ME 103 Experimentation and Measurements
Lab 2 - Fourier Analysis and Filters

Introduction and Objectives

The purpose of this lab is to gain hands-on experience with Fourier analysis and discrete Fourier transforms (DFT and DTFT). Analyzing signals in the spectral (frequency) domain will be applied in order to understand important topics in data sampling, such as aliasing and leakage. Finally, the versatility and practicality of filters will be explored.

Lab Objectives:
- Use the Fourier transform to analyze the frequency content of time-series data.
- Build analog filters and analyze their properties regarding filtering data. Compare them with digital counterparts.
- Observe the influence of aliasing and leakage on data collection.

Equipment

All equipment will be provided in the lab, which include:

- DC power supply
- DG1022 Analog waveform generator
- DDM3058 Digital multimeter
- DS1102E Oscilloscope
- National Instruments USB-6211 DAQ Extension
- Breadboard
- Coaxial cables + Banana plugs
- Resistors and capacitors of various values
- Texas Instrument TLV2374I Quad CMOS Op-Amp

Datasheets/manuals for these pieces of measurement equipment are provided in bCourses under Files > Datasheets and Manuals.

Deliverables

It is to *your benefit* to look at the questions *in advance* to know what you are measuring and why. With your group follow the steps below to complete the lab. You **must** typeset your answers in LaTeX(We recommend using Overleaf with the template provided, but you can also edit locally if you prefer). Upload a single pdf file to Gradescope per team. Everyone should be contributing equally and writing on the document equally.

The lab is due 1 minute before your next lab section i.e the week of October 13th. For example, if you have lab on Monday at 8:00 AM, it is due the following Monday at 7:59 AM.

0 Debugging Checkoff

Note: You are about to complete the **Circuit Debug Checkoff**, an assessment designed to mirror the format and expectations of problems that will appear on the lab midterm. Accuracy, preparation, and proper measurement procedure are expected. Read all instructions carefully before beginning. Failure to demonstrate correct understanding of the circuit and measurements may result in delays to your lab progress.

Procedures

You are provided with a circuit at your station intended to operate as a voltage divider powered by a 5V supply. However, the circuit has been intentionally miswired, and component values are unknown. Your task is to:

- Correct the wiring and address any missing or improperly connected components so the circuit functions as a proper voltage divider.
- Verify correct circuit operation using appropriate measurement techniques.

During the checkoff

When requesting checkoff, you must be prepared to present the following measurements and calculations clearly and confidently:

1. The **measured resistance of each resistor**, reported with correct significant figures.
2. The **total equivalent resistance** seen by the 5V supply.
3. The **current flowing through the circuit**, as measured using your DMM.

Be prepared to explain how each value was obtained. Measurements that cannot be justified may be rejected. The GSI may also ask every group member a different brief conceptual questions about your circuit or measurements during checkoff.

Checkoff Logistics

- When ready, sign in at bearqueue.com to request a GSI.
- A GSI will come to your station to inspect the circuit and verify your measurements.
- You may proceed with the remainder of the lab **only after** successful completion of the checkoff.

Students who are unprepared or whose circuits are not functioning correctly will be asked to continue debugging before receiving checkoff approval.

Warning: Completion of this checkoff is **mandatory**. You must successfully complete the Circuit Debug Checkoff and receive GSI approval **before proceeding** with the remainder of the lab. Work completed beyond this point without checkoff approval may be rejected and may need to be repeated.

1 The Fourier Transform

First, let's experiment with the Fourier transform to understand its application. To provide some initial background, the Fourier transform expresses data that was originally in the time domain (i.e. a signal as a function of time) in the frequency domain. This part of the lab should allow you to discover what the frequency domain is, as well as how the Fourier transform relates the time and frequency domains.

- Download `lab2.vi` from bCourses. Open the virtual instrument and explore the interface.
  - Notice that this VI allows the user to control some of its characteristics (sampling rate and number of samples) before you run it.
  - **Q1**: If the sampling frequency is 500 Hz, how many samples must be taken so that the VI runs for 10 seconds?
  - Set the sampling frequency to 500 Hz and the number of samples to the number you found in **Q1**.
- Turn on an analog waveform generator and use it to produce a 10 Hz, $V_{pp} = 2$V **sinusoidal** signal. Don't forget to turn the output on.
- Use an appropriate coaxial cable and jumper cables to read the signal from step 2 into channel `AI1` of the DAQ Extension.
  - Note that the VI for this lab has been set up to use **differential** or **double-ended pins**. This means there is no longer a universal ground pin like Lab 1. Instead, each channel `AI1` through `AI8` has its own ground pin, directly to the right of itself. For example, `the` ground pin for `AI1` (pin 17) is `AI9` (pin 18).
- Run the VI (remember this will take 10 seconds) and study both the Time Domain and Frequency Domain graphs. Note that the Frequency Domain graph is created simply by passing the Time Domain data through a Fourier transform.
  - Make sure to zoom to an appropriate viewing window for both graphs. It may be helpful to first "Autoscale" each graph and then zoom in if necessary.
  - Additionally, you may turn off the Voltage 2 display for now by right-clicking it and unchecking "Plot Visible." The signal closely traces that for Voltage 1 due to electromagnetic induction.
  - Take a screenshot; make note of features in this plot, you will use this to answer **Q2** and **Q3**.
- Turn off the output of your analog waveform generator before adjusting the settings so that it produces a 10 Hz, $V_{pp} = 2$ V, square signal. Turn on the output and then run the VI again.
  - Study the Time Domain and Frequency Domain plots. Again, make sure to zoom to an appropriate viewing window.
  - Take a screenshot; make note of features in this plot, you will use this to answer **Q4**, **Q5** and **Q6**.

2 The Limits of Sampling Time-varying Data

In the previous part of the lab, we explored how sampling and the Fourier transform work in an ideal scenario. In this part of the lab, we will explore the limitations of the Fourier transform as well as the process of sampling itself.

- Set the sampling frequency for the VI to 100 Hz. Set the number of samples so that the VI will still collect 10 seconds worth of data.
- Based on your findings in Section 1 of the lab, fill in the prediction column of the following table if sinusoidal signals are measured into the DAQ. This corresponds to **Q7**.

Signal frequency	Observed frequency (Prediction)	Observed frequency (Actual)
25 Hz		
35 Hz		
45 Hz		
55 Hz		
65 Hz		
75 Hz		

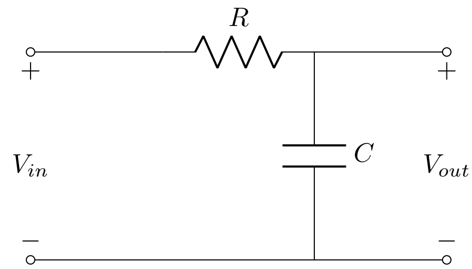
Table 1: Comparison of predicted and actual observed frequencies

- Use the analog waveform generator to test your predictions (produce sinusoidal waves with $V_{pp} = 2$ V and the above frequencies). Fill in the actual column of Table 1. This will correspond to **Q8** .
  - Study your plots and use your data in Table 1 to answer **Q9** and **Q10**.
- Use the analog waveform generator to produce a sinusoidal wave with frequency 50 Hz. Measure this wave by running the VI several different times. Note any differences between runs. You will use this to answer **Q11**.
- Experiment to determine at what frequency phase ambiguity occurs. Try measuring a 49 Hz signal, a 49.9 Hz signal, and a 49.99 Hz signal. When does phase ambiguity begin to occur? Use this to answer **Q12**.
- Now set the sampling frequency of the VI back to 500 Hz. Set the number of samples to 50.
- Measure a 20 Hz sinusoidal signal with these settings. Try changing the number of samples to 100 or 500 as well. Notice that the resolution of frequencies changes depending on the number of samples. Use this to answer **Q13**.
- Set the number of samples to 100 and measure a 22 Hz sinusoidal signal. What is different about the frequency content this run? Do you have a hypothesis as to why this leakage occurs? Feel free to test different signal frequencies or frequency resolutions to test your hypothesis. Use this to answer **Q14**.

3 Filtering Signals

In the final part of this lab, you will explore analog electronic filters through the lens of sampling time-varying data. By the end of the lab, you should understand how you can combine your discoveries about filters and Fourier transforms to tailor a measurement system to your needs.

3.1 Low Pass Filter



A circuit diagram for a passive RC low-pass filter (LPF) is shown above. A passive RC high-pass filter (HPF) is the same as above, but with the capacitor and resistor swapped! For a given resistance R in Ohms and capacitance C in Farads, the corner/cutoff frequency in Hz for both LPF and HPF is given by

$$f_c = \frac{1}{2\pi RC}$$

We can also implement a LPF and HPF using an RL circuit. RL filters are generally used instead of RC filters in high-power applications, such as power supply conditioning, because they handle higher currents and dissipate less power, avoiding the energy losses associated with resistors. RL filters are also preferred when low impedance paths are required for specific frequencies. For a given resistance R in Ohms and inductance L in Henry's, the corner/cutoff frequency in Hz for both LPF and HPF is given by

$$f_c = \frac{R}{2\pi L}$$

For an ideal LPF, all frequencies above f_c are completely eliminated, but physical LPFs can rarely if ever achieve this. Instead, signals of various frequencies have their amplitudes attenuated, or reduced according to the following theoretical formula where f_s is the signal frequency

$$\left|\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}\right| = \left(1 + \left(\frac{f_s}{f_c}\right)^2\right)^{-1/2}$$

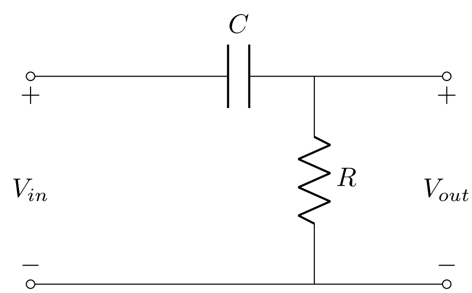
- Design a LPF xcircuit with a corner frequency of approximately 340 Hz. Notice you have an infinite number of choices of R and C combinations (or in lab, a large number of choices).
- Set the sampling frequency of the VI to 10 kHz and the number of samples so that it runs for 10 seconds.
- Use the analog waveform generator to produce sinusoidal signals with $V_{pp} = 2$V and the following frequencies: 10 Hz, 34 Hz, 100 Hz, 340 Hz, 1 kHz, 3.4 kHz. Measure the unfiltered signal (V_{in}) into channel `AI1` of the DAQ and the filtered signal (V_{out}) into channel `AI2` of the DAQ.
  - Record the approximate amplitude attenuation ratio $|V_{\text{out,amp}}/V_{\text{in,amp}}|$ for each frequency in the table by reading the graph on the VI. This corresponds to **Q16**.
  - Also calculate the attenuation in decibels (dB). Fill out the table below.

Frequency [Hz]	$\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}$ (absolute)	$\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}$ (dB)
10		
34		
100		
340		
1000		
3400		

Table 2: Measured amplitude ratio versus frequency for LPF

- By filling out the table above, you will have answered **Q16**. Using the filled in table, you will now answer **Q17**and **Q18**.

3.2 High Pass Filter



- Design an HPF with corner frequency approximately 340 Hz.
- Using the same VI sampling settings (10 kHz, 10 s) as above, repeat the same frequency sweep (10, 34, 100, 340, 1000, 3400 Hz).
  - Recall to measure unfiltered into `AI1` and HPF output into `AI2`.
  - Record the approximate amplitude attenuation ratio $|V_{\text{out,amp}}/V_{\text{in,amp}}|$ for each frequency in the table by reading the graph on the VI. This corresponds to **Q16**.
  - Also calculate the attenuation in decibels (dB). Fill out the table below.

Frequency [Hz]	$\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}$ (absolute)	$\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}$ (dB)
10		
34		
100		
340		
1000		
3400		

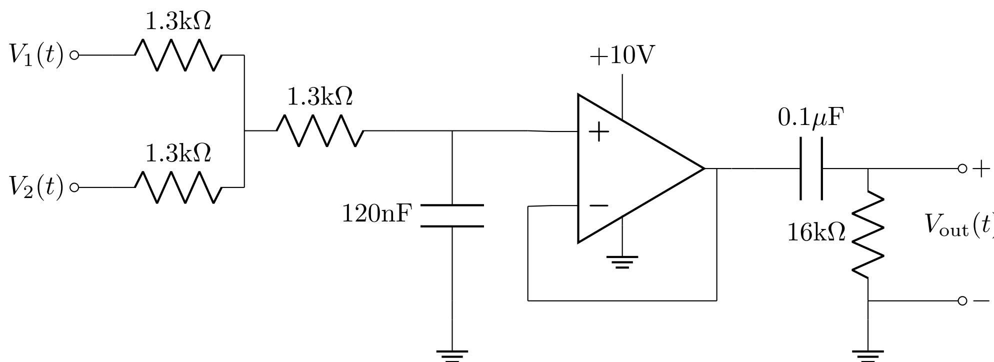
Table 3: Measured amplitude ratio versus frequency for HPF

  - By filling out the table above, you will have answered **Q19**. Use this to answer **Q20**.

3.3 Band-pass Filter

In this section, you will be creating an analyzing an analog band-pass filter, as well as assessing its uses. You will be applying a signal with an offset and additional noise. The signal will pass through a low pass filter, which will attenuate the noise. After the signal will pass through a high pass filter, which will attenuate the offset. In an ideal world your final signal should look like the original signal generated, perfectly untampered. You will also analyze in this section why this perfect attenuation may or may not be the case, and why.

- Turn on the analog waveform generator and use it to produce a 10 kHz, $V_{pp} = 2$V sinusoidal signal in Channel 1 that will act as the noise. Produce a 250 Hz, $V_{pp} = 5$ V, sinusoid signal in Channel 2 with an offset voltage $V_{DC} = 6$V. Each of these channels should be connected to `common GND` and to each the parallel resistors at the beginning of the circuit.



- Turn on the power supply and set the voltage to 10V with a maximum current of 0.5A. Connect the appropriate terminals of the op-amp to the power supply. Look at the TLV2374 data sheet on bCourses for a pinout.
- After confirming that your circuit is powered and your waveform generator is running correctly, verify that the filter is functioning as intended.
- Connect oscilloscope probes with **Channel 1** probing the input signal after averaging the signal and **Channel 2** probing the output i.e. the filtered signal.
- Set the **reference ground**; ensure both probe grounds are connected to the common ground on your breadboard.
- Take a picture/bitmap of your oscilloscope with both the signal after the averager and after the first filter stage i.e the low-pass filter.
- Take a picture/bitmap of your oscilloscope with both the signal after the averager and after the second filter stage i.e the high-pass filter. Use the cursor to show the $|\Delta Y|$.

Additional Filters: In this lab, you constructed basic passive low-pass, high-pass, and band-pass filters. However, many practical measurement and signal-processing systems use more advanced filters to achieve sharper transitions, better control over gain, or precise removal of specific unwanted frequencies.

Some additional filters include: active low-pass and high-pass filters, cascaded filters, band-stop and notch filters, nth-order Butterworth filters, Chebyshev Filters (Types I and II), and the Sallen-Key Filter Topology for 2nd order filter implementation. For more information, you may read the appendix. Some of these filters may appear in **homework**, **midterms**, or other courses!

4 Questions

Section 1 Questions

1. If the sampling frequency is 500Hz, how many samples must be taken so that the VI runs for 10 seconds?

   SOLUTION: The number of samples is $N = f_s \times T = 500\text{Hz} \times 10\text{s} = 5000$ samples.

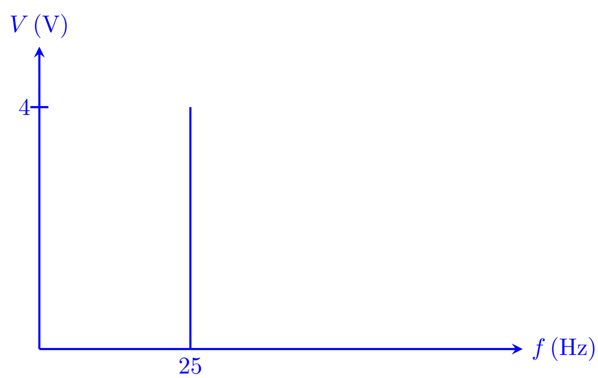
2. You are likely familiar with the plots like the Time Domain one. Describe what you see in the Frequency Domain plot and how various features might correspond to aspects of the Time Domain plot.

   SOLUTION: Students should describe a large peak at 10 Hz (and possibly some smaller peaks scattered elsewhere). Students might say that the peak at 10 Hz corresponds to the frequency of the signal in the time domain, and that the height of the peak corresponds to the amplitude of the signal in the time domain.

   An example response could be: In the frequency domain plot, we see a single peak at 10Hz that spans from 9.9 to 10.1 Hz. The height of the peak in the frequency domain is our amplitude in the time domain. If there are multiple peaks in the frequency domain then the signal in the time domain consists of multiple sinusoids.

3. Predict what the Frequency Domain plot would look like if you instead measured a 25Hz, $V_{pp} = 4\text{V}$ sinusoidal signal (sketch an approximate graph).

   SOLUTION: Students should present a predicted graph similar to the one shown below. The prediction need not be accurate (note the peak shown should have a height of 2V instead of 4V).



4. Describe the frequency content of (what frequencies are present in) the signal. Is there a pattern? Is there a pattern in the amplitudes of the frequencies?

   SOLUTION: The frequency content of the signal is 10 Hz, 30 Hz, 50 Hz, and so on. In other words, all odd multiples of 10 Hz. The amplitude decreases as the frequency increases.

5. In MATLAB or the graphing software of your choice, graph the sum of several sine waves of the form

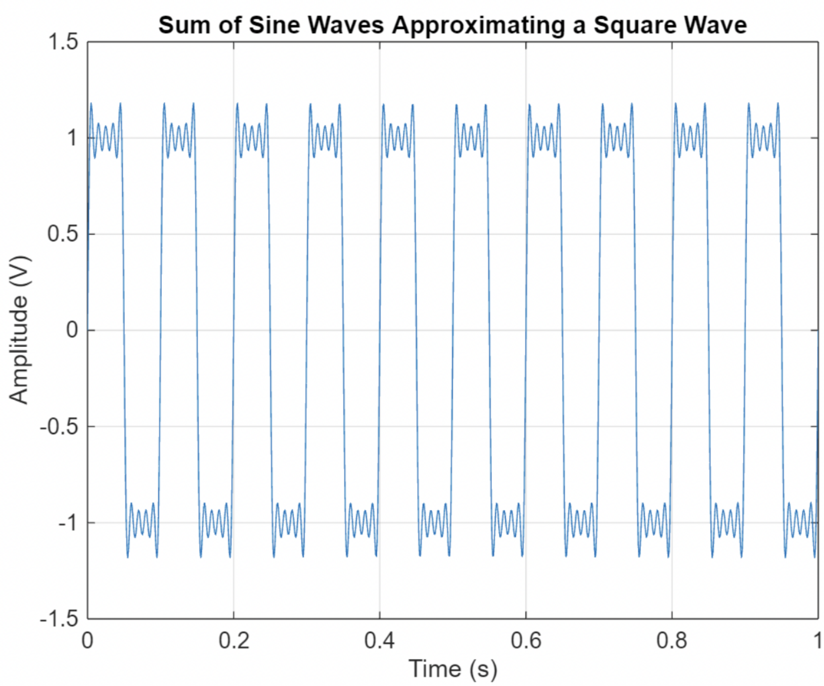
$$s(t) = \frac{2V_{pp}}{\pi n} \sin(2\pi f_{\text{square}} n t)$$

   where f_{square} is the frequency of the square wave, and values of n are chosen to match the frequency content you observed. Start with only a few terms (perhaps even just 1) and slowly add more terms. What happens to your plotted signal as more terms are added?

   SOLUTION: Students should show the graphs they obtain. They should, however, observe that as the number of summed terms increases, the wave appears more and more similar to a square wave.

```
Vpp = 4;
f_square = 25;
```

```
t = 0:1e-4:0.2;

harmonic_sets = [1 3 5 15 51];

figure;
for k = 1:length(harmonic_sets)

    Nmax = harmonic_sets(k);
    s = zeros(size(t));

    for n = 1:2:Nmax
        s = s + (2*Vpp)/(pi*n) * ...
            sin(2*pi*f_square*n*t);
    end

    subplot(length(harmonic_sets),1,k)
    plot(t, s, 'LineWidth',1.5)
    grid on
    ylabel('Amplitude V')
    title('Sum of Sine Waves Approximating a Square Wave')

    if k == length(harmonic_sets)
        xlabel('Time (s)')
    end
end
```



6. In your own words, describe what the Fourier transform does; recall that the Fourier transform is what generates the Frequency Domain plot.

   SOLUTION: The Fourier transform converts a signal in the time domain into data in the frequency domain. It extracts the frequencies which are present in the time domain signal and represents them as values which correspond to the amplitude of that frequency in the time domain.

Section 2 Questions

7. Based on your findings in Section 1 of the lab, fill in the prediction column of the following table if sinusoidal signals are measured into the DAQ.

   SOLUTION: Students should fill out the prediction column here, likely as follows. The frequencies need not be correct.

Signal frequency	Observed frequency (Prediction)	Observed frequency (Actual)
25 Hz	25 Hz	
35 Hz	35 Hz	
45 Hz	45 Hz	
55 Hz	55 Hz	
65 Hz	65 Hz	
75 Hz	75 Hz	

8. Fill in the actual column of the table from above.

   SOLUTION: Students should fill out the actual column here, likely as follows. The prediction column need not be correct, but the actual column should.

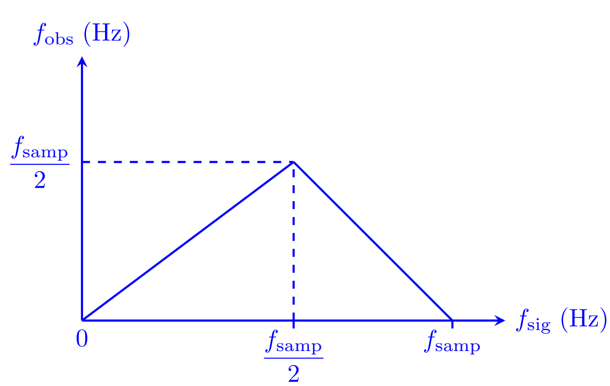
Signal frequency	Observed frequency (Prediction)	Observed frequency (Actual)
25 Hz	25 Hz	25 Hz
35 Hz	35 Hz	35 Hz
45 Hz	45 Hz	45 Hz
55 Hz	55 Hz	45 Hz
65 Hz	65 Hz	35 Hz
75 Hz	75 Hz	25 Hz

9. Is there anything interesting about the observed frequencies? (*Hint*: do you notice reflective symmetry about a specific frequency? Feel free to test other frequencies to test your hypothesis.) This phenomenon is called aliasing and maximum resolvable frequency is called the **Nyquist frequency**.

   SOLUTION: The frequencies are accurately observed up to 50 Hz, half the sampling frequency. Moreover, there is reflective symmetry about 50 Hz, i.e. a signal at, say, 70 Hz is observed at 30 Hz instead. In general, a frequency $50\text{Hz} \le f < 100\text{Hz}$ is observed at $50\text{Hz}-(f-50\text{Hz})$. Students do not need to explicitly state the general equation

10. Using your data and the understanding you gained in Question 9, draw a plot of the observed frequency as a function of the signal frequency for the full range of signal frequencies between 0 Hz and the sampling frequency. Label any important points.

    SOLUTION:



11. What do you notice about successive runs? Can you explain why this occurs? This phenomenon is called *phase ambiguity*.

    SOLUTION: The measured amplitude of the signal changes each time. This is a result of the data collection beginning at an essentially random phase shift relative to the signal, depending on when the user begins collecting data.

    Example: On successive runs, the voltage oscillates in time domain thus presenting a sinusoid with different amplitudes, and thus produces a noisy wave in the frequency domain. One odd case was when everything went chaotic and there was no sense of direction in the frequency domain. Due to the nature of the device, the two phase values differ since the DAQ is hard capped at 50 so it does not provide consistent readings. Another reason could be the differences in the alignment between the signal's cycle and the start of sampling can lead to different phase measurements in successive runs, even if the frequency and amplitude are the same.

12. When does phase ambiguity begin to occur? Can you resolve frequencies arbitrarily close to the Nyquist frequency, i.e. does the Fourier transform have an infinite resolution?

    Students should observe that phase ambiguity begins to occur at frequencies closer to 50 Hz than 49.9 Hz (in this case only 49.99 Hz), indicating that we cannot resolve frequencies arbitrarily close to the Nyquist frequency, and the Fourier transform does not have infinite resolution.

13. Develop a formula to calculate the frequency resolution using the sampling frequency and number of samples.

    SOLUTION: $\Delta f = \frac{f_s}{N}$, where f_s is the sampling frequency, and N is the number of samples.

14. What is different about the frequency content this run? Do you have a hypothesis as to why this leakage occurs? Feel free to test different signal frequencies or frequency resolutions to test your hypothesis.

    Students should observe that the "peak" is no longer a sharp peak and is instead spread out across several different frequencies. Students can write a variety of different hypotheses for why leakage occurs, but the true reason is that with a finite number of samples, if a non-integer number of signal periods is measured, leakage occurs.

    Example: The frequency content is not equal to the actual inputted 22Hz; it is not linear around the input frequency and does not behave symmetrically. Leakage most likely occurs when the signal is not perfectly periodic within the measurement window and the signal's frequency does not align exactly with the discrete frequency bins of the Fourier Transform due to a lack of samples.

Section 3 Questions

Low-pass filter

15. Would all choices of R and C behave equally? In what scenarios might you prefer a larger value or smaller value for one of the quantities, say R?

    Not all choices of R and C behave equally. The rest of this question is relatively open-ended. In short, the overall impedance of the circuit varies depending on the choice of R and C. For a larger R, there is a larger overall impedance, which may be desirable if a voltage drop is desired and the internal resistances of various measurement devices need to be made negligible. Alternatively, if the LPF is being used in a scenario where the signal must be passed to another controller or receiver, a very small voltage drop may be desired, in which case a smaller choice of R and a smaller overall impedance is desirable.

    Example: They would not all behave equally. Larger choices of R and C would result in a smaller capacitance frequency, thus resulting in a smaller output amplitude voltage. We would prefer larger R values since we want to eliminate higher voltages. A smaller R would allow us to view the larger changes in the output voltage.

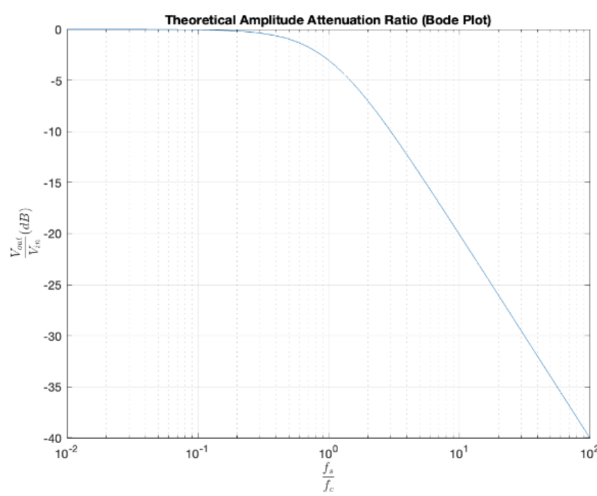
16. Record the approximate amplitude attenuation ratio $\left|\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}\right|$ for each frequency in the table by reading the graph on the VI. Also calculate the attenuation in dB (decibels). The conversion from an absolute value to dB is $Q_{dB} = 20\log_{10}(Q_{\text{abs}})$ where Q_{dB} is a quantity in decibels and Q_{abs} is the absolute quantity.

    SOLUTION:

Frequency [Hz]	$\left\lvert\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}\right\rvert$ (absolute)	$\left\lvert\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}\right\rvert$ (dB)
10	0.9996	-0.003
34	0.9950	-0.043
100	0.9594	-0.360
340	0.7071	-3.010
1000	0.3219	-9.846
3400	0.0995	-20.043

17. Create a bode plot of the amplitude attenuation ratio vs. the f_s/f_c ratio (this plots the attenuation in dB on the vertical axis vs. the log of the frequency ratio on the horizontal axis). Plot both your observed attenuation ratios and the theoretical attenuation ratio. Why do you think the corner frequency is named such? A synonym for the corner frequency is the -3dB frequency. Why do you think this is?

    SOLUTION: The following bode plot should be presented.



    The corner frequency is that at which the bode plot shows a dramatic bend, similar to a corner. Alternatively, it is the frequency at which the response is attenuated to -3dB.

18. Combining what you have learned about measurements of time-varying signals and filters, how could a filter assist in measuring a complex signal with many different frequencies present? Specifically, which phenomenon explored in section 2 of the lab could be mitigated, and how would you select the corner frequency of your filter to do so?

    SOLUTION: It would be possible to mitigate aliasing using a LPF. Specifically, the corner frequency of the LPF could be chosen to be equal to or slightly less than the Nyquist frequency, therefore attenuating signals with higher frequencies and preventing them from being aliased and interpreted as actual frequencies.

Students do not explicitly need to state this, but it would be important to remember that depending on what the corner frequency is chosen to be, different frequencies would be attenuated by different amounts, so some higher frequencies may still show up in a measured signal, but at lower amplitudes.

High-pass filter

19. Fill in an attenuation table (absolute and dB) for the HPF, analogous to the LPF table you already have.

SOLUTION:

Frequency [Hz]	$\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}$ (absolute)	$\frac{V_{\text{out,amp}}}{V_{\text{in,amp}}}$ (dB)
10	0.0275	-31.2
34	0.09	-20.91
100	0.25	-12.04
340	0.642	-3.84
1000	0.918	-0.739
3400	1	-0

20. Create a bode plot of the amplitude attenuation ratio vs. the f_s/f_c ratio for the HPF (this plots the attenuation in dB on the vertical axis vs. the log of the frequency ratio on the horizontal axis). Compare its shape to the LPF.

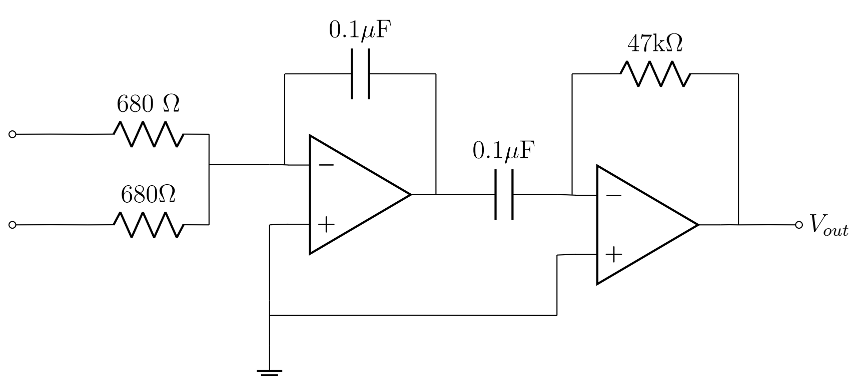
SOLUTION: Same Bode plot as 17 except rising slope.

Band-pass filter

21. Submit 2 photos of your bitmap showing a) the signal after the averager and the low-pass filter, b) the signal after the average and the high-pass filter.

SOLUTION: If it looks like the blue is a lot cleaner after LPF, and offset remove full credit.

22. In this section, we used a voltage-follower or a unity-gain op-amp in our circuit. An alternative way to construct a band-pass filter is to cascade two op-amp stages as follows



In our lab setup, the TLV2374 op-amps are powered from a single supply (0–10 V). Explain why this circuit may not operate correctly without additional modifications.

23. In this circuit, a low-pass filter stage is followed by a high-pass filter stage to create a band-pass response. Explain why the order of these stages matters in our lab implementation. What practical issues could arise if the high-pass filter were placed first?

24. The 250 Hz input signal is given a DC offset of approximately 6 V before being applied to the filter. Why is this offset necessary when operating the op-amp from a 0–10 V supply, and why might it be chosen to be closer to 6 V rather than exactly mid-supply?

Appendix

Active Low-Pass and High-Pass Filters

The filters built in this lab were **passive** filters using only resistors, capacitors, and inductors. An **active filter** uses an operational amplifier together with passive components.

Active filters offer several advantages:

- They can provide **signal gain** instead of attenuation.
- They prevent loading effects by buffering stages.
- They allow sharper filter responses using higher-order designs.

Active LPFs and HPFs behave similarly to passive versions but provide improved performance and flexibility in real measurement systems. Below are 4 possible ways using inverting and non-inverting op-amps

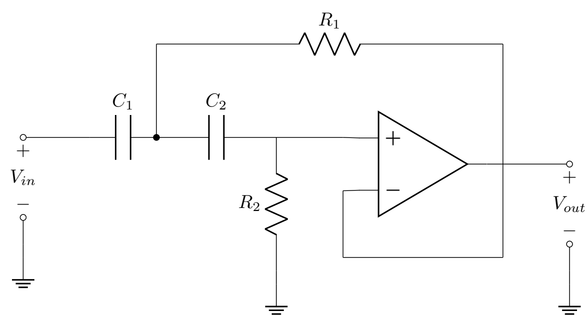


Figure 1: Active Non-Inverting High-Pass Filter

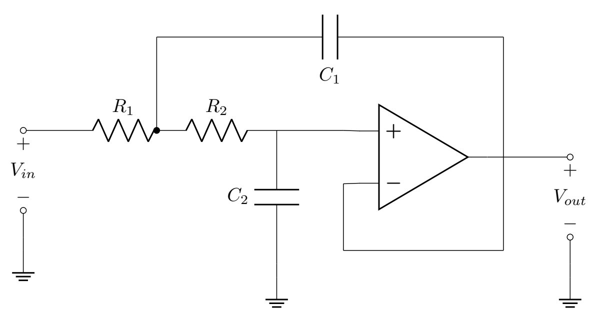


Figure 2: Active Non-Inverting Low-Pass Filter

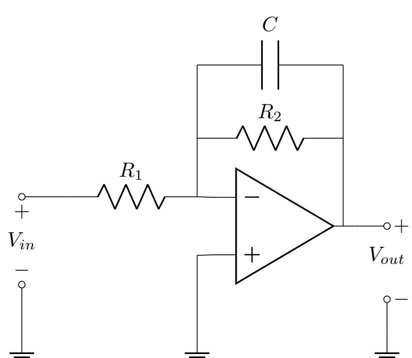


Figure 3: Active Inverting Low-Pass Filter

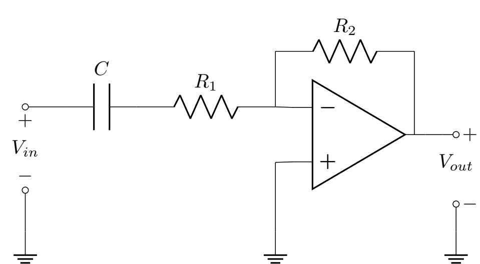


Figure 4: Active Inverting High-Pass Filter

Cascaded Filters

Higher-order filters are often built by **cascading** multiple simpler filters. For example:

- Two HPFs in series create a steeper high-pass response.
- Two LPFs in series create a steeper low-pass response.

Each stage increases attenuation outside the desired frequency band, improving noise rejection. Many practical filters are built as cascades of second-order stages. An example of this is in Q22 in the Section 3 Questions.

Band-Stop and Notch Filters

A **band-stop filter** attenuates a range of frequencies while allowing frequencies both above and below that band to pass.

A special case is the **notch filter**, which removes a *very narrow* band of frequencies. Notch filters are commonly used to eliminate known interference sources such as:

- motor or switching noise,
- mechanical vibration frequencies.

Notch filters are especially useful when a single unwanted frequency contaminates a signal.

Butterworth Filters

A **Butterworth filter** is designed to have a **maximally flat response** in the passband. This means:

- No ripple occurs in the passband,
- The response is smooth and monotonic,
- Transition from passband to stopband is moderate.

Butterworth filters are commonly used when smooth amplitude response is more important than having the sharpest cutoff.

Chebyshev Filters

Chebyshev filters allow a sharper cutoff than Butterworth filters but introduce ripple.

Two common versions exist:

- **Chebyshev Type I**: ripple occurs in the **passband**.
- **Chebyshev Type II**: ripple occurs in the **stopband**.

These filters are useful when faster attenuation outside the desired band is needed and small amplitude variations are acceptable.

Sallen–Key Filter Circuits

A very common active filter structure is the **Sallen–Key** topology. It uses an op-amp with resistors and capacitors to implement a compact **second-order filter stage**.

Sallen–Key circuits are popular because:

- They are easy to design and build,
- They provide stable filter characteristics,
- They can implement LPF, HPF, band-pass, or band-stop responses,
- They are widely used in instrumentation and audio systems.

Many practical filters are constructed by cascading several Sallen–Key stages to achieve higher-order filter behavior.