Analog Circuits
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Lecture-21
Active RC Filters

We start with the filters as we said largest last time. The word is active the FT filter essentially stand for the word there is a active device during filtering. So, some property of active device has been used otherwise we shall see soon that even a normal RC, or LRC filters can do as much a good job as otherwise but why active we will see later.

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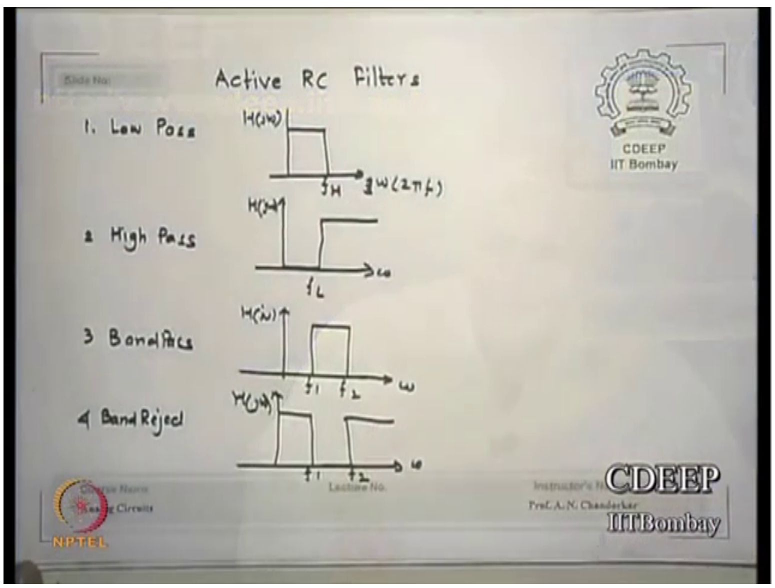

I have typical filter requirements as I shown you last day was we have 4 kinds of filter we may use. One is something which passes almost everything except up to a frequency we call it low pass. Something which passes all of it from starting from some frequency below which it is nothing is passed so it is called high pass. You are allowed to pass between two frequencies the signal but otherwise block everywhere or do not pass anything there here or here.

And the finally you are passing everywhere except some band of frequencies. So, these band reject, band pass, high pass and a low pass. These are the 4 filters which we will be using in any of these circuits which you will use in your whole career okay. We are right now; either one filter says does not say whether it should be analog filter or it can be a digital filter you can actually either kind of them.

Normally even if I am doing a analog a sorry digital filter we actually convert partly to digital in a half way and what we call switch capacitor filters. These are essentially digital filters but actually copy out most of the analog functions okay. So, the basic filtering still could be said to be analog filter.

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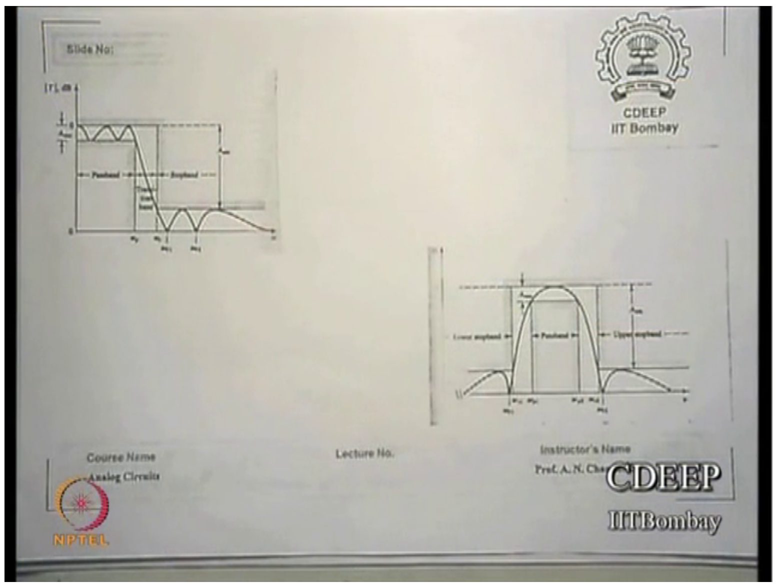

Of course there are many digital interesting filter your signals and system theory, a communication system theory will deal with many more of them later in your career or latter in your classes. Here is a typical low pass filter shown to you maybe I will Center it out first yeah this is typically ideal filter. This is what I am expecting ideally but when I actually pass through any active filtering system.

What I see is there is certain frequency called Omega P the certainly the transfer function does not show a constant value but ripples up that is keeps on changing and till this point which is essentially a pole which is essentially a pole. At this point the gain starts falling or the transfer function starts following its value because that is the pole we said. And it ideally it should I reach to 0 at this point but it did not it actually crossed that point.

And before it actually settles to 0 again ripples okay again ripples. So, the way we defined is this is of course called the ripple which is passing through it we may assume it is a universal one frequency repel in real life that also is not to repel itself in our different components. But right now assumes single frequency ripples ideally what did we want; we want flat response here and sharp fall them.

This is ideally a filter low pass filter was asking for what we are essentially getting a fixed value and + ripple up to a frequency where the pole starts that band we call pass band. That means where the transfer function is such that everything is passing. Beyond this depends on the number of poles this may be 20 DB down 40 DB down or 60 DB down whatever it is. The values start going down and when it reaches the so called ideal filter value.

We essentially say that we are reached there is a band between the past band and before everything becomes 0. Let us say this band is called transition band that means from high to low the frequency range is called transition band. Ideally what was what do we really need transition band, ideally we want transition band should be 0 okay that is what ideally we are looking but what we are seeing in real life there is a transition band.

There is a fall time or fall this up to which it will keep falling. Beyond this we would have expected something to become constant but it did not. But it also repels and maybe finally it settles to the value which you are expecting. The band in which nothing is passed as for the low pass requirement we call it stop band. So, for any low pass filter we are interested in what is the ripple magnitude, ripple frequency, what is the expected pass band you are getting, what is the transition band you are getting and what is the stop band you are getting.

And of course as I say ripple frequency itself will be varying but I assume right now it is single frequency ripple occurs. So, this is a definition for any filter this; so ideally for a low pass I have expect A max be 0 okay that means this should have come here I want transition band frequency Omega this low - and Omega P should be as 0 as possible and again this A max be 0. Ideally this is what was expected and if that occurs we say we have a ideal low pass filter.

If you look at the other bank bass requirement it is similar a typical good band pass filter would have expected to have this transition. Between this frequency and this frequency the transfer function should have a value which is constant and everything is passing. Below this frequency we do not want anything to pass. So, we say lower stop band beyond this frequency nothing should pass we say it upper storm.

And so there is a lower side stop band and upper side stop band in between there is a pass band. However in real life this is not starting from here it actually again the purse and some kind of a Gaussian function one gets rather Gaussian function it gets and then that means beyond this Val stop band a I mean pass band a this and but there is a ripple over it here and it going down so there is a pole there is a somewhere 0 starting here.

There is a pole going down and one can say it again the purse before it actually reaches to 0. So, essentially in our design what are we expected to do that here it starts very tense and what is this ripple magnitude is going to be is that clear that is what we must have and what should we do minimize them as much as possible. Same way this is some transition band there is a transition when I also want transition meant to be as close to 0 as it is possible.

We already said that a pass this band pass filter if essentially a combination of low pass and high pass. If I said this is my FL and if I say this is my FH, please remember I repeat listen that is why I gave the name separately this is my FL this is my FH that means first there is some high pass starts and then there is a low pass starts. So, in between both are passing and so done the band pass. If I separate the upper frequency if I make this as a FH and if I make that as FL that means high pass sorry low pass followed by after certain frequency high pass.

There is that frequency band in which nothing is passing so we said say band reject. So, basic design of a filter is essentially this and this. This is just a setting of the frequencies to suit your band pass or band reject requirement. So, in design because in this course we should have done time is not enough so we will only do one good low pass and one good high pass and if we get that then we say okay. This is the circuit which can be then manipulated to create a band pass as well as a band reject filter.

Now having shown you this is my requirement and this is what ideally and non ideal I am going to get I want to come closer to idea. But before we go to act networks active networks, I just want to see a network itself a normal network itself.

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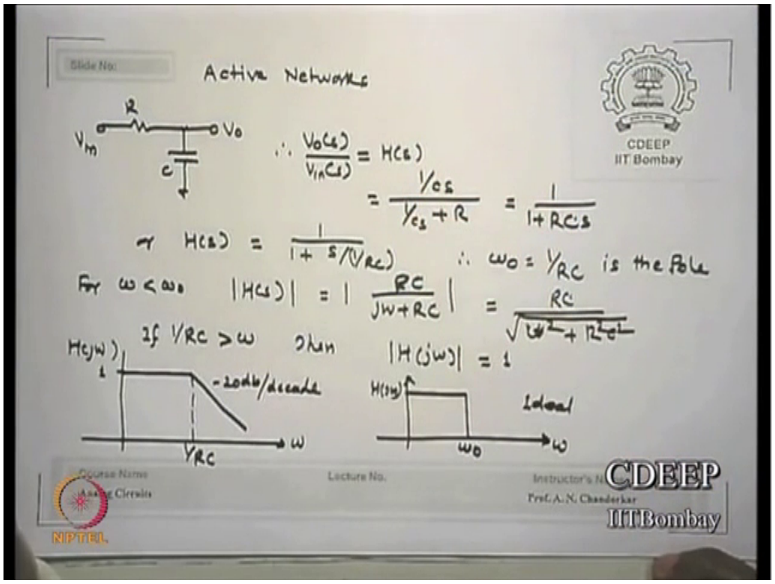

We are done this earlier is well, if you are a RC network 1R and 1C, if I take a transfer function of this V0 by Vin is 1 upon CS upon 1 upon CS + R which is 1 + RCS or it can be written as 1 upon 11 by RC, if I define a frequency cutoff frequency as 1 upon RC this is nothing but 1 upon S by Omega0. What does that mean there is a pole in negative half2 - Omega 0 and if I plot the Bode plot therefore at this the gain will start falling okay.

So, it simple RC network actually acts like a which will turn low pass filter. Now, one can think little r interestingly, if you see now ideally I wanted this sharp fall, is that correct. I must all my design why I am doing active filters all my; in any passive filter this is going to occur is that clear. So, what I say can I do some mischief ok or close to this I come closer to ideal and the active part should help me in getting these as close to the ideal value as is possible.

If I do that then I will actually create a filter which is close to ideal low pass. So, why I active part is necessary because I want to ship the slop down. If I do so something else may happen I will say whether I can tolerate that or not I will check that but at least I want fall to be as sharp as possible. Let us say it is 120 DB down at this point is that clear. So, it was short very sharp is that clear.

So, essentially I am trying to see I have done this pole 0 theory and I am now started looking here. Someday there should be multiple poles coming there. If they are they occur at very close faced at the same point then my fall can be; but if I look at the other filter equivalence of this which is interesting okay.

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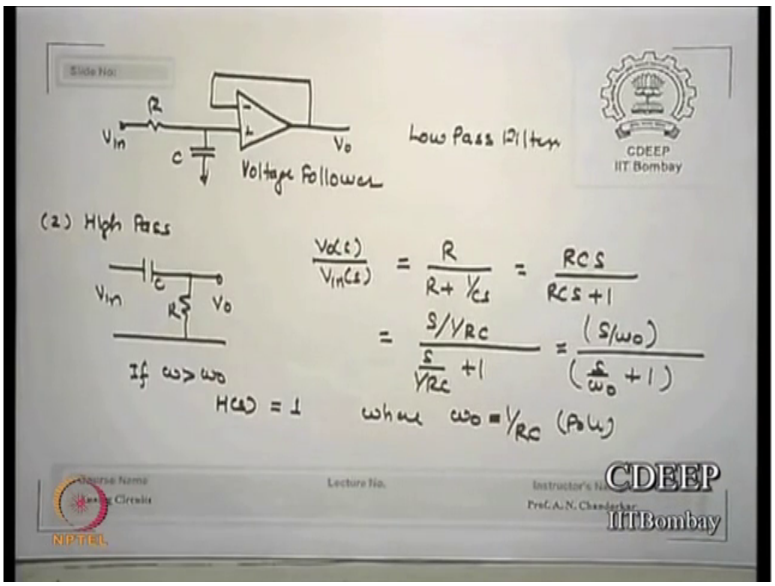

So, basically active low pass filter I really could be a RC circuit as I just now do followed by a voltage follower is that kind of followed by a voltage follower. So, whatever is the voltages here will be transferred here because a voltage follower. So, this is the easiest way of making a low pass filter we are not seen as transfer function so far. So, we do not know whether it is going closer to ideal but this should do as a low pass filter with an active device.

What is the purpose follower here, if any load occurs at the capacitor it will load that out this buffer will separate your input response from the output loads is that clear. That is why I put a buffer in which is my voltage follower. By doing this I am still achieving 1 upon RC kind of requirements and I am getting low pass with of course 20 DB downs gain falling. I can do similar thing for and as I say high pass is just the opposite of low pass and I say okay replace R by C and C by R wherever there.

So, I put a C here, R here again derive this transfer function which I get S by Omega0 upon S by Omega0 + 1 where 1 upon RC is Omega0, now you see how many things that thing here how many poles are there how many 0's are there, there is a 0 at S = 0 itself and there is a pole at S is = - Omega0 is that clear. So, there is a 0 and there is a pole. So, till Omega0 which will dominate? The 0 will start dominating is that correct.

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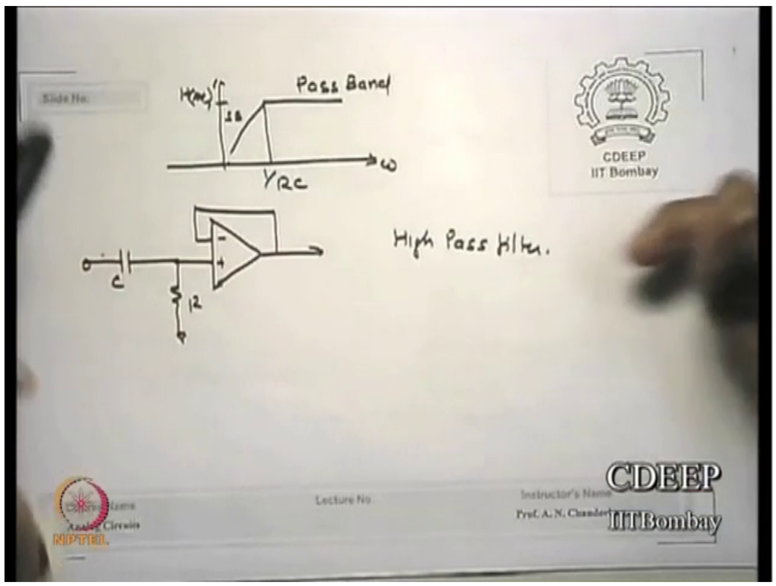

So, initially what could occur you can see from here 20 DB, 0 means 20 DB per decade rise start rising okay. At the pole what is the fall should be -20 DB but what is it rising already with 0, +20 DB. So, at this point onwards 20 – 20 is 0 DB gain becomes our output function becomes constant is that high pass requirement at frequency above 1 upon RC, I want transfer function to our unity value all no attenuation it just passes out inputs is that correct.

So, how do I create in this case, you are the CR here pass through a follower and this will act like a high pass filter, this is 0, this is 0 and polled together constant. So, obviously now one thing you must have got from there if I had no pass I would avoid 0's is that actually 0, should be at infinite okay. If I am looking for high passed I will bring 0 wherever I want flatness to start, is that clear. So, that means I have now a trip that if I have a S in the numerator I am creating 0's + something also.

But S is in the numerator and S+ something in the denominator if that kind of function I create then what I will create high pass. If only 1 upon S + something I create I will create a low pass is that point clear, this is the proof of filtering that okay. Create properly 0's and poles such that you actually achieve low pass and high pass values. The only thing trick is how much to get whether this value should be one ort it should have gained function.

If I want a gain function what should I do? I should do R here then it will also give me some gain out of it. Like a normal amplifier I may amplify it also, now these tricks are always played in many of the filter designs. But using this what is the main problem we saw that that 20 DB is the all that we could get in you cannot have fast rise or fast Falls, so this; this is a filter low pass high pass can be always realized by this.

Try yourself one and you will see yes it will cut off somewhere and it will show you but the response will not be closer to ideal. But why should we always have ideal yeah the reason is okay so there is something you must remember. Many circuits do not mind for example that it should actually be cut off completely. As far as any way it is like a noise going on a DPAMP matter. So, anyway I do not care very much what happened.

But if I do care single end system I will worry about it. So, depending on the what is going is going to dry or wear from the next stage will be. Your choice may not be you should not see that I should have a repel I should not have now. Way I have figured out and that is all work today and not necessarily I will complete. I have figured out basically if I see a function as I am getting I know any conic section take any conic section okay straight line parabola, hyperbola, circle, ellipse or for that matter any curve for that matter.

Can always be represented by a very simple functions what is that function is called polynomials. If I write A0 + A1x + A2x square up to n polynomials I assure you that any curve can be fitted okay. It may require thousand terms to actually get closer to the curl you are looking for it is a fit function technique. So, I figured out that if I can generate some functions which also used to be light those 0s and poles which I am looking.

Then I can have a this equivalent of a filter and repeat what I said I create a polynomial or I create a function whose numerator and denominator looks like 0's and poles but their values can be tailored then I can say I am right now creating a filter for that function is that clear. That response will be something closer to filter is that correct. This is the technique which most designers or most circuits in the analog we use.

That we actually create some kind of in known functions whose values mathematically we can derive and then try to fit it on those on a circuit and we say how close we are getting these two filters which we shall see little later one is called Butterworth filter the other is called Chebyshev filters. Why I am showing you because in Butterworth filter the ripple can be minimized what is the advantage of a Butterworth filter that the ripples are very small.

And therefore it is called maximally flat. So, all Butterworth filters are so designed that their repulse are very 0 or close to small value of 0, kind. It is a flat here but the price I pay because of the function I am using to create the ripple 0 or ripple small there will be a transition band is that correct there will be a transition band if I want to reduce transition band then I say the number of poles which I will require higher and higher.

That number of remember each pole will require let us say one circuit of this kind equivalently saying I just now showed you each pole requires one such circuit if there are n such pole I will require n such circuits. So, I figure out that if I want I use a Butterworth filter then I can maximize the flatness but to reduce this transition man I may require larger number of poles is that correct.

On the contrary Chebyshev filter can more this they say we cannot reduce very much the ribbon okay but for a smaller number of poles I can reduce the transition band is the two difference clear. In the maximally flat filters but air filters the ripple is smaller of small very small. At the cost if I want to reduce the transition band I will require larger number of poles, a larger number of sections as I call.

In the case of Chebyshev, I say I cannot reduce this very much I will tolerate ripple okay but for a smaller number of posts I will give it a transition or the transition at least center on this and I would have gotten better word okay with a smaller number. What is the advantage of smaller number of sections or smaller poles circuit, a smaller numbers? What is the cost? Cost reduces if your number of sections goes down the price goes down.

But price is going down at what cost the ripple is that correct. So, if you want a very good filter then you pay price okay higher price is that correct. So, maximally flat filters will require larger sections but will care much less ripples is that correct. This is something all designers must keep in mind (FL) All IC designers are always told that anything I use because we have a silicon area unknown area.

Let us say few millimeters by a few millimeters say one centimeter at best I may have good science so many millions and I am putting. If I do not put additional hardware I can put something else there, is that clear. That means the price of silicon is very, very high area silicon area what does them equivalently we say like in Mumbai we say it is a real estate, silicon is a real estate any small micron by micron you should throw you are losing money okay.

So, whole frickin indeed circuit designs are how to minimize area or increased density is that correct. So, if you are doing the same function with larger circuit, it will do better but you paid price higher price for that okay. so, one must understand some chips we sell very heavily high at high price if someone like young man says I want exactly this filter this should do this think. (FL) This man says no, no I do not have only this much money. (FL)

So, the trick of design is what is the customer wants okay. So, in all of our engineering this part is never explained till maybe in your graduation, will learn in fourth year we are never telling that all this our theory etc is as good as the money in your pocket, that is it. So, this has to be understood why we learn all the tricks because we do not know what customer and then will give me a response. (FL)

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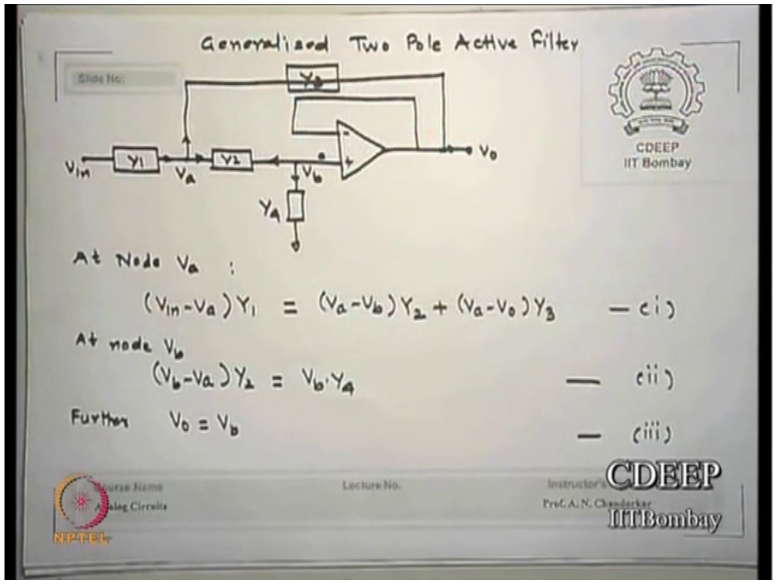

So, I have now suggest you here is a circuit active filter which is a generalized to coal filter how much how many what you mean by two poles how many capacitor RC's it should be 2 RC's system okay. This is generalized why I say generalize what does that mean generalize means, all four filter should be clear can be created out of this is that clear generalized I never said it is low pass, high pass, band pass.

From this circuit by proper choice of the y1 and y2 and y3 and y4 I should be able to create low-pass high-pass, band pass, band reject, is that clear to you. What is the advantage of such generalize network then because this is one time created any user during design can just manipulate this and say okay here is a low pass. So, he does not what is the other cost on design do you know what is the cost of design.

Is the time taken by so many engineers are called man years, man hours is the cost larger the circuit redesigned by you every block then it will be costly every time. So, if I have a diamond block which can be manipulated to create different things I has saved a lot of money that is essentially called semi custom. Something is prepared and used again and again. So, this is something equivalent of a semi custom design that means a generalized block can be configured as either low pass, high pass or any other is that clear to you.

Why I shown you this is something which all designers should know analysis why this is not very great this is same as what low pass high pass very big. But here is some analysis I will show. I have an input V in to one of the conductance, it can be admittance or conductance Yi g + JB Y is G is the conductance base acceptance admittance. So, Y = as we said Z = R + JX same way it is the opposite of that. So, I say ok here is a Y1 component, Y2 component, Y3 component and feedback.

This is to feedback this is our follower which I am not disturbing. Please remember this follower I have kept as it is. There is another feedback which I brought here to output. Let us say Kirchhoff's law at node VA what are the current what is the current entering V in - VA into Y1 is that current, is the current passing through Y networks very simple trivial. (FL) from VA to VB and VA to V0 is that clear. These are two more branch currents.

So, branch current at VA must be 0, total current. So, this current must be this + this, so I write VA - VB into Y2, VA - V0 into Y3 they must be equal Kirchhoff's law at node V8 we cannot summed up okay. Same way I sum at VB, I said no VB, VB - VA (FL) or what is current here? No current, no current enters OPAMP that property I used. So, I say VB - VA into Y2 = VB by Y then the third equation I see it since this is a follower, this is V0 but that should be same as VB.

By what is the that is active and did you get but what is active part that I used OPAMP properties have been used in actually writing the networks is that correct. These equation I could not have written if I have not used OPAMP properties what properties are used same with no current enters same potentials both properties are used and I use this is their follower okay.

So, I use this property of OPAMP and then could write such equations. If I solve this, you solve yours, I have solved but you solve yourself it, then you can do analysis.

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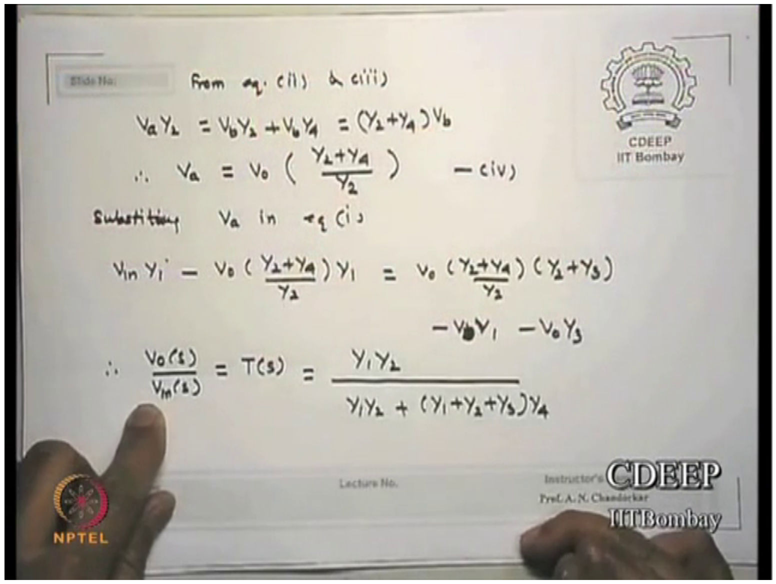

I did just look at the final transfer function you write and verify V0 S by Vin S which is my; either you write PS or you can also write HS whichever way you are writing either T is the transfer function okay. Normally I have been using H as the transfer function but I do not know from wherever earlier and there was a T, so I use T. So, maybe your colleges, so I get Y1 Y2 upon Y1 Y2 this kind of function.

Please note down this function rest we do yourself Y1 Y2 upon Y 1 Y2 + Y1 + Y2 + Y3 into Y this is the transfer function I got for this circuit I have given there, is that clear. Simple (FL) expressions are V0 by Vin is; I will now explain a specific example which will tell do you want me to do we will see little later but let us do this example.

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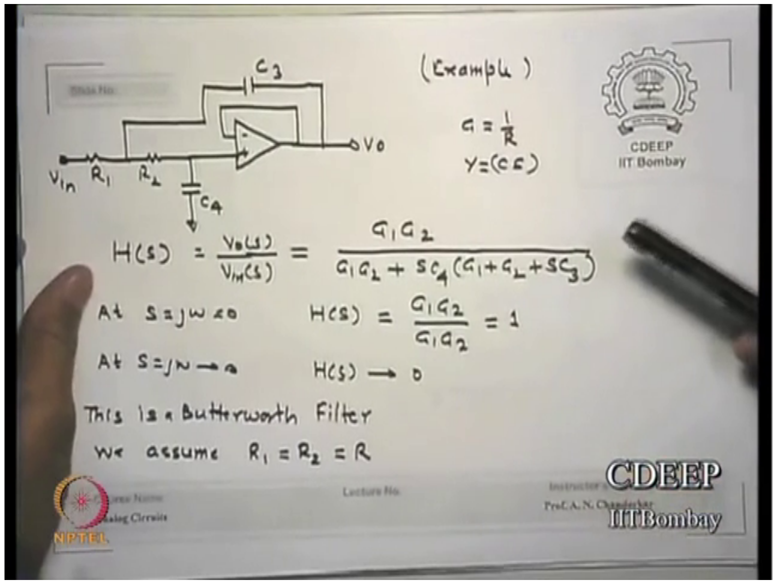

Now in this I create the first Y1 Y2 I put it as R1 R2 and this Y3 I4 I put as two capacitors in our case G is 1 upon R and Y is CS okay B rather not but these are Y only, this is capacitance. So, CS is the admittance of that and conductance of this is 1 upon R1 and 1 upon R2, this is the expression I just got further transfer function G1 G2 upon G1 + G2 + SC 4 G1 G2 SC 3. So, now you look at it at S = 0, what is the value of HS. (FL)

Please remember all that I have done is I substituted here the actual G's and C's okay. So, I got G1 G2 upon G1 G2 + SC4 G1 + G2 + SC3 is HS okay. At S = 0, what is the value of transfer function (FL) G1 G2 by G1 G2 so 1, what does that what does that means at S = 0 there is already a transfer function has a value, 1 means it could be normalized to some other 0, so it has a value is that correct. So, which filter I am looking for now.

Initially there is a value which filter I am looking? Low pass? I have a value of the function of the output gain available at this value is that clear. Now when S tends to infinity, so there is a S = J Omega, when Omega tends to infinity. One can see the transfer function will go to 0 is that correct that means finally it is going to 0 that I see it is one and finally it is going to 0. So, (FL) this is what we say, initial or a loop and then it is going towards 0, so obviously I am looking for a low pass.

So, can you now get this by just substituting here the proper values of this I can create a low pass filter. How many poles you are seeing here? Two poles say at least fall will be better than one single pole system, is that correct. It will be better than the single poles (FL) or it will become an high pass filter okay. So, a generalized filter (FL) so, I can create a different filter of my choice of different cutoff.

Why different because RNC is in my value (FL) so, generalized systems (FL) why it is not so good because it is now once made at best you only those two poles to manipulate additional (FL) is that clear you cannot actually go to this person requirement charge (FL) so, this tolerance of yours is how much will be depending on I said how much money you keep in your pocket. That is the way it is. (FL)

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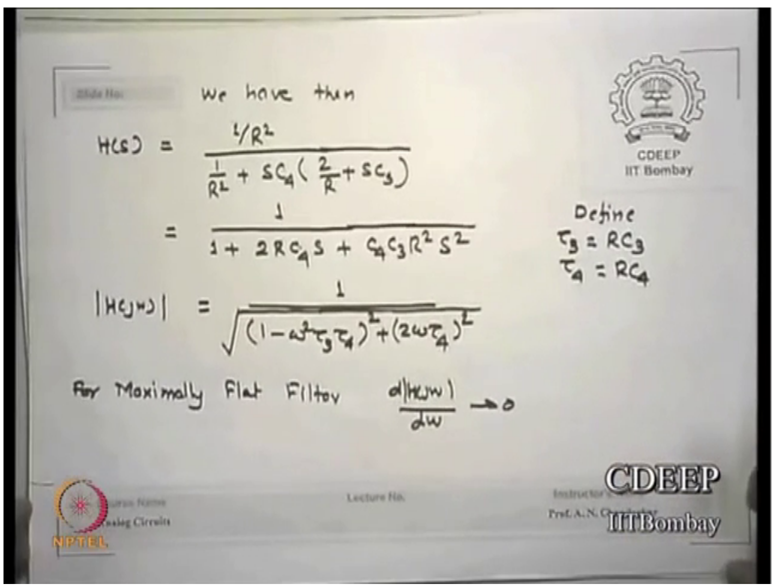

So, I wrote the transfer function as it is (FL) is it clear but I just substitute G 1 upon R and twice GS and I get this function if I manipulate it better and I give a definition that tau3 is RC3 and let us assume R's are equal. My assumption is R's are equal but I never said C's are equal, R's are equal but see why I do not want to make C is equal because then the time there will same poles will come. So, I okay there are two different poles one is greater RC3 the other related to RC4.

So, this is the transfer function I want please note down HS is 1upon 2 RC4S + C4 C3 R square S square (FL) if I define this I can write H J Omega magnitude (FL) you can also think how I have written (FL) 1 upon S - A into 1 upon S - V (FL) A = A1 into A2 is that correct (FL) is that point clear. I repeat a transfer option H1 is using one pole a transfer function H2 using second put is that correct. I can create two separate functions. Output of the first is given to the input so the transfer function will get multiplied so 1 upon S - A into 1 upon into S – V automatically will appear. So, is that trick clear to you?

Once you have two poles you actually can create single pole sections and then keep output of that should be given to the next input keep doing. So, as many sections you will put those many poles you can create is that correct. If you put 7 sections that is why a word I use sections, each one pole will create one section I have a number of sections. So, I can create as many poles I want and the trick is or they get always multiplied by H1, H2, H3, H4.

This is the way we actually will implement higher pole functions. Anyway (FL) d by this function with d Omega should be 0. (FL)

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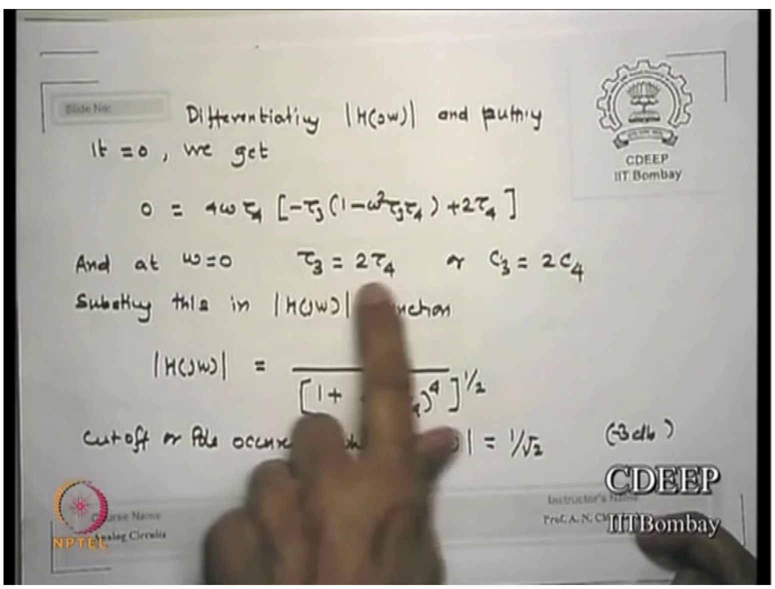

Okay, I can get a condition tau 3 is tau T4 and C3 is 2C4 (FL) is that one clear. I differentiated that below that cutoff and I say at this point if I want no ripple then what should happen if I meet these conditions then I would say I am going to have a maximally flat situation is that correct. (FL) We can remove by actually choice off proper tau’s and C’s is that clear, all other C’s, C3 should be twice of C4 in our case.

If we substitute back in normal function the cutoff of this transfer function value will occur at 3 DB point. What do we 3 DB point means, normal in bode plot what is the 3 DB point is? The corner, we start right there 20 DB but in real life that is not the point where it started. Where a 3 DB below it starts falling that is a continuous curve it is Bode what did he a suggest that instead of that continuous curve you actually make two points and then you say it is your cutoff point.

So, you can actually see if I substitute here this is 1 upon root 2 DB at Omega = Omega T for a 1 upon T4 that is the corner frequency of the bode, is that correct.

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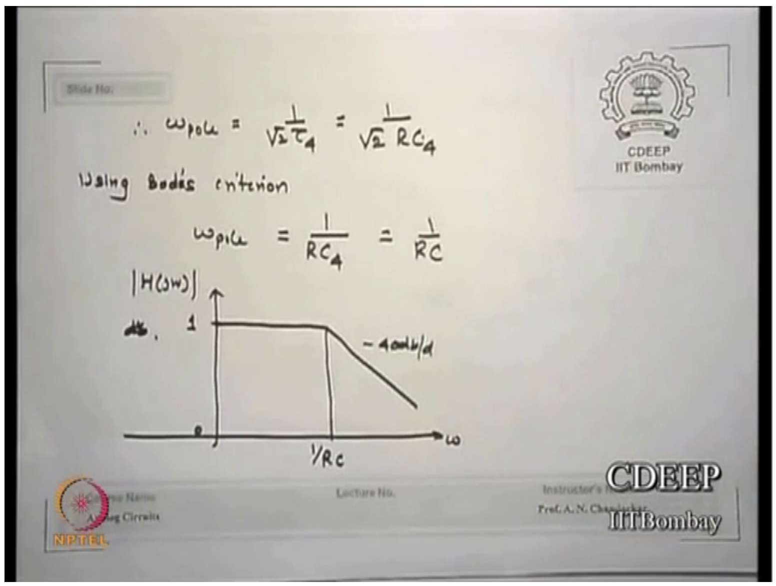

Say if I get the pole you can see this. (FL) 1 upon root T4 1 upon root 2 RC4 (FL) why we said except in real life actual curve will be something like this sorry and yet ready be is it not (FL) so, RC4 okay so one can see that I can create a low pass and how much is now fall will start from because I made that condition tau 1 is to tau 4, right here the gain will starts following by how many DB's (FL) under what condition under this condition the two poles are matching is that clear.

And the maximally flat situation attained so I got maximally flawed falling down by 40 DB. So, at least 20 DB (FL) and also what did I do? (FL) number of poles should keep on increasing that means number of sections will keep on increasing that is the problem with larger amount of money you spend. (FL) Now if I say I do not use that condition of maximally flat then I must say can I get sharpness at the lower number of poles and then I will call they are Chebyshev filters will come back to Butterworth in more details.

But this theory has to be understood what we are really trying the design of it is that point clear. Please remember only two filters will design low pass high pass okay. (FL) what is the all pass filter which we have not discussed maybe here we can quickly show you what is it about? Before we go to the next; (FL)

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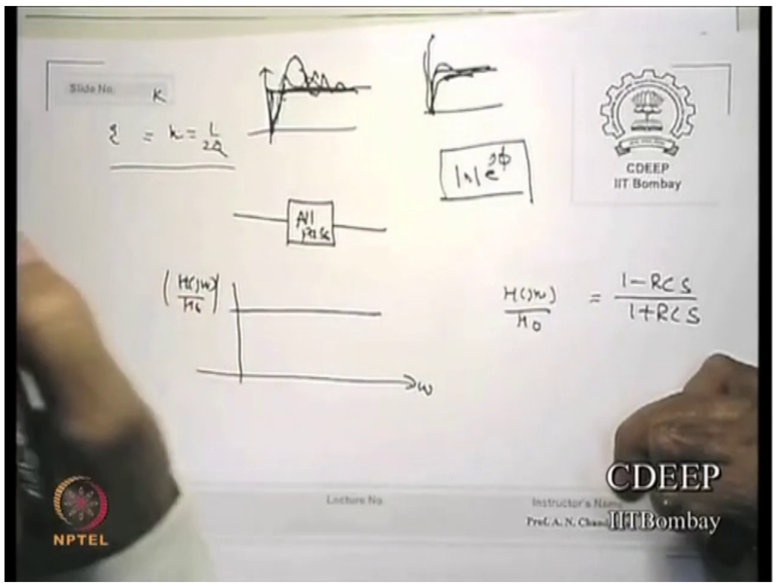

This is not the exact one what is it trying to tell you this transfer function (FL) said every frequency you get flat is that correct. However even if you see this function (FL) phase can vary depending on RC values phase can vary from 0 to 180 degree, so whenever the transfer function filters you create and you want a particular phase at the output between in and out then you connect the last stage of your filter can be all pass filter with proper phase requirements is that clear to you.

What is the advantage of this (FL) is that correct, so the next phase at the output compared to the input need not be just a product of their magnitude H1, H2, H3, H4 but also I can adjust my final phase with the input is that correct. Many requirements the output should mean should be either 180 degree or less than 180 for the stability case. So, we actually create the last stage as all pass filters okay though it does not change Your whatever H1, H2, H3, H4 broad everything it will pass anyway.

But it will give you a phase bit whatever phase difference between the final stages of filter to the actual stage where you are connecting. So, please remember the all pass filter as such words seems to be funny why are if you everything is passing what is being filtered okay. Nothing is actually what is phase is also; please remember any complex system will have something to the power J Phi magnitude into e to the power J Phi.

So, it is the magnitude may be one unity everywhere but the phase is not is that clear and that phase variations can be attained by proper choice of RC's is that clear this essentially is the trick of phase controls. We do something called phase control oscillators okay. We do use some of some such property there is that correct phase control oscillators okay.

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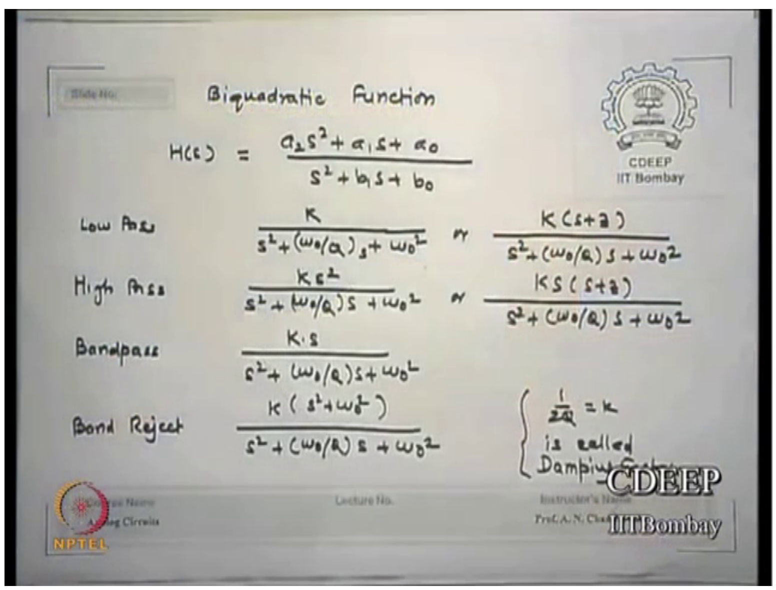

All the filters which we are going to talk are essentially in real life this can be a polynomial of n by N or M by n it can be H cube, H4, HN same way is just S0 is something more up to SN, (FL) so in filter we actually try to see whether both Butterworth or Chebyshev functions are bi quadratic nature is that correct. We are looking for bi quadratic functions, what is that now I am talking about is called implementation (FL)

So, we say okay let us look functions, if I have a bi quadratic function like this a2 S square + a1 S by a0 upon S square + b1 S + b0 this is also given in Sedra Smith and me, I can have I can be part of this function I can create into low pass, by pass, band pass ,high pass whatever it is. (FL) something called K, if that functions are K upon S square Omega 0 by QS is this + Omega 0 square have any time you heard of the word Q (FL) Omega l by R, 1 upon omega RC they are also quality.

Now this quality factor (FL) if I apply a step input to any amplifier any circuit the output essentially does not respond instantaneously is that one clear what did I say. If I give okay maybe that figure which I have I can still use if I apply a step input something like this, this is my input so the output does not respond immediately it starts rising and let us say trance function at the value 1 it should have actually reached here it did not.

So, what it did what is this word we call ringing okay ringing we start ringing of course it settles (FL) K which is the damping factor or in some books it will be zeta it will be called zeta. So, zeta is 1 upon 2, so what does that essentially means that ideally I want something to happen like this okay depending on higher value. So, you can see if I put a very high (FL) it may reach at asymptotic only to the actual. (FL)

So, I will have to adjust my Zeta so that very close to that value I will get flatter response is that clear. So, damping factor has something to do with the quality factor by a function which is this okay. So, if the transfer function has a nature of k upon S, S square Omega 0 by Omega there is the pole Omega R cutoff frequency into S sorry offer + Omega 0 is it quadratic S square + S + something.

So, it is a quadratic term numerator does not have any quadratic term constant (FL) however if the numerator has a term which contains S square because you run over multiple 0 at 0 multiple to 0 that 0 what the deviation (FL) in band pass instead of S square you now want afterward another cut off to occur (FL) you can do this analysis but if you have a function which is k dot remember the denominator is same everywhere.

And in the band reject you should cut off at some other points so it is S square + Omega 0 square (FL) so it will be band reject so if I can create by these two names I give Butterworth and Chebyshev I functions closer to this then I can say I am creating low pass, high pass, band pass, band reject. (FL) I say even this function should give me low pass why did I say so. It has a 0 okay so initial value can be pumped up okay but then after the pole actually starts falling down anyway. (FL) is that clear.

So, the; generally this we may not used but this is also possible in designs we always use these four functions got easiest to make okay and these are the four functions we will realize in either Butterworth form or in Chebyshev form, what are the advantage of that two Chebyshev gives me lesser number of poles circuit to create sharper fall at the cost I will get written. Butterworth will not give me any repel very little repel but it will take larger number of poles to get the same as a Chebyshev.

If I shall let us say five section but arithmetic seven sections or eight sections is that clear so look at the cost of money real cost I may create Butterworth even for higher poles but ideally I say I can choose either Chebyshev or Butterworth to my specifications. If this is my quadratic form let us take the low pass one.

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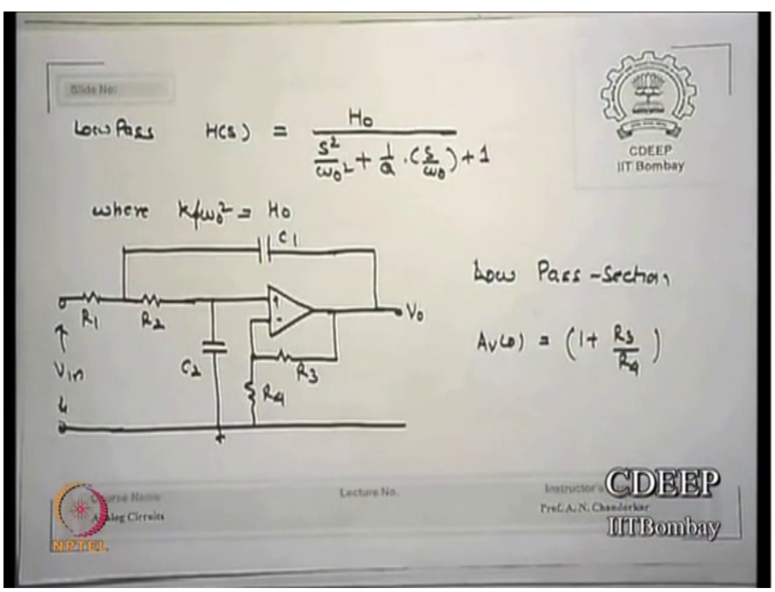

It is H0 upon some constant K (FL) so, it is H0 S square upon Omega 0 square 1 upon QS upon Omega 0 + 1 (FL) this circuit has an amplifier here is that correct R3 R4 (FL) is that point clear (FL) it follows the similar pattern then what do we say bi-quadratic form (FL) is that clear there is saying no even buffer will do the same as well as this buffer. (FL) Even OPAMP does the same thing what buffer does the only difference between buffer and OPAMP is that we do not want to use any additional component.

Because I do not need gain there, if I need gain I will put resistors. So, essentially if R is shorted here I am going to get a unity gain that is what I am saying. Even in OPAMP that R0 still lower only thing is now it will be a function of these values there it with the intrinsic value is that the difference clear. Now R3 R4 will also affect that R0 value is that clear. So, R0 will not be same as what we naturally device was giving it will get modified by these feedback factors is that correct.

So, there is slight difference but the nature is same as you know R0 will go down and RI will increase will remain same independent of that is that clear to you. Only thing we are there, we are not interested in actually getting gain out of it we only want resistor transfers natural resistance is available so put a buffer in all unit again follower okay. (FL)

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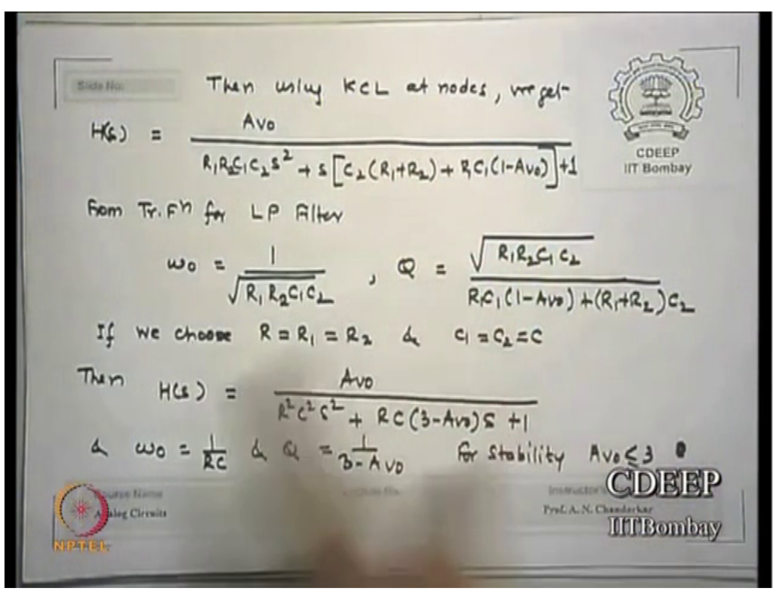

R1 R2 C1 C2 S square + S time bigger bracket C2 + C2 into R1 R2 + R1 C1 into 1 - AV 0 bracket close + 1, let us say R1 and R2 are same and that is R and let us say C1 and C2 is also same which is C okay. Then I get this time function is R square C square S square RC 3 - AV 0 S + 1 (FL) what is this why I am doing this please look at my transfer function for loop S square Omega 0 by Q into S + Omega 0 square Y (FL)

If I create Q is equal to this and Omega 0 equal to this then I am actually doing bi quad is that clear. If I choose my Q which is under root R1 R2 C1 C2 divided by this R1 C1 and I choose my Omega 0 is 1 upon R1 R2 C1 C2 under root of that then I am essentially converting this function into S square + Omega 0 by Q S + Omega 0 square which is my bi quad transfer function for low pass is that clear. (FL)

I converted it equivalently in this form this we already said by theory now if you can see the expression between the two is identical. They are identical only thing the values correspondingly should be like this if these values are there these two functions are identical is that correct. That means with these values this function will represent a low pass filter is that clear. You see this but bi quad function which is this s square Omega.

This we have already proved by theory this is bi quad though poles (FL) please remember what did I do I had to implement this function. So, I said okay here is a circuit with me I tried this circuit by my earlier theory also but I know I put this circuit I evaluated this transfer function is that okay. Then I say if I compare with my bi quad function I get these values and if I get these values that this function and that bi quad function is identical is that correct.

Since that when a low pass filter this also should do me a low pass action is that clear with cutoff frequency now how much 1 upon RC R square C square means 1 upon RC and Q is RC divided by this much is that clear. So, if I get this the cutoff frequency is 1 upon RC, Q is 1 - 3 upon AV0 and for this can you think why I put this condition Q is AV 0 should be less than 3 what will happen if AB0 is greater than 3.

What will happen look at this function (FL) is that now condition clear this low pass filter will keep acting as a low pass as long as AV 0 is less than 3, AV 0 is decided I bought what is AV0 value there 1 + R2 by R1 or which is a - R3 by R2 R3 by R2 (FL) ratio should never be greater than 3, is that correct you should be never be greater than 3. If that value is never greater than 3 then stability will achieve.

This function will implement a low pass filter is that correct at this cutoff. What is the fall it will start at this cut off a double pole (FL) is that correct (FL)

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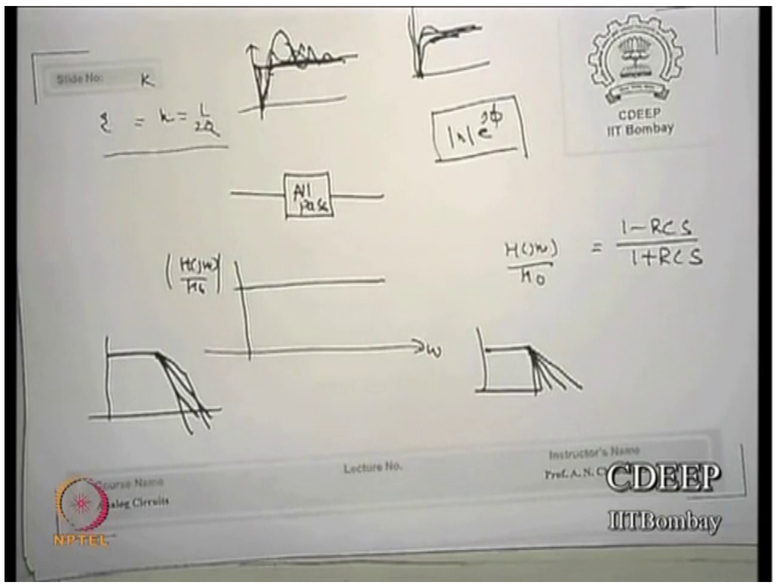

What I am trying is (FL) then I am not sure whether face abilities guarantee, so what make (FL) is that clear I want ideally this so (FL) I want to come as close to the ideal value (FL) is that clear.

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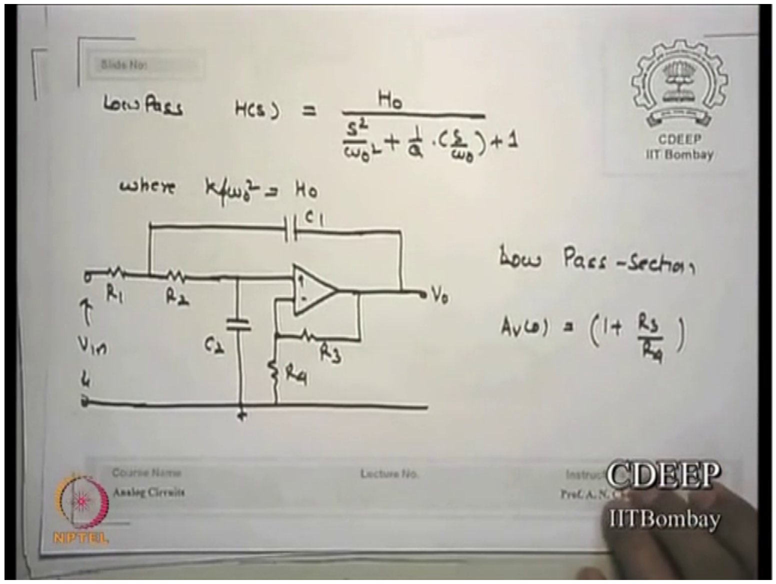

So, definition (FL) is that correct this will become an high pass okay (FL) is that clear (FL) some other day I show you in a given circuit how 0s are introduced. I already explained you how 0 is (FL) because if you are two parts out of phase at a given frequency they have same value then the 0 is created output goes to 0 is that correct that condition always can be created by proper choices of frequency as well as the components you use. (FL)

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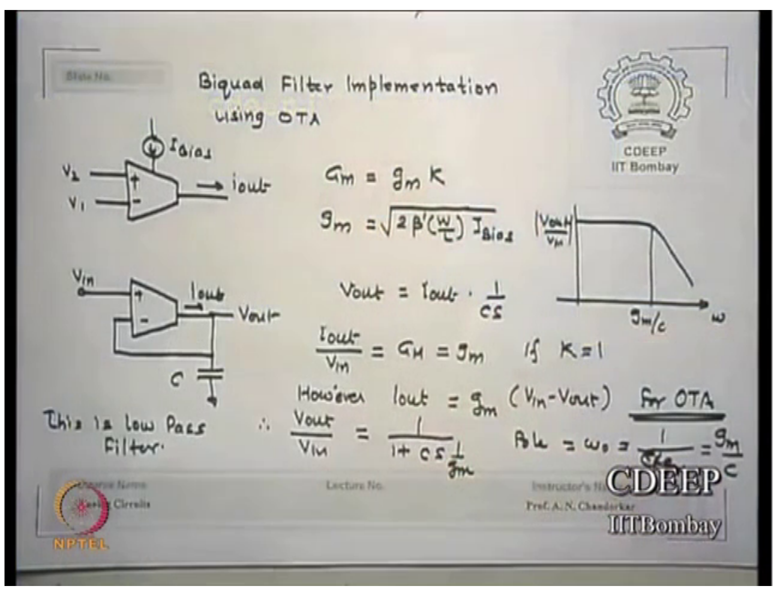

This is another which is an integrated circuit filter (FL) what is the difference between operational amplifier and operational transconductance amplifier (FL) see if I were an amplifier whose output current is controlled by input signal then I have a operational transconductance amplifier typically OPAMP (FL) such is called OTA, operational transconductance amplifier forget about what is inside this.

Just keep properties (FL) what is the difference you see (FL) okay so you have a different signal of V2 - V1 or V1 - V2 and there is an output current and there is a bias current called I bias (FL) K is a multiplying factor so I can improve the transconductance by this K factor okay. (FL) K is essentially a size factor okay if amplifier (FL) so, this is the function (FL) so what is V out current; please remember is any current can go here, no current can enter OPAMP.

So, whatever is coming here must go through the capacitance is that point clear no current can enter OPAMP (FL) can only pass through capacitor as it cannot get into this input OPAMP okay. (FL) so I out into 1 upon CS is my V out, so I out (FL) V in is capital GM which is what the transconductance amplifier give I assume right now K is 1 okay. So, GM is GM however I out by transconductance (FL)

So, Vin - V out is the difference potential so GM times Vd is the actual current which is coming out for collect the terms (FL) So, Vout by V in is 1 upon CS upon 1 upon GM (FL) so when 1 upon GM is resistance, so 1 upon RCS low pass filter (FL) 1 up 1 + RCS pole is ID 1 upon R C but RCs is 1 upon GM, so GM upon C this is the pole is that correct. (FL) Let us say I want there low pass which has a very high cutoff frequency.

So, what does that mean that is a filter as a bandwidth of 10 megahertz 400 megahertz (FL) C should be small but C can be created in a silicon circuit whatever smallest point 0 one purple someone femtofarad, tens of femtofarad (FL) is that correct why GNC filters are not very, very popular. So, they are used where they will use less than a megahertz cutoff say I have 100 of kilo Hertz, 500 kilo Hertz yes simple filter uses. (FL)

So, you have to now worry in a chip how much power I will be given to dissipate what is the smallest value of capacitor I can create during whatever processing I have and therefore what is the maximum size I will be allowed to because the area constrained. So, maximum frequencies can never be greater than a megahertz is that correct. Why megahertz (FL) whatever number we can create that is Math's. (FL)

But in reality such OTA filters can be used only for limited frequency range up to a megahertz maximum preferably 500 kilohertz (FL) so, it is difficult to reduce C, net C difficult increase GM there we say GM by C is limited to a megahertz up literally 500 megahertz. Most of the low frequency continuous filters are OTA filters. Most of the low frequency up to a 100 kilos guaranteedly it works fantastic.

So, GMC filter whatever advantage on the low power relatively very few components so very small chip, very cheap chip okay and can filter it up to few kilos hertz, 100 kilos hertz. These filters are extensively used in almost every hardware where audio signal frequencies are in nature. Whatever system you are looking mostly an audio then these filters are ideal. (FL) is that correct, is that correct. This is also a bi quad because it also uses the same low pass function. (FL)

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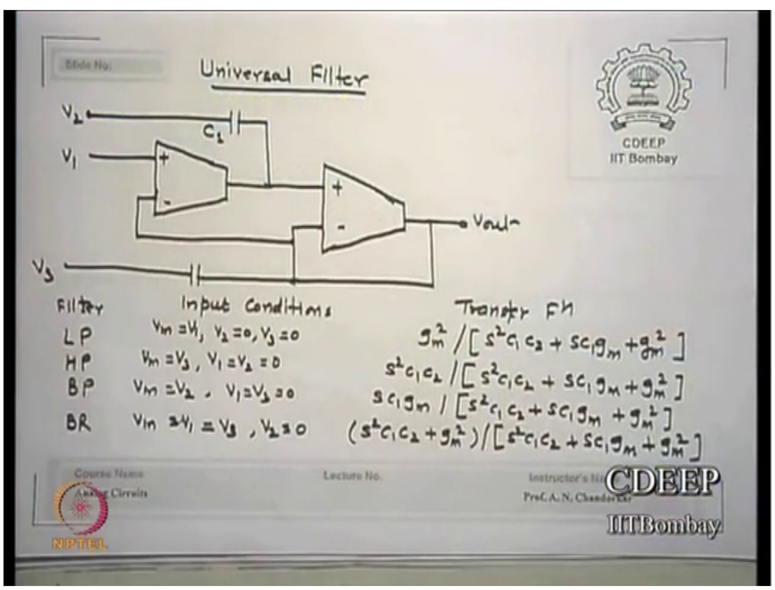

Two OTA’s and two capacitors can be manipulated by different signals at V1, V2, V3, 01, 01 (FL) and you can create all four filters out of that we will come back to it next. I will start on this just want to show you (FL) but none of them will be good (FL) just to show you that same block can be utilized to make universal.