Electronics

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Module No. # 07

Differential & Operational Amplifiers

Lecture No. # 40

Frequency Response of Intigrator

We were discussing the Frequency Response of an Intigrator.

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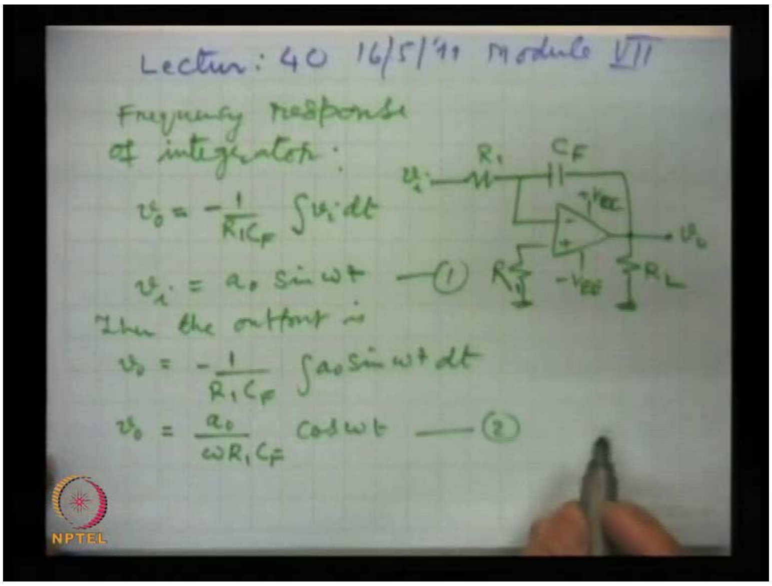

The basic integrating circuit, integrating amplifier using operation amplifier, we have already discussed and it was this, (No audio from 00:44 to 01:04) this is the input signal v i and resistance R 1, capacitor C F and of course, these power supply is there. This is load resistor across which, we take the output, and this is to reduce the offset effects, about offset effects we will definitely talk. So, this is around R 1 and we have seen that the output of this is minus 1 by R 1 C F integration v i d t, where v i is the input signal.

And if we take that input signal v i is expressed as, a 0 sin omega t, then the output is then the output of the integrator is v 0 minus 1 by R 1 C F and integration a 0 sin omega t d t, and this gives v 0 equal to a 0 omega R 1 C F cos omega t. This is we take as first and this is the second (Refer Slide Time: 02:56).

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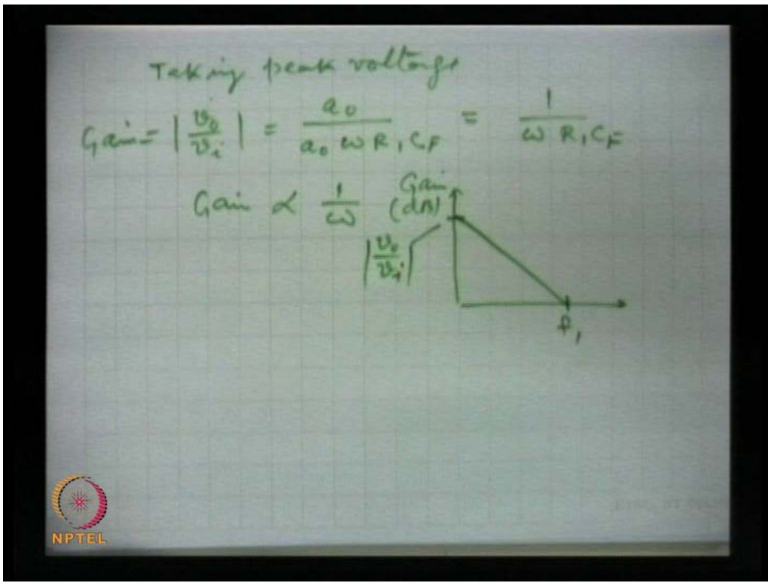

Then we can write the ratio of the two for the peak voltages, taking peak voltages both at the input and output, the ratio which is gain v 0 by v i its magnitude is, a 0 by a 0 omega R 1 C F and so, it is equal to 1 by omega R 1 C F and this is gain, gain of the intigrator.

So, what we are seeing that, the gain is frequency dependent. As the frequency is, this angular frequency is omega, which is 2 pi F. So, as the frequency F increases, the gain falls. So, gain is proportional proportional to 1 by omega, the frequency and the plot for this is, this is the frequency f 1 (Refer Slide Time: 04:26), at which the gain falls to 0, this is gain in d B, and this value this represents v 0 by v i this magnitude of this quantity.

Now, the gain is falling and it can be shown that, this fall is minus 20 d B per decade change in frequency. So, if frequency becomes 10 times, then gain becomes one-tenth that is the meaning and the gain falls. And here, the f 1 this frequency, this is frequency f 1 is given by 1 by 2 pi R 1 C F, this we can get from here actually, this is gain and when at the frequency at which gain falls to 0 is given by this (Refer Slide Time: 05:41). And so, this is this is the frequency response of the basic integrator, this integrator circuit.

And now we have discussed that, there is a problem with this circuit. And the problem arises, because of the offset, offset currents and voltages and the capacitor, gets charged and it is amplified and there is a output, even when there is no input; and these problems can be tackled by attaching a resistor along with the capacitors in shunt.

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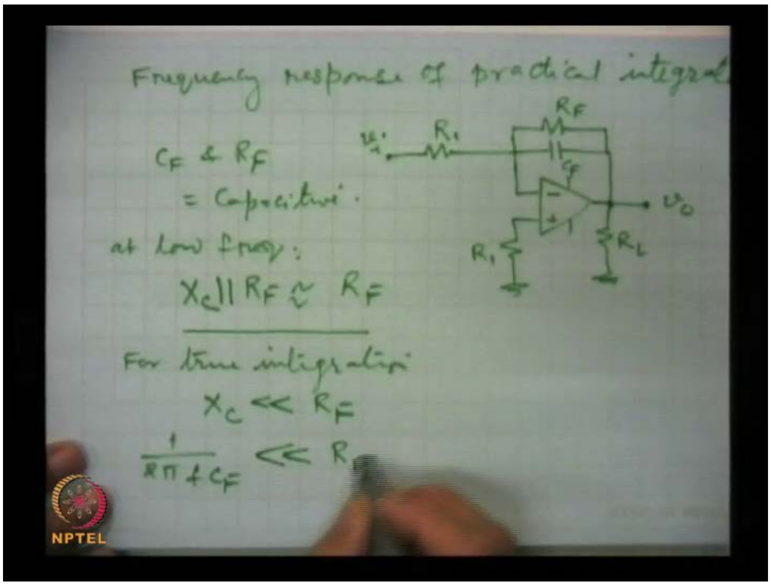

So now, we discussed the frequency response of a practical integrator, frequency response of practical integrator. So, the circuit which we are considering is now this (Refer Slide Time: 07:02), (No audio from 07:03 to 07:31) this is R F the resistance we have included; and this is C F and this is connected here and this is load across, which we take the output and here, we connect the input. So, this is the practical integrator.

Now, remember the basic that, true integration occurs when the when the impedance of this combination of C F and R F is capacitive, and this will be capacitive only when, this the effect of this capacitance dominates (Refer Slide Time: 08:29). So, and we know that at low frequencies, at low frequencies the parallel combination of the reactance of this capacitance in parallel with R F at low frequency, this will be very high impedance reactive impedance of the capacitance will be very high, and this parallel combination will give the effective value as R F.

So, that means at low frequency is this, inclusion of this resistance R F, this limits the lower frequency for integration; that means, at very low frequencies we had this condition is satisfied, the output will not be the true integration of the of the input. And then, so for true integration, for true integration this reactance should be very small as compared to R F; that means, 1 by 2 pi f C F, this should be this should be very small as compared to R F.

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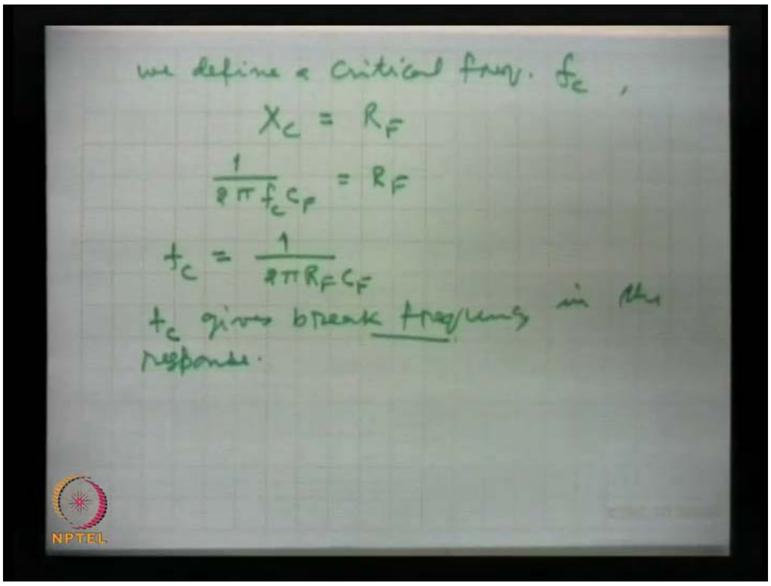

And we can define a critical frequency beyond which, integration will be alright. So, we define a critical frequency, f c we define a critical frequency f c such that, X c is equal to R F. And substituting the value of this, 1 by 2 pi f c now into C F, this is equal to R F or the frequency f c is equal to, from here 1 by 2 pi R F C F, this is the critical frequency. And this will give; f c give a break frequency gives break frequency in the response in the response.

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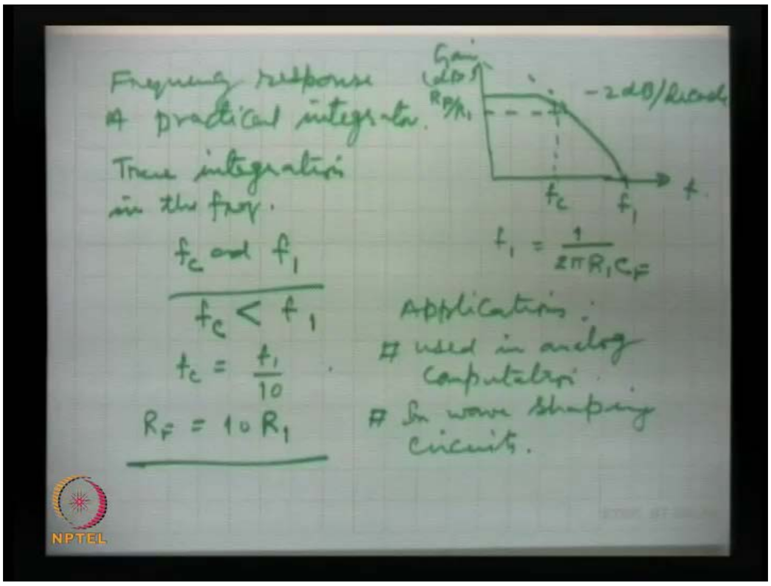

And the response is this, (No audio from 11:42 to 11:53) this is frequency, this frequency is f 1 and this is at, this is the 20 d B (( )) and here (Refer Slide Time: 12:16), this is the 3 d B line and this gives, f c this cut of frequency which we have written this f c, this is this value and for this f 1 we have written earlier, f 1 is equal to were gain falls to 0 d B, this is gain in d B and this is R F by R 1 gain. So, and this gain is this frequency is 1 by 2 pi R 1 C F.

There is a difference in two break frequencies, f 1 this is because this resistance will be smaller than the resistance, which is used with the capacitor. So, this contains the first resistance here, this is R F and R F is definitely higher at least 10 times, normally 10 times of this (Refer Slide Time: 13:33). So, this is the frequency response of this is the frequency response of practical integrator.

And what we see, that below f c frequency which we can calculate from this expression, below f c true integration will not occur. So, true integration true integration in the frequency limit, in the frequency between the frequency f c and f 1. So, actually f 1 is chosen quite high as compared to f c; f c we choose very high as compared to f 1 and in it is a thumb rule that, f c is one-tenth of f 1, and and in that case R F will be ten times of R 1.

This resistance is ten times of this resistance in the design (Refer Slide Time: 15:02). So, that is this, integration will occur in this frequency region, the integrator is not used below the frequencies f c. About the applications applications of the integrator, these are used in analog computation analog computation and in wave shaping circuits and of course, they are similar other uses. So, this was about the integrator or integrating amplifier using (( )).

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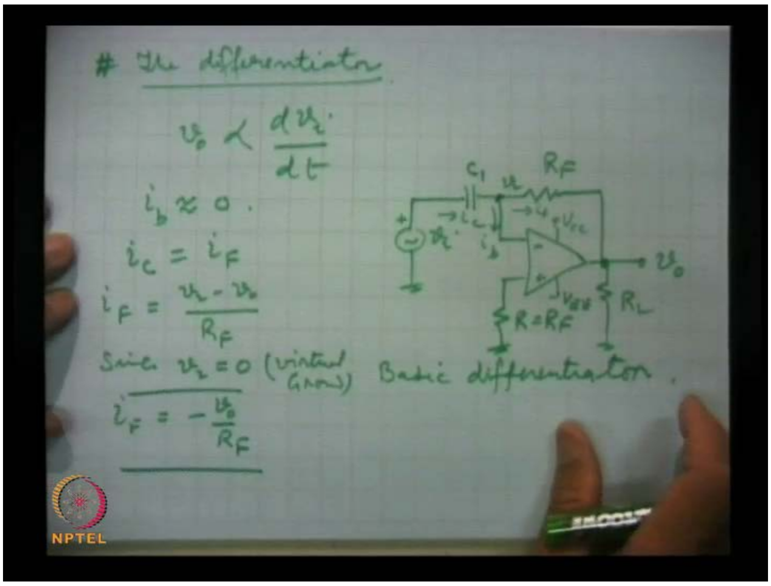

Next, we take a differentiator that is the, next in which we are considering the differentiator. In differentiator circuit, the output is proportional to the rate of change of the input at that instant. I repeat that, output in the differentiator is proportional to the rate of change of the input. And rate of change is as we will see; that means, what I am saying is this, is proportional to when input signal is v i and output is proportional to this and this is this circuit is called differentiating differentiator or differentiating amplifier.

And this can be again realized in inverting amplifier. Inverting amplifier, the capacitor R 1 the the resistor R 1 is replaced by the capacitance, then the circuit works as a differentiator. So, if the resistance R 1 of inverting amplifier is replaced by the capacitance capacitor, then this is the circuit (Refer Slide Time: 17:48), (No audio from 17:49 to 17:59) this is R F, this is c 1 and here this is v I, the resistance which is equal to roughly R F and here, is the load resistor across which we take the output, this is the basic differentiator, basic.

And of course, this power supplies are there, which are not essential to be shown, because it is implied that, they are they are no electronic repeatedly I am saying, no electronic circuit will function, unless the transistors of this in the device are properly biased and for biasing we need the d c supply. So, this is these are two supplies here.

This is the charging current i c, which the signal will send v i will send and here is current i f, this is i f, this is the small current i b; and i b this repeatedly is being said that, i b because, the input impedance is very high of the device it is in 1 or 2 (( )). So, i b can be taken as 0, if i b is taken 0; that means, no current is going in this arm, in the inverting input then, we can write i c equal to i f i c equal to i f.

And how much is i f, i f this current two voltages, this is v 2 this is v 0. So, v 2 minus v 0 by R F this is this. But we know that, v 2 the inverting input is at virtual ground potential, so since v 2 is 0 virtual ground everywhere we are seeing, the advantages in inverting amplifier, the advantages of virtual ground, because the concept of virtual ground makes the analysis much simpler. So, this is virtual ground. Then i f is equal to minus v 0 by R F, this is one expression (Refer Slide Time: 21:23).

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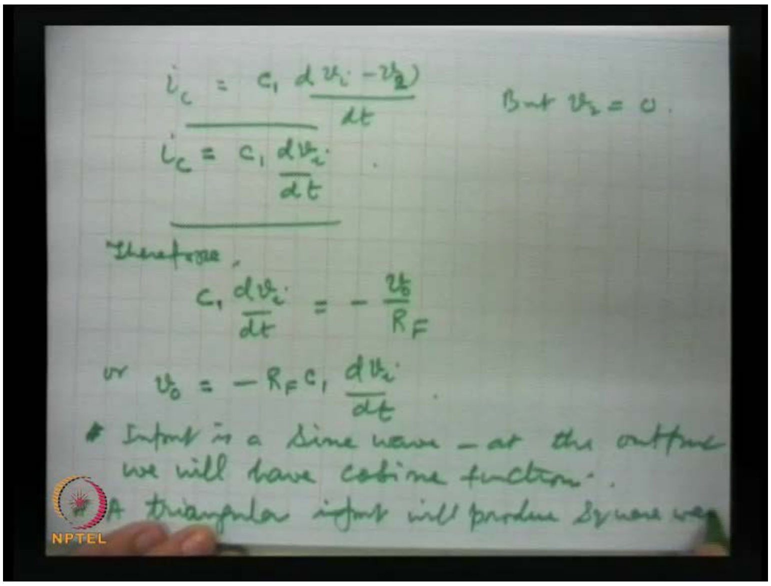

And how much is i c? The charging current charging current we can write i c, this is equal to the capacitance into, from simple capacitors theory and voltage difference, so v i minus v 0 by d t and sorry the two voltages, this is at v i and this is at v 2. So, this is 2 (Refer Slide Time: 22:10), v 2 which is 0. But, v 2 is 0 and hence i c equal to c 1 d v i d t hence, because the two currents are identical here, i c is equal to i f.

So, these two expressions we we have already got for i f and here this is for i c and so, we can write therefore, therefore c 1 d v i by d t, this is equal to minus v 0 by R F or v 0 is equal to minus R F c 1 d v i by d t. This is what we said in the beginning that, this is all constant and this sine inversion is does not make a difference this also we have been talking and so, the output is proportional to the rate of change of input.

So, let us take two cases, when input is a sine wave, input is a sine wave output will appear as cosine, at the output we will have cosine function cosine wave. And for triangular input, a triangular triangular input will produce a square wave output will produce square wave output. Now, we talk about again, because for these integrators and differentiators, frequency response is very important. When the input frequency thus, the frequency of the input signal is changing then, how the output will vary that is of significance.

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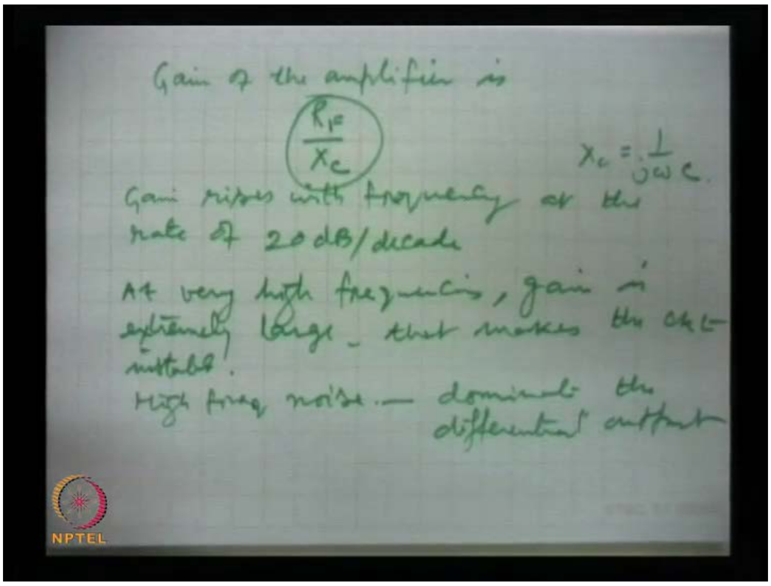

Now here, we we should note in this basic circuit of differentiator. There are two problems, one is that gain of the amplifier gain of the amplifier is R F by X c, where X c is the this is you know for the inverting amplifier, the gain used to be R F by R 1, R 1 was the resistance here (Refer Slide Time: 26:07). Now, here it is the capacitance. So, the resistance will replaced by the reactance; and this reactance will go low very fast with at higher frequency, because this is X c is 1 by j omega c.

So, at low frequencies this is at high frequencies, this is very low and therefore, gain rises with frequency with frequency at the rate of at the rate of 20 d B per decade of frequency; that means, when frequency becomes ten times, then it changes ten times, so 20 d B. So, this is the change, and at very high frequencies, this will be very low and gain will become very large. At very high frequencies, gain is extremely large and that makes the circuit (( )) that makes the circuit (( )) and that way, the frequencies noise they will also may get amplified. Now because of this, there is a high frequency noise which will be heavily amplified, and that will dominate the differential output differential output. So, it will it will not be a accurate differentiation at very high frequencies.

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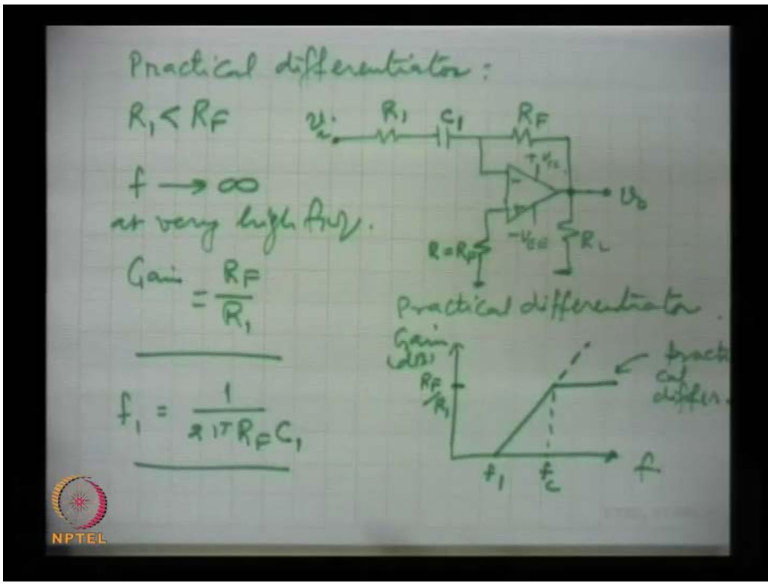

So, to take care of this fact, that gain becomes exceedingly high, when the frequencies are very high. We attach a small value resister and we get a practical differentiator practical differentiator is this (Refer Slide Time: 29:37), (No audio from 29:38 to 29:54) this is R F, this is the resistance R 1 which we have attached, and R 1 is quite a smaller as compared to R F; and this is c 1 and here, we attach the input signal, this resistance is equal to R F and these are the supplies. This is the practical differentiator practical differentiator.

And here, now at very low frequencies frequency of the signal tending towards infinity at very high frequencies at very high frequencies. Here (Refer Slide Time: 31:09), the gain was rising very fast with the fall in X c at in this case, now when this is almost is a short, then gain will be restricted to this. And this normally may be 10, 12, 15, so gain will be this at very high frequencies; and the frequency response of this practical differentiator is this and here, this was the gain going up very fast with frequency, this is frequency (Refer Slide Time: 31:54), and this is in d B.

Now here, for the practical one, the gain will state very high frequencies when this capacitor is almost acts as a short at those frequencies, the gain will be this here; this is R F by R 1. So, this is the response of a practical differentiator, this is practical differentiator, this was for the ideal one, this is for the the basic which we have showed the integrator, basic integrator and this is the practical integrator.

And here (Refer Slide Time: 32:51), this frequency is f 2 or f c we can say and this is f 1, f 1 is the frequency, where the gain will fall to 0. And f 1 will be given by, f 1 is 1 by 2 pi R F c 1 at the frequency at which this becomes a the gain becomes 0. So, this is the frequency.

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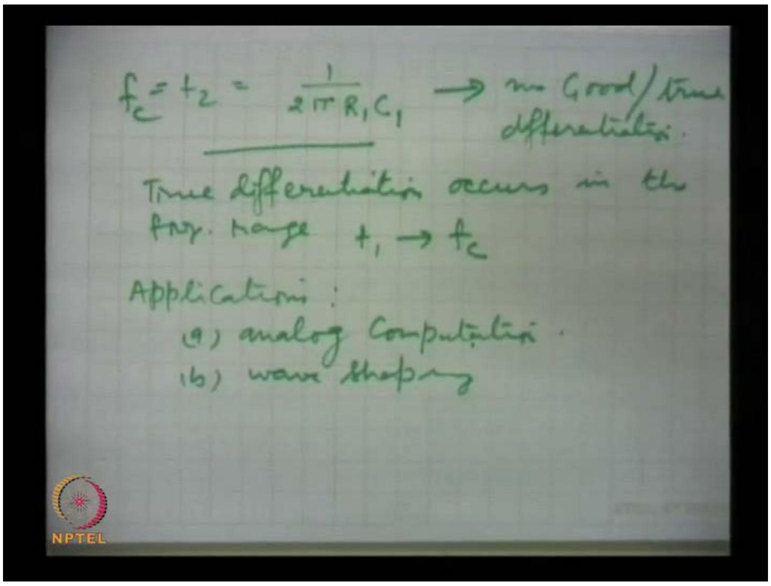

And the inclusion of R 1 that has restricted that limits the high frequency range of the differentiator, and this f c will be given by, f c is given by 2 pi R 1 C 1. So, again as in differentiator we have two frequencies, f 2 which is same as f c which I have written here, this we can write f 2 or f c, this is this frequency (Refer Slide Time: 34:09), and this is the the difference is of R 1 and R F. So, beyond this frequency f 2, no no good differentiation, no true differentiation of the input will appear.

And so, differentiation is there in the frequency, true differentiation occurs in the frequency range f 1 to f c or f 2 whatever you write. And this frequency is set f c is set quite high in comparison to f 1, so that there is a good range of frequencies over which true differentiation will be possible about applications, this is we have discussed.

So, the basic circuit is not used for differentiator, but this circuit is used by inclusion of resistance R 1 and this is the frequency response of the practical differentiator, the inclusion of R 1 puts a limit on the high frequency; that means, a safes the circuit from (( )) of very high gain; and so, a true differentiation occurs between the range f 1 and f c.

And about the applications of differentiator, they are the same as the integrator that, they are used in analog analog computation and wave shaping. So, this is all about the differentiator. So, we have taken several applications, we continue with few more and that will show the versatility of the operation amplifier. So, after these summing circuits and differentiator, integrators, sine changer and we now go for another class of of applications, and these are active filters.

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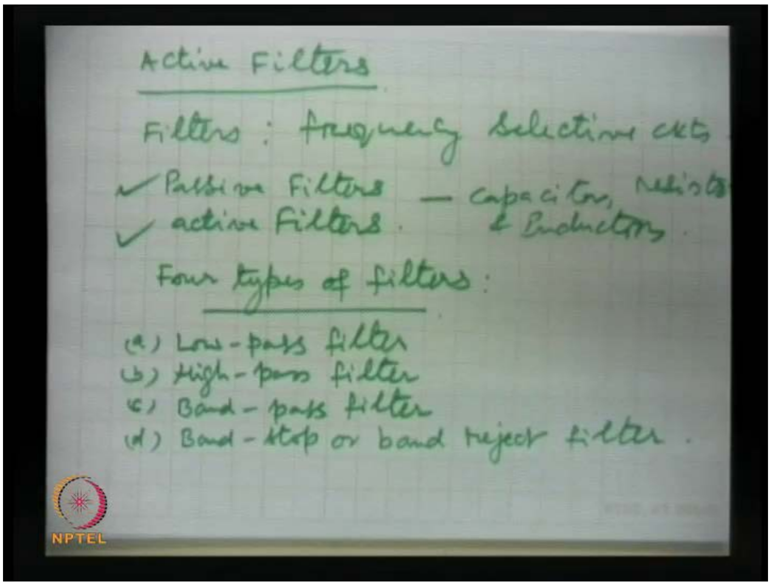

Active filters first thing is, what is a filter? Filter is a frequency selective circuit. Frequency selective; that means, you have a spectrum of frequencies and you want to select for your system that your system should respond to certain frequencies. So, how to get them? Frequency selection can be done in by using filters, filter circuits.

Now, there are two types of filters, passive filters and active filters. Filters, these are frequency selective circuits, and they can be two types, passive filters and active filters passive filters and active filters. Now, passive filters can be realized frequency selection circuits you can make by using capacitors, resistors and inductors, you must have done, you have done a resonance circuit.

What is a resonance circuit? Resonance circuit is a frequency selective circuit, and resonance can be obtained by the combination of capacitor, resistance and inductors. But, these filters the passive, these are called passive filters, they are inefficient; inefficient in the sense, there are two points, one is that these components will dissipate certain signal. So, signal is attenuated in passive filters number 1. Second thing is that frequency response of passive filters is not as good as in the case of active filters.

So, active filters are the ones, which make use of the active device, transistors or (( )) to make filters, operational amplifiers are very widely used in fact, they have replaced B J T and all other circuits, for making active filters; because, handling of operation amplifiers is not at all difficult plus, they are not expensive. And hence, active filters make use of operation amplifiers and these circuits are very widely used.

Now, there are four types of filters, four types, four types of filters. What are these types? First I name them and then, I will elaborate them. One is low pass filter low pass filter then, we have high pass filter, we have band pass filter and then finally, we have band stop band stop or band reject filter or band reject filter. These are the four different kinds of filters, depending on the selection of frequencies for the working of the circuit at which frequencies we want to choose. And in signal processing, this filtering of frequencies is a very important process and so, we now first talk about these filters.

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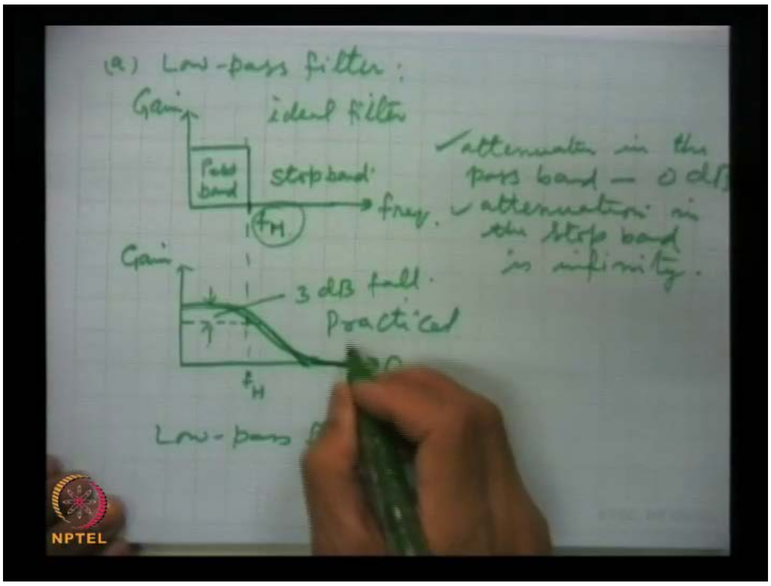

The low pass filter, first we take low pass filter. Low pass filter is the one, which is supposed to pass, by pass we mean that they will go in the circuit for further processing. So, certain frequencies from very low to a certain high frequency, they will be allowed to propagate beyond that frequency signals will attenuate heavily very heavily and they will be prohibited from going for further processing, so that will be low pass filter.

The ideal first I take ideal low pass filter, then the actual low pass filter practical. So, this is the case, this is gain and this is frequency, this is the pass band, this is frequency, so this is pass band (Refer Slide Time: 44:17), and all these is stop band and let us, call this frequency as f H and this is ideal.

So, what is the low pass filter, from 0 frequency to upto a maximum of say f H frequency, it will pass; that means, signal will pass through the circuit without attenuation and it will be it has been selected in this frequency range it will be selected and it will be further processed. And in ideal, this is the case of ideal filter, in which gain the attenuation in the pass band attenuation in the pass band in the ideal filter is 0 d B that means no attenuation; while attenuation in the stop band, stop band is infinity infinitely high attenuation. So, this is for ideal filter.

Now, electronically we cannot have this ideal gain frequency response, because there is no process by which this break can be that sharp. So, the practically what we have is this (Refer Slide Time: 46:25), (No audio from 46:26 to 46:39) here (No audio from 46:42 to

46:51) we have this frequency is f H. And so here, this gain this is 3 d B fall 3 d B fall and this is frequency (Refer Slide Time: 47:19).

So, this is the case that gain will start falling here and then, it will go and fall very fast here and upto this frequency 3 d B fall will be corresponding very close to H f here (Refer Slide Time: 47:36). The the fall will start and 3 d B has falling by this frequency, cut off frequency and beyond that, again in the ideal case, it was infinite attenuation here, that at loss the attenuation will be high, but it continues for certain frequency region.

So, this is the response and we will drive an expression for this cut off. What is the highest cut off, for the low pass filter? In this, we can design we can choose it can be 5 kilo hertz; it can be 50 kilo hertz depending on our requirement. So, this is the case of low pass filter, this is the practical filter, this is practical case practical filter.

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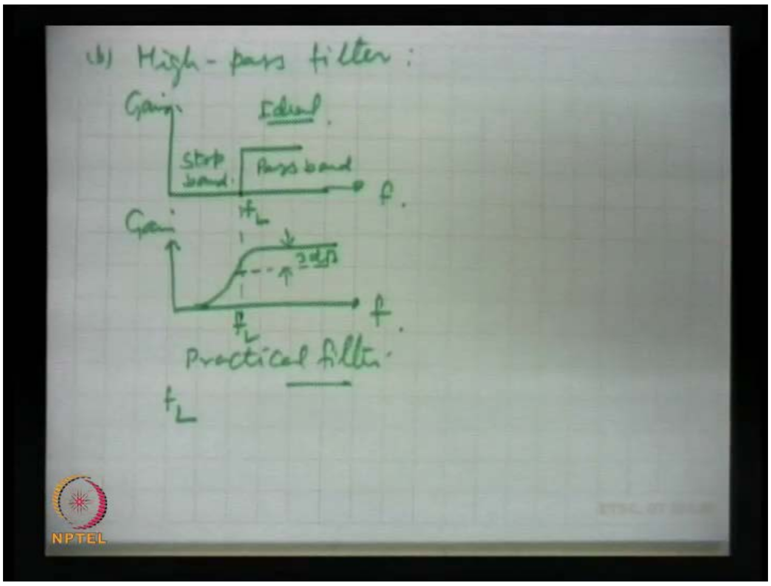

Then we go for high pass filter high pass high pass filter the ideal case is here, this is pass band (Refer Slide Time: 49:06), and this is stop band, this is gain and here, this is of course, frequency. So, this is the ideal ideal filter, where the attenuation in the stop band is infinity; and from this frequency, which is the lowest frequency from where the pass band start, the attenuation is 0. But, this 0 and infinity again, these are ideal and this sharp boundary between stop band and pass band, this is also only ideal.

In the practical case, this circuit this response for the the actual filter will be like this (Refer Slide Time: 50:12), this is gain and this gain is 3 d B, and this frequency will be f L say, this is frequency. So, this is the response of the high pass filter, this is the practical. And we will arrive at an expression for this f L that is lowest frequency in this we can choose as I said, depending on the circuit components.

Now, the active filters as I said, will make use of (( )) capacitors and resistors, just few capacitors and resistors and filter design will be complete, the inductors are not used; there are reasons, and these reasons are inductors are bulky and heavy, and they are not (( )) and hence, inductors are not used with active filters. So, this is the case of high pass filter.

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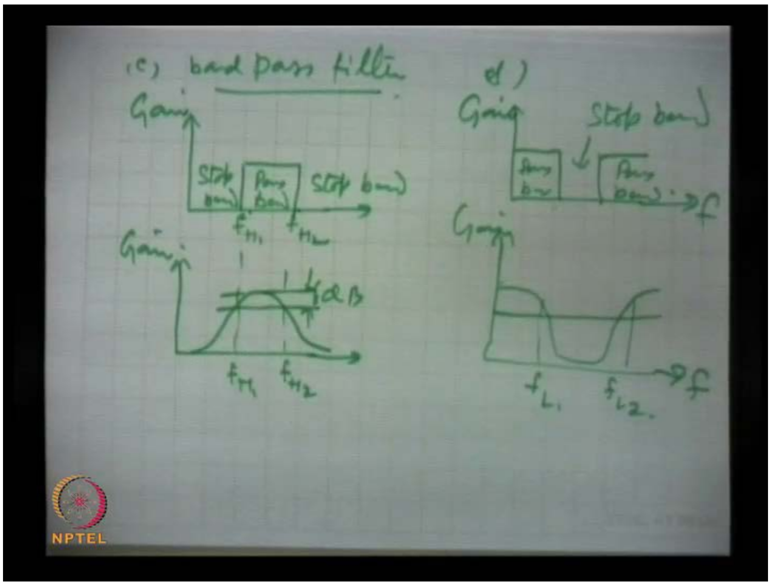

And then, we can have the band pass filter, c band pass filter. The band of frequencies is allowed before that, they it is not permitted, after that it is not permitted; and for the ideal, this is the case (Refer Slide Time: 52:04), this is gain and this is pass band, this is stop band, this is also stop band. But practically, these are the frequencies f H 1 and f H 2; that means, for this these frequencies line between the pass band, the attenuation will be very low; very heavy in the smaller frequency region and very heavy in the higher frequency region.

And for the practical one, this plot is like that (Refer Slide Time: 52:44), and here this is 3 d B change again, 3 d B and this is f H 1, f H 2 gain. So, this is the the band pass, band of frequencies will pass, rest will be stopped.

And finally, we have the last one, the band reject or band stop. This is ideally it is this (Refer Slide Time: 53:27), this is pass band, this is pass band, this is the stop band, this is gain, this is frequency and this is that. And practically this will be like this (Refer Slide Time: 53:50), where this is gain, this is frequency, and these are those two frequencies, L 1, f L 2. So, this band will not be permitted to go in the circuit for further processing. So now, we will go one by one about these in details, the circuits and their analysis.