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)

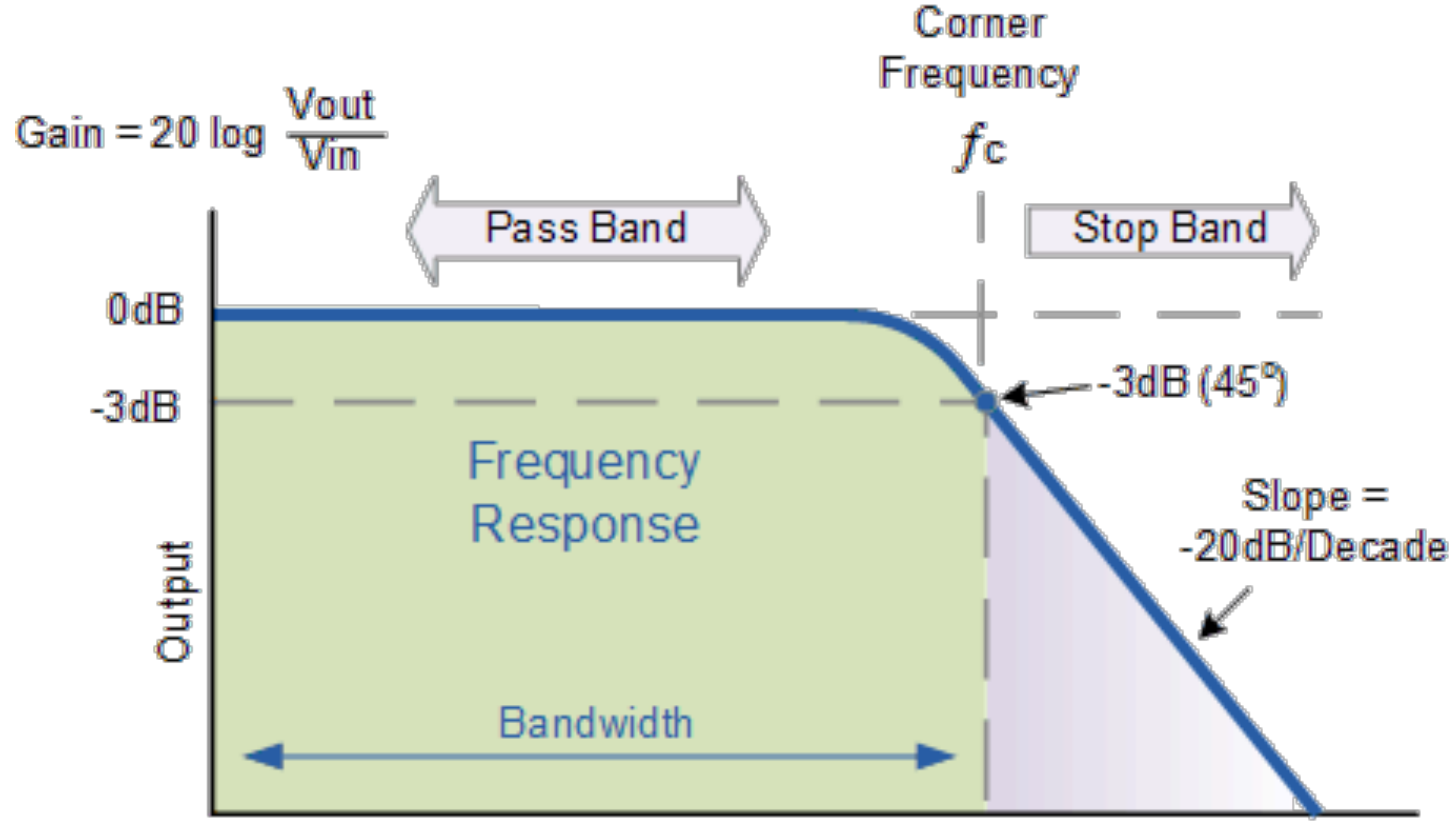
Comb

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.

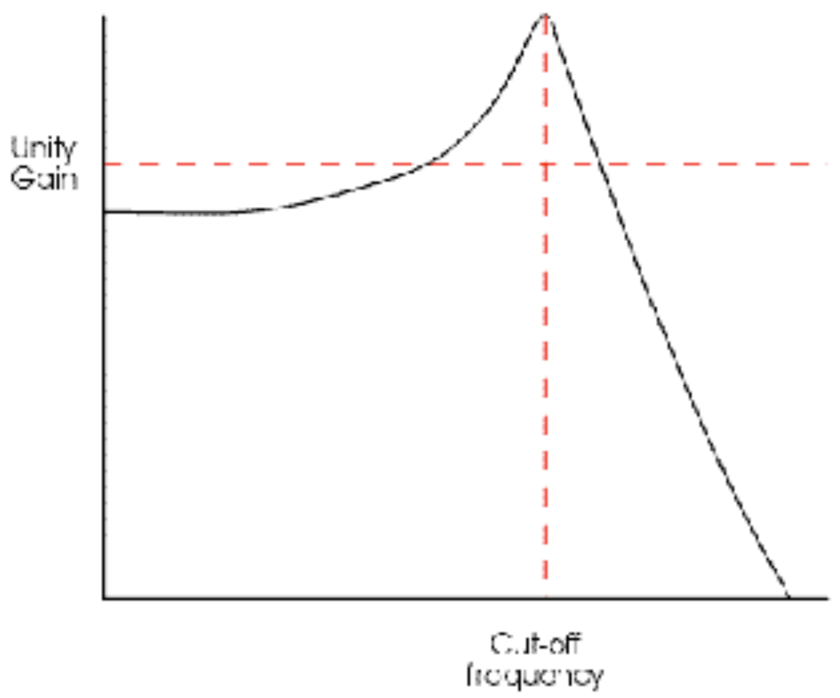


Figure 13: A typical resonant low-pass filter response.

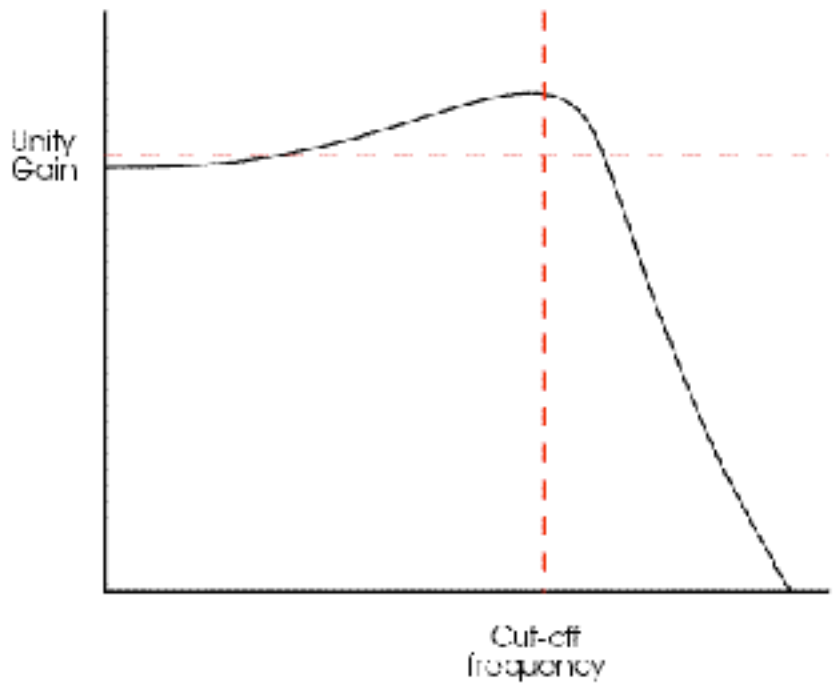


Figure 14: Resonant low-pass filter with low Q.

The Parameters of Filters

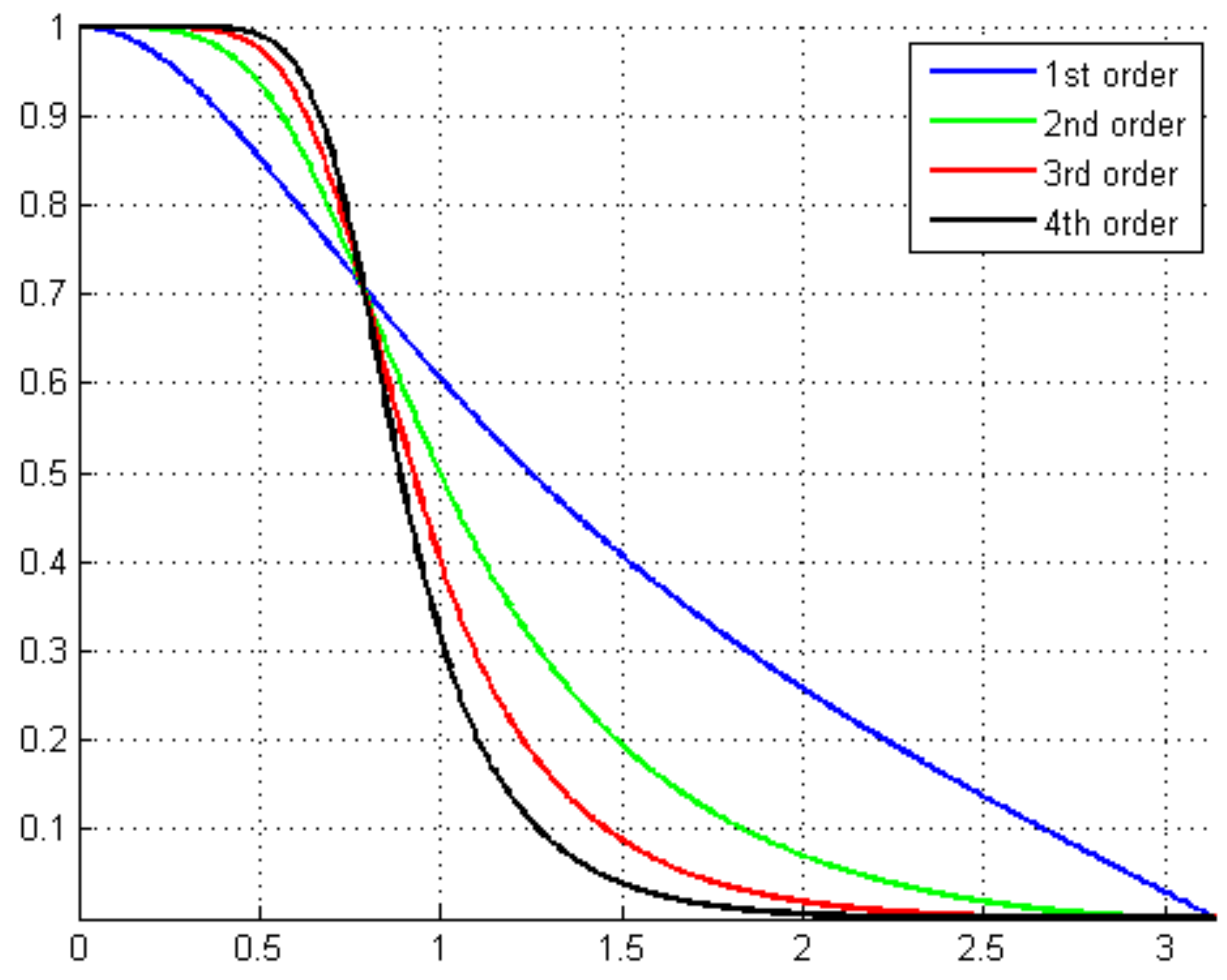
Descriptive/Behind-the-Scenes Parameters:

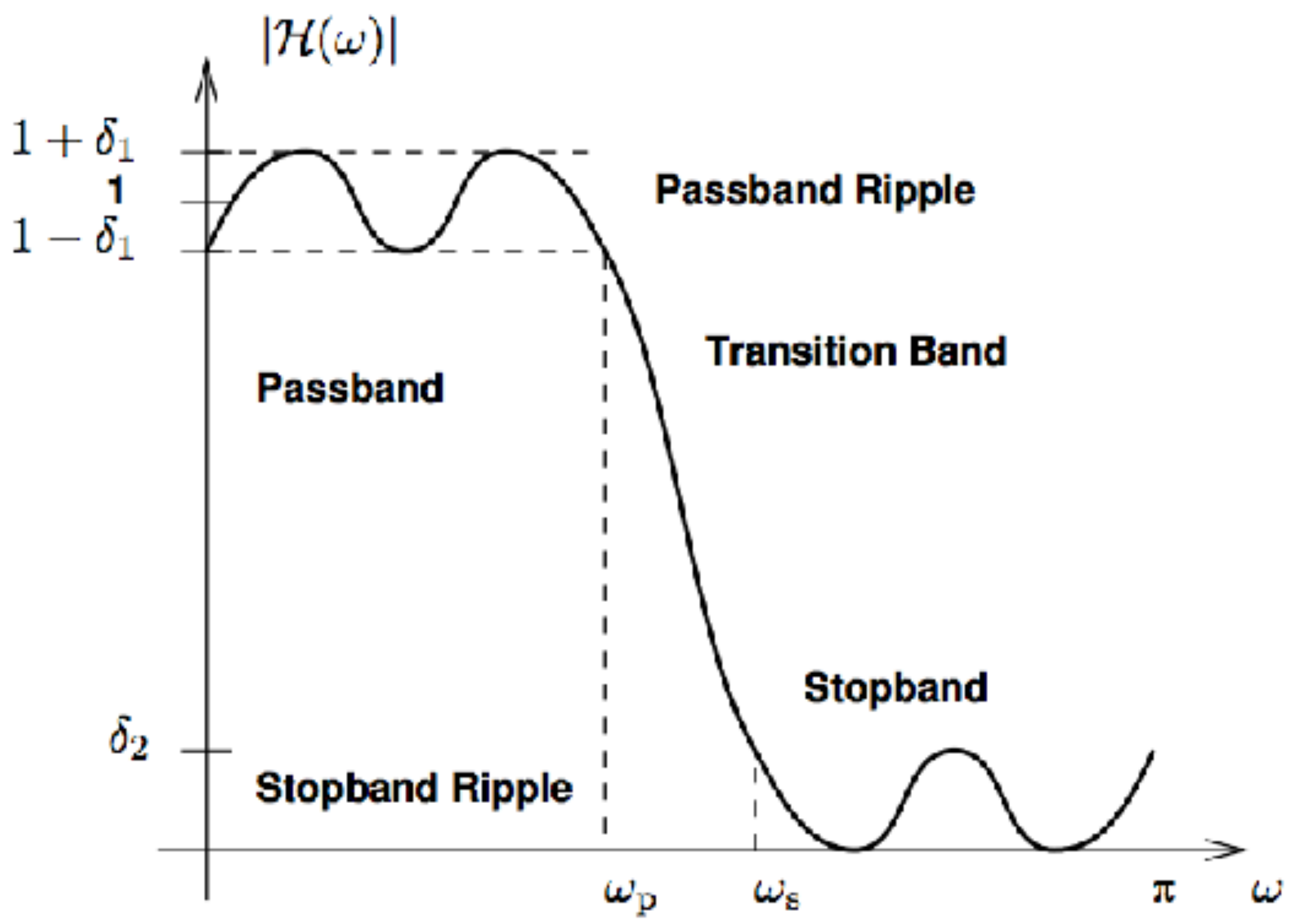
Order - a number describing the highest sample delay in a filter, often corresponding to the steepness of a filter's slope

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 (%)

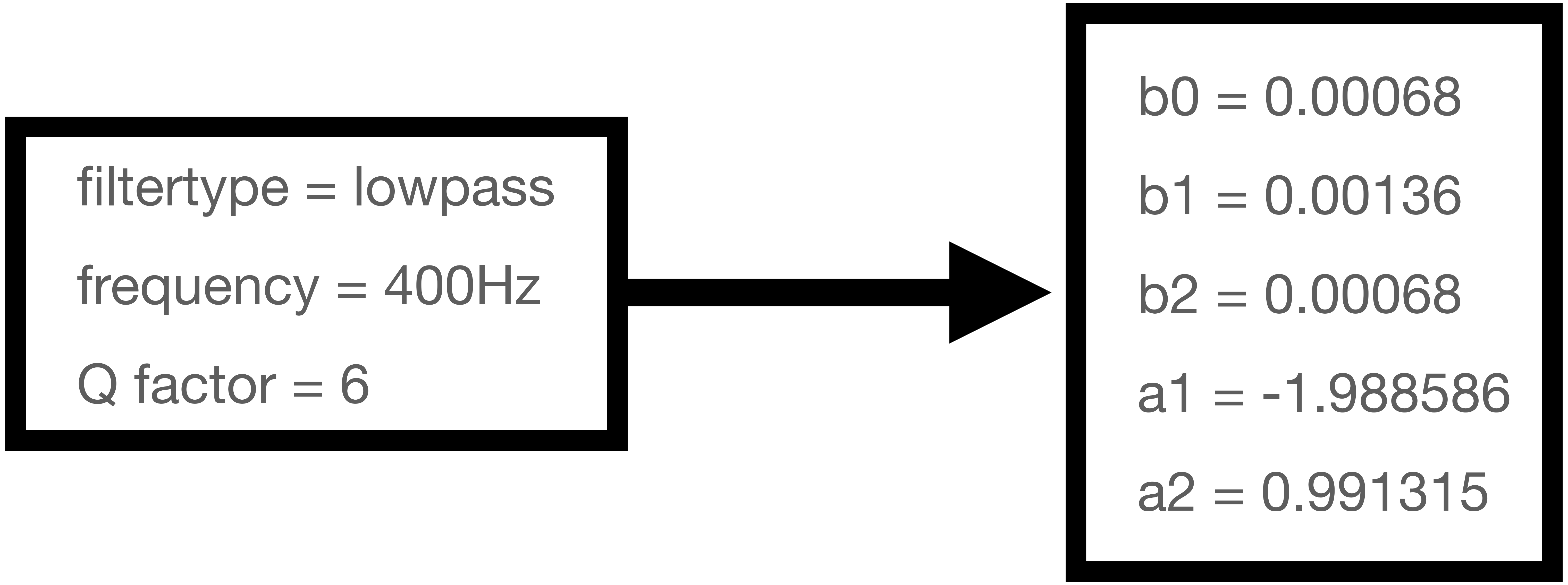




Designing Filters from the Top-Down

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.:



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

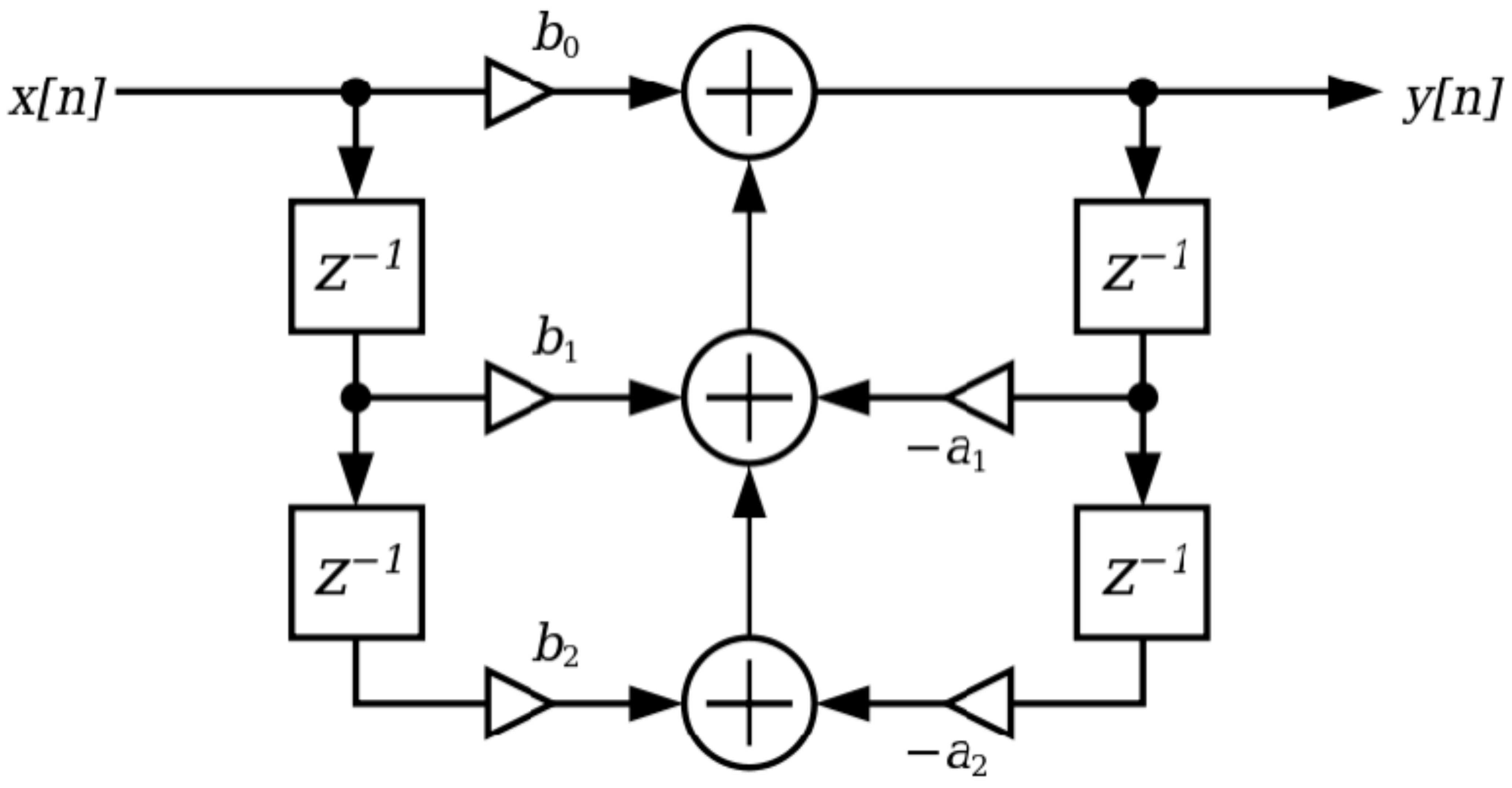
Bessel Function: most linear phase response, with no consideration of the frequency magnitude response

Biquad(ratic) Filter

DIFFERENCE EQUATION

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

SIGNAL FLOW DIAGRAM



FREQUENCY RESPONSE

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

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)

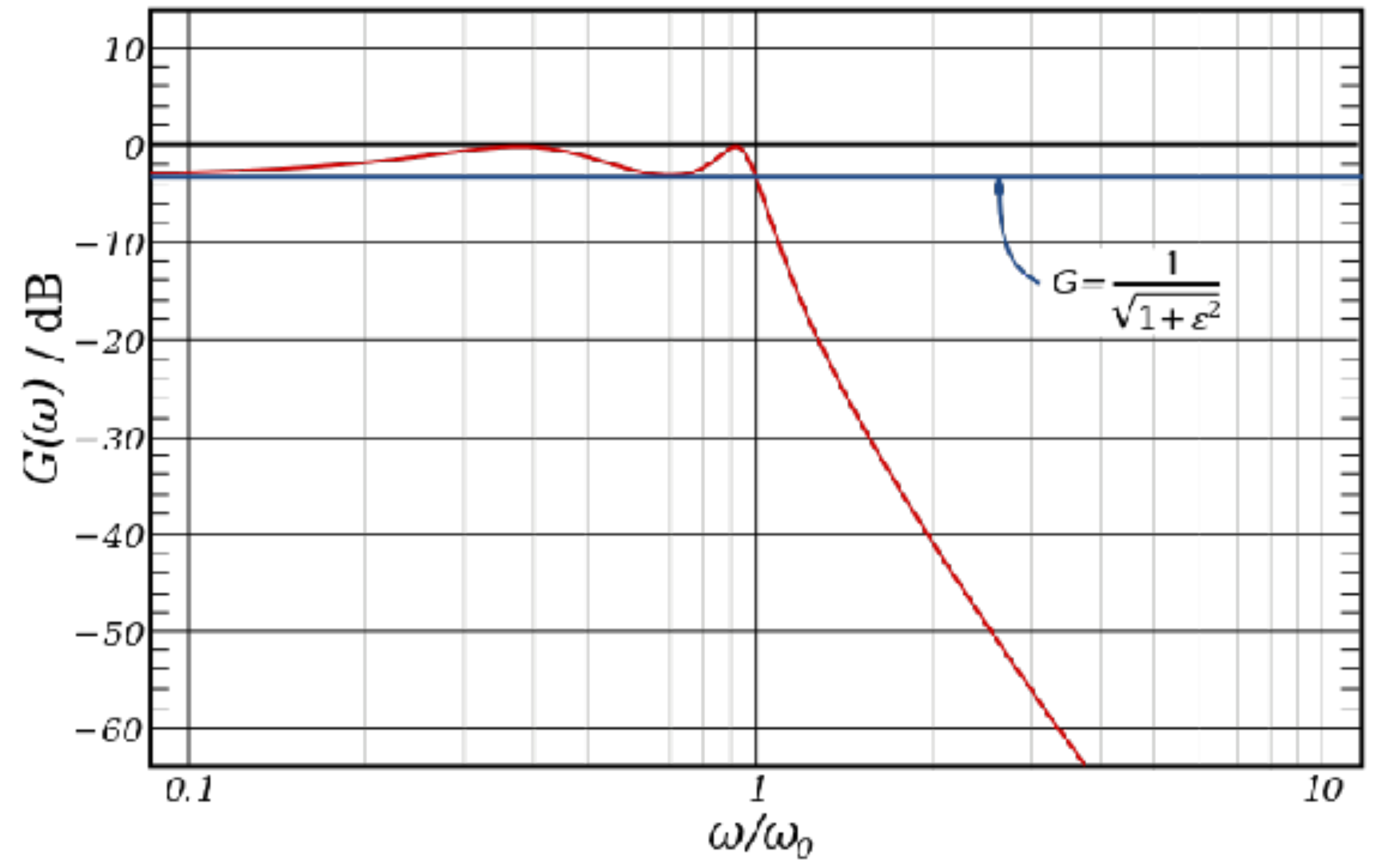
Chebyshev Function

Characteristic: best approximation to the ideal response of any filter for a specified order…but has ripple in passband (Type I) or stopband (Type II)

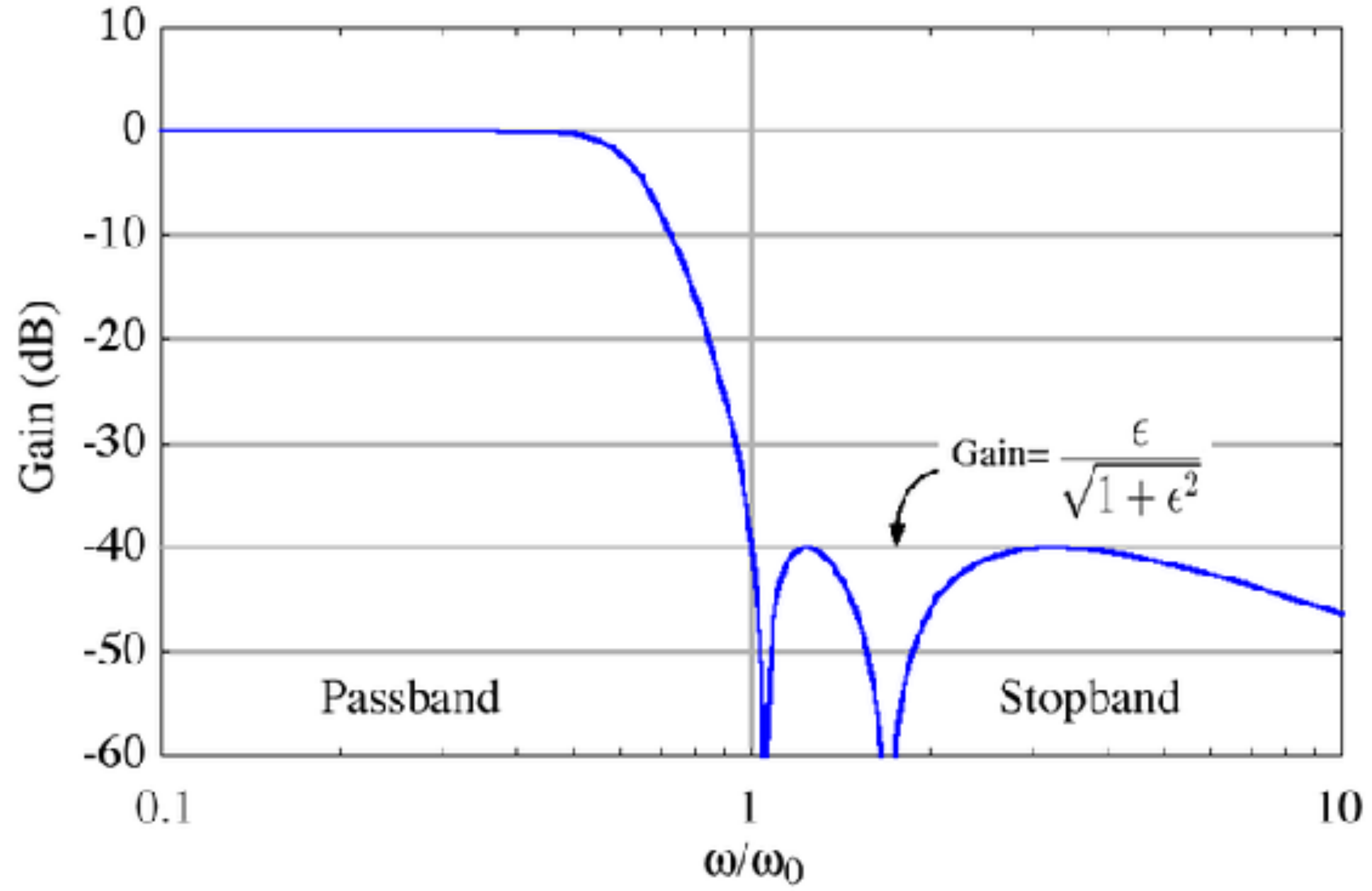
MATLAB CODE

```
[b,a] = cheby1(n,Rp,Wp)
returns the transfer function coefficients of an nth-order
lowpass digital Chebyshev Type I filter
with normalized passband edge frequency Wp and Rp decibels of
peak-to-peak passband ripple.
```

DEMO (IN MAX)



TYPE I



TYPE II

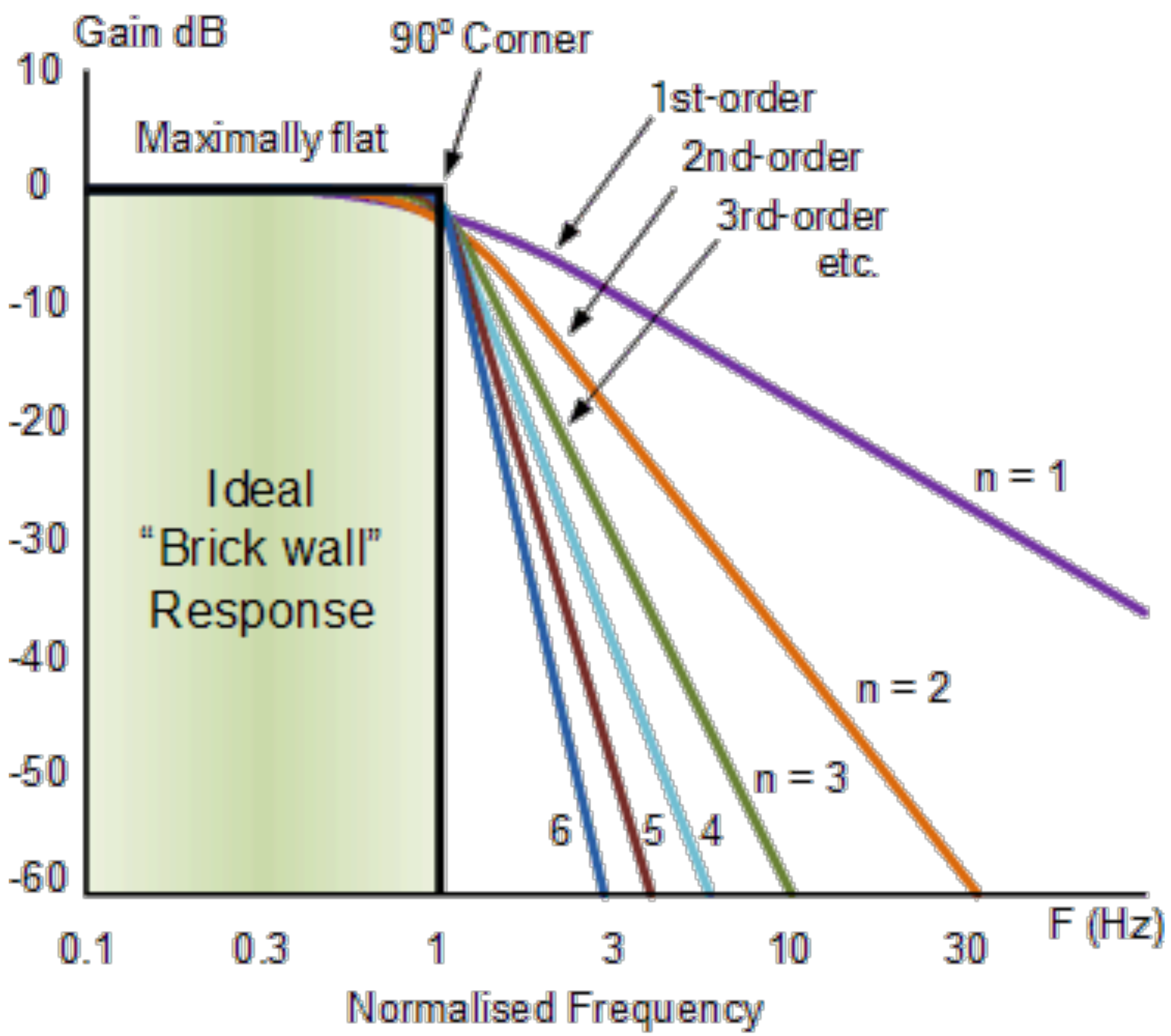
Butterworth Function

Characteristic: maximally flat frequency response in the passband (wow, that's butter!)

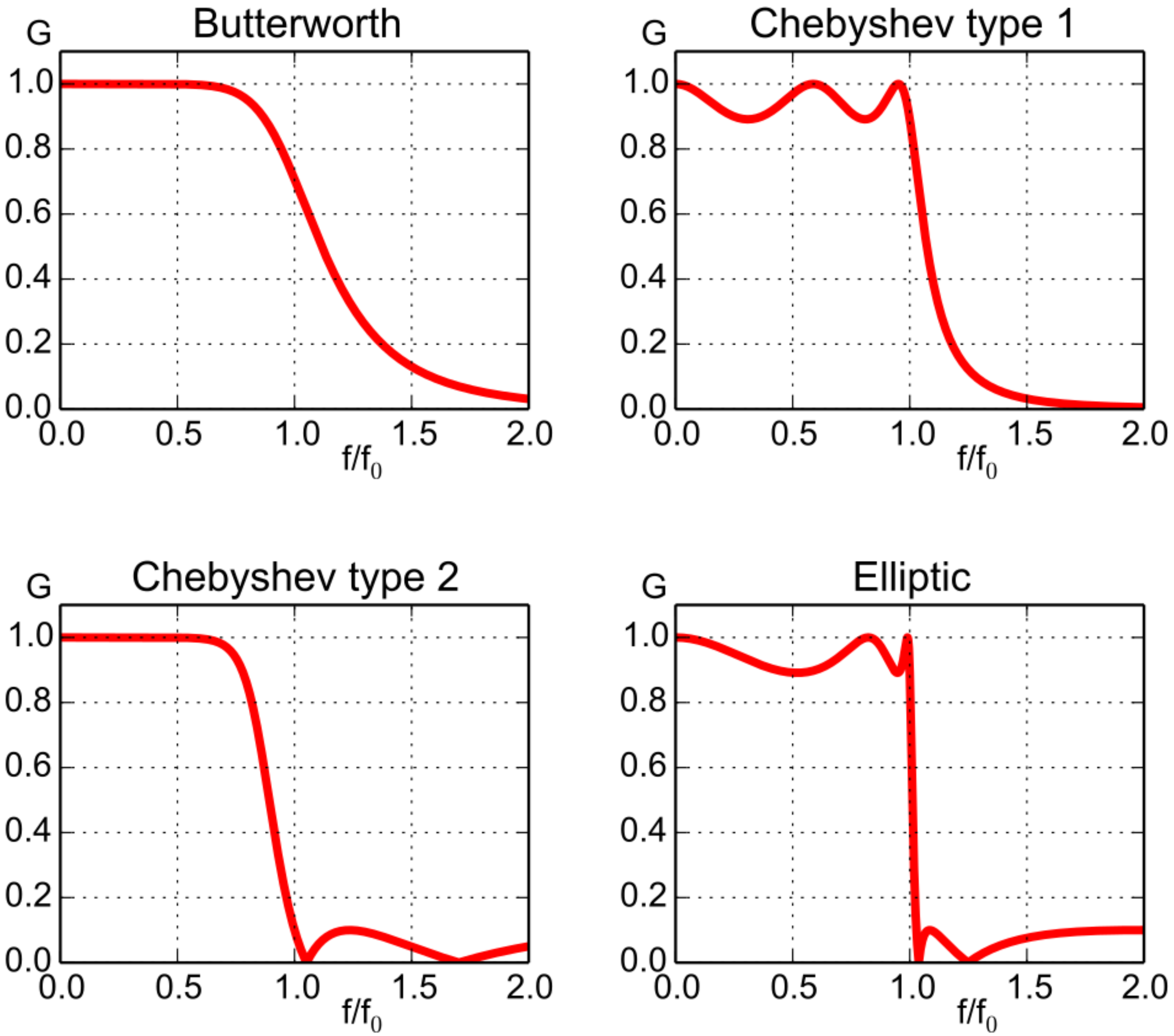
MATLAB CODE

```
[b,a] = butter(n,Wn) returns the transfer function
coefficients of an nth-order lowpass digital Butterworth
filter with normalized cutoff frequency Wn.
```

DEMO (IN MAX)



Comparison of Filter Topologies



Comparison of Filter Topologies

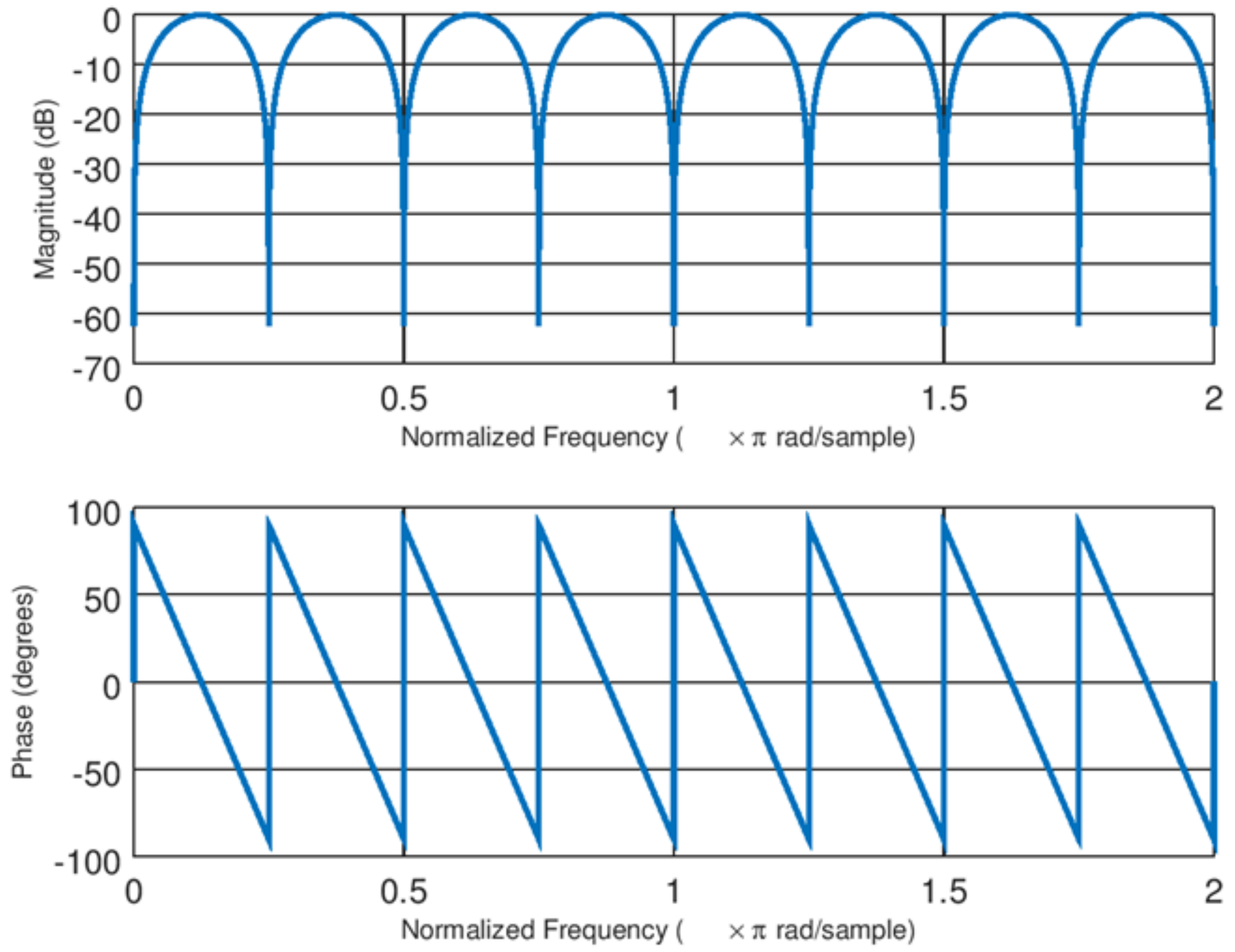
	PASSBAND	TRANSITION REGION	STOPBAND
BUTTERWORTH	Maximally flat magnitude response in pass-band.	Steeper than Bessel, not as steep as Chebyshev or Inverse Chebyshev filters.	No ringing
INVERSE CHEBYSHEV	Flat magnitude response in pass-band.	Steeper than Butterworth, Bessel, and Chebyshev filters.	More ringing than other filters.
CHEBYSHEV	Ripple in the pass-band.	Steeper than Butterworth and Bessel filters, but not as steep as the Inverse Chebyshev filter.	No ringing
BESSEL	Flat magnitude response in pass-band.	Slower than the Butterworth, Chebyshev or Inverse Chebyshev filters.	No ringing

Comb Filter

Characteristic: demonstrates constructive and destructive interference caused by phase offset.

Result is equally spaced resonant cascaded notch filters (resembling a “comb”).

Uses: flanging (when delay time is modulated by a low frequency oscillator (LFO)), harmonic “ringing out” as an effect.

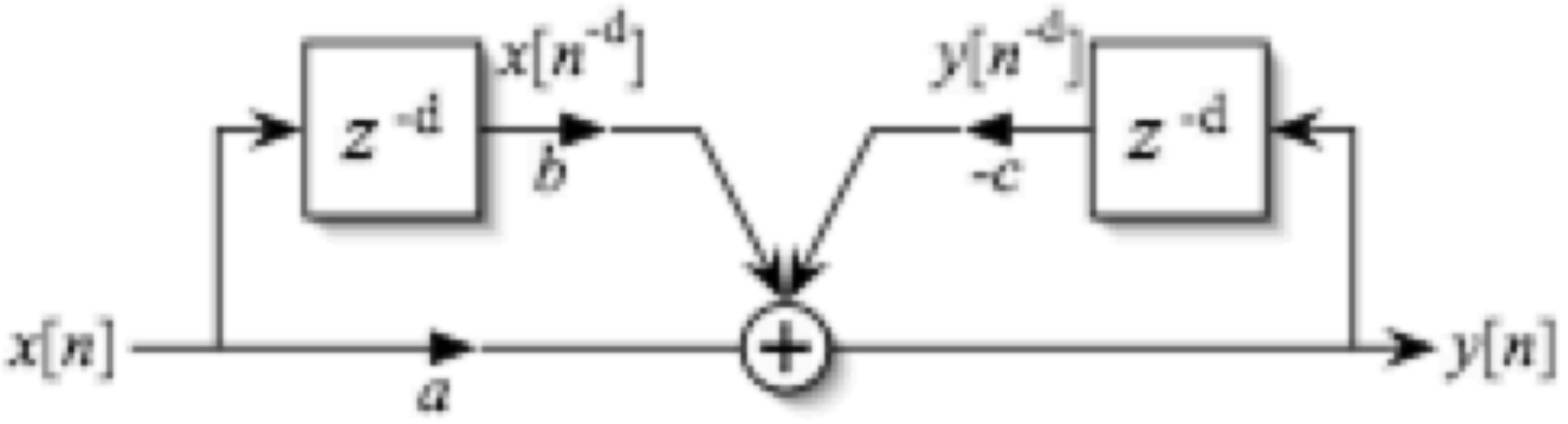


Comb Filter

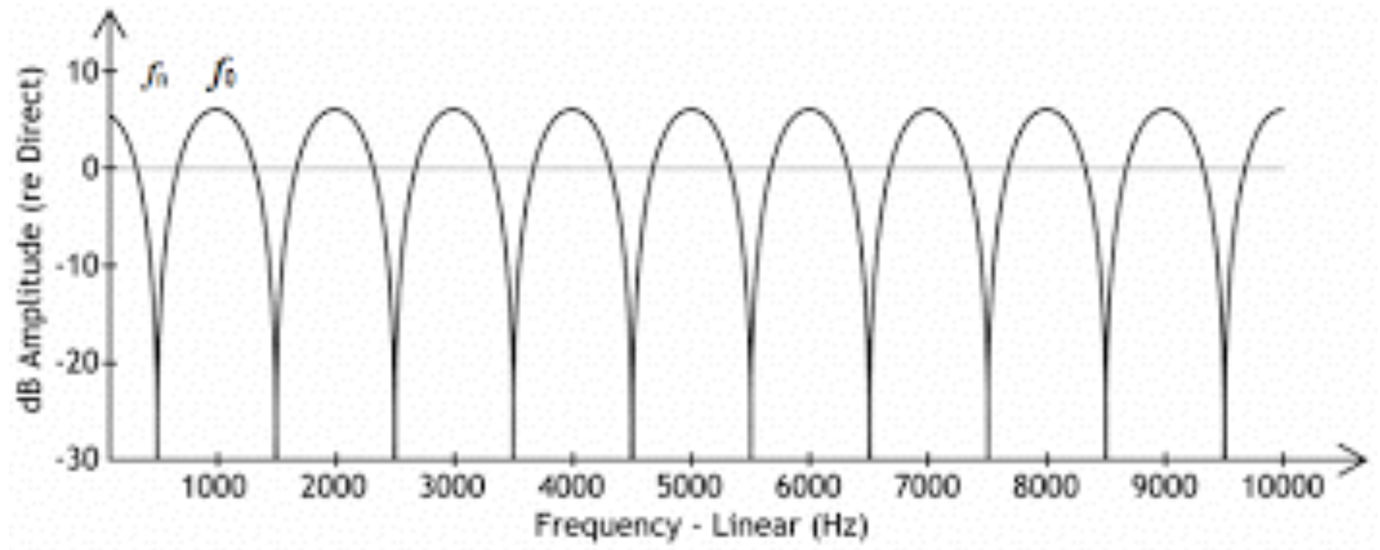
DIFFERENCE EQUATION

$$y(n) = a\,x(n) + b\,x(n\text{-}d) - c\,y(n\text{-}d)$$

SIGNAL FLOW DIAGRAM



FREQUENCY RESPONSE



MATLAB CODE

```
d  = fdesign.comb('notch','N,BW',8,0.02);
```

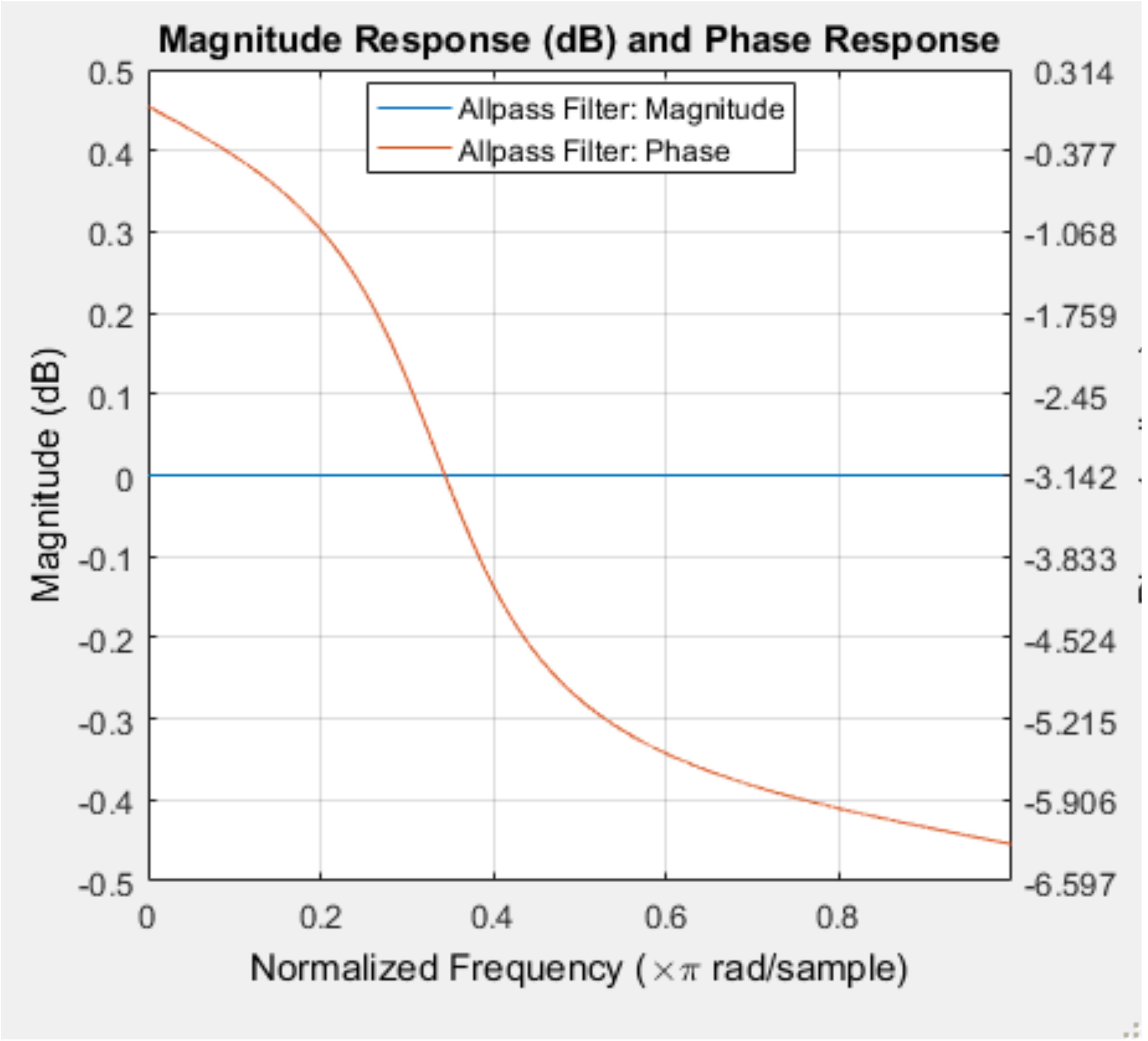
DEMO (IN MAX)

Allpass Filter

Characteristic: magnitude response is unity over its entire frequency range (***passes all frequencies***), but whose phase response is variable.

As their phase response can be variable, allpass filters can be put in series before filters with problematic phase response, ultimately correcting phase issues (neat!)

Uses: Makes a significant appearance in algorithmic reverb design (which we'll talk about shortly)

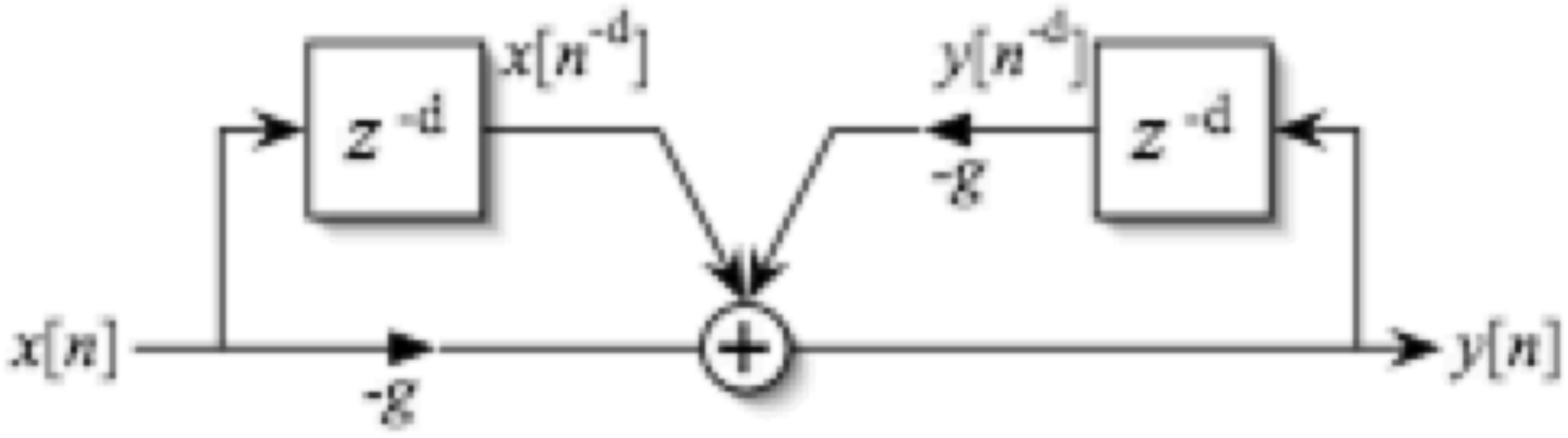


Allpass Filter

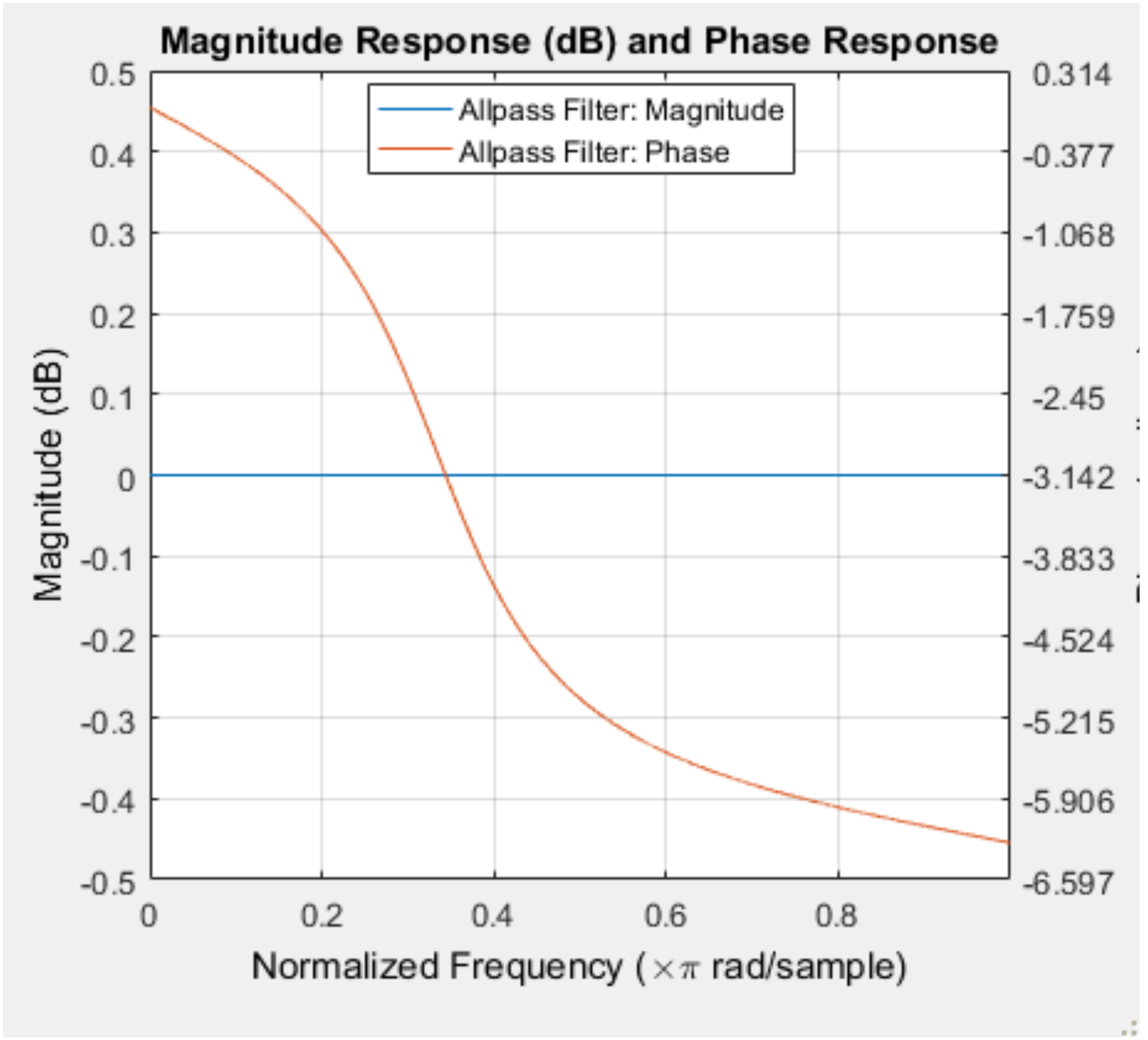
DIFFERENCE EQUATION

$$y(n) = -g\,x(n) + x(n-d) - g\,y(n-d)$$

SIGNAL FLOW DIAGRAM



FREQUENCY RESPONSE



MATLAB CODE

```
c = [1.5, 0.7];
% Create a second-order dfilt object.
hd = dfilt.allpass(c);
```

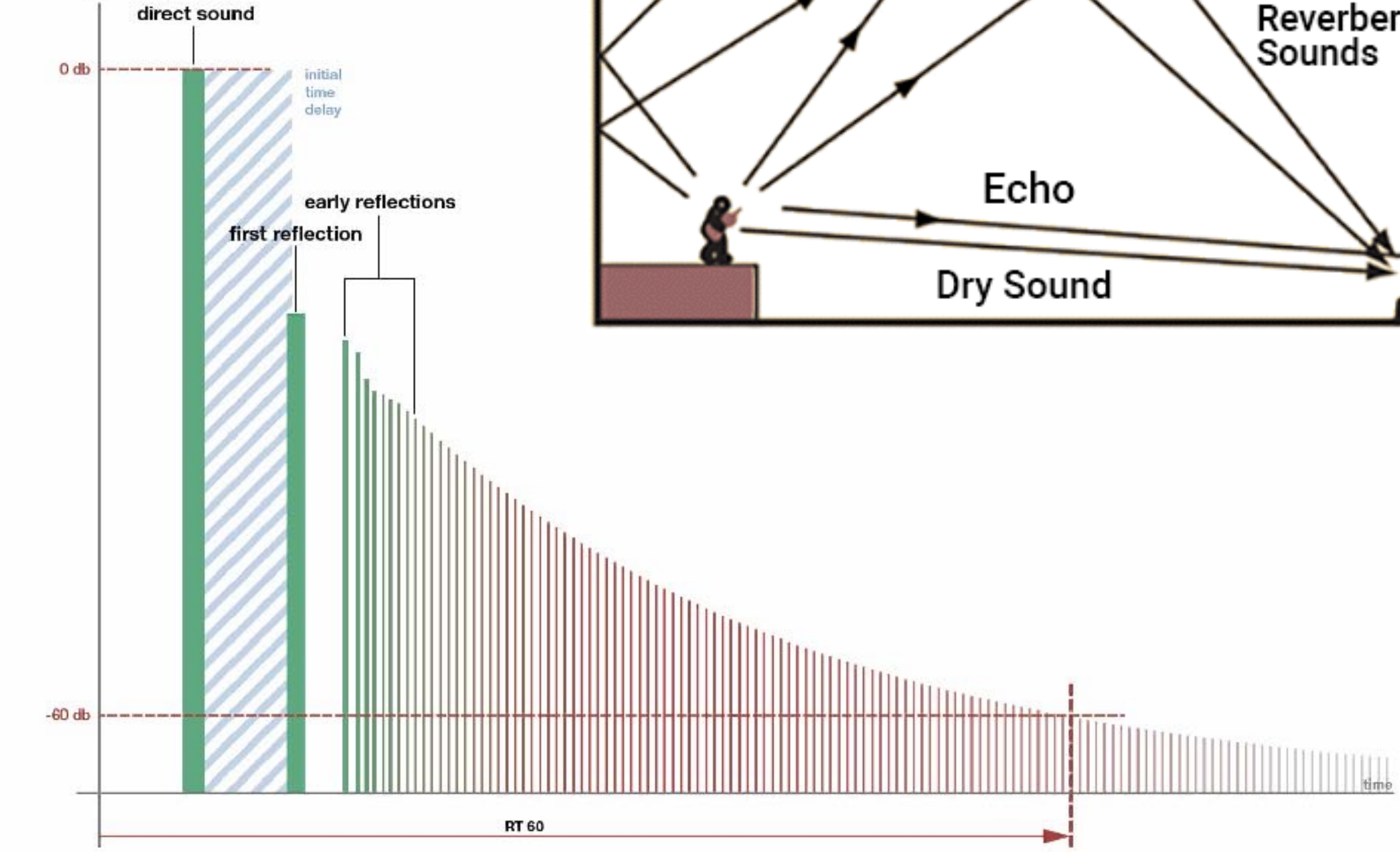
DEMO (IN MAX)

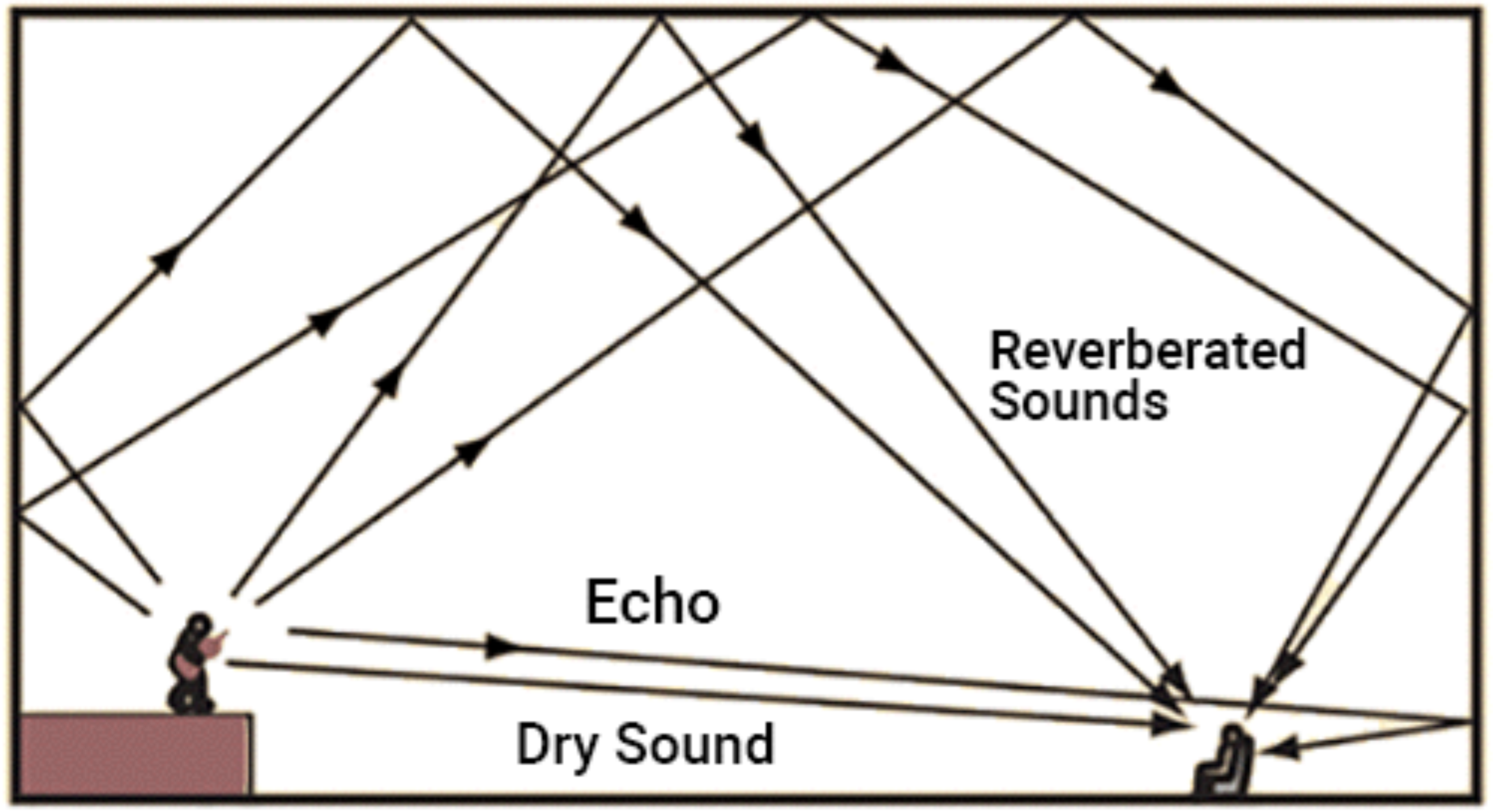
Digital Algorithmic Reverb

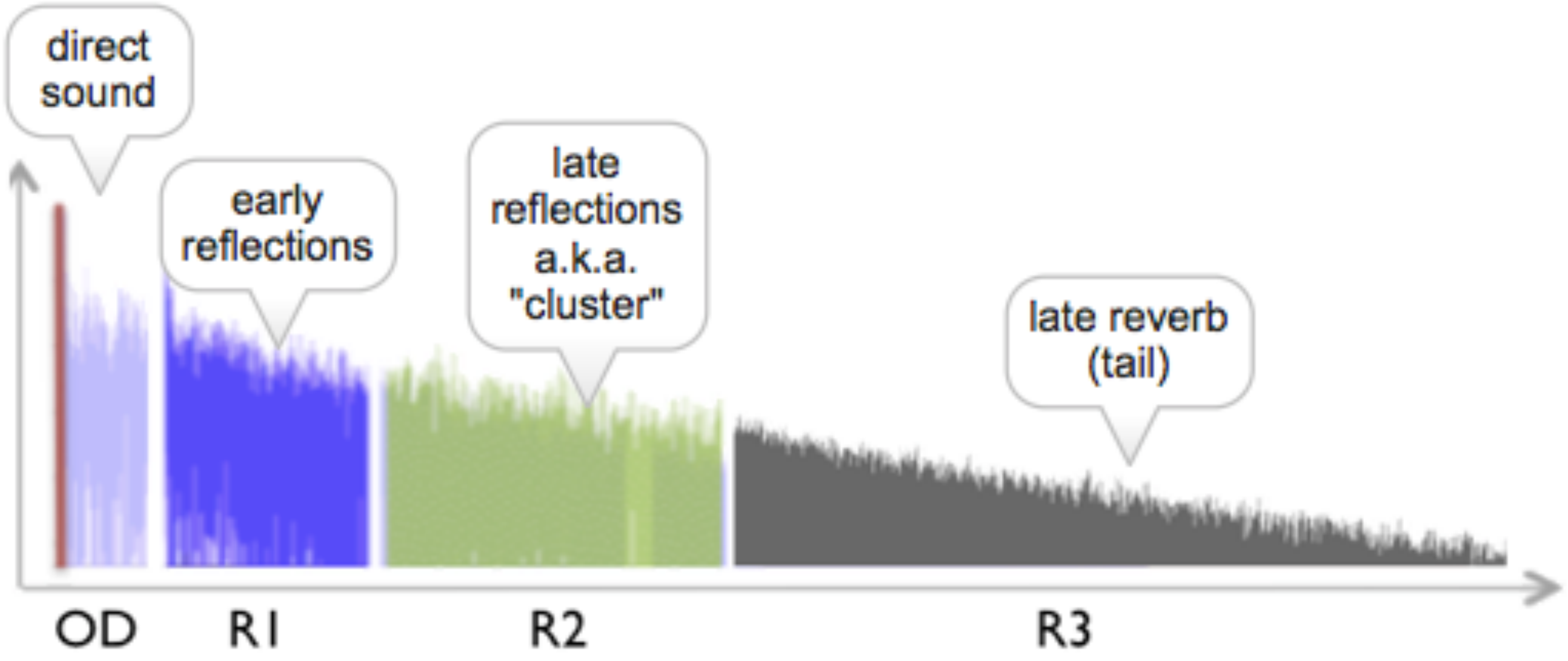
Definition: The digital simulation of virtual reverberance through networks of filters (as opposed to convolution or impulse response-based digital reverb simulation)

Exceedingly important invention to music production, as it allows for the application of super-flexible simulated reverb to dry signals, without the need for re-recording (recording a version of a signal played back into a reverberant space)

What are the Components of Reverb?







Reverb Typologies

Plate: simulates playing the sound electromagnetically at one side of a plate and recording it on the other side (used for Mastering)

Spring: simulates playing the sound electromagnetically at one side of a spring and recording it on the other side

Room: simulates an acoustic space (short RT60)

Hall/Chamber/Arena: simulates an acoustic space (longer decay)

Inverted: reversed reverberation

Gated: reverberation that is gated (i.e. attenuates amplitude after it dips below a certain threshold

Algorithmic Reverb: Parameters

Room Size: simulated distance between walls in the virtual acoustic space (affects overall length of delays in the network)

RT60 (Decay): how long it takes a sound put through the reverb to decay 60dB.

Damping: simulated absorption properties of surfaces in the virtual acoustic space (e.g. tile, wood, carpet, etc.) (affects lowpass filtering applied to the network)

Wet/Dry: relative amplitude levels of the input signal and the output of the reverberation (run in parallel with inverse gain of each other)

Density/Diffusion: depends on implementation, but often results in perceived prominence of early/late reflections (echo-y) vs. tail (smooth)

Pre-Delay: amount of time between the direct sound and the start of the early reflections

Algorithmic Reverb: Examples

Some Versions:

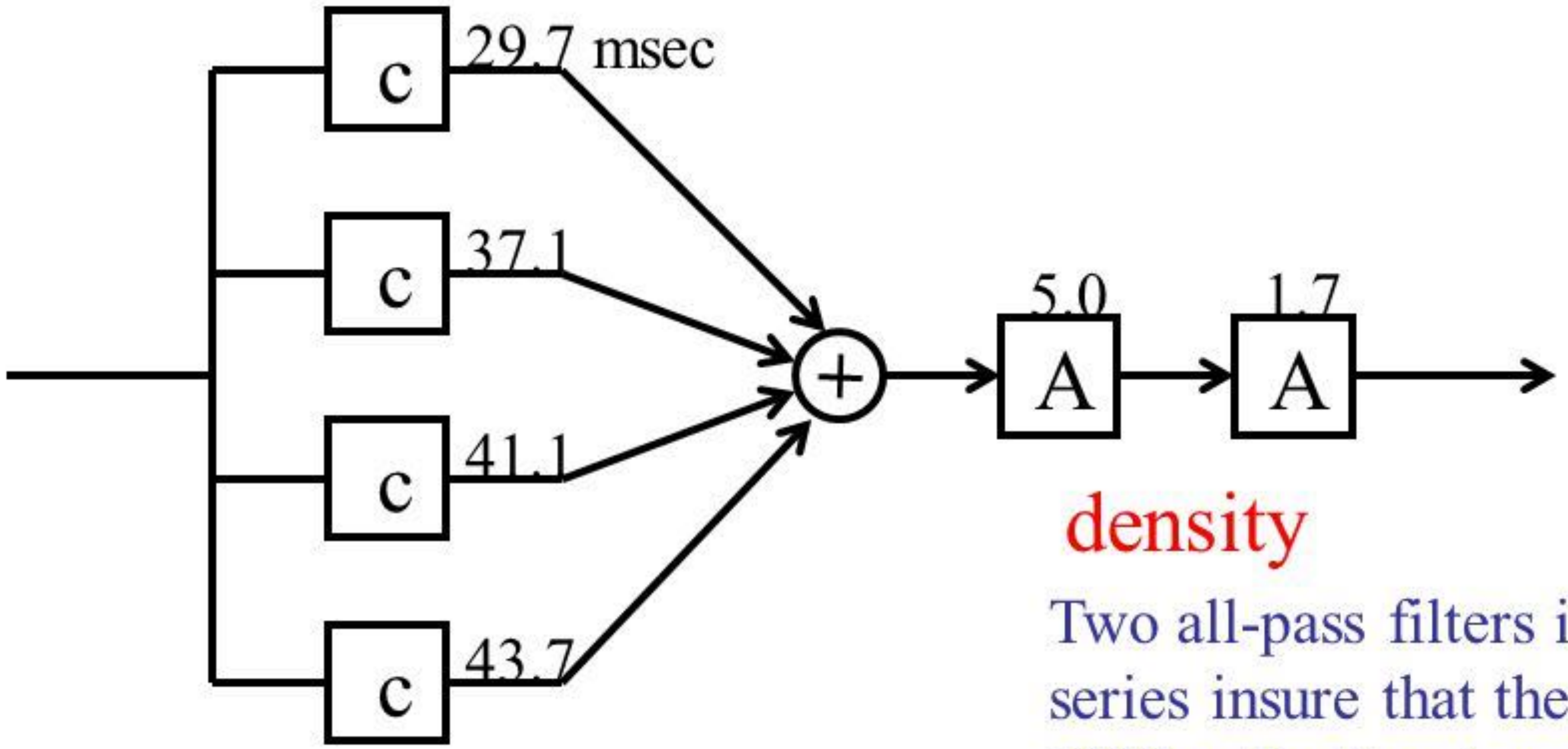
Schroeder

Griesinger

Moorer

Freeverb3

Schroeder Reverberator



decay

Four recursive comb filters in parallel determine the reverb time

density

Two all-pass filters in series insure that there are 1000 reflections/sec.