Op-Amp Practical Applications: Design, Simulation and Implementation
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Lecture – 02
Introduction/ Summary on Op-amps Contd

Welcome to this module. And, this is in continuation with our first module regarding understanding the op-amps. The idea of these particular two modules is to summarize; what are the parameters, characteristics of an operational amplifier, and here now we will see filters and oscillators. Because we will be implementing this particular theory that we are understanding in this two modules into various circuits. So, let us see what we are learning in this particular module.

To start with we will start with filters.

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So, when you talk about filters, filters are what they are circuits. That are capable of passing certain signal and blocking another. So, what do you mean by passing certain signals? The band of frequencies that you want to pass; we can design a circuit that will allow to pass those band of frequencies our desired band of frequencies, while blocking the undesired signals, right; so this property of the filters is called frequency selectivity. Now if you divide the filters, filters can be active filters and filter can be passive filters. So, what do you mean by passive filters? Passive filters are filters that use R C that is

resistor and capacitor resistor and inductor or all combinations of resistor capacitor and inductor all 3 combinations; while active components or active filters are the one that employs op-amp in addition to resistors and capacitors right.

Now what are the advantages of active filters over passive filters? So, active filters can design or can be designed to provide gain and hence no attenuation as in case of passive filters. We have seen that in the earlier course. Same way there is no loading problem and active filters are cost effective. What are the applications? Applications are it is used in communication and signal processing circuits. they are also employed in a wide range of applications, such as entertainment and medical electronics. Of course, there are lot of applications that is why we have written, etcetera.

Now, 4 basic categories of active filters: low pass, high pass, band pass and band reject, right. And further, we can also classify active filters into digital filters and analog filters. So, digital filters are implemented using a digital computer or special purpose digital hardware; while analog filters can be classified as either active or passive implemented with R L and C as well as op-amps.
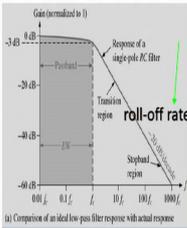
So, we have then seen each of those filters which are low pass, high pass, band pass band reject. So, let us see once again low pass filter.

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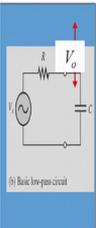What are low pass filter? Low pass filter allows for easy passage of low frequency signals from source to load and difficult passage of high frequency signals. So, now, if you see the circuit actual response should be ideal response, if you see at the right side and actual response is on the left side. So, when you see ideal response right this is like a brick like a brick wall, where actual response is little bit different, then ideal response and here one new definition comes into picture is called roll off rate; roll off rate, right.

So, low pass filter is a filter that passes frequency from 0 hertz to critical frequency what is the critical frequency; frequency at which you want to stop the or frequency until which you want to pass the signal in case of low pass filter. So, f c here is the frequency cutoff frequency or critical frequency above which we do not want the signals to pass. And that is why you see that after f c in the left side graph actual response plot, you see that after the f c the signal starts attenuating right, ok. What other bands, we can see a pass band, right, then here we see, this is your bandwidth this is your transition region this is your response of single pole RC filter.

Single pole is when there is one r and one c here you can see 1 R and 1 C, right. So, roll off rate roll off rate is about minus 20 dB per decade and then there is a stop band, right. So, the y axis y axis is gain x axis is your frequency, right.

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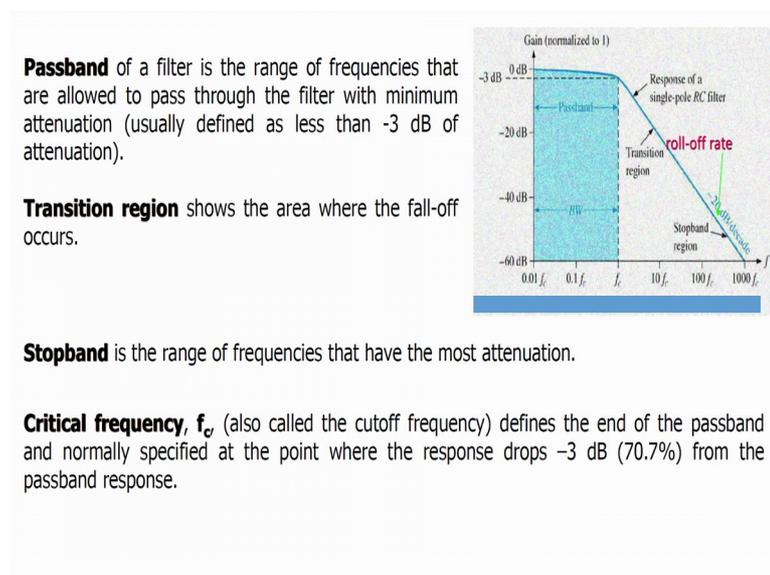
Now, passband; what is passband; passband? This passband is the range of frequencies that are allowed to pass through the filter with minimum attenuation while transition

region you can see here transition region is shows a area where they fall of a curves now. So, from here after the passband is over the fall off starts the attenuation starts and this is your transition region, then there is a stop band right is the range of frequency that have most attenuation. Finally, there is a critical frequency f c, right also called cutoff frequency as we have as I have just described in the last slide and it defines the end of passband. And normally specified the point where the response drops to minus 3 dB or 70.7 percent from the passband response, right.

So, that is your critical frequency f c ok.

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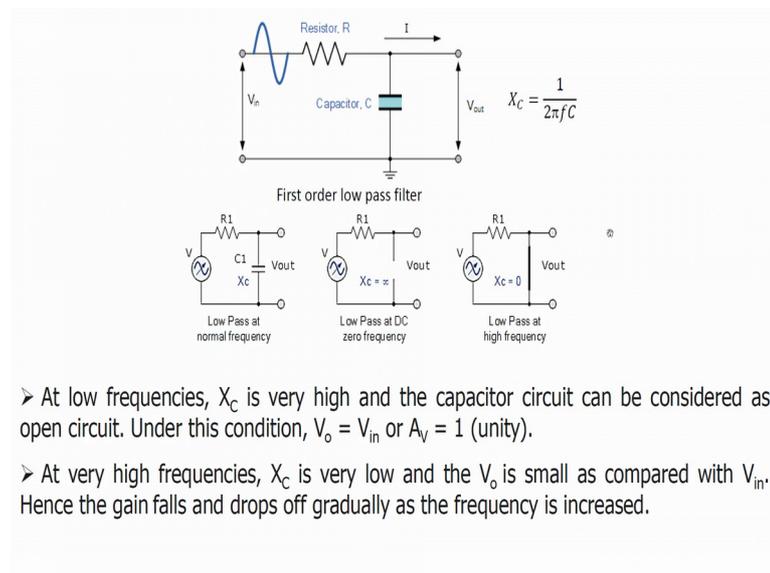
Now, when you want to design a low pass filter first order low pass filter resistor comes at the starting and then there is a capacitor. So, signal pass through the resistor and capacitor in this particular fashion. So, if I have X C; X C is nothing but 1 by 2 pi f c, right. So, in a low pass at normal frequency what will happen at low frequency your X C right X C would be very high and the capacitor circuit can be consider as an open circuit. So, here if you see in the center right this one in this case if your frequency is low your X C can be considered as open circuit and all the frequency, this is low frequency can pass and we can we can get at the output.

If the frequency is high if frequency is high right, then X C would be considered very low X C would be low, right, if f is high X C would be low if X C is low, then V o is small as compared to V in. Hence the gain and drop falls gradually as the frequency is

increased. This how the this is how the R C filter works, right a lower frequency the circuit allows the signals to pass at higher frequency or the frequency above cutoff frequency the circuit will stop the signal to pass through it.

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> The **bandwidth** of an **ideal** low-pass filter is equal to f_c:

$$BW = f_c$$

> The critical frequency of a low-pass RC filter occurs when **X_C = R** and can be calculated using the formula below:

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

So, the bandwidth of an ideal low pass filter bandwidth of an ideal low pass filter f c right is equal to bandwidth is equal to f c whatever critical frequency we have or cutoff frequency. We have that is equal to your bandwidth the critical frequency of a low pass filter occurs when X C equals to R, right and it can be calculated using f c equals to 1 by 2 pi RC.

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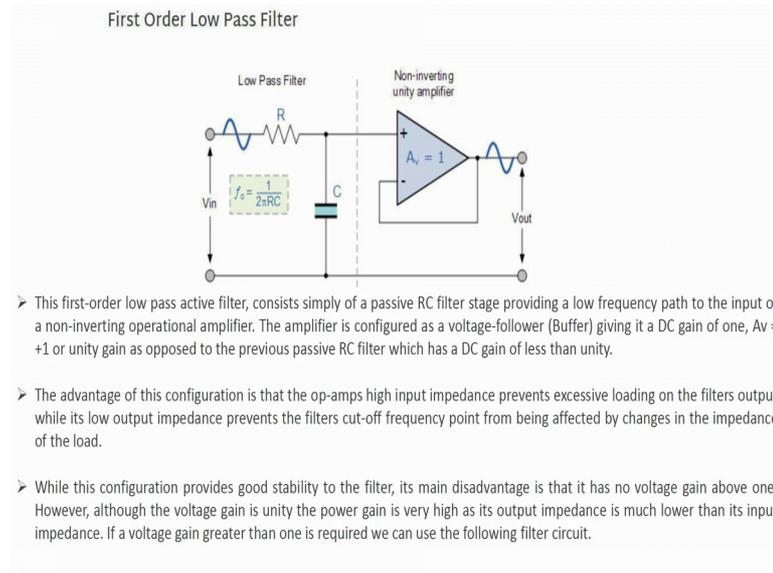
So, first order low pass filter if I use active filter active filter is when you use amplifier you use op-amp, right. Here we are using op-amp and as you can see op-amp is just in a non-inverting non inverting configuration and is a unity amplifier unity amplifier, right. So, the first order low pass active filter consists of simple passive R C filter and a low frequency pass to the input of non inverting operation amplifier. Amplifier is configured as voltage follower unity gain or what you call non inverting unity gain amplifier.

Right, where is DC gain is equal to 1 right DC gain is equal to 1 the advantage of this particular configuration is that the op-amp has high input impedance, we have seen op-amp input impedance is extremely high. So, this prevents excessive loading of filters output while its low output impedance prevents the filter cutoff frequency point from being affected by the changes in the frequency of the load, right. This advantage of using the operational amplifier as unity gain amplifier or voltage follower.

We are refreshing again what we have been learning in the last course and people who have not registered for my last course they may see the videos, and come up with the understanding about this particular course. So, for those people who have not attended my last course, we are repeating kind of summarizing the concepts in it just two modules, alright.

Active Low Pass Filter with Amplification

> The frequency response of the circuit will be the same as that for the passive RC filter, except that the amplitude of the output is increased by the pass band gain, A_F of the amplifier. For a non-inverting amplifier circuit, the magnitude of the voltage gain for the filter is given as a function of the feedback resistor (R_2) divided by its corresponding input resistor (R_1) value and is given as:

$$DC\ gain = \left(1 + \frac{R_2}{R_1}\right)$$

Therefore, the gain of an active low pass filter as a function of frequency will be:

Gain of a first-order low pass filter

•Where:
• A_F = the pass band gain of the filter, (1 + R2/R1)
• f = the frequency of the input signal in Hertz, (Hz)
• fc = the cut-off frequency in Hertz, (Hz)

$$Voltage\ Gain,\ (Av) = \frac{Vout}{Vin} = \frac{A_F}{\sqrt{1 + \left(\frac{f}{fc}\right)^2}}$$

So, if I use amplification if I use amplifier here, right. So, the signal is applied to the non inverting configuration non inverting terminal of the op amp. So, what is the gain? Gain is DC gain is 1 plus R 2 by R 1 1 plus R 2 by R 1, right and my voltage gain is nothing, but V out by V in which is A F by under root of 1 plus f c by f c; f by f c whole square, right, this it is easy to know. So, what happens here what happens here for non-inverting amplifier circuit the magnitude of voltage gain right is given as a function of the feedback resistor by the input resistor.

So, you have R C filter, here at the input stage and then you have a non inverting amplifier. So, here you can amplify the gain you see the gain or signal can be amplified with the whatever gain you have and you can set the gain with the help of R 2 and 1 with the help of R 2 and 1, right.

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Thus, the operation of a low pass active filter can be verified from the frequency gain equation above as:

• 1. At very low frequencies, $f < fc$

$$\frac{Vout}{Vin} \cong A_F$$

• 2. At the cut-off frequency, $f = fc$

$$\frac{Vout}{Vin} = \frac{A_F}{\sqrt{2}} = 0.707\, A_F$$

• 3. At very high frequencies, $f > fc$

$$\frac{Vout}{Vin} < A_F$$

➤ Thus, the **Active Low Pass Filter** has a constant gain A_F from 0Hz to the high frequency cut-off point, f_C. At f_C the gain is $0.707A_F$, and after f_C it decreases at a constant rate as the frequency increases. That is, when the frequency is increased tenfold (one decade), the voltage gain is divided by 10.

➤ In other words, the gain decreases 20dB (= 20log 10) each time the frequency is increased by 10. When dealing with filter circuits the magnitude of the pass band gain of the circuit is generally expressed in *decibels* or *dB* as a function of the voltage gain, and this is defined as:

Magnitude of Voltage Gain in (dB) $\quad Av(dB) = 20log_{10}\left(\frac{Vout}{Vin}\right) \quad \therefore -3dB = 20log_{10}\left(0.707\frac{Vout}{Vin}\right)$

So, that is the advantage for this configuration that is low pass active filter. We can see that at low frequencies f is less than f c and V out by V in is given by A F at the cut off frequency where f equals to f c V out by V in is given by 0.707 A F while at higher frequency where f is greater than f c V out by V in is less than A F.

This will be kind of opposite to the high pass filter and we will see that right and the magnitude of voltage gain. Here if you are using dB, then 20 log 10 V out by V in that is minus 3 dB equals to 20 log 10.707 V out by V in, right.

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Second-order Low Pass Active Filter

As with the passive filter, a first-order low-pass active filter can be converted into a second-order low pass filter simply by using an additional RC network in the input path. The frequency response of the second-order low pass filter is identical to that of the first-order type except that the stop band roll-off will be twice the first-order filters at 40dB/decade (12dB/octave). Therefore, the design steps required of the second-order active low pass filter are the same.

Cascading Voltage Gain

When cascading together filter circuits to form higher-order filters, the overall gain of the filter is equal to the product of each stage. For example, the gain of one stage may be 10 and the gain of the second stage may be 32 and the gain of a third stage may be 100. Then the overall gain will be 32,000, (10 x 32 x 100) as shown below.
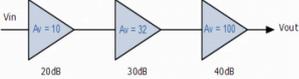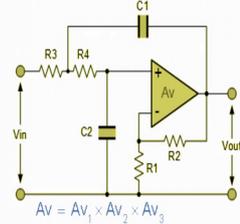
$$Av = 1 + \frac{R_2}{R_1}$$

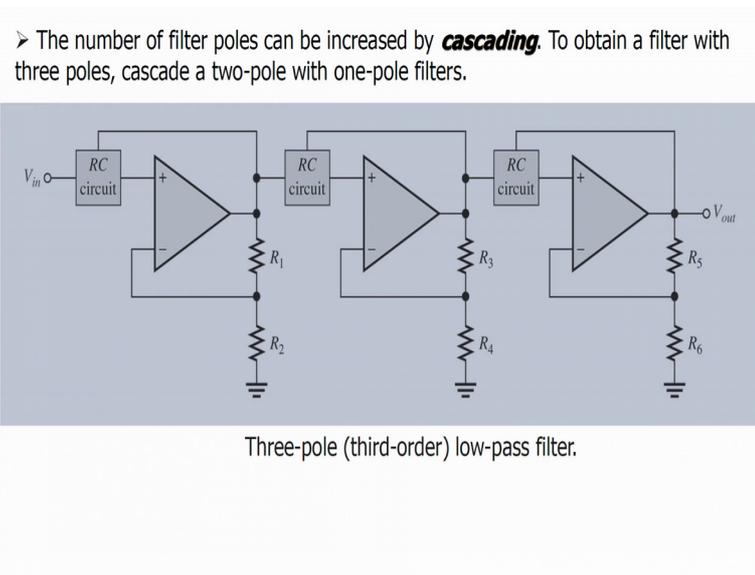$$f_C = \frac{1}{2\pi \sqrt{R_3 R_4 C_1 C_2}}$$

$$Av = Av_1 \times Av_2 \times Av_3$$

$$Av = 10 \times 32 \times 100 = 32,000$$

$$Av(dB) = 20log_{10}(32,000)$$

$$Av(dB) = 90dB$$

$$90dB = 20dB + 30dB + 40dB$$

If I talk about second order low pass signal I have 2 R and 2 C right, 2 R and 2 C. Here, the gain again A v is 1 plus R 2 by R 1 which you can see here while f c would be 1 by 2 pi under root of R 3, R 4, C 1 and C 2, right.

In this case, if I R 3 equals to R 4 and C 1 equals to C 2, I will have f c equals to 1 by 2 pi under root of R square c square. So, it will be it will be nothing, but f c equals to 1 by 2 pi R C. So, the cascading from single pole to two pole filter, we can we can make it by cascading the R and C.

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> The number of filter poles can be increased by **cascading**. To obtain a filter with three poles, cascade a two-pole with one-pole filters.

Three-pole (third-order) low-pass filter.

So, if you keep on cascading the number of filter poles can be increased by cascading to obtain a filter with 3 poles cascade a two pole with one pole filters and so on. So, this is a 3 pole filter, low pass filter, one pole will give you 1 pole will give you minus 20 dB per decade, right. So, 3 pole will give you about minus 60 dB per decade.

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High-Pass Filter Response

➤ A **high-pass filter** is a filter that significantly attenuates or rejects all frequencies **below** f_c and passes all frequencies **above** f_c.

➤ The passband of a high-pass filter is all frequencies above the critical frequency.

➤ The critical frequency of a high-pass RC filter occurs when

➤ $X_C = R$ and can be calculated using the formula:

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

➤ Ideally, the response rises abruptly at the critical frequency, f_L

Now, if I talk about high pass filter high pass filter what is a change high pass filter, we have capacitor at the input and then it goes to the resistors. So, signal goes to capacitor and a resistor right. In this one the high pass filter will significantly attenuate or reject all frequencies below f c correct and passes all frequency above f c. So, here the plot would be that here there will be attenuation at the input and then as the frequency which is close to f c and above f c.

It will start the signal can pass through the high pass filter right, this is ideal response for high pass filter and again formula remain same f c equals to 1 by 2 pi RC.

So, if I want to design first order high pass filter right with operational amplifier and operational amplifier is a unity gain amplifier. Then here I can again design a high pass active filter with the non inverting unity gain amplifier non-inverting, because your signals are applied to the non inverting terminal unity gain amplifier, we know with the circuit, right. So, this is how we can create a non-inverting, sorry, this is how we can create high pass active filter.

Now, if I want to use amplification factor application factor into the into the high pass filter then what I will have the op-amp in a amplifier configuration as an amplifier. So, here you can see the op-amp is used as an non inverting amplifier. So, here I can change here I can change the gain or amplify the signal based on the gain that I can I can adjust right. So, whatever I want I can adjust the gain and how can I do that by changing the value of R 2 and R 1,because the gain for non inverting amplifier is 1 plus R 2 by R 1 the voltage gain here. In this case would be A F into 1 f by f c divided by 1 plus f by f c whole square, but what is A F a f is your pass band gain f is the frequency of input signal and f c is the cut off frequency in hertz or critical frequency.

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Just like the low pass filter, the operation of a high pass active filter can be verified from the frequency gain equation above as:

1. At very low frequencies, $f < fc$
$$\frac{Vout}{Vin} < A_F$$

2. At the cut-off frequency, $f = fc$
$$\frac{Vout}{Vin} = \frac{A_F}{\sqrt{2}} = 0.707\,A_F$$

3. At very high frequencies, $f > fc$
$$\frac{Vout}{Vin} \cong A_F$$

Then, the **Active High Pass Filter** has a gain A_F that increases from 0Hz to the low frequency cut-off point, f_c at 20dB/decade as the frequency increases. At f_c the gain is $0.707A_F$, and after f_c all frequencies are pass band frequencies so the filter has a constant gain A_F with the highest frequency being determined by the closed loop bandwidth of the op-amp.

So, here if you see in case of low pass filter V out by V in was equal to A F and V out by V in case of high frequency was less than A F right here is opposite. So, here V out by V in less than A F V out by V in for f equals to f c is 0.07 and f greater than f c equals to A F. So, one more thing is that the active filter the active high pass filter has a gain A F that increases from 0 hertz to low cut low cutoff frequency right low frequency cut off point and then f c at 20 dB per decade as the frequency increases at f c equals to f equals to f c the gain is nothing, but 0.707 of A F. And then when f is greater than f c their gain would be nothing, but equals to AF.

So, bandpass filter when you talk about band pass filter it has a high pass filter amplification and a low pass filter. Now you want to pass a certain band right you have seen a low pass filter that will pass the low frequency band high pass filter it will pass the high frequency signal. And then we want to fabricate or design a band pass filter it should be a combination of high pass and low pass filter with amplification if it is an active filter.

So, the principal characteristics of band pass filter or any filter for that matter is its ability to pass frequencies relatively unattenuated over a sp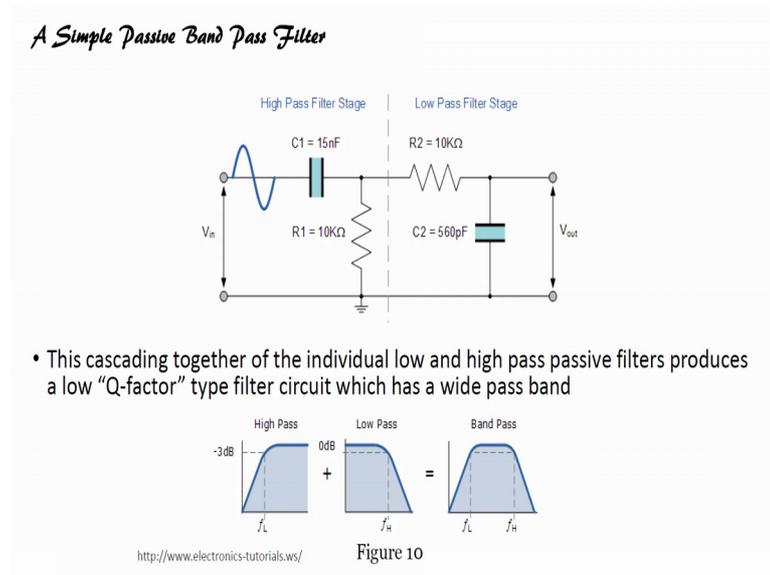ecified band or a spread of frequencies called pass band right. The cut off frequency or corner frequency of low pass filter is higher than the cut off frequency of the high pass filter, right. And difference between the frequencies at minus 3 dB point will determine the bandwidth of the band pass filter.
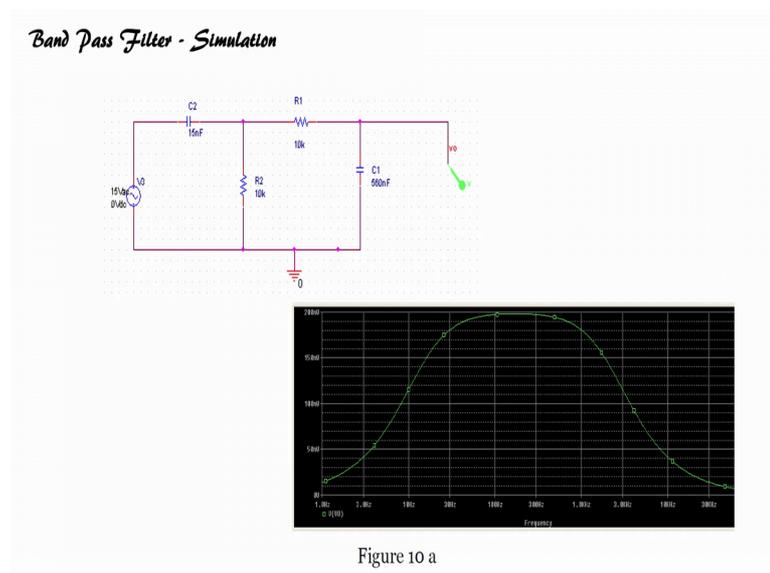
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Figure 10

So, this is very easy to understand low pass filter and high pass filter. So, now, if you see here if you want to fabricate or if you want to design a circuit, if you want to design a circuit right which is band pass filter and that too is passive. Then you can use a combination of high pass filter and low pass filter you see the signal from high pass filter signal from low pass filter. So, if you combine both you will get a band pass filter.

So, cascading together the individual low pass and high pass filter produces a low Q factor type filter which has a wide pass band, it is just a simulation.
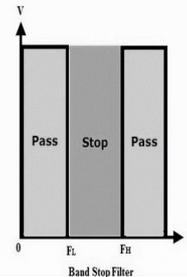
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Figure 10 a

If you want to work on the simulation you can use Simulink, right.

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Then band reject filter. So, as the name suggest it will reject a particular band of frequencies right. So, the figure shows which figure? Figure here it shows right this stop band pass band and pass band. So, this shows the ideal frequency response of band stop filter, this is a band stop filter is also called band reject filter right you can see here. This is the band stop or band reject filter or it is also called notch filter notch filter, alright.

So, do not get confused if it is band stop or band reject or notch everything is same. So, band stop filter is formed by combination of low pass and high pass filters with a parallel connection instead of cascading connection right. In the case of band pass filter we were cascading low pass and high pass in case of band reject filter we are again cascading, but in a parallel configuration the name itself suggests that it will stop a particular band of frequencies right. Since, it eliminates frequency also called band elimination filter or band reject filter or notch filter right.

So, we know that unlike high pass and low pass band pass and band stop filters have two cut off frequencies you can see here F L and F H right; two cut off frequency, it will pass above and below a particular range of frequency whose cut off frequencies are predetermined depending on the value of components used in the circuit design right by changing the value of R and C we can change the value of F L and F H, right or we can we can pass certain band of frequency we can reject certain band of frequency, alright.

Now, any frequency between these two frequencies are attenuated it has two bands and one stop band as you can see from the figure.

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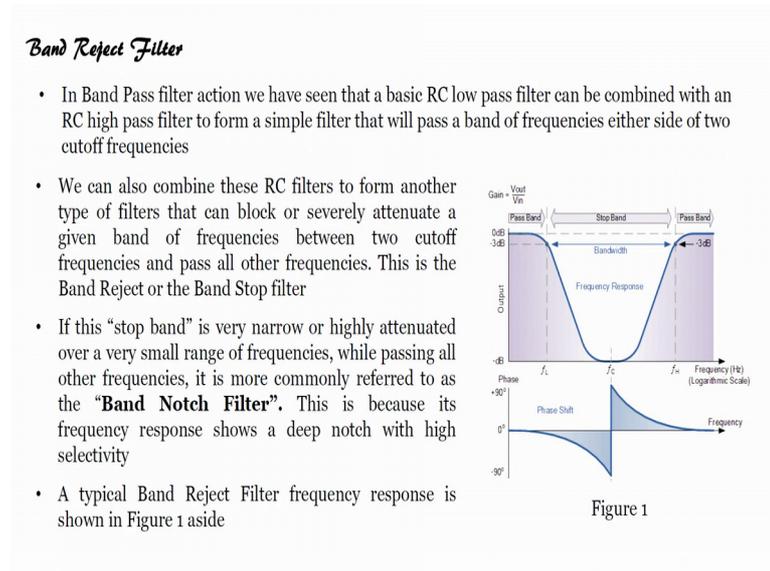
So, band reject filter right it has a R C low pass filter combined with a R C high pass filter. Now we can also combine these R C filters to form another type of filters that can block or severely attenuate a given band of frequencies between two cutoff frequencies and pass all the other frequencies, right.

So, by tweaking the values of R and C we can just allow a certain band of frequency to pass or even a single frequency to just block, right. So, only single frequency to attenuate or band of frequency to attenuate. So, if it is just a small frequency or very low range of frequency or very small range of frequency while it can pass all the other frequency, then it is called notch filter right.

So, if this stop band is very narrow you can see this is stop band is here right and if the stop band is extremely narrow. That means, only a certain frequencies are stopped only small range of frequencies are stopped. Then it will work as a notch filter right because the frequencies response when you see this response it will show like a notch. So, deep notch with high selectivity a typical band reject filter frequency response is shown here you can see here.

Figure 2
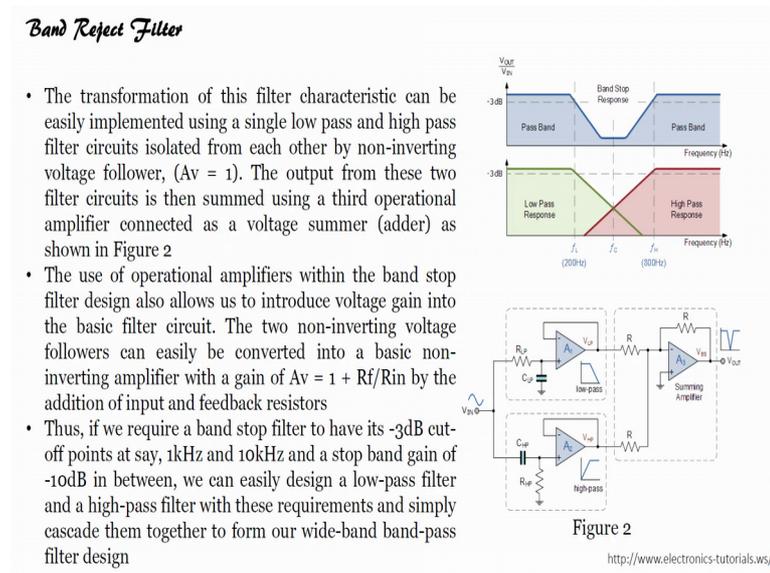
Now, if I want to design the band reject filter then this is the circuit that is used for designing is a low pass filter high pass filter in a parallel configuration and then there is a summing amplifier, right. So, how does it works is the transformation of this filter characteristics can be easily implemented using a low pass and high pass filter isolated from each other by non inverting voltage amplifier voltage follower right see voltage follower is here, it is here right. The use of op-amp with the band stop filter design also allows us to introduce voltage gain into the basic filter then the two non inverting voltage followers can easily converted into basic non inverting amplifier with a gain of A v equals to 1 plus R f by R in.

This non inverting voltage follower here or unit gain amplifier can be converted to a amplifier by changing the R f by R in, right. We can connect R f in R in. Thus if we require a band stop filter to have its minus 3 dB cut off say at 1 kilo hertz and 10 kilo hertz and a stop band gain of minus 10 dB, in between we can easily design a low pass and high pass filter with these basic requirements, right.

Oscillators - Introduction

- Oscillator is a circuit that works on the principle of positive feedback
- The circuit basically works as a generator, generating the output signal which oscillates and generates an output signal which oscillates with a fixed amplitude and frequency. It does not require an input signal
- In short, an oscillator is an amplifier, which uses a positive feedback, and without any external input signal, generates an output waveform at a desired frequency
- Consider a non-inverting amplifier with voltage gain A, as shown in figure below
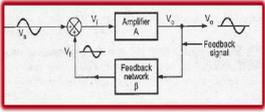
Source: Electronic Devices and Circuits II by A.P. Godse et.al

- There is a feedback network with feedback factor β. The feedback is said to be positive whenever the part of the output that is fed back into the input is in phase with the original input signal applied to the amplifier
- Assume that a sinusoidal input signal V_s is applied at the input and since the amplifier is non-inverting the output signal is in phase with the input. A part of this output signal is fed back into the input of the amplifier through the feedback network shown

Then we have seen oscillators. So, when we talk about oscillators we have seen Barkhausen criteria in what does Barkhausen criteria says Barkhausen criteria says that- let us see one what does it say one is that the phase the output signal that is sent back to the input the feedback signal right should be in phase with respect to input signal right. So, there is a phase shift the phase shift right of the output signal compared to input signal or the feedback signal compared to input signal should be 0 or 360 degree phase second; it says that the mod of a into beta should be greater than equal to 11; two criterias first is that the oscillators when you want to design a operation amplifier in as an oscillator then you have to apply a positive feedback, right.

And as you can see here right that there is a signal V s, there is a input signal V i this input signal V i depends upon V s and the feedback signal V f there is a amplifier which amplify the signal and the output part of the output signal is feedback to the input. And what I said is that the part of the output signal that is fed back to the input should have same phase with respect to the input signal, alright.

So, let us see you can see here that only a part of output signal is feedback, if I increase the beta; if I increase the feedback what I have I have A F extremely high and in case of 0.05 for a gain of 20, I see that the feedback gain is infinite, right. Thus we had to follow a Barkhausen criteria.

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And you have to understand that only a part of the signal is fed back to the input. We have seen this thing in earlier course just quickly repeating in it here. So, that you understand because you will be using this for actual implementation of circuits alright.

So, you can see that the feedback must be positive voltage derived from output using feedback network must be in phase with V I, right, thus if I use an inverting amplifier as you can see here right you can see the signal here and signal here is 180 degree phase shift compared to the input signal correct.

So, if I want to feedback if I want to fed this particular output voltage through the feedback network back to the input than the feedback network should introduce another 180 degree phase shift right, because you see 0, 180, 180 is fed here, if I want to make it 0 or 360, another 180 degree phase shift, I will put I will implement in the feedback network, right. So, feedback should be having 360 degree phase shift or 0 degree phase shift, right.

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Barkhausen Criterion

Consider a fictitious voltage V_i applied at the input of the amplifier. Hence we get,

$$V_o = Av_i$$

The feedback factor ß decides the feedback to be given to input,

$$V_f = ßV_o$$

On substitution,

$$V_f = Aß V_i$$

For the oscillator, we want that feedback should drive the amplifier and hence V_f must act as V_i

Therefore, we can write that V_f is sufficient to act as V_i when,

$$|Aß| = 1$$

And the phase of V_f is same as V_i i.e. feedback network should introduce 180° phase shift in addition to 180° phase shift introduced by inverting amplifier. This ensures positive feedback. So total phase shift around a loop is 360°

In this condition, V_f drives the circuit and without external input, circuit works as an oscillator

The two conditions discussed above, required to work the circuit as an oscillator are called **Barkhausen Criterion for oscillation**

Another point that we discuss is mod of a beta should be equal to 1 or greater than equal to 1 because initially, you want to start the oscillator, then mod of a beta should be greater than 1 and then you can make it equal to 1, alright.

So, this is very important criteria you have to understand and you have to design a circuit that can follow these criteria, otherwise the oscillation should not happen, alright.

Barkhausen Criterion

The Barkhausen Criterion states that:

1. The total phase shift around a loop, as the signal proceeds from input through amplifier, feedback network back to input again, completing a loop, is precisely 0° or 360°

2. The magnitude of the product of the open loop gain of the amplifier (A) and the feedback factor ß is unity i.e $|A \, ß| = 1$

Satisfying these conditions, the circuit works as an oscillator producing sustained oscillations of constant frequency and amplitude

In reality, no input signal is needed to start the oscillations. In practice, Aß is made greater that 1 to start the oscillations and then circuit adjusts itself to get Aß = 1, finally resulting into self-sustained oscillations

We will now see the effect of the product Aß on the nature of oscillations

So, what are these states; I we have already discussed first is the total phase shift around a loop as the signal proceeds from input through amplifier feedback network back to the input completing a loop is precisely 0 or 360 degree. And second is the magnitude of product of open loop gain and feedback factor beta should be 1, alright.

Effect of Magnitude of Aß

$|Aß| > 1$

When the total phase shift around a loop is 0° or 360° and $|Aß| > 1$, then the output oscillates but the oscillations are of growing type. The amplitude of oscillations goes on increasing as shown
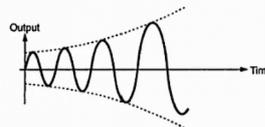
$|Aß| = 1$

As stated by Barkhausen criterion, When the total phase shift around a loop is 0° or 360° ensuring positive feedback and $|Aß| = 1$ then the oscillations are with constant frequency and amplitude called sustained oscillations. Such oscillations are shown below
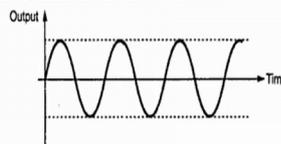
Source: Electronic Devices and Circuits II by A.P. Godse et.al

So, what happens if it is greater than 11, what happens if it is equal to 1 and what happens if it is less than 1 what a into beta a is gain beta is feedback. So, when the total phase shift around a loop is 0 degree or 360 degree and a beta is greater than 1 then the

output oscillates, but the oscillations are growing type you can see the oscillations are growing in nature, right.

But when you talk about a beta equal to 1 then you will see that the total phase around phase shift is 0 or 360 degree. This is still we have to maintain this condition, but if I have a bit equal to 1 then the oscillations are with constant frequency and they are called sustained oscillations are sustaining, right.
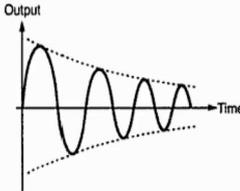
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But if I have a beta less than 1 a beta less than 1 then what will happen then the total phase shift around a loop is 0 degree or 360 degree. See, every time we have to write the sentence phase shift should be 0 degree or 360 degree, but if I have a beta less than 1 then I will see a damping type of oscillation damping type of oscillation right or we can say decaying the oscillations are decaying.

Now the question comes that how oscillators starts because is a beta right is kept at higher than unity and then the circuit adjusts itself to a beta equals to 1 to result sustained oscillations, it is ok. But the obvious question here is that if no input signal is required here we say that right in oscillator. There is no input signal required if input signal is not required, then how the oscillator starts and from where does the starting voltage come.

So, we can see that the starting voltage comes because of the free electron because every resistance has a free electrons under the influence of a normal room temperature this electrons move randomly into various directions, right.

Such electrons will form a voltage called noise voltage and this noise voltage are amplified hence to amplify the such a small noise voltages we need to keep a beta or mod of a beta is slightly greater than 1 or slightly greater than unity right, then once we have this oscillations, we can keep a beta equal to 1 and phase shift 360 degree to have a sustained oscillations right this is a starting of the oscillator.
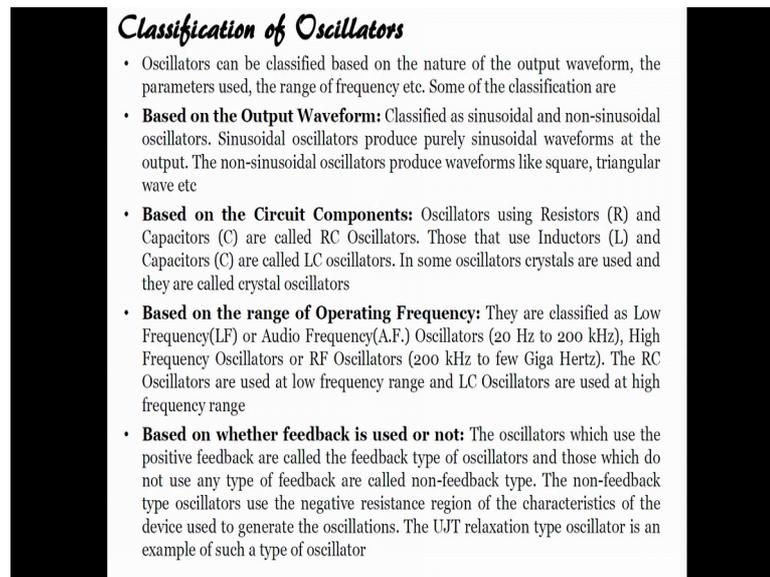
Classification of Oscillators

- Oscillators can be classified based on the nature of the output waveform, the parameters used, the range of frequency etc. Some of the classification are
- **Based on the Output Waveform:** Classified as sinusoidal and non-sinusoidal oscillators. Sinusoidal oscillators produce purely sinusoidal waveforms at the output. The non-sinusoidal oscillators produce waveforms like square, triangular wave etc
- **Based on the Circuit Components:** Oscillators using Resistors (R) and Capacitors (C) are called RC Oscillators. Those that use Inductors (L) and Capacitors (C) are called LC oscillators. In some oscillators crystals are used and they are called crystal oscillators
- **Based on the range of Operating Frequency:** They are classified as Low Frequency(LF) or Audio Frequency(A.F.) Oscillators (20 Hz to 200 kHz), High Frequency Oscillators or RF Oscillators (200 kHz to few Giga Hertz). The RC Oscillators are used at low frequency range and LC Oscillators are used at high frequency range
- **Based on whether feedback is used or not:** The oscillators which use the positive feedback are called the feedback type of oscillators and those which do not use any type of feedback are called non-feedback type. The non-feedback type oscillators use the negative resistance region of the characteristics of the device used to generate the oscillations. The UJT relaxation type oscillator is an example of such a type of oscillator

Now, if you want to talk about oscillators are classified into several categories based on output waveform based on circuit components based on range of operating frequency. And based on whether feedback is used or not these are 4 different components the 4 different classification of the oscillators right and we have seen this. Let us see once again classified as sinusoidal non sinusoidal oscillators produces sinusoidal waveform non sinusoidal produces square wave triangle wave etcetera.

Same way; basic based on circuit components whether we are using R C l LC, right. So, if it R and C, it is a R C oscillator, if it is L and C, it is a different kind of oscillator, we will see different kind of oscillators depending on how many L, how many C, it can be Hartley oscillator, it can be colpitts oscillator, right. So, we have to understand that depending on the circuit components. We can also classify oscillator depending on waveform we can classify oscillators sinusoidal waveform or non sinusoidal waveform. Then, based on operating frequency where low operating frequency audio frequency or it is a higher frequency right. So, if it is R C oscillator, it uses low frequency, if LC oscillator it uses higher frequency alright based on frequency we can classify the oscillator.

Then we have based on whether feedback is used or not if feedback is used, it is called feedback type of oscillator feedback is not used non feedback type of oscillators right. So, we have different classification of oscillators the if you see the screen the non

feedback type of oscillators uses the negative resistance region of characteristics and this is a UJT relaxation type of oscillator. So, UJT relaxation type of oscillator is your non feedback type of oscillator.

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RC Phase Shift Oscillators

A RC Phase Shift Oscillator consists of and amplifier and feedback network made up of resistances and capacitances. The basic RC circuit is as shown below

Let $\qquad V_i = V_m \sin(\omega t)$

The impedance of circuit is, $\quad Z = R - jX_c$

Where, $\qquad X_c = \frac{1}{2\pi f c}, |Z| \angle \varphi$

where $\qquad \tan\varphi = -\frac{X_c}{R} = -\frac{1}{2\pi f c R}$

$$I = \frac{(V_m / \sqrt{2}) \angle 0^0}{|Z| \angle -\varphi} \quad \ldots\ldots(\frac{V_m}{\sqrt{2}} \text{ is r.m.s. value})$$
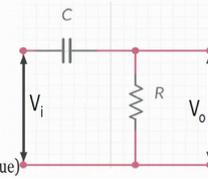
$$\therefore \quad I = |I| \angle + \varphi^0 \text{ A}$$

The positive phase angle $+ \varphi$, indicates the current leads applied voltage by angle ϕ

Now $\qquad\qquad\qquad\qquad V_o = \mathbf{IR}$

As output voltage V_o is drop across the resistance it is phase with current, hence the output voltage leads the input voltage by angle φ

In general, φ is called the phase angle of the circuit and depends on the values of R and C selected. When the value of C is very large as compared to R then the phase angle φ tends to 90^0 but in practice these values are so selected that the phase angle is 60^0

So, if we discussed phase shift oscillators or R C phase shift oscillators then it is in this particular configuration, you can see C and R. Now if I work on the circuit if I work on the circuit and I solve the circuit right what will I have let V i equals to V m sin omega t if this is the input right and impedance of circuit is Z equals to r minus j X C where x is nothing, but 1 by 2 pi f c into Z and the mod of Z should be less than your theta or phi.

So, if I solve this then what will I have; I will have I equals to this particular equation right mod of I angle plus phi degree into a the positive phase angle indicates the current leads applied voltage. You can see here there is a positive phase angle right and this positive phase angle what does it says it says that the current leads applied voltage by angle phi. And now V o equals to IR as output voltage V o is dropped across the resistance and it is in phase with the current hence the output voltage leads the input voltage by an angle phi right. Output voltage leads it by angle phi in general phi is called phase angle of the circuit and depends on the value of R and C. when the value of C is very large as compared to R. Then the phase tends to be 60 degree. So, for a single c and r is the value of C is higher than we can add a phase shift of 90 degree.

But in practice these values are so selected that the phase shift is 60 degree, right. So, if your C is high right compared to R, then we have phase angle of or phase shift of 90 degree, but we have to adjust R and C, such that we have a phase shift of 60 degree why because this R C circuits are used in the feedback are used in the feedback. So, if I have a feedback. That means, that each R C producing 60 degree 3 R C will produced 180 degree, right.

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RC Phase Shift Oscillators

Now if amplifier used causes a phase shift of 180⁰, then feedback network should create a phase shift of 180⁰, to satisfy the Baurkhausen Criterion. Hence in a phase shift oscillators three sections of RC circuits are connected in cascade, each introducing a shift of 60⁰, thus introducing a total phase shift of 180⁰, due to feedback network. Thus RC phase shift of network consisting of three RC sections, used in phase shift oscillators is shown below

The network is also called the ladder network. All the resistance values and all the capacitance values are the same, so that for a particular frequency each section of R and C produces a shift of 60⁰

So, if I have a phase shift of 180 degree at the output compared to the input signal then if I feed this output back to this R C circuit, I have another 180 degree phase shift. So, my total phase shift is 360 degree or 0 degree which will satisfy my Barkhausen criteria; right that is the use of RC.

So, you can see here if the signal is applied to the inverting terminal right; I have a 180 degree phase shift with output I am feeding it to the R C feedback circuit. And I will have another 180 degree. So, 180 plus 180 will be 360 degree and some part of the signal is feedback to the input to sustain the oscillation, right.

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We have seen this we have also derived the equation for the gain and equation for the beta, right.

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Phase Shift Oscillator using Op-Amp contd..

As we know that,

$$V_o = Vf = I_3 R = \frac{V_i R^2 s^3 C^3 * R}{1 + 5sRC + 6s^2C^2R^2 + s^3C^3R^3}$$

$$\therefore \beta = \frac{V_o}{V_i} = \frac{R^3 s^3 C^3}{1 + 5sRC + 6s^2C^2R^2 + s^3C^3R^3}$$

Replacing s by $j\omega$ and dividing numerator and denominator by $-j\omega^3 R^3 C^3$ and replacing $1/\omega RC$ by α we get,

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

To have a phase shift of 180°, the imaginary part in the denominator must be zero.

$$\therefore \alpha(6 - \alpha^2) = 0$$

$$\therefore \alpha^2 = 6$$

$$\therefore \alpha = \sqrt{6}$$

$$\therefore 1/_{\omega RC} = \sqrt{6}$$

$$\therefore \omega = 1/_{RC\sqrt{6}}$$

And what we found is that beta in case of R C phase shift oscillator should be 1 by 29 while gain should be greater than 29, right.

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Phase Shift Oscillator using Op-Amp contd..

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

This is the frequency with which it oscillates

At this frequency,

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

Negative sign indicates phase shift of 180°

The negative sign indicates phase shift of 180°, therefore

$$|\beta| = \frac{1}{29}$$

Now to have oscillations, $|A\beta| \geq 1$

As a result we get

$$|A| \geq \frac{1}{|\beta|} \geq \frac{1}{\frac{1}{29}}$$

$$\Rightarrow |A| \geq 29$$

To have oscillations, thus the gain of the op-amp must be equal to or greater than 29, which can be adjusted by changing the feedback and input resistances
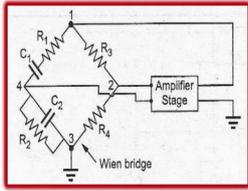
You please, see my earlier lectures right in another course the course name is integrated circuits MOSFETs op-amps and their applications. If you see YouTube videos you will find how we are driving these equations, such as gain equation such as feedback network or feedback back component beta.

So, for R C phase shift oscillator we have beta equals to one by 29 while gain equals to 29. So, if you come back to the screen what we see if we have if we have beta equals to 1 by 29 and gain is greater than equal to 1 by 29 greater than equal to 29. Then mod of a into beta would be greater than equal to 1 and I have phase shift of 0 degree or 360 degree. Thus I am satisfying the Barkhausen criteria thus to have the oscillations the gain of op-amp must be equal to or greater than 29 which can be adjusted by changing the feedback and input resistance.

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Now, let us see another oscillator which is Wien bridge oscillator alright and in Wien bridge oscillator there is no phase shift phase shift is not there. So, Wien bridge oscillator is an audio frequency sine wave oscillator of a high stability and simplicity it has 2 R C amplifier circuit connected in wheatstone you can see here R C here R C here right, 1 is in series another in parallel right the basic circuit shown in figure the output of the amplifier is amplified between terminal 1 and 3; 1 and 3 right the output amplifier is feedback between terminal one and terminal 3 right.

The amplifier input; input is applied between terminal two and four right the resistor values are adjusted, such that the input to the amplifier must not be 0 or non-vanishing right the R C network is responsible for determining the frequency of the oscillator R C network which is here and here determines the frequency of the oscillator. So, here how many R C oscillators here there are 2 R C oscillators right one series R 1 C 1 another

parallel R 2 C 2, right and these are called frequency sensitive arms frequency sensitive arms.

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Wien Bridge Oscillator using Op-amps contd..

- To understand the gain of the feedback network, let us consider the feedback network of the Wien bride oscillator as shown below
- This configuration is also called lead-lag network because it acts like a lead at very low frequencies and while at very high frequencies it acts as a lag network
- To calculate the gain of the network, consider series RC as Z_1 and parallel RC as Z_2
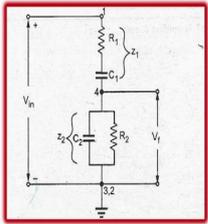
Where,

$$Z_1 = R_1 + \frac{1}{j\omega C_1} = \frac{1 + j\omega C_1 R_1}{j\omega C_1} = \frac{1 + sC_1 R_1}{sC_1}$$

$$Z_2 = R_2 || \frac{1}{j\omega C_2} = \frac{R_2}{1 + j\omega C_2 R_2} = \frac{R_2}{1 + sC_2 R_2}$$

- The simplified circuit is as shown a side which is a voltage divider

$$\therefore \beta = \frac{V_f}{V_{in}} = \frac{Z_2}{Z_2 + Z_1}$$

On substitution we get,

$$\beta = \frac{V_o}{V_i} = \frac{j\omega C_1 R_2}{(1 - \omega^2 R_1 R_2 C_1 C_2) + j\omega (R_1 C_1 + R_2 C_2 + C_1 R_2)}$$

Source: Electronic Devices and Circuits II by A.P. Godse et.al

So, if we consider just feedback then R 1 C 1 R 2 C 2 in this particular configuration I will say R 1 C 1 is Z one R 2 C 2 is Z 2 right and applying input between 1 and 3. So, V in our output is between 2 and 4 sorry 2 and 1. And let us see what is that 1 and 3 is the output feedback from the amplifier and between 2 and 4 between 2 and 4 my feedback is there, right.

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Wien Bridge Oscillator using Op-amps contd..

Rationalising the expression we get,

$$\beta = \frac{V_o}{V_i} = \frac{j\omega C_1 R_2[(1 - \omega^2 R_1 R_2 C_1 C_2) - j\omega (R_1 C_1 + R_2 C_2 + C_1 R_2)]}{(1 - \omega^2 R_1 R_2 C_1 C_2)^2 + \omega^2 (R_1 C_1 + R_2 C_2 + C_1 R_2)^2}$$

$$\beta = \frac{V_o}{V_i} = \frac{\omega^2 C_1 R_2 (R_1 C_1 + R_2 C_2 + C_1 R_2) + j\omega C_1 R_2 (1 - \omega^2 R_1 R_2 C_1 C_2)}{(1 - \omega^2 R_1 R_2 C_1 C_2)^2 + \omega^2 (R_1 C_1 + R_2 C_2 + C_1 R_2)^2}$$

To have zero phase shift of the feedback network, its imaginary part must be zero.

$$\therefore \omega (1 - \omega^2 R_1 R_2 C_1 C_2) = 0$$

$$\omega^2 = \frac{1}{R_1 R_2 C_1 C_2}$$

$$\therefore f = \frac{1}{2\pi \sqrt{R_1 R_2 C_1 C_2}}$$

Hence the frequency of the oscillator shows that the components (i.e. $R_1 R_2 C_1 C_2$) of the frequency sensitive arms are the deciding factors for the frequency

Let $R_1 = R_2 = R$ and $C_1 = C_2 = C$

$$\therefore f = \frac{1}{2\pi RC}$$

So, solving this particular equation solving this what we will get we get f equals to 1 by 2 pi R C the frequency formula for this oscillator is 1 by 2 pi R 1 under root of R 1 R 2 C 1 C 2. But, when R 1 equals to R 2 and C 1 equals to C 2 we have 1 by 2 pi under root of R square C square is nothing but 1 by 2 pi R C right.

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Wien Bridge Oscillator using Op-amps contd..

The gain of the feedback networks becomes,

$$\beta = \frac{V_o}{V_i} = \frac{\omega^2 CR(3RC) + j\omega CR(1 - \omega^2 R^2 C^2)}{(1 - \omega^2 R^2 C^2)^2 + \omega^2 (3RC)2}$$

On substitution of value of ω = 1/RC,

$$\beta = \frac{V_o}{V_i} = \frac{3}{+\frac{1}{(RC)2}(3RC)2} = \frac{3}{9}$$

The positive sign of β indicates that the phase shift by the feedback network is 0°

For sustained oscillations, $|A\beta| \geq 1$

$$|A| \geq \frac{1}{|\beta|} \geq \frac{1}{\frac{1}{3}}$$

$$\Rightarrow |A| \geq 3$$

This is the required gain of the amplifier stage, without any phase shift

So, my beta which is the feedback network here in this case if I solve further what is that beta is nothing but 1 by 3 right and what is my gain my gain should be greater than equal to 3, then only my a into beta would be the greater than equal to 1, because 1 by 3 into 3 is greater than equal to 1, right. So, this is required gain of the amplifier without any phase shift because there is no phase shift in case of Wien bridge oscillator, alright.

Wien Bridge Oscillator using Op-amps contd..

Let $R_1 \neq R_2$ and $C_1 \neq C_2$ then

$$f = \frac{1}{2\pi\sqrt{R_1 R_2 C_1 C_2}}$$

On substitution we get,

$$\beta = \frac{V_o}{V_i} = \frac{C_1 R_2}{(R_1 C_1 + R_2 C_2 + C_1 R_2)}$$

For sustained oscillations, $|A\beta| \geq 1$

$$|A| \geq \frac{1}{|\beta|} \geq \frac{(R_1 C_1 + R_2 C_2 + C_1 R_2)}{C_1 R_2}$$

The advantage of the Wien bridge oscillator is that by varying the two capacitor values simultaneously, different frequency ranges can be provided
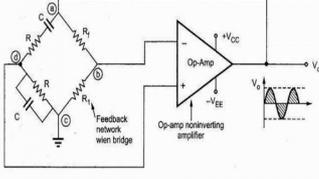
In case of Wien bridge oscillator, there is no phase shift but what if R 1 is not equal to R 2 and C 1 is not equal to C 2, then we cannot have f f equal to 1 by 2 pi R C, right in this case we will have if on substitution we will have C 1 R 2 V o by V i right beta is V o by V i. So, I have this formula and my a gain would be based on this particular equation alright this, but the advantage of Wien bridge oscillator right is that by varying the capacitor varying the capacitor simultaneously right by varying the two capacitor values simultaneously we can have different frequency ranges. This is the advantage of Wien bridge oscillator.

Second is there is no phase shift required in the feedback network right that is the second advantage of the Wien bridge oscillator right.

So, if the amplifier is you see the Wien bridge oscillator is used with an operational amplifier right, in this particular configuration right then what will happen the resistance and capacitors are the components of frequency sensitive arms. You can see here the resistance R f and R 1 are part of the feedback network right R f, and r R f and R 1 could be part of the feedback network for the amplifier, and the gain would be because signal is provided at the non inverting terminal gain would be 1 plus R f by R 1, right.

According to oscillations for Wien bridge oscillator a should be greater than equal to 3; that means, 1 plus R f by R 1 equal to greater than equal to 3 or R f by R 1 should be greater than equal to 2. So, we can adjust the value of R f R 1, such that my value comes greater than or equal to 2 right. So, to for the oscillation to or frequency calculated is f equals to 1 by 2 pi R C feedback is given to the non inverting terminal of the op-amp to ensure the 0 phase shift right frequency feedback is given to the non inverting terminal you can see here. So, that we have a 0 phase shift at the output right.

Wien Bridge Oscillator using Op-amps contd..

- If the amplifier circuit using transistor is replaced by the op-amp with the basic feedback network remains as the Wien bridge circuit, the oscillator is called Wien bridge oscillator using op-amp. The circuit is as shown in the Figure below
- The resistance R and capacitor C are the components of frequency sensitive arms of the bridge. The resistance R_f and R_1 are the part of the feedback path
- The gain of the op-amp is given by

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

- According to the oscillating conditions, $A \geq 3$

$$\Rightarrow 1 + \frac{R_f}{R_1} \geq 3 \Rightarrow \frac{R_f}{R_1} \geq 2$$

- Thus the ratio of R_f and R_1 should be greater than or equal to 2 to provide sufficient loop gain for the circuit to oscillate at the frequency calculated as,

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

- The feedback is given to the non-inverting terminal of the op-amp to ensure zero phase shift

Source: Electronic Devices and Circuits II by A.P. Godse et.al

So, this is about this about this particular module this about this particular module and we have seen how the oscillators can be designed as well as we have also seen how the filters can be designed right. In the next module which is the last module for the summary. So, that we understand how we can implement this things in the practical applications right. We will be looking at few other oscillators that are LC oscillators and we will see that how, we can change the inductors and capacitor for the value of inductor and value of capacitor to obtain the LC oscillators LC oscillators as we have discussed are used for high frequency applications and R C oscillators are used for low frequency applications, right.

So, let us see in the next class; how we can design LC oscillators. And that will be the last module for this particular lecture, right. So, I will see you next class. Just go through this and refresh your understanding about op-amps and its application, then we have to actually implement those op-amps for different applications.

I will see you in the next class. Till then, you take care. Bye.