

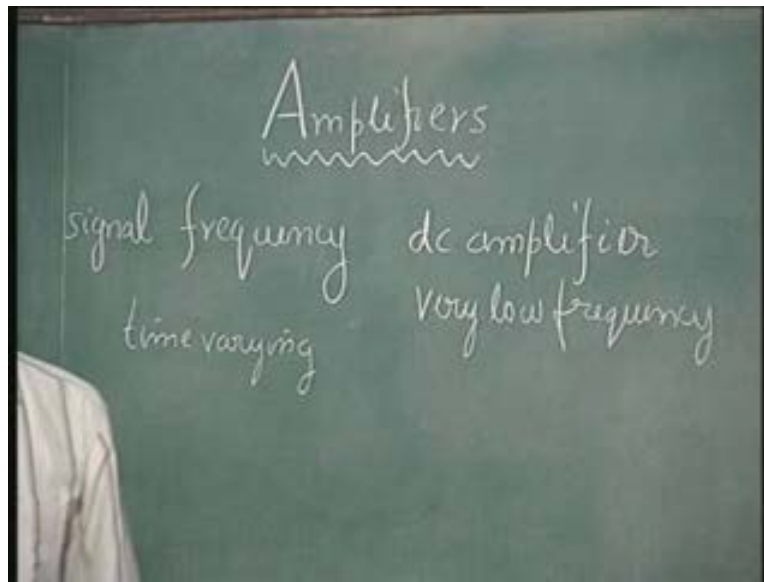
Electronics for Analog Signal Processing - I
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Lecture - 17
Amplifier Applications

Today, we will discuss definition of amplifiers based on their applications. So far, we have discussed various types of amplifiers, basic amplifiers. Today, let us classify the amplifiers based on applications. Now, we have to bring in another term called frequency, the signal frequency. So, amplification we were now referring to was corresponding to signal, which was a voltage or a current; and it was in turn acting as a voltage controlled or current controlled input block, and correspondingly, was available as voltage source or current source. So, we had four types of amplifiers.

Now, based on signal frequency, we can classify amplifiers. Let us see how. The lowest frequency amplification, this is called, dc amplifier, d c. Here, in our case, please remember that signal in electrical engineering, useful signal, which is to be amplified, always means time varying. So, signal means actually time varying. If it is constant with respect to time, there is no information content; because, already we know that it is constant at a certain value. There is no variation with respect to time; so, there is no information content. The moment the signal varies with respect to time, then only, we consider it worthwhile signal that has to be amplified.

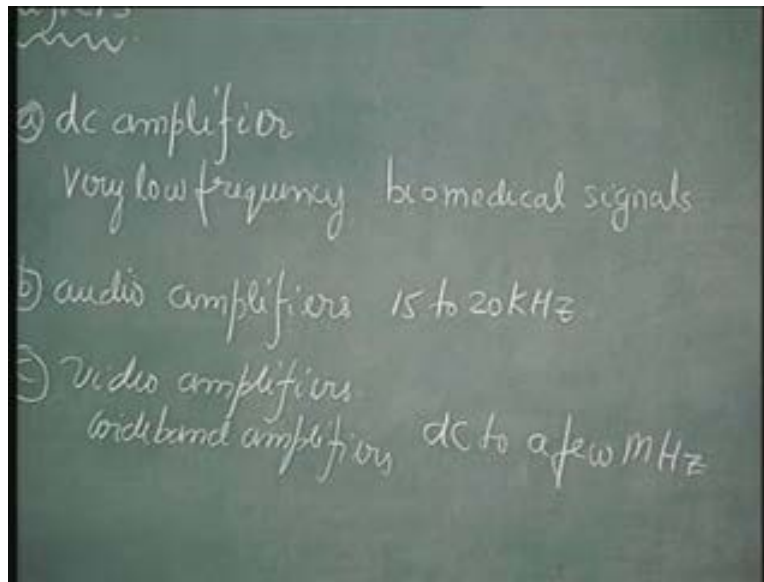
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Therefore, dc here means very low frequency; like in the case of biomedical signals; may be, EEG, ECG, EMG, all these signals - these are all very low frequency signals. So, such signals, if have to be amplified, must be amplified by what is called as a dc amplifier; dc means, zero frequency. But actually, in our application, it always means low frequency. Please remember this. So low, that it is almost equal to zero, you can say.

Then we have audio amplifiers; audio range. Good audio, quality signal, may be up to about 15 to 20 Kilo hertz. So, this **this** range of signals, when amplified, need audio amplifiers. Like radio receiver, gramophone player, cassette player, cd player, all these things will require high quality audio amplifiers. In fact, hobbies, normally electronic hobbies, normally, concentrate primarily on building audio amplifiers. So, we have about 15 to 20 Kilo hertz frequency range. Then, video amplifiers, or also called, wide band amplifiers; which will mean dc to a few megahertz; picture signal, television.

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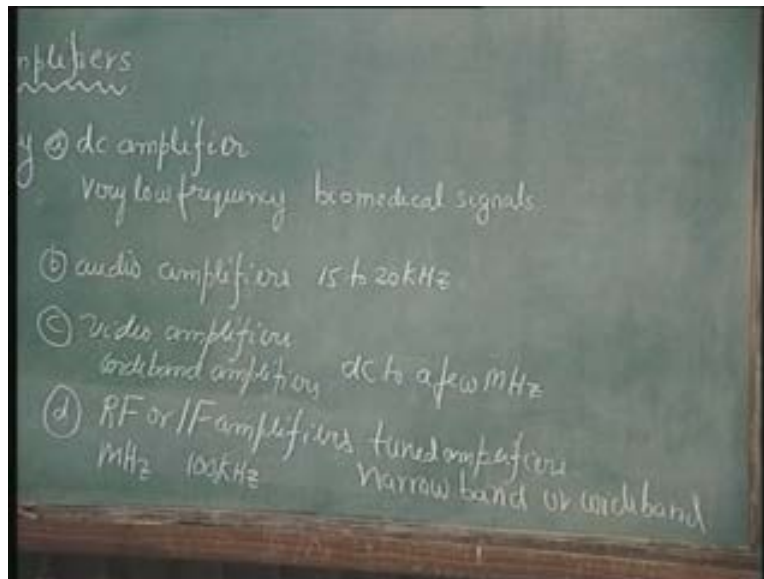


This is of primary interest to us and we want high quality wide band amplifiers for amplifying this picture signal. This is called video amplifier. In fact, therefore, our primary concern in most of the amplifier design is going to be discussion as to how to design such amplifiers; very low frequency amplifiers, audio amplifiers, video amplifiers. So, we can see the frequency range and appreciate the difficulties involved in various amplifier designs.

Apart from this, we also need to design amplifiers for already coded signals; either the signal may be FM or put in the form of FM or AM. These high frequency amplifiers which are used for FM or AM signals; these are the ones which are of interest. These are called RF or IF amplifiers. These are basically what are called as tuned amplifiers of the order of megahertz to hundreds of Kilo hertz; megahertz to hundreds of Kilo hertz. And tuned amplifiers - they may be narrow band or wide band amplifiers.

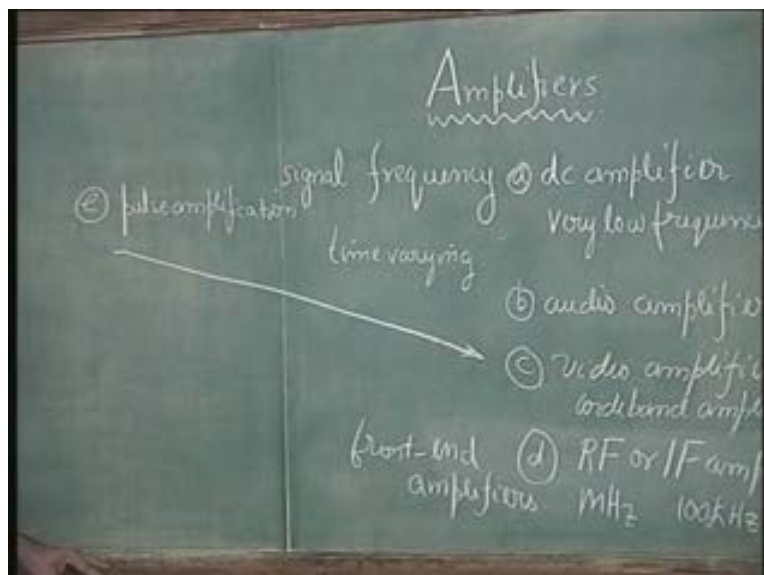
In order to make these selective, select only the carrier that you would like to amplify. These are made tuned amplifiers, narrow band or wideband, to select a band of signals and then amplify this. So, RF - radio frequency, IF - intermediate frequency amplifiers.

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These are found as front end amplifiers in all your receivers; radio receivers or television receivers. So, these are basically front end amplifiers. These are the amplifiers which are working for the highest frequency ((... Refer Slide Time: 09:40)) and then we have... This wideband amplifier can also be used for what is called as pulse amplification. These are also the other name because these are amplifying from very low frequency; dc to very high frequency.

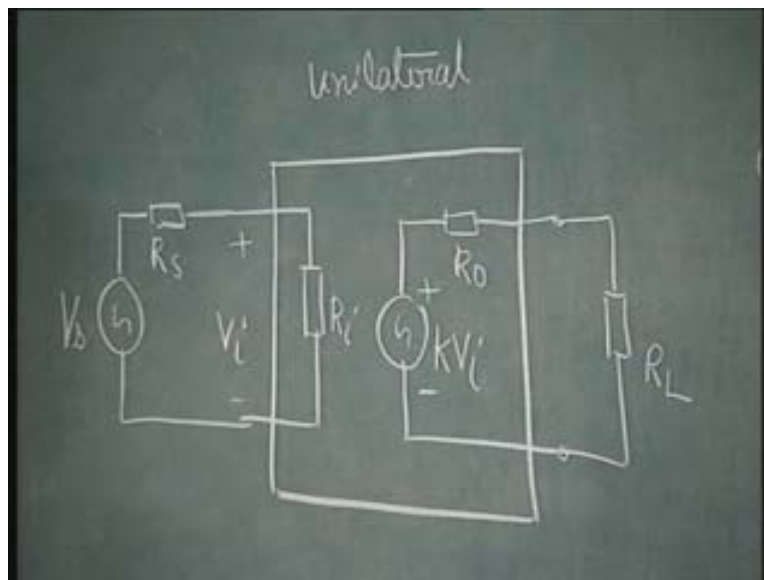
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So, they are able to amplify pulses of signal. So, these are also called pulse amplifiers. So, depending upon the range of frequency, we have the various categories of these amplifications.

Now, we have to understand how frequency comes into picture ((in this... Refer Slide Time: 10:35)). We have already understood this much; that, in an amplifier block which is close to ideal, we have this unilateral. So, we are considering that unilateral amplifier where the reverse transmission is zero. It may have a sort of input resistance R_i , output resistance R_o . I am considering arbitrarily some category of amplifier, let us say, which has finite input resistance and finite output resistance; and then, let us consider open loop voltage gain, A_v is K times V_i ; it is K times V_i , open loop voltage gain, and this is what we had considered earlier. This is connected to R_L . Now, it is driven by a source. This was the first amplifier we considered; non-ideal amplifier. Voltage controlled voltage source amplifier with voltage gain equal to K .

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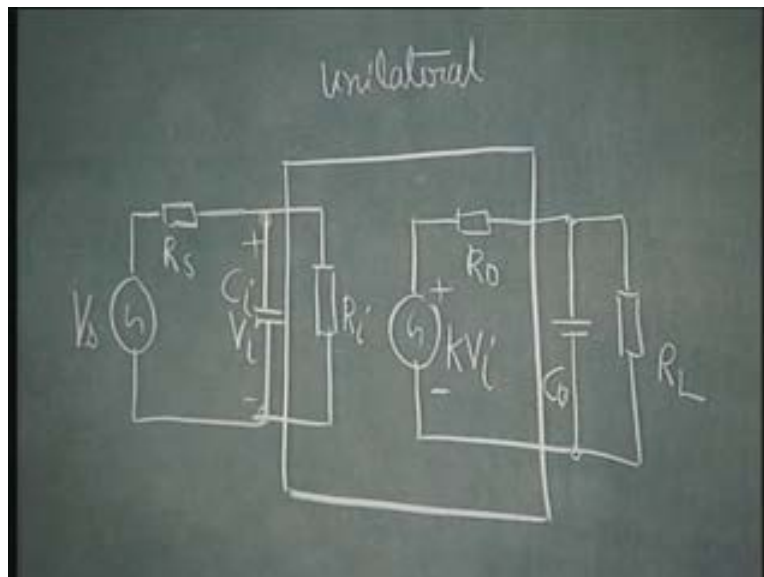


Now, let us consider: how does the frequency dependence come to this amplifier? Normally, this wiring of the amplifier and all the other connections cause what is called stray capacitors to come into picture at the input. The whole lump effect of the capacitor

at the input can be put as C_i . This is unwanted, but this comes. Like it, this was unwanted, if it is voltage controlled, R_i should have gone to infinity; but it is there. Similarly, in general, we have the capacitor also, although unwanted, coming as stray element between the input terminals.

Similarly, at the output terminal, we have C_{out} as the effective output capacitance. This may take into account the capacitance due to load. Here also, it may take into account the capacitance due to source. All put together, let us say, we put it as a single capacitor C_i at the input and C_{out} at the output.

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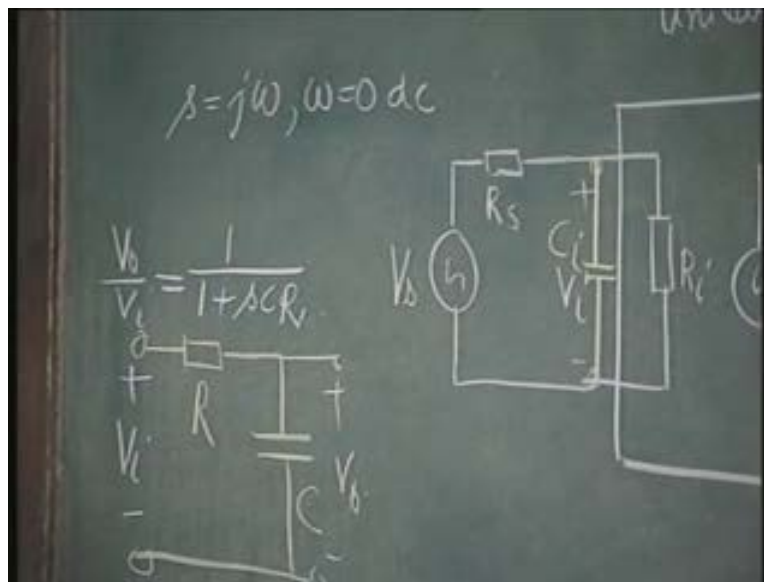
So, what does this capacitor do? So, it becomes fairly simple for us that this capacitance is going to cause V_i to become much less than what it otherwise would have been because of high frequency effect. Now, at high frequencies, this capacitor will shunt this R_i and therefore, most of the current which would have gone into amplifier, would have then gone into C_i . So, we are going to lose some amount of signal because of C_i .

Similarly, we are going to lose some amount of signal because of C_{out} ; takes away some amount of signal current. So, what is the effect of this? This, we can easily say that,

when we have a network like this, R and C; if this is V_i and this is V_o for this network, V_o/V_i from network, you know, is going to be equal to $1/(1 + sCR)$, 1 by 1 plus sCR .

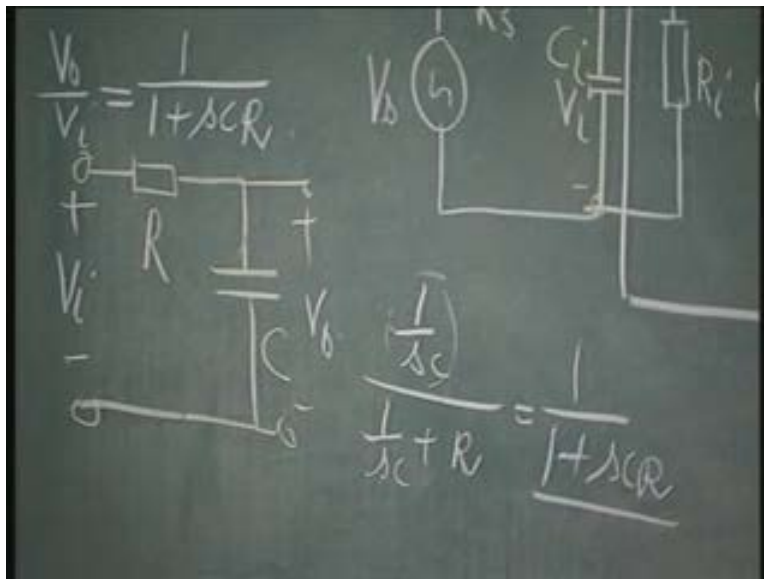
How did you do that? Because, for capacitance, you can consider that it is an open circuit at low frequencies. So, $1/s$ is equal to, put s is equal to $j\omega$ and ω equal to zero; that is dc.

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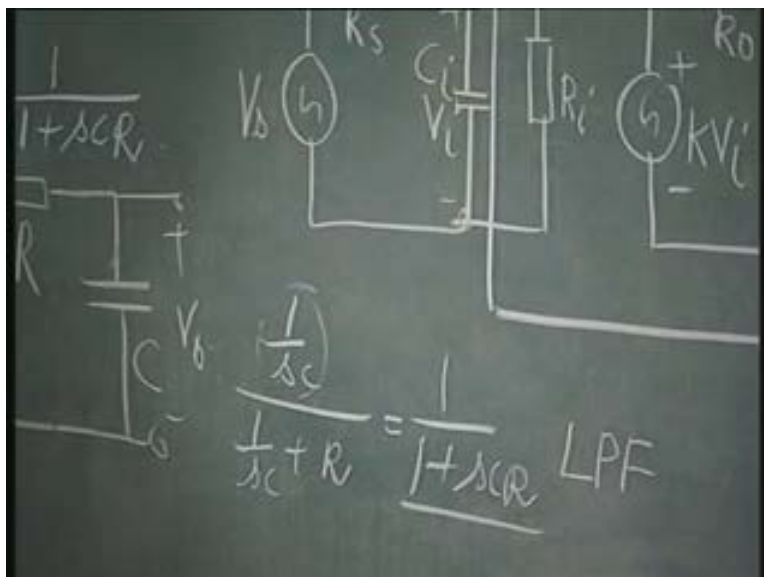
Then you get this as 1; that is true, because no attenuation occurs due to this. So V_o is equal to V_i . So, this is the low frequency response of this. At high frequencies, obviously, the capacitive reactance comes into picture. $1/(j\omega C)$ or $1/sC$ divided by R plus $1/sC$, is what this is; which is written as $1/(1 + sCR)$. So basically, it is $1/sC$ divided by $1/sC$ plus R which is written as $1/(1 + sCR)$. This is the impedance of the capacitor.

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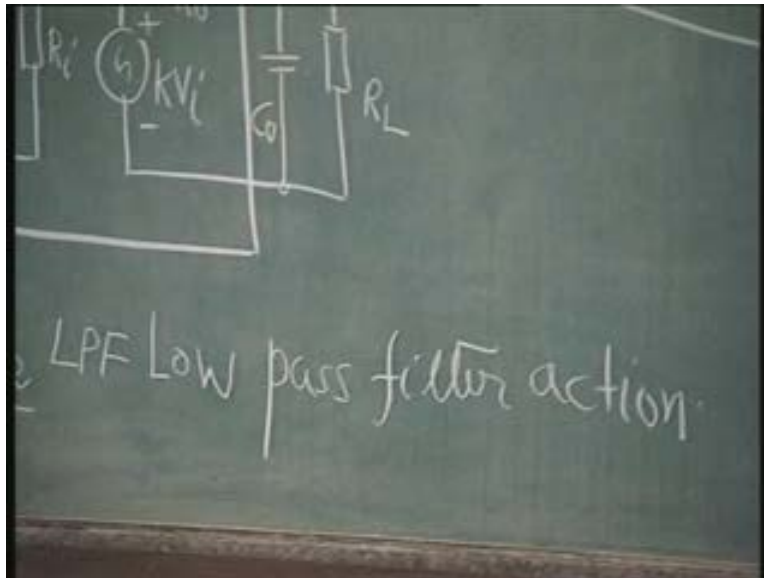
For obtaining the frequency response, you have to put s is equal to j omega. All these things you have learned in your networks course. So, this is called what? This is called a low pass filter action, low pass, low pass.

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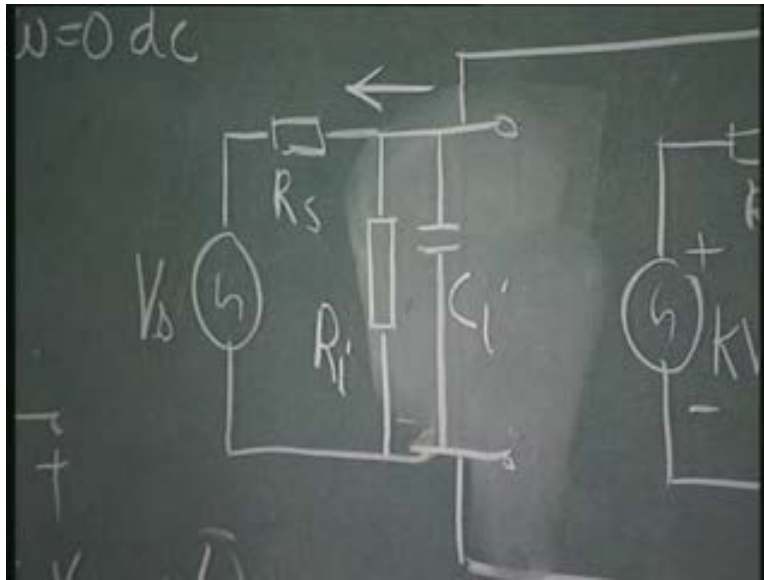
So, the stray capacitor at the input of an amplifier results in low pass filter action.

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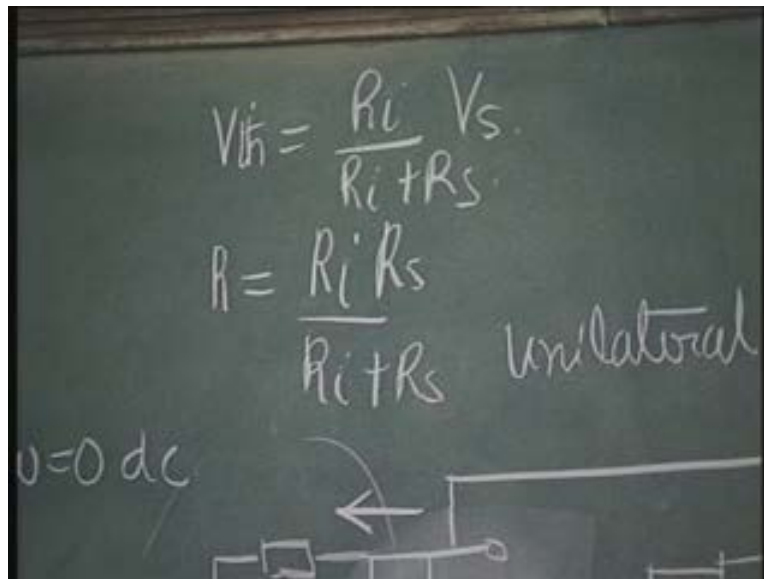
What it means is, it is going to attenuate the signal. Now, what is this R ? In our case, we can replace this equivalent here by a Thevenin's equivalent. What is that Thevenin's equivalent? You can just put down this circuit at the input as R_i here, C_i . So, look at this from this side. This circuit remains the same. I have not changed anything. Instead of putting C_i this side, I have put C_i this side. This R_i , I am making it become part of the input.

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So, this is a new voltage source now, which has R_i , R_s and V_s in this arrangement. That means, I can put down the Thevenin's equivalent for this. The Thevenin's resistance is R equal to R_i R_s by R_i plus R_s . How do I get it? Short circuit this and short circuit any voltage source and open circuit any current source and find out the effective impedance. So, effective impedance is nothing but this. So, this R is equal to R_i parallel R_s ; and therefore, you have then the Thevenin's voltage, which is, $V_{Thevenin}$ is going to be nothing but R_i divided by R_i plus R_s times V_s . So, I can always convert this in this following fashion; that it is a Thevenin's voltage, V source, with a new source resistance R at the input. Is this point understood?

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Handwritten equations on a chalkboard:

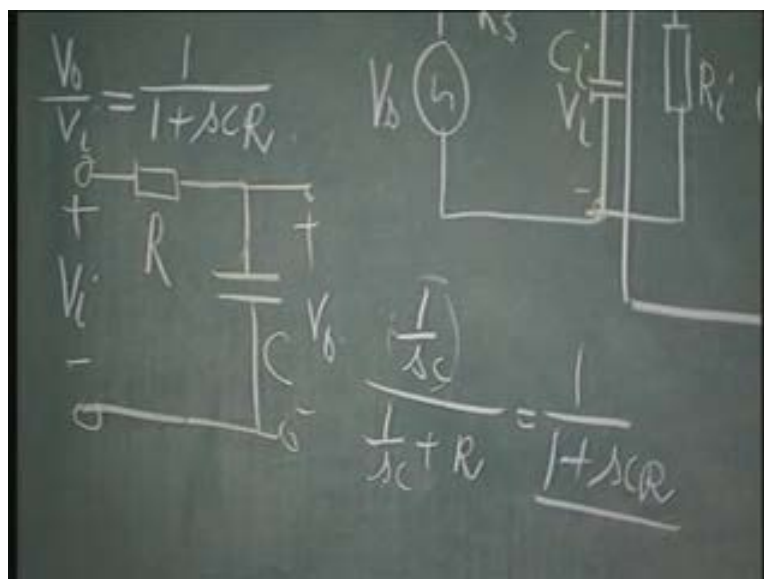
$$V_{th} = \frac{R_i}{R_i + R_s} V_s$$

$$R = \frac{R_i R_s}{R_i + R_s} \quad \text{unilateral}$$

Below the equations, it is noted: $\omega = 0 \text{ dc}$. A partial circuit diagram is visible at the bottom, showing a voltage source and a resistor.

Now, if that is the case, this is nothing but this new circuit here, which we have discussed.

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Handwritten equations and circuit diagrams on a chalkboard:

Left side: A circuit diagram showing an input voltage V_i connected to a resistor R in series with a parallel combination of a capacitor C and a load resistor R_L . The output voltage is V_o . The transfer function is given as:

$$\frac{V_o}{V_i} = \frac{1}{1 + sCR}$$

Right side: A circuit diagram showing a voltage source V_s in series with a resistor R_s connected to a parallel combination of a capacitor C_i and a resistor R_i . The output voltage is V_o . Below this, a simplified equation is shown:

$$\frac{\frac{1}{sC}}{\frac{1}{sC} + R} = \frac{1}{1 + sCR}$$

That means, its transfer function should be the same as 1 by 1 plus sCR with C being equal to C i and R being equal to the Thevenin's resistance here. So, I do not have to work out this structure for its transfer function. I can simply look at it and say that if the

input capacitance is C_i , the effective resistance is found out by finding out the effective resistance across the capacitor as the Thevenin's resistance between the two terminals. That means, first, short circuit all the voltage sources and open circuit all the current sources, and find out the resistance. That resistance will come in the denominator.

So, we have the Thevenin's voltage here as R_i by R_i plus R_s . So effectively, we have R_i by R_i plus R_s . This we had got earlier also, was the attenuation of this V_s when it comes here as V_i . Apart from this, we have 1 plus sC into R equivalent; this is R , which is $R_i R_s$ divided by R_i plus R_s .

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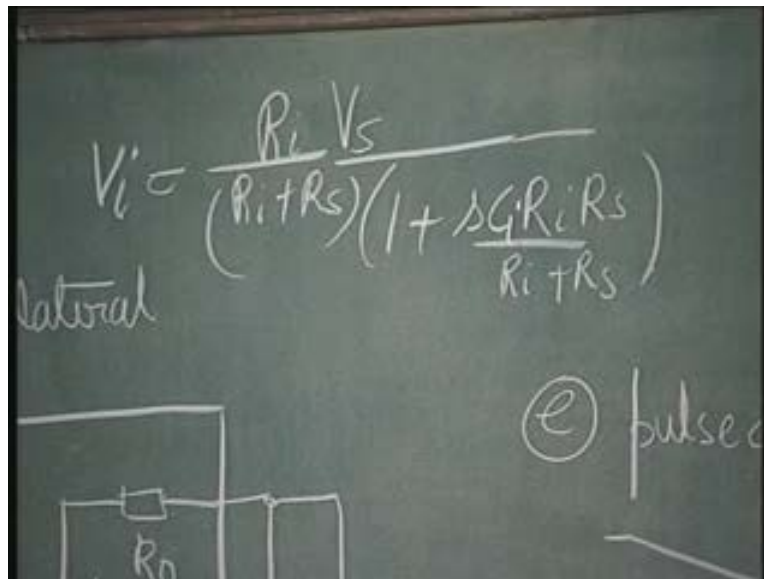
The image shows a chalkboard with the following handwritten equation:

$$\frac{R_i V_s}{(R_i + R_s) \left(1 + \frac{sC R_i R_s}{R_i + R_s} \right)}$$

Below the equation, there is a circled 'e' and the word 'pulse'.

So, this is what is called as a transfer function between V_i and V_s . This, please remember; in the earlier evaluation, this factor was not at all there. It was only R_i by R_i plus R_s into V_s . Now, a new factor of attenuation has come because of capacitor.

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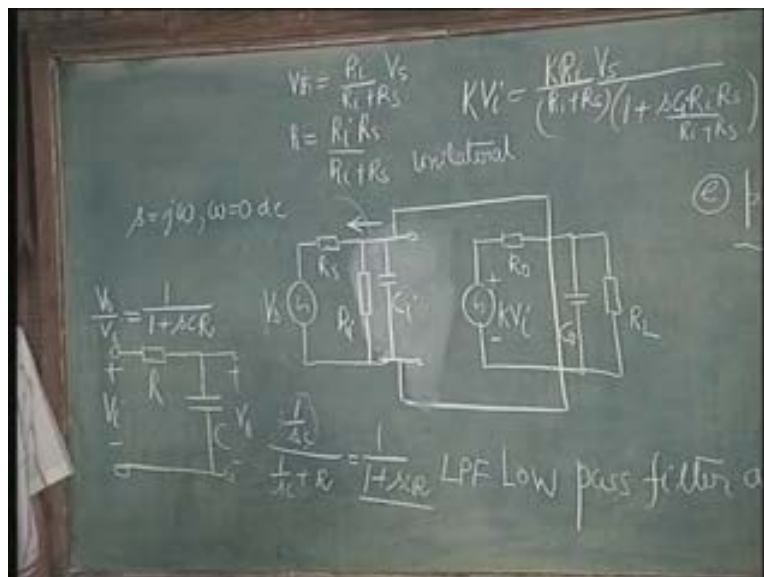
Handwritten equation on a chalkboard:

$$V_i' = \frac{R_i V_s}{(R_i + R_s) \left(1 + \frac{s C_i R_i R_s}{R_i + R_s} \right)}$$

Below the equation, the word "natural" is written. To the right, there is a circled 'e' followed by "pulse". Below the equation, a partial circuit diagram is visible, showing a resistor labeled R_0 .

And this is frequency dependent; C is equal to C_i . This is frequency dependent and therefore the V_i decreases as frequency increases. Then, this voltage is going to appear here as K times V_i . So, V_i is this; K times V_i is this; so that gets, remains unaffected.

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Handwritten equations and circuit diagrams on a chalkboard:

Top left equation:

$$V_i' = \frac{R_i V_s}{R_i + R_s}$$

Top right equation:

$$K V_i' = \frac{K R_i V_s}{(R_i + R_s) \left(1 + \frac{s C_i R_i R_s}{R_i + R_s} \right)}$$

Below the top left equation, the word "unilateral" is written. Below the top right equation, there is a circled 'e' followed by "pulse".

Bottom left equation:

$$\frac{V_o}{V_i} = \frac{1}{1 + s C R}$$

Bottom right equation:

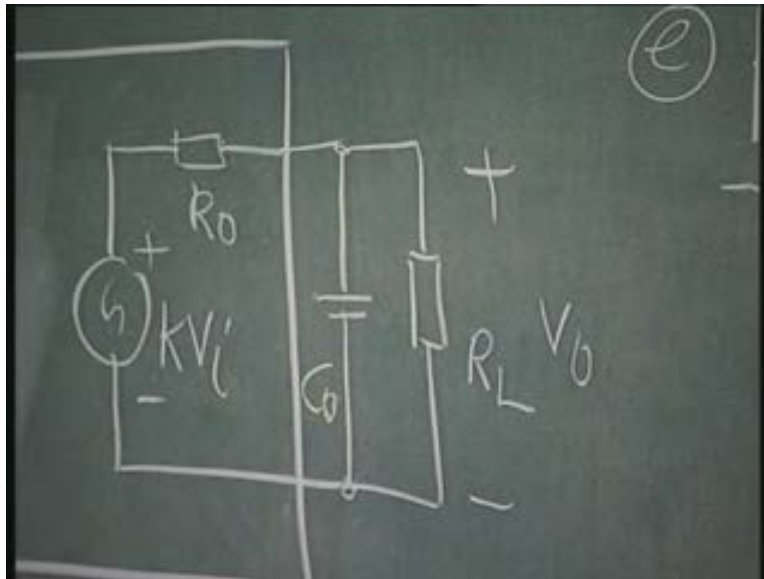
$$\frac{1}{s C + R} = \frac{1}{1 + s C R}$$

Bottom right text: "LPF Low pass filter"

Circuit diagrams are shown in the center and bottom left. The central diagram shows a voltage source V_i connected to a resistor R_i and a capacitor C_i in parallel, which is then connected to a load resistor R_L . The bottom left diagram shows a voltage source V_i connected to a resistor R and a capacitor C in parallel.

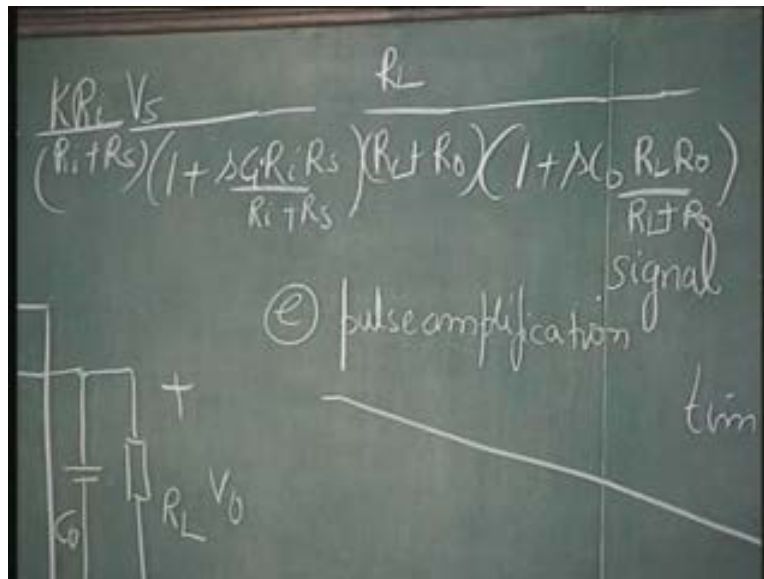
Now, I have to find out the voltage V_{naught} ultimately. How do I get V_{naught} ? V_{naught} is across C_{naught} . By a similar argument, I would leave this as an exercise for you; we can treat this as a source in series with the resistance R_{naught} shunted by a capacitance C_{naught} across R_L .

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So, this situation is exactly similar to this. So, you can again convert this into a Thevenin's equivalent where the equivalent resistance is R_L parallel R_{naught} ; and the capacitance is C_{naught} , and find out the voltage here. So, without doing any analysis, I can do that. It is going to be equal to R_L by R_L plus R_{naught} , the low frequency thing, R_L by R_L plus R_{naught} , times $K V_i$; this into 1 plus $s C_{naught}$, $R_L R_{naught}$ divided by R_L plus R_{naught} .

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The chalkboard shows a handwritten equation for the output voltage V_o across a load resistor R_L :

$$V_o = \frac{K R_i V_s}{(R_i + R_s) \left(1 + \frac{s G R_i R_s}{R_i + R_s}\right) (R_L + R_o) \left(1 + \frac{s C_o R_L R_o}{R_L + R_o}\right)}$$

Below the equation is a circuit diagram of a voltage divider. It consists of a voltage source V_s in series with a resistor R_s , connected to a load resistor R_L . The output voltage V_o is measured across R_L . To the right of the diagram, the text "pulse amplification" is written, with "signal" and "time" written vertically next to it.

I am writing it straightaway because we know that it is exactly similar in nature, except that C_i is replaced by C_{naught} ; R_i is replaced by R_L ; R_s is replaced by R_{naught} and V_s is replaced by K times V_i okay. So, the analysis is similar; and therefore it will come out with another attenuation like this. So, the total gain is going to be, this whole thing is going to be, equal to V_{naught} .

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The chalkboard shows a handwritten equation for the output voltage V_o across a load resistor R_o :

$$V_o = \frac{K R_i V_s}{(R_i + R_s) \left(1 + \frac{s G R_i R_s}{R_i + R_s}\right) (R_o + R_o) \left(1 + \frac{s C_o R_L R_o}{R_L + R_o}\right)}$$

Below the equation is a circuit diagram of a voltage divider. It consists of a voltage source V_s in series with a resistor R_s , connected to a load resistor R_o . The output voltage V_o is measured across R_o . To the right of the diagram, the text "pulse" is written.

So, what has happened? Earlier, we know that V_o over V_s was equal to simply K times R_i by R_i plus R_s into R_L by R_L plus R_o . This was the earlier expression. You can refer to your notes and verify this was our earlier expression for voltage gain V_o over V_s . Now, that gets modified by 1 plus $sC_i R_i$ by R_i plus R_s . This is called input time constant.

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$$\frac{V_o}{V_s} = \left(\frac{K R_L}{R_i + R_s} \times \frac{R_L}{R_L + R_o} \right) \frac{V_o}{V_s} = \frac{1}{1 + sC_i \left(\frac{R_i R_L}{R_i + R_s} \right)}$$

$s=0$ dc

C_i is the input capacitor; $R_i R_L$ by R_i plus R_s is the effective resistance across the capacitor. So, this is called the input time constant; we call this as τ input time constant.

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$$\frac{C_i R_i R_s}{R_i + R_s} = Z_i'$$
 input time constant
 $s = j\omega, \omega = 0 \text{ dc}$

$$\frac{V_o}{V_s} = \frac{1}{1 + s Z_i'}$$

$$\frac{V_o}{V_s} = \left(\frac{R_L}{R_L + R_o} \right) \frac{1}{1 + s Z_i'}$$

So, we can instead of keeping on writing this, we will put it as simply input time constant τ_i . C into R is the dimension of time.

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$$\frac{C_i R_i R_s}{R_i + R_s} = Z_i'$$
 input time constant
 $s = j\omega, \omega = 0 \text{ dc}$

$$\frac{V_o}{V_s} = \frac{1}{1 + s \tau_i}$$

$$\frac{V_o}{V_s} = \left(\frac{R_L}{R_L + R_o} \right) \frac{1}{1 + s \tau_i}$$

So, this one... the other one is therefore called the output time constant; and output time constant is τ_o , which is $C_o R_o \parallel R_L$ by $R_o \parallel R_L$.

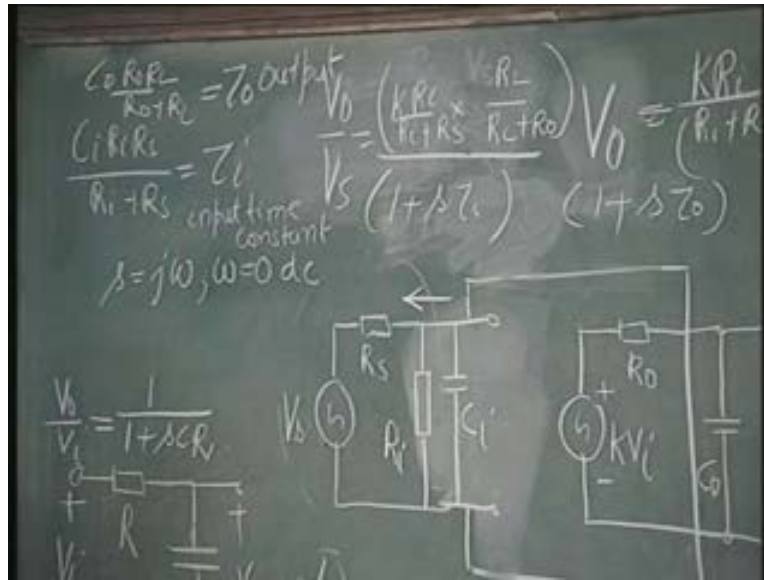
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Handwritten equations on a chalkboard:

- $\tau_o = \frac{C_o R_o R_L}{R_o + R_L}$
- $\tau_i = \frac{C_i R_i R_s}{R_i + R_s}$ (labeled "input time constant")
- $\frac{V_o}{V_s} = \left(\frac{K R_o}{R_i + R_s} \right) \left(\frac{R_o \parallel R_L}{R_o + R_L} \right)$
- $s = j\omega, \omega = 0 \text{ dc}$

For any amplifier, the input time constant and output time constants are always the same as what is indicated here. Whether it is a FET amplifier, or an operational amplifier, or a bipolar junction transistor amplifier, these definitions remain the same; input time constant and output time constant bring about degradation in performance of the amplifier at higher frequencies. This gain is the low frequency gain and these are the high frequency attenuation factors, τ_i and τ_o . So, τ_o is the output time constant.

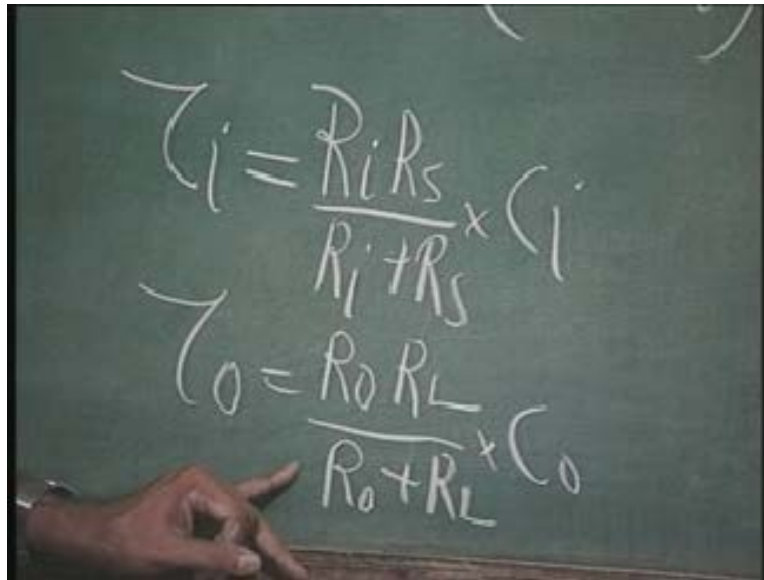
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Now, summarizing what we had just discussed previously, the dc gain K_{naught} , which is independent of frequencies; whatever term that is independent of frequency that we call as dc gain, V_{naught} over V_s . So, R_i by R_i plus R_s , the attenuation constant at the input into open loop gain K into R_L by R_L plus R_{naught} , attenuation constant at the output. Attenuation at the input, attenuation at the output and the open loop gain, overall gain, is therefore determined by this.

Just by looking at the amplifier circuit, you should be able to evaluate the overall gain without really putting down any equations. Similarly, the time constant τ_i and τ_{naught} , if they are known, τ_i being the input time constant, τ_{naught} being the output time constant, τ_i is defined as C_i into effective resistance across the capacitor at the input; and τ_{naught} is C_{naught} into effective resistance across the capacitor at the output.

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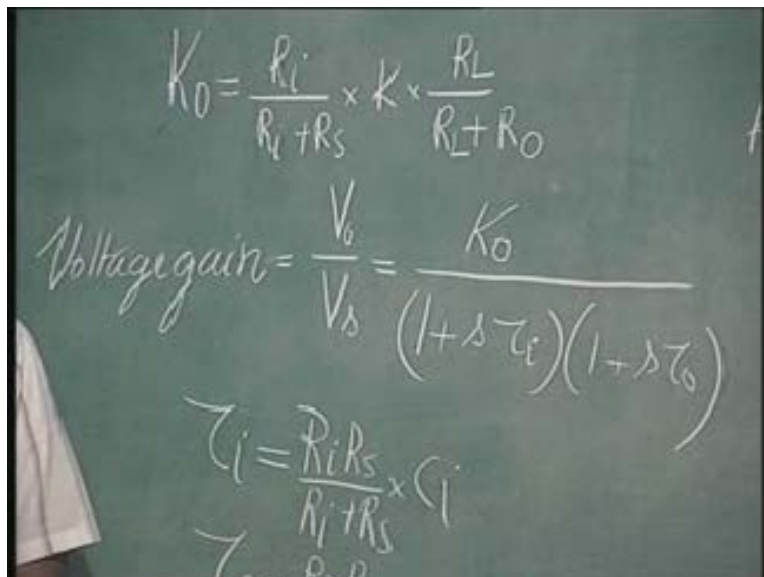


$$\tau_i = \frac{R_i R_s}{R_i + R_s} \times C_i$$

$$\tau_o = \frac{R_o R_L}{R_o + R_L} \times C_o$$

So, if you know this, you can straightaway write down the transfer function V_o/V_s , without really solving any equation. That is what you should practice. K_o divided by $1 + s\tau_i$ into $1 + s\tau_o$.

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$$K_o = \frac{R_i}{R_i + R_s} \times K \times \frac{R_L}{R_L + R_o}$$

$$\text{Voltage gain} = \frac{V_o}{V_s} = \frac{K_o}{(1 + s\tau_i)(1 + s\tau_o)}$$

$$\tau_i = \frac{R_i R_s}{R_i + R_s} \times C_i$$

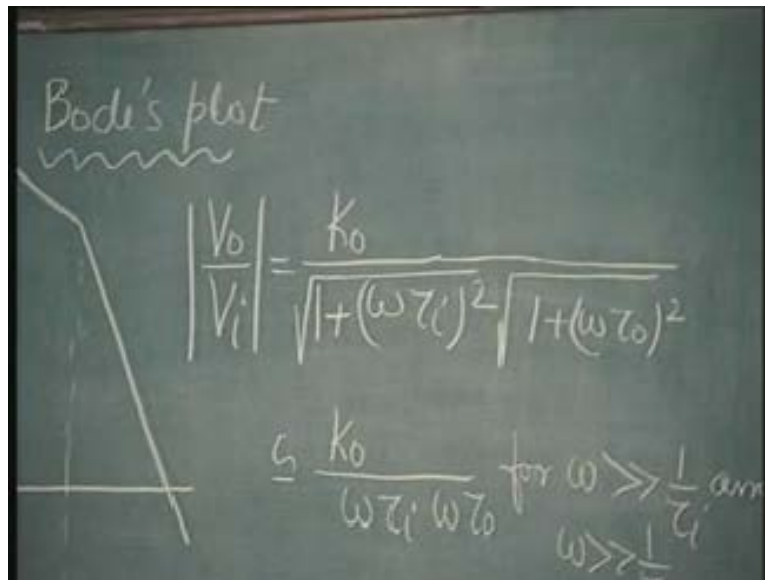
$$\tau_o = \frac{R_o R_L}{R_o + R_L} \times C_o$$

It is always going to be the transfer function in terms of τ_i and τ_o , irrespective of the amplifier configuration. Whether it is a voltage controlled voltage source, current

controlled current source, or any of the other two types, as long as it is non-ideal, this is the way it is going to be defined. So, this is a general term.

Now, let us see how this looks like. This has made the amplifier frequency dependent and that frequency dependent, that dependence, is plotted here in these two diagrams. These are, this is called, popularly known as Bode's plot.

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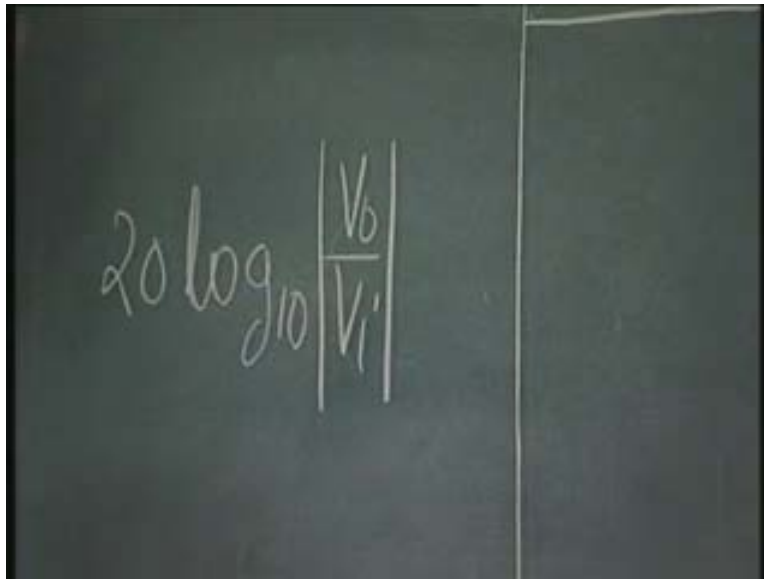


You must have come across this in your controls. However, it is nothing but magnitude of V_o over V_i plotted as a function of ω , phase of V_o over V_i plotted as a function of ω . This is magnitude of V_o over V_i which is nothing but K_o divided by root of $1 + \omega \tau_i$ whole square, where s is replaced by $j\omega$, and then you find out the magnitude.

So, $1 + \omega \tau_i$ whole square, root of that; into, $1 + \omega \tau_o$ whole square, root of that. That is the magnitude function. That magnitude function, plotted as a function of ω . Now, how is it plotted? It is plotted as $20 \log$ of this. So actually, what you are going to do is, $20 \log$ of this is plotted so that, this is in terms of decibels.

So, this is what we have earlier also indicated. $20 \log$ to the base 10 of V_o over V_i . That means, this will be in terms of decibels, so many decibels. Suppose K is 100; then, this is going to be $20 \log$ to the base 10 100, which is nothing but 40 decibels. So, this will start with 40 decibels of gain.

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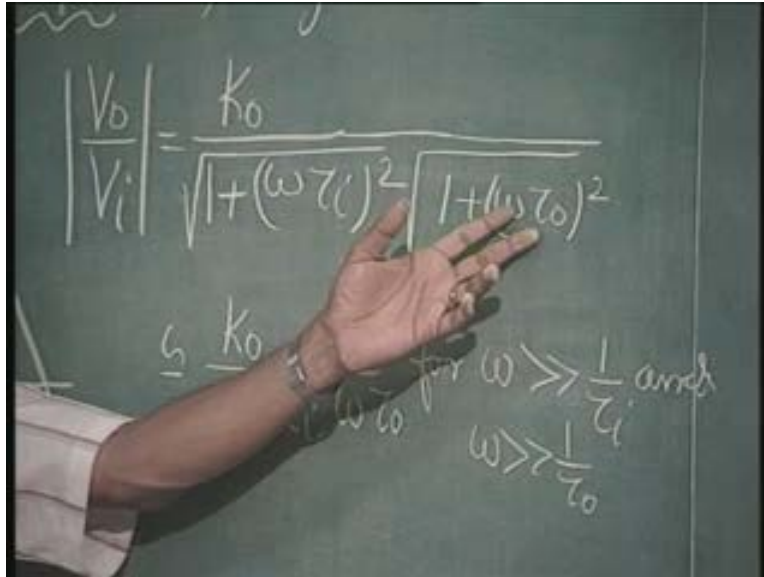


$$20 \log_{10} \left| \frac{V_o}{V_i} \right|$$

So, the gain is reducing. That means it is falling here. This fall is indicated here as an asymptote, asymptotic fall. How do you get that? For ω much greater than $1/\tau$, this factor is very dominant compared to 1; so you can ignore 1. So, this whole thing becomes ω into τ . It is inversely proportional to frequency ω here.

This factor, let us consider, is not coming into picture now. That means it is still far away from this omega.

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A hand is pointing to a chalkboard. The chalkboard contains the following equation:

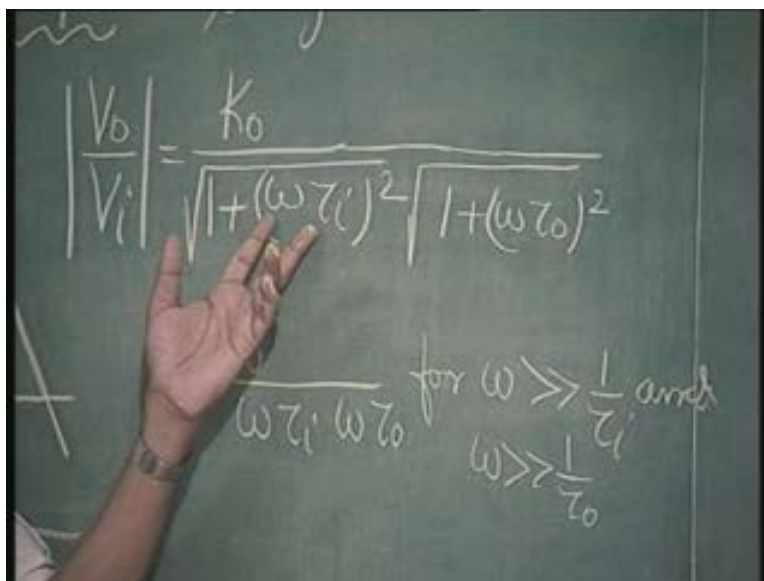
$$\left| \frac{V_o}{V_i} \right| = \frac{K_o}{\sqrt{1 + (\omega \tau_i)^2} \sqrt{1 + (\omega \tau_o)^2}}$$

Below the equation, there is a note:

$\approx \frac{K_o}{\omega \tau_o}$ for $\omega \gg \frac{1}{\tau_i}$ and $\omega \gg \frac{1}{\tau_o}$

Then, this is small compared to 1. Only this has become very large compared to 1.

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A hand is pointing to a chalkboard. The chalkboard contains the following equation:

$$\left| \frac{V_o}{V_i} \right| = \frac{K_o}{\sqrt{1 + (\omega \tau_i)^2} \sqrt{1 + (\omega \tau_o)^2}}$$

Below the equation, there is a note:

$\approx \frac{K_o}{\omega \tau_i \omega \tau_o}$ for $\omega \gg \frac{1}{\tau_i}$ and $\omega \gg \frac{1}{\tau_o}$

In this range therefore, we can say that this is dominant; time constant τ_i is said to be dominant. τ_i is much greater than τ_0 , let us take.

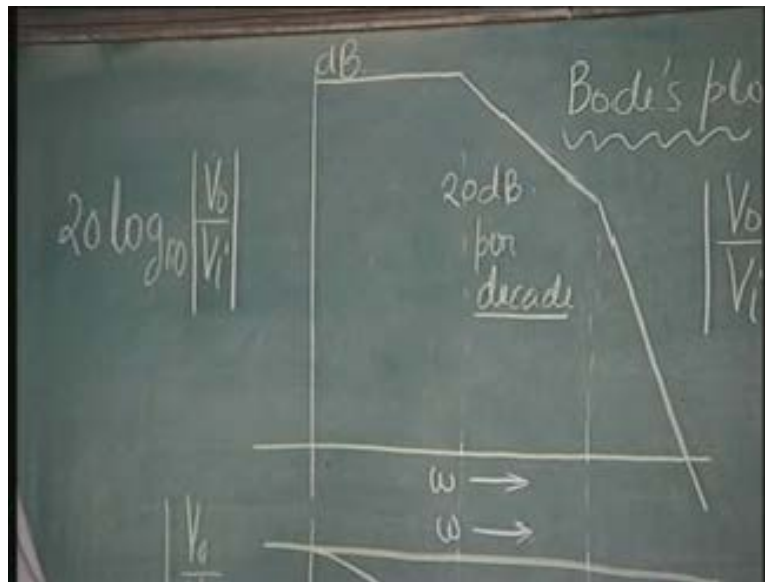
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let us take $\tau_i \gg \tau_0$
 τ_i dominant
 $s = j\omega$
 $\frac{|V_o|}{|V_i|} = \frac{K_0}{\sqrt{1+(\omega\tau_i)^2} \sqrt{1+(\omega\tau_0)^2}}$
 $\approx \frac{K_0}{\omega\tau_i \omega\tau_0}$ for $\omega \gg \frac{1}{\tau_i}$ and $\omega \gg \frac{1}{\tau_0}$

This may not be the case; but if we take τ_i much greater than τ_0 , then I can explain to you what exactly happens, because each of these time constants. They are coming almost independently. τ_i is much greater than τ_0 . First, τ_i makes its appearance; $\omega\tau_i$ becomes much greater than 1. So, this is K_0 by $\omega\tau_i$. This factor is very negligible; say 1. So, this is K_0 by $\omega\tau_i$. It is inversely proportional to ω .

If you take $20 \log$ of this factor, then, you can call this as an attenuation, at 20 decibels per decade.

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This term, this is a commonly used term. The attenuation is 20 decibels per decade. How? We know that K naught by $\omega \tau_i$ is the attenuation in this range and let ω change by a factor of decade; that means, 10. Then, the attenuation from whatever frequency you are starting and whatever frequency you are going, if it is a decade apart, then the attenuation is 20 decibels because it is $20 \log_{10} 1/10$, $20 \log_{10} 1/10$ it becomes, which is minus 20 decibels per decade.

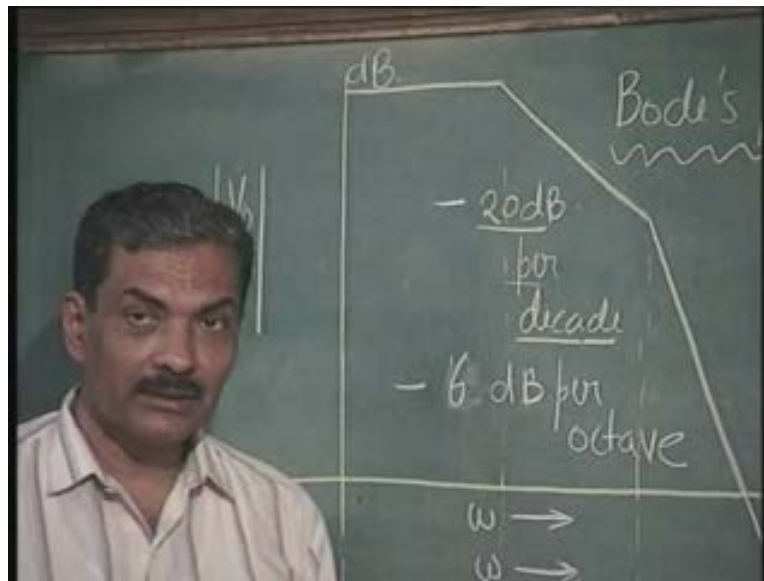
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$$\approx \frac{K_0}{\omega \tau_i \omega \tau_o} \text{ for } \omega \gg \frac{1}{\tau_i} \text{ and } \omega \gg \frac{1}{\tau_o}$$

$$\frac{K_0}{\omega \tau_i}$$

So, due to one corner frequency which is dominant, the attenuation is 20 decibels per decade. This is always the case. Irrespective of, again any specific amplifier configuration, if one time constant becomes dominant, we can make a statement, general statement that there is going to be an attenuation of 20 decibels per decade. Or, this is also called as 12 decibel, that is, actually, we are calling, instead of decibels per decade, we will also have a terminology called octave. What is octave? A factor of 2. So, $20 \log 1$ over 2, which is, minus $20 \log 2$. $\log 2$ is point 3. $20 \log 2$ is 6 decibels. So, you have 6, minus 6, minus 20, decibels per octave.

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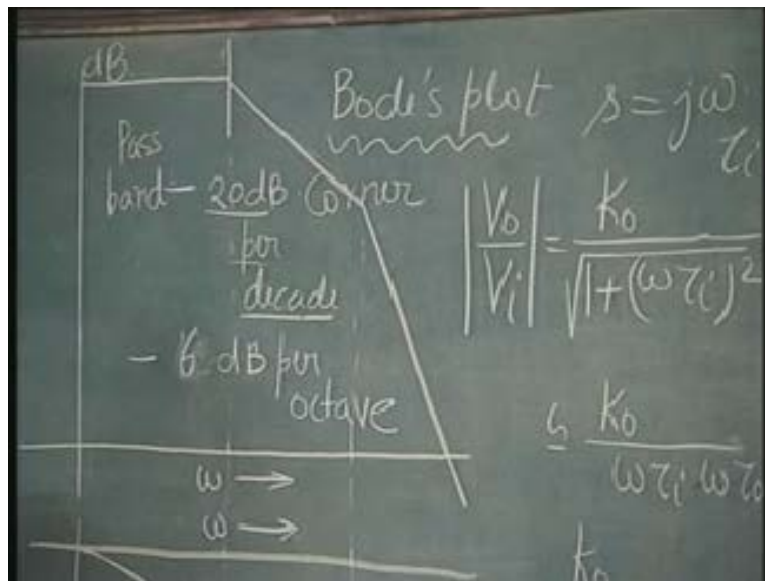


As electronic engineers, this terminology you must never forget. Because of one frequency dependence, because of one time constant, the amplifier is going to have an attenuation of, always, any amplifier is going to have an attenuation of, 20 decibels per decade or 6 decibels per octave. If another time constant comes, then, it will have a further attenuation of 20 decibels per decade or total attenuation of 40 decibels per decade or 12 decibels per octave.

So, N number of time constants will cause 20 into N decibels of attenuation per decade or 6 into N decibels of attenuation per octave. This is universal. So, you can therefore see

how attenuation occurs in the so called out of the pass band. This is called the pass band. This is the band of frequencies which is permitted to pass the amplifier without much of an attenuation, with respect to frequency. This is the pass band. This is the corner frequency. These are all called corner frequencies. These asymptotes correspond to this ω , $1/\omega$; this is $1/\omega^2$.

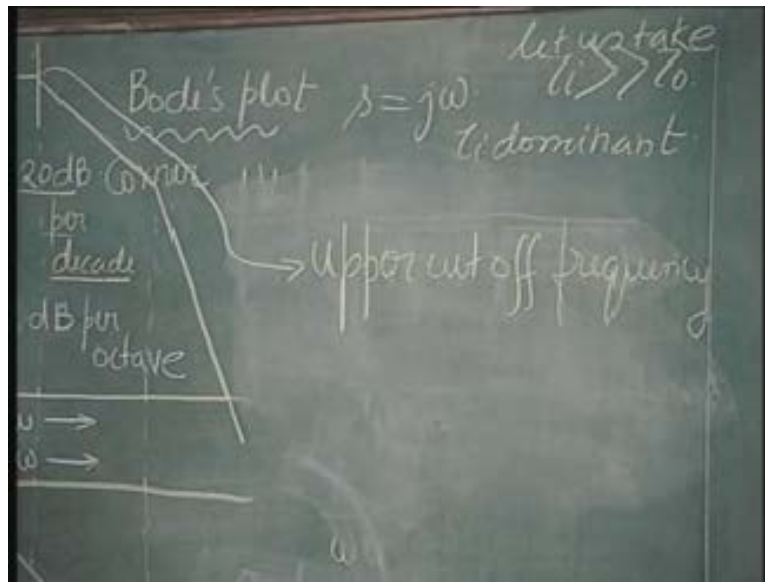
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So, this is 20 decibels per decade; this is 40 decibels per decade. There is next, another one coming that will be 60. But here, it is a two time constant system; so it can only cause an attenuation of 40 decibels per decade.

Now, something regarding the ((phase shift of asymptote Refer Slide Time: 35:25)). Therefore, these non-idealities come into picture in any amplifier. This is also a performance factor, which is as important as input resistance, output resistance and gain. So, this is one of another performance factors. It tells us, up to what frequency it can be used. This is called the upper cut off frequency. This is the first corner frequency in control terminology; or, this is also called upper cut off frequency, upper cut off frequency.

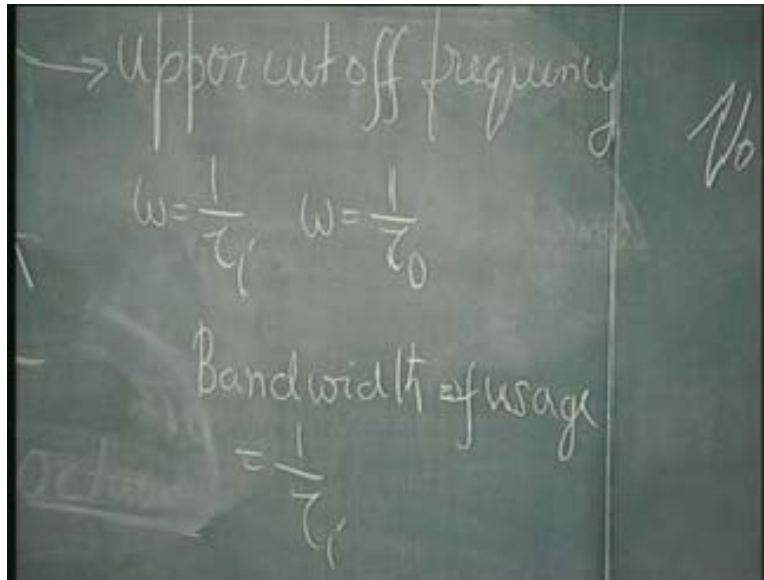
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What is this? In this case, it is equal to, ω equal to, $1/\tau_i$, exactly. This corner frequency corresponds to ω equal to $1/\tau_i$; and this corner frequency is going to be ω equal to $1/\tau_{naught}$. As long as τ_i is far away, $1/\tau_i$ is far away from $1/\tau_{naught}$, this is what happens. Otherwise, they will be actually related to one another; upper cut off and the other cut off.

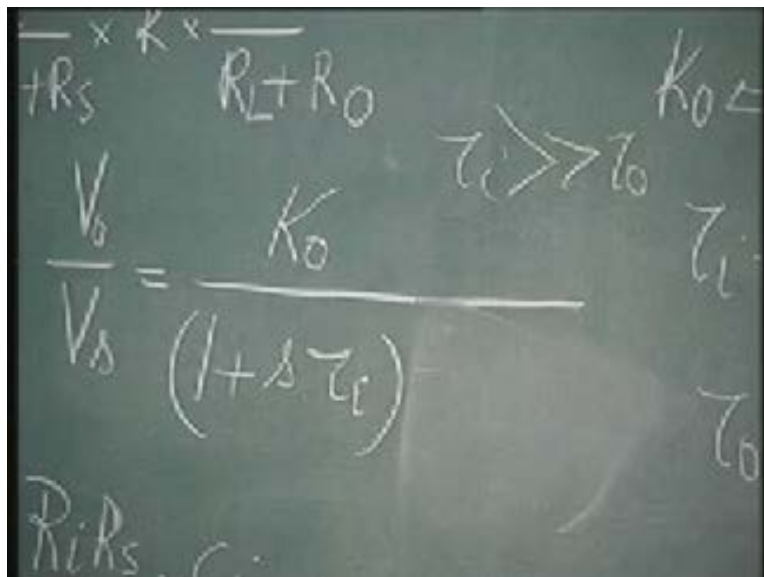
Now, upper cut off frequency is very important; it determines the band width of usage. So, the bandwidth of usage of this amplifier, for example, where τ_i is the dominant time constant, is equal to $1/\tau_i$. So, bandwidth is equal to $1/\tau_i$.

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What it simply means is, if tau i is dominant, this can be ignored. If tau i is much greater than tau naught, this can be ignored because this is of no significance, in the frequency range of interest to us.

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Why? What is the frequency range of interest to us? It is this pass band. That is the frequency range of interest to us. In that frequency range, it has no influence. So, this is a

single time constant system. Is this clear? Dominantly determined by τ_i ; so this is a single time constant system. If both the τ_i and τ_o were of same order, then, this is a second order system. If this is negligible, as negligible effect in the pass band, then this is called, this is approximated as a first order system.

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Handwritten notes on a chalkboard:

$$\frac{V_o}{V_s} = \frac{K_o}{(1+s\tau_i)(1+s\tau_o)}$$

$\tau_i = \text{input time constant}$
 $\tau_o = \text{output time constant}$

first order $\tau_i \gg \tau_o$
 2nd order $\tau_i \approx \tau_o$ comparable

$\frac{R_i R_s}{R_i + R_s} \times C_i$
 $\frac{R_o R_L}{R_o + R_L} \times C_o$

Is this clear – definition? This is a first order system if τ_i is much greater than τ_o . This is a second order system if τ_i and τ_o are comparable to one another. If τ_i is much greater than τ_o , then the bandwidth of this system is $1/\tau_i$; it is called the corner frequency of the system, first corner frequency of the system. It is also called upper 3 dB point.

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upper cut off frequency
 $\omega = \frac{1}{\tau_i}$ $\omega = \frac{1}{\tau_o}$
 Bandwidth = f usage
 $= \frac{1}{\tau_i}$ Upper 3dB point

Why this? Let us understand that. If this is negligible, then, this is the one that fixes up the attenuation; and s is equal to $j\omega$, you are going to put.

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$R_i + R_s$ $R_L + R_o$ $\tau_i \gg \tau_o$

$$\frac{V_o}{V_s} = \frac{K_o}{(1 + j\omega\tau_i)}$$

 $= R_i R_s$ first order

And therefore, the magnitude function for this is going to be root of 1 plus omega square tau i square. Say, omega equal to omega over tau i. This factor becomes equal to 1.

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Handwritten derivation on a chalkboard. At the top, it says $R_L + R_S$ and $R_L + R_O$. The main equation is
$$\left| \frac{V_o}{V_s} \right| = \frac{K_o}{\sqrt{1+1}}$$
 with the condition $\tau_i \gg \tau_o$ written to the right. Below the equation, it says "first order". A hand holding a phone is visible in the bottom left corner.

So, this factor becomes K naught divided by root 2. And, if you compare it with K naught, it has reduced by 1 over root 2. It is therefore $20 \log 1 \text{ over root } 2$, which is $10 \log 1 \text{ over } 2$ or minus $10 \log 2$ which is 3 dB, minus 3 dB. So, this is called upper 3 dB point. So, this is equal to K naught by root 2.

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Handwritten derivation on a chalkboard, continuing from the previous slide. It shows the same equation
$$\left| \frac{V_o}{V_s} \right| = \frac{K_o}{\sqrt{1+1}}$$
 with the condition $\tau_i \gg \tau_o$. Below this, it simplifies the expression to
$$= K_o / \sqrt{2}$$
. At the bottom, it says $= \frac{R_L R_S}{R_L + R_S} \times C_i$ and "first order".

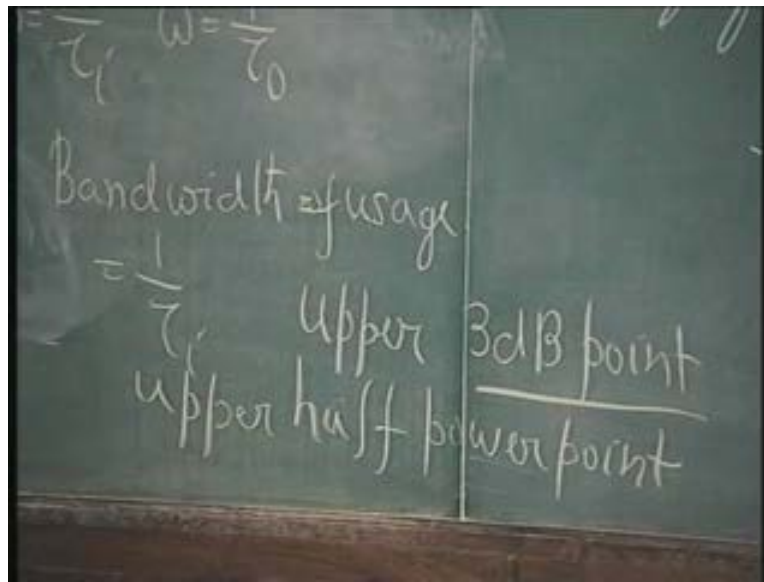
Or, this corner point, the attenuation, actual attenuation curve will be going like this. So here, it will be suffering an actual attenuation of 3 decibels.

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At every corner frequency, it will suffer an additional attenuation of 3 decibels. So, at this point, it would definitely suffer, may be, an additional attenuation of 3 decibels, apart from whatever it has further suffered due to the first corner frequency. So, these are the important definitions. If voltage gain reduces by $1/\sqrt{2}$, power gain reduces by another factor of $1/\sqrt{2}$. It is square; so, power gain is halved. This is also called upper half power point.

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Now, I am giving you all these definitions which are involved for the same thing. Upper half power point is same as upper 3 dB point is same as what? - first corner frequency. All these things are same things, defined in different ways. This is in terms of voltage ratio; this is in terms of power. So, this is important in estimating the useful frequency range of other amplifier. Any other point would have also been chosen; but they have conveniently chosen the half power point for reference. Therefore, this corresponds to half power point.

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And, the useful range of bandwidth of your amplifier is up to the upper cut off frequency, first upper cut off frequency really; so, first corner frequency. If this bandwidth is of the order of, let us say, tens of Kilohertz, then it is audio amplifier; if this bandwidth is of the order of megahertz, then it is video amplifier.

So, that is why, in classifying the amplifier, it is important you understand how this bandwidth comes about. This bandwidth comes about because of shunt capacitors which are unavoidable in circuits. And therefore, in order to maintain the bandwidth of an amplifier very high, we have to make these time constants very low. It is obvious.

This also gives us design features for wide band amplifiers. In order to make these time constants very low, we have to have obviously C_i very low, C_{naught} very low; also, we have to have R_i and R_s very low. This is important.

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$$\tau_i = \frac{R_i R_s}{R_i + R_s} \times C_i \quad = k_0 / \sqrt{2}$$

$$\tau_o = \frac{R_o R_L}{R_o + R_L} \times C_o$$

R_i and R_s also must be very low in order to have... or R_o and R_L ; any one of this. Either R_i or R_s should be low in this; here, either R_o or R_L should be low. So, these are important in designing wide band amplifiers. So, wide band amplifiers also are called pulse amplifiers. So, even if you are designing a pulsed amplifier, same design features are valid. You have to make the capacitors very low. You have to make the... either source resistance or input resistance very low. If you make source resistance very low, you are calling it voltage source driven. If R_i is made very low, it is current controlled. So, either it can be voltage controlled or current controlled; either it can be a voltage source or current source.

As long as you are using ideal amplifiers which are driven very nicely, then, you are going to achieve wide band for your amplification. Therefore, this is one of the important performance factors of an amplifier; the band width of an amplifier; and that is determined by this Bode's plot.

Now, the phase information. This is also important. How do you get the phase? In this expression that we have, $1 + s\tau_i$, $1 + s\tau_o$... As frequency increases, the phase lag occurs because, both are in the denominator. So, the phase is, $\tan^{-1}$. That

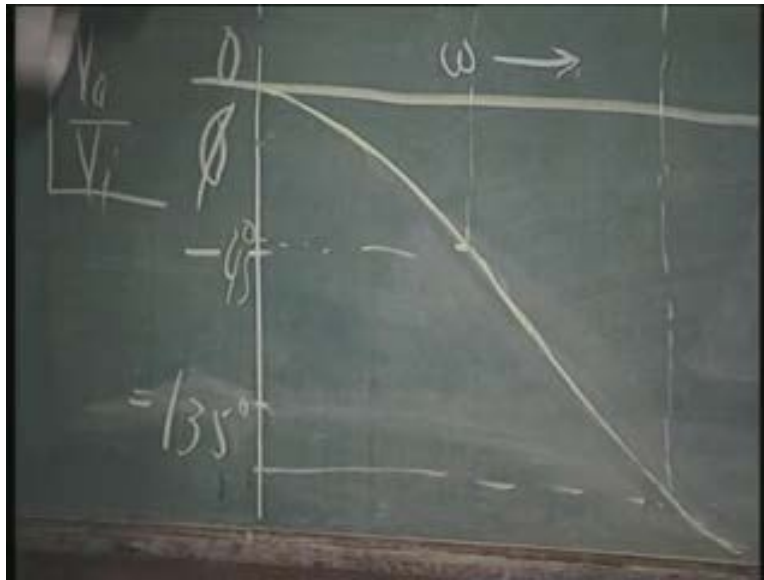
is, ϕ , phase lag, $\tan^{-1} \omega \tau_i$ plus $\tan^{-1} \omega \tau_0$. That is the phase lag associated with this transfer function. At very low frequencies, there is no phase lag; so, this is going to be zero, ϕ . Starting with zero, gradually there is going to be phase lag. At τ_i , τ_i being dominant, we have $\omega \tau_i$ equal to 1. It becomes $1 + j$; $1 + j$ means how much? A phase shift of 45 degrees. So, around this point, we have a phase shift, phase lag of 45 degrees.

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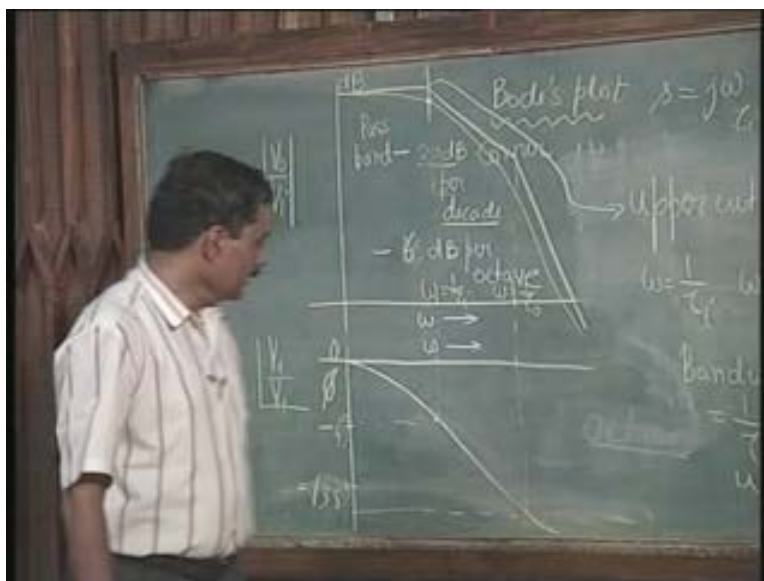
Then, as this progresses, the dominant time constant would have already contributed to your full phase because $\omega \tau_i$ becomes dominant compared to 1. That means, it is phase shift of 90 degrees it would have given. Then, when we come to other frequency, ω equal to $1/\tau_0$ is this; **ω equal to $1/\tau_0$ is this**; at this frequency, additional phase of 45 degrees can be given. So, we have a phase shift of minus 135 degrees. So, it is going from 45 degrees at one corner frequency to minus 135 degrees at the next corner frequency.

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Ultimately, it will go to... both will contribute to a phase shift of 90 degrees, and it will go to 180 degrees, at infinite frequency. So, this is capable of going to a phase lag of 180 degree only at infinite frequency. So, this is the phase plot of the circuit; together, these plots are called Bode's plot. These are important in characterizing amplifier very well.

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So, both these distortions: the magnitude distortion as well as the phase distortion, together contributes to the distortion signals; different frequency components get subjected to different phase shifts and therefore there is a distortion in the signal. Therefore, it is required that we restrict our field of operation within the pass band.

Now, as far as tuned amplifiers are concerned, basically, the load is selected in such a manner that this particular frequency response characteristic is not like this - flat from the low frequency end, coming down; it is starting with zero, reaches a peak at certain point and comes down like this. So, the tuned amplifier characteristic is different from these wide band amplifiers, or, low frequency amplifiers.