

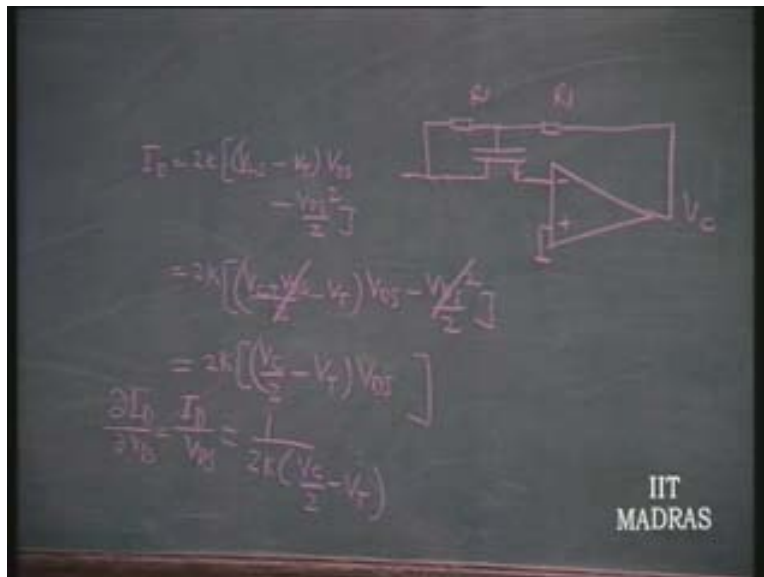
**Analog ICs**  
**Prof. K. Radhakrishna Rao**  
**Department of Electrical Engineering**  
**Indian Institute of Technology, Madras**  
**Lecture - 23**  
**Self Tuned Filter**

So, in the last few classes we had discussed about variety of techniques of designing multipliers. A multiplier is nothing but a voltage controlled amplifier which can also be designed by introducing MOSFET as a resistor determining the gain of the stage.

We had discussed earlier how MOSFET in the triode region can be used as a linear resistor. Therefore whenever we would like to design multipliers we can also take **recourse** to this technique of designing multipliers. There is one important technique of multiplication which uses a very novel method for the purpose of multiplication.

Let us consider this; I am using FET as a voltage dependent resistor. Let us say we can put the linearizing blocks here so that it is a linear voltage dependent resistor because the characteristic for this FET is in the triode region  $I_D$  is equal to  $2K$  times  $V_{GS}$  minus  $V_T$  into  $V_{DS}$  minus  $V_{DS}$  square by 2. If I now make this as the control voltage then we know that this is equal to  $2K$  into  $V_c$  plus  $V_{DS}$  by 2 this voltage is  $V_c$  plus  $V_{DS}$  by 2 minus  $V_T$  into  $V_{DS}$  minus  $V_{DS}$  square by 2 so you can see how nicely it gets linearized. We also have found out ways and means of doing this without the help of resistors also.

(Refer Slide Time: 00:05:00)



So in any case this demonstrates a particular method of making the FET linear, the  $V_{DS}$ .

(Refer Slide Time: 00:05:21)

Handwritten derivation on a chalkboard showing the drain current  $I_D$  and its derivative with respect to  $V_{GS}$ . The equations are:

$$I_D = 2k \left[ (V_{GS} - V_T) V_{DS} - \frac{V_{DS}^2}{2} \right]$$

$$= 2k \left[ \left( \frac{V_C}{2} - V_T \right) V_{DS} - \frac{V_{DS}^2}{2} \right]$$

$$= 2k \left[ \left( \frac{V_C}{2} - V_T \right) V_{DS} \right]$$

$$\frac{\partial I_D}{\partial V_{GS}} = \frac{I_D}{V_{DS}} = 2k \left[ \frac{V_C}{2} - V_T \right]$$

The IIT MADRAS logo is visible in the bottom right corner.

So we get  $\Delta I_D$  by  $\Delta V_{DS}$  is same as  $I_D$  by  $V_{DS}$  is equal to  $2K(V_C \text{ by } 2 \text{ minus } V_T)$  or  $R$  is equal to  $1$  by  $2K(V_C \text{ by } 2 \text{ minus } V_T)$  is equal to  $1$  by  $K(V_C \text{ minus } 2V_T)$ .

(Refer Slide Time: 05:41)

Handwritten derivation on a chalkboard showing the output resistance  $R_k$  and its derivative with respect to  $V_{GS}$ . The equations are:

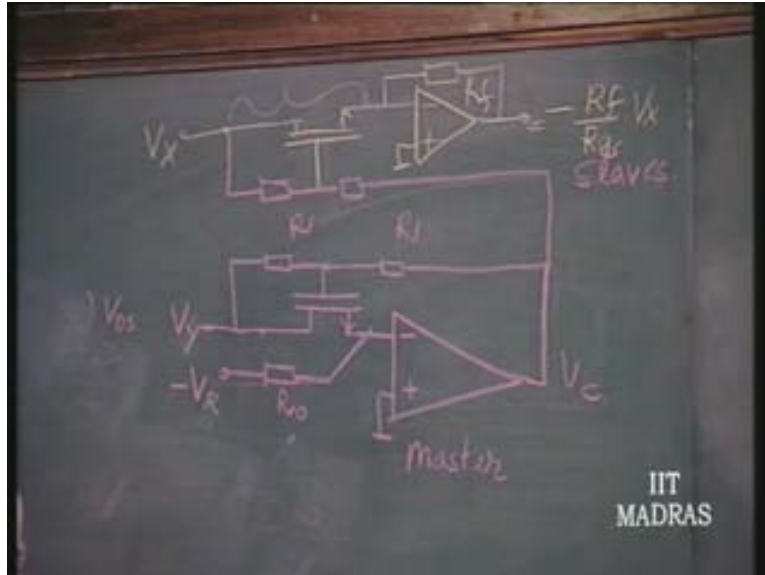
$$R_k = \frac{1}{2k \left( \frac{V_C}{2} - V_T \right)}$$

$$= \frac{1}{k(V_C - 2V_T)}$$

The IIT MADRAS logo is visible in the bottom right corner.

So how do I establish this resistance that I want to be simulated?  
That can be done by feedback.

(Refer Slide Time: 00:11:02)



Now what I do is, I put a resistance here; I make this  $R_0$  reference resistance and apply a voltage minus  $V$  here and  $V$  here. So this is  $V_i$ , I put minus  $V_i$  here then what happens here, this resistance has to automatically adjust itself because of the negative feedback such that this current which is  $V_i$  by  $R_0$  is same as this current. That means this  $R_{ds}$  now is going to be automatically adjusted to be equal to  $R_0$ . So this is a negative feedback circuit which will adjust the FET at all times irrespective of its value of  $K$  and  $V_T$ . That means even if  $K$  and  $V_T$  were dependent upon temperature this control circuit will see to it that FET resistance is exactly adjusted to  $R_0$  at all times. That means now you have got a perfectly temperature compensated FET resistance which remains absolutely constant as constant as this fixed value of resistor.

Now, for example, you call this as a master then this master says, I will make sure that, you FET behave yourself properly so that the resistance value you simulate is constant. Now once it is made sure then I can use this same control voltage for all the slaves. FETs which are exactly identical and the circuit which is exactly identical, these are all slaves any number of such slaves you can have and all of which get the same control voltage.

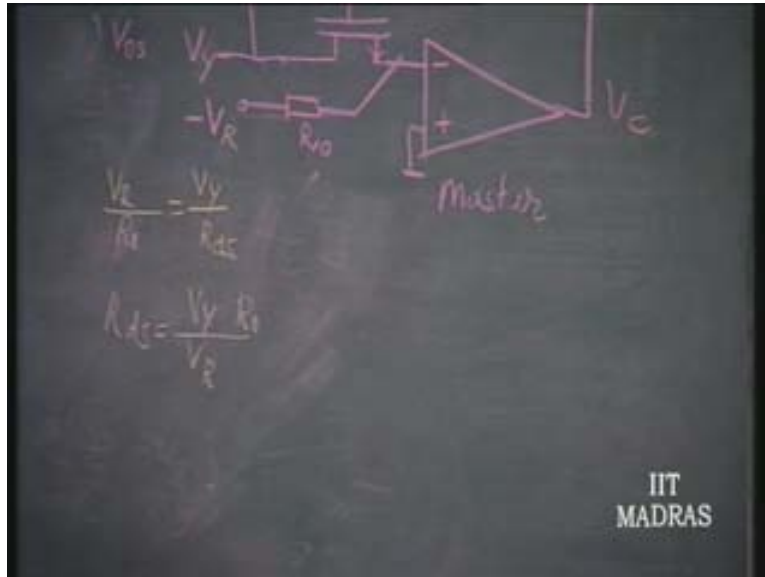
You can independently feed any voltage here, let us called this as  $V_X$  and put a negative feedback resistance so that this is nothing but an inverting amplifier designed using the FET resistance which has been made to be constant by another independent negative feedback circuit so it will remain constant forever. Let us therefore have this as  $R_f$  feedback and this resistance that it is going to simulate is going to be  $R_0$  so the gain of this is minus  $R_f$  by  $R_0$  into  $V_X$ . And  $R_0$  itself is nothing but  $R_{ds}$ , this has been made equal.

So let us see what this  $R_0$  is?

This  $R_0$  is determined by in fact the voltage  $V_i$ . I do not have to make this and this is the same. So what I will do is I will keep this as  $V_R$  and keep this as  $V_Y$ . Now we know that this is always correct, the  $R_{ds}$  of this.  $R_{ds}$  of this is as same as  $R_{ds}$  of this. Earlier when I

had these two as the same  $R_{ds}$  was same as  $R_0$ . Now I can write down the equation here,  $V_R$  by  $R_0$  is the current drawn and that should be same as equal to  $V_Y$  is positive and  $V_R$  is positive if it is so it is  $V_Y$  by  $R_{ds}$ .

(Refer Slide Time: 00:12:11)

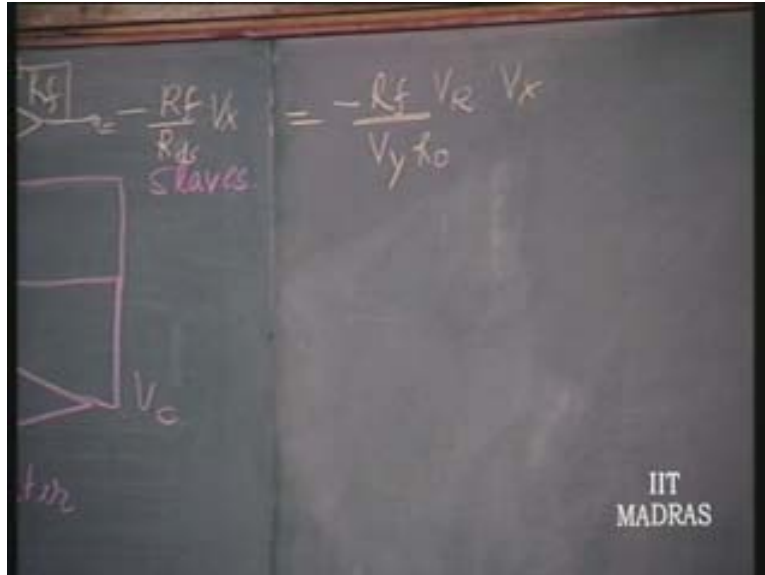


It is necessary that  $V_Y$  can only take on the positive polarity in this case if this is my negative. If this is positive this has to be negative, this is the requirement. Under this situation  $V_R$  by  $R_0$  is equal to  $V_Y$  by  $R_{ds}$  or this can be written as  $R_{ds}$  you replace  $R_{ds}$  is equal to  $V_Y V_R$  by  $R_0$ . If you now replace this  $R_f$  by  $R_{ds}$  which is  $V_Y$  into  $V_R$  by  $R_0$  into  $V_X$  this becomes a divider in this case.

You want to make a multiplier out of it what do you do?

You interchange  $V_R$  and  $V_Y$ . If you interchange  $V_R$  and  $V_Y$  this you make as a reference voltage, this you make it as variable voltage then you get a multiplier. You can design either a multiplier or a divider simply by using this principle of master slave concept which is similar to the current mirror. In the current mirror we had a transistor which was connected as a diode so that is a feedback structure which was learning to develop the voltage required to pass a current and this voltage was used to bias all the other transistors at the same current.

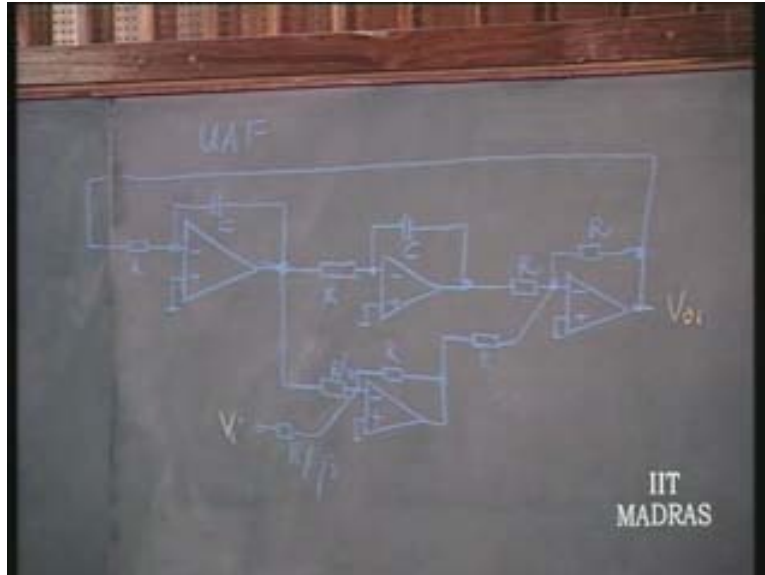
(Refer Slide Time: 00:12:29)



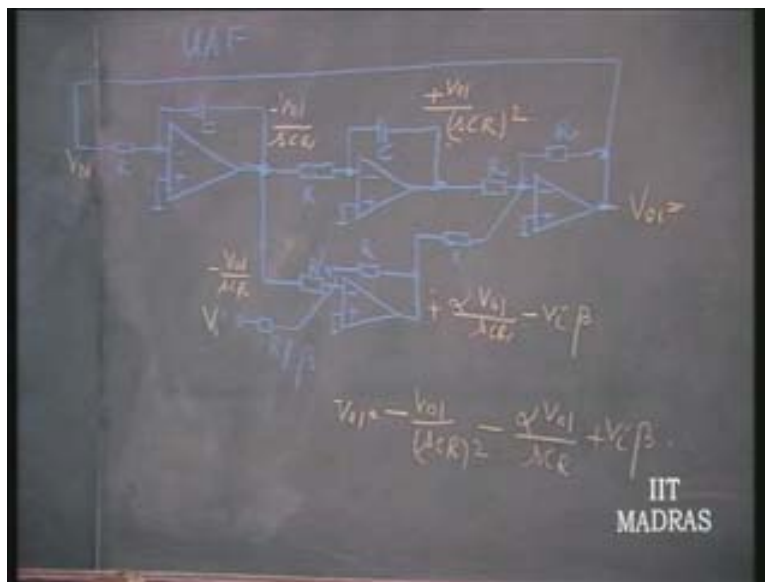
So the same thing is used here in the design of a FET multiplier. In this it is not necessary to use voltage ampere characteristics of the FET. Voltage ampere characteristic of the FET is of no consequence strictly speaking. This relationship of the equal resistance being valid at all times is true irrespective of the volt ampere characteristics. The only thing is that here I am assuming that this current is the current that is passing through this and that current is determined by this divided by that. In that assumption I am considering that the FET is linear. So this is another technique of designing multipliers. Particularly this technique is very popular in regular laboratories where you can use two matched FETs to design a fairly good multiplier. This is not four quadrants but two quadrants because  $V_x$  can take on two polarities but  $V_y$  can take on only one polarity.

Now we will see another application of this multiplier in what is called a very important block that is universal active filter block. Universal active filter blocks are available as ICs. Now-a-days these are being used as VLSI sub cells in realizing higher order analog filters. So this itself can become a sub cell of a very large scale integrated circuit. Therefore we would like to have some control over these filter parameters. Let us therefore quickly analyze this filter.

(Refer Slide Time: 00:16:13)



(Refer Slide Time: 00:17:49)



Let us see if  $V_i$  is the input applied here we can see that we will call this output as  $V_{o1}$  and let us express everything in terms of  $V_i$  and  $V_{o1}$ . This is  $V_{o1}$  and therefore this output is  $V_{o1}$  by sCR with a negative sign and this output is minus  $V_{o1}$  because plus  $V_{o1}$  by (sCR) square and here we have minus  $V_{o1}$  by sCR and  $V_i$ . So this  $V_{o1}$  by sCR will appear in this case as alpha times plus alpha times  $V_{o1}$  by sCR minus  $V_i$  into beta. That is as far as this is concerned. Therefore  $V_{o1}$  has output resulting due to these two inputs so  $V_{o1}$  is going to be equal to, this will appear as an inverting thing, so  $V_{o1}$  is minus  $V_{o1}$  by (sCR) square and then minus of this.

(Refer Slide Time: 00:18:51)

$$\frac{V_{o1}}{V_i} = \frac{\beta}{\left[1 + \frac{\alpha}{sCR} + \frac{1}{(sCR)^2}\right]}$$

$$= \frac{(sCR)^2 \beta}{[(sCR)^2 + \alpha sCR + 1]}$$

$$= \frac{\left(\frac{\Delta}{\omega_0}\right)^2}{\left(\frac{\Delta}{\omega_0}\right)^2 + \frac{\Delta}{\omega_0 Q} + 1}$$

IIT MADRAS

We have quickly seen how this can be simply analyzed and we get  $V_{o1}$  by  $V_i$  from that equal to beta (1 plus alpha by sCR plus 1 by sCR) square or [(sCR) square beta by (sCR) square alpha sCR plus 1] which according to us can be written as s by omega<sub>0</sub> square s by omega<sub>0</sub> square plus s by omega<sub>0</sub> Q plus 1. Wherein omega<sub>0</sub> is equal to 1 by CR and Q is equal to 1 by alpha by comparison. And obviously here we have beta coming into picture in the gain of the stage. This is what is called the high pass output.

(Refer Slide Time: 00:19:25)

$$\omega_0 = \frac{1}{CR}$$

$$Q = \frac{1}{\alpha}$$

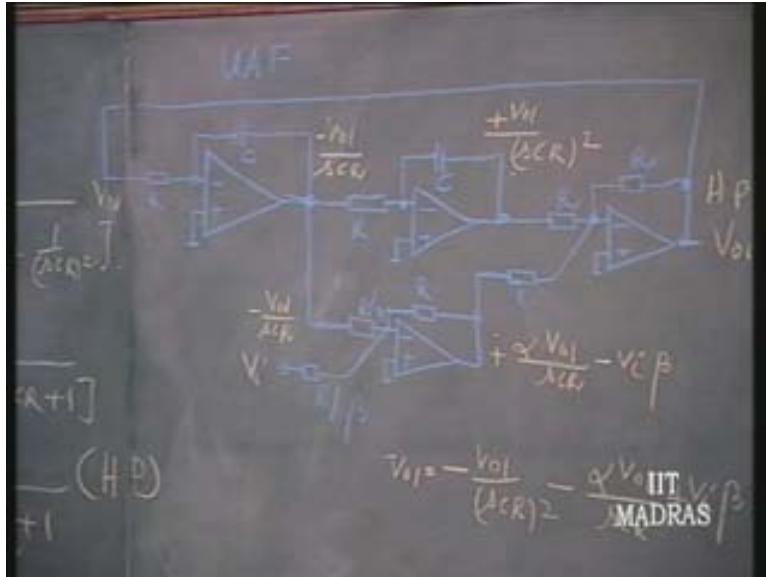
$$\frac{V_{o1}}{V_i} = \frac{\beta}{\left[1 + \frac{\alpha}{sCR} + \frac{1}{(sCR)^2}\right]}$$

$$= \frac{(sCR)^2 \beta}{[(sCR)^2 + \alpha sCR + 1]}$$

$$= \frac{\left(\frac{\Delta}{\omega_0}\right)^2 \beta}{\left(\frac{\Delta}{\omega_0}\right)^2 + \frac{\Delta}{\omega_0 Q} + 1} \quad (HP)$$

IIT MADRAS

(Refer Slide Time: 00:19:39)



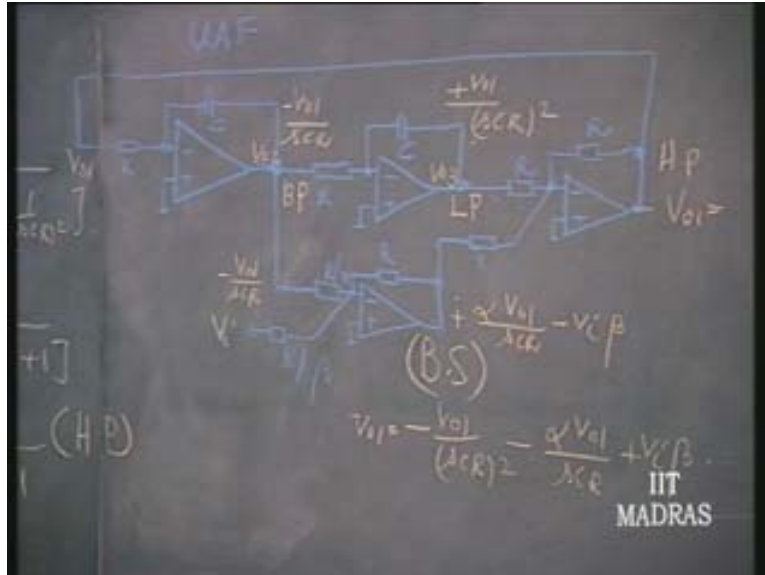
If you take the output here this is then the high pass and that high pass output into 1 by  $sCR$  or  $\omega\omega_0$  by  $s$  minus which will give you band pass. So  $V_{02}$  by  $V_i$  is nothing but minus  $s$  by  $\omega\omega_0$  beta divided by the same denominator which is band pass.

(Refer Slide Time: 00:20:28)





(Refer Slide Time: 00:21:05)



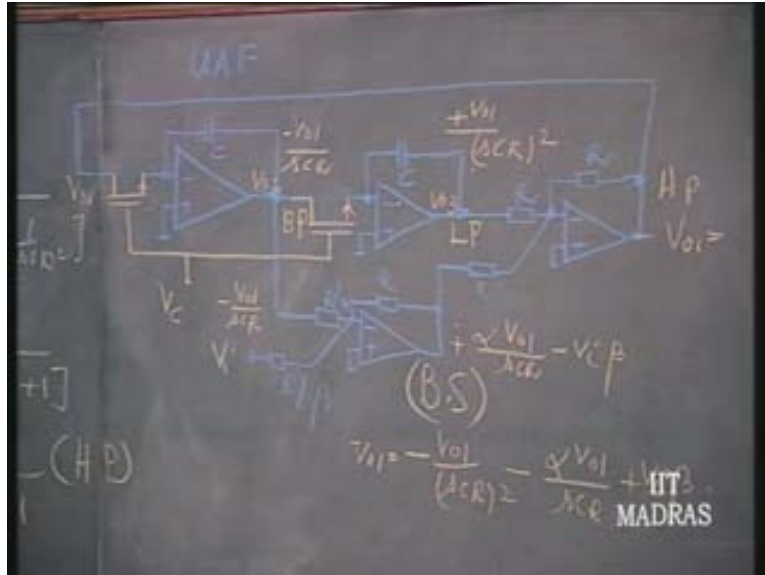
So this output is band pass. And again if you integrate this you will get low pass which is  $V_{03}$ , this will be  $V_{02}$  and this will be  $V_{03}$ . So  $V_{03}$  by  $V_i$  which is again minus beta divided by the same denominator which is low pass. There is some universal nature about it because band pass, high pass, low pass if you can add or subtract suitably you can get any second order system.

Any second order system block can be got by adding suitably or subtracting suitably these outputs. Therefore you might need one more op amp for doing this. If you have here an op amp this output is going to be band stop output. It is nothing but alpha times  $V_{01}$  by  $sCR$  minus  $V_i$  beta, it will be nothing but band stop output. That is why this circuit is very important because we can just use it for even biomedical applications where you want an ECG waveform which contains lot of your 50Hz, the output of this is going to be the same ECG waveform devoid of the 50Hz. So this is a nice setup for that kind of filtering in biomedical applications.

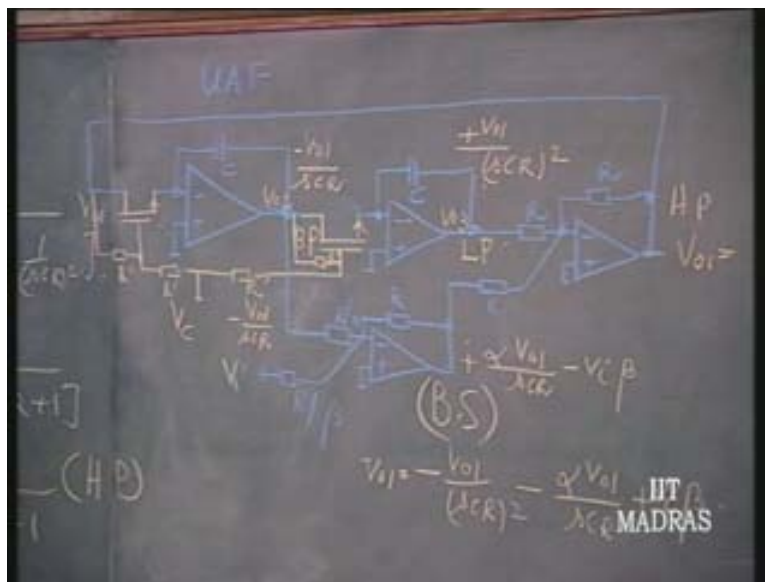
Let us now see how this can be converted into voltage controlled filter. We can do this very simply by replacing this integrator and this integrator because here  $R_C$  times constant means the 1 over frequency, omega is equal to 1 by  $R_C$ . So, if I make both these resistances depend upon a voltage that means if this is made voltage control or if I put a multiplier here it becomes voltage controlled filter.

Instead of that I can just say that it can be made using a FET here and using a FET here. If I use this kind of an arrangement it is not linearized, if you want to linearize you can do it by putting resistors. So  $R, R'$  let us put it to distinguish from this resistance  $R'$  prime linearizing resistors and control voltage is connected here. Now this will be nothing but same as omega<sub>0</sub> is equal to 1 by  $CR$ 's, the  $Q$  remains the same as 1 by alpha undisturbed and the gain also remains undisturbed so you are now able to change omega<sub>0</sub> dependent upon  $R_{ds}$ .  $R_{ds}$  is equal to 1 by  $2K$  or  $K$  into  $V_c$  minus  $2V_T$ .

(Refer Slide Time: 00:22:03)



(Refer Slide Time: 00:23:46)



Using these FETs which we had linearized we can make this particular filter voltage controlled filter.

(Refer Slide Time: 00:24:31)

The image shows handwritten mathematical equations on a chalkboard. The equations are:

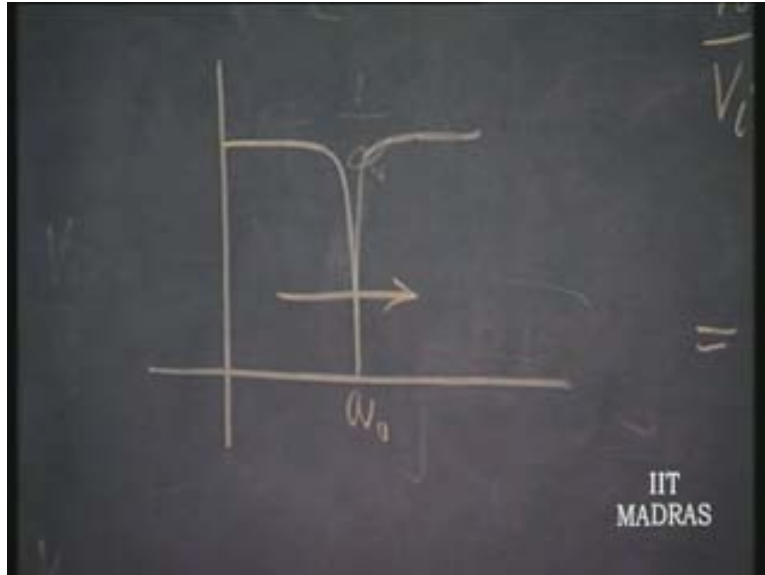
$$\omega_0 = \frac{-K(V_c - 2V_T)}{C}$$
$$Q = \frac{1}{\alpha}$$
$$\frac{V_{o2}}{V_i} = \frac{-\frac{\Delta}{\omega_0} / \beta}{[ \quad ]} \text{ B.P.} = \frac{(s\omega_0)}{[ \quad ]}$$

In the bottom right corner of the chalkboard, the text "IIT MADRAS" is visible.

This works beautifully as long as the resistance of the FET can be varied from a certain value  $V_{c1}$  to some  $V_{c2}$  depending upon the current that you are permitted to have through the FET over this range. Now this linear voltage controlled filter can be tuned based on this control voltage. For example, let us take the same example, if it is some specific frequency you want to get rid of you can make  $Q$  fairly high to 100 or so. Therefore you can now obtain a characteristic here if you take the output at this point where the output is going to have a notch at  $\omega_0$  which can be varied. So you can get rid of the unwanted frequency.

Another application for such a thing is, you have a given wave form which is periodic and you want to find out the harmonic content in the given waveform. So you can put a notch at the fundamental and measure the power at rest of the output. So that will give you the harmonic content of the given waveform. Or if you take the band pass output you can find out the fundamental or the harmonics so it can become a spectrum analyzer. So, it can be used as a harmonic analyzer or as a spectrum analyzer the sweep voltage being given as  $V_c$ . So it has considerable amount of application in signal processing.

(Refer Slide Time: 00:25:47)

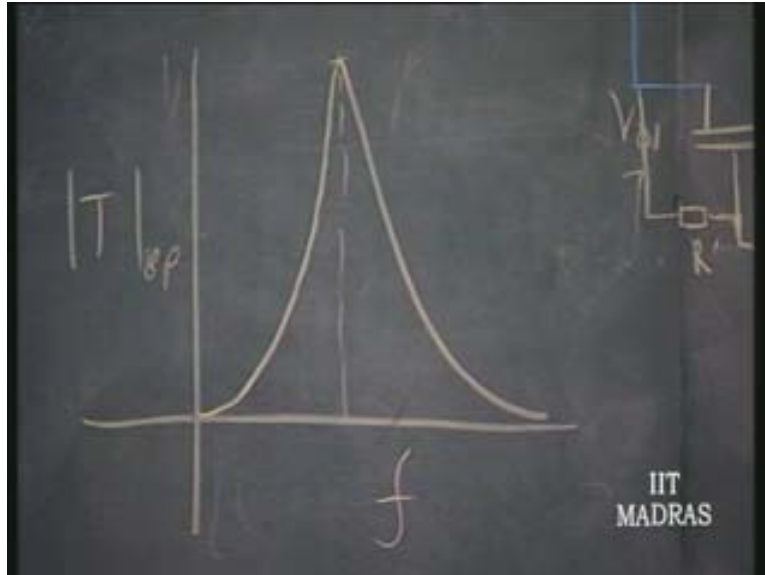


Now I want to make this filter self tuned, what does it mean?

I am feeding a certain frequency here let us say 50Hz then I want it to tune itself to 50Hz. If I change this frequency to 200Hz its center frequency or its pole frequency should go to 200Hz if it is band pass or not or whatever it is. So I want the pole frequency to be now tuned automatically.

Let us give this as a problem on tuning for a student; this is given to you in the lab. You are given this frequency as the incoming frequency or reference frequency then you are asked to tune this filter to the incoming frequency. Obviously we will change the  $V_c$ . We might take any of the outputs for finding out whether it is tuned to the incoming frequency or 0.

(Refer Slide Time: 00:30:07)



Now in this particular case, suppose you take the band pass output, how will you know that it is tuned? How will you know that it is tuned to the incoming frequency 50Hz or 100Hz how will you know?

Here gain is determined by beta, what is the band pass output characteristic?

It will have a peak. Peak mathematically means the magnitude does not vary with respect to frequency band pass. Whereas in order to find out whether around that frequency it is tuned or not then something has to vary by a great extent. This is something that even though is visible for an IQ circuit this is visible as a technique of tuning, this is the wrong method of tuning whereas we should make use of another parameter that is phase variation with respect to frequency.

How does phase vary with respect to frequency for a band pass?

Let us have these characteristics  $s$  by  $\omega_0$  minus  $\beta$   $s$  by  $\omega_0$  square  $s$  by  $\omega_0$   $Q$  plus 1. The phase of this is 90 numerator and the denominator is the one which causes phase variation with respect to frequency which is nothing but  $\tan^{-1} \frac{\omega}{\omega_0} \frac{1}{Q}$  by  $1 - \frac{\omega^2}{\omega_0^2}$ .

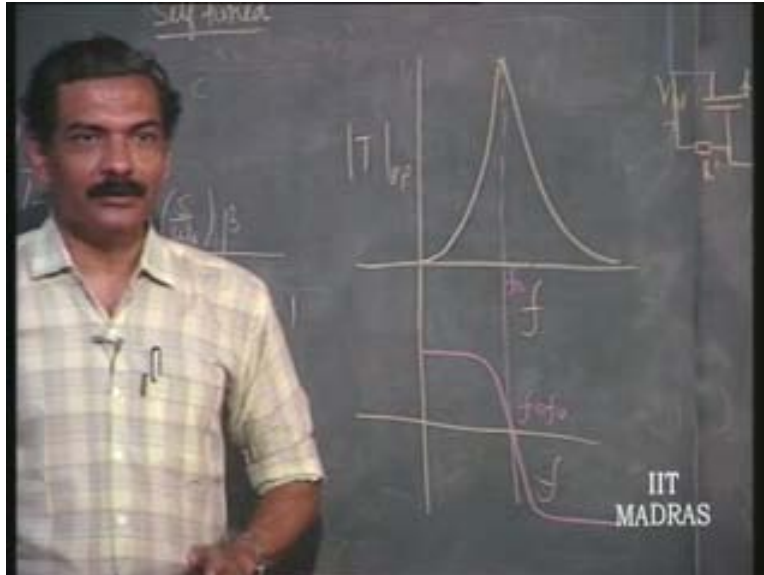
(Refer Slide Time: 00:32:29)

The image shows a chalkboard with handwritten mathematical expressions. The top part shows the phase shift  $\phi_{out} = \tan^{-1} \frac{\omega}{\omega_0 S}$  over  $1 - (\frac{\omega}{\omega_0})^2$ . Below this, the derivative  $\frac{d\phi}{d\omega}$  is shown to be equal to  $Q$  at  $\omega = \omega_0$ . The expression for  $Q$  is given as  $-\frac{(\frac{S}{\omega_0}) \beta}{(\frac{S}{\omega_0})^2 + \frac{S}{\omega_0} + 1}$ . The IIT MADRAS logo is visible in the bottom right corner of the chalkboard image.

That is the phase variation with respect to frequency. And if you maximize this find out  $\Delta \phi$  by  $\Delta \omega$  and you will find that this will be maximum at  $\omega$  is equal to  $\omega_0$  and is going to be equal to, at  $\omega_0$  phase shift is  $S$  and  $S$  gets cancelled, this is going so phase shift is 0 or 180 degrees in this case it is 180 degrees. Phase shift at the resonant frequency is 0 that is known to you for the band pass output 0 or 180 degrees. But phase variation is directly proportional to  $Q$ .

What is  $\Delta \phi$  by  $\Delta \omega$  at  $\omega$  is equal to  $\omega_0$ ? Find out that it is directly proportional to  $Q$  of the circuit. That means if I plot the phase variation initially the phase shift is starting at minus 90 or whatever and going over to 0 going to plus 90 and it is this slope that we are talking of which is  $\Delta \phi$  by  $\Delta \omega$  and that is highest at  $\omega$  is equal to  $\omega_0$  and is dependent upon  $Q$ . Higher the  $Q$  steeper this is going to be. Therefore if you want to check whether you have exactly located your center frequency or not you have to find out the phase difference because the phase is going to vary the maximum around that frequency whereas the magnitude is not going to vary at all by definition because it is flat because it is the maximum.

(Refer Slide Time: 00:33:14)

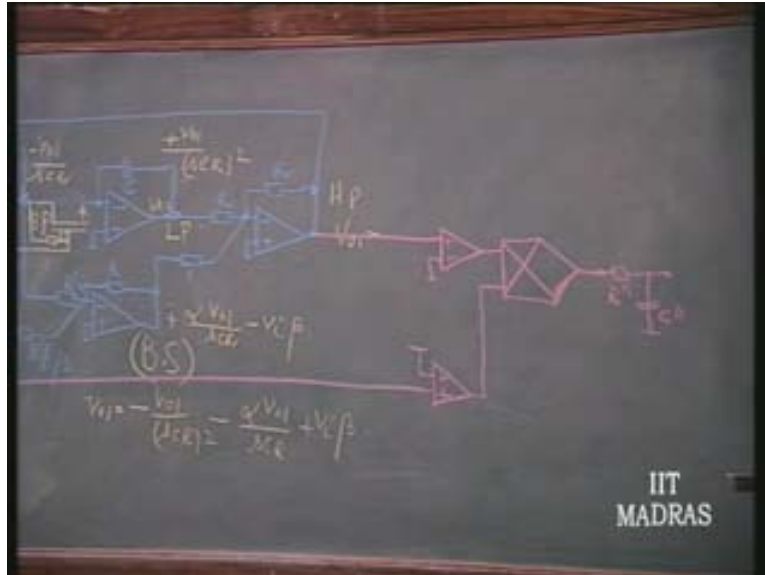


Phase variation is maximum, in any circuit therefore if you want to adjust the frequency exactly you must use always the phase variation with respect to frequency and not the magnitude. So, phase detector is invariably used and not magnitude detector. That means I need a phase detector, let us now therefore use a phase detector, but then the phase detector that we used was capable of detecting only phase not from 0 but from 90 degrees.

The multiplier type of phase detector we used earlier or AND OR gate phase detector was not capable of detecting phase that is leading and lagging phase if it is starting from 0 we had a quiescent phase shift of 90 degrees. That means it is advisable not to use the output of the band pass but we must use additional phase shift which is already available to us. Therefore if you use additional phase shift of 90 degrees that means either low pass or band pass in this structure we can use it in combination with our phase detector to find out how exactly I have to shift the frequency in order to tune it exactly to the incoming frequency.

In a given case if you have only band pass output available you must shift it by additional phase of 90 degrees or put an integrator or differentiator. But in a given circuit if you have low pass and high pass available or high pass available then you can take the output from these.

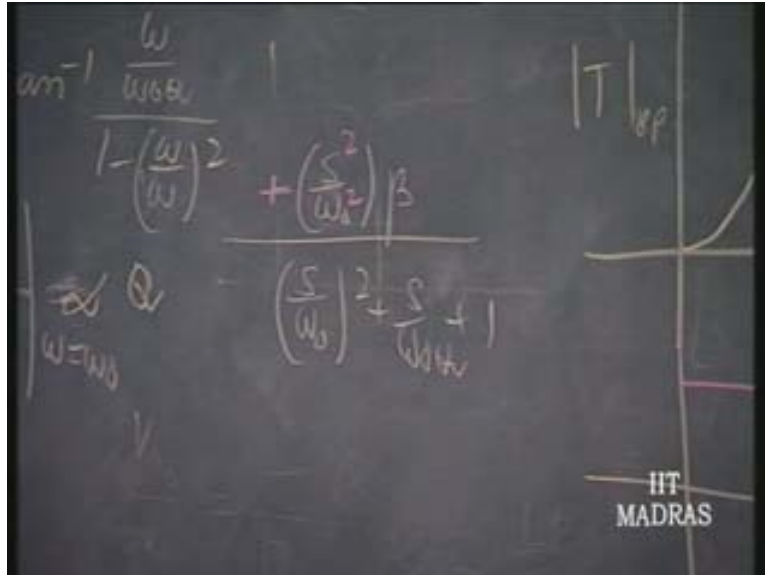
(Refer Slide Time: 00:36:39)



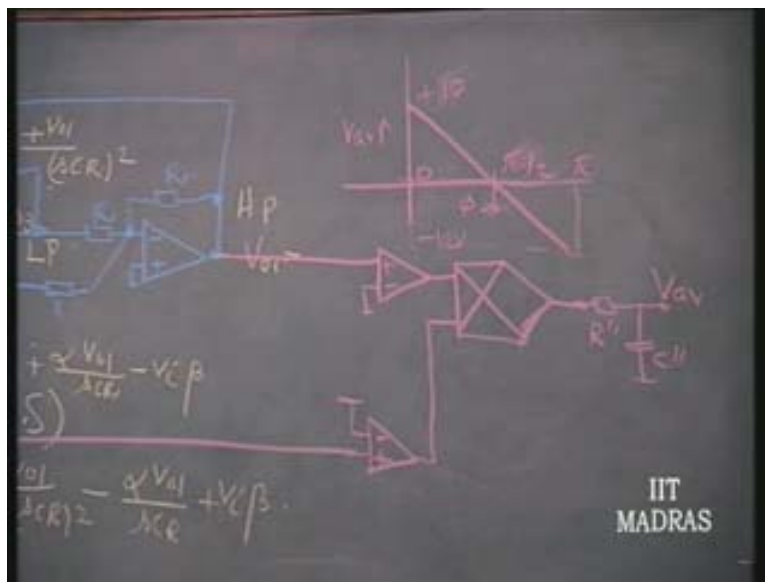
Therefore let us take the output from this the high pass and the input from this and put a phase detector, how is the phase detector designed, we put a multiplier a linear phase detector, we do not have to put a linear phase detector but for simple analysis we have put a linear phase detector now. It will become a phase detector primarily when I take the DC out of this the average and this average will tell me whether it is tuned to, since I have taken the high pass the high pass output corresponds to plus  $s$  square by  $\omega_0$  square. And you can see if it is exactly tuned to  $\omega_0$   $\omega$  is same as  $\omega_0$ , the incoming frequency  $\omega$  is same as  $\omega_0$ , this will get cancelled and I will get a phase shift of 90 degrees exactly at that frequency. When there is phase shift of 90 degrees for this circuit the output is going to be 0. The phase characteristic was going like this from plus  $V_s$  to, or plus 10V to minus 10V to 0.



(Refer Slide Time: 00:36:58)



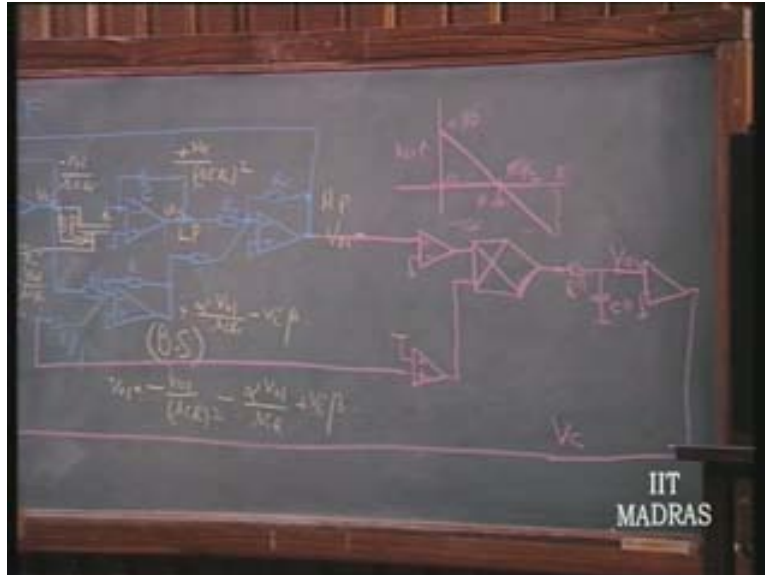
(Refer Slide Time: 00:38:00)



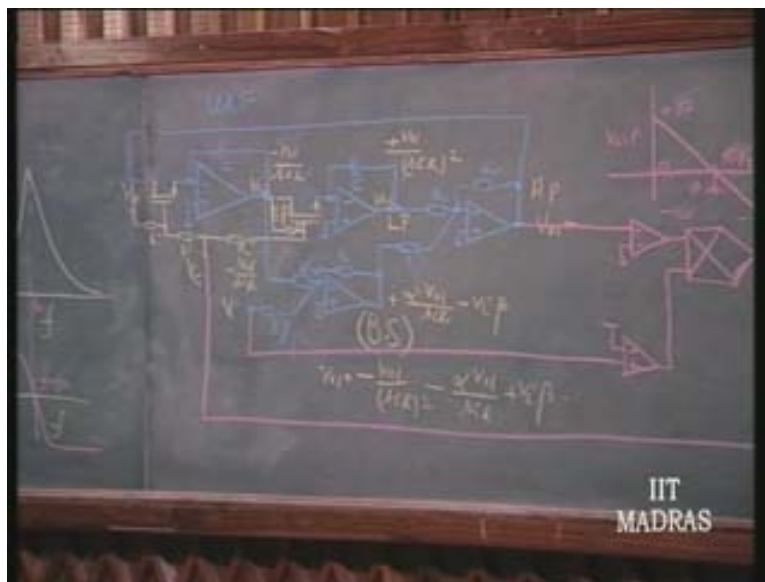
This was the phase characteristic with respect to, this is  $\pi$  by 20. So we can find out whether it is tuned to the incoming frequency or not the moment we get a zero average voltage at this point. Now how do I therefore make sure that this is going to be the case automatically?

I put a comparator here and connect this  $R$ , this is the reason.

(Refer Slide Time: 00:38:35)



(Refer Slide Time: 00:39:28)

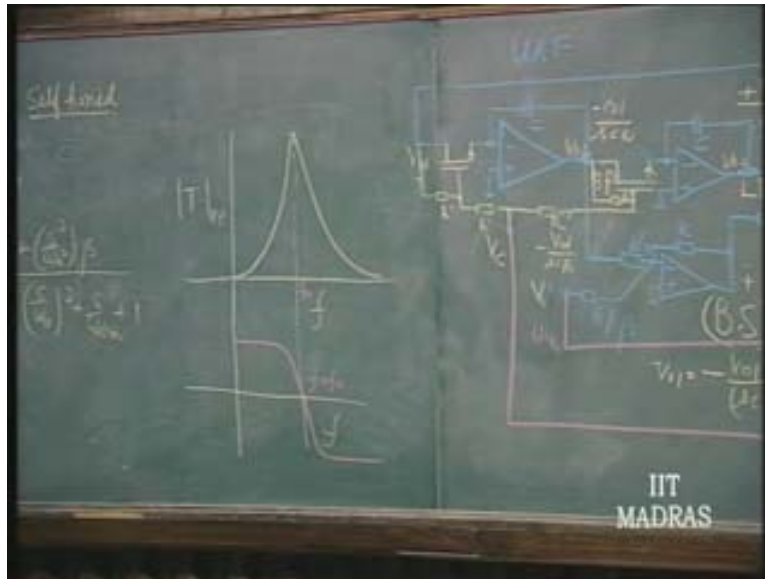


This is an automatic frequency locking scheme, it is automatically self tuned. This voltage has to automatically become equal to 0. If this is a negative feedback system since I am using an op amp here this output should adjust itself so that this voltage and this voltage become one and the same if it is a negative feedback system which you have to make sure. The proper polarity has to be put here to see that it is a negative feedback system.

We have to find out whether this particular scheme works as a negative feedback structure. When this is at omega is equal to omega R if the original frequency to which it

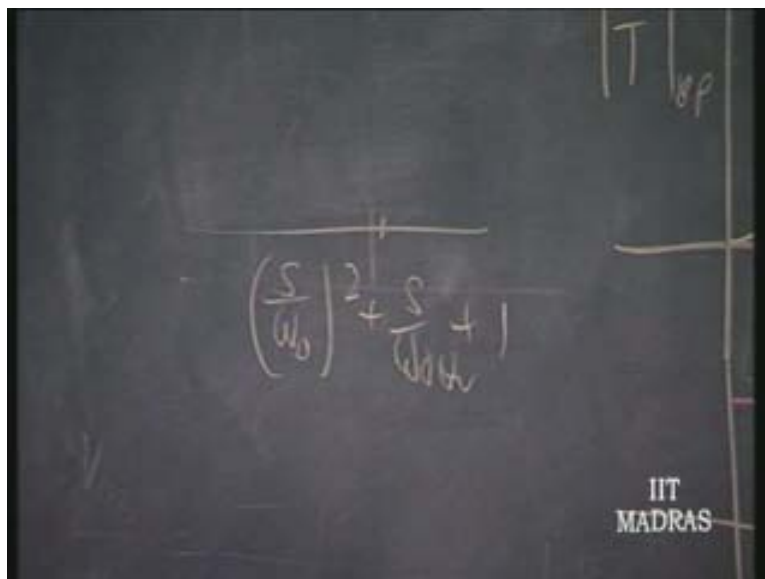
was tuned was less than  $\omega R$  now the feedback should be such that it should be brought towards  $\omega R$  that is negative feedback system. For this particular case I am using a n-channel MOSFET sometime to find out whether it is negative feedback or not. Now I have not marked the polarity of the last amplifier and I will change that depending upon whether it is going to be negative feedback.

(Refer Slide Time: 00:40:45)



This is  $\omega R$  and this is the incoming reference frequency, now I know how phase varies for this particular thing with reference to  $\omega R$ .

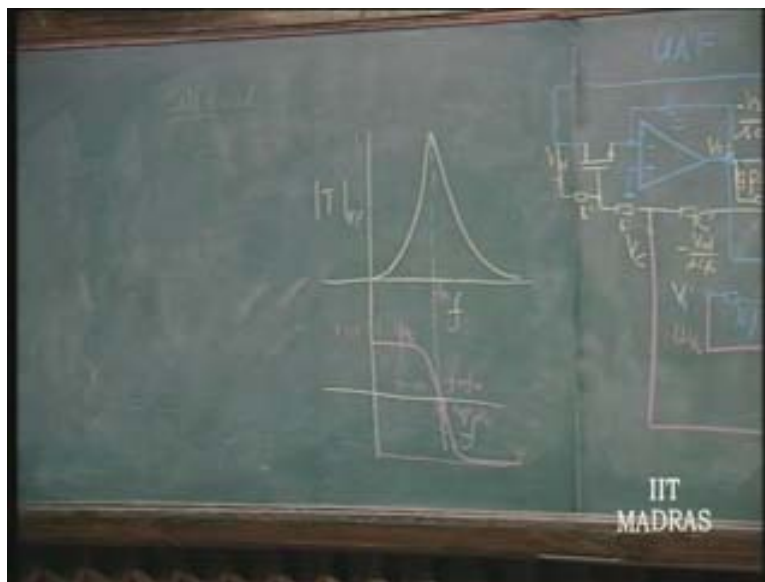
(Refer Slide Time: 00:41:28)



You can just put this as  $s$  is equal to  $j\omega$  and  $\omega$  is equal to  $\omega R$ . Therefore, as far as the numerator is concerned we are not bothered but as the denominator is concerned we know that as  $\omega R$  is very low what is the phase shift?

This quantity is going to be very low, the phase shift is determined primarily that it is minus 90 degrees from the denominator. The numerator itself is going to give you a phase shift of 180 degrees and you can ignore that. Therefore afterwards as  $\omega$  comes close to  $\omega_0$  this quantity vanishes, it becomes exactly equal to 90 degrees. As  $\omega$  goes very high you can see that this becomes negligible and this quantity becomes dominant so it goes to 180 degrees. As far the numerator is concerned this magnitude itself is going to go close to 0 when  $s$  is equal to 0.

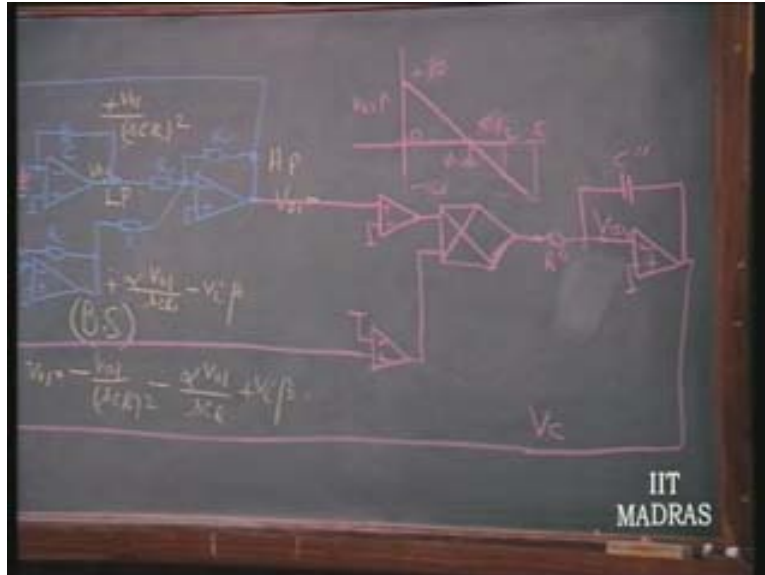
(Refer Slide Time: 00:44:23)



So the phase variation of this if you plot we have already plotted it here, it is going to vary in this fashion, this is going to be  $\pi/2$ , the initial phase variation is primarily determined by the numerator rather than the denominator because  $s$  is  $j\omega R$  and  $\omega$  is very small so it is starting with a phase of 180 degrees and going to minus 90 degrees and going to 0 degrees like this. Whether is a minus or plus it does not make any difference, so 180 or 90 or 0 and that is how it is changed. Therefore you should know this nature of variation. If this is the nature of variation at, this is  $\omega R$  is equal to  $\omega_0$  the phase shift in this.

Let us say  $\omega R$  is less than  $\omega_0$  then the phase shift is closer to 180 degrees this is important and it is changing from higher phase shift to this. Now, as far as the phase detector is concerned if the phase shift is closer to 180 degrees we have started with  $\omega_R$  less than  $\omega_0$ , so  $\omega_R$  is somewhere here in this region. When the phase shift is greater than  $\pi/2$  you are getting a negative value so this voltage is negative. If this is negative let us put this as negative just for **argument sake** then this would have gone to more positive voltage here or less negative.

(Refer Slide Time: 00:45:56)



More positive means the resistance is going to decrease, this is n-channel so the resistance is going to decrease. If the resistance decreases the frequency is going to increase and the  $\omega R$  is remaining the same but  $\omega a_0$  is going to increase so this is positive feedback. In order to make it negative feedback what should we do? We can change any one of these things, either we make this because I would like to change this rather than anything else, instead of n-channel I make it p-channel and retain this as minus because I would like to not use large capacitors here instead I would like to use miller effect and make this an integrator. This is what is called a self tuned filter which is now going to work satisfactorily.

What does it mean? Where is it used?

This is used when the signal to noise ratio is very high, that is the signal is buried in noise. But we know that the signal is almost a single frequency, that is, the bandwidth of the signal required is very small. Therefore I make a high very high Q self tuned filter and let it get tuned to the incoming signal then it will eliminate all the other noise around it if it gets locked on to the signal. This is one way of searching whether there is a signal anywhere in space.

We do not know who is sending our information in space but we know that this information is a highly narrow band signal. So we can keep scanning the entire range of frequencies using this kind of self tune filter. What you do is you can skip ranges by making  $V_c$  get disconnected and making it a saw tooth and search in different ranges of frequency and see whether anything is getting locked. Once it is locked it will keep on tracking. So this is really used in space applications for finding out signals from unidentified sources. And we just keep recording this for years and years to see whether there is any intelligible information coming from these sources. So in such amplifiers the self tuned filters are used.