# TECH 350: DSP

## Class V: Describing/Designing Filters, Algorithmic Reverb Design

# Categorizing Filters

## How can we categorize filters?

***First***, by their **frequency response**

(magnitude and phase response)

***Second***, by **how they are implemented**

(underlying functions, order, etc.)

***Third***, by **contextualizing them vis-a-vis filter parameters**/characteristics

(phase response, ripple, etc.)

# An Aside: The Psychoacoustics of Time Delay

## What effect do different delay lengths have on perception?

Precedence (or Haas) effect: <40ms delay time, copies are “fused”

Digital filters: microscopic delays, measured in samples (~0.02ms for 44.1kHz $F_S$)

(alter perceived timbre of sounds)

Modulated Delays: Phaser < Flanger < Chorus: 1-100ms+ delays

Echo / Algorithmic Reverb: 25-1000ms+ delays

# Categorizing Filters: Frequency Response

**Echo (delay > 40ms)**

**Low-pass / High-pass (inverses)**

**Band-pass / Band-stop (inverses)**

**Peak (or band, or bell)**

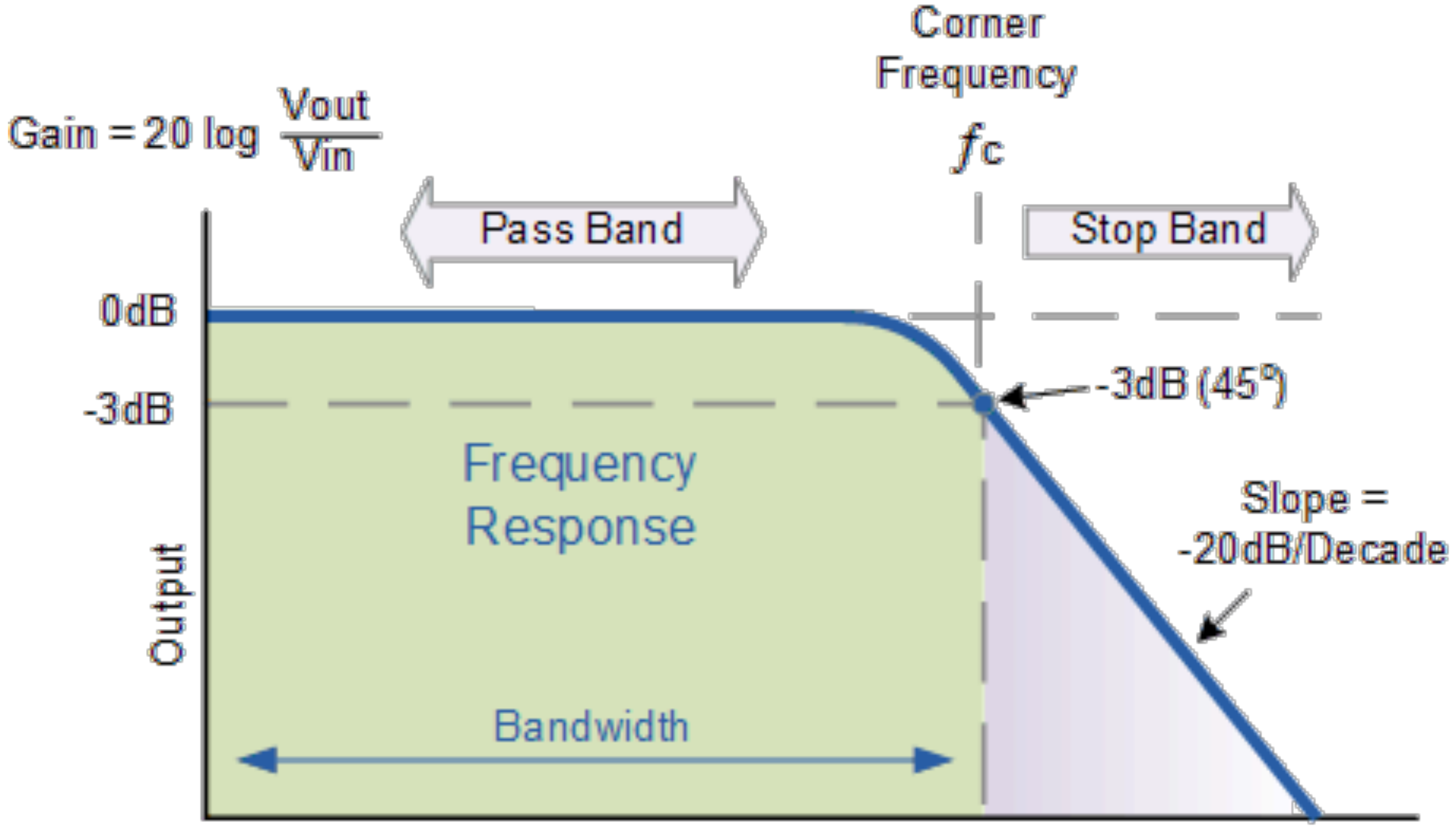
**Comb**

**All-pass**

# The Parameters of Filters

## Intuitive Parameters:

***Cutoff (or Corner) Frequency*** - Boundary in a filter's frequency response determined by a -3 dB change in magnitude response relative to a peak passband value




***Slope*** - rate at which a filter's magnitude response attenuates frequencies, measured in dB/octave or /decade (decade = ten-fold increase in frequency)

***Resonance (also Q Factor)*** - boosting (above unity) magnitude response around the cutoff frequency. Some filters have (resonant), others do not.

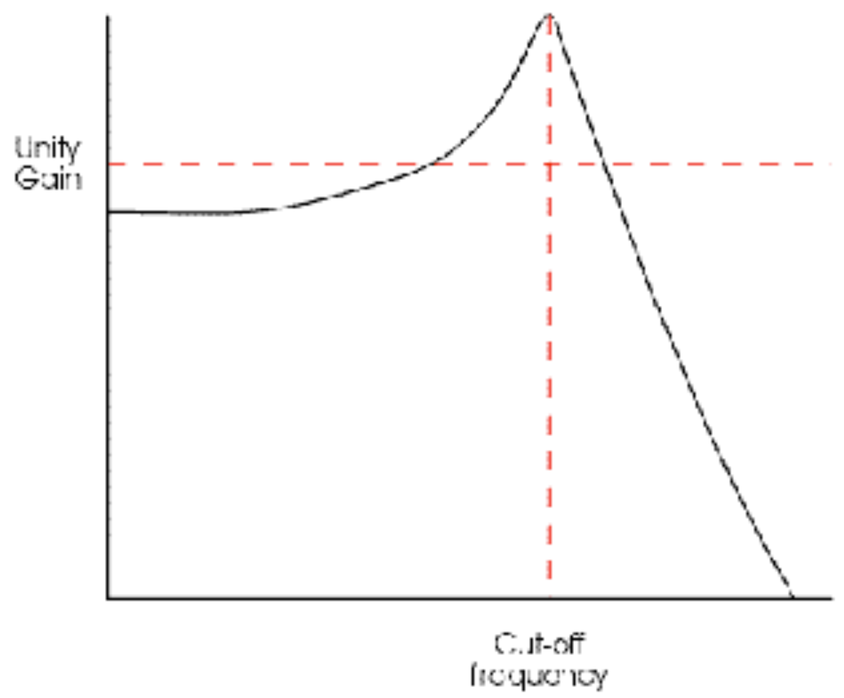



**Figure 13:** A typical resonant low-pass filter response.

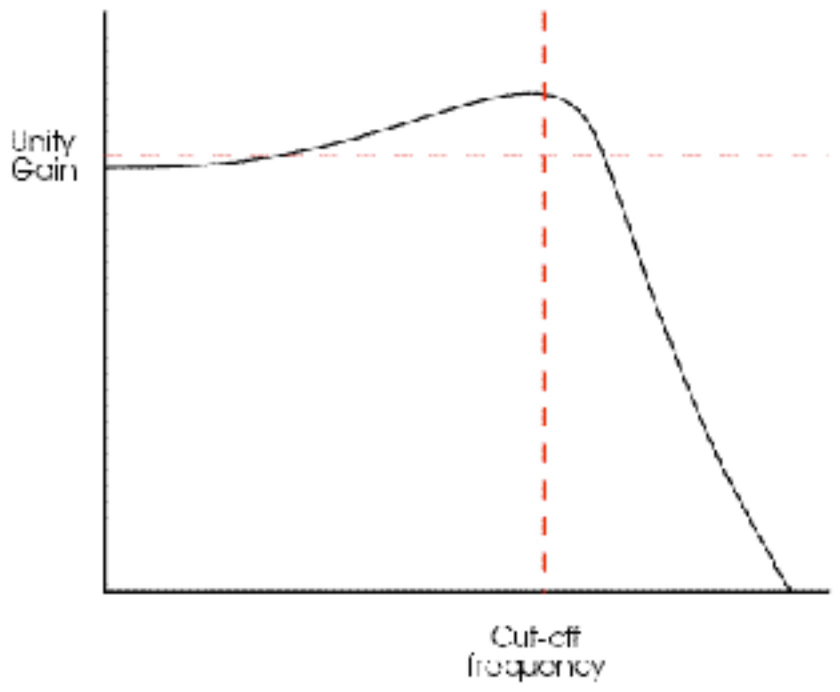



**Figure 14:** Resonant low-pass filter with low Q.

# The Parameters of Filters

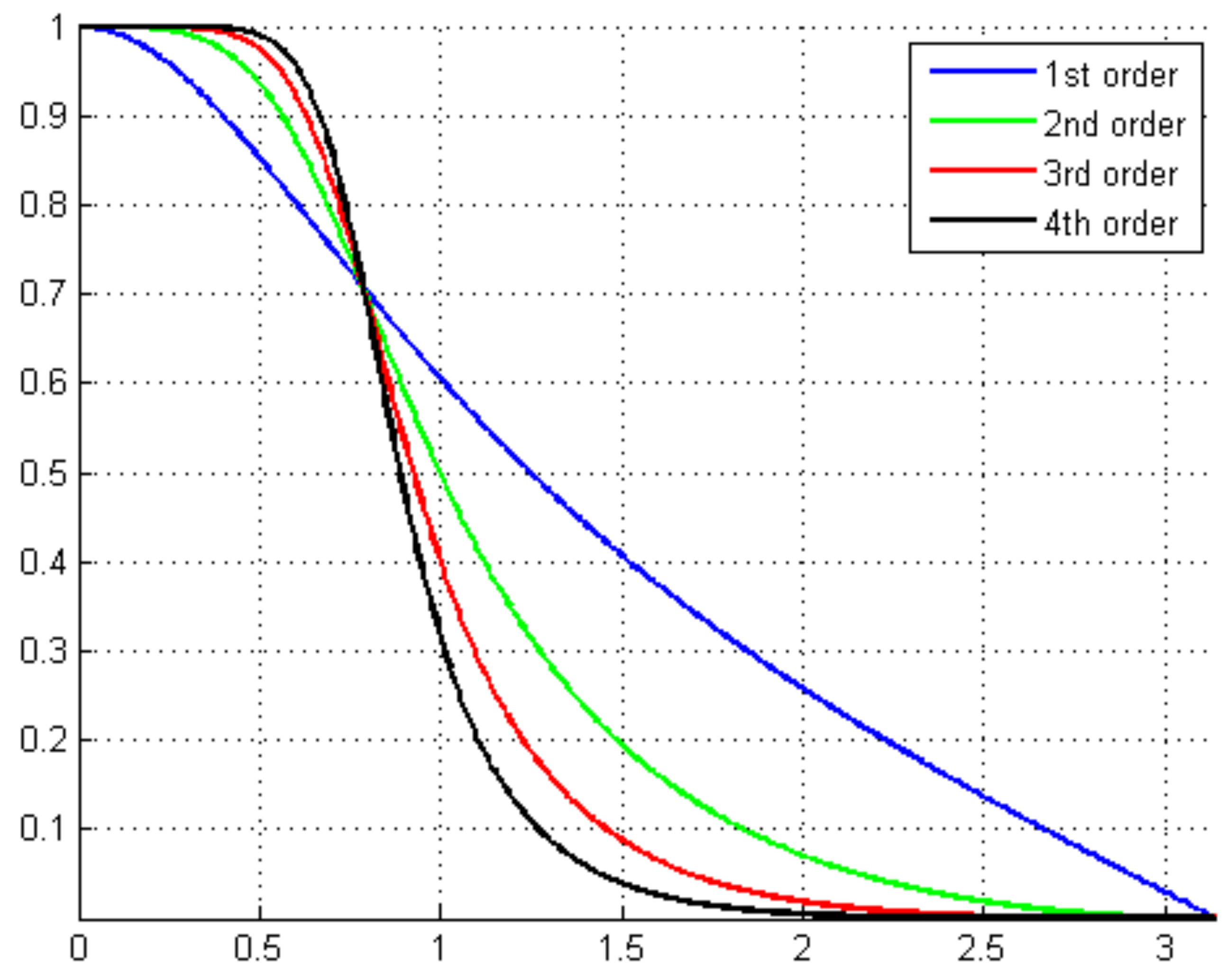
## Descriptive/Behind-the-Scenes Parameters:

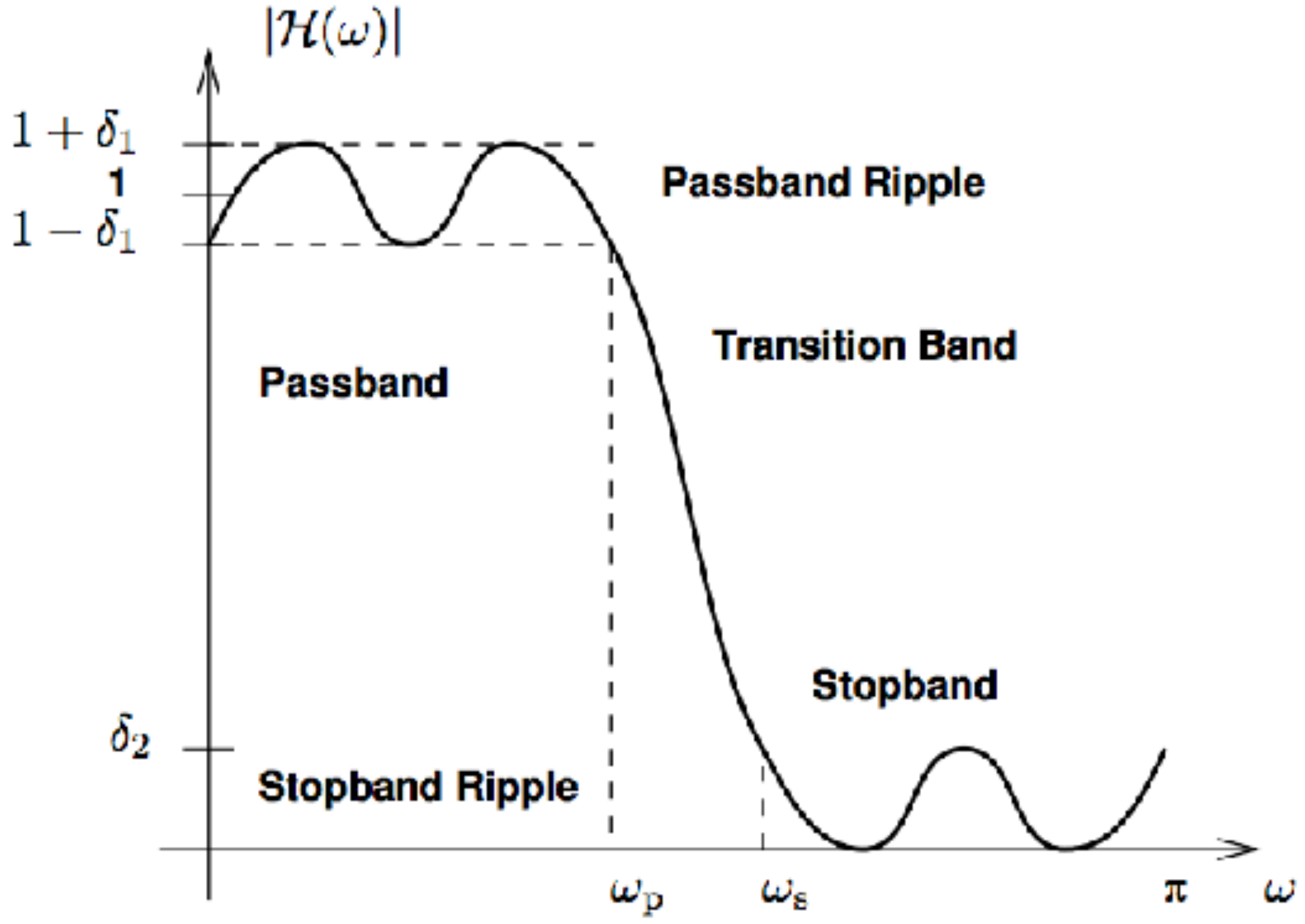
***Order*** - a number describing the highest sample delay in a filter, often corresponding to the steepness of a filter's slope

***Coefficients*** - multipliers within a filter's difference equations, controlling the amount of feedback and feedforward of delayed versions of the signal

***Transition band*** - frequency range between the passband and the stopband, measured in Hz

***Ripple*** - fluctuations, or variations, in the magnitude response within the passband of a filter, measured in percent (%)



# Designing Filters from the Top-Down

***Ideally, we want to be able to design a filter by how it attenuates certain frequencies (magnitude response) and then be given its different equation coefficients, rather than the other way around…***

***In other words, we want to be able to design a filter with intuitive parameters, and then resolve the correct behind-the-scenes parameters to match those intuitive parameters, e.g.:***

filtertype = lowpass

frequency = 400Hz

Q factor = 6

→

b0 = 0.00068

b1 = 0.00136

b2 = 0.00068

a1 = -1.988586

a2 = 0.991315

# Some Underlying Filter Functions/Topologies

## (and the characteristics they maximize)

***Biquadratic Filter:*** efficient, but effective, second-order filter

***Chebyshev Function:*** best approximation to the ideal response of any filter for a specified order (but has significant ripple)

***Butterworth Function:*** maximally flat frequency response

***Bessel Function:*** most linear phase response, with no consideration of the frequency magnitude response

# Biquad(ratic) Filter

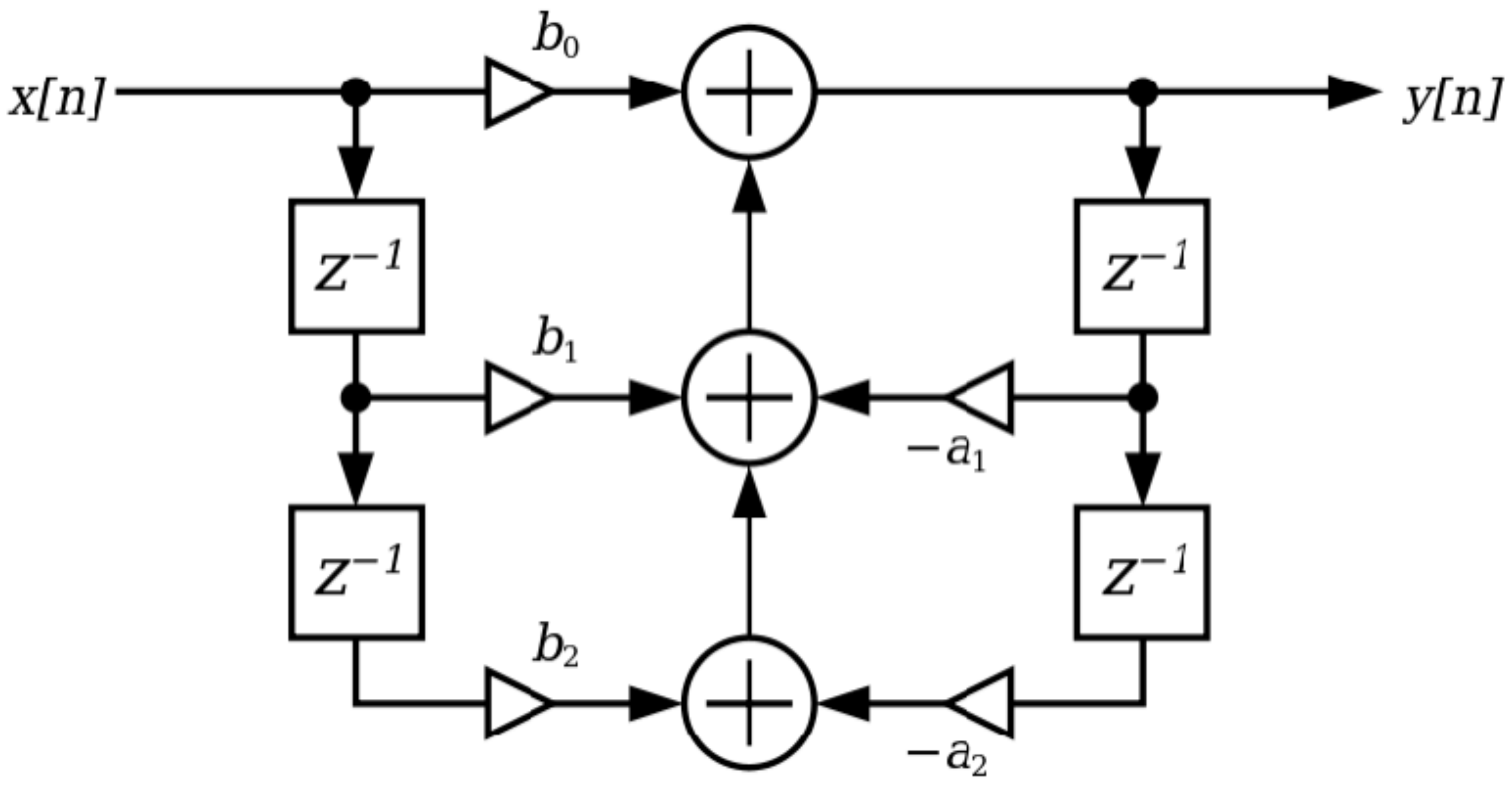
DIFFERENCE EQUATION

$$y(n) = b_0\, x(n) + b_1\, x(n-1) + b_2\, x(n-2) - a_1\, y(n-1) - a_2\, y(n-2)$$

FREQUENCY RESPONSE

low/high-pass, band-pass/stop, resonant, all-pass

SIGNAL FLOW DIAGRAM




MATLAB CODE

```
filter(b,a,x)

# where x is the input vector, b is the
list of feedforward coefficients, and a
is the list of feedback coefficients
```

DEMO (IN MAX)