Introduction:
The last experiment builds on the ideas from the lab 3. Again, the topic is time-dependent signals in RC circuits. However, here you will deal with the steady-state response of an RC circuit to sinusoidally varying signals at various frequencies. You will represent these signals in the time and frequency domain and investigate the circuit’s frequency dependence. This frequency dependence can be put to use in analog electronics for designing filters. You will study the basic properties of such filters, build your own audio filter and test it on the music of your choice. By the end of the lab, you should be able to:

- Understand sinusoidal signals and phasors
- Measure amplitude gain and phase shift of an RC filter
- Measure the frequency response of a first order filter
- Design a filter with desired characteristics

Topics from the lecture you need to be familiar with:
- Sinusoidal signals, phasors and phasor diagrams
- Impedance
- Frequency dependence
- First order RC and RL filters
- Amplitude and phase response of a filter
- Bode plots

Pre-lab questions (hand in before lab starts):
1. Give at least three reasons why phasors are useful to describe AC circuits.
2. Explain why the maximum voltage across a capacitor in an RC circuit depends on frequency even if the peak voltage of the input signal does not vary with frequency.
3. Explain in words what the 3dB frequency in a high pass filter circuit is.
4. Can an RC circuit be used as both high and low pass filter? How?
5. What is a bandpass filter? Why can a simple RC circuit not be used as a bandpass filter?
1. Resistive circuits

a) Build the following circuit on your circuit board. Use the function generator as input source and set it to sinusoidal signal. **NOTE:** Try to keep the connecting wires short for good high-frequency performance.

![RC Filter Diagram]

\[ R=22k\Omega \]
\[ C=10nF \]
\[ V_{in,max}=2V \]
\[ f=2kHz \]

b) Monitor the input waveform on the scope’s channel 1 and the output waveform on channel 2, with both channel inputs set to AC coupling. The sensitivity of both channels (V/div) should be the same. The trigger source selector should be set to channel 1, the trigger coupling to AC, and the trigger slope to positive. You should be able to obtain the following display:

![Scope Display]

Carefully establish the zero level for each waveform on the oscilloscope screen by using the ground setting for each channel. Align this level with a main graticule horizontal line using the vertical position control.
c) Vary the frequency of the input signal over a wide frequency range (keep the input amplitude $V_{in,max}$ on the oscilloscope fixed by readjusting the function generator output if necessary) and observe qualitatively the output amplitude $A_{out}$. Do your observations justify the term low-pass filter? Why?

d) Record the values of $V_{in,max}$ and $V_{out,max}$ over a wide frequency range between DC (0 Hz) and 100 kHz (suggested steps: 1, 2, 5, 10, 20, 50, …). Ensure that $V_{in,max}$ is 2 V on the oscilloscope screen for each frequency setting.

e) Make a Bode plot of the (amplitude) gain magnitude $G$. That is, plot $\log(G) = 20\log(V_{out,max}/V_{in,max})$ versus $\log(f)$. Add a curve for the theoretically expected curve to your plot. Label the y-axis both in absolute $G$-values and in decibels (dB). Does your experiment agree with the theoretical prediction?

f) Find the 3 dB frequency. If necessary, take more data in the relevant frequency range to get an accurate value. Does your result agree with theory?

2. Phase response

a) For the same frequency range and using the same frequency steps as in part 1d), measure the phase difference $\Phi$ between input and output signals. Use the time difference of the zero crossings for both signals. Note that you can get more accurate results by choosing a shorter time scale.

b) Make a graph of $\Phi$ (linear scale) versus $\log(f)$. Also add the theoretically expected curve to your plot. Do your measurements agree with theory?

c) Set the frequency to the 3dB frequency found in part 1f). Determine the phase difference $\Phi_{3dB}$. Do you obtain the expected result?

3. Phasors

a) Using your results for amplitude and phase transfer from parts 1 and 2, determine the output voltage phasors at the following frequencies:
   $F=1$ Hz, 10 Hz, 100 Hz, 1kHz, 10kHz, and100kHz
   Assume that your input signal phase is $\Phi_{in}=0$ in each case.

b) Does your input voltage phasor depend on frequency? Why or why not?

c) Plot the input and output voltage phasors at the above frequencies in a phasor diagram. Describe what happens to the phasor vectors as the frequency increases.

4. High-pass filter

a) Exchange R and C in your circuit. Monitor the voltage across the resistor on the scope.

b) Repeat the qualitative measurements of part 1c) and describe why this circuit is called high-pass filter.
Part 2: Design of an audio filter

For this part of this experiment, you will use the audio speakers. For each circuit you build, ensure that you start off with an input voltage of 0 V. Slowly increase the input amplitude to protect the speakers. Keep the noise in the lab at a reasonable level. You can use your own music device or the computer’s audio player and you should bring some music of your choice for this experiment. You can obtain the best results with music that has few instruments and covers a wide frequency range.

1. Single frequency response

Connect the audio speakers to the output of the circuit shown below. \( R = 10 \Omega \), \( C = 10 \mu F \)

![Circuit Diagram]

a) Set the function generator frequency to 1kHz and start increasing the amplitude slowly until you can hear the sound from the speaker. Monitor the input signal on the scope.

b) Vary the frequency of the function generator while keeping the input amplitude the same. What do you observe? At which frequency do you notice a decrease in output volume? Compare with the 3dB frequency you measured earlier.

2. Audio filter

a) Design an RC low-pass filter which passes only sound with a frequency below 5kHz, i.e. \( f_{3dB} = 5kHz \). Draw a circuit diagram of your filter and identify the parts you need.

b) If you want to make a more interesting filter, design a filter that is tunable over a wide range of frequencies in the audio range; for example between 1.5 and 15 kHz. You will need a potentiometer to do this.

c) Build the following audio amplifier circuit that allows you to listen to music in your device or the PC’s music player over the speakers using the headphone outputs.

![Audio Amplifier Circuit]

Slowly increase the volume of your source until you can comfortably hear the music. Monitor the input signal of the speakers on the scope.
c) Now insert your filter between the power amplifier and the speakers as shown below:

![Circuit Diagram](image)

Describe what happens to the sound and what you observe on the scope.

d) Set the function generator frequency to 1kHz and start increasing the amplitude until you can hear the sound from the speaker. Monitor the input signal on the scope. 

NOTE: It is possible that your output signal is in saturation due to an unwanted DC contribution. If you observe this, add a DC offset to the function generator signal (you can add this offset on the function generator itself).

e) If you designed a tunable filter, change the cutoff frequency and observe the change in sound and scope image. This is a simple treble control.

f) If you have time: Change your filter circuit to a high-pass filter (part 1.4.) and record your observations of speaker output and scope image.