10/09/19.
Tuesday

UNIT-IV

ACTIVE FILTERS

→ Filter is nothing but separation and gives desired comp quantities matter. It separates unwanted particles from desired particles.

→ power supply is nothing but converter of AC signal into pure DC signal.

→ Here the characters of signal are frequency, phase, amplitude here signals are separated from based on frequencies.

Filters are 3 types. They based on components. they are

(i) passive and active filter.

Based on signal

(ii) Analog and digital filter.

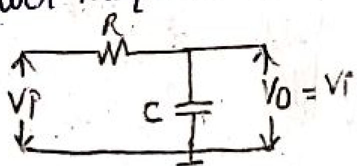
Based on frequency range

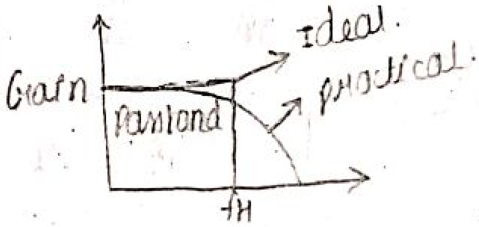
(iii) Audio & radio frequency filters.
(20-20K)   (>20KHZ)

→ Here filters are mainly is active and passive filters. If the filter is constructed by resistor, inductor transistor, capacitor then it is called passive filter and they filter is constructed by op-amp, resistor and capacitor then it is called active filter.

→ If we use inductor, at lower frequencies we use want high value of inductance this increases the winding and resulting it increases size & power dissipation now this is one drawback in passive filters.

→ In passive low pass filter it allows lower frequencies and (gain is maximum) and rejects lower frequencies.





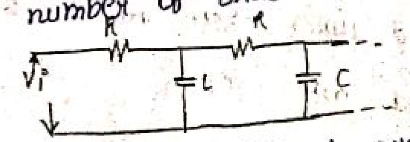
$X_C = \frac{1}{2\pi fC}$ $f_H = \frac{1}{2\pi RC}$

In ideal output is same as input at lower frequencies but in practical, output has some drops and output is gradually reduces after cut-off frequency.

→ Here Gain is also not adjusted in passive filter.

These are another drawbacks of passive filter.

→ Here for ideal characteristics, the output must be sharply reduce its gain. Here so we want high gain rolloff for high gain rolloff we use want to increase number of orders.

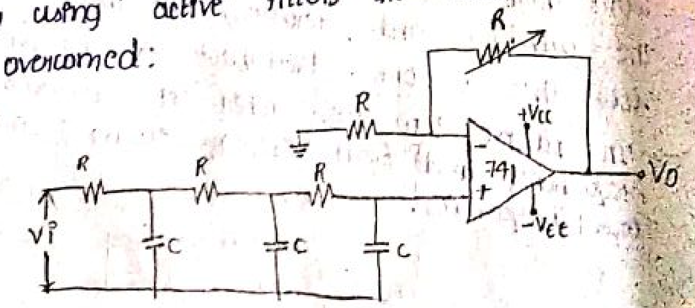


So if we use number of orders then drop across each element and leads to decrease in output voltage. So the o/p is decreases then it leads to loading effect.

Drawbacks of passive filters:

1. circuit size is increases.
2. power dissipation is large.
3. practical characteristics are doesn't match with the ideal characteristics.
4. Loading effect is occurs.
5. Impedance is dismatched. 6. Gain is not adjustable

By using active filters the above all drawbacks are overcomed:



In the above ckt, the power dissipation and size are decreased, because we didn't use any inductor coil here. So the first two drawbacks are overcomed.

And here the output characteristics are nearly ideal characteristics. So the third drawback is also overcomed.

For op-amp the input impedance is very large i.e infinity. So for any low amount of input the ckt can able to drive the load. So the fourth drawback is also overcomed.

Here the output impedance of op-amp is low so it can easily connected to load so to drive the load. Hence impedance is matched to 5th drawback is overcomed.

In the above ckt we use one rheostat. By changing the rheostat position we can adjust the Gain of the ckt. So the 6th drawback is also overcomed.

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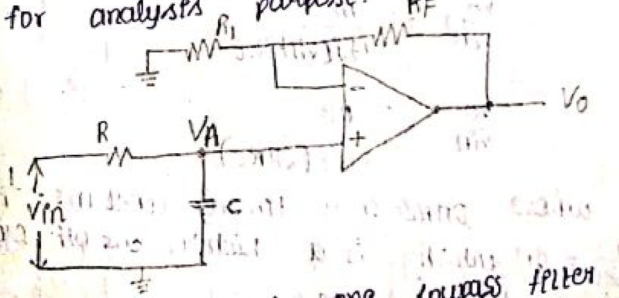
The mostly used filters are
(i) Lowpass (ii) highpass (iii) Bandpass (iv) Band stop / Band rejection (v) allpass filter.

Firstorder active lowpass filter:

Here we use butterworth approximation techniques for analysis purpose.



Here we use only one lowpass filter order at non inverting terminal. And the phase difference between output and input is 0.

→ To know the whether the ckt acts as a LPF or not we must know its frequency response

→ To know frequency response we know gain change with respect to frequency.

→ To know gain we know about output voltage.

→ Here the input for op-amp is V_A and output is V_0. And this is a non-inverting amplifier so

• Gain is $1+\frac{R_F}{R_1}$. The overall gain of the ckt is $\frac{V_0}{V_{in}}$.

$$V_0 = \left(1+\frac{R_F}{R_1}\right)V_A$$

$$A_F = \frac{V_0}{V_A} = \left(1+\frac{R_F}{R_1}\right) \rightarrow ①$$

By using voltage division rule we can measure V_A

$$V_A = \frac{V_{in}\frac{1}{j\omega C}}{R+\frac{1}{j\omega C}}$$

$$V_A = \frac{V_{in}}{1+j\omega RC}$$

from ①; $V_0 = A_F \frac{V_{in}}{1+j\omega RC}$

$$\frac{V_0}{V_{in}} = \frac{A_F}{1+j\omega RC}$$

where $\omega = 2\pi f$

$$\frac{V_0}{V_{in}} = \frac{A_F}{1+j(2\pi f)RC}$$

$$\frac{V_0}{V_{in}} = \frac{A_F}{1+jf(2\pi RC)}$$

where $2\pi RC$ is a timing constant. $\frac{1}{T}$ = frequency

so $\frac{1}{2\pi RC} = f_H$ which is a higher cut-off frequency.

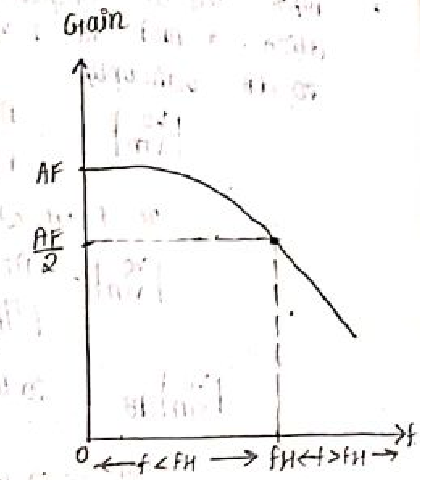
$$\frac{V_0}{V_{in}} = \frac{A_F}{1+j\left(\frac{f}{f_H}\right)}$$

$$\left|\frac{V_0}{V_{in}}\right| = \frac{A_F}{\sqrt{1+(f/f_H)^2}}$$

$$\phi = -\tan^{-1}\left(\frac{f}{f_H}\right)$$

$(f_H = 1KHZ)$

f	Gain	Gain in dB
0		AF
1HZ		AF
100HZ		AF
$f = f_H$		$(AF/\sqrt{2})$
$f > f_H$		Gain is decreases.



* Design a LPF with cut-off frequencies 10KHZ and gain is 2. & draw the frequency response

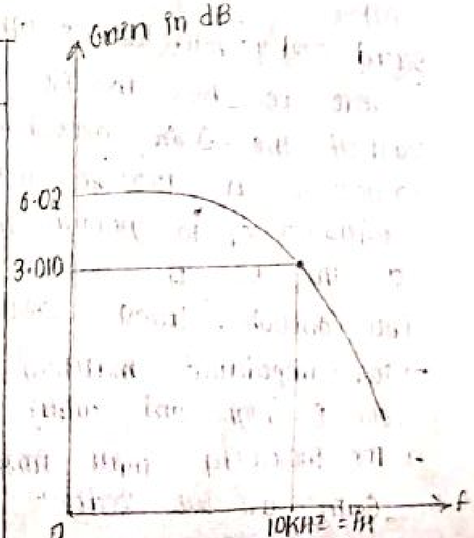
Sol: WKT $\left|\frac{V_0}{V_{in}}\right| = \frac{A_F}{\sqrt{1+(f/f_H)^2}}$

Given $A_F = 2$ and $f_H = 10KHZ$

$$\left|\frac{V_0}{V_{in}}\right| = \frac{2}{\sqrt{1+(f/10K)^2}}$$

$f_H = 10KHZ$

f	Gain	Gain in dB.
0 10HZ	2	6.0205
10HZ	1.999	6.0201
100HZ	1.999	6.02
5 $\cancel{\text{5}}$00HZ	1.9975	6.009.
1KHZ	1.99007.	5.98
5KHZ	1.7889.	5.0517.
10KHZ	1.4142	3.0102
20KHZ	0.8944	-0.969.
30KHZ	0.6324	-3.979.
50KHZ	0.3922	-8.129.1



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* After cut-off frequency we want have maximum attenuation. And so here we for 1 decade it reduces 20 dB Gain only.

$$\left|\frac{V_0}{V_{in}}\right| = \frac{A_F}{\sqrt{1+(f/f_H)^2}}$$

If $f > f_H$, neglect 1 then

$$\left|\frac{V_0}{V_{in}}\right| = \frac{A_F}{(f/f_H)}$$

$$\left|\frac{V_0}{V_{in}}\right|_{dB} = 20\log\left(\frac{A_F}{(f/f_H)}\right)$$

$$= 20\log\left(A_F\left(\frac{f}{f_H}\right)^{-1}\right) \quad \left[\begin{array}{l} f = 10K \;\&\; f_H = 1K \\ -1 \text{ decade} \end{array}\right]$$

$$\left|\frac{V_0}{V_{in}}\right|_{dB} = -20\log\ (\text{constant Gain}).$$

so for every 1 decade of frequency Gain reduces (roll off) by 20 dB.

→ so here the ideal and practical characteristics are not matched due to not having any sharp cutting reduction of Gain.

so we have to go for second order active low pass filter.

Second order Active lowpass filter:

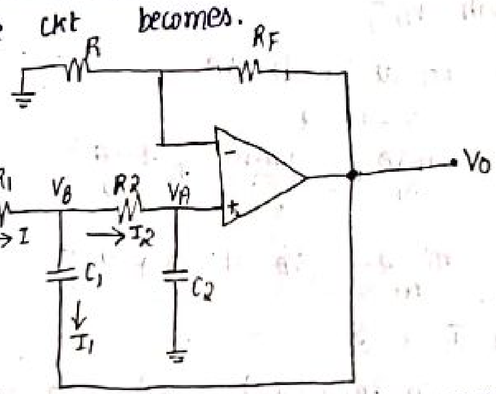
here we have increase another order at input to roll off the Gain maximum Gain.

→ Here we use two RC sections. If we connect the capacitor C_1 to ground the input of the impedance of the ckt is reduces due to maximum current flow through lowest resistance path.

→ For impedance matching the input resistance of ckt is high and output resistance is small.

→ For increasing input impedance and for better Gain we use feedback from C_1 to output.

then the ckt becomes.

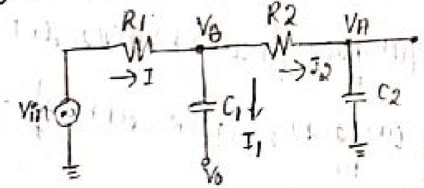


Analysis: Here we have to calculate frequency response to know whether the ckt acts as LPF or not. For frequency response, we have to calculate output Gain changes with respect to frequency.

→ For Gain we have to calculate output voltage Vo.

$$V_0 = A_F V_A.$$

Here V_A for V_A analysis the ckt is complex to analysis. So we have to redraw it.



By voltage division rule

$$V_A = \frac{V_B\left(\frac{1}{j\omega C_2}\right)}{R_2+\left(\frac{1}{j\omega C_2}\right)}.$$

$$V_A = \frac{V_B\left(\frac{1}{SC_2}\right)}{R_2+\left(\frac{1}{SC_2}\right)}.$$

$$V_A = \frac{V_B}{1+SR_2C_2}.$$

$$V_B = V_A(1+SR_2C_2). \longrightarrow ①$$

[Here directly finding V_B is difficult because we have to apply superposition theorem. so we calculate it through V_A.]

Apply KCL at node V_B

$I = I_1 + I_2$

$$\frac{V_{in}-V_B}{R_1} = \frac{V_B-V_o}{\left(\frac{1}{SC_1}\right)} + \frac{V_B-V_A}{R_2}$$

$$\frac{V_{in}-V_B}{R_1} = (V_B-V_o)(SC_1) + \frac{V_B-V_A}{R_2} \longrightarrow ⑤$$

from ① & ②

$$⑤ \Rightarrow \frac{V_{in}-V_A(1+SR_2C_2)}{R_1} = (V_A(1+SR_2C_2)-V_o)(SC_1) + \frac{V_A(1+SR_2C_2)-V_A}{R_2}$$

$$\frac{V_{in}-V_A-V_ASR_2C_2}{R_1} = SC_1V_A + V_AS^2R_2C_2C_1 - V_oSC_1 + \frac{V_A[1+SR_2C_2-1]}{R_2}$$

$$\frac{V_{in}-V_A-V_ASR_2C_2}{R_1} = SC_1V_A + V_AS^2R_2C_2C_1 - V_oSC_1 + \frac{SR_2C_2V_A}{R_2}$$

$$\frac{V_{in}-V_A-V_ASR_2C_2}{R_1} = SC_1V_A + V_AS^2R_2C_2C_1 - V_oSC_1 + SC_2V_A$$

$$\frac{V_{in}}{R_1} + V_oSC_1 = V_A\left[\frac{(1+SR_2C_2)}{R_1}\right] + SC_1V_A + V_AS^2R_2C_2C_1 + SC_2V_A$$

$$\frac{V_{in}}{R_1} = V_A\left[\frac{(1+SR_2C_2)}{R_1} + SC_1 + S^2R_2C_2C_1 + SC_2 - \frac{V_oSC_1}{V_A}\right]$$

$$V_A = \frac{\frac{V_{in}}{R_1}}{\frac{(1+SR_2C_2)+SR_1C_1+S^2R_1R_2C_1C_2+SC_2R_1}{R_1} - \frac{V_o}{V_A}SC_1R_1}$$

$$V_A = \frac{V_{in}}{(1+SR_2C_2)+SR_1C_1+S^2R_1R_2C_1C_2+SC_2R_1 - \frac{V_o}{V_A}SC_1R_1}$$

$$V_{in} = V_A\left\{(1+SR_2C_2)+SR_1C_1+S^2R_1R_2C_1C_2+SC_2R_1 - \frac{V_o}{V_A}SC_1R_1\right\}$$

$$V_{in} = V_A\left\{1+SR_2C_2+SR_1C_1+S^2R_1R_2C_1C_2+SC_2R_1 - A_FSC_1R_1\right\}$$

$$\left[\because \frac{V_o}{V_A} = A_F\right]$$

$$V_A = \frac{V_o}{A_F}$$

$$V_{in} = \frac{V_o}{A_F}\left\{1+SR_2C_2+SR_1C_1+S^2R_1R_2C_1C_2+SC_2R_1-A_FSC_1R_1\right\}$$

$$\frac{V_o}{V_{in}} = \frac{A_F}{S^2R_1R_2C_1C_2+S(R_1C_1+R_1C_2+R_2C_2-A_FR_1C_1)+1} \longrightarrow ③$$

Divide with $R_1R_2C_1C_2$ then

$$\frac{V_o(S)}{V_{in}(S)} = \frac{A_F/R_1R_2C_1C_2}{S^2+S\frac{(R_1C_1+R_1C_2+R_2C_2-A_FR_1C_1)}{R_1R_2C_1C_2}+\frac{1}{R_1R_2C_1C_2}} \longrightarrow ④$$

The standard form of any 2nd order system is Given by

$$\frac{V_o(S)}{V_{in}(S)} = \frac{A}{S^2+2\xi\omega_nS+\omega_n^2} \longrightarrow ⑤$$

$\xi \to$ damping ratio(or) damping factor.

$\omega_n \to$ natural frequency here ω_H(higher cut off frequency)

compare Eqns ④ & ⑤ then

$$\omega_n^2 = \frac{1}{R_1R_2C_1C_2}$$

$$\omega_H^2 = \frac{1}{R_1R_2C_1C_2} \longrightarrow ⑥$$

$$(2\pi f_H)^2 = \frac{1}{R_1R_2C_1C_2}$$

$$f_H^2 = \frac{1}{(2\pi)^2R_1R_2C_1C_2}$$

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

If $R_1 = R_2 = R$ and $C_1 = C_2 = C$ then

$$f_H = \frac{1}{2\pi\sqrt{R^2C^2}}$$

$$\boxed{f_H = \frac{1}{2\pi RC}}$$

and Eqn ③ becomes.

$$\frac{V_o(S)}{V_{in}(S)} = \frac{A_F}{S^2(RC)^2+SRC(3-A_F)+1}$$

From ⓑ ; $\omega_H^2 = \frac{1}{R_1R_2C_1C_2} = \frac{1}{(RC)^2}$

$(RC)^2 = \frac{1}{(\omega_H)^2}$ and $s = j\omega$.

$$\frac{V_0(s)}{V_{in}(s)} = \frac{A_F}{\left(\frac{j\omega}{\omega_H}\right)^2 + \frac{(j\omega)}{\omega_H}(3-A_F)+1} \rightarrow ⓐ$$

Here we consider the butterworth approximation techniques for better plot response.

The standard form of butterworth approximation for second order is

$$\frac{V_0(s)}{V_{in}(s)} = \frac{A}{s^2+s\sqrt{2}+1} \rightarrow ⓑ$$

comparing ⓐ & ⓑ we get

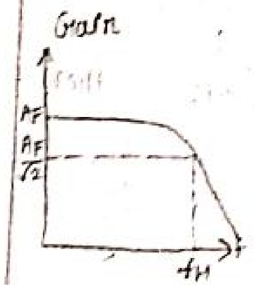
$3 - A_F = \sqrt{2}$.

From ⓐ ; $\frac{V_0(s)}{V_{in}(s)} = \frac{A_F}{-\frac{(\omega)^2}{(\omega_H)} + \sqrt{2}\frac{j\omega}{\omega_H}+1}$

$$\frac{V_0(s)}{V_{in}(s)} = \frac{A_F}{1-\left(\frac{\omega}{\omega_H}\right)^2 + \sqrt{2}\frac{j\omega}{\omega_H}}$$

$$\left|\frac{V_0(s)}{V_{in}(s)}\right| = \frac{A_F}{\sqrt{\left[1-\left(\frac{\omega}{\omega_H}\right)^2\right]^2 + \left[\sqrt{2}\left(\frac{\omega}{\omega_H}\right)\right]^2}}$$

$$= \frac{A_F}{\sqrt{1+\left(\frac{\omega}{\omega_H}\right)^4 - 2\left(\frac{\omega}{\omega_H}\right)^2 + 2\left(\frac{\omega}{\omega_H}\right)^2}}$$

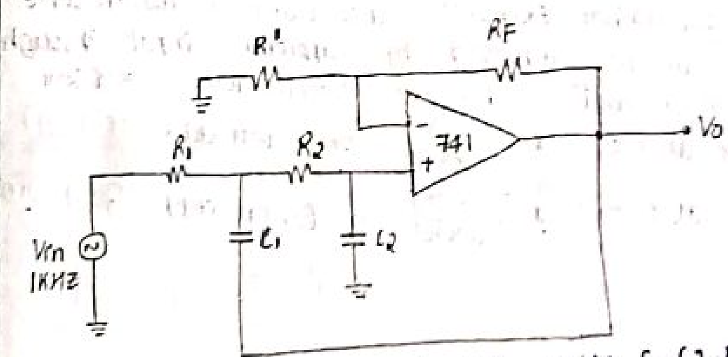


$$\left|\frac{V_0(s)}{V_{in}(s)}\right| = \frac{A_F}{\sqrt{1+\left(\frac{\omega}{\omega_H}\right)^4}}$$

Here the Gain roll off is 40dB because in denominator we have 4th degree. So here gain is reduces better than first order active low pass filter.

* Design a second order active lowpass filter with cut-off frequency $f_H = 1KHz$. $3 - A_F = \sqrt{2}$

Sol:



Here we have to calculate $R_1, R_2, C_1, C_2, R', R_F$

WKT $f_H = \frac{1}{2\pi\sqrt{R_1R_2C_1C_2}}$

If $R_1 = R_2 = R$ and $C_1 = C_2 = C$ then

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

Let $\boxed{C = 0.01\mu F}$ then

$R = \frac{1}{2\pi(1K)(0.01\mu)} \simeq 15.91K\Omega$

$\boxed{R \simeq 15K\Omega}$

We know that

$3 - A_F = \sqrt{2}$.

$A_F = 3 - 1.414$

$A_F = 1.586$

WKT $1 + \frac{R_F}{R'} = A_F \Rightarrow 1.586 - 1 = \frac{R_F}{R'}$

$0.586 R' = R_F$, assume $R' = 1K\Omega$

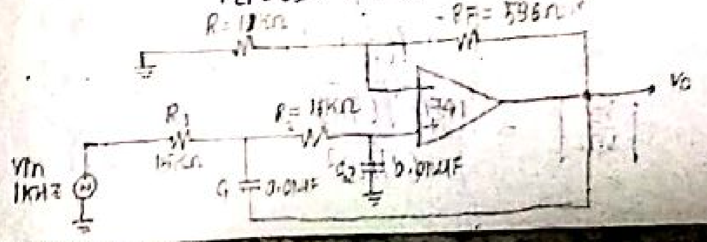
$R_F = 0.586 \times 15K$ 1KΩ

$R_F = 8.79 K\Omega$ $R_F = 0.586K\Omega$

$R_F \simeq 10K\Omega$ $R_F \simeq 586\Omega$

$R_1 = R_2 = R = 15K\Omega$ and $R_F = 10K\Omega$, $R' = 1K\Omega$

$C_1 = C_2 = C = 0.01\mu F$



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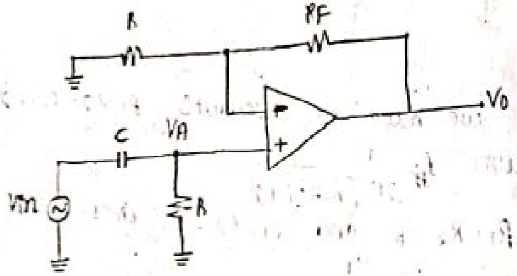
* **First order Highpass filter:**

For lower frequencies, the Gain must be minimum and for higher frequencies, the Gain is maximum. This can be achieved by appling input through capacitor and output is taken across resistor.

at $f=0$; $X_C = \frac{1}{2\pi(0)C} = \infty$ (open ckt). (Gain = 0)

at $f=\infty$; $X_C = \frac{1}{2\pi\infty C} = 0$ (short ckt). (Gain = max)



Analysis:

Here we have to calculate Gain for frequency response. For Gain, we should calculate Vo.

$$V_O = A_F V_A$$

For Vo, we calculate VA

$$V_A = \frac{V_{in} R}{R + \left(\frac{1}{j\omega C}\right)}$$

$$V_A = \frac{V_{in}\, j\omega RC}{1 + j\omega RC}$$

$$V_O = A_F \left[\frac{V_{in}\, j\omega RC}{1 + j\omega RC}\right]$$

$$V_O = A_F \left[\frac{V_{in}\, j(2\pi f)RC}{1 + j(2\pi f)RC}\right]$$

where RC is timing constant. $1/T = f$, so $1/2\pi RC$ produces one frequency i.e. f_L (because of HPF).

$$\boxed{V_O = A_F \left[\frac{V_{in}\, j\left(\frac{f}{f_L}\right)}{1 + j\left(\frac{f}{f_L}\right)}\right]}$$

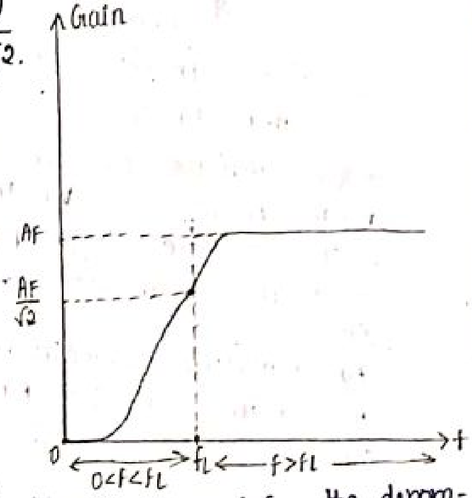
$$\left|\frac{V_O}{V_{in}}\right| = \frac{\sqrt{(A_F)^2}\sqrt{(f/f_L)^2}}{\sqrt{1 + (f/f_L)^2}}$$

$$\left|\frac{V_O}{V_{in}}\right| = \frac{A_F\,(f/f_L)}{\sqrt{1 + (f/f_L)^2}}$$

$$\phi = -\tan^{-1}(f/f_L)$$

Let $f_L = 1\text{KHz}$

I/P freq (f)	Gain
0	0
10	$\simeq 0$
100	$\simeq 0$
1K	$A_F/\sqrt{2}$
10K ($f>f_L$)	Gain maintains constant (A_F).



After f_L i.e $f>f_L$ the denominator increases with incr correspondingly with increase in numerator. So Gain maintains a constant value as A_F (neglect 1 in denominator. $\left|\frac{V_O}{V_{in}}\right| = \frac{A_F(f/f_L)}{(f/f_L)} = A_F$

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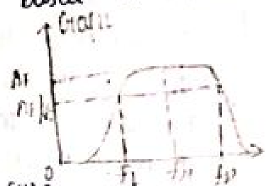
* **Bandpass filter:**

It allows the certain range of frequencies and rejects all other frequencies. So we have two stop bands and one passband. Pass band allows frequencies and stop bands means the rejected low and high frequencies before passband and after passband respectively.



→ Bandpass filters are two types based on quality factor

$$Q = \frac{f_H}{B.W.}$$

$Q>10$; then it is narrowband passfilter

$Q<10$; then it is wideband pass filter.

(1) wide bandpass filter ($Q \le 10$).

(2) narrow bandpass filter ($Q>10$).

$$Q = \frac{f_r}{B.W.}$$

B.W = uppercutoff frequency - lowercut off frequency

$$B.W = f_H - f_L$$

→ For calculate f_r we didn't use arithmetic mean because of varying components with respect to time we didn't use arithmatic mean so we use geometric mean for time varying factors.

By calculating geometric mean for f_H and f_L we obtain f_r. here f_H and f_L are only two factor so $f_r = \sqrt{\quad}$ $f_r = [f_H f_L]^{1/2}$.

$$f_r = \sqrt{f_H . f_L}$$

Now we determined both f_r and B.W so we easily calculate quality factor and for from Q we easily said whether the filter is wide (or) narrow band filter.

$$Q = \frac{f_r}{B.W.}$$

$$Q = \frac{\sqrt{f_H f_L}}{\sqrt{f_H - f_L}} = \frac{\sqrt{f_H f_L}}{f_H - f_L}$$

→ In wideband pass filter we the more frequencies are allowed to output. So we simply select some frequencies. so it is a less selective circuit.

→ For narrowband pass filter the selection is difficult to select some of the frequencies only. so it is a more selective circuit.

Widepass

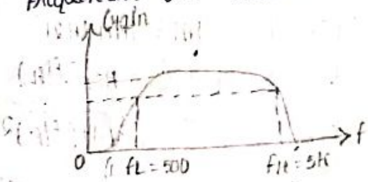
wideband pass filter:

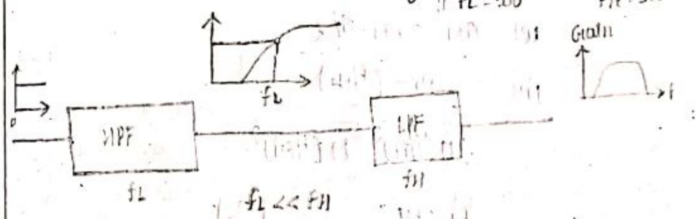
Wideband pass filter is constructed by using HPF and LPF. highpass filter followed by lowpass filter is wideband pass filter.

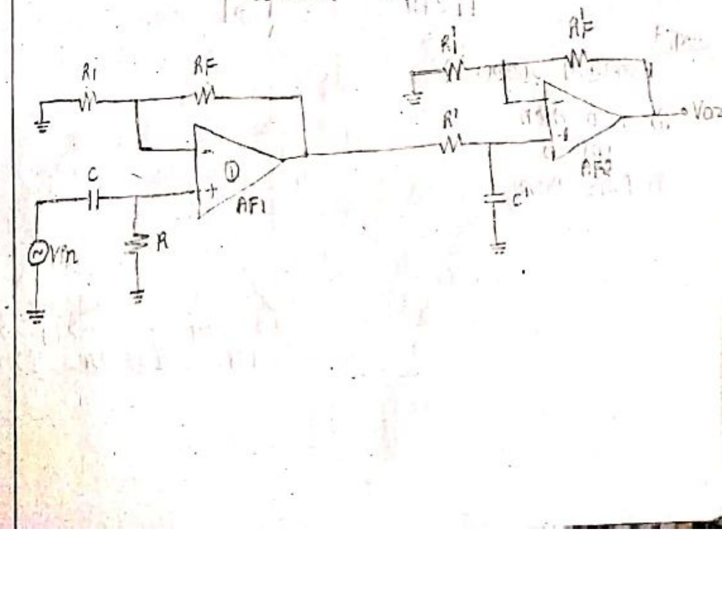
→ By selecting proper values of components, we fix the cut-off frequency f_L. From f_L onwards the HPF allows the frequencies. If f_L is 500 then HPF allows all frequencies greater than 500 and rejects upto 500.

Now this output is given to lowpass filter and by proper values of R and C we fix f_H. let f_H is 5K then LPF allows all frequencies upto 5K and rejects above 5K frequencies.

The output of LPF contains only 500 to 5K frequencies because below 500 frequencies are rejected by HPF and above 5K frequencies are rejected by LPF then the response is equivalent to wide bandpass filter.







Analysis:

Highpass filter:

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

$$A_{F1} = 1 + \frac{R_F}{R_1}$$

$$|A_1| = \left|\frac{V_{o1}}{V_{in}}\right| = \frac{A_{F1}(f/f_L)}{\sqrt{1+(f/f_L)^2}}$$

Lowpass filter:

$$f_H = \frac{1}{2\pi R'C'}$$

$$A_{F2} = 1 + \frac{R_F'}{R_1'}$$

$$|A_2| = \left|\frac{V_{o2}}{V_{o1}}\right| = \frac{A_{F2}}{\sqrt{1+(f/f_H)^2}}$$

Here both HPF and LPF are in cascaded connection. So total gain is the multiplication of individual gains.

$$|A| = |A_1||A_2|$$

$$= \frac{A_{F1}(f/f_L)}{\sqrt{1+(f/f_L)^2}} \cdot \frac{A_{F2}}{\sqrt{1+(f/f_H)^2}}$$

Let $A_{FT} = A_{F1} \cdot A_{F2}$.

$$|A| = \frac{A_{FT}(f/f_L)}{\sqrt{1+(f/f_L)^2}\sqrt{1+(f/f_H)^2}}$$

$f_L < f_H$

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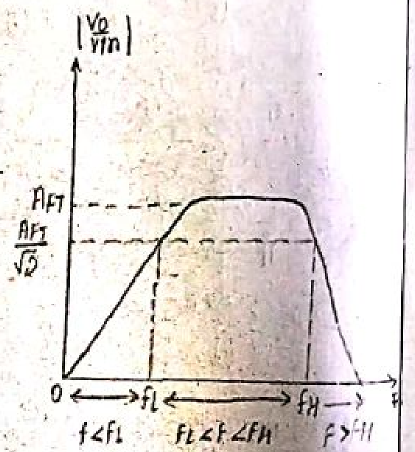
Frequency response:

(i) $f = 0$ then

$|A| = 0$.

(ii) $f < f_L$ then



Band rejection (or) Band elimination (or) Band stop filter: It has 2 passbands and one stopband. It rejects the certain band of frequencies and used in biomedical applications.

There are two types based on quality factor:

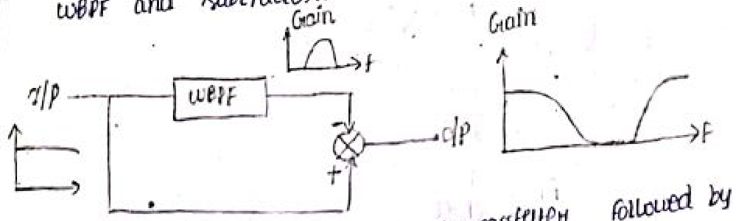
(i) wide band rejection filter WBRF ($Q<10$)

(ii) Narrowband rejection filter NBRF ($Q>10$)

(i) **Wideband rejection filter: Method 1:**

In this the we reject more number of frequencies and here we have less selection.

→ wideband rejection filter is constructed by using a WBPF and subtractor.



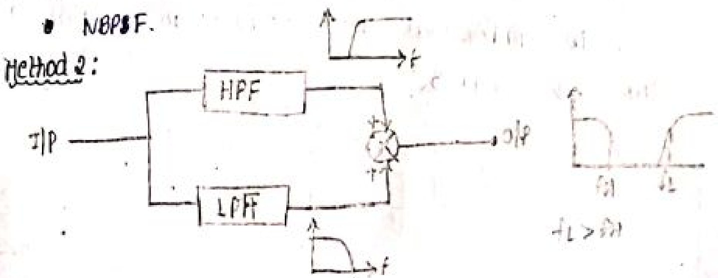
So here we require one act lowpassfilter followed by highpass filter and one subtractor.

→ Here for subtractor, the for negative terminal we have to apply WBPF output and for positive terminal we apply the input which has all frequencies.

→ then subtractor subtracts WBPF o/p from all pass frequencies then the output response becomes the WBRF response.

→ For WBRF we use WBPS and for NBRF we use NBPSF.

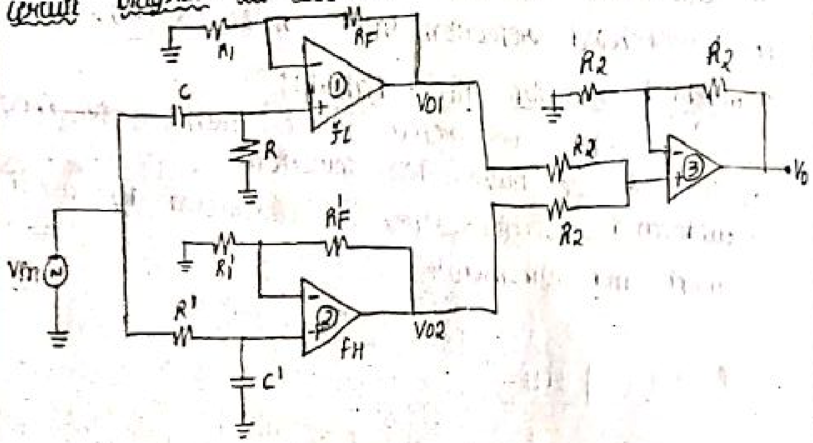
Method 2:



If we want to construct WBRF by using LPF and HPF then we should make $f_L > f_H$. For WBPS $f_H > f_L$.

→ In BRF, we require one adder inorder to make add the responses of LPF and HPF. Then the output response is the response of BRF.

Circuit Diagram for method 2:



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

$$f_H = \frac{1}{2\pi R'C'}$$

~~$f_L < f_H$~~ $f_L > f_H$

$$A_F = 1 + \frac{R_F}{R_1}$$

$$V_{01} = \frac{V_{in} A_F j\left(\frac{f}{f_L}\right)}{1 + j\left(f/f_L\right)}$$

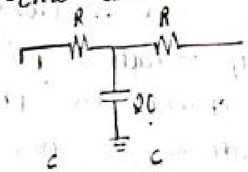
$$V_{02} = \frac{V_{in} A_F}{1 + j\left(f/f_H\right)}$$

For adder, all the resistors are same. so that $\boxed{V_o = V_{01} + V_{02}}$

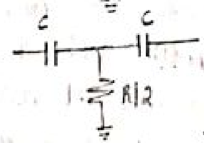
(ii) Narrow band rejection filter (or) Notch filter:

It is constructed by using Twin T followed by voltage follower (or) notch filter is implemented by cascading connection of twin T and voltage follower.

→ Here T-shape is implemented by passive components and we have follow some rules.

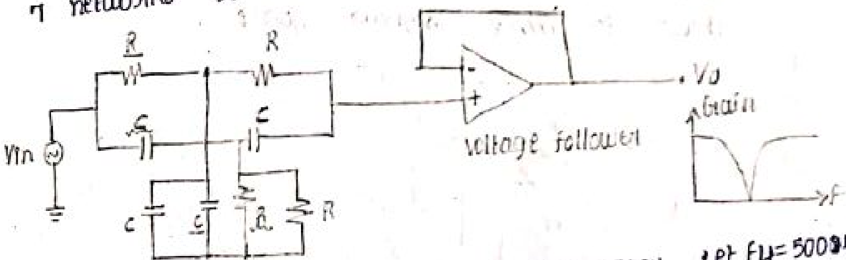


→ Here the two resistor values must be equal the capacitor value is double the actual capacitor value.



→ Here the two capacitors have same value and resistor value is half of actual resistor.

→ In the construction of NBRF we use the above two T networks so that we call it as Twin T network.



→ Here the R and C acts as lowpass filter. Let $f_H = 500Hz$ then upto f_H it allows all frequencies. Except f_H.

→ And R and C acts as highpass filter and let $f_L = 500H$ then after f_H it allows all frequencies. Except f_H 500H is rejected. In above

→ so in overall response is rejected. Hence it is only one frequency is rejected. called as narrow band rejection filter.

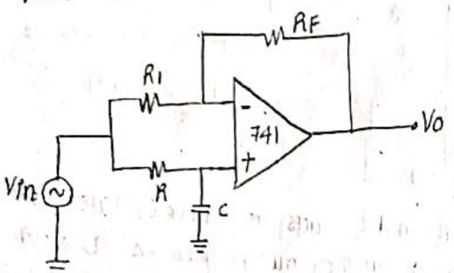
26/09/19 **All pass filter:**

It allows all frequencies i.e the signal strength is maximum at all frequencies.

→ In general without using any device the signal strength is constant.

→ All pass or filter are also called as phase comparator (or) phase delay (or) phase corrector.

→ In the transmission of sig we want to transmit that in freespace then if any obstacle is present in freespace then the signal changes its phase i.e the signal bends and continues the transmission. At receiver the transmitted sig phase and received sig phase are not same. so we use all pass filter at receiver. this filter mai doesn't change amplitude and frequency but it produces corresponding phase to make signals equals

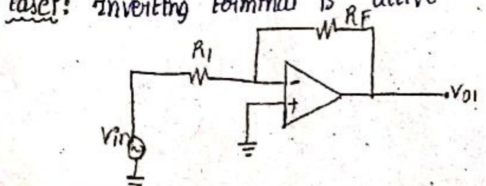


Analysis:

Here we have to prove the above ckt has unity gain and because it didn't change the amplitude and frequency.

Here both inverting and non-inverting terminals so we are in active. So we use superposition theorem

case I: Inverting terminal is active

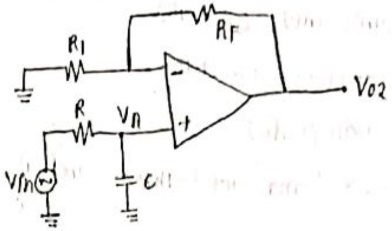


$$V_{o1} = \left(-\frac{R_F}{R_1}\right) V_{in}.$$

If $R_F = R_1$ then

$$V_{o1} = -V_{in}$$

case II: non-inverting terminal is in active



$$V_{o2} = \left(1+\frac{R_F}{R_1}\right) V_A.$$

If $R_F = R_1$ then

$$V_{o2} = 2V_A$$

where $V_A = \dfrac{V_{in}\left(\frac{1}{j\omega C}\right)}{R+\left(\frac{1}{j\omega C}\right)}$

$$V_A = \frac{V_{in}}{1+j\omega RC}$$

$$V_{o2} = 2\left[\frac{V_{in}}{1+j\omega RC}\right]$$

Total output voltage $V_o = V_{o1} + V_{o2}$

$$V_o = V_{o1} + V_{o2}.$$

$$V_o = -V_{in} + \frac{2V_{in}}{1+j\omega RC}$$

$$V_o = V_{in}\left[-1 + \frac{2}{1+j\omega RC}\right]$$

$$\frac{V_o}{V_{in}} = \left[\frac{2-(1+j\omega RC)}{1+j\omega RC}\right]$$

$$\frac{V_o}{V_{in}} = \frac{1-j\omega RC}{1+j\omega RC}$$

$$\left|\frac{V_o}{V_{in}}\right| = \frac{\sqrt{1+(\omega RC)^2}}{\sqrt{(1+(\omega RC)^2}}$$

$$\left|\frac{V_o}{V_{in}}\right| = 1$$

Here output voltage is equal to input voltage So it acts as all pass filter.

$$\phi = \tan^{-1}(-\omega RC) - \tan^{-1}(\omega RC)$$

$$\phi = -\tan^{-1}(\omega RC) - \tan^{-1}(\omega RC)$$

$$\phi = -2\tan^{-1}(\omega RC)$$

and $\omega = 2\pi f$ and let $f = 1KHz$ and $R = 1K\Omega$ and $C = 0.1\mu F$ then

$$\phi = -2\tan^{-1}[2\pi \times 1K \times 1K \times 0.1\mu]$$

$$\phi = -64.28$$

By choosing the proper values of R and C we can change the phase angle

→ Here we get lagging phase if you want leading phase then we can just interchange R and C in this manner we can correct the phase angle by changing R and C and interchanging also.

→ Let us consider the phase angle is -90° then the input and output waveforms be [let vin is sinusoidal s/g

→ Here we didn't change the amplitude and frequency but we only phase is changes.