EE 218 LINEAR SYSTEMS II LABORATORY
Missouri University of Science & Technology

LAB 6: HIGH-PASS FILTER FREQUENCY RESPONSE

Aim: In this lab you will construct a high-pass filter and measure its amplitude response using a variety of methods.

Introduction: Build a simple R/C high-pass filter in the lab. Design the filter to have a cutoff frequency of 1 kHz. You should have learned how to select the RC time constant in circuits or linear systems courses. If you want this filter to be close to ideal, you need to think about some of the following things:

- Use commonly available capacitor and resistor values. It is easy to find low-power resistors from a few ohms to MOhms. Small, inexpensive, capacitors are usually in the range of of 100's of pF to a few uF.

- Use unpolarized capacitors, such as ceramic disk caps. Do not use electrolytic caps.

- In classes we often assume the signal source is an ideal voltage source. In practice, every signal source will have an output impedance. You can ignore the output impedance of the signal source if the input impedance of your filter is much higher (10 times higher or more). The series combination of the source-output-impedance and the filter-input-impedance will still look like a voltage divider, but the voltage drop across the output impedance will be very small.

- In classes we assume the output of the filter is connect to something with infinite input impedance. All practical devices have a finite input impedance. You can ignore this, if the measuring-device-input-impedance is much higher than the output impedance of the filter. If you are not sure this is the case, insert a unity-gain op-amp circuit between your filter and any other device.

- Resistor and capacitor values differ from their specified values. Depending on the precision specified on the device, the actual resistance/ capacitance may be up to 20% different from the one marked on the device.

Procedure: Once you have constructed the filter, measure it's frequency response. There are a variety of ways to do this. Use at least one of the following methods (i.e. either 1. or 2.):

1. Generate an impulse, measure the impulse response, then Fourier Transform the result. You cannot generate a true Dirac Impulse function in the lab, but you can approximate it with a very narrow square pulse. You may be able to generate this narrow pulse using a pulse generator, the impulse output at the back of the Fourier Analyzers, generate an impulse in Matlab and send it out through the sound card, or generate a signal

with periodic impulses and burn a CD. In all of these cases you will want to generate a series of impulses. Make sure the time between the impulses is much longer than the impulse response of your filter. You can record the impulse response on a digital scope, or with the sound card in the PC.

2. Generate a step, measure the step response, calculate the Fourier Transform of the step response. An impulse is the derivative of a step, so you can use Fourier Transform Theorems to calculate the Fourier Transform of the impulse response from the Fourier Transform of the step response. You can generate steps using a square wave. Make sure the time the square wave spends high/low is much longer than the impulse response of the filter. The two methods mentioned above are sometimes difficult to use. Noise in the measurement may goof up your result. In addition, we sometimes only care about the amplitude response of the filter, and not the phase response. In these cases, it is often easier, and more reliable, to use the following methods:

3. Swept Sine Wave. Using a signal generator, or the PC, generate a sine wave that sweeps linearly from a frequency that is 10% of the filter cut-off frequency to 10 times the cut-off frequency. Drive your filter with this signal. Measure the amplitude of the output signal, as a function of the input frequency. You can measure this with an envelope detector -which is a diode, resistor and capacitor. See your instructor for a description of how to build one of these devices in either hardware or Matlab.

4. Noise Measurement. Feed white noise into your device, and measure the spectrum the same way you measured the spectrum of white noise. Be sure to use averaging. You can do this either in Matlab, or using the spectrum analyzer in the lab.

For Your Report:

- Parts 1-4: Make sure to write down the amplitude spectrum data for each method you use.

- Make sure to include the calculations on how you chose R and C to build the desired high pass filter.

- Calculate and plot the theoretical amplitude response on the same figure as the amplitude responses you measured during lab. Make sure to scale the plots so they line up correctly. How does the theoretical amplitude response compare to the measured amplitude responses?

- Make sure you tabulate the 3dB point of your filter for each method you used.

- Perform an error analysis on the filter. See how the theoretical amplitude spectrum changes if RC is 20 % too high or too low. Include these plots on the same figure with the theoretical amplitude spectrum in your report. What range of 3dB points do you obtain? Do the 3dB points from your measured data fall inside this range?
