



OLLSCOIL NA GAILLIMHE  
UNIVERSITY OF GALWAY

## School of Engineering Report/Assignment Cover Page

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Class and Year (4 <sup>th</sup> Electronic & Computer Engineering etc.):	4BP1
Subject Code and Name: (e.g. EI140 Fundamentals of Engineering)	<b>EE445 Digital Signal Processing</b>
Lecturer Name:	Edward Jones
Title of Report/Assignment:	<b>DSP-2 Digital Filter Design</b>
Submission Deadline:	11:59 pm on Friday, November 29 <sup>th</sup> , 2024
Submission Date:	27/11/2024
Name(s) of Lab Partner(s) (where applicable)	N/A

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Student's signature

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# Question 1 - IIR Filter

Using the `cheby1`, `cheb1ord` and `butter` functions build a chebyshev low pass filter matching the following specifications and an equivalent butterworth of the same order.

*Specifications:*

- Sampling Freq:  $F_{\text{samp}}$ : 10Khz
- Cutoff Freq:  $F_c$ : 3khz
- Transition BW: 400hz
- Freq sampled:  $F_s$ : 3.4khz
- Stop Band Attenuation: 35dB: i.e. 35dB @ 3.4khz
- Pass band Ripple: 0.1dB

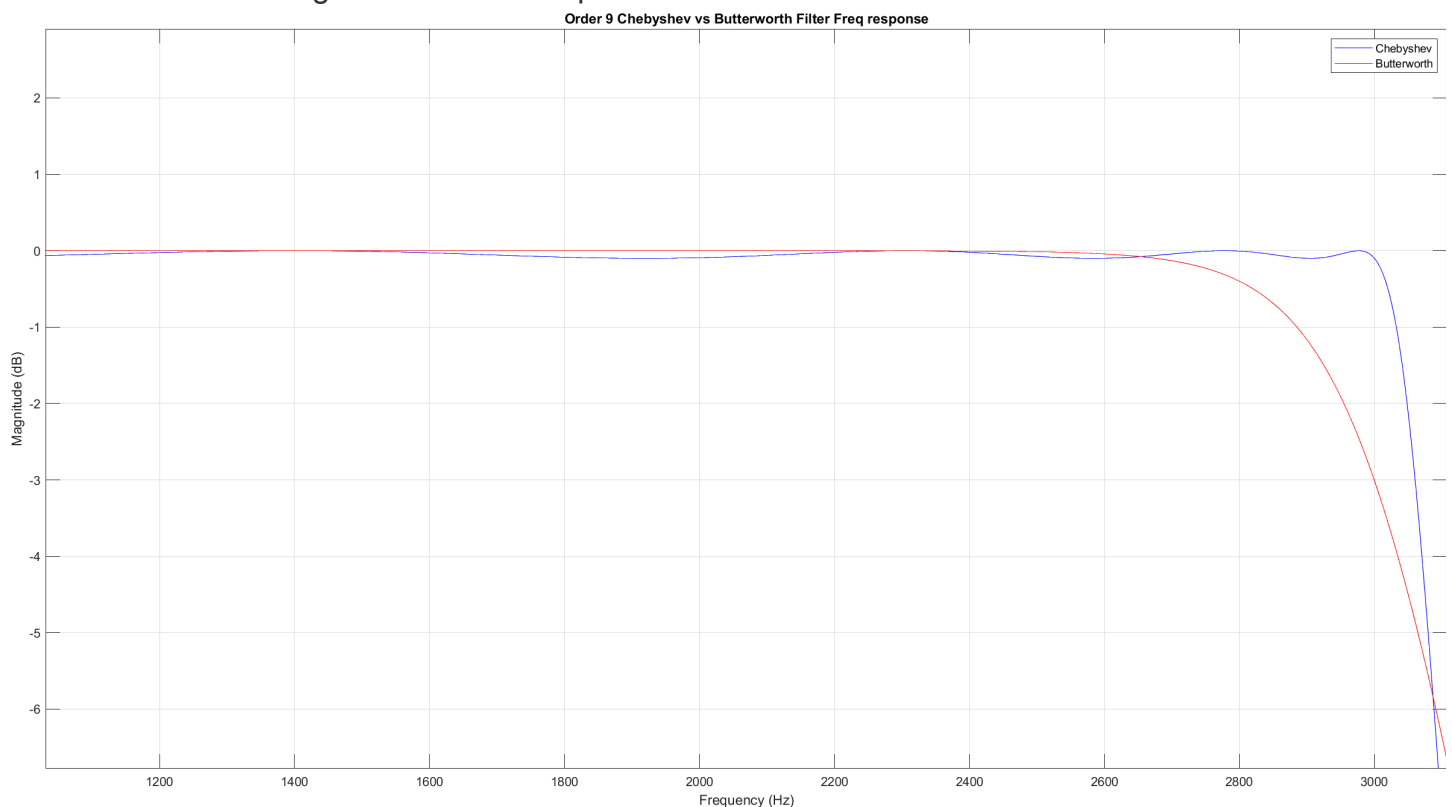
*Butterworth Alignment:*

There are two main ways to align the Butterworth filter with the Chebyshev,

1. Set the cutoff frequency the same for each frequency.
2. Have the 0dB attenuation move at the same point. i.e 3000hz before the filters drop.

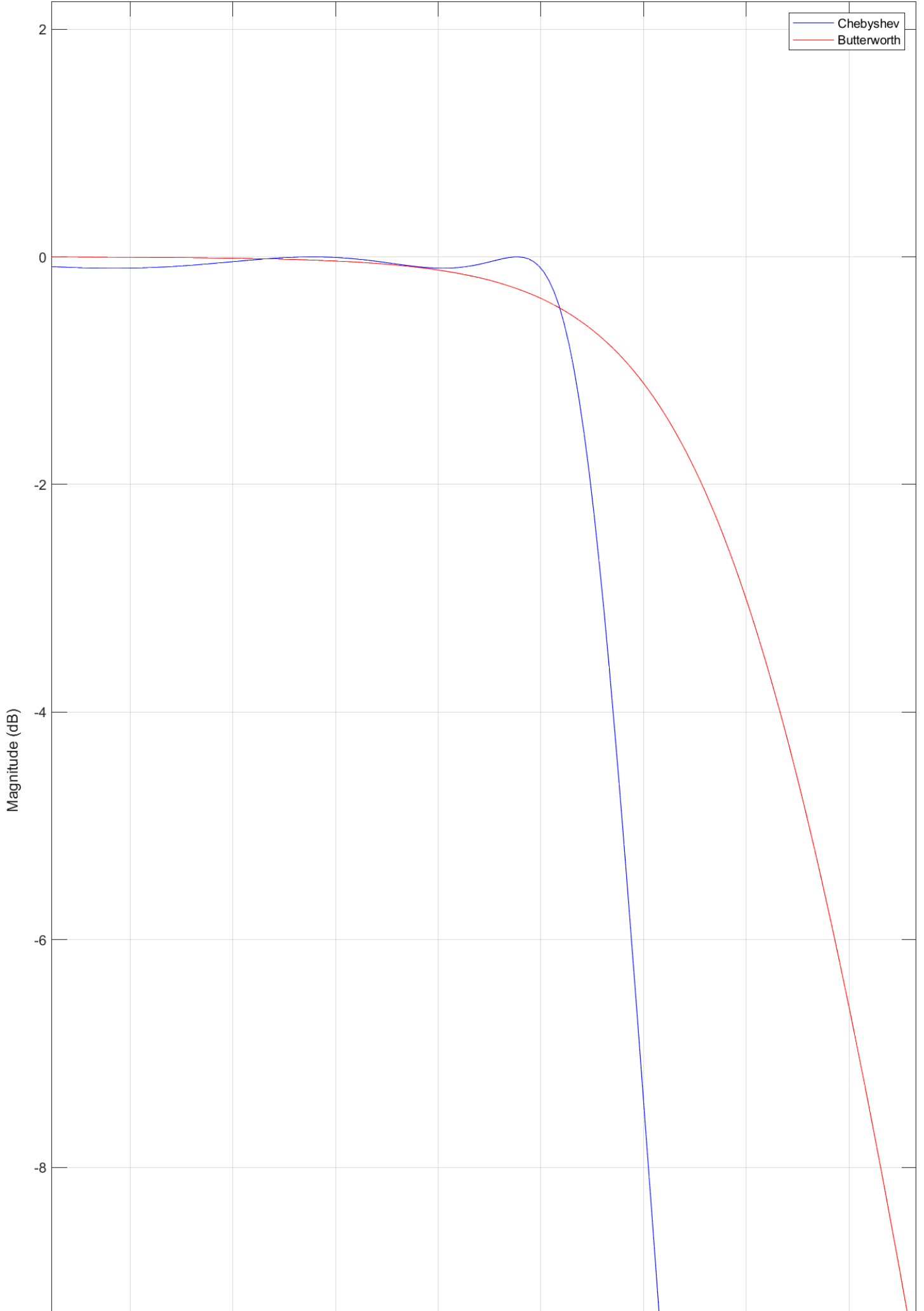
Cutoff frequencies are calculated slightly differently for a Chebyshev and Butterworth, where Chebyshev has the cutoff frequency as the point of the last ripple and butterworth has the cutoff frequency as the point where the function hits -3dB.

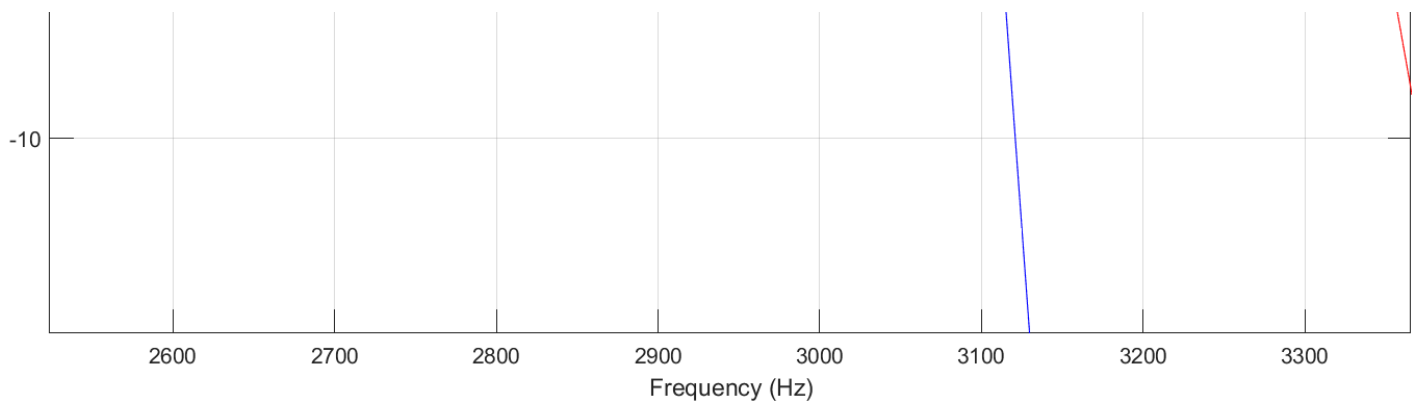
The graphs below show the two different options, one with butter cutoff at 3000hz and one with butter cutoff at 3200hz to align the attenuation points.



The Butterworth filter hits -3dB at 3000hz in the image above. This function reaches the transition bandwidth closer than than the below option.

Order 9 Chebyshev vs Butterworth Filter Freq response



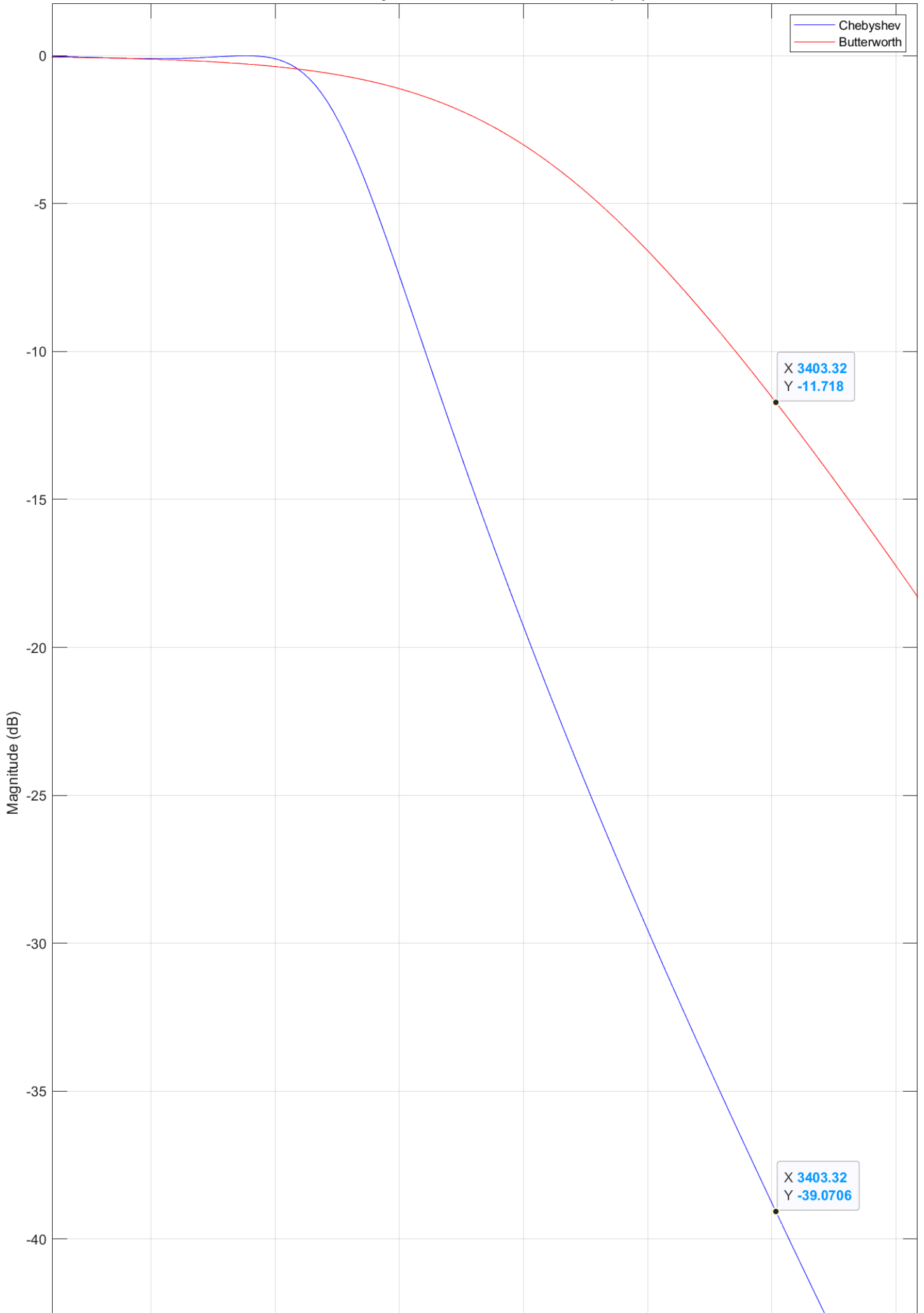


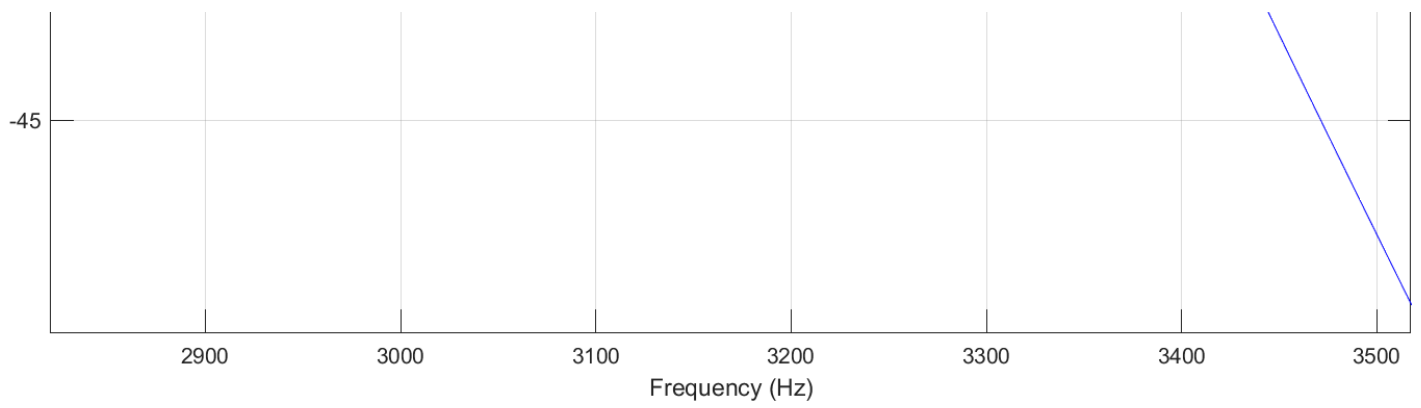
The Butterworth filter hits -0.35dB at 3000hz. The 2 functions match with attenuation drop off at the same points. This allows for better comparisons with how quickly the functions drop off.

The second option will be used for comparison purposes to keep the attenuations comparable.

*Filter Sharpness*

Order 9 Chebyshev vs Butterworth Filter Freq response

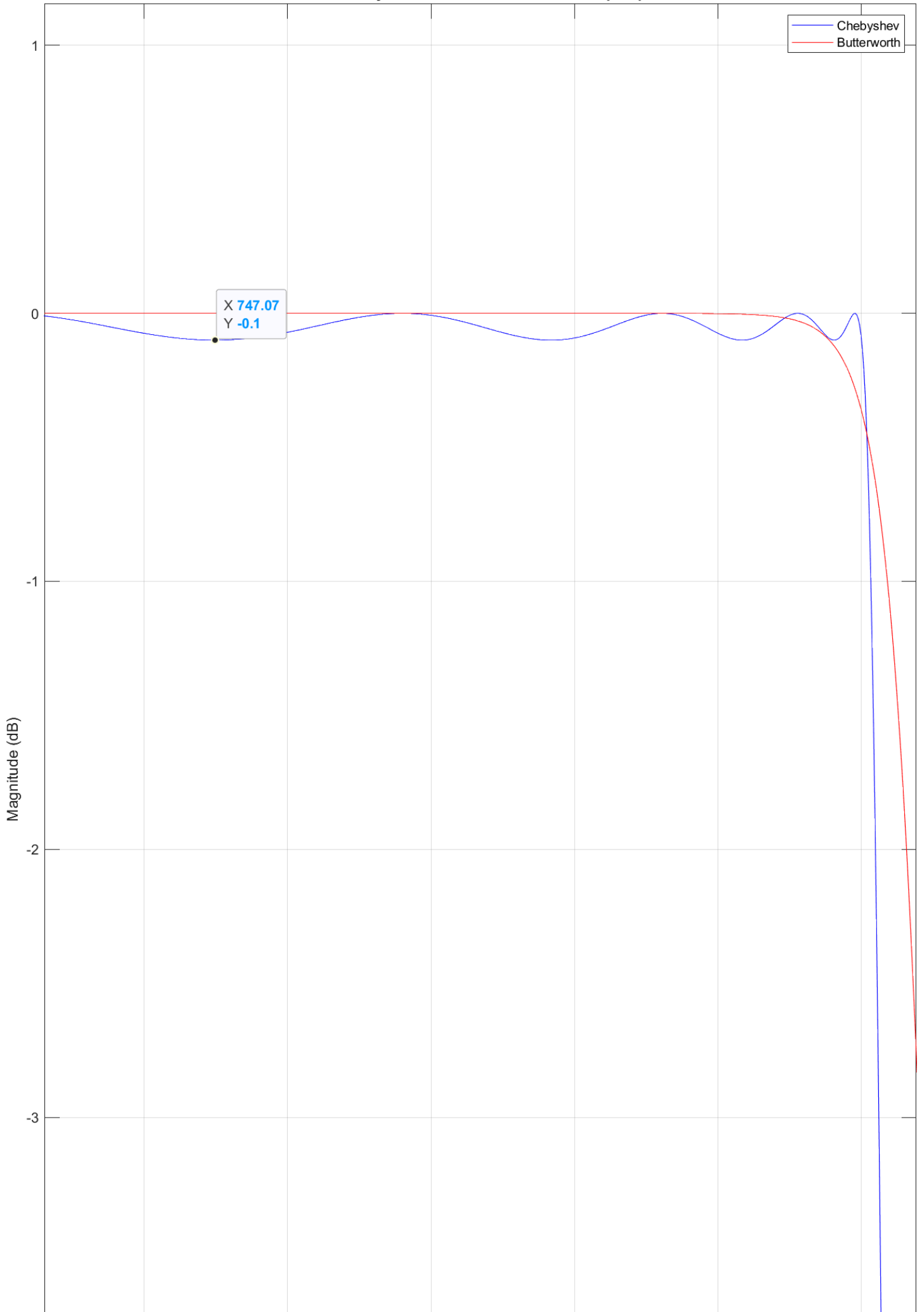




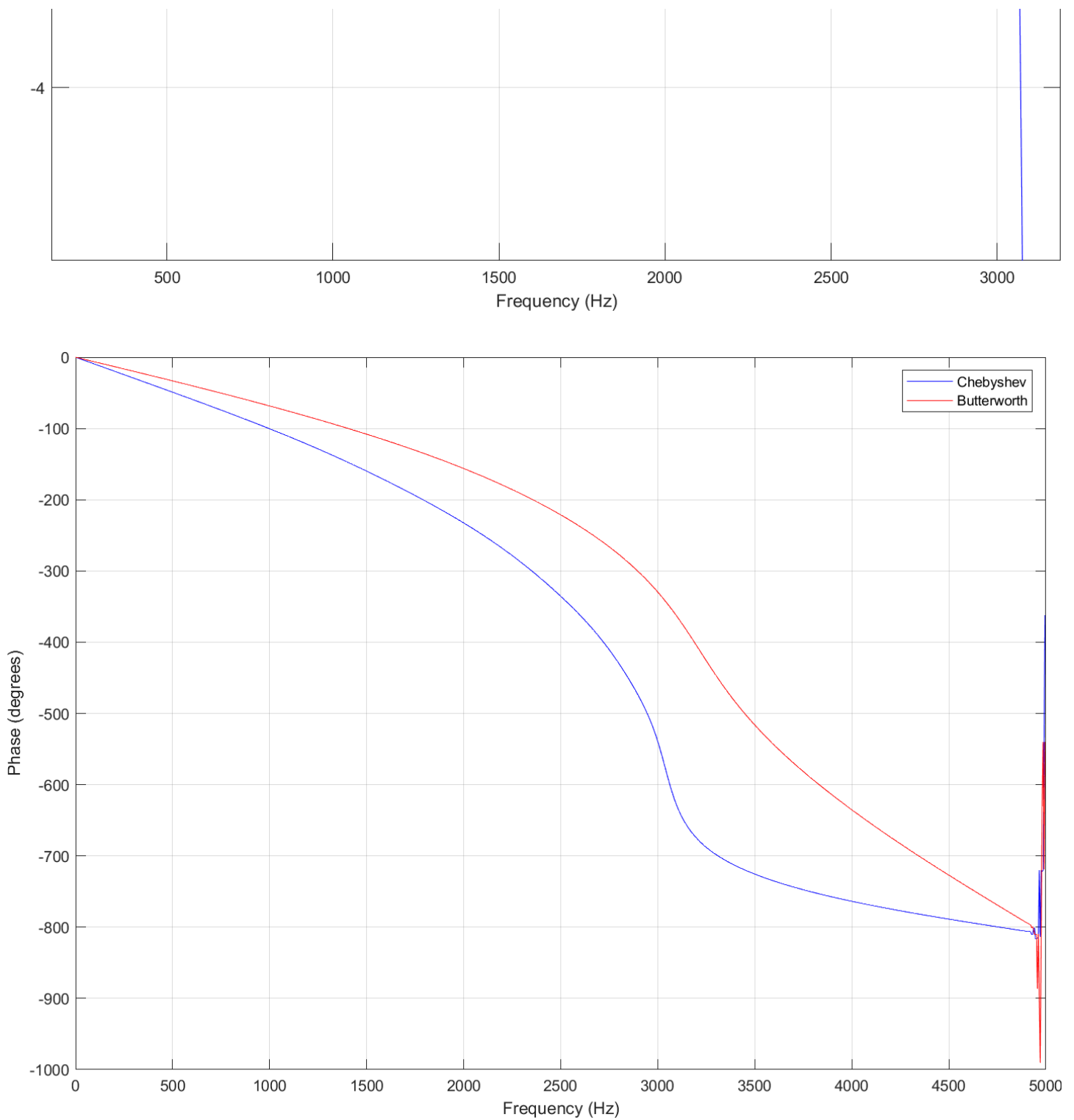
The above image shows how much both filters have attenuated at the transition bandwidth point. The difference is 27dB or an absolute magnitude of 500x.

*Pass Band distortion*

Order 9 Chebyshev vs Butterworth Filter Freq response







The images above show the ripple amplitude in the pass band and the phase distortion of the two filters respectively. The Chebyshev has much steeper phase distortions than the Butterworth.

```

clear;
Fsamp = 10e3;
Fc = 3e3;
Fs = 34e2;
Wc = Fc/(Fsamp/2);
Ws = Fs/(Fsamp/2);
Wb = (Fc+200)/(Fsamp/2);
ftb = 400;
Stop_att = 35;
Pass_rip = 0.1;
[n, Wp] = cheb1ord(Wc, Ws, Pass_rip, Stop_att);

[b,a] = cheby1(n , Pass_rip, Wp);
[c, d] = butter(n, Wb);

[H1, W1] = freqz(b, a, 1024);
[H2, W2] = freqz(c, d, 1024);

F1 = W1*Fsamp/(2*pi);
mag1_db = 20*log10(abs(H1));

F2 = W2*Fsamp/(2*pi);
mag2_db = 20*log10(abs(H2));

% Calculate phase responses in degrees
phase1 = unwrap(angle(H1))*180/pi;
phase2 = unwrap(angle(H2))*180/pi;

% Create subplot layout
subplot(2,1,1);
plot(F1, mag1_db, 'blue');
hold on;
plot(F2, mag2_db, 'red');
hold off;
grid on;
xlabel('Frequency (Hz)');
ylabel('Magnitude (dB)');
title('Order 9 Chebyshev vs Butterworth Filter Response');
legend(["Chebyshev", "Butterworth"]);

subplot(2,1,2);
plot(F1, phase1, 'blue');
hold on;
plot(F2, phase2, 'red');
hold off;
grid on;

```

```

xlabel('Frequency (Hz)');
ylabel('Phase (degrees)');
legend(["Chebyshev", "Butterworth"]);

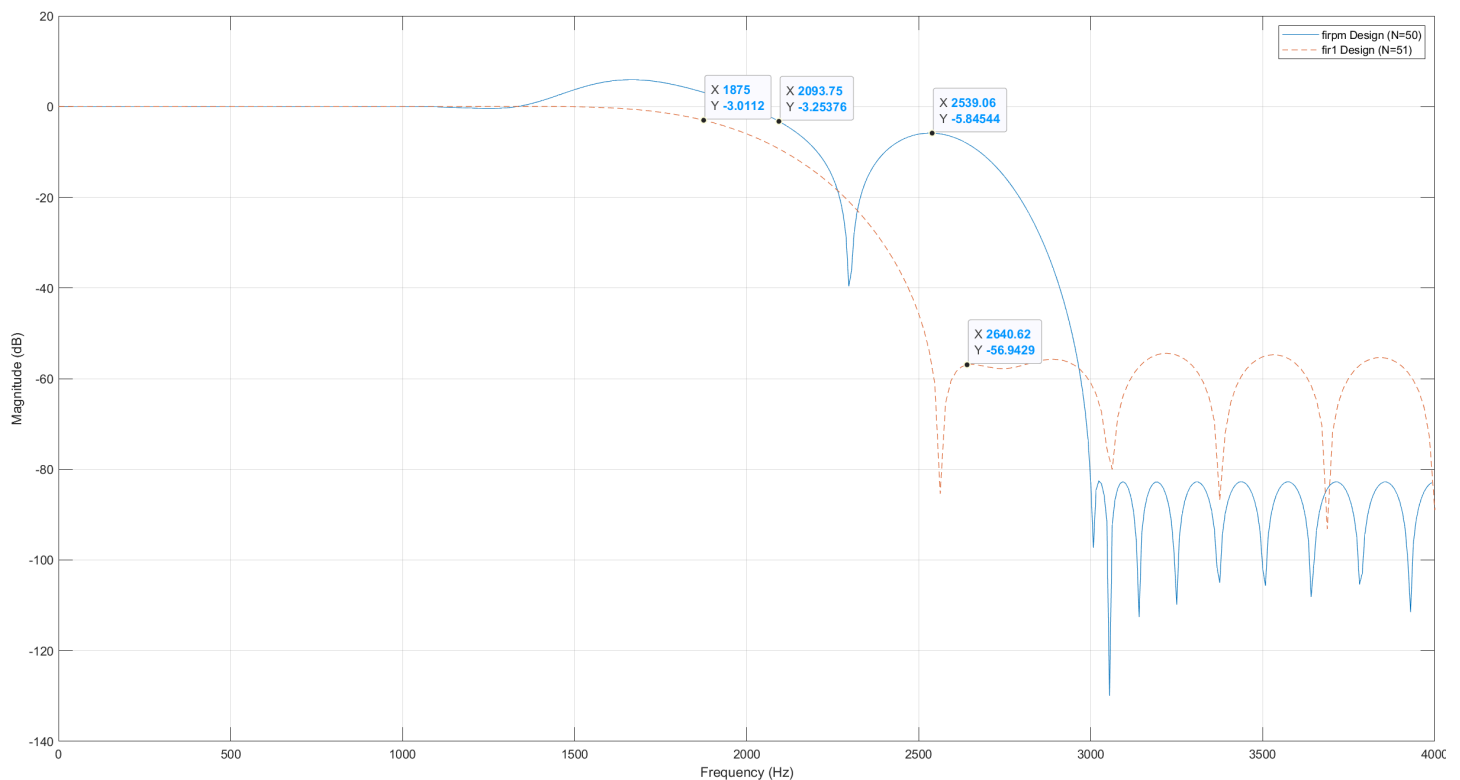
```

## Question 2 - FIR Filter

Using the `firpm` and `fir1` functions design a digital low pass filter with the following specifications:

- Sampling Frequency:  $F_{\text{samp}}$ : 8kHz
- Cutoff Frequency:  $F_c$ : 2kHz
- Transition Bandwidth: 300Hz
- Stopband Gain: 0 (Desired)

*Transition bandwidth*



The graph above shows the transition bandwidths for both the `firpm` function and the `fir1` function with the same order.

**firpm:**

- Bandwidth: ~450Hz
- Attenuation difference: ~2.6dB

**fir1**

- Bandwidth: ~765Hz
- Attenuation difference: ~54dB

Both filters have their own merit, with the `firpm` filter having a narrower transition bandwidth but attenuation difference not as steep as the `fir1` filter. The attenuation for each ripple after the first ripple is larger for the `firpm` filter compared to the `fir1` filter.

Finite impulse response filters have a linear phase property, which can be given by the following function for even order N:

$$\phi(\omega) = -(N/2 + 1)\omega$$

```
clear;
close;
Fs = 8e3;
Fc = 2e3;
Fbw = 300;
F = [0, 1000, 2000, 2300, 3000, 4000]; % Input frequencies
A = [1, 1, 1, 0, 0, 0]; % Input attenuations for given frequencies
Frat = F / (Fs/2); % Ratio of freq to nyquist

legend_strings = {};
for n = 20:5:50 % Order of filter where filter is n+1
    b = firpm(n, Frat, A);
    [h1, w1] = freqz(b, 1, 512);
    mag1 = 20*log10(abs(h1));
    plot(w1/pi*Fs/2, mag1);

end
legend_strings{end+1} = sprintf('firpm Design (N=%d)', n);
hold on;
n = 51;
c = fir1(n, Fc/Fs, 'low', hamming(n+1));
[h2, w2] = freqz(c, 1, 512);
mag2 = 20*log10(abs(h2));
plot(w2/pi*Fs, mag2, '--');
legend_strings{end+1} = sprintf('fir1 Design (N=%d)', n);

grid on;
legend(legend_strings);
xlabel('Frequency (Hz)');
ylabel('Magnitude (dB)');
xlim([0, 4000]);
```