Active Filter Trainer
NV6504

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Learning Material
Ver 1.1

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Active Filter Trainer
NV6504
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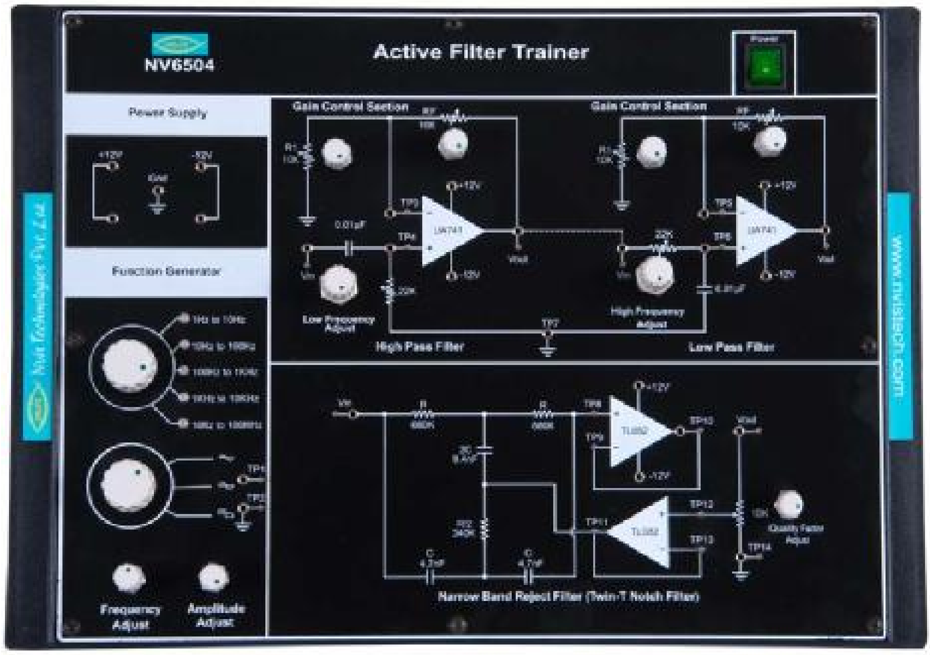
Introduction

NV6504 Active Filter Trainer is remarkable and frequently used laboratory equipment which is designed to explain basic fundamentals of filter design and their working. Electronic filters are electronic circuits which perform signal processing functions, specifically intended to remove unwanted signal components and/or enhance wanted ones. Electronic filters can be :

- Passive or Active
- Analog or Digital
- Linear or Non-Linear

The most common types of electronic filters are linear filters, regardless of other aspects of their design. The trainer is designed keeping in mind that a student can comprehend with each block of Active Filter in a very easy way.



Features

- **A low cost trainer demonstrating all the basic concepts of Active Filters**
- **Exclusive presentation and easy illustration of each part of the Filter Circuit**
- **Designed with considering all the Safety Standards**
- **Provided with inbuilt Function Generator**
- **Provided with inbuilt Power Supply**
- **Selectable frequency range of Function Generator**
- **Manual creation of Band Pass Filter using high pass and low pass filter**
- **Twin-T Notch Filter with Quality Factor adjust**
- **Learning Material CD**

Technical Specifications

1. Frequency range of Function Generator : Selectable

- 1 Hz to 10Hz
- 10Hz to 100Hz
- 100Hz to 1KHz
- 1 KHz to 10 KHz
- 10 KHz to 100 KHz

2. Amplitude controlled output

3. Filters :

- Accurate Frequency Response
- Variable Cutoff Frequencies
- Adjustable Gain of output

4. Power Supply : 230 V ±10%, 50 Hz

5. Fuse : 350 mA

Theory

A Brief Introduction to the Filters :

"In this digital age, who needs continuous-time filter?" Such an obvious question and one which deserves an immediate response. True, we do live in a digital age-digital computer, digital communication, digital broadcasting. But much through digital technology may bring us advantages over analog system, at the end of the day a digital system must interface with the real world-the analog world. For example to gain the advantage that digital signal processing can offer, that processing must take place on band limited signals, if unwanted aliasing effects are not to be introduced. After the processing, the signals are returned to the real analog world after passing through a reconstruction filter. This is but one example-but any system that interfaces with the real world will find use for continuous filters.

The term continuous time perhaps needs some explanation. There was the time when analog filters were just that-they processed analog signal in real time, in contrast to the digital filters which performed the filtering operations on digital representation of samples of signals, often not in real time. Then in 1970s, analog came sampled data filters. Sampled data filter did not work with the digital representation of sampled signal, but operated on the samples themselves. Perhaps the best known example of such an approach is that of switched-capacitor filter, which as the name suggests, use switches (usually transistor switches) together with the capacitors and active devices to provide filter functions. Note that these filters are discontinuous in time as the result of the switching which takes place within the circuits; indeed continuous band limiting and reconstruction filter are needed as a result. Much research took place in 1970's and 1980's on switched capacitor as a result of the advantages for integrated circuit realization that they promised. There was so much stress on research in this area that development of the more conventional analog filters received relatively little attention. However when switched capacitor filter failed to provide all the solutions, the attention once again turned to the more traditional approaches and the name continuous-time filter was coined to differentiate them for their digital and sampled data counterparts.

The classic LCR filters built with the inductors, capacitors and resistors are such filters, of course, and indeed are still much in use. However these filters are unsuitable to implement in the integrated circuits, since no satisfactory way of making inductors on chip has been found. That is why so much attention has been paid to active continuous –time filter over the year. Active filters offer the opportunity to integrate complex filters on chip, and do not have the problem that the relatively bulky, lossy, and expensive inductors bring, in particular their stray magnetic field that can provide unwanted coupling in a circuit or system.

As just mentioned, active filters have been around for some time as a means of overcoming the disadvantages associated with passive filter (of which the use of inductance is one).

Electronic Filter Topology :

An Electronic Filter Topology is an electronic analog filter circuit in which the values of the components remain undefined. A particular Topology is then characterized entirely by the manner in which the components are connected, and not by their values.

There are many different types of electronic filters and they are characterized by their transfer function, but not by any particular Topology. Once the transfer function for a filter is chosen, it remains to select the particular Topology to implement that filter. For example, one might choose to design a Butterworth filter using the Sallen Key Topology.

List of Topologies :

- Cauer Topology – Passive
- Sallen Key Topology – Active
- Multiple Feedback Topology – Active
- State Variable Topology – Active
- Biquadratic Topology – Active

Cauer Topology :

Wilhelm Cauer proposed a number of electronic filter designs or circuit topologies for realization of driving point impedance.

Cauer's forms of driving point impedances :

Cauer's first form of driving point impedance consisted of shunt capacitors and series inductors. If we take an output at the far end of the driving point impedance so-realized, we end up with a Low Pass Filter.

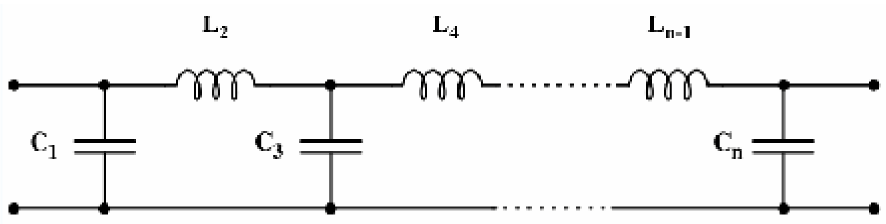


Figure 1

Cauer's second form consisted of series capacitors and shunt inductors. If we take an output at the far end of the driving point impedance so-realized, we end up with a High Pass Filter.

Sallen Key filter :

A Sallen and Key filter is a type of active filter, particularly valued for its simplicity. The circuit produces a 2-pole (12dB/octave) lowpass or highpass response using two resistors, two capacitors and (usually) a unity-gain buffer amplifier. Higher-order filters can be obtained by cascading two or more stages. This filter Topology is also known as a voltage controlled voltage source (VCVS) filter. It was introduced by R.P. Sallen and E. L. Key of MIT Lincoln Laboratory in 1955.

Although the filters depicted here have a passband gain of 1 (or 0 dB), not all Sallen and Key filters have a gain of 1 in the passband. Non-unity-gain buffers can also be used

Multiple Feedback Topology (Electronics) :

Multiple Feedback Topology is an Electronic Filter Topology which is used to implement an electronic filter, by adding two poles to the transfer function. A diagram of the circuit Topology for a second order Low Pass Filter is shown in the figure 2.

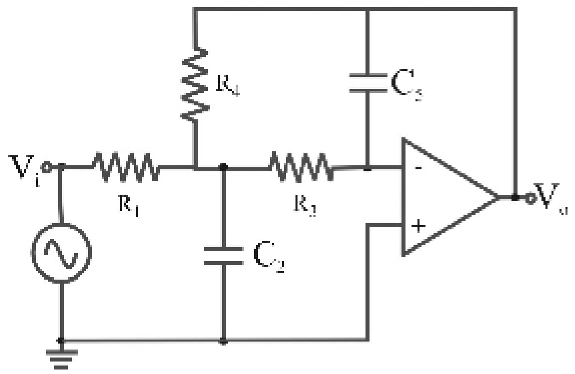


Figure 2

State Variable Topology :

State Variable Topology refers to filters with two pole filter Topology. State variable filters are available in lowpass, highpass, bandpass, and notch versions. The notch version is not recommended, because it requires too many amplifiers. It is one of three topologies that offer complete and independent control over gain, frequency, and type (Butterworth, Chebyshev, and Bessel). Signal input is at one place, and the output is taken from different places. The State Variable filter has the unique characteristic of producing low pass, high pass, band pass and notch outputs simultaneously.

Classification of Filters :

Real-world signals contain both wanted and unwanted information. Therefore, some kind of electronic signal filtering technique must separate the two before processing and analysis can begin. Every electronic design project produces signals that require electronic signal filtering, processing, or amplification, from simple gain to the most complex Digital-Signal Processing (DSP). According to the requirement and their functioning filters are classified in different categories which can be shown graphically as follows.

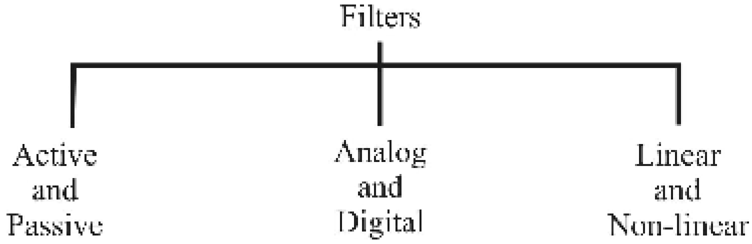


Figure 3

A. Active and Passive Filters

Active Filters : An active filter is a type of analog electronic filter, distinguished by the use of one or more active components i.e. voltage amplifiers or buffer amplifiers. Typically this will be a vacuum tube, transistor or operational amplifier.

There are two principal reasons for the use of active filters. The first is that the amplifier powering the filter can be used to shape the filter's response, e.g., how quickly and how steeply it moves from its pass band into its stop band. (To do this passively, one must use inductors, which tend to pick up surrounding electromagnetic signals and are often quite physically large.) The second is that the amplifier powering the filter can be used to buffer the filter from the electronic components it drives. This is often necessary so that they do not affect the filter's actions.

Passive Filters : Passivity is a property of engineering systems, most commonly used in electronic engineering and control systems. A passive component, depending on field, may either refer to a component that consumes (but does not produce) energy, or to a component that is incapable of power gain. A component that is not passive is called an active component. An electronic circuit consisting entirely of passive components is called a passive circuit (and has the same properties as a passive component). A passive filter is a kind of electronic filter that is made only from passive elements -- in contrast to an active filter, it does not require an external power source (beyond the signal). Since most filters are linear, in most cases, passive filters are composed of just the four basic linear elements-- resistors, capacitors, inductors, and transformers. More complex passive filters may involve nonlinear elements, or more complex linear elements, such as transmission lines.

A passive filter has several advantages over an active filter :

- Guaranteed stability
- Passive filters scale better to large signals (tens of amps, hundreds of volts), where active devices are often impractical
- No power consumption (aside from possibly taking some power out of the signal)

- Cheap
- For linear filters, generally, more linear than filters including active (and therefore non-linear) elements

B. Analog and Digital Filters :

Analog Filters : An analog filter handles analog signals or continuous-time signals, whether electric potential, sound waves, or mechanical motion directly. This is opposed to a digital filter that operates on discrete-time signals.

Digital Filters : In electronics, a digital filter is any electronic filter that works by performing digital mathematical operations on an intermediate form of a signal. This is in contrast to older analog filters which work entirely in the analog realm and must rely on physical networks of electronic components (such as resistors, capacitors, transistors, etc.) to achieve the desired filtering effect.

Digital filters can achieve virtually any filtering effect that can be expressed as a mathematical function or algorithm. The two primary limitations of digital filters are their *speed* (the filter can't operate any faster than the computer at the heart of the filter), and their *cost*. However as the cost of integrated circuits has continued to drop over time, digital filters have become increasingly common place and are now an essential element of many everyday objects such as radios, cell phones, and stereo receivers.

C. Linear and Nonlinear Filters :

Linear Filters : A linear filter applies a linear operator to a time-varying input signal. Linear filters are very common in electronics and digital signal processing, but they can also be found in mechanical engineering and other technologies. They are often used to eliminate unwanted frequencies from an input signal or to select a desired frequency among many others. There are a wide range of types of filters and filter technologies. Linear filters can be divided into two classes: Infinite Impulse Response (IIR), and Finite Impulse Response (FIR) filters.

Nonlinear filter : A non-linear filter is a signal-processing device whose output is not a linear function of its input. Non-linear filters are ideal for removing very short wavelength, but high amplitude features from data. It is often thought of as a noise spike-rejection filter, but it can also be effective for removing short wavelength geological features

A brief Introduction to some basic Filters :

Chebyshev filter :

Chebyshev filters are analog or digital filters having a steeper roll-off and more passband ripple (type I) or stopband ripple (type II) than Butterworth filters. Chebyshev filters have the property that they minimize the error between the idealized filter characteristic and the actual over the range of the filter, but with ripples in the passband. This type of filter is named in honour of Pafnuty Chebyshev because their mathematical characteristics are derived from Chebyshev polynomials.

Because of the passband ripple inherent in Chebyshev filters, filters which have a smoother response in the passband but a more irregular response in the stopband are preferred for some applications.

Butterworth filter :

The Butterworth filter is one type of electronic filter design. It is designed to have a frequency response which is as flat as mathematically possible in the passband. Another name for them is 'maximally flat magnitude' filters. The Butterworth type filter was first described by the British engineer Stephen Butterworth.

The frequency response of the Butterworth filter is maximally flat (has no ripples) in the passband, and rolls off towards zero in the stopband. When viewed on a logarithmic Bode plot, the response slopes off linearly towards negative infinity. For a first-order filter, the response rolls off at −6 dB per octave (−20 dB per decade) (All first-order filters, regardless of name, have the same normalized frequency response). For a second-order Butterworth filter, the response decreases at −12 dB per octave, a third-order at −18 dB, and so on. Butterworth filters have a monotonically changing magnitude function with ω. The Butterworth is the only filter that maintains this same shape for higher orders (but with a steeper decline in the stopband) whereas other varieties of filters (Bessel, Chebyshev, elliptic) have different shapes at higher orders.

Bessel filter :

In electronics and signal processing, a Bessel Filter is a variety of linear filter with a maximally flat group delay (linear phase response). Bessel filters are often used in audio crossover systems. Analog Bessel filters are characterized by almost constant group delay across the entire passband, thus preserving the wave shape of filtered signals in the passband. The filter is named in honour of Friedrich Bessel, a German mathematician (1784–1846).

Here we will discuss the Active Filters of following type :

- Low Pass Filter
- High Pass Filter
- Band Pass Filter

Low Pass Filters :

A Low-Pass Filter is a filter that passes low frequency signals but attenuates (reduces the amplitude of) signals with frequencies higher than the cutoff frequency. The actual amount of attenuation for each frequency varies from filter to filter. It is sometimes called a high-cut filter, or treble cut filter when used in audio applications.

The concept of a Low-Pass Filter exists in many different forms, including electronic circuits, digital algorithms for smoothing sets of data, acoustic barriers, blurring of images, and so on. There are different types of filter circuits, with different responses to changing frequency. The frequency response of a filter is generally represented using a Bode plot.

A first-order filter, for example, will reduce the signal amplitude by half (about –6 dB) every time the frequency doubles (goes up one octave). The magnitude Bode plot for a first-order filter looks like a horizontal line below the cutoff frequency, and a diagonal line above the cutoff frequency. There is also a "knee curve" at the boundary between the two, which smoothly transitions between the two straight line regions.

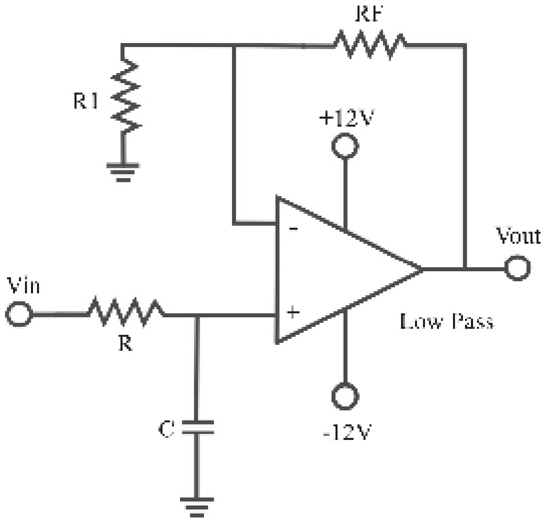


Figure 4

Frequency Response Curve of Low Pass Filter :

Ideal Frequency Response Curve :

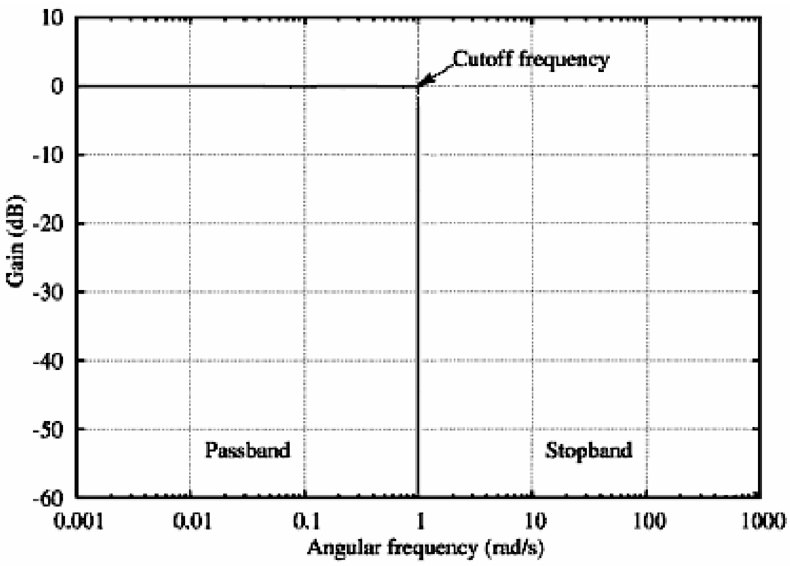


Figure 5

Practical Frequency Response Curve :

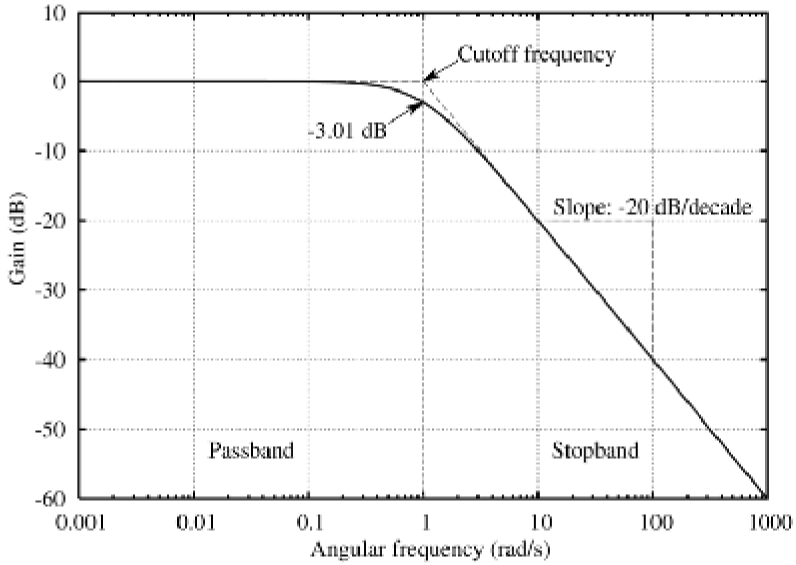


Figure 6

Mathematical Calculations :

$$V_1 = \frac{-jX_c}{R - jX_c} Vin \qquad \text{.................(1)}$$

where $j = \sqrt{-1}$

and $$jX_c = \frac{1}{j2\pi f_c} \qquad \text{.................(2)}$$

On putting this value in equation (1) we get

$$V_1 = \frac{Vin}{1 + j2\pi fRC} \qquad \text{.................(3)}$$

and $$Vo = \left(1 + \frac{Rf}{Rl}\right) V1 \qquad \text{.................(4)}$$

=> $$Vo = \left(1 + \frac{Rf}{Rl}\right)\left(\frac{Vin}{1 + j2\pi fRC}\right) \qquad \text{.................(5)}$$

$$\frac{Vo}{Vin} = \frac{Af}{1 + j\left(\frac{f}{fh}\right)} \qquad \text{.................(6)}$$

f/fh is an imaginary term so we can write the above equation as

$$\frac{Vo}{Vin} = \frac{Af}{\sqrt{1+\left(\frac{f}{fh}\right)^2}} \qquad \text{………………(7)}$$

where

Vo/Vin	=	Gain of filter in frequency function
Af	=	1 + Rf / R1
f	=	Frequency of input signal
fh	=	1 / 2πRC = High cutoff frequency

So, we can see that

| Vo/Vin |	=	Af	(when f < fh)
| Vo/Vin |	=	Af /√2 = 0.707Af	(when f = fh)
| Vo/Vin |	<	Af	(when f > fh)

Examples of Low Pass Filters :

Acoustic :

A stiff physical barrier tends to reflect higher sound frequencies, and so acts as a low-Pass Filter for transmitting sound. When music is playing in another room, the low notes are easily heard, while the high notes are attenuated.

Electronic :

Electronic Low-Pass Filters are used to drive subwoofers and other types of loudspeakers, to block high pitches that they can't efficiently broadcast.

Radio transmitters use Low Pass Filters to block harmonic emissions which might cause interference with other communications.

An integrator is another example of a low-pass filter.

DSL splitters use Low-Pass and high-pass filters to separate DSL and POTS signals sharing the same pair of wires.

Low-Pass Filters also play a significant role in the sculpting of sound for electronic music as created by analogue synthesisers.

Multi Order Filters :

There are a great many different types of filter circuits, with different responses to changing frequency. The frequency response of a filter is generally represented using a Bode plot.

A first-order filter, for example, will reduce the signal amplitude by half (about –3 dB) every time the frequency doubles (goes up one octave). The magnitude Bode plot for a first-order filter looks like a horizontal line below the cutoff frequency, and a diagonal line above the cutoff frequency. There is also a "knee curve" at the boundary between the two, which smoothly transitions between the two straight line regions.

A second-order filter does a better job of attenuating higher frequencies. The Bode plot for this type of filter resembles that of a first-order filter, except that it falls off more quickly. For example, a second-order Butterworth filter will reduce the signal amplitude to one fourth its original level every time the frequency doubles (–6 dB per octave). Other second-order filters may roll off at different rates initially depending on their Q factor, but approach the same final rate of –12 dB per octave. Third and higher-order filters are defined similarly. In general, the final rate of rolloff for an n-order filter is 6n dB per octave.

Experiment 1

Objective :

Study of Active Low Pass Filter and to Evaluate :

- **High cutoff frequency of Low Pass Filter**
- **Pass band gain of Low Pass Filter**
- **Plot the frequency response of Low Pass Filter**

Equipments Needed :

1. Oscilloscope
2. Digital Multimeter

Circuit Diagram :

Circuit used to study Active Low Pass Filter shown in figure 7.

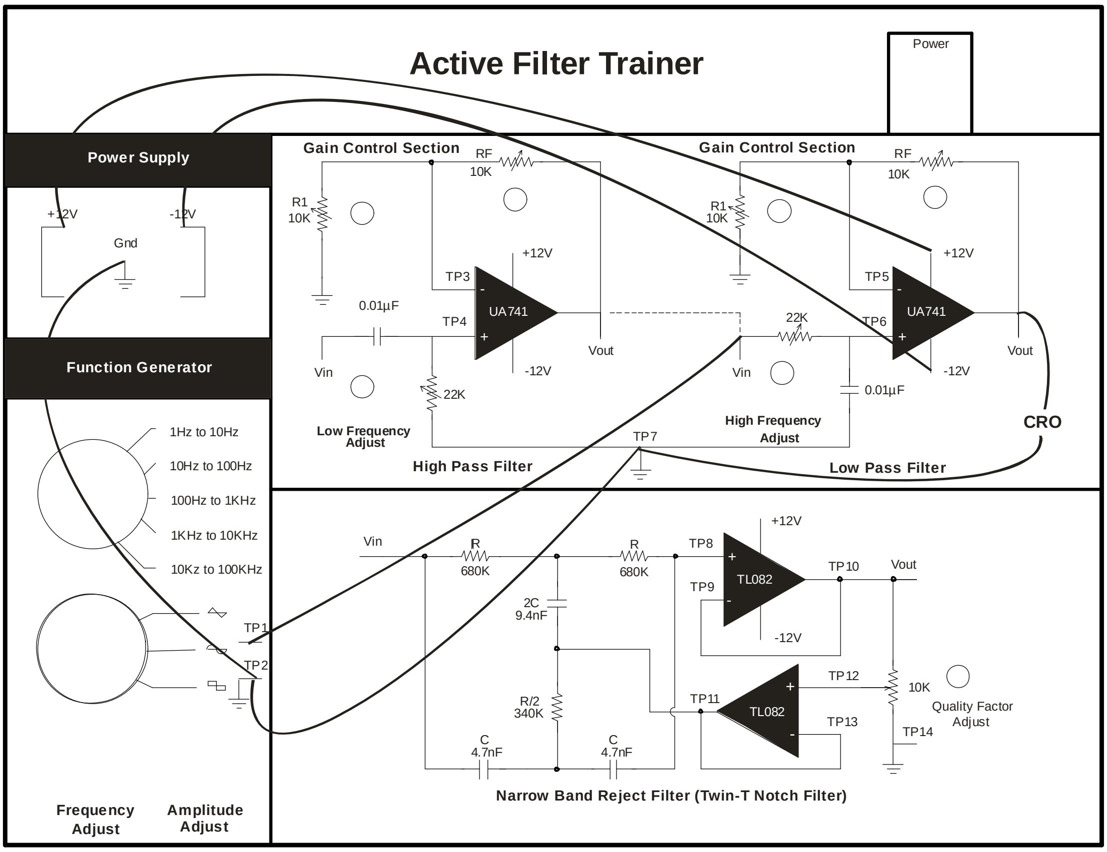


Figure 7

Procedure :

1. Initially rotate potentiometers R1 and Rf in fully clockwise direction in order to make R1 = Rf = 10K, so that according to the formula given below

   $$Vo = (1+Rf/R1)\ Vin$$

   The gain of the output will be twice of the input.
2. The high cutoff frequency is given by the formula:

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

3. Connect an Ohmmeter between Vin of Low Pass Filter and TP6. Adjust resistance value at 1.59K by varying the potentiometer of 22K to set the high cutoff frequency (fh) at 10K.
4. Connect +12V and –12V DC power supplies at their indicated position from Power Supply section.
5. Connect all the ground test points using patch chords.
6. Switch ‘On’ the Power Supply.
7. Set the output of function generator at 2 volt, 500 Hz using Oscilloscope with sinusoidal waveform.
8. Connect TP1 with Vin of Low Pass Filter to give a sinusoidal signal of amplitude 2Vpp of frequency 500Hz to Low Pass Filter.
9. Observe output on Oscilloscope.
10. Similarly you can give the triangular and square wave instead of sinusoidal wave, from the function generator section.
11. As we know according to the formula, Output gain is directly proportional to the Rf and inversely proportional to the R1. So, we can adjust the gain of the output by increasing the value of Rf as well as by decreasing the value of R1.
12. So, change the values of R1 and Rf and observe the change in output.
13. Increase the frequency of input signal step by step and observe the effect on output Vout on Oscilloscope.
14. Tabulate the values of Vout, gain, gain (db) at different values of input frequency shown in Observation Table.

Observation Table :

S. No.	Input frequency (Hz)	Vout	|Vout/Vin| = gain	Gain(db) = 20 Log |Vout/Vin|
1	500			
2	1 K			
3	5 K			
4	10 K (f_H)			
5	15 K			
6	20 K			
7	30 K			

15. Plot the frequency response of Low Pass Filter using the data obtained at different input frequencies.

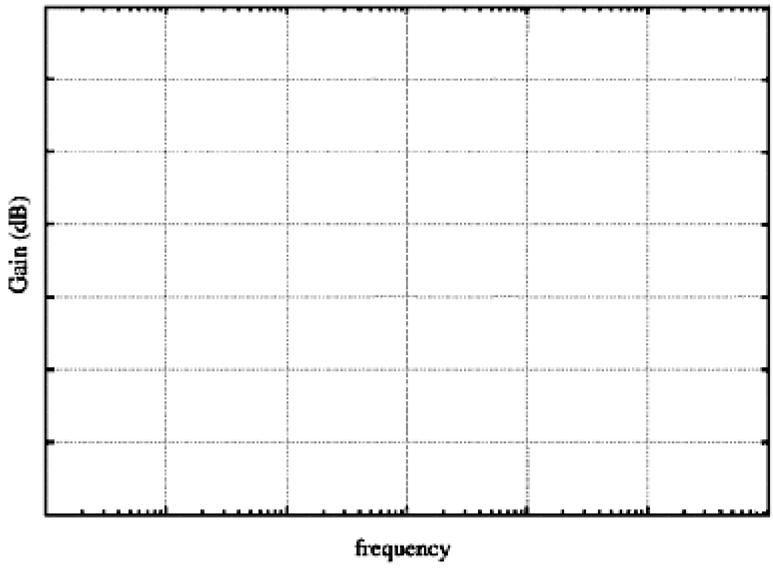


Figure 8

16. As we know, according to the formula

$$Vout = (1+Rf/R1)\ Vin$$

where

Rf is directly proportional to the Vout

And R1 is inversely proportional to Vout

So by varying the pots Rf & R1 we can adjust the gain of the output.

17. Perform the same procedure at different Cutoff frequencies as shown below:

Resistance (W)	**Capacitance (mF)**	**fh-high cutoff frequency (Hz)**
800	0.01	20K
1.59 K	0.01	10K
15.9 K	0.01	1K

Theoretical Calculations:

Calculate all the following values

1. Pass band gain of Low Pass Filter $A_F = 1 + R_F / R_1$

=…………………………………………..

2. Pass band gain (db) = 20 log |Vout/Vin|

=…………………………………………..

3. 3 db frequency $f_H = 1/2\pi RC$

=…………………………………………..

4. Gain at 3 db frequency $f_H = 0.707 * A_F$

5. Gain (db) at 3 db frequency f_H = 20 log |Vout/Vin|

where Vout = $(2)^{1/2}$ * Vin

High Pass Filter :

A high-pass filter is a filter that passes high frequencies but attenuates (or reduces) frequencies lower than the cutoff frequency. The actual amount of attenuation for each frequency varies from filter to filter. It is sometimes called a low-cut filter; the terms rumble filter is also used in audio applications. A high-pass filter is the opposite of a low-pass filter.

It is useful as a filter to block any unwanted low frequency components of a complex signal while passing the higher frequencies. Ofcourse, the meanings of 'low' and 'high' frequencies are relative to the cutoff frequency chosen by the filter designer.

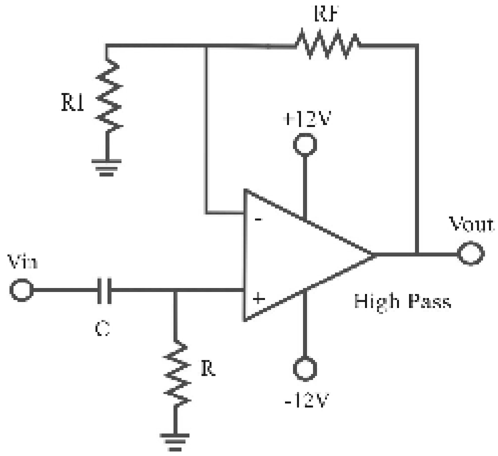


Figure 9

Here also the lower cutoff frequency is given by the same formula that is:

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

Frequency Response Curve Of Low Pass Filter :

Ideal Frequency Response Curve :

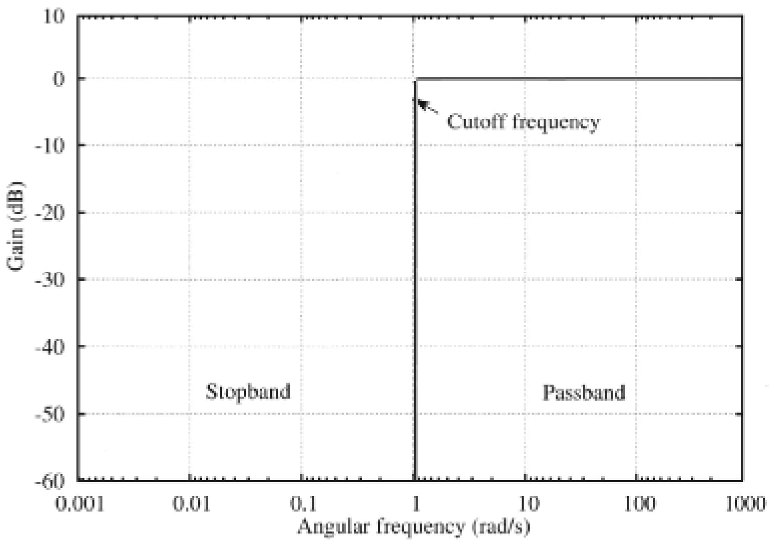


Figure 10

Practical Frequency Response Curve :

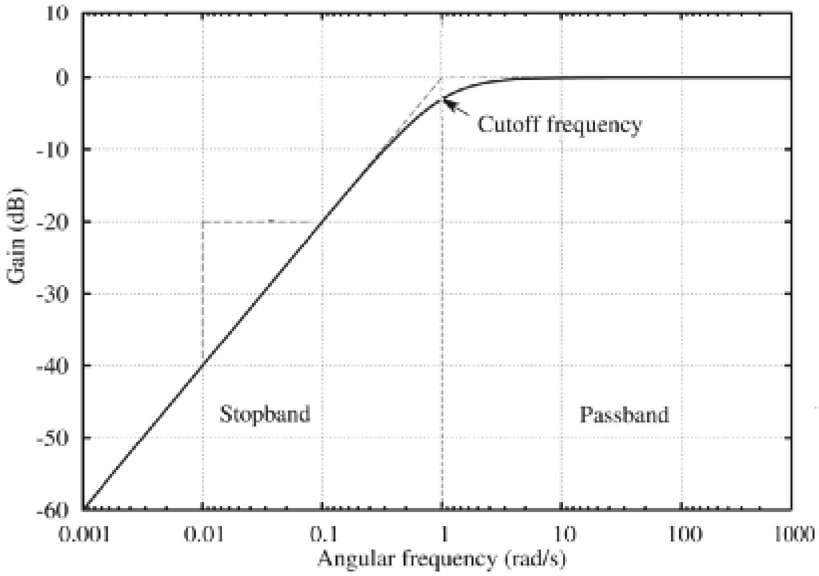


Figure 11

Applications :

Such a filter could be used to direct high frequencies to a tweeter speaker while blocking bass signals which could interfere with or damage the speaker. A low-pass filter, using a coil instead of a capacitor, could simultaneously be used to direct low frequencies to the woofer.

High-pass and low-pass filters are also used in digital image processing to perform transformations in the spatial frequency domain.

Most high-pass filters have zero gain (-in dB) at DC. Such a high-pass filter with very low cutoff frequency can be used to block DC from a signal that is undesired in that signal (and pass nearly everything else). These are sometimes called DC blocking filters.

Experiment 2

Objective :

Study of Active High Pass filter and to Evaluate :

- **Low cutoff frequency of High Pass Filter**
- **Pass band gain of High Pass Filter**
- **Plot the frequency response of High Pass Filter**

Equipments Needed :

1. Oscilloscope

2. Digital Multimeter

Circuit Diagram :

Circuit used to study Active High pass filter shown in figure 12.

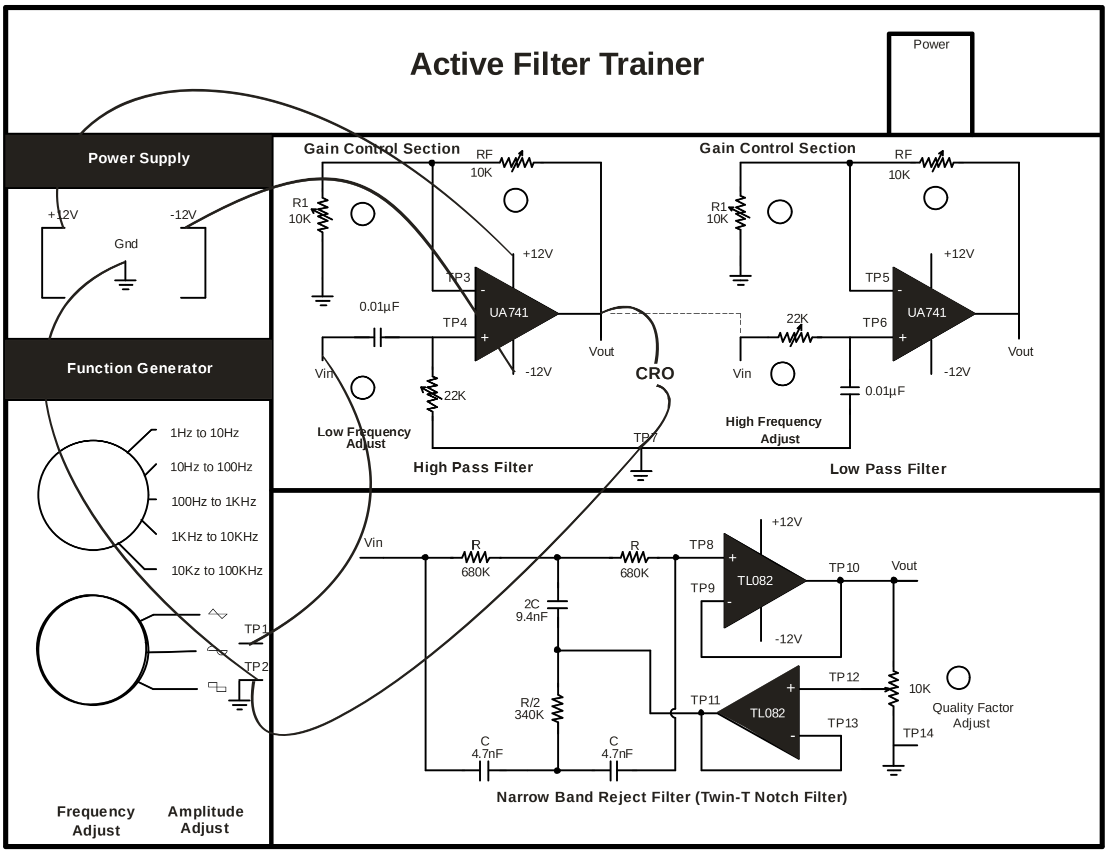


Figure 12

Procedure :

1. Initially rotate potentiometers R1 and Rf in fully clockwise direction in order to make R1 = Rf = 10K, so that according to the formula given below

$$Vo = (1+Rf/R1)Vin,$$

The gain of the output will be twice of the input.

2. The Low cutoff frequency is given by the formula:

$$f1 = \frac{1}{2\pi RC}$$

3. Connect an Ohmmeter between TP4 and TP7. Adjust resistance value at 15.9K by varying the potentiometer of 22K to set the Low cutoff frequency (f_l) at 1K.

4. Connect +12V and –12V DC power supplies at their indicated position from Power Supply section.

5. Connect all the ground test points with each other using patch chords.

6. Switch ‘On’ the Power Supply.

7. Set the output of function generator at 2 volt, 100 Hz using Oscilloscope with sinusoidal waveform.

8. Connect TP1 with Vin of High Pass Filter to give a sinusoidal signal of amplitude 2V (p-p) of frequency 100Hz.

9. Observe output on Oscilloscope.

10. Similarly you can give the triangular and square wave instead of sinusoidal wave, from the function generator section.

11. As we know according to the formula, Output gain is directly proportional to the Rf and inversely proportional to the R1. So, we can adjust the gain of the output by increasing the value of Rf as well as by decreasing the value of R1.

12. So, change the values of R1 and Rf and observe the change in output.

13. Increase the frequency of input signal step by step and observe the effect on output Vout on Oscilloscope.

14. Tabulate the values of Vout, gain, gain (db) at different values of input frequency shown in Observation Table.

Observation Table :

S. No.	Input frequency (Hz)	Vout	|Vout/Vin| = gain	Gain(db) = 20 Log |Vout/Vin|
1	100			
2	200			
3	500			
4	1K(f_L)			
5	5 K			
6	10 K			
7	15 K			
8	20 K			

15. Plot the frequency response of high pass filter using the data obtained at different input frequencies.

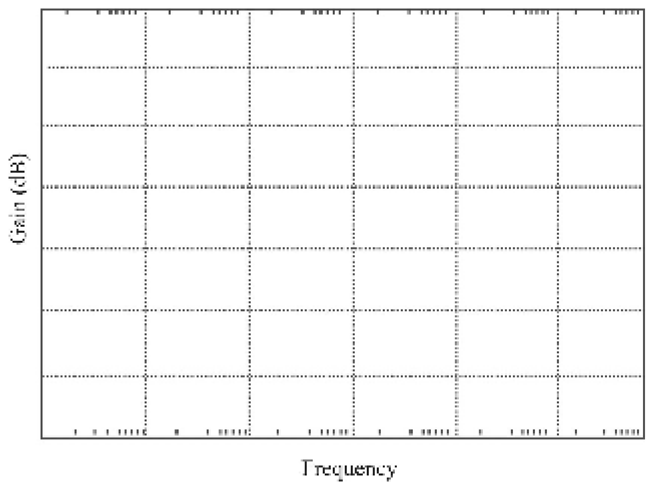


Figure 13

16. As we know, according to the formula

$$Vout = (1+Rf/R1)*Vin$$

where

Rf is directly proportional to the Vout

And R1 is inversely proportional to Vout

So by varying the pots Rf & R1 we can adjust the gain of the output

17. Perform the same procedure at different Cutoff frequencies as shown below:

Resistance (W)	**Capacitance (mF)**	**3 db frequency (Hz)**
800	0.01	20K
1.59 K	0.01	10K
15.9 K	0.01	1K

Theoretical Calculations:

Calculate all the following values

1. Pass band gain of Low Pass Filter $A_F = 1 + R_F / R_1$

=…………………………………………..

2. Pass band gain (db) = 20 log |Vout/Vin|

=…………………………………………..

3. Low cutoff frequency $f_L = 1/2\pi RC$

=…………………………………………..

4. Gain at Low cutoff frequency $f_L = 0.707 * A_F$

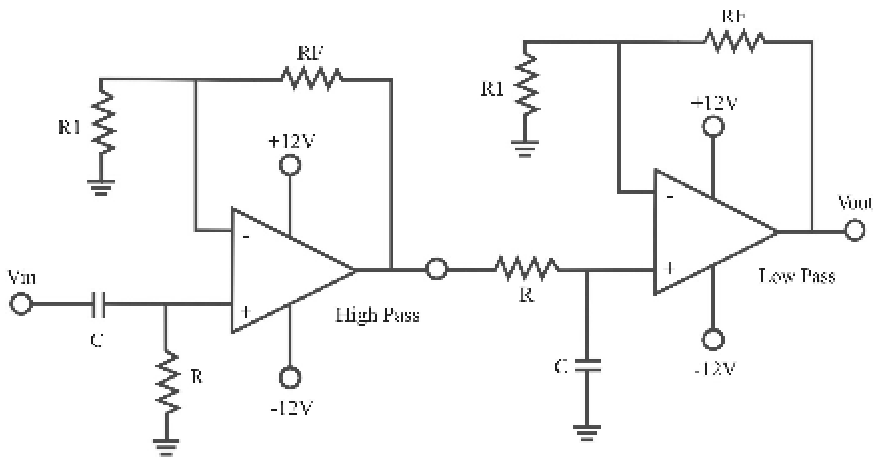
5. Gain (db) at Low cutoff frequency f_H = 20 log |Vout/Vin| where Vout = $(2)^{1/2}$ * Vin

Band Pass Filter :

A band-pass filter is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range. An example of an analogue electronic band-pass filter is an RLC circuit (a resistor–inductor–capacitor circuit). These filters can also be created by combining a low-pass filter with a high-pass filter.

An ideal filter would have a completely flat passband (e.g. with no gain/attenuation throughout) and would completely attenuate all frequencies outside the passband. Additionally, the transition out of the passband would be instantaneous in frequency. In practice, no bandpass filter is ideal. The filter does not attenuate all frequencies outside the desired frequency range completely; in particular, there is a region just outside the intended passband where frequencies are attenuated, but not rejected. This is known as the filter roll-off, and it is usually expressed in dB of attenuation per octave or decade of frequency. Generally, the design of a filter seeks to make the roll-off as narrow as possible, thus allowing the filter to perform as close as possible to its intended design. Often, this is achieved at the expense of pass-band or stop-band ripple.

The bandwidth of the filter is simply the difference between the upper and lower cutoff frequencies. The shape factor is the ratio of bandwidths measured using two different attenuation values to determine the cutoff frequency, e.g., a shape factor of 2:1 at 30/3 dB means the bandwidth measured between frequencies at 30 dB attenuation is twice that measured between frequencies at 3 dB attenuation.



Band Pass Filter

Figure 14

Frequency Response Curve of Band Pass Filter :

Ideal Frequency Response Curve :

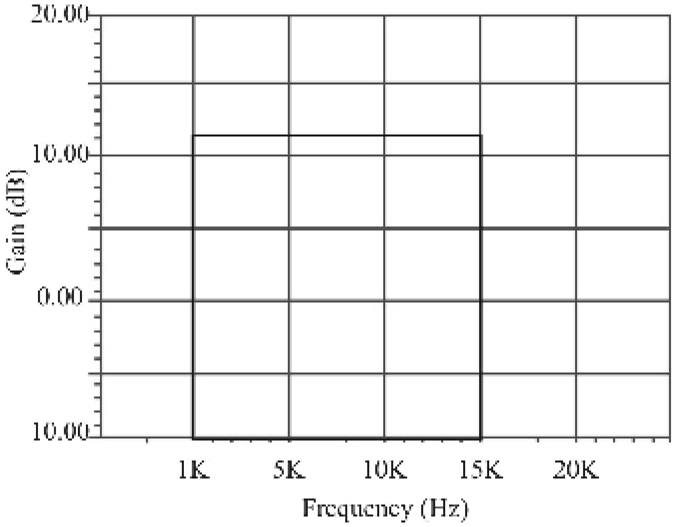


Figure 15

Practical Frequency Response Curve :

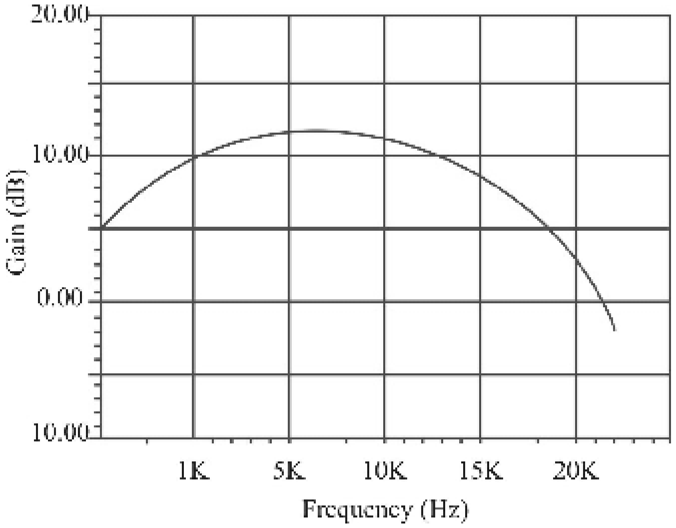


Figure 16

Experiment 3

Objective :

Study of Active Band Pass Filter and to Evaluate :

- **Low cutoff and High cut off frequency of Band Pass Filter**
- **Pass band gain of Band Pass Filter**
- **Plot the frequency response of Band Pass Filter**

Equipments Needed :

1. Oscilloscope

2. Digital Multimeter

Circuit Diagram :

Circuit used to study Active High Pass Filter shown in figure 17.

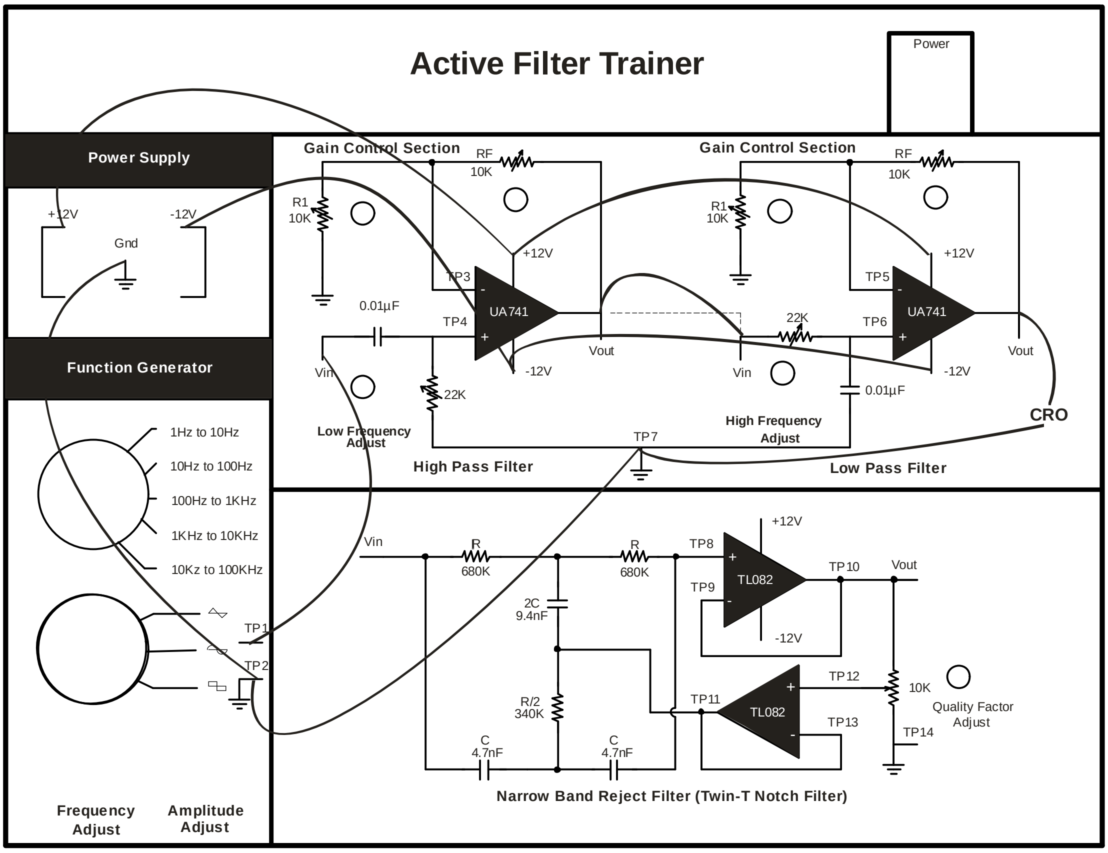


Figure 17

Procedure :

1. Connect the Output of High Pass Filter to the input of Low Pass Filter in order to make a Band Pass Filter.
2. Connect an Ohmmeter between TP4 and TP7 (Gnd) Adjust resistance value to 15.9K by varying the potentiometer 22K of High Pass Filter to set the Low cutoff frequency (f_L) at 1K.
3. Connect Ohmmeter between Vin of Low Pass Filter and TP6. Adjust resistance value to 1.59K by varying the potentiometer 22K of High Pass Filter to set the Low cutoff frequency (f_L) at 10K.
4. Connect +12V and −12V DC Power Supplies at their indicated position from Power Supply section.
5. Switch 'On' the Power Supply.
6. Set the output of function generator at 2 Volt, 100 Hz using Oscilloscope with sinusoidal waveform.
7. Connect TP1 with Vin of High Pass Filter to give a sinusoidal signal of amplitude 2 Vpp of frequency 100 Hz.
8. Observe output on Oscilloscope.
9. Similarly you can give the triangular and square wave instead of sinusoidal wave, from the function generator section.
10. Increase the frequency of input signal step by step and observe the effect on output Vout on Oscilloscope.
11. Tabulate the values of Vout, gain, gain (db) at different values of input frequency shown in Observation Table.

Observation Table :

S. No.	Input frequency (Hz)	Vout	|Vout/Vin| = gain	Gain(db) = 20 Log |Vout/Vin|
1	100			
2	200			
3	500			
4	1K(F_L)			
5	2 K			
6	5 K			
7	6 K			
8	9 K			
9	10 K(Fh)			
10	15 K			
11	20 K			
12	30 K			

12. Plot the frequency response of high pass filter using the data obtained at different input frequencies.

13. Similarly perform the experiment on the different values of Fh and Fl.

Narrow Band-Reject Filter :

The narrow band-reject, often called the notch-filter, is commonly used for the rejection of a single frequency, such as the 50 Hz power line frequency hum. The most commonly used notch filter is the twin-T network shown in figure 18. This is a passive filter composed of two T-shaped networks. One T-network is made up of two resistors and a capacitor, while the other uses two capacitors and a resistor.

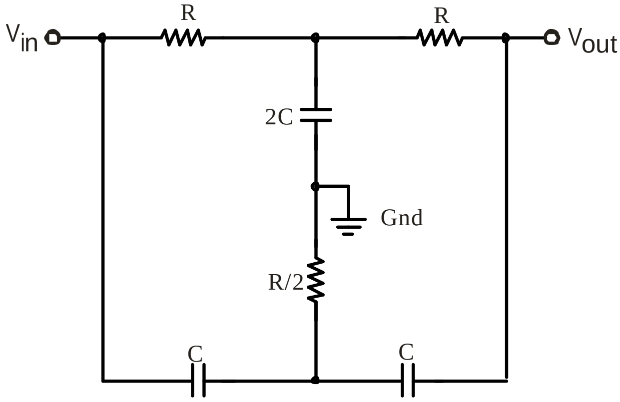


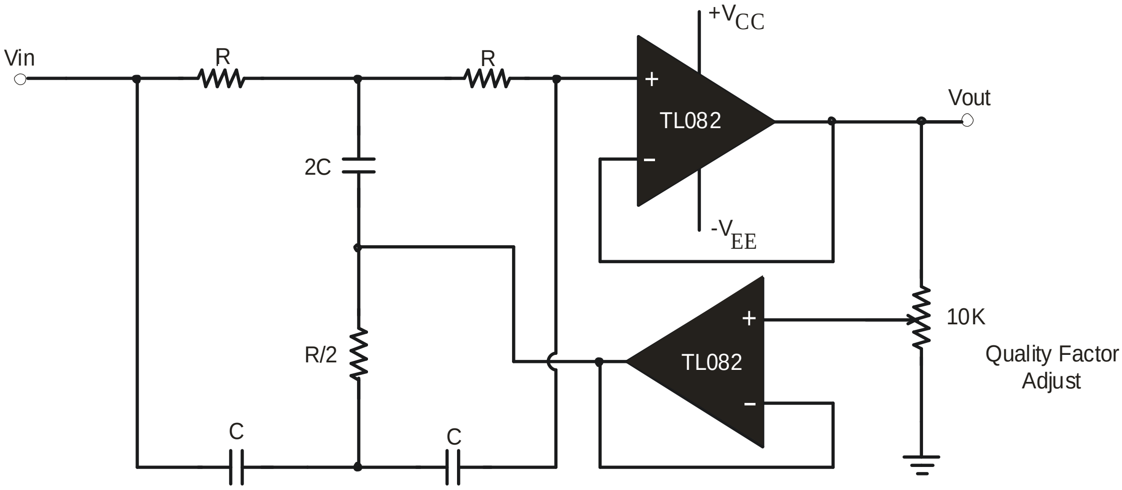
Figure 18

Passive Twin-T Notch Filter :

The notch-out frequency is the frequency at which maximum attenuation occurs, it is given by

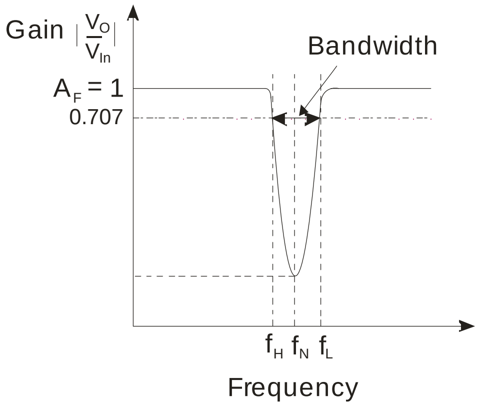
$$f_N = \frac{1}{2\pi RC} \qquad \text{………………..(8)}$$

Unfortunately, the passive twin-T network has a relatively low figure of merit Q. The Q of the network can be increased significantly if it is used with the voltage follower as shown in figure 19. The frequency response of the active notch filter of figure 19 is shown in figure 20. The most common use of notch filters is in communications and biomedical instruments for eliminating undesired frequencies. To design an active notch filter for a specific notch-out frequency f_N, choose the value of $C \leq 1\mu F$ and then calculate the required value of R from equation 8.



Active Twin-T Notch Filter

Figure 19



Frequency Response of Active T-Notch Filter

Figure 20

Experiment 4

Objective :

Study of Narrow Reject T-Notch Filter and to Evaluate :

- **Notch-out frequency of T-Notch Filter**
- **Plot the frequency response of T-Notch Filter**

Equipments Needed :

1. Oscilloscope
2. Digital Multimeter

Circuit Diagram :

Circuit used to study Narrow Reject T-Notch Filter shown in figure 21.

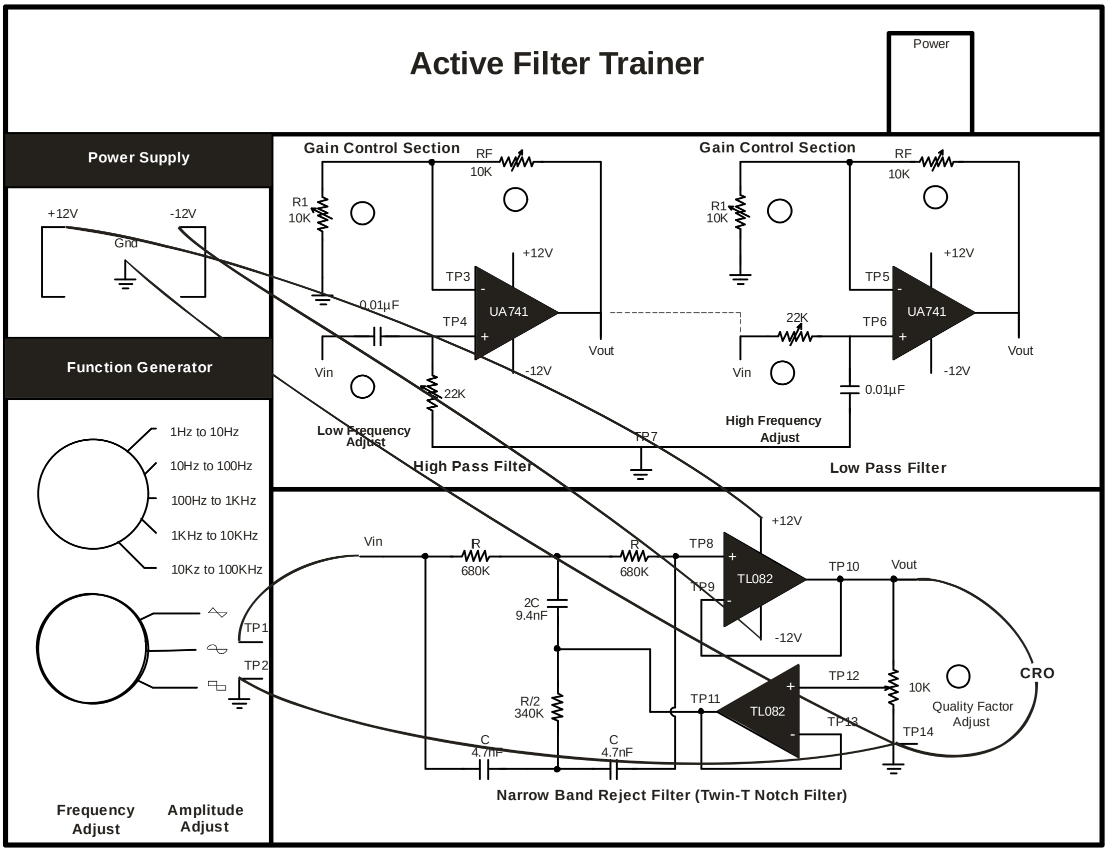


Figure 21

Procedure :

1. Initially rotate potentiometer of Quality Factor Adjust anticlockwise to set it at some lower value of resistance (around 680 Ω).

2. The notch frequency of the filter is given by

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

For this circuit R = 680K and C = 4.7nF, so f_N = 49.823 Hz ≅ 50 Hz.

3. Connect +12V, –12V and Gnd terminals from Power Supply Section at their indicated position of Twin-T Notch Filter.

4. Switch 'On' the Power Supply.

5. Set the output of Function Generator for sinusoidal signal at 20 Hz with a fixed amplitude (say 8 V) using Oscilloscope.

6. Connect TP1 with Vin of Twin-T Notch Filter to give a sinusoidal signal of fixed amplitude of frequency 20Hz to Twin-T Notch Filter.

7. Connect Gnd terminal of Function Generator to Gnd terminal of Twin-T Notch Filter i.e. connect TP2 to TP14.

8. Observe output (Vout) on Oscilloscope and note down the output voltage in the Observation Table.

9. Increase the frequency of input signal step by step and observe the effect on output (Vout) on Oscilloscope. Tabulate the corresponding value of Vout at different values of input frequency shown in Observation Table.

Observation Table :

S.No.	Input frequency (Hz)	Vout	|Vout / Vin| = A, Gain
1.	20		
2.	30		
3.	40		
4.	50(f_N)		
5.	60		
6.	70		
7.	80		
8.	90		
9.	100		
10.	110		
11.	120		

10. Plot the frequency response (i.e. graph between Gain and Input Frequency) of Twin-T Notch Filter using the data obtained at different input frequencies.

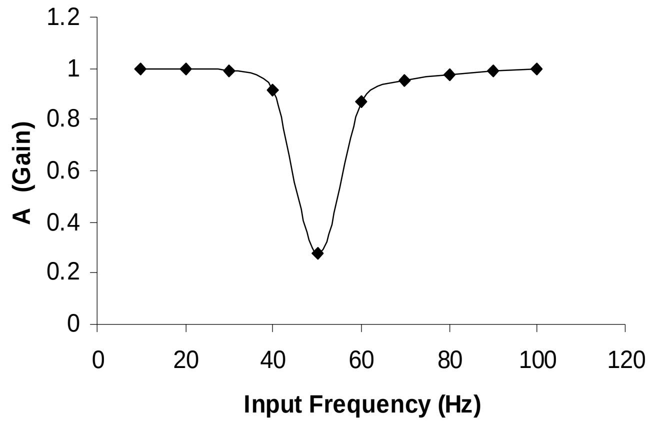


Figure 22

Frequency Response of Active T-Notch Filter

11. Now rotate the Quality Factor Adjust potentiometer clockwise to increase the value of resistance. As the resistance increases Quality Factor decreases. The higher the value of Q, the narrower the bandwidth and the lower the value of Q, the wider the bandwidth. So value of Q is an indication of the selectivity for a band filter.

12. Repeat steps 3 to 11.

13. Compare the frequency responses at different positions of Quality Factor Adjust potentiometer.

Warranty

1) We guarantee the product against all manufacturing defects for 24 months from the date of sale by us or through our dealers. Consumables like dry cell etc. are not covered under warranty.

2) The guarantee will become void, if

- **a)** The product is not operated as per the instruction given in the learning material.
- **b)** The agreed payment terms and other conditions of sale are not followed.
- **c)** The customer resells the instrument to another party.
- **d)** Any attempt is made to service and modify the instrument.

3) The non-working of the product is to be communicated to us immediately giving full details of the complaints and defects noticed specifically mentioning the type, serial number of the product and date of purchase etc.

4) The repair work will be carried out, provided the product is dispatched securely packed and insured. The transportation charges shall be borne by the customer.

List of Accessories

1. 2mm Patch Cord 8”……………………………………………………15 Nos.

2. Mains Cord………………………………………………………………1 No.

3. Learning Material CD………………………………………………….1 No