Physics 364, Fall 2014, Lab #5 **Name:** ______________________

(scope probe, RLC resonant filter, more on filters)

Monday, September 15 (section 401); Tuesday, September 16 (section 402)

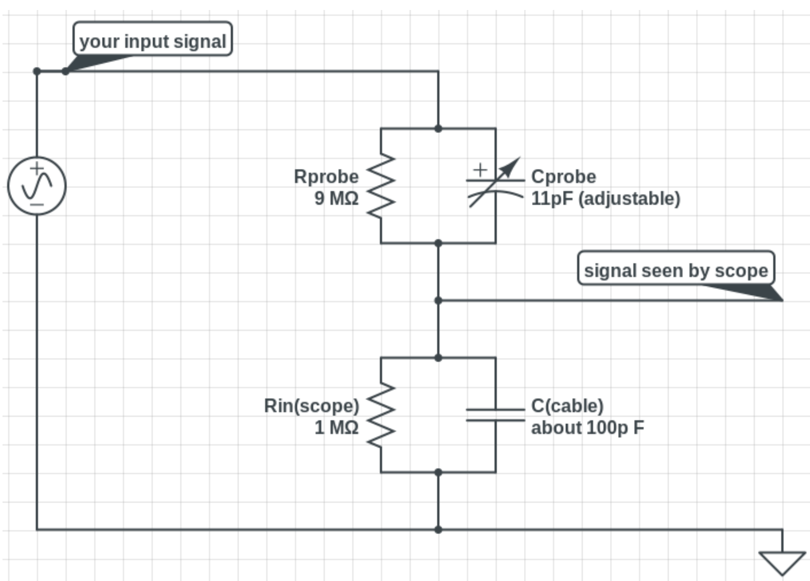
Course materials and schedule are at `positron.hep.upenn.edu/p364`

Part 1 **Start Time:** ____________

scope probe "compensation" (time estimate: 15 minutes)

As you read in this past weekend's notes (and saw briefly the weekend before as well), a scope probe achieves two aims. First, it increases the scope's R_{in} from 1 MΩ to 10 MΩ. More importantly, it cancels out the effect of the $\sim$ 100 pF capacitance of the $\sim$ 1 m-long cable.[1] The capacitance (about 30 pF/foot) of the cable plugged into your scope appears in parallel with the scope's 1 MΩ input resistance, since the coaxial cable consists of two concentric conductors: the outer conductor is grounded, and the inner conductor is observed by the scope. At frequencies around 1 MHz, the impedance of the cable capacitance is $|Z_C| \sim 1.6$ kΩ, which would be a disastrously small input impedance for the scope. The unwanted result is a frequency-dependent attenuation and phase shift.

The cure is to make the scope probe's impedance (both real and imaginary parts) be precisely 9× that of the scope+cable. That implies that the probe needs 9× the resistance of the scope and $\frac{1}{9}\times$ the capacitance of the cable. The probe's capacitance can be adjusted[2] to get the ratio just right, so that $\frac{\mathbf{Z}_{\text{scope+cable}}}{\mathbf{Z}_{\text{probe}} + \mathbf{Z}_{\text{scope+cable}}} = \frac{1}{10}$ is purely real, with no phase shift or frequency dependence.



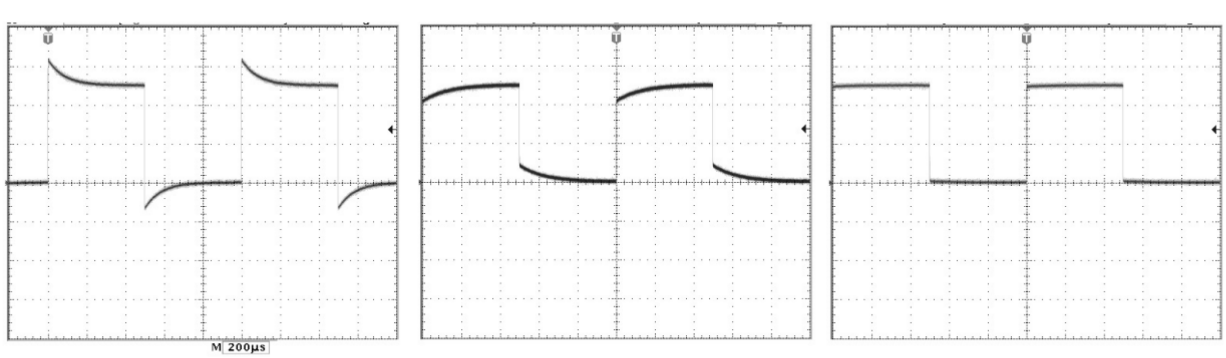
[1]This section parallel's Tom Hayes's notes 3N.2.

[2]In some probes, the adjustable capacitance is near the probe tip, as I've drawn in the figure. In other probes, the adjustable capacitance is next to the scope. Our scope probes are of the second type. In either case, the key idea is to turn the ratio of top and bottom impedances into a purely real number, by adjusting a small capacitor.

1.1 Connect the probe as shown in the photo above. The lower-right clip on the scope outputs a 1 kHz square wave, while the upper-right clip is grounded. Adjust the probe with a small screwdriver until the square wave looks square. First try turning the screw in one direction to exaggerate the high frequencies, then in the other direction to attenuate the high frequencies, and then finally to get the square wave just right.

Be sure to check and (if needed) adjust both of your scope probes.

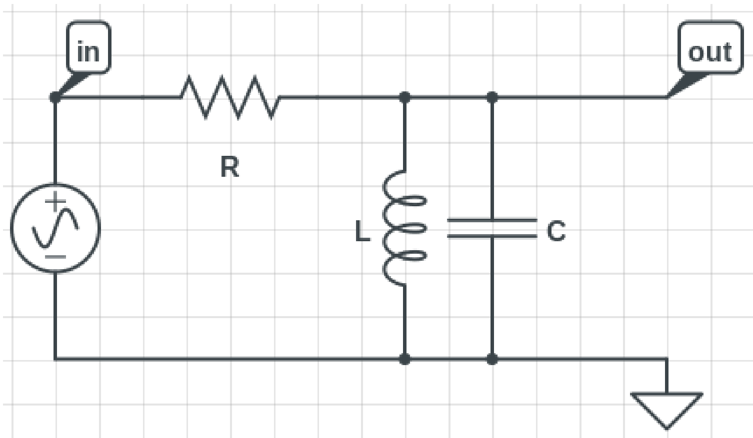
Try to imagine how happy Joseph Fourier would be to see you turning a tiny screwdriver to tweak a capacitor, until your eye confirms that the f, $3f$, $5f$, $7f$, $\ldots$ components of a square wave appear in the perfect 1, $\frac{1}{3}$, $\frac{1}{5}$, $\frac{1}{7}$, $\ldots$ proportions! **Now that you know how to adjust your scope probe, you'll want to use it, in $10\times$ mode, for your measurements going forward.**

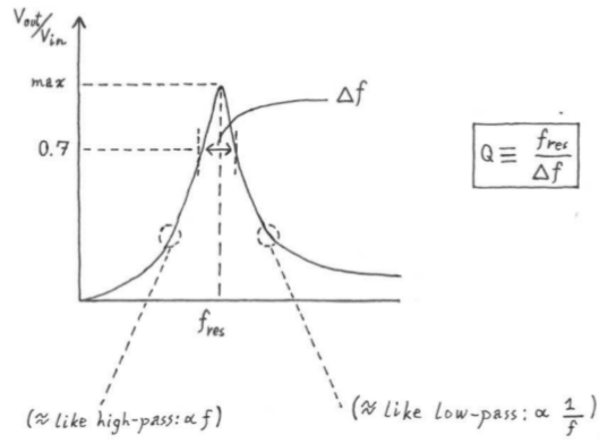


overstates high frequencies... ...understates high frequencies... ...treats highs and lows alike (keeps 10%)

Part 2
RLC band-pass filter

Start Time: ________________
(time estimate: 60 minutes)





On page 7 of last week's notes ("`reading02.tex`" due 9/7), we saw the *band-pass* filter shown above (left). It again looks like a generalized voltage divider, where the impedance on top is $\mathbf{Z}_1 = R$ and the impedance on the bottom is

$$\mathbf{Z}_2 = \mathbf{Z}_L \parallel \mathbf{Z}_C = j\omega L \parallel \frac{1}{j\omega C} = \frac{j\omega L}{1 - \omega^2 LC}.$$

The full expression for

$$\frac{\mathbf{V}_{\text{out}}}{\mathbf{V}_{\text{in}}} = \frac{\mathbf{Z}_2}{\mathbf{Z}_1 + \mathbf{Z}_2} = \frac{\mathbf{Z}_2}{R + \mathbf{Z}_2}$$

is messy, but it's easy to look at a few limiting cases. You can see that $\mathbf{Z}_2 \to 0$ for both $f \to 0$ and $f \to \infty$, so V_{out} should vanish at both of these limits. And you can see that $|\mathbf{Z}_2| \to \infty$ at $\omega = \frac{1}{\sqrt{LC}}$, so V_{out} should be maximized at the resonant frequency

$$f_0 = \frac{1}{2\pi\sqrt{LC}}.$$

Ideally $\frac{|V_{\text{out}}|}{|V_{\text{in}}|} = 1$ at f_0, but real-world inductors have some finite series resistance, which makes the peak wider and less tall. With ideal components, the bandwidth (illustrated above, right) is

$$\Delta f \approx \frac{1}{2\pi RC}$$

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2.1 First build the bandpass filter from the previous page, using $R = 100\ \text{k}\Omega$, $L = 10$ mH, $C = 10$ nF. Now imagine your (resonant) filter to be the Liberty Bell (before it cracked) and whack it with a big 10 V_{pp} square wave at about 100 Hz. At what frequency does it ring? Is it reasonably close to your calculated f_0? How quickly does it decay to $1/e \approx 37\%$ of its initial amplitude? Is the decay time reasonably close to RC? (Probably τ will be somewhat shorter than RC because of the inductor's internal series resistance.)

2.2 Now measure your filter's frequency response using a sine wave input. Start out at $f \approx f_0$, and use the knob on the function generator to move the frequency up and down in small steps. You should be able to see where V_{out} is maximal, and how far you have to move f to get V_{out} to fall to $\frac{1}{\sqrt{2}} \approx 0.707$ of its maximum value. Does it peak near the expected f_0? Is Δf about right (e.g. $\Delta f = 1/(2\pi\tau)$, where τ is the decay time found in part 2.1)? (When I tried it, I got OK but not great agreement for Δf.) Sketch a quick (not too elaborate) graph of $\|\frac{V_{\text{out}}}{V_{\text{in}}}\|$ vs. f, indicating f_0 and Δf. (Next page is blank to leave room for this.)

(This page is blank in case you need the space to draw a graph.)

2.3 Try replacing the 100 kΩ resistor with 10 kΩ and then with 2 kΩ, and see if the effect on Δf (or on $Q = f_0/\Delta f$) makes sense. Go back to 100 kΩ when you're done.

2.4 Make sure you're back to the original $R = 100$ kΩ resistor choice. Next, try adding one or more additional capacitors in parallel with C and see how it affects f_0. In a future lab, we're going to tune this circuit (but set up for $f_0 \approx 1$ MHz) to find a nearby AM radio station.[3] Reducing f_0 by about 10% is most easily done by increasing C by about 20% (because of the square-root), etc. Remember that capacitors add in parallel.

[3]The AM broadcast band goes from 540 kHz to 1610 kHz, in 10 kHz steps.

2.5 The function generator has a handy feature for studying frequency-dependent circuits: it can repeatedly sweep through a specified range of frequencies.[4] By watching the output of the circuit during one sweep, you can see the shape of V_{out} vs. f.

On the Rigol function generator, select a 5 V_{pp} Sine. Attach a BNC cable from the *sync* output on the back of the FG and plug this cable into CH3 or CH4 of the scope. Only CH1 of the FG is synchronized with the *sync* output, so use CH1 for $V_{\text{in}}(t)$ to your RLC circuit.

To activate the sync feature, you have to turn it on using the "utility" menu. Select "sync on." Note that this selection is deactivated every time you turn off the FG.

To get a better understanding of signal produced by the FG during a single sweep, use a BNC cable to connect CH1 of the FG to CH1 of the scope. Display the sync signal as well, which should be a 5 V square pulse. **Set the scope to trigger on the sync signal.**

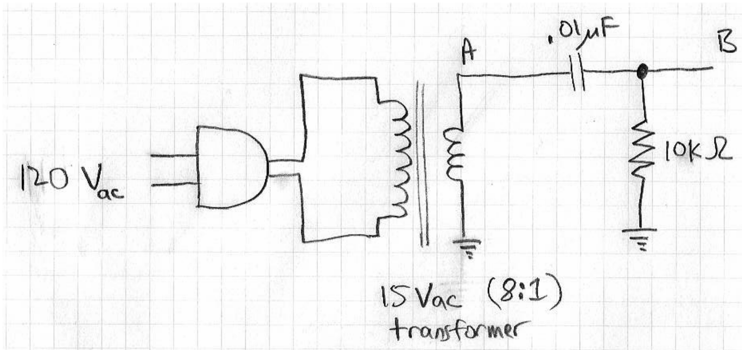
Now select Sweep on the FG. Select a Start frequency of 100 Hz, a Stop frequency of 400 Hz, and a Time of 100 ms. Set the horizontal scale of the scope to 20 ms/div. You should see a sine wave the starts with a frequency of 100 Hz on the left-hand side of the display with a frequency that increases to 400 Hz as the waveform progresses from left to right.

Now attach the FG to the input of your RLC circuit and sweep from 5 kHz to 25 kHz (not Hz) in a time interval of 100 ms. Again look with 20 ms/div on the oscilloscope. Display the output of your circuit on CH2. Find the time that corresponds to the peak in the trace that you observe. If this time is, for example, 50 ms, then this time would correspond to a frequency of 15 kHz. In a similar way, check that the bandwidth roughly agrees with the values you found above. For the bandwidth measurement, you may want to sweep over a narrower frequency range.

[4] Text for this section adapted from Joe Kroll's 2013 Lab5.

Part 3 **Start Time:** ________________

"Garbage detector" for AC line voltage (time estimate: 15 minutes)[5]



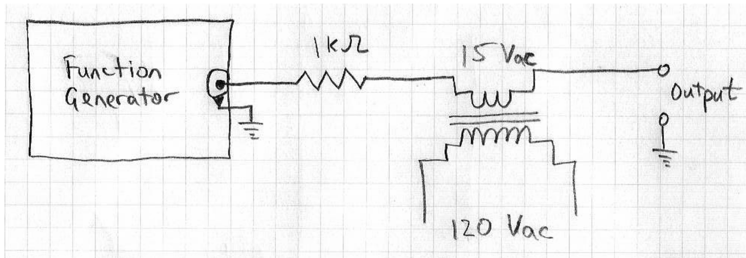
3.1 The circuit shown above will let you look (safely!) at the waveform from the 120-volt power line. (The nominal rms voltage for U.S. electric mains is $120 \pm 6\mathrm{V_{ac}}$, which implies an amplitude of 170 V, or 339 $\mathrm{V_{pp}}$.) The 8:1 transformer reduces 120 $\mathrm{V_{ac}}$ to a safer 15 $\mathrm{V_{ac}}$, and it "isolates" the circuit we're working on from the potentially lethal power-line voltage. You'll need to plug the 15 $\mathrm{V_{ac}}$ transformer into the power strip under your desk. **Be very careful** not to short the two transformer leads together, as this will quickly melt the insulation off of the transformer's coiled wire, destroying the transformer. We need at least 10 transformers to survive both of this week's labs. Looking at point **A**, you can see the power-line waveform ($\times\frac{1}{8}$). It should look more or less like a classical sine wave. What is its frequency? Does the amplitude make sense to you, for 15 $\mathrm{V_{ac}}$?

[5]Part 3 is borrowed from Lab 2L.2.3 of Tom Hayes's *Physics 123*.

3.2 To see glitches, wiggles, and other high-frequency imperfections in this waveform, look at point **B**, the output of the high-pass filter. A variety of interesting stuff should appear, some of which may come and go with time. (Jose says he can see SEPTA trains go by.) You'll get a clearer picture if you push acquire $\rightarrow$ mode $\rightarrow$ hi res on your scope. What do you predict for the filter's attenuation at 60 Hz? No complex arithmetic is needed here: just use the fact that for an RC highpass filter at $f \ll f_{3dB}$, the amplitude ratio approaches $|\frac{V_{\rm out}}{V_{\rm in}}| \rightarrow \frac{f}{f_{3dB}}$. Does the observed amplitude of the 60 Hz signal at point **B** make sense? If you have time, you might also try out the scope's FFT mode.

Part 4 **Start Time:** ______________

Selecting signal from signal + background (time estimate: 30 minutes)[6]



4.1 Now let's try using first a high-pass filter and then a low-pass filter to prefer one frequency range or the other in a composite signal, formed as shown in the figure above. The transformer adds a 60 Hz sine wave, about 42 V_{pp}, to the output of the function generator. Set the function generator to around 10 kHz. To choose the R value for your filter, below, you will need to determine the "output impedance" (a.k.a. R_{thev}) for the signal source you have constructed, i.e. function generator + transformer. The function generator's R_{out} is 50 Ω. The series impedance of the transformer winding is negligible at the frequencies of interest to us. The 1 kΩ resistor is included, incidentally, to protect the function generator in case the composite signal accidentally is shorted to ground. First make a reasonable guess for R_{out} (a.k.a. R_{thev}) of your composite voltage source (labeled "output"). Then check your guess by loading the output with a resistor value that you expect to divide the output signal roughly in half. (Remember that if $R_{load} = R_{thev}$, then V_{out} becomes $\frac{1}{2}V_{OC}$.) When you're done checking, remove the load resistor.

[6] Borrowed from Tom Hayes's lab 2L.2.4.

4.2 Now **design** a high-pass filter (to place just after the output of your composite signal source) that will keep most of the 10 kHz "signal" and get rid of most of the 60 Hz "background." As you design, consider (a) what is an appropriate f_{3dB}, and (b) what should be the minimum input impedance Z_{in} of your filter, to obey our $\times 10$ rule of thumb.

Draw the schematic diagram for your HPF design. Run the composite waveform ("signal" plus "background") through your high-pass filter. Do you like the output of your filter? Is the attenuation of the 60 Hz waveform about what you would expect? (In many real-life circuits, the 60 Hz power lines are a common source of unwanted "background" or "noise.")

4.3 Now let's change assumptions. Let's suppose that we consider the 60 Hz to be the "signal," and the function generator's 10 kHz to be the "background." Design a low-pass filter (to replace your high-pass filter) that will keep most of the "signal" and get rid of most of the "background." Draw your low-pass filter design. Now run the composite signal through your LPF and see if you like the result. If not, fix your design!

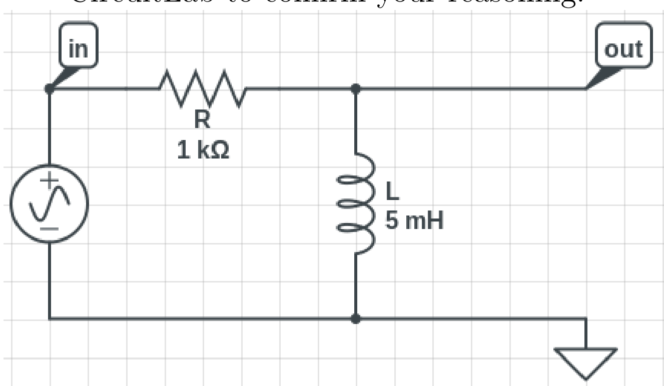
Part 5 **Start Time:** ______________

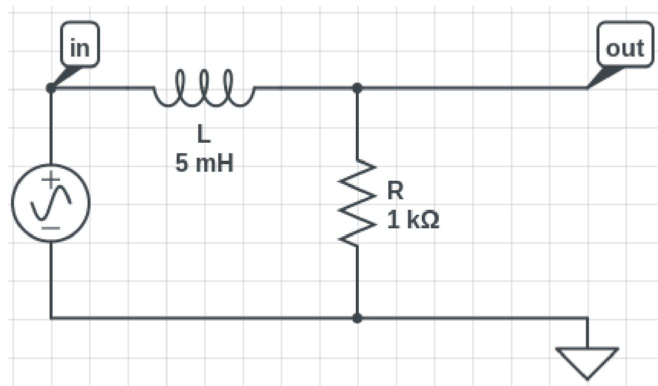
CircuitLab (time: aim for 30 minutes)

Three options: do either 5.1 or 5.2 or 5.3, whichever you find most interesting. Using either your own computer or one of the lab's notebook computers, go to `www.circuitlab.com` in your web browser and sign up for a free student account using your `upenn.edu` email address. The engineering school has a university-wide site license through 3/2015. CircuitLab lets you draw and simulate a wide range of electronic circuits, without the hassle of installing any software on your own computer. We aim to use it often in this course. Real-life circuit designers always simulate their circuit designs before building them, to be sure that they understand how the circuit they want to build is *supposed* to work. You will probably want our help getting started.

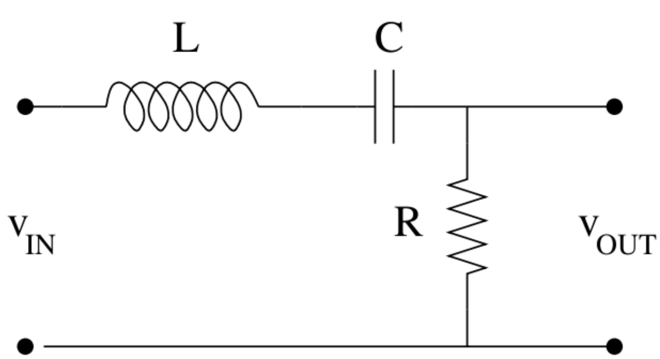
5.1 Option 1: Repeat the key parts of Part 2 (RLC filter) in the CircuitLab simulator.

5.2 Option 2: In CircuitLab, study the following two filters, along the lines of the RC filters in Lab 4. The time constant is now $\tau = L/R$ instead of $\tau = 1/RC$, so $f_{3dB} = \frac{R}{2\pi L}$. Which one should be the LPF / "integrator," and which one should be the HPF / "differentiator?" (What happens to Z_L as $f \to 0$? What about $f \to \infty$?) Build and study the circuits in CircuitLab to confirm your reasoning.





5.3 Option 3: In CircuitLab, study the series version (shown below) of the resonant RLC circuit, using the same component values as in Part 2.



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