UNIT-II

APPLICATIONS OF OPERATIONAL AMPLIFIER

Inverting summer:

The configuration is shown in fig. 2. With three input voltages v_a, v_b & v_c. Depending upon the value of R_f and the input resistors R_a, R_b, R_c the circuit can be used as a summing amplifier, scaling amplifier, or averaging amplifier.

Again, for an ideal OPAMP, $v_1 = v_2$. The current drawn by OPAMP is zero. Thus, applying KCL at v_2 node

$$i_1 + i_2 + i_3 = i_f$$

$$\frac{V_a}{R_a} + \frac{V_b}{R_b} + \frac{V_c}{R_c} = -\frac{V_o}{R_f}$$

$$V_o = -\left(\frac{R_f}{R_a} V_a + \frac{R_f}{R_b} V_b + \frac{R_f}{R_c} v_c\right)$$

If in the circuit shown, $R_a = R_b = R_c = R$

$$V_o = -\frac{R_f}{R}(V_a + V_b + V_c)$$

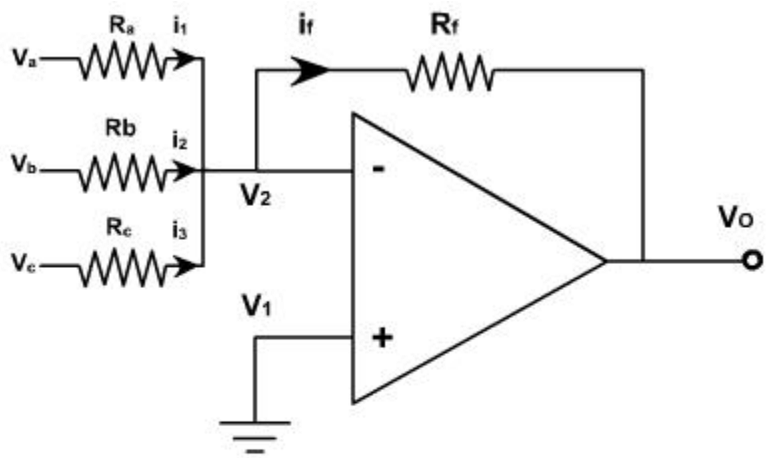


Fig. 2

This means that the output voltage is equal to the negative sum of all the inputs times the gain of the circuit R_f/ R; hence the circuit is called a summing amplifier. When R_f= R then the output voltage is equal to the negative sum of all inputs.

$v_o = -(v_a + v_b + v_c)$

If each input voltage is amplified by a different factor in other words weighted differently at the output, the circuit is called then scaling amplifier.

$$\frac{R_f}{R_a} \neq \frac{R_f}{R_b} \neq \frac{R_f}{R_c}$$

$$V_o = -\left(\frac{R_f}{R_a} V_a + \frac{R_f}{R_b} V_b + \frac{R_f}{R_c} v_c\right)$$

The circuit can be used as an averaging circuit, in which the output voltage is equal to the average of all the input voltages.

In this case, $R_a = R_b = R_c = R$ and $R_f / R = 1 / n$ where n is the number of inputs. Here $R_f / R = 1 / 3$.

$v_o = -(v_a + v_b + v_c) / 3$

In all these applications input could be either ac or dc.

Noninverting configuration:

If the input voltages are connected to noninverting input through resistors, then the circuit can be used as a summing or averaging amplifier through proper selection of R_1, R_2, R_3 and R_f. as shown in [fig. 3](#).

To find the output voltage expression, v_1 is required. Applying superposition theorem, the voltage v_1 at the noninverting terminal is given by

$$v_1 = \frac{R/2}{R+R/2} v_a + \frac{R/2}{R+R/2} v_b + \frac{R/2}{R+R/2} v_c$$

$$V_1 = \frac{V_a + V_b + V_c}{3}$$

Hence the output voltage is

$$V_o = \left(1+\frac{R_f}{R_1}\right) V_1 = \left(1+\frac{R_f}{R_1}\right)\left(\frac{V_a+V_b+V_c}{3}\right)$$

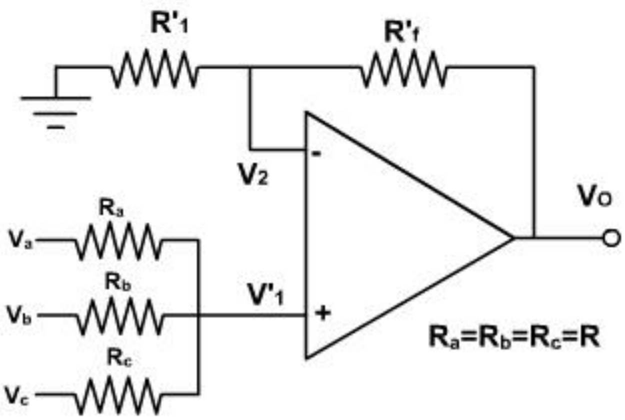


Fig. 3

This shows that the output is equal to the average of all input voltages times the gain of the circuit ($1+ R_f / R_1$), hence the name averaging amplifier.

If ($1+R_f/ R_1$) is made equal to 3 then the output voltage becomes sum of all three input voltages.

$v_o = v_a + v_b + v_c$

Hence, the circuit is called summing amplifier.

Subtractor:

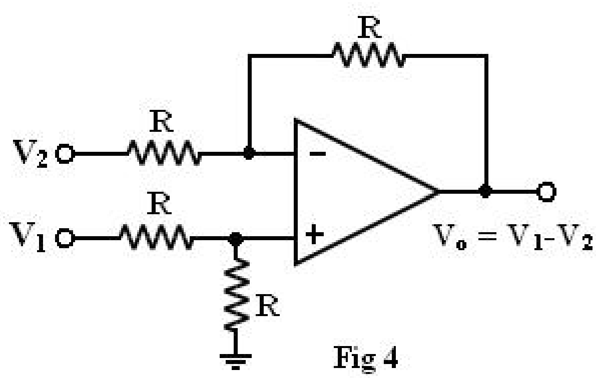


Fig 4

A basic differential amplifier can be used as a subtractor as shown in the above figure. If all resistors are equal in value, then the output voltage can be derived by using superposition principle.

To find the output V_{01} due to V_1 alone, make $V_2 = 0$.

$$V_{o1} = \frac{V_1}{2}\left(1 + \frac{R}{R}\right) = V_1$$

Similarly the output V_{o2} due to V_2 alone (with V_1 grounded) can be written simply for an inverting amplifier as

$$V_{o2} = -V_2$$

Thus the output voltage V_o due to both the inputs can be written as

$$V_o = V_{o1} + V_{o2} = V_1 - V_2$$

ADDER-SUBTRACTOR CIRCUIT:

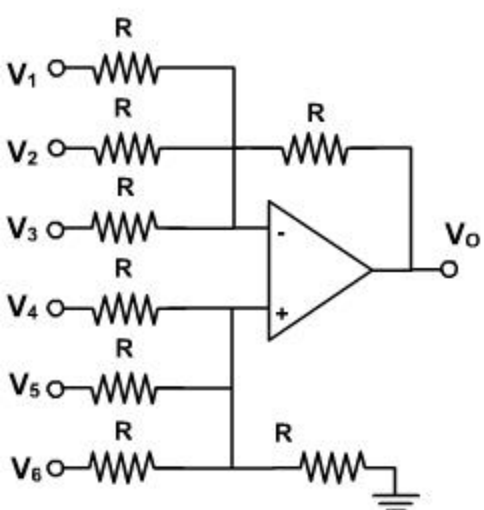


Fig. 5

Let's consider of V_1 (singly) by shorting the others i.e. the circuit then looks like as shown in **fig. 6**.

The current flowing through the resistor R into the i/e.

$$I = \frac{V_i - V_1}{R}$$

$$V_1 \approx 0.1 = \frac{V_1}{R}$$

The current when passes through R, output an operational value of

$$-\frac{R}{R} V_1 = -V \qquad (1)$$

$$\text{the net output } V' = -(V_1 + V_3 + V_5) \qquad (2)$$

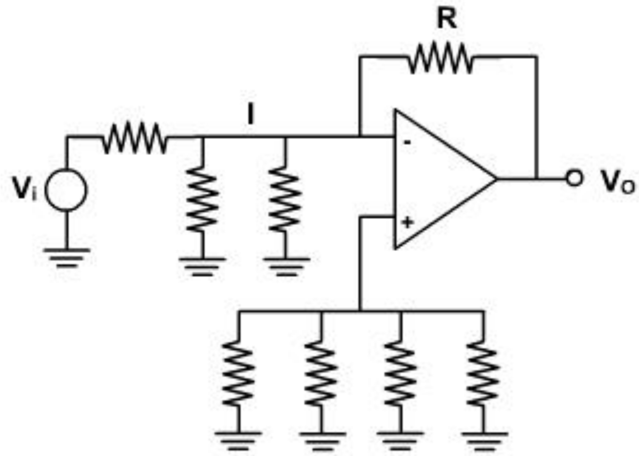


Fig. 6

Let as now consider the case of V_2 with other inputs shorted, circuit looks like as shown in **fig. 7**.

Now V_O is given by

$$V_N\left(1 + \frac{R}{R_3}\right) = V_0 - V_N \ (4)$$

$$V_N = \frac{V_2}{R + R/3} \times R/3 = \frac{V_2}{4}$$

$$V_0 = V_N(4) - \frac{V_2}{4} \times 4 = V_2$$

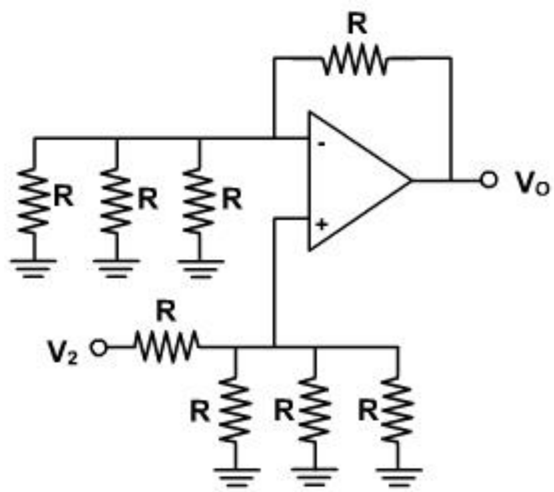


Fig. 7

same thing to V_4 and V_6

net output $V'' = V_2 + V_4 + V_6$ (3)
From (2) & (3)

$V' + V'' = (V_2 + V_4 + V_6) - (V_1 + V_3 + V_5)$
So $\mathbf{V_O = V_2 + V_4 + V_6 - V_1 - V_3 - V_5}$.

Integrator:

A circuit in which the output voltage waveform is the integral of the input voltage waveform is called integrator. **Fig. 4**, shows an integrator circuit using OPAMP.

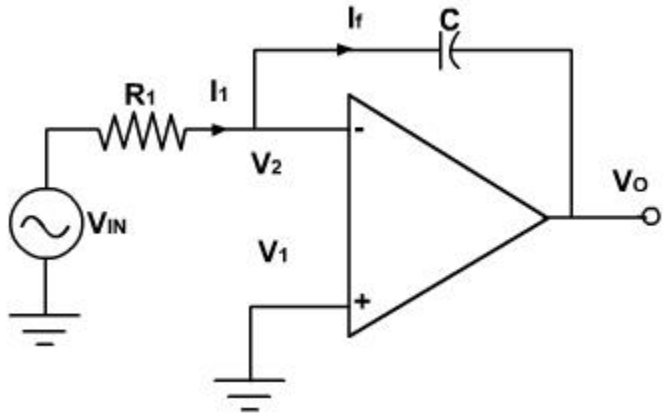


Fig. 4

Here, the feedback element is a capacitor. The current drawn by OPAMP is zero and also the V_2 is virtually grounded.

Therefore, $i_1 = i_f$ and $v_2 = v_1 = 0$

$$\frac{v_{in} - 0}{R} = C\,\frac{d(0 - v_o)}{dt}$$

Integrating both sides with respect to time from 0 to t, we get

$$\int_0^t \frac{v_{in}}{R}\,dt = \int_0^t C\,\frac{d(-v_o)}{dt}\,dt$$

$$= C(-v_o) + v_o\Big|_{t=0}$$

$$\text{if } V_O\Big|_{t=0} = 0\text{ V, then}$$

$$v_o = \frac{-1}{R}\int_0^t v_{in}\,dt$$

The output voltage is directly proportional to the negative integral of the input voltage and inversely proportional to the time constant RC.

If the input is a sine wave the output will be cosine wave. If the input is a square wave, the output will be a triangular wave. For accurate integration, the time period of the input signal T must be longer than or equal to RC.

Consider the output equation of an ideal integrator,

$$\therefore \quad V_o(t) = -\frac{1}{R_1 C_f}\int_0^t V_{in}\, dt$$

Assume the initial voltage $V_o(0)$ as zero.

Taking Laplace transform of this equation, we get,

$$V_o(s) = -\frac{1}{s R_1 C_f} V_{in}(t)$$

To get the frequency response replace s by $j\omega$.

$$\therefore \quad V_o(j\omega) = -\frac{1}{j\omega R_1 C_f} V_{in}(j\omega)$$

Hence the gain of the integrator is,

$$\frac{V_o(j\omega)}{V_{in}(j\omega)} = -\frac{1}{j\omega R_1 C_f}$$

To get the frequency response, obtain the magnitude of the gain, which is

$$A = \left|\frac{V_o(j\omega)}{V_{in}(j\omega)}\right| = \left|-\frac{1}{j\omega R_1 C_f}\right|$$

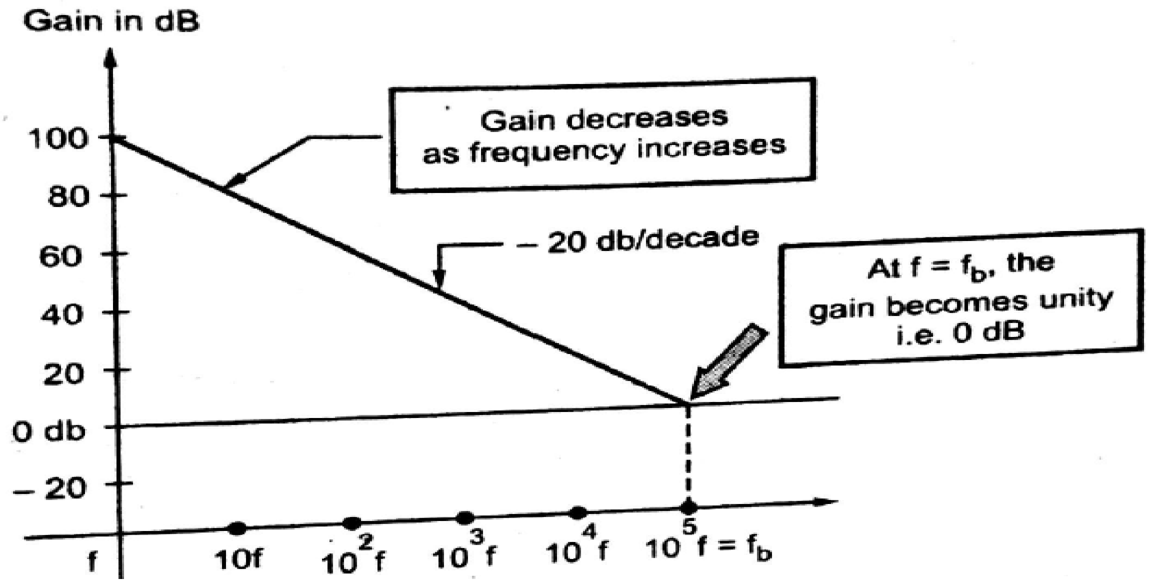
$$A = \frac{1}{\omega R_1 C_f} = \frac{1}{2\pi f R_1 C_f}$$

At very low frequencies such as d.c. ($f = 0$), the gain becomes infinite. This is because the capacitive reactance becomes very high for low frequencies. Hence it acts as an open circuit and thus op-amp works in open loop configuration. And the open loop gain of the op-amp is almost infinite. As the frequency increases, the gain drops. The response is similar to that of low pass filter.

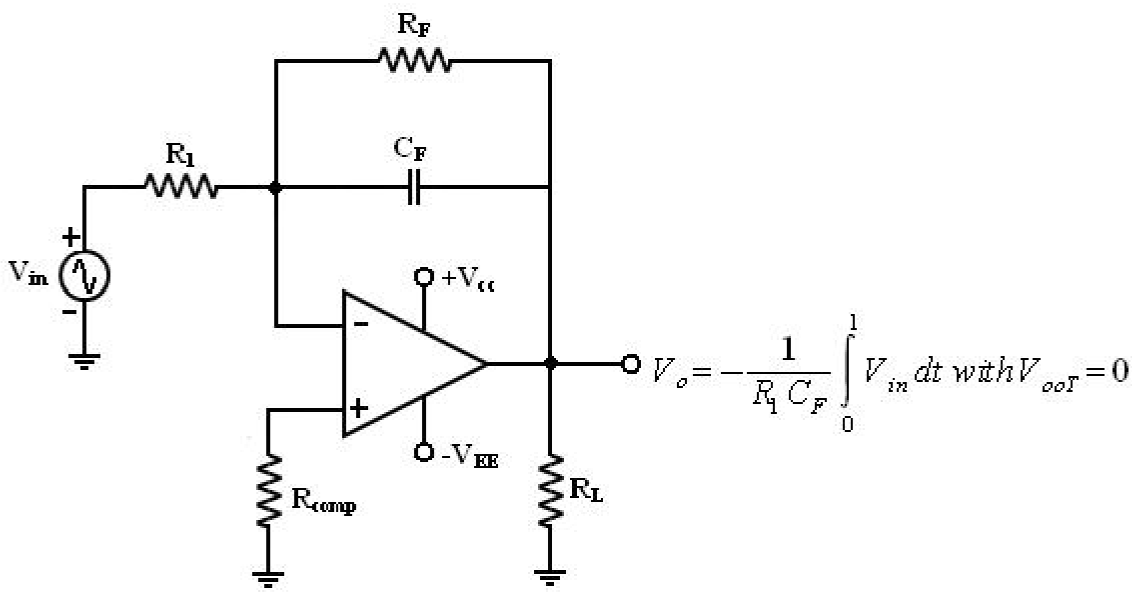
Let $f = f_b$ be the frequency at which gain of the op-amp becomes one i.e. 0 dB.

At $f = f_b$, $$20 \log A = 20 \log \frac{1}{2\pi f_b R_1 C_f} = 0 \text{ dB}$$

i.e. $$A = \frac{1}{2\pi f_b R_1 C_f} = 1 \qquad \text{... (21)}$$



Practical Integrator:



Practical Integrator to reduce the error voltage at the output, a resistor R_F is connected across the feedback capacitor C_F .

Thus R_F limits the low frequency gain and hence minimizes the variations in the output voltages. Both the stability and low frequency roll-off problems can be corrected by the addition of a resistor R_F in the practical integrator.

Stability -> refers to a constant gain as frequency of an input signal is varied over a certain range.

Low frequency -> refers to the rate of decrease in gain roll off at lower frequencies.

From the fig of practical Integrators,

$$I = \frac{V_{in} - V_A}{R_1} = \frac{V_{in}}{R_1}$$

Similarly $$I_1 = C_f \frac{d(V_A - V_o)}{dt} = -C_f \frac{dV_o}{dt}$$

And $$I_2 = \frac{V_A - V_o}{R_f} = \frac{-V_o}{R_f}$$

At node A, applying KCL

$$I = I_1 + I_2$$

$$\therefore \quad \frac{V_{in}}{R_1} = -C_f \frac{dV_o}{dt} - \frac{V_o}{R_f}$$

Taking Laplace of this equation,

$$\frac{V_{in}(s)}{R_1} = -s\,C_f\,V_o(s) - \frac{V_o(s)}{R_f}$$

$$\frac{V_{in}(s)}{R_1} = -V_o(s)\left[s\,C_f + \frac{1}{R_f}\right]$$

$$\frac{V_{in}(s)}{R_1} = \frac{-V_o(s)\,[1 + s\,C_f\,R_f]}{R_f}$$

$$V_o(s) = -\frac{R_f}{R_1\,(1 + s\,C_f\,R_f)}\,V_{in}(s)$$

$$V_o(s) = -\frac{1}{\left(s\,R_1\,C_f + \frac{R_1}{R_f}\right)}\,V_{in}(s)$$

$$V_o(s) = -\frac{1}{s R_1 C_f} V_{in}(s)$$

$$\boxed{V_o(t) = -\frac{1}{R_1 C_f}\int V_{in}(t)\,dt \qquad \text{as } \frac{1}{s} = \int dt}$$

$$\frac{V_o(s)}{V_{in}(s)} = -\frac{R_f/R_1}{1 + s C_f R_f}$$

Replacing s by jω in the steady state,

$$\frac{V_o(j\omega)}{V_{in}(j\omega)} = -\frac{R_f/R_1}{1 + j\omega C_f R_f}$$

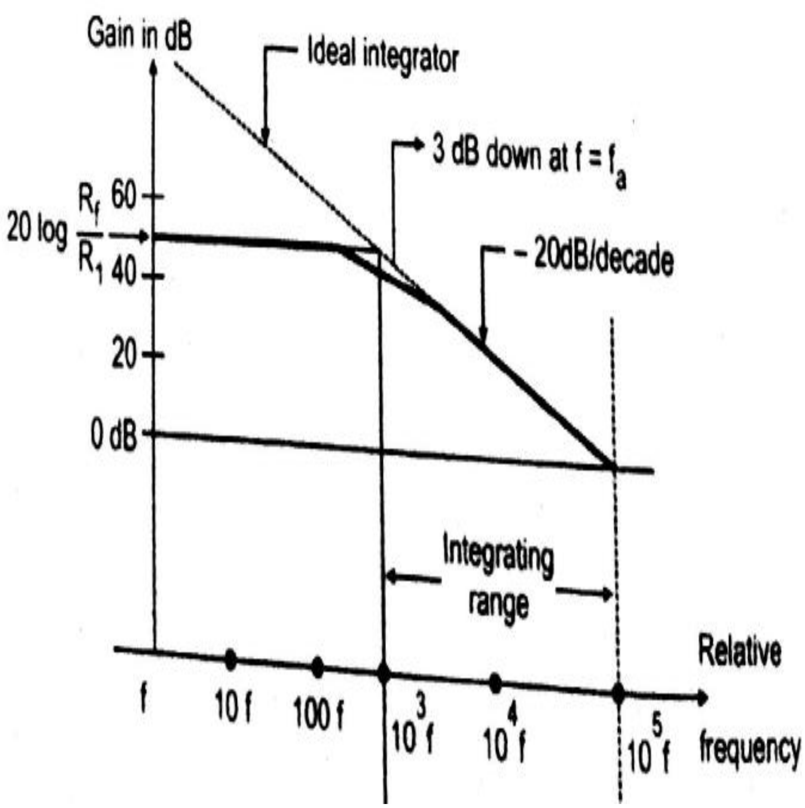
$$\therefore \quad A = -\frac{R_f/R_1}{1 + j\,2\pi f C_f R_f} =$$

$$\therefore \quad A = \frac{R_f/R_1}{1 + j\frac{f}{f_a}}$$

Where $$f_a = \frac{1}{2\pi C_f R_f}$$

The magnitude of the gain A is,

$$\boxed{|A| = \frac{R_f/R_1}{\sqrt{1 + \left(\frac{f}{f_a}\right)^2}}}$$



f is some relative operating frequency and for frequencies f to fa to gain R_F / R_1 is constant. After fa the gain decreases at a rate of 20dB/decade or between fa and fb the circuit act as an integrator. Generally the value of fa and in turn R_1 C_F and R_F C_F values should be selected such that fa<fb. In fact, the input signal will be integrated properly if the time period T of the signal is larger than or equal to R_F $C_{F,}$ (i.e)

Uses:

Most commonly used in analog computers.

ADC

Signal wave shaping circuits

Differentiator:

One of the simplest of the op-amp circuits that contains capacitor in the differentiating amplifier.

Differentiator:

As the name implies, the circuit performs the mathematical operation of differentiation (i.e) the output waveform is the derivative of the input waveform. The differentiator may be constructed from a basic inverting amplifier if an input resistor R_1 is replaced by a capacitor C_1 .

$$I_1 = C_1 \frac{d(V_{in} - V_A)}{dt} = C_1 \frac{dV_{in}}{dt}$$

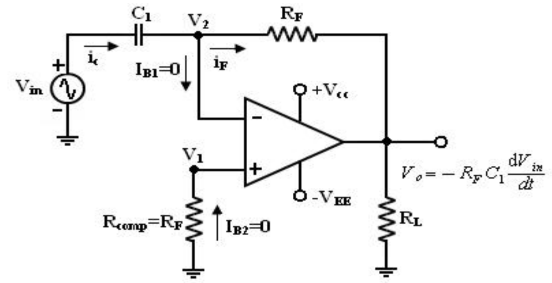
From the output side we can write,

$$I = \frac{(V_A - V_0)}{R_f} = -\frac{V_0}{R_f}$$

Equating the two equations,

$$C_1 \frac{dV_{in}}{dt} = -\frac{V_0}{R_f}$$

$$\boxed{V_0 = -C_1 R_f \frac{dV_{in}}{dt}}$$



The –sign => indicates a 180^0 phase shift of the output waveform V_0 with respect to the input signal.

Consider the output equation of an ideal differentiator which is,

$$V_o(t) = -R_f C_1 \frac{dV_{in}}{dt}$$

Taking Laplace transform of this equation, we get

$$V_o(s) = -s R_f C_1 V_{in}(s)$$

To get the frequency response, replace s by jω

$$V_o(j\omega) = -j\omega R_f C_1 V_{in}(j\omega)$$

∴ Hence the gain of the differentiator is,

$$\frac{V_o(j\omega)}{V_{in}(j\omega)} = -j\omega R_f C_1 \qquad \dots (12)$$

To get the frequency response obtain the magnitude of the gain which is,

$$A = \left|\frac{V_o(j\omega)}{V_{in}(j\omega)}\right| = \left|-j\omega R_f C_1\right|$$

$$A = \omega R_f C_1 = 2\pi f R_f C_1 \qquad \dots (13)$$

So at very low frequency such as d.c. (f = 0) the gain is zero. And as frequency increases, gain also increases.

The expression of the gain can be written as

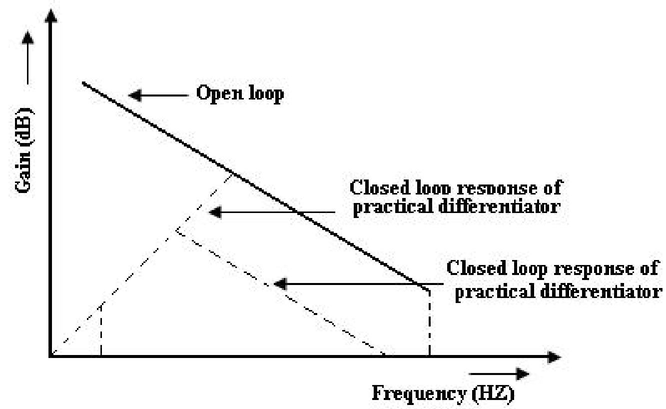
$$A = \frac{f}{f_a} \qquad \dots (14)$$

Where $$f_a = \frac{1}{2\pi R_f C_1} \qquad \dots (15)$$

The circuit will not do this because it has some practical problems.

The gain of the circuit (R_F /XC_1) R with R in frequency at a rate of 20dB/decade. This makes the circuit unstable.

Also input impedance XC_1 S with R in frequency which makes the circuit very susceptible to high frequency noise.



Practical differentiator:

From the above fig, f_a = frequency at which the gain is 0dB and is given by,

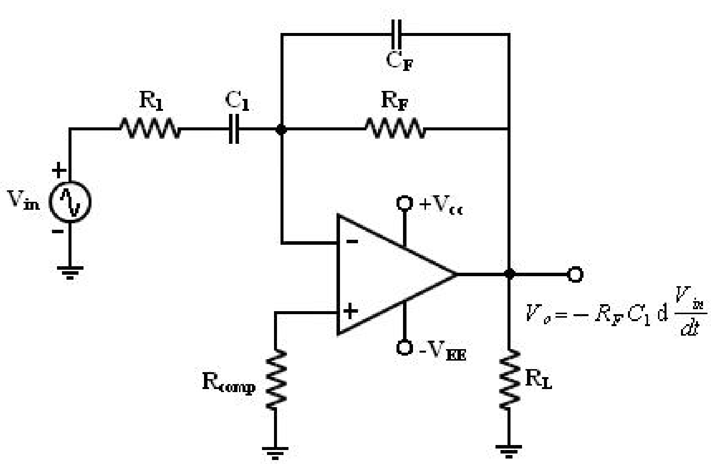
Both stability and high frequency noise problems can be corrected by the addition of 2 components. R_1 and C_F . This circuit is a practical differentiator.

From Frequency f to feedback the gain Rs at 20dB/decade after feedback the gain S at 20dB/decade. This 40dB/ decade change in gain is caused by the R_1 C_1 and R_F C_F combinations. The gain limiting frequency f_b is given by,

Where $R_1 C_1 = R_F C_F$

R_1 C_1 and R_F C_F => helps to reduce the effect of high frequency input, amplifier noise and offsets. All R_1 C_1 and R_F C_F make the circuit more stable by preventing the R in gain with frequency. Generally, the value of Feedback and in turn R_1 C_1 and R_F C_F values should be selected such that

The input signal will be differentiated properly, if the time period T of the input signal is larger than or equal to R_F C_1 (i.e) $T > R_F C_1$



For the current I, we can write

$$I = \frac{V_{in} - V_A}{Z_1} = \frac{V_{in}}{Z_1}$$

where $Z_1 = R_1$ in series with C_1

So in Laplace domain we can write,

$$Z_1 = R_1 + \frac{1}{s C_1} = \frac{1 + s R_1 C_1}{s C_1}$$

$$\therefore \quad I = \frac{s C_1 V_{in}(s)}{(1 + s R_1 C_1)}$$

Now the current I_1 is,

$$I_1 = \frac{V_A - V_o}{R_f} = -\frac{V_o}{R_f}$$

$$I_2 = C_f \frac{d(V_A - V_o)}{dt} = -C_f \frac{d V_0}{dt}$$

$$I_2 = -s C_f V_o(s)$$

Applying KCL at node A,

$$I = I_1 + I_2$$

$$\frac{s C_1 V_{in}(s)}{(1 + s R_1 C_1)} = -\frac{V_o(s)}{R_f} - s C_f V_o(s)$$

$$\frac{s C_1 V_{in}(s)}{(1 + s R_1 C_1)} = -V_o(s)\left[\frac{1 + s R_f C_f}{R_f}\right]$$

$$V_o(s) = \frac{-s R_f C_1 V_{in}(s)}{(1 + s R_1 C_1)(1 + s R_1 C_1)}$$

If $R_f C_f = R_1 C_1$ then

$$V_o(s) = \frac{-s R_f C_1 V_{in}(s)}{(1 + s R_1 C_1)^2}$$

The time constant $R_f C_1$ is much greater than $R_1 C_1$ or $R_f C_f$ and hence the equation (11) reduces to,

$$V_o(s) = -s R_f C_1 V_{in}(s)$$

i.e.

$$\boxed{V_o(t) = -R_f C_1 \frac{d V_{in}(t)}{dt} \qquad \text{as } s = \frac{d}{dt}} \qquad \ldots (12)$$

From the equation (11) we can write,

$$\frac{V_o(s)}{V_{in}(s)} = \frac{-s R_f C_1}{(1 + s R_1 C_1)^2} \qquad \text{where } R_f C_f = R_1 C_1$$

Replacing s by jω in the steady state,

$$\frac{V_o(j\omega)}{V_{in}(j\omega)} = \frac{-j\omega R_f C_1}{(1 + j\omega R_1 C_1)^2}$$

$$\therefore \quad \frac{V_o(j\omega)}{V_{in}(j\omega)} = \frac{-j 2\pi f R_f C_1}{(1 + j 2\pi f R_1 C_1)^2}$$

Now let, $$f_b = \frac{1}{2\pi R_1 C_1}$$

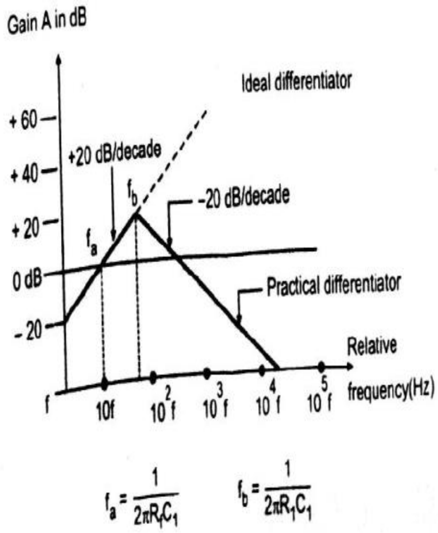
And $$f_a = \frac{1}{2\pi R_f C_1}$$

$$\therefore \quad \frac{V_o(j\omega)}{V_{in}(j\omega)} = \frac{-j\left(\frac{f}{f_a}\right)}{\left(1 + j\frac{f}{f_b}\right)^2}$$

The frequencies f_a and f_b are two break frequencies.

$$\therefore \quad A = \left|\frac{V_o(j\omega)}{V_{in}(j\omega)}\right| = \frac{(f/f_a)}{\left[\sqrt{1+\left(\frac{f}{f_b}\right)^2}\right]^2}$$

$$\therefore \quad \boxed{A = \frac{f/f_a}{1+\left(\frac{f}{f_b}\right)^2}}$$



Practical Differentiator design steps:

A workable differentiator can be designed by implementing the following steps.

1. Select fa equal to the highest frequency of the input signal to be differentiated then assuming a value of $C_1 < 1\mu f$. Calculate the value of R_F .

2. Choose fb = 20fa and calculate the values of R_1 and C_F so that $R_1 C_1 = R_F C_F$.

Uses:

Its used in waveshaping circuits to detect high frequency components in an input signal and also as a rate of change and detector in FM modulators.

Instrumentation Amplifier:

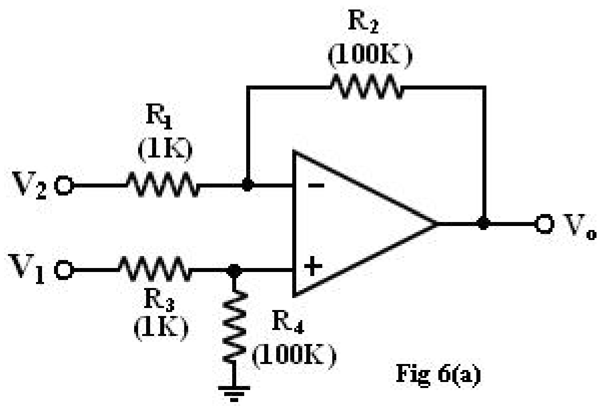


Fig 6(a)

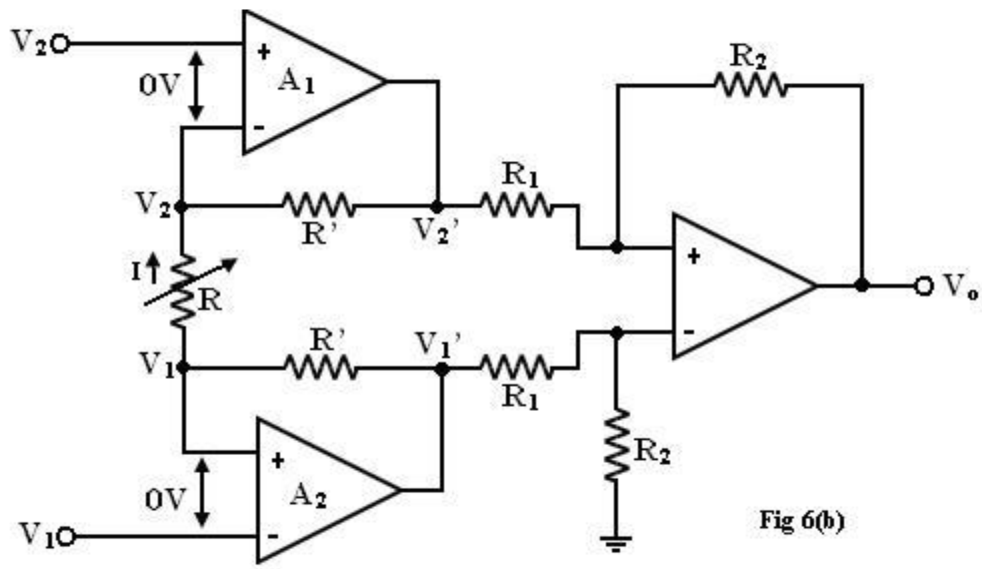


Fig 6(b)

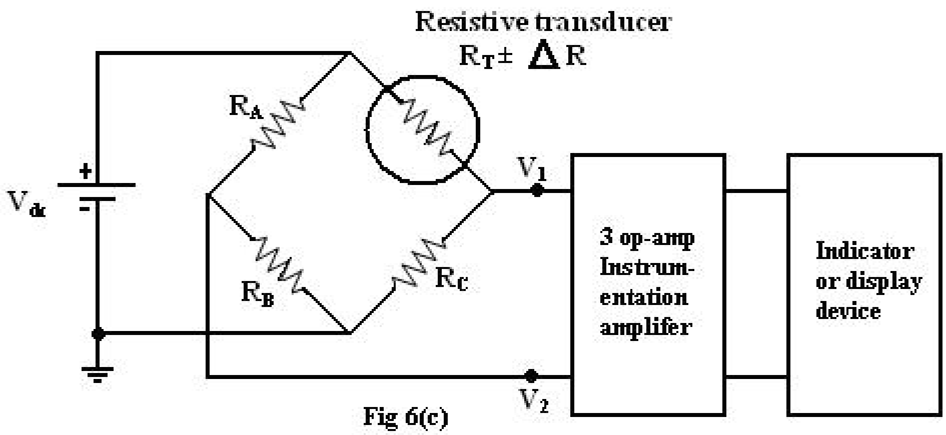


Fig 6(c)

In a number of industrial and consumer applications, one is required to measure and control physical quantities.

Some typical examples are measurement and control of temperature, humidity, light intensity, water flow etc. these physical quantities are usually measured with help of transducers.

The output of transducer has to be amplified so that it can drive the indicator or display system. This function is performed by an instrumentation amplifier. The important features of an instrumentation amplifier are

1. high gain accuracy
2. high CMRR
3. high gain stability with low temperature coefficient
4. low output impedance

Consider the basic differential amplifier as shown in figure 6(a).

In the circuit of figure 6(a), source V_1 sees an input impedance = R_3+R_4 (=101K) and the impedance seen by source V_2 is only R_1 (1K). This low impedance may load the signal source heavily.

Therefore, high resistance buffer is used preceding each input to avoid this loading effect as shown in figure 6(b).

The op-amp A_1 and A_2 have differential input voltage as zero. For $V_1=V_2$, that is, under common mode condition, the voltage across R will be zero. As no current flows through R and R‘ the non-inverting amplifier.

A_1 acts as voltage follower, so its output $V_2‘=V_2$. Similarly op-amp A_2 acts as voltage follower having output $V_1‘=V_1$. However, if $V_1 \neq V_2$, current flows in R and R‘, and $(V_2‘-V_1‘)>(V_2-V_1)$. Therefore, this circuit has differential gain and CMRR more compared to the single op-amp circuit of figure 6(a).

The output voltage V_o can be calculated as follows

we have,

$$V_o = -\frac{R_2}{R_1} V_2' + \left(1 + \frac{R_2}{R_1}\right)\left(\frac{R_2 V_1'}{R_1 + R_2}\right)$$

$$= \frac{R_2}{R_1}\left(V_1' - V_2'\right) \tag{4.18}$$

Since, no current flows into op-amp, the current I flowing (upwards) in R is $I = (V_1 - V_2)/R$ and passes through the resistor R'.

$$V_1' = R'I + V_1 = \frac{R'}{R}(V_1 - V_2) + V_1 \tag{4.19}$$

and

$$V_2' = -R'I + V_2 = -\frac{R'}{R}(V_1 - V_2) + V_2 \tag{4.20}$$

Putting the values of V_1' and V_2' in Eq. (4.18), we obtain,

$$V_o = \frac{R_2}{R_1}\left[\frac{2R'}{R}(V_1 - V_2) + (V_1 - V_2)\right]$$

$$V_o = \frac{R_2}{R_1}\left(1 + \frac{2R'}{R}\right)(V_1 - V_2)$$

The difference gain of this instrumentation amplifier R, however should never be made zero, as this will make the gain infinity. To avoid such a situation, in a practical circuit, a fixed resistance in series with a potentiometer is used in place of R.

Figure 6(c) shows a differential instrumentation amplifier using Transducer Bridge. The circuit uses a resistive transducer whose resistance changes as a function of the physical quantity to be measured.

The bridge is initially balanced by a dc supply voltage V_{dc} so that $V_1=V_2$. As the physical quantity changes, the resistance R_T of the transducer also changes, causing an unbalance in the bridge ($V_1 \neq V_2$). This differential voltage now gets amplified by the three op-amp differential instrumentation amplifier.

There are number differential applications of instrumentation amplifier with the transducer bridge, such as temperature indicator, temperature controller, and light intensity meter to name a few.

Current to voltage converter:

The circuit shown in fig. 1, is a current to voltage converter.

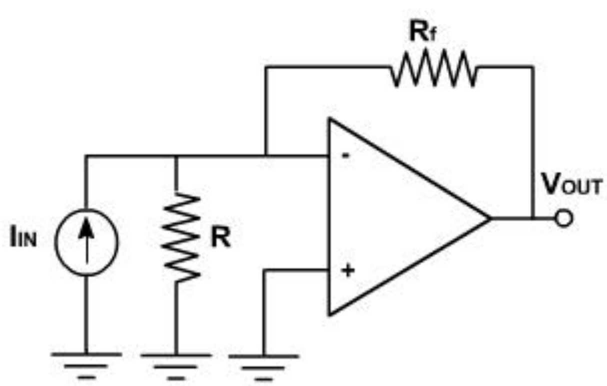


Fig. 1

Due to virtual ground the current through R is zero and the input current flows through R_f. Therefore,

$v_{out} = -R_f * i_{in}$

The lower limit on current measure with this circuit is set by the bias current of the inverting input

Voltage to current converter:

Fig. 3, shows a voltage to current converter in which load resistor R_L is floating (not connected to ground).

The input voltage is applied to the non-inverting input terminal and the feedback voltage across R drives the inverting input terminal. This circuit is also called a current series negative feedback, amplifier because the feedback voltage across R depends on the output current i_L and is in series with the input difference voltage v_d.

Writing the voltage equation for the input loop.

$v_{in} = v_d + v_f$

But v_d » since A is very large,therefore,

$v_{in} = v_f$
$v_{in} = R\, i_{in}$
$i_{in} = v_{in} / R.$

and since input current is zero.

$i_L = i_{in} = v_{in} ./ R$

The value of load resistance does not appear in this equation. Therefore, the output current is independent of the value of load resistance. Thus the input voltage is converted into current, the source must be capable of supplying this load current.

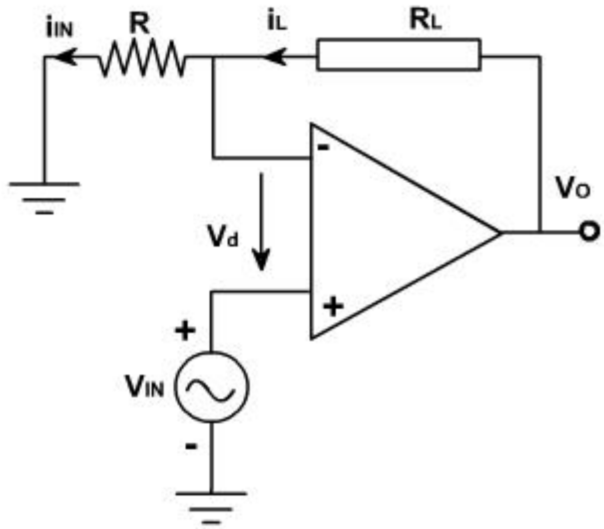
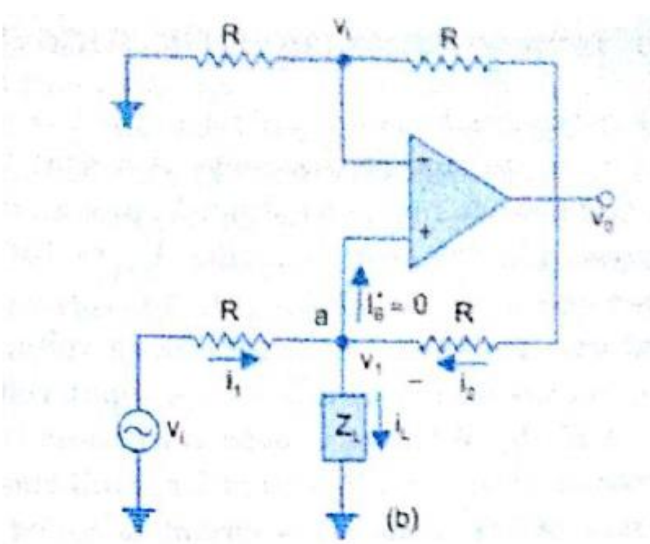


Fig. 3

Grounded Load:

If the load has to be grounded, then the above circuit cannot be used. The modified circuit is shown in fig



$$i_1 + i_2 = i_L$$

$$\frac{v_i - v_1}{R} + \frac{v_o - v_1}{R} = i_L$$

$$v_i + v_o - 2\,v_1 = i_L R$$

$$v_1 = \frac{v_i + v_o - i_L R}{2}$$

Since the op-amp is used in non-inverting mode, the gain of the circuit is $1 + R/R = 2$. The output voltage is,

$$v_o = 2\,v_1 = v_i + v_o - i_L R$$

that is,

$$v_i = i_L R$$

or,

$$i_L = \frac{v_i}{R}$$

Filters:

A filter is a frequency selective circuit that, passes a specified band of frequencies and blocks or attenuates signals of frequencies out side this band. Filter may be classified on a number of ways.

1. Analog or digital
2. Passive or active
3. Audio or radio frequency

Analog filters are designed to process only signals while digital filters process analog signals using digital technique. Depending on the type of elements used in their consideration, filters may be classified as passive or active.

Elements used in passive filters are resistors, capacitors and inductors. Active filters, on the other hand, employ transistors or OPAMPs, in addition to the resistor and capacitors. Depending upon the elements the frequency range is decided.

RC filters are used for audio or low frequency operation. LC filters are employed at RF or high frequencies. The most commonly used filters are these:

1. Low pass filters
2. High pass filter
3. Band pass filter
4. Band reject filter.
5. All pass filter

Fig. 1, shows the frequency response characteristics of the five types of filter. The ideal response is shown by dashed line. While the solid lines indicates the practical filter response.

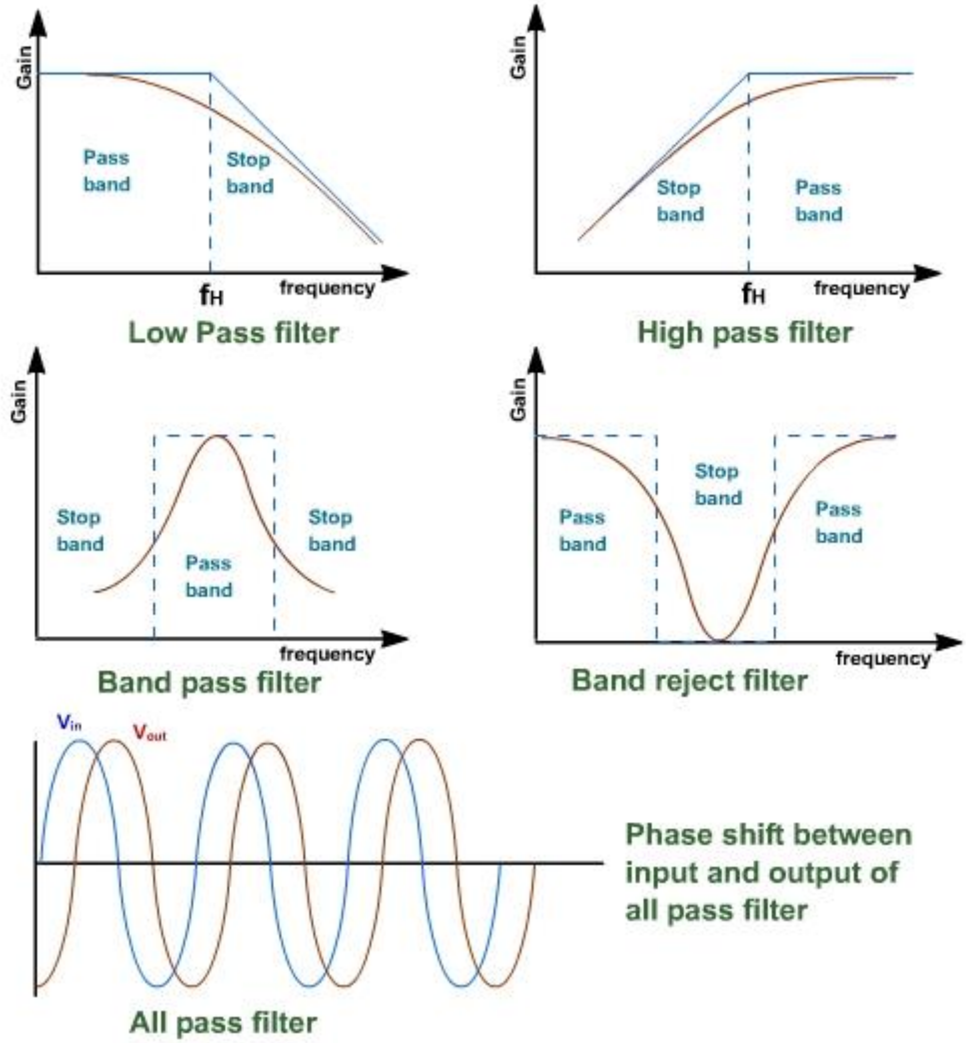


Fig. 1

A low pass filter has a constant gain from 0 Hz to a high cutoff frequency f_H. Therefore, the bandwidth is f_H. At f_H the gain is down by 3db. After that the gain decreases as frequency increases. The frequency range 0 to f_H Hz is called pass band and beyond f_H is called stop band.

Similarly, a high pass filter has a constant gain from very high frequency to a low cutoff frequency f_L. below f_L the gain decreases as frequency decreases. At f_L the gain is down by 3db. The frequency range f_L Hz to ∞ is called pass band and bleow f_L is called stop band.

Active filters offers the following advantages over a passive filters:

1. Gain and Frequency adjustment flexibility:

Since the op-amp is capable of providing a gain, the i/p signal is not attenuated as it is in a passive filter. [Active filter is easier to tune or adjust].

2. No loading problem:

Because of the high input resistance and low o/p resistance of the op-amp, the active filter does not cause loading of the source or load.

3. Cost:

Active filters are more economical than passive filter. This is because of the variety of cheaper op-amps and the absence of inductors.

First Order Low Pass Filter:

Fig. 2, shows a first order low pass Butter-worth filter that uses an RC network for filtering, opamp is used in non-inverting configuration, R_1 and R_f decides the gain of the filter.

According to voltage divider rule, the voltage at the non-inverting terminal is:

$$V_1 = \frac{-jX_C}{R - jX_C} V_{in} \quad \text{where} \quad X_C = \frac{1}{2\pi f C}$$

$$V_1 = \frac{-j\frac{1}{2\pi f C}}{\frac{2\pi f CR - j}{2\pi f C}} V_{in}$$

$$= \frac{-j}{2\pi f CR - j} V_{in}$$

$$= \frac{1}{1 + j2\pi f CR} V_{in}$$

$$V_1 = \frac{V_{in}}{1 + j2\pi f CR} = V_2$$

$$V_0 = \left(1 + \frac{R_f}{R_1}\right) V_1$$

$$= \left(1 + \frac{R_f}{R_1}\right) \frac{V_{in}}{1 + j2\pi f CR}$$

$$\frac{V_0}{V_{in}} = \frac{A_f}{1 + j\left(\frac{f}{f_H}\right)}$$

where V_0/V_{in} is the gain of the filter as a function of frequency.

$$A_f = \left(1 + \frac{R_f}{R_1}\right) = \text{Pass band filter gain}$$

f = frequency of input signal

$$f_H = \text{Cutoff frequency of the filter} = \frac{1}{2\pi RC}$$

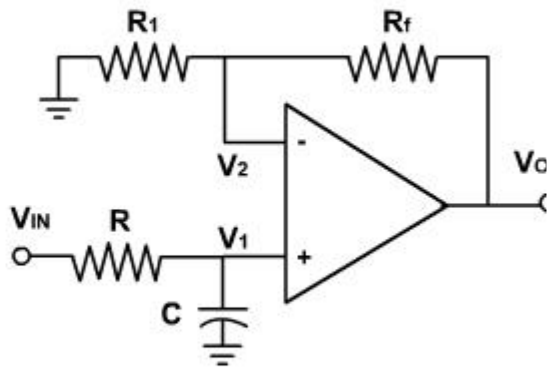


Fig. 2

$$\text{Magnitude of the gain of low pass filter} = \left|\frac{V_0}{V_{in}}\right| = \frac{A_f}{\sqrt{1 + \left(\frac{f}{f_H}\right)^2}}$$

$$\text{and phase angle } \varphi = -\tan^{-1}\left(\frac{f}{f_H}\right)$$

$$\text{At low frequencies} \quad \left|\frac{V_0}{V_{in}}\right| = A_f$$

$$\text{At } f = f_H, \quad \left|\frac{V_0}{V_{in}}\right| = \frac{A_f}{\sqrt{2}} = 0.707 A_f$$

$$\text{and at } f > f_H \quad \left|\frac{V_0}{V_{in}}\right| < A_f$$

Thus the low pass filter has a nearly constant gain A_f from 0 Hz to high cut off frequency f_H. At f_H the gain is 0.707 A_f and after f_H it decreases at a constant rate with an increases in frequency. f_H is called cutoff frequency because the gain of filter at this frequency is reduced by 3dB from 0Hz.

Filter Design:

A low pass filter can be designed using the following steps:

1. Choose a value of high cutoff frequency f_H.
2. Select a value of C less than or equal to 1 µF.

3. Calculate the value of R using $R = \frac{1}{2\pi f_H C}$.

4. Finally, select values of R1 and RF to set the desired gain using $A_F = 1 + \frac{R_F}{R_1}$.

Example - 1

Design a low pass filter at a cutoff frequency of 1 kH z with a pass band gain of 2.

Solution:

Given f_H = 1 kHz. Let C = 0.01 μF.

Therefore, R can be obtained as

$$R = \frac{1}{2\pi \times 10^3 \times 0.01 \times 10^{-6}}$$
$$= 15.9\,k\Omega$$

A 20 kΩ potentiometer can be used to set the resistance R.

Since the pass band gain is 2, R_1 and R_F must be equal. Let $R_1 = R_2$ = 10 kΩ.

Second Order Low-Pass Butterworth filter:

A stop-band response having a 40-dB/decade at the cut-off frequency is obtained with the second-order low-pass filter. A first order low-pass filter can be converted into a second-order low-pass filter by using an additional RC network as shown in fig. 1.

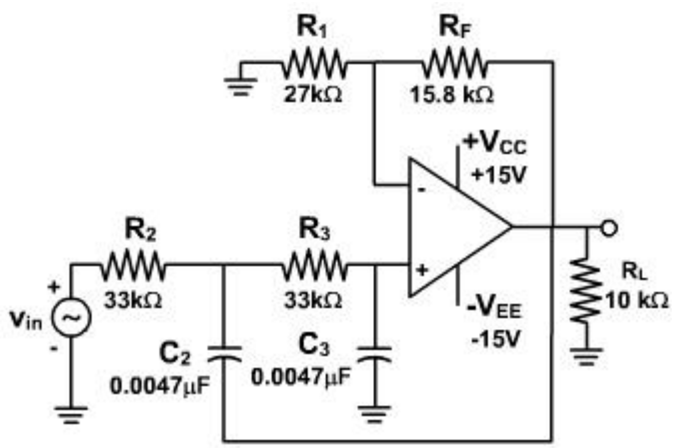


Fig. 1

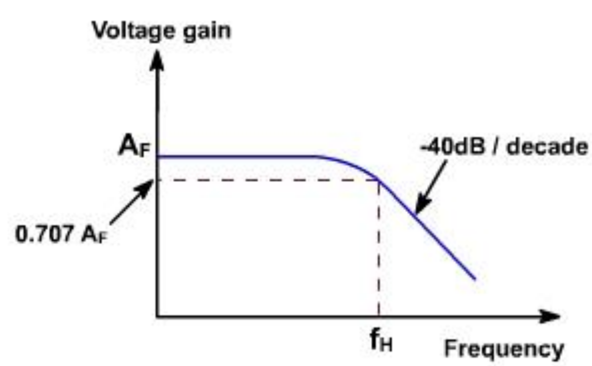


Fig. 2

The gain of the second order filter is set by R_1 and R_F, while the high cut-ff frequency f_H is determined by R_2, C_2, R_3 and C_3 as follows:

$$f_H = \frac{1}{2\pi\sqrt{R_2 R_3 C_2 C_3}}$$

Furthermore, for a second-order low pass Butterworth response, the voltage gain magnitude is given by

$$\left|\frac{V_o}{V_{in}}\right| = \frac{A_F}{\sqrt{1+(f/f_H)^4}}$$

where,

$$A_F = 1 + \frac{R_F}{R_1} = \text{passband gain of the filter}$$

$$f = \text{frequency of the input signal}$$

Except for having the different cut off frequency, the frequency response of the second order low pass filter is identical to that of the first order type as shown in fig. 2.

Filter Design:

The design steps of the second order filter are identical to those of the first order filter as given bellow:

1. Choose a value of high cutoff frequency f_H.
2. To simplify the design calculations, set $R_2 = R_3 = R$ and $C_2 = C_3 = C$. Then choose a value of C less than 1 μF.
3. Calculate the value of R using $R = \frac{1}{2\pi f_H C}$.
4. Finally, because of the equal resistor ($R_2 = R_3$) and capacitor ($C_2 = C_3$) values, the pass band voltage gain A_F has to be equal to 1.586. This gain is necessary to guarantee Butterworth response. Therefore, $R_F = 0.586\ R_1$. Hence choose a value of R_1= 100 kΩ and calculate the value of R_F.

First Order High Pass Butterworth filter:

Fig. 3, shows the circuit of first order high pass filter.This is formed by interchanging R and C in low pass filter.

The lower cut off frequency is f_L. This is the frequency at which the magnitude of the gain is 0.707 times its pass band value. All frequencies higher than f_L are pass band frequencies with the highest frequency determined by the closed loop bandwidth of the OPAMP.

$$V_0 = \left(1 + \frac{R_f}{R_1}\right) * V_1$$

$$= \left(1 + \frac{R_f}{R_1}\right) * \frac{j2\pi f RC}{1+j2\pi fRC} * V_{in}$$

$$\frac{V_0}{V_{in}} = A_f\left[\frac{j\left(\frac{f}{f_L}\right)}{1+j\left(\frac{f}{f_L}\right)}\right]$$

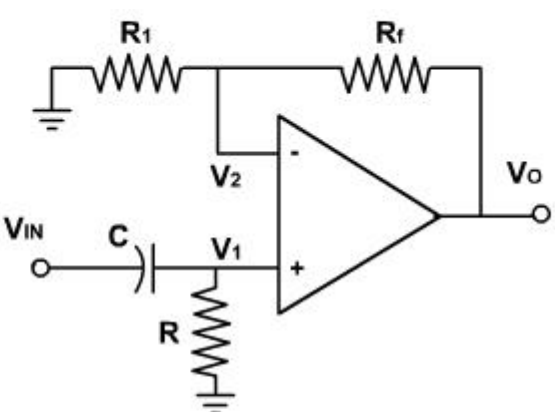


Fig. 3

The magnitude of the gain of the filter is

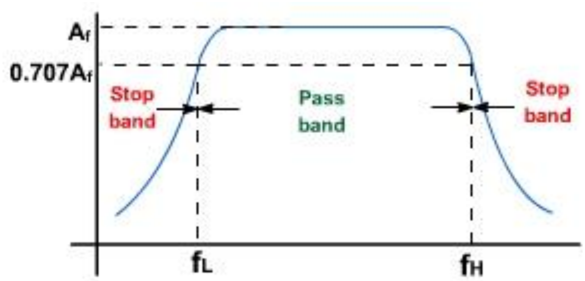
$$\left|\frac{V_o}{V_{in}}\right| = A_f \frac{\left(f/f_L \right)}{\sqrt{1 + \left(f/f_L \right)}}$$

$$\text{at } f < f_L \quad \left|\frac{V_o}{V_{in}}\right| < A_f$$

$$\text{at } f = f_L \quad \left|\frac{V_o}{V_{in}}\right| < A_f/\sqrt{2}$$

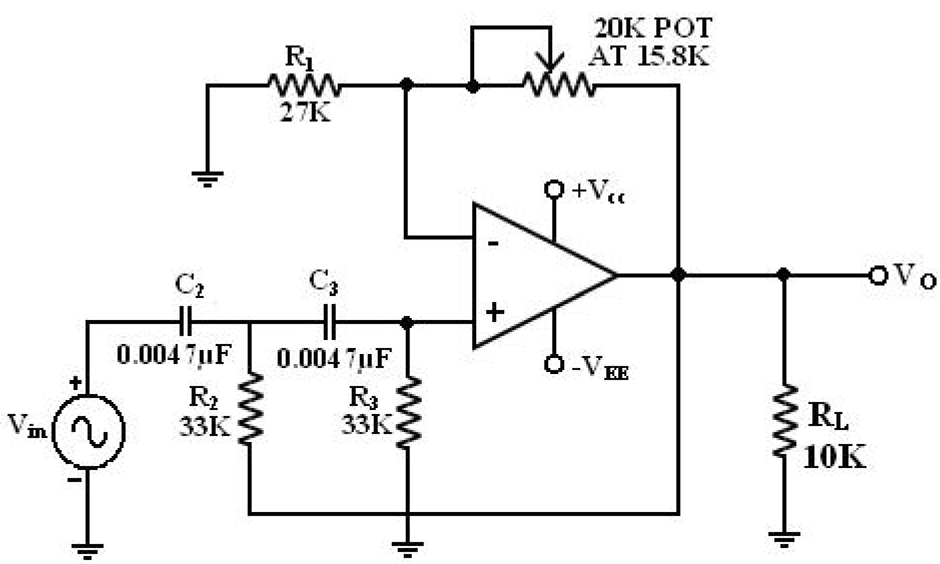
$$\text{at } f > f_L \quad \left|\frac{V_o}{V_{in}}\right| = A_f$$

If the two filters (high and low) band pass are connected in series it becomes wide band filter whose gain frequency response is shown in fig. 4.

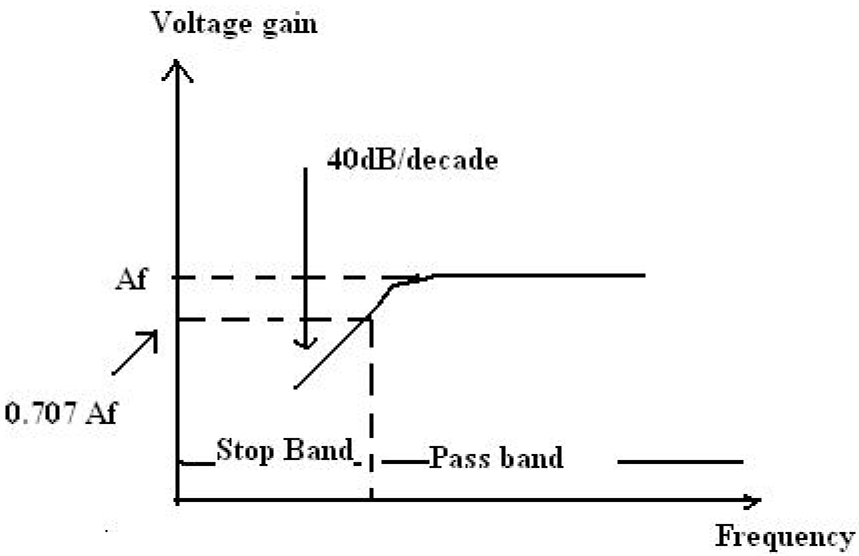


Second – order High Pass Butterworth Filter:

I order Filter, II order HPF can be formed from a II order LPF by interchanging the frequency – determine resistors and capacitors.



II order HPF



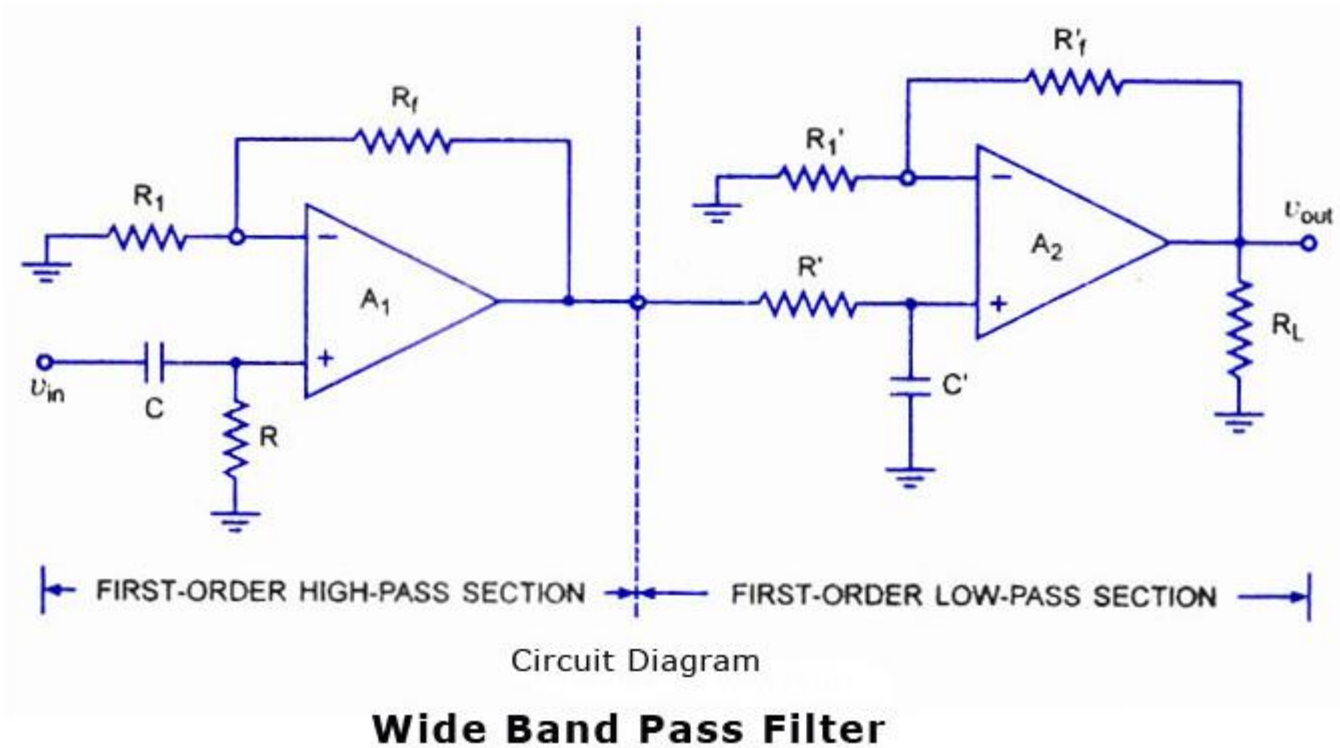
Band Pass Filter

A band-pass filter is a circuit which is designed to pass signals only in a certain band of frequencies while attenuating all signals outside this band. The parameters of importance in a bandpass filter are the high and low cut-off frequencies (f_H and f_l), the bandwidth (BW), the centre frequency f_c, centre-frequency gain, and the selectivity or Q.

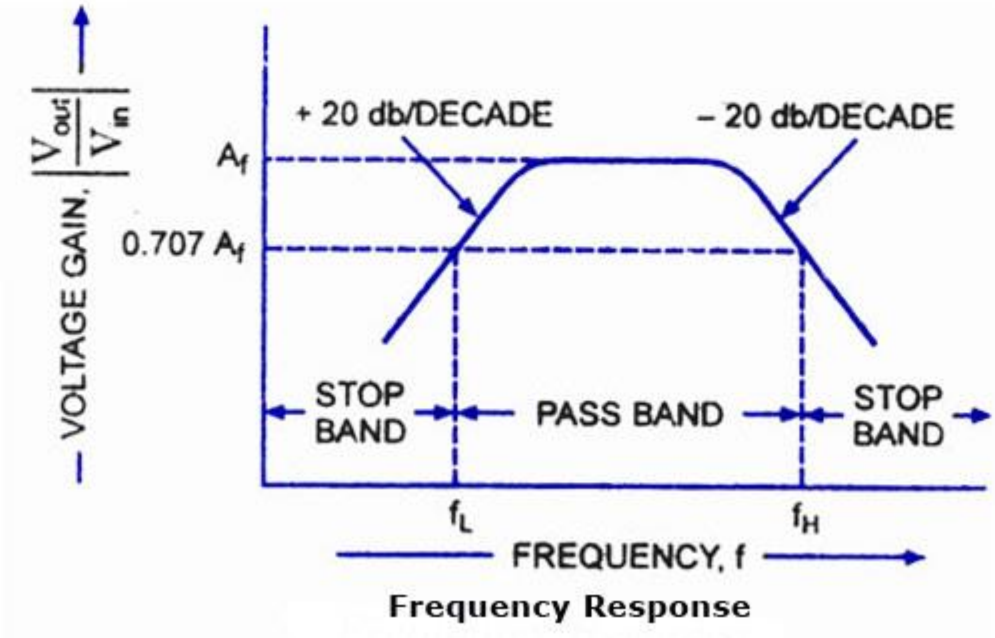
There are basically two types of bandpass filters viz wide bandpass and narrow bandpass filters. Unfortunately, there is no set dividing line between the two. However, a bandpass filter is defined as a wide bandpass if its figure of merit or quality factor Q is less than 10 while the bandpass **filters with Q > 10 are called the narrow bandpass filters.** Thus Q is a measure of selectivity, meaning the higher the value of Q the more selective is the filter, or the narrower is the bandwidth (BW). The relationship between Q, 3-db bandwidth, and the centre frequency f_c is given by an equation

For a wide bandpass filter the centre frequency can be defined as where f_H and f_L are respectively the high and low cut-off frequencies in Hz.In a narrow bandpass filter, the output voltage peaks at the centre frequency f_c.

Wide Bandpass Filter



A wide bandpass filter can be formed by simply cascading high-pass and low-pass sections and is generally the choice for simplicity of design and performance though such a circuit can be realized by a number of possible circuits. To form a ± 20 db/ decade bandpass filter, a first-order high-pass and a first-order low-pass sections are cascaded; for a ± 40 db/decade bandpass filter, second-order high- pass filter and a second-order low-pass filter are connected in series, and so on. It means that, the order of the bandpass filter is governed by the order of the high-pass and low-pass filters it consists of.

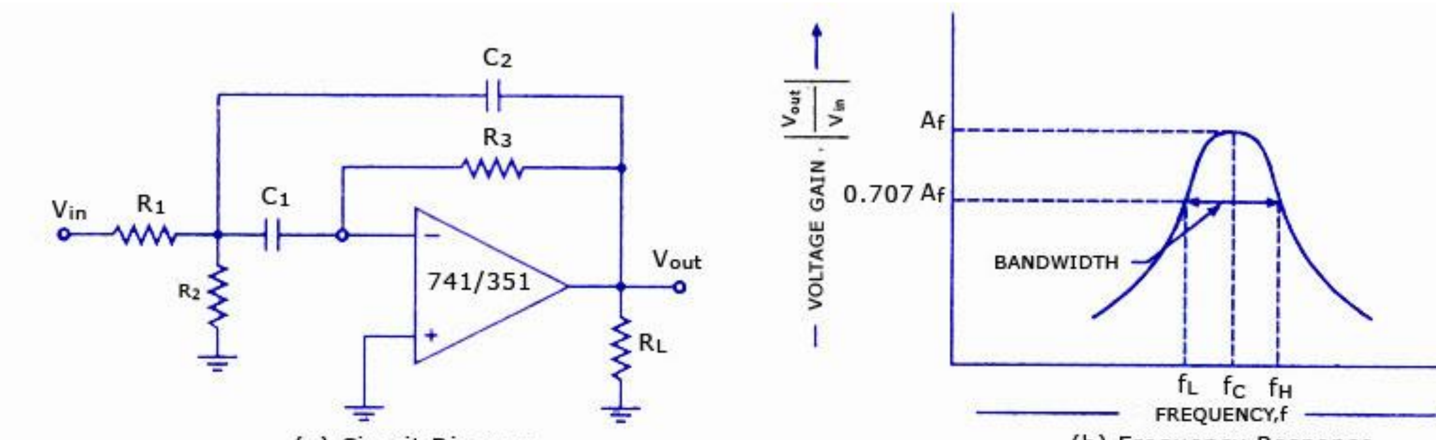


Frequency Response wide bandpass filter

www.CircuitsToday.com

A ± 20 db/decade wide bandpass filter composed of a first-order high-pass filter and a first-order low-pass filter, is illustrated in fig. (a). Its frequency response is illustrated in fig. *(b).*

Narrow Bandpass Filter.



(a) Circuit Diagram

(b) Frequency Response

Multiple Feedback Narrow Band-Pass Filter

Narrow Bandpass Filter

www.CircuitsToday.co

A narrow bandpass filter employing multiple feedback is depicted in figure. This filter employs only one op-amp, as shown in the figure. In comparison to all the filters discussed so far, this filter has some unique features that are given below.

1. It has two feedback paths, and this is the reason that it is called a multiple-feedback filter.

2. The op-amp is used in the inverting mode.

The frequency response of a narrow bandpass filter is shown in fig(b).

Generally, the narrow bandpass filter is designed for specific values of centre frequency f_c and Q or f_c and BW. The circuit components are determined from the following relationships. For simplification of design calculations each of **C_1 and C_2 may be taken equal to C.**

$R_1 = Q/2\prod f_c CA_f$

$R_2 = Q/2\prod f_c C(2Q^2-A_f)$

and $R_3 = Q / \prod f_c C$

where A_f, is the gain at centre frequency and is given as

$A_f = R_3 / 2R_1$

The gain A_f however must satisfy the condition $A_f < 2 Q^2$.

The centre frequency f_c of the multiple feedback filter can be changed to a new frequency f_c‘ without changing, the gain or bandwidth. This is achieved simply by changing R_2 to R'_2 so that

$R'_2 = R_2 [f_c/f'_c]^2$

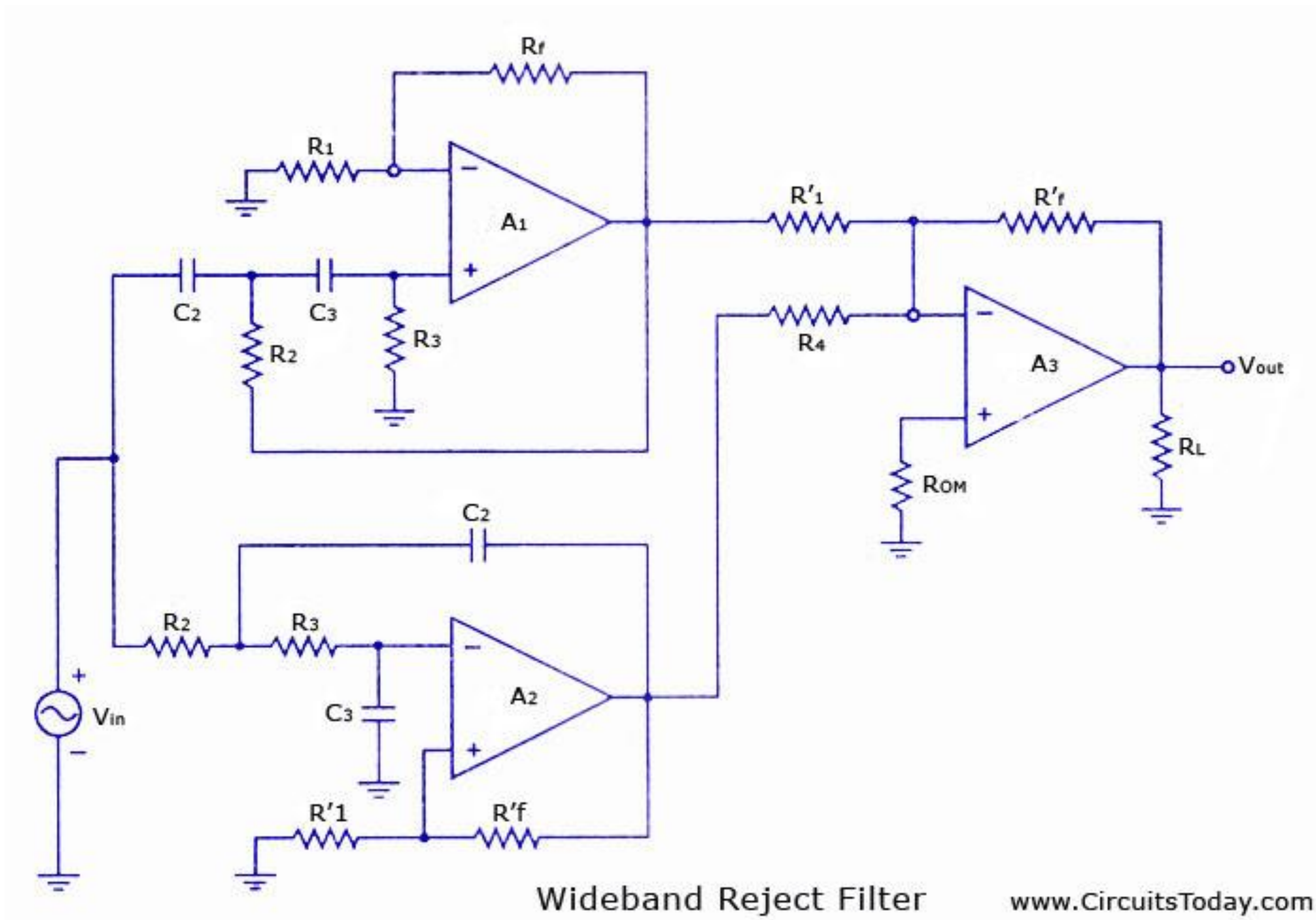
Band-Stop (or Reject) Filter

The bandpass filter passes one set of frequencies while rejecting all others. The band-stop filter does just the opposite. It rejects a band of frequencies, while passing all others. This is also called a band-reject or band-elimination filter. Like bandpass filters, band-stop filters

may also be classified as (i) wide-band and (ii) narrow band reject filters.

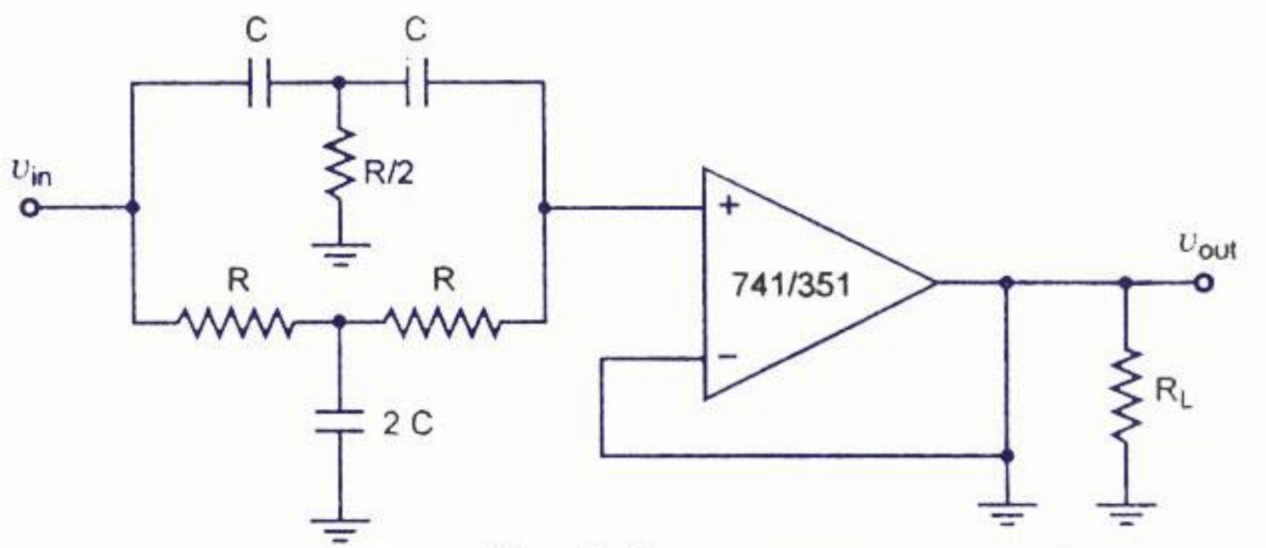
The narrow band reject filter is also called a notch filter. Because of its higher Q, which exceeds 10, the bandwidth of the narrow band reject filter is much smaller than that of a wide band reject filter.

Wide Band-Stop (or Reject) Filter.

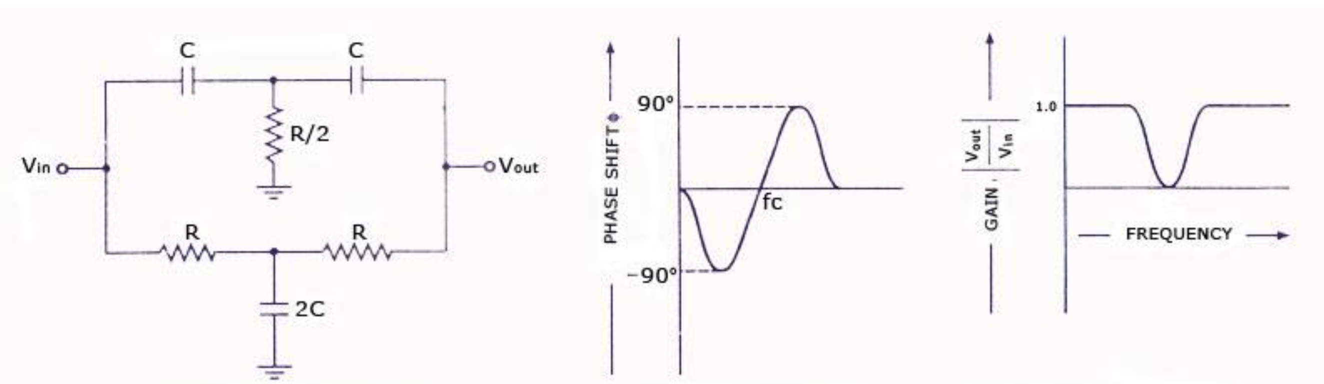


A wide band-stop filter using a low-pass filter, a high-pass filter and a summing amplifier is shown in figure. For a proper band reject response, the low cut-off frequency f_L of high-pass filter must be larger than the high cut-off frequency f_H of the low-pass filter. In addition, the passband gain of both the high-pass and low-pass sections must be equal.

Narrow Band-Stop Filter.



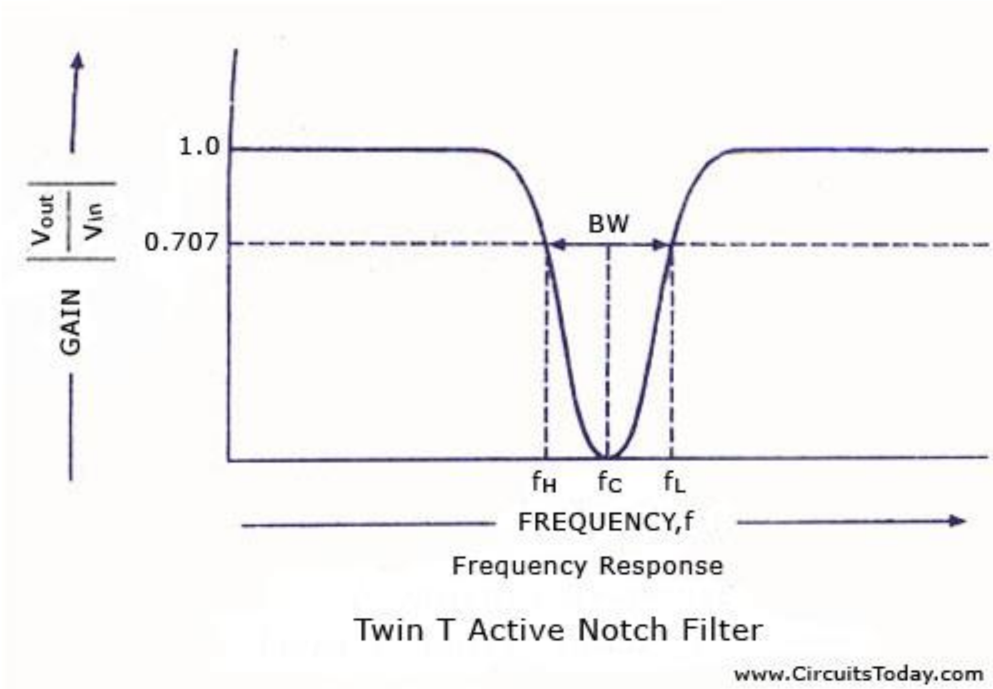
Circuit Diagram
Twin T active notch filter



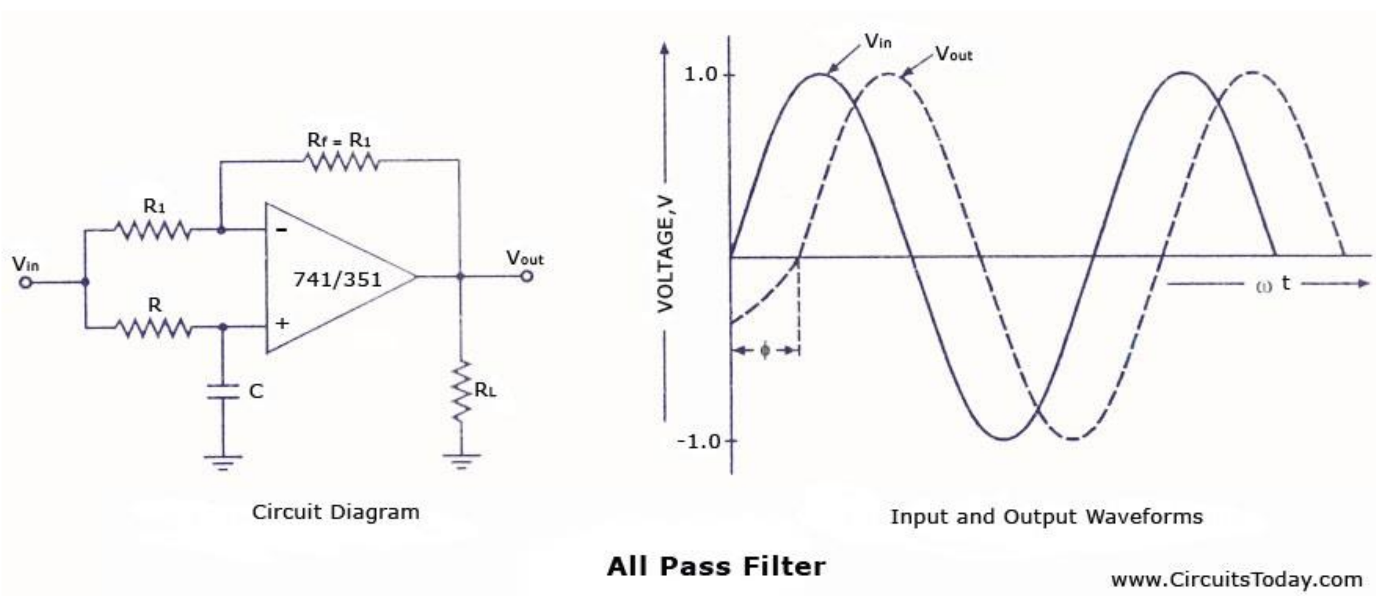
(a) Circuit Diagram (b) Phase Shift (c) Voltage Gain

Twin T Notch Passive Filter

This is also called a notch filter. It is commonly used for attenuation of a single frequency such as 60 Hz power line frequency hum. The most widely used notch filter is the twin-T network illustrated in fig. (a). 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 is made of two capacitors and a resistor.One drawback *of* above notch filter (passive twin-T network) is that it has relatively low figure of merit Q. However, Q of the network can be increased significantly if it is used with the voltage follower, as illustrated in fig. (a). Here the output of the voltage follower is supplied back to the junction of R/2 and 2 C. The frequency response of the active notch filter is shown in fig *(b).*



An all-pass filter is that which passes all frequency components of the input signal without attenuation but provides predictable phase shifts for different frequencies of the input signals. The all-pass filters are also called delay equalizers or phase correctors. An all-pass filter with the output lagging behind the input is illustrated in figure.



The output voltage v_{out} of the filter circuit shown in fig. (a) can be obtained by using the superposition theorem

$$v_{out} = -v_{in} + [-jX_C/R-jX_C]2v_{in}$$

Substituting $-jX_C = [1/j2\prod fc]$ in the above equation, we have

$$v_{out} = v_{in}[-1 + (2/j2\prod Rfc)]$$

or $$v_{out} / v_{in} = 1- j2\prod Rfc/1+ j2\prod Rfc$$

where / is the frequency of the input signal in Hz.

From equations given above it is obvious that the amplitude of v_{out} / v_{in} is unity, that is $|v_{out}| = |v_{in}|$ throughout the useful frequency range and the phase shift between the input and output voltages is a function of frequency.

Oscillators:

An oscillator may be described as a source of alternating voltage. It is different than amplifier.

An amplifier delivers an output signal whose waveform corresponds to the input signal but whose power level is higher. The additional power content in the output signal is supplied by the DC power source used to bias the active device.

The amplifier can therefore be described as an energy converter, it accepts energy from the DC power supply and converts it to energy at the signal frequency. The process of energy conversion is controlled by the input signal, Thus if there is no input signal, no energy conversion takes place and there is no output signal.

The oscillator, on the other hand, requires no external signal to initiate or maintain the energy conversion process. Instead an output signals is produced as long as source of DC power is connected. Fig. 1, shows the block diagram of an amplifier and an oscillator.

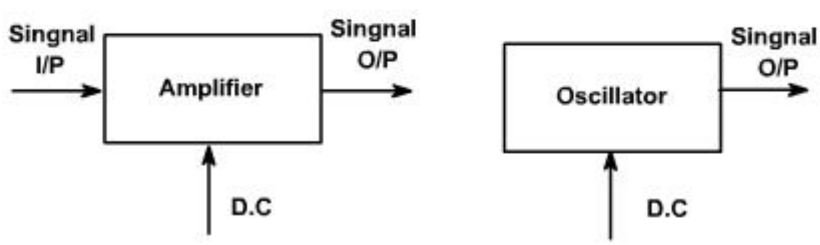


Fig. 1

Oscillators may be classified in terms of their output waveform, frequency range, components, or circuit configuration.

If the output waveform is sinusoidal, it is called harmonic oscillator otherwise it is called relaxation oscillator, which include square, triangular and saw tooth waveforms.

Oscillators employ both active and passive components. The active components provide energy conversion mechanism. Typical active devices are transistor, FET etc.

Passive components normally determine the frequency of oscillation. They also influence stability, which is a measure of the change in output frequency (drift) with time, temperature or other factors. Passive devices may include resistors, inductors, capacitors, transformers, and resonant crystals.

The RC Phase Shift Oscillator:

At low frequencies (around 100 KHz or less), resistors are usually employed to determine the frequency oscillation. Various circuits are used in the feedback circuit including ladder network.

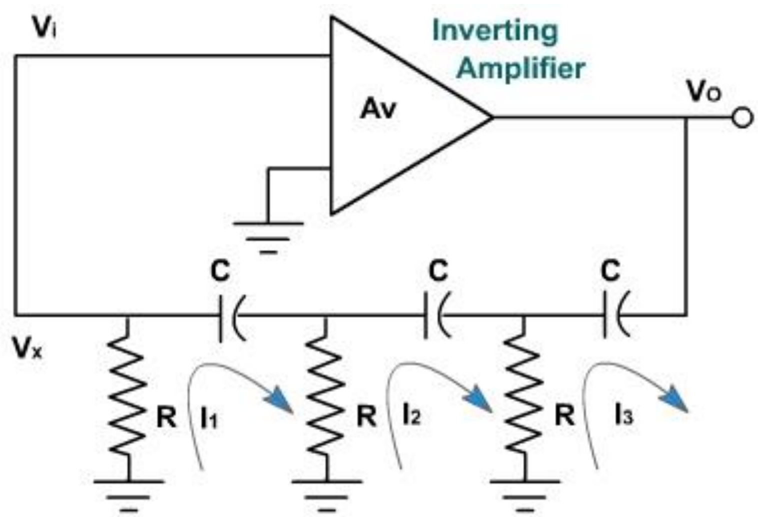


Fig. 4

A block diagram of a ladder type RC phase shift oscillation is shown in fig. 4. It consists of three resistor R and C capacitors. If the phase shift through the amplifier is 180º, then oscillation may occur at the frequency where the RC network produces an additional 180 phase shift.

To find the frequency of oscillation, let us neglect the loading of the phase shift network. Writing the KV equations,

$$V_x = -I_1 R$$

$$X_c = \frac{1}{\omega c}$$

$$I_1 R - jI_1 X_c + (I_1 - I_2) R = 0$$

$$(I_2 - I_1)R - jI_2 X_c + (I_2 - I_3)R = 0$$

$$(I3 - I_1)R - jI_3 X_c + V_0 = 0$$

$$I_1 = -\frac{V_x}{R}$$

$$I_2 = \frac{I_1}{R}(2R - jX_c) = -\frac{V_x}{R^2}(2R - jX_c)$$

$$I_3 = \frac{1}{R}\left\{(2R - jXC)^2\left(-\frac{V_x}{R^2}\right) + VX\right\}$$

$$= -\left\{\frac{3R^2 - XC^2 - j4RX_c}{R^2}\right\}VX$$

$$-V_o = -(R - jX_c)\left\{\frac{3R^2 - X_c^2 j4RXC}{R^2}\right\}V_x + \frac{V_x}{R}(2R - jX_c)$$

$$-V_0 = VX\left\{\frac{\left(3R^3 - RX_c^2 - j4R^2X_c - j3R^2X_c^3 - 4RX_c^2\right) + 2R^3 - jR^2XC}{R^3}\right\}$$

$$\frac{V_x}{V_0} = \frac{R^3}{R^3 - 5RX_c^2 - 6jR^2X_c + jX_c^3}$$

putting $X_c = \dfrac{1}{\omega C}$, we get

$$\frac{V_x}{V_0} = \frac{R^3}{R^3 - \dfrac{5R}{\omega^2C^2} - \dfrac{6jR^2}{\omega C} + \dfrac{j}{\omega^3C^3}}$$

For phase shift equal to 180o between V_x and V_O, imaginary term of V_x / V_O must be zero.

$$\frac{j}{\omega^3C^3} - \frac{6jR^2}{\omega C} = 0$$

$$\omega^2C^2 = \frac{1}{6R^2}$$

$$\omega = \frac{1}{RC\sqrt{6}}$$

Therefore, $$f = \frac{1}{2\pi RC\sqrt{6}}$$

This is the frequency of oscillation. Substituting this frequency in V_x / V_O expression.

$$\frac{V_x}{V_o} = \frac{R^3}{R^3 - 5R.6R^2} = -\frac{1}{29} = \beta$$

This shows that 180° phase shift from Vo to Vx can be obtained if

$$f = \frac{1}{2\pi RC\sqrt{6}}$$

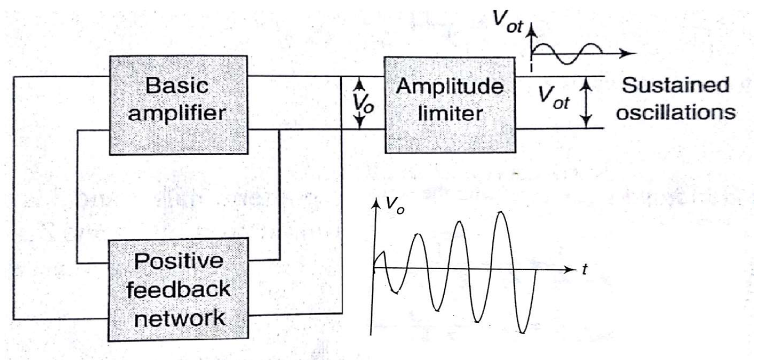
and the gain of it feedback circuit becomes $1/29$.

In order to ensure the oscillation, initially $|A\beta| > 1$ and under study state $A\beta = 1$. This means the gain of the amplifier should be initially greater than 29 (so that $A\beta > 1$) and under steady stat conditions it reduces to 29.

Oscillators:

- The oscillator is a circuit which produces oscillations at desired frequency without giving any input signal.
- The circuit used to generate an a.c voltage without an ac input signal is called an oscillator.
- The circuit requires a dc source to generate ac signal.
- If the output signal is a sine wave then it is called as sinusoidal oscillator or harmonic oscillator. Otherwise it is called as relaxation oscillator.

Oscillators:



Block diagram of Oscillator

Oscillators:

- Oscillator contains both active & passive components, Passive components decides the frequency of oscillations.

Condition for oscillations or Barkhausen criterion:

The conditions to maintain oscillations are

1. $|A\beta| = 1$
2. The total phase shift around the closed loop is 0^0 or 360^0

Classification of oscillators:

Oscillators are classified as

1. According to the waveforms generated

 (a) Sinusoidal oscillator

 (b) Relaxation oscillator

2. According to the fundamental mechanisms

 (a) Negative resistance oscillators

 (b) Feedback oscillators

Classification of oscillators:

3. According to the frequency generated

(a) Audio Frequency Oscillator (AFO): up to 20KHz

(b) Radio Frequency Oscillator (RFO): 20KHz to 30MHz

(c) Very High Frequency Oscillator: 30MHz to 300MHz

(d) Ultra High Frequency Oscillator: 300MHz to 3GHz

(e) Microwave Frequency Oscillator: Above 3GHz

Classification of oscillators:

4. According to the type of circuit used, sine wave oscillators are classified as

(a) RC Oscillators

(b) LC Oscillators

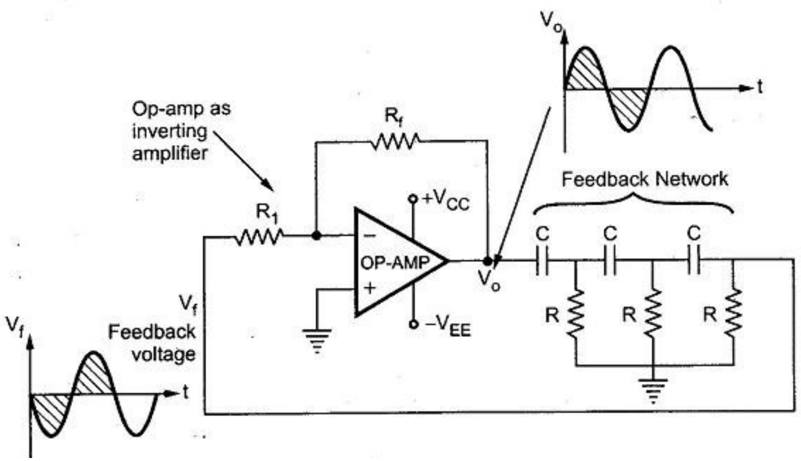
RC Oscillators:

- RC oscillators are popularly used to produce **low frequencies**. There are two important RC oscillators, they are

  1. RC Phase shift oscillator

  2. Wien Bridge oscillator

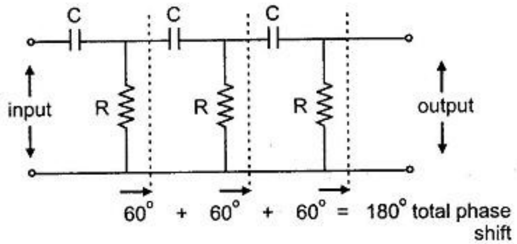
The RC Phase Shift Oscillator:



- Circuit diagram of RC phase shift oscillator

The RC Phase Shift Oscillator:

- OP-amp provides 180^0 phase shift & feedback network provides 180^0 phase shift.
- Each RC section provides a phase shift of 60^0 .



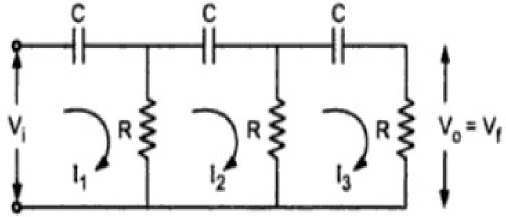
$$f = \frac{1}{2\pi\sqrt{6}\,RC}$$

$$|A| \geq \frac{R_f}{R_1} \geq 29 \text{ for oscillations}$$

$$R_f \geq 29\, R_1$$

Derivation for frequency of oscillations:

- Consider the feedback network
- Apply KVL to the loops



$$I_1\left(R + \frac{1}{j\omega C}\right) - I_2 R = V_i \quad \text{--- (1)}$$

$$-I_1 R + I_2\left(2R + \frac{1}{j\omega C}\right) - I_3 R = 0 \quad \text{--- (2)}$$

$$0 - I_2 R + I_3\left(2R + \frac{1}{j\omega C}\right) = 0 \quad \text{--- (3)}$$

Contd..

- Replace jw as S and using matrix form,

$$\begin{bmatrix} R + \frac{1}{sC} & -R & 0 \\ -R & 2R + \frac{1}{sC} & -R \\ 0 & -R & 2R + \frac{1}{sC} \end{bmatrix} \begin{bmatrix} I_1 \\ I_2 \\ I_3 \end{bmatrix} = \begin{bmatrix} V_i \\ 0 \\ 0 \end{bmatrix} \quad \text{--- (4)}$$

- Solving the above matrix for I_3 using Cramer's rule.

$$I_3 = \frac{D_3}{D} = \frac{V_i\, R^2 s^3 C^3}{1 + 5\,sRC + 6\,s^2C^2R^2 + s^3C^3R^3} \quad \text{--- (5)}$$

$$V_o = V_f = I_3\, R = \frac{V_i\, R^3 s^3 C^3}{1 + 5\,sRC + 6\,s^2C^2R^2 + s^3C^3R^3} \quad \text{--- (6)}$$

Contd..

$$\beta = \frac{V_o}{V_i}$$

$$= \frac{R^3 s^3 C^3}{1 + 5\,sCR + 6\,s^2C^2R^2 + s^3C^3R^3}$$

Replacing s by $j\omega$, s^2 by $-\omega^2$, s^3 by $-j\omega^3$

$$\beta = \frac{-j\omega^3 R^3C^3}{1 + 5\,j\omega CR - 6\,\omega^2C^2R^2 - j\omega^3C^3R^3}$$

Dividing numerator and denominator by $-j\omega^3 R^3C^3$ and using,

$$\boxed{\alpha = \frac{1}{\omega RC}}$$

$$\boxed{\beta = \frac{1}{(1 - 5\alpha^2) + j\alpha(6 - \alpha^2)}}$$

Contd..

- Equating imaginary part to zero to find the frequency of oscillations.

$$\alpha(6-\alpha^2) = 0$$
$$\alpha^2 = 6$$
$$\alpha = \sqrt{6}$$
$$\frac{1}{\omega RC} = \sqrt{6}$$
$$\omega = \frac{1}{RC\sqrt{6}}$$
$$\boxed{f = \frac{1}{2\pi RC\sqrt{6}}}$$

Where, f is frequency of oscillations.

- **Advantages of RC Phase Shift Oscillator:**
- The circuit is simple to design.
- Can produce output over audio frequency range.
- Produces sinusoidal output waveform.
- It is a fixed frequency oscillator.
- **Disadvantages of RC Phase Shift Oscillator:**
- Frequency can be varied by simultaneous changing of all RC components. Hence considered as fixed frequency oscillator.

Contd..

- At this frequency

$$\beta = \frac{1}{1-5\times(\sqrt{6})^2} = -\frac{1}{29} \qquad |\beta| = \frac{1}{29}$$

$$|A|\,|\beta| \geq 1$$
$$|A| \geq \frac{1}{|\beta|} \geq \frac{1}{\left(\frac{1}{29}\right)}$$
$$\boxed{|A| \geq 29}$$

- For oscillations to occur, the gain of the op-amp must be ≥29. which can be adjusted by the resistors R_f & R_1

Problem:

1. Design a phase shift oscillator for the frequency 500Hz with +/-12V supply voltage

Solution : As f is less than 1 kHz, Use op-amp 741 with $I_b(max) = 50$ nA.

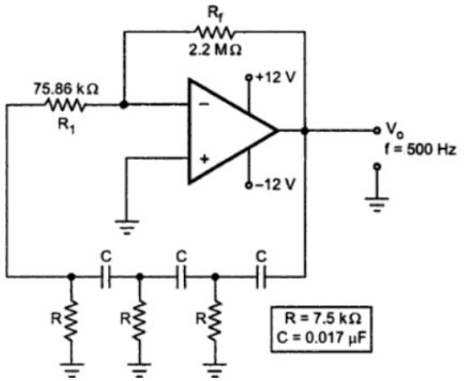
$$I_1 = 100\, I_b(max) = 5\ \mu A$$
$$R_f = \frac{V_o(sat)}{I_1} \quad \text{where } V_o(sat) = 12 - 1 = 11\ V$$
$$= \frac{11}{50\times10^{-6}} = 2.2\ M\Omega \text{ (standard value)}$$
$$A_{CL} = \frac{R_f}{R_1} \geq 29$$
$$R_1 = \frac{R_f}{29} = \frac{220\times10^3}{29} = 75.86\ k\Omega$$

Contd..

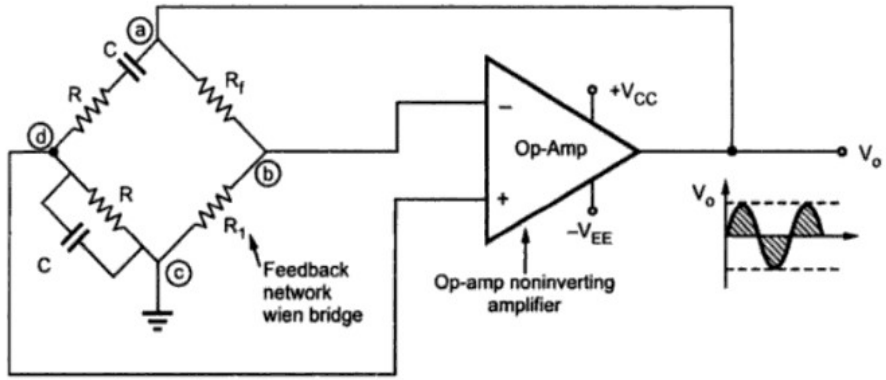
Use standard value of 75.86 kΩ

$$R = \frac{R_1}{10} = \frac{75.8\times10^3}{10} = 7.5\ \text{k}\Omega\ \text{(standard value)}$$

$$C = \frac{1}{2\pi\sqrt{6}\,f R} = \frac{1}{2\pi\sqrt{6}\times500\times7500} = 0.017\ \mu\text{F}$$



Wien Bridge Oscillator :



- In wien bridge oscillator the basic amplifier is working as non inverting amplifier, so that no phase shift is necessary through feedback network.

Contd..

- The two arms of the bridge namely R_1C_1 in series and R_2 C_2 in parallel are called frequency sensitive arms.
- The type of feedback is called lead-lag network.
- The gain of the non inverting amplifier can be adjusted using the resistor R_f and R_1.

$$\boxed{A = 1 + \frac{R_f}{R_1}}$$

Contd..

- To satisfy the Bharkhausen condition Aβ≥ 1. it is necessary that the gain of non inverting amplifier must be min ‘**3**’.

$$|A| \ \geq\ 3 \quad \text{i.e}\quad 1 + \frac{R_f}{R_1} \geq 3$$

$$\frac{R_f}{R_1} \ \geq\ 2$$

- So the ratio of R_f and R_1 must be ≥ 2.
- The frequency of oscillations is given by,

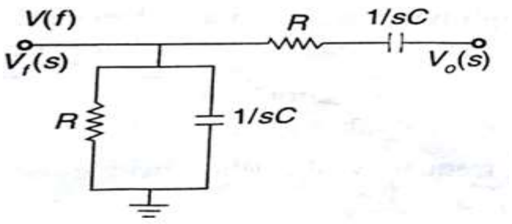
$$\boxed{f = \frac{1}{2\pi RC}\ \text{Hz}}$$

Contd..

- If the resistors and capacitors used in the bridge are not equal then the frequency of oscillations is given by. $f = \dfrac{1}{2\pi\sqrt{R_1R_2C_1C_2}}$ Hz

- Derivation for frequency of oscillations:

- The equivalent circuit of feedback network is

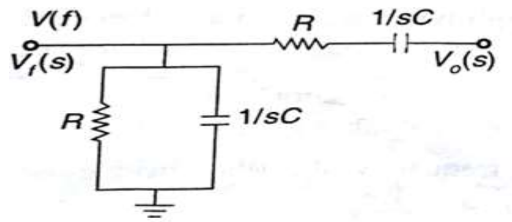


Contd..

- If the bridge is balanced.

$$\frac{R_3}{R_4} = \frac{R_1+XC_1}{\frac{R_2XC_2}{R_2+XC_2}}$$ X_{c1} and X_{c2} are the reactance of the capacitors C_1 and C

$$= \frac{R_1+\left(\frac{1}{j\omega c_1}\right)}{\frac{R_2\left(\frac{1}{j\omega C_2}\right)}{R_2+\frac{1}{j\omega C_2}}} = \frac{R_1-j\left(\frac{1}{\omega c_1}\right)}{\frac{R_2\left(\frac{-j}{\omega C_2}\right)}{R_2-\frac{j}{\omega C_2}}}$$



$$= \frac{\frac{\omega C_1R_1-j}{\omega C_1}}{\frac{\frac{-R_2j}{\omega C_2}}{\frac{\omega C_2R_2-j}{\omega C_2}}} = \frac{\frac{\omega C_1R_1-j}{\omega C_1}}{\frac{-R_2j}{\omega C_2R_2-j}}$$

Contd..

$$= \frac{\omega c_1R_1-j}{\omega c_1} x \frac{\omega c_1R_1-j}{-R_2j}$$

$$= \frac{\omega c_1R_1\,\omega c_2R_2 - j\omega c_1R_1 - j\omega c_2R_2 - 1}{-j\omega c_1R_2}$$

$$= \frac{j[\omega c_1R_1\omega c_2R_2 - j\omega c_1R_1 - j\omega c_2R_2 - 1]}{\omega c_1R_1}$$

$$= \frac{j\omega c_1R_1\omega c_2R_2 + \omega c_1R_1 + \omega c_2R_2 - j}{\omega c_1R_2}$$

$$= \frac{\omega c_1R_1 + \omega c_2R_2 + j[\omega^2R_1R_2c_1c_2 - 1]}{\omega c_1R_2}$$

To get the frequency of the oscillation imaginary part on both the sides made equal to zero

Contd..

$$\omega^2R_1R_2c_1c_2 - 1 = 0$$

$$\omega^2R_1R_2c_1c_2 = 1$$

$$\omega^2 = \frac{1}{R_1R_2c_1c_2}$$

$$\omega = \frac{1}{\sqrt{R_1R_2C_1C_2}}$$

$$f_0 = \frac{1}{2\pi\sqrt{R_1R_2C_1C_2}}$$

If $R_1 = R_2 = R$ and $c_1 = c_2 = c$ then

$$\boxed{f_0 = \frac{1}{2\pi RC}}$$ Hz

Contd..

From the equivalent circuit of Wien bridge oscillator,

$$V_f(s) = V_0(s) . \frac{R \| \frac{1}{SC}}{R + \frac{1}{SC} + R \| \frac{1}{SC}}$$

$$= V_0(s) . \frac{\frac{R}{1+SRC}}{R + \frac{1}{SC} + \frac{R}{1+SRC}}$$

$$= V_0(s) \frac{SRC}{S^2R^2C^2 + 3SRC + 1}$$

Therefore, feedback factor is

$$\beta = \frac{V_f(s)}{V_0(s)} = \frac{SRC}{S^2R^2C^2+3SRC+1}$$

- **Advantages:**
- Easy to adjust frequency
- Perfect sine wave oscillator
- Best audio frequency oscillations

Contd..

We know that $A\beta = 1 \rightarrow A = \frac{1}{\beta} = \frac{S^2R^2C^2+3SRC+1}{SRC}$

Substitute $S = j\omega_0$ to the above equation

We know that $\omega_0 = \frac{1}{RC} \left[f_0 = \frac{1}{2\pi RC} \right]$

$$\boxed{A = 3 = \frac{1}{\beta} \qquad \therefore \beta = \frac{1}{3}}$$

Problem:

- Design a wien bridge oscillator for f_0 = 10 kHz

Solution : Choose C = 0.01 μF

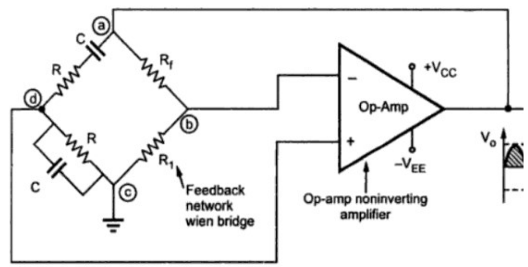
$$R = \frac{1}{2\pi f C} = \frac{1}{2\pi \times 10 \times 10^3 \times 0.01 \times 10^{-6}} = 1.5915 \text{ k}\Omega$$

Choose standard value of 1.5 kΩ

$$R_f = 2 R_1$$

$$R_1 = 1 \text{ k}\Omega$$

$$R_f = 2 \text{ k}\Omega$$



IC Applications

UNIT-II

Topic: Instrumentation Amplifier

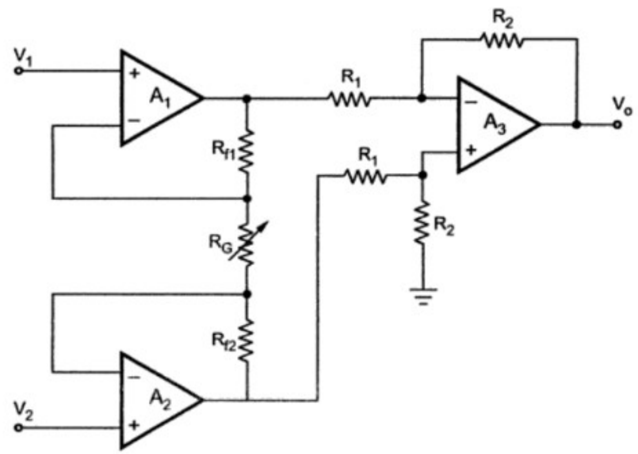
Instrumentation Amplifier:

- In a number of industrial and consumer applications, one is required to measure and control physical quantities.
- Some typical examples are measurement and control of temperature, humidity, light intensity, water flow etc.
- These physical quantities are usually measured with help of transducers.
- The output of transducer has to be amplified so that it can drive the indicator or display. This function is performed by an instrumentation amplifier.

Instrumentation Amplifier:

- Def: **An instrumentation amplifier is used to amplify very low-level signals, rejecting noise and interference signals. Examples can be heartbeats, blood pressure, temperature, earthquakes and so on.**
- The important features of an instrumentation amplifier are
  - High gain accuracy
  - High CMRR
  - High gain stability with low temperature coefficient
  - Low output impedance

Instrumentation Amplifier:

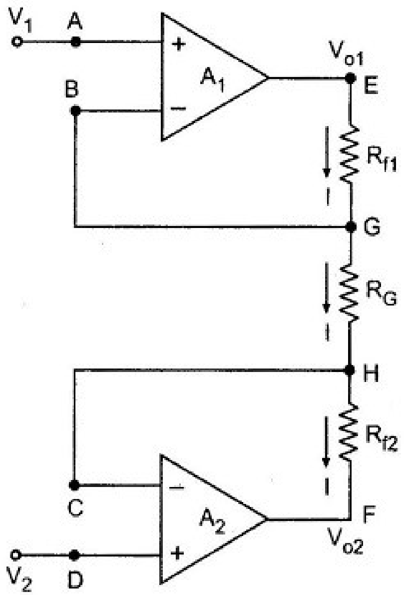


- Three op-amp instrumentation amplifier
- A_1 and A_2 are the non-inverting amplifiers
- A_3 is the difference amplifier

Analysis of Three Op Amp Instrumentation Amplifier:

- The output of the op-amp A_1 is V_{o1} and the output of the op-amp A_2 is V_{o2},

$$V_o = \frac{R_2}{R_1}(V_{o2} - V_{o1})$$



Applying Ohm's law between the nodes E and F we get,

$$I = \frac{V_{o1} - V_{o2}}{R_f + R_g + R_{f2}}$$

Let

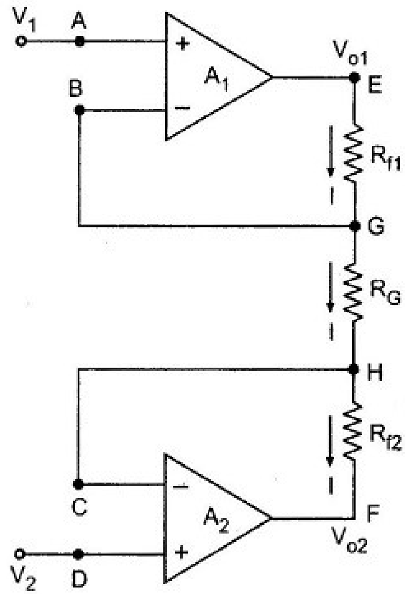
$$R_{f1} = R_{f2} = R_f$$

$$I = \frac{V_{o1} - V_{o2}}{2R_f + R_g}$$

Contd..

- Now from the Observation of nodes G and H,

$$I = \frac{V_G - V_H}{R_G} = \frac{V_1 - V_2}{R_G}$$



Equating the two equations

$$\frac{V_{o1} - V_{o2}}{2R_f + R_G} = \frac{V_1 - V_2}{R_G}$$

$$\frac{V_{o2} - V_{o1}}{2R_f + R_G} = \frac{V_1 - V_2}{R_G}$$

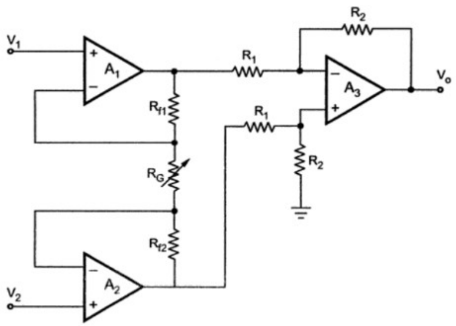
$$V_{o2} - V_{o1} = \frac{(2R_f + R_G)(V_2 - V_1)}{R_G}$$

Contd..

- W.k.t $V_o = \frac{R_2}{R_1}(V_{o2} - V_{o1})$
- Sub $V_{02} - V_{01}$ Value

$$V_o = \frac{R_2}{R_1}\cdot\left[\frac{2R_f + R_G}{R_G}\right](V_2 - V_1)$$

$$\boxed{V_o = \frac{R_2}{R_1}\cdot\left(1 + \frac{2R_f}{R_G}\right)(V_2 - V_1)}$$



Advantages of Instrumentation Amplifier :

- With R_G, the gain can be easily varied.
- Gain can made stable by selecting high quality resistances.
- The input impedance is high.
- The output impedance is very low.
- The CMRR of the op-amp A_3 is very high and most of the common mode signal will be rejected.

Instrumentation amplifier using transducer bridge:

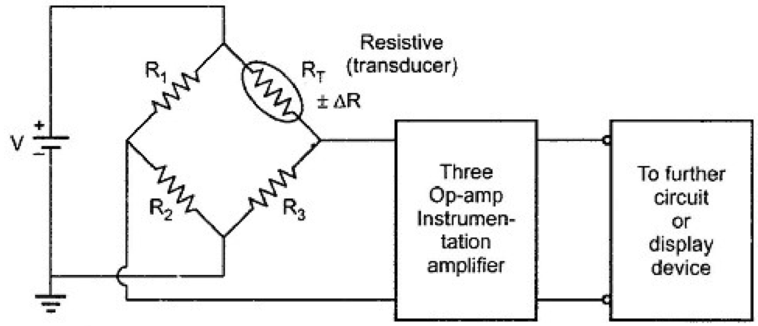
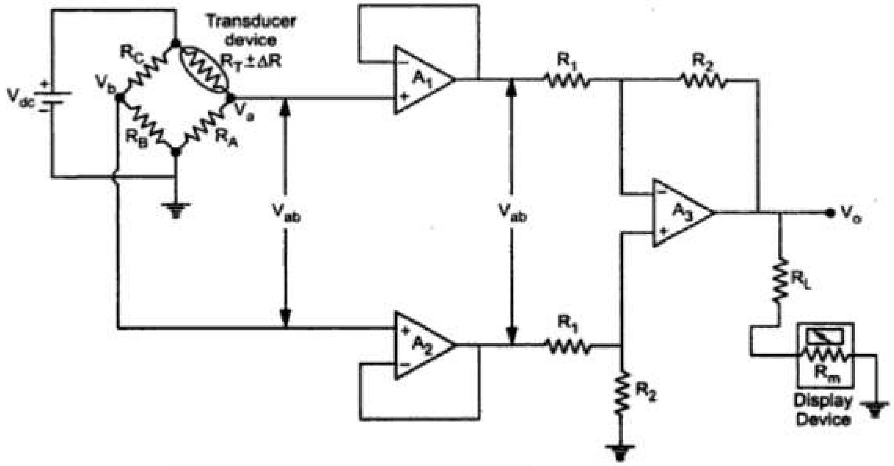


Fig. 2.54 Instrumentation amplifier with transducer bridge

- Single chip instrumentation amplifiers such as AD521, AD524, LH0036, LH0037 used in industries.

Instrumentation amplifier using transducer bridge:



$$\boxed{V_0 = \frac{A\,\Delta R}{4\,R} V_{dc}}$$

Contd..

- The circuit uses a resistive transducer whose resistance changes as a function of the physical quantity to be measured.
- The bridge is initially balanced by a dc supply voltage V_{dc} so that $V_1=V_2$.
- As the physical quantity changes, the resistance R_T of the transducer also changes, causing an unbalance in the bridge ($V_1 \neq V_2$).

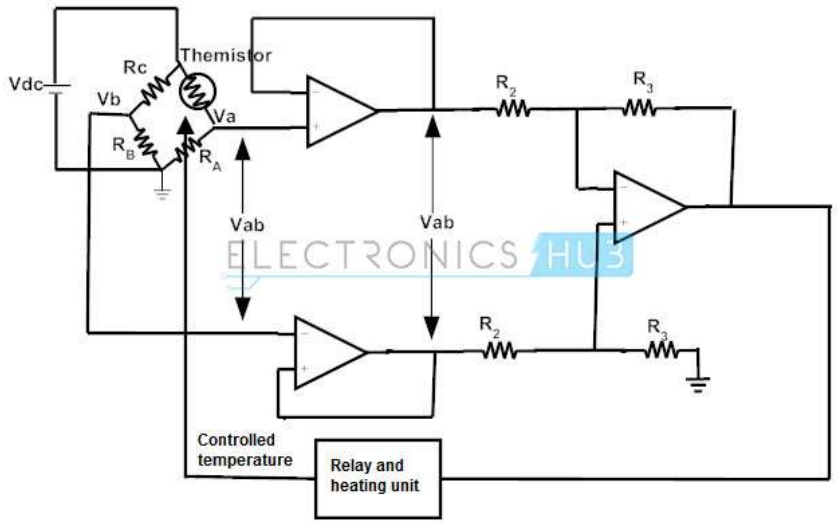
Contd..

- This differential voltage now gets amplified by the three op-amp differential instrumentation amplifier.
- There are number differential applications of instrumentation amplifier with the transducer bridge, such as :
- Temperature indicator, temperature controller, and light intensity meter etc.,

Applications of Instrumentation amplifier:

- Used in data acquisition from small o/p transducers like thermocouples, strain gauges, measurements of Wheatstone bridge, etc.
- Used in navigation, medical, radar, etc.
- These amplifiers are used to enhance the S/N ratio (signal to noise) in audio applications
- **Temperature Controller**
- **Temperature Indicator**
- **Light Intensity Meter**

Temperature Controller:



Contd…

- A simple temperature controller system can be constructed using a thermistor as the transducer device, in the resistive bridge, as shown in the figure above.
- The resistive bridge is kept balanced for some reference temperature. For any change in this reference temperature, the instrumentation amplifier will produce an output voltage, which drives the Relay which in turn turns ON/OFF the heating unit, thereby controlling the temperature.

• Temperature Indicator

- The circuit shown for temperature controller can also be used as a temperature indicator.
- The resistive bridge is kept balanced for a particular reference temperature when V_o = 0V.
- The temperature indicating meter is calibrated to reference temperature, corresponding to this reference condition.

Light Intensity Meter:

- The same circuit can be used to detect variations in the intensity of light, by replacing the thermistor by a Light Dependent Resistor (LDR).
- The bridge is set to a balanced condition in darkness.
- When light falls on the LDR, its resistance changes and unbalances the bridge. This causes the amplifier to produce a finite output, which in turn drives the meter.