Introduction to active filtering and the SR560 low-noise preamplifier

Eric D. Black
California Institute of Technology

October 29, 2014

1 Prelab exercises

1. Read Horowitz and Hill, Chapter 1, especially Sections 1.03, 1.07, 1.08, "Impedance and Reactance," starting on Page 29, Sections 1.18, 1.19, 1.20, 1.21, and 1.22. You will also find 1.09 and 1.14 exceedingly useful in future electronics work, but they are not essential for this lab. Note that they use the convention $j = \sqrt{-1}$, which is standard in electrical engineering. As a physicist, this should not confuse you, and you should be able to mentally switch between i and j as necessary, depending on the language required in the company you keep.

2. Starting with an input voltage $V_{in} = V_0 e^{j\omega t}$, calculate the *transfer function* V_{out}/V_{in} of the passive low-pass filter shown in Figure 1. Plot both the magnitude and phase of this transfer function as a function of frequency. You will find it useful to get into the habit of plotting the magnitude on a log-log scale, in units of decibels, and the phase, in units of degrees, on a semi-log scale, as shown in Figure 2. Define

$$f_0 = \frac{1}{2\pi RC}$$

This form is known as a *Bode plot*, after H. W. Bode of Bell Labs, its inventor (yes, you can get credit for inventing a way to plot things), and it will be quite useful later on, when you are analyzing transfer functions and feedback control.

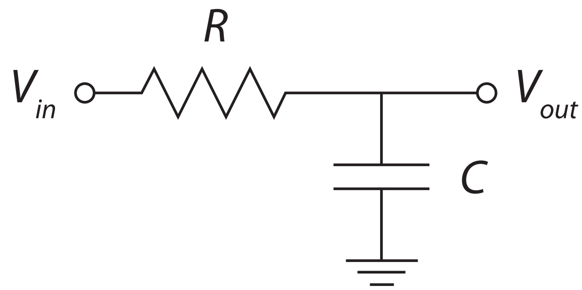

Figure 1: A passive low-pass filter.

3. Now do the same for a *high-pass* filter. If you are unfamiliar with high-pass filters, you can get one by swapping the resistor and capacitor in Figure 1.

4. What are the limiting behaviors of the magnitudes and phases of the transfer functions you just plotted in the regimes $f \gg f_0$ and $f \ll f_0$? (*Hint:* The magnitudes should limit to straight lines on your graphs in all four cases.) How would you express the slopes of these lines in units of decibels per octave? (An octave is a factor of two in frequency.)

5. Now imagine stringing two low-pass filters together, feeding the output of one to the input of the next. What would the transfer function of this "double filter" be? Is it the same as the product of the two individual-filter transfer functions? If not, why not?

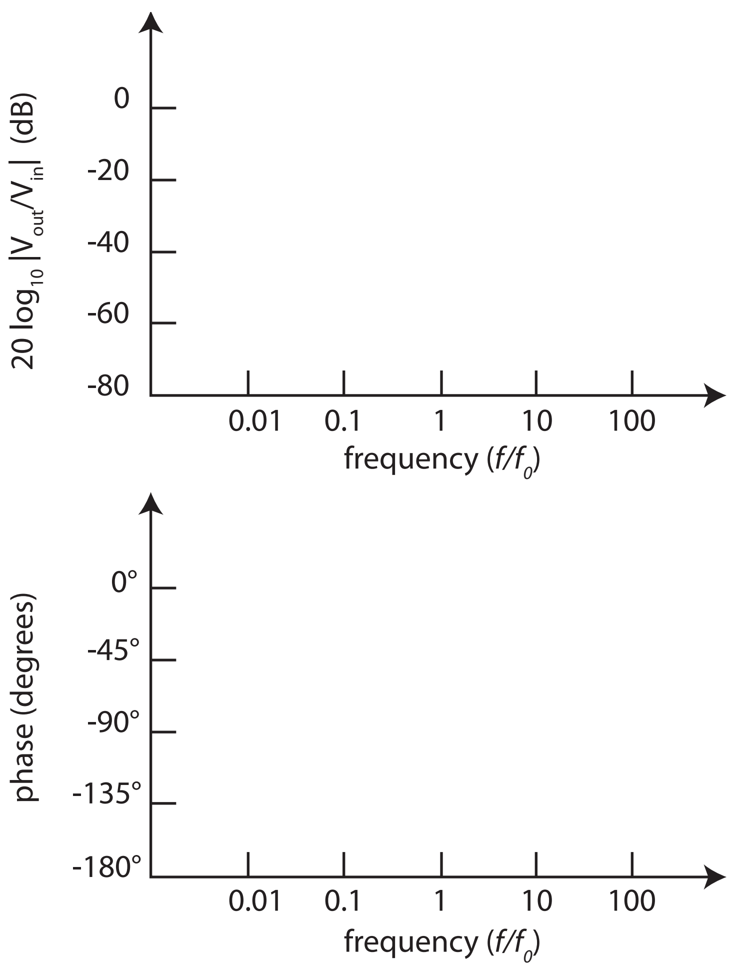

Figure 2: Standard form for a *Bode plot*, often used in communications and control engineering to describe the magnitude and phase of complex transfer functions.

2 Laboratory exercises

1. Examine the front panel of the SR560. Its operation should be obvious, if not completely then at least mostly. Explore the front and rear panels of the instrument, and make sure you know what everything does. If you have any questions, look to the manual first. Contact your TA if anything in the manual is not clear. This is how you would do it in a research setting, except there you often don't have anybody around to serve as a TA, short of calling the company. This is sometimes necessary and is done more often than you would think. You can call the company with questions for this lab if you want (it's free and makes for fine practice), but for now all you need to be aware of is that it is an option. Find the appropriate tech support phone number, and write it down in your lab notebook. Find the number to call for repairs, if it is different, and write that one down, too.

2. Program a basic low-pass filter with a single pole at 1 kHz and a gain of 10. Use the SR770 to measure its transfer function, including phase. Take a screenshot of this, and tape it in your lab notebook. Ideally, you want to use a dual screen display for this, so that you get a Bode plot directly on the screen.

3. Use the cursor to measure the 3dB breakpoint (whose definition you should remember from your Horowitz and Hill reading), and verify that it is where you expect it to be. Measure the amplitude of the signal at low frequency, and verify that the gain is what you asked for. Verify the slope of the amplitude at high frequencies (above the rolloff), and verify that it is what you expect. Check the phase in the limits of low frequency, high frequency (compared with the pole frequency), and exactly at the 3dB breakpoint. Does it agree with what you expect?

4. Change the number of poles from one to two (6dB/oct rolloff to 12dB/oct). How does this change the transfer function?

5. Now make a crude bandpass filter. Program a high-pass filter with a cuttoff frequency of 30 Hz, and a low-pass filter with a cuttoff of 300 Hz. What amplitude do you get in the "middle" of this range, *i.e.* 100 Hz. What is the maximum amplitude of your transfer function, and

how does it compare with the expected gain? What is the behavior of the phase in the passband?

6. Look up the specification for the input-referred noise of the SR560 in its manual, and make a plot of the expected noise at the output vs. gain in the passband. Do this for gain values starting at 1 and going up to as high as you can get without overloading. Try this with and without a 50 Ω terminator on the input. Does it make a difference?

7. Measure the noise floor of the SR560 as a function of gain, and compare it with your prediction. In what regime does your measurement agree with the SR560's specs? Make sure you are really measuring the noise of the SR560, and not the input noise of the SR770!

8. Read Horowitz and Hill, Chapter 5, starting with the introductory material on Page 263 and continuing with Sections 5.01, 5.04, and 5.05, but stop at "Active Filter Circuits" on Page 272. This is an introduction to Butterworth, Chebyshev, and Bessel filters. Note that the authors feel the advanced-filtering material is optional, whereas the sections on oscillators is essential. I would argue the reverse, that those priorities have switched in the years since the book was written. Today, every experimentalist I know either buys their oscillators commercially, or has a staff of electrical engineers associated with his or her project to design and build them. Advanced filters, in my experience, are more widely used now than in the past, but like oscillators we don't build them ourselves. You should understand what they are and what they can do for you, so that you can specify the purchase of one when you need it.

   Locate the SR640 programmable elliptic filter in the lab. Look up the formula that describes the behavior of an elliptic filter, and note the conditions under which it replicates that of a Butterworth. (The manual may also be of some use to you here.) See if you can program a single-pole Butterworth filter, and measure its transfer function using the Network Analyzer. Compare your data with your prediction.

   If you are interested and want to build one of these active filters it's not hard. Horowitz and Hill describe how to do it starting on Page 272, using voltage-controlled voltage-source (VCVS) op-amp circuits. You are welcome to do it if you like, but in my opinion it is of limited educational value for a practicing physicist. Unless you can prototype it really quickly, you're better off concentrating on the math behind what these filters do.

9. Imagine you are a physicists attached to a big project and have to specify what kind of filter to use in a particular subsystem. What design would you choose to achieve a maximally-flat passband? How about a "sharpest knee" in the amplitude response as a function of frequency? A least-disruptive time delay?

References

[1] Paul Horowitz and Winfield Hill, *The Art of Electronics, Second Edition*, Cambridge University Press, Inc., (1989).

[2] *Model SR560 Low-Noise Preamplifier*, Stanford Research Systems, Sunnyvale, California (1989). Available online at http://www.thinksrs.com/downloads/PDFs/Manuals/SR560m.pdf.

[3] *Model SR640 Dual-Channel Filters*, Stanford Research Systems, Sunnyvale, California (1989). Available online at http://www.thinksrs.com/downloads/PDFs/Manuals/SR640m.pdf.