Lecture 22: Design of FIR / IIR Filters

Foundations of Digital Signal Processing

Outline

- Designing FIR Filters with Windows
- Designing FIR Filters with Frequency Selection
- Designing FIR Filters with Equi-ripples
- Designing IIR Filters with Discrete Differentiation
- Designing IIR Filters with Impulse Invariance
- Designing IIR Filters with the Bilinear Transform
- Related Analog Filters

News

Homework #9

- Due <u>on Thursday</u>
- Submit via canvas

Coding Assignment #6

- Due <u>on next Monday</u>
- Submit via canvas

News

- Exam #2 Great Job!
 - Mean: 86.3
 - Median: 87.5





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Question: Consider a length-M symmetric, causal filter. What condition must be satisfied?

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What condition must be satisfied?

•
$$x[n] = \pm x[-n + (N-1)] = \pm x[N-1-n]$$

- **Positive:** Even symmetry
- **Negative:** Odd symmetry

Question: Consider a length-M symmetric, causal filter. What is the phase response?

Question: Consider a length-M symmetric, causal filter. What is the phase response? Assume M is even.

Even Symmetry

 $X(z) = a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_1 z^{-(M-2)} + a_0 z^{-(M-1)}$

• Odd Symmetry $X(z) = a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots - a_1 z^{-(M-2)} - a_0 z^{-(M-1)}$

Question: Consider a length-M symmetric, causal filter. What is the phase response? Assume M is even.

Even Symmetry

$$X(z) = a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_1 z^{-(M-2)} + a_0 z^{-(M-1)}$$

= $z^{-\frac{(M-1)}{2}} \left[a_0 z^{\frac{(M-1)}{2}} + a_1 z^{\frac{(M-1)}{2}-1} + \dots + a_1 z^{1-\frac{(M-1)}{2}} + a_0 z^{-\frac{(M-1)}{2}} \right]$

Odd Symmetry

$$X(z) = a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots - a_1 z^{-(M-2)} - a_0 z^{-(M-1)}$$

= $z^{-\frac{(M-1)}{2}} \left[a_0 z^{\frac{(M-1)}{2}} + a_1 z^{\frac{(M-1)}{2}-1} + \dots - a_1 z^{1-\frac{(M-1)}{2}} - a_0 z^{-\frac{(M-1)}{2}} \right]$

Question: Consider a length-M symmetric, causal filter. What is the phase response? Assume M is even.

Even Symmetry

• Odd Symmetry $X(z) = a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots - a_1 z^{-(M-2)} - a_0 z^{-(M-1)}$ $= z^{-\frac{(M-1)}{2}} \left[a_0 z^{\frac{(M-1)}{2}} + a_1 z^{\frac{(M-1)}{2}-1} + \dots - a_1 z^{1-\frac{(M-1)}{2}} - a_0 z^{-\frac{(M-1)}{2}} \right]$ $G(\omega) = |X(\omega)| e^{j\Theta(\omega)}$

Question: Consider a length-M symmetric, causal filter. What is the phase response? Assume M is even.

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Question: Consider a length-M symmetric, causal filter.
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$$\begin{aligned} X(z) &= a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_1 z^{-(M-2)} + a_0 z^{-(M-1)} \\ &= z^{-\frac{(M-1)}{2}} \left[a_0 z^{\frac{(M-1)}{2}} + a_1 z^{\frac{(M-1)}{2}-1} + \dots + a_1 z^{1-\frac{(M-1)}{2}} + a_0 z^{-\frac{(M-1)}{2}} \right] \\ &= z^{-\frac{(M-1)}{2}} \sum_{k=0}^{M/2-1} a_k \left[z^{\frac{(M-1)}{2}-k} + z^{-\left[\frac{(M-1)}{2}-k\right]} \right] \end{aligned}$$

Question: Consider a length-M symmetric, causal filter.
What is the phase response? Assume M is even.

$$\begin{aligned} X(z) &= a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_1 z^{-(M-2)} + a_0 z^{-(M-1)} \\ &= z^{-\frac{(M-1)}{2}} \left[a_0 z^{\frac{(M-1)}{2}} + a_1 z^{\frac{(M-1)}{2}-1} + \dots + a_1 z^{1-\frac{(M-1)}{2}} + a_0 z^{-\frac{(M-1)}{2}} \right] \\ &= z^{-\frac{(M-1)}{2}} \sum_{k=0}^{M/2-1} a_k \left[z^{\frac{(M-1)}{2}-k} + z^{-\left[\frac{(M-1)}{2}-k\right]} \right] \end{aligned}$$

• Notice that
$$X(z) = z^{-(M-1)}X(z^{-1})$$

Question: Consider a length-M symmetric, causal filter.
What is the phase response? Assume M is even.

$$\begin{aligned} X(z) &= a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_1 z^{-(M-2)} + a_0 z^{-(M-1)} \\ &= z^{-\frac{(M-1)}{2}} \left[a_0 z^{\frac{(M-1)}{2}} + a_1 z^{\frac{(M-1)}{2}-1} + \dots + a_1 z^{1-\frac{(M-1)}{2}} + a_0 z^{-\frac{(M-1)}{2}} \right] \\ &= z^{-\frac{(M-1)}{2}} \sum_{k=0}^{M/2-1} a_k \left[z^{\frac{(M-1)}{2}-k} + z^{-\left[\frac{(M-1)}{2}-k\right]} \right] \end{aligned}$$

Pole-zero plot property?
$$X(z) = z^{-(M-1)}X(z^{-1})$$

Question: Consider a length-M symmetric, causal filter. What is the phase response? Assume M is even.



Causality

Question: How do we describe causal filter magnitude?



Causality

Question: How do we describe causal filter magnitude?



Causality

Question: How do we describe causal filter magnitude?



Lecture 21: Design of FIR Filters

Foundations of Digital Signal Processing

Outline

- Review Downsampling & Upsampling
- Causality in Filters
- Designing FIR Filters with Windows
- Designing FIR Filters with Frequency Selection
- Designing FIR Filters with Equi-ripples

Question: How can I design an FIR filter from an ideal filter?



Question: How can I design an FIR filter from an ideal filter?



Question: How can I design an FIR filter from an ideal filter?



Answer: Window the response!

Question: How can I design an FIR filter from an ideal filter?



Answer: Window the response!

Question: How can I design an FIR filter from an ideal filter?



Answer: Window the response!

Different Filters









Different Filters





 $|W(\omega)|$ [hamming]

2



 ω

Different Filters



 ω

ω

















Windowing the sinc impulse response







w[n] [hamming]



Windowing the sinc impulse response





-2

 $|H(\omega)^*W(\omega)|$

2



 ω


































Windowing the sinc impulse response





-60

-2

 $|H(\omega)^*W_{\mathcal{G}}\omega)|$ in dB

 ω

w[n] [hamming]



























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Option 2: Work backwards with constraints

Consider the DFT:

$$h[n] = \sum_{k=0}^{N-1} H[k] e^{j\frac{2\pi}{N}nk}$$

such that H[k] = H[N - k]

Option 2: Work backwards with constraints

$$h[n] = \frac{1}{N} \sum_{k=0}^{N-1} H[k] e^{j\frac{2\pi}{N}nk}$$

such that
$$H[k] = H[N - k]$$

$$h[n] = \frac{1}{N} \left[H[0] + \sum_{k=1}^{(N-1)/2} H[k] e^{j\frac{2\pi}{N}nk} + \sum_{k=(N+1)/2}^{N-1} H[k] e^{j\frac{2\pi}{N}nk} \right]$$

Option 2: Work backwards with constraints

$$h[n] = \frac{1}{N} \sum_{k=0}^{N-1} H[k] e^{j\frac{2\pi}{N}nk}$$

such that
$$H[k] = H[N - k]$$

$$h[n] = \frac{1}{N} \left[H[0] + \sum_{k=1}^{(N-1)/2} H[k] e^{j\frac{2\pi}{N}nk} + \sum_{k=1}^{(N-1)/2} H[k] e^{j\frac{2\pi}{N}n(N-k)} \right]$$

Option 2: Work backwards with constraints

$$h[n] = \frac{1}{N} \sum_{k=0}^{N-1} H[k] e^{j\frac{2\pi}{N}nk}$$

such that
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Consider the DFT:

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An inverse DFT that forces time-symmetry

$$h[n] = \frac{1}{N} \left[H[0] + 2 \sum_{k=1}^{(N-1)/2} H[k] \cos\left(\frac{2\pi}{N}nk\right) \right]$$

An inverse DFT that forces time-symmetry

$$h[n] = \frac{1}{N} \left[H[0] + 2 \sum_{k=1}^{(N-1)/2} H[k] \cos\left(\frac{2\pi}{N}nk\right) \right]$$

Example: Consider the desired 9-sample frequency response with the first half defined by [1 1 0 0]

Compute the frequency sampled filter

An inverse DFT that forces time-symmetry

$$h[n] = \frac{1}{N} \left[H[0] + 2 \sum_{k=1}^{(N-1)/2} H[k] \cos\left(\frac{2\pi}{N}nk\right) \right]$$

Example: Consider the desired 9-sample frequency response with the first half defined by [1 1 0 0]

Compute the frequency sampled filter

$$h[n] = \frac{1}{2} \left[1 + 2\cos((2\pi/9)n) \right]$$

- Example: Consider the desired 9-sample frequency response with the first half defined by [1 1 0 0]
 - Compute the frequency sampled filter

 $h[n] = \frac{1}{2} \left[1 + 2\cos((2\pi/9)n) \right]$



- Example: Consider the desired 9-sample frequency response with the first half defined by [1 1 0 0]
 - Compute the frequency sampled filter

$$h[n] = \frac{1}{2} \left[1 + 2\cos((2\pi/9)n) \right]$$

In practice, this should be circularly shifted so that the h[n]



An inverse DFT that forces time-symmetry

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

Example: Consider the desired 9-sample frequency response with the first half defined by [1 1 0 0]

Compute the frequency sampled filter

$$h[n] = \frac{1}{2} \left[1 + 2\cos((2\pi/9)n) \right]$$

- Example: Consider the desired 9-sample frequency response with the first half defined by [1 1 0 0]
 - Compute the frequency sampled filter

 $h[n] = \frac{1}{2} \left[1 + 2\cos((2\pi/9)(n - 8/2)) \right]$



- Example: Consider the desired 17-sample frequency response with the first half defined by [11110000]
 - Compute the frequency sampled filter

 $h[n] = \frac{1}{17} \left[1 + 2\cos((2\pi/19)n_c) + 2\cos((4\pi/19)n_c) + 2\cos((6\pi/19)n_c) \right]$ $n_c = n - \frac{16}{2}$

- Example: Consider the desired 17-sample frequency response with the first half defined by [11110000]
 - Compute the frequency sampled filter





- Example: Consider the desired 41-sample frequency response with the first 10 values defined by 1
 - Compute the frequency sampled filter



- Example: Consider the desired 401-sample frequency response with the first 100 values defined by 1
 - Compute the frequency sampled filter
 - Note that in practice, this needs to be circularly shifted to the center



Note: The definition can be slightly modified

Our definition:

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

Final Definition

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

Side note: This is very closely related to the discrete cosine transform

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Design with Equi-ripples

Previously derived:

$$X(z) = z^{-\frac{(M-1)}{2}} \sum_{k=0}^{M/2-1} a_k \left[z^{\frac{(M-1)}{2}-k} + z^{-\left[\frac{(M-1)}{2}-k\right]} \right]$$

$$X(\omega) = e^{-j\omega} \frac{(M-1)}{2} \sum_{\substack{k=0\\M/2-1}}^{M/2-1} a_k \left[e^{j\omega \left[\frac{(M-1)}{2} - k \right]} + j\omega^{-j\omega \left[\frac{(M-1)}{2} - k \right]} \right]$$
$$= 2e^{-j\omega} \frac{(M-1)}{2} \sum_{\substack{k=0\\k=0}}^{M/2-1} a_k \cos \left(\omega \left[\frac{M-1}{2} - k \right] \right)$$

Design with Equi-ripples

Equi-ripple design $X(\omega) = 2e^{-j\omega} \frac{(M-1)}{2} \sum_{k=0}^{M/2-1} a_k \cos\left(\omega \left[\frac{M-1}{2} - k\right]\right)$

 Goal: Find the optimal a_ks that satisfies passband / stopband ripple constraints.



Design with Equi-ripples



Equals: $\frac{\delta_2}{\delta_1}$ for ω in pass band 1 for ω in stop band δ_2 = stopband ripple δ_1 = passband ripple

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Designing IIR Filters

- No easy ways to design digital IIR filters
- So let us start from analog filters

Designing IIR Filters

- No easy ways to design digital IIR filters
- So let us start from analog filters

Option 1: Preserve the difference equation!

Question: What is a derivative in discrete-time?

In continuous-time

$$\frac{dx(t)}{dt} \to sX(s)$$

In discrete-time

Question: What is a derivative in discrete-time?

In continuous-time

$$\frac{dx(t)}{dt} \to sX(s)$$

In discrete-time

$$\frac{dx(t)}{dt} = \lim_{\Delta T \to 0} \frac{x(t) - x(t - \Delta T)}{\Delta T}$$

$$\frac{dx(t)}{dt} \Big|_{t=nT} = \frac{x(nT) - x(nT - T)}{T} = x[n] - x[n - 1]$$
Question: What is a derivative in discrete-time?

In continuous-time

$$\frac{dx(t)}{dt} \to sX(s)$$

$$\frac{dx(t)}{dt} = \lim_{\Delta T \to 0} \frac{x(t) - x(t - \Delta T)}{\Delta T}$$
$$\frac{dx(t)}{dt} \bigg|_{t=nT} = \frac{x(nT) - x(nT - T)}{T} = \frac{1}{T} (x[n] - x[n - 1])$$

Question: What is a derivative in discrete-time?

In continuous-time

$$\frac{dx(t)}{dt} \to sX(s)$$

$$\frac{dx(t)}{dt} = \lim_{\Delta T \to 0} \frac{x(t) - x(t - \Delta T)}{\Delta T}$$

$$\frac{dx(t)}{dt} \bigg|_{t=nT} = \frac{x(nT) - x(nT - T)}{T} = \frac{1}{T} (x[n] - x[n - 1])$$

$$\frac{dx(t)}{dt} \bigg|_{t=nT} \to \frac{1}{T} (1 - z^{-1}) X(z)$$

Question: What is a second-derivative in discrete-time?

In continuous-time

$$\frac{d^2 x(t)}{dt^2} \to s^2 X(s)$$

$$\frac{d^2 x(t)}{dt^2} = \frac{dx(t)}{dt} \left[\frac{dx(t)}{dt} \right]$$
$$\frac{dx(t)}{dt} \bigg|_{t=nT} = \frac{x(nT) - x(nT - T)}{T}$$

Question: What is a second-derivative in discrete-time?

In continuous-time

$$\frac{d^2 x(t)}{dt^2} \to s^2 X(s)$$

$$\frac{d^2 x(t)}{dt^2} = \frac{dx(t)}{dt} \left[\frac{dx(t)}{dt} \right]$$
$$\frac{d^2 x(t)}{dt^2} \bigg|_{t=nT} = \frac{[x(nT) - x(nT - T)]/T - [x(nT - T) - x(nT - 2T)]/T}{T}$$

Question: What is a second-derivative in discrete-time?

In continuous-time

$$\frac{d^2 x(t)}{dt^2} \to s^2 X(s)$$

$$\frac{d^2 x(t)}{dt^2} = \frac{dx(t)}{dt} \left[\frac{dx(t)}{dt} \right]$$
$$\frac{d^2 x(t)}{dt^2} \bigg|_{t=nT} = \frac{x(nT) - 2x(nT - T) + x(nT - 2T)}{T^2} \to \frac{x[n] - 2x[n-1] + x[n-2]}{T^2}$$

Question: What is a second-derivative in discrete-time?

In continuous-time

$$\frac{d^2 x(t)}{dt^2} \to s^2 X(s)$$

$$\begin{aligned} \frac{d^2 x(t)}{dt^2} &= \frac{dx(t)}{dt} \left[\frac{dx(t)}{dt} \right] \\ \frac{d^2 x(t)}{dt^2} \bigg|_{t=nT} &= \frac{x(nT) - 2x(nT - T) + x(nT - 2T)}{T^2} \to \frac{x[n] - 2x[n-1] + x[n-2]}{T^2} \\ \frac{dx(t)}{dt} \bigg|_{t=nT} \to \frac{1}{T^2} (1 - 2z^{-1} + z^{-2}) X(z) = \frac{1}{T^2} (1 - z^{-1})^2 X(z) \end{aligned}$$

Question: What is a derivative in discrete-time?

Translate continuous-time to discrete-time

$$\frac{\frac{d^k x(t)}{dt^k} \to s^k X(s)}{\frac{d^k x(t)}{dt^k}} \xrightarrow{k} \frac{1}{T} (1 - z^{-1})^k X(z)$$

Question: What is a derivative in discrete-time?

Translate continuous-time to discrete-time

$$\frac{\frac{d^k x(t)}{dt^k}}{\frac{d^k x(t)}{dt^k}} \xrightarrow{} s^k X(s)$$

$$\frac{\frac{d^k x(t)}{dt^k}}{\frac{d^k x(t)}{dt^k}} \xrightarrow{} \frac{1}{T} (1 - z^{-1})^k X(z)$$

$$s \to \frac{1}{T}(1-z^{-1})$$

Example:
$$s \rightarrow \frac{1}{T}(1-z^{-1})$$

 Use the derivative conversion to transform the following biquad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

Example:
$$s \rightarrow \frac{1}{T}(1-z^{-1})$$

 Use the derivative conversion to transform the following bi-quad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

$$H(z) = \frac{1}{\left(\frac{1}{T}(1-z^{-1})+0.1\right)^2 + 9} = \frac{T^2}{T^2 \left[\left(\frac{1}{T}(1-z^{-1})+0.1\right)^2 + 9\right]}$$

$$= \frac{T^2}{\left((1-z^{-1})+0.1T\right)^2 + 9T^2} = \frac{T^2}{\left((1+0.1T)-z^{-1}\right)^2 + 9T^2}$$

Example:
$$s \rightarrow \frac{1}{T}(1-z^{-1})$$

 Use the derivative conversion to transform the following bi-quad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

$$H(z) = \frac{T^2}{((1+0.1T) - z^{-1})^2 + 9T^2}$$

$$((1+0.1T) - z^{-1})^2 + 9T^2 = 0$$

$$((1+0.1T) - z^{-1})^2 = -9T^2$$

$$(1+0.1T) - z^{-1} = \pm 3Tj$$

$$z^{-1} = (1+0.1T) \mp 3Tj$$

$$Z = \frac{1}{1+(0.1 \mp 3j)T}$$

Example:
$$s \rightarrow \frac{1}{T}(1-z^{-1})$$

 Use the derivative conversion to transform the following bi-quad filter into the discrete-time domain.

$$z = \frac{1}{1 + (0.1 + 3j)T}$$
 Poles

• Example:
$$s \to \frac{1}{T} (1 - z^{-1})$$

 $z = \frac{1}{1 + (0.1 + 3j)T}$



Question: What is a derivative in discrete-time?

Translate continuous-time to discrete-time

$$s \to \frac{1}{T}(1-z^{-1})$$

Pros:

Relatively simple

Cons:

- Very limiting
- Stable continuous-time poles can only be mapped to low frequencies



Lecture 22: Design of FIR / IIR Filters

Foundations of Digital Signal Processing

Outline

- Designing FIR Filters with Windows
- Designing FIR Filters with Frequency Sampling
- Designing FIR Filters with Equi-ripples
- Designing IIR Filters with Discrete Differentiation
- Designing IIR Filters with Impulse Invariance
- Designing IIR Filters with the Bilinear Transform
- Related Analog Filters

Designing IIR Filters

- No easy ways to design digital IIR filters
- So let us start from analog filters

Option 2: Preserve the impulse response!

$$H(s) = \prod_{k=1}^{K} \frac{1}{s - p_k}$$

$$H(s) = \prod_{k=1}^{K} \frac{1}{s - p_k} = \sum_{k=1}^{K} c_k e^{p_k t}$$

$$H(s) = \prod_{k=1}^{K} \frac{1}{s - p_k} = \sum_{k=1}^{K} c_k e^{p_k t}$$

$$h(t) = \sum_{k=1}^{K} c_k e^{p_k t}$$

$$H(s) = \prod_{k=1}^{K} \frac{1}{s - p_k} = \sum_{k=1}^{K} c_k e^{p_k t}$$

$$h(t) = \sum_{k=1}^{K} c_k e^{p_k t}$$

$$h(nT) = h[n] = \sum_{k=1}^{K} c_k e^{p_k nT} = \sum_{k=1}^{K} c_k [e^{p_k T}]^n$$

$$H(s) = \prod_{k=1}^{K} \frac{1}{s - p_k} = \sum_{k=1}^{K} c_k e^{p_k t}$$



• Example:
$$H(z) = \sum_{k=1}^{K} \frac{c_k}{1 - e^{p_k T_z - 1}}$$

 Use impulse invariance to transform the following biquad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

• Example:
$$H(z) = \sum_{k=1}^{K} \frac{c_k}{1 - e^{p_k T_z - 1}}$$

 Use impulse invariance to transform the following biquad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

Poles:

$$(s + 0.1)^2 + 9 = 0$$

 $s = \pm 3j - 0.1$

• Example:
$$H(z) = \sum_{k=1}^{K} \frac{c_k}{1 - e^{p_k T_z - 1}}$$

 Use impulse invariance to transform the following biquad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9} = \frac{1/2}{s+3j+0.1} + \frac{1/2}{s-3j+0.1}$$

Poles:

$$(s + 0.1)^2 + 9 = 0$$

 $s = \pm 3j - 0.1$

• Example:
$$H(z) = \sum_{k=1}^{K} \frac{c_k}{1 - e^{p_k T_z - 1}}$$

 Use impulse invariance to transform the following biquad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9} = \frac{1/2}{s+3j+0.1} + \frac{1/2}{s-3j+0.1}$$
$$H(z) = \frac{1/2}{1 - e^{(-3j-0.1)T}z^{-1}} + \frac{1/2}{1 - e^{(3j-0.1)T}z^{-1}}$$

Example:
$$H(z) = \sum_{k=1}^{K} \frac{c_k}{1 - e^{p_k T} z^{-1}}$$

 $H(z) = \frac{1/2}{1 - e^{(-3j - 0.1)T} z^{-1}}$
 $+ \frac{1/2}{1 - e^{(3j - 0.1)T} z^{-1}}$



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Option 3: Preserve the definition of z!

Question: How are z and s related?

From continuous-time to discrete-time

$$s = j\Omega$$

 $e^{st} = e^{snT} = z^n$
 $z = e^{sT}$
Taylor Series
Expansion / Approximation

Building an approximation ($e^x \approx 1 + x$)

$$z = \frac{e^{\frac{sT}{2}}}{e^{-\frac{sT}{2}}} \approx \frac{1 + sT/2}{1 - sT/2}$$

Question: How are z and s related?

From continuous-time to discrete-time

$$s = j\Omega$$

$$e^{st} = e^{snT} = z^n$$

$$z = e^{sT} \qquad s = \frac{1}{T}\ln(z)$$

Building an approximation

$$s \approx \frac{2}{T} \frac{z-1}{z+1}$$

Bilinear Expansion / Approximation

The Bilinear Transform

Continuous-time to discrete-time

$$s \to \frac{2}{T} \frac{1 - z^{-1}}{1 + z^{-1}}$$

Discrete-time to continuous-time

$$z \to \frac{1 + sT/2}{1 - sT/2}$$

• Example:
$$s \to \frac{2}{T} \frac{1-z^{-1}}{1+z^{-1}}, \quad z \to \frac{1+sT/2}{1-sT/2}$$

 Use the bilinear transform to transform the following biquad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

• Example:
$$s \to \frac{2}{T} \frac{1-z^{-1}}{1+z^{-1}}, \quad z \to \frac{1+sT/2}{1-sT/2}$$

 Use the bilinear transform to transform the following biquad filter into the discrete-time domain.

$$H(s) = \frac{1}{(s+0.1)^2 + 9}$$

$$H(z) = \frac{1}{\left(\frac{2}{T}\frac{1-z^{-1}}{1+z^{-1}} + 0.1\right)^2 + 9}$$

$$= \frac{(1+z^{-1})^2}{\left(\frac{2}{T}(1-z^{-1}) + 0.1\right)^2 + 9(1+z^{-1})^2}$$

Example:
$$s \to \frac{2}{T} \frac{1-z^{-1}}{1+z^{-1}}, \quad z \to \frac{1+sT/2}{1-sT/2}$$

$$H(z) = \frac{(1+z^{-1})^2}{\left(\frac{2}{T}(1-z^{-1})+0.1\right)^2 + 9(1+z^{-1})^2}$$



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Multi-pole Filters

- **Butterworth:** Maximally flat passband
- **Chebyshev:** Faster cutoff with passband ripple
- Elliptic: Fastest cutoff with passband and stopband ripple



Foundations of Digital Signal Processing Lecture 22: Designing FIR / IIR Filters
Butterworth Filter

Butterworth Filter of order N

$$|H(j\omega)| = \frac{1}{\sum_{k=1}^{N} (s - s_k)}$$
 $s_k = e^{\frac{j(2k+N-1)\pi}{2N}}$

Butterworth Filter

Butterworth Filter of order N

 N equally spaced poles on a circle on the left-hand-side of the splane



Foundations of Digital Signal Processing Lecture 22: Designing FIR / IIR Filters

Butterworth Filter

Properties of the Butterworth Filter

- It is maximally flat at $\omega = 0$
- It has a cutoff frequency $|H(\omega)| = \frac{1}{\sqrt{2}}$ at $\omega = \omega_c$
- For large *n*, it becomes an ideal filter

Chebyshev Filter

Chebyshev Filter of order N

$$|H(j\omega)| = \frac{1}{\sqrt{1+\epsilon^2 C_n^2(\omega)}}$$

 $C_n^2(\omega)$ is an nth-order Chebyshev Polynomial

 ϵ^2 controls ripple

Chebyshev Filter

Chebyshev Filter of order N



Foundations of Digital Signal Processing Lecture 22: Designing FIR / IIR Filters

Chebyshev Filter

Properties of the Chebyshev Filter

- It has ripples in the passband and is smooth in the stopband.
- The ratio between the maximum and minimum ripples in the passband is

$$(1 + \epsilon^2) - 1/2$$

- If e is reduced (i.e., the ripple size is reduced), then the stopband attenuation is reduced.
- It has a sharper cut-off than a Butterworth filter, but at the expense of passband rippling

Elliptic Filter

Elliptic Filter of order N

$$|H(j\omega)| = \frac{1}{\sqrt{1 + \epsilon^2 R_n^2(\omega)}}$$

 $R_n^2(\omega)$ is an nth-order elliptic function

 ϵ^2 controls ripple

Elliptic Filter

Elliptic Filter of order N



Foundations of Digital Signal Processing Lecture 22: Designing FIR / IIR Filters

Elliptic Filter

Properties of the Elliptic Filter

- It has ripples in the passband and the stopband
- The ratio between the maximum and minimum ripples is larger than the Chebyshev filter, but it has an even quicker transition from passband to stopband
- It has poles and zeros, but they are much more difficult to compute compared with the Butterworth and Chebyshev filters