Introduction to Data Acquisition, Filter Design and Digital Circuits (Electronics)

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The experiment gives you an introduction to electronics. Our main objective in this experiment is to filter out noise in electric circuits. This experiment is divided into sections, such that each section introduces one of the key concepts, and finally this culminates to our final objective.

KEYWORDS

Breadboard · Data Acquisition · Sampling · Signal reconstruction · Nyquist Theorem · Filters · Composite Signal · Frequency Spectrum

APPROXIMATE PERFORMANCE TIME 4 hours

1 Conceptual Objectives

In this experiment, we will,

1. learn how to implement circuits,
2. practice data acquisition,
3. understand the inter-relationship between mathematical expressions and graphs,
4. learn the use of electric test and measurement equipment,
5. appreciate the concept of frequencies,
6. observe the conversion between analog and digital signals, and
7. practice how to extract useful information from graphs.

2 Experimental Objectives

The experimental objectives are presented at the head of each Section. There are nine sections comprising this Lab manual.

1. Introduction to the history of electronics
2. Breadboard layout and its internal connections
3. Data acquisition system
4. Understanding the frequency concept
5. Verifying the Nyquist theorem
6. Filter design

3 Introduction to the History of Electronics

The history of modern electronics can be traced back to 1883, when Edison discovered that electrons flow from one metal conductor to another through vacuum. This is known as *thermionic emission*.

In 1987, J.J. Thomson developed a vacuum tube to carefully investigate the nature of cathode rays. He showed that the cathode rays were made up of particles, which he named "corpuscles". This marked the discovery of the electron. Thomson received the Nobel Prize in Physics 1906.

In 1904, John Fleming applied Edison's thermionic emission to invent a two-element electron tube called the *diode*. This was followed by Lee De Forest's discovery, in 1906, of the three-element tube, the *triode*. These vacuum tubes made possible the amplification and transmission of electrical signals.

In 1947, John Bardeen and Walter Brattain, working at Bell Telephone Laboratories, were trying to understand the nature of the electrons at the interface between a metal and a semiconductor. They realized that by making two point contacts very close to one another, they could make a three terminal device called the *transistor*, the semiconducting analog of the triode.

The invention of the transistor, initiated the electronics revolution of the twentieth century. The drive was to build more transistors on a single chip. In 1965, Gordon Moore, co-founder of Intel, observed that the number of transistors per square inch on integrated circuits had doubled every year since the invention of the *integrated circuit*. Moore predicted that this trend would continue for the foreseeable future. In subsequent years, the pace slowed down a bit, but data density has still doubled approximately every 18 months.

Today, as the trend and need towards miniaturization is gaining momentum, there is also a growing realization that the physical limits of the transistor fabrication have been achieved.

Figure 1: John Bardeen, William Shockley, and Walter Brattain.

So, new fields have now emerged, such as *quantum computing*, *spintronics*, *nanoelectronics*, and so on.

4 Breadboard Layout and its Internal Connections

4.1 Objective

The objective of this section is to familiarize you with the internal connections of the *breadboard*.

4.2 Breadboard

A breadboard is used to make up **temporary circuits** for testing or to trying out an idea. No soldering is required, so it is easy to change connections and replace electronic components

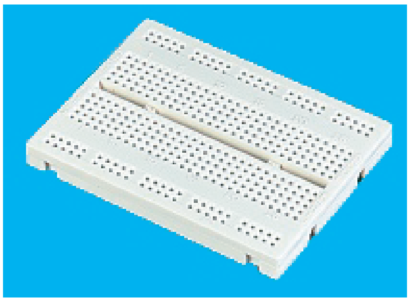


Figure 2: breadboard.

4.3 Internal connections

Figure 3 shows the layout and internal connections of the breadboard. The holes in black are used for inserting the electronic components. The line joining holes shows their serial connection.

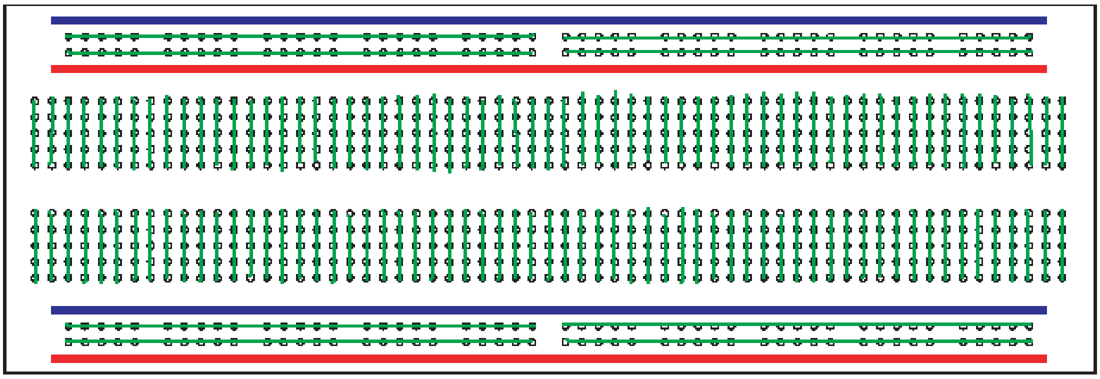


Figure 3: Breadboard Internal Connections.

4.4 IC placement on the breadboard

Figure 4 shows how to place an integrated circuit (IC) chip on a breadboard.

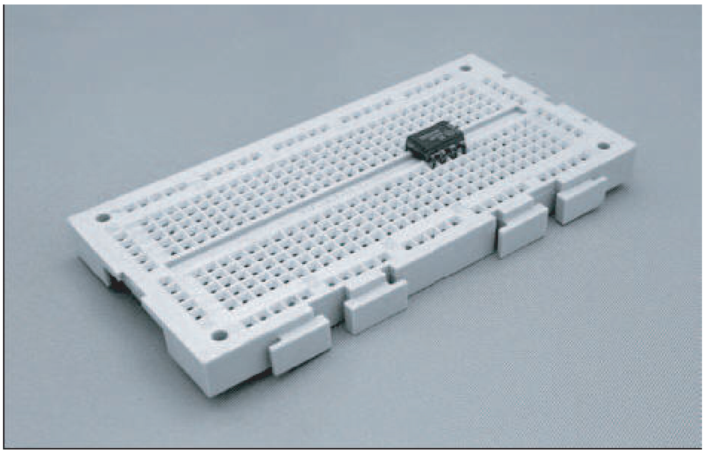


Figure 4: IC on a breadboard.

5 Data Acquisition System

5.1 Objective

This section describes the experimental setup, particularly the data acquisition system, which is being used in different activities of this experiment.

5.2 Experimental layout

The basic layout of the Data Acquisition System for our experiment is shown in Figure 5.

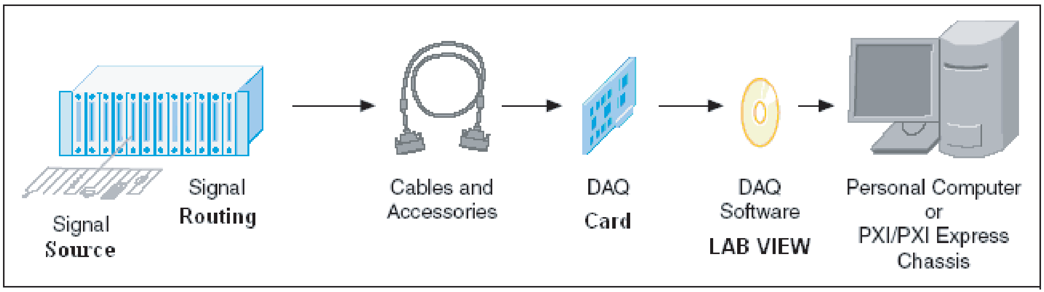


Figure 5: Data Acquisition System.

1. **Signal source**

   The signal is provided by the signal generator (*GW-INSTEK*). Sine wave, square wave and triangular shaped signals can be generated using the signal generator.

2. **Signal routing**

   The signal routing and conditioning module (*National Instruments SCC-69*) is used for signal routing. The signal is taken from the breadboard using the provided hook-up wires. The SCC-68 module is shown in Figure 6.

Figure 6: NI SCC-68 Module.

3. **Cables and accessories**

   The SCC-68 module is provided with the cable. It consists of a male connector which is further attached to the DAQ card.

4. **DAQ card**

   The DAQ card (*NI PCCI 6221*), shown in Figure 7 acquires the signal. If the incoming signal is analog, then DAQ performs *analog to digital conversion*. This is done using a chip called the *analog to digital converter* (ADC). The ADC is located on the DAQ card.

   For, analog to digital conversion we have to *sample* the incoming signal. The *sampling frequency* must be greater than twice the fundamental frequency of the incoming signal. This criterion is stated in the **Nyquist Theorem**. Sampling and the Nyquist theorem will be discussed further in Section 7.

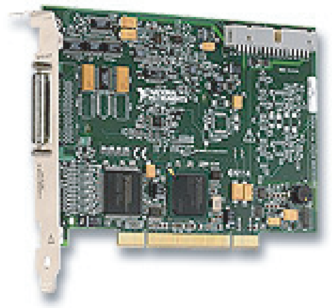


Figure 7: NI PCI 6221 DAQ Card.

5. **LABVIEW**

   The DAQ system is controlled through Labview software. We are using Labview version 8.5.1 on our computers.

6 Understanding the Frequency Concept

6.1 Objective

This section gives insight into the relationship between *frequency* and *time period*.

6.2 Definitions

1. **Frequency**

   It is a measure of the number of occurrences of a repeating event per unit time. It is usually denoted by f and its units are Hertz (Hz), cycles per second.

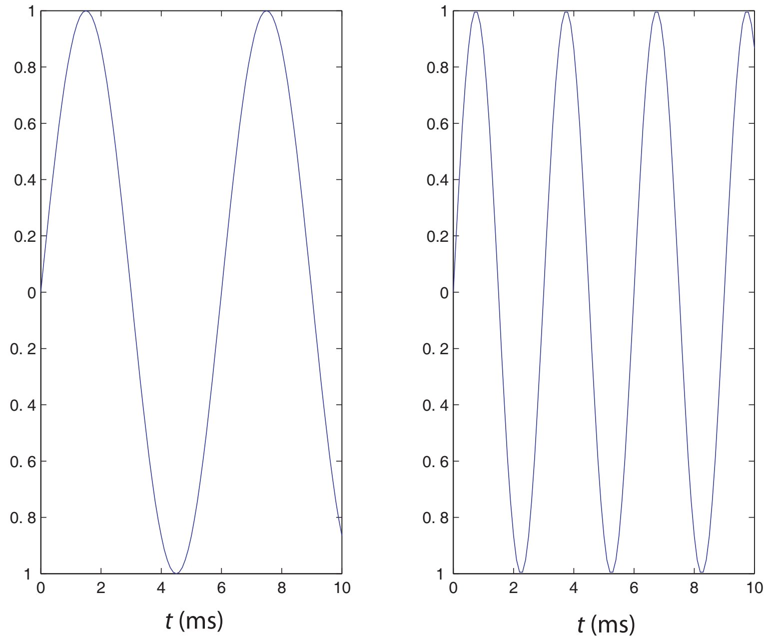
2. **Time period**

   It is defined as the time required for a single cycle in a repeating event. It is denoted by T and the SI units are seconds. So we notice that frequency is the reciprocal of the time period. Mathematically, this is expressed as,

$$f = \frac{1}{T}. \tag{1}$$

6.3 Example

Figure 8 shows the two sine waves. Figure 8(a) shows a sine wave with time period of 6 ms, whereas Figure 8(b) shows a sine wave whose time period is 3 ms. Thus the second sine wave has half the time period but double the frequency.



(a) (b)

Figure 8: Sine Waves.

7 Verifying the Nyquist Theorem

7.1 Objective

This section describes some of the key concepts in data acquisition and signal processing—sampling and analog-to-digital conversion.

7.2 Sampling

Sampling a signal means to take the value of signal at discrete intervals of time, thus making a discrete time signal out of a continuous one.

Sampling rate is defined as the number of samples taken per second.

7.3 Example

Figure 9(a) shows a simple continuous sine wave with a time period of 6 ms. Figure 9(b) shows the digitized, sampled version of this signal, with a sampling rate of 5 KHz or a sampling interval of 200 μs.

An analog input signal is fed into the DAQ card. The ADC on the DAQ card samples the analog voltage and stores the sampled points in the memory. The sampled points will have an amplitude distribution resembling the one in Figure 9(b).

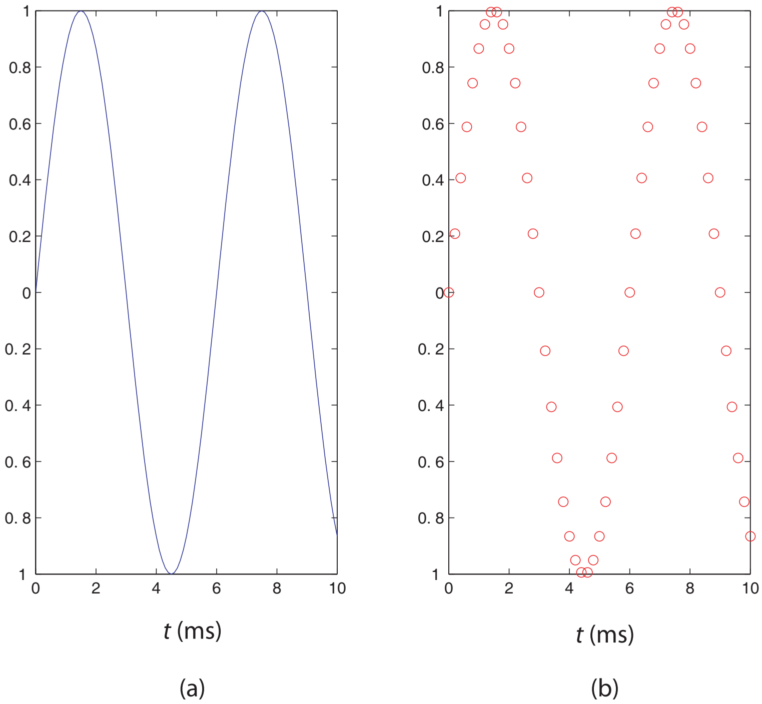


Figure 9: (a) Continuous time analog sine wave and (b) its sampled version.

7.4 Block diagram of an ADC

The functional block diagram of an ADC is shown in Figure 10. It takes the analog signal as an input and returns the sampled signal.

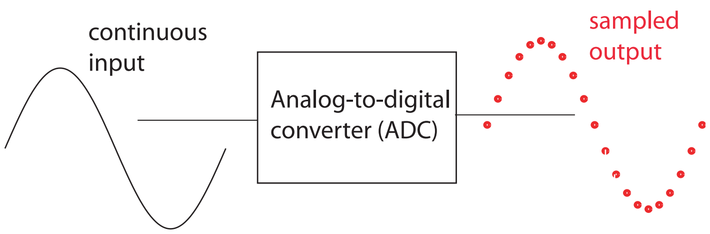


Figure 10: ADC.

7.5 Curve fitting

After sampling, it might be required to fit the samples to a mathematical function. This could be done by using the curve fitting techniques, which we have learnt in the Matlab tutorials. Figure 11 shows how the acquired samples are fitted by Matlab command **lsqcurvefit** that employs the least square curve fitting algorithm

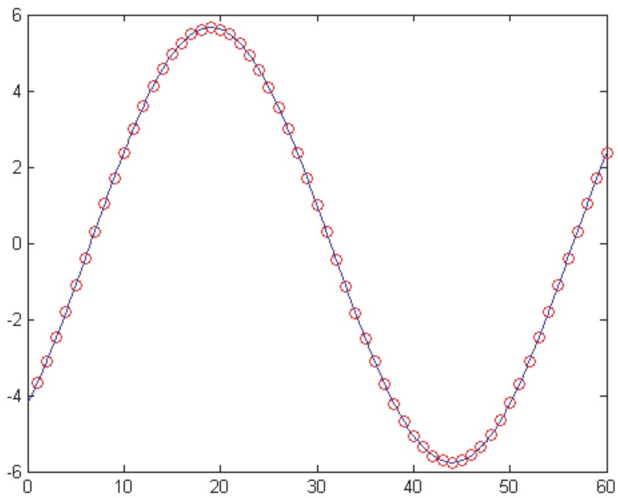


Figure 11: Least Square curve fitted waveform.

7.6 Signal reconstruction

The result of data acquisition is an array of sampled points. The analog, continuous nature of the signal has been lost. After the process of sampling, the resulting sample values can be used for re-constructing the original signal. But, we have to determine the appropriate sampling rate that allows for reconstruction. If the sampling rate is not high enough, it is impossible to get the original signal back. This is the essence of the *Nyquist Theorem*.

7.7 Nyquist sampling theorem

The theorem states that for the perfect reconstruction of a signal the sampling frequency must be greater than twice the maximum frequency of the signal being sampled.

★ **Q** 1. Can you guess what is the minimum number of samples per second required to reconstruct the 1 Hz sine wave?

7.8 Experiment

We now use our DAQ system to generate and acquire a signal and sample it at different rates. We observe the appearance of the corresponding reconstructed signal and determine the frequency of the reconstructed signal. We will comment on the accuracy of the reconstruction process, Thus, we will experimentally verify the Nyquist Theorem, and of course, in the process we also learn about sampling.

7.9 Procedure

Q 2. Carry out the following procedure.

1. Familiarize yourself with the apparatus. Tell instructor what each component does.
2. Connect the signal generator to DAQ.
3. Generate a sine wave as an input signal using the signal generator.
4. Set the otput frequency on the signal generator to 10 Hz.
5. Run the **nyquist.vi** file.
6. In the **Block Diagram** window (Ctrl+E), enter the **rate**. It is the sampling rate, at which ADC samples the input signal (start with 1000 Hz).
7. Samples are stored in **nyq.lvm** file, and this file must be in the current directory of the MATLAB. (In the block diagram right-click the **Write-to-measurement** block and select properties to change directory).
8. In the **front panel** window click the **Run** button (shown by arrow key).
9. Data acquisition starts, the output starts appearing on the waveform graph (stop after a few cycles appear).
10. Now start *Matlab*. Ensure the workspace directory is same as **nyq.lvm**.
11. In the Matlab window, type,

    $\gg$ **nyquist**
12. This command asks you to input the sampling rate, which then returns the reconstructed waveform.
13. Now observe the outputs and complete the following table on your note book, with sampling rates of 3, 4 , 8, 15, 20, 30, 50, 70, 100, 500, 1000 Hz. Start from 1000 Hz. After each acquisition clear the labview output chart (right-click on the chart, go to 'data operations' and select 'clear chart').
14. Using cursors on Matlab determine frequency. Show cursor in printout. (Bonus: Use subplot).

Sampling Rate	Frequency of the reconstructed waveform (Hz)
3 . .	

Table 1: Illustrating Nyquist Theorem

★ **Q** 3. Write your observations and inferences on your note books.

8 Filter Design

8.1 Objective

Having understood what is meant by frequency, the present section introduces the concept of a *filter*, its operation and design.

8.2 Frequency content of signals

A signal may be composed of one or more frequencies. Thus each signal could be expressed in terms of its frequencies as well.

8.3 What is a filter?

Filter is an electronic device which allows a range of certain frequencies to pass through and blocks the rest.

8.4 Cutt-off or corner frequency

The cut-off frequency of the filter is the frequency at which the power output of the filter is reduced to half of its maximum. This frequency is denoted by f_c.

Figure 12 shows a filter made from a resistor R and a capacitor C. This filter is an RC filter. This can be used for making both *low pass* as well as *high pass filters*. We will describe these terms shortly. The input signal comes into the left and the output is taken from the right.

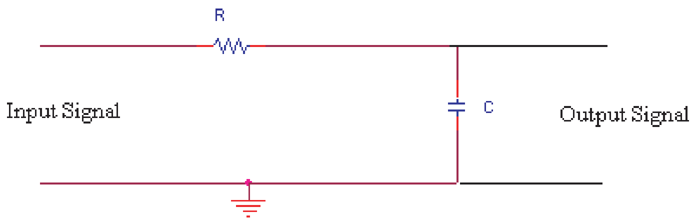


Figure 12: An RC Filter.

8.5 Filter diagrams

Figure 13(a) shows a low pass filter, (b) shows a high pass filter and (c) shows a band pass filter. For example, a low pass filter allows low frequencies to pass through and blocks high frequencies. The transition between the blocked and the unblocked frequencies, however, is not sharp. The output power decreases smoothly. At an applied frequency equal to f_c, the output power is reduced to half as compared to the maximum.

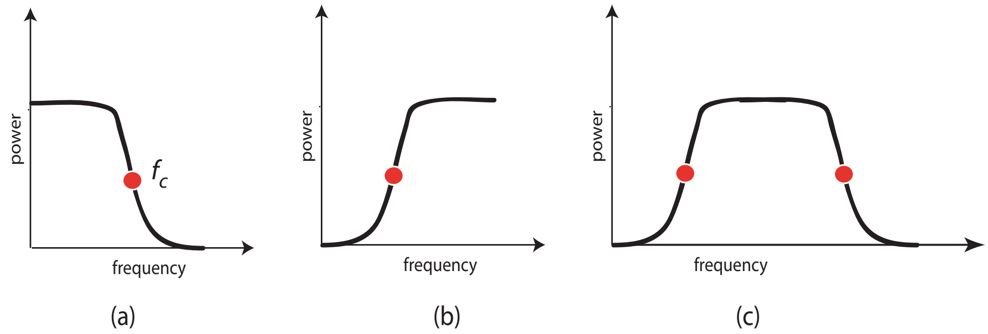


Figure 13: Filter Diagrams; the output power from the filter is plotted with respect to frequency.

8.6 Design considerations

For RC circuits, cut-off frequencies are determined using the formula,

$$f_c = \frac{1}{2\,\pi\,R\,C}. \tag{2}$$

8.7 Calculational example

1. Select a suitable cut-off frequency for the filter.
2. Take any standard value of the resistor available.
3. Then, using Equation 2, determine the value of capacitor, to achieve the desired f_c.

8.8 Low pass filter

In order to make a low pass filter, we take the output across the capacitor, as shown in Figure 12. This will *attenuate* all the high frequencies present in the input signal and will allow the low frequency components to pass. Attenuation means that the filter will reduce the output amplitude for the high frequency components.

8.9 High pass filter

In order to make a high pass filter, we take the output across the resistor, as shown in Figure 14. This will attenuate all the low frequencies present in the input signal, allowing the high frequency components to pass.

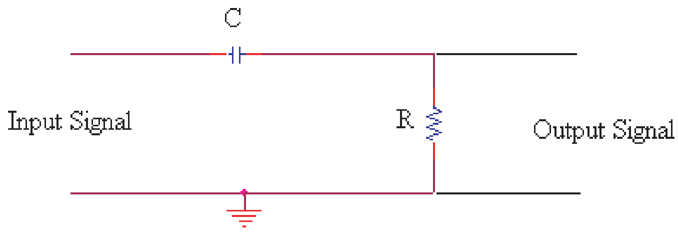


Figure 14: High Pass Filter.

8.10 Mathematical explanation

As we know a resistor has a property called the *resistance* R. The resistance quantifies the opposition to the flow of electrons, Similarly, a capacitor has an analogous property called the *reactance* denoted as X_c. The capacitive reactance is

$$X_c \propto \frac{1}{\omega C} = \frac{1}{2\,\pi\,f\,C}. \tag{3}$$

The above relation tells that capacitive reactance is inversely related to frequency and capacitance. Thus by decreasing frequency, reactance increases. When the frequency is zero, i.e., the input voltage is constant, the reactance becomes infinite, indicating an open circuit, as shown in Figure 15(b). This means that no current will flow through the circuit and the voltage drop across the resistor R will be zero ($I\,R = 0$). But according to Kirchoff's rules, the output voltage must be equal to the input voltage. Hence the drop across the capacitor will be equal to the input voltage and hence, maximum.

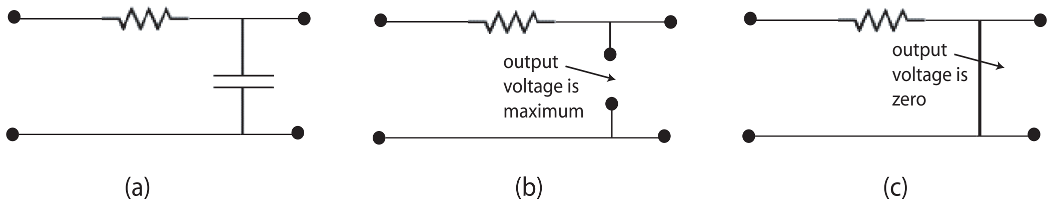


Figure 15: (a) Behaviour of the low pass filter under the conditions of (b) DC input voltage and (c) very high frequencies.

Likewise, for extremely high frequencies, the reactance of the capacitor drops to very small values and it behaves like a short circuit—an ordinary piece of wire. This is shown in Figure 15(c). The drop across the capacitor becomes zero and hence the output voltage is zero.

★ **Q** 4. Draw an illustration similar to Figure 15 for the high pass filter of Figure 14 and describe how this circuit attenuates low frequencies.

★ **Q** 5. When a DC voltage is applied to a capacitor, does the capacitor conducts current?

★ **Q** 6. When a sinusoid of any frequency is applied, the capacitor conducts current. However, in between the plates of the capacitor, we have air that is a poor conductor. How does the current pass through the insulator?

8.11 Procedure

★ **Q** 7. Carry out the following procedure.

1. Design a low pass filter with cut-off frequency of approximately 1.3 KHz using the provided components (Hint: Use $R > 250\,\text{ohms}$).
2. Run the **filter.vi** file.
3. In the **front panel** window click the **Run** button (shown by the arrow key).
4. Generate a sine wave using the signal generator. This is going to be used as an input signal for the rest of our experiment.
5. Use labview to set the peak input voltage V_{in} to 5 V (Connect the input signal directly to the output of DAQ).
6. Set up your designed filter on breadboard.
7. Input sine wave into filter.
8. Data acquisition starts, the filter's output starts appearing on the waveform graph.
9. Observe the waveforms and carry out the following procedures.

8.12 Low Pass Filter

1. Take the output across the capacitor resulting in a low pass filter. All grounds must be on the same node.
2. Now, you will successively change the input frequency, f_{in} to 0.1, 0.2, 0.4, 0.6, 0.8, 1.0, 1.2, 1.4, 1.6, 2.0, 3.0, 4.0, 5.0, 6.0, 7.0, 8.0, 9.0 and 10.0 KHz.
3. Complete Table 2 in your note books.
4. Use Matlab to plot the graph between f and V_{out}. Use **semilogx** plot and a normal plot of V_{out} versus f.
5. Use Matlab to plot the graph between f and G that is called the *gain* of the filter. Use **semilogx**.
6. Matlab command to compute the Gain is G = 20*log10(V_{out}./Vin)

Frequency f_{in} (KHz)	Peak output voltage V_{out} (V)	$G = 20 \times \log\left(\frac{V_{out}}{V_{in}}\right)$ (dB)
0.1 ⋮		

Table 2: Suggested format for tabulating the experimental results for the low pass filter.

8.13 High Pass Filter

★ **Q** 8. For the high pass filter, carry out the following procedure.

1. Take the output across the resistor resulting implementing the high pass filter. All grounds on the same node.
2. Successively change the input frequency, f_{in} to 0.2, 0.4, 0.6, 0.8, 1.0, 1.2, 1.4, 1.6, 1.8, 2.0, 3.0, 4.0, 5.0, 10.0, 15.0 and 20.0 KHz.
3. Complete Table 3 in your note books.
4. Use Matlab to plot the graph between f and V_{out}. Use **semilogx** plot and a normal plot of V_{out} versus f.
5. Use Matlab to plot the graph between f and G. Use **semilogx**.

Frequency f_{in} (KHz)	Peak output voltage V_{out} (V)	$G = 20 \times \log\left(\frac{V_{out}}{V_{in}}\right)$ (dB)
0.2 ⋮		

Table 3: Suggested format for tabulating the experimental results for the high pass filter.

★ **Q** 9. Based on the plotted results, describe your observations on the low and high pass filters.

★ **Q** 10. How do the semi-log plots (plots between the gain and frequency) compare with the plots between output voltage and frequency?

9 Filtering a Composite Signal

9.1 Objective

This section introduces the use of an *operational amplifier* commonly abbreviated as the *opamp*. It also represents a more illuminating verification of the concept of filtering introduced in the previous Sections.

9.2 Composite Signal

Two or more signals are added to make, what we call a *composite signal*. In this section we are using an operational amplifier to add the two signals.

A signal may possess different frequency components. When two signals are added the resulting signal possess the frequency components of both signals. Thus in order to filter out a particular signal, we design a suitable corresponding filter (low pass, high pass, or band pass) to extract the desired components and leave the rest.

9.3 Block Diagram

The Block diagram shown in Figure 16, describes the flow chart of the experiment, in which two signals with different frequencies are added together using an opamp to produce a composite signal which is then filtered.

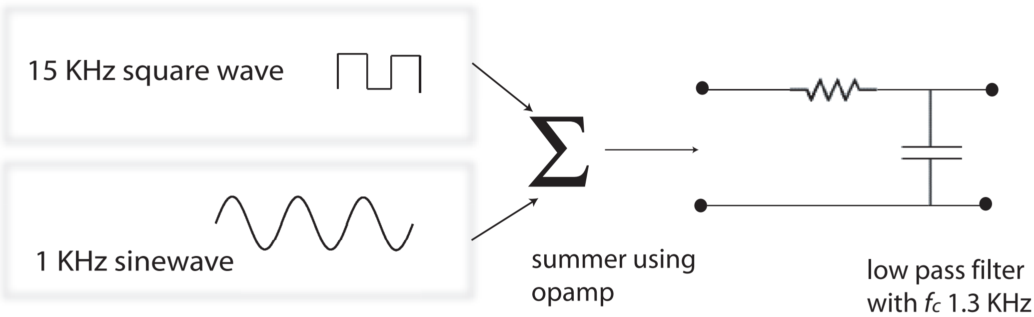


Figure 16: Block diagram showing low pass filtering of the composite signal.

9.4 Frequency Spectrum of Signals

★ **Q** 11. Carry out the following procedure to see the frequency spectrum .
Note down all the observations on your note books.

1. Generate a sine wave of 1 KHz.
2. Run the Labview file **composite.vi**.
3. In the **front panel** window click the **Run** button (shown by the arrow key).
4. See the sine wave on the waveform graph.
5. Start Matlab and type
   ≫ **signalspect**

6. It shows the frequency spectrum of the signal. You can see a single peak on the frequency axis at 1 KHz.

Q 12. Repeat the same procedure to see the frequency spectrum of a square wave of 1 KHz. It consists of frequencies of 1KHz, 3KHz, 7KHz, and so on. It consists of odd harmonics of fundamental frequency (1KHz in this case).

★ **Q** 13. Print the composite signal waveforms (both in time and frequency domain)

★ **Q** 14. Print the square wave signal after low pass filtering (both in time and frequency domain).

9.5 Procedure

★ **Q** 15. Carry out the following procedural steps. The circuit diagram is shown in Figure 17.

1. The first input signal is taken from the signal generator. This input is a square wave of frequency 1 KHz.
2. Take another sine wave of frequency 1 KHz from signal generator.
3. Setup the summer circuit shown in Figure (17). You give the input signals at proper places on the breadboard.
4. Google image the UA741 IC and find out where the $+V_{cc}(+12\,\mathrm{V})$ and $-V_{cc}(-12\,\mathrm{V})$ are connected.
5. The output composite signal can be viewed from pin no. 6 of the opamp on the breadboard.
6. Run the Labview file **composite.vi**.
7. In the **front panel** window click the **Run** button (shown by the arrow key).
8. Data acquisition starts, the filter's output starts appearing on the waveform graph.
9. You can see the different frequencies in the composite signal.
10. Start Matlab and type
    $\gg$ **signalspect**
11. Now use your low pass filter of cut off frequency approximately 1.3 KHz. Assemble it on the breadboard and filter the composite signal.
12. Observe the filtered output.
13. Place all the components back in their racks.

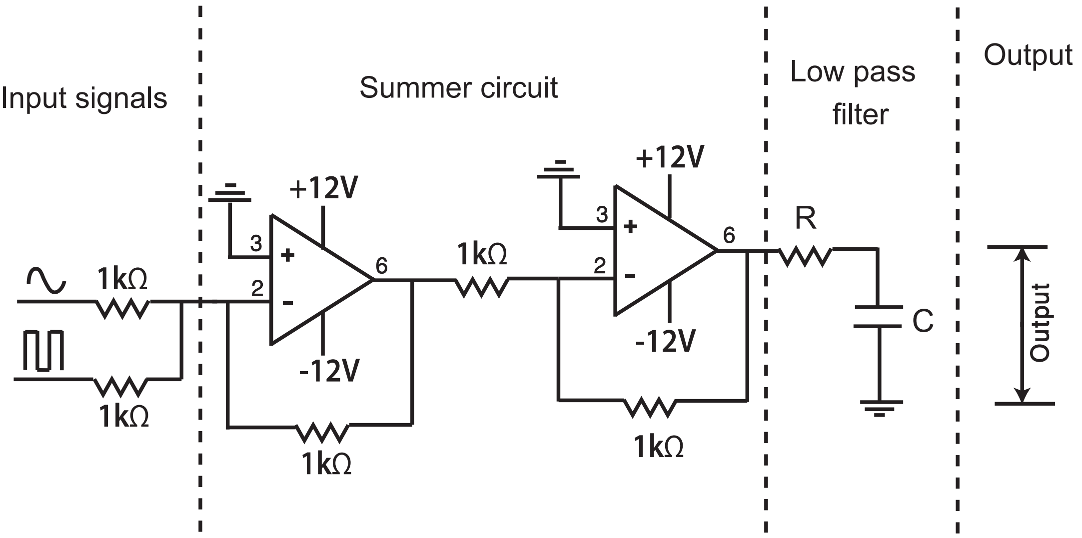


Figure 17: Circuit diagram for generation and low pass filtering of the composite signal.

10 Idea Experiments

1. How do we reduce noise? One method is to acquire more and more samples points, the signal grows more rapidly than the noise, increasing the signal-to-noise ratio. Design a simple experiment to verify this [4].

2. Determine the Boltzmann constant using measurements of Johnson noise [5].

3. The electrocardiogram (ECG) is an example of a scenario where the signal of interest can be buried in a high level of noise. Design a circuit that measures the ECG from a human body, enhancing the signal. Also filter out the interfering noise from the 50 Hz mains [6].

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