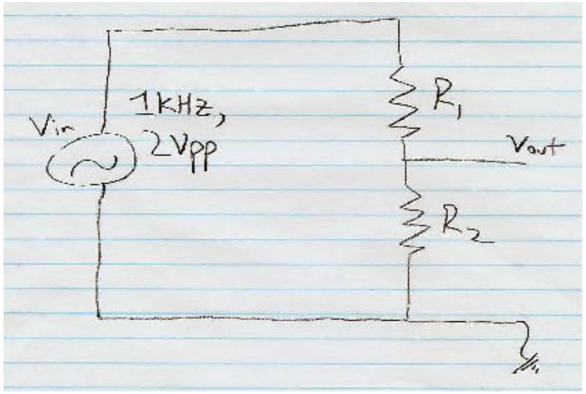
Physics 364 – fall 2010 – lab #2 – due by lecture, Monday 2010-09-27

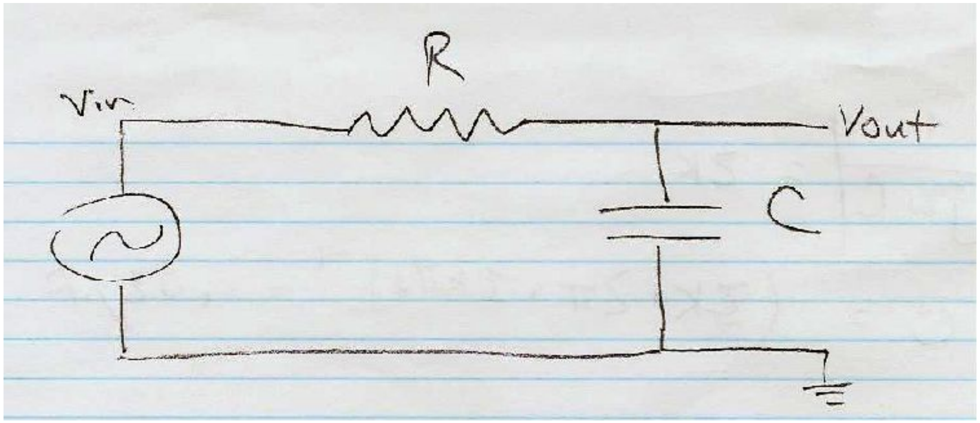
The main focus of this week's lab is to generalize the purely resistive voltage dividers we met last week to build filter circuits, whose response varies with frequency. We generalize resistance (a real quantity) to *impedance* (a complex quantity), which allows us to include capacitors and inductors in a straightforward way. We'll also build a few more circuits that use diodes, for which V=IZ does not hold. After building a variety of circuit fragments in the early parts of Lab 2, you will piece together several of these fragments – a bandpass filter, a diode rectifier, a lowpass filter – to make a simple AM radio receiver. The idea is to show you that these circuits have some relevance in the real world.

<u>Part 1.</u> Build the voltage divider shown below, using R1=R2=2.2K. (Note that the 2.5K used in the homework is not a standard component value. If you prefer to use your homework calculations unmodified, then feel free to combine two resistors to make 2.5K. Be sure to note in your write up whether you are using 2.2K or 2.5K.) Sketch the input and output waveforms. If your function generator and oscilloscope do not agree on the magnitude of Vin, check that the function generator is configured to drive a "high impedance" load rather than a 50 ohm load.



<u>Part 2.</u> Now replace R2 with a .068μF capacitor, to form a *low pass* filter. Measure |Vout/Vin| for f=0Hz (i.e. for DC), f=100Hz, f=1kHz, f=10kHz, f=1MHz. What is f_{3dB}? Measure |Vout/Vin| at 0.5×, 2×, 4×, 8× f_{3dB}. Do you see that beyond f_{3dB}, the response falls (asymptotically) by *6dB/octave*, or *20dB/decade*? (An octave is a factor of two in frequency, as in music, and a decade is a factor of ten.)

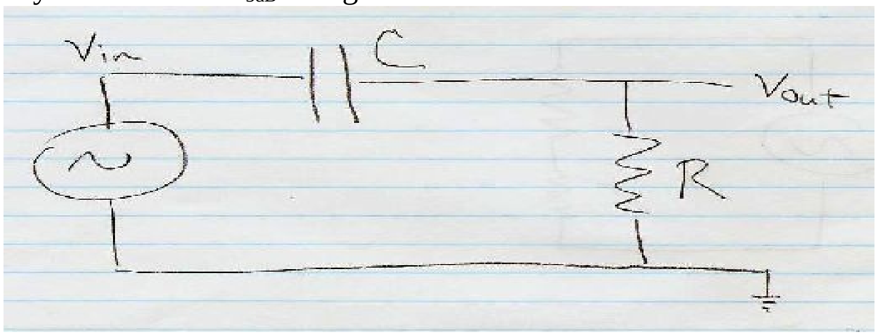
Look on the oscilloscope at the relative phase between Vin and Vout. Which one leads at $f \ll f_{3dB}$? Which one leads at $f \gg f_{3dB}$? How does the phase difference vary from low f to f_{3dB} to high f?



Part 3. What is the output impedance of the low-pass filter from part 2? What happens to |Vout/Vin| if you place a 220K load resistor in parallel with C? Try this for a range of frequencies. Now what happens if you place a 2.2K load resistor in parallel with C? What is |Vout/Vin| at DC now? What is f_{3dB} now? Can you understand the qualitative difference between the 220K case and the 2.2K case by sketching a voltage divider between the filter's output impedance and the filter's load resistor?

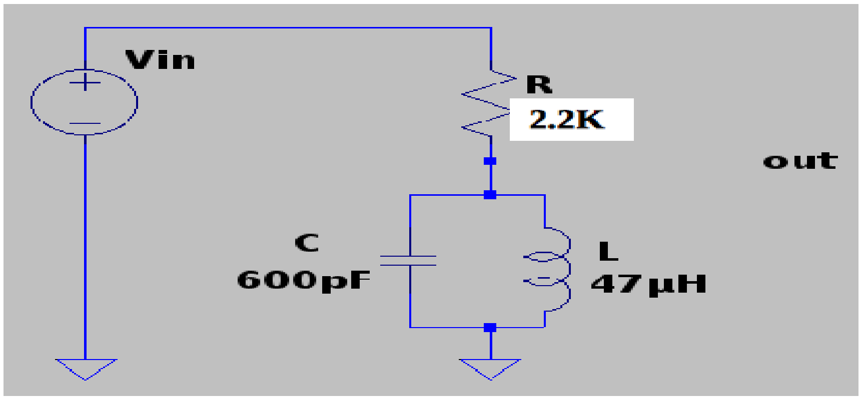
Part 4. Now interchange R and C from part 2, so that you have a *high-pass* filter, with R=2.2K and C=.068μF. Measure |Vout/Vin| for DC, f=100Hz, 1kHz, 10kHz, 1MHz. What is f_{3dB} ?

Measure |Vout/Vin| at 2×, 1×, 0.5×, 0.25×, and 0.125× f_{3dB}. Look on the oscilloscope at the relative phase between Vin and Vout. Which one leads at $f \ll f_{3dB}$? Which one leads at $f \gg f_{3dB}$? How does the phase difference vary from low f to f_{3dB} to high f?



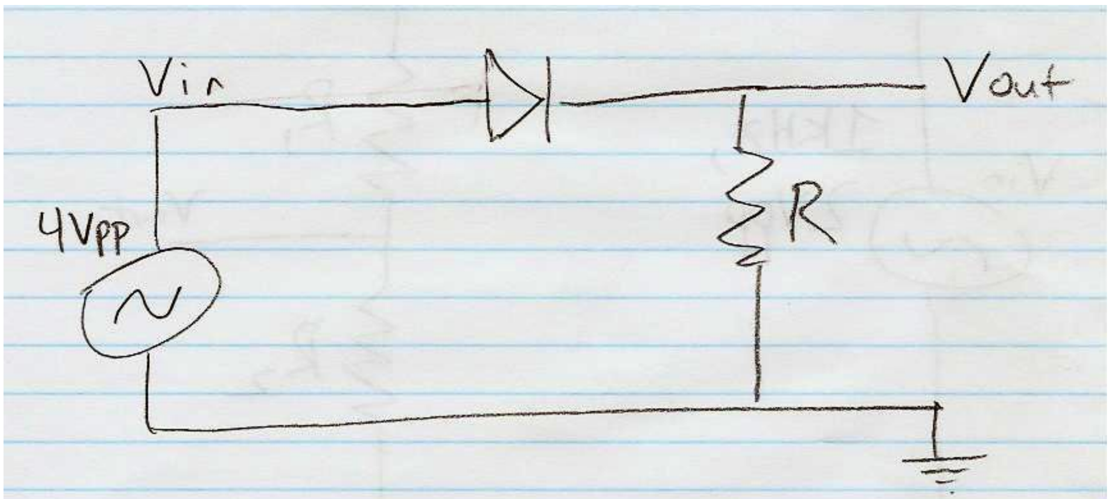
Part 5. Now make a *band-pass* filter, by replacing R2 in part 1 with the parallel combination of a capacitor and an inductor. Take R=2.2K, C=600pF, L=47μH. Measure f_0 , where |Vout/Vin| is maximal. Measure the 3dB bandwidth, i.e. Δf between the two frequencies at which |Vout/Vin| is 3dB below the peak. Try changing R, L, and C, one at a time, by a factor of two (in whichever direction is convenient), and observe the change in the peak position and in the bandwidth.

If you want to try to do this in a quick graphical way, try experimenting with *sweep* mode of the function generator. If you do this, you'll also need to trigger the scope with the function generator's *SYNC* output.



Part 6. Let's again modify the voltage divider from part 1, but this time replacing R1 with a diode, such as 1N914 or 1N5817. (The latter should have a smaller voltage drop, which is handy for rectifying small signals or large currents.) Using a 4Vpp sine wave (i.e. 2 volts in amplitude) for Vin, measure Vin and Vout, and compare with the sketch you made in Homework #2.

On page 41 of the HP33120A user guide (available at positron.hep.upenn.edu/wja/p364 if you haven't found it elsewhere), you will find instructions for generating an amplitude-modulated waveform. Use a 950kHz carrier, 4Vpp, and a 1kHz modulation frequency with 25% modulation depth. If you set the horizontal scope scale to about 500μs/division, can you see the modulation on both input and output waveforms? How far can you dial down the amplitude and still see a signal?



Part 7. Now let's piece together several of the circuit fragments from this lab to make a simple AM radio receiver. For starters, keep the rectifier and AM signal generator from part 6 intact. Put the bandpass filter from part 5 between the signal generator and the rectifier. Move the carrier frequency up and down a bit and look at the response to make sure that the peak response is around 950 kHz – or at least somewhere in the AM radio band 540—1600kHz. Now replace the rectifier's load resistor with a lowpass filter with f3dB ~ 10kHz. Probe with the scope before and after the lowpass filter to check that the filter separates the 1kHz modulation from the radio frequency carrier. If you see a 1kHz signal at the end of the chain, try connecting a speaker. You should hear a tone pretty close to the B two octaves above middle C. If this all works, you're ready to try replacing the signal generator with Bill's ad-hoc radio antenna.

Ask Bill or Jose when you're ready for the speaker and antenna.

Note that we deliberately do not provide a schematic diagram for this part, so that you have to think for yourself about how the pieces go together.

Note also that I (Bill) cheated a bit and amplified the signal coming out of the antenna (using a circuit that you will build in Lab 3), because my antenna's output on its own was not large enough to rectify easily – i.e. its amplitude was smaller than a diode drop.

Here is a puzzle for you to ponder: Would the lowpass filter separate the radio-frequency carrier from the audio-frequency modulation without the rectifier? Why does it work after the rectifier? (You don't have to write this part up – just think about it.)

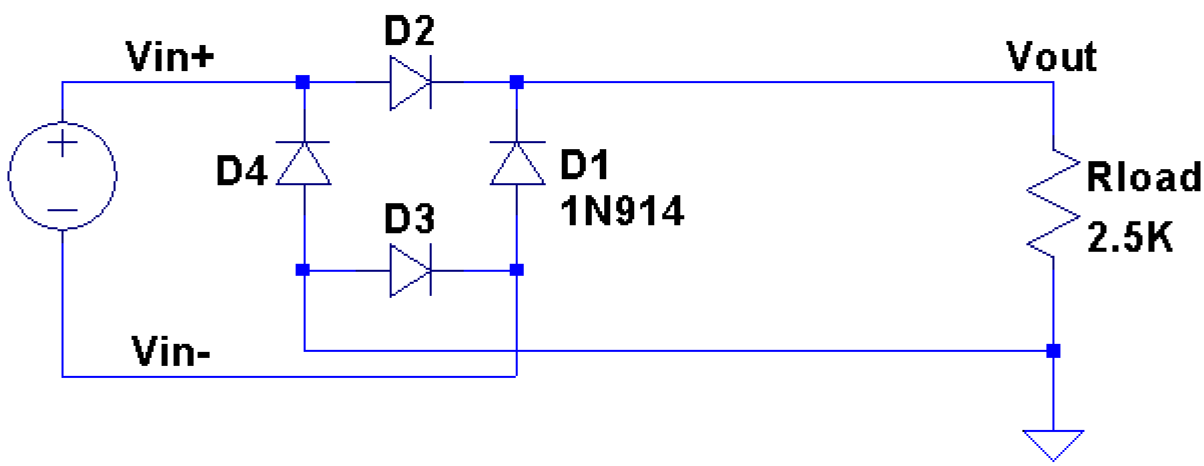
Here is a hint: $\int dt\ \{1-A\cdot\cos(2\pi t\cdot 1kHz)\}\cdot\cos(2\pi t\cdot 1MHz)\cdot\cos(2\pi t\cdot 1kHz)=0$,
while $\int dt\ \max[0, \{1-A\cdot\cos(2\pi t\cdot 1kHz)\}\cdot\cos(2\pi t\cdot 1MHz)]\cdot\cos(2\pi t\cdot 1kHz)\neq 0$.

There is no 1kHz component in the original waveform: its fourier transform shows peaks only at 949kHz, 950kHz, and 951kHz. (The 949 and 951kHz peaks are called *sidebands*.) The nonlinear diode *intermodulates* the frequencies that appear in the original signal, producing frequencies that are sums and differences of frequencies in the original spectrum – i.e. 1kHz. A linear circuit will only respond at frequencies at which it is driven. Only a nonlinear circuit can produce frequencies in the output that do not appear in the input.

Part 8. The electricity that comes out of most wall outlets in North America takes the form of a sinusoidal voltage source, 110 volts rms, 60 Hz. Imagine that you are trying to turn this AC line voltage into 5 volts DC to power your mp3 player. In real life, we would first use a transformer to get the AC line voltage into a more convenient (and safer) range, say 10 volts in amplitude. Instead, you will start by programming your function generator to output a 60 Hz sine wave, 10 volts in amplitude.

Now we need to turn this into a voltage whose average value exceeds +5V. (You'll learn later in the course how to turn an unregulated 6 or 7 volt power supply into a regulated 5 volt supply. You actually saw such a device inside Mystery Box #2 in Lab 1.)

We could start with the circuit in part 6, but it has a significant drawback: it uses only half of each 60Hz cycle. Instead, build the *full-wave bridge rectifier* drawn below. Measure and sketch Vin and Vout.



Now add a capacitor (the *filter* capacitor) in parallel with the load. Try a few capacitor values – starting from the value you calculated in Homework #2 and going up and down by a factor of 2 or more – and see how the amplitude of the 120kHz droop is affected. How do your measurements compare with what you calculated in the homework? The measured droop should be a bit smaller than you calculated, if in the homework you approximated the rectified sine wave as a series of narrow spikes.

If you reduced Rload by a factor of two, by how much would you have to change the filter capacitor in order to achieve the same (or smaller) droop?