need to transform an analog design into a digital design
  - How to go from the s-plane to the z-plane?

$$H(z) = H_a(s)|_{s=f(z)}$$

- Demands on the mapping?
  - Stable analog filters need to be mapped to stable digital filters

$$Re\{s\} < 0 \Rightarrow |z| < 1$$

- Imaginary axis in s-plane mapped to unit circle in z-plane

$$Re\{s\} = 0 \Rightarrow |z| = 1 \Leftrightarrow j\Omega \to e^{j\omega}$$

- The bilinear tranformation satisfies these conditions
- Alternative method is to sample analog impulse respose

7

# Impulse invariance method

Sample analog impulse response

$$h[n] = h_a(t)|_{t=nT} \overset{\mathcal{F}}{\leftrightarrow} H(\Omega T) = \frac{1}{T} \sum_{k=-\infty}^{\infty} H_a\left(\Omega - \frac{2\pi k}{T}\right)$$

- Frequency mapping:  $\omega = \Omega T$ 
  - Simple and linear
  - Suffers from potential aliasing  $\Rightarrow$  not useful for highpass
- Transfer function:

$$H_a(s) = \sum_{k=1}^{N} \frac{c_k}{s - p_k} \rightarrow H(z) = \sum_{k=1}^{N} \frac{c_k}{1 - e^{p_k T_{z^{-1}}}}$$

• No explicit mapping s = f(z), but mapping of poles

$$s_k = p_k \to z_k = e^{p_k T}$$

# Impulse invariance method...

- Procedure for transforming  $H_a(s)$  to H(z):
  - 1. Find poles of  $H_a(s)$ ,  $p_k$
  - 2. Express  $H_a(s) = \sum_{k=1}^{N} \frac{c_k}{s p_k}$
  - 3. Finally,  $H(z) = \sum_{k=1}^{N} \frac{c_k}{1 e^{p_k T} z^{-1}}$

9

#### Bilinear transformation...

• The bilinear transform, a conformal mapping, provides an explicit mapping between *s*-plane and *z*-plane

$$S = \frac{2}{T} \frac{z-1}{z+1}$$
, or  $z = \frac{\frac{2}{T} + S}{\frac{2}{T} - S}$ 

• Setting  $s = \sigma + j\Omega$  and  $z = e^{j\omega}$ , we get the frequency mapping  $\omega = 2\arctan\frac{\Omega T}{2} \text{ or } \Omega = \frac{2}{T}\tan\frac{\omega}{2}$ 

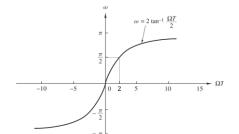
The discrete-time filter's system function is given by

$$H(z) = H_a(s) \Big|_{s = \frac{2z - 1}{Tz + 1}}$$

# Bilinear transformation...

• Example: Fill in the table using  $z = \frac{\frac{2}{T} + s}{\frac{2}{T} - s}$  and  $\omega = 2 \arctan \frac{\Omega T}{2}$ 

S	Z	ω
0		
∞		
$\frac{2j}{T}$		
$-\frac{2}{T}$		

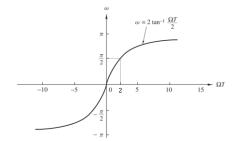


11

# Bilinear transformation...

Example: Fill in the table using  $z = \frac{\frac{2}{T} + s}{\frac{2}{T} - s}$  and  $\omega = 2 \arctan \frac{\Omega T}{2}$ 

S	Z	ω
0	1	0
8	-1	π
$\frac{2j}{T}$	j	$\pi/2$
$-\frac{2}{T}$	0	N/A



#### Bilinear transformation...

- Substitute  $s = \sigma + j\Omega$  in  $|z| = \frac{\left|\frac{2}{T} + s\right|}{\left|\frac{2}{T} s\right|}$ , look at  $\sigma < 0, \sigma = 0, \sigma > 0$ 
  - For  $\sigma < 0 \Rightarrow |z| = \left| \frac{\frac{2}{T} + \sigma + j\Omega}{\frac{2}{T} \sigma j\Omega} \right| < 1$
  - For  $\sigma = 0 \Rightarrow |z| = \begin{vmatrix} \frac{2}{T} + j\Omega \\ \frac{2}{T} j\Omega \end{vmatrix} = 1$
  - For  $\sigma > 0 \Rightarrow |z| = \left| \frac{\frac{2}{T} + \sigma + j\Omega}{\frac{2}{T} \sigma j\Omega} \right| > 1$
- Entire left half-plane maps into the inside of unit circle
- Imaginary axis maps onto the unit circle

13

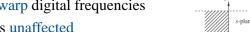
#### Bilinear transformation...

- The mapping satisfies the conditions for stability and mapping of the  $j\Omega$ -axis to the unit circle
- Reversible mapping of frequency axis, i.e.,

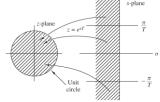
$$\Omega \in (-\infty,\infty) \longleftrightarrow \omega \in (-\pi,\pi]$$

Nonlinear relation between analog and digital frequencies

Need to pre-warp digital frequencies



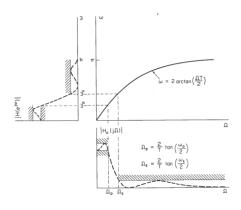
- Magnitude levels unaffected
- No aliasing
  - Can design all filter types



#### Bilinear transformation...

• Transformation of  $H_a(s)$  to H(z):

$$H(z) = H_a(s)|_{s=\frac{2}{T}\frac{1-z^{-1}}{1+z^{-1}}}$$



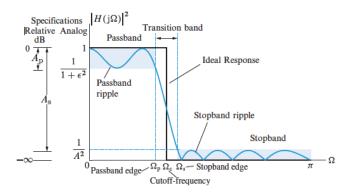
15

# Bilinear transformation...

• Example: Transform  $H_a(s) = \frac{s+1}{s^2+5s+6}$  into a digital filter using the bilinear transformation. You may choose T=1

$$H(z) = H_a(s)|_{s=2\frac{1-z^{-1}}{1+z^{-1}}} = \frac{2\frac{1-z^{-1}}{1+z^{-1}} + 1}{\left(2\frac{1-z^{-1}}{1+z^{-1}}\right)^2 + 5\left(2\frac{1-z^{-1}}{1+z^{-1}}\right) + 6}$$
$$= \frac{2\frac{1-z^{-1}}{1+z^{-1}} + 1}{\left(2\frac{1-z^{-1}}{1+z^{-1}}\right)^2 + 5\left(2\frac{1-z^{-1}}{1+z^{-1}}\right) + 6} = \frac{3+2z^{-1}-z^{-2}}{20+4z^{-1}}$$

# **Analog filter specifications**

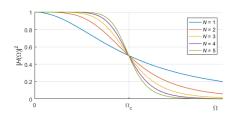


1

# Three classes of IIR filters

- Butterworth filters
  - In Matlab: butter
  - No ripples (oscillations) in  $|H(\omega)|$ , maximally flat
  - Smoothest transition from passband to stopband
- Chebyshev filters (two types)
  - cheby1 and cheby2 commands in Matlab
  - Ripples in either passband or stopband
- Elliptic filters
  - ellip in Matlab
  - Ripples in both passband and stopband
  - Sharpest transition from passband to stopband for a given order

#### **Butterworth filter**



• Frequency response:

$$|H(\Omega)|^2 = \frac{1}{1 + (\Omega/\Omega_c)^{2N}} = \frac{1}{1 + \epsilon^2 (\Omega/\Omega_n)^{2N}}$$

- N poles lying at a circle with radius  $\Omega_c$  in the s-plane
- Notice:  $|H(0)|^2 = 1$ ,  $|H(\Omega_c)|^2 = 0.5$  for all N $|H(\Omega)|^2$  monotonically decreasing
- Choose filter order depending on flatness of passband and how rapid decay in stopband

19

# **Butterworth filter...**

• How to find H(s):

$$|H(\Omega)|^2 = H(\Omega)H^*(\Omega) = H(s)H(-s)|_{s=j\Omega} = \frac{1}{1 + (-s^2/\Omega_c^2)^N}\Big|_{s=j\Omega}$$

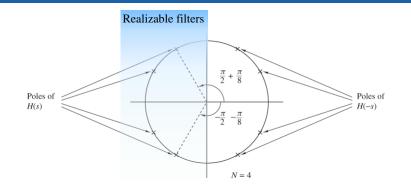
Poles can be found from

$$1 + \left( -p_k^2/\Omega_c^2 \right)^N = 0 \ \Rightarrow p_k = \Omega_C e^{\frac{j\pi}{2}} e^{\frac{j(2k+1)\pi}{2N}}, k = 0, \dots, 2N-1$$

• Poles in H(s):  $p_k$  in the left half-plane k = 0, ..., N - 1

$$H(s) = \frac{1}{(s-p_0)(s-p_1)...(s-p_{N-1})}$$

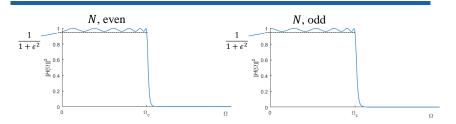
#### **Butterworth filter...**



$$p_k = \Omega_C e^{\frac{j\pi}{2}} e^{\frac{j(2k+1)\pi}{2N}}, k = 0, \dots, N-1$$

21

# **Chebyshev I**

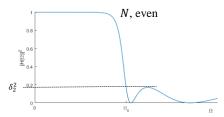


• Frequency response:

$$|H(\Omega)|^2=\frac{1}{1+\epsilon^2T_N^2(\Omega/\Omega_c)},$$
  $(T_N~(x)~N\text{th-order Chebyshev pol.})$ 

- Parameter  $\epsilon$  decides ripple in passband
- Poles lying on an ellipse in the *s*-plane

# **Chebyshev II**



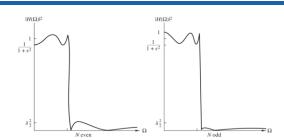
• Frequency response:

$$|H(\Omega)|^2 = \frac{1}{1+\epsilon^2 \left[T_N^2(\Omega_{\rm s}/\Omega_p)/T_N^2(\Omega_{\rm s}/\Omega)\right]}$$

- Parameter  $\epsilon$  decides ripple in stopband
- Poles on an ellipse in s-plane
- Zeros on the imaginary axis  $(j\Omega$ -axis)

23

# **Elliptic filter**



• Frequency response:

$$|H(\Omega)|^2 = \frac{1}{1+\epsilon^2 U_N^2(\Omega/\Omega_{\rm c})} (U_N(x))$$
 Nth-order Jacobi elliptic function)

- Parameter  $\epsilon$  decides ripple in passband
- Sharpest transition from passband to stopband among discussed filters
- Zeros on the imaginary axis  $(j\Omega$ -axis)

# **Example: Bandpass filter**

· Design three digital IIR bandpass filters with resonance frequencies

$$\omega_{r_1}=\frac{\pi}{4}$$
,  $\omega_{r_2}=\frac{\pi}{2}$ , and  $\omega_{r_3}=\frac{3\pi}{4}$ 

by converting the analog filter with system function

$$H_a(s) = \frac{s+0.1}{(s+0.1)^2+9}$$

using the the bilinear transformation

25

# **Example: Bandpass filter...**

Poles of analog filter

$$H_a(s) = \frac{s+0.1}{(s+0.1)^2+9} = \frac{s+0.1}{(s-[-0.1+j3])(s-[-0.1-j3])} = \frac{s+0.1}{(s-p_r)(s-p_r^*)}$$

reveals analog resonance frequency  $\Omega_r = 3 \text{ rad/s}$ 

• Use frequency relation between  $\Omega$  and  $\omega_{r_i}$  to obtain  $T_i$ 

$$\Omega = \frac{2}{T_i} \tan \frac{\omega_i}{2} \Rightarrow T_i = \frac{2}{\Omega} \tan \frac{\omega_i}{2}$$

$$T_1 = \frac{2}{3} \tan \frac{\pi}{8}, T_2 = \frac{2}{3}, \text{ and } T_3 = \frac{2}{3} \tan \frac{\pi}{8}$$

# **Example: Bandpass filter...**

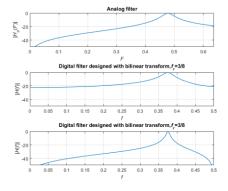
• For each  $T_i$  apply the bilinear transform:

$$\begin{split} H_i(z) &= H_a(s) \big|_{s = \frac{2}{T_i} \frac{1 - z^{-1}}{1 + z^{-1}}} = \frac{\left(\frac{2}{T_i} \frac{1 - z^{-1}}{1 + z^{-1}}\right) + 0.1}{\left(\frac{2}{T_i} \frac{1 - z^{-1}}{1 + z^{-1}} + 0.1\right)^2 + 9} \\ &= \frac{\left(2T_i + 0.1T_i^2\right) + 0.2T_i^2 z^{-1} + \left(0.1T_i^2 - 2T_i\right)}{(4 + 0.4T + 9.01T^2) + (18.02T^2) z^{-1} + (4 - 0.4T + 9.01T^2) z^{-1}} \end{split}$$

27

# Analog filter Digital filter designed with bilinear transform, f,=1/4 Digital filter designed with bilinear transform, f,=1/8 Digital filter designed with bilinear transform, f,=1/8





29

# **Summary**

- Today we discussed:
  - IIR filter design
- Next:
  - Wiener filters

# **Matlab: fdatool**

• Type fdatool at Matlab command prompt:

