

TTT4120 Digital Signal Processing Fall 2017

Lecture: Filter Properties and Inverse Systems

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Lecture in course book*

- Proakis, Manolakis Digital Signal Processing, 4th Ed.
 - 5.4.2 Lowpass, highpass, and bandpass filters
 - 5.4.3 Digital resonators
 - 5.4.4 Notch filters
 - 5.4.5 Comb filters
 - 5.4.6 All-pass filters
 - 5.4.1 Ideal filter characteristics
 - 10.2.1 Symmetric and antisymmetric FIR filters
 - 5.5 Inverse systems and deconvolution

*Level of detail is defined by lectures and problem sets

Contents and learning outcomes

- Some simple filter properties
- Why linear phase?
- Minimum-phase and inverse systems

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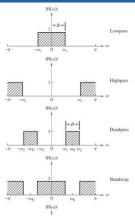
Ideal filter characteristics

• Systems may be designed to amplify or attenuate parts of the input signal (e.g., remove noise, suppress interference)

$$x[n]$$
 $y[n] = h[n] * x[n]$ $Y(\omega) = H(\omega)X(\omega)$ $Y(z) = H(z)X(z)$

• Frequency response $H(\omega)$ shapes the spectrum of the input signal to have a desired form

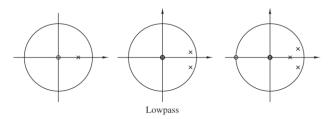
Linear time-invariant systems...



- Passband, stopband, cutoff frequencies
- Cannot get this kind of shapes using a causal impulse response with a finite number of coefficients (later)

Lowpass

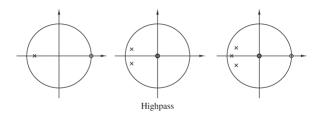
• Poles close(r) to z = 1 and zeros close(r) to z = -1. Why?



• Example: $H(z) = \frac{1-a}{2} \cdot \frac{1+z^{-1}}{1-az^{-1}}$

Highpass

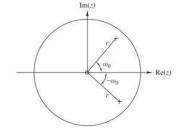
• Poles close(r) to z = -1 and zeros close(r) to z = 0

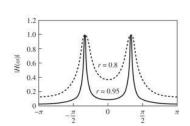


- · Reflect poles-zeros of lowpass around imaginary axis
- Frequency translation: $H_{\rm hp}(\omega) = H_{\rm lp}(\omega \pi)$
- Example: $H_{lp}(z) = \frac{1-a}{2} \cdot \frac{1+z^{-1}}{1-az^{-1}} \to H_{hp}(z) = \frac{1-a}{2} \cdot \frac{1-z^{-1}}{1+az^{-1}}$

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Digital resonator



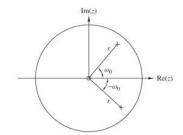


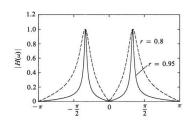
• Complex-conjugate poles $p_{1,2} = re^{\pm j\omega_0}$ close to |z| = 1

$$H(z) = \frac{b_0}{(1 - re^{j\omega_0}z^{-1})(1 - re^{-j\omega_0}z^{-1})}$$

- Resonant peak can be computed: $\omega_r = \cos^{-1}\left(\frac{1+r^2}{2r}\cos\omega_0\right)$
- For $r \approx 1$, $\omega_r \approx \omega_0$

Digital resonator...





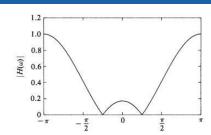
• Complex-conjugate poles $p_{1,2}=re^{\pm j\omega_0}$ and zeros $z_{1,2}=\pm 1$

$$H(z) = \frac{(1+z^{-1})(1-z^{-1})}{(1-re^{j\omega_0}z^{-1})(1-re^{-j\omega_0}z^{-1})}$$

· Exact location of resonant peak harder to find analytically

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Notch filter

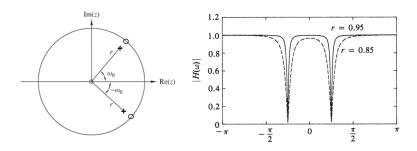


- · A filter that contains deep notches in its frequency response
- Removing powerline frequency disturbance
- Create nulls by complex-conjugate zeros on the unit circle

$$H(z) = b_0 \big(1 - e^{j\omega_0} z^{-1} \big) \big(1 - e^{-j\omega_0} z^{-1} \big)$$

• Large bandwidth is a problem with FIR notch filters

Notch filter...

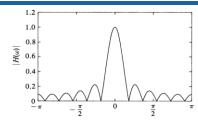


• Introduce poles close to unit circle reduces notch bandwidth

$$H(z) = \frac{b_0(1 - e^{j\omega_0}z^{-1})(1 - e^{-j\omega_0}z^{-1})}{(1 - re^{j\omega_0}z^{-1})(1 - re^{-j\omega_0}z^{-1})}$$

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Comb filter



- · Notch filter with nulls periodically spaced across frequency
- Simple moving average (FIR) filter

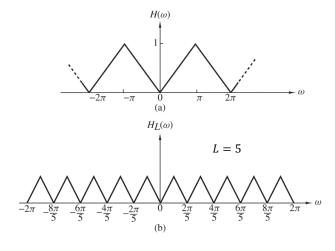
$$H(z) = \sum_{k=0}^{M} z^{-k} = \frac{1 - z^{-(M+1)}}{1 - z^{-1}}$$

• We may also construct a comb filter by replacing z with z^L

$$H_L(z) = \textstyle \sum_{k=0}^M h[k] z^{-Lk} \Longleftrightarrow H_L(\omega) = H(L\omega)$$

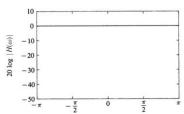
Comb filter...

$$H_L(z) = \sum_{k=0}^{M} h[k] z^{-Lk} \iff H_L(\omega) = H(L\omega)$$



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All-pass filters



- · All-pass filter has constant magnitude response
- Can be used to compensate poor phase characteristics

$$H(z) = \frac{a_N + a_{N-1}z^{-1} + \dots + a_1z^{-N+1} + z^{-N}}{1 + a_1z^{-1} + \dots + a_{N-1}z^{-N+1} + a_Nz^{-N}} = \frac{z^{-N}A(z^{-1})}{A(z)}$$

Assuming real coefficients

$$H(z) = \frac{z^{-N}A(z^{-1})}{A(z)} \iff |H(\omega)|^2 = H(z)H(z^{-1})|_{z=e^{j\omega}} = 1$$

Linear phase filters...

• Why linear phase filters, i.e., $\angle H(\omega) = a + b\omega$?

$$H(\omega) = |H(\omega)|e^{j \not = H(\omega)}$$

Compare the two ideal lowpass specifications

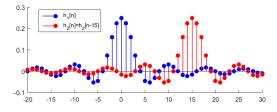
$$H_1(\omega) = \begin{cases} 1, & |\omega| \le \omega_c \\ 0, & |\omega| \le \omega_c \le \pi \end{cases}$$

$$H_2(\omega) = \begin{cases} e^{-jn_d\omega}, & |\omega| \le \omega_c \\ 0, & |\omega| \le \omega_c \le \pi \end{cases}$$

• How about the time-domain pulses?

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Linear phase filters...



Matlab n = -20:30; wc = 2*pi*1/8; x = wc/pi; nd = 15; stem(n,x*sinc(x*n),'r') hold on stem(n,x*sinc(x*(n-nd)))

How about the time-domain pulses?

$$\begin{split} h_{1[n]} &= \frac{w_c}{\pi} \frac{\sin[w_c n]}{w_c n} \\ h_{2}[n] &= \frac{w_c}{\pi} \frac{\sin[w_c (n - n_d)]}{[w_c (n - n_d)]} \Longrightarrow \ h_{2}[n] = h_{1}[n - n_d] \end{split}$$

• Delays the output signal with n_d samples, no signal distortion!

Linear phase filters...

• Filter design in general (later in the course):

$$\min_{a,b} ||E(z)|| = \min_{a,b} \left\| H_{\text{des}}(z) - \frac{\sum_{k=0}^{M-1} b_k z^{-k}}{\sum_{k=0}^{N-1} a_k z^{-k}} \right\|$$

• Consider FIR filters having a frequency response of the form

$$H(\omega) = H_{\rm r}(\omega)e^{-j(\omega d + c)}, H_{\rm r}(\omega)$$
 real-valued

- We want a pure signal delay in passband
- Obtained by choosing h[k] real and $h[k] = \pm h[M-1-k]$
 - Symmetric or antisymmetric

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Linear phase filters...

• Example: FIR with $M = 5 \Rightarrow N = 2$

$$H(z) = h[0] + h[1]z^{-1} + h[2]z^{-2} + h[3]z^{-3} + h[4]z^{-4}$$

$$= z^{-2} \{ h[0]z^{2} + h[1]z + h[2] \pm h[1]z^{-1} \pm h[0]z^{-2} \}$$

$$= z^{-2} \{ h[2] + h[0][z^{2} \pm z^{-2}] + h[1][z^{1} \pm z^{-1}] \}$$

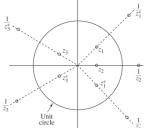
• Frequency response symmetric filter (take the '+' signs):

$$H(z)\Big|_{e^{j\omega}} = e^{-j2\omega} \{h[2] + 2h[0]\cos 2\omega + 2h[1]\cos \omega\}$$

• Frequency response antisymmetric filter (take the '-' signs):

$$H(z)\Big|_{e^{j\omega}} = je^{-j2\omega} \{2h[0]\sin 2\omega + 2h[1]\sin \omega\}$$

Linear phase filters...



- Zeros of H(z) occur in reciprocal pairs
- Example (cont.): Symmetric FIR with M = 5 (N = 2)

$$\begin{split} H(z) &= h[0] + h[1]z^{-1} + h[2]z^{-2} + h[3]z^{-3} + h[4]z^{-4} \\ &= z^{-2}\{h[0]z^2 + h[1]z + h[2] + h[1]z^{-1} + h[0]z^{-2}\} \\ &= z^{-2}\{h[2] + h[0][z^2 + z^{-2}] + h[1][z^1 + z^{-1}]\} \\ &= z^{-4}H(z^{-1}) \end{split}$$

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Inverse and minimum-phase systems

$$x[n]$$
 $X(z)$
 $y[n] = h[n] * x[n]$
 $Y(z) = H(z)X(z)$

- What if we are given y[n] and want to determine x[n]?
 - Information signal passing through communication channel

Inverse and minimum-phase systems

$$x[n] \longrightarrow h[n] \qquad y[n] \longrightarrow h_I[n] \qquad y_I[n] = (h_I[n] * h[n]) * x[n]$$

$$X(z) \longrightarrow H_I(z)H(z)X(z)$$

• If system \mathcal{T} is invertible, x[n] and can be recovered from y[n]

$$x[n] = \mathcal{T}^{-1} \{y[n]\} = \mathcal{T}^{-1} \{\mathcal{T}[x[n]]\}$$

· Linear time-invariant systems

$$h[n] * h_I[n] = \delta[n] \stackrel{Z}{\leftrightarrow} H(z)H_I(z) = 1$$

• Solving for $h_I[n]$ usually simpler in z-domain, especially if H(z) is rational, i.e., H(z) = B(z)/A(z)

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Inverse and minimum-phase systems...

- Example: Determine inverse system $h[n] = \delta[n] \frac{1}{2}\delta[n-1]$
- Time-domain solution ($h_I[n]$ causal and stable)

$$h[n] * h_I[n] = \delta[n] \Leftrightarrow \sum_{k=0}^n h[k] h_I[n-k] = \delta[n]$$

• Solution via z-transform

$$H(z)H_I(z) = 1 \Leftrightarrow H_I(z) = 1/H(z)$$

Remember to specify ROC of $H_I(z)$ (two possibilities)!

Inverse and minimum-phase systems...

• Solution via z-transform

$$H(z)H_I(z) = 1 \Leftrightarrow H_I(z) = 1/H(z)$$

Remember to specify ROC of $H_I(z)$ (two possibilities)!

$$H(z) = 1 - \frac{1}{3}z^{-1}$$
 ROC: $|z| \neq 0 \rightarrow H_I(z) = \frac{1}{1 - \frac{1}{3}z^{-1}}$

Corresponds to either

$$h_I[n] = -\left(\frac{1}{3}\right)^n u[-n-1]$$
, ROC: $|z| < \frac{1}{3}$ (anti-causal unstable) or $h_I[n] = \left(\frac{1}{3}\right)^n u[n]$, ROC: $|z| > \frac{1}{3}$ (causal stable)

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Inverse and minimum-phase systems...

- In general we have that if H(z) is stable and causal then poles $|p_k| < 1 \ \forall k$ and ROC: $|z| > \max_k |p_k|$
 - \Rightarrow There exists a stable and causal inverse $H_I(z) = 1/H(z)$ if zeros of H(z) are within the unit circle, i.e., $|z_k| < 1 \,\forall k$
- Definition: A system is called minimum-phase if all zeros and poles are inside the unit circle
 - ⇒ a stable pole-zero system that is minimum phase has a stable inverse that is also minimum phase

Summary

Today:

- Some simple filter types and their properties
- Linear phase systems
- Inverse and minimum-phase systems

Next:

• Correlation and energy spectrum density