

## Laboratory Session 2 – The Low Pass Filter

Name: \_\_\_\_\_ Name: \_\_\_\_\_ Name: \_\_\_\_\_

### Materials:

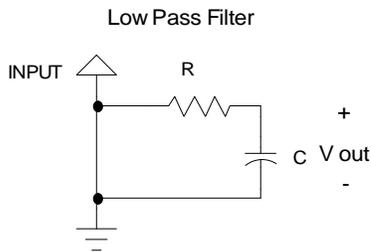
0.67 and 6.7  $\mu\text{F}$  capacitors  
 1  $\text{k}\Omega$  resistor  
 10  $\text{k}\Omega$  resistor  
 corrupted music file (*test.mp3*)  
 Headphones  
 1/8 inch mono headphones  
 1/8 inch mono male jack with leads

1/8 inch mono female jack with leads  
 Protoboard  
 Function Generator  
 Spectrum Analyzer  
 Wires  
 Wire Stripper

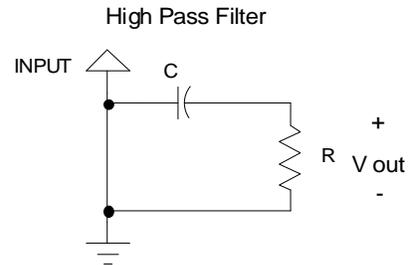
### Introduction:

Low pass filters find a lot of applications in practice. We have seen in lecture how low-pass, band-pass, and high-pass filters are employed in the design of loudspeakers' crossover networks. Your typical audio equalizer comprises of many of these filters. Telephone companies use low-pass filters to limit the voice bandwidth to 3 kHz, allowing more long-distance calls to be transmitted simultaneously.

Another application of filtering is in audio processing. Often times, music can be corrupted by unwanted frequencies due to poor transmission or poor maintenance. By applying and adjusting filters, we can block certain frequencies in our music so that we can experience a more pleasing listening experience. In this lab we will study the usefulness of the Low Pass Filter, the High Pass Filter, and the Band Pass Filter and observe its effects on of sound.



$$w_c = \frac{1}{RC}$$



$$w_c = \frac{1}{RC}$$

1. Download the files *test.mp3* onto your local computer.
2. Plug your headphones into the computer's CD player's output jack, play *test.mp3*, and listen to the song. What do you notice about it? You should notice that the song sounds corrupted, containing some unwanted high frequency.
3. Remove the headphones from the CD player's output jack replacing it with the provided 1/8 inch male jack and insert the two leads of the 1/8 male jack into your protoboard, using wires. Then insert your headphone jack into the 1/8 inch female jack and connect its two leads into the protoboard so that the both signal (red) wires of the jacks are in the same row, and both ground (black) wires are in the same row. You should still be able to hear the song now. Note that the jacks are mono, therefore you will only hear the sound in one ear.

4. Power on the function generator and set the frequency to 440 Hz and the amplitude to 0.5 V. Disconnect the CD player's wires from your headphones and connect your signal generator's signal and ground to your headphones' wires. You should be able to hear the sinusoidal wave in your headphones. You can think of the sinusoidal frequency as the pitch and amplitude as the volume of the sound. Try some other values for the frequency. Be careful not to raise the amplitude too much higher (limit to 1 V) due to large volume. (typical range for hearing is about 20Hz to 20kHz)
5. So how do you fix this noisy problem? We can try to use something called a low pass filter. The circuit for the low pass filter is shown above. Create the LPF (Low Pass Filter) circuit shown on page 1 with values for  $R = 1 \text{ k}\Omega$  and  $C = 6.7 \mu\text{F}$ . The INPUT should be the signal from the signal generator with a frequency of 10 kHz.
6. Connect your headphones to INPUT and ground. Then connect your headphones to  $V_{\text{out}}$  and ground. Compare the strength of the original signal (at INPUT) to that after it has passed through the LPF (at  $V_{\text{out}}$ ). Try different values for the frequency.
7. Now test your LPF on the sound file *test.mp3*. The INPUT signal should be the red wire coming from your CD player and Ground should be the black wire coming from your CD player. Play *test.mp3* again. The low pass filter should significantly reduce the strength of (or block) the higher frequencies in the signal, while passing the lower frequencies.
8. Calculate the cutoff frequency  $w_c$  for your values of R and C using the equation given for the low pass filter shown directly below the diagram.

Calculated  $w_c =$  \_\_\_\_\_

9. Why is it that we can still hear some frequencies that are above the cutoff frequency  $w_c$  and why has the overall volume has been reduced? Quickly sketch the signal due to your real Low Pass Filter, freq vs. magnitude (no numbers need to be labeled).
10. Note that the corruption to the sound file was due to the addition of an unwanted 10 kHz frequency. If we were designing a filter to block this frequency using an **ideal** Low Pass Filter, what is the value of R that we should use (assume  $C = 6.7 \mu\text{F}$  still).

R = \_\_\_\_\_

11. If time permits, try creating a High Pass Filter (HPF) now, with  $R = 10 \text{ k}\Omega$  and  $C = 6.7 \mu\text{F}$ . Can you hear the difference the high pass filter made in the sound? A simple Band Pass Filter can be created by combining the High Pass Filter and a Low Pass Filter. No need to implement it in lab, but what would you think would happen ideally, if  $w_{c,\text{LPF}} < w_{c,\text{HPF}}$ ?

