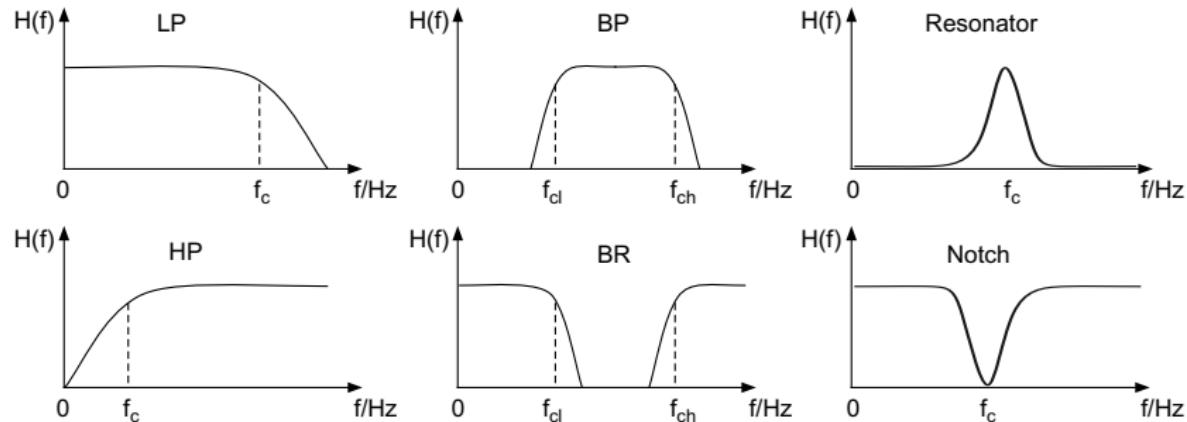


E85.2607: Lecture 2 – Filters

1 Basic IIR filters

2 Applications

Basic filters



f_c cutoff or center frequency

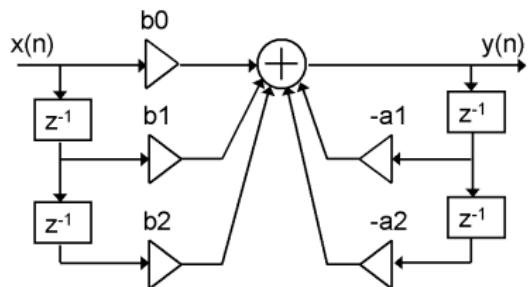
f_b bandwidth

Q “quality factor” $Q = \frac{f_b}{f_c}$

IIR filter structures: Digital biquad

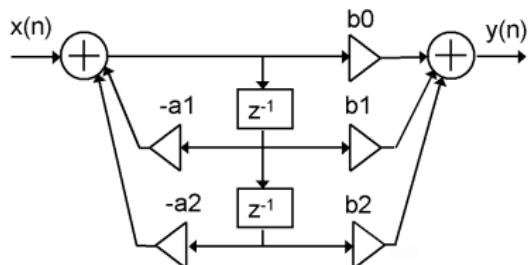
$$H(z) = \frac{B(z)}{A(z)} = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

Direct form 1



zeros on left, poles on right

Direct form 2



canonical (minimum delays)

Building blocks: Allpass filter

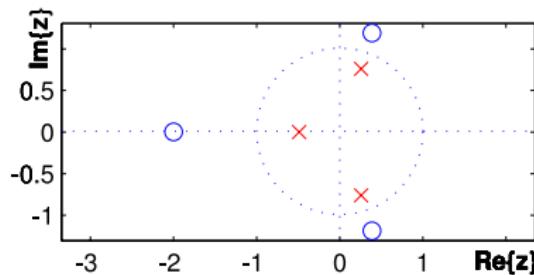
$$\begin{aligned} H(z) &= \frac{B(z)}{z^{-N}B(z^{-1})} \\ &= \frac{b_0 + b_1 z^{-1} + \dots + b_{N-1} z^{N-1} + b_N z^{-N}}{b_N + b_{N-1} z^{-1} + \dots + b_1 z^{N-1} + b_0 z^N} \end{aligned}$$

Where are the poles and zeros?

Building blocks: Allpass filter

$$\begin{aligned} H(z) &= \frac{B(z)}{z^{-N}B(z^{-1})} \\ &= \frac{b_0 + b_1 z^{-1} + \dots + b_{N-1} z^{N-1} + b_N z^{-N}}{b_N + b_{N-1} z^{-1} + \dots + b_1 z^{N-1} + b_0 z^N} \end{aligned}$$

Where are the poles and zeros?



What about the frequency response?

Parametric first order allpass filter

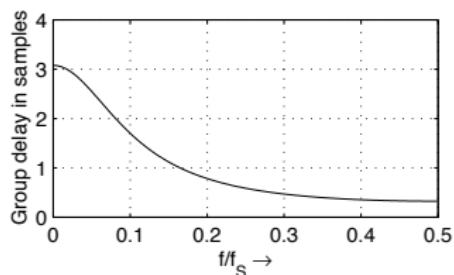
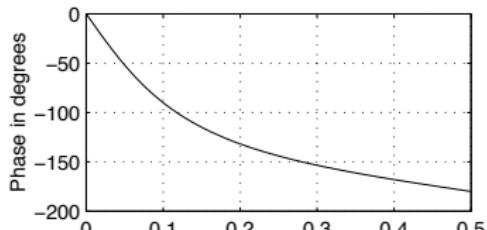
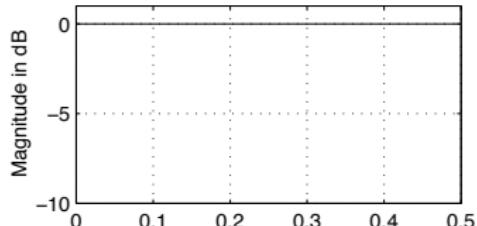
$$A(z) = \frac{c + z^{-1}}{1 + c z^{-1}}$$

$$c = \frac{\tan(\pi f_c/f_s) - 1}{\tan(\pi f_c/f_s) + 1}$$

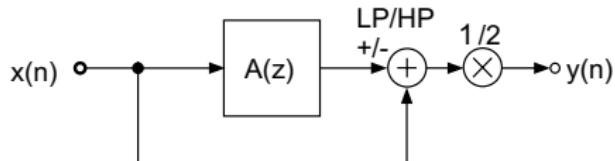
Flat magnitude response, but introduces phase distortion

$$\text{Group delay} = -\frac{\partial}{\partial \omega} \angle H(e^{j\omega})$$

Magnitude Response, Phase Response, Group Delay



Tunable lowpass/highpass filters



$$H(z) = \frac{1}{2}(1 \pm A(z)) \quad (\text{LP/HP } +/-)$$

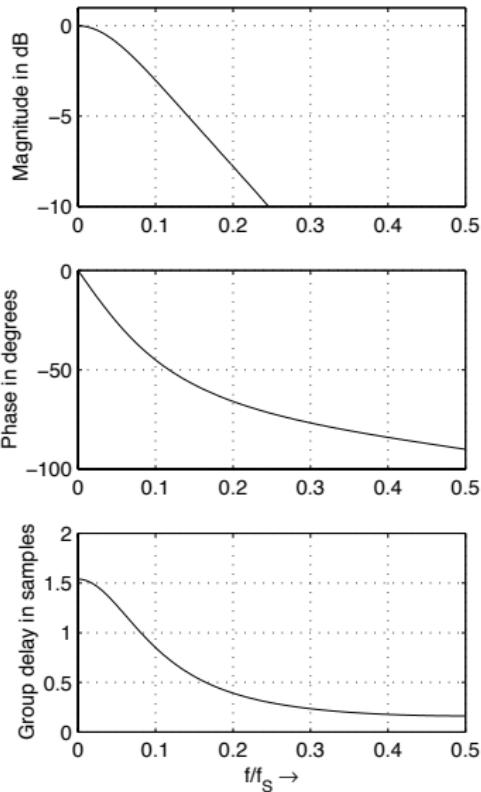
$$A(z) = \frac{z^{-1} + c}{1 + cz^{-1}}$$

$$c = \frac{\tan(\pi f_c/f_s) - 1}{\tan(\pi f_c/f_s) + 1},$$

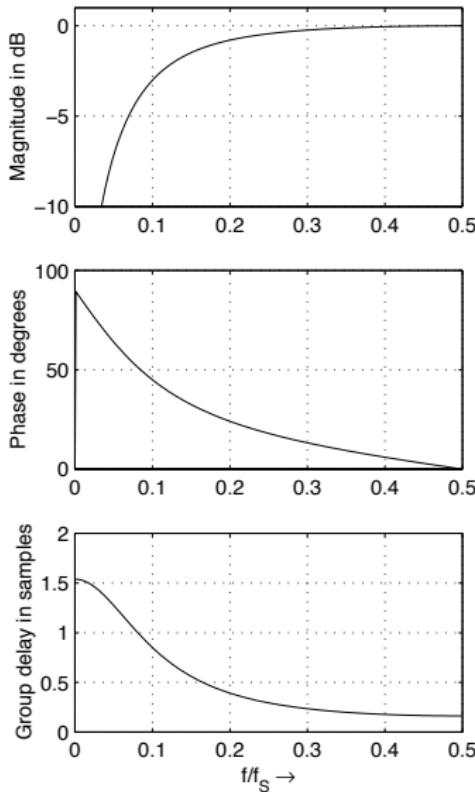
Why does this work?

LP/HP frequency response

Magnitude Response, Phase Response, Group Delay



Magnitude Response, Phase Response, Group Delay

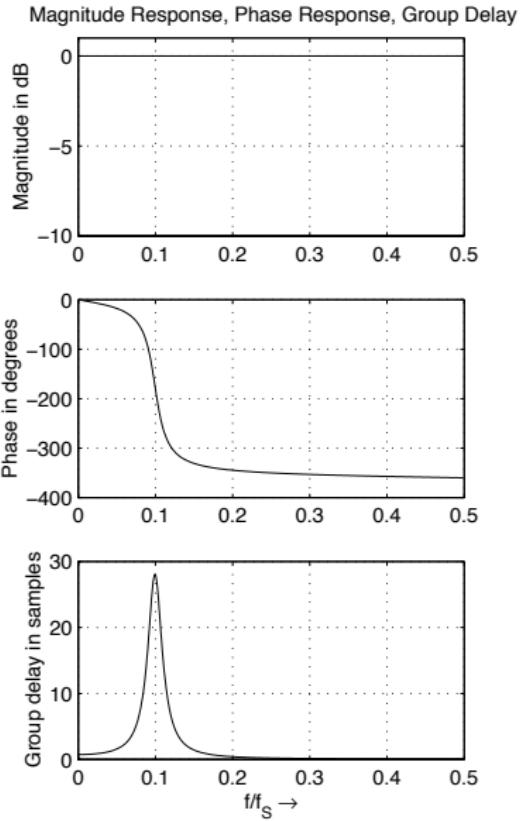


Second order allpass

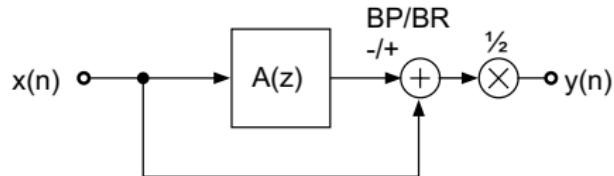
$$\begin{aligned} A(z) &= \frac{-c + d(1 - c)z^{-1} + z^{-2}}{1 + d(1 - c)z^{-1} - cz^{-2}} \\ c &= \frac{\tan(\pi f_b/f_s) - 1}{\tan(\pi f_b/f_s) + 1} \\ d &= -\cos(2\pi f_c/f_s). \end{aligned}$$

c controls bandwidth

d controls cut-off



Tunable bandpass/bandreject filters



$$H(z) = \frac{1}{2} [1 \mp A(z)] \quad (\text{BP/BR } -/+)$$

$$A(z) = \frac{-c + d(1 - c)z^{-1} + z^{-2}}{1 + d(1 - c)z^{-1} - cz^{-2}}$$

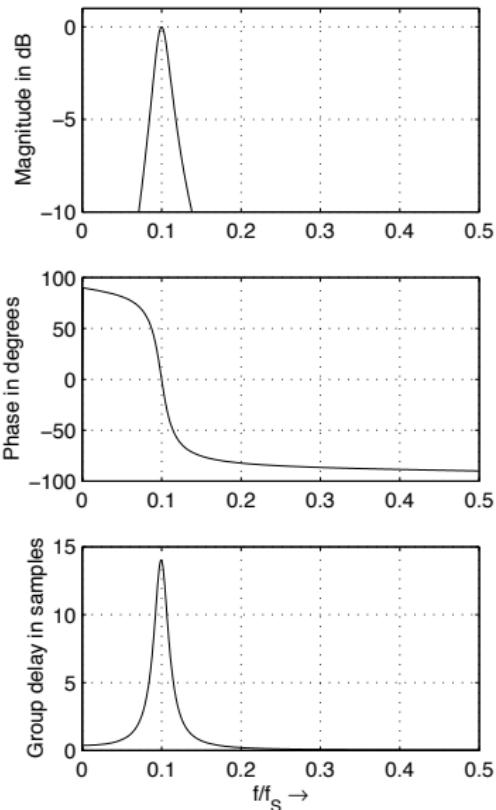
$$c = \frac{\tan(\pi f_b/f_s) - 1}{\tan(2\pi f_b/f_s) + 1}$$

$$d = -\cos(2\pi f_c/f_s),$$

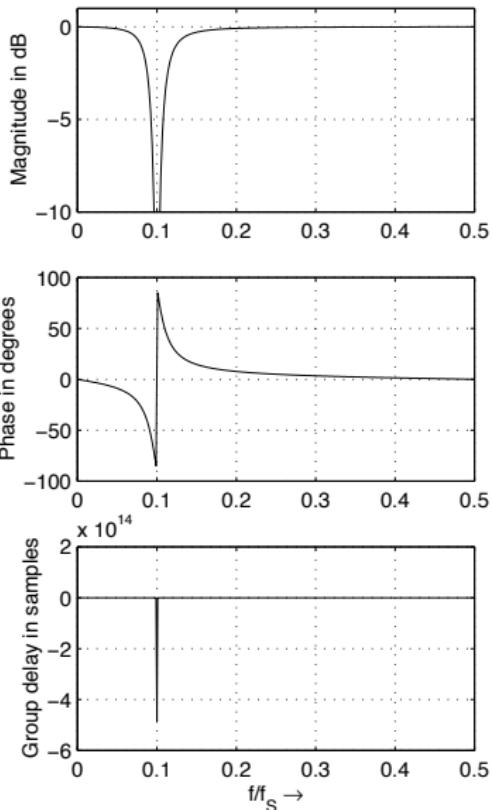
typo

BP/BR frequency response

Magnitude Response, Phase Response, Group Delay

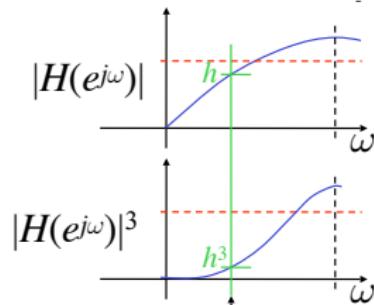
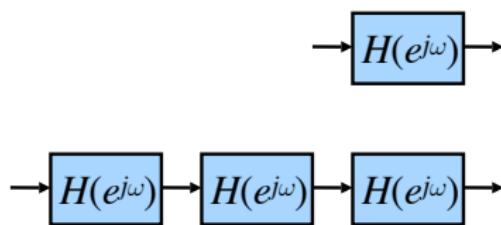


Magnitude Response, Phase Response, Group Delay

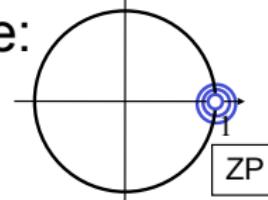


Cascading filters

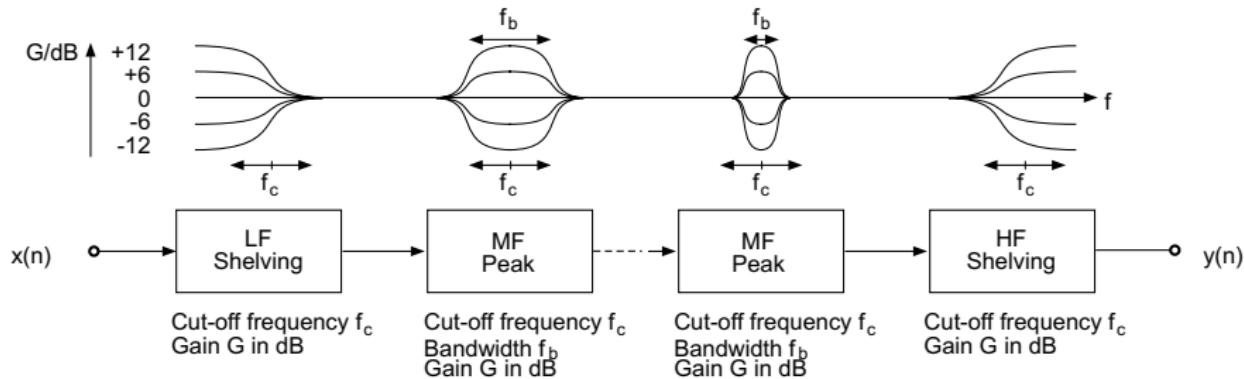
Make the frequency response sharper by passing the signal through the same filter multiple times



- Repeated roots in z-plane:

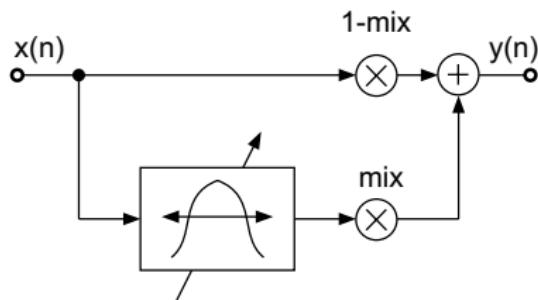


Equalizers



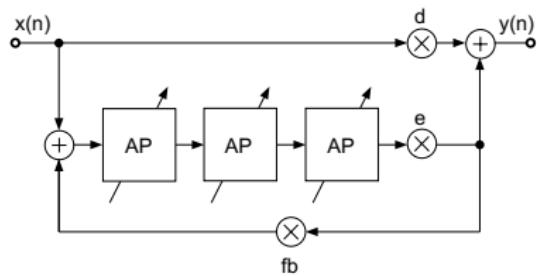
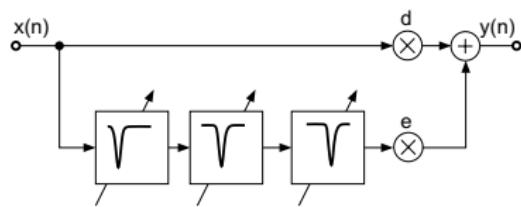
- Chain of simple filters to shape spectrum
- Low/high shelf $H(z) = 1 + (10^{G/20} - 1) H_{LP/HP}(z)$
- Peak $H(z) = 1 + (10^{G/20} - 1) H_{BP}(z)$
- Parameters: Gain, center frequency, bandwidth (Q)
- Applications: mixing, compensate for room acoustics, genre controls

Time-varying filters: Wah



- Bandpass filter with time varying center frequency
- Mimics formant resonances in speech

Time-varying filters: Phaser



- Notches with time varying center frequency
- Controlled by a low frequency oscillator

Reading

- *Introduction to Digital Filters*
 - Elementary Audio Digital Filters
- *DAFX*, Chapter 2 (if you have it)